CHAPTER - VI

CONCLUSIONS AND FUTURE DIRECTION

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6.1. CONCLUSION

This thesis makes use of two recursive algorithms, known as Spectral observer and Functional expansion, for digital instrumentation schemes, estimation of harmonics and measurements of R and X of electrical or electronic networks.

Interpolation of signal samples with harmonic trigonometric interpolating function is equivalent to a discrete Fourier transform. An interpolating observer produces samples of the correct interpolating function, from which Fourier Coefficients may be determined. For convinience, harmonic triogonometric interpolating observers are termed Spectral Observer . Spectral observers perform a recursive discrete Fourier transform where progressive incorporation of new samples, discarding the old as in running Fourier transformation, requires only one iteration. These filters are especially useful for sampling rate other than the Nyquist rate. The response of these interpolating observers to samples corrupted by noise is clear : noisy or not, the samples are interpolated [13]. There is the possibility of placing the observer eigenvalues at positions other than the origin of the complex plane to achieve a different noise performance. Non-deadbeat observation will modify the updating behaviour of the interpolator to one involving a fading memory of all previous samples. Non-uniform sampling rate can be easily incorporated into the observer without much computational overhead, but with the availability of multiple and fast processors and floating point micro-chips the computational over heads for unevenly spaced samples gets reduced. Imprecision which results in placement of the observer eigenvalues at positions other than the origin may be reduced asymptotically to zero by recirculation of the data provided that the error is not so great as to result in system instability [15]. By varying the observer gains, it is also possible to generate signal estimates with successively higher order harmonic content, if desired. This spectral observer is well suited to a state variable format.

This proposed interpolating Spectral Observer is a powerful and versatile method for computing voltage, current, phase angle, active and reactive power, positive and negative-sequence voltage in an electrical or electronic network. It is also quite suitable for computing apparent impedences and harmonic contents from digitized voltage and current signals. Measurements of voltage, currents, etc. from signal samples placed arbitrarily in time result in inexpensive A/D converter without sample and hold device. Further, for computing apparent impedences and harmonic contents in a signal, there is freedom in choosing the signal model and some flexibility in selecting the size of the data window and time reference for fault computations. This freedom of choosing the system model allows prespecified harmonics and decaying d.c. components. Round off and truncation errors are taken care of by recirculating the signal samples.

For harmonic estimation Spectral observer yields accurate results within 3/4 of a cycle based on 60/50 Hz waveform. In case of measurements of apparent impedences in an electrical network under fault condition, the fault detection can be achieved within 1/2 to 3/4 of a cycle based on 60/50 Hz waveform. Though this Spectral observer algorithm does not computationally rival the fast Fourier transform for complete transformation, it is especially attractive for progressive sample incorporation, as in running (or moving window) Fourier transformation.

The recursive Functional expansion algorithm, is based upon discrete-time filter theory, provides a simple yet powerful general program for efficient recursive computation of arbitrary discrete linear transforms.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 recrusive least squares 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 are no restriction on the basis function except finiteness. The recursive computations are ongoing. This technique is equally versatile in computing voltage, current, active and reactive powers, etc. for digital instrumentation scheme, and for estimation of apparent impedances and harmonic contents of an electrical/electronic networks. This algorithm also produces correct results, for harmonic estimation, within 3/4 of a cycle based on 60/50 Hz waveform. In case of electrical transmission networks, the measurements of apparent impedence show that the fault detection can be obtained within 1/2 to 3/4 of a cycle based on a 60/50 Hz waveform. There is a clear, non-catastrophic indication at any step when a signal sample is linearly dependent upon previously used samples (or nearly so), i.e., when a sample yields no new information about the transform Coefficients [12]. Sampling rate other than the Nyquist rate can be suitably accomodated in this algorithm.

The fast computational capability of these algorithms make them suitable for microprocessor applications. These algorithms have been tested in off-line and real-time conditions using an LSI-11/23 microcomputer system alongwith a data aquisition interface. The software for the digital instrumentation scheme is developed in Macro assembler and a second order recursive Butterworth low-pass digital filter program is used to filter the laboratory samples. It is found that better filtering can be achieved with a higher order recursive Butterworth low-pass digital filter. Further, this thesis presents the results of both filtered and unfiltered laboratory data showing the oscillatory tendency of the measured voltage and current phasors. The sampling rate and the placement of observer poles have a significant impact on the instrumentation scheme. Higher the sampling rate, better is the performance of the digital-filter, at the cost of more computational overheads in real-time implementation. To vary the sampling rate a programmble clock is used for real-time work. It is observed from both the theoritical and practical applications that an ideal sampling rate for digital instrumentation scheme is 12 per voltage and current in an electrical/electronic network. For transformer and electrical transmission line network the ideal sampling rate is 12 based on 60/50 Hz waveform. The variance of noise and its Gaussian distribution has a significant impact on the measurement technique. For the Spectral observer, the observer pole at -.5 achieve the optimal result in suppressing the noise components.

An important aspect of this processing in electrical/ electronic networks is the inclusion of sub-harmonic and d.c. components in the signal model. Such signals occur in electrical transmission networks, in power transformer network, electronic/electric circuits with switched capacitors and inductors. These filters can filter these sub-harmonic and d.c. components and yield accurate results for signal processing and measurement.

6.2 FUTURE DIRECTIONS

These filters need further refining for real-time implementation in situation requiring fast computations. The computational burden seems to be higher for practical electrical/electronic systems and requires the use of a fast processor. Also, detailed studies are required for unevenly spaced samples and multirate sampling. The noise rejection performance of these filters should be improved for practical case studies. Frequency deviation can be estimated by these filters to study the system instability in electrical networks.

Field tests are needed for final acceptance of these techniques for practical digital instrumentation schemes.