SYNOPSIS

MICROPROCESSOR BASED DIGITAL SIGNAL PROCESSING FOR ELECTRONIC INSTRUMENTATION

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By

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Developments in microprocessors and microcomputers in recent years have aroused keen interest amongst scientists and engineers for their application in monitoring, control and protection of electrical/electronic networks. In most of the electrical/electronic networks, the fundamental frequency wave of voltage and current include noise with wide spectrum. This level of noise increases when disturbances in the networks occur. Therefore, it is desirable to develop a digital technique to estimate the fundamental component of voltage, current, power, frequency deviations, etc. in the presence of noise in off-line and real-time conditions. Also the measurements of both positive - and negative - sequence voltages, current and power (active and reactive) in an electrical network assume importance in evaluating system losses and performances during unbalanced operating conditions.

Many alternative algorithms have been suggested in present years for this purposes. A powerful algorithm using Kalman filter [9] has been recently proposed for optimal estimation of voltage phasors frequency and rate of change of frequency in electrical networks, which yields a fairly accurate estimate of the measurable quantities.

Since proliferation of electronic control of equipments by semiconductor devices generate harmonics, every major industrial process is, therefore, subject to the presence of harmonics. These devices inherently display a poor power factor which is compensated by the addition of power capacitors which, in turn, may create a parallel resonant condition with the utility source. Because of network pollution by harmonics and with the complexity of modern installation, it is essential to have a harmonic study performed in most instances.

The other important parameters in electrical/ electronic networks are their resistance and reactance (inductive or capacitive or both). Under normal condition of operation of the networks, the R and X are found to be steady with time, but under abnormal conditions they are found to vary drastically and these drastic changes can be used to detect any abnormal situation in an electrical network.

Except for the harmonic estimations, the most important factor for this instrumentation scheme is to extract the fundamental component of voltage and current from the signal corrupted with noise. Keeping this in view, two recursive algorithms have been proposed for computation of voltage, current, powers, positive - and negative sequence voltages, phase angle, apparent impedences and harmonics present in an electrical networks. Energy estimations of the inrush and fault currents in a transformer under open and loading conditions have been done digitally by an auto-correlation method which can be computed by additions only. Both these algorithms are based on generation of discrete Fourier coefficients from input data samples.

The Spectral observer (algorithm No.1), which is an interpolating observer, performs a recursive discrete Fourier transform where progressive incorporation of a new sample, discarding the effect of the oldest sample as in running Fourier transformation, requires only one iteration. There is no restriction to the sampling rate, the non-Nyquist rate of sampling can easily be accommodated. Non-uniform sampling rate can be easily incorporated without much of computational overhead. Errors due to truncation and round off may be corrected by recirculating the signal samples. This algorithm is also useful for unevenly spaced data samples. By placing the observer eigenvalues at positions other than the origin of the complex plane, the speed and the performance of the observer can be changed.

The recursive Functional expansion (algorithm No.2), which is based on the theory of discrete-time filter, provides a simple yet a powerful general program for efficient recursive computation of arbitrary discrete linear transform. Each filter iteration produces a completely new set of nth order transform coefficients based upon the most recent n input samples. The solution is similar in structure to recursive least square but, in contrast, develops the exact solution to the most recent n linearly independent equations rather than a 'best fit' of all previous equations. There is no restriction on the basis function except finiteness and the recursive computations are ongoing. Sampling rates other than the Nyquist rate of sampling can be easily accommodated.

The fast computational processing capability of these two algorithms make them suitable for microprocessor applications. Software programs have been developed for digital instrumentation scheme, for estimation of harmonics and calculation of apparent impedances in electrical/electronic networks, and they have been tested in an LSI-11/23, 16-bit microcomputer in off-line and real-time conditions. For digital instrumentation scheme, the soft-ware was also developed in Macro-assembler language. Further, the decaying d.c. terms have been incorporated in the signal model in most of the situations. Energy estimation of inrush currents in a transformer in open and loading condition has been done by an autocorrelation process which is fast and involves only addition. Variation of noise level, placement of poles of the observer, sampling rates have been highlighted.

The results clearly indicate that the performances of the two techniques are quite suitable for microprocessor application and the results are accurate and fast. These algorithms produce correct results, for harmonic estimation, within 3/4 of a cycle based on 60/50 Hz waveform. The measurements of apparent impedances in electrical transmission networks show that the fault detection can be achieved with 1/2 to 3/4th of a cycle based on 60/50 Hz waveform.

Though the spectral observer algorithm does not rival the speed of computation of fast Fourier transform (FFT) for complete transformation, it is especially attractive for progressive incorporation of data samples, as in running Fourier transform.