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АНАЛИЗ ПРИМЕНИМОСТИ СИСТЕМЫ MATLAB ДЛЯ СИНТЕЗА УПРАВЛЯЕМЫХ ЦИФРОВЫХ РЕКУРСИВНЫХ БИХ-ФИЛЬТРОВ БАТТЕРВОРТА

В ряде приложений цифровой обработки сигналов применяются управляемые цифровые рекурсивные БИХ-фильтры. Под словом «управляемые» имеются в виду фильтры, коэффициенты структуры которых явно зависят от частоты среза или граничных частот. Управляемые БИХ-фильтры могут быть синтезированы с помощью различных средств для расчета традиционных, неуправляемых БИХ-фильтров. В статье был рассмотрен синтез неуправляемых БИХ-фильтров и проанализирована целесообразность представления результатов синтеза для построения управляемых БИХ-фильтров. Были описаны и объяснены методы проектирования на основе MATLAB (2021a) и фундаментальные концепции цифровых фильтров БИХ-Баттерворта. Составной сигнал был обработан анализируемым фильтром, чтобы определить, соответствует ли он критериям прохождения фильтрации. Для проверки рассчитанных фильтров использовался прототип Симулинк, а также инструмент FDA инструментария обработки сигналов. На основании полученных результатов делается вывод о применимости системы MATLAB для синтеза управляемых цифровых рекурсивных БИХ-фильтров Баттерворта. Проанализированная техника была более эффективной, быстрой, уменьшила количество задач и обнаружила, что результаты удовлетворительны.

Цифровой БИХ-фильтр; Баттерворт; управляемый; проектирование; синтез; Инструмент FDA; Simulink; Программное обеспечение MAT LAB.

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ANALYSIS OF MATLAB SYSTEM APPLICABILITY FOR SYNTHESIS OF CONTROLLED BUTTERWORTH DIGITAL RECURSIVE IIR FILTERS

A number of digital signal processing applications use controlled IIR digital recursive filters. The word " controlled " refers to filters whose structure ratios clearly depend on the cut rate or boundary frequencies. Controlled IIR filters can be synthesized using a variety of tools to compute traditional, uncontrolled IIR filters. The article dealt with the synthesis uncontrolled IIR filters and analyzed the suitability of the presentation of the results of synthesis for the construction of controlled IIR filters. The design techniques based on MATLAB (2021a) and the fundamental concepts of IIR-Butterworth digital filters were described and explained. The composited signal was processed by the analyzed filter to find whether it met the filtering progress criteria. To check the calculated filters, the Simulink prototype was used, as well as FDA tool of signal processing toolbox. Based on the results obtained, a conclusion was made about the applicability of the MAT LAB (2021a) system for the synthesis of controlled digital recursive IIR-Butterworth filters. The analyzed technique was more efficient, faster, decreased the tasks and found the results are satisfying.

Digital IIR filter; Butterworth; controlled; design; synthesis; FDA tool; Simulink; MATLAB software.

I. Introduction. Filters are most important and influential elements in the framework of signals and systems, because they are practically presented to filter out the desired signal per the prerequisite to be managed in various areas of interest, varying from speech via image to video and audio processing. i.e., to eliminate or improve particular frequency components in the signal [1]. Tunable digital filters are used in a large number of technical applications. These are, first of all, various adaptive systems: adaptive compression, adaptive sampling, adaptive filters, as well as optimal signal reception against the background of interference, and many others [2]. FIR (Finite Impulse Response) and

IIR (Infinite Impulse Response) filters are the two most common types of digital filters seen in a variety of applications. Signal filtering in the time domain is performed by digital filters. While Spectrum Analyzer is another form of system represent signals in the frequency domain [5, 6].

IIR filter handles specified properties like widths of both passband and stopband, as well as maximum allowable ripples at both passband and stopband [3, 4]. These properties can be used to create a preferred IIR filter design [5]. In comparison to a similar FIR filter, IIR can achieve a specified filtering characteristic with not much memory and computations. For the same filter, IIR needs a lower order than FIR. It is capable of obtaining all of the desired specifications at a low operating complexity. As a result, the IIR filter is the best option for signal filtering [7]. The powerful computational resources of MATLAB allow the realization and simulation testing of digital filters to be completed quickly and efficiently. Simulink, as one of the MATLAB signal processing boxes, has useful features and a user-friendly interface, and the modular design of Simulink and MATLAB allows users to create simulation models and monitor simulation results rather easily and efficiently [8].

II. The basic concept OF IIR digital filter. An impulse response of an IIR digital filter contains an infinite amount of non-zero specimens. The existence of feedback, which can produce instability during the operation of the processed filters besides cause them to carry out nonlinear phase features, is the main explanation for their infinite response characteristic. This will initiate oscillations appearing in the response of the IIR filter, resulting in erroneous output that may be difficult to notice and adjust. However, the fundamental advantage of IIR filters is their capacity to complete tasks with less computational power and memory. [9]. The output and input sequences of an IIR filter are defined as the ratio of two polynomials in most cases, is given by the equation (1):

$$y(n) = \sum_{i=0}^{M} b_i\, x(n-i) - \sum_{k=1}^{N} a_k\, y\,(n-k). \tag{1}$$

Where: x(n) and y(n) are the input signal & output signal of the recursive filter, $\{a_1, a_2, \ldots, a_N\}$ are coefficients values of feedback, {b0 ,b1, …., bM} are coefficients values of feed-forward and N as well as M are the numbers of both poles and zeros respectively, (usually N larger than M), they are determined the order of the IIR filter.

In essence, the above equation explain that the current output is a weighted sum total of previous inputs and outputs. The transfer function, or difference equation, of a digital filter is its defining feature. The transfer function can be mathematically analyzed to determine how it will respond to any input. As a result, creating specifications suitable to the problem (for example, a second-order low pass filter with a particular cut-off frequency) and then generating a transfer feature that satisfies the requirements is the process of developing a filter [10]. Through a transfer function for a linear, time invariant, the digital filter in the z-domain can be indicated. Each non-unit coefficient in a recursive system has a causal interpretation, it has a transfer function as in (2):

$$H(z) = \frac{Y(z)}{X(z)} = \frac{b_0 + b_1 z^{-1} + \cdots + b_M z^{-M}}{1 + a_1 z^{-1} - a_2 z^{-2} - \cdots - a_N z^{-N}}. \tag{2}$$

The digital filter is generally performed in nonlinear process by using the z-transform to change the transfer function into a linear difference equation with a constant coefficient. A block diagram (workable diagram) of the IIR filter specification can be built directly from equation (2) and shown in Figure (1); this is referred to as a direct form realizing.

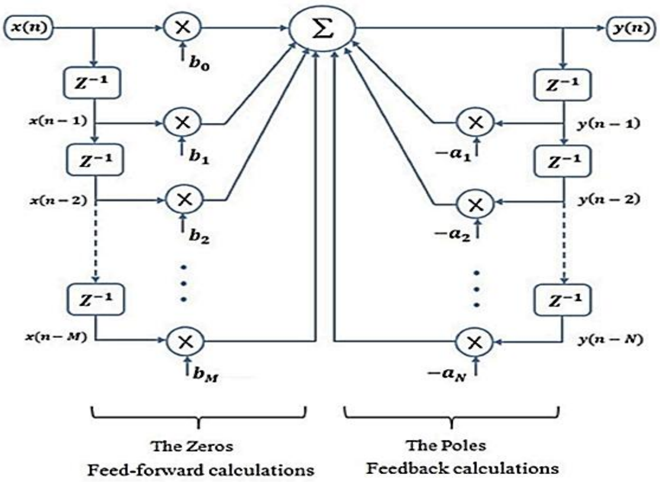


Fig. 1. IIR filter block diagram

A. Advantages of IIR filter over FIR filter. Both IIR and FIR filters can be developed using a variety of techniques. As a result, one can wonder which filter, IIR or FIR, is best for a given requirement and which technique should be used to design an acceptable filter. Conversely, some significant distinctions between these two filters are feasible. The IIR filter has several benefits over the FIR filter, which are mentioned below:

1. IIR uses less memory and calculations to attain the required filtering characteristic than a comparable FIR filter.
2. In the stopband of IIR filter it contains a smaller number of side lobes.
3. IIR filters have a lesser or no time delay than FIR filters.
4. IIR needs a lower order than FIR when implementing the same filter.
5. IIR can meet all of the requirements at a low operating complexity.
6. When sharp cutoff filters and fast response are needed, IIR filters produce fewer coefficients than FIR filters.

B. The basic steps of digital filter design. There are five steps to designing a digital filter [11]:

1) Filter specifications: they are determined based on real needs.

2) Coefficient calculation: select one of approximation method and calculate the value of a_N and b_M.

3) Realization: including a finite accuracy process to convert the transfer function into an appropriate filter form.

4) Evaluation of errors: due to the calculation of coefficients and using a finite quantity of bits.

5) Implementation: which entails organizing the software and/or hardware.

C. Butterworth filters. Have no ripples, gain decreases steadily over the pass and transition bands. The gain gradually declines to $1-\sqrt{1/2}$ (-3 dB) within the passband. Outside the passband, it diminishes by a factor of 2N per octave (N 20 dB/decade) asymptotically [12]. With increasing N, the phase response of Butterworth filter develops increasingly nonlinear. The cutoff frequency and the number of poles is the only two mathematical factors that define this filter [13].

Butterworth approximation is a common tool for developing analog filters [14]. The magnitude squared response of low pass Butterworth filter is explained by means of equation (3):

$$H(jw) = \frac{1}{1+(\frac{w}{wc})^{2N}}. \tag{3}$$

Where selectivity of filters is provided with (4),

$$F_s = \frac{N}{2\sqrt{2w_c}}. \tag{4}$$

As well as their attenuation by (5),

$$A = 10\log(1 + (\frac{w}{w_c})^{2N}). \tag{5}$$

In addition, In most cases, the frequency response is maximal flatness in the passband with 0% rolls-off in the stopband [15]. Comparing with Chebyshev type I and/or Elliptic filters, the Butterworth filters have a relatively slow roll-off around the cutoff frequency without ripple, requiring a higher order to achieve a specified stopband requirement as well as a further linear phase response in the passband [16, 17]. As noticed in figure (2), where each of these filters is in the fifth order.

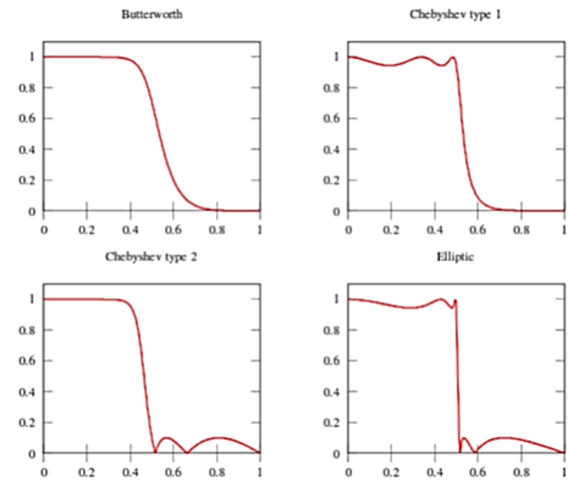


Fig. 2. The phase response of different types of IIR filters

D. Design method and parameter confirmation. IIR filters are designed to determine coefficients of the transfer sequence or attenuation characteristic that satisfy specifications. The approximation Butterworth digital filter model is used to implement the design process. A filter is used to eliminate high-frequency components from a signal.

Based on the current Nyquist-Shannon sampling formula, the sampling frequency F_s ought to be higher than or equal to the twice of the highest frequency in the entire signal frequencies.

Low-pass filter structure requirement was proposed IIR design method, minimum, order select, frequency in units of hertz, where: sampling frequency of 200; stopband and passband frequencies are 40 and 30 respectively. Where stopband attenuation of 20 dB and passband ripple of 1dB. The menu options "Analysis" views the amplitude as well as frequency responses, zero-pole assignment, filter coefficient, and various filter features. After the design is finished, the result will be saved with a .fda extension.

Initially, a Butterworth lowpass filter was chosen to filter the signal to satisfy the operational needs of the input signal. Signal processing toolbox comprises functions to build all of these classic IIR filters in lowpass, high pass, band pass, and band stop setups across both the analog and digital realms (except Bessel, which is only supported in the analog domain) [19].

The filter design abilities programmed directly in MATLAB 2021a by the use of the Filter Design and Analysis Tool (FDAT). It has several powerful features can be listed as following:

1. It is a user interface for modeling, quantifying, and evaluating filters.
2. It had a very straightforward and adaptable process.
3. It includes methods for considering the behavior and characteristics of filters, such as phase and magnitude responses and pole-zero plot.
4. Additional features from other MathWorks products are universally integrated [18].
5. It can export filters as single input or output block to a Simulink window [19].

E. Interface design based on FDA tool. Utilizing MATLAB, the way of organizing a filter is fairly self-helpful [20]. FDAT method was used to design IIR filter. After typing fdatool or filterDesigner in the command window and command is run, the graphical user interface (GUI) would open in the default design mode [21].

The designer needs to select design method of IIR filter (Butterworth, Chebyshev Types I and II, elliptic or other), response type and other essential performing index. Also, toward decide value of frequency and magnitude specifications. When all requirements are satisfied, select the design filter [22]. as shown in figure (3).

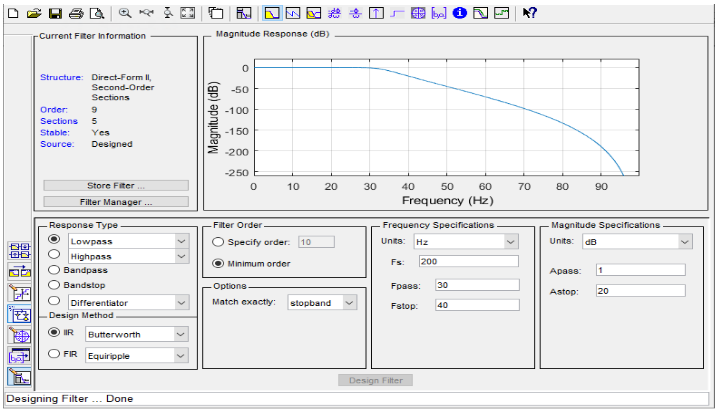


Fig. 3. Filter design and analysis tool

Also, the MATLAB method for IIR filter design is a two-command process; first, to determine the order and critical frequencies, second to compute the filter coefficients. For a Butterworth filter are achieved in following code to ensure that it met the requirements.

The program executed with "evaluation selection" instruction or F9 in the MATLAB 2021a command window. And all variables' values of the designed filter were listed in Workspace window.

```
fs = 200;
Wp=2*30/fs;
Ws=2*40/fs;
Apass = 1;
Astop = 40;
[N,Wn]=buttord(Wp,Ws,1,20);
[B,A]=butter(N,Wn);
fvtool(B,A)
```

The necessary parameters, including the passband frequency, the stopband frequency, and the passband and stopband ripples, are chosen in such systems to obtain the best possible results in the magnitude response, phase response, impulse response, pole-zero plotting, and coefficient determination.

The stability of the realized structure is demonstrated by the pole-zero plotting, which forces the poles of a stable IIR filter within the unit circle. On the other hand, the instability of the filter in the case of errors during filtering is due to the fact that the recursive filters are based on unstable links - integrators. Therefore, it is sometimes said that the integrator is on the verge of stability [23]. From figure.4 to figure.6, the performance graphs were shown.

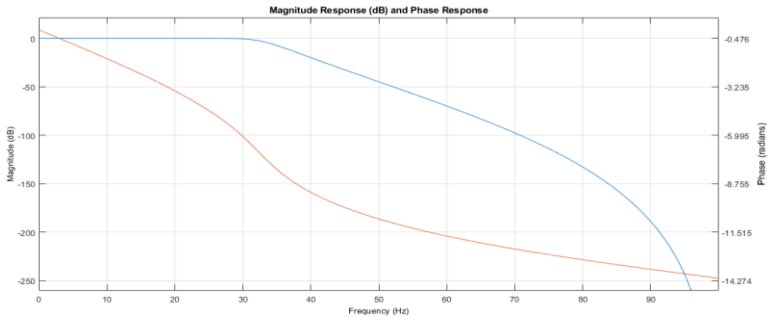


Fig. 4. Butterworth filter lowpass filter Magnitude and phase responses

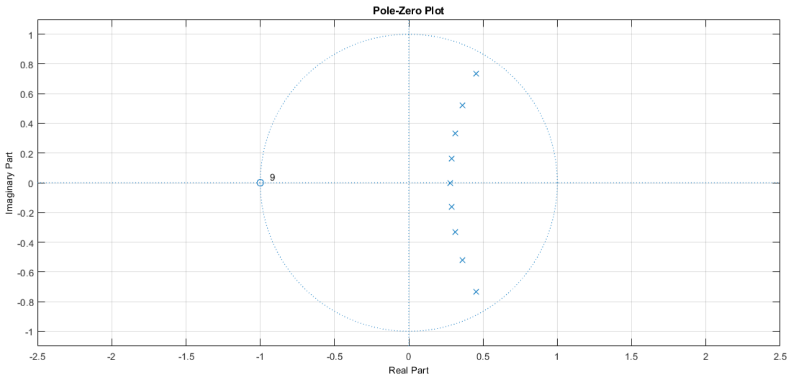


Fig. 5. Butterworth filter low pass filter pole-zero plot

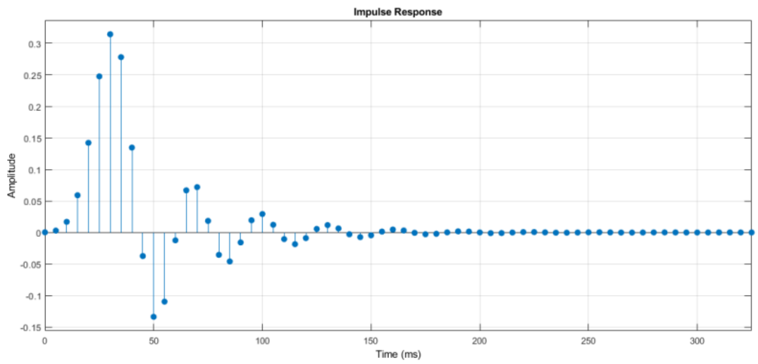


Fig. 6/ Butterworth filter low pass filter Impulse response

III. Simulation based on simulink model. Simulink is another significant MATLAB toolbox, with the primary goal of pattern identification, emulation, and evaluation. Which may include pre-simulation and review of the device before it was built. The Simulink library bowser dialog box was opened after Simulink syntax was typed in the command screen.

In the Simulink environment, the paper low pass digital filter “Butterworth” is the output of exporting it as of GUI to Simulink. Also, three block parameters: sine wave, vector scope in addition to add part of sum process library scope, which can be saved as new simulation file (.slx). Without having to compile code, a prototype of a system block illustration can be quickly created [24]. As shown in figure .7, each element was linked on the way to form a Simulink block diagram of the filtering process.

We constructed following: a group of signals units were set with frequency value (in hertz also) the same as: 20, 45 and 60 sinusoidal signals, specifications for the digital filter interface element as: F_s=200; f_{pass}=30; f_{stop}=40, besides sample time set = 1/200. Add module with specify character vector containing +++ for each input port. The time domain waveforms of the filtering effect were produced when the completed simulation block in figure .7, was performed with the run choice.

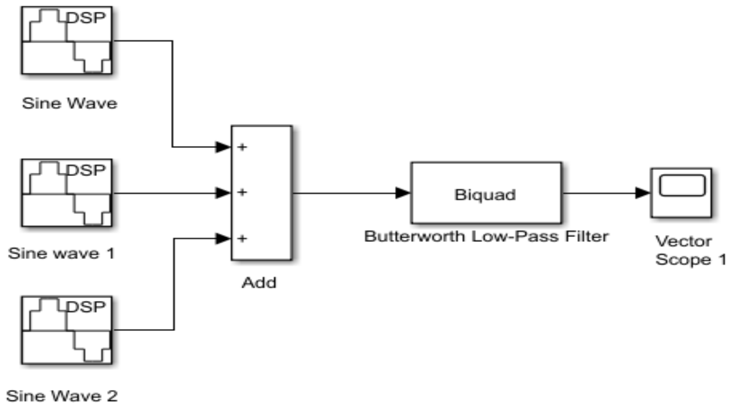
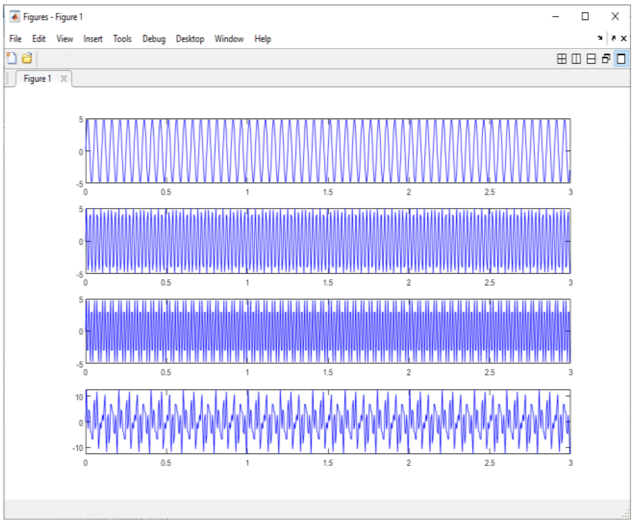


Fig. 7. Simulink block diagram

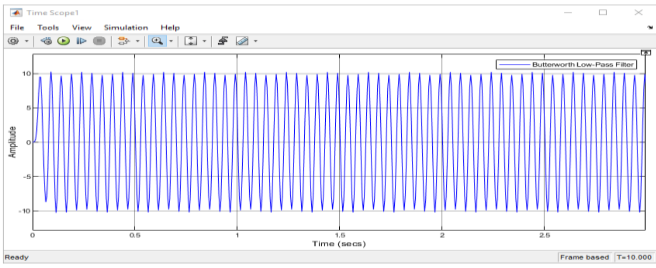
The input signal was comprised with 3 types of frequency components in the time domain waveform and simulated with subsequent MATLAB instructions as shown in figure.8 (a and b) [25][26]. After filtering, the two higher frequencies were damped. Whereas f_1 frequency sine wave signals pass through the low pass IIR filter, f_2 and f_3 frequency sine wave signals are greatly suppressed. Before filtering, the waveform of input signal (x_k) was disoriented, but it was ordered after filtering realization [8]. The simulation results demonstrate that, the different evaluation indexes of the IIR filter design fulfilled the required specifications, and the design process was intuitive and straightforward.

```
f1=20;f2=45;f3=60;
Fs=200;
Ts=1/Fs;
t=0:Ts:2-Ts;
Xk1=5*sin(2*pi*f1*t);
Xk2=5*sin(2*pi*f2*t);
```

```
Xk3=5*sin(2*pi*f3*t);
Xkn= Xk1+ Xk2+ Xk3;
subplot(4,1,1);
plot(t,Xk1,'b');
subplot(4,1,2);
plot(t,Xk2,'b');
subplot(4,1,3);
plot(t,Xk3,'b');
subplot(4,1,4);
plot (t,Xkn,'b');
```



a – Original discrete three input signals and add signal waveforms



b –The discrete wave after filterig. (vector scope 1)

Fig. 8. Time domain waveforms of the utilized signal before (a) and after (b) filtering

Conclusion. The IIR Butterworth filter was successfully analyzed and implemented with the MAT LAB (2021a) environment and the Simulink model of a filter. To design controlled filters, the system (transfer) function of an ordinary (uncontrolled) filter is required, which is presented as a cascading connection of second-order links. If the filter order as a whole was odd, then plus one more link of the first order. The digital filter was easily and simply simulated, reducing both the programming complexity and the amount of work required. In practice, it was very reliable. Concurrently, using MATLAB tools, filter analysis can be done easily to meet requirements and model whole systems more accurately. The comparison of the results of filter synthesis with the method of designing controlled digital filters [2] allows us to conclude that, the MATLAB system is applicable for designing digital controlled IIR Butterworth filters.

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СРАВНИТЕЛЬНЫЙ АНАЛИЗ ДВУХ СПОСОБОВ ФИЛЬТРАЦИИ ДЛЯ УСТРАНЕНИЯ ШУМА В ИЗОБРАЖЕНИИ РАЗНОЙ СТЕПЕНИ ЗАШУМЛЕННОСТИ

В современной технике фото- и видеосъемки любое изображение в процессе его формирования искажается под действием различных видов шумов. Существуют различные виды шумов, но на практике наиболее часто встречаются модели импульсного и гауссовского шума. Ослабление действия шумов достигается путём фильтрации. На данный момент не существует универсального фильтра, подавляющего данные типы шумов при различных интенсивностях искажения. Поэтому важным аспектом является определение области применения каждого вида фильтра при подавлении шумов в изображении и создании типа фильтра, состоящего из синтеза сочетающего различные методы фильтрации для оптимальной очистки изображения. В статье представлен сравнительный анализ медианной фильтрации и фильтрации Винера для устранения импульсного и гауссовского шума в изображении при разной степени зашумленности. Для моделирования использовалось одно изображение, искаженное отдельно импульсным и отдельно гауссовским шумом с вероятностями искажения пикселей от 1 % до 99 % включительно. Фильтрация производилась с окнами, равными 3x3 и 5x5. В результате были получены численные оценки качества фильтрации изображений на основе пикового отношения сигнал-шум (PSNR). На основе полученных данных была проанализирована область применения исследуемых фильтров, их модификации, достоинства и недостатки, а также приведены рекомендации по их использованию. В результате сравнительного анализа исследуемых видов фильтрации для зашумленных изображений было установлено, что медианный фильтр с окном 3x3 лучше справляется с очисткой изображения от импульсного шума малой интенсивности и с окном 5x5 – с очисткой изображения средней интенсивности зашумления. Также медианный фильтр лучше справляется с фильтрацией гауссовского шума при его средних и высоких значениях среднеквадратичного отклонения. Фильтр Винера с окнами 3x3 и 5x5 лучше фильтрует гауссовский шум при малых его значениях его среднеквадратичного отклонения. Также фильтр Винера лучше справляется с импульсным шумом относительно высокой интенсивности зашумления.

Обработка изображений; импульсный шум; гауссовский шум; фильтры; медианный фильтр; фильтр Винера.