Aims

By the end of the course students should:

- be able to apply properties of LTI systems
- understand sampling, aliasing, convolution, filtering, spectral estimation
- be able to process signals in time and frequency domains
- be able to implement, apply and evaluate simple DSP applications in MATLAB

Content

1. Statistics. Means and standard deviation, signals, the histogram, the normal distribution, digital noise.
2. ADC and DAC. Quantization, the Sampling theorem, aliasing.
3. Linear systems. Requirements, examples, superposition, common decompositions.
4. Convolution and properies of convolution.
5. The Discrete Fourier Transform. Analysis-Discrete Fourier Transform, Synthesis-Inverse Discrete Fourier Transform.
6. Properties of the Fourier transform. Linearity, periodicity, compression and expansion.
7. The Fast Fourier Transform.
8. Filters. FIR and IIR filters.
9. Applications. Speech signal processing. Short-term Fourier analysis. Speech recognition.

Bibliography

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4. Lyons, R. G., Understanding Digital Signal Processing, Prentice Hall, Upper Saddle River, New Jersey, 2004.

5. Hesselman, Norbert, Digitális jelfeldolgozás, Műszaki Könyvkiadó, 1985.

6. Huang, X., Acero, A., Hon, H.-W., Spoken Language Processing, Prentice Hall, 2001.