Estimation of Multi-Sinusoidal Signals: A Deadbeat Methodology

Boli Chen, Peng Li, Gilberto Pin and Thomas Parisini

Abstract— The problem of estimating the n unknown amplitudes, frequencies and phases of the components of a multisinusoidal signal is addressed in this paper. The proposed methodology theoretically allows the exact identification of the above unknown parameters within an arbitrarily small finite time in the noise-free scenario. The measured signal is processed by a bank of Volterra integral operators with a suitably designed kernel, that yields a set of auxiliary signals which are computable on-line by causal linear filters. These auxiliary signals are in turn used to estimate the frequencies in an adaptive fashion, while the amplitudes and the phases estimates can be calculated by means of algebraic formulas. The effectiveness of the estimation technique is evaluated and compared with other existing finite-time estimators via numerical simulations.

I. INTRODUCTION

The parametric estimation of a signal composed by a given number of sinusoids is one of the fundamental issues arising in several areas of engineering, such as, for instance, vibration diagnostics and prognosis, power quality monitoring and periodic disturbance rejection. Several methods are available in literature for the adaptive estimation of the amplitude, frequency and phase (AFP) of a single sinusoid (see, for example, [1], [2], [3], [4], [5], [6], and the references cited therein), while the AFP problem for a multisinusoidal signal has recently received renewed attention. Besides the well-known Fast Fourier Transform (FFT), which is far the most common tool used for harmonic extraction, several algorithmic alternatives have been conceived, being the Phase-Locked-Loop (PLL) and Adaptive Notch Filtering (ANF) the most successful methods for their ease of implementation. Although both PLL and ANF in their original formulation only apply to single-sinusoidal signals, multiple PLLs or ANFs can be combined to address the estimation problem in the multi-sinusoidal scenario. In [7], n enhanced-PLL (EPLL) units (see [8]) are deployed to extract the n harmonics and inter-harmonics of a multi-sinusoidal signal. Analogously, in [9] a bank of n ANF modules is used for the same task, with the advantage of being less computationally intensive than [7]. The problem becomes more challenging in case of an input with two frequencies that are close to each other. It has been shown in [10] that any two nearby frequencies can be discriminated by a couple of PLLs equipped with a "de-correlation" module. An alternative solution is given in [11], where the estimates from two identifiers are separated by enforcing a minimum frequency interval. However, such methods with de-correlation are hardly applicable for a number of sinusoids larger than two.

Another family of methodologies to track multiple frequencies relies on adaptive observers. These techniques are interesting since global or semi-global stability is ensured in most cases (see [12], [13], [14], [15] [16], [17], [18] and [19]). In particular, [16] and [17] deal with direct adaptation mechanisms for the squares of the frequencies with semiglobal stability guarantees.

Despite the large number of AFP techniques, relatively few deadbeat AFP estimation techniques are available in the literature. This type of estimators are needed in scenarios where the estimates are required to converge in a neighborhood of the true values within a predetermined finite time, *independently from the unknown initial conditions*. A deadbeat AFP estimation method is firstly addressed in [20] based on the concept of algebraic derivatives. However, reinitialization may be needed due to the presence of singularities. This issue has been tackled in [21] and [22] by recursive least squares algorithms. In [23], the algebraic identification approach is further extended to address the parameter estimation of two sinusoidal signals. Moreover, a modulating function-based approach is presented in [24], which allows non-asymptotic frequency detection by processing the input with truncated periodic functions. A new tool for finitetime estimation has been recently proposed in [25], [26], where Volterra operators with a suitably designed kernel function allow to annihilate in finite-time the effect of the unknown initial conditions on the estimate. Compared with the algebraic identification method, the kernel-based one features internal stability, thus not requiring periodic reinitialization. Resorting to the said kernel-based design, a novel finite-time frequency identifier is presented in [27], where Volterra operators are paired with a robust slidingmode adaptation law.

The present paper presents a deadbeat AFP estimator that employs Volterra operators with novel kernel functions. Compared to the previous kernels proposed by the authors in [26] and [27], yielding to Linear Time Varying (LTV) filters, the new ones admit a linear time invariant (LTI) realization. Moreover, the new kernel functions do not annihilate the initial conditions, that instead take part to the estimation as extended parameters, allowing for the retrieval of both the amplitude and the phase of the sinusoidal components.

II. PROBLEM STATEMENT AND PRELIMINARIES

Consider the following multi-sinusoidal signal

$$
y(t) = \sum_{i=1}^{n} A_i \sin(\vartheta_i(t)), \ \dot{\vartheta}_i = \omega_i, \ \vartheta_i(0) = \phi_i, \quad (1)
$$

where $A_i \in \mathbb{R}_{>0}$ and $\omega_i \in \mathbb{R}_{>0}$ denote respectively the unknown amplitudes and the angular frequencies, verifying the inequality $\omega_i > 0$, $\omega_i \neq \omega_j$ for $i \neq j$, while ϕ_i denotes the initial phase of each sinusoid. As mentioned in the Introduction, our objective consists in estimating A_i , ω_i and ϕ_i within an arbitrarily small finite time.

The signal (1) can be thought of as being generated by the following observable autonomous marginally-stable

B. Chen is with Imperial College London (UK) (boli.chen10@imperial.ac.uk); P. Li is with the Imperial College London (UK) (peng.li13@imperial.ac.uk); G. Pin is with Electrolux Professional S.p.A., Italy (gilberto.pin@electrolux.it); T. Parisini is with the Dept. of Electrical and Electronic Engineering at the Imperial College London, UK, and also with the Dept. of Engineering and Architecture at University of Trieste, Italy. (t.parisini@gmail.com).

dynamical system:

$$
\begin{cases} \dot{\mathbf{w}}(t) = \mathbf{A}_w \mathbf{w}(t) \\ y(t) = \mathbf{c}_w^{\top} \mathbf{w}(t) \end{cases} , \qquad (2)
$$

where $\mathbf{w}(t) \triangleq [w_0(t) \dots w_r(t) \dots w_{2n-1}(t)]^{\top} \in \mathbb{R}^{2n}$,

$$
\mathbf{A}_{w} \triangleq \begin{bmatrix} \mathbf{J}_{1} & 0 & \cdots & 0 \\ 0 & \mathbf{J}_{2} & \ddots & 0 \\ \vdots & \ddots & \ddots & \vdots \\ 0 & \ddots & \ddots & \mathbf{J}_{n} \end{bmatrix}, \quad \mathbf{c}_{w} \triangleq \begin{bmatrix} c_{1} \\ c_{2} \\ \vdots \\ c_{n} \end{bmatrix},
$$

$$
\mathbf{J}_{i} \triangleq \begin{bmatrix} 0 & 1 \\ -\omega_{i}^{2} & 0 \end{bmatrix}, \quad c_{i}^{\top} \triangleq \begin{bmatrix} 1 & 0 \end{bmatrix},
$$

and with initial conditions

$$
w_{2i-2}(0) = A_i \sin \phi_i, \ \ w_{2i-1}(0) = A_i \omega_i \cos \phi_i, \n\forall i \in \{1, ..., n\}.
$$
 (3)

The associated characteristic polynomial, having purely imaginary roots occurring in complex-conjugate pairs, is given by

$$
P(s) = \prod_{i=1}^{n} (s^2 + {\omega_i}^2)
$$

= $s^{2n} + {\alpha_{n-1}} s^{2n-2} + \dots + {\alpha_1} s^2 + {\alpha_0}$ (4)

where s is Laplace variable, $(\alpha_0, \alpha_1, \dots, \alpha_{n-1})$ are the coefficients of the characteristic polynomial, simply determined by the unknown frequencies ω_i , $i = 1, 2, \dots, n$.

Being (2) observable, the state vector $w(t)$ admits a linear transformation of coordinates $z(t) = Tw(t)$ with T is defined later in (7), such that the signal generator of $y(t)$ can be rewritten in an observer canonical form. Consider

$$
\mathbf{z}(t) = \begin{bmatrix} z_0(t) & z_1(t) & \dots & z_r(t) & \dots & z_{2n-1}(t) \end{bmatrix}^\top \in \mathbb{R}^{2n}
$$

the canonical system evolving from the unknown initial state $\mathbf{z}(0) = \mathbf{T} \mathbf{w}(0)$, is given as follows:

$$
\begin{cases} \dot{\mathbf{z}}(t) = \mathbf{A}_z \, \mathbf{z}(t), \\ y(t) = \mathbf{c}_z^\top \mathbf{z}(t), \quad t \in \mathbb{R}_{\geq 0} \end{cases} \tag{5}
$$

where $\mathbf{A}_z = \mathbf{T} \mathbf{A}_w \mathbf{T}^{-1}, \mathbf{c}_z^{\top} = \mathbf{c}_w^{\top} \mathbf{T}^{-1}$ are given by

$$
\mathbf{A}_{z} = \begin{bmatrix} a_{2n-1} & 1 & 0 & \cdots & 0 \\ a_{2n-2} & 0 & 1 & \ddots & 0 \\ \vdots & \vdots & \ddots & \ddots & \vdots \\ a_{1} & 0 & 0 & \cdots & 1 \\ a_{0} & 0 & 0 & \cdots & 0 \end{bmatrix}, \quad \mathbf{c}_{z} = \begin{bmatrix} 1 \\ 0 \\ \vdots \\ 0 \end{bmatrix}.
$$
 (6)

with $a_{2i+1} = 0$ and $a_{2i} = -\alpha_i$, $\forall i = \{0, 1, \dots, n-1\}$. The transformation matrix T is determined by:

$$
\mathbf{T} = \mathbf{M}\mathcal{O} \tag{7}
$$

,

where $\mathcal O$ is the observability matrix of (2) and M is a triangular matrix with respect to $a_1, a_2, \cdots, a_{n-1}$

In the following, a deadbeat algorithm is introduced to address the identification of the unknown system parameters α_i and the initial conditions $z(0)$. Thereby the frequency are computed as the zeros of the characteristic polynomial $P(s)$, while the amplitudes and phases are determined by inverting (3) with the current frequency estimates.

Letting $x(t) \in \mathbb{R}, \forall t \geq 0$ be an *i*-th order differentiable signal, in this paper we denote by $x^{(1)}$ the *i*-th order derivative signal. Moreover, given a kernel function $K(\cdot, \cdot)$ in two variables, its i -th order derivative with respect to the second argument will be denoted as $K^{(i)}(t, \tau)$, $i \in \mathbb{Z}_{\geq 0}$.

Consider a Volterra integral operator (see [26] for a detailed review on the subject) with respect to a kernel function $K(\cdot, \cdot)$

$$
\left[W_Kx\right](t) \triangleq \int_0^t K(t,\tau)x(\tau)d\tau, \ \ t \in \mathbb{R}_{\geq 0} \,. \tag{8}
$$

For the sake of practical implementability, it is worth to point out that the transformed signal $[V_K x](t)$, for $t \geq 0$, can be obtained as the output of a dynamic system described by the following scalar integro-differential equation:

$$
\begin{cases}\n\xi^{(1)}(t) & = K(t,t)x(t) + \int_0^t \left(\frac{\partial}{\partial t}K(t,\tau)\right)x(\tau)d\tau \\
[V_Kx](t) & = \xi(t)\n\end{cases}
$$
\n(9)

where $\xi(0) = \int_0^0 K(0, \tau) x(\tau) d\tau$ and $\xi^{(1)}(0) = 0$.

The following result is useful in dealing with the application of Volterra operators to the derivatives of a signal.

Lemma 2.1: [26] For a given $i \geq 0$, consider a signal $x(\cdot) \in \mathcal{L}^2(\mathbb{R}_{\geq 0})$ that admits a *i*-th weak derivative in $\mathbb{R}_{\geq 0}$ and a kernel function $K(\cdot, \cdot) \in \mathcal{HS}$, admitting the *i*-th derivative (in the conventional sense) with respect to the second argument. Then, it holds that:

$$
\[V_K x^{(i)}\] (t) = \sum_{j=0}^{i-1} (-1)^{i-j-1} x^{(j)}(t) K^{(i-j-1)}(t, t)
$$

+
$$
\sum_{j=0}^{i-1} (-1)^{i-j} x^{(j)}(0) K^{(i-j-1)}(t, 0) + (-1)^i \big[V_{K^{(i)}} x \big] (t)
$$
 (10)

that is, the function $[V_K x^{(i)}] (\cdot)$ is non-anticipative with respect to the lower-order derivatives $x(\cdot)$, $x^{(1)}(\cdot), \ldots, x^{(i-1)}(\cdot).$

The properties of the Volterra operator depend significantly on the shape of the kernel function. In this connection, we define a class of kernel functions that plays an important role in this framework.

Definition 2.1: If a kernel $K(\cdot, \cdot) \in \mathcal{HS}$ which is at least $(i - 1)$ -th order differentiable with respect to the second argument, verifies the condition

$$
K^{(j)}(t,t) = 0, \forall j \in \{0,1,\ldots,i-1\}
$$
 (11)

then, it is called an i -th order Bivariate (strict) Causal Kernel $(BC-K)$.

Here, we introduce a BC-K that fulfills (11):

$$
K(t,\tau) = e^{-\beta(t-\tau)} \left(1 - e^{-\beta(t-\tau)}\right)^N \tag{12}
$$

with the parameter $\beta \in \mathbb{R}_{>0}$. Indeed, the condition (11) up to the N-th order is met by the factor $(1 - e^{-\beta(t-\tau)})^N$.

III. FINITE-TIME AMPLITUDE, FREQUENCY AND PHASE ESTIMATION

For the sake of further discussion, it is worth to introduce the differential-constraint model of (5):

$$
\begin{cases}\ny^{(2n)}(t) = \sum_{i=0}^{2n-1} a_i y^{(i)}(t), \ \forall t \in \mathbb{R}_{\geq 0}, \\
y^{(2i)}(0) = y_0^{(2i)}, \ i \in \{0, \dots, n-1\}\n\end{cases}
$$
\n(13)

where $y_0^{(i)}$, $i \in \{0, \ldots, 2n-1\}$ represent the unknown initial conditions on hidden output derivatives. Notably, the state-variables of the observer canonical realization can be expressed as a linear combination of the output derivatives:

$$
z_r(t)=y^{(r)}(t) - \sum_{j=0}^{r-1} a_{2n-r+j} y^{(j)}(t), r \in \{0, 1, \cdots, 2n-1\}
$$
\n(14)

where we have used the convention $\sum_{j=0}^{k} {\{\cdot\}} = 0$, $\forall k < 0$.

Assuming that $K(\cdot, \cdot)$ is a 2n-th order Bivariate Causal kernel function satisfying the condition (11), thanks to Lemma 2.1, it is immediate to show that

$$
\[V_K y^{(i)}\] (t) = \sum_{j=0}^{i-1} (-1)^{i-j} y^{(j)}(0) K^{(i-j-1)}(t,0) + (-1)^i [V_{K^{(i)}} y](t) \quad (15)
$$

for all $i \in \{0, 1, \dots, 2n\}.$

Consider the case $i = 1$, from (15) we have that

$$
[V_{K^{(1)}} y](t) = -y(0)K(t,0) - [V_K y^{(1)}](t).
$$

Moreover, performing the substitution of y with $y^{(2n-1)}$ we have that also the following integral equation holds

$$
\left[V_{K^{(1)}}y^{(2n-1)}\right](t) = -y^{(2n-1)}(0)K(t,0) - \left[V_{K}y^{(2n)}\right](t).
$$

Therefore, owing to the I/O relationship (13), it holds that

$$
\left[V_{K^{(1)}}y^{(2n-1)}\right](t) = -y^{(2n-1)}(0)K(t,0) - \sum_{i=0}^{2n-1} a_i \left[V_Ky^{(i)}\right](t)
$$

which can be rearranged as

$$
(-1)^{2n-1} \left[V_{K^{(2n)}} y \right](t) = -y^{(2n-1)}(0) K(t,0)
$$

$$
- \sum_{j=0}^{2n-2} (-1)^{2n-1-j} y^{(j)}(0) K^{(2n-j-1)}(t,0)
$$

$$
- \sum_{i=0}^{2n-1} a_i \left(\sum_{j=0}^{i-1} (-1)^{i-j} y^{(j)}(0) K^{(i-j-1)}(t,0) + (-1)^i \left[V_{K^{(i)}} y \right](t) \right).
$$

After some cumbersome algebra, we get

$$
(-1)^{2n-1}[V_{K^{(2n)}}y](t) + \sum_{i=0}^{2n-1} a_i (-1)^i [V_{K^{(i)}}y](t)
$$

=
$$
-\sum_{r=0}^{2n-1} K^{(2n-r-1)} (-1)^{2n-r-1}
$$

$$
\times \left(y^{(r)}(0) - \sum_{j=0}^{r-1} a_{2n-r+j} y^{(j)}(0)\right)
$$
(16)

that, thanks to (14), can be written in a compact form

$$
\left[V_{K^{(2n)}}y\right](t) = \sum_{i=0}^{n-1} \alpha_i \left[V_{K^{(2i)}}y\right](t) + \sum_{r=0}^{2n-1} \gamma_r(t) z_r(0) \tag{17}
$$

where $\gamma_r(t) = K^{(2n-r-1)}(t,0)(-1)^{2n-r-1}$.

Noting that the right-hand side of (17) is linear with respect to the parameters α_i and the initial state $z_r(0)$, it can be recast in vector form

$$
\left[V_{K^{(2n)}}y\right](t) = \nu(t)^{\top} \boldsymbol{\theta} \tag{18}
$$

where $\theta \triangleq [\alpha_0, \alpha_1, \ldots, \alpha_{n-1}, z_0(0), z_1(0), \ldots, z_{2n-1}(0)]^\top$ is an extended parameter vector that contains, besides the model model parameters, also the initial conditions of output derivatives, while

$$
\nu(t) \triangleq \left[[V_K y](t), [V_{K^{(2)}} y](t), \ldots, [V_{K^{(2n-2)}} y](t), \ldots, \gamma_0(t), \gamma_1(t), \ldots, \gamma_{2n-1}(t) \right]^{\top}
$$

is a vector of known signals. For the sake of the further discussion, let us partition $v(t)$ as follows: $v(t) = [\mathbf{z}_e(t), \gamma(t)]$ where $z_e(t)$ contains signals obtainable by processing $y(t)$ by Volterra operators, while $\gamma(t)$ contains known (kerneldependent) functions of time. In the following, we show that $z_e(t)$ can be obtained by processing the measurable output through a stable linear filter.

Consider a BC-K in the form of (12) with $N = 2n + 1$:

$$
K(t,\tau) = e^{-\beta(t-\tau)} \left(1 - e^{-\beta(t-\tau)} \right)^{2n+1}
$$
 (19)

with the the design parameter $\beta \in \mathbb{R}_{>0}$.

For any $i \in \{0, 1, 2, \ldots, 2n\}$, the *i*-th derivative of the designed kernel with respect to the second argument can be expressed as:

$$
K^{(i)}(t,\tau) = \sum_{j=1}^{2n+2} e^{-j\beta t} f_{i,j}(\tau).
$$
 (20)

Let $K_{i,j}(t, \tau) \triangleq e^{-j\beta t} f_{i,j}(\tau)$, then we have

$$
\frac{\partial}{\partial t}K_{i,j}(t,\tau) = -j\beta e^{-j\beta t}f_{i,j}(\tau).
$$

Moreover, by the linearity of the Volterra operator, it follows that $[V_{K^{(i)}} y](t) = \sum_{j=1}^{2n+2} [V_{K_{i,j}} y](t)$. Defining the internal state vector

$$
\boldsymbol{\xi}(t) = [\xi_{0,1}(t), \xi_{0,2}(t), \dots, \xi_{0,2n+2}, \xi_{2,1}(t), \dots, \xi_{2n,2n+2}]^{\top},
$$

with
$$
\xi_{i,j}(t) \triangleq [V_{K_{i,j}}y](t).
$$
 Then the augmented signal vector

$$
\mathbf{z}_a(t) \triangleq \left[\mathbf{z}_e(t) \left[V_{K^{(2n)}}y\right](t)\right]^{\top}
$$
 can be computed by the
following stable LTI system:

$$
\begin{cases} \xi^{(1)}(t) = \mathbf{G}_{\xi}\xi(t) + \mathbf{E}y(t) \\ \mathbf{z}_a(t) = \mathbf{H}\xi(t) \end{cases}
$$
 (21)

with $\xi(0) = 0 \in \mathbb{R}^{(n+1)\times(2n+2)}$ and where \mathbf{G}_{ξ} is a diagonal, time invariant and Hurwitz matrix, defined by G_{ξ} = blockdiag[G, \ldots, G], with G $diag(-\beta, -2\beta, \ldots, -(2n+2)\beta)$, and **H** is defined by $H =$ blockdiag $[1^{\top}, \dots, 1^{\top}]$, with 1^{\top} denotes a row vector of ones with $2n + 2$ elements. Finally, the vector $\mathbf{E} = [\mathbf{E}_0, \mathbf{E}_2, ..., \mathbf{E}_{2n}]^\top$ can be obtained as described in the following lines. Since the functions K_i , $j(t, \tau)$, evaluated for $\tau = t$,

$$
K_{i,j}(t,t) = \lambda_{i,j} \triangleq (-1)^{j-1} \begin{pmatrix} 2n+1 \\ j-1 \end{pmatrix} (j\beta)^i
$$

are constant, then \mathbf{E}_i is given by \mathbf{E}_i = $[\lambda_{i,1}, \lambda_{i,2}, \ldots, \lambda_{i,2n+2}]^\top$. In order to form a well-posed algebraic system based on (18) conventional augmentation tools used in system's identification can be employed. The covariance filtering technique is adopted here to construct a linear algebraic system. Let us multiple $v(t)$ on both sides of (18), leading to:

$$
\mathbf{S}(t) = \mathbf{R}(t)\boldsymbol{\theta} \tag{22}
$$

where $\mathbf{S}(t) \triangleq \boldsymbol{\nu}(t)[V_{K^{(2n)}}](t) \in \mathbb{R}^{3n \times 1}$ and $\mathbf{R}(t) \triangleq$ $\nu(t)\nu^{\top}(t) \in \mathbb{R}^{3n\times 3n^{\mathsf{T}}}.$

Note that rank $(\mathbf{R}(t)) = 1, \forall t > 0$, hence we apply to both sides of the (22) a low-pass filtering operation, obtaining

$$
\begin{cases}\n\dot{\mathbf{S}}_f(t) = -g\mathbf{S}_f(t) + \mathbf{S}(t) \\
\dot{\mathbf{R}}_f(t) = -g\mathbf{R}_f(t) + \mathbf{R}(t)\n\end{cases}
$$
\n(23)

where $S_f(0) = 0 \in \mathbb{R}^{3n \times 1}$, $\mathbf{R}_f(0) = 0 \in \mathbb{R}^{3n \times 3n}$. Now, let

$$
\mathcal{F}_{r,j} = (-1)^{j-1} \begin{pmatrix} 2n+1 \\ j-1 \end{pmatrix} (-j\beta)^{2n-r-1}, \quad (24)
$$

it is worth noting that $\gamma_r(t)$, $r = 0, 1, \ldots, 2n - 1$ contained in the regressor $v(t)$ can be represented as the sum of exponential functions

$$
\gamma_r(t) = \sum_{j=1}^{2n+2} e^{-j\beta t} \mathcal{F}_{r,j}
$$

which decay to zero as $t \to \infty$. The following technical result characterizes a specialized persistency of excitation condition (PE) on signal $v(t)$ that is needed to prove the convergence of the proposed algorithm.

Lemma 3.1: (Finite-time persistency of excitation) Given the multi-sinusoidal measurement $y(t)$ (see (1)) and the designed kernel (19), there exist some $\epsilon \in \mathbb{R}_{>0}$, $t_{\epsilon} \in \mathbb{R}_{>0}$ and $T \in \mathbb{R}_{>0}$ such that

$$
\int_{t-t_{\epsilon}}^{t} \nu(\tau) \nu^{\top}(\tau) d\tau \geq \epsilon \mathbf{I}, \ \forall t \in [t_{\epsilon}, t_{\epsilon} + T]. \tag{25}
$$

Proof: Let us split $\nu(t)$ into two vector signals $\nu_1(t) \in$ \mathbb{R}^n and $\nu_2(t) \in \mathbb{R}^{2n}$, such that

$$
\mathscr{L}{\lbrace \nu_1(t) \rbrace} = \mathbf{G}_1(s) \mathscr{L}{\lbrace y(t) \rbrace},
$$

and

$$
\boldsymbol{\nu}_2(t) = \mathbf{G}_2 \boldsymbol{\psi}_2(t)
$$

where

$$
\mathbf{G}_1(s) = \begin{bmatrix} \kappa_0(s) & \kappa_2(s) & \cdots & \kappa_{2n-2}(s) \end{bmatrix}^\top \in \mathbb{C}^n
$$

with
$$
\kappa_i(s) \triangleq \sum_{j=1}^{2n+2} \frac{\lambda_{i,j}}{s+j\beta}
$$
, $i = 0, 2, ..., 2n - 2$, and

$$
\psi_2(t) \triangleq \begin{bmatrix} e^{-\beta t} & e^{-2\beta t} & \cdots & e^{-(2n+2)\beta t} \end{bmatrix}^\top, \quad (26)
$$

$$
\mathbf{G}_2 = \left[\begin{array}{cccc} \mathcal{F}_{0,1} & \mathcal{F}_{0,2} & \cdots & \mathcal{F}_{0,2n+2} \\ \mathcal{F}_{1,1} & \mathcal{F}_{1,2} & \cdots & \mathcal{F}_{1,2n+2} \\ \vdots & \vdots & \ddots & \vdots \\ \mathcal{F}_{2n-1,1} & \mathcal{F}_{2n-1,2} & \cdots & \mathcal{F}_{2n-1,2n+2} \end{array} \right] \in \mathbb{R}^{2n \times (2n+2)}.
$$

Since $y(t)$ takes on the multi-sinusoidal form (1), it can be concluded that $y(t)$ is sufficient rich of order $2n$. Thanks to the linear independence of the complex vectors $\mathbf{G}_1(j\omega_1)$, \cdots , $\mathbf{G}_1(j\omega_n)$ on the complex space \mathbb{C}^n , $\nu_1(t)$ is PE for all $t \geq 0$ ([28, Chapter 2]).

Moreover, for the signal $\psi_2(t)$ defined in (26), there always exists a finite time interval $[t,\bar{t}]$ with $\bar{t} > t$ over which the elements of $\psi_2(t)$ are linearly independent functions [29]. It also implies that for any $t > t$, there exist a constant $\epsilon_2 \in \mathbb{R}_{>0}$, such that

$$
\int_{t-\underline{t}}^t \boldsymbol{\psi}_2(\tau) \boldsymbol{\psi}_2^\top(\tau) d\tau \geq \epsilon_2 \mathbf{I} \,.
$$

Then, in view of (24), G_2 is full row rank of $2n$. Hence, we have

$$
\int_{t-\underline{t}}^{t} \nu_2(\tau) \nu_2^{\top}(\tau) d\tau = \mathbf{G}_2 \int_{t-\underline{t}}^{t} \psi_2(\tau) \psi_2^{\top}(\tau) d\tau \mathbf{G}_2^{\top}
$$
\n
$$
\geq \underline{g}_2^2 \epsilon_2 \mathbf{I} \quad (27)
$$

where we denote by g_2 the minimum singular value of \mathbf{G}_2 . The inequality (27) implies $\nu_2(t)$ PE over an interval $[\underline{t}, \overline{t}]$.

By using the fact that the sinusoidal functions in $\nu_1(t)$ and the exponential functions in $\nu_2(t)$ are linearly independent, it can be concluded that also for $v(t)$ there exist some $\epsilon \in$ $\mathbb{R}_{>0}$, $t_{\epsilon} \in \mathbb{R}_{>0}$ and $T \in \mathbb{R}_{>0}$, such that the finite-time PE condition (25) holds, thus ending the proof.

Owing to (23) and (25), it is straightforward to show that

$$
\mathbf{R}_{f}(t) \geq \int_{t-t_{\epsilon}}^{t} e^{-g(t-\tau)} \nu(\tau) \nu^{\top}(\tau) d\tau
$$

$$
\geq e^{-gt_{\epsilon}} \epsilon \mathbf{I}, \ t \in [t_{\epsilon}, t_{\epsilon} + T]
$$

which in turn implies that, under the PE condition, the filtered auto-covariance matrix $\mathbf{R}_f(t)$ is invertible within a time interval $t_{\epsilon} \le t \le t_{\epsilon}+T$. In this connection, the unknown parameter vector θ can be estimated by

$$
\hat{\theta}(t) = \begin{cases} \n\theta_0, & t < t_{\epsilon}, \\ \n\mathbf{R}_f(t)^{-1}\mathbf{S}_f(t), & t_{\epsilon} \le t \le t_{\epsilon} + T, \\ \n\mathbf{R}_f(t_{\epsilon} + T)^{-1}\mathbf{S}_f(t_{\epsilon} + T), & t > t_{\epsilon} + T \n\end{cases}
$$

where θ_0 is a guessed parameter vector. It is worth noting that the algorithm is switched off after $t = t_{\epsilon}+T$ by freezing the estimates.

Given $\hat{\theta}$, the estimates of the α_i , $i \in \{0, ..., n-1\}$ and the initial states $z_r(0)$, $r \in \{1, \ldots, 2n-1\}$ are computable. Thanks to the the characteristic polynomial (4) that is parametrized by α_i , the frequencies $\omega_0, \ldots, \omega_{n-1}$ are computed by letting $P(s) = 0$. From (3), using the

initial states $w_i(0)$, obtained by $\mathbf{w}(0) = \mathbf{T}^{-1}\mathbf{z}(0)$ and ω_i , we finally get

$$
(\omega_i w_{2i-2}(0))^2 + w_{2i-1}(0)^2 = A_i^2 \omega_i^2
$$

which yields

$$
A_i = \sqrt{\left((\bar{\omega}_i w_{2i-2}(0))^2 + w_{2i-1}(0)^2 \right) / \bar{\omega}_i^2}
$$

with $\bar{\omega}_i \triangleq \max(\omega_{\min}, \omega_i)$, ω_{\min} is a *known* lower bound of the input frequencies. Finally,

$$
\phi_i = \tan^{-1} \left(\frac{\omega_i w_{2i-2}(0)}{w_{2i-1}(0)} \right)
$$

for all $i \in \{1, ..., n-1\}$.

IV. NUMERICAL EXAMPLE

In this section, a few numerical examples are carried out to examine the behavior of the proposed methodology, the performance of which is also compared with another algebraic algorithm proposed in [23].

A. Identification of two sinusoidal signals

Let us first consider the example used in [23], assuming $y_1(t) = \sum_{i=1}^2 A_i \sin(\omega_i t + \phi_i)$ where $A_1 = 2$, $A_2 = 5$, $\omega_1=1.4\pi\approx 4.4$ rad/s, $\omega_2=0.6\pi\approx 1.89$ rad/s, $\phi_1=1$ rad and $\phi_2 = 0.5$ rad.

The parameters of the algebraic algorithm [23] are set as $\epsilon = 1s$, $\zeta = 0.707$, $\omega_n = 31.4$ rad/s, while the proposed method is tuned by $\beta = 1$, $q = 1$. All the estimates are initialized by zero. According to Fig. 1, in the noise-free scenario, both methods are able to capture the sinusoidal parameters precisely in finite-time.

Instead of using a pure sinusoidal signal, let us assume the input $y_1(t)$ is corrupted by a bounded disturbance $d(t)$ with uniform distribution in the interval $[-0.1, 0.1]$. Keeping the tuning parameters unchanged, the behavior of the two methods in the presence of $d(t)$ are shown in Fig. 2. It is observed that the kernel-based method succeeds in AFP detection with fast convergence speed and slightly better noise immunity than [23]. It is worth noting that the algebraic estimator may be susceptible to numerical problems in the noisy scenario, due to its internal instability (as shown in the results at $t = 3.8$ s).

B. Identification of three sinusoidal signals

In the second example, we investigate the behavior of the proposed method in the presence of three sinusoidal $\sum_{i=1}^{3} A_i \sin (\omega_i t + \phi_i)$ where $A_1 = 2$, $A_2 = 5$, $A_3 = 4$, components with nearby frequencies. Consider $y_2(t)$ = $\omega_1 = 2$ rad/s, $\omega_2 = 2.2$ rad/s, $\omega_3 = 5$ rad/s, $\phi_1 = 1, \phi_2 = 1$ 0.7 rad and $\phi_3 = 0.2$ rad.

The estimation results in the noise-free scenario with tuning gains tuning $\beta = 1$ and $q = 1$ are reported in Fig. 3, showing the two nearby frequencies (ω_1 and ω_2) are precisely discriminated, while the other sinusoidal parameters are accurately identified as well. Moreover, Fig. 4 shows the estimated parameters when the measurement is perturbed by a bounded disturbance within [−0.5, 0.5]. Despite some degradation, the proposed methodology can provide satisfactory estimates within a short period of time.

V. CONCLUDING REMARKS

In this paper, the problem of AFP identification for a multi-sinusoidal signal has been addressed. A novel estimator is designed to provide reliable amplitude, frequency and phase estimates in finite-time. While relying upon strong theoretical foundations referring to the algebra of integral operators, the estimator ends up with a simple linear filter that, applied to the measured signal, produces an array of auxiliary signals that are exploited to retrieve all the parameters of the sinusoidal components in one shot.

REFERENCES

- [1] B. Wu and M. Bodson, "A magnitude/phase-locked loop approach to parameter estimation of periodic signals," *IEEE Trans. on Automatic Control*, vol. 48, no. 4, pp. 612–618, 2003.
- [2] G. Pin, "A direct approach for the frequency-adaptive feedforward cancellation of harmonic disturbances," *IEEE Trans. on Signal Processing*, vol. 58, no. 7, pp. 3513–3530, 2010.
- [3] B. Chen, G. Pin, W. M. Ng, C. K. Lee, S. Y. R. Hui, and T. Parisini, "An adaptive observer-based switched methodology for the identification of a perturbed sinusoidal signal: Theory and experiments," *IEEE Trans. on Signal Processing*, vol. 62, no. 24, pp. 6355–6365, 2014.
- [4] L. Hsu, R. Ortega, and G. Damm, "A globally convergent frequency estimator," *IEEE Trans. on Automatic Control*, vol. 44, no. 4, pp. 698– 713, 1999.
- [5] A. A. Bobtsov, D. Efimov, A. A. Pyrkin, and A. Zolghadri, "Switched algorithm for frequency estimation with noise rejection," *IEEE Trans. on Automatic Control*, vol. 57, no. 9, pp. 2400–2404, 2012.
- [6] G. Fedele and A. Ferrise, "Non adaptive second order generalized integrator for identification of a biased sinusoidal signal," *IEEE Trans. on Automatic Control*, vol. 57, no. 7, pp. 1838–1842, 2012.
- [7] M. Karimi-Ghartemani and M. R. Iravani, "Measurement of harmonics/inter-harmonics of time-varying frequencies," *IEEE Trans. on Power Delivery*, vol. 20, no. 1, pp. 23–31, 2005.
- [8] M. Karimi-Ghartemani, H. Mokhtari, and M. R. Iravani, "Wavelet based on-line disturbance detection for power quality applications," *IEEE Trans. on Power Delivery*, vol. 15, no. 4, pp. 1212–1220, 2000.
- [9] M. Mojiri, M. Karimi-Ghartemani, and A. Bakhshai, "Processing of harmonics and interharmonics using an adaptive notch filter," *IEEE Trans. on Power Delivery*, vol. 25, no. 2, pp. 534–542, 2010.
- [10] G. Pin and T. Parisini, "A direct adaptive method for discriminating sinusoidal components with nearby frequencies," in *Proc. of the IEEE American Control Conference*, O'Farrell Street, San Francisco, CA, USA, 2011, pp. 2994–2999.
- [11] X. Guo and M. Bodson, "Frequency estimation and tracking of multiple sinusoidal components," in *Proc. of the IEEE Conf. on Decision and Control*, Maui, Hawaii, USA, 2003.
- [12] R. Marino and P.Tomei, "Global estimation of n unknown frequencies," *IEEE Trans. on Automatic Control*, vol. 47, no. 8, pp. 1324– 1328, 2002.
- [13] X. Xia, "Global frequency estimation using adaptive identifiers," *IEEE Trans. on Automatic Control*, vol. 47, no. 7, pp. 1188–1193, 2002.
- [14] M. Hou, "Parameter identification of sinusoids," *IEEE Trans. on Automatic Control*, vol. 57, no. 2, pp. 467–472, 2012.
- [15] A. A. Bobtsov and A. A. Pyrkin, "Cancelation of unknown multiharmonic disturbance for nonlinear plant with input delay," *Int. J. Adapt. Control Signal Process*, vol. 26, no. 4, pp. 302–315, 2012.
- [16] B. Chen, G. Pin, and T. Parisini, "An adaptive observer-based esti-mator for multi-sinusoidal signals," in *Proc. IEEE American Control Conference*, Portland, OR, USA, 2014.
- [17] G. Pin, Y. Wang, B. Chen, and T. Parisini, "Semi-global direct estimation of multiple frequencies with an adaptive observer having minimal parameterization," in *Proc. of the Conference on Decision and Control*, Osaka, Japan, 2015, pp. 3693–3698.
- [18] S. Aranovskiy, A. Bobtsov, R. Ortega, and A. Pyrkin, "Improved transients in multiple frequencies estimation via dynamic regressor extension and mixing," in *12th IFAC International Workshop on Adaptation and Learning in Control and Signal Processing*, Eindhoven, Netherlands, 2016.
- [19] D. Carnevale and A. Astolfi, "A hybrid observer for frequency estimation of saturated multi-frequency signals," in *Proc. of the IEEE Conf. on Decision and Control and European Control Conference*, Orlando, FL, USA, 2011, pp. 2577–2582.
- [20] J. R. Trapero, H. Sira-Ramirez, and V. F. Battle, "An algebraic frequency estimator for a biased and noisy sinusoidal signal," *Signal Processing*, vol. 87, no. 6, pp. 1188–1201, 2007.
- [21] D. Liu, O. Gibaru, and W. Perruquetti, "Parameters estimation of a noisy sinusoidal signal with time-varying amplitude," in *Proc. 19th Mediterranean conference on Control and automation (MED'11)*, Corfu, 2011, pp. 570 – 575.

Fig. 1. Time behavior of the AFP estimates in noise-free scenario (estimates are overlapped for both methods).

Fig. 2. Time behavior of the AFP estimates in noisy scenario.

Fig. 3. Time behavior of the AFP estimates in noise-free scenario.

Fig. 4. Time behavior of the AFP estimates in noisy scenario.

- [22] A. Luviano-Juarez, J. Cortes-Romero, and H. Sira-Ramirez, "Parameter identification of a discretized biased noisy sinusoidal signal," *Measurement*, vol. 60, pp. 129–138, 2015.
- [23] J. R. Trapero, H. Sira-Ramirez, and V. F. Battle, "On the algebraic identification of the frequencies, amplitudes and phases of two sinusoidal signals from their noisy sum," *International Journal of Control*, vol. 81, no. 3, pp. 507–518, 2008.
- [24] G. Fedele and L.Coluccio, "A recursive scheme for frequency estimation using the modulating function method," *Applied Mathematics and Computation*, vol. 216, no. 5, pp. 1393–1400, 2010.
- [25] G. Pin, M. Lovera, A. Assalone, and T. Parisini, "Kernel-based nonasymptotic state estimation for linear continuous-time system," in *Proc. of the 2013 American Control Conference*, Washington, DC,

2013, pp. 3123–3128.

- [26] G. Pin, A. Assalone, M. Lovera, and T. Parisini, "Non-asymptotic kernel-based parametric estimation of continuous-time linear systems," *IEEE Trans. on Automatic Control*, vol. 61, no. 2, pp. 360–373, 2016.
- [27] G. Pin, B. Chen, and T. Parisini, "Deadbeat kernel-based frequency estimation of a biased sinusoidal signal," in *Proc. of the IEEE European Control Conference*, Linz, 2015.
- [28] S. Sastry and M. Bodson, *Adaptive Control: Stability, Convergence, and Robustness*. Prentice-Hall, 1994.
- [29] G. Sansone, *Orthogonal Functions: Revised English Edition*. Dover Publications, 1991.

1.5 ϕ_1 (Algebrai $\phi_2(\rm Algebraic\ meth$ ϕ_1 (Kernel method) ϕ_2 (Kernel method) 1 \Box φ**0.5** 0^{1}_{0} **0 1 2 3 4 5** Time [s]

