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Receiver Designing and Channel Estimation

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***Abstract*-With new advances in DSP technologies, OFDM has become popular for the reasons of efficient bandwidth usage and ease of synthesis with new DSP technology. However, it is sensitive to synchronization error and has a relatively large peak to average power ratio. This thesis will provide an overall look into Receiver Designing and Channel Estimation systems and its developments. It will also look into the challenges OFDM faces and concentrate on two main aspects of an OFDM receiver design, channel estimation.**

This will explore concepts and designs, which have already been established and look into some newer technologies. However, we will concentrate on what really makes an OFDM system work, for which a certain degree of knowledge and understanding of signal processing and digital communications is necessary. I hope you will find the explanation clear and insightful!

OFDM is a modulation technique in that it modulates data onto equally spaced sub-carriers. The information is modulated onto the sub-carrier by varying the phase, amplitude, or both. Each sub-carrier then combined together by using the inverse Fast Fourier Transform to yield the time domain waveform that is to be transmitted. To obtain a high spectral efficiency the frequency response of each of the sub-carriers are overlapping and orthogonal. This orthogonality prevents interference between the sub-carriers (ICI) and is preserved even when the signal passes through a multipath channel by introducing a Cyclic Prefix, which prevents Inter-symbol Interference (ISI) on the carriers. This makes OFDM especially suited to wireless communications applications.

Keywords-Wireless Network, Mobile Network.

1. Introduction

OFDM is currently used in European Digital Audio Broadcasting (DAB). OFDM is used in DSL (Digital Subscriber Line) where it is known as DMT (Discrete Multitone). It is used in the European standard Hyperlan/2 and in IEEE 802.11a.

A well know challenge in OFDM systems is that of synchronization in time, to find which instance to sample and to find where the cyclic prefix starts synchronization in frequency, the frequency offset must be very small.

For Equalization, one area which would be interesting to look into is finding TEQ (Time domain Equalizers) to shorten the effective channel impulse response. This is useful in the case where carrier orthogonality is compromised when the channel impulse response grows larger than the length of the cyclic prefix.

It is most common for OFDM equalizers to used in a DFE (Decision Feedback Equalization) structure. I would encourage future students to look into this and why this is such a popular technique and the intricacies behind it.

There is also work to be done on what some people call pre-equalization. These are recovery techniques done at the transmitter using some kind of feedback link. Optimizations in adaptive loading has also been mentioned as a tool for engineers to use in OFDM systems.

The channel estimators considered in this thesis have all been block based channel estimators. In order to obtain better spectral efficiency, we can investigate the performance of channel estimation based on pilot patterns where pilot symbols occur only on some sub carriers. This would free the sub carriers to carry data. This can then be compared to the performance of block based estimators.

The LMMSE and low rank approximate estimators in this thesis only considers the frequency correlation of the signal. In order to maximize the performance, we should consider both time and frequency correlation in the channel estimator. The problem with this is that it is very high complexity due to two correlations. One possible solution to this is to use a seperable filter to consider each correlation separately from the other, which will result in a small performance degradation, but substantial reduction in complexity.

Other possible areas for future study include the study of channel estimation under fast fading environments, where the channel changes faster than the symbol, or estimation under non-sample spaced channels.

My motivation was to use the parameters of the European standard Hyperlan/ 2 and see how they affect an OFDM system. The Hyperlan/2 standard for wireless LAN transmissions in the 5.2 GHz frequency band makes use of OFDM modulation with a TDMA access scheme to efficiently exploit time dispersive channels with frequency selective fading.

Looking at OFDM formulas in textbooks and papers intrigued me to find what these signals actually look like in band pass transmission. This is what this simulation will show.

There was no real reason for choosing Hyperlan/2 over 802.11a, which is also another standard for wireless LAN transmissions. Their parameters are quite similar but they work on different frequencies, have different synchronization protocol and others. The main purpose of the simulation is to see what these signals look like at various stages in the transmission process.

Important OFDM parameters for Hyperlan/2 are summarized in Table below. These parameters are incorporated into our OFDM model.

So, here I will present my OFDM model based on Hyperlan/2 parameters, modulating baseband QAM symbols onto a carrier at the 2.4GHz frequency. Hyperlan/2 actually runs at on 5.2GHz but that would mean I had to sample at more than 10.4GHz which had taken considerable time and strained my computer.

2. OFDM Simulation Model

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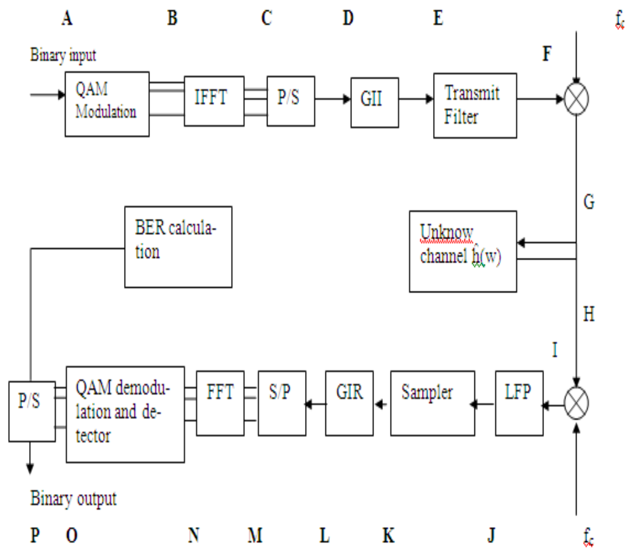

Figure 1. OFDM Simulation Block Diagram

2.1. The Simulation

To begin, we should look at a diagram of the implementation to get a general idea of what is begin simulated.

The exact model of our simulation is given in figure 1. Each part will be explained carefully in all its intricacies. At this point I should also mention that two down conversion methods were implemented. Down conversion is done to take a signal from the band pass to the baseband. This can be done in many ways but the two-implemented techniques are the LPF method and the Hilbert method.

2.2. Channel Estimation

We have taken an OFDM symbol from baseband to band pass and back again. This model has provided us with realizations of the signals at each stage of OFDM syntheses. We have been able to analyses the signals both in time and frequency to see the implications of the parameters we have used.

In a wireless environment, the channel is much more unpredictable than a wire channel because of a combination of factors such as multi-path, frequency offset, timing offset, and noise. This results in random distortions in amplitude and phase of the received signal as it passes through the channel. These distortions change with time since the wireless channel response is time varying. Channel estimation attempts to track the channel response

By periodically sending known pilot symbols, which enables it to characterize the channel at that time. This pilot information is used as a reference for channel estimation. The channel estimate can then be used by an equalizer to correct the received constellation data so that they can be correctly demodulated to binary data.

Modulation can be classified as differential or coherent. For differential, information is encoded in the difference between two consecutive symbols so no channel estimate is required. However, this limits the number of bits per symbol and results in a 3-dB loss in SNR (figure 2). Coherent modulation allows the use of arbitrary signalling constellations, allowing for a much higher bit rate than differential modulation and better efficiency. \

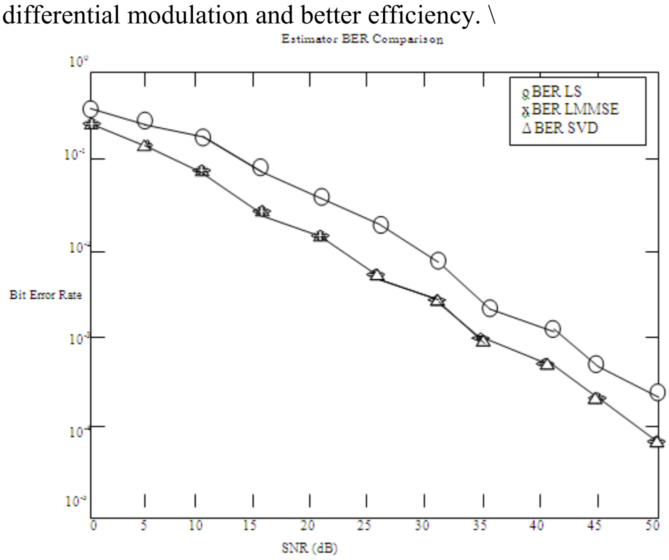

Figure 2. Average bit error rate for the Channel Estimators

2.3. Equalization

There are several methods of equalization commonly used. These are Maximum Likelihood (ML) sequence detection - Optimal in the sense of error probability.

Linear Equalization - filters with adjustable coefficients.

Decision Feedback Equalization (DFE) - uses previously detected symbols to suppress ISI in the symbol presently being detected.

It is clear that equalization in OFDM can be very simple. This is one of the major advantages of using OFDM over single carrier systems. Channel equalization in OFDM actually can be done by just a simple division in the frequency domain. This is because the channel as a filter is convolved with the input signal in the time domain on transmission. This operation is equivalent to multiplication in the frequency domain and thus undoing the effects of the channel is just a division. This optimization follows the Zero-Forcing criteria (ZF) which will be discussed in the following section.

Now, Equalizers like the zero-forcing and MMSE require channel estimation,. This can introduce additional errors in equalization. To test if we can combat this, I have implemented adaptive complex RLS (Recursive Least Squares) and LMS (Least Mean Squares) equalizers. These algorithms can use the statistical averages of correlated signals to derive a representation of the channel and undo its effects accordingly.

The following sections contain somewhat long derivations of the LMS and RLS algorithms. To see what each step does (not just briefing over them) has taken considerable time. I thought it was only befitting to include these since so much time was spent in reading these and again in revision for the report. I believe I have been able to explain them again in simpler terms.So let's get into it and see this of the work have been doing.

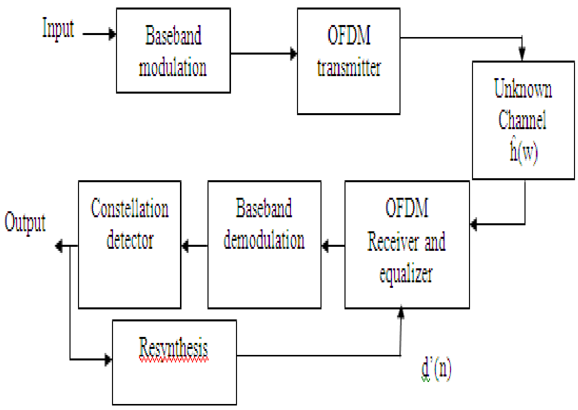

Figure 3. Equalizer block diagram

2.3.1. Linear Equalization

For an input sequence v[n] and the output as an estimation of y[n], the estimate of the nth symbol for a 2N + 1 tap filter is

$$\bar{y}[n] = \sum_{i=-N}^{N} wi\ vn\text{-}I \qquad (1)$$

Here, wi are the complex equalizer tap weights selected based on some optimization criterion. We will continue to denote wi as the equalizer coefficients. This is because they are done so in the Matlab code.

There are two basic criterion of optimization for which the following equalizers are made.

Peak Distortion Criterion - for which the Zero-Forcing Equalizer is derived.

Mean-Square-Error (MSE) Criterion - for which the LMMSE equalizer is derived

2.3.2. Zero-Forcing Equalizer

The zero-forcing equalizer minimizes the peak distortion, which is simply defined as the worst-case symbol interference at the equalizer output.

It is important to mention that our equalizer takes place in the frequency domain. The transversal filters mentioned above and take place in the time domain.

The derivation below of the zero-forcing filter coefficients are derived. I will attempt to explain it more simply because the author likes to word things in a difficult way. His solution results from a z transform which can be easily changed to a frequency perspective on the substitution of z = ejw.

The linear channel filter with coefficients fhng and the equalizer with coefficients fwng can be expressed as a single filter simply by the convolution of the two

$$qn = \sum_{j=-\infty}^{\infty} wj\ hn\text{-}j \qquad (2)$$

We are assuming that the equalizer has infinite taps and then look at the situation where it is finite. The output at the kth sampling can be expressed as

$$\hat{y}_k = q_0\, y_k + \sum_{n \neq k} y_n\, q_{k-n} + \sum_{j=-\infty}^{\infty} w_j\, \eta_{k-j} \qquad (3)$$

where {yk} is the transmitted information sequence and ηk-j is a noise term. And as established before the second term is the ISI term. Now, the peak value of this interference is called the peak distortion.

$$D(w) = \sum_{N=-\infty,\, n\neq 0}^{\infty} |\, qn\, |$$

$$= \sum_{N=-\infty,\, n\neq 0}^{\infty} \left|\, \sum_{j=-\infty}^{\infty} wj\ \eta k\text{-}j \,\right| \qquad (4)$$

Now what we do is select a finite number of taps to force D(w) to zero, that is D(w) = 0 and qn = 0 for all n except n = 0. That's why it's called zero forcing as we are completely eliminating the inter-symbol interference. The condition for this is

$$qn = \sum_{j=-\infty}^{\infty} wj\ \ hn\text{-}j \qquad (5)$$

Finally, if we take the z transform of this, we get

Q(z) = W(z)H(z) = 1

in other words

$$W(z) = 1/H(z) \qquad (6)$$

So, what this means is that the filter which satisfies the zero forcing criterion has the set of coefficients W(z) which is just the inverse of of the channel coefficients H(z). This what is called a zero forcing equalizer.

Table 1. Parameters

Parameter	Value
Sampling rate F0 = 1/T	20 MHz
Carrier central frequency fc	5.2 GHz
FFT size N	64
Useful symbol part duration TU	64T = 3.2μs
Cyclic prefix duration TCP	16T = 0.8μs (optional 8T = 0.4μs)
Symbol interval TS = TU + TCP	80T = 4.0μs (optional 72T = 3.6μs)
Number of data sub-carriers NSD	48
Number of pilot sub-carriers NSP	4
Total sub-carriers NST = NSD + NSP	52
Sub-carrier spacing F = 1=TU	0.3125 MHz
Nominal bandwidth B = NSTF	16.25 MHz
Data symbol constellations	BPSK, QPSK, 16-QAM, 64-QAM

3. RELATED WORK

In 1966 Robert W. Chang published a paper on the synthesis of bandlimited orthogonal signals for multichannel data transmission [7]. It describes a method in which signal can be simultaneously transmitted through a bandlimited channel without ICI (Interchannel Interference) and ISI (Intersymbol Interference). The idea of dividing the spectrum into several channels allowed transmission at a low enough data rate to counter the effect of time dispersion in the channel. As the subchannels are orthogonal they can overlap providing a much more efficient use of the available spectrum.

In 1971, S.B. Weinstein and P.M. Ebert introduced the DFT (Discrete Fourier Transform) to perform the baseband modulation and demodulation [9]. This replaced the traditional bank of oscillators and multipliers needed to create and modulate onto each subcarrier.

In 1980, A. Peled and A. Ruiz introduced the cyclic prefix [8]. This takes the last part of the symbol and attaches it to the front. When this extension is longer than the channel impulse response, the channel matrix is seen as cirulant and orthogonality of the subcarriers is maintained over the time dispersive channel.

OFDM is currently used in European Digital Audio Broadcasting (DAB). OFDM is used in DSL (Digital Subscriber Line) where it is known as DMT (Discrete Multitone). It is used in the European standard Hyperlan/2 and in IEEE 802.11a.

This thesis will explore concepts and designs, which have already been established and look into some newer technologies. However, we will concentrate on what really makes an OFDM system work, for which a certain degree of knowledge and understanding of signal processing and digital communications is necessary

4. CONCLUSIONS

From the 1960s to today, we can see that OFDM is another tool for which the engineer can use to overcome channel effects in a wireless environment. There are many advantages in OFDM, but there are still many complex problems to solve.

We provided a basic simulation tool for future students to use as a starting point in their theses. It is our motivation that by using the parameters of a working system, a much clearer and insightful explanation of the fundamentals of OFDM have been presented.

Channel Estimation is an important part of an OFDM receiver, especially in wireless environments where the channel is unpredictable and changing continuously. A good channel estimation will allow the equalizer to correct the fading effects of the channel. Of the three channel estimators studied, the low rank approximate estimator seems to be the most practical in terms of good performance and low complexity. The LS does not perform well in low SNR environments while the LMMSE estimator complexity seems to high for a small performance improvement.

In OFDM equalization, it seems that the adaptive algorithms used in the OFDM did not add many special benefits. It's adaptive capability allowed the equalizer coefficients to change with time but it is done on the basis on resythesised symbols for which noise and rounding errors may accumulate. These algorithms did not exploit OFDM characteristics which the zero forcing and LMMSE did. It may be wise to incorporate LMMSE design into a DFE. Because the zero forcing and LMMSE equalizers exploit the OFDM design by equalizing in the frequency domain, it is very simple, especially compared the complexities of the adaptive algorithms.

The conclusion of the matter is, It has enabled us to learn for ourselves and digest textbooks, which would have boggled our eyes and minds before.

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