École Polytechnique Fédérale de Lausanne

Final Project Report
OFDM for Acoustic Transmission

EE-442: Wireless Receivers (Prof. Andreas Burg)

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CHAPTER 1

INTRODUCTION

1.1 BACKGROUND AND MOTIVATION

Frequency-Division Multiplexing (FDM) is a modulation scheme in which multiple signals are gathered for transmission on a single communication channel. Each of these signals are assigned to a different subchannel with a specific frequency within the main channel. To avoid Inter Symbol Inference, FDM divides assigned frequency bands by ranges of unused frequencies called guard bands. Therefore, the bandwidth is not fully used and there can be bandwidth waste [1].

A better scheme is Orthogonal-Frequency-Division Multiplexing (OFDM), where carriers are orthogonal. The data stream is split into parallel streams. Each stream is transmitted over a narrow frequency band [2]. The OFDM symbols are separated by a guard interval. If this interval is longer than the channel impulse response, the system is free of intersymbol interference (ISI), yielding a very efficient bandwidth utilisation.

OFDM has the advantage of enabling more efficient data bandwidth usage and less interference.

1.2 PROJECT DESCRIPTION

In the project, we design and implement an OFDM system for acoustic transmission. This report presents the transmitter architecture, receiver architecture design, results obtained and some conclusions for further improvements.

CHAPTER 2

TRANSMITTER

This section describes the different components of the transmitter as shown in Fig. 2.1

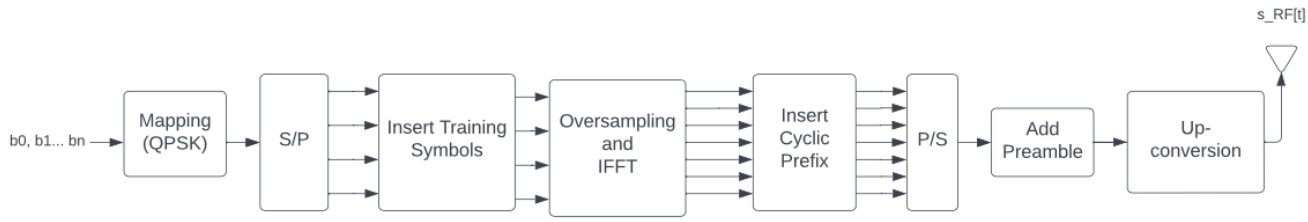


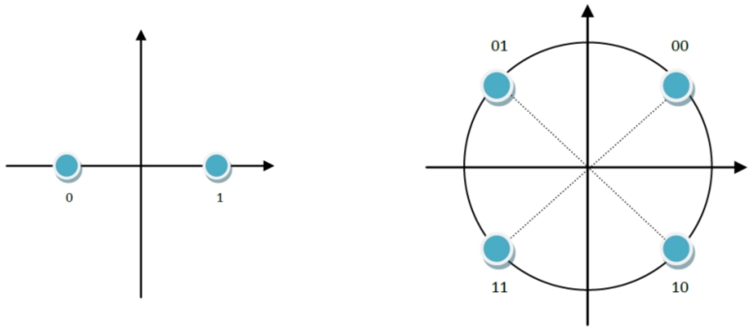
FIGURE 2.1
Transmitter Diagram

2.1 MAPPING

Modulation is performed to map the bit sequence $b_0, ..., b_n$ into a sequence of symbols, also referred as constellation points. In this project, two types of modulation are considered: Binary Phase Shift Keying (BPSK) modulation for the preamble and the training symbols and Quaternary Phase Shift Keying (QPSK) modulation for the data. In PSK modulation, the mapped sequence of symbols are positioned such that they have uniform spacing around a circle. In BPSK, two points are used and they are separated with 180 degrees (–180° and +180°). Each constellation point is encoded as a single bit (a logical 0 or 1). In QPSK, four phase-shift values which are separated from each other by 90° (–45°, –135°, +45° or +135°) is used. Each constellation point is encoded in two bits (00, 01, 11 or 10). Fig. 2.2 shows BPSK and QPSK modulation constellation diagrams.

To transmit all symbols with the same energy, the constellations were normalised. Assuming there are M constellation points a_i and all symbols have the same likelihood, the average transmitted energy (P) is given as

$$P = \frac{1}{M} \sum_{i=1}^{M} |a_i|^2$$



(A) BPSK Modulation (B) QPSK Modulation

FIGURE 2.2
Modulation Schemes

Therefore, for PSK modulations,

$$P = \sqrt{\frac{2}{3}(M-1)}$$

Normalised points $\hat{a}_i$ is given as

$$\hat{a}_i = \frac{a_i}{\sqrt{\frac{2}{3}(M-1)}}$$

For BPSK (M=2),

$$\hat{a}_i = a_i$$

which means symbols can take values +1 or -1.

For QPSK (M=4),

$$\hat{a}_i = \frac{a_i}{\sqrt{2}}$$

which means symbols can take values $\frac{+1+i}{\sqrt{2}}, \frac{+1-i}{\sqrt{2}}, \frac{-1+i}{\sqrt{2}}, \frac{-1-i}{\sqrt{2}}$

2.2 TRAINING SYMBOLS

Before the data, for each frame, a training symbol is inserted. It is used to estimate the channel response and the phase offset later in the receiver. The sequence is known at the receiver and the transmitter.

The training symbol must be a pseudo random sequence in order to prevent the system from a high Peak to Average Power Ratio (PAPR). Otherwise, the transmitter would not be able to generate a signal with such high power. Furthermore, the magnitude of the training symbols should be uniform for all in order to minimize the error in the channel estimation. Therefore, the training symbol should be a uniformly distributed BPSK sequence. It is inserted before the data. Since our system uses Viterbi block-training (described in 3.6), a training symbol is inserted every few symbols.

2.3 CYCLIC PREFIX

As mentioned in the Introduction, the OFDM symbols are separated by a guard interval. It is possible to transmit nothing in this interval. Like many other systems, our transmitter, however, transmits a cyclic prefix. The process is that the last samples (length of N_{cp}) of each OFDM symbol are copied and inserted in front of the symbol as shown in Fig. 2.3. The reason of this cyclic extension is that it helps OFDM achieve no ISI. With the prefix inserted, the received signal contains all the channel multipath effects. Thus, the channel estimate is sufficient to restore the spectrum to its pre-channel state. Furthermore, the cyclic prefix gives the receiver added robustness against timing error [3].

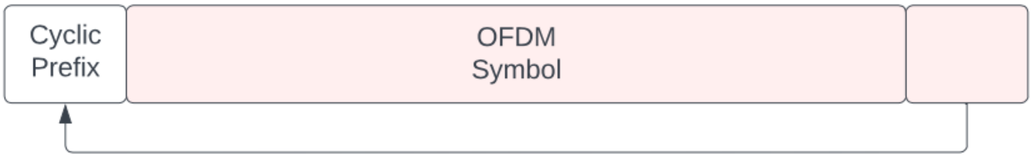


FIGURE 2.3
Cyclic Prefix

2.4 INVERSE DISCRETE FOURIER TRANSFORM AND OVERSAMPLING

The data symbols are transformed into time domain by Inverse Discrete Fourier Transform (IDFT). In this project, we were already provided with the OSIFFT function which performs the inverse transform and the over sampling. Oversampling is performed due to the fact that the symbol transmission rate is different from the sampling rate.

2.5 PREAMBLE

To allow the timing synchronization of the received and transmitted signals, a sequence of symbols which is known to both the receiver and the transmitter, is inserted in the beginning of the data. The receiver uses the autocorrelation function to detect the preamble to signal the start of the data. In order to observe a clear peak in the output of this function and make the detection easier at the receiver end, a pseudo-random sequence is preferred as the preamble [2]. This is added at the beginning of the signal sequence. Before adding the preamble, energies of the preamble and OFDM symbols were normalised.

2.6 UP CONVERSION

Up to this point, the signal $s(t)$ is a complex baseband signal, because it is complex-valued and its spectrum is centered at f = 0. In order to transmit the signal over the channel, it must be transformed into a real-valued signal and its spectrum must be shifted to a carrier frequency f_c which is greater than the bandwidth of the signal. This is done simply by,

$$s_{RF}(t) = Re\{s(t)e^{i2\pi f_c t}\}$$

2.7 TRANSMISSION

The signal is now transmitted in the channel. The frame structure, which we designed using the architecture described in this section, looks like Fig. 2.4

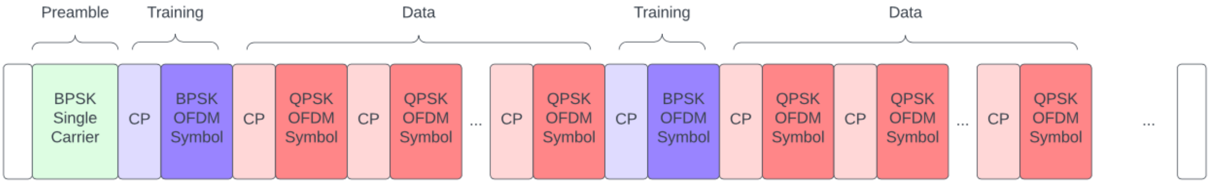


FIGURE 2.4
Frame Structure

Chapter 3

Receiver

This section describes the different components of the receiver as shown in Fig. 3.1

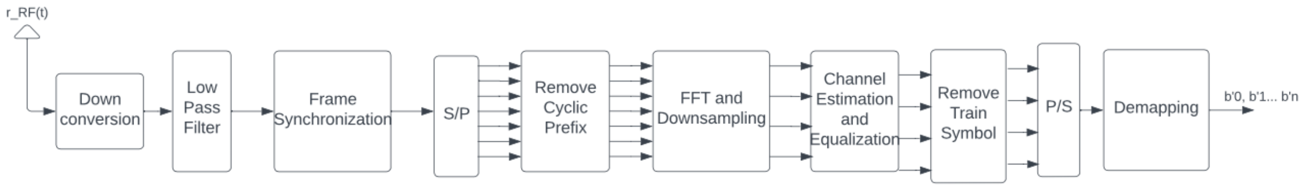


Figure 3.1
Receiver Diagram

3.1 Down Conversion

The received continuous time signal is sampled at the sampling frequency. The receiver converts the signal back into the baseband by

$$r_{RF}(t) = r(t)e^{-i2\pi f_c t}$$

3.2 Low Pass Filter

The signal is filtered with a lowpass filter in order to eliminate the spectral components around $2f_c$. A low pass filter function was already provided to us as part of the project. The filter cut-off point was set slightly lower than the base-band bandwidth.

3.3 Frame Synchronization

The receiver must synchronize itself to the OFDM symbols. To do this, it must be aware at which sample a new OFDM symbol is starting. This is done by the usual frame synchronization by detecting the preamble.

To detect the preamble, the correlation function is used. Since we used a pseudo-random sequence preamble, the correlator can have a clear peak in its output. The start index of the data is determined by setting a threshold and examining the normalised magnitude of the correlator function. Since the received signal is oversampled, once the threshold is reached in the correlator output, oversamples also are input in the correlator function and the start index is determined as the one corresponding to the highest magnitude of the output.

When setting the threshold, a low value of it will cause the receiver to signal the start of a sequence with no actual data sent from the transmitter. A high value will cause receiver to ignore signals sent.

3.4 CYCLIC PREFIX REMOVAL

After the detection of the start of the data, the cyclic prefix is removed from the received sequence.

3.5 DEMODULATION AND DOWNSAMPLING

The data symbols are transformed into frequency domain by Fourier Transform. Moreover, the sample stream should be converted back to symbol rate by discarding oversampled points. In this project, we were already provided with the OSFFT function which performs this transform and the down sampling.

3.6 CHANNEL ESTIMATION

The channel distorts the transmitted signal in its phase and magnitude before it reaches the receiver. Therefore, the effect of the channel on the signal must be estimated. This is done by a training pilot symbol (a symbol known in the receiver and the transmitter). Using Maximum Likelihood Estimation, channel can be estimated for each subcarrier as,

$$\hat{H}_i = argmax_{H_i}|Y_i - H_iT_i|^2$$

where for each subcarrier i, $\hat{H}_i$ is the estimator of channel, H_i is the transfer function of the channel, Y_i is the received symbol and T_i is the training pilot symbol.

The signal can then be corrected as

$$\hat{a}_i = \frac{Y_i}{|\hat{H}_i|}e^{-iarg\{H_i\}}$$

However, channel does not stay the same through the transmission. Therefore, this will likely not result in high accuracy since the same estimator is used to correct all received symbols. To yield higher accuracy, continuous channel tracking can be used to update the estimator more frequently.

1. Block Training: The pilot training symbol is sent every few data symbols to update the estimator more frequently. The frequency of the update depends on the channel conditions.

2. Viterbi-Viterbi Block Training: The pilot training symbol is sent every few data symbols to update the estimator more frequently and Viterbi-Viterbi algorithm is applied. By applying Viterbi-Viterbi, changes that occur before the arrival of next training symbol.

The frequency of the update depends on the channel conditions.

In this project, we apply Viterbi-Viterbi block training.

After the channel estimation is done, the training symbol is removed from the data. The phases and the magnitude of the received signal is corrected with the channel estimate.

3.7 DEMAPPING

Next, signal is transformed from parallel to series. And finally, it is decoded from QPSK to binary. The decision is based on which quadrant of the complex plane the symbols are in and it is done by inspecting the signs of the real and imaginary components of the symbols. Depending on the signs, inverse of what is described in Part 2.1 was performed.

CHAPTER 4

RESULTS

4.1 CHANNEL MODELING FOR THE MODEL CHARACTERIZATION

As the transmission process through a real channel does not allow to easily change the parameters such as SNR or phase shift, the channel was modelled and added to the project in order to carry some experiments. Some results presented in this chapter were achieved using this modeled channel. Our system uses the following configuration :

- 256 sub-carriers
- 128 symbols for cyclic prefix length
- Viterbi-Viterbi algorithm
- Carrier frequency of 6kHz

4.1.1 IMPACT OF CHANNEL VARIATION ON BER

Figure 4.1 and 4.2, show the ability of the receiver to correct the effects of the channel. This correction is effective for angle variations of less than 0.0007 radians and an SNR higher than 15dB.

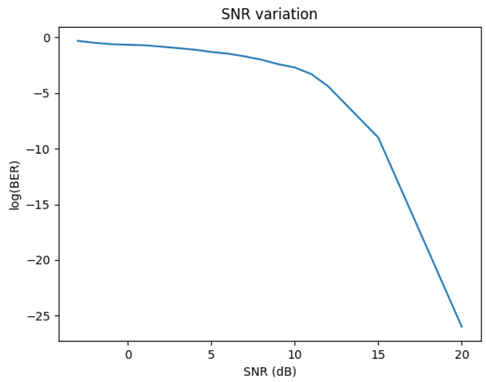


FIGURE 4.1
Noize variation thought a modeled channel (bypass)

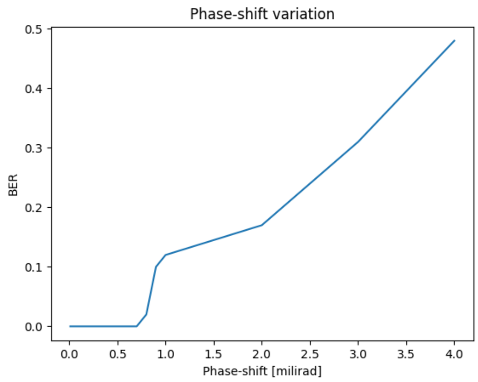


FIGURE 4.2
Phase-shift variation through a modeled channel (bypass)

4.1.2 CHANNEL CORRECTION RESULTS

Figure 4.3 shows the difference between the two correction methods implemented, we can see that the bigger the space between the training, the more significant is the gap between the bit error rate. Below 200 blocks of data per training, the difference is insignificant. This represents a gain of less than 0.5% in transmission speed and will generate additional computations. In this channel condition, the Viterbi-Viterbi algorithm is not really relevant.

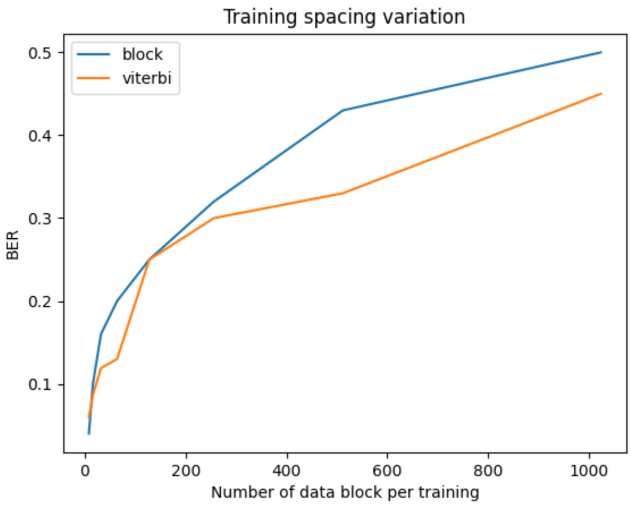


FIGURE 4.3
Comparison between two correction methods using modeled channel (bypass)

4.1.3 EXPERIMENTS WITH CYCLIC PREFIX LENGTH

In order to be effective, the cyclic prefix length must be larger than the channel impulse response. It is important to keep its size to a minimum as it causes losses in transmission speed. Figure 4.4 shows the relation between the cyclic prefix length and the error on transmission.

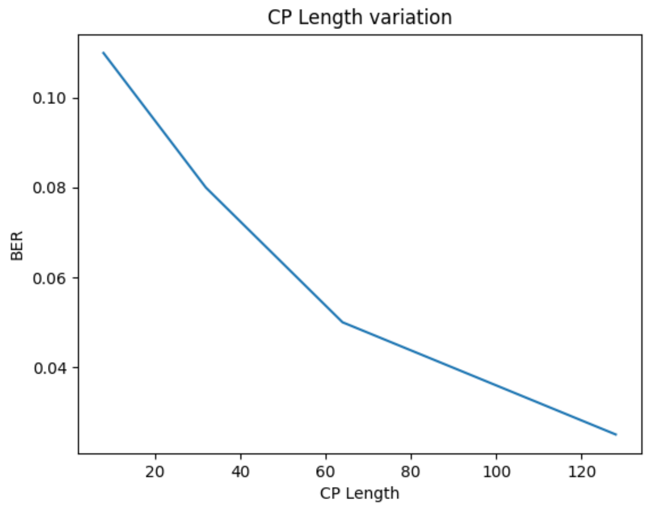


FIGURE 4.4
Cyclic-Prefix length variation through
a reel channel (matlab)

4.1.4 EXPERIMENTS WITH NUMBER OF SUB-CARRIERS

To speed up the transmission, it is important to use many sub-carriers. As the bandwidth is defined by the product between the number of sub-carriers and their width, keeping the same bandwidth and increasing the number of sub-carriers implies reducing the space between these sub-carriers. Figure 4.5 shows the BER evolution with the sub-carrier spacing.

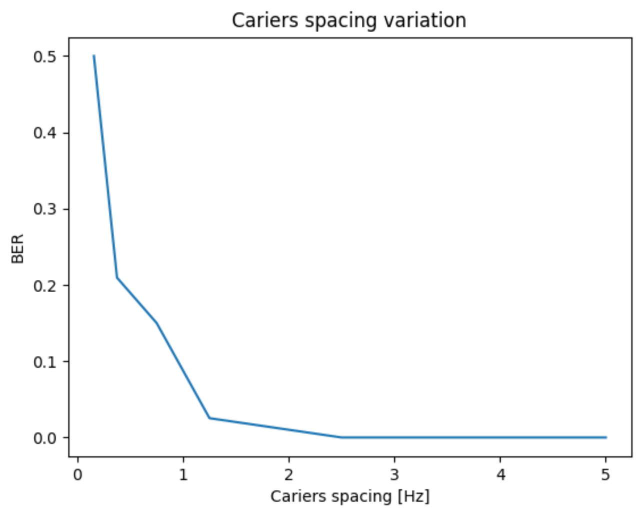


FIGURE 4.5
Sub-carrier spacing variation through
a reel channel (matlab)

The bit error is low for a sub-carrier spacing of 2Hz which allows using 512 sub-carriers for a bandwidth of 1024Hz.

4.2 REAL DATA TRANSMISSION USING SOUND WAVES

The system is capable of transmitting data like an image, random bits are added at the end of the frame to send a number of symbols multiple of the number of sub-carriers. The number of added bits is transmitted on the first 32 bits of the data stream so that the receiver can remove them.

An image of Matterhorn was used to test our system. Figure 4.6 shows the image after the transmission with sound waves.

FIGURE 4.6
Image recovered through a real channel (matlab)

Chapter 5

Conclusion

The system developed is capable of transmitting data as an image through the sound channel. However, even though channel correction methods have been implemented, the model is still very dependent on the channel conditions which may introduce errors if it is not ideal. It would be interesting to implement algorithms to detect and correct these errors by adding bits to the data frame. The transmission speed is also low, to improve it, it would be interesting to use a 16-QAM modulation. To keep an acceptable error rate, it would, however, be necessary to implement a more robust correction method to estimate the amplitude variations at each data block. This is possible by combining the training with the data blocks.

CHAPTER 6

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