Inverse Filter Optimisation Applied to Sonic Booms Reproduction

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Introduction

The continuing interest in the human hearing response to sonic booms encourages us to experimentally establish the relationship between the subjective evaluation of the annoyance felt by someone in-situ and the characteristic parameters of sonic-boom signatures. Following experiments [1] and [2], synthesized signals have been computed, allowing us to play freely with the characteristic parameters of the waveforms. Nevertheless, without signal pre-processing, these sounds are barely heard as sonic-boom explosions as their reproduction is distorted. For example, the perceived difference between two different reproduced N-waveforms is weak as the distortions are predominant in the resulting hearing sensations. The method of inverse pre-processing of the signal to eliminate the transducer distortions can be applied to sonic-boom high-fidelity reproductions. This process allows us to actually perceive two different sonic-boom signatures as different.

Reproducing System Modelisation

The system is composed of electrodynamic large-bandwidth closed-headphones (Sennheiser, HD280Pro) coupled to a dummy-head in which two control microphones (B&K-4192-B-001) have been fixed. Figure 1 schematizes the system identification procedure by directly measuring the frequency response H(f) of the system to a white noise and swept-tones for one ear.

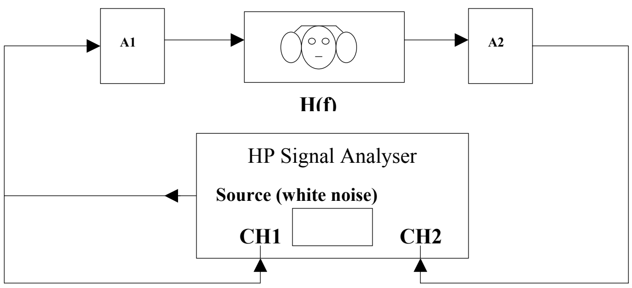


Figure 1: Block diagram of the system identification procedure by measuring Ĥ(f) with the signal analyser (Hewlett Packard 35665A). Note that A1 and A2 are two power amplifiers which ensure a perfect electrical impedance adaptation and have no effect on the measurement of Ĥ(f). They have been respectively set to 0 and 20 dB gains.

H(f) involves the frequency response of the left headphones transducer, the frequency response of the control microphone and the filtering function of the closed-volume cavity defined between them. The measured frequency response Ĥ(f) is given by averaging 1000 samples, among which 50% of the samples correspond to a white noise excitation and 50% to a swept pure-tones excitation in order to extend the accuracy of the measurement in the low-frequency range. The measured frequency response Ĥ(f) and its corresponding impulse response ĥ(t) have been plotted on Figures 2 and 3 respectively.

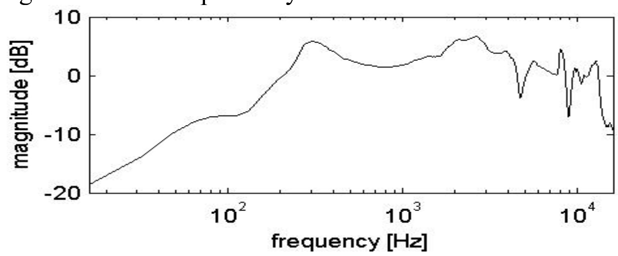


Figure 2 : Measured Frequency response Ĥ(f) with a coherence up to 99% from 16 Hz to 16 kHz (Δf=16 Hz).

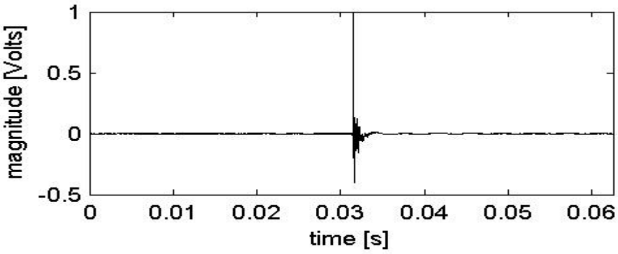
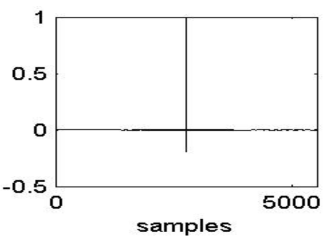


Figure 3 : Computed impulse response ĥ(t) from Ĥ(f) by the Inverse Fast Fourier Transform. Note that ĥ(t) has been time-centred in order to have a causal IIR filter to eliminate the delay involved in the following deconvolution process.

As long as $\hat{H}^{-1}(f)=1/\hat{H}(f)$, the computed inverse Fourier transform of $\hat{H}^{-1}(f)$ corresponds to the pursued inverse impulse response $\hat{h}^{-1}(t)$. The efficiency of this method can be estimated according to the high-fidelity condition defined by

$$|| \hat{h}^{-1}(t) * \hat{h}(t) - \delta(t) || < \varepsilon \quad (1)$$

Then, as explained on Figure 4.a) and Figure 4.b), the remaining error of the measurement process can be observed.



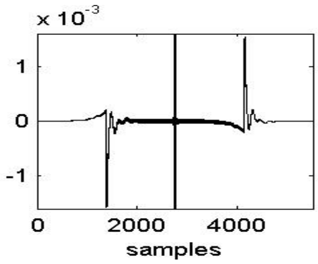


Figure 4.a) : Convolution of $\hat{h}^{-1}(t)*\hat{h}(t)$ **Figure 4.b):** Focussed view of Figure 4.a). Checking of the high-fidelity condition : two side-peaks of acceptable relative level of 0.15% correspond to the error in the model determined by the measurement.

This condition is the most important point of this study since it refers to the difference between the ideal model h(t) of the system and its measured model ĥ(t). After several investigations to improve the measurement process (zero-padding of $\hat{h}(t)$, LMS pseudo-random noise excitation signals [3], SVD [4]), the method exposed here corresponds to the best obtained result according to the high-fidelity condition.

Linear Deconvolution Process

The linear deconvolution process will be described for the reproduction of a synthesised signal extracted from the experiment detailed in [1]. The goal is to accurately reproduce the desired signal y(t). Without pre-processing of this signal, the measured acoustic output signal of the reproduction system can be modelled as the convolution of the input signal y(t) by the actual impulse response of the system h(t). Thus,

$$y_h(t) = h(t) * y(t) \neq y(t) \quad (2)$$

The pre-processing method consists in deconvoluting the desired signal y(t) with the measured impulse response of the system $\hat{h}(t)$ in order to eliminate the linear filtering effects of the reproducing system. This new input signal $y_t(t)$ is then defined by

$$y_t(t) = y(t) * \hat{h}^{-1}(t) \quad (3)$$

Once reproduced, $y_t(t)$ is convoluted by h(t) to obtain the new acoustic output signal $y_{th}(t)$ which is similar to the desired signal y(t) under its acoustical form.

$$y_{th}(t) = y_t(t) * h(t) = [\, y(t) * \hat{h}^{-1}(t)\,] * h(t) \quad (4)$$
$$y_{th}(t) = y(t) * [\, \hat{h}^{-1}(t) * h(t)\,] \cong y(t)$$

As $h(t) \cong \hat{h}(t)$, equation (1) becomes $\hat{h}^{-1}(t) * h(t) \cong \delta(t)$, which is fulfilled at 99.85%. In order to evaluate this pre-process, the temporal waveforms of these signals and their spectrum have been respectively plotted on Figure 5 and Figure 6.

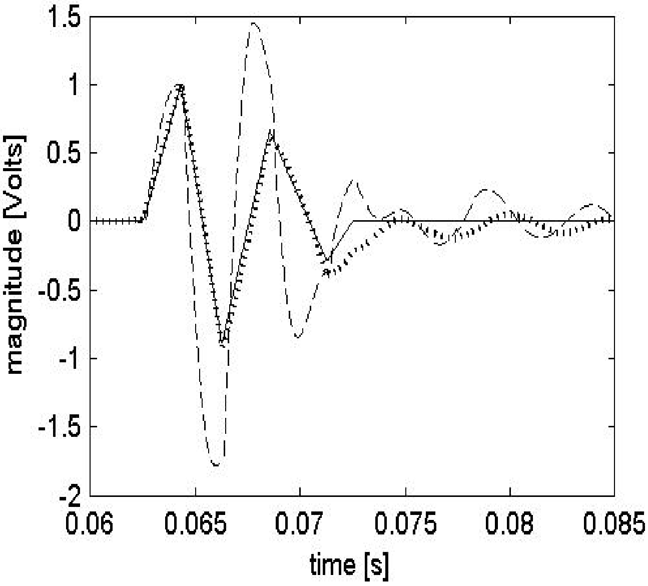


Figure 5 : The original signal y(t) has been plotted in solid line. Without pre-processing of y(t), the acoustical signal $y_h(t)$ is obtained (dashed line). With pre-processing, the acoustical signal $y_{th}(t)$ which tends towards y(t) is obtained (dotted line).

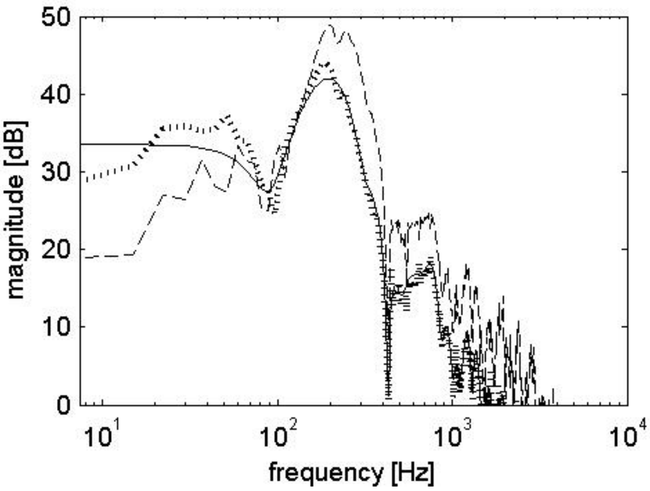


Figure 6 : The spectrum of the reference signal y(t) has been plotted in solid line. Without pre-processing (dotted line), the spectrum of $y_h(t)$ shows us the filtering effects of the reproducing system. Above 100 Hz, the system increases the spectrum of y(t) in magnitude, while under 100 Hz the spectrum is decreased. With linear pre-processing, the frequency-range up to 100 Hz has been perfectly corrected. In the low-frequency range, only an approximated equalization has been made.

Psychoacoustics Validation

The results of a psychoacoustics test enhance the validity of this pre-processing method. First, the high-fidelity reproduction of sonic-boom signatures allows us to actually listen to sounds of explosions whereas without pre-processing the sensation mainly corresponds to rough and distorted sounds. Secondly, we are now able to search which characteristic parameters of the waveform are perceived and mainly take part in the felt annoyance of sonic-booms. It should be noted that this work investigates only the audible part of sonic-booms at a relative acoustical pressure level. For longer signal reproduction, as the spectrum is shifted to the very low frequency range, only the remaining noise of the signal can be subjectively studied.

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