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Improved Wireless Communication System using Interleaved Encoder with QAM and PSK

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Abstract - Wireless media for sharing information worldwide is getting popular day by day, and there is need to improve the wireless communication mechanism to provide seamless connectivity without losing any information of links. Technology needs improvements everyday and wireless technology needed utmost than past, to deliver quality of service and high data rates with excellent reliability. In the same context this work working towards increasing the reliability of the wireless system to deliver better quality of service with minimal loss of data which is presented in the simulation results using bit error rate performance vs signal to noise ratio. The proposed methodology utilizes the randomizer with convolutional encoder with differnt FFT sizes to mitigate the information loses and achieve better bit error rate. The proposed design also reduces the requirement of signal power than the existing system.

Keywords - Wireless communication, encoder, signal to noise ratio, BER, QAM and PSK.

I. INTRODUCTION

Wireless communication has become increasingly important not only for professional applications but also for many activities in our daily routine and in the consumer electronics industry. In recent years, wireless communication technology has been delivering high data speeds, and it provides a relatively low cost, high-quality services. With the development of advanced wireless communication technology, new services have been developing.

Various modern communication systems employ time, frequency and spatial multiplexing techniques to achieve the various goals of efficient communication. A new platform shall be developed for the interoperability of all current and emerging standards. One of such platform is that of reconfigurable, or agile radios where signals can be transmitted using different modulation schemes on different carrier frequencies and then be identified and demodulated using advanced Automatic Modulation Recognition(AMR) schemes at the receiver. Recently, automatic modulation recognition schemes have been developed using Wavelet transforms. However, two vital elements for the functioning of an agile transceiver system are Channel Estimation and Channel Equalization. In an ideal communication channel, the received information is identical to that transmitted. However, this is not the case for real communication channels, where signal distortions take place. A channel can interfere with the transmitted data through three types of distorting effects: power degradation and fades, multi-path time dispersions and background thermal noise [2]. Equalisation is the process of recovering the data sequence from the corrupted channel samples. A typical baseband transmission system is depicted in Figure 1.1, where an equalizer is incorporated within the receiver.

Within telecommunication channels multiple paths of propagation commonly occur. In practical terms this is equivalent to transmitting the same signal through a number of separate channels, each having a different attenuation and delay. Consider an open-air radio transmission channel that has three propagation paths, as illustrated in Fig 1.2. These could be direct, earth bound and sky bound.

Figure 1.2b describes how a receiver picks up the transmitted data. The direct signal is received first whilst the earth and sky bound are delayed. All three of the signals are attenuated with the sky path suffering the most.

Multipath interference between consecutively transmitted signals will take place if one signal is received whilst the previous signal is still being detected [2]. In Figure 1.2 this would occur if the symbol transmission rate is greater than 1/t where, t represents transmission delay. Because bandwidth efficiency leads to high data rates, multi-path interference commonly occurs.

Since equalisers are designed to invert the channel distortion process they will in effect model the channel inverse. The minimum phase channel has a linear inverse model therefore a linear equalisation solution exists. However, limiting the inverse model to m-dimensions will approximate the solution and it has been shown that non-linear solutions can provide a superior inverse model in the same dimension. A linear inverse of a non-minimum phase channel does not exist without incorporating time delays.

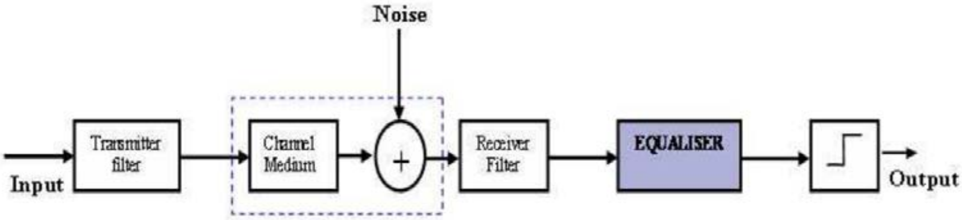

Figure 1.1 a baseband Communication System

A time delay creates a convergent series for a non-minimum phase model, where longer delays are necessary to provide a reasonable equaliser.

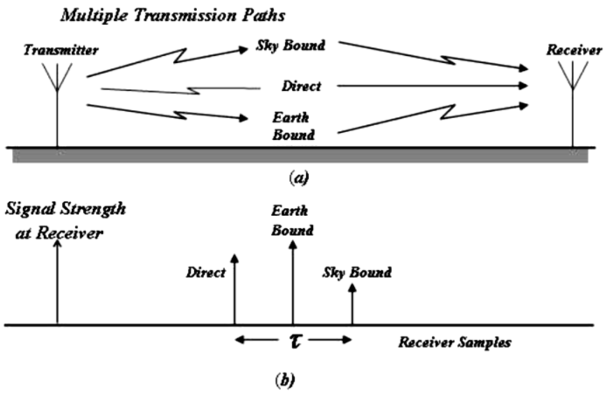

Figure 1.2 Impulse Response of a transmitted signal in a channel which has 3 modes of propagation, (a) The signal transmitted paths, (b) The received samples

II. SYSTEM MODEL

In a conventional serial data system, the symbols are transmitted sequentially, one by one, with the frequency spectrum of each data symbol allowed to occupy the entire available bandwidth. A high rate data transmission supposes a very short symbol duration, conducing at a large spectrum of the modulation symbol. There are good chances that the frequency selective channel response affects in a very distinctive manner the different spectral components of the data symbol, hence introducing the ISI. The same phenomenon, regarded in the time domain consists in smearing and spreading of information symbols such, the energy from one symbol interfering with the energy of the next ones, in such a way that the received signal has a high probability of being incorrectly interpreted. Intuitively, one can assume that the frequency selectivity of the channel can be mitigated if, instead of transmitting a single high rate data stream, transmit the data

Simultaneously, on several narrow-band subchannels (with a different carrier corresponding to each subchannel), on which the frequency response of the channel looks “flat”. Hence, for a given overall data rate, increasing the number of carriers reduces the data rate that each individual carrier must convey, therefore lengthening the symbol duration on each subcarrier. Slow data rate (and long symbol duration) on each subchannel merely means that the effects of ISI are severely reduced. Unlike the classical frequency division multiplexing technique, OFDM will provide much higher bandwidth efficiency. This is due to the fact that in OFDM the spectra of individual subcarriers are allowed to overlap. In fact, the carriers are carefully chosen to be orthogonal one another. As it is well known, the orthogonal signals do not interfere, and they can be separated at the receiver by correlation techniques. The orthogonality of the subcarriers accounts for the first part of the OFDM name figure 2.1 demonstrate the basic block diagram of digital communication system.

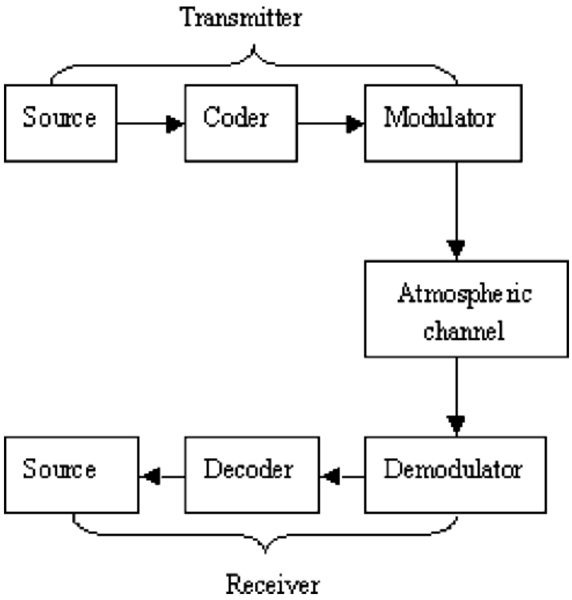

Figure 2.1 Basic wireless systems.

III. PROPOSED METHODOLOGY

The proposed system is based on the QPSK/4- QAM modulator with convolution encoder inter leaver figure 3.1 demonstrate the block diagram of the propos and the flow of the synthesis is illustrated in figure 3.2 proposed system is more efficient as compare to the existing system. Proposed system can be segmented in three sections transmitter, receiver, and channel.

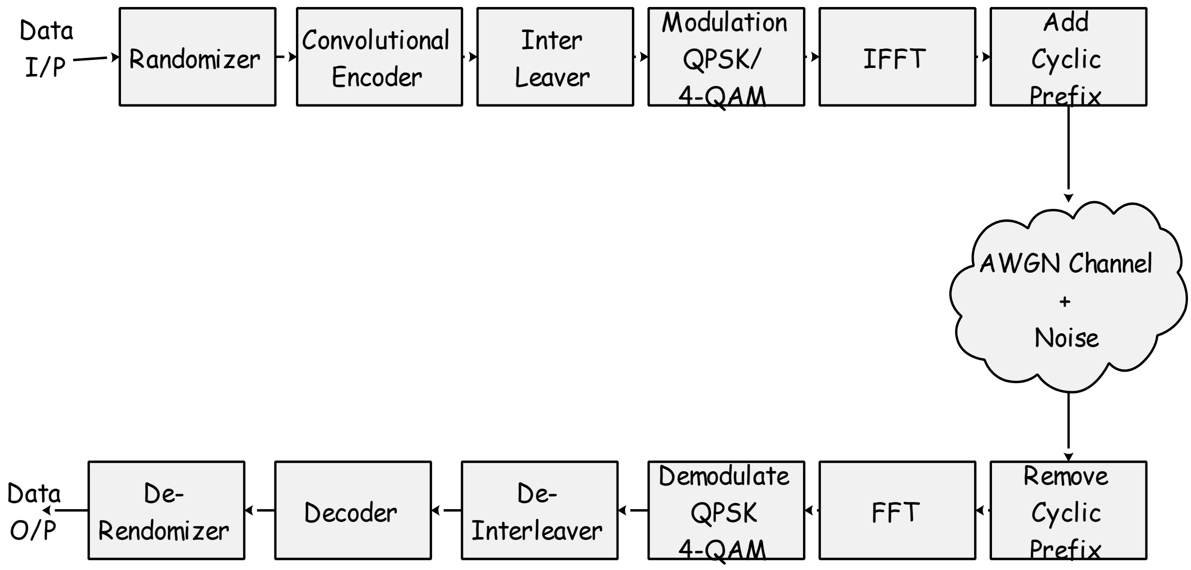

Figure 3.1 Block diagram of the Proposed System.

1. Transmitter

Input data sequence is baseband modulated using digital modulation scheme as demonstrated in figure. Various modulation schemes could be employed such as BPSK, QPSK (also with their differential form) and QAM with several different signal constellations. An randomizer transform the bit sequence in to logic sequence then convolutional encoder encode the randomized sequence and The size of the interleaver determines the length of the codeword. And 4QPsk modulation applied on the modulating signal. IFFT transform and add cyclic prefix. Leave the signal through channel for transmission.

2. Channel

An AWGN channel model is used for the transmission of the test signal where adaptive white or wide band Gaussian noise is added to signal. The signal is received by the receiver end.

3. Receiver

At the receiver end noisy signal is received by receiving antennas just inverse process of transmitter has done here. First of all cyclic prefix is removed from the received noisy signal. Fast Fourier transform applied to reverse IFFT. Demodulate with QPSK QAM. De-Interlever and decode de-randomize and produce output noise free test signal. The simulation and result analysis of the proposed system has give in figure 4.1 to figure 4.3.

4. Flow of proposed system.

Create Simulation Environment using Simulink MatLab Simulator. Generate test signal to transmit.

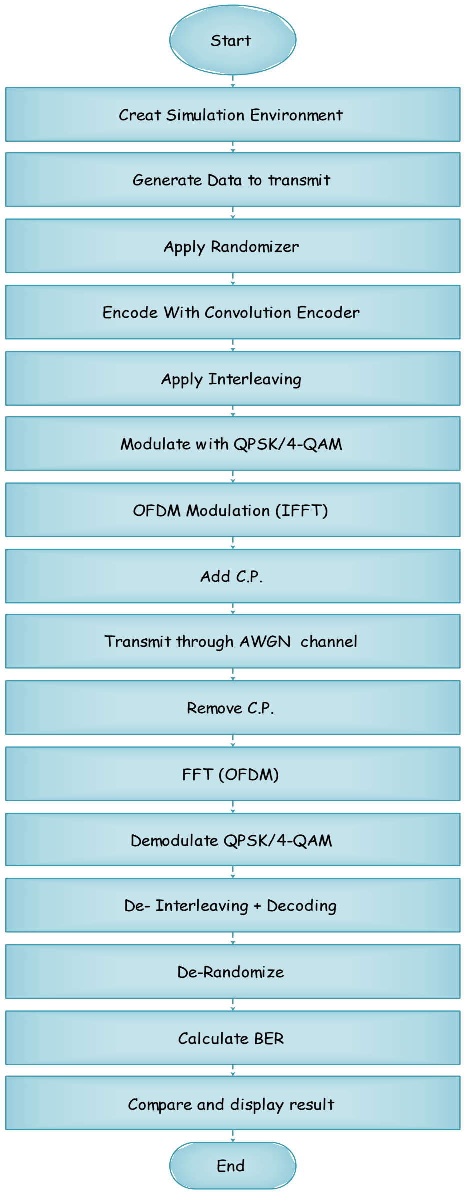

Figure 3.2 Flow of proposed work.

Apply randomizer on test signal which is to be transmitting. Apply convolution encoding Interleaving on randomized test transmitting signal. Modulate with QPSK 4-QAM OFDM modulation with inverse fast Fourier transform Add cyclic prefix on ready to transmit signal. Transmit prepared signal with AWGN channel model where certain random noise is added in it.

Remove cyclic prefix from received signal at receiver end. Apply Fast Fourier transformer. Demodulate QPSK /4QAM. De-interleve and decode demodulated signal. De-randomize and calculate bit error rate from received signal. And finally compare and display result.

IV. SIMULATION RESULTS

Proposed system has simulated on Matlab. simulation results of proposed system has given in figure 4.1 to 4.3 on different parameters figure 4.1 illustrate performance of proposed system using 2-FFT with convolution encoder with PSK QAM modulation. similarly with 8 FFT and 16 FTT correspondingly in figure 4.2 and 4.3 .

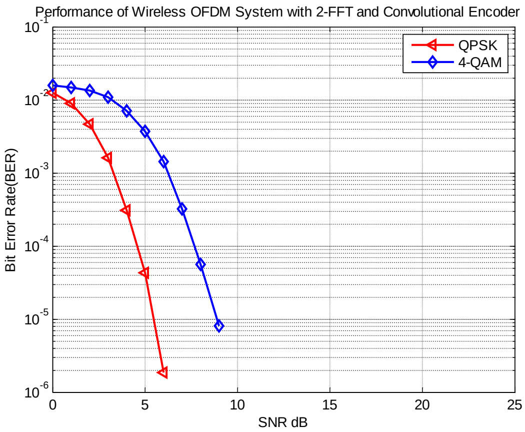

Fig 4.1 Proposed System Performance using 2-FFT using Convolutional Encoder with PSK and QAM Modulation

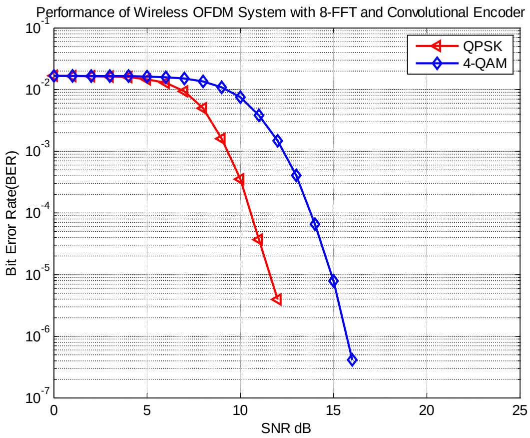

Fig 4.2 . Proposed System Performance using 8-FFT using Convolutional Encoder with PSK and QAM Modulation

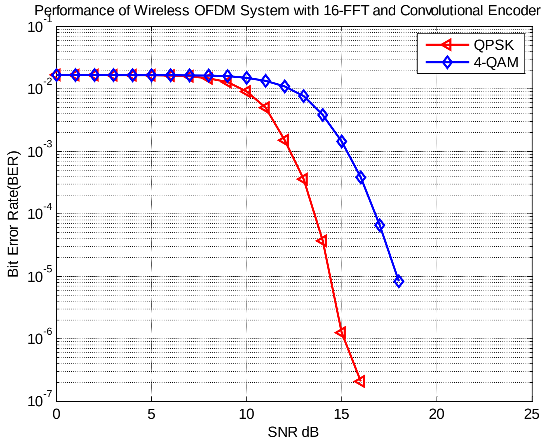

Fig. Proposed System Performance using 16-FFT using Convolutional Encoder with PSK and QAM Modulation

V. CONCLUSION AND FUTURE SCOPE

The proposed system is implemented on simulation tool and the error rate of the system calculated to show the efficiency over previous systems. The BER achieved is 2 x 10^{-7} with QPSK modulation scheme and 16-FFT points with convolutional encoder. The proposed system consumes less power to transmit signals than the previous system and even though getting better results. The system can be equipped with the detection methodologies for future enhancements. The system discussed in this work has less complex modulation techniques which can be replaced with the higher order modulation techniques which will enhances the performance as well as complexity of the system which can be overcome by integration of digital filters.

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