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Method based on independent component analysis for harmonic extraction from power system signals

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ABSTRACT

This paper presents a new application of independent component analysis for harmonic component extraction from power system signals (voltage and current). The harmonics to be extracted can be time varying and the method does not require synchronous sampling, which means it is able to work in off-nominal frequency. The proposed method has shown to be simple in the operational stage. The method was tested using both simulated and real signals, and performance was evaluated using measures in both frequency and time domains. Results were compared with another method available in the literature.

1. Introduction

The increase of nonlinear loads and power electronic equipment in the power system results in high levels of distortion in the power system signal caused mainly by harmonics [1]. The presence of harmonics may cause serious problems, both for consumers and utilities. Therefore, the analysis of harmonics in the electrical signal becomes extremely important before taking actions to eliminate or mitigate the problems caused in the power grid.

Moreover, the extraction of fundamental and harmonic components from the electrical signal becomes indispensable in control strategies for inverters connected to the grid or for active compensation such as the Shunt Active Power Filter (APF) [2]. Several standards address this issue for the purpose of monitoring harmonic limits and protection of industrial electrical systems. The IEC (International Electrotechnical Commission) standardizes the technical test norms and measurement of harmonics and other PQ disturbances [3]. The IEEE (Institute of Electrical and Electronics Engineers) specified in IEEE Std 1531-2003 components, protection and control of harmonic filters [4]. In IEEE Std 519-1992, limits of harmonics and methods for controlling them are determined [5].

In this context, several techniques have been developed in recent years for harmonics analysis. When the analyzed signal is stationary and the sampling frequency is an integer multiple of the fundamental frequency (synchronous sampling), the algorithms used are based on the Fourier transform [6], since they lead to minor estimation errors. However, these methods lose accuracy when the signal is non-stationary or has non-synchronous sampling.

In power system signal analysis, non-stationary signals are those that experience some type of disturbance like sag, frequency variation, transients and time varying harmonic. To deal with non-stationary signals, new methods have been developed based on short time Fourier transform (STFT) [7], wavelets [8], filter banks [9], Phase-Locked Loop (PLL) [10], among others [11]. The STFT and wavelets are special cases of filter banks. STFT uses a set of filters with same bandwidth which generates a complex output signal whose magnitude corresponds to the amplitude of the harmonic components. The main disadvantage of this method is the difficulty in efficiently designing a band-pass filter that minimizes

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the spreading spectrum effect. Though the wavelet transform uses filters with real coefficients, the common wavelet mothers do not have good magnitude response and the traditional binary tree structure of discrete wavelet transform (DWT) is not suitable for separating the harmonic components. In this case the Wavelet Packet can be used, but to obtain good harmonic separation a high order wavelet must be used and consequently higher convergence time is expected. The Phase-Locked Loop (PLL) has also been used for extracting time-varying harmonic components. However, this method only works well in the case where a few time varying harmonics are present at the input signal. Otherwise the energies of adjacent harmonics spread over each other, mixing the spectrum. Besides the convergence time is considered to be long.

This work proposes the use of independent component analysis (ICA) [12] for extracting harmonics from the power system signal. ICA is a technique that, based on a model of mixtures (here formed by harmonic components), estimates the individual harmonic components that generate the power signal without any prior knowledge of the power system parameters. The proposed method results in a simple structure to be used and does not require coherent sampling (synchronous sampling) and can be used to extract time-varying harmonics. The method was tested using both simulated and real signals, which take into account power electronics and power quality applications. Both voltage and current signals were used to test the ICA-based method.

Regarding ICA applications for harmonic estimations, in [13], the use of the popular FastICA algorithm is proposed to determine harmonic orders by using virtual observed channels. The least square method is used to determine amplitudes and phases of harmonics. In [14], a multichannel signal constructed through a cycle-spinning method is presented to the ICA algorithm. The ICA outputs are clustered according to their frequency by using k-means, and mapped back to the observation domain. Unlike [13] and [14], the method proposed here applies the ICA algorithm only in the design stage (off-line) and in the operational stage ICA-based FIR filters are used to extract the harmonics. Also, it does not require using virtual signals, least square method and the clustering task.

2. Problem statement

As the electrical signal $x(t)$ is formed by a mixture of the fundamental component and its harmonics, it can be interpreted as a linear mixture of components. As harmonics are orthogonal and therefore independent of each other, the decomposition of the electric signal into fundamental and harmonic components can be seen as an ICA problem [12].

ICA is one of the most popular BSS (Blind Source Separation) methods and differs from others because the original components to be separated are considered statistically independent and follow a non-Gaussian distribution.

In the present work, a single mixture is available, which is the electrical signal monitored through a given point of the power system. The problem of ICA with a single mixture (a single measurement channel) is known as single channel ICA (SCICA) [15]. The work [15] addresses this problem in detail, with applications in electroencephalogram and electrocardiogram. In the present work, the single channel ICA is applied to the problem of harmonics in electrical signals in a simplified form.

3. Single channel ICA

To apply the ICA technique in a single channel, it is necessary to form a multi-channel data representation, which can be done by generating vectors with time delays from the observed discrete signal $x[n]$, obtained from continuous signal $x(t)$. Thus, one gets an observation matrix, which comprises discrete time-delayed versions of the observed signal:

$$\mathbf{x}[n] = [x[n],\ x[n-1],\ \ldots,\ x[n-M+1]]^T, \tag{1}$$

where the superscript T denotes transposition and $M = D + 1$ is the dimension of $\mathbf{x}$ for D time-delayed versions of $x[n]$. It is important to mention that in the context of this work, the observed signal $x[n]$ is the monitored power signal.

Thus, the standard formulation of ICA (which is applied to problems where the number of mixtures is equal to the number of sources) can be used:

$$\mathbf{x}[n] = \mathbf{A}\mathbf{s}[n], \tag{2}$$

where $\mathbf{s}[n] = [s_1[n], s_2[n], \ldots, s_M[n]]^T$ is the matrix of statistically independent components at the sample n, and $\mathbf{A}$ is a $M \times M$ matrix, which performs the linear combination of the sources to form the observed signal, and it is called the mixture matrix.

The ICA algorithms estimate the original signal sources blindly, that is, using only the observed signals. The estimated sources are obtained by

$$\mathbf{y}[n] = \mathbf{W}\mathbf{x}[n], \tag{3}$$

where $\mathbf{y}[n] = [y_1[n], y_2[n], \ldots, y_M[n]]^T$ represents the matrix of estimated independent components and $\mathbf{W}$ is the separation matrix (or demixing matrix), which is an estimate of $\mathbf{A}^{-1}$. Thus, $y_i[n]$, for $i = 1, \ldots, M$, are the estimates of the independent original sources (components) $s_i[n]$.

There are different approaches to obtain the separation matrix $\mathbf{W}$, from mixtures using certain statistical properties of the source signals such as non-Gaussianity, temporal structure and cross-cumulants [12]. From these properties, several algorithms have been proposed in the literature. In this work, the Second Order Blind Identification (SOBI) algorithm [16] is used, once it exploits temporal information of the mixtures.

However, in order to ensure that the sources can be separated successfully following the SCICA approach, it is necessary that they have disjoint spectra [15]. Thus, this process can be represented by the sum of convolutive series, wherein M signals $a_i[n]$ are convolved with the impulse responses of the M filters $s_i[n]$. In this case, it is assumed that the impulse responses are statistically independent and represent the original sources (denoted by the matrix $\mathbf{s}[n]$ in Eq. (2)) of mixture $x[n]$ and a_i are the columns of the mixture matrix $\mathbf{A}$.

The vectors a_i tend to be individual delayed filters, such that all of them associated with a particular subset will have similar spectral content. Therefore, the estimation of each independent component can be given by one of the filtered versions of the mixture $x[n]$, where the coefficients of each filter are given by the rows of the demixing matrix $\mathbf{W}$, or by grouping together signals with the same spectral content. In [15], signals with the same spectral information are clustered using the k-means algorithm.

Unlike [15], we propose to use the rows of $\mathbf{W}$ as filters to extract the harmonics, avoiding thus the clustering task, which leads to a simpler approach.

4. Proposed method

4.1. Design stage

The design of the proposed method corresponds to obtaining the best separation row $\mathbf{w}$ (among the rows of the demixing matrix $\mathbf{W}$) for each harmonic component. The capacity of the separation row to extract a given harmonic component with good quality is directly related to the number of delayed versions of the monitored signal to be used. Furthermore, the use of few delayed versions