# Chapter 5
# Phasor Measurement Improvement Using Digital Filter in a Smart Grid

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## ABSTRACT

*During a transient operation condition of power smart grid, line current may include unwanted components that may cause unnecessary tripping of protection system. The disturbance mainly appears in a form of harmonics and sub-harmonics. In this case of signal waveforms including harmonics, the low pass filter may be used. However, this type of filter does not provide the ability to reject sub-harmonics. This chapter presents the digital filtering design issue based on optimization approach for removing sub-harmonics and hence improving the measurement. The first point of view is to reach an unified accurate phasor measurement algorithm that is immune to nearly all disturbances (sub-harmonics) in power grid including FACT devices and renewable energy sources, simultaneously with required speed of convergence. The second point focuses on reducing the computational requirement and algorithm complexity through designing recursive digital filter with reduced order.*

## INTRODUCTION

An electric power system is a large interconnected system that produces, transmits and distributes an electric energy to different consumers. Stability of the smart power system is of a great concern, since it is subjected to different disturbances that may cause false measurement and hence mal-operation of protective and control systems which may lead to complete system collapse. Therefore, many techniques have been developed to mitigate the disturbance effects by introducing the digital filtering for computing

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the fundamental phasors of line voltages and currents. Several digital algorithms have been proposed in the literature for estimating the fundamental phasors, such as the discrete Fourier transform, cosine algorithm, least square algorithm, Kalman filter and wavelet transform. The discrete Fourier transform (DFT) filter which has proven reliability and robustness in relaying application was considered as the most widely used and most popular algorithm for estimating phasors. However, during a transient time of smart grid condition, line fault current may include unwanted components that are associated to the fault impedance level as well as to the power transmission line type. The first type of disturbance mainly appears in a form of dc exponentially decaying component associated with the line fault current signal waveforms (Boutora, S., Bentarzi, H., & Ouadi, A., 2011). In this case of signal waveforms including dc components, phasors estimated through application of the DFT filter may have large errors during the first cycles. This problem stems from the fact that the signal containing dc exponential decaying components has non integer harmonics (using the Fourier transform of this dc component) while the DFT filter does not provide the ability to reject non integer type of harmonics. The second kind of disturbance that may occur in power transmission line including series compensation devices in presence of relatively high fault impedance, where a significant distortion is introduced in the faulted line current waveforms. This is due to the behavior of non-linear components integrated in series compensated devices. A considerable exponentially decaying sub-harmonic or sub-synchronous resonance is associated with fundamental and integer harmonic of the line current waveform. This type of fault is more severe since it reduces the speed of convergence of the DFT algorithm to correctly estimate the fundamental phasors.

The digital filtering design issue based on optimization approach is aimed to achieve different perspectives. The first point of view is to reach a unified accurate phasor measurement algorithm that is immune to nearly all disturbances in power grid, (including other type of FACT devices) introducing considerable and different kind of waveform distortions affecting both line voltage and current signals, simultaneously with required speed of convergence. Although, only the line current waveform distortion filtering is of concern and considered in this work. The second point, focus on reducing the computational requirement and algorithm complexity through designing recursive digital filter with reduced order.

Digital filters exist in two types: Finite impulse response (FIR) and Infinite impulse response (IIR) or recursive. FIR filters suffer from the problem of high order (hence implementation and performance issues) if strict requirements are imposed at the design stage. Furthermore, IIR filters can have smaller group delay than its equivalent FIR filters (Ouadi, A., Bentarzi, H.; & Recioui, A., 2014; Ouadi, A., Bentarzi, H., & Zitouni A.,2018). The optimal design of an infinite impulse response (IIR) filter consists in choosing a set of coefficients of the filter to have a frequency response that optimally approximates the desired response (Ouadi, A., Bentarzi, H.; & Recioui, A., 2013; Kaur, R. Dhillon, J.S. Singh, D., 2010; Kezunovic, M., & Kasztenny, B., 2000).

Different techniques exist for the design of digital filters. Windowing method; in which the ideal impulse response is multiplied by a window function, is the most popular. There are various kinds of window functions (Butterworth, Chebyshev, Kaiser etc.), depending on the requirements on ripples in the passband and stopband, stopband attenuation and the transition width. These various windows limit the infinite length impulse response of ideal filter into a finite window to design an actual response. Furthermore, windowing methods do not allow sufficient control of the frequency response in the various frequency bands and other filter parameters such as transition width. The designer always has to compromise between the design specifications (Ouadi, A., Bentarzi, H.; & Recioui, A., 2014; Ouadi, A., Bentarzi, H., & Zitouni A.,2018).