FI:PA190 Digital Signal Processing
Syllabus 2025

1. Introduction to DSP.
   Signal, spectrum. Continuous time signal, periodic signal and its representation using Fourier series, spectrum of rectangular signal. Spectral representation of non-periodic signal, Fourier transform.

2. Discrete time signal, sampling, A/D, D/A conversion.
   Continuous time signal sampling, sampling theorem, aliasing. Signal discretization in values, analog-to-digital converter. Continuous time signal reconstruction from samples, digital-to-analog converter.

3. Linear time invariant (LTI) system.
   LTI system description in time domain, unit impulse response. Convolution and its properties, calculating system response using convolution. LTI systems commutativity.

4. LTI signal description by the means of signal flow diagram.
   Signal flow basic blocks. Using signal flow for LTI system response calculation. Finite signal response (FIR) system, infinite signal response (IIR) system, and their comparison in the signal flow.

5. LTI system description in the operator domain, Z-transform.
   Z-transform definition, properties, and region of convergence. Z-transform of convolution, system transfer function. Signal flow diagram in operator domain. Mutual transition between transfer function and signal flow.

6. LTI system analysis in the frequency domain.
   Complex sinusoidal signal and its representation by a rotating phasor. LTI system frequency response, its polar representation, and properties. Real valued sinusoidal signal as an LTI system input.

7. Discrete time signal spectral analysis.
   Discrete time Fourier transform (DTFT) and its properties, inverse DTFT. Periodic discrete time signal and its spectral representation using discrete Fourier series (DFS). Incoherent sampling of a continuous time periodic signal. Discrete Fourier transform (DFT), inverse DFT. DFT and DFS comparison.

8. Fast Fourier transform (FFT).
   FFT and its main motivation. DFT matrix representation, Cooley – Tookey algorithm.

9. DFT/FFT typical usage.
   Spectrum estimation, FFT (DFT) calculated spectrum interpretation on frequency axis and on magnitude/phase axis. Sinusoidal signal DFT spectrum – coherent and incoherent sampling, leakage and its prevention using windowing. Increasing DFT resolution by the means of zero padding. Fourier filtering.

10. IIR filter design.
    Principle of filter design using an analog prototype, bilinear transform. IIR filter stability, frequency warping and pre-warping. Cascading filters.

11. FIR filter design.
    FIR filter symmetry, causality. Direct FIR filter design from the desired frequency response, shifting the response and shortening the response by a window.

12. MATLAB and Python DSP libraries.
    Basic MATLAB and Python functions for DSP and their usage. Convolution, filtering, FFT, bilinear transform, computing analog and digital filter frequency response, Filter Designer.

13. DSP implementation on microcontrollers.
    DSP on Microchip 8 bits microcontrollers ATmega. DSP support on ARM based 32 bits STM M series microcontrollers.