Time-Varying Signal Analysis Method Based on Linear Interpolation

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Abstract. To solve the problem of harmonic analysis of time-varying signals, an analysis method based on linear interpolator is proposed. Firstly, the noise in measuring equipment is filtered by input filter, and then the sample number of signals in a period is guaranteed to be an integer power of 2 by linear interpolator, and then the cascade subtraction of filters is carried out. The number of sampled values and the cosine roll down window are used to prevent the energy from spreading to the side lobe. Since the fundamental frequency of the system changes, frequency control is used to estimate the frequency and feedback to the linear differentiator module to adjust the time interval of the difference to ensure that the output signal sampling value is strictly an integer power of 2. Finally, the FFT transformation of the voltage and current data is carried out to analyze the current, voltage and impedance. The comparison results show that this method can be applied to harmonic analysis, especially in high-frequency harmonic analysis has a better performance and certain practicability.

1. Introduction

With the development of power electronics technology, a large number of new power electronics equipment have been introduced into the distribution network system. These new power electronics equipment will introduce a large number of high frequency harmonics into the power grid because of their high switching frequency. Therefore, it is necessary to measure high frequency harmonics accurately. At present, fast FFT is widely used in harmonic analysis Change. However, fast FFT requires synchronous sampling of signals, i.e. the integer-multiple relationship between the sampling frequency and the signal period. However, the actual grid frequency is constantly changing, which results in the problem of sampling asynchrony. If the sampling is not synchronous, it will cause spectrum leakage and spectrum aliasing, which will be used for harmonic measurement, especially for harmonic measurement. High frequency harmonic measurement results in error.

There are two methods for the measurement of time-varying signals: resampling and interpolation. The method of resampling is to measure the frequency of the fundamental wave first, and resample the signal according to the frequency. Because the fundamental frequency needs to be measured in real time, once the fundamental wave is a time-varying signal, the real-time tracking of the fundamental frequency may be problematic, which will easily lead to loss of synchronization and error. After calculating the fundamental frequency, interpolation algorithm is used to interpolate and recombine the sampled data,
so that the re-obtained data approximate the ideal data, and then analyze the method. This paper presents a method of analyzing time-varying signals by using linear interpolator, and compares its effectiveness with data analysis.

2. Time-varying signal analysis method

The method consists of six parts: input filter, interpolator, filter cascade, cosine roll-off window, and fast FFT transform and frequency controller.

The sampled signal is filtered by input filter first, then the sample number of the signal in a period is guaranteed to be an integer power of 2 by linear interpolator, and then the number of sampled values and cosine roll-off window are reduced by cascade filter to prevent energy from dispersing to sidelobes. Since the fundamental frequency of the system changes, a frequency control section is required to estimate the frequency and feed back to the linear differentiator module to adjust the time interval of the difference to ensure that the output signal sample value is strictly an integer power of 2. Finally, the original data of voltage and current can be obtained by FFT transformation, and further analysis of current, voltage and impedance can be carried out. Interpolator and cosine roll down window are 2 important parts of the method.

When the input signal enters the measurement system, it will contain all kinds of frequency noise. In order to reduce the error caused by noise, we choose to use an IIR Chebyshev low-pass filter. Compared with FIR filter, IIR Chebyshev low-pass filter has fewer coefficients, faster operation speed and less storage space.

![Figure 1. Algorithm flowchart and Amplitude frequency response of IIR filter](image)

Because of the fluctuation of the frequency of the grid, the periodic length of the fundamental wave will change. The FFT algorithm needs to use the time of fundamental wave period, which will change with the change of grid frequency. In order to make the FFT operation meet the requirement, the input of FFT operation is worth the integer power of 2. The function of the interpolator is to ensure that the number of FFT input values is constant. The linear interpolator is used in this method, because the number of samples to be processed is very large, so the implementation complexity can be reduced by using the linear method.

The design of the interpolator is based on the following expressions:

\[
T_{in} \leq T_{out} \leq 2 \cdot T_{in} \quad (1)
\]

\[
n \leq k \cdot \frac{T_{out}}{T_{in}} \leq n + 1 \quad (2)
\]

\[
o[k] = \left(\frac{n+1}{T_{in}}\right)^{T_{out}} \cdot i[n] + \frac{kT_{out}-nT_{in}}{T_{in}} \cdot i[n + 1] \quad (3)
\]
\[k = 0, 1, 2\ldots u\]

Tin represents the time interval between two input values and Tout represents the time interval between two output values of the interpolator. \(I[n]\) is the input signal and \(o[k]\) is the output signal.

Before the signal performs FFT operation, it passes through a window function which prevents the signal energy from dispersing to the sidelobe. The cosine roll down window is used to suppress the sidelobe. The modified method is used cosine roll-off window to suppress sidelobes. The calculation of the cosine roll-off window is shown in the following formula:

\[
block_{new} = \block_{old} \cdot \frac{\cos\left(\frac{\pi}{2} n\right)}{1, 2, 3\ldots n}
\]  

(4)

Because the length of the basic period of FFT execution depends on the current frequency of the power grid, a frequency controller is needed to track and adjust the length of FFT. The controller mainly adjusts the basic cycle length of the interpolator. In the measurement system, the frequency controller adopts proportional integral controller PI. The step response method is used to determine the parameters of the controller. The amplification factor is set to 0.25, and the reset time is set to 200ms. The deviation of control is to calculate the difference between the angle of the first harmonic of the previous FFT and the angle of the current FFT, and the expected value of the difference is 0 for the ideal case. The output of the frequency controller can be used to calculate interpolator's new interpolation interval. This method is similar to the principle of PLL.

3. Comparison of high frequency parameter acquisition results

In order to compare the effectiveness of this method for data analysis, we use household energy-saving lamp (9 W) and incandescent lamp as the test equipment. The following figure shows the capture result (20 ms) of one cycle signal (voltage and current) of the selected high-precision energy-saving lamp in the market.
For the above measurement data, the method proposed in this paper is compared with the power quality tester on the market.

![Frequency component analysis of energy-saving lamp and incandescent lamp](image)

Figure 4. Frequency component analysis of energy-saving lamp and incandescent lamp

The results of frequency component analysis of voltage signals are shown in the diagram. The Y axis is dB, and the X axis is Hz, with a range of 0 to 2500 Hz. The blue line is the test equipment on the market, and the green line is the test equipment of this method. The comparison shows that the analysis results of the two sets of measuring equipment are almost the same in the low frequency part, but in the high frequency part, the method shows a more reasonable analysis results.

4. Summary

According to the analysis results, there are two main conclusions.

(1) In this paper, a method of using linear interpolator to analyze the asynchronous sampling signal is proposed. The frequency controller is used to control the interval of interpolator to realize the function similar to PLL, and the cosine roll-off window is added to reduce the spectrum leakage. This method is feasible in principle.

(2) According to the data analysis and comparison with other equipments, it can be found that the method is consistent with the market power quality analyzer in the low frequency part, and the validity of the method can be determined. The data analysis structure of the linear interpolator in the high frequency part is more reasonable.

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