Performance Evaluation of WiMAX System OFDM Physical Layer for Flat-Fading and Multipath-Fading Channel Environments

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ABSTRACT

This paper represents an end-to-end baseband model of the physical layer of a WiMAX network according to the IEEE® 802.16 standard. The model supports all mandatory and optional data rates 27 Mb/s. The model also illustrates adaptive modulation and coding over a flat-fading and dispersive multipath fading channel, whereby the simulation varies the data rate dynamically The channel estimation technique uses training sequence inserted at the beginning of each transmission. Theory and methods are presented in details. The system model is designed for WiMAX IEEE 802.16 transceiver in MATLAB Simulink and the performances are evaluated using software Matlab.

***Keywords*-Adaptive modulation, Multipath Fading, Equalizer, OFDM, WiMAX.**

I. INTRODUCTION

In recent years, Mobile WiMAX has received wide interest for next generation wireless communications. Two key technologies, Multiple Input Multiple Output (MIMO) and Orthogonal Frequency Division Multiplexing (OFDM) have been adopted in Mobile WiMAX standard, IEEE 802.16e (Mobile WiMAX) which is an extension of the IEEE 802.16-2004, which enable high data rate transmission over multipath and frequency selective fading channels Accurate channel state information (CSI) is required to perform coherent detection of OFDM signals and estimating mobile channels for high data rate communications has proved to be very challenging [10].

Channel estimation of MIMO-OFDM systems has been widely investigated by researchers and two kinds of approaches have been introduced, frequency and time domain channel estimation. In these methods, a complete OFDM frame composed of training symbols is first sent in order to estimate the channel parameters. The channel is therefore assumed to be constant for the next OFDM blocks until a new estimation is performed. In fast fading environment, performances degradation would be noticed due to the outdated channel estimation. Normally, estimation performed in time domain would be more appropriate as impulse responses contain fewer parameters than frequency responses.

Simulations were conducted based on the specifications given in the IEEE 802.16e standard for WiMAX and under Stanford University Interim (SUI) channel models.

II. SYSTEM MODEL

OFDM system works on the principle of division of the available frequency spectrum into several subcarrier channel frequencies. The subcarriers are orthogonal in nature and hence overlapping is possible to achieve better spectral efficiency. Even though the signal passes through a time dispersive fading channel, orthogonality can be completely maintained with a smallprice in a loss in SNR[11].

A block diagram of a WiMAXOFDMPHY system is shown in Fig. 1[4]. The random binary signal information is first generated and grouped in symbols, then coded for error correction. Adaptive modulation system is used because different modulation scheme is needed for different data rates. After that guard band is inserted and the Inverse Fast Fourier Transform (IFFT) block transforms the data sequence into time domain. If multiple antennas are used, then space-time diversity encoder is also implemented. Then a cyclic prefix is used which is chosen larger than the expected delay spread to avoid inter-symbol and inter-carrier interferences (ISI and ICI). The D/A converter contains low-pass filters with bandwidth $1/T_S$, where T_S is the sampling interval. The channel is considered to be a multipath fading channel followed by addition of white Gaussian noise.

At the receiver, after passing through the analog-to-digital converter (ADC) and removing the CP, the FFT is used to transform the data back to frequency domain. Adaptive filtering technique is used for channel estimation. Lastly, the binary information data is obtained back after the demodulation and channel decoding.

1.1 Forward Error Correction Code

Digital communications need forward error correction (FEC) code for reducing errors due to noise, fading, interference, and other channel impairments. It is also called channel coding or error control coding. It introduces controlled redundancies to a transmitted signal sothat they may be exploited at the receiver. The redundancy of the

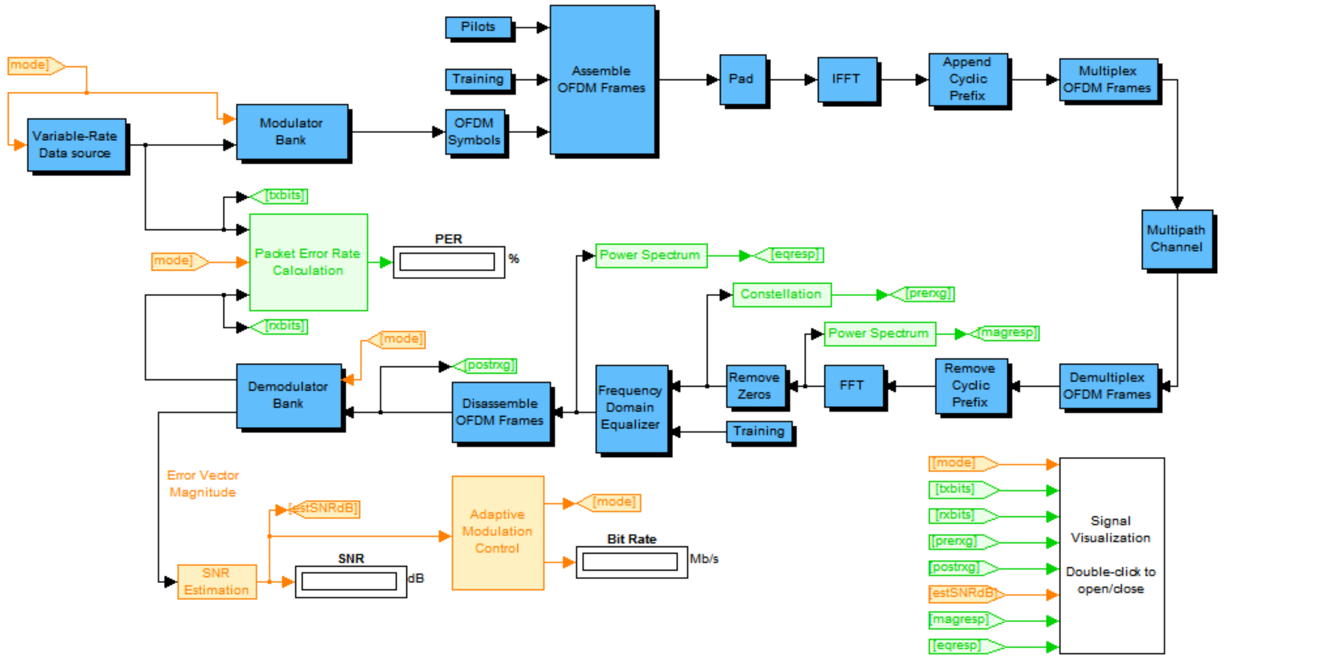

Fig. 1 IEEE 802.16 WiMAX - OFDM PHY system model in MATLAB Simulink[4].

encoded data is quantified by the ratio of b_d data bits per b_c encoded bits

$$R_c = \frac{b_d}{b_c} \text{ where } (b_d < b_c) \quad (1)$$

This ratio is known as the code rate, which basically means for every b_d data bits input into the encoder, there will be b_c code bits output from the encoder. Generally, correction codes are categorized into two main types: block codes and convolutional codes. Block coding is basically generating a “codeword” of b_c coded bits that would be algebraically related to a sequence of b_d data bits. Convolutional coding is generated by the discrete-time convolution of a continuous sequence of input data bits. Both block code and convolutional code are employed in wireless systems. However, convolutional codes have proven superiority over block codes for a given degree of decoding complexity[14].

1.2 Block Interleaving

The decoder operates under the assumption that the errors will be random or spaced apart. However, in fading channels, deep fades may cause a long sequence of errors, which may render the decoder ineffective. In order to alleviate bit correlation, the encoded bits are scrambled with a block interleaver (fig.2). Interleaving can either be done before symbol mapping, and is known as bit interleaving, or it can be done after symbol mapping, and is known as symbol interleaving . The purpose of interleaving is to minimize the bit correlation, while data symbols are essentially groups of bits. Maintaining the bits in groups, in a sense, adds to the bit correlation, especially when using larger constellations[11].

1.3 Modulation Schemes

Adaptive modulation systems improve the rate oftransmission. The implementation of adaptive modulation isaccording to the channel information that is present at the transmitter.

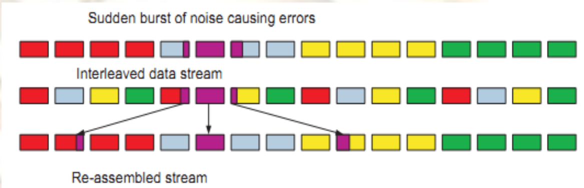

Fig. 2 Transmission with a burst error and interleaving.

TABLE I. IEEE 802.16 TRANSMISSION MODES

Mode or Rate_ID	Modul-ation	Interle a-ving	Code Rate	Data Rate (Mbps)	Simulation-Threshold SNR (dB)
0	BPSK	6 x 8	1/2	3	0
1	BPSK	6 x 8	3/4	4.5	10
2	QPSK	12 x 8	1/2	6	11
3	QPSK	12 x 8	3/4	9	14
4	16-QAM	12 x 16	1/2	12	18
5	16-QAM	12 x 16	3/4	18	22
6	64-QAM	18 x 16	2/3	24	26
7	64-QAM	18 x 16	3/4	27	28

The method of making adaptive modulation in this model is according to the estimated SNR, a bit rate will be specified and then data source generates binary data according to the specified data rate in adaptive modulation control. The transmission modes supported in the IEEE 802.16 standard are described in the Table 1.

1.4 OFDM Transmitter

OFDM converts serial data stream into parallel blocks of size N and modulates these blocks using inverse fast Fourier transform (IFFT). Time domain samples of an OFDM symbol can be obtained from frequency domain data symbols [15] as

$$x(i,n) = IFFT_N[X(i,k)] = \frac{1}{N}\sum_{k=0}^{N-1} X(i,k)\exp\{\frac{j2\pi nk}{N}\} \quad (2)$$

where X(i,k) is the transmitted data symbol at the k^{th} subcarrier of the i^{th} OFDM symbol, N is the fast Fourier transform (FFT) size. After the addition of cyclic prefix (CP) and D/A conversion, the signal is passed through the radio channel. The channel is assumed to be constant over an OFDM symbol, but time-varying across OFDM symbols.

At the receiver, the signal is received along with noise. After synchronization, down-sampling, and removal of the CP, the simplified baseband model of the received samples can be formulated[15] as

$$y(n) = \sum_{l=0}^{L-1} x(n-l)h(l) + w(n), \quad (3)$$

where L is the number of sample-spaced channel taps, w (n) is the additive white Gaussian noise (AWGN) sample with zero mean and variance of σ^2, and the time domain channel impulse response (CIR) for the current OFDM symbol, h (l), is given as a time-invariant linear filter. Note that perfect time and frequency synchronization is assumed. In this case, after taking FFT of the received signal y(n), the samples in frequency domain can be written as

$$Y(i,k) = X(i,k)H(i,k) + W(i,k) \quad (4)$$

where H and W are FFTs of h and w respectively.

1.5 Pilot Insertion

An OFDM system is equivalent to a transmission of data over a set of parallel channels. At the receiver, there are equalizers for compensation of channel characteristics which is achieved by sending pilot symbols from transmitter. Two types of methods can be used for this as shown in figure 3.The fading channel of the OFDM system can be viewed as a 2D lattice in a time-frequency plane, which is sampled at pilot positions and the channel characteristics between pilots are estimated by using pilot information by the channel estimator algorithm. The art in designing channel estimators is to solve this problem with a good trade-off between complexity and performance. The first one, block-type pilot channel estimation, is developed under the assumption of slow fading channel, and it is performed by inserting pilot tones into all subcarriers of OFDM symbols within a specific period. The second one, comb-type pilot channel estimation, is introduced to satisfy the need for equalizing when the channel changes even from one OFDM block to the subsequent one[11].

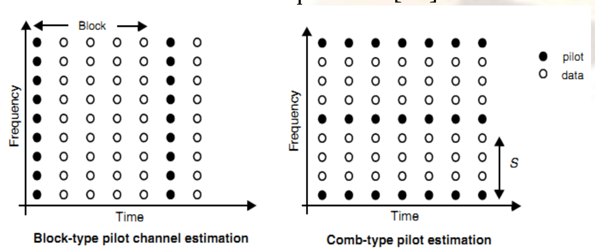

Fig. 3 Two Basic Types of Pilot Arrangement for OFDM Channel Estimations

1.6 Training Sequence Preamble

The first symbol of each downlink subframe is dedicated as a preamble in OFDMA mode of 802.16 standard. The preamble is generated by modulating each third subcarrier using boosted binary phase shift keying (BPSK) with a specific pseudo noise (PN) sequence. Hence, the time domain preamble consists of three repeating parts. This preamble is used for initial estimation of time-varying channel[15].

1.7 Equalization

A symbol-spaced linear equalizer consists of a tapped delay line that stores samples from the input signal. Once per symbol period, the equalizer outputs a weighted sum of the values in the delay line and updates the weights to prepare for the next symbol period. This class of equalizer is called *symbol-spaced* because the sample rates of the input and output are equal. Below is a schematic of a symbol-spaced linear equalizer with N weights, where the symbol period is T.

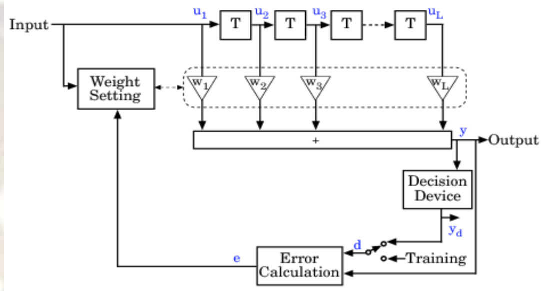

Fig. 4 A linear equalizer with tapped delay line and N weights.

1.8 Radio Channel

There are three basic types of channels considered for this work. Performance of three channels, viz., AWGN, Rayleigh Fading Channel, Rician Fading Channel in communication environment is evaluated through simulation. Multipath fading is a significant problem in communications. In a fading channel, signals experience fades (i.e., they fluctuate in their strength). When the signal power drops significantly, the channel is said to be in a fade. This gives rise to high bit error rates (BER)[7].

1.9 Channel Estimation

Time-dispersive channels can cause inter-symbol interference (ISI), a form of distortion that causes symbols to overlap and become indistinguishable by the receiver. For example, in a multipath scattering environment, the receiver sees delayed versions of a symbol transmission, which can interfere with other symbol transmissions. An equalizer based on LMS channel estimation algorithm attempts to mitigate ISI and improve receiver performance.

III. SIMULATIONS

In the realistic scenario where the channel state information is not known at the receiver, this has to be extracted from the received signal. We assume that the channel estimator performs this using orthogonal pilot signals that are prepended to every packet. It is assumed that the channel remains unchanged for the length of the packet (i.e., it undergoes slow fading).

Scatter plots of the received signal before and after equalization are shown in each simulation. From the plot of the equalized signal, modulation type the system can be recognized, because the plot resembles a signal constellation of 2, 4, 16, or 64 points. The power spectrum of the received signal before and after equalization, in dB. The dynamics of the signal's spectrum before equalization depend on the fading mode parameter in the Multipath Channel block. The estimate of the SNR based on the error vector magnitude. The bit rate of the transmission is shown. Table III shows parameter values for different simulations.

TABLE II. SIMULATION PARAMETERS IN ACCORDANCE WITH WIMAX 802.16 STANDARD

Carrier Frequency	2.3, 2.5, 3.5 GHz
Channel Model	LOS/Non-LOS
Raw Bit Rate	1.0-75.0 Mbps
Modulation	BPSK, QPSK, 16QAM, 64QAM
OFDM subcarriers	256
Channel Bandwidth	1.75, 3.5, 3, 7, 5.5, 10, 20 MHz
Frame Duration	5 ms
Number of Frames (per second)	200
IFFT/FFT	256 point
Data Tone	192
Pilot tone	8
Cyclic Prefix	64
Guard Interval/Symbol Interval	1/4, 1/8, 1/16, 1/32 (or 64, 32, 16,8 samples)
Decoder	Viterbi
Fading Channel	No fading; flat-fading; and Rayleigh or Rician Multipath fading
Noise	AWGN

Fig. 5 shows the signal constellation and power spectrum of received and equalized signal when there is no fading due to channel. The SNR chosen is 20 dB and resultant bit rate is 18 Mbps but there are no packet errors. The received signal is having distortion in amplitude only.

Fig. 6 shows the effect of flat fading channel with maximum Doppler shift of 50 Hz. The signal is distorted in amplitude as well as in phase but after equalization we can recognize the signal. The power spectrum of received signal is flat. As we vary the SNR values, bit rate is changed and error is generated at 0 dB. Fig. 7 also shows the effect of flat fading channel but maximum Doppler shift (200 Hz) is more than fig. 6 which increases the packet error (18%), so BER performance is relatively poor.

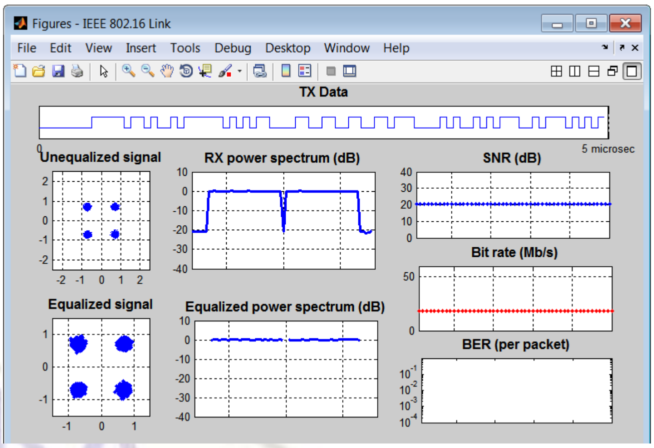

Fig: 5 Effect on received and equalized signal when there is no fading.

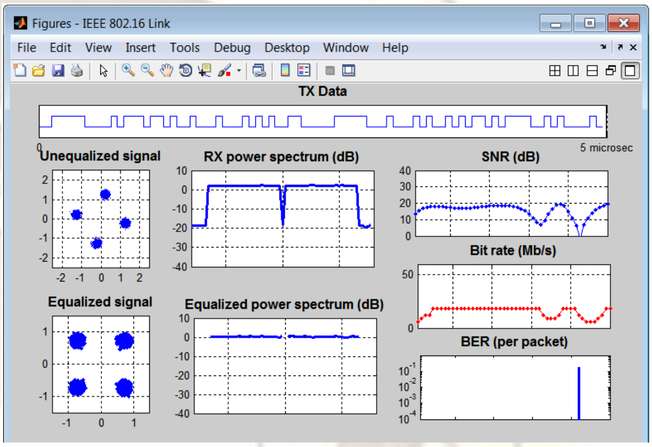

Fig: 6 Effect of flat-fading channel on received and equalized signal

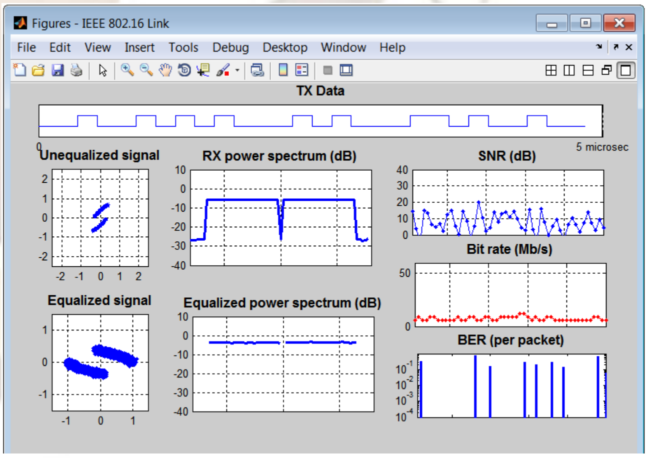

Fig: 7Effect on received and equalized signal when there is flat- fading channel with more Doppler shift.

TABLE III. SIMULATION RESULTS FOR VARYING PARAMETERS IN WIMAX SYSTEM MODEL

Case No.	OFDM symbols/block	OFDM symbols/training sequence	SNR (dB)	Doppler Shift(Hz)	Fading Mode	Bit Rate(Mb/s)	PER(%)
1.	200	8	20.44	200	No Fading	18	0
2.	200	8	19.67	50	Flat Fading	18	4
3.	200	8	4.119	200	Flat Fading	6	18
4.	50	8	13.89	200	Dispersive	12	14
5.	200	20	17.73	50	Dispersive	18	20
6.	500	8	15.89	50	Dispersive	12	34
7.	200	8	4.334	200	Dispersive	6	74

Fig. 8 through fig. 11 are having dispersive channel, which means there is a multipath fading effect introduced in these simulations. In fig. 8, the number of OFDM symbols/block (50) is less as compared to other cases (200), so the number of bits which are erroneous is least in this case because each burst is having separate fading and noise effects. When the length of block increases, a noise can deteriorate whole burst block, so more number of data bits become erroneous.

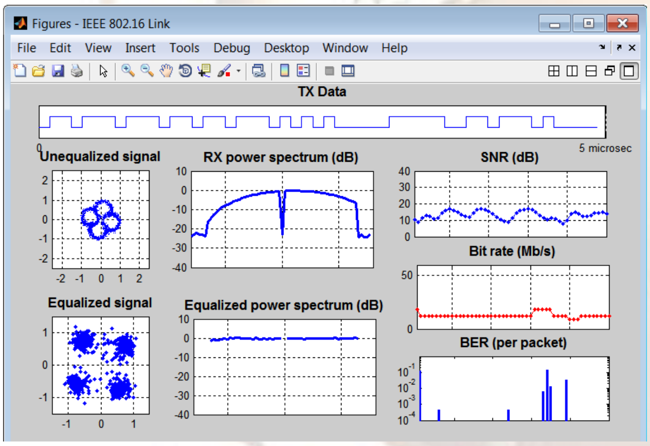

Fig: 8Effect of dispersive channel on received and equalized signal

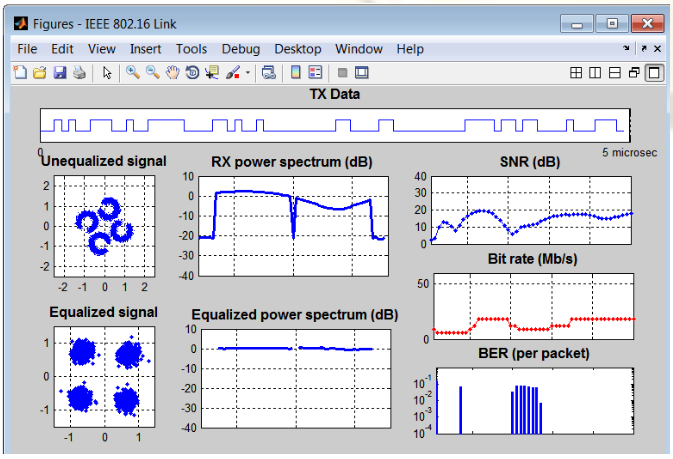

Fig: 9 Effect of multipath-fading channel on received and equalized signal with more training symbols.

In fig. 9 and 10, the received signal is distorted in amplitude and phase due to multipath fading effects and the received power spectrum is also varying. But after equalization, signal constellation can be recognized and power spectrum is balanced. Due to dispersive channel, SNR values continuously changes, so bit rate is also changed. Number of training symbols are more in case 5 (20) as compared to case 6 (8),so equalization is better in case 5 and resultant erroneous bits are less here.

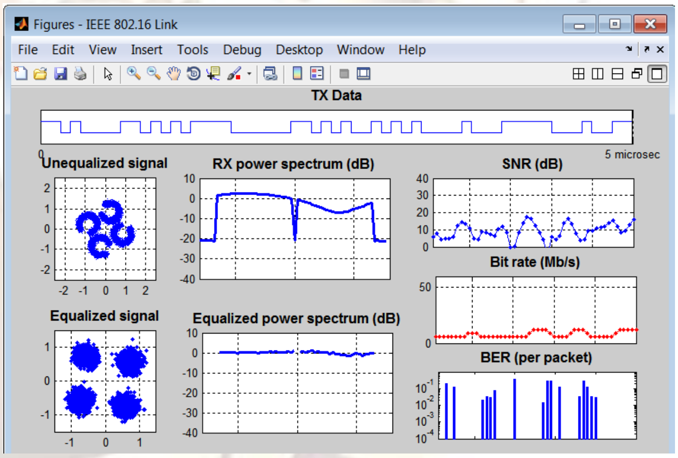

Fig: 10Effect of multipath-fading channel on received and equalized signal with more OFDM symbols per block.

Fig. 11 shows the effect of signal to noise ratio upon bit rate and packet error. When the SNR value of the signal goes below a threshold level, the modulation scheme is changed for a transmission with relatively lower bit rate. If the SNR value of the received signal is too low, even after equalization the packet error rate is very high as shown in the figure. The SNR value (4.334 dB) is minimum in case 7, so packet error (74%) is maximum here.

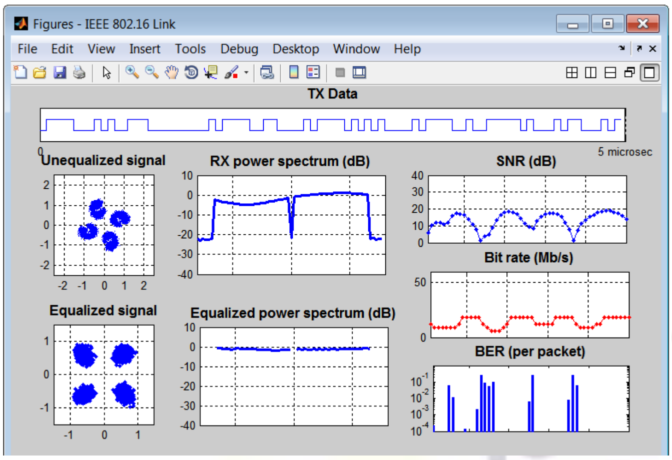

Fig: 11 Effect of signal to noise ratio upon bit rate and packet error.

IV. CONCLUSIONS

A MATLAB simulation is carried out in order to analyze baseband processing of the transceiver. For this, IEEE 802.16 standard is applied which allows transmission data rates from 3 up to 27 Mbps. Channel behavior is analyzed in terms of BER performance for radio environments having flat and multipath fading effects. Simulation results are compared for analyzing system behavior in terms of error probability. For channel estimation, the received signal is splitted up into the training symbols and data symbols.The following blocks display numerical results:

1. The PER block shows the packet error rate as a percentage.
2. The SNR block at the top level of the model shows an estimate of the SNR based on the error vector magnitude. The SNR block in the Multipath Channel subsystem shows the SNR based on the received signal power.
3. The Bit Rate block shows which of the bit rates specified in the standard is currently in use.

As a conclusion, it can be said that channel estimation and equalization are necessary for multipath fading channels so that the packet error rate can be reduced. Using more training symbols give better results in terms of accuracy but also increases the overhead.

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