MIDDLE EAST TECHNICAL UNIVERSITY
DEPARTMENT OF ELECTRICAL AND ELECTRONICS ENGINEERING
EE430 Digital Signal Processing, 2019-2020 Fall

EE430 Term Project

In this project, you are going to design and implement a system that will transmit voice from one computer to another by using acoustic waves as the communication medium. The transmitter-side computer will capture the voice using a microphone, process it and transmit acoustic waves from its speakers. The receiver-side computer will receive the transmitted acoustic waves with its microphone, process it and play back the reconstructed voice from its speakers.

The project consists of 3 parts. The due date of each part, as announced in the syllabus, is as follows:

Part-1	System requirements, block structure and discussions of the chosen architecture	20/10/2019
Part-2	Basic system implementation in MATLAB	01/12/2019
Part-3	Final System implementation and demo	15/01/2020

Part-1

The main goal of part-1 is to do a preliminary work to design the overall architecture of the system using many subsystems and to determine the requirements of each subsystem. In particular, you need to include at least the following items in your report:

- **Block Diagram:** Provide a block diagram of the overall system.
- **Subsystems**: Explain the purpose and tasks of each subsystem in your block diagram, including the hardware and/or software components of each subsystem.
- **Discussion**: Discuss the following issues, determine the feasible ranges of parameter values, by explaining your specific choices
    a. The bandwidth, sampling-rate and any related filters for the sampling of the voice at the transmitter side.
    b. Quantization of the sampled voice signal.
    c. The advantages/disadvantages of the following modulation types for the digital communication system using acoustic waves. Use the following links and references to obtain information about the modulation types.
        - M-FSK: The symbols are transmitted with $\cos(2\pi f_1 t)$, $\cos(2\pi f_2 t)$, … , $\cos(2\pi f_M t)$. As there are M different transmit waveforms, one symbol consists of $log_2(M)$ bits.
          https://en.wikipedia.org/wiki/Frequency-shift_keying
          https://en.wikipedia.org/wiki/Multiple_frequency-shift_keying
          Proakis, John G., and Masoud Salehi. *Fundamentals of communication systems*, 2^{nd} Edition, Pearson Education, 2014.

        - BPSK: The symbols are transmitted with $\cos(2\pi f_c t + n\pi)$, $n = 0{,}1$. One symbol consists of one bit (which is either 1 or 0)
          https://en.wikipedia.org/wiki/Phase-shift_keying#Binary_phase-shift_keying_(BPSK)
          Proakis, John G., and Masoud Salehi. *Fundamentals of communication systems*, 2^{nd} Edition, Pearson Education, 2014.

- QPSK: The symbols are transmitted with $\cos\left(2\pi f_c t + (2n-1)\pi/4\right), n = 1,2,3,4$. As there are four possible transmit waveforms, one symbol consists of two bits, which can be 00, 01, 10 or 11.
  https://en.wikipedia.org/wiki/Phase-shift_keying#Quadrature_phase-shift_keying_.28QPSK.29
  Proakis, John G., and Masoud Salehi. *Fundamentals of communication systems*, 2^{nd} Edition, Pearson Education, 2014.

d. The bandwidth of the communication system, carrier frequency(ies).
e. The minimum and maximum separation between the carrier frequencies if frequency shift keying (FSK) is to be used as modulation type.
f. The minimum and maximum carrier frequencies if BPSK or QPSK is to be used as modulation type.
g. The distance between the transmitter-side speaker and receiver-side microphone.
h. The bitrate of the communication system.
i. Loudspeaker, microphone characteristics, room acoustic response.
j. Let's say that the transmitted signal from the loudspeaker arrives at the microphone at the receiver through two paths and the difference between the lengths of the two paths is 3.4 cm. This will cause inter-symbol interference (ISI), for which the received symbols are affected by the previous symbols due to time delay between the arrival times of the different paths. What should be the maximum symbol rate so that a received symbol contains interference from no more than 10 previous symbols? (You may assume that the speed of sound signal is ~340 m/s)

Your report should be uploaded as a pdf file to ODTUClass before the due date.

Part-2

The main goal of part-2 is to implement (in Matlab) and demonstrate a preliminary version of the system designed in part-1 to transmit a recorded voice signal to the receiver without any constraints on the communication performance of the system. In other words, the voice signal may have a duration of 3 seconds but the communication may take longer, such as 50 seconds.

You are required to implement the following modulation methods:

- Frequency-shift keying (FSK):
- Binary phase-shift keying (BPSK):
- Quadrature phase-shift keying (QPSK):

The complexity and the performance (in "certain" terms) of the above modulation types increase from FSK to QPSK.

When you submit your Part-2, you need to supply your MATLAB codes and written report which includes the plots during your experiments. Both your codes and report should be packed in a zip file and uploaded to ODTUClass before the due date.

Part-3

The main goal of part-3 is to improve the design and implementation in part-2 so that the transmission time is as small as possible with respect to the duration of the sound transmitted. You are expected to employ the data acquisition toolbox in MATLAB to be able to simultaneously receive data from the microphone at the receiver and process the received samples. This is to minimize the possible delay to process the received samples to detect the transmitted bits. An example code for this task is provided in the following link:
https://www.mathworks.com/help/daq/examples/getting-started-acquiring-data-using-audio-in-session.html
In addition, the system should have the following properties:

1. The sampling rate of the voice can be chosen by the user. At least the following choices should be provided to the user: 1 kHz, 2 kHz, 4 kHz, 8 kHz, 16 kHz.
2. The quantization level of the sampled voice signal in bits per sample (bps) can be chosen by the user. The provided choices for the quantization levels should range from 24 bits/sample (bps) (or the maximum possible by the hardware) to 1 bps.
3. The sampled voice signal can be filtered with a low-pass filter that has variable bandwidth, stopband attenuation and passband ripple size. Each of these parameters can be adjusted by the user. While there is no restriction on the type of the filter, the use of optimum filters in this project may help you get more credit.
4. Time domain speech signal plot, frequency domain signal plot, filter frequency response plot can be included in the user interface.
5. There should be a user interface which allows the selection of system parameters.

When you submit your Part-3, you need to supply your MATLAB codes and written report which includes the plots during your experiments. Both your codes and report should be packed in a zip file and uploaded to ODTUClass before the due date.