Digital Signal Processing

1. **Course number and name:** 020TNSES3 Digital Signal Processing

2. **Credits and contact hours:** 4 ECTS credits, 2x1:15 contact hours

3. **Name(s) of instructor(s) or course coordinator(s):** Hadi Sawaya

4. **Instructional materials:** Course handouts; lab experiments; slides; in-class problems

   References:
   - Discrete-Time Signal Processing, A. V. Oppenheim, R. W. Schaffer, Pearson, 3rd Edition, 2010
   - 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 – Structure of IIR and FIR filters - Discrete Fourier Transform DFT, Fast Fourier Transform FFT, Windowing and effects on spectrum - Analog filter design (Butterworth, Tchebychev, Bessel) - IIR filter design methods: Impulse invariance, bilinear transformation - FIR filter design: Windowing - Real-time DSP card Implementation: Matlab and Simulink

   b. **Prerequisites:** 020THSES2 Signal Theory

   c. **Required** for CCE Telecommunication Networks option students; **Selected Elective** for CCE Software Engineering option students

6. **Educational objectives for the course**
   a. **Specific outcomes of instruction:**
      - 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 to meet specific magnitude requirements.
      - Design FIR and IIR filters 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. **PI addressed by the course:**

PI	1.1	1.2	1.3	6.1	6.2	6.3	6.4
Covered	x	x	x	x	x	x	x
Assessed			x	x	x	x	x

7. Brief list of topics to be covered

- 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) (2 lectures)
- Upsampling, Downsampling (1 lecture)
- Digital Filters, Finite Impulse Response (FIR) filters, Infinite Impulse Response (IIR) filters, Time and frequency domain, Z-transform of digital filters, Causality and stability, Zeros and poles effects on filter function, Linear phase FIR filters: impulse response, positions of zeros and poles (5 lectures)
- RII and RIF filter structure (2 lectures)
- Discrete Fourier Transform (DFT), Fast Fourier Transform (FFT) principle and algorithms: Decimation in time and Decimation in frequency (3 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 (2 lectures)
- Digital Infinite Impulse Response (IIR) filter design methods: Impulse invariance and bilinear transformation (3 lectures)
- Digital Finite Impulse Response (FIR) filter design methods: Windowing and frequency sampling (2 lectures)
- 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 (3 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 (3 lab hours)