Week	Lectures	References
1.	**1. Discrete-Time Signals and Systems. Summary**	
	1.1. Linear Discrete-Time Signals and Systems (LDTS). Basic Definitions Some elementary discrete-time signals, input-output description of LDTS, static versus dynamic systems, time-invariant versus time-variable systems, linear versus non-linear systems, causal versus non-causal systems, stable versus unstable of systems.	[1] pp. 35-88. [3] pp. 62-73. [3] pp. 101-123.
	1.2. Time-Domain Representation of LDTS Unit-impulse signal (unit-sample), (unit) impulse response, convolution (convolution sum). Unit-step (Heaviside step sequence), (unit) step response. Finite impulse response (FIR) LDTS, infinite impulse response (IIR) LDTS, recursive and non-recursive LDTS.	[1] pp. 35-88. [3] pp. 62-73. [3] pp. 101-123.
	1.3. Frequency-Domain Representation of Discrete Signals and LDTS Complex sinusoidal input sequence (complex-valued function), frequency response of LDTS, magnitude response of LDTS, phase response of LDTS, group delay function of LDTS, expression of frequency response in terms of impulse response.	[3] pp. 62-73. [3] pp. 101-123.
2.	*1.4. Transform-Domain Representation of Discrete Signals and LDTS* Transfer function of LDTS, poles, zeros, pole-zero plot, transfer function and stability of LDTS.	[3] pp. 62-73. [3] pp. 101-123.
	2. Introduction to Digital Filters	
	2.1. Definitions of Basic Terms. Filtering and filter, analogous and digital filters.	[2] pp. 242-245.
	2.2. Filter Specifications. Low-pass, high-pass, band-pass, band-stop and multi-band filters, all-pass filters, differentiators, Hilbert transformers.	[2] pp. 245-252. [2] pp. 263-264. [2] pp. 284-298.
	3. FIR Digital Filter. Introduction	
	3.1. Introduction On FIR digital filters.	[3] pp. 163-164. [4] pp. 75-77.
	3.2.Frequency Response of Linear Phase FIR Digital Filters Four types of linear phase FIR digital filters.	[1] pp. 331-335.
3.	**4. Linear-Phase FIR Digital Filter Design**	
	4.1. Windows (Windowing) Method Basic principles and algorithms, Gibbs phenomenon, rectangular and non-rectangular windows (Bartlett, Hann, Hamming, Hanning, Blackman, Kaiser).	[2] pp. 284-298. [3] pp. 174-189. [4] pp. 88-94.
	4.2. Frequency-Sampling Methods	
	4.2.1. Non-Uniform Frequency-Sampling Method Basic principles, algorithms and properties.	[1] pp. 331-335
4.	*4.2.2. Uniform Frequency-Sampling Method 1.* Basic principles, algorithms and properties. Specification of samples frequency response samples.	[1] pp. 331-342. [4] pp. 108-110.
	4.2.3. Uniform Frequency-Sampling Method 3. Nonrecursive FIR Filter Design by DFT Applications.* Basic principles, algorithms and properties. Selection of samples of frequency response.	[1] pp. 559-571.
	4.2.4. Uniform Frequency-Sampling Method 3. Recursive FIR Filter Design by DFT Applications.* Basic principles, algorithms and properties. Selection of samples of frequency response, recursive structures for filter implementations, stability problem solution.	[1] pp. 559-571. [4] pp. 105-107.
5.	*4.2.5. Design of Equiripple Linear-Phase FIR Digital Filter* Basic principles (Chebyshev approximation problem), algorithm (Remez exchange algorithm) and properties (equiripple magnitude response).	[1] pp. 571-580.
	4.3. Comparison of Design Methods for Linear-Phase FIR Digital Filter	[1] pp. 595-596.

		References
6.	**5. Design of IIR Digital Filters from Analogous Filters**	
	5.1. Introduction	[1] pp. 596-597.
	Analogous filter description, required properties of analogous-to-digital	[3] pp. 293-302.
	transformation, a review of basic principles of analogous-to-digital transformation.	[4] pp. 210-212.
	5.2. Bilinear Transformation Method	[1] pp. 608-612.
	Basic principles, algorithms and properties.	[3] pp. 294-296.
		[4] pp. 219-224.
	5.3. Impulse-Invariant Method (Impulse Invariant Transformation)	[1] pp. 603-608.
	Basic principles, algorithms and properties.	[3] pp. 296-300.
		[4] pp. 216-219.
7.	*5.4. The Matched Z-Transform*	[1] pp. 612.
	Basic principles, algorithms and properties.	[4] pp. 224-226.
	5.5. Method of Approximation of Derivatives (Differentials)	[1] pp. 598-603.
	Basic principles, algorithms and properties.	[3] pp. 300-302.
		[4] pp. 212-216.
	6. Frequency Transformations. A Review	[1] pp. 631-635.
8.	**7. Digital Filter Realization**	
	7.1. Direct Realizations	[2] pp. 389-396
	Direct and transposed direct realizations of IIR filters, symmetric and anti-symmetric FIR filters.	
	7.2. Parallel Realizations	[2] pp. 396-398
	7.3. Cascade Realizations	[2] pp. 399-402
	Advantages, pairing in cascade realization, coupled cascade realizations.	
9.	*7.4. FFT-Based Realizations of FIR Filters*	[2] pp. 148-151
	Linear convolution by FFT, overlap-add convolution.	
	7.5. Robust Digital Filter Structures	[3] pp. 419-490
	The state-space concept, structural passivity, wave digital filters, lattice filters.	
10.	**8. Digital Filter Implementation**	
	8.1. Overview of hardware for digital filters implementation	[3] pp. 728-732
	Digital Signal Processors (DSP), Multiply and Accumulate (MAC) unit, fixed-point, floating-point arithmetic.	[3] pp. 337-344
	Field Programmable Arrays (FPGA), principle of distributed arithmetic.	
11.	*8.2. The Finite Word Length Problems*	[2] pp. 433-437
	Problems of zero-input limit cycles in digital filters, dead zones, quantization noise.	
	8.3. Coefficient Quantization in Digital Filters	
	Quantization effects on poles and zeros, quantization effects on the frequency response.	[2] pp. 412-419
12.	*8.4. Scaling in Fixed-Point Arithmetic*	[2] pp. 419-426
	Time-domain scaling, Frequency-domain scaling, scaling in parallel and cascade realizations.	

References

[1] Proakis, J. G. - Manolakis, D. G.: Introduction to Digital Signal Processing. Macmillan Publishing Company, New York. Collier Macmillan Publishers, London. 1988. (pp. 35-88, 331-342, 559-580, 595-612, 631-635).

[2] Porat, B.: A Course I Digital Signal Processing. John Wiley&Sons, Inc. 1996. (pp. 242-252, 263-264, 284-298).

[3] Mitra, S. K. - Kaiser, F. J.: Handbook for Digital Signal Processing. John Wiley&Sons, Inc.1993. (pp. 62-73, 101-123, 163-164, 174-189, 293-302).

[4] Rabiner, L. R.-Gold, B.: Theory and Application of Digital Signal Processing. Prentice-Hall, Englewood Cliffs, NJ, 1975. (pp. 75-77, 88-94, 105-110, 210-226).