

Digital Signal Processing

1. **Course code and name:** 020TNSES3 Digital Signal Processing
2. **Credits and contact hours:** 4 ECTS credits, 2x1:15 contact hours + 4x3 lab hours
3. **Instructor's or course coordinator's name:** Rabih Moawad
4. **Text book:**
 - a. **Other supplemental materials:**
Handout and lecture slides

References

Digital Signal Processing: Principles, Algorithms, and Applications, J. G. Proakis, O. G. Manolakis, Prentice-Hall, 4th Edition, 2007.
Digital Signal Processing, S.K. Mitra, McGraw Hill, 3rd Edition, 2006.

5. Specific course information

a. Catalog description:

Digital signals and systems, sampling and reconstruction, quantization, SNR, truncation-Digital Filters FIR and IIR, time and frequency response, Z transform, filter stability-Discrete Fourier Transform DFT, Fast Fourier Transform FFT, Windowing and effects on spectrum-Analog filter design (Butterworth, Tchebychev, Bessel)-FIR filter design methods: Windowing, frequency sampling-IIR filter design methods: Impulse invariance, bilinear transformation-Real-time DSP card Implementation: Matlab and Simulink

b. Prerequisites or co-requisites: 020THSES2 Signal Theory

c. Required: Elective for CCE students; required for CCE telecommunication networks option students

6. Specific goals for the course

a. At the completion of the course, students will be able to:

- Analyze FIR and IIR filters in time and frequency domains, their stability and performances.
- Determine the spectrum of a digital signal using FFT with the suitable parameters (sampling frequency, window type, window length, FFT length)
- Design Analog filters by hand to meet specific magnitude requirements.
- Design FIR and IIR filters by hand to meet specific magnitude and phase requirements.
- Design and analyze digital filters, generate and process signals, create and analyze digital systems using Matlab and Simulink.
- Implement filters and systems on real-time DSP cards.

b. KPI addressed by the course:

KPI	a1	a2	b2	b3	c2	c3	e3	k2	k3
Covered	x	x	x	x	x	x	x	x	x
Assessed	x	x	x	x	x	x	x	x	x

7. Topics and approximate lecture hours:

- Introduction to digital signals and systems, their classification and parameters (amplitude, power, energy...). Study the digital system block diagram, and evaluate each block's limitation (sampling frequency, quantization, Signal to Noise Ratio SNR, processing, truncating and signal reconstruction) (3 lectures)
- Digital Filters. Finite Impulse Response (FIR) filters. Infinite Impulse Response (IIR) filters. Time and frequency domain, Z-transform of digital filters. Causality and stability. Filter architecture. Zeros and poles effects on filter function (4 lectures)
- Discrete Fourier Transform DFT, Fast Fourier Transform FFT principle and algorithms: Decimation in time and Decimation in frequency (4 lectures)
- Using windows for spectrum computing: choosing the right window (Rectangular, Bartlett, Hann, Hamming, and Blackman) and its parameters (2 lectures)
- Analog filter design methods: Butterworth, Tchebychev and Bessel (3 lectures)
- Digital Finite Impulse Response (FIR) filter design methods: Windowing and frequency sampling (4 lectures)
- Digital Infinite Impulse Response (IIR) filter design methods: Impulse invariance and bilinear transformation (3 lectures)
- Digital filters design and spectral analysis and estimation using Matlab (fdatool, sptool). Description of the DSP card TMS320C6713 (1 lecture)
- Labs on DSP cards TMS320C6713 using Matlab and Simulink on different topics:
 - **Lab1:** Digital filter design and analysis for separating low and high frequency components in audio signals as well as trying to separate voice from music (3 lab hours)
 - **Lab2:** Echo and Reverberation modeling and analysis (especially RT60). Spectral analysis and estimation (4:30 lab hours)
 - **Lab3:** AM modulation and demodulation using different circuits and creating a graphic user interface for changing the modulation and demodulation parameters in real time (4:30 lab hours)