Implementation of a Wireless Multi-Carrier Acoustic Modem using Error Control Coding

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Abstract

In Digital Communication Systems, especially high-speed modems, it is desirable to increase the bit rate while maintaining the same bandwidth without seriously impairing transmission quality. Single-Carrier systems provide a decent bit rate, but Multi-Carrier systems give bit rates that are several orders higher. Asymmetric Digital Subscriber Line (ADSL) services and Orthogonal Frequency Division Multiplexing (OFDM) are multi-carrier methods that are suited for this type of application. ADSL is used in two-wire line telephone services and OFDM is used in Wireless applications. Work done in a modem application that employs OFDM concepts and Reed-Muller Error Correcting codes is presented in this paper. The modem was implemented using the MATLAB numerical computation software package.

1. INTRODUCTION

Digital Communication enters our daily lives in so many ways that it is very easy to overlook the multitude of its facets. It involves the transmission of information in digital form from a source that generates the information to one or more destinations [1]. The telephone at our hands, the radios and televisions in our living rooms, the computer terminals in our offices and homes, and our newspapers are all capable of providing rapid communications from every corner of the globe. Communication provides the senses for ships on the high seas, aircraft in flight, and rockets and satellites in space. Communication through a cordless telephone keeps a car driver in touch with the office or home miles away; it also keeps a weather forecast informed of conditions measured by a multitude of sensors [2]. Nowadays, the applications of communication systems seem to be endless.

Today we are witnessing a significant growth in the introduction and use of communications systems, including voice, data, and video transmission. One of the newest features is wireless transmission of information. The development of wireless communications stems from the works of Oersted, Faraday, Gauss, Maxwell, and Hertz during the nineteenth century. Since these systems are getting so sophisticated, the specifications for their designs are becoming more demanding. In the work done in this paper it was desired to increase the bit rate using the same bandwidth and at the same time maintaining wireless system. The system that was designed was a modem. The word modem comes from MOdulator / DEModulator and it allows one computer to talk to another one through a communication channel, e.g. telephone line, air, etc.

The system can be designed to be single-carrier or multi-carrier. A single-carrier system provides decent bit rates, but multi-carrier systems give bit rates that are several orders of magnitude higher. ADSL and OFDM are methods that are suited for high bit rate applications. For wireless applications OFDM is the method to be used. OFDM is a useful digital modulation method for digital terrestrial broadcasting systems, including TV broadcasting [3]. It can be used for transmission of a continuos stream of data (broadcasting applications) and in bursty data as in a wireless local area networks (WLAN) [4]. OFDM is very robust against multipath reception and is useful for channels presenting strong linear distortions. The OFDM signal is especially sensitive to sine waves spurious and to frequency deviations in the receiver [5].

2. SYSTEM CONCEPTUALIZATION

A multi-carrier system superimposes several carrier modulated waveforms to represent an input bit stream. The spectrum of this signal is shown in figure 1.

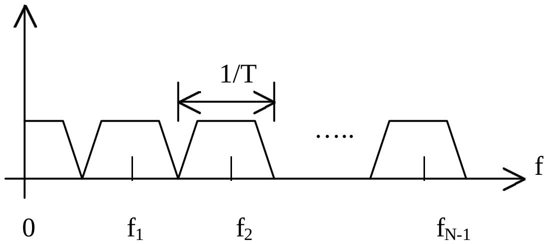


Figure 1: Spectrum of a multi-carrier signal.

The multi-carrier transmit signal is the sum of N independent sub-signals each of equal bandwidth, 1/T, and with center frequency f_i , i = 0,1,…,N-1. The number of sub-carriers is chosen from a set of numbers that are always a power of two, $N\in\{128,256,512,1024,\ldots\}$. Each of these sub-signals can be considered to be a Quadrature Amplitude Modulated (QAM) signal [6]. If we try to imagine the first part of a one-shot (transmission of only one symbol per subchannel) multi-carrier system using a single-carrier system, we would have a system like that shown in figure 2,

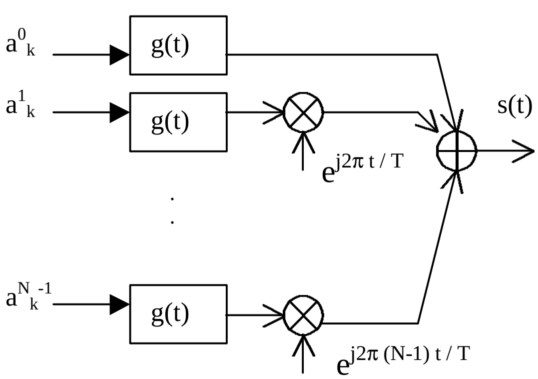


Figure 2: Multi-carrier system.

where T is the symbol period, a^n_K is the k^{th} symbol transmitted through the n^{th} subchannel, and g(t) is a rectangular pulse of amplitude 1. In equation form s(t) would be:

$$(t) = (a^{0}_{k} + a^{1}_{k}\, e^{j2\pi t/T} + .. + a^{N-1}_{k}\, e^{j2\pi(N-1)t/T})\, g(t)$$

$$= \sum_{i=0}^{N-1} a_k^i e^{\frac{j2pit}{T}} g(t)\,.$$

If we sample the continuous-time signal, s(t), we obtain a discrete-time signal s_k. From the sampling theorem [2] we know that we must sample s(t) at a rate that satisfies $fs \geq 2f_m$, where f_m is the maximum frequency content in the signal s(t) and fs is the sampling rate. In our system fm = (N - -0.5) / T, so fs ≥ 2(N – 0.5) / T. Let fs = 2N/T = q / T and Ts = T / q , where q = 2N. So, to obtain $s_{K,}$ we sample s(t) at time t = kTs. Doing this we obtain the following equation:

$$s_k = s(t)|_{t = kTs} = \sum_{i=0}^{N-1} a_k^i e^{\frac{j2p\cdot i\cdot(kTs)}{T}} g(kTs)$$

$$= \sum_{i=0}^{N-1} a_k^i e^{\frac{j2p\cdot i\cdot k(T/q)}{T}} = \sum_{i=0}^{N-1} a_k^i e^{\frac{j2p\cdot i\cdot k}{q}}$$

In the last part of this equation the term g(kTs) is omitted because g(t) is a rectangular pulse of value 1. Recalling the definition of an Inverse Discrete Fourier Transform (IDFT) of a discrete signal [7], we notice that the equation for s_K looks like the IDFT of the sequence of symbols a_K, but of length q = 2N. The sequence a_K is of length N, to extend the sequence to a length of q = 2N we use a cyclic prefix. The cyclic prefix of a sequence X_i is defined as:

$$Y_i = \begin{cases} X_i, \rightarrow i = 1,\ldots,N-1 \\ real(X_N) \rightarrow i = 0 \\ imag(X_N) \rightarrow i = N \\ X^*_{q-1} \rightarrow i = N+1,\ldots,q-1 \end{cases}$$

The IDFT of the expanded sequence results in a sequence that contains only real values, the imaginary part of this sequence is equal to zero. The IDFT is used in a modulation technique called Discrete Multi-Tone Modulation (DMT). DMT is a common form of multi-carrier modulation and it's the modulation method used in OFDM and ADSL. Peled and Ruiz introduced it in 1980 to take advantage of digital signal processors and the FFT [6]. The basic transmitter part of a multi-carrier system is shown in figure 3.

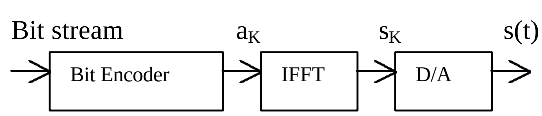


Figure 3: Basic transmitter part of a multi-carrier system.

In figure 3, s(t) represents the continuous-time signal that will be transmitted. An input bit stream of data rate R_b bits/second is buffered into blocks of $b = R_b * T$ bits. Of these b bits, b_i are intended for use in the i^{th} subchannel and

$b = \sum_{i=1}^{N} b_i$. In DMT systems, the number of bits transmitted through each of the subchannels, b_i , does not necessarily have to be the same, but in the work presented in this paper b_i was the same for each subchannel. Since each signal in the subchannels can be visualized as a QAM signal, the bit rate in each subchannel, R_{bi} , can be defined as $R_{bi} = b_i / T$ and the total bit rate would be $R_b = \sum_{i=1}^{N} R_{bi} = N(b / T)$. In the DMT encoder the b_i bits for each of the subchannels are translated into a complex symbol to form the sequence a_K . Then the IDFT of the resulting sequence is taken. After the IDFT, the resulting real discrete-time sequence is passed through a D/A to generate the signal s(t) that will be transmitted through the channel.

A basic multi-carrier receiver is shown in figure 4.

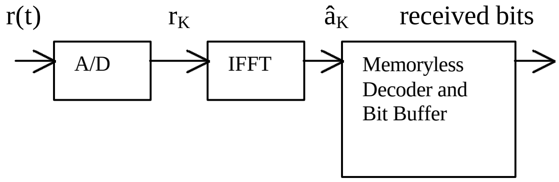
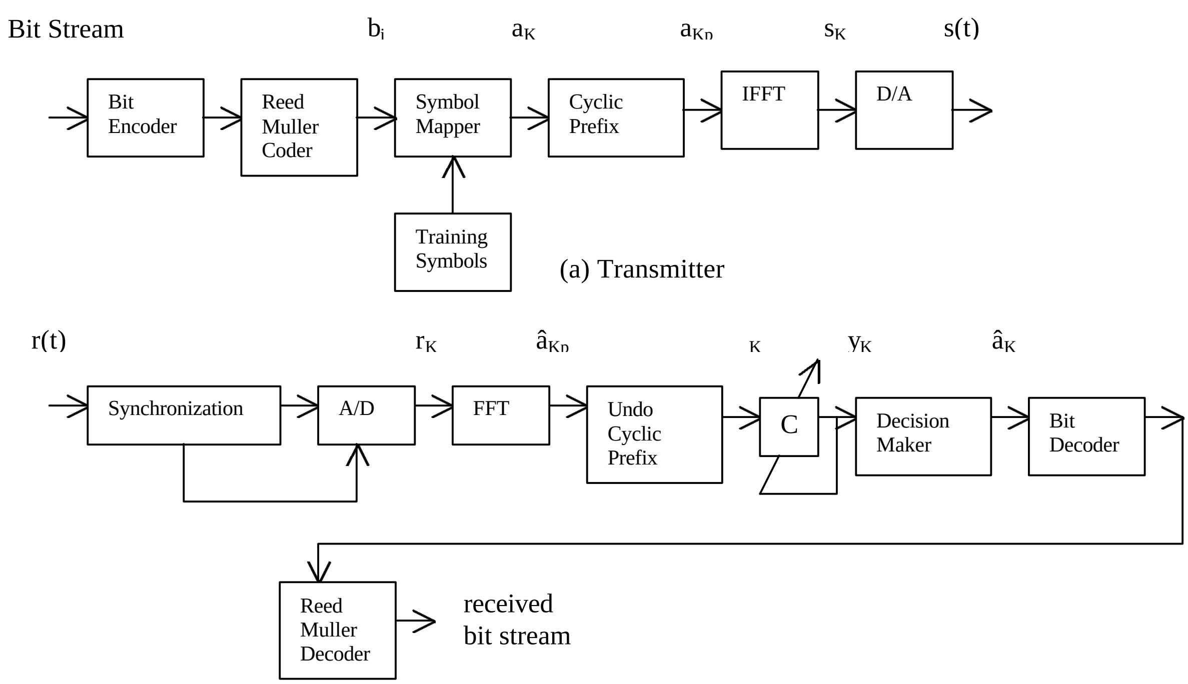


Figure 4: Basic receiver for a multi-carrier system.

The received signal, r(t), is applied to a A/D to produce the discrete-time sequence r_K . Then the Discrete Fourier Transform of the sequence is taken to yield the sequence $\hat{a}_K$, which would be composed of the transmitted symbols. Then $\hat{a}_K$ is passed through a memoryless decoder and buffered back into bits to yield the transmitted bit stream.

3. SYSTEM IMPLEMENTATION

The system was implemented in MATLAB using the transmitter and receiver presented in figure 5. The information that was being transmitted was an image. The image was first converted to bits, and these bits were used as the input bit stream. The number of subchannels used was N = 128



(b) Receiver

Figure 5: Transmitter and Receiver Part

and the number of symbols of the M-ary QAM system that were used was M = 16. At the transmitter, the bits are first passed through a (8,4) Reed-Muller code. Since M = 16, this means that 4 bits are transmitted through each subchannel; the code inserts 4 parity bits to these bits that will be used for correction at the receiver. This new bit stream is then converted to symbols to produce part of the sequence a_K. To complete the other part of the sequence some training symbols are inserted. These training symbols are inserted for synchronization at the receiver. After this the sequence a_K is cyclic prefixed to extend it to q = 2N = 256 symbols to produce the sequence a_{Kp}. Then the IFFT of the sequence is computed and the resulting real sequence passed through a D/A and transmitted by sound through the channel.

At the receiver the first step is to perform synchronization. This is done by creating a matched filter with the known training symbols. The received signal is then filtered with the matched filter. With the signal resulting from the filtering we determine t0, which tells us at what time the information begins in the received signal r(t). Using t0 we sample r(t) to obtain the discrete-time sequence r_K. Now the FFT of r_K is taken to yield the sequence $\hat{a}_{Kp}$. The cyclic prefix is then removed from this signal producing y_K, which has a length of N. The signal $_K$ is composed of the original signal plus noise. In mathematical form it can be written as: $_K = a_K +$ noise. To remove the noise from the signal an adaptive equalizer is used. The equalizer is generated using the least-mean-square (LMS) algorithm. The training sequence is used to generate the coefficients C in figure 5 (b). The LMS algorithm uses the equation:

$$C_{K+1} = C_K - \mu \text{ x } e \text{ x } X^* ,$$

where C_K is the vector of coefficients, e is an error signal and is determined by :

$$e = C_K \text{ x } X - \text{(desired response)},$$

and X^* is the complex conjugate of the signal that is being corrected. The desired response in this case are the values of the training symbols that the receiver knows and X is the training sequence that we receive. After the coefficients C are obtained and multiplied by the sequence $_K$ the sequence y_K is produced. This sequence is passed through a decision-maker that employs maximum likelihood to determine which symbol was transmitted. The sequence $\hat{a}_K$ contains the symbols received. These symbols are then coded back into bits and the passed through an (8,4) Reed-Muller Decoder to produce the bit stream received. Since the information we were transmitting was an image, these bits are converted into pixels and the received image is displayed.

4. RESULTS

The original and received image are shown in figure 6. The Bit Error Rate for this signal was BER = 0.11. A plot of the received constellations is shown in figure 7. It can be seen that the received message is similar to the transmitted.

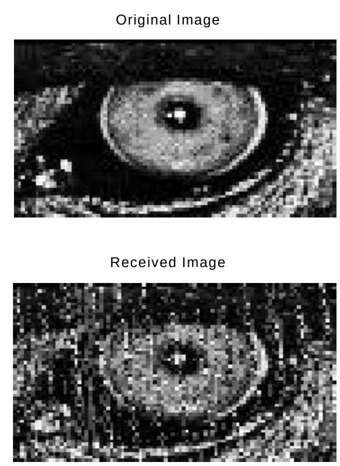


Figure 6: Original and Received images

5. CONCLUSIONS

From the results obtained in the implementation of the modem it can be seen that the way in which multi-carrier transmission was implemented, a combination of ADSL and OFDM, is an efficient manner in which digital multi-carrier transmission can be accomplished. Also, Error control Coding is a useful technique for improving BER in Digital Communications.

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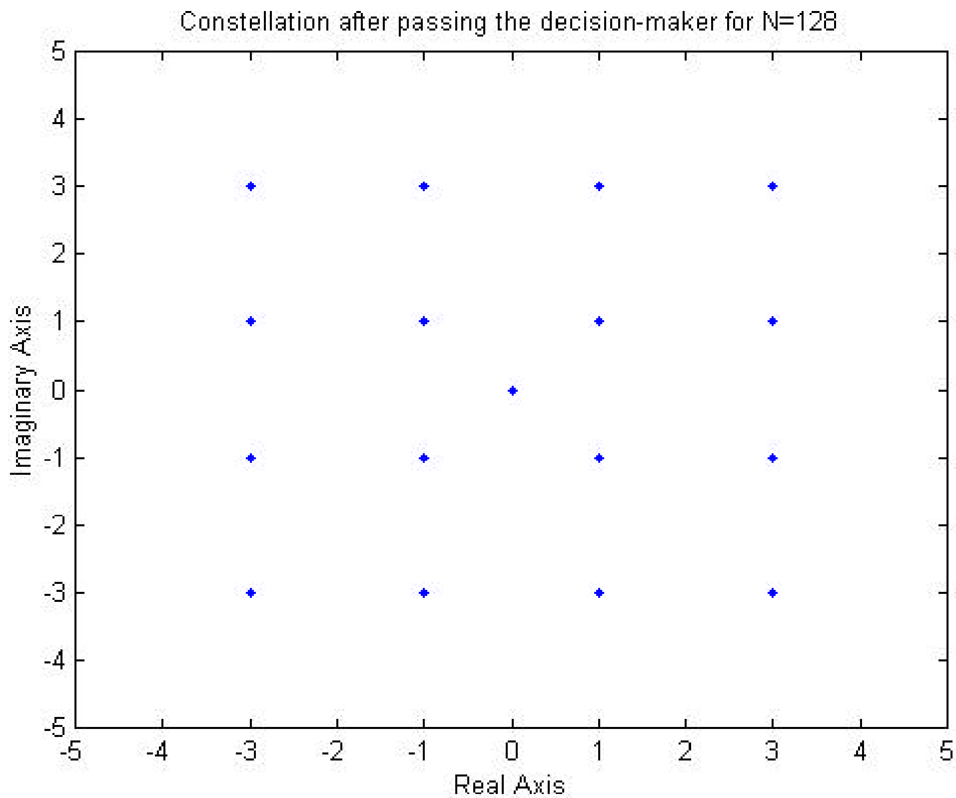

Figure 7: Constellation of the received signals