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A Simple Approach for Power Signal Frequency Determination on Virtual Instrument Platform

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Abstract: A simple approach to the zero crossing technique for the purpose of frequency determination of power signal is presented. The Fourier algorithm is used for digital filtering in order to extract the cosine and sine parts of the fundamental frequency component. Then, the zero crossing technique is applied to the cosine or sine components of the signal. The derived algorithm is developed in a PC-based signal analyzing platform for its implementation. The computer is equiped with the DAQ device and LabVIEW program. As the acquired signal was stored in the databank of the program, zero crossing algorithm is implemented to determine the frequency of the power signal. A burst sinusoidal signal with its high harmonic is stimulated by the signal generator to evaluate the accuracy for this developed system. This experiment can approve that the zero-crossing algorithm is an effective method for the frequency estimation.

Keywords: Zero-Crossing Fourier Algorithm, DAQ, LabVIEW

1 Introduction

The frequency of a power network is an important opertional parameter for the safety, stability and efficiency of the power system. As the nonlinear devices loaded in the power system increase, the harmonic phenomena of the power signal is increased and resulted in the unbalance between the power systems. These effects also introduce the frequency shift and make the damage on the power system [1,2,3]. Due to the current advanced technological applications, livelihood equipments are ever increasing. And humans have the convenience of living life. Those electrical equipment will also relatively increase the electricity consumption and make the frequency changes, because of changes in load characteristics of the design and supply systems [4].

In power system operation, the frequency is to explain whether the electricity supply and demand balance. Security and economy are very important indices for system reliability. When the frequency is lower than the nominal value, it means the system is in overload conditions; if the frequency is higher than the reduction value, indicating an oversupply of generating capacity. By observing the frequency change, one can clearly understand operation conditions of a power system. Reliable frequency measurement is prerequisite for effective power control, load shedding, loas restoration and system protection. Therefore, there is a need for fast and accurate estimation of the frequency of the power network using voltage waveform which may be corruped by noise and harmonics components.

A few methods for the frequency determination have been proposed in the past fw decades. The discrete Fourier transformation, least error squares and Kalman filter are known signal processing techniques, used for the frequency measurement [5,6,7,8,9]. The bilinear form approach [10] seems to be a very efficient method for both small frequency deviation and offnominal frequency estimation. An adative algorithm for frequency measuring over a wide range is suggested by Moore et al [11].

In this paper, a very simple algorithm with acceptable accuracy is derived. The Fourier algorithm is used as digital filter to extract cosine and sine parts of the fundamental frequency component, and the zero crossing technique is applied to cosine and sine parts of the signal for frquency estimation.. The original signal may be corrupted by noise and then the higher harmonic components. Nonrecursive Fourier algorithm is well known from the signal processing theory and it is used to

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provide cosine and sine component of the fundamental harmonic of the signal with high accuracy. The algorithm we derived is very simple and requires abridged resources for implementation.

2 The Proposed Fourier Method

A measured signal $v(t)$ (arbitary voltage or current), shown in Fig. 1, could be expressed as

$$v(t) = V\cos(\omega t + \phi) + R(t) \tag{1}$$

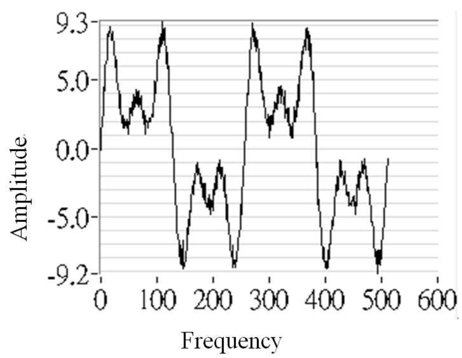

Fig. 1: Signal with its 3th and 5th harmonic and white noise components.

Where v is magnitude of the fundamental harmonic, $\omega = 2\pi f$ is angular frequency of the fundamental harmonic, f is fundamental harmonic frequency, ϕ is phase of the fundamental harmonic, $R(t)$ is signnal which contains higher harmonics and zero mean white noise.

If the exact value of ω is unknown and assumed as ω_a , v_a (discrete Fourier series parameter) can be estimated using discrete Fourier series (DFS) as the folowing:

$$\overline{v} \cong \frac{2}{m}[\sum_{n=1}^{m} v_n cos(\frac{\omega_a T_a}{m}n) - j\sum_{n=1}^{m} v_n sin(\frac{\omega_a T_a}{m}n)] = A + jB \tag{2}$$

where $v_a^2 = A^2 + B^2$, ω_a -assumed angular frequency in Fourier series (In formula (2) gives exact value of V only for $\omega_a = \omega$), m -munber of samples in assume period of the first harmonic of the processing signal, $v_n - n$ -th sample of the signal, and $T_a = \frac{2\pi}{\omega_a}$.

Assumed signal is sampled at frequency $f(s) = \frac{m}{T_a} = \frac{1}{T_s}$, where T_s is sampling period. It means that the assumed frequency of the discrete ourier series is defined by the sampling period and number of samples in the assumed period of the fundamental harmonic. For the frequency estimation it is enough to observe the cosine, sine or both components of the original signal. The cosine and sin components, $A(t)$ and $B(t)$ can be calculated as the following relation:

$$A(t) = \frac{2}{m}\sum_{n=1}^{m} v_n \cos(\varphi n) \tag{3}$$

$$B(t) = \frac{2}{m}\sum_{n=1}^{m} v_n \sin(\varphi n) \tag{4}$$

where $\alpha = \frac{\omega_a T_a}{m} = \frac{2\pi}{m}$.

If data window sweeps along the measured signal, formula (3) and (4) could gives a grup of points corresponding to the periodic time functions $A(t)$ and $B(t)$. The cosine and sine omponents, A and B , are periodic functions of time. If the frequency of fundamental harmonic of signal is equal to assumed fundamental frequency of Fourier series ($T_s \cdot m = T_a = T = \frac{1}{f}$), then $A(t)$ and $B(t)$ are ortogonal periodic functions of frequency f . If $T_A \neq T$, functions $A(t)$ and $B(t)$ are not pure sine and cosine waves but the frequency of their fundamental harmonic is f.

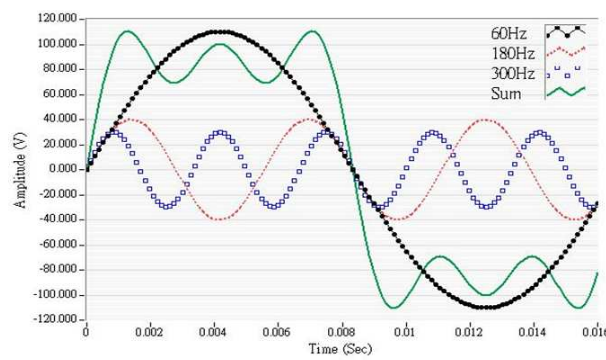

Fig. 2: The distored signal of the fundamenttal signal with 3th and 5th harmonic components.

Figure. (2) shows a fundamental signal with a frequency of 60 Hz, which contains 100% of the first, 40% of the third, 30% of the fifth harmonic and its cosine component $A(t)$. For a practical application of formula (3), the auxiliary vectors of cosine, sine and signal samples are required. The length of all the vectors is m . The cosine and sine vectors are shown as

$$\cos = \frac{2}{m}[\cos(\psi), \cos(2\psi), \ldots \cos((m-1)\psi), 1]^T \tag{5}$$

$$\sin = \frac{2}{m}[\sin(\psi), \sin(2\psi), \ldots \sin((m-1)\psi), 1]^T \tag{6}$$

The vector of signal samples is

$$SAMs = [v_1 v_2 v_3 v_4 v_5 \ldots\ldots v_n]^T \tag{7}$$

By using the auxiliary vectors COS and SIN, the alculation in (3) and (4) requires only multiplication and addition, without calculation of trigonometric functions and without division. After the acquisition of a new sample v_{new} , it is necessary to reorder the vector SAMs as $v_1 = v_2, v_2 = v_3, \ldots, v_n = v_{new}$. Therefore,the moving data window and samples of the processed signal are considered as scalars. For each data window the corresponding points of the cosine (A) and sine (B) components are calculated according to relation (3) and (4).

Previous developed algorithm has some limitations. When the sampling frequency f_s is not exactly an integer multiple of the fundamental frequency f , the period of cosine and sine coponents, A and B , also do not contain integer number of sampkes. For high sampling frequency the error of f and $f_a(=\frac{f_s}{m})$ is small. But for low sampling frequency, the error will be great. Therefore it is nessary to apply the zero crossing algorithm to modify the sampling number m to be ineger, where is number of samples in assuming period $T_a = \frac{1}{f_a}$.

3 Zero-Crossing Algorithm

Zero-crossing Algorithms [12] is the most direct method of frequency detection. Zero-crossing algorithm calculation method is simple, fast execution, without mathematical formulas complex operations. For a periodic signal, the time slot between two zero-crossing points is half a period of the signal. The time unit between two zero-crossing points is calculated and countdown converted into the actual frequency of the signal [13,14]. The algorithm of the estimation process is described as follows:

For the sine component of a power signal, shown in Figure. (3), it could be described by the sinusoidal wave function as

$$y(t) = A\sin(\omega t) \tag{8}$$

where A is amplitude, $\omega = 2\pi f$ is angular frequency and f is frequency. When the signal is expressed in discrete mode, and let $t = kT_s$, formular (1) is expressed as

$$y(k) = A\sin(\omega k T_s) \tag{9}$$

According to the sampling theorem, the angle difference [12,13,14,15] between any continuous points is calculated

$$\Delta\theta = \omega T_s = 2\pi f T_s \tag{10}$$

where T_s is sampling time interval.

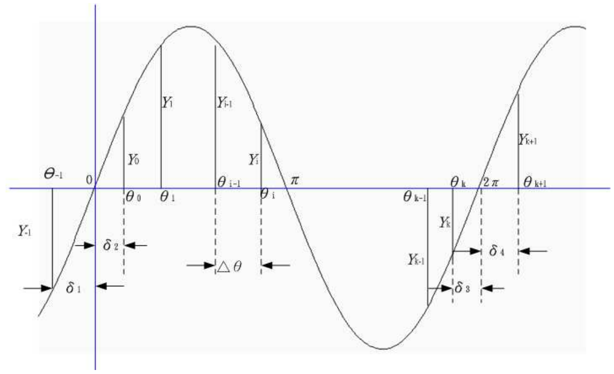

Fig. 3: The diagram for Zero-Crossing Algorithm.

Assuming there are k sampling points in a full period of a periodical signal, the estimated frequency for the signal is derived by

$$f = \frac{1}{kT_s} \tag{11}$$

For the practical application, we do not really let the frequency of the measured signal as the integer multiples of the sampling frequency and resulting in a non-integral N.

Let us observe the first posive sample y_0 and the last sample y_k in the concerned period.

The complete period of the sine component should expressed as

$$T_a = kT_s + \delta_2 + \delta_3 = NT_s = N\Delta\theta \tag{12}$$

where $T_s = \Delta\theta$.

In the zero crossing area cosine function can be presented by linear function and the N can be derived as the following equations

$$N = k + \frac{\delta_2}{\Delta\theta} + \frac{\delta_3}{\Delta\theta} = k + \frac{\delta_2}{\delta_1+\delta_2} + \frac{\delta_3}{\delta_3+\delta_4} \tag{13}$$

Where $\delta_2 = \theta_0$ is the angle difference of the zero point between the first sampling point in the period, $\delta_3 = 2\pi - \theta_k$ is the angle difference of the zero point between the last sampling point in the period, and $\Delta\theta = T_s$ is the angle difference between two adjacent sampling points, which is interpreted as the sampling period $T_s(= f_s)$. All samples between 0 and 2π which correspond to one period are equal to $N = k + \frac{\delta_2}{\Delta\theta} + \frac{\delta_3}{\Delta\theta}$. Summation of $\delta_2 + \delta_3$, from the same period can be different from one sampling period T_s . Therefore, the number of samples n of one whole period of the sine components can be a fraction. After this modification, the algorithm becomes much more accurate.

Similar triangle theory [16] can be applied on Figure. 3, and formula (12) can be simplified as

$$N = k + \frac{|y_0|}{|y_0| + |y-1|} + \frac{|y_k|}{|y_k| + |k+1|} \quad (14)$$

where $\frac{\delta_2}{\delta_1+\delta_2} = \frac{|y_0|}{|y_0|+|y-1|}$ and $\frac{\delta_3}{\delta_3+\delta_4} = \frac{|y_k|}{|y_k|+|k+1|}$

Then the accurate estimation for the acquired signal frequency is

$$f' = \frac{1}{NT_S} \quad (15)$$

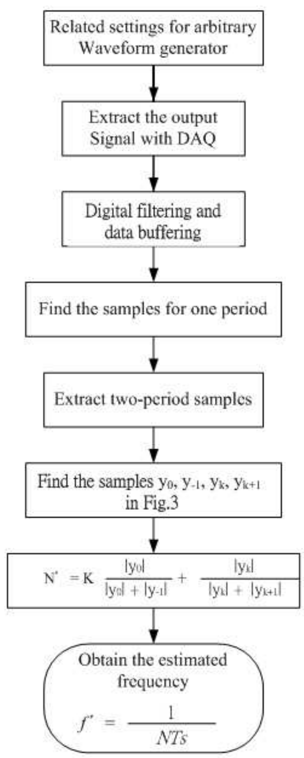

Fig. 4: Program flow chart for frequency determination.

The program flow chart for the frequency determination is shown in Figure. 4. Once the signal was acquired from the arvitrary function gerator, it was proceed with the fourier algorithm to filter out its high order harmonic and white noise to extract the sine component. The data matrix of sine component was stored in the memory the computer. Then the zero crossing agoritm was implemented to determine the actual frequency of the sine component.

The deatil was described as the following:

(1) Assuming much more than 3 cycles of sine data was processied in the program, the k value in equation (12) was obtained by determining the number between first maximun and the secnd maximun value in the data bank.

(2) Then, we could find out the first minimun value in the data bank and store the subsequent $3k$ samples for the subsequent process.

(3) Therefore, we could have the center k samples, whicn means the k^{th} sample to the $(2k+1)^{th}$ sample, as the observe sine data shown in Figure. 3, where the k^{th} sample is loacated at θ_{-1} and the $(2k+1)^{th}$ sample is located at θ_{k+1} .

(4) Finally, the δ_2 and δ_3 can be determined and the N value and frequency f' were obtained.

4 Experiment and Discussions

LabVIEW (NI Instruments Co.,11500 N Mopac Expwy Austin, TX 78759-3504) is implemented to integrated instruments such as oscilloscope, digital meter, functional generator, data acquisition (DAQ) device, etc to establish a PC-based instrumentation system for measurement and computation.

Figure. 5 is the schematic diagram for signal frequency determination and the experimental set-up was shown in Figure. 6.

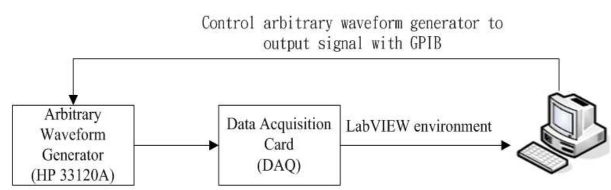

Fig. 5: Schematic diagram for signal frequency determination.

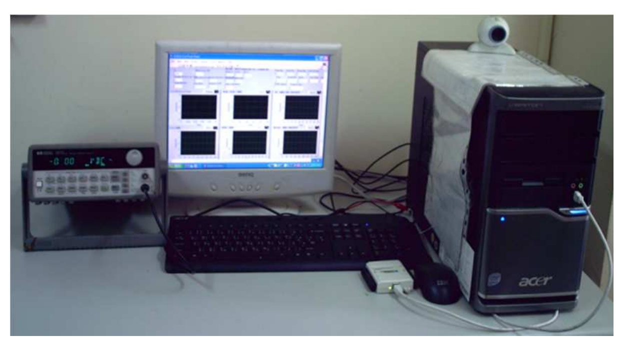

Fig. 6: The experimental set-up for frequency determination.

The simulated power signal is a 4-cycle signal which contains a fundamental signal with a frequency of 59.8 Hz,

, 40% of the third and 30% of the fifth harmonic and white noise with 1% deviation.

The signal was generated by HP 33120A arbitrary functional generator, which is instrumented through GPIB. Then the simulated signal was acquired through DAQ (NI USB-6009) and then stored in file. By using the Labview application software, the zero-crossing program (Figure. 7) was developed to determine the signal frequency.

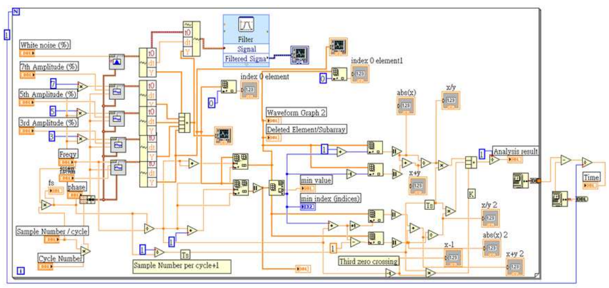

Fig. 7: LabVIEW program for zero-crossing algorithm.

In our experiment, the DAQ sampling frequency (f_s) is set at 15.308k Hz, and the signal acquired by the DAQ is shown in Figure. 8. The sine component was shown in Figure. 9 after the digital filtering procedure. Due to the fourier series transformation, the signal shows transient phenomena and it depent on the sampling points and the filter characteristic. The sine component with exact 3 cycles (Figure. 10) was captured from Figure. 9 to Engaged in the zero-crossing determination.. The N is calculated by Triple the points between two maximun peaks of the signal as described above.

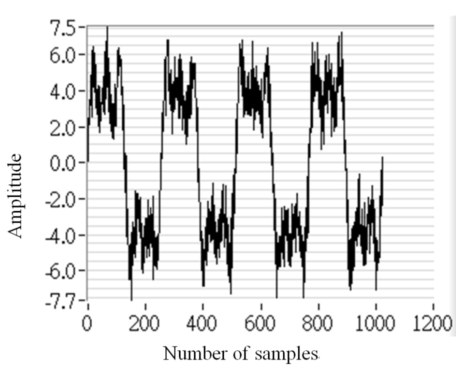

Fig. 8: Signal acquired by DAQ.

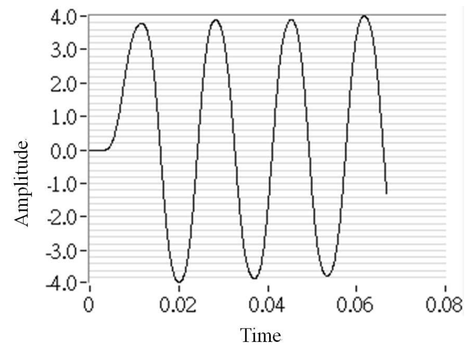

Fig. 9: Sine component extract from the signal acquired by DAQ.

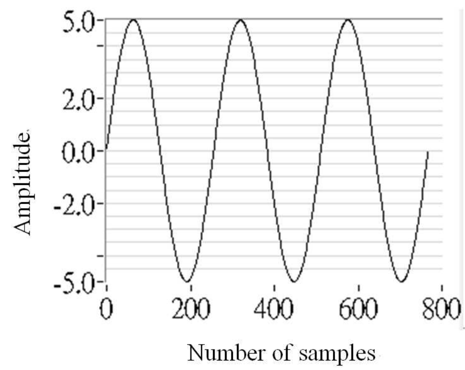

Fig. 10: 3-cycle Sine component extract from Figure. 9

As the sampling points was about 128 points for one period. The experiment results are shown in Table **??** and the error derivations were calculated and shown in Table 2. The experiment results approve the performance of zero crossing algorithm and the measurement accuracy is maintained.

Table 1: Experiment Results

Simulated Frequency. (Hz)	Calculated Frequency First meas.(Hz)	Calculated Frequency Second meas.(Hz)	Calculated Frequency Third meas.(Hz)	Calculated Frequency Fourth meas.(Hz)
58	57.9623	58.0992	58.0967	58.0906
59	59.0373	59.0291	59.018	59.0406
60	60.0068	60	59.9753	59.9865
61	60.9795	60.9772	60.964	60.9646

5 Conclusions

A simple approach to the zero crossing technique for the purpose of frequency determination of power signal is

Table 2: Experiment Results (derivation)

Simulated Frequency. (Hz)	Calculated Frequency First meas.(Hz)	Calculated Frequency Second meas.(Hz)	Calculated Frequency Third meas.(Hz)	Calculated Frequency Fourth meas.(Hz)
58	0.065	0.171034	0.166724	0.156207
59	0.0632203	0.049322	0.030508	0.068814
60	0.0113333	0	0.041167	0.0225
61	0.0336066	0.037377	0.059016	0.058033

presented. This approach demostrates a simple and powerful algorithm of an aceptable accuracy for freqency measurement. The Fourier algorithm is used for digital filtering in order to extract the cosine and sine parts of the fundamental frequency component. Then, the zero crossing technique is applied to the cosine or sine components of the signal. The derived algorithm is developed in a PC-based signal analyzing platform for its implementation. The computer is equiped with the DAQ device and LabVIEW program. Verification of the proposed algorithm has been carried out by performing the computer simulations, and then the laboratory experiment. As the acquired signal was stored in the databank of the program, zero crossing algorithm is implemented to determine the frequency of the power signal. A burst fundamental sinusoidal signal with its high order harmonic is stimulated by the signal generator to evaluate the accuracy for this developed system. This experiment can approve that the zero-crossing algorithm is an effective method for the frequency estimation. The results of the experiments confirmed that the algorithm could be a useful application for power system protection.

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