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A spectral estimation method for nonstationary signals analysis with application to power systems

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ABSTRACT

In this paper we analyze a new method for estimating the spectral content of frequency and amplitude modulated waveforms in the context of power line signals where, as well known, the frequency of the fundamental component (50 Hz or 60 Hz) slightly changes over time. The method, proposed here in two distinct implementations, is based on a different choice of the harmonic functions that are usually used in the DFT as a basis to analyze signals. Indeed their frequency is allowed to change over time, according to the output of an instantaneous frequency detector. The orthogonality of the basis is preserved by performing a change of reference in the time. The theory is developed in continuous time and subsequently implemented in discrete time. Simulations and experimental results are provided in order to verify and characterize preliminarily the proposed method.

1. Introduction

Voltages and currents of Power Systems (PS) are affected by non-idealities and distortions that may have detrimental effects on power distribution and utilization and that should be quantified by means of power quality analyses [1–11]. Nowadays the accurate measurement of PS parameters has become of great interest also in Smart Grids and in renewable energy production [12–20], where sudden changes in energy production and consumption threaten grid stability and security. Frequency, phase and amplitude of the 50 Hz (or 60 Hz) component of the mains are among the parameters which may change over time and that should be measured. Moreover, the quantification of harmonics is important because they produce energy losses and malfunctions. Due to the importance of the topic, several methods have already been proposed to measure the spectral content of power line signals with digital instruments. One reason for this abundance of methods is that the use of the simple DFT tool for spectral analysis poses many difficulties; among them, the most important are:

(1) *The signal is nonstationary*. In this case, even though the DFT is a biunivocal representation of a sampled signal, its analysis in terms of windowed sinusoids is not immediately informative about the evolution of the signal. This happens in PS where the waveforms can be modeled as frequency- and amplitude-modulated sinusoids, for which an estimation of the modulating functions would be more compact and informative. In general, there are several different ways of representing signals which are best adapted to different cases. An approach is to take into account the particular application for which the analysis is carried on and to use parametric models [21–22]; another one is to resort to unambiguos mathematical definitions, such as analytic signal and instantaneous frequency.

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(2) *The sampling should be coherent*, i.e. an integer number of cycles should be acquired for each record processed by the DFT. This is important, even in the simple case of a perfectly periodic signal, in order to avoid spectral leakage. There are two different strategies to achieve coherent sampling: changing the sampling frequency and resampling an already acquired signal [23]. The former can be realized using a digital Phase Locked Loop (PLL) to control the sampling clock [24]. Digital PLLs can be implemented in various ways [25–26]. They can also be designed according to the Kalman filter theory [27]. The latter strategy is based on sampling at constant rate and subsequent resampling according to the signal frequency estimated, for example, by the zero-crossing method [28]. In this way the resampled record contains an integer number of signal cycles. Among the techniques for frequency estimation it should mentioned also the interpolated DFT [29–31] and leakage correction algorithms such as the one proposed in [32].

Other approaches are known in literature to analyze nonstationary signals, such as the Welch method [33] and many other windowing techniques [34]. In [35] the interpolated DFT is used to estimate harmonics and interharmonics, while in [36] the Interpolated Dynamic DFT is proposed for synchrophasor and frequency measurement. In [37] the spectral components are measured by using quadrature demodulators. Compressive sensing is used in [38] to enhance frequency resolution. A parametric model of the time evolution of the phasor, based on its Taylor series expansion, has been exploited in [39]; the same concept is at the basis of the least squares estimation method for synchrophasor measurement described in [40]. Dynamic phasor measurement is also obtained with rotational invariance techniques and propagator method in [41] and with a least squares iterative method in [42]. In [43] a Prony adaptive filter is proposed to provide instantaneous estimates of the first derivative of amplitude and phase, useful to stability system assessment. Algorithms for the measurement of phase-angle jumps associated to voltage dips are proposed and compared in [44]. The performance of several synchrophasor measurement methods and frequency estimation algorithms are compared in [45] and [46], respectively. A parametric method based on min-norm and nonparametric methods based on Wigner and Choi-Williams distributions are proposed and compared in [47]. The idea of adapting the basis to the signal has been researched many times in the literature [48–49].

In this paper, we address the above mentioned issues, namely nonstationary signal and noncoherent sampling, by redefining the basis functions used in the DFT to analyze the signal, as preliminarily described in [50]. The basic idea is to estimate the instantaneous phase of the fundamental component and resample the harmonic functions used to decompose the signal according to it. There are two main assumptions involved for the applications of the method: (i) the fundamental frequency-modulated component exists and is stronger than the other components; (ii) its amplitude is modulated by a lower-frequency signal. The method presented here is characterized by its simplicity, the possibility of real-time implementation, and the easy and straightforward interpretation of the results in terms of a modified DFT.

This paper is structured as follows: in Section 2 the proposed method is presented in continuous time, in Section 3 the practical application to sampled signals is described; in Sections 4 and 5 simulations and experimental results are illustrated, respectively. Several aspects are finally discussed in Section 6.

2. Proposed method

In this section, the idea at the basis of the proposed method is explained; let $x(t)$ be a real multisine signal modulated in frequency and amplitude,

$$x(t) := \sum_{k=-K}^{K} A_k(t) e^{jk\omega_0 g(t)} \qquad (1)$$

where $2K+1$ is the number of complex components in the signal, and $A_k(t) = A^*_{-k}(t)$. We choose ω_0 with the meaning of nominal frequency of the fundamental component, with $g(t)$ approximately equal to t. It should be noted that if we rescale $g(t)$ and ω_0 in (1) while keeping constant their product, the value of $x(t)$ does not change. For simplicity $g(\cdot)$ is assumed to be strictly increasing and continuous, hence invertible. We assume also that $A_k(t)$ are slow time-varying functions. As will be better explained in Section 3.1, the band of $A_1(t)e^{j\omega_0 g(t)}$ should be sufficiently narrow and separate from the band of the other components to perform accurate instantaneous phase detection.

The frequency components in (1) are harmonically related and exhibit the same frequency modulation. Harmonics can be generated for example, by nonlinear loads connected to the grid and by the non-ideal behavior of power inverters. Slow frequency and phase fluctuations are often associated with variations in the rotation speed of electrical machines, and can be purposely introduced in electrical grids as a mean for regulating power flows. Some components are not included in the model (1): (a) high-frequency components not harmonically related to the fundamental, e.g. those due to the use of power-line communication in medium voltage [51] and (b) interharmonics with a modulation different from $g(t)$, which require an increase in the spectral resolution in order to be detected, as further discussed in Section 6.

In this paper the problem of estimating the parameters of the model (1) from the measurement of $x(t)$ is analyzed.

If $A_k(t)$ and $\omega(t)$ were constant, the estimation of A_k would have been obtained by measuring ω_0 and performing a simple Fourier series analysis of a cycle $2\pi/\omega_0$ of $x(t)$. However two obvious and well known issues arise when the Fourier series is applied to the model (1):

- the first one is that $A_k(t)$ are slow functions of time and then the use of the short-time Fourier transform is more appropriate;
- the second one is that, usually, $\omega(t)$ is not constant or, in other terms, $g(t) \neq t$, hence a proper technique to take intervals containing an integer number of signal cycles