Electrical Power and Energy Systems 56 (2014) 140–146

Contents lists available at ScienceDirect

Electrical Power and Energy Systems

journal homepage: www.elsevier.com/locate/ijepes

Selective estimation of harmonic components in noisy electrical signals for protective relaying purposes

J. Lázaro [a,*], J.F. Miñambres [b], M.A. Zorrozua [b]

[a] Department of Applied Mathematics, Faculty of Engineering (Bilbao), University of the Basque Country UPV/EHU, Spain
[b] Department of Electrical Engineering, Faculty of Engineering (Bilbao), University of the Basque Country UPV/EHU, Spain

ARTICLE INFO

Article history:
Received 22 October 2012
Received in revised form 17 October 2013
Accepted 6 November 2013

Keywords:
Digital filtering
Digital relaying
Noisy signal
Phasor estimation algorithm

ABSTRACT

Digital filters used in protection relays must comply with a series of requirements. Among its most important features are: low computational load, good behaviour in the presence of harmonic and decaying dc components, accuracy against noise in the input signal and good behaviour in the transition from steady state to fault period.

Before implementing a digital filter in a protection relay, it is necessary to subject it to stringent tests simulating a large array of real fault conditions. Electrical signals undergo a series of transformations before reaching the digital filter. As a result of this, the analysed signal carries errors and noise with it. The filter will only be apt for use in a digital relay if it rapidly and precisely responds to these types of signals. The objective of this paper is to present a new digital filter for selective estimation of harmonic components in noisy electrical signals for protective relaying purposes.

1. Introduction

All power systems must be provided with a protection system that effectively contributes to the reliability of electrical energy supply. Different technologies have been developed throughout time with the aim of providing protection relays with the capacity to carry out two basic tasks. On the one hand, they must be able to detect any faults occurring in the power system, while on the other hand, they must be able to give the necessary orders to isolate the faulty circuit and avoid the fault propagation. Both tasks must be carried out in as short a time as possible to minimise the impact of the fault on the power system.

Digital technologies along with the need to optimise resources have currently led to the development of interesting additional features, such as fault location and data recording. Due to this, digital relays have become a fundamental tool for protecting power systems, thanks to their multi-functional features.

In a digital relay (Fig. 1), signal processing can be divided schematically into the three following stages:

- *Measurement and preconditioning of signals*: The analogical waves, from the Current Transformer (CT) and Voltage Transformer (VT), pass through an antialiasing (low pass) filter that eliminates the high frequency components. Then, an Analogic Digital Converter (ADC) provides the discrete samples that will be numerically processed in the following step.
- *Digital filtering of the sampled signals*: This process is carried out by a Digital Signal Processor (DSP). The digital filter implemented in the DSP estimates the harmonic components (in most cases the fundamental component) that will be used by the protection functions.
- *Protection functions*: A set of logical and mathematical criteria which, based on the output data of the digital filter, determine the correct decision to be taken by the relay.

The protection relay must take the correct decision in the shortest possible time. Therefore, two fundamental features of the digital relay are precision and speed in its response, especially during the transient period of the input signal. The key element to achieve both features is the digital filter.

Typical algorithms used in digital filters to carry out phasor estimation are based on the Discrete Fourier Transform (DFT) [1,2]. But signals containing aperiodic components produce significant oscillations in the output of the DFT. Due to the fact that decaying dc-offset components are very common in transient periods, several authors have proposed modifications in the DFT to improve its convergence [3–7]. These methods are based on estimating and eliminating the exponential component and then applying the DFT to the resulting signal. One important drawback of these methods is that they require knowing the fault instant. As a consequence, an additional algorithm is needed to detect the

* Corresponding author. Address: Faculty of Engineering, Alda. Urquijo, s/n, 48013 Bilbao, Spain. Tel.: +34 946014150.
E-mail address: jorge.lazaro@ehu.es (J. Lázaro).

0142-0615/$ - see front matter © 2013 Elsevier Ltd. All rights reserved.
http://dx.doi.org/10.1016/j.ijepes.2013.11.007

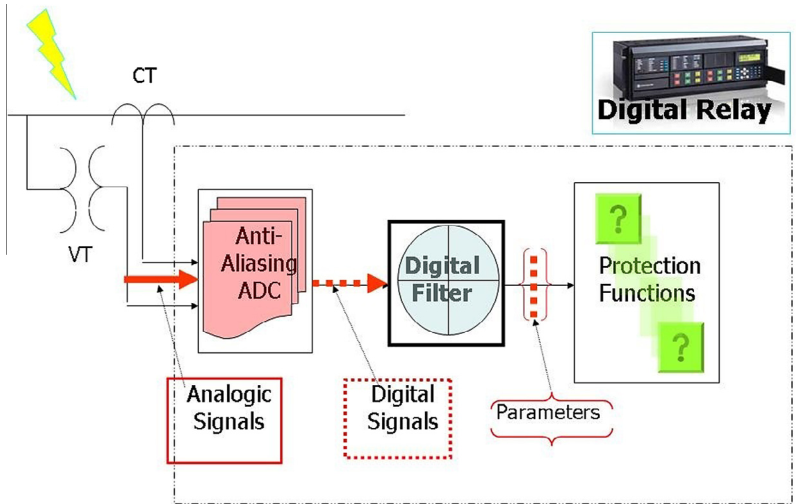

Fig. 1. Diagram showing signal processing in a digital relay.

fault instant. By this reason, these methods have the disadvantage of significantly increasing the computational load [8,9].

Other methods for phasor estimation are based on the Least Error Square (LES) technique [10–13]. In these methods, a system of equations is defined according to the components considered a priori in the analysed signal. The greater the number of components considered, the greater is the precision but also the greater is the computational load. Different solutions have been proposed to limit this computational load. Some authors propose the use of different algorithms according to the status (steady state/transient) of the input signal. However, this option requires knowing the fault instant [10], which has the disadvantage of having to implement an algorithm for fault detection. Other authors propose an application of the LES method to a limited set of harmonic components [11], which means losing precision in the results when the input signal is more complex.

This paper proposes a new digital filter that can be implemented in digital relays. Its basic characteristics are the following:

- *Precision*: The accuracy provided in the output enables the use of the proposed filter in digital relaying.
- *Low computational load*: The mathematical techniques involved are well below the limits imposed by the DSP hardware for real time applications.
- *Independence of the fault instant*: The proposed digital filter does not require knowing the fault instant, as it is applied in the same manner in all possible conditions (steady state/transient, pre-fault/fault, all fault types).

In addition, it is also necessary to guarantee the correct performance of the filter in the presence of distortions due to errors of discretization and electromagnetic noise. In the case of the ADC [14,15], the errors depend on the maximum desired range of measurement and the number of bits of the ADC. Moreover, there are additional errors such as: offset error of the analogical circuits, measurement error of the internal transformers of the relay, error of derivation by temperature, electromagnetic noise and others.

These errors of internal origin are random and difficult to predict. Therefore, the existence of distortions in the signals processed is inherent to the operation of all the digital relays. Consequently, the response of a digital filter must be both fast and accurate in the presence of noisy signals and decaying dc-offset in order to be successfully used for protective relaying purposes.

2. The new filtering technique: s-CharmDF definition

The proposed method is a CharmDF-based modified digital filter. In comparison to the CharmDF [16,17], the proposed filter improves precision of results due to its flexibility of application. This flexibility enables it to optimise the estimation of any harmonic component.

To illustrate the definition of this new digital filter a generic signal $y(t)$ is used. This signal has the typical characteristics of electrical signals incoming to digital relays. According to the existing technical literature on the subject [18–21], the following signal is considered:

$$y(t) = C_0 + Ce^{-t/\tau} + \sum_{r=1}^{n} A_r \cos(\omega_r t + \alpha_r) + z(t) \quad (1)$$

where C_0 is a constant offset, C and τ are the amplitude and time constant of decaying dc offset, A_r and α_r are the amplitude and phase of the fundamental ($r = 1$) and harmonic components and, finally, $z(t)$ is the noise component of the signal.

The proposed methodology is based on defining an auxiliary wave $x_s(t)$ that contains all the information regarding the periodic components of $y(t)$. The auxiliary signal $x_s(t)$ has been denominated s-Characteristic HARMonic wave (s-Charm wave) due to the fact that it is defined using samples of $y(t)$ separated by a sample slip of s samples.

$$x_s(t) = f_s(t) + \sum_{r=1}^{n} X_r \cos(\omega_r t + \beta_r) \quad (2)$$

where $f_s(t)$ is a residual signal that includes the influence of aperiodic component and noise. Moreover, X_r and β_r are the amplitude and phase of the fundamental ($r = 1$) and harmonic components of the s-Charm wave.

The definition of the s-Charm wave implies a biunivocal relationship between the harmonic spectra of signals $x_s(t)$ and $y(t)$. For an original signal recorded with a sampling rate of N samples per cycle, the relationship between the respective amplitudes and phases can be expressed through the following correction factors:

$$Ks_r = \frac{A_r}{X_r} = \frac{1}{2\sin\left(\frac{180sr}{N}\right)} \quad (3)$$

$$\theta s_r = \beta s_r - \alpha_r = 90\left(\frac{N - 2sr}{N}\right) \quad (4)$$