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IFESIS: Instantaneous frequencies estimation via subspace invariance properties of wavelet structures

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ABSTRACT

According to the proposed method, a set of wavelet transforms of the signal is first obtained, using a structure of Complex Shifted Morlet Wavelets. No specific constraints are imposed on the center frequencies and the bandwidths of the individual wavelets, as well as on the number of wavelets used. In this way, a set of complex signals result in the time domain, equal to the number of the wavelets used. Then, the instantaneous frequencies of the signals are estimated by applying an appropriate subspace algorithm (as for e.g. ESPRIT), to the entire set of the resulting complex wavelet transforms, exploiting the corresponding subspace rotational invariance property of this set of complex signals. Since the method proposes the application of the subspace algorithm after the signal has been appropriately transformed by an appropriate wavelet structure, contrary to the classical subspace methods which are applied to the signal itself, the desired time–frequency features of the signal are enhanced, while simultaneously, the undesired frequency components, as well as the noise, are suppressed. In this way, the method combines the advantages of the Complex Shifted Morlet Wavelets with the advantages of subspace based approaches. Moreover, the method provides a means for estimating the number of the resulting harmonic components, using as a relevant indicator the number of the non-zero singular values of the corresponding singular value decomposition problem. It should be noted that the resulting singular values are also time dependent, providing valuable information on the possibly time variable dynamic structure of the signal.

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1. Introduction

The estimation of the instantaneous frequency content of non-stationary multicomponent harmonic signals is of paramount importance in all signal processing applications of mechanical systems, including condition and health monitoring, operational modal analysis or even real time control. Equally important is such a need also in the broad range of signal

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processing applications, including for e.g. communications, speech and image processing, biomedical and environmental signal analysis, etc.

For this reason, this task has been adressed long ago. Boashash [1,2] contributed an interpretation of the IF concept as well as a comprehensive review of the literature about IF estimation methods. In general, the IF estimation methods can be classified into five main categories: (a) phase difference-based IF estimators; (b) zero-crossing IF estimators; (c) linear-prediction-filter-based adaptive IF estimators; (d) IF estimators based on the moments of time–frequency distributions (TFDs); and (e) IF estimators based on the peak of TFDs. An indicative brief review of relevant approaches can be found among others in [3,4]. Of special importance and effectiveness are methods based on single component variants of subspace based approaches, due to their enhanced resolution and robustness to noise. Moreover, among the TFD methods, far more interesting are IF estimations based on the Wavelet Transform [4], due to the established overall advantages of the Wavelet Transform over the other TFD methods.

Parallel, during the same time period, significant achievements have been accomplished in the field of parameter estimation of multicomponent harmonic signals. Among others, some indicative well-known milestone contributions in the field include all variants of subspace based approaches such as MUSIC [5], ESPRIT [6] and minimum norm [7], matching pursuit algorithms based on wavelet dictionaries [8], multiband variants of the well-known Energy Operator Separation Algorithm (EOSA) [9], Periodic algebraic separation and energy demodulation (PASED) [10], etc. Other general purpose approaches can be also used for the same purpose, such as Autoregressive Moving Average (ARMA) approaches [11,12], Empirical Mode Decomposition [13] or appropriate extensions of the Hilbert Transform [14,15].

Although a vast number and variety of relevant methods have been proposed, some key problems of such methods still exist. As it has been demonstrated for e.g. in [3], oversampling in relation to noise may lead to significant errors, even in the case of well established procedures. Therefore, the sampling frequency must be selected appropriately in relation to the noise level and the frequency content of the signal, which is a cumbersome task in the case of broadband signals. Moreover, if a signal containing multiple harmonics is interpreted as a single component signal, characteristic spikes appear in the time waveform of the instantaneous frequency, which resemble “outliers” or “impulsive noise” and consequently, they are often misinterpreted as such. The real nature of this phenomenon can be only understood by a careful analysis of the analytical form of the instantaneous frequencies of signals including multiple harmonic components, as for e.g. those presented in Appendix A of [16], or more analytically in [17]. Thus, although a number of post processing procedures have been proposed [2] for smoothing the waveform of the instantaneous frequency, the effects of this phenomenon can be only mitigated but not alleviated.

Consequently, the problem of multiple instantaneous frequency estimation is closely related to the problem of the proper estimation and interpretation of the number of harmonic contained in a signal, a task which by itself is far from being trivial. Not only a variety of methods exist for estimating the number of harmonic components in a signal, but also the answer may differ according to the decomposition approach selected [18,19]. Moreover, as it will be shown in this paper, the answer to this question may be time dependent, as it should be anticipated in time variable systems.

In this paper, a method for multicomponent instantaneous frequency estimation is proposed, combining the advantages of the Complex Shifted Morlet Wavelets with the advantages of subspace based approaches. As summarized elsewhere [4], Complex Shifted Morlet Wavelets (CSMW) combine a number of advantages. First, since Morlet Wavelets are modulated in the time domain by a Gaussian shaped time window (and for this reason are also known as Gabor wavelets), they present the optimal resolution simultaneously in the time and in the frequency domain. Second, compared to the Discrete Wavelet Transform (DWT), they allow for the continuous (and thus more accurate) approximation in both time and frequency domains. Additionally, they are not related to spectral leakage effects. Third, contrary to the traditional real number representation of the Morlet Wavelet Transform, CSMW results to complex wavelet coefficients in the time domain, thus enabling the direct simultaneous calculation of the signal instantaneous amplitude and frequency. Fourth, contrary to the classical concept of “scaling” the wavelet in the time or frequency domain, which allows the identification of only just one parameter – the wavelet scale – the concept of shifting the Morlet wavelet in the frequency domain allows for the simultaneous optimal selection of both the wavelet parameters necessary to identify it as a proper filter in the frequency domain: The center frequency and the bandwidth.

Finally, as it is shown in [4], the recovery of the signal frequency can be performed accurately, without the requirement that the wavelet center frequency coincides to the signal frequency, contrary to the accurate recovery of the signal amplitude, which requires additionally this last condition. This constitutes a definite advantage of CSMW over other TFD based methods, which require efficient adaptation strategies. Parallel, parametric and especially signal eigenspace or subspace base methods, offer increased frequency resolution and robustness to noise [3], especially in the case of multiple harmonic components.

According to the proposed method, first a set of wavelet transforms of the signal are obtained, using a structure of Complex Shifted Morlet Wavelets. In principle, there are no constraints on the center frequencies and the bandwidths of the individual wavelets, as well as on the number of wavelets used, apart from the fact that the number of wavelets should be greater or equal to the number of harmonic components anticipated. In this way, a number of complex signals result in the time domain, equal to the number of Morlet wavelets selected, each one of which contains only that part of the signal frequency range allowed by the corresponding wavelet.

However, since multiple frequencies may coexist in the frequency band of a single wavelet, or equivalently, a time dependent instantaneous frequency may coexist in different wavelet bands during the time, the task of effectively

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