

Adaptive Modulation and Coding in COFDM for WiMAX Using LMS Channel Estimator

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Abstract

The OFDM technique is predominantly used during the implementation of WiMAX Physical layer. This paper focuses on the PHY-layer design aspects, namely, modulation and coding techniques associated. OFDMA, an extension of OFDM, makes use of Adaptive Modulation and Coding techniques to improve efficiency, fairness, and throughput in WiMAX. To achieve higher data rates and smaller BER's channel coding can be carried out in OFDM, called COFDM. The channel state information is fed back to the transmitter by the channel estimator. The simulation analysis presented includes comparison of BER vs. SNR for different modulation schemes. Here, LMS channel estimator is used.

Keywords: WiMAX, OFDM, COFDM, OFDMA, CSI, LMS

1. INTRODUCTION

WiMAX (Worldwide Interoperability for Microwave Access) is a wireless broadband access technology that offers performance similar to Wi-Fi (IEEE 802.11) networks with the coverage and QoS (quality of service) of mobile networks. WiMAX can offer broadband wireless access (BWA) up to 50 km for fixed stations (called as Fixed WiMAX (IEEE 802.16d)), and 5-15 km for mobile stations (called as Mobile WiMAX (IEEE 802.16e-2005)). In contrast, the Wi-Fi/802.11 wireless local area network standard provides access up to 30-100m only. WiMAX radio antenna technology coupled with the underlying benefits of OFDM (Orthogonal Frequency Division Multiplexing) /OFDMA (The Orthogonal Frequency Division Multiple Access) based radios can be in range and bandwidth capacity. WiMAX provides a pair of mechanisms that insure good QoS. First, modulation (64-QAM/16-QAM/QPSK) and coding schemes insure steady signal strength though the distance increases. Secondly, Dynamic Bandwidth Allocation (DBA) is a mechanism that supervises the network and, when interference or other detraction occurs to signal strength, the base station apportions more bandwidth and power for the affected stream.

Following this introduction the rest of the paper is organized as follows. Section 2 gives the interpretation of OFDM and COFDM. This section explains the concept behind OFDM and shows how the COFDM is better than OFDM. In Section 3, the details of OFDMA and AMC system and explains different algorithms for channel estimation. Then, in Section 4, Simulation results have been discussed. Finally, conclusions are provided in Section 5.

OFDM is expended in various broadcast technologies like WLAN, WiMAX, DVB and DAB. OFDM is a multicarrier system which employs a prominent number of close spaced carriers that are regulated with low data rate. In order to avoid mutual interference, the signals are made as orthogonal to each other. OFDMA is strategy applied to provide a multiple access capability for applications such as mobile telecommunications when using OFDM technologies. COFDM [3] is a kind of OFDM where error correction coding is comprised into the signal. Because of the merged benefits of multicarrier modulation and coding, COFDM system is able to attain excellent performance on frequency selective channels.

The multicarrier nature of the OFDMA transmission reaches better results if incorporated with additive techniques in order to achieve higher efficiency in terms of throughput and error rate. Adaptive Modulation and Coding (AMC) [4] allows OFDMA systems to choose the most appropriate Modulation and Coding Scheme (MCS) depending on the propagation considerations of the communication channel. This paper aims to develop techniques that improve the system performance in terms of suitable QoS metrics: keep the error probability below a specified threshold or maximize the data throughput. The MCS selection is based on the amount of received packets with errors, having as target that of minimizing that number and reducing the implementation complexity by using different algorithms.

2. Interpretation of OFDM AND COFDM

The basic principle of OFDM is to split the wideband high data rate incoming stream into narrowband data rate stream and transmitted over a number of subcarriers at the same time. OFDM utilizes multiple sinusoidal having frequency separation $1/T$ ('T' is Symbol duration). Where each sinusoidal is modulated by independent data. The used sinusoidal in OFDM can be determined as,

$$g_k(t) = \frac{1}{\sqrt{T}} \exp \frac{j2\pi kt}{T} w(t) \quad (1)$$

$k=0,1,\dots,K-1$ correspond to the frequency of the sinusoidal and $w(t)=u(t)-u(t-T)$ is a rectangular window over $[0,T)$.

The information a_k the corresponding carrier $g_k(t)$ and the sum of such modulated sinusoidal form the transmit signals. Mathematically, the transmit signal is,

$$s(t) = \frac{1}{\sqrt{T}} \sum_0^{K-1} a_k \exp \frac{j2\pi kt}{T} \quad (2)$$

The interpretation of the above equation is as follows:

- (a) Each information signal a_k multiplies the sinusoidal having frequency of k/T .
- (b) Sum of all such modulated sinusoidal are summed and the resultant signal is transmitted out as $s(t)$.

The sampled version of above equation is,

$$s(nT) = \frac{1}{\sqrt{T}} \sum_0^{K-1} a_k \exp \frac{j2\pi knT}{T} w(nT) \quad (3)$$

It is reasonable to understand that above operation corresponds to an inverse Discrete Fourier Transform (IDFT) operation.

In addition to the multipath channel, the received signal gets disturbed by noise, typically referred to as Additive White Gaussian Noise (AWGN). The values of the noise follow the Gaussian probability distribution function,

$$p(x) = \frac{1}{\sqrt{2\pi\sigma^2}} \exp \frac{-(x - \mu)^2}{2\sigma^2} \quad (4)$$

Mean $\mu=0$ and variance $\sigma^2 = N_0/2$. The received signal is $y[k]=s[k] \otimes h[k]+n$. \otimes is the convolution operator.

The probability of BER (P_b) for different modulation scheme (BPSK, QPSK, 16QAM, etc.) in the AWGN channel is given by [7],

$$P_{bBPSK} = P_{bQPSK} = Q(\sqrt{2\gamma_b}) \quad (5)$$

$$P_{bMQAM} = \frac{2}{\log_2 M_a} \left(1 - \frac{1}{\sqrt{M_a}}\right) Q\left(\sqrt{\frac{3\gamma_b \log_2 M_a}{M_a - 1}}\right), \quad (6)$$

where M_a is order of Modulation scheme, $Q =$ Complementary error function, $\gamma_b =$ Signal to Noise ratio.

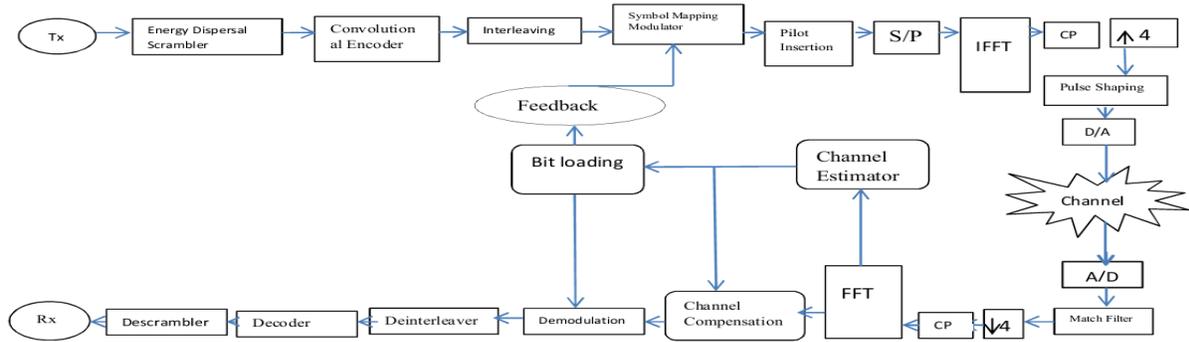


Figure 1. COFDM Transmitter and Receiver block diagram.

If FFT is used at the receiver and compute correlation values with the center of frequency of each subcarrier, it is possible to recover the transmitted data with nearly no crosstalk. The COFDM system comprises of three main elements. These are Guard interval/cyclic prefix, Channel coding/interleaving and IFFT. These technical aspects build the system tolerant to multipath fading and ISI.

Description of WiMAX-PHY layer with COFDM is listed below:

1. Information Source - To generate the information bits randomly.
2. Energy dispersal scrambler - To ensure appropriate energy dispersal in the transmitted signal, the single inputs of the energy dispersal scramblers will be scrambled by a modulo-2 addition with a pseudo-random binary sequence (PRBS), prior to convolutional encoding.

3. Convolutional encoder To encode the information data with coding rate $R = 1/2, 2/3,$ or $3/4$, matching to the wanted data rate. The convolutional en-coder utilizes the industry-standard generator Polynomials, [133, 171] of rate $R = 1/2$ [6].
4. Interleaving To eliminate the effects of selective fading. It o sets any deep fades that happen in the wireless channel by disseminating the data bits over the sub-carrier channels. It is essential for proper function of forward error correction [7].
5. Modulation Use different modulation schemes (BPSK, QPSK, 8QAM, 16QAM, 32QAM, and 64QAM) convert the given binary bits into complex signals.
6. Performed serial to parallel conversion.

7. FFT and IFFT To do linear mappings between N complex data symbols and N complex OFDM symbols result in hardness against multipath channel fading [2]. Length of FFT/IFFT is taken to be 1024 (Mobile WiMAX specifications).. Length of FFT/IFFT is taken to be 1024 (Mobile WiMAX specifications).
8. Guard time insertion -For making OFDM symbols tolerant to inter symbol interference (ISI). It copies last samples (equal to guard interval) from each OFDM symbol and place it at the starting of the OFDM symbol.
9. Performed parallel to serial to conversion.
10. The signals are carried over an AWGN (Additive White Gaussian Noise) channel.
11. At the receiver side, do reverse processes to decode the received sequences of data bits.
12. Count the number of spurious bits by equating the decoded bit sequences with original one.
13. Compute the Bit Error Probability (BER) against different values of signal to noise ratio (SNR) and plot it accordingly.

3. OFDMA - AMC System

In this section the proposal of AMC systems exploiting the OFDMA potentialities is described considering that the modulation order and coding rate of each complex symbol c_k linked to the corresponding k_{th} subcarrier ($0 \leq k \leq 1023$) could be changed according to the physical channel state. This paper considers LMS method [8] as the channel estimator and gives the information the transmitter about the channel through feedback. The channel estimator provides the knowledge on the Channel Impulse Response (CIR) to detectors. The channel estimation is based on the known sequence of bits which is singular for a particular transmitter and which is reoccurred in every transmission burst. The transmitted symbols going through the radio channel can be posed as a circular convolution between the channel impulse response (CIR) and the transmitted data block i.e., $[s(m)*w(\square, t)]$. Since the channel coefficient is normally unknown to the receiver, it demands to be efficiently estimated while maintain low computational complexity. At the

receiver, reverse operation can be performed. After synchronization, cyclic prefix samples are removed and the remaining N samples are processed by the FFT to recover the complex constellation symbols transmitted over the orthogonal sub-channels. The received signals are remapped and equalizer is applied to compensate for the radio channel frequency selectivity. After IFFT operation, these received signals are extracted and the corresponding bits are processed to the decoder. The decoder analyses the structure of received bit pattern and tries to rebuild the original signal. In order to achieve better performance the receiver has to know the impact of the channel.

3.1. Least Mean Square (LMS) Algorithm

The basic principle of LMS algorithm is stochastic gradient and it applies one tap LMS adaptive filter at every pilot frequency. The first value is found directly through LS and the remaining values are computed based on the previous estimation and the current channel output. The LMS estimator is used mainly for the tracking of the channel and is usually clumped with an equalizer or a decision feedback equalizer[9].

$$e(m) = S^T(m)w(m) + z(m) - S^T(m)h(m) \quad (7)$$

$$h(m+1) = h(m) + \eta S(m)e(m) \quad (8)$$

Where η is step size, $S(m)$ is the transmitted diagonal matrix at sampling time m , $h(m)$ is the adaptive filter coefficient, and $e(m)$ is the estimation error. The filter coefficients are modified using an estimate of the cost function gradient, $[\eta S(m)e(m)]$. When the noise is involved in the received sequence, interference will also in the coefficients adaption process through the term $[\eta S(m)e(m)]$.

The step size parameter, η decides the convergence rate of the algorithm and higher value renders faster convergence. Also a higher value of η results in higher variations in the tap weight vector estimate after the initial convergence phase. The LMS estimator applies one tap LMS adaptive filter at every pilot frequency. The first value is found directly through LS and the remaining values are computed based on the previous estimation and the current channel output. The LMS estimator is used mainly for the tracking of the channel and is usually clumped with an equalizer or a decision feedback equalizer.

3.2. Normalized LMS (NLMS) Algorithm

The primary trouble given by the LMS CE algorithm is that it is sensible to the scaling of its input signals. This causes it very hard to select h that guarantees stability of the algorithm. The NLMS is a variant of the LMS algorithm that resolves this problem by annealing with the power of the input signal. The NLMS algorithm can be resumed as,

$$h(m+1) = h(m) + \eta e(m) [S^T(m)S(m)]^{-1} S(m) \quad (9)$$

when a constant scalar step size is maintained in the LMS/NLMS algorithm, there is a trade off among the steady state error-convergence towards the true channel coefficients, which averts a fast convergence when the step size is favored to be small for small output estimation error.

3.3. Variable Step Size (VSS)-LMS Algorithm

The VSS algorithm is,

$$\eta(m+1) = \alpha \eta(m) + \gamma p^2(m) \quad (10)$$

$$p(m) = \beta p(m) + (1 - \beta) e^T(m) e(m-1), \quad (11)$$

where $0 < \alpha < 1$, $0 < \beta < 1$ and $\gamma > 0$.

Control parameters α and β need to be corrected for a better performance.

3.4. Recursive Least Squares (RLS) Algorithm

This algorithm is frequently used for rapid convergence and improved MSE performance [9]. The standard RLS algorithm is,

$$R(m) = B(m-1)S(m)[\lambda + S^T(m)B(m-1)S(m)]^{-1} \quad (12)$$

$$B(m) = \lambda^{-1}B(m-1) - \lambda^{-1}R(m)S^T(m)R(m-1) \quad (13)$$

$$h(m+1) = h(m) + S(m)e(m)R(m), \quad (14)$$

where λ is the exponential forgetting factor with $0 < \lambda < 1$. The smaller value of λ leads to faster convergence rate as well as larger variations in the weight signal after the initial convergence. But then too small λ value causes this algorithm unstable.

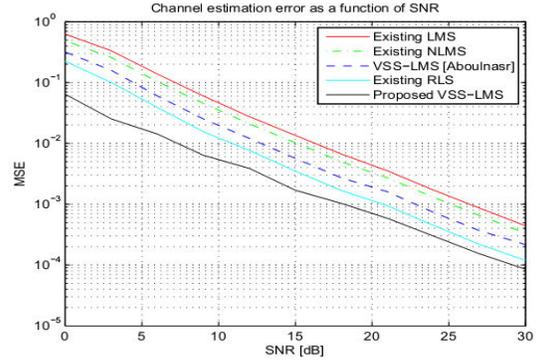


Figure 2: Comparison of MSE behaviors of all CE algorithms for $f_d = 100$ Hz

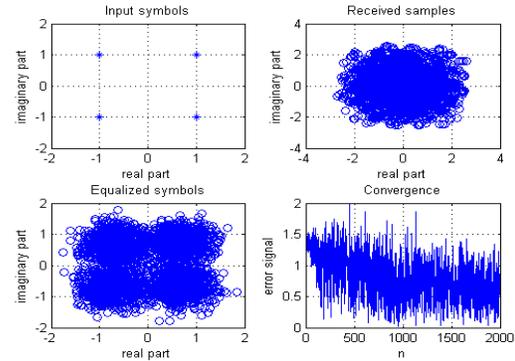


Figure 3: LMS Channel Equalization.

4. SIMULATION RESULTS

This paper considers an OFDM binary transmission with $N=1024$ and channel length $L=16$ and each packet contains 8 pilots only. In this section BER vs. SNR performance is compared for various modulation schemes (BPSK, QPSK, 8PSK, 16QAM, 32QAM, 64QAM and 256QAM) with Coded OFDM and uncoded OFDM. Fig. 4 shows the gap-to-capacity is denigrated with respect to a (3, 1) repetition code or the suboptimal un-coded code word. In this paper, the gap-to-capacity (given by Shannon's theoretical limit) of $1/3$ code rate (also can be modified for $1/N$) is minimized. Fig. 5 Shows, plot between BER vs. SNR for different modulation schemes with code rate $1/2$, $3/4$ in the AWGN channel for COFDM. The results show that COFDM is well suited for high speed data transmission in mobile WiMAX and interleaving is essential for reducing bit error rate for high speed transmission. The convolutional code is considered for its simplicity and generality, though the better strategies are available (such as LDPC and turbo codes) [10]. Turbo code doesn't perform a proper performance in multipath

channels, Convolutional Coding is considered due to its less complexity, but in packet error probabilities Turbo code is preferred.

Table 1. Parameters Definition

Parameters	Values
Number of Sub channels (N)	1024
Number of Pilots (P=N/8)	128
Num of Data Subcarriers (N-P)	896
Guard Interval Length (GI=N/4)	256
Pilot position Interval	8
Channel length	16
Channel	AWGN
Digital Modulations	BPSK, QPSK, 8PSK, 16QAM, 64QAM, 256QAM

The applied OFDM parameters are listed in Table. 1. In adaptive modulation technique modulation scheme (BPSK, QPSK, 8QAM, 16QAM, 64QAM and 256QAM) and convolution code rate are altered according to the variation in the communication channel. The transmitter will choose the appropriate modulation scheme and code rate is depend upon the SNR threshold or BER value.

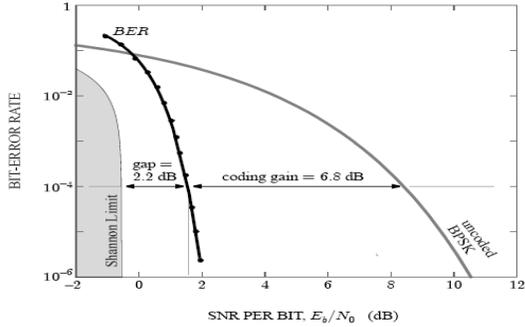


Figure 4: BER performance for coded and Uncoded OFDM [9].

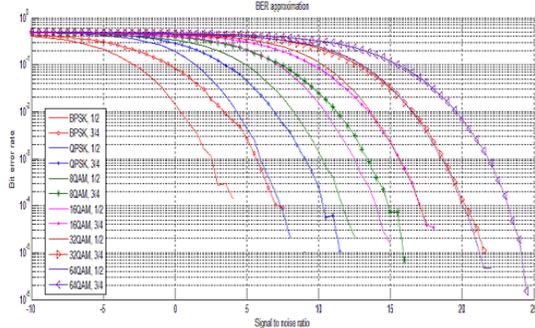


Figure 5: BER performance for different modulation schemes with CC rate 1/2, 1/3.

Table 2. Obtained results for LMS channel estimator

SNR (db)	BPSK (BER)	QPSK (BER)	8PSK (BER)	16QAM (BER)	64QAM (BER)	256QAM (BER)
0	0.09	0.07	0.776	0.895	0.979	0.9956
3	0.1	0.08	0.724	0.872	0.977	0.9955
6	0.063	0.09	0.676	0.853	0.972	0.994
9	0.025	0.1	0.630	0.831	0.966	0.9934
12	0.012	0.050	0.588	0.803	0.959	0.9933
15	0.01	0.015	0.549	0.794	0.954	0.9931
18	0.003	0.012	0.537	0.769	0.944	0.990
21	0.002	0.01	0.524	0.762	0.939	0.989
24	0.001	0.003	0.514	0.760	0.935	0.988
27	0.0007	0.001	0.512	0.758	0.933	0.986

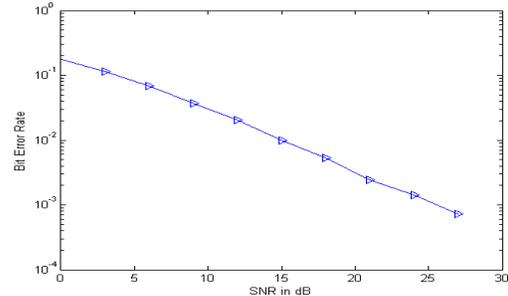


Figure 6: performance of LSE algorithm for BPSK modulation in OFDM channel estimation.

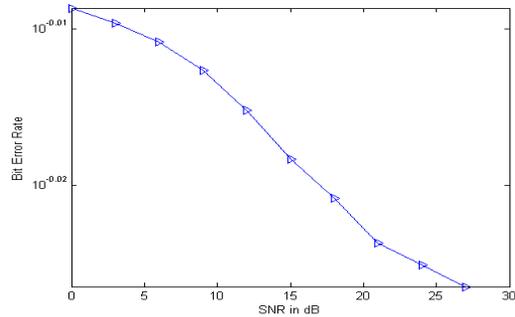


Figure 7: Performance of LSE algorithm for 64QAM modulation in OFDM channel estimation.

Table. 2 has been prepared for LMS channel estimator by varying modulation schemes order 'M'. Table 2 shows that 'M' increases BER also increases for a given SNR value and as SNR increases BER value are reduced for a given 'M' value. Based on this SNR or

BER value, required modulation scheme is selected by channel estimator.

5. CONCLUSION

This paper compares the Coded OFDM and Uncoded OFDM performance in terms of BER for different modulation schemes over an AWGN channel. COFDM is powerful modulation technique to achieve higher bit rate, long range data access and to eliminate ISI. So, the COFDM has chosen for high speed digital communications like Wi-Fi, WiMAX, etc. At lower SNR values BPSK is preferred because it gives lower BER values compared to remaining techniques. As modulation order increases, BER also increases for a given SNR value. This paper discusses the importance of Adaptive Modulation Coding schemes in an OFDMA based system. LMS channel estimator is considered because of its low complexity. The performance of AWGN channel is measured in terms of BER and SNR. The most suitable MCS is chosen at the transmitter on the basis of channel estimator in terms of BER.

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