MIDDLE EAST TECHNICAL UNIVERSITY

ELECTRICAL AND ELECTRONICS ENGINEERING

EE430 Term Project Report

Part 1

Prepared by

Rafet KAVAK - Ersin KESKİN

2166783 - 2166817

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1) Introduction

As the technology develops further rapidly, the electronic applications in our daily life continue to shift to the digital domain. One of the reasons behind this is, undoubtedly, the ease of the digital data manipulation. Moreover, the construction and the design of digital hardware is rather cost efficient and consumes less time to execute than the analog counterparts. A microprocessor for example, can handle the digital data for many discrete applications very rapidly. Thanks to its fast execution, in real time applications where the amount of delay from input to output is critically important, microprocessors are preferred as they are easy to program via a supporting software.

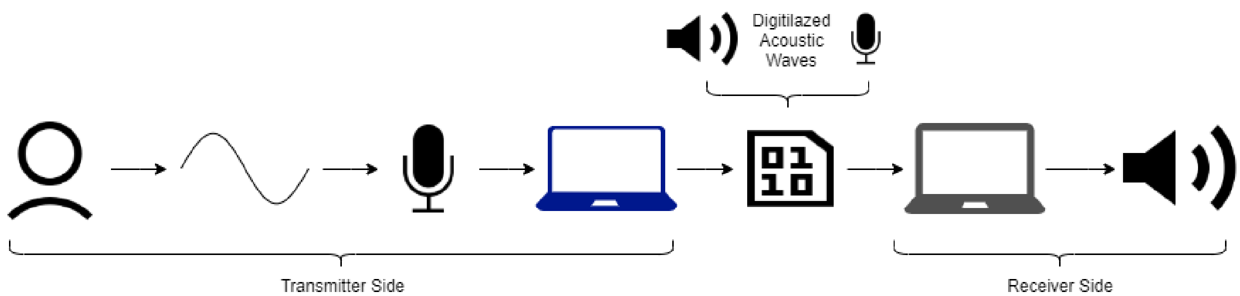

Figure 1: Architecture of the system

In EE430 Term Project, a program that can sense an acoustic wave, process the data, and reconstruct the detected acoustic wave in real time will be written using MATLAB. An illustrative diagram of the project is included in Figure 1. The detection of the autistic wave will be done with a computer which has a microphone to sense the data, and a speaker to convey the digitally processed data. The transmitted data will be detected, processed and broadcasted with a secondary computer that also has installed a microphone and a speaker.

In this report, a discussion on the work to design the overall architecture of the system, and the specifications of the subsystems will be included. The outline of the report is as follows. In Section 2, the construction of the system with its subsystems will be presented. A more detailed discussion on the requirements and the techniques that will be used is provided in Section 3. In Section 4, the conclusory arguments will be discussed. For further reading, References is presented in Section 5.

2) Subsystems

In this section, purpose of the subsystems will be introduced. Mainly, these subsystems can be grouped into two parts namely transmitter and receiver, whose illustrative block diagram can be seen in

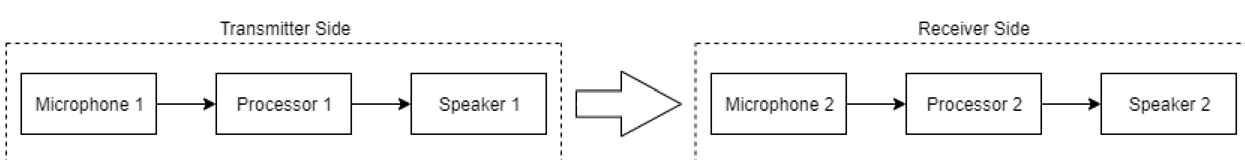

Figure 2: Block diagram of the system

the Figure 2. The aim of the transmitter side is detecting of acoustic waves, digitalizing the data and conveying it to the other side. The detection will be done with a microphone and the transmission of the data will be done through a speaker. The input of the transmitter side is an acoustic wave which is expected from a person. The data will be modulated in MATLAB and the corresponding digitalized output will be transmitted to the secondary side. The receiver side will detect this modulated signal with Microphone 2 and re-transform it to the acoustic waves by broadcasting it with Speaker 2. The main aim of this project is playing back the reconstructed voice with minimum distortion and minimal delay.

Notice that, there will be always some amount of delay arising from both software and hardware of the system. In terms of software, MATLAB will be used in this project and it has a non-zero execution time. Moreover, on the hardware perspective, the detection and transmission will be done with pressure based devices which have significant amount of latency. Consequently, the real time application is impossible, but amount of distortion and delay will be decreased as much as possible.

3) Design Specifications

In this section, some essential background information about the project, and the specifications and the requirements about the design will be discussed. The starting step of the project is the detection of the acoustic signal with a microphone. The microphone, as a component, is a physical device that is sensitive to the disturbances in the air pressure. Thus, as well as its input, the output of the mic is a continuous time signal. However, hence the computer can only handle digital signals, the signal at the output side of the mic needs to be converted into a digital signal. Thus, the continuous signal ought to be sampled through the Equation 1. The sampling process is demonstrated in Figure 3 for convenience.

$$x_d(t) = x_c(nT) \sum_k \delta(t - nT) \tag{1}$$

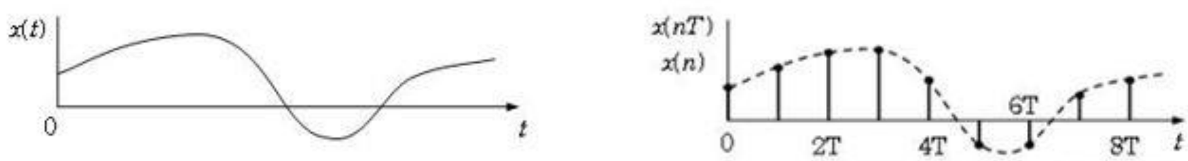

Figure 3: Sampling of continuous signal

Notice here that, in order to be able to reconstruct the sensed wave completely the sampling rate shall obey an important rule, which is nothing but the Nyquist rate. For a proper sampling of a continuous time signal, the sampling rate should be greater than twice of the maximum frequency component of the signal

$$f_{sampling} = 2f_{ct,max} \tag{2}$$

that is to be sampled, which is summarized in Equation 2. If this is not the case, the signal will be distorted which is called aliasing, and the complete reconstruction of it will not be possible. Assuming that a person can produce a sound with maximum frequency of **4kHz**, the sampling frequency should be greater than **8kHz**. If the sampling frequency satisfies Nyquist Rate, the next is to digitalize the signal in order for a computer to process it. This is done by quantizing the sampled signal, that is, approximating the discrete values to the corresponding nearest quantization values. The quantization process is demonstrated in Figure 4. Having digitalized the sensed data, the information is now stored in the successive harmonic signals. These harmonic signals repeat themselves, and they also shall obey the Nyquist Criterion. Here, the inclusion of a Low-Pass Filter may help to increase the intelligibility of the signal manipulation as to guarantee that the signal obeys Nyquist Criterion. An ideal Low-Pass pass filter may be desirable in this

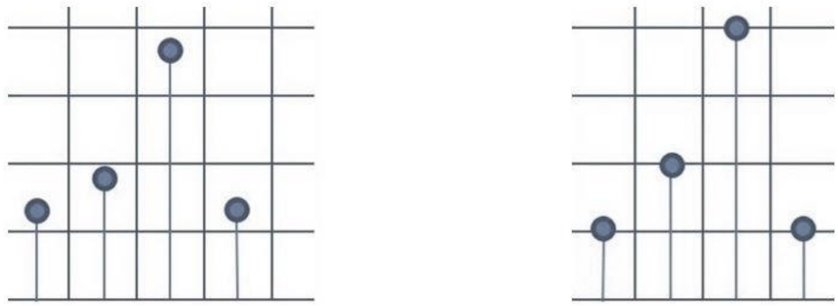

Figure 4: Quantization of a sampled signal

regard; however, providing an ideal filter response is not feasible. The common method is to use prototypes of filters such as Chebyshev and Butterworth filters. Chebyshev filter is an advantageous chose if a narrow transmission band, however it exhibits ripples in either in passband or stopband. On the other hand, Butterworth filter is a more suitable choice if no ripple is desired. A comparison of the transmission

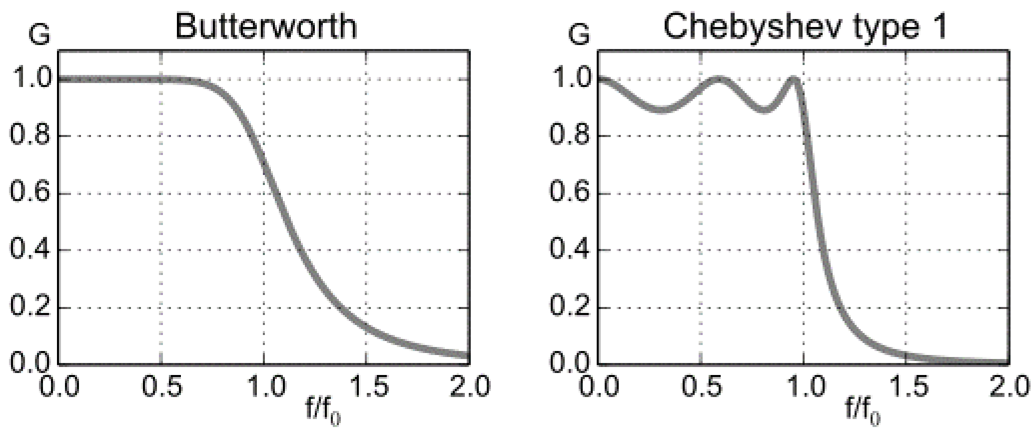

Figure 5: Comparison of the transmission characteristics

response of the corresponding prototypes when the gain is adjusted to unity are included in Figure 5. After filtering the data, in order to send the signal over we need to modulate it beforehand. Modulation is the process of changing the properties of the carrier signal with a modulating signal (input signal) that normally contains data to be transferred. In the 20th century, most of the time amplitude modulation (AM) or frequency modulation (FM) have been used. However, currently, digital modulation techniques are preferred for their capacity to convey larger amounts of data than analog modulation techniques. Also, digital modulation techniques provide more information capacity, high data security and quicker system availability with better quality communication. In digital modulation, an analog carrier signal is modulated by a discrete signal, which has 3 different types namely Amplitude modulation, Frequency modulation, Phase modulation. In these modulation types, the names imply which component of the signal is being modulated. That is, frequency of the signal that is to be transmitted is changed while its amplitude and phase are kept constant. While dealing with acoustic waves, M-FSK which is an example for frequency modulation, BPSK and QPSK who are examples for phase modulation are some of the widely used techniques. In the following sections, some digital modulation types will be introduced alongside with their advantages and disadvantages.

MFSK

Frequency Shift Keying (FSK) is the digital modulation technique in which the frequency of the carrier signal varies according to the digital signal changes. This modulation technique uses two different carrier frequencies to represent binary 1 and binary 0. If more than two frequencies are used, this modulation technique is called Multiple Frequency-Shift Keying (MFSK). As can be seen in the Figure 6, carrier frequency f_1 represents binary data 1 and carrier frequency f_2 represents binary data 0. Notice that amplitude and phase of the carrier signal remain constant while carrier frequency is changed.

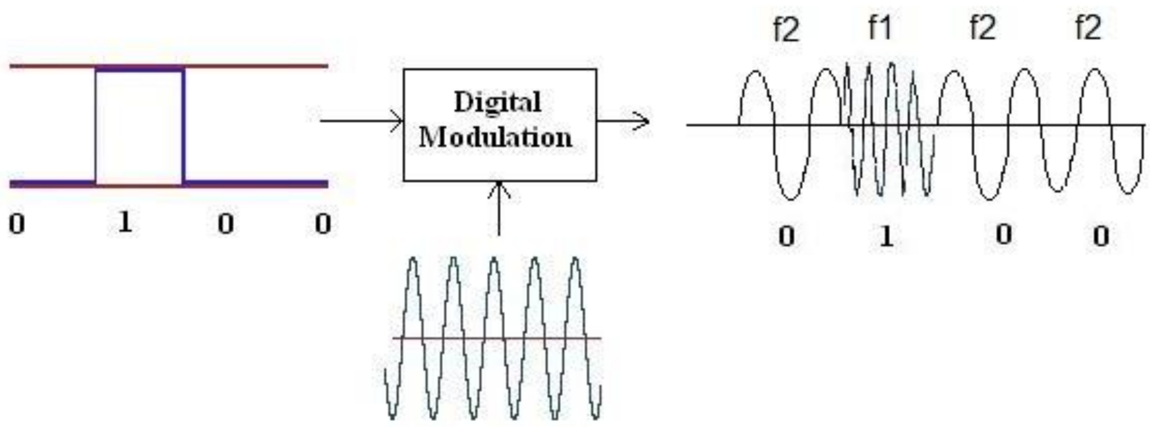

Figure 6 Representation of the FSK

This modulation technique can be represented by following equations,

$$x(t) = Acos(2\pi f_1 t)\ for\ binary\ 1 \tag{3}$$

$$x(t) = Acos(2\pi f_2 t)\ for\ binary\ 0 \tag{4}$$

If more than two frequencies are used, these equations can be increased up to f_M. It is important to note here that, hence the different frequency components caries distinct messages, the carrier frequencies (f_1, f2, …) shall be distinguishable by the physical system. The general human voice is said to be 255Hz maximally, thus, the minimum frequency difference between two distinct carrier frequencies shall be greater than 255Hz. If a transmission of the data with a greater resolution is needed, the frequency may be increased provided that no aliasing occurs. Taking into consideration of some discrepancies, and to give the demodulation system a relaxation time, the generally used frequency difference is 316Hz for a 64kbps rate. It is convenient to discuss what this rate stands for. In data communications, bit-per-second (bps) is a common measure of data speed. 64kbps is suitable for our sampling rate as explained in Equation 5.

$$Bit\ Rate = \frac{Bits\ per\ Sample}{Time\ per\ Sample} = \frac{8}{1/8kHz} = \frac{8}{125 * 10^{-6}} = 64k \tag{5}$$

Advantages of MFSK

- It has lower probability of error.
- It provides high Signal to Noise Ratio.
- It has higher immunity to noise due to constant envelope. Hence it is robust against variation in attenuation through channel.
- FSK transmitter and FSK receiver implementations are simple for low data rate application.

Disadvantages of MFSK

- It uses larger bandwidth compare to other modulation techniques such as ASK and PSK. Hence it is not bandwidth efficient.
- The Bit Error Rate performance in AWGN channel is worse compare to PSK modulation.

PSK (BPSK and QPSK)

Phase Shift Keying (PSK) is the digital modulation technique which uses phase of the analog carrier to represent digital binary data. For instance, phase of the carrier wave is changed according to the binary inputs (1 or 0). In two level PSK, there are 180 phase difference between binary 1 and binary 0 and this can be seen in the Figure 7.

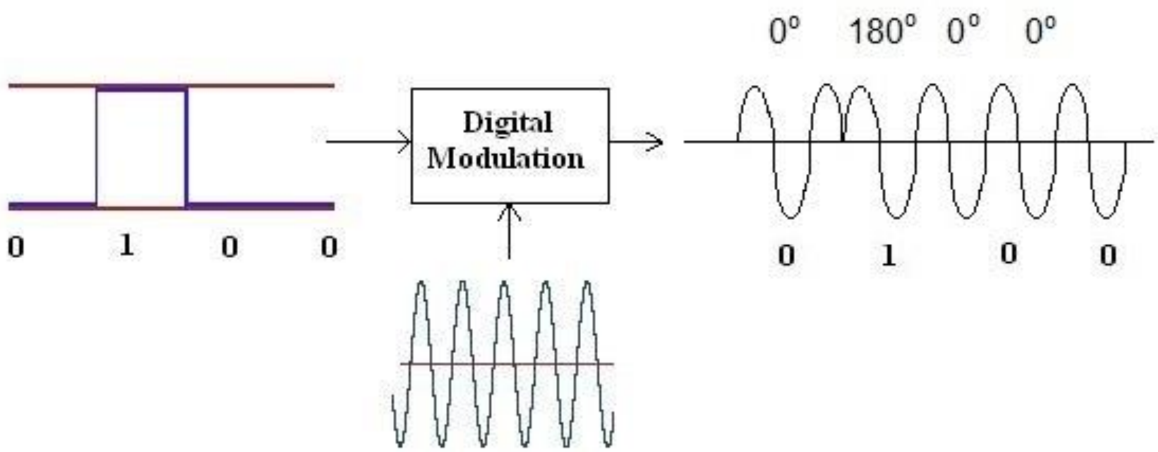

Figure 7 Representation of the PSK

There are many different PSK techniques, but only Binary Phase Shift Keying (BPSK) & Quadrature phase shift keying (QPSK) and their advantages & disadvantages will be mentioned in this report. BPSK represents single bit (0 or 1) by each signalling elements while QPSK represents two bits (00,01,10,11) by each signalling elements. Each of these PSK modulation types have different advantages and disadvantages.

BPSK can be represented by following equation,

$$x(t) = Acos(2\pi f_c t + n\pi), n = 0,1. \tag{6}$$

Also, QPSK can be represented as,

$$x(t) = Acos\left(2\pi f_c t + (2n-1)\frac{\pi}{4}\right), n = 1,2,3,4. \tag{7}$$

<u>Advantages of BPSK</u>

- It is a robust technique.
- Modulated waves can travel longer.
- In order to demodulate signal, less decision is required.
- It requires simpler receiver according to QPSK technique.

<u>Disadvantages of BPSK</u>

- Data rate is smaller than other PSK techniques.
- This technique is not efficient in terms of bandwidth.

<u>Advantages of QPSK</u>

- It has higher data rate.
- It has very good noise immunity.
- Information transmission rate is higher than BPSK.

<u>Disadvantages of QPSK</u>

- Complexity of the system compared to the BPSK.
- It is not power efficient modulation technique.

It is also convenient to note that, while using BPSK and QPSK modulation techniques, the carrier frequency is reversely proportional to bit rate of the transmission. The bit rate shall be adjusted for the carrier frequency to lay on the range of the proper operation of the installed filters and the physical foundation of the system.

Having discussed on several modulation techniques, a very important feature which is bandwidth will be described. Bandwidth is a key concept in several technological fields. In signal processing, it describes the difference between upper and lower frequencies in transmission signals like radio signals, speech signals, etc. For example, a signal with a bandwidth of 20KHz means that signal has frequency components from 0Hz (DC) up to 20KHz. The measurement of the bandwidth of a signal is done in hertz (Hz).

Most of the built-in microphones can sense up to 20kHz. Therefore, carrier frequency must be in the limits of the receiver. However, for the sake of precision, carrier frequency can be selected up to 10 or 15kHz. Speed of the system is another important issue, so optimization of the maximum limit of the carrier frequency will be done according to the laptops' microphone specifications.

The distance between the transmitter-side speaker and receiver-side microphone will be adjusted according to the noise and echo in the environment. We cannot place them too close because of the echo and we cannot place them too far away from each other because of the noise.

Also, laptop speakers will be used during the project. At first, experiments can be conducted in the controlled environment because of to reduce noise. Then according to the acoustic response of the room, different experiments can be performed.

Lastly, inter-symbol interference (ISI) and its causes will be mentioned. In digital communications, symbol rate is the number of waveform changes in the transmission medium. In other words, it can be called as modulation rate. When one symbol interferes with subsequent symbols, it causes distortion. This is an unwanted phenomenon and can be considered like noise. For instance, if the transmitted signal arrives to the microphone through two paths and the difference between the lengths of the two path is 3.4 cm and path difference as a time can be calculated as,

$$\frac{3.4 \times 10^{-2} m}{340 m/s} = 10^{-4}\ sec \tag{8}$$

If maximum 10 symbols reach to the receiver in this time interval, the condition will be satisfied in the part j. Therefore. the maximum symbol rate must be

$$\frac{10\ symbols}{10^{-4}\ seconds} = 10^{5}\ symbols\ per\ second \tag{9}$$

4) Conclusion

In this project, it is desired to create a program using MATLAB to form a system which transfers acoustic waves from one terminal to another. The milestones are sampling and performing operations on sensed sound data, filtering and modulation of the signal, transmitting the resulting signal and replay it from the destination platform. As the project is highly integrated with the digital signal processing, it is beneficial for the participating students in a way that it is possible to test their theoretical backgrounds and have a hands-on experience.

This report is devoted as a starting step for real time application of transmission of acoustic waves. Although a perfect real time execution is not feasible, the aim for this project is to converge a real time operation as much as possible for human hearing. The notices and conclusions of this report will illuminate the way of the actual implementation of MATLAB code that will be written in the second part of this project.

5) References

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