Filtering of Large Signal Sets: An Almost Blind Case

Anatoli Torokhti, Phil Howlett
CIAM, School of Inf. Techn. & Math. Sci.
University of South Australia,
Adelaide, SA 5095, Australia
Email: anatoli.torokhti@unisa.edu.au, phil.howlett@unisa.edu.au

Hamid Laga
PBRC, University of South Australia
ACPFG,
Adelaide, SA 5095, Australia
Email: hamid.laga@unisa.edu.au

***Abstract*—We propose a new technique which allows to estimate any random signal from a large set of noisy observed data on the basis of information on only a few reference signals. The conceptual device behind the proposed estimator is a linear interpolation of an optimal incremental estimation applied to random signal pairs interpreted an extension of the Least Squares Linear (LSL) estimator. We consider the case of observations corrupted by an arbitrary noise (not by an additive noise only) and design the estimator in terms of the Moore-Penrose pseudo-inverse matrix. Therefore, it is always well defined. The proposed estimator is justified by establishing an upper bound for the associated error and by showing that this upper bound is directly related to the expected error for an incremental application of the optimal LSL estimator. It is shown that such an estimator is possible under quite unrestrictive assumptions.**

Keywords–large signal sets; filtering; least squares linear estimator.

I. Introduction

A. Motivation

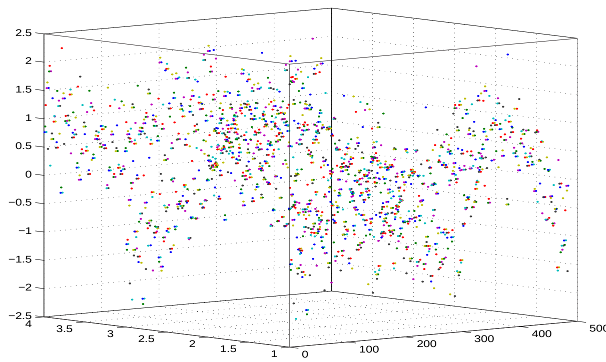
We write $\boldsymbol{x}_\omega = \boldsymbol{x}_\omega(t)$ for a stochastic vector $\boldsymbol{x}_\omega(t)$ associated with a random outcome ω and time $t \in T = [a, b] \subset \mathbb{R}$. A rigorous notation is given in the section that follows.

In many applications associated with a difficult environment, *a priori* information on a large set of signals of interest, $\mathcal{K}_x = \{\boldsymbol{x}_\omega(t)\}$, can only be obtained for a few signals $\{\boldsymbol{x}_\omega(t_j)\}_1^p \subset \mathcal{K}_x$ where p is a small number. Typical examples are devices and equipment exploited in the stratosphere, underground and underwater such as those in defence and the mining industry. Signals $\boldsymbol{x}_\omega(t_1), \ldots, \boldsymbol{x}_\omega(t_p)$, are associated with given times, $t_1, t_2, \ldots, t_p$, respectively, such that

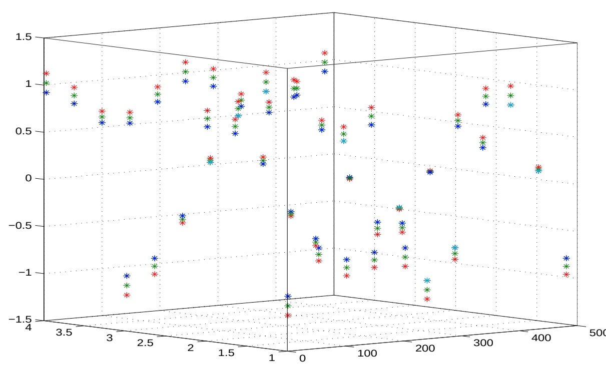
$$a = t_1 < t_2 < \cdots < t_{p-1} < t_p = b. \tag{1}$$

A choice of signals $\boldsymbol{x}_\omega(t_1), \ldots, \boldsymbol{x}_\omega(t_p)$ might be beyond our control (in geophysics and defence tasks, for instance). At the same time, it is required to estimate each reference signal in the set $\mathcal{K}_x$ from the corresponding set of noisy observations. Thus, all we can exploit to develop an associated filter is observed noisy data and a sparse information on reference signals.

Example 1: Suppose we need to process a set $\mathcal{K}_y$ of $N = 121$ random signals over set $T = [\tau_1, \tau_2, \ldots, \tau_N]$ so that each input signal from this set, $\boldsymbol{y}(t, \cdot)$, enters a filter at time $t = \tau_k$



(a) Observed signals from the set $\mathcal{K}_y$.



(b) Samples of reference signals $\tilde{X}^{(j)}$ at times t_j for $j = 1, \ldots, 11$.

Fig. 1. Signals and samples considered in Example 1.

where $\tau_1 = 0$ and $\tau_{k+1} = \tau_k + 0.05$, for $k = 1, \ldots, 120$. At time τ_k, for $k = 1, \ldots, N$, the observed signal $\mathbf{y}(\tau_k, \cdot)$, is represented by its realizations as a 4×4 matrix

$$Y^{(k)} = \{y_{\ell,r}^{(k)}\}_{\ell,r=1}^4 = [\mathbf{y}(\tau_k, \omega_1), \ldots, \mathbf{y}(\tau_k, \omega_4)]. \tag{2}$$

A column of matrix $Y^{(k)}$, $\mathbf{y}(t_k, \omega_i) \in \mathbb{R}^4$, represents the realization of the signal $\mathbf{y}(t, \omega_i)$ at time $t = \tau_k$ generated by the random event ω_i, for each $i = 1, 2, \ldots, 4$. Thus, all observed signals are given by the 4×484 matrix $Y = [Y^{(1)}, \ldots, Y^{(121)}]$ represented in Fig. 1 (a).

Suppose that, for $j = 1, \ldots, p$, information on the references signals can only be obtained at some times $t_1 = \tau_1$,

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$t_{j+1} = \tau_{12j+1}$ where $j = 1, \ldots, 10$ (see (1)) in the form of samples given by 4×4 matrices

$$\tilde{X}^{(j)} = [\tilde{\mathbf{x}}(t_j, \omega_1), \ldots, \tilde{\mathbf{x}}(t_j, \omega_4)] = \{\tilde{x}^{(k)}_{\ell,r}\}^4_{\ell,r=1}. \quad (3)$$

Fig. 1 demonstrates a typical situation with noisy observed signals and sparse information on the reference signals. In Example 2 below we show that, under certain conditions, the proposed technique allows us to estimate the signals of interest with an acceptable accuracy.

B. Formalization of the problem

To formalize the problem, we write $\{\Omega, \Sigma, \mu\}$ for a probability space where Ω is the set of all experimental outcomes, $\Sigma \subset \Omega$ is a sigma-algebra of measurable sets known as the event space and μ is a non-negative probability measure with $\mu(\Omega) = 1$. We denote by $\mathcal{K}_x = \{\boldsymbol{x}_\omega \mid \omega \in \Omega\}$ a set of reference stochastic signals and by $\mathcal{K}_y = \{\boldsymbol{y}_\omega \mid \omega \in \Omega\}$ a set of observed signals.

In an intuitive way, $\mathbf{y}$ can be regarded as a noise-corrupted version of $\mathbf{x}$. For example, $\mathbf{y}$ can be interpreted as $\mathbf{y} = \mathbf{x} + \mathbf{n}$ where $\mathbf{n}$ is white noise. We do not restrict ourselves to this simplest version of $\mathbf{y}$ and assume that the dependence of $\mathbf{y}$ on $\mathbf{x}$ and $\mathbf{n}$ is arbitrary. Note that, theoretically, $\mathcal{K}_x$ and $\mathcal{K}_y$ are infinite signal sets. In practice, however, sets $\mathcal{K}_x$ and $\mathcal{K}_y$ are finite and large, each with, say, N signals.

To each random outcome $\omega \in \Omega$ we associate a unique signal pair $(\boldsymbol{x}_\omega, \boldsymbol{y}_\omega)$ where $\boldsymbol{x}_\omega : T \to \mathcal{C}^{0,1}(T, \mathbb{R}^m)$ and $\boldsymbol{y}_\omega : T \to \mathcal{C}^{0,1}(T, \mathbb{R}^n)$. The space $\mathcal{C}^{0,1}(T, \mathbb{R}^p)$ is the set of vector-valued Hölder continuous functions $\boldsymbol{f}$ of order 1 with $\boldsymbol{f}(t) \in \mathbb{R}^p$ and $\|\boldsymbol{f}(s) - \boldsymbol{f}(t)\| \leq K|s-t|$ (see [1], p. 96.) Write

$$\mathcal{P} = \mathcal{K}_x \times \mathcal{K}_y = \{(\boldsymbol{x}_\omega, \boldsymbol{y}_\omega) \mid \omega \in \Omega\} \quad (4)$$

for the set of all such signal pairs. For each $\omega \in \Omega$, the components $\boldsymbol{x}_\omega = \boldsymbol{x}_\omega(t), \boldsymbol{y}_\omega = \boldsymbol{y}_\omega(t)$ are Lipschitz continuous vector-valued functions on T [1].

We wish to construct an estimator $F^{(p-1)}$ that estimates each reference signal $\boldsymbol{x}_\omega(t)$ in $\mathcal{P}$ from related observed input $\boldsymbol{y}_\omega(t)$ under the restriction that *a priori* information on only a *few* reference signals, $\boldsymbol{x}_\omega(t_1)$, ..., $\boldsymbol{x}_\omega(t_p)$, is available where $p \ll N$.

In more detail, this restriction implies the following. Let us denote by $\mathcal{K}_x^{(p)}$ a set of p signals $\boldsymbol{x}_\omega(t_1), \ldots, \boldsymbol{x}_\omega(t_p)$ for which *a priori* information is available. A set of associated observed signals $\boldsymbol{y}_\omega(t_1)$, ..., $\boldsymbol{y}_\omega(t_p)$ is denoted by $\mathcal{K}_y^{(p)}$. Then for all $\boldsymbol{y}_\omega(t)$ that do not belong to $\mathcal{K}_y^{(p)}$, $\boldsymbol{y}_\omega(t) \notin \mathcal{K}_y^{(p)}$, estimator $F^{(p-1)}$ is said to be the *blind* estimator [2], [3], [4], [5] since no information on $\boldsymbol{x}_\omega(t) \notin \mathcal{K}_x^{(p)}$ is available. If $\boldsymbol{y}_\omega(t) \in \mathcal{K}_y^{(p)}$ then $F^{(p-1)}$ becomes a *nonblind* estimator since information on $\boldsymbol{x}_\omega(t) \in \mathcal{K}_x^{(p)}$ is available. Thus, depending on $\boldsymbol{y}_\omega(t)$, estimator $F^{(p-1)}$ is classified differently. Therefore, such a procedure of estimating reference signals in $\mathcal{K}_x$ is here called the *almost blind* estimation.

C. Differences from known techniques

We would like to note that the *almost blind* estimation is different from known methods such as nonblind [6]–[18], semiblind and blind techniques [2]–[5], [19]–[22].Indeed, at each particular time $t \in T$, the input of the *almost blind* estimator $F^{(p-1)}$ developed below in this paper, is a random vector $\boldsymbol{y}_\omega(t)$. Thus, for different $t \in T$, the input is a different random vector $\boldsymbol{y}_\omega(t)$ but we wish to keep *the same estimator* $F^{(p-1)}$ for any $t \in T$, i.e., for any observed signal $\boldsymbol{y}_\omega(t)$ in the set $\mathcal{K}_y$. The literature on these subjects is very abundant. Here, we listed only some related references.

By known techniques in [2]–[16] and [19]–[22], an estimator (here, we choose the united term 'estimator' to denote an equalizer or a system) is specifically designed for *each* particular input–output pair represented by random vectors. That is, for different inputs (observed signals) $\boldsymbol{y}_\omega(t)$, known techniques require different specified estimators and the number of estimators should be equal to a number of processed signals. In the case of *large signal sets*, such approaches become inconvenient because the number of signals N can be very large as it is supposed in this paper. For example, in problems related to DNA analysis, $N = \mathcal{O}(10^4)$. Therefore, the inconvenient restriction of using *a priori* information on only p reference signals, with $p \ll N$, is quite significant. At the same time, beside difficulties that this restriction imposes on the estimation procedure, we use it in a way that allows us to avoid the hard work associated with known techniques applied to large signal sets. To the best of our knowledge, the exception is the methodology in [17], [18], where, for estimation of a set of signals, the single estimator is constructed. The estimation techniques in [17], [18] exploit information in the form of a vector obtained, in particular, from averaging over signals in $\mathcal{K}_x^{(p)}$.

Further, the semiblind techniques are not applicable to the considered problem because they require a knowledge of some 'parts' of each reference signal in $\mathcal{K}_x$ (e.g., see [3], [5], [19]) but it is not the case here. Although the blind techniques allow us to avoid this restriction, it is known that they have difficulties related to accuracy and computational load. In the problem under consideration, the advantage is a knowledge of some (small) part of the set of reference signals. It is natural to use this advantage in the estimator structure and we will do it in Section II.

Nonblind estimators [6]–[16] are not applicable here because they require *a priori* information on each reference signal in $\mathcal{K}_x$ (e.g., a knowledge of covariance matrix $E[\boldsymbol{x}_\omega \boldsymbol{y}_\omega^T]$ where E is the expectation operator). In particular, it is known that there are significant advantages in adaptive or recursive estimators (e.g., associated with Kalman filtering approach [23]) and it may well be possible to embed our estimator into such an environment—but that is not our particular concern here. Further, we note that much of the literature on piecewise linear estimators [24]–[28] seems to be *not directly relevant* to the estimator proposed here. In the first instance papers such as [24]–[28] are mostly concerned with the theoretical problems of approximation by piecewise linear functions on multi-dimensional domains which is *not the case here.*

Also, unlike many known techniques, we consider the case of observations corrupted by an arbitrary noise (not by an

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additive noise only) and design the estimator in terms of the Moore-Penrose pseudo-inverse matrix [29]. Therefore it is always well defined.

II. The main results

In this section, we outline the rationale for the proposed estimator and state the main results.

A. Some preliminaries

The proposed estimator $F^{(p-1)}$ is adaptive to a sparse set $\mathcal{K}_x^{(p)}$.

The conceptual device behind the proposed estimator is a linear interpolation of an optimal incremental estimation applied to random signal pairs $(\boldsymbol{x}_\omega(t_j), \boldsymbol{y}_\omega(t_j))$ and $(\boldsymbol{x}_\omega(t_{j+1}), \boldsymbol{y}_\omega(t_{j+1}))$, for $j = 1, \ldots, p-1$, interpreted an extension of the Least Squares Linear (LSL) estimator (see, for example, [6], [11], [16]). Although this idea may seem reasonable, the detailed justification of the new estimator is not straightforward and requires careful analysis. We shall do this by establishing an upper bound for the associated error and by showing that this upper bound is directly related to the expected error for an incremental application of the optimal LSL estimator. In Section II-B below, we will show that such an estimator is possible under quite unrestrictive assumptions.

Since the estimator proposed below is based on an extension of the LSL estimator it is convenient to sketch known related results here. Consider a *single* random signal pair $(\boldsymbol{x}(\omega), \boldsymbol{y}(\omega))$ where $\boldsymbol{x} \in L^2(\Omega, \mathbb{R}^m)$ and $\boldsymbol{y} \in L^2(\Omega, \mathbb{R}^n)$ with zero mean $(E[\boldsymbol{x}], E[\boldsymbol{y}]) = (\mathbf{0}, \mathbf{0})$, where $\mathbf{0}$ is the zero vector. Note that here, $\boldsymbol{x}$ and $\boldsymbol{y}$ do not depend on t as above. The estimate $\widehat{\boldsymbol{x}}$ of the reference vector $\boldsymbol{x}$ by the optimal least squares linear estimator is given by

$$\widehat{\boldsymbol{x}}(\omega) = E_{\boldsymbol{x}\boldsymbol{y}} E_{\boldsymbol{y}\boldsymbol{y}}^{\dagger}\, \boldsymbol{y}(\omega) \tag{5}$$

where $E_{\boldsymbol{x}\boldsymbol{y}} = E[\boldsymbol{x}\boldsymbol{y}^T]$ and $E_{\boldsymbol{y}\boldsymbol{y}} = E[\boldsymbol{y}\boldsymbol{y}^T]$ are known covariance matrices and $E_{\boldsymbol{y}\boldsymbol{y}}^{\dagger}$ is the Moore-Penrose pseudo-inverse of $E_{\boldsymbol{y}\boldsymbol{y}}$. By the LSL estimator, matrices $E_{\boldsymbol{x}\boldsymbol{y}}$ and $E_{\boldsymbol{y}\boldsymbol{y}}^{\dagger}$ should be specified for each signal pair $(\boldsymbol{x}(\omega), \boldsymbol{y}(\omega))$.

Further, for a justification of our estimator, we need some more notation as follows. It is convenient to denote $\boldsymbol{x}(t,\omega) = \boldsymbol{x}_\omega(t)$ and $\boldsymbol{y}(t,\omega) = \boldsymbol{y}_\omega(t)$ so that $\boldsymbol{x}(t,\omega) \in \mathbb{R}^m$ and $\boldsymbol{y}(t,\omega) \in \mathbb{R}^n$.

B. The piecewise LSL interpolation estimator

For each signal pair (or vector function pair) in the set $\mathcal{P}$, $(\boldsymbol{x}(t,\omega), \boldsymbol{y}(t,\omega))$, we assume that $(E[\boldsymbol{x}(t,\cdot)], E[\boldsymbol{y}(t,\cdot)]) = (\mathbf{0}, \mathbf{0})$. To begin the estimation process we need to find an initial estimate $\widehat{\boldsymbol{x}}(t_1,\omega)$. It is assumed this can be found by some known method. Further, let us consider a signal estimation procedure at $t_2, \cdots, t_p$. We use an inductive argument to define an incremental estimation procedure. Consider a typical interval $[t_j, t_{j+1}]$ and define incremental random vectors

$$\boldsymbol{v}_j(\omega) = \boldsymbol{x}(t_{j+1},\omega) - \boldsymbol{x}(t_j,\omega) \in \mathbb{R}^m, \tag{6}$$

$$\boldsymbol{w}_j(\omega) = \boldsymbol{y}(t_{j+1},\omega) - \boldsymbol{y}(t_j,\omega) \in \mathbb{R}^n \tag{7}$$

and construct the optimal linear estimate

$$\widehat{\boldsymbol{v}}_j(\omega) = E_{\boldsymbol{v}_j\boldsymbol{w}_j} E_{\boldsymbol{w}_j\boldsymbol{w}_j}^{\dagger}\, \boldsymbol{w}_j(\omega) \tag{8}$$

of the increment $\boldsymbol{v}_j(\omega)$ for each $j = 1, \ldots, p-1$. We will write

$$B_j = E_{\boldsymbol{v}_j\boldsymbol{w}_j} E_{\boldsymbol{w}_j\boldsymbol{w}_j}^{\dagger} \in \mathbb{R}^{m\times n}. \tag{9}$$

Define the estimate at point t_{j+1} by setting $\widehat{\boldsymbol{x}}(t_{j+1},\omega) = \widehat{\boldsymbol{x}}(t_j,\omega) + \widehat{\boldsymbol{v}}_j(\omega)$. Thus we have

$$\begin{aligned}\widehat{\boldsymbol{x}}(t_{j+1},\omega) &= \widehat{\boldsymbol{x}}(t_j,\omega) + B_j[\boldsymbol{y}(t_{j+1},\omega) - \boldsymbol{y}(t_j,\omega)] \\ &= \boldsymbol{\epsilon}_j(\omega) + B_j\boldsymbol{y}(t_{j+1},\omega)\end{aligned} \tag{10}$$

where we write

$$\boldsymbol{\epsilon}_j(\omega) = \widehat{\boldsymbol{x}}(t_j,\omega) - B_j\boldsymbol{y}(t_j,\omega). \tag{11}$$

Note that this definition can be rewritten more suggestively as

$$\widehat{\boldsymbol{x}}(t_j,\omega) = \boldsymbol{\epsilon}_j(\omega) + B_j\boldsymbol{y}(t_j,\omega) \tag{12}$$

for each $j = 1, \ldots, p-1$.

The formula (10) shows that on each subinterval $[t_j, t_{j+1}]$ the estimate of the reference signal at t_{j+1} is obtained from the initial estimate at t_j by adding the optimal LSL estimate of the increment.

Now, consider a signal estimation at any $t \in [a,b]$. The first step is simply to extend the formulæ (10) and (12) to all $t \in [t_j, t_{j+1}]$ by defining

$$\widehat{\boldsymbol{x}}(t,\omega) = \boldsymbol{\epsilon}_j(\omega) + B_j\boldsymbol{y}(t,\omega). \tag{13}$$

Thus, the incremental estimation across each subinterval is extended to every point within the subinterval. Because of determining estimate $\widehat{\boldsymbol{x}}(t_{j+1},\omega)$ in the form (8)–(10) we interpret this procedure as the *LSL piecewise interpolation*.

The incremental estimations are collected together in the following way. For each $j = 1, 2, \ldots, p-1$, write

$$F_j[\boldsymbol{y}(t,\omega)] = \boldsymbol{\epsilon}_j(\omega) + B_j\boldsymbol{y}(t,\omega) \tag{14}$$

for all $t \in [t_j, t_{j+1}]$ and hence define the *piecewise LSL interpolation estimator* by setting

$$F^{(p-1)}[\boldsymbol{y}(t,\omega)] = \sum_{j=1}^{p-1} F_j[\boldsymbol{y}(t,\omega)][u(t-t_j) - u(t-t_{j+1})] \tag{15}$$

for all $t \in [a,b]$ where $u(t) = \begin{cases} 1 & \text{for } t > 0 \\ 0 & \text{otherwise.}\end{cases}$ is the unit step function. Thus we can now use the estimate

$$\widehat{\boldsymbol{x}}(t,\omega) = F^{(p-1)}[\boldsymbol{y}(t,\omega)] \tag{16}$$

for all $(t,\omega) \in T \times \Omega$. The idea of a piecewise LSL interpolation estimator on T seems intuitively reasonable for signals with a well defined gradient over T.

We note that by (9)-(16), the estimator $F^{(p-1)}$ is adaptive to a variation of signals in $\mathcal{K}_x^{(p)}$. A change of signals in $\mathcal{K}_x^{(p)}$ implies a change of determinations of sub-estimators B_j in (9) and keep the same structure of the $F^{(p-1)}$.

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C. Justification of the LSL interpolation estimator

We wish to justify the proposed estimator by establishing an upper bound for the associated error.

To explain the technical details we introduce some further terminology.

Let us denote $\|\boldsymbol{x}(t,\cdot)\|^2_\Omega = \int_\Omega \|\boldsymbol{x}(t,\omega)\|^2 d\mu(\omega)$. Assume that for all $t \in T$, we have

$$\|\boldsymbol{x}(t,\cdot)\|^2_\Omega < \infty \quad \text{and} \quad \|\boldsymbol{y}(t,\cdot)\|^2_\Omega < \infty, \tag{17}$$

where $\|\boldsymbol{x}(t,\omega)\|$ and $\|\boldsymbol{y}(t,\omega)\|$ are the Euclidean norms for $\boldsymbol{x}(t,\omega)$ and $\boldsymbol{y}(t,\omega)$ for each $(t,\omega) \in T \times \Omega$, respectively. Thus we will say that the signals are square integrable in ω and write $\boldsymbol{x}(t,\cdot) \in L^2(\Omega)$ and $\boldsymbol{y}(t,\cdot) \in L^2(\Omega)$ for each fixed $t \in T$.

For each $t \in T$, let $\mathcal{F} = \{\boldsymbol{f} : T \times \Omega \to \mathbb{R}^m \mid \boldsymbol{f}(t,\cdot) \in L^2(\Omega, \mathbb{R}^m)\}$ and define

$$\begin{aligned}\|\boldsymbol{f}\|_{T,\Omega} &= \frac{1}{b-a}\int_{T\times\Omega} \|\boldsymbol{f}(t,\omega)\|\; dt\, d\mu(\omega)\\ &= \frac{1}{b-a}\int_T E[\|\boldsymbol{f}(t,\cdot)\|]\, dt\end{aligned}$$

for each $\boldsymbol{f} \in \mathcal{F}$ where $\|\boldsymbol{f}(t,\omega)\|$ is the Euclidean norm of $\boldsymbol{f}(t,\omega)$ on $\mathbb{R}^m$ for all $(t,\omega) \in \mathbb{R}^m$. Suppose that for all $(\boldsymbol{x},\boldsymbol{y}) \in \mathcal{P}$ there exist constants $\gamma_j, \delta_j > 0$ such that

$$\|\boldsymbol{x}(s,\omega) - \boldsymbol{x}(t,\omega)\| \le \gamma_j |s-t|, \tag{18}$$

$$\|\boldsymbol{y}(s,\omega) - \boldsymbol{y}(t,\omega)\| \le \delta_j |s-t| \tag{19}$$

for all $(s,\omega), (t,\omega) \in [t_j, t_{j+1}] \times \Omega$, i.e. we suppose that the Lipschitz constants in (18) are independent of ω.

The error bound for the piecewise LSL interpolation estimator is established in Theorem 1 below.

Theorem 1: If condition (18) is satisfied then the error $\epsilon_p = \|\boldsymbol{x} - F^{(p-1)}[\boldsymbol{y}]\|_{T,\Omega}$ associated with the piecewise LSL interpolation estimator satisfies the inequality

$$\epsilon_p \le \max_{j=1,\dots,p-1} \{(\gamma_j + \|B_j\|_2 \delta_j)|t_{j+1} - t_j| \tag{20}$$

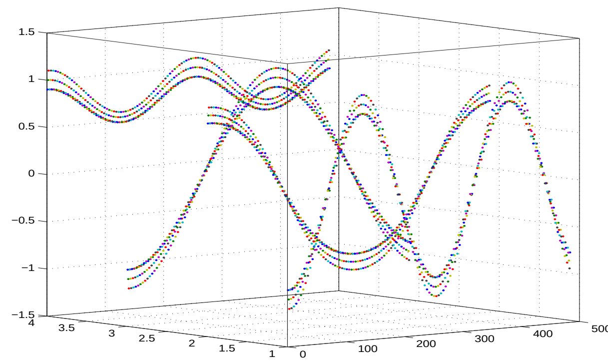
$$+ \left[\|E^{1/2}_{\boldsymbol{v}_j,\boldsymbol{v}_j}\|^2_F - \|E_{\boldsymbol{v}_j\boldsymbol{w}_j}(E^{1/2}_{\boldsymbol{w}_j\boldsymbol{w}_j})^\dagger\|^2_F\right]^{1/2}\} \tag{21}$$

where $\|B_j\|_2$ denotes the 2-norm given by the square root of the largest eigenvalue of $B_j^T B_j$ and $\|\cdot\|$ denotes the Frobenius norm.

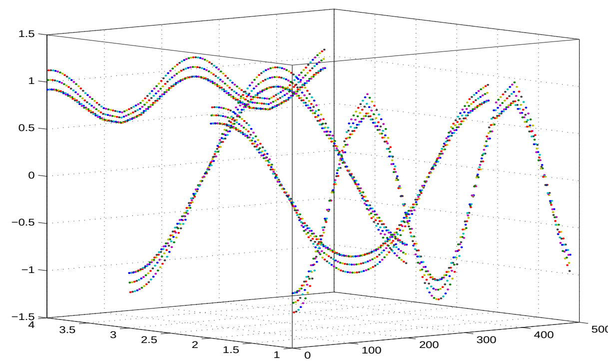
Example 2: The time interval T is the same as in Example 1. At each time τ_k, for $k = 1,\dots,N$, the training reference signal $\mathbf{x}(\tau_k,\cdot)$ is represented by its realizations as a 4×4 matrix

$$X^{(k)} = [\mathbf{x}(\tau_k,\omega_1),\dots,\mathbf{x}(\tau_k,\omega_4)] = \{\tilde{x}^{(k)}_{\ell,r}\}^4_{\ell,r=1}. \tag{22}$$

where $x^{(k)}_{1,1} = 0.918\eta^{(k)}_1, x^{(k)}_{1,2} = 1.02\eta^{(k)}_2, x^{(k)}_{1,3} = 1.122\eta^{(k)}_3, x^{(k)}_{1,4} = 0.918\eta^{(k)}_4, \eta^{(k)}_1 = -\cos(2k),\ \eta^{(k)}_2 = \sin(\cos(k)), \eta^{(k)}_3 = -\cos(k), \eta^{(k)}_4 = \cos(\sin(k))$. All training reference signals are simulated as a 4×484 matrix $X = [X^{(1)},\dots,X^{(121)}]$ shown in Fig. 2 (a). Note that in (3), due to measurement errors the values of $\tilde{x}^{(k)}_{\ell,r}$ are different from the values of $x^{(k)}_{\ell,r}$. The observed signals in Example 1 were simulated from X by adding random noise.



(a) Training reference signals.



(b) Their estimates by filter $F^{(10)}$.

Fig. 2. Training signals and their estimates considered in Example 2.

The estimates of the reference signals by filter $F^{(p-1)}$, for $p = 11$, obtained on the basis of the information represented in Fig. 1 (b) are given in Fig. 2 (b). The covariance matrices are estimated from samples $Y^{(j)}$ and $\tilde{X}^{(j)}$ taken at times t_j, for $j = 1,\dots,11$ (see Example 1). The averaging polynomial filter [16] gives much worse accuracy.

III. Conclusion

The piecewise least squares linear (LSL) interpolation estimator was developed to estimate a large set of random signals of interest from the set of observed data. The distinctive feature is that *a priori* information can be obtained on only a *few* reference signals in the form of samples. Since no information of the major part of the set of reference signals is known, such a procedure is called *almost blind* estimation.

The proposed estimator mitigates to some extent the difficulties associated with existing estimation approaches such as the necessity to know information (in the form of a sample, for instance) on *each* random reference signal; invertibility of the matrices used to define the estimators; and demanding computational work.

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