Module (Course Syllabus) Catalogue

2023-2024

College/ Institute	**Erbil Technology College**	
Department	**AIT**	
Module Name	**Digital Signal Processing**	
Module Code	**DSP504**	
Degree	**Technical Diploma** ☐ **Bachelor** ☒ **High Diploma** ☐ **Master** ☐ **PhD** ☐	
Semester	**5th Semester**	
Qualification		
Scientific Title	**Assist.Lecturer**	
ECTS (Credits)	**6**	
Module type	**Prerequisite** ☐ **Core** ☐ **Assist.** ☐	
Weekly hours		
Weekly hours (Theory)	**(2)hr Class**	**(162)Total hrs Workload**
Weekly hours (Practical)	**(2)hr Class**	**( )Total hrs Workload**
Number of Weeks	**12**	
Lecturer (Theory)	**Jabbar Majeed Sadeq**	
E-Mail & Mobile NO.	**jabbar.sadeq@epu.edu.iq**	
Lecturer (Practical)	**Jabbar Majeed Sadeq**	
E-Mail & Mobile NO.		
Websites	Google Account, ResearchGate, LinkedIn	

Course Book

Course Description	The course covers the theory and practices of digital signal processing and the fundamental concepts guiding the design and analysis of discrete-time systems as signal processing apparatuses. Review of z-transforms, Fourier transforms, and discrete-time linear, time-invariant systems.
Course objectives	• To teach students to fundamental methods for developing and putting into practice digital signal processing systems. • To master fundamental spectrum analysis techniques. • To investigate the systems for data communication. • To instruct students on digital filter design. • To provide knowledge about the most crucial concerns in sampling and reconstruction. • They will also be able to use MATLAB to solve problems involving digital processing and create presentations.
Student's obligation	Students are expected to attend classes regularly. In case of missing an in-lab activity, a student \ should perform additional work submitted to the instructor within a week aftera class was missed. Every topic involves an assignment. Students should provide a written report on the assignment within two weeks of receiving the list of problems. The final mark will rely on the same \grading policy as for the final exam

<table>
<tr><td>Required Learning Materials</td><td colspan="5">Software: MATLAB</td></tr>
<tr><td rowspan="14">Evaluation</td><td colspan="2">Task</td><td>Weight (Marks)</td><td>Due Week</td><td>Relevant Learning Outcome</td></tr>
<tr><td colspan="2">Paper Review</td><td></td><td></td><td></td></tr>
<tr><td rowspan="6">Assignments</td><td>Homework</td><td></td><td></td><td></td></tr>
<tr><td>Class Activity</td><td></td><td></td><td></td></tr>
<tr><td>Report</td><td></td><td></td><td></td></tr>
<tr><td>Seminar</td><td></td><td></td><td></td></tr>
<tr><td>Essay</td><td></td><td></td><td></td></tr>
<tr><td>Project</td><td></td><td></td><td></td></tr>
<tr><td colspan="2">Quiz</td><td></td><td></td><td></td></tr>
<tr><td colspan="2">Lab.</td><td></td><td></td><td></td></tr>
<tr><td colspan="2">Midterm Exam</td><td></td><td></td><td></td></tr>
<tr><td colspan="2">Final Exam</td><td></td><td></td><td></td></tr>
<tr><td colspan="2">Total</td><td></td><td></td><td></td></tr>
<tr><td colspan="5"></td></tr>
<tr><td>Specific learning outcome:</td><td colspan="5">1- by the end of the course, Students will have learned how to create digital filters, modify and extract characteristics from digital signals, and design digital filters.
2- Students will be able to: program in the MATLAB scripting language.
3- Use MATLAB tools.
4- By the course’s end, students can use expert tools to entirely self-develop DSP applications.
5- Describe the steps in the D/A process using ideal samples, and why an ideal analog reconstruction filter is necessary.
6- Create block diagrams of the decimation and interpolation of signals and describe how they work.</td></tr>
<tr><td>Course References:</td><td colspan="5">1. Richard G. Lyons. Understanding Digital Signal Processing, Third Edition, Pearson Education, Inc, 2012. p.667. ISBN-13: 978-0-13-702741-5, ISBN-10: 0-13- 702741-9
2. 2. A. V. Oppenheim and R. W. Schafer. Discrete-Time</td></tr>
</table>

	Signal Processing (Prentice-Hall Signal Processing Series) 3rd Edition, 2021. p.861, ISBN-13: 978-0131988422, ISBN-10: 0131988425 3. 3. Dick Blandford, John Parr. Introduction to Digital Signal Processing. Pearson Education, Inc, 2013, ISBN: 978-0-13-139406-3 4. 4. Jonathan (Y) Stein. Digital Signal Processing: A Computer Science Perspective. John Wiley & Sons, Inc ISBN:9780471295464 5. 5. Michael Weeks. Digital Signal Processing Using MATLAB & Wavelets. Jones & Bartlett Publishers, 2011. p.492

Course topics (Theory)	Week	Learning Outcome
1- Basic Concepts of Digital Signal Processing	1	To understand the basic concept of DSP
2-Basic Digital Signal Processing Examples in Block Diagrams, Digital Filtering, Signal Frequency (Spectrum) Analysis	2	To understand signal block diagram, digital filtering, and how to analyse them
3-Digital Signal Processing Applications	3	To get knowledge about DSP applications in the real life
4-Signal Sampling and Quantization	4	To understand sampling and quantization and make difference between them
5-Sampling of Continuous Signal	5	Be able to sample a continuous signal

6-Signal Reconstruction, Anti-Aliasing Filtering, Anti-Image Filter, and Equalizer.	6	To understand Reconstruction, Anti-Aliasing Filtering, Anti-Image Filter, and Equalizer.
7- Analog-to-Digital Conversion, Digital-to-Analog Conversion, and Quantization.	7	Be able to do ADC and DAC
8- Digital Signals and Systems, Digital Signals, Common Digital Sequences, Generation of Digital Signals	8	To understand Digital Signals and Systems, Digital Signals, Common Digital Sequences, Generation of Digital Signals
9- Linear Time-Invariant, Causal Systems, Linearity, Time Invariance, Causality	9	To understand Time-Invariant, Causal Systems, Linearity, Time Invariance, Causality
10- Recursive and Non-recursive discrete-time systems, Convolution sum, and impulse response.	10	Be able to make difference between Recursive and Non-recursive discrete-time systems, Convolution sum and impulse response,
11 -Difference Equations, Format of the Difference Equation, System Representation Using Its Impulse Response	11	To know about Difference Equations, the Format of the Difference Equation, and System

		Representation Using Its Impulse Response
12- Sampling continuous signals and spectral properties of sampled signals	12	Be able to understand Sampling continuous signals and spectral properties of sampled signals
13- Amplitude Spectrum and Power Spectrum, Decimation, and Interpolation	13	To understand Amplitude Spectrum and Power Spectrum, Decimation, and Interpolation
14-To design and implement IIR and FIR digital filter	14	Be able to design and implement IIR and FIR digital filter
Practical Topics	**Week**	**Learning Outcome**
1. Introduction to DSP tools.	1	To understand the DSP laboratory tool
2. GENERATION OF BASIC SIGNALS USING MATLAB	2	Be able to implement the basic signals
3. Generate continuous time sinusoidal signal, Discrete-time cosine signal.	3	To generate CTS and DST signals and make difference between them
4. FREQUENCY RESPONSE	4	Be able to understand frequency response
5. IMPULSE RESPONSE OF A GIVEN SYSTEM	5	Be able to understand impulse response
6. Design of FIR filters of Low pass filter using	6	Be able to

MATLAB commands.		implement a low pass filter
7. Design of FIR filters of high pass filters using MATLAB commands	7	Be able to design high pass filter
8. Designing Band Pass Filter	8	Be able to design a bandpass filter
9. Designing Notch Filter	9	Be able to design notch filter

Questions Example Design

Ministry of Higher Education& Scientific Research
Erbil Polytechnic University
Erbil Technology College
Dept. of E.& C.

EPU

Academic year: 2021 – 2022
(Final Exam/ 2nd Attempt)

Class: 4th semester
Subject: Digital Signal Processing
Code: DSP403
Time: 2 hours
Date: 14/ 6 /2022

- Q1/ **(A) Define** the following transforms and **explain** how they are related to each other:

10mark

- Discrete-time Fourier transform **(DTFT)**
- Discrete Fourier transforms **(DFT)**

- **B**- **(A)** Consider the system shown in **Figure (1)** if the signal [**y (t)** = **X (t) cos ω_ct**] Determine whether it is:

10mark

- Memoryless
- Causal.

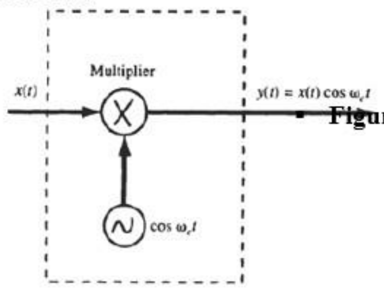

Figure (1)

- **Q2/ Choose the correct answer:** **20mark**

- One-dimensional signal is a function of
  - Multiple independent variables
  - Single independent variable
  - Multiple dependent variables
  - Single dependent variable

- The scaling of a sequence x[n] by a factor α is given by:
- $y[n] = \alpha\, [x[n]]^2$
- $y[n] = \alpha\, x[n^2]$
- $y[n] = \alpha\, x[n]$

d. $y[n] = x[n]x[-n]$

| P a g e **1**

1- Correlation is used for:
   a. Computation of average power in waveforms
   b. Climatographic
   c. Identification of binary code word in PCM systems
   d. Quantization

2- n the frequency response characteristics of FIR filter, the number of bits per coefficient should be --------------- in order to maintain the same error.
   a. Increased
   b. Constant
   c. Decreased
   d. None of the above

3- In Overlap-Add Method with linear convolution of a discrete-time signal of length L and a discrete-time signal of length M, for a length N, zero padding should be of length
   a. L, M > N
   b. L, M = N
   c. L, M < N
   d. L, M < N2

Q3/ Consider the analog signal **$x(t) = 3\cos(100\pi t)$**: **20mark**

(i) Determine the minimum required sampling rate to avoid aliasing.

(ii) Suppose that the signal is sampled at the rate **Fs= 200Hz**. What is the discrete time signal obtained after sampling?

Q4/ Let **X(t)** be a zero-mean **WSS** process with $R_{X}(T)= e^{-|T|}$. X(t) is input to an **LTI** system with:

$$|H(f)| = \begin{cases} \sqrt{1+4\pi^2 f^2} & |f| < 2 \\ 0 & \text{otherwise} \end{cases}$$

20mark

let **Y(t)** be the output:

1- Find $\mu_Y(t)$= [E Y(t)]
2- Find $R_Y(T)$
3- Find $E[Y(t)^2]$

Q5/ Design a **6-tap** FIR band reject filter with a lower cutoff frequency of **1000Hz**, an upper cutoff frequency of **2500Hz** and a sampling rate of **8000Hz** using the Hamming window method:

20mark

Good Luck

Extra notes:

External Evaluator

1- Asst. Prof. Dr. Ilham Kadim Onees
2- Lecturer. Sevan Hussein Ali