HITEC University, Taxila
Department of Computer Engineering

BS Computer Engineering Program

Course Title:	**EC-390 Digital Signal Processing 4 (3+1)**
Batch / Semester:	Batch 2020 / 6th Semester
Instructor:	Dr Raza Ali Shah
Target PLO:	**CLO3:** Cognitive Domain (Level 4): Design and formulate different digital filtering techniques.

Complex Engineering Problem

Design and Implementation of Digital Filter

Marks Distribution:

Filter Design (Software Implementation)	: 50%
Report	: 10%
Viva	: 30%
Demo	: 10%

Objective:

The objective of this assignment is to design a digital filter that utilizes various digital signal processing techniques such as windowing, frequency sampling, and design methods like Butterworth, Chebyshev, or Finite Impulse Response (FIR). The primary purpose of the filter is to process a digital signal by selectively attenuating or amplifying specific frequency components according to desired specifications. The design process draws upon the knowledge acquired in laboratory and lecture sessions on digital signal processing, aiming to create an efficient and effective filter that can be implemented digitally.

Note:
The assignment has to be done in Groups. Each group can have a maximum of 3 students.

Project Details:

You have been tasked with designing an audio equalizer for a music production studio. The studio wants to enhance the sound of their recordings by adjusting specific frequency bands. Your job is to design a digital filter that can achieve this.

Define the requirements:
a) The equalizer should have five adjustable frequency bands: low, mid-low, mid, mid-high, and high.
b) Each frequency band should have a user-defined gain control ranging from -12 dB to +12 dB.

c) The equalizer should maintain a linear phase response to avoid phase distortion.

Design approach:
To design the digital filter, you decide to use a Finite Impulse Response (FIR) filter due to its linear phase characteristics and ease of implementation.

Implementation steps:
a) Determine the desired frequency response for each frequency band. Let's assume the center frequencies are: 50 Hz, 200 Hz, 800 Hz, 3 kHz, and 12 kHz.
b) Express the desired frequency response of each band as a linear combination of complex exponentials using the Fourier series.
c) Design the filter coefficients based on the desired frequency response using methods such as windowing, least squares, or frequency sampling.
d) Implement the digital filter in a software or hardware platform suitable for audio signal processing.

Testing and fine-tuning:
a) Test the equalizer with various audio signals and adjust the gain controls for each frequency band to achieve the desired sound enhancement.
b) Fine-tune the filter coefficients and gain controls based on subjective evaluation and feedback from sound engineers or musicians.

By following this design process, you can create a digital filter that meets the studio's requirements for an audio equalizer. The Fourier series representation of the desired frequency response allows you to manipulate specific frequency bands and achieve the desired sound enhancement for professional music production.

Estimated Time:

Assignments can vary greatly in terms of complexity and scope, so it's challenging to provide an accurate time estimation without specific details about the assignment. However, I can provide a general approach to dividing tasks and an estimated time range for each task based on a typical assignment related to designing an audio equalizer.

Understanding the Requirements and Research (2-4 hours):

Familiarize yourself with the assignment instructions and the requirements of the audio equalizer.

Research digital filter design techniques, FIR filters, and frequency response manipulation.

Frequency Response Design and Fourier Series Representation (2-3 hours):

Determine the desired frequency response for each frequency band based on the assignment requirements.

Express the desired frequency response using the Fourier series representation.

Filter Coefficient Design (3-6 hours):

Select an appropriate design method (e.g., windowing, least squares, frequency sampling) based on the desired frequency response.

Design the filter coefficients using the chosen method.

Implementation (4-8 hours):

Implement the digital filter using a software or hardware platform suitable for audio signal processing.

Test the implementation with various audio signals.

Testing and Fine-Tuning (2-4 hours):

Test the equalizer with different audio signals and adjust the gain controls for each frequency band to achieve the desired sound enhancement.

Fine-tune the filter coefficients and gain controls based on subjective evaluation and feedback.

Please note that these time estimations are approximate and can vary depending on your familiarity with the topic, your programming skills, and the complexity of the assignment. It's important to allocate additional time for unforeseen challenges and to review and refine your work before submission.