MIDDLE EAST TECHNICAL UNIVERSITY

ELECTRICAL AND ELECTRONICS ENGINEERING

EE430 Term Project Report

Part 3

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1) Introduction

In the previous part, three modulation techniques which are Frequency-Shift Keying (FSK), Binary Phase-Shift Keying (BPSK) and Quadrature Phase-Shift Keying (QPSK) were explained detailed with the necessary MATLAB codes and figures. Also, all three modulation techniques were successfully completed.

In the third part of the project, the modulation codes were rearranged again because they cannot work as real time. Firstly, the codes were divided into two parts which are modulation and demodulation parts. Then, necessary drivers were installed in order to use Data Acquisition Toolbox. This toolbox has a crucial role on the real time process, therefore we spend a lot of time to analyze and learn the example codes. After executing the fft example, we converted our modulation and demodulation codes into functions to begin the real time process. However, we faced a lot of difficulties while integration of our previous codes into functions. Although spending most of our time to analyze LPC codes, we could not find the reasons of errors, but we found some functions like try & catch and by using them, we successfully run our codes.

In order to real time transmission, we tried to transmit "430_dnm.wav" audiofile at first by using BPSK modulation technique, because our BPSK and QPSK techniques are more robust than the FSK technique in the previous part. Also because of the BPSK technique is simpler that the QPSK we choose the BPSK. We successfully completed the overall transmission with BPSK. However, we have known that the length of the "430_dnm.wav" and we arranged our code in this manner, but our code did not work for the speech signal because we could not run the LPC codes. Also, our modulation types are not working perfectly while the transmission of "430_dnm.wav" file but we have tried to succeed the transmission from the air.

Although we cannot finalize the part 3 completely, some design parameters and specifications will be explained in this report. Firstly, all time durations to send one second of the audio file will be presented for each modulation type. For the second case, we could not calculate the maximum time delay between the instant that a word is spelled out to the microphone and reconstruction of it at the speaker at the receiver side, because we could not complete the transmission of the speech signal. And finally, in terms of successful transmission and robust communication, all modulation techniques were inspected.

2) Design Specifications

1)

Table 1 The time duration (in seconds) it takes to send one second of audio file for each modulation type

Modulation Types	The Time Duration to Send One Second of Audio File
FSK	15.63 sec
BPSK	2.63 sec
QPSK	1.57 sec

As can be seen in the Table 1 above, BPSK and QPSK modulation technique were more successful than the FSK modulation technique.

2) We could not achieve a complete system therefore we could not obtain the maximum time delay between the instant that a word is spelled out to the microphone and reconstructed at the speaker at the receiver side.

3) We were able to transmit sound data using BPSK and QPSK modulation topologies when the data is extracted from a file. Although QPSK provided us with a faster transmission, the SNR ratio was higher than that of BPSK. Consequently, successful operation for LPC_rx was not so frequent.

4) BPSK was the most efficient topology of ours, because it has the highest probability of a successful operation. However, its data transmission duration was 2.63 seconds for an acoustics data of 1 second. The main algorithm to make transition error-free is demonstrated in Figure ??, where a single bit increased

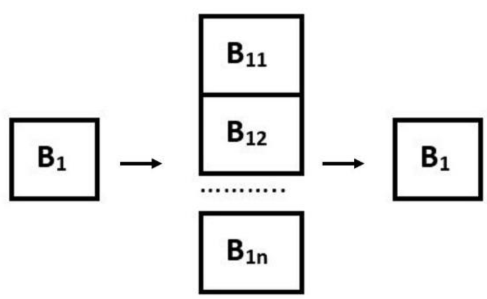

Figure 1: Algorithm to reduce transmission bit error rate

to a number of same bits before transmitting. The resultant data in the receiver side was sampled and filtered to get the original bit. This significantly reduced the bit error rate while transmitting; however, it introduces a significant delay to the operation. In order to increase the execution speed of BPSK, we decreased number of repetitive bits, n, to 3. Unfortunately, it did not help us to reduce the transmission of 1 s data under 2.63 seconds. Disregarding this algorithm and transmitting the data bit per bit was not a convenient choice either due to increased SNR.

3) Conclusion

In the third part of the project, it was desired to implement the code developed through the first two stages on two different computers. The primary computer is used as a transmitter terminal and the secondary computer is used as a receiver terminal that are utilized to operate in real time. The process of the operation on both terminals are presented to the user via two different GUI windows where the user is allowed to change several parameters of the system such as sampling rate and cut-off frequencies of the filters used.

Three different modulation topologies; namely, Frequency Shift Keying (FSK), Binary Phase Shift Keying (BPSK) and Quadrature Shift Keying are used while transferring the acoustic data. The main goal was to increase the execution speeds and the robustness of the corresponding topologies so that a real time application is achieved; however, we were only able to transfer the sound data using BPSK.

To sum up, 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. Lastly, we would like to express our thanks to everyone for their supports, patience and guidance.