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Sound source characterization on a high speed train from microphone array measurements

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ABSTRACT

Environmental noise is a major concern for the railway transportation. The characterization of the main noise sources of a train during its pass-by is needed to develop the most relevant noise reduction solutions. Microphone antenna measurement and beamforming signal processing have been used by SNCF to provide source positions, spectra and levels for more than 10 years. It consists in delaying and summing the signals measured by several microphones to estimate the sound radiated at a focus point. The accuracy of the estimation depends strongly on the spatial repartition of the sensors. As the movement of the sources induces a Doppler effect (both frequency and amplitude are modulated), two approaches are considered to suppress it. The first method based on dedopplerization adjusts the focus of the antenna during the pass-by to follow a mesh grid of the moving train. This method enables to provide a short stationary time-signal, emitted by the sources, for spectrum analysis. The other method focuses the antenna straightforward on different heights. The output signal is observed using a time-frequency representation. The potential of the two methods to characterize the sources are illustrated with results from a dedicated measurement campaign involving an adapted microphone array to process the acoustic signals emitted by a TGV high speed train.
Keywords: Beamforming, high speed train, array processing

1. INTRODUCTION

Transport is a major contribution to the urban noise. Prediction methods [1] provide pass-by sound samples of trains to investigate and optimize the noise reduction solutions. To increase the accuracy of the results of the prediction, the characteristics of the sources are deeply investigated (position, power, spectrum).

Microphone array methods have been used by the SNCF since 1995 [2] to provide those characteristics. They are particularly adapted to the railway domain, where the measurement of the acoustic nearfield is impossible due to security restrictions and vehicle dimensions. Simulations could provide aeroacoustic sources, and rolling noise, which is the most important contribution to the total noise emitted by the train, but validation is only accessible in rolling conditions.

This paper deals with the use of beamforming to estimate the noise source characteristics on the TGV, considering the context of high speed motion and large dimensions of the object. In a first part, the time-domain beamforming algorithm is introduced. Then, the specificities of moving sources are presented and also how the beamforming algorithm is adapted to those specific measurement conditions. Two main methods are used: filtering and dedopplerization. As a conclusion, a commercial TGV is characterized and comparisons between the results of the two approaches are performed. Both methods are compared in terms of computation time and accuracy of the results.

The array used in this paper is optimized to process the dedopplerization, its directivity pattern variations are minimized using the genetic optimization defined in [3]. 40 microphones are used to

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cover the [800 6000] Hz frequency band. The array is located at 4 m from the rail.

2. BEAMFORMING ALGORITHM

A plane array, as presented in Figure 1, is composed of N microphones. Let us consider N_i unknown acoustic sources $s_i(t)$, propagating in a homogeneous and isotropic medium. The signal measured by the n^{th} microphone of the array is

$$p_n(t) = \sum_{i=1}^{N_i} 1/(4\pi r_{in}) s_i(t - r_{in}/c), \tag{1}$$

where r_{in} is the distance between s_i and the n^{th} microphone and c the sound celerity in the medium. The source signals are delayed and attenuated following a spherical wave propagation law.

Beamforming consists in time-shifting and summing the outputs of the microphones to estimate the signal at an arbitrary focus point M [4]. The output of the beamforming is then

$$x_M(t) = \sum_{n=1}^{N} (4\pi r_{nM}) p_n(t + r_{nM}/c), \tag{2}$$

with r_{nM} the distance between M and the n^{th} microphone. From Equation 1, it comes

$$x_M(t) = \sum_{n=1}^{N} (4\pi r_{nM}) \sum_{i=1}^{N_i} 1/(4\pi r_{in}) s_i(t - r_{in}/c + r_{nM}/c), \tag{3}$$

if M is chosen such as $r_{nM}=r_{in}$, we can assume that $x_M(t)=s_i(t)$. Beamforming operates a bandpass spatial filter around the focus point. The quality of the filtering operation depends mainly on the number and the location of the microphones toward the focus point [4]. Figure 1 details the position of the array toward the reconstruction plane.

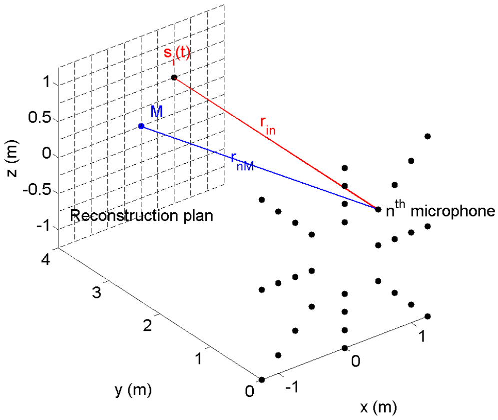

Figure 1 –Representation of a star-shaped array toward the sources and the reconstruction plane

By focusing on several points, a reconstruction plane is created. The beamforming provides the estimation of the acoustic contribution of each point of this plane. The signals are transformed into the frequency domain. Figure 2 presents the example of two harmonic sources at 4000 Hz detected using a star-shaped array. The sources are estimated in an accurate way in terms of acoustic levels and positions (real positions are represented by red points) on the plane. The reconstruction plane is polluted by ghost sources due to the secondary lobes of the array response.

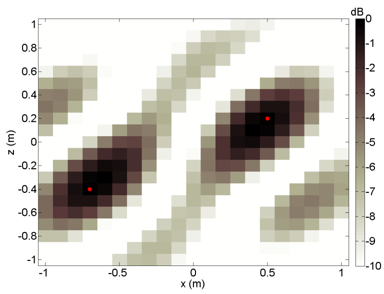

Figure 2 – Reconstruction plane showing the acoustic level estimation of two harmonic sources

3. TAKING THE MOTION INTO ACCOUNT

As the sources are in uniform non accelerated motion of speed $\vec{v}$, a Doppler effect is introduced on the microphone measurements. The distance r_{im} depends now on time, as shown in Figure 3. An amplitude modulation

$$A_{in}(t) = 1/\left(r_{in}(t)\left(1 - Ma\cos\left(\vartheta_e(t)\right)\right)\right) \tag{4}$$

and a frequency modulation

$$f_{in}(t) = f_0/\left(1 - Ma\cos\left(\vartheta_e(t)\right)\right) \tag{5}$$

are introduced on the source terms of Equation 1 [5]. $Ma = |\vec{v}|/c$ is the Mach number and ϑ_e the Mach angle between the source velocity and the observer's position seen from the source, as shown in Figure 3. When r_{in} is minimum, $A_{in} = 1/r_{in}$ and $f_{in} = f_0$, the Doppler effect is suppressed; the frequency acquired by the array is equal to that of the moving source.

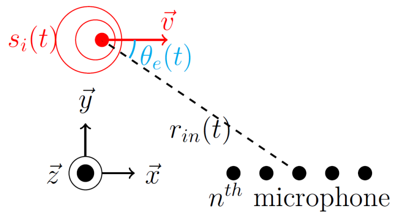

Figure 3 – Displacement of the source toward the array, which generates the Doppler effect

Figure 4 a) plots the time evolution of an harmonic source at 4000 Hz passing by a listening point at a speed of 300 km/h. The time frequency representation of the signal, in Figure 4 b), shows the frequency modulation. When the source comes closer to the listening point, the perceived frequency is pitched high, when the source goes away from the listening point, the perceived frequency is pitched down.

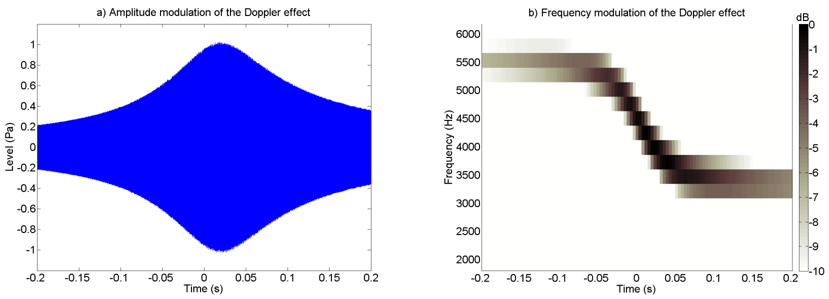

Figure 4 – Time and time-frequency representation of a Dopplerized sinus at 4000 Hz emitted by a moving source at 300 km/h. The amplitude and frequency are modulated proportionally to the source-listener distance.

3.1 Filtering the Doppler effect

This approach considers beamforming as an optimal filter; the focus points are on a vertical mesh in front of the array where the Doppler effect can be considered as a linear modulation [2]. Then, the output of the beamforming processing can be considered as an indicator of the source acoustic levels. By projecting the output on a time-frequency domain, using the short time Fourier transform (STFT):

$$STFT\{x_M(t)\} \equiv X_M(\tau, f) = \int_{-\infty}^{\infty} x_M(t)h(t-\tau)e^{-j2\pi ft}dt, \tag{6}$$

the time evolution of the source spectrum is estimated. h is a weighting window (typically a Hann window), which slides along the time axis for each value τ. The window is $\Delta\tau$ long and determines the spatial resolution of the mesh along the train length, which is $|\vec{v}| \times \Delta\tau$; the frequency resolution is equal to $1/(\Delta\tau)$. As the filtering method cannot provide accurate results in both the space and frequency domains, a compromise must be found.

Figure 6 a) plots the time-frequency representation of the filtered Doppler effect of a harmonic source at 4000 Hz at the speed of 300 km/h. The Doppler effect is not totally suppressed, but the source spectrum and its position can be estimated. Figure 6 b) presents reconstruction plane and the detection of the moving source. Ghost sources are observed and pollute the estimation. The spatial accuracy along the abscissa is about 0.6 m, and depends on the speed of the sources and the desired accuracy on frequency.

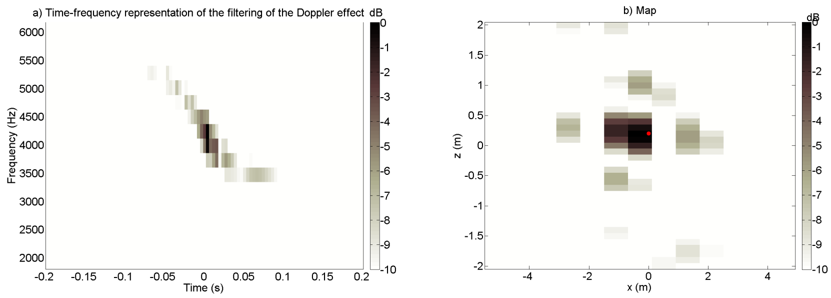

Figure 6 – Time-frequency representation of the Doppler effect and detection of the source on the reconstruction plane.

3.2 Dedopplerization method

The dedopplerization method is an adaptation of classical beamforming for acoustic sources in motion proposed by Barkisow [6] in 1988. It consists in considering the displacement of the focusing mesh at the same speed of the moving object. Then, Equation 1 becomes

$$x_M(t) = \sum_{n=1}^{N} \left(4\pi r_{nM}(t)\right) p_n\left(t + r_{nM}(t)/c\right). \quad (7)$$

The Doppler effect is suppressed from the signals recorded by the microphone array and then beamforming processing is applied to the data, the time signal becomes stationary as shown in Figure 7 a), for a source at 4000 Hz moving at 300 km/h. A classical spectral density estimator (*e.g.* Welch estimator) is performed to retrieve the source position on the map, as in Figure 7 b).

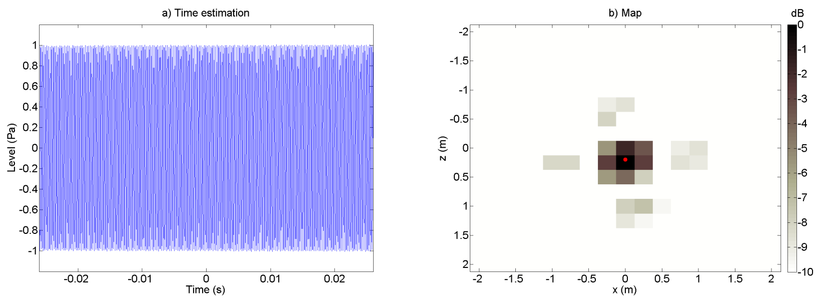

Figure 7 – Reconstruction of the source signal in time and detection of the source on the reconstruction plane

Properties of the array evolve strongly while the focus point follows the sources. To minimize these variations, the incidence angle is limited to ±30°. Then the reconstructed signal x_M is limited to a short duration. This duration constrains to a low frequency resolution in the Fourier domain.

4. APPLICATION: IDENTIFICATION OF THE SOURCES ON A TGV POWER CAR

In order to make comparisons between the methods, both processing techniques are applied to the same measurements (*i.e.* same train and same array). The TGV is passing by the array at the speed of 300 km/h. Results are presented using a TGV map to identify noise sources for the one-third octave bands centered on 1005 Hz, 1272 Hz, 1587 Hz and 2005 Hz.

4.1 Noise maps

Figure 8 plots the noise maps calculated with the linear filtering method and Figure 9 the noise maps calculated using the dedopplerization method. The maximum level of Figures 8 and 9 is set to 0 dB and the dynamic range is limited to 10 dB.

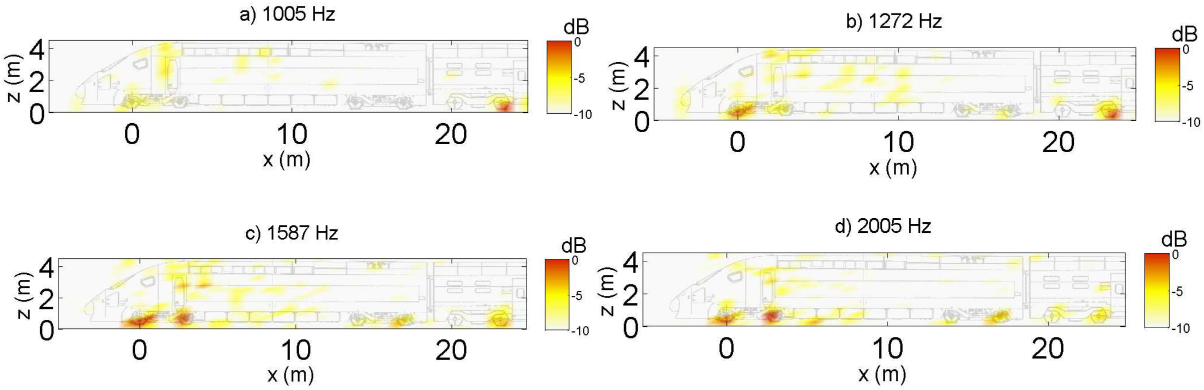

Figure 8 – Identification of the main noise sources on the train using the filtering method

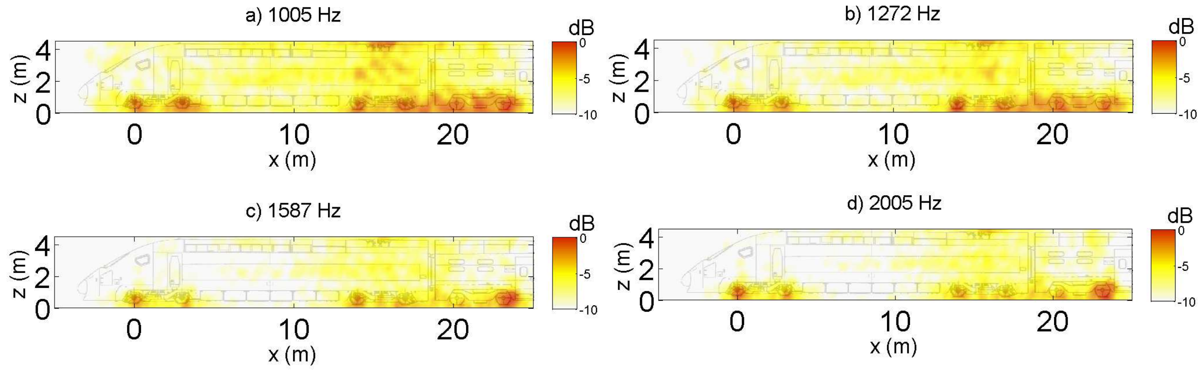

Figure 9 – Identification of the main noise sources on the train using the dedopplerization method

Important noise sources (from 0 dB to -5 dB) can be seen, using the dedopplerization method on the pantograph cavity and on the louvers at the one-third octaves centered on 1005 Hz and 1272 Hz [see Figures 9 a) and b)]. Those sources are not detected on the maps resulting from the filtering method [or with a lower level, like the louvers in Figure 8 b)]. Their localization indicates they are aeroacoustic sources.

Other aeroacoustic sources can be seen in Figures 8 a) and 9 a), at the one-third octave centered on 1005 Hz, on the door of the driving cab, the bogies and the nose of the train by both methods.

Wheel and rail emissions are retrieved on all maps. The acoustic level is estimated with the same relative power using the filtering method, Figures 8 a), b) c) and d), or the dedopplerization technique, Figures 9 a), b), c) and d).

4.2 Estimations of the spectrum around a wheel

Let us consider the first wheels of the train in Figures 8 and 9. Both methods are used to estimate the spectrum of the radiated sound at the center of the wheel. As previously mentioned, aeroacoustic sources (around 1000 Hz) and rolling noise are dominant noise sources in this area. Figure 10 plots the estimated spectrum in narrow band, using the filtering method (in red) and the dedopplerization method (in blue).

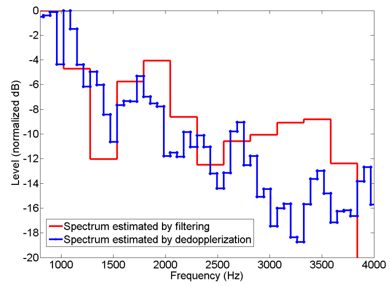

Figure 9 – Estimation of the spectrum around the first wheel

Tendencies are observed on both curves, same decreasing, approximately same relative level. The frequency resolution of the filtering method limits the detection of resonant frequencies. Peaks are detected with the dedopplerization method around 1700 Hz, 2700 Hz, 3500 Hz and 3900 Hz, and around 1700 and 3500 Hz with the filtering method.

4.3 Comparisons of the methods

Results are closely linked. Most of the main sources are retrieved at the same positions in the same one-third octave bands and the same power. However filtering method does not retrieve large aeroacoustic sources such as the pantograph cavity and louvers of the TGV.

Processing have been performed using Matlab and a personal computer with an Athlon 64 3400+, 2.3 GHz processor. The linear filtering method took 200 s and the dedopplerization took 12,000 s. Moreover, the dedopplerization method requires to know the train speed accurately to be performed.

Concerning the accuracy of the results, filtering method has a spatial mesh step of 0.68 m and a frequency precision of 256 Hz, whereas the dedopplerization method provides an accuracy of 0.25 m and a frequency precision of 64 Hz using Welch's estimator (23 means).

The filtering method seems to provide more accurate spatial results but, due to the STFT, spatial and frequency resolution are conversely dependent. The dedopplerization method presents maps with lower signal to noise ratio, due to stronger hypothesis on the array response, the array properties evolve while focusing.

The spatial and frequency accuracies are uncorrelated in the dedopplerization method, thus the accuracy reached can be higher than the filtering method but very time consuming.

5. CONCLUSIONS

A better knowledge of the noise sources on the train is required to propose more adapted reduction solution. Beamforming is a suitable array processing to provide the characteristics of those sources on high speed trains. Arrays and methods must be adapted to the railway context.

Two methods are presented in this paper to estimate the source characteristics, in terms of position, power and spectra, considering a high speed motion.

These methods are based on different assumptions on the array; both states that the array is a perfect spatial filter and that the dedopplerization requires an invariant array response toward the focusing points. According to [3], the array response can be adapted to the signal processing.

With the growing needs of accuracy and correctness on the estimation of the source parameters, the dedopplerization method seems relevant. According to the one-third octave band representations, the filtering method offers quite good results on positions and power of the main sources, but seems to underestimate aeroacoustic sources (pantograph cavity and louvers). For more accurate spectral and spatial estimation, the dedopplerization method is required.

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