Laboratory of Digital Control Techniques

Exercise 2

Designing and testing the properties of filters with finite impulse response

I. Aims

1. Learning the methods of designing digital filters with finite impulse response.
2. Synthesis and implementation of a digital filter with given properties.
3. Analysis of the properties of the designed filter.

II. Framework

1. Determine the transmittance of the digital non-recursive filter in the following form:

$$G(z) = \sum_{k=0}^{N-1} a(k) z^{-k}$$

– in the design process use a method based on the discretization of predetermined frequency characteristic of a digital filter (use the inverse discrete Fourier transform)

- design a low-pass filter
- cut-off frequency of the designed digital filter: fgc = (200 + ('group number') * 50) Hz.
- sampling frequency: fp = (900 + ('group number') * 100) Hz.
- assume an ideal, rectangular shape of the amplitude characteristic (in the range from 0 to fp).
- assume a constant value of the group delay (linear phase characteristic).
- consider two variants of the length of the windows.
- after determining the discretized frequency characteristic of the designed filter, use the Matlab environment (ifft function) to determine the filter window coefficients.

2. Using the Matlab / Simulink environment:

- examine the time responses of the filter for different input signals (perform spectral analysis of the signals before and after filtration). When selecting the frequency of the input signals, consider the shape of the obtained frequency characteristic of the filter obtained.

3. Apply the selected smoothing windows (Hamming, Hanning, Blackman) to the filter obtained in point 1 and perform the tests as in point 2.

4. Realize the filtration by implementing the difference equation of the obtained filter in the Matlab environment.

5. Synthesize the SOI filter as described in step 1, but with a non-rectangular shape of the amplitude characteristic.

6. Based on the filter with the window obtained in point 3. determine the coefficients of the high-pass or the band-pass filters. Use the following formulas for transformation.

Conversion to high-pass filter:

$$f_{gGP} = f_p / 2 - f_{gDP}\,; \quad h_{GP}(m) = (-1)^m \cdot h_{DP}(m)\,.$$

Conversion to band-pass filter:

$$f_{d1SP} = f_0 - f_{gDP}\,; \qquad f_{g2SP} = f_0 + f_{gDP}\,; \qquad h_{SP}(m) = 2 * \cos(2 \cdot \pi \cdot (m-1) \cdot f_0 / f_p) \cdot h_{DP}(m)\,;$$

where:
m - filter window coefficient number, m = 1,2,…, N;
f_{gGP} *- cut-off frequency of the high-pass filter,*
f_{gDP} *- cut-off frequency of the low-pass filter,*
$h_{GP}(m)$ *- high-pass filter window coefficients,*
$h_{DP}(m)$ *- low-pass filter window coefficients,*
f_{d1SP} *- the lower limit frequency of the band-pass filter,*
f_{g2SP} *- the upper limit frequency of the band-pass filter,*
f_0 *- center frequency of the bandpass filter,*
f_p *- sampling frequency,*