NATIONAL UNIVERSITY OF ENGINEERING
COLLEGE OF ELECTRICAL AND ELECTRONICS ENGINEERING
ELECTRONICS ENGINEERING PROGRAM

SYLLABUS - EE612 DIGITAL SIGNAL PROCESSING

I. GENERAL INFORMATION

CODE	**:**	EE612
SEMESTER	**:**	
CREDITS	**:**	3
HOURS PER WEEK	**:**	4 (Theory – Practice)
PREREQUISITES	**:**	EE610 Analysis of Signals and Systems
CONDITION	**:**	Elective
INSTRUCTOR	**:**	Armando Cajahuaringa

II. COURSE DESCRIPTION

The course trains the student in applying digital signal processing concepts, methods and techniques so that he or she is capable of using basic mathematical techniques for the digital signal analysis treatment using Fourier concepts, as well as the FIR and IIR digital filter design theory using Laplace and Z-Transform, complemented with computer-aided simulations. Engineering application problems are solved, and specialized software is used.

III. COURSE OUTCOMES

1. Learn and understand analog to digital conversion and signal reconstruction processes taking into consideration every studied method's advantages and disadvantages.
2. Analyze and understand the signal quantization process and measure its performance using subjective (perception) and objective (signal-to-noise ratio) methods.
3. Analyze and model discrete time-frequency signals using Fourier transform.
4. Mathematically describe, analyze and model time, frequency and Z domain discrete systems.
5. Understand discrete Fourier transform and its implementation through efficient computational algorithms.
6. Design and implement FIR and IIR digital filters using several structures and methods.
7. Implement several signal processing algorithms using computer programming.

IV. LEARNING UNITS

1. INTRODUCTION TO SIGNAL PROCESSING

Introduction to digital signal processing/ Global review of tools provided by modern DPS systems.

2. ANALOG TO DIGITAL SIGNAL CONVERSION

Analog to digital signal conversion / Sampling theorem/ Problem of signal binary qualification/ Quantization noise and its effect on digital signal processing/ Theory of discrete signal / Exponential and sinusoidal signals. Typical functions/ Sampling rate change/ Linear time-invariant systems/ Convolution/ Difference equations and its applications/ AR, MA, and ARMA models/ Analog circuits simulation using discrete schemes.

3. BASIC CONCEPTS OF FOURIER ANALYSIS

Fourier series (FS)/ Fourier transform (FT)/ Discrete-time Fourier transform (DTFT)/ Discrete Fourier transform (DFT)/ Inverse Fourier discrete transform (IFDT)/ Fourier transform properties/ fast Fourier transform (FFT)/ Fundamentals/ FFT algorithm and methods/ Decimation-in-time and decimation-in-frequency.

4. Z-TRANSFORM AND ITS RELATION WITH DIGITAL FILTERS

Z-Transform/ Inverse Z-Transform/ Z-transform properties/ Z-plane, Unit circle and transfer function in terms of Z/ Amplitude and phase response according to the frequency/ Digital filter concept.

5. RECURSIVE AND NONRECURSIVE DIGITAL FILTERS

Types of digital filters. Recursive and non-recursive filters/ Adaptive and non-adaptive filters/ Digital filters structures, Nonrecursive digital filters design/ Design using the windowing method/ Rectangular window/ Hann window/ Hamming window/ Kaiser window/ Research on other windows/ Frequency modification/ Low-pass to high-pass filter conversion/ Low-pass to band-pass filter conversion/ Band-pass to band-reject or band-stop filter conversion/ Digital multiband filter, Recursive digital filter design/ Filtration operation/ Bilinear or Tustin transform/ Butterworth recursive filter/ Chebyshev recursive filter/ Recursive digital filter design/ Frequency shift/ Introduction to adaptive filters.

6. UNSTEADY SIGNAL SPECTROGRAM

Discrete Fourier transform and its application in Spectrum analysis/ Application in unsteady signals. Radar, Sonar, voice and medical signals.

V. LABORATORY EXPERIENCES

Lab 1: Analog to digital system conversion.
Lab 2: Application of Fourier transform, Fast Fourier transform.
Lab 3: Z-transform and its application in digital filtering. Nonrecursive digital filters,
Lab 4: Application of recursive digital filters, Application of digital filtering and spectrum analysis.

VI. METHODOLOGY

The course is carried out in computing lab, theory and practice sessions. In theory sessions, the instructor introduces concepts, theorems and applications. In practice sessions, several problems are solved, and their solutions are analyzed. In lab sessions, Matlab simulation software is used to solve problems and analyze their solutions. In all sessions student’s active participation is encouraged.

VII. EVALUATION FORMULA

The average grade NF is calculated as follows:

$$PF = (EP + EF + (P1+P2+P3+P4)/4 + (L1+L2+L3+L4)/4) / 4$$

EP: Mid-Term Exam EF: Final Exam
P#: Quizzes L#: Labs

VIII. BIBLIOGRAPHY

1. **PROAKIS**
   Digital Signal Treatment (Spanish)
   4th Edition, Prentice Hall, Madrid, 2007

2. **DINITZ, SILVA AND NETTO**
   Digital Signal Processing
   1st Edition, Cambridge University Press, USA, 2002

3. **S.M. BOZIC**
   Digital And Kalman Filtering
   2nd Edition, Butterworth Heineman Publishing, England