OFDM Project

Wireless Receivers: algorithms and architectures

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1. Introduction

Developing a single carrier system for transmitting a real acoustic signal, we realised the Inter Symbol Interference was so significant that the channel estimation and the equalization was extremely complex. In this project, we try to solve this problem by using Orthogonal Frequency Division Multiplexing.

On the one hand, single carrier systems consist of Time Division Multiplexing, which means that every symbol is transmitted one after another using the whole bandwidth. On the other hand, multicarrier systems consist of Frequency Division Multiplexing, which allows sending multiple symbols (carriers) at the same time but at different frequencies. In other words, the bandwidth is divided into multiple parallel channels of narrower bandwidth but longer time as we can see in Fig.1

For avoiding ISI (Inter Symbol Interference), guard bands are introduced in FDM between every carrier. However, this makes that the entire spectrum cannot be used for data transmission given that the guard bands do not contain data information. We can solve this problem forcing the carriers to be orthogonal, which means that at every subcarrier peak, the rest of subcarriers are exactly zero. This orthogonality features allows the overlapping and therefore saves a lot of spectrum as can be observed in Fig. 2

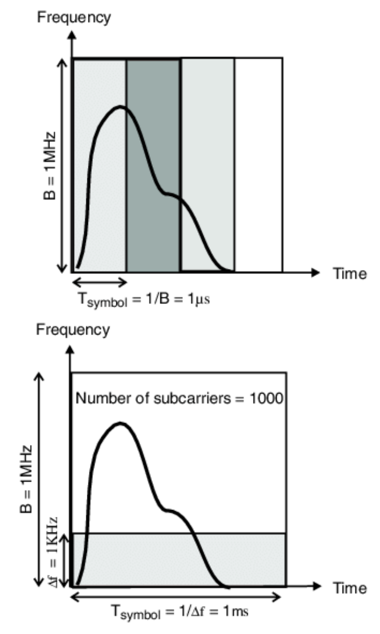


Figure 1: TDM vs FDM

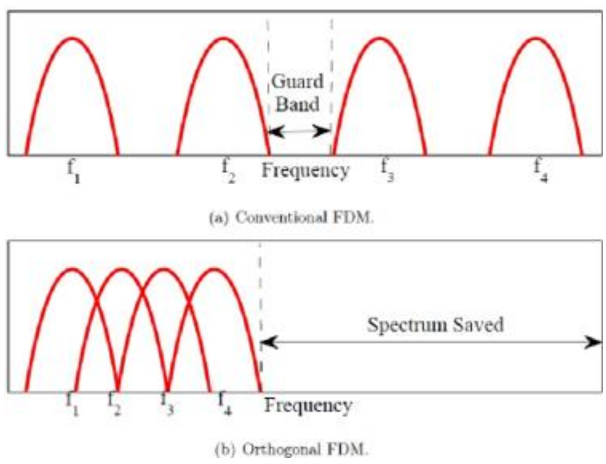


Figure 2: FDM vs Orthogonal FDM

2. System Implementation

Having explained the advantages of OFDM, we will know explore and analyse its implementation. The transmitted signal is composed of a BPSK preamble using a single carrier, followed by a BPSK OFDM "training" symbol and finally several QPSK OFDM data symbols. This signal will be transmitted in an acoustic channel using external speakers and it will be received by an external microphone.

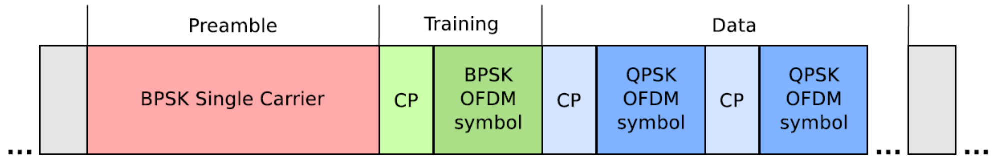


Figure 3: Transmitted frame

2.1 Transmitter

The binary source data is provided as input to the transmitter so that it can prepare the signal to transmit. The bit-stream is converted to 2-bits symbols using QPSK modulation. After this, the symbols are rearranged into N parallel streams as illustrated in Fig.4, so that they can easily be transformed into time domain with an Inverse Fourier Transform, which is equivalent to creating orthogonal subcarriers with a frequency spacing.

After performing the IDFT, the cyclic prefix is inserted between every OFDM symbol. The idea behind it is very similar to the guard band in FDM: in order to diminish the inter symbol interference not between subcarrier but between OFDM symbols. It is called cyclic prefix because the last Ncp samples of each OFDM symbol are copied in front of the symbol. The advantage is that the when transforming back to frequency domain, this operation is equivalent to a circular convolution and the orthogonality is partially recovered. Finally, the baseband signal is up-converted to a high pass-band frequency for easier signal reconstruction.

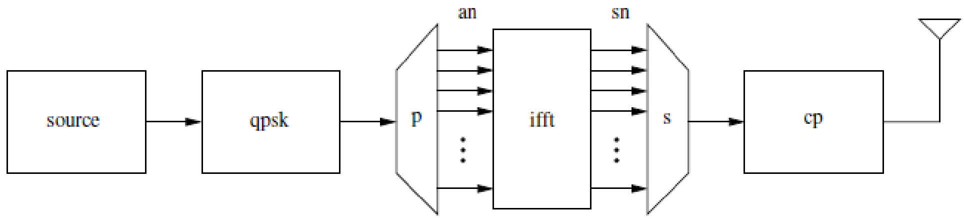


Figure 4: Simple OFDM transmitter architecture

2.2 Receiver

Once the transmitted acoustic signal arrives to the microphone, the task of the receiver is to reconstruct the original bit-stream from the received signal. First of all, the signal is down-converted to baseband again. In the single carrier system, we used a matched filter but, in this case we can use a simple low-pass filter as depicted in Fig.5, therefore making the channel equalizer much simpler. Next, we need to perform the usual frame synchronization where the pulsed shaped BPSK preamble is detected using a correlator.

After the preamble, we use the first symbol received for "training" the channel model: since the training data are known to the receiver, the channel frequency response is estimated, so that we know how the acoustic channel impacts the magnitude and the phase of the different subcarriers over the bandwidth. Making use of this estimation, we can correct the magnitude and phase of the rest of the received OFDM data symbols. Finally, the data is demodulated from QPSK to binary digits and the Bit Error Rate is computed

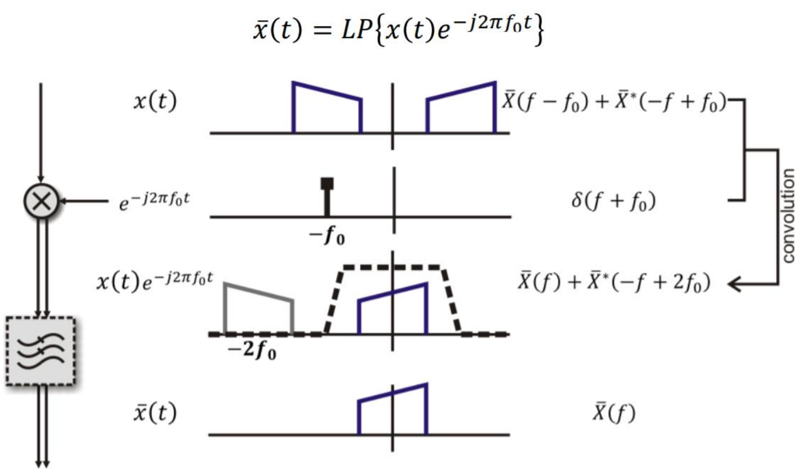
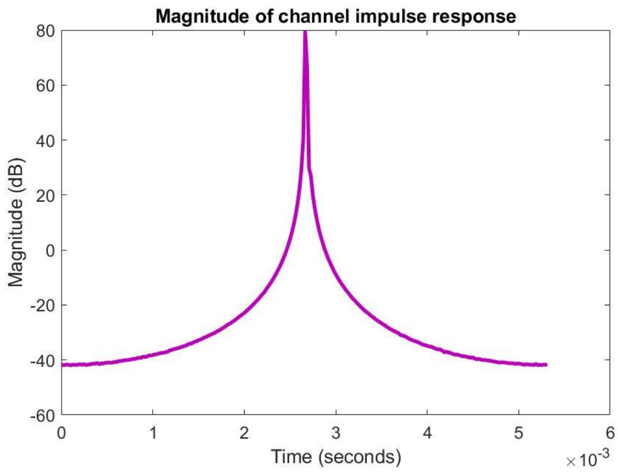


Figure 5: Down-conversion and low-pass filtering

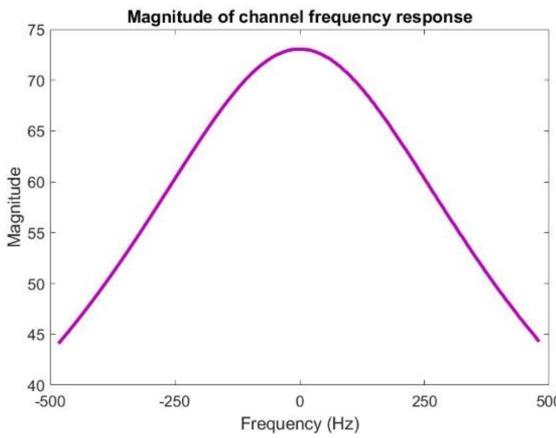
3. Channel Analysis

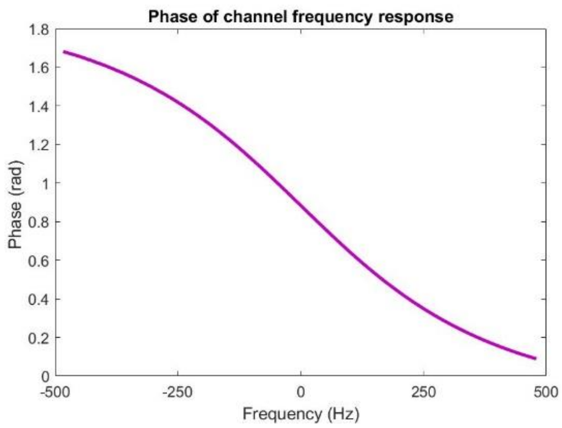
Channel estimation and analysis are the most crucial part for this wireless system to work properly. In this section, we study the signal transmission and reception on an ideal case (bypass mode) and then we experiment with real acoustic channel conditions (matlab mode) by evaluating the channel impulse response.

3.1 Bypass mode



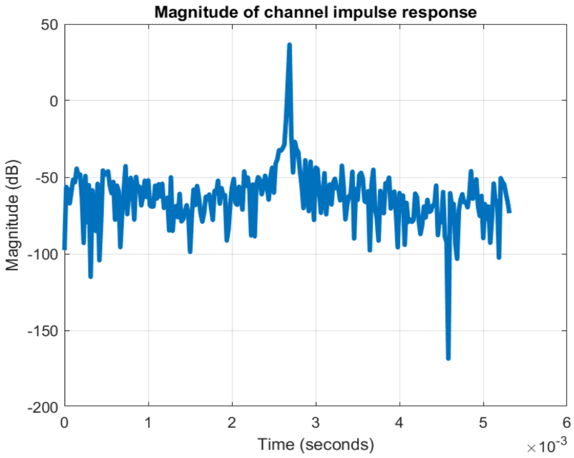
In this mode, the input of the receiver simply equals the output of the transmitter. Therefore, we can analyse and check if everything is working on the ideal case. We can observe that the channel impulse response shows only one central peak which means there is no fading at all, only a direct path. Moreover, the magnitude and phase channel frequency response show the shape of a low-pass filter, which matches with the fact that there is a low-pass filter included at the receiver with a bandwidth of 1.5 times the signal bandwidth, which gives almost 1 MHz of filter bandwidth.



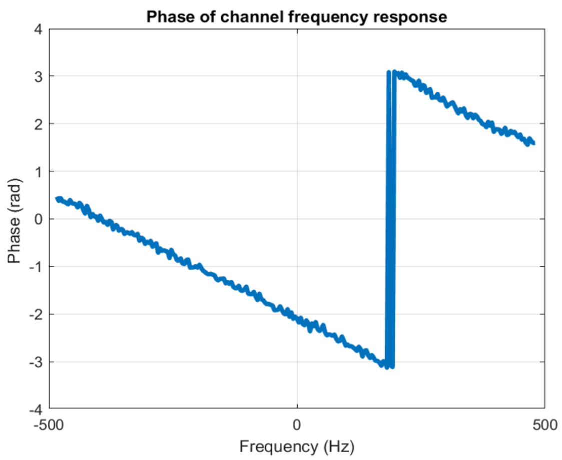


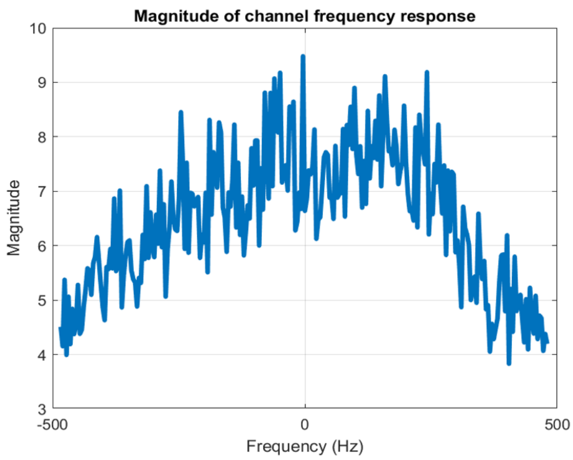
3.2 Matlab Mode

In this mode, the input of the receiver equals the acoustic signal received by the microphone so that we can see how the channel really impacts the signal.



First of all, we try a simple setup: we place the speakers and the microphone only one meter apart and make sure there is nothing in the immediate proximity. As a result, we obtain a channel impulse response with clear central peak and lots of other negligible peaks, which means there is only a direct path and some random noise in the channel. In the frequency domain, again we notice this noise in the numerous fluctuations of the magnitude spectrum and the shifting of the phase spectrum. However, in general terms, we continue to see a low-pass behaviour as expected because of receiver architecture.





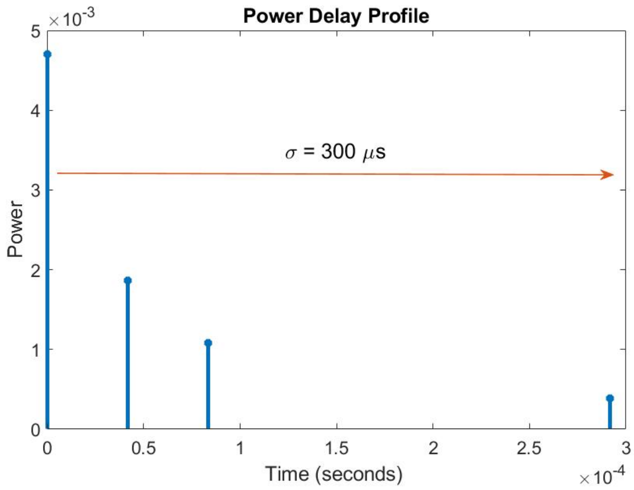
Additionally, we perform another channel estimation with a different setting: we place the speakers and the microphone few meters apart and we also place different objects in between them. In this case, the obtained channel impulse response has multiple peaks which mean signal fading and secondary paths because of the reflections. We will discuss and analyse the consequence of these multiple paths in the next section using the power delay profile

4. Results discussion

After having explained the how the transmitter, the receiver and the channel works, we will know explain other key aspects to take into consideration using the obtained results.

4.1 Power delay profile

The power delay profile quantifies the delay spread of the signal over the channel. It can be obtained by setting a threshold in the channel impulse response above which a peak is considered to be a signal path. In the scenario with reflective objects as explained in the previous section, we obtain a power delay profile which shows fading and multipath propagation.



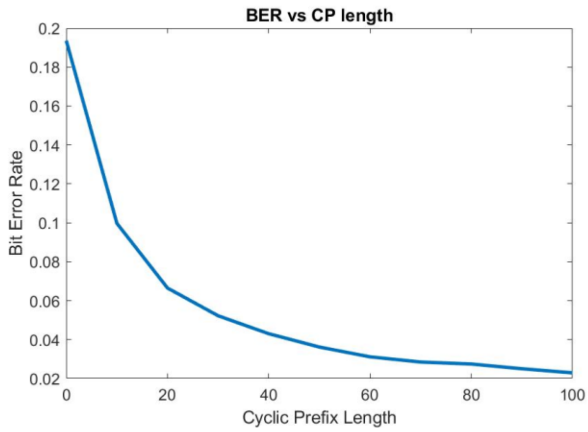
We observe the direct path at time zero and the least power- contributing path at around 300 us. Therefore, we can state that the maximum channel delay spread is 300 us. There are alternative methods for computing the delay spread such as using the root mean square, but we will stick with the maximum delay spread for choosing the appropriate size of the cyclic prefix. The time duration of each cyclic prefix has to be longer than the delay spread for avoiding Inter Symbol Interference. Therefore, given a 300 us delay spread, 48 kHz of sampling frequency and 5 Hz of spacing frequency, the minimal size of the cyclic prefix should be 15 subcarriers.

4.2 System efficiency

Defining N_{data} as the number of data subcarriers and N_{prefix} the number of cyclic prefix subcarriers, the efficiency of the system is computed as follows:

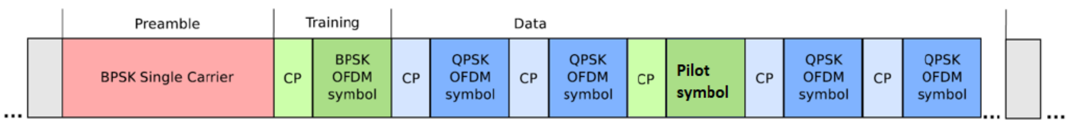
$$\varepsilon = \frac{N_{data}}{N_{data} + N_{prefix}}$$

It is clear that the maximal efficiency is obtained with the minimal cyclic prefix size which in our case is 15 subcarriers as previously explained. Setting the data subcarriers to be 256, the maximal efficiency is 94%. Nonetheless, there is a trade-off between the Bit Error Rate and the system efficiency, which can be understood with the shown figure. Given that with only 15 subcarriers we obtain around 14% of inaccuracies, we decide to set a higher cyclic prefix length of 60 subcarriers, decreasing the Bit Error Rate to 0.03 and decreasing the system efficiency to 81%



4.3 Continuous phase tracking

During the previous analysis the channel was assumed as constant through-out the signal transmission. However, in real situations is rather common that a channel changes its configuration and properties. In order to tackle channel property changes, a continuous phase tracking is required and, in our case, we decide to opt for introducing pilot tones as shown below.



Pilot tones are simply new training symbols known by the receiver so that after a few transmitted OFDM data symbols, the channel can be estimated again and any change can be corrected. In this case there is also a trade-off between the number of pilots and the system efficiency: the higher the pilot rate, the better the data reception but the worse the system efficiency.

5. Conclusions

Thanks to this project we have learnt how OFDM can be very useful to avoid Inter Symbol Interference and to use the spectrum in a more efficient way. Moreover, although the complexity of the receiver is greatly reduced compared to the Single Carrier systems, the composition of the transmitted signal plays a key role in the success of this system. For example, one of the problems we faced at the beginning was the fact of understanding that the preamble is actually transmitted using a single carrier even if it is transmitted along with the other OFDM symbols.

Additionally, the trade-offs between the cyclic prefix length, the pilot tones rate, the Bit Error Rate and the system efficiency are crucial for a successful design. There is not a specific solution or method for properly tuning all the parameters because it really depends on the considered application. We have realised the importance of properly studying the channel where the transmission will be carried out and we have really discovered the link between the channel impulse response and the real characteristics of the channel in the real world.