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DESIGN AND IMPLEMENTATION OF A DOPPLER REMOVAL METHOD FOR PARAMETERS EXTRACTION AND AURAL SIMULATION OF TRAFFIC NOISE

Fernando A. Marengo Rodriguez[a], Federico Miyara[b]

Laboratorio de Acústica y Electroacústica, Facultad de Ciencias Exactas, Ingeniería y Agrimensura, Universidad Nacional de Rosario. Riobamba 245 bis, Rosario, Argentina,
http://www.fceia.unr.edu.ar/acustica/

[a]*fmarengorodriguez@yahoo.com.ar, fmarengo@eie.fceia.unr.edu.ar*

[b]*fmiyara@fceia.unr.edu.ar*

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Abstract. Noise effects and traffic noise impact at an urban location can be assessed by means of aural simulation. This simulation could be performed from the knowledge of the noise produced by each vehicle individually, the source movement effects (such as geometric divergence and Doppler shift), and sound reflections from façades. In theory, the noise sample from each vehicle could be obtained by recording it with a microphone that moves along with the source. However, this approach would be impractical due to turbulence noise that would be generated on the microphone. In order to overcome this drawback, a technique based on the recording of the noise sample at a fixed station, followed by a movement-compensation algorithm is proposed and tested with experimental data. In order to carry out the compensation stage, a pilot tone is radiated from the vehicle during the passby in order to estimate the instantaneous frequency shift of the source. In this way, the movement is coded in the phase modulation of the tone, which can be retrieved through a ridge searching algorithm applied to the input spectrogram. Therefore, the distortion affecting the pilot tone can be easily compensated for by reconstructing its original phase from the recorded one, using an interpolation algorithm. The tone frequency is selected so that there is negligible spectral overlap with the spectrum of the source. This stage gives a variable-step temporal sequence which is uniformly interpolated with the input time vector for Doppler compensation of the whole data set. As will be seen, due to the selection of the tone frequency, this technique is quite robust and allows the determination of the source speed over a long time interval. In the data caracterization stage, the compensated noise is separated into tonal and non-tonal components locally, i.e., taking into account their energy contribution to the whole noise as a function of the vehicle position. Non-tonal information is simulated by white noise filtered by a reduced set of appropriate Linear Prediction Coefficients (LPC) filters in order to represent accurately the spectral envelope of non-tonal spectrum evolution. This compensation and caracterization method is applied to experimental data and assessed by aural simulation tests over an arbitrary time interval for an open virtual profile urban street.

1 INTRODUCTION

Several traffic noise prediction methods have been proposed for the evaluation of the acoustical impact produced by the existing and projected roads and highways. Many of these methods are based upon the statistical description of noise sources, traffic intensity and surrouding conditions such as neighboring surfaces, buildings and ground absorption (Barry and Regan, 1978; US Department of Transportation Research and Special Programs Administration, 1995; Hankard et al,. 2006). The models currently in use enable the prediction of the expected noise equivalent levels, which are usually represented graphically by means of noise maps. Regardless of this fact, these techniques do not allow the prediction of the noise inside the buildings, which is even more relevant than the noise outdoors. Also, the information given by these models can only be interpreted by specialists.

There is a new concept based on the digital aural simulation of traffic noise, which enables the qualitative evaluation by non-specialists (Miyara et al., 2003; Tanaka et al., 2004, 2008; Marengo Rodriguez and Miyara, 2008b). Furthermore, by combining this technique with the filtering effect produced by façades, it is possible to simulate the indoor noise. As a side result, the usual quantitative indicators both outside and inside buildings can be easily determined.

Aural simulation techniques were developed by Miyara et al. (2003) in order to elucidate the effect of façades on traffic noise. More recently, a new traffic road simulation scheme was proposed taking into consideration the contributions made by exhaust, tyre and engine noise for a few kinds of vehicles (Tanaka et al., 2004, 2008). Even though this model enables aural simulation for different traffic noise situations, it does not take into account the aerodynamic noise as well as variations in the engine noise produced when the vehicle is loaded.

In Marengo Rodriguez and Miyara (2008b) a novel methodology for carring out aural simulation of traffic noise was proposed, under the basis of the extraction and parametrization of the most relevant features of the noise signal produced by each vehicle individually. An anechoic sample of such noise (free from Doppler effect) would be obtained by means of a microphone travelling along with the vehicle, but too much turbulence noise would corrupt the signal of interest, turning it useless. For this reason, the noise was recorded from a fixed position station close to the vehicle path and subsequently processed by a Doppler and divergence compensation algorithm. The compensated sequence was described in terms of its spectral tonal and nontonal components so as to aurally simulate the noise produced by the vehicle during its passby, including Doppler and divergence effects for any speed. As this methodology does not ignore the contribution of any noise sources, it would constitute a more complete description of the vehicle noise. Also, it is worth mentioning that the compensated signal was analised over a small time interval around the instant in which the vehice is closer to the observer, also referred by Marengo Rodriguez and Miyara (2008b) as the reference instant.

In the present work, an improvement of the technique mentioned in the previous paragraph is presented. In comparison to that method, the one proposed here is simpler, more precise, more robust and takes into consideration the directional radiation of the source.

2 PROPOSED METHOD

The methodology proposed here starts with noise recording from a receiver station located at a fixed position close to the rectilinear vehicle path. If $p_0(t_e)$ is the sound pressure at a reference distance d_0 that corresponds to the emission time t_e, $p(t)$ is the sound pressure at the receiver at time t, and d is the distance between them, the signal detected at the receiver is given by

$$p(t) = p_0(t_e)\ .\ d_0\,/\,d(t), \tag{1}$$

where $t_e = t - d(t)/c$, and c is the sound speed. In case the emitted sound contains a pure tone, its frequency is affected by Doppler shift caused by the relative movement between source and receiver (Pierce, 1989). As the vehicle does not emit such a tone, the addition of a pilot tone from the vehicle is proposed here. Its frequency is adopted so that there is little spectral overlapping between the added signal and the main information of the source noise.

After recording the noise signal with the pilot tone, it is processed by a Doppler shift and geometric divergence compensation algorithm. To this aim, the spectrogram of the input signal is computed and the temporal evolution of the tone frequency is detected by means of a ridge searching algorithm. Then, the temporal evolution of the tone Doppler-distorted frequency is mapped onto the undistorted one, resulting a non-uniformily sampled temporal sequence. Doppler shift compensation is carried out by interpolation of the mapped recorded sequence over the uniformly sampled discrete time. Divergence compensation is easily obtained by multipling the sequence obtained in the previous stage by a factor directly proportional to the distance between the pilot tone source and the receiver. This distance is determined from its minimum value (measured in the experiment) and from the instantaneous speed of the source which is coded in the temporal evolution of the pilot tone frequency.

The following step consists in analysing the compensated signal as follows: *a*) extraction of the most intense tonal components of the spectrogram, and *b*) representation of the remaining input information by a linear prediction model (LPM), which consists in filtering white noise with a linear prediction coefficients (LPC) filter.

3 IMPLEMENTATION

In order to carry out our technique, we propose digitally recording the sample noise of the source S from the station O as described in the experimental setup depicted in **Figure 1**. The source moves along a straight path in an open profile street so as to avoid unwanted reflections from façades (Miyara et al., 2007).

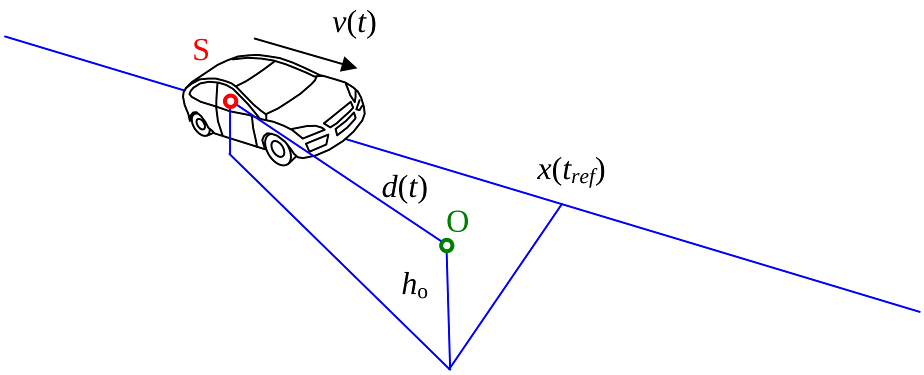

Figure 1: Experimental setup for the proposed method.

3.1 Compensation stage

With the aim of removing Doppler shift and geometric divergence from the recorded signal, the instantaneous frequency of the pilot tone is detected. Such detection is carried out by means of a ridge searching algorithm (Miyara et al., 2002, 2005) of the input spectrogram. Then, the associated phase ϕ is obtained by integration of the frequency time history calculated in the previous step. In order to remove the Doppler shift, this phase is mapped onto the Doppler-undistorted one. The input phase is expressed as

$$\phi(t_k) = 2\pi f_0 \cdot t_{ek}, \tag{2}$$

where f_0 is the pure tone frequency, t_k is the uniformly sampled discrete reception time and t_{ek} is the non-uniformly sampled discrete emission time. Therefore, phase mapping consists in obtaining the sequence t_{ek} from Eq. (2) followed by cubic interpolation of the input sequence at a uniformly spaced time sequence in order to recover the original sampling rate. An example of such implementation is shown in **Figure 2**. The input signal affected by Doppler shift, together with its uniformly spaced samples (associated with $\phi(t_k)$) are depicted in **Figure 2** panel (a). Panel (b) shows the mapped signal and its non-uniformly spaced samples (associated with t_{ek}), and panel (c) illustrates the resulting mapped tone uniformily sampled.

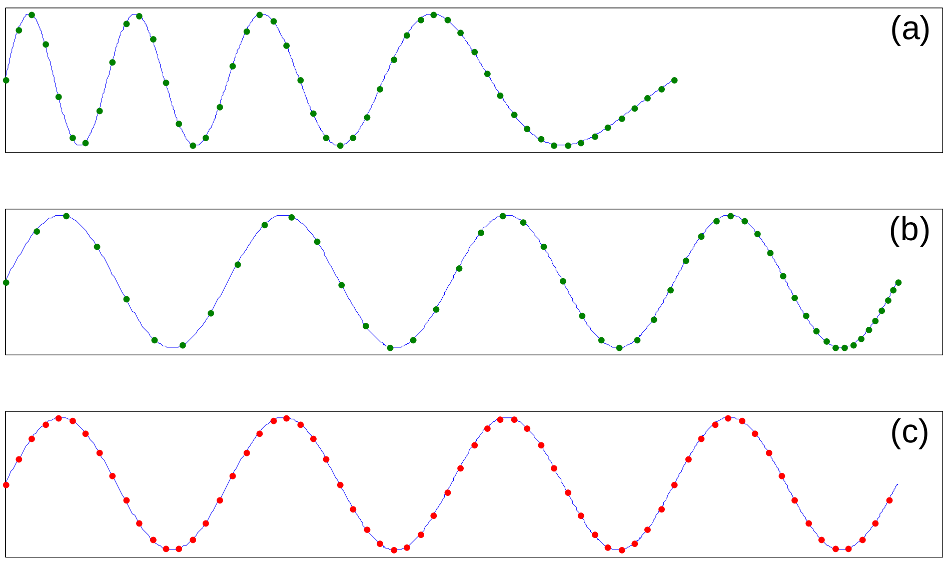

Figure 2: Pilot tone for illustrating Doppler compensation: (a) input signal and its equally spaced samples; (b) output signal and its non-uniformily spaced samples; (c) output signal uniformily sampled.

It is important to note that this Doppler compensation stage *a*) removes frequency distortion due not only to Doppler effect, but also to any other cause, for instance wind or velocity variations; and *b*) operates in the phase domain rather than in the frequency domain, avoiding the need for numerical differentiation, and hence adding robustness to this method.

As for divergence attenuation compensation, the pilot tone frequency detected is associated with the theoretical expression of Doppler-distorted frequency $f_D(t)$ given by (Pierce, 1989)

$$f_{\mathrm{D}}(t) \;=\; f_0 \frac{c}{c + v(t)\cdot \dfrac{x(t)}{\sqrt{x(t)^2 + R^2}}}, \tag{3}$$

where $x(t)$ is the linear coordinate of the vehicle position assumed as low-order polynomial, $v(t)$ is the instantaneous vehicle speed (see **Figure 1**), and R is the minimum distance between the pilot tone source and the receiver (measured in the experiment). The obtained frequency sequence was least-square fitted to Eq. (3), and the linear coordinate was decoded from this fit. By knowing the numerical value of R, the distance $d(t)$ is determined and multiplied by the Doppler-compensated signal in order to obtain the signal completely compensated.

3.2 Characterization stage

The signal obtained in the previous step is analised in terms of the temporal evolution of its spectrum, since the detected noise is composed by several sources located at different parts of the vehicle under study. Hence, the model which describes this noise is both directional and parameter distributed. From the input spectrogram, the most intense tonal components are retrieved by means of a multitracking filter, which consists of a ridge searching algorithm followed by masking of the detected components recursively applied to the spectrogram. The result of this step is a set of pseudotones $\{P_n(t)\cdot\cos\phi_n(t)\}$, where each one is associated with its analytic signal $P_n(t)\cdot\exp[j\phi_n(t)]$ (where j is the imaginary unit) by means of its Hilbert transform (Hahn, 1996). Consequently, these tones are resynthesised from a few samples of their instantaneous amplitude and phase. The remaining wideband noise of the input signal is associated with a set of LPMs for different positions of the vehicle path. Each LPM consists of white noise filtered by an N-th-order LPC filter, where N is the filter order, conveniently selected to represent the input noise with high quality. The synthesis of the wideband noise is normalised in terms of a few samples of its instantaneous rms value.

4 NUMERICAL RESULTS

4.1 Experiment and data compensation

The experiment proposed here was carried out using a Suzuki Fun automobile 2003 model furnished with a 1100 cm^3 4 cilinder engine moving along a straight path in an open profile street. Its average speed was about 35 km/h. The pilot tone of f_0=1000 Hz was generated at a sample rate of 44100 Hz and reproduced with a standard MP4 device and an autoamplified battery-supplied general purpose speaker located by the side window of the vehicle (h_s=1,5 m). The tone emision level was moderate and its frequency was sufficiently high, so there was no peaking at the receiver and no overlap with the main spectral information of the source noise.

At the receiver station located at 2,4 meters from the vehicle path, data were digitally recorded at a sampling rate of 44100 Hz using a Rion NL-15 sound level meter connected to a digital recorder Zoom H4 furnished with a memory card located at a height of h_o=1,42 m. The wind speed during the whole experiment was less than 1,5 m/s and a windscreen was used in order to minimise signal contamination. Also, the ambient temperature was 32,9 ºC.

At the signal processing stage, a temporal interval of 8 s around the reference instant was selected so as to analise spectral components of the noise over nearly 80 m of the vehicle path. The signal studied is illustrated in **Figure 3**.

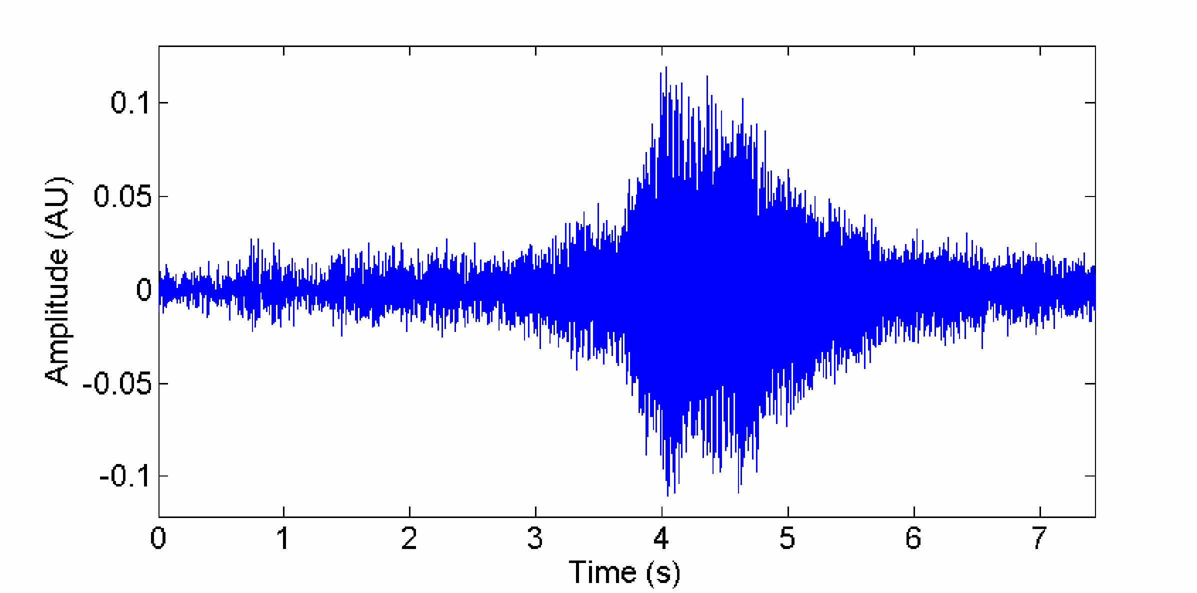

Figure 3: Selected input signal.

The input signal was divided into 4096-sample frames weighted by a Hanning window with an overlap factor of 0,5, and each segment was Fourier transformed with a resolution of 4096 points. Then, the ridge searching algorithm was applied to this spectrogram in order to compensate for Doppler shift. This algorithm sequentially computes the maxima of the 2D signal in the neighborhood of the pilot tone frequency, taking into account that the resulting temporal signal cannot change abruptly. Afterwards, in order to compensate for divergence attenuation, data from this sequence were least-square fitted with the Doppler-shifted frequency signal that would be produced by a linearly accelerated rectilinear movement of the source, i.e., $x(t) = x_0 + v_0 t + a_0/2 \cdot t^2 + b_0/3 \cdot t^3$. **Figure 4** illustrates the input spectrogram around the tone frequency; its maxima, determined by the ridge searching algorithm, and its least-square fit. As this figure clearly shows, the pilot tone frequency detected resembles the theoretical Doppler-distorted one. As shown in the spectrogram as a function of time (**Figure 5**) or position (**Figure 6**), the compensated signal is free from Doppler and geometric divergence distortion, and demonstrates the accuracy of the algorithm proposed here. Its robustness is evident near the borders where the tone amplitude reduction reaches about 1,5 % and signal to noise ratio is quite poor. Finally, the average speed of the vehicle estimated from the fitted frequency was 33,75 km/h, being this result in agreement with the speed measured during the experiment.

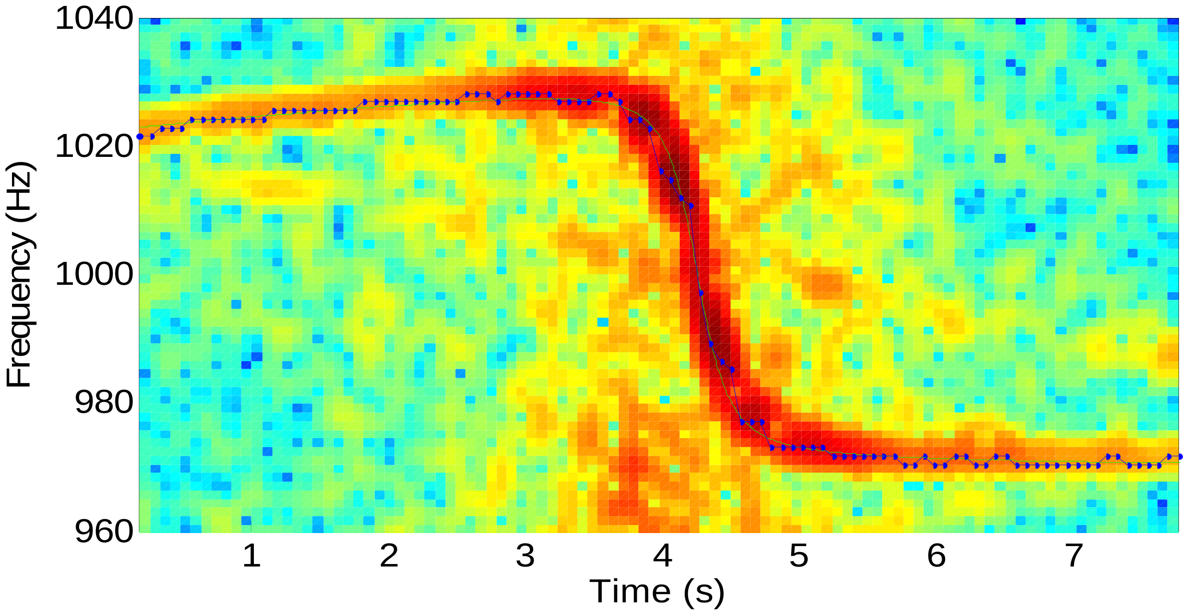

Figure 4: Input spectrogram around the pilot tone frequency, its maxima given by the ridge searching algorithm (dotted blue line) and its least squares fit (solid green line).

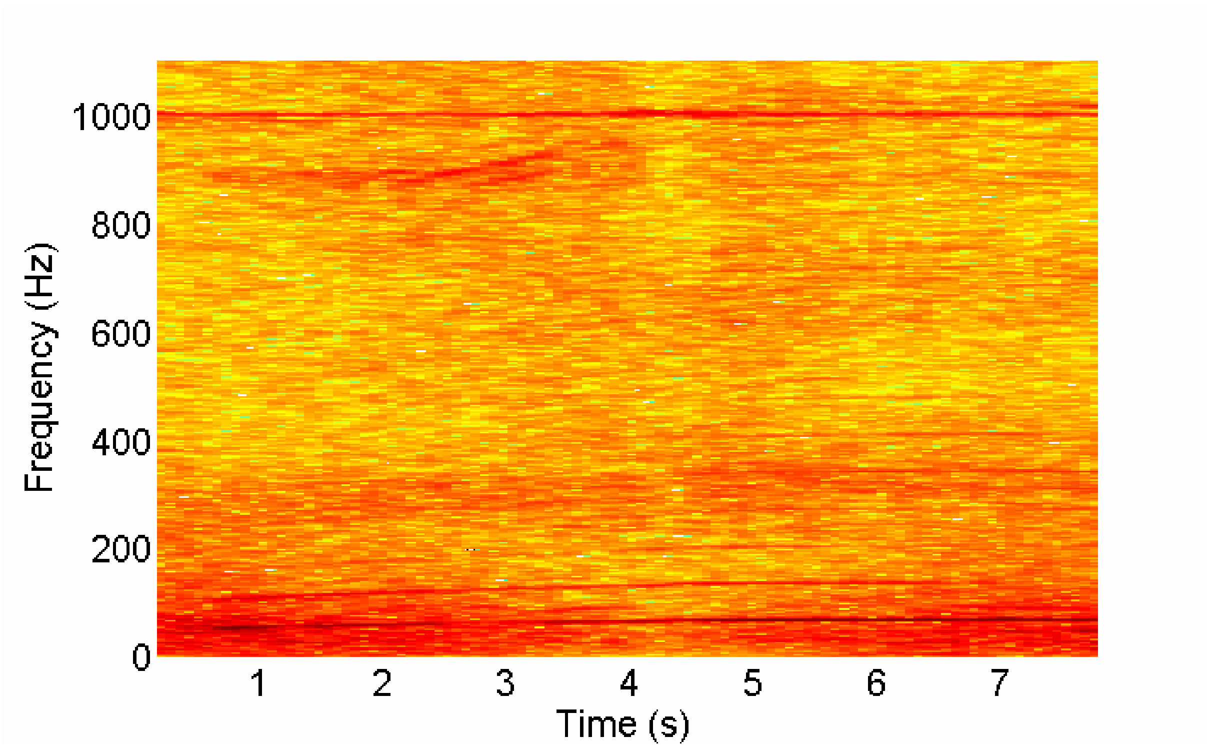

Figure 5: Spectrogram of the compensated signal including the pilot tone information.

4.2 Parametrization

The spectrogram of the compensated signal was computed with the same 2D resolution as in the recorded signal, and it was processed by the previously mentioned multitracking filter. Although some parameters were driven heuristically for this filter (such as the quantity of tones to detect and their estimated bandwidths), a complex of 5 pseudotones in the audio frequency band were successfully discriminated in the input sequence (see **Figure 6** and **Figure 7**).

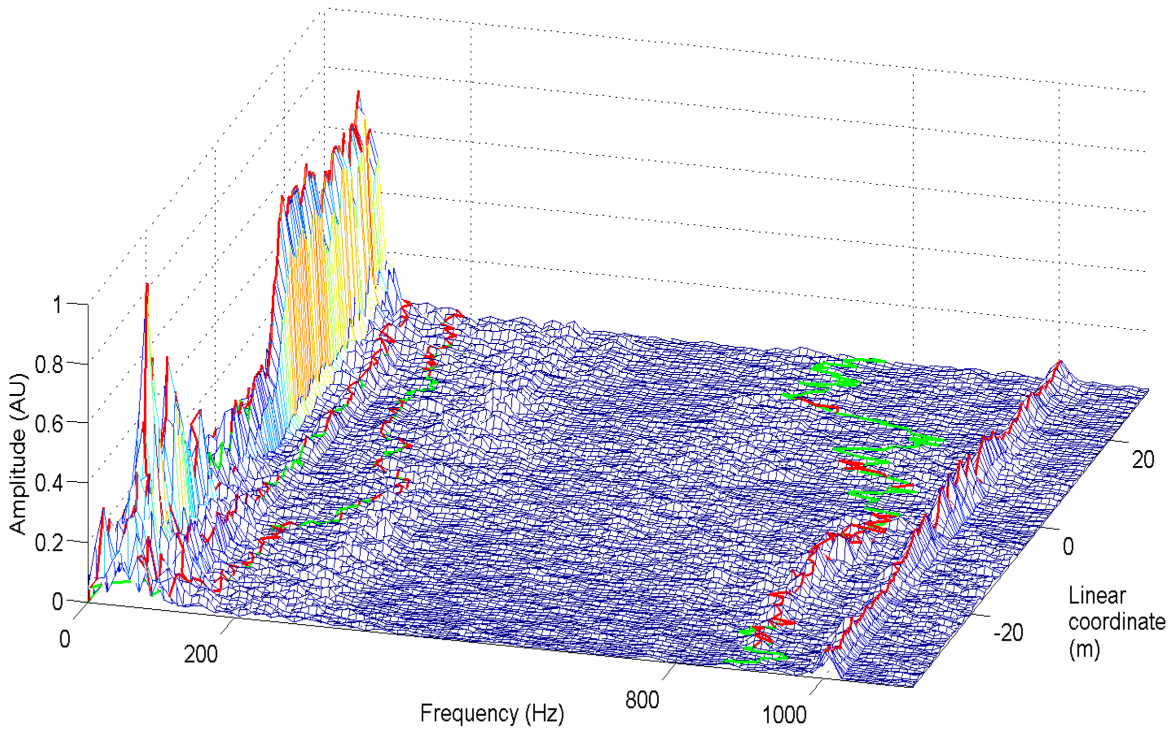

Figure 6: Spectrogram of the compensated signal and amplitude-position-frequency plot for each tone detected (solid lines). Low amplitude values for these tones are shown in green lines. Pilot tone information is included to illustrate the accuracy of the compensation algorithm proposed here.

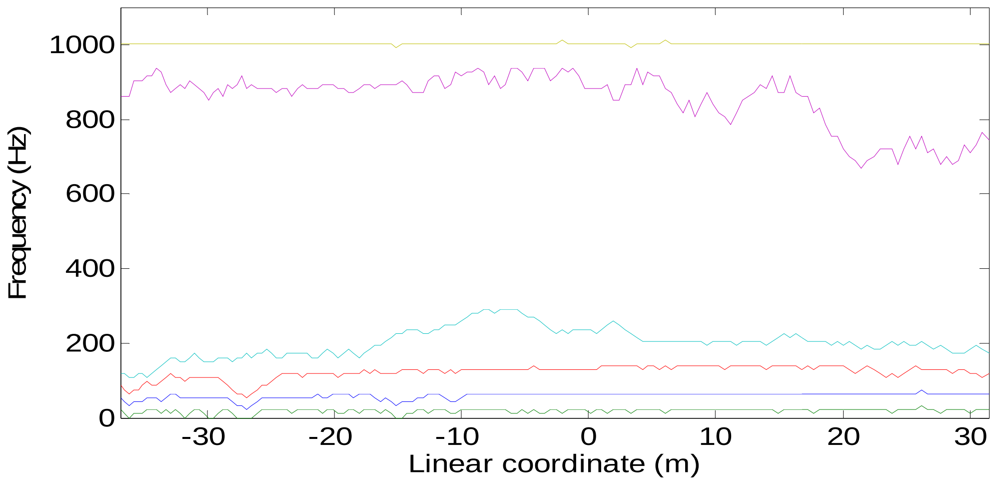

Figure 7: Frequency evolution for each tone as a function of the vehicle position, including pilot tone information.

The extracted polyharmonic signal (shown in **Figure 8** (a)) was split into a set of functions through a set of appropriate bandpass filters. As expected, each one of them is a purely oscillating zero-mean function due to its narrow bandwidth. These tones were added to their respective Hilbert transformed signals multiplied by the imaginary unit in order to compute the set of the associated analytic signals. Among several algorithms for calculating the Hilbert transform are available, the *hilbert* function in the MATLAB Signal Processing Toolbox (MATLAB, 1996) was used due to its simplicity, effectiveness and computational load (Marengo Rodriguez et al., 2008a). The analytic signals obtained herein code the amplitude and phase information for each tone and for several positions of the vehicle along its path.

In order to reconstruct the multitonal input signal, first amplitudes and phases were obtained by cubic interpolation to get $P_n(t_k)$ and $\varphi_n(t_k)$, subsequently the signal was computed as

$$p_T(t_k) = \sum_{n=1}^{5} P_n(t_k)\cos\left(\varphi_n(t_k)\right). \tag{4}$$

The sequence so obtained is depicted in **Figure 8** (b), and in comparison with the sum of the input tones obtained by filtering, the accuracy of the proposed method is evident.

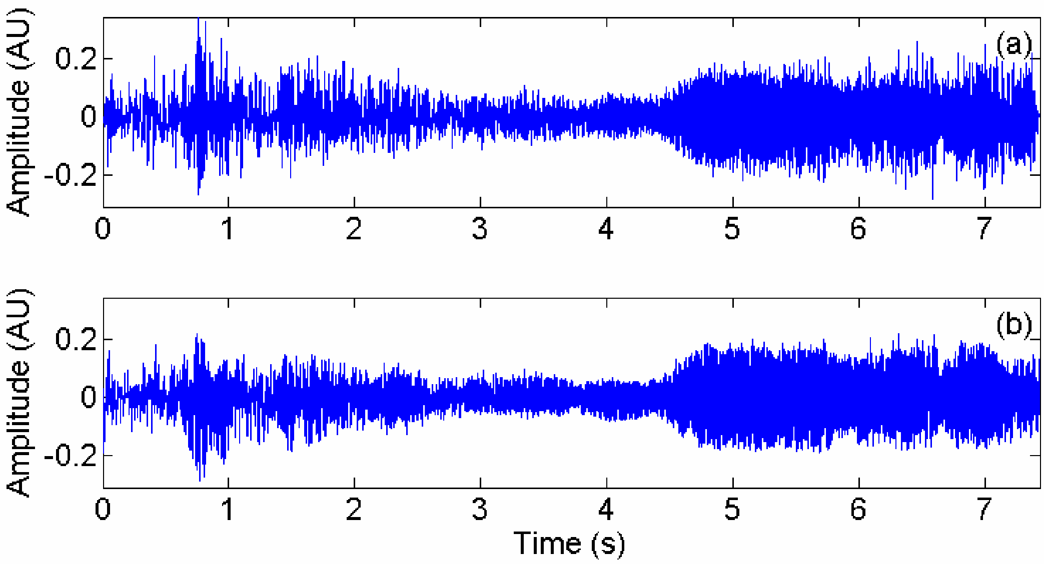

Figure 8: Temporal evolution of the (a) input and (b) synthesised multitonal components.

The background wideband noise detected in the compensated signal was analised spectrally (see **Figure 9**), and although little temporal changes were visually detected, aural experiments revealed relevant psychoachoustic changes. Therefore, five LPMs were determined for different representative positions of the vehicle during its pass-by (see **Figure 10** (a)). The LPC filters used for these LPMs were of order N=20, and Hanning window with an overlap factor of 0,5 was applied to the sound synthesised by each one of these models. The signal obtained was multiplied by a reduced set of rms input values, with the result depicted in **Figure 10** (b).

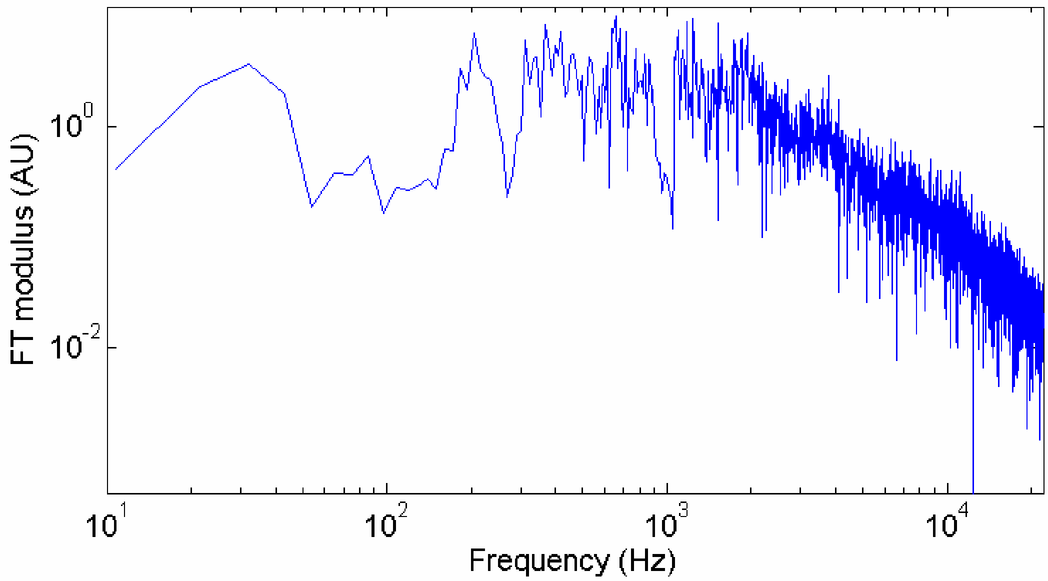

Figure 9: Wideband noise spectrum for a time interval around the reference instant.

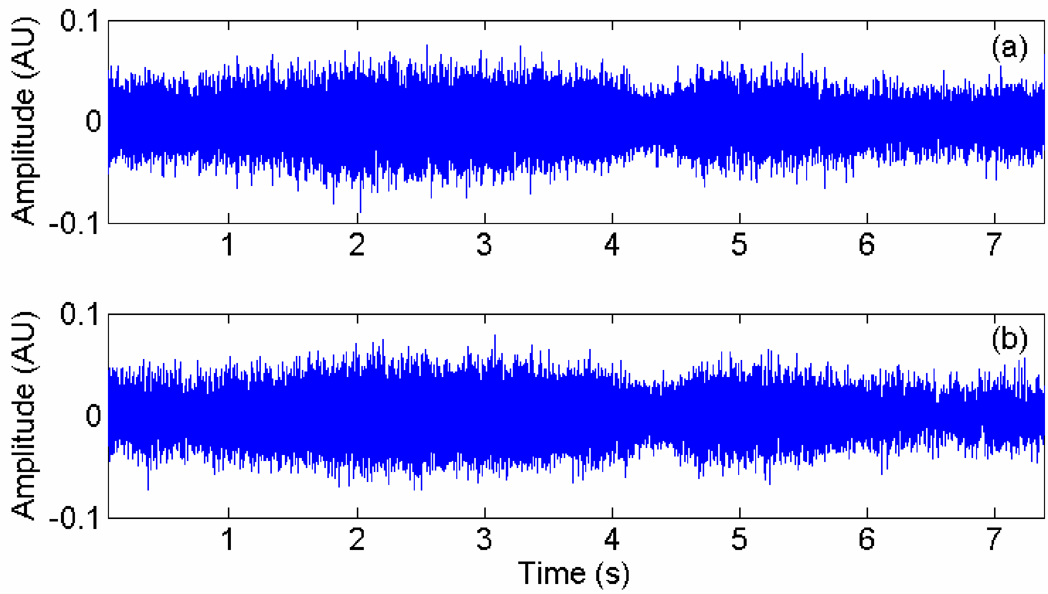

Figure 10: Wideband component: (a) input signal; (b) simulated sequence.

4.3 Aural simulation

The synthesis of the input signal was carried out by using the input data $x(t)$ (detected in section 4.1) and cubic interpolating them with a Doppler-and-amplitude distortion procedure described in Miyara et al., (2003). The synthesised signal is shown in **Figure 10**, and by comparing it with the recorded one visually and aurally, the performance of the method proposed in this paper is verified successfully.

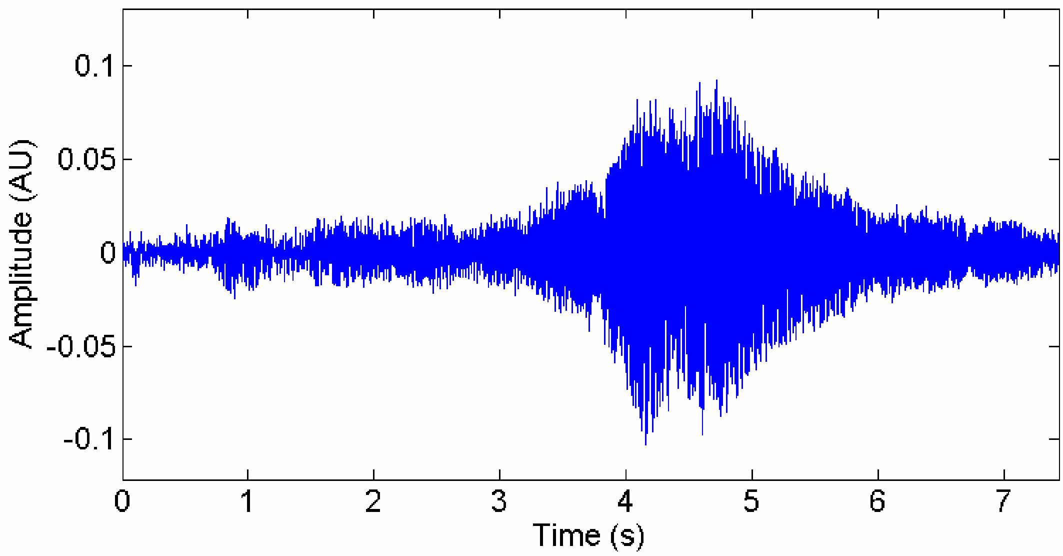

Figure 11: Temporal evolution of the simulated signal.

5 CONCLUSIONS

In this paper, a new approach for synthesising the noise produced by a vehicle during its passby was presented and tested experimentally. This method relies on the information that would be obtained by a microphone moving along with the vehicle but avoids contaminating turbulence noise that would be recorded. Instead, the sample noise was recorded from a fixed station and the recorded signal was compensated by a proper algorithm through a mapping stage. Signal compensation takes into account both Doppler shift and amplitude distortion, and is based on the information contained in a pilot tone added to the vehicle noise. Ridge searching was applied to the input spectrogram in order to retrieve the temporal evolution of the tone frequency, and by least-square fit it was possible to recover the linear coordinate that gives information about the vehicle noise for several positions. Subsequent analysis of the compensated noise via spectrogram and multitracking filter made it possible to discriminate tonal components from nontonal information with reasonably low resolution. Detected tones were associated to their corresponding analytic signals through the Hilbert transform, and wideband noise was represented by a reduced set of low-order LPC filters. In the aural simulation stage, polyharmonic information was reconstructed by interpolating the reduced set of amplitude and phase data, and the remaining wideband noise was sinthesised with white noise filtered by the previously mentioned set of LPC filters weighted by input rms values. The results presented in this work show that the proposed method is robust since it works in the phase domain rather in the frequency domain and avoids the need for numerical differentiation. Moreover, this approach is cheap, easy to implement, effective and requires little computational load. Its accuracy was successfully tested by aurally simulating the input noise. The results obtained here could be extended to include the effects of façades, and a more complex traffic noise situation could be simulated for both open and U virtual profile streets. Theses cases are of great concern in the international community and the method developed here makes it possible to assess its acoustical impact.

6 AKNOWLEDGEMENTS

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