

A Noise-Robust Method with Smoothed L1/L 2 Regularization for Sparse Moving-Source Mapping

Mai Quyen I Pham, Benoit Oudompheng, Jerome I. Mars, Barbara Nicolas

▶ To cite this version:

Mai Quyen I Pham, Benoit Oudompheng, Jerome I. Mars, Barbara Nicolas. A Noise-Robust Method with Smoothed L1/L 2 Regularization for Sparse Moving-Source Mapping. Signal Processing, 2016, 135 (June 2017), pp.96-106. 10.1016/j.sigpro.2016.12.022 . hal-01426251

HAL Id: hal-01426251 https://hal.science/hal-01426251v1

Submitted on 4 Jan 2017

HAL is a multi-disciplinary open access archive for the deposit and dissemination of scientific research documents, whether they are published or not. The documents may come from teaching and research institutions in France or abroad, or from public or private research centers. L'archive ouverte pluridisciplinaire **HAL**, est destinée au dépôt et à la diffusion de documents scientifiques de niveau recherche, publiés ou non, émanant des établissements d'enseignement et de recherche français ou étrangers, des laboratoires publics ou privés.

A Noise-Robust Method with Smoothed ℓ_1/ℓ_2 Regularization for Sparse Moving-Source Mapping

Mai Quyen Pham^{a,*}, Benoit Oudompheng^{a,b}, Jérôme I. Mars^a, Barbara Nicolas^{a,c}

^a Université Grenoble-Alpes, GIPSA-lab, F-38000 Grenoble, France ^b MicrodB, 28 Chemin du Petit Bois, BP 36, 69131 Écully Cedex, France ^c Université de Lyon, CREATIS, CNRS UMR5220; Inserm U1044; INSA-Lyon; Université Lyon 1, France

Abstract

The method described here performs blind deconvolution of the beamforming output in the frequency domain. To provide accurate blind deconvolution, sparsity priors are introduced with a smoothed ℓ_1/ℓ_2 regularization term. As the mean of the noise in the power spectrum domain depends on its variance in the time domain, the proposed method includes a variance estimation step, which allows more robust blind deconvolution. Validation of the method on both simulated and real data, and of its performance, are compared with two well-known methods from the literature: the deconvolution approach for the mapping of acoustic sources, and sound density modeling.

Keywords: Smoothed ℓ_1/ℓ_2 regularization, Sparse representation, Proximal forward-backward, Acoustic moving-source localization, Beamforming blind deconvolution, Robustness algorithms.

19

21

22

23

24

25

26

27

28

29

30

31

32

33

34

35

36

37

38

39

1 1. Introduction

Blind deconvolution has a central role in the field 2 of signal and image processing. It has many applica-3 tions in communications [1], nondestructive testing 4 [2], image processing [3, 4, 5], medical imaging pro-5 cessing [6], and in acoustics [7]. Moreover, in under-6 water acoustics, blind deconvolution methods have 7 already been proposed to estimate simultaneously 8 the transfer function of the environment and the un-9 known source signal in multipath underwater sound 10 channels [8, 9, 10]. In many realistic scenarios, the 11 blurring kernel (or the system) is imprecise or not 12 known. Thus, the deconvolution problem becomes 13 blind and underdetermined, and often requires ad-14 ditional hypotheses. 15

One possible additional hypothesis is the sparsity
of the signal, which is an extensively studied topic
in signal processing. The main idea is to find the

*Corresponding author. Email addresses:

mai-quyen.pham@gipsa-lab.grenoble-inp.fr

Preprint submitted to Signal Processing

most compact representation of a signal that consists of only a few nonzero elements. In acoustic signal processing, sparsity can be introduced either in the system or the signal domain (input). In the experimental context of this paper, the goal is to perform source mapping in a moving-source context¹. The measurements are the pressures recorded on a horizontal line array during the pass-by experiment, and the signal of interest is the source locations inside the global vehicle. The positions of sources can be considered as sparsely distributed on a calculation grid. The question is then which measure can be used to evaluate the sparsity of a signal? In [11], Pereira used ℓ_2^2 -norm as a penalty to stabilize inverse problem solutions, which can be achieved using an adapted Tikhonov regularization method. However this penalty is not adapted for the considered case of sparse source positions. An ℓ_1 -norm is popular to restore the sparsity of the solution, as proposed in [12, 13]. However, in [14], Benichoux et al. showed that the use of the ℓ_1 norm suffers from

⁽Mai Quyen Pham), benoit.oudompheng@grenoble-inp.org (Benoit Oudompheng),

jerome.mars@gipsa-lab.grenoble-inp.fr (Jérôme I. Mars), barbara.nicolas@creatis.insa-lyon.fr (Barbara Nicolas)

¹In this paper the terms "source localization" and "source mapping" are equivalent. These refer to the goal of the paper, which is to map noise sources inside a global vehicle (here a boat) during a pass-by experiment.

scaling and shift ambiguities due to the nonlinear 1 relation between the blurring kernel and the signal, 2 as also discussed in [15, 16]. Felix et al. extended 3 this result for the case of ℓ_p , (p < 1)-norm in [17]. 4 In particular, both of these articles showed that us-5 ing the ℓ_1/ℓ_2 function can overcome this difficulty. 6 However, the ℓ_1/ℓ_2 function creates some difficulties 7 when solving the nonconvex and nonsmooth mini-8 mization problems that prevent the use of such a 9 penalty term in current restoration methods. In the 10 present paper, we propose to use the smoothed ℓ_1/ℓ_2 11 ratio mentioned in [18] as a penalty. This penalty 12 also overcomes the scaling and shift-ambiguity is-13 sues. Moreover, this ratio can be used to force the 14 sparse representation of the signal in a blind de-15 convolution of moving-source mapping. Note that 16 this is a different problem from classical blind de-17 convolution in underwater acoustics, as we consider 18 several sources in a nonmultipath environment. The 19 problem is presented in the power spectrum domain. 20 where the noise variance is a parameter that has 21 to be estimated jointly with the autospectra and 22 the blur. This fundamental problem was not taken 23 into account with the original SOOT algorithm in 24 [18], while it is explicitly dealt with in our proposed 25 method. 26

This paper is organized as follows. Following this 27 Introduction, Section 2 is devoted to a review of the 28 related framework for moving-source mapping us-29 ing deconvolution. Section 3 presents the proposed 30 forward model, and Section 4 describes the mini-31 mization problem, the proposed algorithm, and some 32 mathematical tools that are essential to this method-33 ology. The performance of the proposed method 34 is assessed in Section 5, where we detail the cho-35 sen optimization criteria and provide comparisons 36 with two methods: the deconvolution approach for 37 the mapping of acoustic sources (DAMAS-MS) and 38 the sound density modeling (SDM) methods. The 39 proposed methodology is first evaluated on realistic 40 synthetic data, and then it is applied to real data 41 recorded in Lake Castillon (Verdon Gorges, France). 42 The conclusions and perspectives are drawn up in 43 Section 6. 44

45 2. Related work

In this section, we briefly present the classical
methods that have been developed for acousticsource localization.

⁴⁹ Many methods have been developed to solve this ¹⁰¹ ⁵⁰ problem based on array processing. The most clas- ¹⁰² sical one is beamforming [19], which has been extensively used due to its robustness against noise and environmental mismatch. However, classical beamforming cannot be used for pass-by experiments, where the 'vehicle' is moving and the goal is to map the different acoustic noise sources in the vehicle. Here instead, source mapping is achieved by the extension to beamforming for moving sources (BF-MS) [20].

Nevertheless, the spatial resolution of BF-MS is limited, as the image of a point source is the array transfer function, which is comprised of a main lobe and secondary lobes. Consequently, many improvements have been proposed to overcome this problem, including a hardware strategy to reduce the sidelobe levels, where the resolution of the main lobes is through optimization of the antenna geometry. In particular, several optimizations of the sensor positions of linear antennas have been proposed through the use of pseudo-random distributions [21, 22, 23]. Furthermore, a numerical strategy classically uses the weighting coefficients, which shade the array aperture and thus taper the side lobes, and as a consequence, also enlarge the main lobe [24]. Another common approach is to use deconvolution methods.

Recently, Sijtsma proposed an extended version of the deconvolution method CLEAN [25, 26] for moving sources, which is known as CLEAN-SC (*i.e.*, CLEAN based on spatial-source coherence) [27]. This method follows an approach similar to the matching pursuit method [28]. CLEAN-SC provides satisfactory results for high signal-to-noise ratios (SNRs), but requires *a-priori* knowledge of the number of sources which is not always known in practical cases. Brooks and Humphreys developed another approach, known as DAMAS [29], and its extensions [20, 30]. These algorithms use the iterative Gauss-Seidel method for solving the linear inverse problem under the nonnegative constraint on source powers. A particular extension was dedicated to moving sources, as DAMAS-MS [20], which improves moving-source mapping in the context of a high SNR. Another popular method that was also developed for moving sources is the SDM method of [31], which is based on a gradient-descent optimization technique. This represented the first use of optimization techniques with a noise prior and constraints on the signal. These two methods (*i.e.*, DAMAS-MS and SDM) will be used as the references for comparison with our proposed method.

In the case of low SNRs, the problem is difficult to solve, and thus some other approaches need to

51

52

53

54

55

56

57

58

59

60

61

62

63

64

65

66

67

68

69

70

71

72

73

74

75

76

77

78

79

80

81

82

83

84

85

86

87

88

89

90

91

92

93

94

95

96

97

98

99



Figure 1: Modeling of the forward problem

be developed. In array processing, Swindlehurst 1 and Kailath [32] proposed a first-order perturbation 2 analysis of the multiple signal classification (MU-3 SIC) and root-MUSIC algorithms for various model 4 errors. Another possibility is to include a regular-5 ization term to stabilize the solution. The Tikhonov 6 regularization is applied to jet noise-source localization [33]. The sparse distribution of sources is also 8 commonly used, as in [12, 34, 35, 13, 36, 37, 38]. q These methods were developed for fixed-source lo-10 calization and have not currently been extended to 11 moving sources. 12

The goal of the present paper is to propose a 13 new blind deconvolution method that is applied to 14 BF-MS results to improve moving-source mapping. 15 The strategy is to formulate the forward prob-16 lem as an optimization problem, with constraints 17 that are derived from the physical context. The 18 proposed cost function contains several parts: (i) a 19 data-fidelity term that accounts for the noise char-20 acteristics; (ii) the smoothed ℓ_1/ℓ_2 ratio [18] that 21 promotes sparsity in the moving-source locations; 22 and (iii) the knowledge of some physical properties 23 of the sources and the system, and of the variance 24 noise, introduced through indicator functions. 25

²⁶ 3. Observation model

27 3.1. Beamforming for moving sources

Beamforming for moving sources compensates for 28 the Doppler effect and back-propagates the pressures 29 measured by the M sensor array to a calculation 30 grid of N points, which correspond to the possible 31 source locations. We consider the classical case of 32 pass-by experiments, in the far-field, with sources 33 that share the same global movement and have low 34 Mach numbers, $||Ma'|| \ll 1$. For these conditions, 35 some assumptions can be made over short time 36 intervals of duration T, which are referred to as 37 snapshots [20], whereby: 38

- 1) The sources are in fixed positions;
- 2) The Doppler effect is negligible at the frequencies and speeds of interest (*i.e.*, it does not exceed the frequency resolution defined for the localization results).

Under these assumptions, BF-MS can be implemented in a simple way in the frequency domain. The measured acoustic pressures are temporally sliced into K snapshots that are indexed by k. The calculation grid of N points is defined for the snapshot k using the *a*-priori known global trajectory of the vehicle. Note that this grid moves according to this trajectory.

For the snapshot k, the pressures measured by these M sensors at time $t \in [1, T]$ are denoted as $\tilde{\mathbf{p}}_t^k \in \mathbb{R}^M$, which can be divided into two parts:

$$\check{\mathbf{p}}_t^k = \check{\overline{\mathbf{p}}}_t^k + \check{\mathbf{r}}_t^k \tag{1}$$

in which $\check{\mathbf{p}}_t^k$ are the pressures measured by the M sensors at time t for the ideal case without noise, and $\check{\mathbf{r}}_t^k$ is an additive noise in the recording domain. This defines the vectors:

$$\begin{split} \check{\mathbf{P}}^{k} &= \left[(\check{\mathbf{p}}_{1}^{k})^{\top}, \, (\check{\mathbf{p}}_{2}^{k})^{\top}, \, \dots, \, (\check{\mathbf{p}}_{T}^{k})^{\top} \right]^{\top} \in \mathbb{R}^{MT}, \\ \check{\overline{\mathbf{P}}}^{k} &= \left[(\check{\overline{\mathbf{p}}}_{1}^{k})^{\top}, \, (\check{\overline{\mathbf{p}}}_{2}^{k})^{\top}, \, \dots, \, (\check{\overline{\mathbf{p}}}_{T}^{k})^{\top} \right]^{\top} \in \mathbb{R}^{MT}, \\ \check{\mathbf{R}}^{k} &= \left[(\check{\mathbf{r}}_{1}^{k})^{\top}, \, (\check{\mathbf{r}}_{2}^{k})^{\top}, \, \dots, \, (\check{\mathbf{r}}_{T}^{k})^{\top} \right]^{\top} \in \mathbb{R}^{MT}, \end{split}$$

where \cdot^{\top} denotes the transpose.

We now consider the pressures measured in the frequency domain between ζ_1 and $\zeta_F Hz$, which is related to a vector $\mathcal{F} = [\zeta_1, \ldots, \zeta_F] \in \mathbb{R}^F$. We define the Fourier transforms of $\check{\mathbf{P}}^k, \, \check{\mathbf{P}}^k$ and $\check{\mathbf{R}}^k$ for each snapshot k, as following:

$$\begin{aligned} \mathbf{P}^{k} &= \left[(\mathbf{p}_{1}^{k})^{\top}, \, (\mathbf{p}_{2}^{k})^{\top}, \, \dots, \, (\mathbf{p}_{F}^{k})^{\top} \right]^{\top} \in \mathbb{C}^{MF} \\ \overline{\mathbf{P}}^{k} &= \left[(\overline{\mathbf{p}}_{1}^{k})^{\top}, \, (\overline{\mathbf{p}}_{2}^{k})^{\top}, \, \dots, \, (\overline{\mathbf{p}}_{F}^{k})^{\top} \right]^{\top} \in \mathbb{C}^{MF} \\ \mathbf{R}^{k} &= \left[(\mathbf{r}_{1}^{k})^{\top}, \, (\mathbf{r}_{2}^{k})^{\top}, \, \dots, \, (\mathbf{r}_{F}^{k})^{\top} \right]^{\top} \in \mathbb{C}^{MF} \end{aligned}$$

where for every $f \in \{1, \ldots, F\}$, \mathbf{p}_{f}^{k} , $\mathbf{\overline{p}}_{f}^{k}$, and \mathbf{r}_{f}^{k} are the vectors that contain the Fourier transform coefficients $p_{f}^{k}(m), \mathbf{\overline{p}}_{f}^{k}(m)$, and $r_{f}^{k}(m)$ of the vectors $[\mathbf{\breve{p}}_{1}^{k}(m), \mathbf{\breve{p}}_{2}^{k}(m), \ldots, \mathbf{\breve{p}}_{T}^{k}(m)]^{\top}$, $[\mathbf{\breve{p}}_{1}^{k}(m), \mathbf{\breve{p}}_{2}^{k}(m), \ldots, \mathbf{\breve{p}}_{T}^{k}(m)]^{\top}$, and $[\mathbf{\breve{r}}_{1}^{k}(m), \mathbf{\breve{r}}_{2}^{k}(m), \ldots, \mathbf{\breve{r}}_{T}^{k}(m)]^{\top}$ at the frequency $\zeta_{f} \in \mathcal{F}$ (ζ_{f} is the f^{th} element of vector \mathcal{F}), for

70

68

46

47

48

49

50

51

52

53

54

55

56

57

58

60

61

62

63

¹ every $m \in \{1, \ldots, M\}$, respectively.

² The BF-MS computed for the n^{th} calculation ³ point at the frequency $\zeta_f \in \mathcal{F}$, and for the snapshot ⁴ $k, b_f^k(n)$, is given by:

$$b_f^k(n) = |\left(\mathbf{w}_{f,n}^k\right)^{\mathsf{H}} \mathbf{p}_f^k|^2 \tag{2}$$

36

37

38

41

42

43

44

45

46

48

49

50

51

54

55

56

57

58

59

60

61

62

63

64

65

67

69

70

71

⁵ where \cdot^{H} is the conjugate transpose, and $\mathbf{w}_{f,n}^{k}$ is the ⁶ steering vector of length M between the M sensors ⁷ and the n^{th} calculation point. The m^{th} element ⁸ $w_{f,n}^{k}(m)$ of $\mathbf{w}_{f,n}^{k}$ is:

$$w_{f,n}^{k}(m) = \left(\sum_{m'=1}^{M} \left(\frac{1}{d_{n,m'}^{k}}\right)^{2}\right)^{-1} \frac{\exp(-j\zeta_{f}d_{n,m}^{k})}{d_{n,m}^{k}}$$
(3)

⁹ where j is the square root of -1, and $d_{n,m}^k$ is the ¹⁰ distance between the m^{th} sensor and the n^{th} calcu-¹¹ lation point during the snapshot k. We then define ¹² the vector $\mathbf{b}_f \in \mathbb{R}^N$ with its n^{th} element $b_f(n)$ as ¹³ the estimate of the BF-MS output for the n^{th} cal-¹⁴ culation point, through averaging over all of the K¹⁵ snapshots; *i.e.*,

$$b_f(n) = \frac{1}{K} \sum_{k=1}^{K} b_f^k(n).$$
 (4)

¹⁶ Note that in the case considered, the receiver array ¹⁷ is a linear array along the x-axis. Consequently, ¹⁸ BF-MS is performed along the x dimension, and the ¹⁹ calculation grid is a one-dimension vector of length ²⁰ N along x. For more details on these computations, ²¹ we refer the reader to [39, 40].

22 3.2. Inverse problem formulation

²³ We set the following assumptions:

 24 (H1) : The sources are random variables that are mutually independent and stationary;

- $_{26}$ (H2) : The number M of the sensors is greater than
- the number N_s of the sources $(i.e., M > N_s)$, and these N_s sources are sparsely distributed on the calculation grid;

²⁹ On the calculation grid,

35

(H3) : The noise components are mutually indepen dent, and independent of the sources.

- ³² Using the expression of the BF-MS, and assuming
- that the sources are located at the N points of the
- $_{34}$ calculation grid, it is possible to express the BF-MS

output at a given frequency ζ_f , $\mathbf{b}_f \in \mathbb{R}^N$, by:

$$\mathbf{b}_f = \mathbf{A}_f \overline{\mathbf{q}}_f + \mathbf{z}_f \tag{5}$$

where $\mathbf{A}_f \in \mathbb{R}^{N \times N}$ (Fig. 1, middle) is the array transfer function matrix that contains the beamforming point-spread functions. The $(n, n') \in$ $\{1, \ldots, N\}^2$ element $a_f(n, n')$ of \mathbf{A}_f is:

$$a_f(n,n') = \frac{1}{K} \sum_{k=1}^{K} \left| \sum_{m=1}^{M} \left(w_{f,n}^k(m) \right)^{\mathsf{H}} \frac{\exp(-j\zeta_f d_{n',m}^k)}{d_{n',m}^k} \right|^2$$
(6)

 $\mathbf{z}_f \in \mathbb{R}^N$ is the measurement noise, and $\overline{\mathbf{q}}_f \in \mathbb{R}^N$ is the autospectra of the possible sources located at the *N* calculation points (Fig. 1, right), which are the unknowns to be estimated. This expression is frequently used in deconvolution [20, 27, 31, 41], although it needs the knowledge of matrix \mathbf{A}_f related to the environment and to the array to perform the deconvolution. Nowadays, some research projects are focused on uncertain cases with partially known or unknown ocean environments and experimental configurations [42]. Consequently, when \mathbf{A}_f is unknown, we propose in this paper to formulate the BF-MS output at the frequency ζ_f as a blind deconvolution problem:

$$\mathbf{b}_f = \mathbf{h}_f * \overline{\mathbf{q}}_f + \mathbf{z}_f \tag{7}$$

where $\overline{\mathbf{h}}_f \in \mathbb{R}^N$ is an unknown blur kernel, which needs to be estimated, as well as the autospectra of the sources. * is a discrete-time convolution operator (with appropriate boundary processing). We now turn our attention to the term \mathbf{z}_f , which corresponds to the additional noise. In the literature, several methods have been proposed with \mathbf{z}_f as a Gaussian noise with zero mean (which is not adapted to the BF-MS signal). We propose to introduce the noise in the time recording domain and to model its transformation throught BF-MS. In acoustics, the noise components $\check{\mathbf{r}}_{m,t}^k$ in the time domain can commonly be considered to be Gaussian, with zero mean and variance $\bar{\sigma}^2$. From Equations (1) and (2), and assumptions (H1) - (H3), we have:

$$b_f(n) = \frac{1}{K} \sum_{k=1}^{K} \left(|\left(\mathbf{w}_{f,n}^k\right)^{\mathsf{H}} \overline{\mathbf{p}}_f^k|^2 + |\left(\mathbf{w}_{f,n}^k\right)^{\mathsf{H}} \mathbf{r}_f^k|^2 \right)$$
(8)

Using Equation (8), we assume in this paper that the observation noise \mathbf{z}_f can be divided into two terms:

$$\mathbf{z}_{f} = \frac{1}{K} \left(\sum_{k=1}^{K} \| \mathbf{w}_{f,n}^{k} \|^{2} \right) \overline{\sigma}^{2} \mathbf{1}_{N} + \mathbf{e}_{f}$$
(9)

 $_1$ $\,$ where $\|\cdot\|$ is the $\ell_2\text{-norm}$ (which is also known as $\,_{21}$

the Euclidean norm), $\mathbf{1}_N$ is a vector of ones of length N, and $\mathbf{e}_f \in \mathbb{R}^N$ represents the remaining unknown

effects, where the amplitude of these remaining ef-

⁵ fects is much lower than that of the variance $\overline{\sigma}^2$ of

6 the Gaussian noise. Note that:

$$\|\mathbf{w}_{f,n}^k\|^2 = \left(\sum_{m=1}^M \left(\frac{1}{d_{n,m}^k}\right)^2\right)^{-1}$$

⁷ Consequently, Equation (7) can be expressed as the

following nonlinear problem in the standard form:

$$\mathbf{B} = \mathbf{H} \circledast \mathbf{Q} + \overline{\sigma}^2 \delta \mathbf{1}_{NF} + \mathbf{E}$$
(10)

9 where

$$\begin{split} \mathbf{B} &= \begin{bmatrix} \mathbf{b}_{1}^{\top}, \, \mathbf{b}_{2}^{\top}, \, \dots, \, \mathbf{b}_{F}^{\top} \end{bmatrix}^{\top} \in \mathbb{R}^{NF}, \\ \overline{\mathbf{H}} &= \begin{bmatrix} \overline{\mathbf{h}}_{1}^{\top}, \, \overline{\mathbf{h}}_{2}^{\top}, \, \dots, \, \overline{\mathbf{h}}_{F}^{\top} \end{bmatrix}^{\top} \in \mathbb{R}^{NF}, \\ \overline{\mathbf{Q}} &= \begin{bmatrix} \overline{\mathbf{q}}_{1}^{\top}, \, \overline{\mathbf{q}}_{2}^{\top}, \, \dots, \, \overline{\mathbf{q}}_{F}^{\top} \end{bmatrix}^{\top} \in \mathbb{R}^{NF}, \\ \delta &= \frac{1}{K} \sum_{k=1}^{K} \left(\sum_{m=1}^{M} \left(\frac{1}{d_{n,m}^{k}} \right)^{2} \right)^{-1}, \\ \mathbf{E} &= \begin{bmatrix} \mathbf{e}_{1}^{\top}, \, \mathbf{e}_{2}^{\top}, \, \dots, \, \mathbf{e}_{F}^{\top} \end{bmatrix}^{\top} \in \mathbb{R}^{NF}, \end{split}$$

and the discrete-time convolution operator \circledast between $\overline{\mathbf{H}}$ and $\overline{\mathbf{Q}}$ is defined as follows:

$$\overline{\mathbf{H}} \circledast \overline{\mathbf{Q}} = \left[(\overline{\mathbf{h}}_1 \ast \overline{\mathbf{q}}_1)^\top, (\overline{\mathbf{h}}_2 \ast \overline{\mathbf{q}}_2)^\top, \dots, (\overline{\mathbf{h}}_F \ast \overline{\mathbf{q}}_F)^\top \right]^\top.$$

¹² 4. Proposed method

13 4.1. Criterion to be minimized

The purpose of this study is to identify $(\overline{\mathbf{H}}, \overline{\mathbf{Q}}, \overline{\sigma}^2)$ from **B** through Equation (10), which leads to an inverse problem. To solve this, we propose the following optimization problem:

Find
$$(\widehat{\mathbf{H}}, \widehat{\mathbf{Q}}, \widehat{\sigma}^2) \in \underset{\mathbf{H} \in \mathbb{R}^{2NF}, \mathbf{Q} \in \mathbb{R}^{2NF}, \sigma^2 \in \mathbb{R}_+}{\operatorname{argmin}} \theta(\mathbf{H}, \mathbf{Q}, \sigma^2) \overset{45}{_{46}}$$
(11)

18 where:

$$\theta(\mathbf{H}, \mathbf{Q}, \sigma^2) = \psi(\mathbf{H}, \mathbf{Q}, \sigma^2) + \rho(\mathbf{H}, \mathbf{Q}, \sigma^2).$$
(12)

The first term of Equation (12) can be split into two
new terms

$$\psi(\mathbf{H}, \mathbf{Q}, \sigma^2) = \phi(\mathbf{H}, \mathbf{Q}, \sigma^2) + \varphi(\mathbf{Q}), \qquad (13)$$

where $\phi : \mathbb{R}^{NF} \times \mathbb{R}^{NF} \times \mathbb{R}_+ \to \mathbb{R}$ is a data fidelity term that is related to the observation model. In this case, we choose the least-squares objective function, *i.e.*,

$$\phi(\mathbf{H}, \mathbf{Q}, \sigma^2) = \frac{1}{2} \left\| \mathbf{H} \circledast \mathbf{Q} + \sigma^2 \delta \mathbf{1}_{NF} - \mathbf{B} \right\|^2.$$
(14)

 φ models a regularization function that accounts for the sparsity of the solution. In the present paper, we propose to use a new regularization function, the smoothed ℓ_1/ℓ_2 ratio, as proposed by [18]; *i.e.*, for every $\mathbf{Q} \in \mathbb{R}^{NF}$, $(\lambda, \alpha, \beta, \eta) \in [0, +\infty]^4$:

$$\varphi(\mathbf{Q}) = \lambda \log\left(\frac{\ell_{1,\alpha}(\mathbf{Q}) + \beta}{\ell_{2,\eta}(\mathbf{Q})}\right)$$
(15)

30 with,

22

23

24

25

26

27

28

29

31

32

33

35

37

38

39

40

41

42

43

48

49

50

$$\ell_{1,\alpha}(\mathbf{Q}) = \sum_{f=1}^{F} \sum_{n=1}^{N} \left(\sqrt{q_f(n)^2 + \alpha^2} - \alpha \right)$$
$$\ell_{2,\eta}(\mathbf{Q}) = \sqrt{\sum_{f=1}^{F} \sum_{n=1}^{N} q_f(n)^2 + \eta^2}.$$

Note that empirically, the SOOT algorithm provides better results if the condition $\beta < \eta^2/\alpha$ is satisfied. The second term of Equation (12), $\rho : \mathbb{R}^{NF} \times \mathbb{R}^{NF} \times \mathbb{R}_+ \to \mathbb{R}$ is a regularization term that is related to some *a-priori* constraints on the solution. In the following, we assume that ρ can be split into three new terms that concern the three quantities to be estimated:

$$\rho(\mathbf{H}, \mathbf{Q}, \sigma^2) = \rho_1(\mathbf{H}) + \rho_2(\mathbf{Q}) + \rho_3(\sigma^2)$$

where ρ_1 , ρ_2 and ρ_3 are (not necessarily smooth) proper, lower semicontinuous, convex functions [49, Ch. 1], that are continuous on their domain, and which introduce the prior knowledge on the kernel blur (system), **H**, the source autospectra, **Q**, and the noise variance, σ^2 . Due to these properties, the problem can be addressed with the block coordinate variable metric forward-backward algorithm [44]. Moreover, in practice, **H**, **Q** and σ^2 have different properties, and this choice allows the *a-priori* information to be taken into account independently from the searched quantities.

4.2. Proposed algorithm

The objective here is to provide a numerical solution to the optimization problem of Equation (12),

which is a nonlinear blind deconvolution with three 47 1 unknowns $(\mathbf{H}, \mathbf{Q}, \sigma^2)$. One class of popular solutions 2 to solve this problem is the alternating minimiza-49 3 tion algorithm, which iteratively performs the three 50 4 steps: (i) updating **H** given **Q** and σ^2 ; (ii) updating 5 **Q** given **H** and σ^2 ; and (iii) updating σ^2 given **H** 6 and \mathbf{Q} [43]. Furthermore, the criterion to minimize, which is formed as the sum of the smooth and non-8 smooth functions, can be addressed with a block 9 alternating forward-backward method [44, 45]. This 10 method combines explicitly the (forward) gradient 56 11 step with respect to the smooth (not necessarily 57 12 convex) functions and the proximal (backward) step 58 13 with respect to the nonsmooth functions. The con-14 vergence of the algorithm can be accelerated using a 15 majorize-minimize approach [46, 44, 47]. In this pa-16 per, we extend the smoothed one-over-two (SOOT) 17 algorithm proposed in [18] by including a step for 18 the noise variance estimation. This algorithm of 19 noise-robust SOOT (NR-SOOT) is proposed, as pre-20 sented in Algorithm 1. As previously mentioned, the 21 block-variable metric forward-backward algorithm 22 combines two steps of the process that requires two 23 optimization principles. We now recall the defini-24 tion of these: The first is related to the choice of a 25 variable metric that relies upon the majorization-26 minimization properties [47]; *i.e.*, 27

Definition 1 Let $\psi : \mathbb{R}^N \to \mathbb{R}$ be a differentiable function. Let $x \in \mathbb{R}^N$. Let us define, for every 28 29 $x' \in \mathbb{R}^N$: 30

$$\varrho(x',x) = \psi(x) + (x-x')^{\top} \nabla \psi(x) + \frac{1}{2} (x-x')^{\top} U(x) (x - x')^{\frac{67}{68}},$$

where $U(x) \in \mathbb{R}^{N \times N}$ is a semidefinite positive ma-31 trix. Then, U(x) satisfies the majoration condi-32 tion for ψ at x if $\varrho(\cdot, x)$ is a quadratic majorant 33 of the function ψ at x; i.e., for every $x' \in \mathbb{R}^N$, 34 $\psi(x') \le \varrho(x', x).$ 35

A function ψ has a μ -Lipschitzian gradient on a 36 convex subset $C \in \mathbb{R}^N$, with $\mu > 0$, if for every 37 $(x, x') \in C^2$, $\|\nabla \psi(x) - \nabla \psi(x')\| \le \mu \|x - x'\|$. Then, 38 for every $\mathbf{x} \in C$, a quadratic majorant of ψ at \mathbf{x} is 39 easily obtained taking $U(\mathbf{x}) = \mu \mathbf{I}_N$, where \mathbf{I}_N is the 40 identity matrix of $\mathbb{R}^{\tilde{N} \times \hat{N}}$. 41

The second optimization principle is the definition 42 of the proximity operator of a proper, lower semi-43 continuous, convex function, relative to the metric 44 induced by a symmetric positive definite matrix, 45 which is defined in [48] as follows: 46

Definition 2 Let $\rho : \mathbb{R}^N \to]-\infty, +\infty]$ be a proper, lower semicontinuous, and convex function, let $U \in \mathbb{R}^{N imes N}$ be a symmetric positive definite matrix, and let $x \in \mathbb{R}^{N}$. The proximity operator of ρ at x relative to the metric induced by U is the unique minimizer of $\rho + \frac{1}{2}(\cdot - x)^{\top}U(\cdot - x)$, and it is denoted by $\operatorname{prox}_{U,\rho}(x)$. If U is equal to I_N , then $\operatorname{prox}_{\rho} := \operatorname{prox}_{I_N,\rho}$ is the proximity operator originally defined in [50].

The convergence property of the NR-SOOT algorithm can be derived from the general results established in [44]:

Proposition 1 Let $(\mathbf{Q}^l)_{l\in\mathbb{N}}, (\mathbf{H}^l)_{l\in\mathbb{N}}$ and $(\sigma^{2,l})_{l\in\mathbb{N}}$ be sequences generated by Algorithm 1. Assume that:

1. There exists $(\underline{\nu}, \overline{\nu}) \in]0, +\infty[^2 \text{ such that, for all }$ $l \in \mathbb{N}$.

$$(\forall i \in \{0, \dots, I_l - 1\})$$

$$\underline{\nu} \mathbf{I}_{NF} \preceq G_1(\mathbf{H}^{l,i}, \mathbf{Q}^l, \sigma^{2,l}) \preceq \overline{\nu} \mathbf{I}_{NF},$$

$$(\forall j \in \{0, \dots, J_l - 1\})$$

$$\underline{\nu} \mathbf{I}_{NF} \preceq G_2(\mathbf{H}^{l+1}, \mathbf{Q}^{l,j}, \sigma^{2,l}) \preceq \overline{\nu} \mathbf{I}_{NF},$$

$$\underline{\nu} \preceq G_3(\mathbf{H}^{l+1}, \mathbf{Q}^{l+1}, \sigma^{2,l}) \preceq \overline{\nu}.$$

2. Step-sizes $(\gamma_1^{l,i})_{l \in \mathbb{N}, 0 \leq i \leq I_l-1}, (\gamma_2^{l,j})_{l \in \mathbb{N}, 0 \leq j \leq J_l-1}$ and $(\gamma_3^l)_{l \in \mathbb{N}}$ are chosen in the interval $[\underline{\gamma}, 2 - \underline{\gamma}]$ $\overline{\gamma}$] where $\underline{\gamma}$ and $\overline{\gamma}$ are some given positive real constants.

3. ρ is a semi-algebraic function.²

Then, the sequence $(\mathbf{H}^l, \mathbf{Q}^l, \sigma^{2,l})_{l \in \mathbb{N}}$ converges to the critical point $(\widehat{\mathbf{H}}, \widehat{\mathbf{Q}}, \widehat{\sigma}^2)$ of Equation (11). Moreover, $(\theta(\mathbf{H}^l, \mathbf{Q}^l, \sigma^{2,l}))_{l \in \mathbb{N}}$ is a nonincreasing sequence that converges to $\theta(\widehat{\mathbf{H}}, \widehat{\mathbf{Q}}, \widehat{\sigma}^2)$.

In NR-SOOT algorithm 1, ∇_1 , ∇_2 , and ∇_3 are the partial gradients of ψ with respect to the variables \mathbf{H}, \mathbf{Q} , and σ^2 . G_1, G_2 , and G_3 are the semidefinite positive matrix used to build the majorizing approximations of ψ with respect to **H**, **Q**, and σ^2 , and their expressions are given by the following proposition, as established in [18]:

18

51

52

53

54

55

60

62

63

66

69

70

71

72

²Semi-algebraicity is a property satisfied by a wide class of functions, which means that their graph is a finite union of sets defined by a finite number of polynomial inequalities.

Algorithm 1 The NR-SOOT algorithm.

For every $l \in \mathbb{N}$, let $I_l \in \mathbb{N}^*$, $J_l \in \mathbb{N}^*$. Let $(\gamma_1^{l,i})_{0 \leq i \leq I_l-1}, (\gamma_2^{l,j})_{0 \leq j \leq J_l-1}$, and γ_3^l be positive sequences. Initialize with $\mathbf{H}^0 \in \operatorname{dom}(\rho_1)$, $\mathbf{Q}^0 \in \operatorname{dom}(\rho_2)$, and $\sigma^{2,0} \in \operatorname{dom}(\rho_3)$. **Iterations:**

For l = 0, 1 $\mathbf{Q}^{l,0} = \mathbf{Q}^l, \ \mathbf{H}^{l,0} = \mathbf{H}^l,$ For $i = 0, \dots, I_l - 1$ $| \widetilde{\mathbf{H}}^{l,i+1} = \mathbf{H}^{l,i} - \gamma_1^{l,i} G_1(\mathbf{H}^{l,i}, \mathbf{Q}^l, \sigma^{2,l})^{-1} \nabla_1 \psi(\mathbf{H}^{l,i}, \mathbf{Q}^l, \sigma^{2,l})$ $\mathbf{H}^{l,i+1} = \operatorname{prox}_{(\gamma_1^{l,i})^{-1}G_1(\mathbf{H}^{l,i},\mathbf{Q}^l,\sigma^{2,l}),\rho_1}(\widetilde{\mathbf{H}}^{l,i+1})$ $\mathbf{H}^{l+1} = \mathbf{H}^{l,I_l}$ For $j = 0, ..., J_l - 1$ $\begin{aligned} \mathbf{\tilde{Q}}^{l,j+1} &= \mathbf{Q}^{l,j} - \gamma_2^{l,j} G_2(\mathbf{H}^{l+1}, \mathbf{Q}^{l,j}, \sigma^{2,l})^{-1} \nabla_2 \psi(\mathbf{H}^{l+1}, \mathbf{Q}^{l,j}, \sigma^{2,l}) \\ \mathbf{Q}^{l,j+1} &= \operatorname{prox}_{(\gamma_2^{l,j})^{-1} G_2(\mathbf{H}^{l+1}, \mathbf{Q}^{l,j}, \sigma^{2,l}), \rho_2}(\widetilde{\mathbf{Q}}^{l,j+1}) \\ \mathbf{\tilde{Q}}^{l+1} &= \mathbf{Q}^{l,J_l} \\ \widetilde{\sigma}^{2,l} &= \sigma^{2,l} - \gamma_3^l G_3(\mathbf{H}^{l+1}, \mathbf{Q}^{l+1}, \sigma^{2,l})^{-1} \nabla_3 \psi(\mathbf{H}^{l+1}, \mathbf{Q}^{l+1}, \sigma^{2,l}) \\ \sigma^{2,l+1} &= \operatorname{prox}_{(\gamma_3^l)^{-1} G_3(\mathbf{H}^{l+1}, \mathbf{Q}^{l+1}, \sigma^{2,l}), \rho_3}(\widetilde{\sigma}^{2,l}) \end{aligned}$

Proposition 2 For every $(\mathbf{H}, \mathbf{Q}, \sigma^2) \in \mathbb{R}^{NF} \times \mathbb{R}^{NF}$ 1 $\mathbb{R}^{NF} \times \mathbb{R}_+, let:$ 2

$$G_{1}(\mathbf{H}, \mathbf{Q}, \sigma^{2}) = \mu_{1}(\mathbf{Q}, \sigma^{2}) \mathbf{I}_{NF},$$

$$G_{2}(\mathbf{H}, \mathbf{Q}, \sigma^{2}) = \left(\mu_{2}(\mathbf{H}, \sigma^{2}) + \frac{9\lambda}{8\eta^{2}}\right) \mathbf{I}_{NF}$$

$$+ \frac{\lambda}{\ell_{1,\alpha}(\mathbf{Q}) + \beta} G_{\ell_{1,\alpha}}(\mathbf{Q})^{23}_{24}$$

$$G_{3}(\mathbf{H}, \mathbf{Q}, \sigma^{2}) = \mu_{3}(\mathbf{H}, \mathbf{Q}),$$

$$G_{4}(\mathbf{H}, \mathbf{Q}, \sigma^{2}) = \mu_{3}(\mathbf{H}, \mathbf{Q}),$$

where: 3

$$G_{\ell_{1,\alpha}}(\mathbf{Q}) = \text{Diag}\left(\left((q_f(n)^2 + \alpha^2)^{-1/2}\right)_{\substack{1 \le f \le F, \ 1 \le n \le N^3\\(16)}}\right)_{\substack{n \le f \le F, \ 1 \le n \le N^3\\(16)}}$$

and $\mu_1(\mathbf{Q}, \sigma^2)$, $\mu_2(\mathbf{H}, \sigma^2)$, and $\mu_3(\mathbf{H}, \mathbf{Q})$ are theLipschitz constants for $\nabla_1 \phi(\cdot, \mathbf{Q}, \sigma^2),$ 5 $\nabla_2 \phi(\mathbf{H}, \cdot, \sigma^2)$, and $\nabla_3 \phi(\mathbf{H}, \mathbf{Q}, \cdot)$, respectively.³ 6 $G_1(\mathbf{H}, \mathbf{Q}, \sigma^2),$ $G_2(\mathbf{H}, \mathbf{Q}, \sigma^2)),$ Then, and 7 $G_3(\mathbf{H}, \mathbf{Q}, \sigma^2)$ satisfy the majoration condition 8 for $\psi(\cdot, \mathbf{Q}, \sigma^2)$ at $\mathbf{H}, \psi(\mathbf{H}, \cdot, \sigma^2)$ at \mathbf{Q} , and 9 $\psi(\mathbf{H}, \mathbf{Q}, \cdot)$ at σ^2 , respectively. 10

To conclude, we have proposed a blind deconvo-11 lution method to apply to the BF-MS that imposes 12 sparsity on the noise acoustic-source locations. This 13 method is validated in the next section, and com-14 pared to the classical methods of DAMAS-MS and 15 SDM, used in acoustics for moving-source deconvo-16 lution. 17

5. Results

We consider synthetic and real data for the method validation. The synthetic data allow the use of quantitative indicators, whereas real data only provide subjective results. For both cases, we perform comparative evaluation with the standard algorithms DAMAS-MS and SDM. In practice, the kernel blur related to the array transfer function has finite energy, and thus ρ_1 can be chosen as an indicator function of set C = $\left\{ \mathbf{H} \in [h_{\min}, h_{\max}]^{NF} \mid \|\mathbf{H}\| \le \kappa \right\}$ (equal to 0 if $\mathbf{H} \in$ C, and $+\infty$ otherwise), where $\kappa > 0$, and h_{\min} , and $h_{\rm max}$ are the minimum and maximum values of $\overline{\mathbf{H}}$, respectively. In the real data case, we choose $h_{\min} = 0$ and $h_{\max} = 1$. As mentioned before, the



Figure 2: Simulated configuration of a pass-by experiment. Black, source S_1 ; green, source S_2 ; red, calculation grid; blue arrow, global movement of the sources.

³These Lipschitz constants are straightforward to derive since ϕ is a quadratic cost.

autospectra of sources \mathbf{Q} is sparse; moreover, it 1 is limited in amplitude. Then, one natural choice 2 for ρ_2 is the indicator function of the hypercube 3 $[q_{\min}, q_{\max}]^{NF}$, where q_{\min} (resp. q_{\max}) is the lower 4 (resp., upper) boundary of $\overline{\mathbf{Q}}$. In practice, we choose 5 q_{\min} as 0, which leads to a nonnegative constraint 6 on the source power variables, and q_{max} is the max-7 imum value of **B**. Finally, the function ρ_3 related 8 to the constraint on the noise variance is equal to 9 the indicator function of the interval $[\sigma_{\min}^2, \sigma_{\max}^2]$, where $\sigma_{\min}^2 = 0$ and $\sigma_{\max}^2 = 1$. Note that the prox-10 11 imity operators can be easily explicitly expressed 12 (see Appendix). 13

The NR-SOOT algorithm with the penalty 14 smoothed ℓ_1/ℓ_2 function and the classical DAMAS-15 MS and SDM algorithms are applied to the BF-MS 16 result. For every $l \in \mathbb{N}$, the number of inner-loops 17 are fixed as $I_l = 1$ and $J_l = 100$. The NR-SOOT 18 algorithm is launched on 5000 iterations, and it 19 can stop earlier at iteration l if $\|\mathbf{Q}^l - \mathbf{Q}^{l-1}\| \leq$ 20 $\sqrt{NF} \times 10^{-6}$. 21

22 5.1. Synthetic data

The simulated configuration is presented in Fig-23 ure 2. Here, we consider two sources: a random 24 broadband source located at $S_1 = (-4m, 0m, 0m)$ 25 (Fig. 2, black) and a sum of 3 sine functions at fre-26 quencies 1200 Hz, 1400 Hz, and 1800 Hz located at 27 $S_2 = (1 m, 0 m, 0 m)$ (Fig. 2, green), in the coordi-28 nate system where the origin is the center of the 29 moving calculation grid all the time. The sources are 30 moving jointly, and they follow a linear trajectory of 31 length 20 m at constant speed v = 2 m/s. A linear 32 antenna of 21 hydrophones that are equally spaced 33 (with an inter-sensor distance of 0.5 m) records the 34 propagated acoustic signals over D = 10 s. Zero-35 mean white Gaussian noise is added to the recorded 36 signals. To perform BF-MS, the moving calculation 37 grid $X_n(t), \forall n \in \{1, \ldots, N\}$ has a length of 20 m 38 and contains N = 101 points. 39

Concerning the initialization of the methods; \mathbf{Q}^0 40 is the BF-MS output **B**. The initialization of the 41 blur \mathbf{H}^0 for the SOOT and NR-SOOT algorithms is 42 a centered Gaussian filter, such that $\mathbf{H}^0 \in C$. The 43 regularization parameters of SDM, and $(\lambda, \alpha, \beta, \eta) \in$ 44 $]0, +\infty[^4$ (depending on the SOOT or NR-SOOT 45 algorithms) are empirically adjusted, although it 46 can be noted that the method is not too sensitive 47 48 to their initialization.

Figure 3 summarizes the quantitative results in terms of reconstruction error $\overline{\mathbf{Q}}$. The relative error is defined with the ℓ_2 -norm (Fig. 3, top) and



Figure 3: Comparison of the results for input data without noise and for three different signal-to-noise ratios (SNR) $\in \{-10, -5, 0\}$ dB.

 ℓ_1 -norm (Fig. 3, bottom) between the real $\overline{\mathbf{Q}}$ and the estimated $\widehat{\mathbf{Q}}$. It demonstrates that the method can reconstruct accurately in terms of amplitude (observing ℓ_2 -norm) and in terms of sparse source positions (observing ℓ_1 -norm). From Figure 3, we observe that SDM performs better in terms of source localization sparsity than DAMAS-MS for the case considered (the ℓ_1 -norm values of the residual error by SDM are always smaller than those by DAMAS-MS). However, the performance of SDM decreases significantly when the SNR decreases. The NR-SOOT method with $\ell_{1,\alpha}$ as the penalty function, shown to compare $\ell_{1,\alpha}$ with ℓ_1/ℓ_2 approach, gives satisfactory results in terms of sparsity of the source localization, but not in terms of amplitude reconstruction. This result confirms the conclusion of [41], which shows that $\ell_{1,\alpha}$ should not be used in this case. The original SOOT provides very satisfying results compared to DAMAS-MS and SDM. Its performances for cases of high SNR are similar to the proposed method NR-SOOT with smoothed ℓ_1/ℓ_2 . Nevertheless, for the cases of low SNR, the NR-SOOT algorithm with smoothed ℓ_1/ℓ_2 is the only one that provides a satisfactory source localization estimation. To summarize, in all of these cases, the NR-SOOT algorithm with smoothed ℓ_1/ℓ_2 as the penalty function has the smallest error for the source localization estimation in terms of sparsity and amplitude. In the following, for the NR-SOOT method, we only perform it with the smoothed ℓ_1/ℓ_2 penalty function (and call it NR-SOOT).

After this quantitative study, it is necessary to investigate the performance of these methods qualitatively, directly on the localization maps for input

52

53

55

56

57

58

59

60

61

62

63

65

66

67

69

70

71

72

73

74

75

76

77

78

79

80

81

82

83

data without noise and for a SNR of -10 dB. Figure 4 1 and Figure 5 show the results for the DAMAS-MS, 2 SDM, original SOOT, and NR-SOOT algorithms 3 at frequencies of 1400 Hz and 770 Hz, respectively. 4 In these Figures, the green lines (a_1,b_1) represent 5 the theoretical sources to estimate (in terms of po-6 sition and amplitude), the magenta lines represent the BF-MS results, which are the starting points 8 of the DAMAS-MS, SDM, original SOOT, and NR-9 SOOT algorithms. In Figure 4 and Figure 5, the 10 results obtained by DAMAS-MS are in greenish-11 blue, those of SDM are in black (a_2,b_2) , those of 12 original SOOT are in red, and those of NR-SOOT 13 are in blue (a_3, b_3) . 14

At the frequency of 1400 Hz (Fig. 4), for which 15 both sources exist, for the case without noise on 16 the recorded data (Fig. 4a) both the original SOOT 17 and the NR-SOOT algorithms detect the source 18 positions accurately. DAMAS-MS gives some false 19 alarms at x = -3m and x = 2.5m. These false 20 sources have small amplitudes, but they are a real 21 problem because the number of sources is gener-22 ally unknown. SDM locates two sources, but the 23 amplitude estimation is not satisfactory and these 24 sources are spread in the space. For the case of a 25 SNR of -10 dB, DAMAS-MS does not succeed at 26 all, and it shows several false alarms with significant 27 amplitudes. With the SDM method, there is one 28 false alarm around x = -1 m, and the amplitudes 29 are not correct. The original SOOT algorithm gives 30 good results, although there are two false alarms 31 around x = -9m and x = 9m. In contrast, the 32 NR-SOOT algorithm gives perfect results in terms 33 of localization and source amplitude estimation. 34

We now turn our attention to the case at the 35 low frequency 770 Hz (Fig. 5), for which only the 36 source S_1 exists. In this case, DAMAS-MS and SDM 37 give unsatisfactory results, with a spatially extended 38 source and false alarms even in the noise-free case for 39 DAMAS-MS. The original SOOT gives satisfactory 40 results without noise (Fig. $5a_3$), although when the 41 SNR decreases, the original SOOT algorithm creates 42 false alarms (Fig. $5b_3$), while the NR-SOOT algo-43 rithm shows excellent results in terms of position and 44 amplitude (Fig. $5b_3$). The NR-SOOT algorithm is 45 robust against noise. In the following, for the sake of 46 simplicity, and as it always provides the best results, 47 we only consider the NR-SOOT algorithm for blind 48 deconvolution of the two-dimensional illustrations. 49

The two-dimensional localization maps are shown in Figure 7 (without noise) and Figure 8 (SNR of -10 dB), with each Figure showing the initial BF-MS



Figure 4: Comparison of the results at the frequency of 1400 Hz, without noise (a), and with SNR of -10 dB (b).

and the results obtained by DAMAS-MS, SDM, original SOOT, and NR-SOOT. For the case without noise of Figure 7, all of the methods improve the BF-MS output, localize the two sources, and allow identification as one broadboand source and a sumof-sine source. However, the results obtained using DAMAS-MS and SDM are not as good as those using the original SOOT or NR-SOOT algorithms, because the source localizations are spread over several x positions. Moreover, by studying the different zones indicated in the red ellipses in Figure 7 and Figure 8, which are related to the autospectrum of the sine source at the three frequencies of 1200 Hz, 1400 Hz, and 1800 Hz, some other conclusions can be drawn. The results obtained using the DAMAS-MS method are not performing well, as some noise

53

54

55

56

57

58

59

60

61

62

63

64

65

66



Figure 5: Comparison of the results at the frequency of 770 Hz, without noise (a), and with SNR of -10 dB (b).

appears, as indicated by the red arrows in Figure 7b. 1 For the case of a SNR of -10 dB (Fig. 8), DAMAS-2 MS, SDM do not identify the sources and give many 3 false alarms. The original SOOT manages to es-4 timate a point source at the true source positions 5 but also gives many false alarms. In contrast, the 6 NR-SOOT algorithm still gives good results and provides the best performance compared to the three 8 other methods. 9

Figure 6 shows the computational time (in minutes) for the different methods and for different noise levels. The computational time corresponds to the time required to satisfy the stopping criterion, i.e. $\|\mathbf{Q}^{l} - \mathbf{Q}^{l-1}\| \leq \sqrt{NF} \times 10^{-6}$, with the simulations performed on the same CPU. The SDM computational cost is four-fold the SOOT and NR-SOOT



Figure 6: Computational times for input data without noise and for three different signal-to-noise ratios, SNR $\in \{-10, -5, 0\}$ dB.

computational costs. As SDM is an approach that is similar to the forward-backward method, these results confirm the performance of the proposed majorant. In all cases, DAMAS-MS is the fastest algorithm, but the computational costs for DAMAS-MS, SOOT and NR-SOOT are of the same order (from 1-2 min).

To conclude, the NR-SOOT algorithms have better performances than the DAMAS-MS, SDM methods in terms of localization, as the source S_2 is spread over several x positions by DAMAS-MS and SDM, whereas the SOOT and the proposed NR-SOOT algorithm manage to estimate a point source at the true source position. Nevertheless, in term of robustness against noise, the NR-SOOT method is the only one that provides satisfactory results for low SNRs.

5.2. Real data

17

18

19

20

21

22

23

24

25

26

27

28

29

30

31

32

33

34

35

36

37

38

30

40

41

42

43

44

45

46

47

48

49

50

51

52

We finally compare the proposed NR-SOOT algorithm with the classical methods using real data. The experiment was conducted in January 2015 by DGA naval systems at Lake Castillon, a mountain lake in the French Alps with an average depth of 100 m and a maximum width of 600 m. This consisted of towing a 21-m-long scale model of a surface ship. The ship hull included two shakers, S_1 and S_2 , that generated two point acoustic sources outside the hull: a sum of 3 sine functions at frequencies of 1200 Hz, 1400 Hz, and 1800 Hz, located at x = -5.9m, and a random broadband source located at x = 2.3m. A linear antenna of nine hydrophones that were equally spaced by 0.5 m recorded the propagated acoustic signals over D = 14.15 s for the source speed of v = 2 m/s (Fig. 9). We also consider the same configuration with the source speed of v = 5 m/s over D = 5.3 s. (Fig. 10). The coordinate system of the array was used to describe all of

the geometries, with the origin corresponding to the 51 1

array center. The array was immersed at 10 m in 2

depth and was positioned at 2.50 m from the closest 3

point of approach in the y direction. The source 4

trajectory was calculated using a tachymeter system 5 on the idler pulley. The acquisition time considered 6

for the array processing is sufficient, such that the

ship model passed by entirely above the antenna.

8 In these Figures, the zones indicated in the red el-9

lipses correspond to the estimated autospectrum of 10

the sine source at the three frequencies of 1200 Hz, 11

1400 Hz, and 1800 Hz, and the red arrows show the 12

remaining noise or the false alarms. 13

First, we consider the results in the case of the 14 source speed v = 2 m/s (Fig. 9). The three meth-15

ods improve the BF-MS output and identify the 16

sources. DAMAS-MS and NR-SOOT have better 17

performances than SDM in terms of localization. 18

However, for the result of the DAMAS-MS method, 19 there are some false alarms that are indicated by 20

the red arrows in Figure 9b. 21

Secondly, we consider the case with the source speed 22 v = 5 m/s, for which the signal in the recording is 23 more noisy. In this configuration, one new 'natural' 24 source appears at the wake of the ship (Fig. 9d, 25 bottom left). Three methods identify three sources, 26 whereby the sine source is better localized by the 27 NR-SOOT algorithm than the other methods. Both 28 the DAMAS-MS and SDM methods show many false 29 alarms, which are indicated by the red arrows in 30 Figure 10b, c. In particular, the localization of the 31 'natural' source is only possible with the NR-SOOT 32 algorithm. In conclusion, our results from this ex-33 periment remain true to our hypothesis, as well as 34 our predictions. The results shown in Figure 9 and 35 Figure 10 present the best results with perfect source 36 location and improved robustness against noise for 37 the NR-SOOT algorithm. 38

6. Conclusions 39

This paper proposes a new method, known as NR-40 SOOT, that is an extension of the SOOT algorithm 41 [18], for moving-source localization based on blind 42 deconvolution in underwater acoustic data. As the 43 number of sources is small enough and they do 44 not spread spatially, its autospectrum has a sparse 45 representation, and it is possible to obtain more 46 accurate results for blind deconvolution through a 47 regularization function. The smooth approximation 48 of ℓ_1/ℓ_2 shows very good performances in terms of 49 localization and suppression of false alarms, and 102 50

provides better results than DAMAS-MS and SDM, particularly for low SNRs.

Appendix

52

53

54

55

56

57

58

59

60

61

62

63

64

69

70

71

72

73

74

75

76

77

78

79

80

81

82

83

85

86

87

88

89

90

91

92

93

94

95

96

97

98

qq

100

In this appendix, we give the explicit expressions of the proximity operators involved in the NR-SOOT algorithm. For every $(\mathbf{H}, \mathbf{Q}, \sigma^2) \in \mathbb{R}^{NF} \times \mathbb{R}^{NF} \times \mathbb{R}_+$ and $\gamma \in [0, +\infty)$, let G_1, G_2 , and G_3 be the majorant matrix of ψ at **H**, at **Q**, and at σ^2 , respectively, that are given by Proposition 2. Then

1. $\operatorname{prox}_{(\gamma_1)^{-1}G_1,\iota_C} = \Pi_C$,

2.
$$\operatorname{prox}_{(\gamma_2)^{-1}G_2,\iota_{[q_{\min},q_{\max}]^{NF}}} = \prod_{[q_{\min},q_{\max}]^{NF}},$$

3. $\operatorname{prox}_{(\gamma_3)^{-1}G_3,\iota_{[\sigma_{\min}^2,\sigma_{\max}^2]}} = \prod_{[\sigma_{\min}^2,\sigma_{\max}^2]},$

which can be easily computed.

Acknowledgements

This work was financially supported by the Ministère du Redressement Productif (Direction Générale de la Compétitivité, de l'Industrie et des Services) and by the DGA-MRIS, grant RAPID ARMADA Nº122906030.

References

- [1] S. Haykin, Ed., Blind Deconvolution, Prentice Hall, 1994.
- [2]A. K. Nandi, D. Mampel, and B. Roscher, "Blind deconvolution of ultrasonic signals in nondestructive testing applications," IEEE Trans. Signal Process., vol. 45, no. 5, pp. 1382–1390, 1997.
- D. Kundur and D. Hatzinakos, "Blind image deconvo-[3] lution," IEEE Signal Process. Mag., vol. 13, no. 3, pp. 43-64, May 1996.
- M. Kato, I. Yamada, and K. Sakaniwa, "A set-theoretic [4] blind image deconvolution based on hybrid steepest descent method," IEICE Trans. Fund. Electron. Comm. Comput. Sci., vol. E82-A, no. 8, pp. 1443-1449, Aug. 1999.
- A. Ahmed, B. Recht, and J. Romberg, "Blind deconvo-[5]lution using convex programming," *IEEE Trans. Inf. Theory*, vol. 60, no. 3, pp. 1711–1732, Mar. 2014.
- P. Campisi and K. Egiazarian, Eds., Blind Image Decon-[6]volution: Theory and Applications, CRC Press, 2007.
- M. Zibulevsky and B. A. Pearlmutter, "Blind source [7]separation by sparse decomposition in a signal dictionary," Neural Comput., vol. 13, no. 4, pp. 863-882, Apr. 2001
- N. Martins, S. Jesus, C. Gervaise, and A. Quinquis, [8] "A time frequency approach to blind deconvolution in multipath underwater channels", IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP 2002), Orlando, FL, 2002.
- K.-G. Sabra, H.-C. Song, and D.-R. Dowling, "Ray-[9] based blind deconvolution in ocean sound channels" The Journal of the Acoustical Society of America, 127(2), EL42-EL47, 2004.

[10] S.-H. Abadi, H.-C. Song, and D.-R. Dowling, "Broad-67 band sparse-array blind deconvolution using frequency-68 difference beamforming", The Journal of the Acoustical 69 Society of America, 132(5), 3018-3029, 2012. 70

1

2

3

4

5

6

7

8

10

11

12

15

22

28

29

31

32

38

39

46

48

49

50

51

55

56

57

59

61

- [11] A. Pereira, "Acoustic imaging in enclosed spaces", Phd 71 thesis, Laboratoire de Vibro-acoustique, INSA Lyon, 2013
- [12] T. Suzuki, " L_1 generalized inverse beam-forming algorithm resolving coherent/incoherent, distributed and multipole sources", Journal of Sound and Vibration, vol. 330, no. 24, pp. 5835–5851, November 21, 2011.
- N. Chu, J. Picheral, A. Mohammad-Djafari, and N. [13]13 Gac, "A robust super-resolution approach with sparsity constraint in acoustic imaging", Applied Acoustics, vol. 14 76, pp. 197-208, Feb. 2014.
- 16 [14]A. Benichoux, E. Vincent, and R. Gribonval, "A fundamental pitfall in blind deconvolution with sparse and 17 shift-invariant priors," in Proc. Int. Conf. on Acoustics, 18 Speech, and Signal Process., Vancouver, Canada, May, 19 26-31, 2013. 20
- P. Comon, "Contrasts for multichannel blind deconvo-[15]21 lution," Signal Process. Lett., vol. 3, no. 7, pp. 209-211, Jul. 1996. 23
- [16] É. Moreau and J.-C. Pesquet, "Generalized contrasts 24 for multichannel blind deconvolution of linear systems," 25 Signal Process. Lett., vol. 4, no. 6, pp. 182–183, Jun. 26 1997.27
- [17] E. Esser, R. Wang, T. T.Y. Lin and F. J. Herrmann "Resolving scaling ambiguities with the ℓ_1/ℓ_2 norm in a blind deconvolution problem with feedback," in IEEE 30 workshop on Computational Advances in Multi-Sensor Adaptive Processing, Cancun, Mexico, Dec., 2015.
- A. Repetti, M.-Q. Pham, L. Duval, E. Chouzenoux, 99 and J.-C. Pesquet, "Euclid in a Taxicab: Sparse Blind 100 [18]33 34 Deconvolution with Smoothed ℓ_1/ℓ_2 Regularization", ¹⁰¹ 35 IEEE Signal Processing Letters, vol. 62, no. 16, pp. 102 36 539-543, August 15, 2014. 37 103
 - [19] R. J. Urick, "Principles of Underwater Sound", 3rd 104 edition, Mc Graw-Hill Book Company, 1983. 105
- V. Fleury, J. Bulté, "Extension of deconvolution al- 106 [20]40 41 gorithms for the mapping of moving acoustic sources", 107 JASA., vol. 129, no. 3, pp. 1417-1428, March 2011. 42 108
- [21] E. Vertatschitsch and S. Haykin, "Nonredundant ar- 109 43 rays", Proc. IEEE, vol. 74, no. 1, pp. 217–218, January, 110 44 45 1986.111
- [22] A. T. Moffet, "Minimum-redundancy linear arrays", 112 IEEE Trans. Antennas Propagat., vol. 16, no. 2, pp. 113 47 172-175, Mar., 1968.
 - [23] R. P. Smith, "Constant beamwidth receiving arrays for 115 broad band sonar systems", Acta Acustica united with 116 Acustica, vol. 23, no. 1, pp. 21–26, January, 1970. 117
- [24] B.D. Van Veen and K.M. Buckley, "Beamforming: a 118 versatile approach to spatial filtering", *IEEE assp mag-* 119 52 53 azine, vol. 5, no. 2, pp. 4–24, April, 1988. 54 120
 - [25]J. A. Högbom, "Aperture Synthesis with a Non-Regular 121 Distribution of Interferometer Baselines", Astron. As- 122 trophys. Suppl., no. 15, pp. 417–426, 1974. 123
- [26] J. Tsao and B. D. Steinberg, "Reduction of sidelobe 124 58 and speckle artifacts in microwave imaging: the CLEAN 125 technique", IEEE Trans. Signal Process, vol. 36, no. 4, 126 60 pp. 543-556, Apr., 1988.
- P. Sijtsma, "CLEAN based on spatial source coherence", [27]62 128 International Journal of Aeroacoustics, vol. 6, no. 4, pp. 129 63 357-374. 2007. 64
- S. G. Mallat and Z. Zhang, "Matching Pursuits with 131 [28]65 Time-Frequency Dictionaries", IEEE Trans. Signal Pro- 132 66

cess, vol. 41, no. 12, pp. 3397–3415, December 1993. T. F. Brooks, and W. M. Humphreys, "Deconvolution

- [29]approach for the mapping of acoustic sources (DAMAS) determined from phased microphone arrays", Sound and Vibration, vol. 294, pp. 856-879, 2006.
- [30]T. Yardibi, J. Li, P. Stoica, L. Cattafesta, "Sparsity constrained deconvolution approaches for acoustic source mapping", JASA, vol. 123, no. 5, pp. 2631-2642, June 2008.
- [31]S. Brühl, and A. Röder, "Acoustic noise source modelling based on microphone array measurements", Journal of Sound and Vibration, vol. 231, no. 3, pp. 611-617, March, 2000.
- [32] A. L. Swindlehurst and T. Kailath, "A performance analysis of subspace-based methods in the presence of model errors. I. The MUSIC algorithm," IEEE Trans. Signal Process., vol. 40, no. 7, pp. 1758-1774, Jul. 1992.
- [33] V. Fleury, J. Bulté, and R. Davy, "Determination of acoustic directivity from microphone array measurements using correlated monopoles", 29th AIAA, 5-7 May 2008, Vancouver, British Columbia Canada.
- [34]K. Sun, Y. Liu, H. Meng, and X. Wang, "Adaptive Sparse Representation for Source Localization with Gain/Phase Errors", Sensors, vol. 11, no. 5, pp. 4780-4793, May 02, 2011.
- [35]S. Cotter, B. Rao, K. Engan, and K. Kreutz, "Sparse solutions to linear inverse problems with multiple measurement vectors", IEEE Trans. Signal Process, vol. 53, no. 7, pp. 2477-2488, Jully 2005.
- D. M. Malioutov, M. Cetin, and A. S. Willsky, "A sparse [36]signal reconstruction perspective for source localization with sensor arrays", IEEE Trans. Signal Process, vol. 53, no. 8, pp. 3010-3022, Aug., 2005.
- [37] Y. Doisy, L. Deruaz and R. Been, "Interference suppression of subarray adaptive beamforming in presence of sensor dispersions," IEEE Trans. Signal Process., vol. 58, no. 8, pp. 4195-4212, Apr. 2010.
- [38]J. Chen and X. Huo, "Sparse Representations for Multiple Measurement Vectors (MMV) in an Over-Complete Dictionary", ICASSP 2005, Philadelphia, PA, USA., pp. 257–260, March 2005.
- [39] B. Oudompheng, B. Nicolas and L. Lamotte, "Passive synthetic aperture array to improve noise mapping of a moving ship", OCEANS 2015, Genova, Italy, pp. 1-6, May 18-21, 2015.
- "Localisation et contribution de [40]B. Oudompheng, sources acoustiques de navire au passage par traitement d'antenne réduite", Phd thesis, GIPSA-Lab, 2015.
- M.-Q. Pham, B. Oudompheng, B. Nicolas and J.-I. Mars, [41]Sparse deconvolution for moving-source localization", IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP 2016), Shanghai, China, March 20-25, 2016, pp. 355-359.
- [42] S. H. Abadi, D. Rouseff, D. R Dowling, "Blind deconvolution for robust signal estimation and approximate source localization", Journal of the Acoustical Society of America, vol. 131, no. 4, pp. 2599–2610, Apr., 2012.
- [43]J. Bolte, P. L. Combettes, and J.-C. Pesquet, "Alternating proximal algorithm for blind image recovery," in Proc. Int. Conf. Image Process., Hong-Kong, China, Sep. 26-29, 2010, pp. 1673-1676.
- E. Chouzenoux, J.-C. Pesquet, and A. Repetti, "A block [44]coordinate variable metric forward-backward algorithm,"
- J. Global Optimization, Springer Verlag, pp.1-29, 2016. J. Bolte, S. Sabach, and M. Teboulle, "Proximal al-J. Bolte, S. Sabach, and M. Teboulle, [45]ternating linearized minimization fon nonconvex and

72

73

74

75

76

77

78

79

80

81

82

83

84

85

87

88

89

90

91

92

93

94

95

96

- nonsmooth problems," Math. Progr. (Ser. A), Jul. 2013.
 [46] E. Chouzenoux, J.-C. Pesquet, and A. Repetti, "Variable metric forward-backward algorithm for minimizing the sum of a differentiable function and a convex function", J. Optim. Theory Appl., vol. 162, no. 1, pp. 107–132, Jul. 2014.
- 7 [47] M. W. Jacobson and J. A. Fessler, "An expanded theoretical treatment of iteration-dependent Majorize-Minimize algorithms," *IEEE Trans. Image Process.*, vol. 16, no.
 10 10, pp. 2411–2422, Oct. 2007.
 - [48] P. L. Combettes and B. C. Vũ, "Variable metric quasi-Fejér monotonicity," Nonlinear Anal., vol. 78, pp. 17–31, Feb. 2013.

11

12

13

14

- [49] H. H Bauschke and P. L. Combettes, "Convex analysis and monotone operator theory in hilbert spaces," *Springer Verlag New York*, 2011.
- Springer Verlag New York, 2011.
 [50] P. L. Combettes and J.-C. Pesquet, "Proximal splitting methods in signal processing", in Fixed-point algorithms for inverse problems in science and engineering, H. H.
 Bauschke, R. Burachik, P. L. Combettes, V. Elser, D.
 R. Luke, and H. Wolkowicz, Eds., pp. 185–212. Springer
 Verlag, 2011.



Figure 7: Localization in the frequency-distance domain obtained (in the case without noise). (a) Initial BF-MS. (b) DAMAS-MS. (c) SDM. (d) original SOOT. (e) NR-SOOT (the dynamic ranges shown are 15 dB).



Figure 8: Localization in the frequency-distance domain obtained (with SNR of -10 dB). (a) Initial BF-MS. (b) DAMAS-MS. (c) SDM. (d) original SOOT. (e) NR-SOOT (the dynamic ranges shown are 15 dB).



Figure 9: Localization obtained in the frequency-distance domain for the model ship with two artificial sources, traveling at 2 m/s. (a) Initial BF-MS. (b) DAMAS-MS. (c) SDM. (d)

NR-SOOT (the dynamic ranges shown are 15 dB).



Figure 10: Localization obtained in the frequency-distance domain for the model ship with two artificial sources, traveling at 5 m/s. (a) Initial BF-MS. (b) DAMAS-MS. (c) SDM. (d) NR-SOOT (the dynamic ranges shown are 15 dB).