# MCT-421: Digital Signal Processing

SEMESTER PROJECT

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| **Instructor:** | Muhammad Awais Hafeez |
| **Deadlines:** | Project Selection; 5th March. 2023<br>Final Submission; 5th May, 2023 |
| **Place:** | awais.hafeez@uet.edu.pk<br>Computer Lab, MCE Dept. |

## OBJECTIVES

Students will be given a chance to prove their learning of Digital Signal Processing techniques. They shall be able to demonstrate whatever they have learned over the course of the whole semester in form of this project. Projects have been designed in such a way that each encompasses a vast area of signal processing techniques and also encourages students to learn few things on their own, so that it may be a suitable measure of the student's knowledge and understanding of this course. Such projects will help students to develop a sense of formulating signal processing algorithms and implementing them on practical examples.

## DELIVERABLES

Each group, comprising of two students, shall present its project during final lab viva voce accompanied by a report and source code. Send your project, satisfying the requirements, to awais.hafeez@uet.edu.pk. Pack everything into a single .zip file (i.e. your report and all function files). You will get bonus points for the use of good programming techniques.

## GRADING SCHEME

This project comprises of 30% of this lab credit. Failing to submit project before the deadline will result in negative marks.

***Table 1: Project listings***

| Sr. # | Project Title | Remarks |
|---|---|---|
| **1** | Dual Tone Multi-Frequency Tone Detection | Advanced |
| **2** | Speaker Identification | Medium |
| **3** | Artificial Reverberations | Beginner |

## SCHOLASTIC ETHICS

It is *emphasized* that students should submit their own work. If part of an existing code is needed to be used it should properly be credited. Plagiarism will **not** be tolerated. Remember *copy-paste* is worst form of plagiarism.

## PROJECT REPORT AND PRESENTATION

The course instructor, after a thorough review, will approve the final submission in form of a report and accompanying code. The report will comprise of various sections including abstract, introduction, methodology (algorithm), possible future enhancements and conclusions.

All the projects will be presented on Friday, May 5th, 2023.

PROJECT DETAILS

**Project # 1: Dual Tone Multi-Frequency (DTMF) Tone Detection**

Touch-tone telephones use a dual tone multi frequency (DTMF) scheme to encode key presses as audio tones. Thus for every key pressed a unique combination of two distinct audio tones is created, with frequencies specified in the following table:

| Freqs | 1209 Hz | 1336 Hz | 1477 Hz |
|---|---|---|---|
| 697 Hz | 1 | 2 | 3 |
| 770 Hz | 4 | 5 | 6 |
| 852 Hz | 7 | 8 | 9 |
| 941 Hz | * | 0 | # |

For example, when the digit 4 is pressed, tones at 770 Hz and 1209 Hz are generated and summed together.

The aim of this project is to investigate and implement method for decoding key presses, such may occur when a digit of complete telephone number is dialed.

A complete project must have following implemented:

- A GUI in MATLAB presenting a complete dial pad, whose any key when pressed would generate a suitable tone for at least 500 milliseconds.
- Fourier Spectrum of the generated tone should be displayed (For the time being suppose only one key is pressed and analyzed).
- Decode from the Fourier Spectrum that which key was pressed using any suitable algorithm and display the result on the same GUI.
- Optional: You should make three different function files; one for generating a tone whenever a key is pressed, one for displaying the Fourier Spectrum of any tone, and one for decoding the pressed number.

**Project # 2: Speaker Identification**

Intelligence obtained a speech communication recording of a terrorist. However, the signal is corrupted by sinusoidal noises from a siren. As a signal processing expert, you were asked to analyze the recording and remove the corruption as much as possible. Following the step-by-step instructions below and using your DSP knowledge learned in this class, design and implement MATLAB programs to process the signal.

A complete project must have following implemented:

- A GUI in MATLAB giving the user an option to start the procedure.
- You should read the audio file provided and display its Fourier Spectrum.
- Design a filter that will filter out the noise. You should display the Impulse response of your filter and Fourier Spectrum of the filter output.
- Play the output of the filter proving that your filter has filtered the noise to an acceptable extent.
- Optional: You should make three different function files; one for reading the audio file and displaying its Fourier Spectrum, one for displaying the Impulse response of your filter, and one for implementing the filter on your input audio file and displaying and playing the final results.

**Project # 3: Artificial Reverberations**

Reverberation is a very common and often unnoticeable phenomenon in our lives. The surrounding walls in a concert hall and in the office, the walls of the buildings in the street, every object around us reflects the sound that is propagating through the space. Because these reflections, what we hear is not only the information that leaves from the sound source; some additional components are added to the original sound. The reflections modify the perception of the sound, changing its loudness, timbre and its spatial characteristics. The presence of reverberation is especially interesting in music. This effect adds life and a sense of space. The reverberation is associated with the architecture and

acoustic of concert halls, and this is a critical factor for good sounding in concert performances. In recorded music, there has been a great effort in simulating this effect through some electro-acoustic devices. Nowadays, digital signal processing techniques are mostly used for these purposes.
The aim of this project is to investigate and implement methods for generating artificial reverberations.
There are many ways of generating such artificial effects. Many researchers around the world have proposed algorithms for artificial reverberations. In this project, your task is to study and implement Schroeder's reverberation network proposed by Schroeder in the early 1960s. Schroeder proposed a reverberator consisting of parallel **comb filters** and series **allpass filters** providing "an artificial reverberation which is indistinguishable from the natural reverberation of real rooms" (Figure 1).

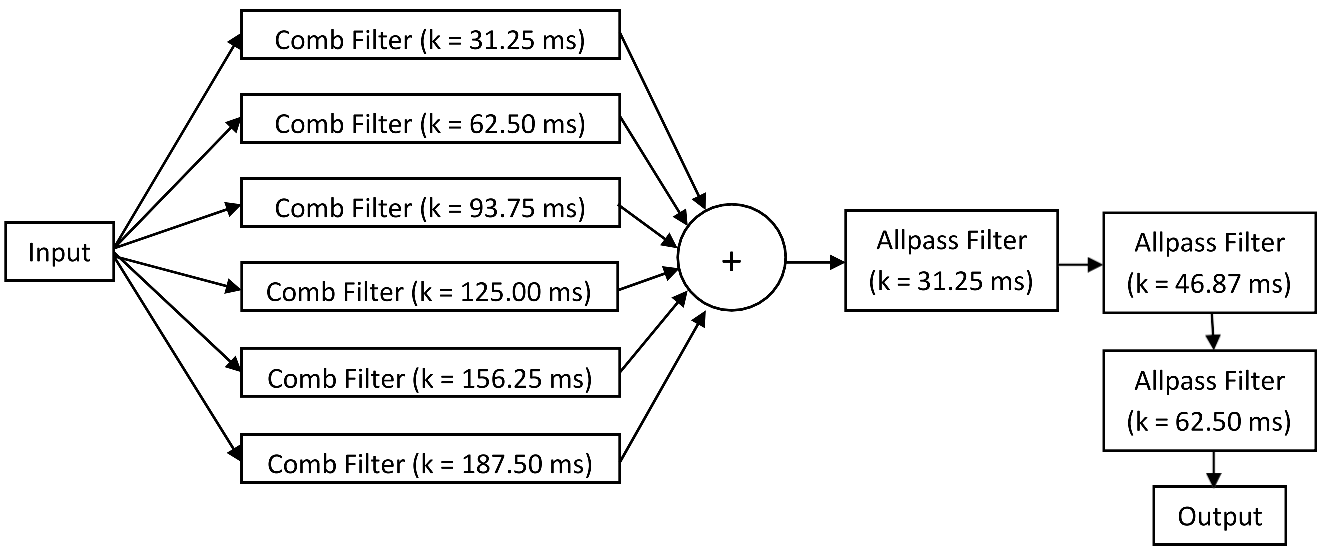


Figure 1: Schroeder's Reverberator Network

A complete project must have following implemented:

- A GUI in MATLAB giving the user an option to start the procedure by allowing him to record a sound lasting for 10 seconds.
- Apply Schroeder's reverberator network on the recorded sound.
- Display the Fourier Spectrum of the original and processed sound.
- Play the output of the network.
- Optional: You should make three different function files; one for recording the audio file and displaying its Fourier Spectrum, one for a comb filter, one for an allpass filter, one for applying Schroeder's network to a recorded sound and playing and displaying the final results.