WIDEBAND ZERO-FORCING MUSIC FOR AEROACOUSTIC SOURCES LOCALIZATION

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ABSTRACT

The application considered in this paper is the localization of the aeroacoustic sources on the body of a car placed in a wind tunnel. This problem addresses a number of nonconventional issues for source localization methods: near-field sources, wideband signals and nonuniform planar antennas. The proposed method is an extension of the Zero-Forcing MUSIC (ZF-MUSIC) algorithm to this case. It is a sequential version of the MUSIC algorithm, where the spectrum of MUSIC is forced to be zero for the previously estimated sources. Simulations are presented using synthetic and real data in order to show the efficiency of the proposed method. It is shown that this technique manifests significant resolution and computational advantages over previous algorithms such as the beamforming, the MUSIC algorithm and the Wideband CLEAN methods, specially in the case of closely spaced sources with different powers.

Index Terms— Acoustic imaging, near-field, wideband, high-resolution, beamforming.

1. INTRODUCTION

In order to improve the comfort of the customer, the manufacturers seek to reduce the acoustic noise during transportation. With the development of quieter engines, the aeroacoustic noise (generated by the wind against the body of the car) becomes an important annoyance inside the vehicule. In addition, the noise level inside does not depend only on the strongest sources but also on the weak ones that are closer to the ear of the passengers. Thus, the localization of the aeroacoustic noise sources and the estimation of their power prove to be essential.

Within the framework of this study, we are interested in an experiment whose principle consists in placing an object in the flow of a wind tunnel, and a microphone array is placed parallel to the flow in order to record the aerodynamic noise on the surface of the object. The goal is to determine the location and the power of the aeroacoustic sources. One of the challenges is to precisely determine at low frequency, close sources with relatively low power compared to the main sources. The reason is that a low power source close to the passengers ears can be considered as noise.

The localization of aeroacoustic sources is subject to a number of issues outside the scope of conventional array processing: near-field sources, wideband signals and nonuniform planar antenna. It is a very active field of research for which methods of aeroacoustic imagery using a planar array of microphones were developed [1] [2] [3] [4]. In order to form high quality acoustic images using a microphone array, the delay and sum beamforming methods [5] were considered. Besides the problem of high sidelobes, another problem arises to the beamforming method in the case of wideband signals: the beamwidth varies according to the frequency. The adaptive methods (i.e. MVDR) present a better resolution and a better ability to suppress the interference between the sources (as opposed to the delay and sum beamforming, which suffers from the problem of sidelobes, specially in the case of nonuniform arrays). However, they suffer from performance degradation when the number of samples is small and the sources are highly correlated [6]. Other methods were proposed to overcome these problems [7] [3]. WideBand (WB) RELAX [8] is a parametric and iterative approach that can be used effectively in the imaging of point sources in order to mitigate the sidelobe effect. Another iterative method is theWB-CLEAN [8] which is non parametric and can be used to locate point or distributed sources. It also aims to reduce the sidelobe effect by deletion of already located sources. Another proposed method [9] consists of combining the robust Capon beamformer [6] with the shading scheme, in order to obtain a constant beamwidth over all frequency bins of interest. In [1], the proposed method is a deconvolution approach for the mapping of acoustic sources.

In this paper, we propose a method based on ZF-MUSIC [10] [11], and we generalize it to the case of wideband signals and nonuniform planar array. The algorithm is a sequential high-resolution MUSIC approach which leaves unchanged the signal subspace but scales appropriately the MUSIC pseudo-spectrum in order to increase the resolution of the sources and their power dynamic. One of the interests of ZF-MUSIC is that it does not perform a deflation of the signal subspace, but it changes directly the MUSIC criterion in order to keep a low computational cost. At each iteration, the location of one source is estimated.

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This paper is organized as follows. Section 2 introduces the signal model used throughout the paper. In section 3, the ZF-MUSIC is introduced and extended to aeroacoustic sources localization. Simulation results are presented in section 4. Section 5 provides our conclusions.

2. PROBLEM FORMULATION

The principle of the experiment is presented in Fig. 1 where the positions of the antennas, the aeroacoustic sources and the direction of the wind are shown. The side edge of the car can be approximated by a plane. Moreover the contribution of the sources that are not in the vertical plane can be neglected because they are mainly masked by the structure of the car. The antenna array is planar and nonuniform. The X axis is oriented in the opposite direction of the wind and its origin is located at the center of the antenna. The Y axis is directed upwards and its origin is on the ground level. We suppose that all the sources are placed in the plane $z = z_0$. The planes of the aeroacoustic sources and the sensors are parallel.

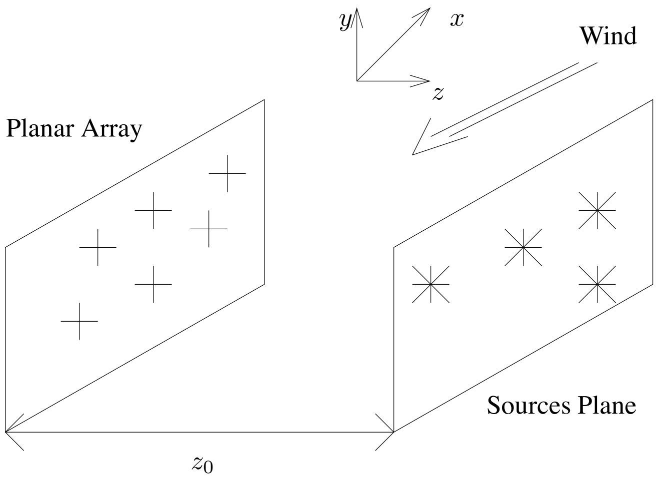

Fig. 1. Principle of the experiment.

Consider an array of M sensors receiving the signal of N near-field wideband sources. Let $s_n(t)$ represent the signal associated to the source n. $\tau_{m,n}$ is the time travel between the source n and the sensor m, $\mathbf{p}_n = [x_n, y_n]$ denotes the coordinates of the n^{th} source in the plane $z = z_0$ and c is the sound speed.

In frequency domain, the model of the measured signal can be written as:

$$\mathbf{r}(\omega) = \mathbf{A}(\mathbf{P}, \omega)\mathbf{s}(\omega) + \boldsymbol{\nu}(\omega), \qquad (1)$$

where $\mathbf{r}(\omega) = [r_1(\omega), \ldots, r_M(\omega)]^\top$, $\mathbf{s}(\omega) = [s_1(\omega), \ldots, s_N(\omega)]^\top$ and $\boldsymbol{\nu}(\omega) = [\nu_1(\omega), \ldots, \nu_M(\omega)]^\top$) are respectively the Fourier transform of the measured signal, the source signal and the measured noise. $\mathbf{A}(\mathbf{P}, \omega)$ is the steering matrix of dimension $M \times N$: $\mathbf{A}(\mathbf{P}, \omega) = [\mathbf{a}(\mathbf{p}_1, \omega), \ldots, \mathbf{a}(\mathbf{p}_N, \omega)]$. $\mathbf{a}(\mathbf{p}_n, \omega)$ is the steering vector of dimension $M \times 1$, commonly used in near-field estimation [8] expressed by:

$$\mathbf{a}(\mathbf{p}_n, \omega) = \left[\frac{1}{\tau_{1,n}.c} e^{-j\omega\tau_{1,n}}, \ldots, \frac{1}{\tau_{M,n}.c} e^{-j\omega\tau_{M,n}} \right]^\top. \quad (2)$$

The matrix $\mathbf{P} = [\mathbf{p}_1, \mathbf{p}_2, \ldots, \mathbf{p}_N]^\top$ contains the 2D coordinates of the incident sources. Using $\mathbf{r}(\omega)$, the purpose is to estimate the positions $\mathbf{p}_n$ of the sources placed in the plane $z = z_0$, and the sources amplitude $\mathbf{s}(\omega)$ in order to find their power.

In order to obtain a robust estimation of the spectral density $\mathbf{R}(\omega) = E\{\mathbf{r}(\omega)\mathbf{r}^H(\omega)\}$, the snapshots are separated into I segments each one containing L snapshots. The Discrete Time Fourier Transform (DTFT) is applied on the segments of length L. The spectral density is then estimated as:

$$\hat{\mathbf{R}}(\omega_l) = \frac{1}{I} \sum_{i=1}^{I} \mathbf{r}_i(\omega_l)\mathbf{r}_i^H(\omega_l). \qquad (3)$$

The number of snapshots L must be chosen sufficiently large in order to guarantee the narrowband hypothesis. Furthermore, the number of segments I must be chosen as $I > M$ to guarantee a good estimation of the spectral density (see chapter 5 of [6]).

3. ZERO-FORCING MUSIC

In this paper, we propose to extend the principle of ZERO-FORCING MUSIC (ZF-MUSIC) [10] [11] to the case of 2D wideband signals. In [10] [11], it is shown that the interest of ZF-MUSIC lies in its capability to increase the resolution of closely spaced sources, specially if they have different powers. ZF-MUSIC is an iterative method which estimates the position of one source at each iteration. To estimate the position of the first source, we calculate the standard narrowband MUSIC criterion for each frequency bin ω_l:

$$S_{\text{MUSIC}} = \frac{1}{\mathbf{a}^H(\mathbf{p}, \omega_l)\hat{\mathbf{E}}_B(\omega_l)\hat{\mathbf{E}}_B^H(\omega_l)\mathbf{a}(\mathbf{p}, \omega_l)}, \qquad (4)$$

where $\hat{\mathbf{E}}_B(\omega_l)$ is an estimate of the basis of the noise subspace. This subspace $\hat{\mathbf{E}}_B(\omega_l)$ is obtained by the $(M - N)$ eigenvectors of $\hat{\mathbf{R}}(\omega_l)$. The position of the first source is estimated by finding the maximum of criterion. To estimate the next source position, the criterion should be forced to zero around the first source. Thus, the MUSIC criterion is modified and a ZF function is added.

To generalize, consider iteration n. Assuming that $(n-1)$ sources have been estimated, the ZF-MUSIC criterion is the MUSIC criterion modified by a ZF function (f_n) in order to force it to zero around the sources already estimated:

$$S_{\text{ZF}}^{(n)}(\mathbf{p}, \omega_l) = \frac{f_n(\mathbf{p}, \omega_l)}{\mathbf{a}^H(\mathbf{p}, \omega_l)\hat{\mathbf{E}}_B(\omega_l)\hat{\mathbf{E}}_B^H(\omega_l)\mathbf{a}(\mathbf{p}, \omega_l)}. \qquad (5)$$

The zero-forcing function f_n is the quadratic function defined by: $f_n(\mathbf{p},\omega_l) = \mathbf{a}^H(\mathbf{p},\omega_l)\mathbf{\Pi}_n^\perp \mathbf{a}(\mathbf{p},\omega_l)$ where $\mathbf{\Pi}_n^\perp = \mathbf{I} - \mathbf{\Pi}_n$ is the orthogonal projector on the space generated by the $(n-1)$ sources and positions already estimated. The matrices $\mathbf{\Pi}_n$ and $\mathbf{A}_n$ are given by:

$$\begin{cases} \mathbf{\Pi}_n = \mathbf{A}_n \left(\mathbf{A}_n^H \mathbf{A}_n\right)^{-1} \mathbf{A}_n^H \\ \mathbf{A}_n = [\mathbf{a}(\hat{\mathbf{p}}_1,\omega_l),\ldots,\mathbf{a}(\hat{\mathbf{p}}_{n-1},\omega_l)] \qquad \text{for } n = 2,\ldots,N \end{cases}$$

where $\hat{\mathbf{p}}_n$ denotes the estimated position of the n^{th} source. It has been shown in [11] that if M is large enough, $f_n(\mathbf{p},\omega_l)$ is null around the $(n-1)$ estimated positions and does not modify the MUSIC criterion elsewhere.

Notice that at each iteration, the only term of S_{ZF} that changes is the ZF-function f_n. This is a great computational advantage over methods based on a deflation of the signal subspace. Also notice that at the first iteration, the ZF-MUSIC criterion is identical to the standard MUSIC one. Fore more details on standard ZF-MUSIC, see [10] and [11].

Since the signal is wideband, $S_{\text{ZF}}(\mathbf{p},\omega_l)$ has to be evaluated for each frequency bin of interest. In order to avoid the maximization of the criterion at each frequency bin, we propose to define a wideband ZF-MUSIC criterion formed by the sum of all narrowband ZF-MUSIC criterion:

$$S_{\text{ZF}}^{(n)} = \sum_{\omega_l \epsilon [\omega_{\min},\omega_{\max}]} S_{\text{ZF}}^{(n)}(\mathbf{p},\omega_l), \qquad (6)$$

where $[\omega_{\min},\omega_{\max}]$ is the band of interest. Another approach would be to estimate the sources at each frequency, but the approach used in this paper is less complex. Besides, since the maximization of the criterion at each frequency bin is avoided, the SNR is increased by averaging the criterion. Nevertheless, the band of interest should not be wide in order to prevent that the lowest frequency degrades the resolution.

The algorithm of the wideband ZF-MUSIC can be summarized as follows:

1. Compute $\hat{\mathbf{E}}_B(\omega_l)$ by eigenvector decomposition of $\hat{\mathbf{R}}(\omega_l)$.
2. Find the first source position by finding the maximum of: $\sum_{\omega_l} S_{\text{MUSIC}}(\mathbf{p},\omega_l)$ where S_{MUSIC} is defined by (4) and $\omega_l \epsilon [\omega_{\min},\omega_{\max}]$.
3. Find the next source position by finding the maximum of: $\sum_{\omega_l} S_{\text{ZF}}^{(n)}(\mathbf{p},\omega_l)$ where $S_{\text{ZF}}^{(n)}$ is defined by (5) and $\omega_l \epsilon [\omega_{\min},\omega_{\max}]$.
4. If $n = N$, stop. Else, $n = n+1$ and go to step 3.

Once the estimation of the positions is achieved, the power of the sources has to be estimated in order to evaluate the contribution of each one to the noise level inside the car. The amplitudes of the sources are estimated by solving the equation (1) using the Least Squares (LS) approach. The solution can be written as:

$$\hat{\mathbf{s}}_i(\omega_l) = \mathbf{A}^\dagger(\hat{\mathbf{P}},\omega_l)\mathbf{r}_i(\omega_l), \qquad (7)$$

where $\mathbf{A}^\dagger = (\mathbf{A}^H\mathbf{A})^{-1}\mathbf{A}^H$ is the pseudoinverse matrix of $\mathbf{A}$, for $l = 1,\ldots,L$ and $i = 1,\ldots,I$.

The power of the source in the band of interest is estimated by:

$$D(\hat{\mathbf{p}}_n) = \sum_{i=1}^{I}\sum_{l=1}^{L} |\hat{s}_{i,n}(\omega_l)|^2. \qquad (8)$$

4. NUMERICAL RESULTS

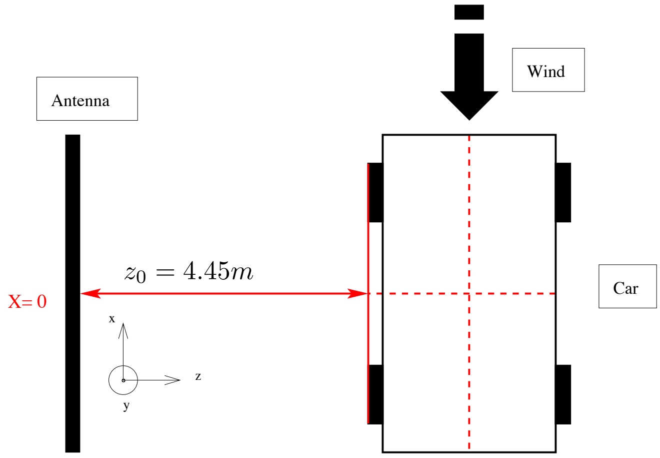

Fig. 2. Principle of the wind tunnel experiment.

In this section, wideband ZF-MUSIC is compared to methods proposed in literature: the delay and sum beamforming (BF), the standard MUSIC modified to be applied to the case of near-field sources and wideband signals and the WB-CLEAN [8]. Two types of data are considered: synthetic and real data.

The application considered is the estimation of aeroacoustic noise sources on the body of the car. The positions of the microphone array and the car are shown in Fig. 2 as well as the direction of the wind. The microphone array used is planar and nonuniform, consisting of $M = 64$ omnidirectional microphones in the plane $z = 0$ (dimension of the array is 4 m x 2 m).

4.1. Synthetic data

In the synthetic simulation data, the sources are not correlated and are generated by low pass filtering the white noise with a cutoff frequency of 10 kHz. The sampling frequency is equal to 25.6 kHz and the acquisition time is equal to 4 seconds. The SNR is 10 dB.

4.1.1. Case of two closely spaced sources with different power

We test the different methods in the case of two closely spaced sources. The sources have the following positions: $\mathbf{p_1} = [-1, 0.5]^\top$ and $\mathbf{p_2} = [-0.9, 0.5]^\top$, with respective powers 0 and -5 dB.

Fig. 3 shows the estimated positions (green circle) using the BF, MUSIC, WB-CLEAN and ZF-MUSIC methods, as well as the contours of the used criteria (in the case of iterative methods, it is the criterion corresponding to the last iteration) and the exact values of the positions (black cross).

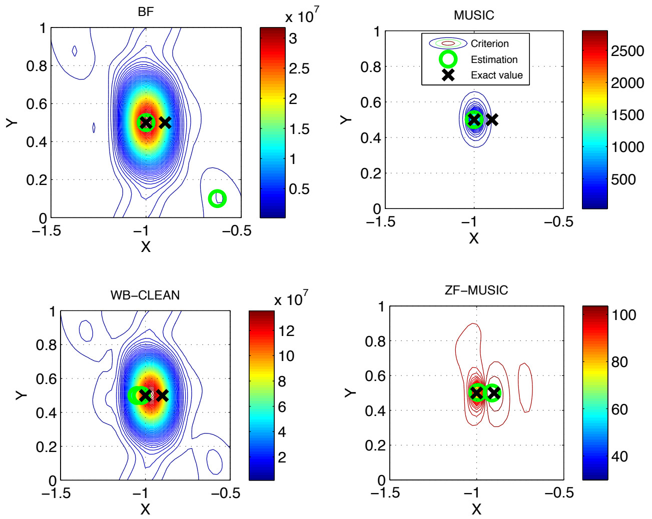

Fig. 3. Exact and estimated positions of two closelyspaced sources with different powers, comparing the criteria of BF, MUSIC, WB-CLEAN and ZF-MUSIC.

This experiment illustrates how ZF-MUSIC is capable of separating two closely spaced sources with different powers. BF and MUSIC estimate only one source, WB-CLEAN estimates two sources, but only one of them is correct.

4.1.2. Case of multiple sources with different powers

Here, ZF-MUSIC is tested and compared to other methods for $N = 20$ sources having different powers. The sources positions are chosen in a sector of 4 m $\times$ 2 m and the acquisition time is 8 seconds. The sources powers are randomly generated in the $[-10, 0]$ dB interval. Fig. 4 represents the estimated positions as well as the exact ones.

ZF-MUSIC estimates accurately all the sources, unlike the other methods. WB-CLEAN was launched on 60 iterations and detected 5 sources only. MUSIC found the position of 15 sources and was not able to separate 5 closely spaced sources. BF was able to find the positions of 11 sources but with low precision. In conclusion, this simulation shows clearly the superiority of ZF-MUSIC over other methods.

Once the sources positions are detected using ZF-MUSIC, the power of the sources can then be estimated (see table 1). These results prove the reliability of the estimation on synthetic data: the goal is achieved by providing accurate position and power estimation of close low power sources selected from a wide power dynamic range.

4.2. Real data

The ZF-MUSIC method was also tested on real data. Data from the wind tunnel experimentations in Renault are used.

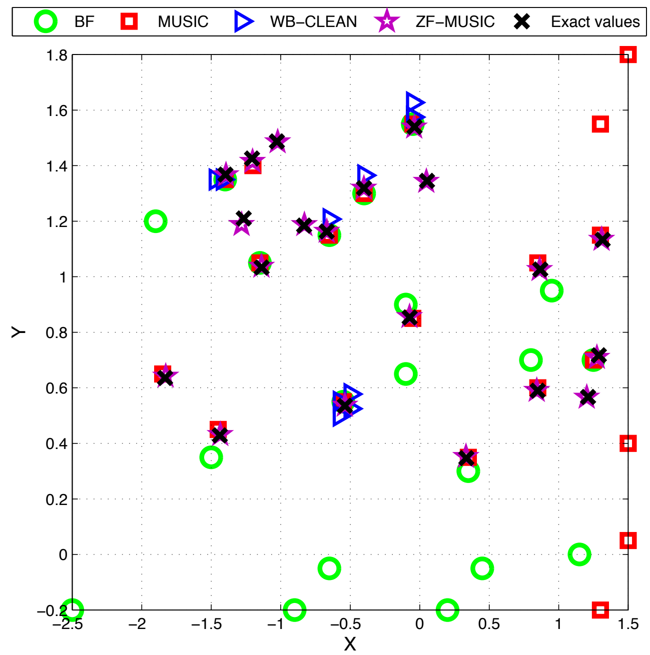

Fig. 4. Comparison of the positions estimated by the BF, MUSIC, WB-CLEAN and ZF-MUSIC methods.

Source Number	1	2	3	4	5	6	7	8	9	10
Exact Power	0.25	0.70	0.68	0.35	0.59	0.69	0.30	0.84	0.55	0.49
Estimation	0.30	0.73	0.70	0.39	0.61	0.70	0.33	0.85	0.057	0.50
Source Number	11	12	13	14	15	16	17	18	19	20
Exact Power	0.36	0.15	0.186	0.16	0.82	0.10	0.12	0.29	0.00	0.22
Estimation	0.39	0.21	0.15	0.24	0.85	0.24	0.15	0.32	0.25	0.27

Table 1. Estimation of the powers of the $N = 20$ sources.

The band of interest is $[2\,\text{kHz}, 3\,\text{kHz}]$. The sampling frequency is equal to 25.6 kHz and the acquisition time is equal to 4 seconds. The data are divided into $I = 204$ intervals each containing $L = 101$ samples. The positions estimated using the methods BF, MUSIC, WB-CLEAN and ZF-MUSIC are shown in Fig. 5.

The BF and MUSIC methods are capable of detecting some noisy sources located on the wheels, the rear-view mirror, the windshield wiper, the antenna and the handle of the doors. But many estimated points are erroneous. On the other hand, WB-CLEAN that was launched on 200 iterations, is only capable of detecting the noise on the wheels but requires long simulation runtimes. ZF-MUSIC is capable of detecting most of the noisy sources with a higher precision, as the ones on the wheels and the rear-view mirror, with less time than the WB-CLEAN.

In conclusion, ZF-MUSIC presents better overall performance than other methods as numerical results showed, specially in the case of closely spaced sources with different powers. It is clearly shown that ZF-MUSIC separates the sources better than MUSIC. It is more precise and it presents a relatively low computational cost. This is an important feature because during the experimentations, the data must be analyzed

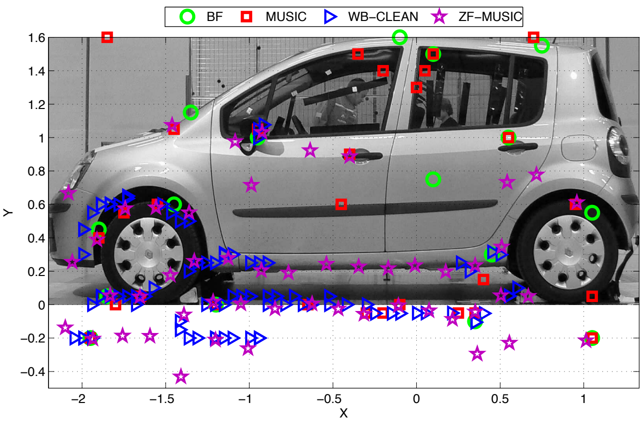

Fig. 5. Comparison of the positions estimated using the BF, MUSIC, WB-CLEAN and ZF-MUSIC criteria.

in semi real time (less than 5 minutes) in order to modify the configurations of the testing.

5. CONCLUSION

In this paper, a new approach is proposed for the aeroacoustic source localization based on a generalization of ZF-MUSIC to the case of wideband signals and nonuniform planar array. The main interest of this method is that it increases the ability to separate sources having different power, one of the main challenges of aeroacoustic source localization in wind tunnel. In addition, ZF-MUSIC has a relatively low computational cost and thus can be used in semi real time experiments. This analysis is supported by numerical results: first, critical scenarios with synthetic data show that wideband ZF-MUSIC is able to separate closely spaced sources having different power and is more precise than other methods. Second, the method can be applied to real data in order to localize the aeroacoustic sources on the body of a car placed in a wind tunnel. In this scenario, the wideband ZF-MUSIC provides accurate results.

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