

MINISTRY OF SCIENCE AND HIGHER EDUCATION OF THE RUSSIAN FEDERATION
Federal State Autonomous Educational Institution for Higher Education
«North-Caucasus Federal University»

APPROVED BY

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ASSESSMENT TOOLS FOR THE DISCIPLINE (MODULE)
«Digital Signal Processing (DSP)»

Field of study	<u>13.04.02 Electric Power Engineering and Electrical Engineering</u>
Specialization (Profile)	<u>Intelligent Power Supply Systems</u>
Year of commencement of studies	<u>2026</u>
Mode of study	<u>full-time</u>
Implemented in semester(s)	<u>2</u>

Stavropol 2026

Foreword

1. Purpose: This assessment tools (AT) document is intended to assess the level of competency formation during ongoing progress control and intermediate assessment of students in the discipline "Digital Signal Processing (DSP)".

2. The AT is an appendix to the syllabus of the discipline "Digital Signal Processing (DSP)".

3. Developer: Dmitry A. Kostyukov, Associate Professor of the Department of Automated Electric Power Systems and Power Supply.

4. The AT has undergone expert review.

Members of the expert group:

Chairperson: Yu. G. Kononov, Doctor of Technical Sciences, Professor, Head of the Department of Automated Electric Power Systems and Power Supply.

Committee members:

D.A. Kostyukov, Candidate of Technical Sciences, Associate Professor, Department of Automated Electric Power Systems and Power Supply.

A.V. Petrov, Candidate of Technical Sciences, Associate Professor, Department of Automated Electric Power Systems and Power Supply.

Representative of the employer organization:

A.S. Filippov, Head of the Leninsky District Department of the Stavropol Interdistrict Branch of the PJSC «Stavropolenergosbyt».

Expert opinion: The AT is recommended for assessing the level of competency formation during ongoing progress control and intermediate assessment of students in the discipline "Digital Signal Processing (DSP)".

5. The validity period of the AT is determined by the period of implementation of the educational program.

1. Description of indicators and criteria for assessment at various stages of their formation, description of grading scales

Competency(ies), indicator(s)	Levels of competency formation			
	Minimum level not achieved (Unsatisfactory) – 2 points	Minimum level (Satisfactory) – 3 points	Medium level (Good) – 4 points	High level (Excellent) – 5 points
PC-3. Capable of managing the power system operating mode of an electrical power supply system using digital technologies				
Learning outcomes for the discipline (module): Indicator: ID-3 PC-3 Applies knowledge of digital technologies in the processes of generation, transmission, and distribution of electrical energy	Does not know the basic concepts of digital signal processing, discrete-time signals and systems, sampling, convolution, correlation, spectral analysis, z-transform, and digital filtering. Cannot interpret spectra, filtering results, or basic signal-processing tasks related to measurement, diagnostics, control, and intelligent power supply systems.	Identifies the main concepts and methods of DSP: discrete-time signals and systems, sampling theorem, aliasing, signal reconstruction, convolution and correlation, DFT/FFT, z-transform, FIR and IIR filters. Performs simple calculations and basic signal-processing tasks using reference materials, with minor errors in interpretation or implementation.	Correctly applies DSP methods for time-domain and frequency-domain analysis, sampling and reconstruction, DFT/FFT-based spectral analysis, z-transform, transfer functions, stability analysis, and FIR/IIR filter design. Interprets signal spectra, filtering results, and quantization or finite-precision effects for engineering measurement, diagnostics, monitoring, and control tasks.	Independently justifies, selects, and implements DSP algorithms for engineering applications in electric power engineering and intelligent power supply systems. Designs and evaluates FIR and IIR digital filters, applies adaptive filtering and multirate signal processing methods, uses software tools for real-time DSP tasks, and critically assesses accuracy, stability, computational complexity, latency, and practical limitations of digital implementation.

ASSESSMENT TOOLS FOR CHECKING THE LEVEL OF COMPETENCY FORMATION

Task No.	Correct Answer	Question Content	Competency
1.	3	Digital Signal Processing (DSP) is primarily concerned with: 1) mechanical transmission of power by rotating shafts 2) manual switching of electrical circuits only 3) analysis, transformation, filtering, and interpretation of digital signals using algorithms 4) design of analog measuring instruments without computation	PC-3
2.	2	A discrete-time signal is best described as: 1) a signal defined for every value of continuous time 2) a sequence of values defined at discrete time instants 3) a signal that cannot be stored in computer memory 4) a signal that always has only two amplitude levels	PC-3
3.	1	According to the sampling theorem, a band-limited signal can be reconstructed without aliasing if the sampling frequency is: 1) greater than twice the highest signal frequency 2) equal to the lowest signal frequency 3) lower than the highest signal frequency 4) independent of the signal spectrum	PC-3

4.	4	Aliasing occurs when: 1) the sampling frequency is much higher than required 2) a signal is represented in the z-domain 3) a digital filter has a finite impulse response 4) different frequency components become indistinguishable after insufficient sampling	PC-3
5.	2	Quantization in analog-to-digital conversion is associated with: 1) changing the sampling interval only 2) representing signal amplitudes by a finite number of levels 3) increasing the sampling frequency to infinity 4) eliminating all noise from the signal	PC-3
6.	3	Linear convolution of two finite-length sequences is used to determine: 1) only the phase spectrum of a signal 2) the region of convergence of the z-transform 3) the output of an LTI system from its input sequence and impulse response 4) the sampling frequency of an ADC	PC-3

7.	1	Correlation analysis is mainly used to: 1) measure similarity between signals or between shifted versions of the same signal 2) replace all frequency-domain methods 3) remove the need for sampling 4) guarantee the stability of any IIR filter	PC-3
8.	2	A difference equation in DSP represents: 1) a graphical spectrum display only 2) an input-output relation involving current and delayed signal samples 3) the physical length of a transmission line 4) the noise temperature of a measuring sensor	PC-3
9.	1	The Discrete Fourier Transform (DFT) converts: 1) a finite sequence of time-domain samples into a finite sequence of frequency-domain components 2) a continuous mechanical process into a thermal model 3) a digital signal into an analog signal without reconstruction 4) a z-transform directly into a sampling theorem	PC-3

10.	3	Which property of the DFT is directly connected with convolution in the frequency domain? 1) Quantization property 2) Causality property 3) Circular convolution property 4) Mechanical inertia property	PC-3
11.	4	For an N-point DFT with sampling frequency f_s, the frequency resolution is: 1) N / f_s 2) $f_s \times N$ 3) $2\pi \times f_s$ 4) f_s / N	PC-3
12.	2	The main advantage of the Fast Fourier Transform (FFT) compared with direct DFT computation is: 1) elimination of sampling requirements 2) reduction of computational complexity 3) removal of all spectral leakage 4) conversion of recursive filters into analog filters	PC-3
13.	1	Spectral leakage appears mainly because: 1) a finite observation interval cuts the signal and may not contain an integer number of periods 2) the z-transform has no region of convergence 3) all FIR filters are unstable 4) sampling theorem is not applicable to digital systems	PC-3
14.	4	Window functions in spectral analysis are used to: 1) increase quantization error deliberately 2) make all signals periodic in continuous time 3) replace the FFT algorithm 4) reduce leakage effects by controlling sidelobes and main-lobe width	PC-3

15.	2	In the z-transform, the region of convergence is important because it helps determine: 1) the physical size of the processor 2) causality and stability properties of a discrete-time system 3) the color of the frequency spectrum plot 4) the number of bits in an ADC only	PC-3
16.	3	For a causal discrete-time LTI system, BIBO stability requires that all poles of the transfer function lie: 1) outside the unit circle 2) exactly on the unit circle 3) inside the unit circle 4) only on the real axis	PC-3
17.	1	The frequency response of a digital system can be obtained from its transfer function by evaluating it: 1) on the unit circle in the z-plane 2) only at $z = 0$ 3) only for negative sampling frequencies 4) at the Nyquist frequency regardless of the system	PC-3
18.	2	A Finite Impulse Response (FIR) filter is characterized by: 1) an infinite impulse response caused by feedback 2) a finite-duration impulse response and usually non-recursive implementation 3) unavoidable instability in all realizations 4) absence of coefficients in its mathematical model	PC-3
19.	3	Linear phase in an FIR filter is commonly achieved when the impulse response is: 1) random and time-varying 2) always complex-valued only 3) symmetric or antisymmetric with respect to its midpoint 4) infinitely long and recursive	PC-3

20.	4	The window method for FIR filter design is based on: 1) placing all poles outside the unit circle 2) using only analog prototype filters 3) eliminating the impulse response completely 4) truncating an ideal impulse response by multiplying it by a selected window function	PC-3
21.	1	The Parks-McClellan algorithm is mainly used for designing: 1) equiripple FIR filters according to specified passband and stopband requirements 2) analog transformers for power systems 3) uncontrolled rectifier circuits 4) random noise generators without frequency specifications	PC-3
22.	2	An Infinite Impulse Response (IIR) filter differs from an FIR filter because it: 1) cannot be described by a transfer function 2) uses feedback and may have an infinite-duration impulse response 3) always has exactly linear phase 4) does not require coefficient values	PC-3
23.	3	Cascade realization of high-order IIR filters is often preferred because it: 1) removes the need for sampling 2) converts the filter into a purely analog circuit 3) improves numerical robustness by using lower-order sections 4) guarantees zero quantization error	PC-3
24.	4	Coefficient quantization in digital filters may lead to: 1) a complete absence of round-off effects 2) automatic improvement of all stopband characteristics 3) elimination of poles and zeros from the model 4) changes in pole-zero locations, frequency response, and possibly stability	PC-3
25.	1	The bilinear transformation in IIR filter design maps the analog s-plane to the digital z-plane and: 1) introduces frequency warping that may require prewarping 2) always causes time-domain aliasing 3) is used only for FIR filters 4) eliminates the need for analog prototypes	PC-3

26.	2	The impulse invariance method for IIR filter design may cause problems because: 1) it cannot preserve any time-domain behavior 2) frequency-domain aliasing may occur after sampling the analog impulse response 3) it is applicable only to non-recursive FIR filters 4) it always places all poles outside the unit circle	PC-3
27.	3	The Least Mean Squares (LMS) adaptive filtering algorithm updates filter coefficients in order to: 1) increase sampling frequency after every iteration 2) force all coefficients to zero permanently 3) reduce the mean-square error between the desired signal and filter output 4) replace convolution with analog modulation	PC-3
28.	4	In adaptive filtering, an excessively large LMS step size can lead to: 1) guaranteed faster convergence without limitations 2) complete removal of all measurement noise 3) automatic reduction of filter order 4) instability or divergence of the adaptation process	PC-3
29.	1	Decimation in multirate signal processing requires preliminary low-pass filtering mainly to: 1) prevent aliasing before reducing the sampling rate 2) increase the signal bandwidth beyond the Nyquist limit 3) remove all filter coefficients 4) transform an FIR filter into an IIR filter	PC-3
30.	2	In real-time DSP for intelligent power supply systems, latency constraints are important because: 1) they define the color of the measurement interface 2) signal processing results must be produced quickly enough for monitoring, diagnostics, or control actions 3) they eliminate the need for feature extraction 4) they make sampling theorem unnecessary	PC-3

2. Description of the grading scale

Within the framework of the rating system, student performance in each discipline is assessed during ongoing progress control and intermediate assessment. The rating system for assessing student knowledge is based on the use of a set of control measures (checkpoints) for testing the material covered, optimally located throughout the entire time interval of studying the discipline. The principles of the rating system for assessing student knowledge are based on the provisions described in the "Regulations on the organization of the educational process based on the rating system for assessing student knowledge" at North Caucasus Federal University.

The rating assessment system is not provided for students studying in master's degree programs, or for students studying in bachelor's degree programs by correspondence or part-time (evening) forms of study.

3. Criteria for assessing competencies

The grade "Excellent" is awarded to a student who has deeply and firmly mastered the program material, presents it exhaustively, consistently, clearly, and logically coherently, is able to closely link theory with practice, handles tasks, questions, and other types of knowledge application with ease, and does not struggle to answer when assignments are modified.

The grade "Good" is awarded to a student who has a solid command of the material, presents it competently and substantively without making significant inaccuracies in answering questions, correctly applies theoretical principles to practical questions and tasks, and possesses the necessary skills and techniques for their execution.

The grade "Satisfactory" is awarded to a student who has knowledge of only the basic material but has not mastered its details, makes inaccuracies, insufficiently correct formulations, and violates logical sequence in presenting the program material.

The grade "Unsatisfactory" is awarded to a student who does not know a significant part of the program material and makes substantial errors.