A new algorithm for improving accuracy and reducing complexity of measuring the frequency of periodic signals

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ABSTRACT

Introduction: Frequency spectrum analysis is now widely used in a variety of application, including communication systems, measurement, instrumentation, electric power systems, mechanical vibration detection, and many other military and civilian uses. However, because of the practical issues that its industrial application encounters, estimating the frequency is a very significant and vital issue. Spectral analysis using the discrete Fourier transform is usually used to estimate frequency. The stationarity assumption, spectral leakage effects, and the finite-grid resolution, however, cause a number of uncertainty contributions in traditional DFT methods. **Purpose of the study:** research a method for determining the frequency by interpolation methods based on fast Fourier transform with windowing function. In this work introduced study a new method of frequency measurement to reduces the possibility of error in frequency evaluation, three spectral lines of the discrete Fourier transform were used, with apply the Kaiser (B=15) window to reduce the spectral leakage resulting from the frequency conversion of the signal into frequency domain by means of the fast Fourier transform. A technique based on estimating the level of the three spectral lines taken into consideration used with the spectral component that carries the highest level of amplitude. **Results:** The proposed new approach to frequency determination made it possible to develop an interpolation method that, unlike most known methods based on the a new correction formula, uses only the three values of amplitude of the spectral lines. **Practical significance:** When implementing the developed method, only one analog-to-digital converter is used, which reduces hardware costs. The results obtained from the analysis of the effect of the error caused by the deviation of the real signal from the harmonic model, make it possible to choose the optimal parameters of the measurement process, depending on the requirements for accuracy and measurement time. **Discussion:** The proposed approach is less complex than the available algorithms and more accurate compared to 4 algorithms. Simulation results indicate that using the approach proposed in this paper the calculation maximum estimation errors are not more than 2.75e-5.

KEYWORDS: fast Fourier transform; spectrum analysis; window; interpolation method; spectral lines; maximum frequency estimation error; spectrum.

Introduction

Frequency spectrum analysis is now widely used in a variety of application, including communication systems, radar and navigation, radio broadcasting, measurement, instrumentation, electric power systems, mechanical vibration detection, information detection, protection and transmission systems, and many other military and civilian uses. However, because of the practical issues that its industrial application encounters, the rapid and accurate estimating the frequency is a very significant and vital issue. Spectral analysis using the discrete Fourier transform is usually used to estimate frequency (DFT). The stationarity assumption, spectral leakage effects, and the finite-grid resolution, however, cause a number of uncertainty contributions in traditional DFT methods [1, 2]. An accurate means of measuring electrical signal parameters, including frequency, providing high levels of measurement speed and accuracy. In addition, the frequency is one of the main types of measurement information carriers for sensors for various purposes. The problem of reducing the signal frequency measurement time is especially acute in monitoring and control system of technological operations [3-6].

In addition to determining the rate of frequency deviation, accurate automatic control in the modern era contributes to the development of many fields such as increasing the accuracy of diagnosis in the medical field and increasing and improving production in the industrial field in addition to reducing the energy required in a number of production and technical factories and others [7].

This need has been addressed in several technological developments in the automation of quality control in recent years, and for this purpose process modeling is used [8]. Process modeling is an effective and economical method for analyzing and diagnosing processes. The extremely useful and detailed information that modeling provides cannot be achieved by any other means. Computer-aided finite element modeling was used to predict control processes [9].

One way to achieve the purpose in this research is to use raw information about the sample of the measurement signal to determine its frequency. At the same time, under certain conditions, the harmonic model is often chosen as the model of the periodic signal [10-11] which leads to subsequent evaluation of the error due to the discrepancy between the model and the real signal. This makes it possible to significantly reduce the measurement time. A further reduction in the frequency measurement time is provided by methods based on interpolation with three or two points of discrete Fourier transform (DFT) and the use of the instantaneous values of the higher voltages present in the signal spectrum [12]. This inevitably leads to an error evaluation of the frequency which is the basic parameter of the harmonic signal.

To increase the speed of algorithms for processing primary data when determining the frequency in the problems of mathematical modeling and control of the modes of electrical systems, it is proposed to perform the transition from discrete instantaneous values of the mode parameters to their representation by generalized vectors [13], this method significantly increases the measurement time. The method is a rather long measurement time, since the exemplary time interval is counted from the moment the input voltage passes through zero.

The (DFT) provides enough resolution for frequency estimation and operates in real-time. The fast Fourier transform (FFT) is an algorithm from the (DFT), which requires even less computational effort and is, therefore, faster. Taking a (FFT) of collected samples is arguably the most common method of making such frequency estimates. Computational or other limitations often restrict the number of samples which may be processed, which correspondingly restricts the resolution of the estimate provided by the (DFT).

Degree of development of the theme

Quinn [14] has developed a simple and effective method for accurately estimating the signal frequency using three-spectral lines interpolation (DFT) samples of the output peak. A second method was shown by Grandke [15], which employ only the DFT output peak and one adjacent sample. Both methods give effective frequency estimates that work well for signal to noise ratio (SNRs) at 0 dBs neither directly gives an adjusted estimate of the value, and both require separation. Jain et al. [16] Proposed a similar method for estimating frequencies, but the operational assumptions are restrictive and again the estimation of frequency requires division. Most modern SSPs do not have effective division instructions, so algorithms that minimize or exclude the use of divisions are preferred.

In [17] Jacobsen a two new approaches have been proposed that could offer benefits over previous approaches: a frequency interpolator that decrease the required number of partition operations with a slight increase in MSE, and a new volume interpolator that requires no spacers. The first approach which is a quadratic fit curve interpolator, delivers frequency estimation performance very close to Quinn's method but requires only one division. In this method it is not possible to estimate the magnitude directly. The second method does not require any divisions, but provides an estimate of the magnitude with only one predetermined frequency. This method is typically used to double the frequency resolution by providing a local estimate of the inter bin amount once the (DFT) output peak is selected, in which the normalized expression for the correction frequency, suitable for rectangular windows, has the form:

$$\delta = \mu \frac{|U(n+1)I - I\,U(n-1)|}{4|U(n)| - 2|U(n+1)| - 2|U(n-1)|} \tag{1}$$

In order to lessen the impact of scattering interference of nearby frequency components, Agrez [18] suggested a correction technique in 2002. As a result, the correction accuracy was somewhat enhanced. The normalized frequency correction formula for a rectangular window has the following structure:

$$\delta = \mu \frac{|U(n+1)| + I\,U(n-1)|}{2|U(n)| + |U(n+1)| - |U(n-1)|} \tag{2}$$

где μ = знак | U (n +1)|-| U(n-1))|.

The dedicated software's frequency calculation programs tell us about the FFT and DFT in all calculations and display the modules' spectrum. Because of this, they are the most practical.

The algorithms suggested by Ding [19], specifically some methods based on the modules of the maximizer Un and its neighbors situated on the spectral lines n-1 and n+1, were selected from the interpolation methods involving three spectral lines for analysis. Ding proposes the following connection as the correction factor:

$$\delta = \frac{|U(n+1)| - |U(n-1)|}{|U(n)| + |U(n+1)| + |U(n-1)|} \tag{3}$$

In this paper we proposed algorithm for measuring the frequency of harmonic signals based on the calculation of the weighted maximum using calculations of the FFT coefficients. To estimate the frequency of the signal, an (FFT) is performed, the number of the sample with the maximum modulus is determined, and the initial calculation of the frequency is performed [20-24].

The frequency of a periodic signal can be estimated from the position of the spectrum components obtained as a result of the (DFT). In this case, the error is determined by the frequency grid step, which is equal to the ratio of the sampling frequency fs to the number of samples N, one of the spectral lines taken into account that has the maximum amplitude as shown in the figure 1. The two highest spectral lines close to the line with the maximum amplitude are taken into consideration, so that the frequency is calculated based on these three spectral lines. This method allows obtaining a weighted average estimate of the position of a peak spectrum energy.

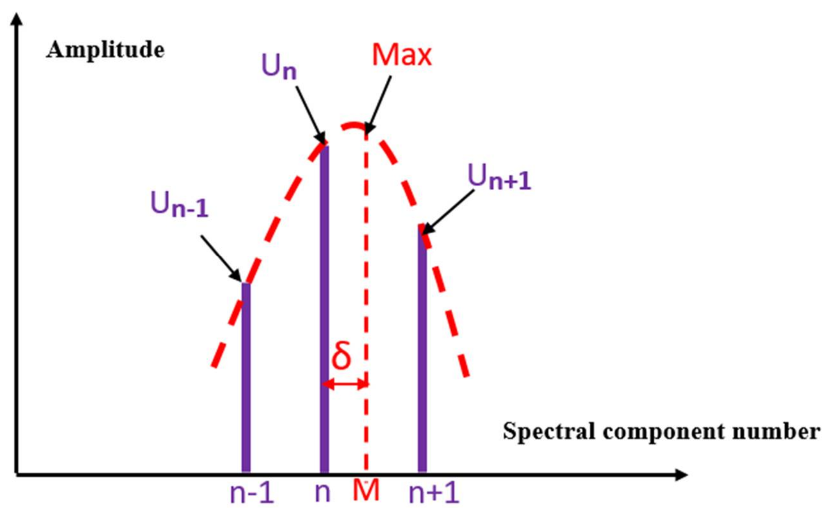

Fig. 1. The three points in the DFT spectrum used for interpolation

Figure 1. Shows the three peak of the (DFT) output component of a periodic harmonic signal. The desired frequency is between the two highest peaks in the DFT, i.e. between n and n+1. Delta is the range of deviation from the desired true frequency. Un, Un-1 and Un+1 are the highest three levels in the (DFT) spectrum.

We would like to show here that the main motive for presenting a new study and approach is the failure or weakness of the above-mentioned interpolation methods, including the Jacobsen algorithm, which is better than the Quinn, Jain, and Dain algorithms. For this reason, we will compare the proposed algorithm in this paper with Jacobsen's algorithm with different sample sizes and using the Kaiser (B=15) window.

In addition to evaluating the frequency of the harmonic signal, the interpolation technique includes additional tasks such as detecting the presence or energy of the signal. There are limitations on the time and power of signal processing, such as the length and method of DFT calculation used. The principles presented for the proposed algorithm apply to any length of DFT and thus all lengths from 16 to 8192 have been studied.

Often, the resolution wanted for frequency estimation cannot be met with the practicably computable DFT length. Using an effective interpolator tool can simply be seen as a way to reconcile conflicting needs. We assume the arrangement of the computational steps to estimate the frequency of the harmonic signal as follows:

1. The ratio of the signal frequency Fc to the sampling frequency Fs, the window function, and the sample size (N) are determined.

2. Convert the analog signal to digital form and collect the N according to the sampling frequency Fs. The time domain representation of a sampled signal with a period T = l/Fc, may be expressed as

$$x(n) = A \sin\left(2\pi Fc \frac{n}{Fs} + \varphi\right) \tag{4}$$

Where A, and φ are the amplitude, and phase respectively, n = 0, 1, 2, …… N - 1.

3. The window function (w (n) = Kaiser (B=15)) is applied to the resulting matrix from step 2.

$$R(n) = x(n)w(n) \tag{5}$$

4. The Fast Fourier transform of the matrix obtained from step 3 is calculated.

$$fft(n) = FFT\big(R(n)\big) \tag{6}$$

By applying a window function, w (n), and the Fast Fourier transform, the discrete approximation sequence of the given signal in the frequency domain X(f) can be obtained.

$$(X(f)\ ) = \sum_{n=-\infty}^{\infty} fft(n)\, e^{-j2\pi fc.n} \tag{7}$$

5. In the complex spectrum obtained from Step 4, determine the number of the component with the maximum amplitude (Un).

6. The level of the two components closest to the maximum component (n) of the spectrum lines is estimated (Un-1 , Un+1).

7. By Proposed formula the (**δ**) is calculated:

$$\boldsymbol{\delta} = \frac{(Un - Un_{-1})\ - (Un - Un_{+1})}{Un} \tag{8}$$

For compare with Jacobsen formula the correction parameter (**δ**) is calculated as:

$$\boldsymbol{\delta} = \mathrm{Re}\left\{\frac{(Un_{+1} - Un_{-1})}{2 * Un - Un_{+1} - Un_{-1}}\right\} \tag{9}$$

8. The frequency of the signal is calculated by equation (7) for both algorithms.

$$F_0 = (F_n + \boldsymbol{\delta}) * \mathrm{Fs/N} \tag{10}$$

10. The frequency evaluation error is calculated by equation (8) for both algorithms.

$$\mathrm{Erest} = (F_0 - \mathrm{Fc})/\mathrm{Fc} \tag{11}$$

The analysis showed that the abrupt discontinuity is due to the switching of the structure of an odd number of components taken into account at the maximum of the energy spectrum with two components. Depending on the ratio of the frequency of the fundamental harmonic of the signal to the sampling frequency, two types of spectrum symmetry and asymmetry can be distinguished (see Figure 2):

- Symmetry structure, when there is a central component close to the desired frequency (figure 2a);

- The structure is asymmetrical, when the desired frequency is located between two components of equal amplitude (figure 2b and 2c).

When the spectrum is symmetric with respect to one component, the position of the three highest-level components with numbers 6, 7, and 8 is taken into account (figure 2a).But with a no symmetry of the spectrum of components 7, 8, and 9 taken into account in the calculations (figure 2c).

This mean when working in the frequency range, a spectrum of an odd or even structure is formed with one or two main components. In a narrow zone of change in the frequency of the signal or sampling, the position of the “center of gravity” on the frequency axis changes abruptly. Figure 2b shows a spectrum whose level of component 7 is slightly higher than 8, so numbers 6, 7 and 8 will be taken into account. In the range of small deviations of the sample duration from an integer number of periods, the error changes smoothly. In the region of symmetry of the even structure of the spectrum, when the levels of the principal components 6 and 7(figure 2b), 7 and 8(figure 2c) change, an error jump is formed, which occurs due to the displacement of the "area" and "weight". The abscissa axis can be the ratio of the signal frequency to the sampling frequency Fc/Fs or an addition to the total sampling time of discrete samples.

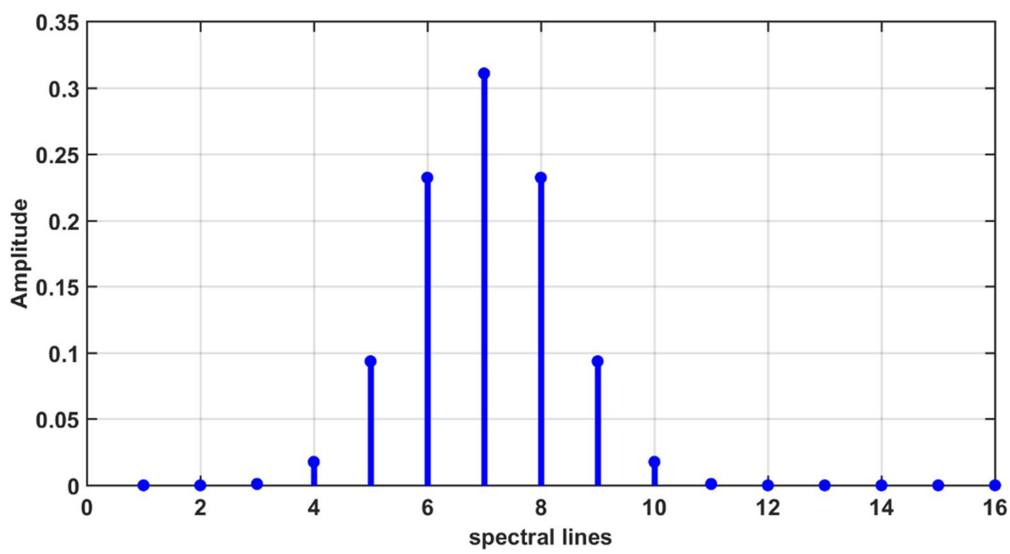

Fig. 2. (a) Symmetry with respect to one spectral lines

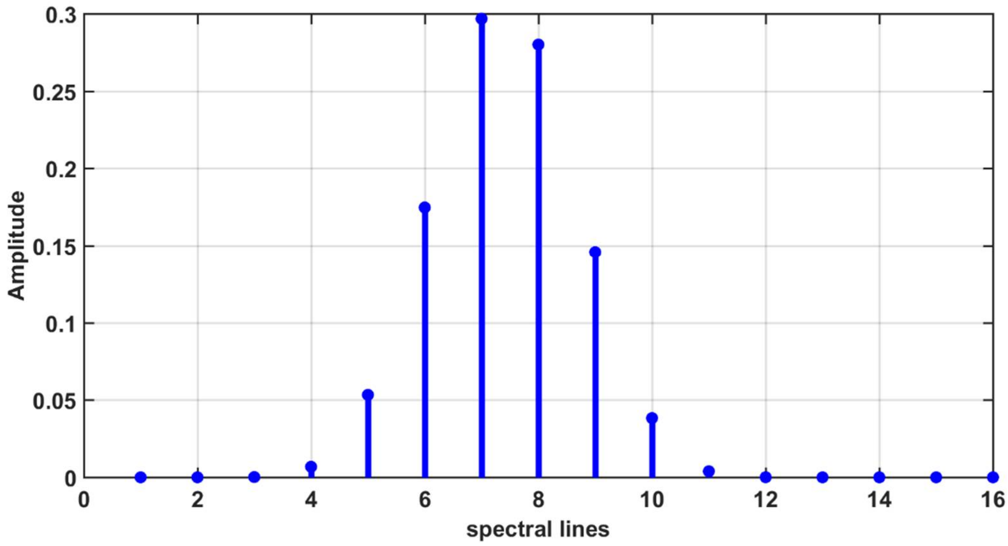

Fig. 2. (b) Asymmetry two spectral lines

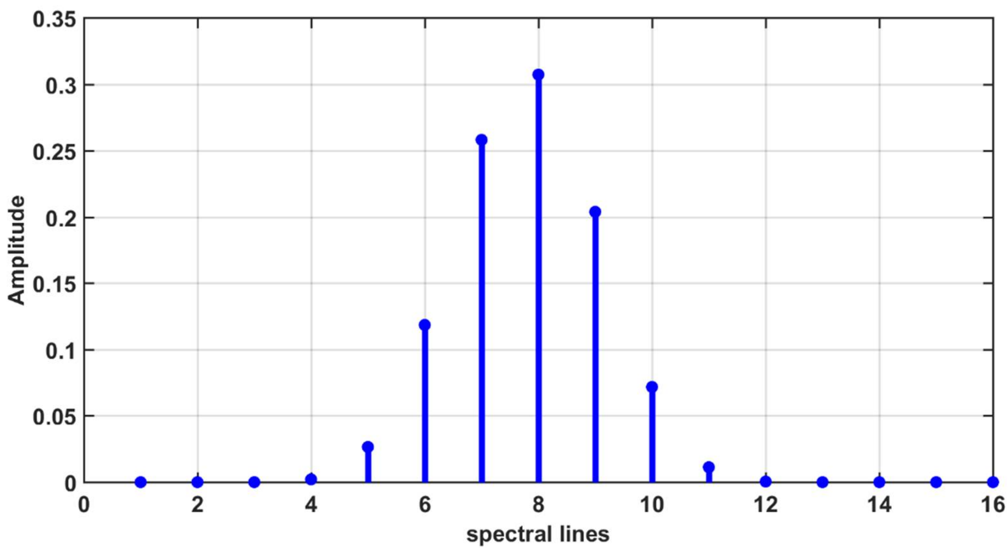

Fig. 2. (c) Asymmetry two spectral lines

Usually when DFT outputs are interpolated to improve in-time performance and reduce computational complexity especially in processor-constrained systems, trade-offs can be made between the system constraints and the algorithms introduced.

Figures 3-5 shows the relative performance of the proposed and Jacobsen estimators with two-period pure signal without additive noise. The lengths were 16,256, and 8192 of (DFT) used with the ratio of signal frequency to sampling frequency 0.25-0.37, 0.25-0.258 and 0.24985-0.25025 respectively, taking into account the symmetry and asymmetry of the largest components of the signal spectrum. The input frequency of the signal Fc was equal to Fs/4, i.e. close to the Nyquist theory, the initial phase was 0.

For in the region of equality between two spectral lines, an error jump occurs, which is caused by the transition from one large component to another large component. For different windows, different jumps and different degrees of smoothing. When considering Jacobsen's algorithm, the error resulting from the residual hop may be 50 times greater than that of the proposed algorithm. The graphs show a reduction in maximum error of about 10 to 50 times with the proposed algorithm, the results of the proposed algorithm provides the lowest error and improves the result. The result shawn in table 1, when the sample size is 16, the number of times improving are 56 times. It is clear that the new estimator provides good approximation performance with potential computational benefit in reduced division requirements. It is noted that the new estimator actually outperforms Jacobsen's performance in all DFT sample sizes.

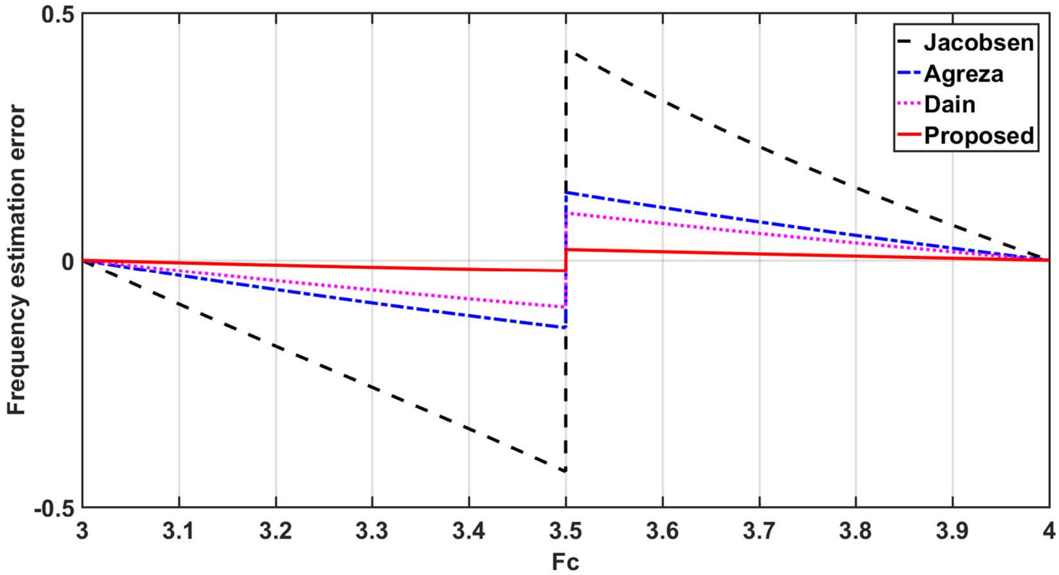

Fig. 3. Kaiser (B=15) window at N=16

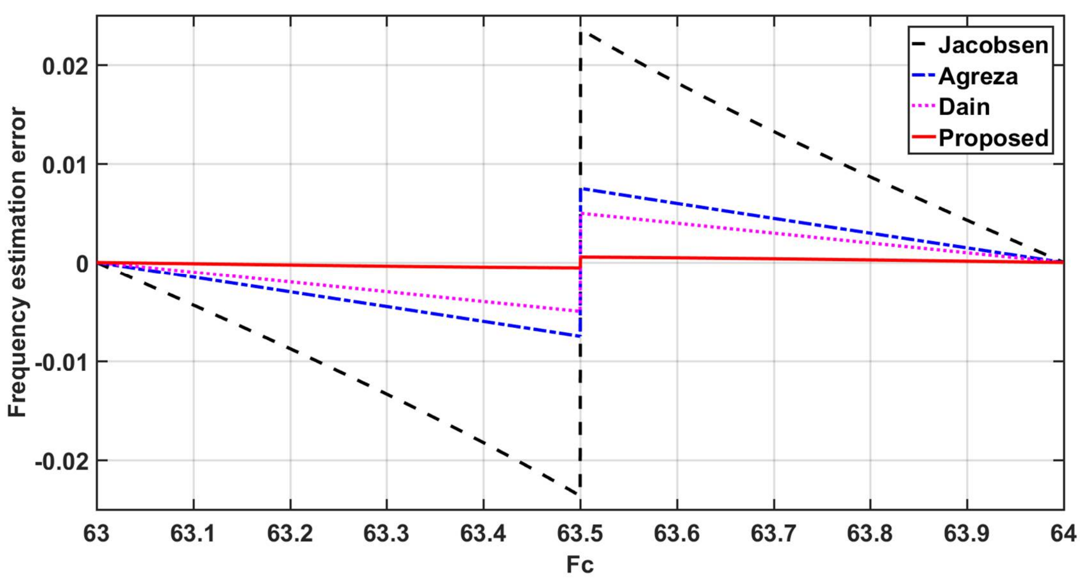

Fig. 4. Kaiser (B=15) window at N=256

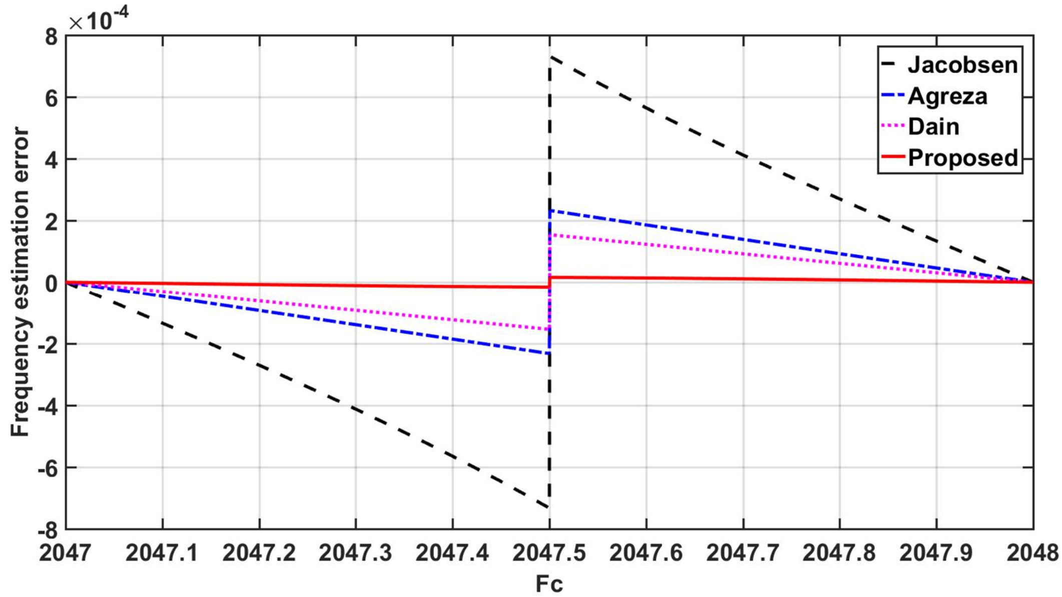

Fig. 5. Kaiser (B=15) window at N=8192

Figure 6-7 displays the error plots for the four algorithms (Jacobsen, Agreza, Dain, and Proposed) using the Kaiser (B=15) window for sample sizes of 16, 256, and 8192, respectively. Graphs demonstrate that the suggested method is superior to Jacobsen's algorithm, which is inferior. The error reduces as the sample size rises from 16 to 8192. The interpolation method's correction precision is denoted by the symbol. If the sign is true, the corrected frequency will get closer to the actual frequency, reducing the frequency error. The corrected frequency will be far from the actual frequency if the sign, or interpolation direction, is incorrect, which will result in a greater error than the uncorrected frequency [25–26].

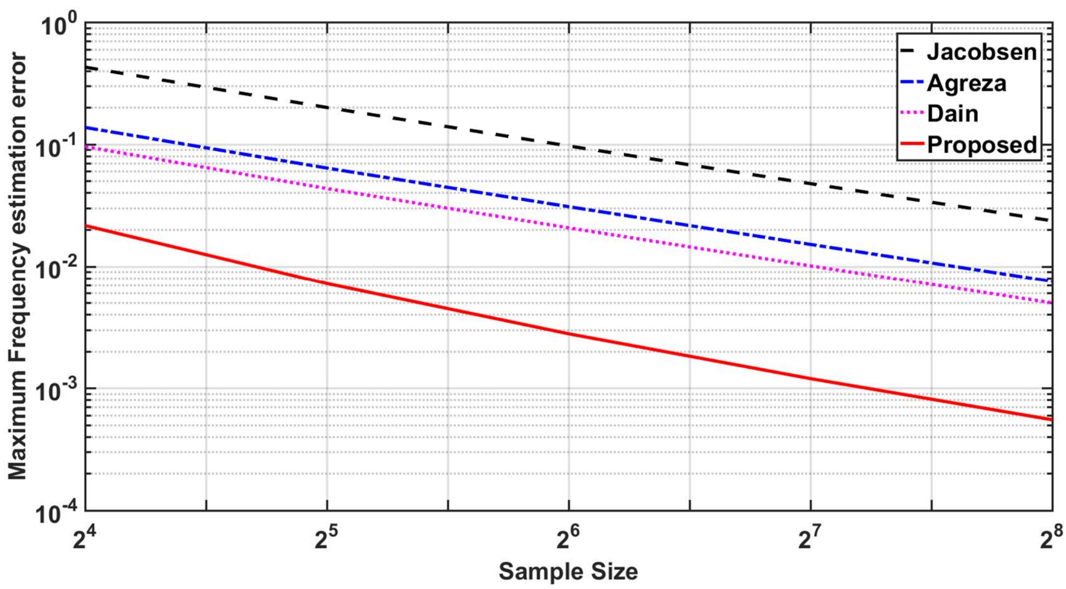

Fig. 6. Kaiser (B=15) window at N=16-256

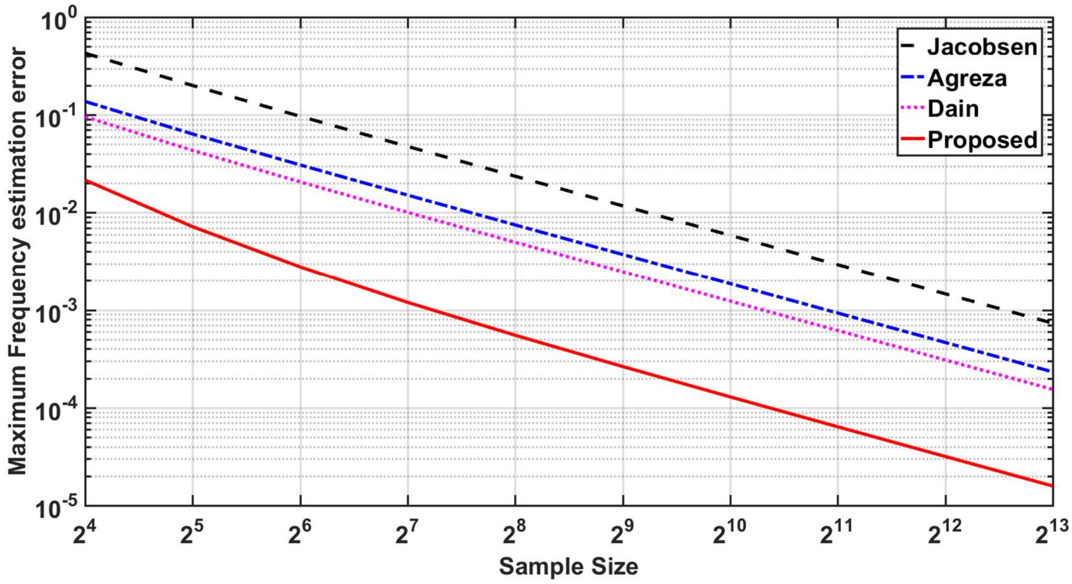

Fig. 7. Kaiser (B=15) window at N=16-8192

Table 1. Improved accuracy for proposed algorithm N=16-8192

Samples size (N)	Maximum Error		
	Jacobsen Algorithm	Proposed Algorithm	improved accuracy
16	1.43e-1	2.51e-3	56-Times
32	6.67e-2	4.37e-3	15-Times
64	3.23e-2	2.88e-3	11-Times
128	1.59e-2	1.61e-3	10-Times
256	7.87e-3	8.43e-4	9-Times
512	3.92e-3	4.31e-4	9-Times
1024	1.96e-3	2.18e-4	9-Times
2048	9.78e-4	1.10e-4	9-Times
4096	4.89e-4	5.50e-5	9-Times
8192	2.44e-4	2.75e-5	9-Times

To work with small errors, using the Kaiser (B=15) window, need to weigh at least 16 samples of the spectrum. The improved accuracy error at a value of 256-8192 is 9 times.

Conclusions

Common interpolation methods currently available fail to perform satisfactorily when working on complex DFT outputs or their detected magnitudes, prompting a search for alternative methods. With Jacobsen interpolator performed on complex data using linear interpolation, is generated a pseudo-null value in the center of the main lobe, which is clearly detrimental to peak and location detection in post-DFT processing. For example, interpolation will never correctly restore nulls between side lobes. It is therefore difficult for interpolators to correctly locate the true peak or to deduce the true shape of the main lobe.

The proposed algorithm offers a solution to all problems available in previously algorithm and also provides an accurate estimation, whereby good performance can be achieved with all sizes of N. The main important features of the proposed algorithm are:

• High performance;
• High measurement accuracy.
• Lower computational complexity

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НОВЫЙ АЛГОРИТМ ПОВЫШЕНИЯ ТОЧНОСТИ И СНИЖЕНИЯ СЛОЖНОСТИ ИЗМЕРЕНИЯ ЧАСТОТЫ ПЕРИОДИЧЕСКИХ СИГНАЛОВ

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АННОТАЦИЯ

Введение: Анализ частотного спектра в настоящее время широко используется в различных приложениях, включая системы связи, измерения, контрольно-измерительные приборы, системы электроснабжения, обнаружение механических вибраций и многие другие военные и гражданские применения. Однако из-за практических проблем, с которыми сталкивается его промышленное применение, оценка частоты является очень важным и жизненно важным вопросом. Спектральный анализ с использованием дискретного преобразования Фурье обычно применяется для оценки частоты (ДПФ). Однако допущение о стационарности, эффекты спектральной утечки и разрешение на конечной сетке вносят ряд вкладов в неопределенность в традиционных методах DFT. **Цель исследования:** исследование метода определения частоты интерполяционными методами на основе быстрого преобразования Фурье с оконной функцией. В данной работе представлен новый метод измерения частоты для снижения вероятности ошибки в оценке частоты, использовались три спектральные линии дискретного преобразования Фурье с применением окна Кайзера (B=15) для уменьшения спектральной утечки, возникающей из-за преобразование частоты сигнала в частотную область с помощью быстрого преобразования Фурье. Методика, основанная на оценке уровня трех учитываемых спектральных линий с использованием спектральной составляющей, имеющей наивысший уровень амплитуды. **Результаты:** Предложенный новый подход к определению частоты позволил разработать метод интерполяции, который, в отличие от большинства известных методов, основанных на новой формуле коррекции, использует только три значения амплитуд спектральных линий. **Практическая значимость:** При реализации разработанного метода используется только один аналого-цифровой преобразователь, что снижает аппаратные затраты. Результаты, полученные при анализе влияния погрешности, вызванной отклонением реального сигнала от гармонической модели, позволяют выбрать оптимальные параметры процесса измерения в зависимости от требований к точности и времени измерения. Обсуждение: Предлагаемый подход менее сложен, чем доступные алгоритмы, и более точен по сравнению с 4 алгоритмами. Результаты моделирования показывают, что при использовании предложенного в данной статье подхода ошибки расчета не превышают 2,75e-5.

Ключевые слова: быстрое преобразование Фурье; спектральный анализ; окно; метод интерполяции; спектральные линии; максимальная ошибка оценки частоты; спектр.

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