Proceedings of the Seventh IASTED International Conference
POWER AND ENERGY SYSTEMS
November 28 - December 1, 2004, Clearwater Beach, FL, USA

REAL TIME MEASUREMENT OF POWER DISTURBANCES – PART 2: IMPLEMENTATION AND EVALUATION

Tao Lin
Department of Electrical and Computer Engineering
University of Florida
Gainesville, FL32611
USA
tlin@ece.ufl.edu

Alex Domijan
Department of Electrical and Computer Engineering
University of Florida
Gainesville, FL32611
USA
alexd@ece.ufl.edu

ABSTRACT
The DL algorithm is an improved signal processing algorithm which can accurately measure the instantaneous electrical variables such as the amplitude, phase angle, frequency and power of the fundamental and harmonic components in power disturbance. This paper briefly reviews the definition and attributes of the DL algorithm, which is introduced in detail in part 1 of the two part series, and then presents the ways to implement the DL algorithm for practical application. Several case studies using practical power disturbances are presented to prove the effectiveness of the DL algorithm.

KEY WORDS
Signal processing, measurement, filter, recursive algorithm, power quality, harmonic

1. Introduction

Accurate and fast measurement of the instantaneous electrical variables in electric power and power electronic systems, such as amplitude, phase angle, frequency and power of the fundamental and harmonic components, forms the basis for power quality analysis, control apparatus, digital relays, event recorders, and modern power instruments / meters. Performance of the techniques used to measure the instantaneous variables directly determines the operation of these devices and their behavior under different service conditions.

The DL algorithm [1], which is proposed in part 1 of the two part series, overcomes the drawbacks of the Fourier Algorithm (FA, or called Fourier Filtering), the Continuous Wavelet Transform (CWT) and other signal processing techniques which rely on hypothetical signal models. The DL algorithm is not sensitive to frequency deviation, harmonic or noise interference. Moreover, the DL algorithm can be conducted recursively along time axis to update output at each sample, where the computational complexity is independent of the sampling frequency. These attributes are highly desirable for real time applications.

This paper presents the ways to implement the DL algorithm for practical use, and case studies using practical power disturbances to prove the effectiveness of the DL algorithm.

2. A Brief Review of The DL Algorithm

2.1 Definition of The DL Filter Bank

Based on a polynomial type real window function

$$g(t)=\left(\frac{\sigma^4t^4}{12}-\frac{\sigma^5t^5}{30}+\frac{\sigma^6t^6}{90}\right)\exp(-\sigma t)\,u(t) \tag{1}$$

which has low pass frequency property, the band pass DL filter bank (complex filter) are built up by introducing modulation [1]:

$$\psi_{a,\omega_n}(t)=\frac{1}{\sqrt{a}}\,g\left(\frac{t}{a}\right)\exp(j\omega_n t) \tag{2}$$

where $\sigma = 2\pi\big/\sqrt{3}$; frequency parameter $\omega_n = 2\pi f_n$, f_n is the nominal frequency of the signal component measured; parameter a is a positive scaling factor, which controls the widths of time window and frequency band of the DL filter bank, here a is 1/40.

Moreover, the band pass filtering based on the DL filter bank can be conducted recursively along time axis as

$$\begin{aligned} W_\psi(a,\omega_n,k) &= s(k) * \psi_{a,\omega_n}(k) \\ &= T\left(\sum_{m=1}^{6}\delta_m\, s(k-m)\right)-\left(\sum_{m=1}^{7}\lambda_m\, W_\psi(a,\omega_n,k-m)\right) \end{aligned} \tag{3}$$

where $s(k)$ is the sampled input real signal at time k ; $W_\psi(a,\omega_n,k)$ is the band pass filtering output at the same time; T is sampling interval; δ_m and λ_m are coefficients which are only related to parameters a , ω_n , σ and T .

Equation (3), which is termed the DL algorithm, updates filtering output at each sample using only six previous sample values and seven previous outputs. This recursive attribute results in a significant reduction of the computational burden and, the computational complexity is fixed rather than being proportion to the sampling frequency. Hence, the DL algorithm can ensure a real time accurate measurement of any signal component with time varying characteristics such as the fundamental component, integer harmonic and inter-harmonic components.

2.2 Measurement of Power Disturbances

Analogous to the FA and CWT, the DL filter bank extracts the rotation phasor vector in its frequency band from a non-stationary power disturbance waveform. Real part $W_R(k)$ and imaginary part $W_I(k)$ of the filtering output $W_\psi(a,\omega_n,k)$ can be used to measure the instantaneous phase angle $\theta(k)$, frequency $f(k)$ and amplitude $A(k)$:

$$W_R(k) = \mathrm{Re}\left(W_\psi(a,\omega_n,k)\right) = s(k) * \mathrm{Re}(\psi_{a,\omega_n}(k))$$
$$W_I(k) = \mathrm{Im}\left(W_\psi(a,\omega_n,k)\right) = s(k) * \mathrm{Im}(\psi_{a,\omega_n}(k))$$
$$\theta(k) = \tan^{-1}\left(W_I(k)/W_R(k)\right)$$
$$f(k) = \frac{1}{2\pi}\,\frac{\theta(k)-\theta(k-1)}{T}$$
$$A(k) = C_f\,(W_R{}^2(k)+W_I{}^2(k))^{1/2} \tag{4}$$

where C_f is a gain for amplitude calibration due to the possible frequency deviation. C_f is related to $f(k)$ and the amplitude frequency properties of the DL filter bank.

In a three-phase power system, the instantaneous symmetrical components can also be measured based on the filtering outputs of three phase voltages or currents:

$$\mathrm{V_P}(k) = \frac{C_f}{3}(W_A(a,\omega_1,k)+W_B(a,\omega_1,k)e^{j2\pi/3}+W_C(a,\omega_1,k)e^{-j2\pi/3})$$
$$\mathrm{V_N}(k) = \frac{C_f}{3}(W_A(a,\omega_1,k)+W_B(a,\omega_1,k)e^{-j2\pi/3}+W_C(a,\omega_1,k)e^{j2\pi/3})$$
$$\mathrm{V_Z}(k) = \frac{C_f}{3}(W_A(a,\omega_1,k)+W_B(a,\omega_1,k)+W_C(a,\omega_1,k)) \tag{5}$$

where $\omega_1 = 2\pi f_1$, f_1 is the rated frequency; $W_A(a,\omega_1,k)$, $W_B(a,\omega_1,k)$ and $W_C(a,\omega_1,k)$ are the instantaneous filtering outputs in phase A, B and C, respectively; $\mathrm{V_P}(k)$, $\mathrm{V_N}(k)$ and $\mathrm{V_Z}(k)$ are the instantaneous phasors of the fundamental positive, negative and zero sequence components, respectively.
The measurement of other instantaneous electrical variables such as active and reactive powers can also be derived according to this principle.

2.3 Broad Applications of The DL Algorithm

Accurate and real time measurement of the instantaneous electrical variables, such as amplitude, phase angle, frequency and power of the fundamental component, symmetrical component and harmonic components is the most basic and key issue in electric power and power electronic systems. Hence the DL algorithm has broad applications in these fields. For example, the DL algorithm can be used (but are not limited) in the following products or services to improve their performance and reliability:
1. Power quality analysis and on line monitoring;
2. Control apparatus for electric power and power electronic systems;
3. Digital protective relays and event recorders for electric power and power electronic systems;
4. Load shedding equipment;
5. Power meters and instruments;
6. Real time monitoring of power flow and harmonic power flow;
7. Condition monitoring of induction motors.

3. Implementation of The DL Algorithm

3.1 Implementation By Software

Most digital power instruments and meters, relays, and controllers are based on microprocessors and realize their measurement functions using the FA by software. Hence, the DL algorithm can be used as a software module and be easily substituted for the FA-based techniques (or other measurement algorithms) into these products or services by just replacing the relevant software module. The procedure for the amplitude calibration in (4) and (5) can be simplified by searching a pre-prepared gain table (an array in software) according to the measured instantaneous frequency. The gain table may be shared for monitoring different signal components.
As an example, the DL algorithm has been implemented in the Texas Instrument (TI) TMS320C6713 Development Start Kit (DSK) module [2]. This DSK module is a low cost standalone development platform that enables users to develop and evaluate their algorithms and applications. This DSK module is base on a TI TMS320C6713 floating point Digital Signal Processor (DSP) operating at 225 MHZ, users can download their applications to the DSK module and upload processing results through the USB port in host PC.
By using the DSP/BIOS [3], which is the real time operation system of TI DSPs, computation of the DL algorithm runs in a Periodic Object (PRD) and is completed within a 1ms time interval (the sampling interval can be changed flexibly). This procedure is illustrated by the Execution Graph (the lower left part) in Figures 1(a) and (b), which are the interface of Code Composer Studio (CCS) [4], the integrated development environment of TI DSPs. The waveform of a simulated

single phase power disturbance with time varying characteristics (sampling interval is 1ms in this study case) is displayed in the main window of CCS, which is the upper right part, in Figure 1(a). This waveform consists of a fundamental component, 20% 3rd and 20% 5th harmonic components; the actual frequency is 63.0 Hz rather than 60.0 Hz, which is the nominal value. Amplitudes of both the fundamental component and harmonics decrease to half of their initial values. As an output of the DL algorithm, the correctly measured amplitude of the fundamental component is presented in the main window in Figure 1(b), which reveals this serious change.

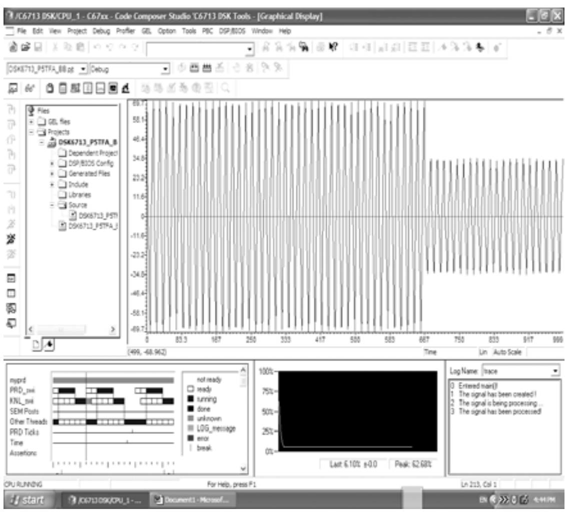

Figure 1(a). The simulated power disturbance waveform.

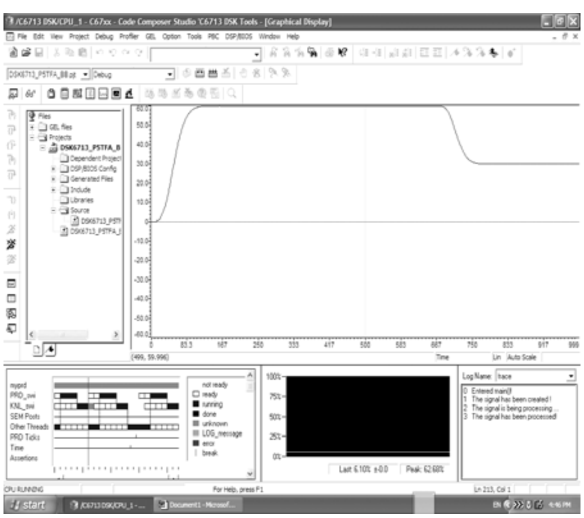

Figure 1(b). Amplitude of the fundamental component measured in real time using the DL algorithm.

3.2 Implementation By Hardware

Because the procedure for the measurement of any signal component in power disturbance waveforms using the DL algorithm is simple and is the same except for the filter coefficients in (3), the DL algorithm can be used for monitoring different signal components using standard hardware, such as FPGA (Field Programmable Gate Array) or CPLD (Complex Programmable Logic Device) which has floating point computation units. The coefficients and gain table for amplitude calibration can be made off-line and be downloaded to the hardware or additional memory like Flash RAM before use. In one hand, the cost can be reduced, in the other hand, the processing speed will rise. The standard hardware and the coefficients / gains memory (if necessary) form an electrical signal processing hardware module. When many more signal components in an electrical signal need to be measured, it can be easily realized by just adding more shunted hardware modules, which share the same data acquisition output.

4. Evaluation of The DL Algorithm

4.1 The Practical Power Disturbances

Performance of the proposed DL algorithm is evaluated using practical power disturbance waveforms. These power disturbances are recorded in 1999 and 2000 as the data benchmarks of existing power quality events in the distribution system in Premium Power Park (PPP) [5], where is in Columbus, OH. Remote power quality monitors were installed at multiple locations within the PPP, such as the point of common coupling (PCC) to different power users and substation 13.2 KV feeder exits. The monitors used are BMI 7100 PQ Nodes, sampling frequency is 7680 Hz (128 samples per cycle), the recording duration is 16-32 cycles (2048 to 4096 samples). The waveshape recording of disturbances / faults provides a simultaneous record of all phase voltages and currents during all significant power disturbances. These recordings provide important information about the nature of each disturbance and can be used to identify the cause of a disturbance and location of a fault. 193 cases are selected to evaluate performance of the DL algorithm using Matlab and DSP programs, and two of them are disclosed here to prove the effectiveness of the DL algorithm and show its advantages to the FA and CWT. It should be pointed out that initial values of the DL algorithm outputs, the derived amplitude and frequency are zero, hence a transient exists in the beginning of the amplitude or frequency estimates because the input of a disturbance waveform to the DL filter at 0.0s can be considered as the input of a step signal to an amplitude estimator. This transient should be ignored or discarded for evaluating the DL algorithm.

4.2 The Evaluation Results

1. Case 1: The first case used for evaluation was recorded on May 20, 2000. The voltage waveform in phase C is shown in Figure 2(a), in which there are slight amplitude

fluctuation and harmonics. The frequency and amplitude of the fundamental component in this disturbance waveform are estimated using the DL algorithm and the FA, respectively. The results are shown and compared in Figures 2(b) and (c). The DL algorithm output (the thick solid line) reveals a slight frequency deviation in the fundamental component, which is about 60.1 Hz in most of times and a slight transient occurring after 0.26s; in contrast, the FA frequency estimate (the thin dashed line) oscillates between 59.5 and 60.5 Hz irregularly, and even drops to about 57.2 Hz after 0.26s. Obviously, the FA frequency estimate is severely affected by frequency deviation and harmonics and hence cannot be accepted as actual value, the DL algorithm output is much more reliable. Similar situation also exists in the amplitude estimation in Figure 2(c), although the dynamic response of the DL algorithm is a bit slower than the FA in time domain due to the relatively longer window $g(t)$, the amplitude estimate is much more stable than the FA output and hence can be used to represent the nature of this voltage disturbance, which has time varying characteristics. The dynamic response of the DL algorithm can be improved by choosing a smaller a.

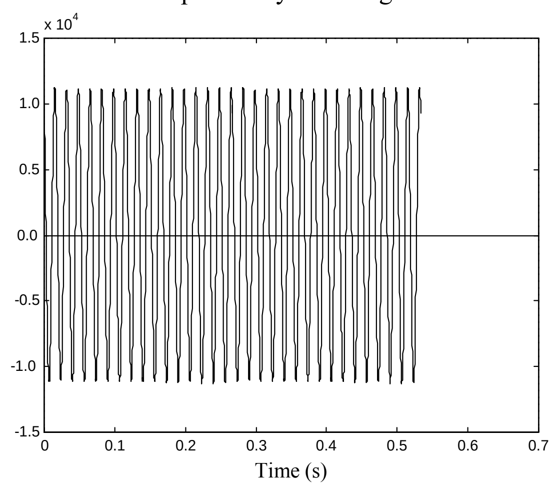

Figure 2(a). The practical voltage disturbance waveform.

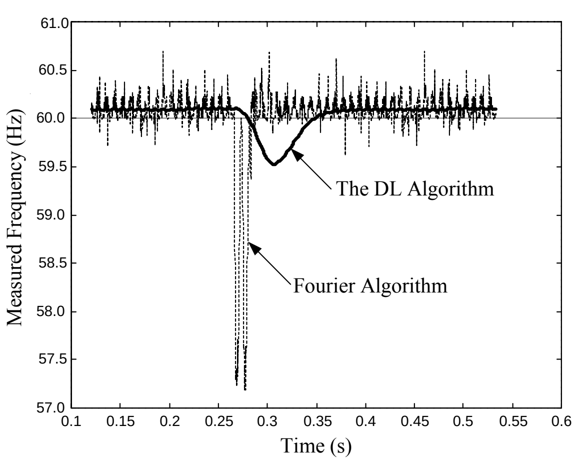

Figure 2(b). The measured fundamental frequency.

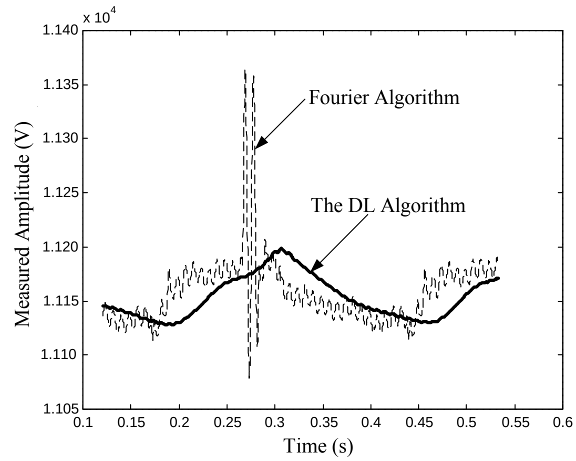

Figure 2(c). The measured amplitude of the fundamental component.

Similar comparison is also made in harmonic measurement. The 3rd and 5th harmonics in the voltage disturbance are estimated using the DL and Fourier algorithms and are shown in Figures 2(d) and (e), respectively. Obviously, the DL algorithm provides a reasonable estimation of these time varying harmonics while the FA output oscillates irregularly in a wide range. In addition, the CWT, which is based on the well known Morlet wavelet, is also used for comparison. Amplitude estimate of the 3rd harmonic component using the CWT (the thin solid line) is shown in Figure 2(d) together with the DL algorithm and FA outputs. It can be proven that the CWT estimate is severely affected by other signal components in the disturbance waveform such as the fundamental component and other harmonics, the CWT estimate even cannot converge.

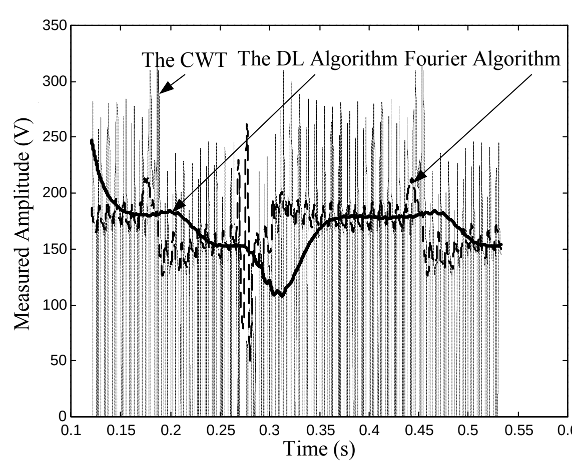

Figure 2(d). The measured amplitude of the 3rd harmonic component.

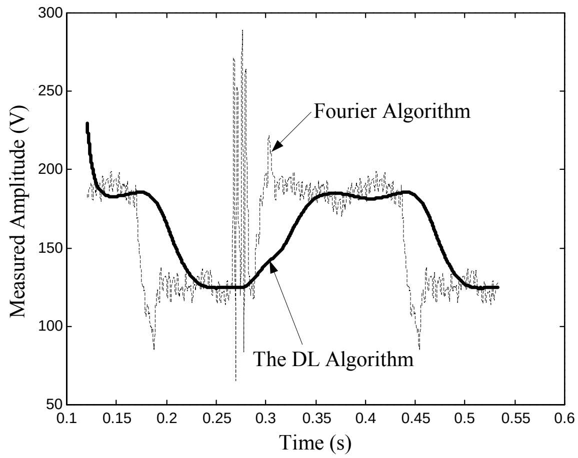

Figure 2(e). The measured amplitude of the 5th harmonic component.

2. Case 2: The second case used for evaluation was recorded on Oct. 2, 2000. The waveforms of three phase currents are shown in Figure 3(a), in which there are slight imbalance and severe harmonic distortion. Like case 1, a slight frequency deviation, which is 60.1 Hz, is revealed by the DL algorithm and is shown is Figure 3(b) while the FA frequency estimate oscillates irregularly between 59.3 and 60.9 Hz. The fundamental positive, negative and zero sequence components in these disturbance waveforms are measured using the DL algorithm and the FA. The estimated amplitude of these components are shown and compared in Figures 3(c), (d) and (e), respectively. Obviously, the DL algorithm outputs are reasonable and much more stable than the FA estimates, which oscillate in a wide range. Hence, the DL algorithm is very helpful to correctly reveal the nature and degree of this non-stationary imbalance disturbance under frequency deviation and severe harmonic distortion circumstance.

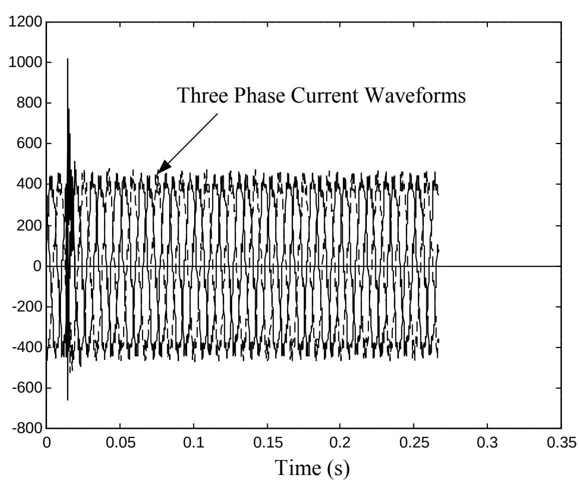

Figure.3(a) The practical power disturbance waveforms.

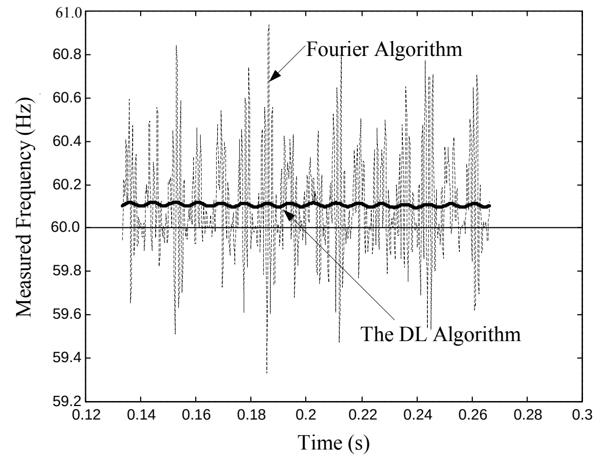

Figure 3(b). The measured fundamental frequency.

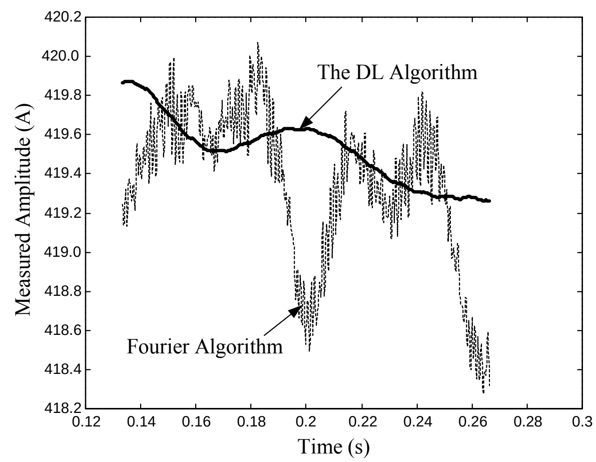

Figure 3(c). The measured amplitude of the fundamental positive sequence component.

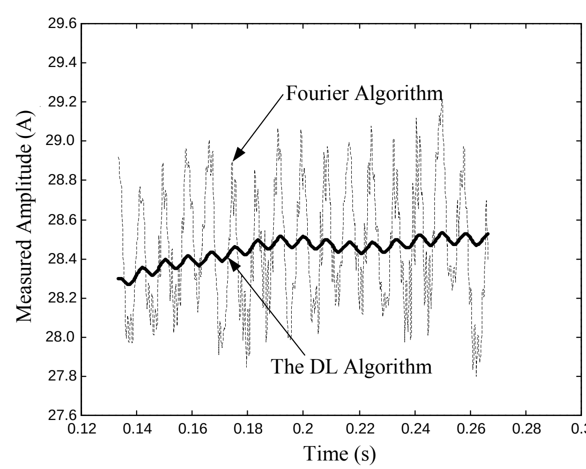

Figure 3(d). The measured amplitude of the fundamental negative sequence component.

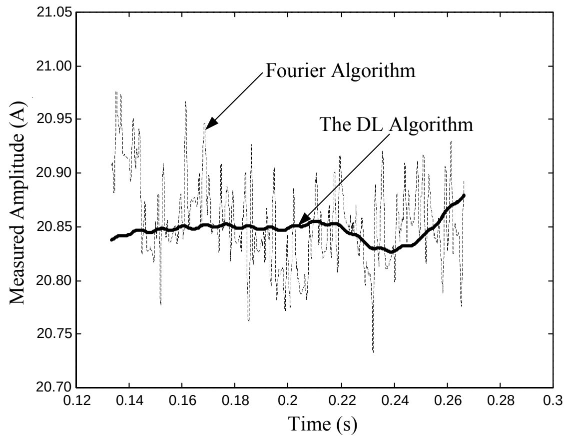

Figure 3(e). The measured amplitude of the fundamental zero sequence component.

5. Conclusion

The DL algorithm can accurately measure the instantaneous electrical variables such as amplitude, phase angle, frequency and power of any time varying signal component in a non-stationary power disturbance, hence it can be broadly used in electric power and power electronic systems. Case studies base on the practical power disturbances prove an excellent performance of the DL algorithm, which is not sensitive to frequency deviation, harmonic interference, or noise contamination. Moreover, the DL algorithm can be conducted recursively and its computational complexity is independent of the sampling frequency. Hence, the DL algorithm is easy to implement by software or hardware.

6. Acknowledgements

The authors would like to acknowledge the National Science Foundation (NSF) for the financial support of this research under grant ECS 014636.

References:

[1] T.Lin, A.Domijan, Real time measurement of power disturbances---Part1: comparative study and a novel complex filter. *Proc. 7th IASTED Int. Conf. on Power and Energy Systems*, Tampa, FL, USA, 2004.
[2] *TMS320C6713 DSK technical reference* (TX: Spectrum Digital, 2003).
[3] *TMS320C6000DSP/BIOS user's guide* (TX: Texas Instrument, 2000).
[4] *Code Composer Studio user's guide* (TX: Texas Instrument, 2000).
[5] *Premium power park (PPP), phase 1 -- Application methodology* (CA: EPRI and OH: American Electric Power, 2000).