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Reduction of noise effect in AWGN channel

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ABSTRACT

In this paper, a new algorithm for noise suppression schemes is applied to specific practical problem mainly the suppression of Binary phase-shift keying (BPSK) level of the modulated signal. This is a practical situation of modulation which is used in different transceivers schemes such as in Digital Video Broadcasting (DVB) or other types of modulation techniques with standard microwave systems. The algorithm is well suited for a modular software radio concept, which we believe, will be more accepted in the future of wireless communication. Further modifications of the scheme necessary for these applications are described, and the results are presented to illustrate performance improvements. A general interpretation of this technique based on multiplying the transmitted data vectors by a generated amplitude matrix (AM) in the transmitter side, and the inverse of amplitude matrix (IAM) in the receiver side algorithm are introduced and discussed. The suggested scheme results are compared with the conventional modem through using BPSK type of modulation. Different gains are obtained from the proposed model relevant with the conventional system in AWGN and flat fading channel, according to the variation of the adjusted parameters.

1. Introduction

Code Division Multiple-Access (CDMA) networks are considered for increasing capacity in mobile communication systems by using frequency bands already allocated in "traditional" narrow-band users [1]–[6]. The performance of such system depends on the modulation type in addition to the other affected parameters. Experimental systems have been built up and tested in a real environment [1]. The results obtained confirm that such a concept is feasible. Interference suppression schemes are used to additionally suppress traditional users in CDMA networks so that after signal despreading, the resulting signal-to-noise ratio (SNR) provides the required system performance even when, at the received front end, the narrow-band signal levels are much higher.

The interference suppression techniques used so far require that narrow-band interference occupy only up to 10%–20% of the CDMA signal bandwidth. Besides the inefficiency, this technique also generates additional administrative problems because in multioperator networks this might require multiple negotiations in order to get permission to implement such a system. In [7], schemes that provide successful suppression of the interferers can be presented by using standard modulation formats in the same frequency band as CDMA signals. These schemes provide not only five to ten times larger capacity, but also reduce the negotiation efforts by making the CDMA overlay concept far more feasible.

The scheme in [8] demonstrates an intuitive approach to feed forward carrier estimation for optical QPSK using DSP. Because of its simple implementation, this scheme can potentially be employed in the coming future. Junghyun Kim, et.al. [9] presented a new analytical approach and experimental verification for the improvement of noise performance in phased-array receivers. For analysis purposes, a multi-channel array system is converted into an equivalent single-channel system, such that the two presents the identical signal and noise powers at the output, respectively. They defined an effective gain, noise figure, and signal-to-noise ratio in the equivalent system. In this paper, we set to find an estimate of the BER for a BPSK modem without using the pilot carriers for channel estimation and compensation.

2. System Model

In this section, we will derive the theoretical equation for bit error rate (BER) with Binary Phase Shift Keying (BPSK) or PSK (2-points) modulation scheme in Additive White Gaussian Noise (AWGN) channel.

With Binary Phase Shift Keying (BPSK), the binary digits 1 and 0 may be represented by the analog levels $+A$ and $-A$ respectively. The system model is as shown in the Fig. (1).

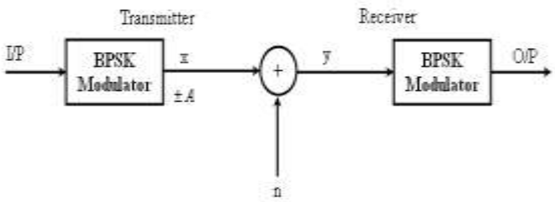

Fig. (1): Simplified block diagram with BPSK transmitter-receiver

2.1 Channel Model

The transmitted waveform gets corrupted by noise n, typically referred to as Additive White Gaussian Noise (AWGN).

Additive**:** As the noise gets 'added' (and not multiplied) to the received signal

White**:** The spectrum of noise if flat for all frequencies.

Gaussian: The values of noise n follow the Gaussian probability distribution function,

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Assuming that $P(r = x_0)$ and $P(r = x_1)$ are equally probablei.e., the threshold 0 forms the optimal boundary decision.

$$P(x) = \frac{1}{\sqrt{2\pi\sigma^2}} e^{\frac{-(x-\mu)^2}{2\sigma^2}} \quad with\ \mu = 0\ and\ \sigma^2 = \frac{N_o}{2} \qquad (1)$$

2.2 Computing the Probability of Error

Using the derivation provided in [10], [11] as references, the received signal,

$y = x_1 + n$ when bit 1 is transmitted and

$y = x_0 + n$ when bit 0 is transmitted, x is the transmitted signal

The conditional probability distribution function (PDF) of y for the two cases are:

$$p(y=0) = \frac{1}{\sqrt{\pi N_0}} e^{\frac{-(y+A)^2}{N_0}}$$

$$p(y=1) = \frac{1}{\sqrt{\pi N_0}} e^{\frac{-(y-A)^2}{N_0}} \qquad (2)$$

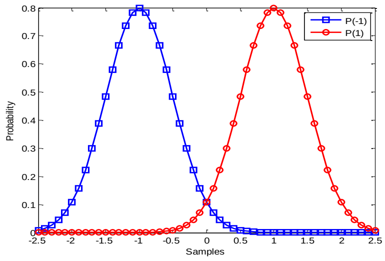

Fig. (2): Conditional probability density function with BPSK modulation

- If the received signal $P(r)$ is greater than 0, then the receiver assumed $P(r = x_1)$ was transmitted.
- If the received signal $P(r)$ is less than or equal to 0, then the receiver assumes $P(r = x_0)$ was transmitted

i.e. $y > 0 \Rightarrow x_1 \quad and\ if\ \ y \le 0 \Rightarrow x_0$

2.3 Simulation of Standard Model

The source code in Matlab is used for computing the bit error rate with BPSK modulation from theory and simulation for fig. (1). The code performs the following:

(a) Generation of random BPSK modulated symbols +1's and -1's of length 32 bit for each vector

(b) Passing them through Additive White Gaussian Noise channel (you can use the Matlab function)

(c) Demodulation of the received symbol based on the location in the constellation mapping of BPSK used in the transmitter side

(d) Counting the number of errors in reference to the transmitted symbols

(e) Repeating the same previous steps for multiple Eb/No value and plot the results by Matlab tools

3. Proposed Model

The conventional model in fig. (1) can be modified by enhancing its performance in AWGN channel as shown in fig. (3). This figure shows that the output data from the BPSK modulator are multiplied by a generated Amplitude Matrix (AM), and it is a sequare matrix of size N*N, where N is the block size of the transmitted data (N=32 in our simulted algorithm). The encoded data is divided into sub-blocks of length 32-bit (N=32), and are direcly multiplied by a generated amplitude matrix (bit by bit) given by [12]:

$$Amplitude\ Matrix = \begin{bmatrix} \mathrm{Real}\left(e^{-j(\frac{2\pi n}{N})k}\right) - A1 & if\ imag\left(e^{-j(\frac{2\pi n}{N})k}\right) < 0 \\ \mathrm{Real}\left(e^{-j(\frac{2\pi n}{N})k}\right) + A2 & if\ imag\left(e^{-j(\frac{2\pi n}{N})k}\right) \ge 0 \end{bmatrix} \qquad (3)$$

Where N=32, n= variable that varies between 1to N, k= between 0 to N-1, the other parameters A1, A2 are:

$$0 < A1 \le 1, \qquad A1 \ne 0\ if\ \ A2 = 0$$

$$0 < A2 \le 1 \qquad A2 \ne 0\ if\ \ A1 = 0$$

Note that the values of A1 and A2 should not be equal to zero, else the inverse of matrix is undetermined. After each multiplication of data bits by AM (32bit data*32 bit AM) the multiplicity values in each 32 bit are added together so that after multiplying each vector of data by 32 column of matrix we have single vector of length 32 bit. Since the amplitude of the signal will increase due to multiplying the transmitted data by AM values, and this will cause a peak to average power ratio (PAPR) problem (especially due to high values in the first column of matrix), so we convert the sign of the odd values in the first column of matrix from positive to negative value (i.e. odd values=negative, even values=positive). This conversion will reduce the PAPR of the multiplied encoded data by amplitude matrix in the first value of each transmitted vector. The output data from AM will be sent to the receiver through a channel which may be AWGN or fading channel.

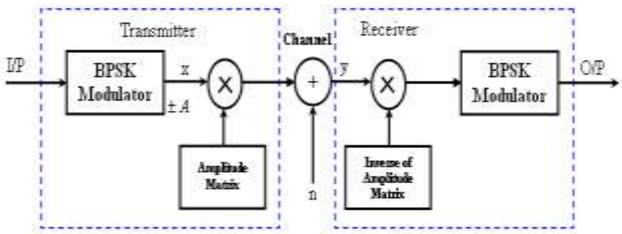

Fig. (3): Block Diagram of the Proposed Model

4. Simulation Results

In this section, the channel is modeled as an Additive White Gaussian Noise for wide range of SNR from 0 dB to 40 dB. The performance of both systems is shown in the same figures in order to simplify the performance comparison between the proposed and the traditional systems. At first time the variance A1 adjusted at 0 and the other variance A2 vary from 0.1 to 1 according to the steps shown in fig. (4). From this figure, it is found that the proposed model based on AM algorithm is better and more significant than the other system for the same type of modulation, where at BER=10^{-4} the SNR is about 8.5 dB for conventional model 15.5 dB at A2=0.1, 10 dB at A2=0.2. As the variance A2 increases the BER decreases in the proposed model. The SNR approaches to 2.5 dB as the variance A2=1, the proposed model outperforms the other one as A2 vary from 0.3 to 1.

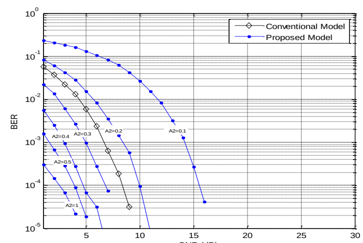

Fig.(4): Performance of the conventional and proposed models at the AWGN channel (A1=0, A2=0.1 to 1)

When A2 is settled at 0 and changing A1 from 0.1 to 1 as shown in fig. (5), the conventional model has better performance than the proposed model as the SNR more than 7 dB. The BER decreases as A1 increases for all range of SNR. The transmitted signal shape in the proposed model is more fluctuated by changing A1 than changing A2, this random variation will increase the BER as in fig. (5). While changing A2 and setting A1 to zero will arrange the shape of the transmitted signal to be similar to the output of the orthogonal vectors from the FFT of IFFT in the OFDM system. The orthogonality property can be exploited for reducing the BER in the conventional model by using the proposed one. The reader can easily remember the matrix of the FFT which is approximately the same as in (3) if we take the exponential part alone. So the generated amplitude matrix has the same values of the FFT matrix but without the imaginary parts if and only if we set the variances A1 and A2 to zero. But in the same time it cannot adjust those to zero as we mentioned previously, the inverse of amplitude matrix is undetermined. So, the proposed model can be considered as an alternative technique to the conventional system based on BPSK type of modulation. The results are extracted without channel equalization techniques, even the performance of both models will be improved after the equalization [13].

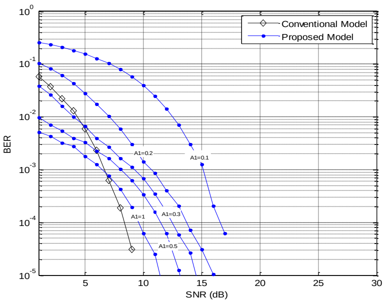

Fig.(5): Performance of the conventional and proposed models at the AWGN channel (A2=0, A1=0.1 to 1)

Fig.(6) shows the performance of both models in flat fading channel at Doppler frequency=5Hz, A1=0 and A2=0.5. This figure confirm the effectiveness of the proposed model in fading channel, there is a wide span gain obtained from the proposed model relative to the conventional system in all range of SNR values. Different gains can be obtained by varying the variances A1 and A2. Finally, the inverse of amplitude matrix can be used at the transmitter side and the amplitude matrix at the receiver side, at this case the system performance will be small affected by changing the variances A1 and A2 and the output results stay in the acceptable range.

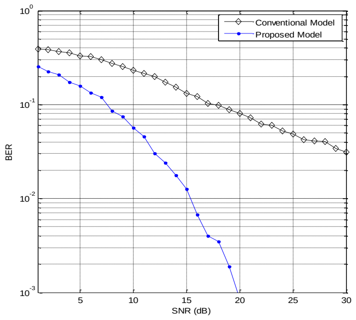

Fig.(6): Performance of the conventional and proposed models flat fading channel (Doppler frequency=5Hz, A1=0, A2=0.5)

5. Conclusion

In this paper, we have proposed an amplitude matrix (square matrix) for enhancing the performance of conventional transceiver in AWGN channel that is based on BPSK type of modulation. We have derived a suitable algorithm composed of real values and two variable parameters (A1, A2). By employing the amplitude matrix algorithm to the proposed architecture, we have significantly reduced the BER of the conventional model by using the new algorithm. Different gains obtained depending on the setting values of A1 and A2 in AWGN and flat fading channel.

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