Blind Channel Estimation & Resolving Phase Ambiguity for OFDM Symbols

by

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Certificate of Approval

It is certified that we have read the project report submitted by **Naeem Shahzad [126-FET/MSEE/F-07]**. It is our judgment that this report is of sufficient standard to warrant its acceptance by the International Islamic University, Islamabad for degree of MS Electronic Engineering (MSEE).

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بسم الله الرحمن الرحيم

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ABSTRACT

The demand for capacity in cellular networks has grown in a literally explosive manner during the last couple of years. In particular, the need for wireless internet access and multimedia application require an increase in information throughput with order of magnitude compared to the data rates made available today. One major technological breakthrough that has made this increase in data rate possible is the use OFDM.

This thesis introduced an approach to estimate the channel blindly without the use of 2^{nd} ordered and higher order statistics. To estimate channel blindly a modified basic maximum likelihood estimator is developed. This can estimate the channel blindly without the use of higher order statistics. This estimation method can successfully recover the magnitude but this doesn't recover the correct phase. To recover the phase a novel approach is established which combine two different modulation schemes (QPSK/3PSK). During coherent demodulation the results are unique for each combination of the combined modulation scheme.

The results of the proposed algorithm have been compared with other techniques reported in literature. The results have been compared using bit error rate & mean square error. The computational comparison also demonstrates the advantages of using the proposed algorithm.

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Chapter 1

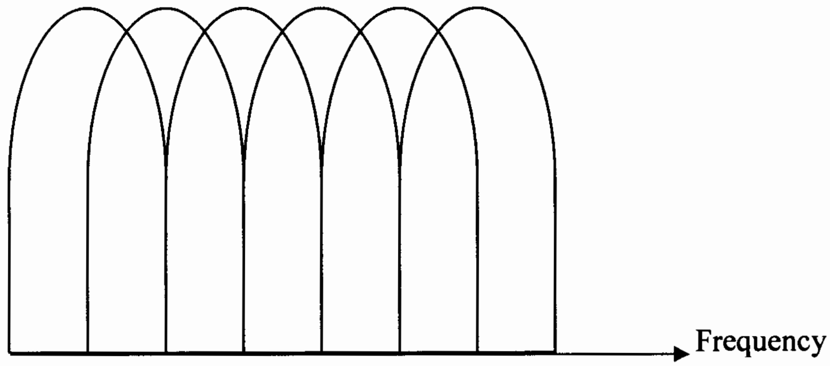
Introduction

Frequency Division Multiplexing (FDM) started some hundred years back, where more than one low rate signal, such as telegraph. It was used on the basis of relative bandwidth. For separation of the signals at the receiver end, the carrier frequencies were spaced with the help of guard interval to minimize the risk of overlap. These guard intervals between the signals guaranteed that they would be separated with the help of these guard intervals. Therefore, the resulting spectral efficiency becomes very low [1].

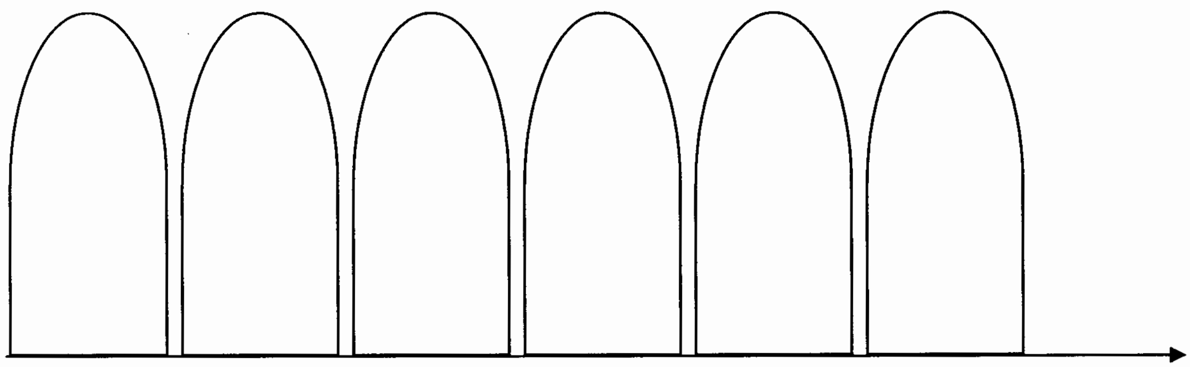
The different frequency carriers can carry different bits of a single higher rate message. The source may be in such a parallel format instead of carrying separate messages, or a serial source can be presented to a serial-to parallel converter whose output is fed to the multiple carriers [1].

1.1 Orthogonal Frequency-Division Multiplexing (OFDM)

In OFDM, the sub-carrier frequencies are chosen so that the sub-carriers are orthogonal to each other. Which simple means that the sub-channels are eliminated without using guard bands. The effects of frequency-selective channel conditions, for example fading caused by multipath propagation, can be considered as constant (flat) over an OFDM sub-channel[]. This makes equalization far simpler at the receiver in OFDM in comparison to conventional single-carrier modulation. The equalizer only has to multiply each sub-carrier by a constant value, or a rarely changed value. Figure 1.1 shows the difference between conventional FDM and OFDM[].



OFDM



Conventional FDM

Fig 1.1 The above figure shows the difference between the conventional FDM and OFDM wave form

1.2 Advantages of OFDM:

The advantages of OFDM are summarized as given below [1]

- Robust against narrow-band example, attenuation of high frequencies
- Robust against ISI
- High spectral efficiency
- OFDM signals are generated using the FFT algorithm

1.3 Disadvantages of OFDM

However, the OFDM exhibits the following disadvantages [1]

- Sensitivity to time synchronization

- Sensitive to Doppler shift
- OFDM requires very accurate frequency synchronization between the receiver and the transmitter

1.4 Basic model for OFDM

The two basic models for OFDM (OFDM transmitter & OFDM receiver) are as given below.

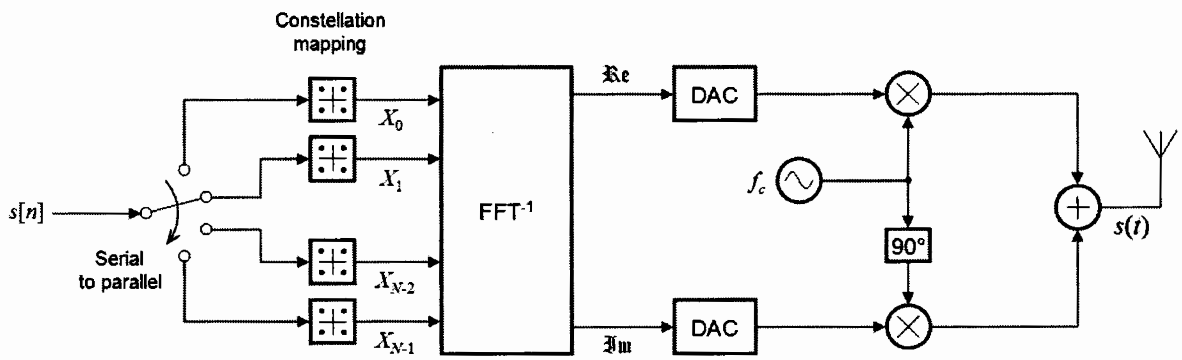


Fig 1.2 Basic OFDM Transmitter using FFT

As shown in figure 1.2 we are receiving binary data in series. At the first step we are converting the serial data into parallel. After that we are sending parallel data to $N-1$ parallel QPSK constellations for mapping. Mapped data which is show as X_0 to X_{N-1} is passed through a FFT. The FFT output two values one is real and second is imaginary. These output then converted to analog form from the digital. It is then transmitted over the carrier frequency as $S(t)$

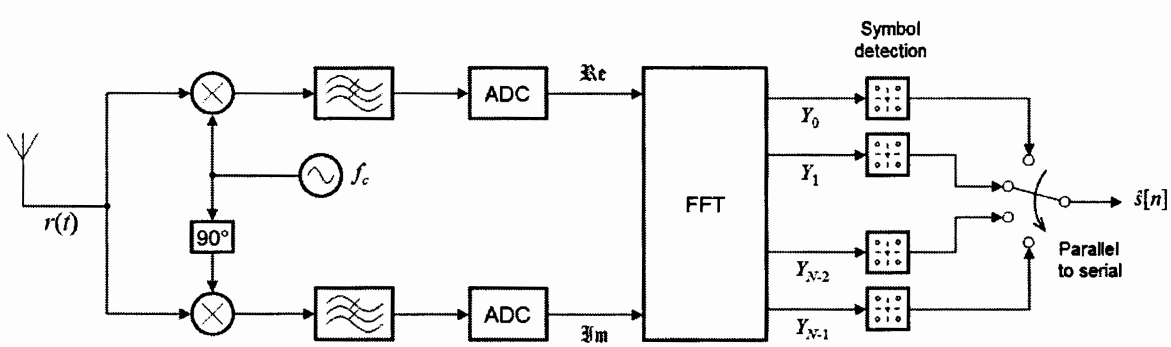


Fig 1.3 Basic OFDM Receiver using FFT

As shown in figure 1.3 we are receiving data in the from of $r(t)$ after passing through a channel. At the first step we take the carrier off. After that we pass the reaming signal through analog to digital converter to get both the imaginary and real part. Real and imaginary parts input to FFT block and the output is coming in the form of symbols Y_o to Y_{N-1}. These N numbers of symbols are detected through symbol detection mechanism. This parallel data is then converted serial data.

1.5 Contribution of Thesis

We have introduced a new approach to resolve the phase ambiguity of the channel estimate. In this paper, we have given a concept that involves obtaining a unique channel estimate at the receiver through the combination of two different modulation schemes on adjacent OFDM carriers. Our emphasis has been on the combinations of quaternary PSK (QPSK) and 3-PSK and we also come up with a new approach to solve the ambiguity two phases QPSK and 3-PSK. Through our approach we have successfully recovered the complex channel gain (amplitude and phase), and we have been able to achieve it without any reference symbols. Hence the blind channel estimation is successfully achieved through the proposed channel estimator.

The first principle develops the ML blind channel estimator. The next section explains how the phase ambiguity of the channel estimate is resolved through the concept of combined PSK modulation schemes. In the last section we have covered the applications and the simulation results for the blind channel estimation method that we have proposed.

Chapter 2

Introduction

Maximum likelihood estimation is a totally analytic maximization procedure. It applies to every form of data, and it is used very wildly in all the method for estimation. Maximum likelihood estimation begins with writing a mathematical expression known as the Likelihood Function of the sample data. Loosely speaking, the likelihood of a set of data is the probability of obtaining that particular set of data, given the chosen probability distribution model. This expression contains the unknown model parameters. The values of these parameters that maximize the sample likelihood are known as the Maximum Likelihood Estimator.

2.1 Maximum Likelihood Estimator

The explanation of the maximum likelihood estimator in the form of mathematical model is as given below

$$\tilde{d} = \left[d_0 \quad {}_1 \cdots\cdots d_{N-1}\right] \qquad |d_n = 1| \tag{1}$$

N sub carriers and QPSK-modulated

d = data symbols

Converting the data series into time domain vector

$$T = \left[T_0 \;\; T_1 \ldots\ldots T_{N-1}\right] \tag{2}$$

Where

$$T_k = \sum_{v=1}^{N-1} d_v e^{\frac{j2\pi vk}{N}} \tag{3}$$

T_k are then passed through a D/A converter it to complex envelope $\tilde{S}(t)$.

$h(t,\tau)$ = time variant channel impulse response.

$$y(t) = \int_{-\infty}^{\infty} h(t, y)\tilde{s}(t-y)dy + n(t) \tag{4}$$

We have convolute the transmitted signal $s(t)$ with channel transfer function $h(t)$ and $n(t)$ is the noise which is white noise Where $y(t)$ is the transmitted signal/waveform. The waveform $y(t)$ is sampled at $t = kT_S$ which form the vector y. The T_s is the duration of one symbol. Therefore, the sampled output form may represent as.

$$y = \left[y_0 \; y_1 \; \cdots\cdots \; y_{N-1}\right] \tag{5}$$

This signal does not contain ICI. Therefore, $h(t,\tau)$ remains approximately constant for the one OFDM symbol duration,

$$r_k = H_k.b_k + N_k \tag{6}$$

N_k is AWGN

b_k is the binary input vector

H_k sampled channel transfer function

$$H_k = H(t, k\Delta w) \qquad H(t, w) = F_\tau\{h(t,\tau)\} \tag{7}$$

$$\Delta w = 2\pi / NT_s \; .$$

Where $d = [d_0 \; d_1 \; . \; . \; . \; .d_{M-1}]$ is a vector of M data symbols with spacing of $k\Delta w$ radians, where k is a positive integer. Therefore, the signal/waveform received at antenna will be $r = [r_0 \; r_1 \; . \; . \; . \; r_{M-1}]$

Let Q_d be the DFT matrix

$$Q_d = [w_{d,0} \; w_{d,1} \; . \; . \; . \; w_{d,L-1}],$$

One vector representation is as follows

$$q_{d,n} = \left[1 \; e^{-jm\Delta wT_s} \; . \; . \; . \; e^{-jmk\Delta wT_s(M-1)T}\right]^T \tag{8}$$

The received vector is written as represented as given below where h is a length- L vector

$$r = DQ_d h + N = DH + N \tag{9}$$

$H = [H_0 \ H_1 \ . \ . \ . \ H_{M-1}]$ is the vector of channel transfer function coefficients, $N = [N_0 \ N_1 \ . \ . \ . \ N_{M-1}]$ is AWGN vector and

$$D = \begin{pmatrix} d_0 & & & \\ & d_1 & & \\ & & \ddots & \\ & & & d_{M-1} \end{pmatrix} \tag{10}$$

Chotikoakamthorn and Suzuki [2] show that the channel can be estimated from a single received OFDM symbol. Theorem 1 provides the underlying basis that makes it possible to apply the ML principle to only one OFDM symbol.

2.2 Theorem 1 (From [2])

The channel parameters and the transmitted symbols are uniquely identifiable up to a scaling factor, if

$$M > P(L-1) \tag{11}$$

With P being the number of b_i / b_j with distinct values for all possible permutations of symbols b_i and b_j of the symbol alphabet S.

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Theorem 1 implies that there is only one vector d and one vector h that can yield the received vector y in the noise-free case. If noise is present, an ML estimator for both d and h can be constructed. If the noise N is white and Gaussian, the ML estimates of d and h are those vectors that minimize the quadratic error from the received sequence r

$$\hat{\theta} = \min \| r - DQ_d h \|^2 \qquad \theta := [h^T, d^T]^T \tag{12}$$

Defining diagonal R matrix

$$R = \begin{pmatrix} r_0 & & & \\ & r_1 & & \\ & & \ddots & \\ & & & r_{M-1} \end{pmatrix} \tag{13}$$

and by exploiting the constant modulus property of PSK signals, (12) reduces to (see [2])

$$\tilde{d} = \max_{d} Tr(R^{*}Q_{d}Q_{d}^{H}Rd^{*}d^{T})$$

$$= \max_{d} d^{T}R^{*}Q_{d}Q_{d}^{H}Rd^{*} \qquad (14)$$

With * denoting the complex conjugate. If d_n have been estimated by solving (14) he channel transfer function can be obtained

$$\tilde{H} = \tilde{D}^{*} r \qquad (15)$$

In the case of PSK, this estimate contains a phase ambiguity, as there are many solutions to (14) that yield the same maximum value. Also note that the calculations involve the received symbols of only one OFDM-symbol.

We have to optimize (14) to obtain one solution which is maximum. A brute force algorithm must exhaust $2^{(M-1)\log_2 q}$ all possibilities for d (where d_0 can be chosen arbitrarily because of the phase ambiguity). However, the algorithm still has high computational complexity, especially for long channel impulse responses and larger values of M. There are two theorems for the noise-free case are introduced

2.3 Theorem 2

By using knowledge of only the received vector r, the channel parameters $H = [H_0 \ H_1]$ and the transmitted symbols $d = [d_0 \ d_1]$ are uniquely identifiable up to a complex scaling factor if (d_0, H_0) and (d_1, H_1) belong to adjacent sub carriers and $|\varepsilon| = |H_1 - H_0|$ is less than half the minimum Euclidean distance between any two received signal points z_i and z_j in the complex plane, where $\varepsilon = H_1 - H_0$. An equivalent requirement is that $|\varepsilon / H_1| < d_{\min}/2$, where $d_{\min}$ is the minimum Euclidean distance between any two signal constellation points d_i and d_j.

Proof: It is sufficient to show that (12) has a unique solution

$$\hat{\theta} = \min \|r - DQ_d h\|^2, \quad \theta := [h^T, d^T]^T$$

We applied change of variable on the above equation and formed the below one

$$\hat{\psi} = \min_{\psi} \|r - DH\|^2, \quad \psi := [H^T, d^T]^T$$

As above mentioned that we have generalized theorem 1 for two OFDM symbols so in the following equations we have supplanted the values of r, D and H in the above equation for two symbols only

$$= \min_{\psi} \|[r_0 - H_0 d_0, \quad r_1 - H_1 d_1]\|^2$$

$$= \min_{\psi} \|[r_0 - H_0 d_0, \quad r_1 - (H_0 + \varepsilon) d_1]\|^2$$

$$= \min_{\psi} \|r_0 - H_0 d_0\|^2 + \|r_1 - (H_0 + \varepsilon) d_1\|^2 \tag{16}$$

$$\tilde{d}_1 = \min_{d_1} \|r_0 - r_0 d_0^* d_0\|^2 + \left\| r_1 - (r_0 d_0^* + \varepsilon) d_1 \right\|^2$$

The first part of the above equation is a constant so we have to minimize the 2nd part of the equation to get best estimate for d_1

$$= \min_{d_1} \|r_1 - (r_0 + \varepsilon) d_1\|^2 \tag{17}$$

2.4 Theorem 3

By knowing only the received vector r, the channel parameters $H = [H_0 \ H_1 \ . \ . \ . \ H_{M-1}]$ and the transmitted symbols $d = [d_0 \ d_1 \cdots\cdots d_{N-1}]$ are uniquely identifiable up to a complex scaling factor for any $M \geq 2$, if $(d_0, H_0) \ . \ . \ . \ .(d_{M-1}, H_{M-1})$ belong to consecutive sub carriers and the channel transfer function coefficients H change slowly in the frequency domain, i.e.,

$$\left|\frac{\varepsilon}{H_n}\right| = \left|\frac{H_n - H_{M-1}}{H_n}\right| < d_{\min}/2, \qquad n = 0....N-1$$

Proof: Theorem 3 can be proved by using Theorem 2 and induction.

Assumption Step:

The channel is uniquely identifiable for $M = 2$ according to Theorem 2.

Induction Step:

Let $H = [H_0 \ H_1 \ . \ . \ . \ H_{M-1}]$

$d = [d_0 \ d_1 \cdots\cdots d_{N-1}]$

It is sufficient to show that by adding the elements H_M and b_M to both of these vectors the uniqueness of the solution is still maintained. If $H_M = H_{M-1} + \varepsilon$

$$\hat{\psi} = \min_{\psi} \|r - DH\|^2$$

$$= \min_{\psi} \|[r_0 - H_0 d_0, \quad r_1 - H_1 d_1 \ . \ . \ . \ . \ r_M - H_M d_M]\|^2$$

$$= \min_{\psi} \|r_0 - H_0 d_0\|^2 + \ . \ . \ . \ + \|r_{M-1} - H_{M-1} d_{M-1}\|^2 + \|r_M - (H_{M-1} + \varepsilon) d_M\|^2 \tag{18}$$

Because of the phase-blindness, the vector d can be modified such that $d_{M-1} = 1$ without loss of generality. Using this fact, and since $H_{M-1} = r_{M-1} d_{M-1}^{*}$ and $H_M = H_{M-1} + \varepsilon$ we have

$$\tilde{d}_M = \min_{d_M} \|r_M - (r_{M-1} + \varepsilon) d_M\|^2 \tag{19}$$

The proof for Theorem 2 establishes a unique solution if $\left|\frac{\varepsilon}{H_M}\right| < d_{\min}/2$. Theorem 3 is thereby proved.

Chapter 3

3.1 Proposed Method

There were many methods which can be used to recover phase and magnitude both but have some drawbacks. Let me throw some light on these existing methods & their drawbacks. To perform coherent demodulation it is necessary to have knowledge of time variant channel transfer function. The channel transfer function is conveniently estimated by two diminution grid of pilot symbols but in this solution channel capacity is wasted [3]. DPSK & differential demodulation successfully implemented but have drawback for loss in E_b / N_0 and larger loss for fading channels [4]. Examples of statistical blind channel estimation techniques include those using correlation methods [5] and cumulant fitting schemes [6] and [7]. 2nd order static's recovers magnitude not phase. Phase can only be recovered if the received signal is cysclostationary only [9] & [10].

We have proposed a solution which recovers both the magnitude and the phase but with out using the 2nd order and higher order statistics.

1. Blind channel estimation with out using higher order statistics (Magnitude)
2. Resolved the phase ambiguity using combined modulation scheme (Phase)

3.2 Blind Channel Estimation

In chapter 2 we have used the theorem 1 in which Chotikoakamthorn and Suzuki [1] show that the channel can be estimated from a single received OFDM symbol. We generalized theorem 1 for two OFDM symbols under the same conditions that bandwidth for the subcarriers is relatively same or constant. Theorem 2 proof that it is possible to estimate channel blindly from 2 received OFDM symbols without using higher order statistics. Where $d_{\min}$ is the minimum Euclidean distance between any two signal constellation points. Theorem 3 is also proved by using theorem 2 induction for M number of received OFDM symbols.

3.3 Combined Modulation Schemes

A new method is explored in this section that restores the phase without using reference symbols. Thus, our proposed method is blind.

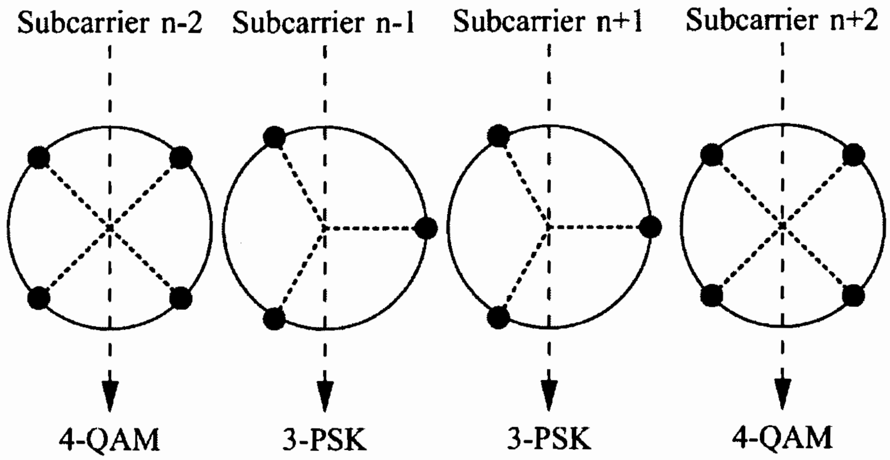

Fig 3.1 Combined Modulation schemes on the OFDM sub carriers

The key concept of the proposed method is that two PSK signal constellations of different order be used within the same OFDM symbol. The two signal constellations are chosen such that the angles between a selected signal point of one constellation and any signal point in the other constellation are unique. For example, QPSK and 3-PSK satisfy this property. If such a waveform is used, a blind channel estimator based on (14) no longer suffers from phase blindness. Other mixtures of signal constellations will also fulfill the above requirement. For example, QPSK can be combined with 5-PSK, and 8-PSK can be combined with 7-PSK or 9-PSK.

The regular QAM-modulation scheme was replaced by the combined QPSK/3-PSK scheme. Code puncturing is used to solve the problem of mapping bits to the 3-PSK and Q-PSK symbols. See Table-1 & Table-2

In general, the distributions of the real and imaginary parts of the resulting ternary symbols have nonzero mean. Therefore, rotating the mapping scheme by 120 does not affect performance or any of the algorithms. The resulting stream of bits and ternary symbols is modulated by the IFFT-block. Attention needs to be paid during the final distribution of the data symbols to the sub carriers, since the QPSK and 3-PSK symbols must alternate.

TABLE -1

CONVERSION OF CODED BITS TO 3-PSK SYMBOL. THE NUMBERS 0, 1, AND 2 REPRESENT THE DIFFERENT SIGNAL POINTS OF A 3-PSK SYMBOL.

Coded Bits	**Ternary Symbols**	**Coded Bits**	**Ternary Symbols**
00	0	01	2
11	1	10	1

TABLE -2

CONVERSION OF CODED BITS TO Q-PSK SYMBOL. THE NUMBERS 0, 1, 2 AND 3 REPRESENT THE DIFFERENT SIGNAL POINTS OF A Q-PSK SYMBOL.

Coded Bits	Ternary Symbols	Coded Bits	Ternary Symbols
00	0	01	2
11	1	10	3

3.4 Basic OFDM Model

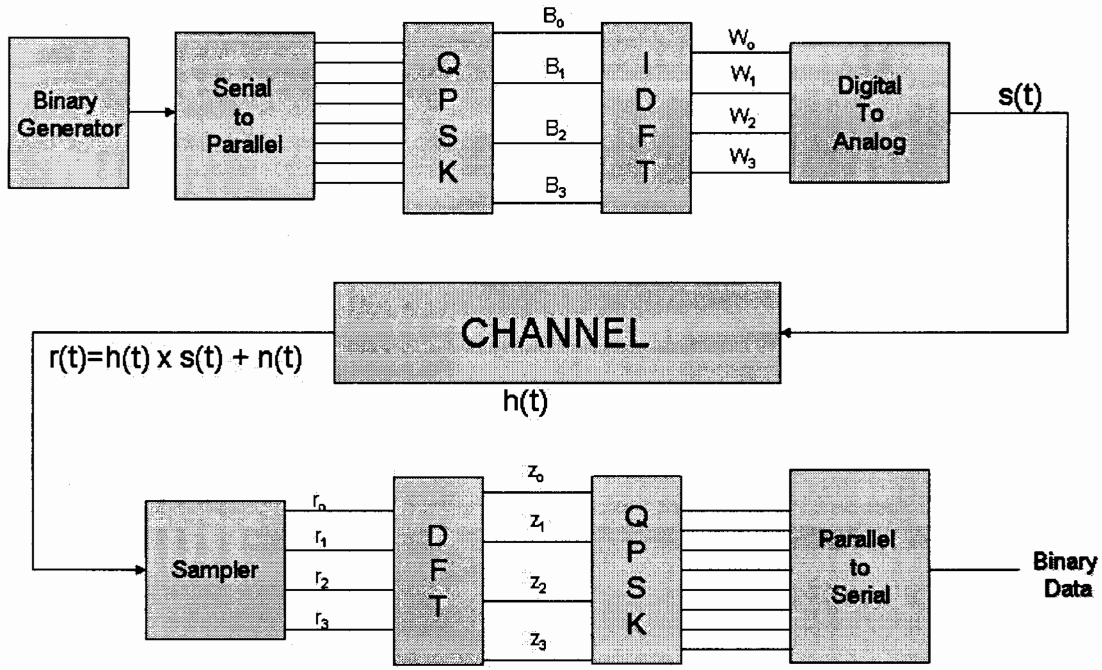

Fig 4.1 Basic model used for blind estimation with QPSK modulation scheme

As shown in figure 4.1 binary generator is the first block which generates binary data as input date. We are receiving binary data in series. At 2nd step we are converting the serial data into parallel. After that we are sending parallel data of 8 bits to parallel QPSK constellations for mapping. Mapped symbols are shown as B_0 to B_3 form for the eight input bits. This is then passed through a FFT block. The FFT output 4 time domain values for each 4 input frequency domain values. These output then converted to analog form the digital. It is then transmitted over the carrier frequency as $S(t)$. Which passed through the channel where $S(t)$ convoluted with channel transfer function and noise is added in it.

At the receiver when $r(t)$ the received signal reaches at 1st step it passed through a sampler. Sampler samples the date at kT intervals and output four symbols to the DFT block to convert it from time domain to frequency domain. The time domain symbols are then demodulate through a QPSK demodulator which produce eight bits of binary data. At last the data is converted from parallel to series form.

3.5 Proposed Model

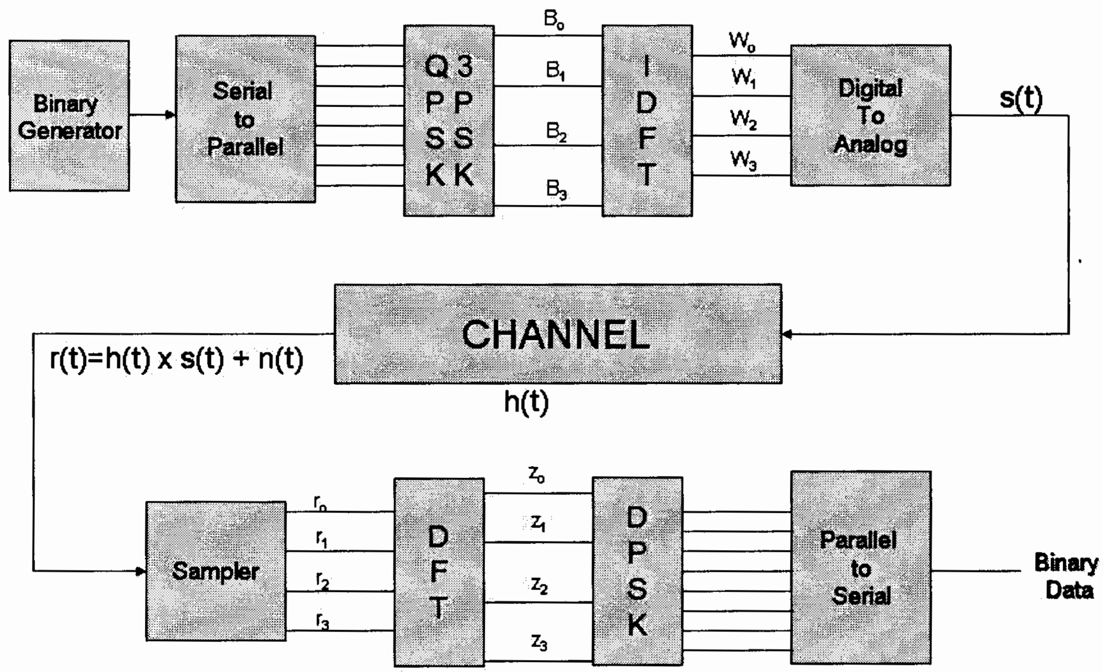

Fig 4.2 Proposed model to resolve phase ambiguity using combined modulation scheme

As shown in figure 4.2 binary generator is the first block which generates binary data as input date. We are receiving binary data in series. At 2nd step we are converting the serial data into parallel. After that we are sending parallel data of 8 bits to parallel 3PSK/QPSK constellations for mapping. Mapped symbols are shown as B_0 to B_3 form for the eight input bits. This is then passed through a FFT block. The FFT output 4 time domain values for each 4 input frequency domain values. These output then converted to analog form the digital. It is then transmitted over the carrier frequency as $S(t)$. Which passed through the channel where $S(t)$ convoluted with channel transfer function and noise is added in it.

At the receiver when $r(t)$ the received signal reaches at 1st step it passed through a sampler. Sampler samples the date at kT intervals and output four symbols to the DFT block to convert it from time domain to frequency domain. The time domain symbols are then demodulate through a 3PSK/QPSK demodulator which produce eight bits of binary data. At last the data is converted from parallel to series form

3.6 Graphical Output

On the receiver side we are detecting symbols by the phase difference. The graphical output of the angle differences is as shown below. Total of 12 possible outcomes to this scenario can uniquely identify on this output graph. Logic to this is also shown in the tabulated form.

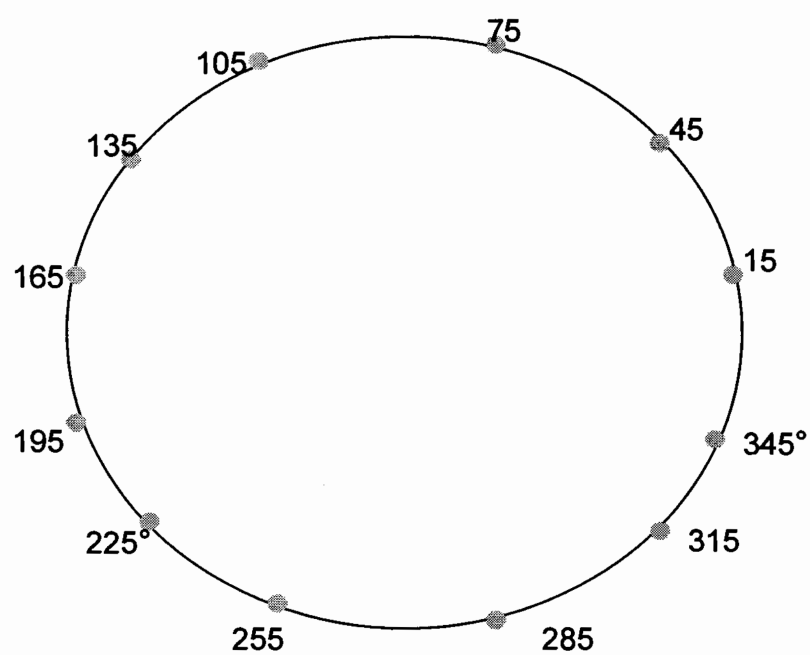

Fig 4.3 Output of possible angle differences according to table -3

TABLE -3

Total possible output after coherent demodulation

3-PSK	QPSK	Difference
0	45	-45
0	135	-135
0	225	-225
0	315	-315
120	45	75
120	135	-15
120	225	-105
120	315	-195
240	45	195
240	135	105
240	225	15
240	315	-75

Table 3 shows total possible outcomes after coherent demodulation. It clearly shows that the possible outcomes are uniquely identifiable. For example we pick a value with difference 75 which means that you send a symbol of 3PSK with angle 120 and for QPSK 45. This uniqueness is resolving the phase ambiguity. This also can be proved that these difference must be unique because 3PSK and 4PSK (QPSK) are prime to each other.

3.7 Complexity Analysis

The suboptimal algorithm for solving (14) can be realized with complexity $O(q^2)$. This is a very small value and also holds if modulation schemes are combined. In the latter case, the q of the higher order modulation scheme determines the complexity. Hence, the central part of the receiver, which is the blind channel estimator itself, consumes a small fraction of the computational resources required to implement the receiver. The remainder of the receiver is no more complex. We believe that our blind channel estimator has low complexity when compared to other blind channel estimation approaches, especially those based on statistics.

Chapter 4

4.1 Results and Discussions

The blind channel estimator was applied to a modified DVB-T system [3]. DVB-T is based on OFDM and uses pilot-based channel estimation for coherent detection of QPSK encoded data symbols. Starting with 1705 sub carriers, all pilots were removed, resulting in a system with only 1512 sub carriers. The regular QPSK modulation scheme was replaced by the combined QPSK/3-PSK scheme. The resulting stream of bits and ternary symbols is modulated by the IFFT-block. This receiver design delivers good performance and also allows for an efficient implementation in hardware because of its simplicity, since all inputs on the different OFDM-symbols can be performed in parallel that also can make the performance better and faster. Attention needs to be paid during the final distribution of the data symbols to the sub carriers, since the QPSK and 3-PSK symbols must alternate and a signal bit position can increase the bit error rate.

The results have been compared using bit error rate & mean square error. In a communication system, both the noise and the interference components can degrade the bit error performance. A minimum mean-square-error (MMSE) detector takes the effects of both multi-user interference and noise into consideration. The formula to calculate the mean square error and bit error rate (BER) are shown below respectively

$$MSE = E\left\{\left(\widehat{W} - r\right)^2\right\}$$

$$\widehat{W} = 1/n\sum_{i=0}^{n}(w_i)$$

Where $\widehat{X}$ is the error mean and X is the error between the transmitted data and the received data

$$BER = 1/n(w - r)$$

Where n is the total number of data symbols w are the transmitted data symbols and r the received data symbols.

4.2 Simulation Assumptions

Following are the assumptions taken for the simulation.

- Two channel coefficients belonging to adjacent sub carriers can be estimated if the channel transfer function does not vary too fast in frequency.

- The resulting stream of bits and ternary symbols is modulated by the IFFT-block. Attention needs to be paid during the final distribution of the data symbols to the sub carriers, since the QPSK and 3-PSK symbols must alternate and a signal bit position can increase the error rate.

4.3 Simulation Results

The fig 4.1 shows the convergence analysis convergence of proposed algorithm over RA channel with $E_b/N_0 = 12$ dB for 3PSK, QPSK and combination of 3PSK/QPSK. The results have been compared using mean square error of the proposed algorithm. The mean square error is 0.45 for the 0^{th} OFDM received symbol at the first point. It gradually decreases to 0.01 for the 16^{th} OFDM received symbol. The mean square error is higher at 0^{th} symbol is because of using 2 bits coding representation of turnery bits as shown in table-1 where the bits 11 and 10 represents the same code which is 1. The simulation results are almost similar for 3PSK abd QPSK modulation schemes. However the QPSK/3PSK scheme shows less mean square error for higher number of OFDM symbols.

The fig 4.2 shows the convergence analysis convergence of proposed algorithm over TU channel with $E_b/N_0 = 12$ dB for 3PSK, QPSK and combination of 3PSK/QPSK. The results have been compared using mean square error of the proposed algorithm. The mean square error is 0.45 for the 0^{th} OFDM received symbol at the first point. It gradually decreases to 0.01 for the 16^{th} OFDM received symbol. The mean square error is higher at 0^{th} symbol is because of using 2 bits coding representation of turnery bits as shown in table-1 where the bits 11 and 10 represents the same code which is 1. The simulation results are almost similar for 3PSK abd QPSK modulation schemes. However the QPSK/3PSK scheme shows less mean square error for higher number of OFDM symbols.

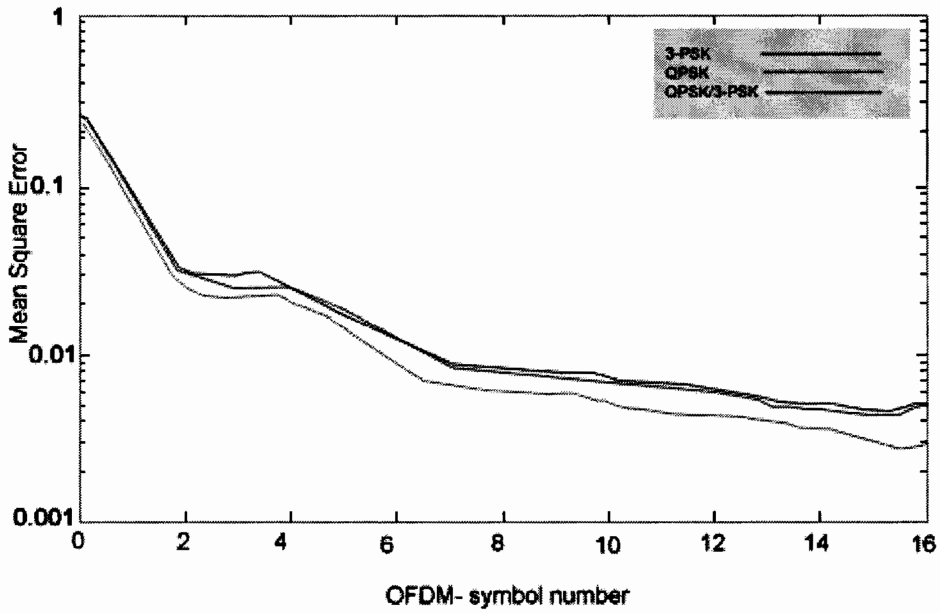

Fig 4.1 Convergence behavior of proposed algorithm over RA channel with E_b/N_0 = 12 dB for 3PSK, QPSK and combination of 3PSK/QPSK

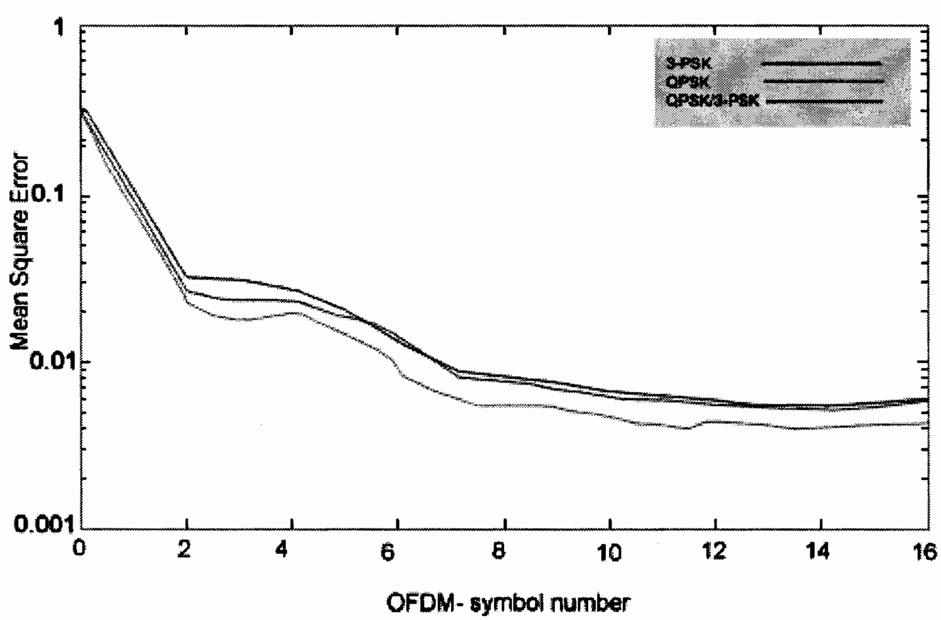

Fig 4.2 Convergence behavior of proposed algorithm over TU channel with E_b/N_0 = 12 dB for 3PSK, QPSK and combination of 3PSK/QPSK

The fig 4.3 shows the convergence analysis convergence of proposed algorithm over TU channel with E_b/N_0 = 12 dB for 3PSK, QPSK and combination of 3PSK/QPSK. The results have been compared using mean square error of the proposed algorithm. The mean square error is 0.45 for the 0^{th} OFDM received symbol at the first point. It gradually decreases to 0.01 for the 16^{th} OFDM received symbol. The mean square error is higher at 0^{th} symbol is because of using 2 bits coding representation of turnery bits as shown in table-1 where the bits 11 and 10 represents the same code which is 1. The simulation results are almost similar for 3PSK and QPSK modulation schemes. However the QPSK/3PSK scheme shows less mean square error for higher number of OFDM symbols. All the above three graphs shows that mean square error is merely same with same Eb/N_0 but has more or less similar impact on changing different channels (RA, BU, TU)

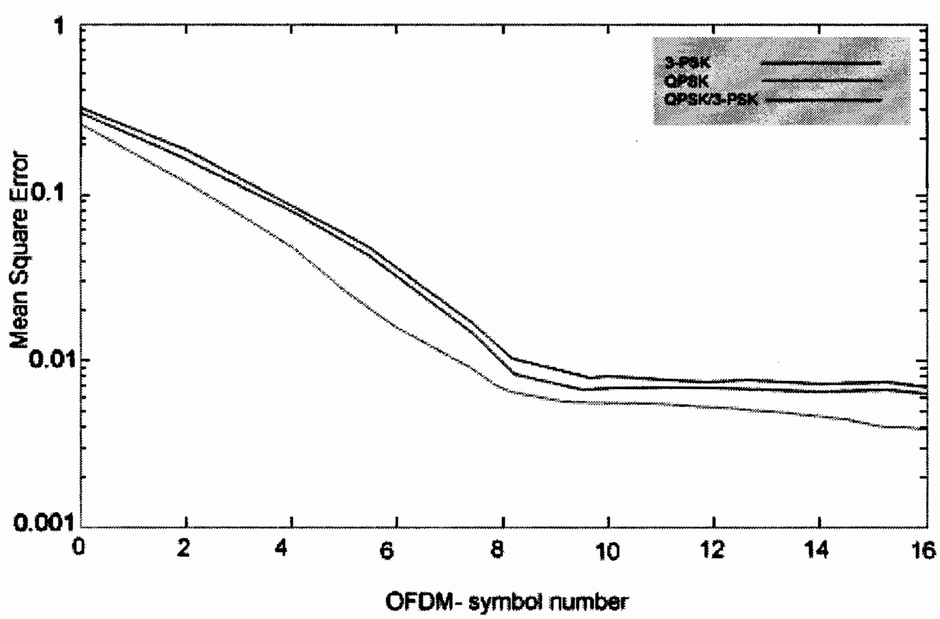

Fig 4.3 Convergence behavior of proposed algorithm over BU channel with E_b/N_0 = 12 dB for 3PSK, QPSK and combination of 3PSK/QPSK

The fig 4.4 shows the convergence analysis convergence of proposed algorithm over TU channel with E_b/N_0 = 12 dB for 3PSK, QPSK and combination of 3PSK/QPSK. The results have been compared using bit error rate of the proposed algorithm. The bit error is 10^{-1} for the 16^{th} OFDM received symbol at the first point. It gradually decreases to 10^{-4} for the 16^{th} OFDM received symbol.. The simulation results are almost similar for 3PSK and QPSK modulation schemes. However the QPSK/3PSK scheme shows less mean square error for higher number of OFDM symbols

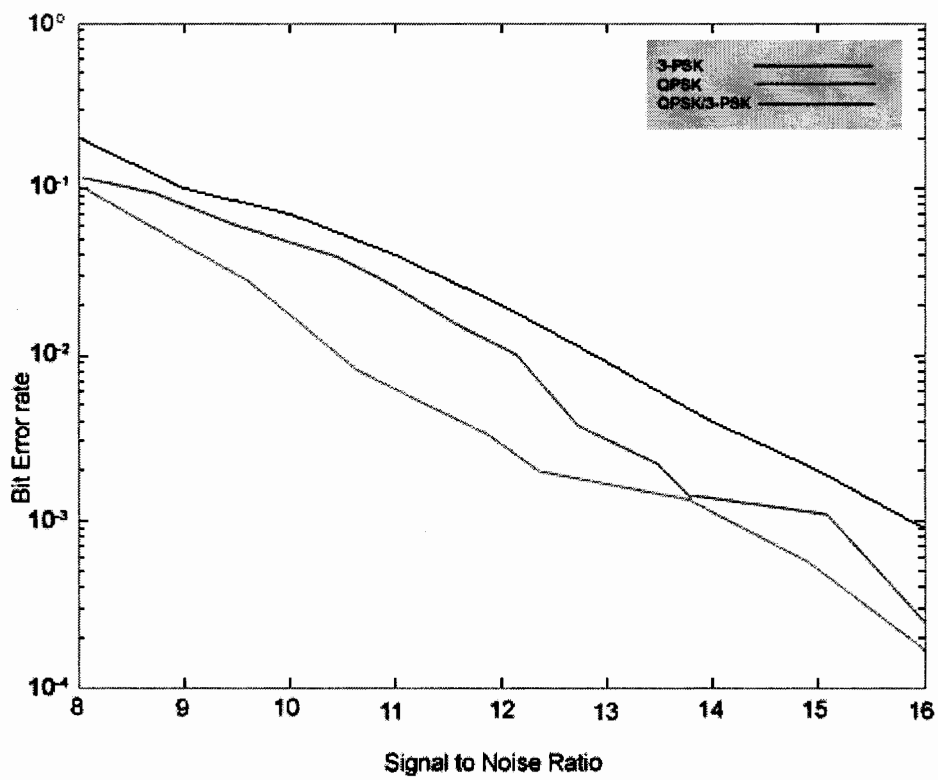

Fig 4.4 Convergence behavior of proposed algorithm over BU channel for 3PSK, QPSK and combination of 3PSK/QPSK with, $f_{D,max} = 193$ Hz

The fig 4.5 shows the convergence analysis convergence of proposed algorithm over RA channel with $E_b/N_0 = 12$ dB for 3PSK, QPSK and combination of 5PSK/QPSK. The results have been compared using mean square error of the proposed algorithm. The mean square error is 0.40 for the 0^{th} OFDM received symbol at the first point. It gradually decreases to 0.01 for the 16^{th} OFDM received symbol. The mean square error is higher at 0^{th} symbol is because of using 2 bits coding representation of turnery bits as shown in table-1 where the bits 11 and 10 represents the same code which is 1. After passing the same data through number of iterations, graphs shows that they converge gradually. The simulation results are almost similar for 5PSK and QPSK modulation schemes. However the QPSK/5PSK scheme shows higher mean square but we achieve these results without any iteration.

The fig 4.6 shows the convergence analysis convergence of proposed algorithm over RA channel with $E_b/N_0 = 12$ dB for 3PSK, QPSK and combination of 5PSK/QPSK. The results have been compared using mean square error of the proposed algorithm. The mean square error is 0.40 for the 0^{th} OFDM received symbol at the first point. It gradually decreases to 0.01 for the 16^{th} OFDM received symbol. The mean square error is higher at 0^{th} symbol is because of using 2 bits coding representation of turnery bits as shown in table-1 where the bits 11 and 10 represents the same code which is 1. After passing the same data through number of iterations, graphs shows that they converge gradually. The simulation results are almost similar for 5PSK and QPSK modulation schemes. However the QPSK/5PSK scheme shows higher mean square but we achieve these results without any iteration.

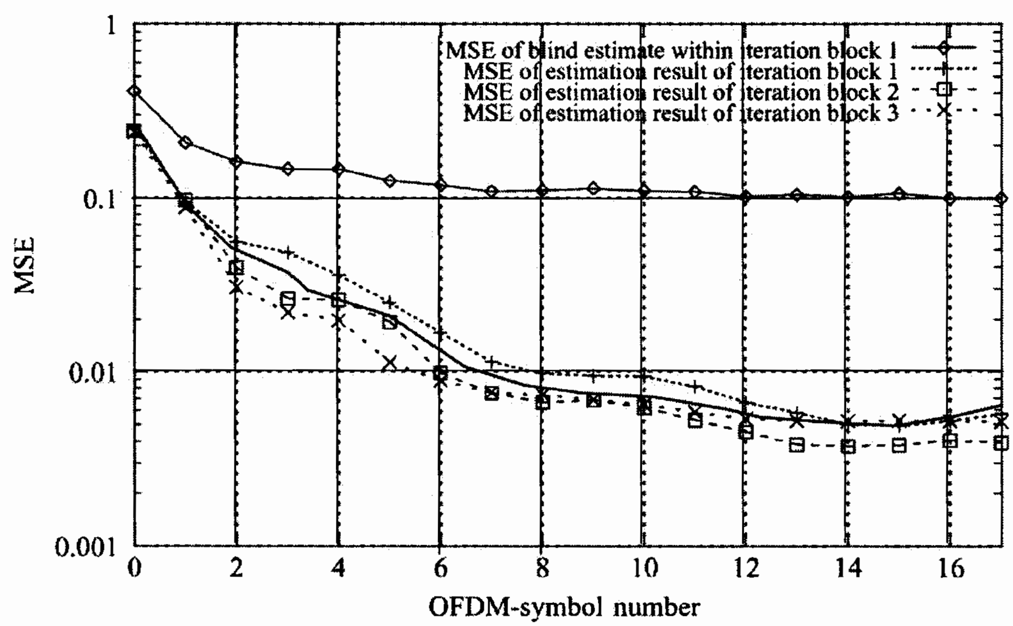

Fig 4.5 Convergence behavior of proposed algorithm over RA channel with $E_b/N_0 = 12$ dB for 5PSK, QPSK and combination of 5PSK/QPSK

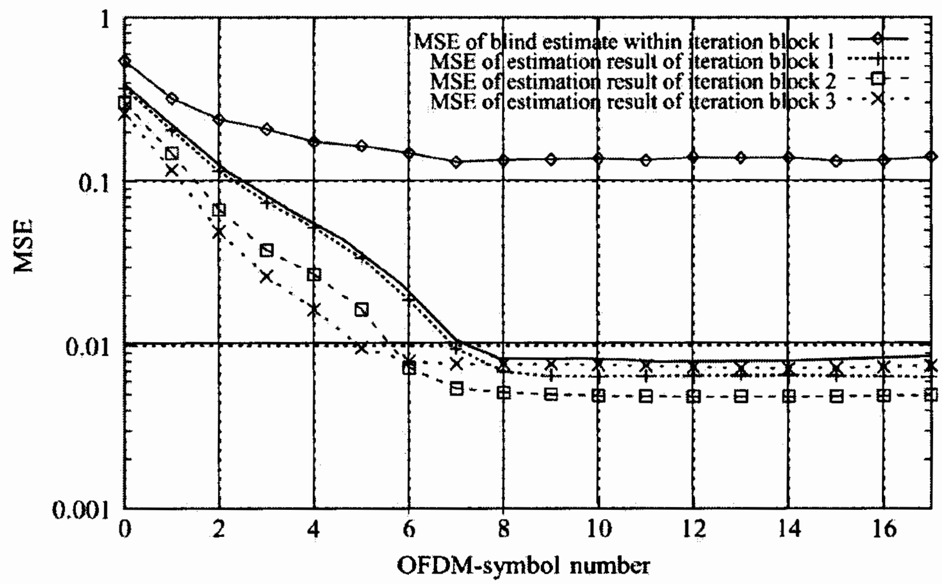

Fig 4.6 Convergence behavior of proposed algorithm over TU channel, E_b/N_0 = 12 dB, QPSK/5-PSK.

The fig 4.7 shows the convergence analysis convergence of proposed algorithm over RA channel with E_b/N_0 = 12 dB for 3PSK, QPSK and combination of 5PSK/QPSK. The results have been compared using mean square error of the proposed algorithm. The mean square error is 0.40 for the 0^{th} OFDM received symbol at the first point. It gradually decreases to 0.01 for the 16^{th} OFDM received symbol. The mean square error is higher at 0^{th} symbol is because of using 2 bits coding representation of turnery bits as shown in table-1 where the bits 11 and 10 represents the same code which is 1. After passing the same data through number of iterations, graphs shows that they converge gradually. The simulation results are almost similar for 5PSK and QPSK modulation schemes. However the QPSK/5PSK scheme shows higher mean square but we achieve these results without any iteration.

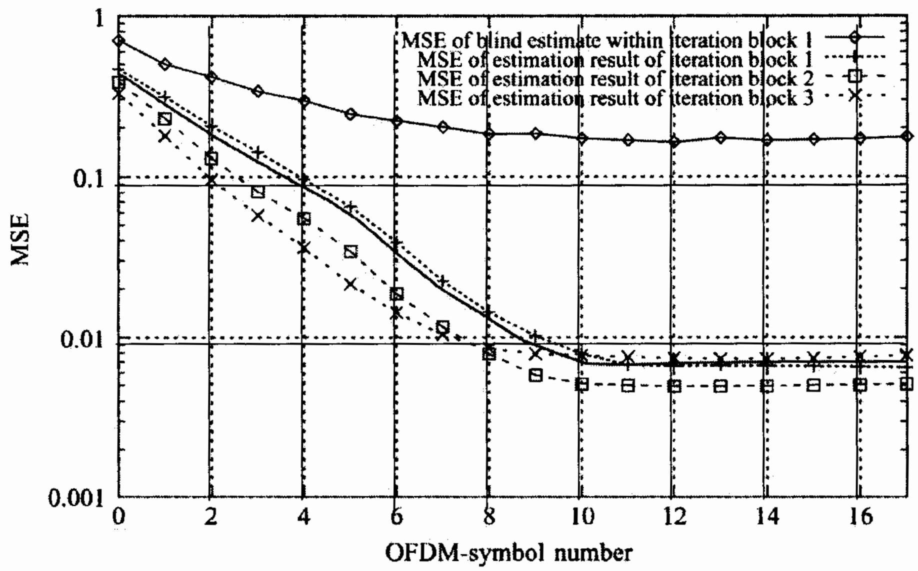

Fig 4.7 Convergence behavior of proposed algorithm over, E_b/N_0 = 12 dB, QPSK/5-PSK.

Chapter 5

5.1 Conclusion

A novel blind channel estimation scheme was presented with some modification. The approach to estimate the channel blindly without the use of 2^{nd} ordered and higher order statistics. To estimate channel blindly a modified basic maximum likelihood estimator is developed. This can estimate the channel blindly without the use of higher order statistics. This estimation method can successfully recover the magnitude but phase ambiguity is still present. To recover the phase a new approach is established which combine two different modulation schemes (QPSK/3PSK). With the combined modulation scheme, the absolute phase of the channel transfer function can be recovered without the need for reference symbols. During coherent demodulation the results are unique for each combination of the combined modulation, which prove it theoretical and statistical by simulations

The results clearly indicate the feasibility of the proposed approach. Finally, the proposed approach maximizes the spectral efficiency by avoiding any reference symbols or pilots, while improving the performance by using a coherent detection rather than differential detection.

The QPSK/5-PSK scheme provides stronger coding than the QPSK/3-PSK scheme. On the other hand, QPSK/5-PSK has smaller minimum distance between any two symbol cancellations. We can use this algorithm and this multi modulation scheme with some other available techniques other than OFDM.

Simulation Code:

```
fft_size = 32;
num_carriers = 32;
k=2;

SNR=[20 23 27 29 32 35];
EbNo=SNR-10*log10(k);

loop=length(SNR);
prob_error=zeros(1,loop);
for simul_num=1:loop


data=round(rand(1000000));
data_copy=data;
```

Modifying data to send only first bit if either 10 or 11 appears in case of 3-PSK

```
data_s=length(data);
for i=1:4:data_s
   if (data(i)==1)
      data=[data(1:i) 1 data(i+1:data_s)];
      data_s=data_s+1;
   end
end

if(rem(data_s,2)~=0)
   data=[data 1];
   data_s=data_s+1;
```

```
end

%%%%%size adjustment%%%%%

new_data_s=ceil(data_s/(2*num_carriers))*2*num_carriers;
data_add=zeros(1,new_data_s-data_s);
data=[data,data_add];
data_s=new_data_s;

num_sym=data_s/2;
data_sym=zeros(1,num_sym);

zerozero=num2str([0 0]);
zeroone=num2str([0 1]);
onezero=num2str([1 0]);
oneone=num2str([1 1]);

for i=2:2:data_s
   check=num2str(data(i-1:i))
   if(rem(i/2,2)==0)
      switch check
         case zerozero
                           data_sym(i/2)=0.7071+0.7071j;
                     case zeroone
                           data_sym(i/2)=-0.7071+0.7071j;
                     case onezero
                           data_sym(i/2)=0.7071-0.7071j;
                     case oneone
                           data_sym(i/2)=-0.7071-0.7071j;
                     otherwise
                           disp('Error detected in switch statment - This should not be
happening.');
```

```
                end
    else
        switch check
            case zerozero
                        data_sym(i/2)=1+0j;
                case zeroone
                        data_sym(i/2)=-0.5+0.866j;
                case onezero                    %no need of this check as data is
preadjusted
                        data_sym(i/2)=-0.5-0.866j;
                case oneone
                        data_sym(i/2)=-0.5-0.866j;
                otherwise
                        disp('Error detected in switch statment - This should not be
happening.');
        end
    end
end

num_chunks = ceil(num_sym/num_carriers);
chunks = zeros(num_chunks,num_carriers);
for i = 1:num_chunks
        chunks(i,:) = data_sym(num_carriers*(i-1)+1:num_carriers*i);
end

td_sets = zeros(num_chunks,fft_size);
for i = 1:num_chunks
        td_sets(i,:) = ifft(chunks(i,:),fft_size);
end

xmit = zeros(1,num_chunks*fft_size);
for i = 1:num_chunks
```

```
        xmit(1,(i-1)*fft_size+1:i*fft_size) = td_sets(i,:);
end
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%Channel%%%%%%%%%%%%%%%%%%%%%%%%%
```

%ch_multipath

```
d1 = 6;                 % delay in units
a1 = 0.30;      % attenuation factor
d2 = 10;        % delay for second multipath signal
a2 = 0.25;      % attenuation factor for second multipath signal
copy1=zeros(size(recv));
for i=1+d1:length(recv)
        copy1(i)=a1*recv(i-d1);
end

copy2=zeros(size(recv));
for i=1+d2:length(recv)
        copy2(i)=a2*recv(i-d2);
end

recv=recv+copy1+copy2;
```

```
%Receive

recv_td_sets = zeros(num_chunks,fft_size);
for i = 1:num_chunks
	recv_td_sets(i,:) = recv(1,(i-1)*fft_size+1:i*fft_size);
end

% perform fft to recover original data from time domain sets
recv_chunks = zeros(num_chunks,num_carriers);
for i = 1:num_chunks
	recv_chunks(i,:) = fft(recv_td_sets(i,:),num_carriers);
end

recv_sym=zeros(1,num_chunks*num_carriers);
for i = 1:num_chunks
	recv_sym(num_carriers*(i-1)+1:num_carriers*i)=recv_chunks(i,:);
end
```

```
%ML detection

recv_data=zeros(1,data_s);
threePSK_test=[1,-0.5+0.866j,-0.5-0.866j];
QPSK_test=[0.7071+0.7071j,-0.7071+0.7071j,0.7071-0.7071j,-0.7071-0.7071j];
for i = 1:2:num_sym

  %3-PSK

  sym_check=recv_sym(i)*ones(1,3);
  distance=abs(sym_check-threePSK_test);
  [value,index]=min(distance);
  switch index
    case 1
      recv_data(1,2*i-1:2*i)=[0 0];
    case 2
      recv_data(1,2*i-1:2*i)=[0 1];
    case 3
      recv_data(1,2*i-1:2*i)=[1 1];
  end

  %QPSK

  sym_check=recv_sym(i+1)*ones(1,4);
  distance=abs(sym_check-QPSK_test);
  [value,index]=min(distance);
  switch index
    case 1
      recv_data(1,2*i+1:2*i+2)=[0 0];
    case 2
```

```
            recv_data(1,2*i+1:2*i+2)=[0 1];
        case 3
            recv_data(1,2*i+1:2*i+2)=[1 0];
        case 4
            recv_data(1,2*i+1:2*i+2)=[1 1];
    end

end

recv_len=length(recv_data);
error=sum(((recv_data-data))~=0);
prob_error(simul_num)=error/recv_len;

%Deleting the dummy bits inserted for 3-PSK
i=1;
while (i<recv_len)
    if (recv_data(i:i+1)==[1 1])
        recv_data=[recv_data(1:i) recv_data(i+2:recv_len)];
        recv_len=recv_len-1;
        i=i-1;
    end
    i=i+4;
end

end
```

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