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Measuring the sound absorption properties of noise barriers with inverse filtered maximum length sequences

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ABSTRACT

Loudspeaker characteristics can have an appreciable influence on the sound absorption properties obtained with CEN/TS 1793-5 [1]. Although sound sources with omnidirectional radiation properties are favored, these often hold the problem of long impulse responses due to their design potentially incorporating ports [2]. In this paper, the inverse filtering approach is applied to two different sound sources. It is shown that it is a valuable measure to reduce the influence of the loudspeaker characteristics on the obtained sound absorption values.

Furthermore, an attempt is made to measure the absorption properties according to CEN/TS 1793-5 without the need for the subtraction procedure of a free-field measurement. Thereby, as long as only high frequencies are considered, reasonable values can be obtained.

1. Introduction

For the non-destructive determination of the sound absorption properties of building elements such as noise barriers or road surfaces, impulse response measurement techniques [3–9] are a frequently used measure to separate the direct and reflected components of a measured impulse response. These measurement techniques have strict requirements for the sound source used, as the impulse response of the direct component should be as short as possible to permit a good separation from the reflected component in the time domain. Moreover, the directivity of the sound source can have an influence on the measured sound absorption properties for measurements under oblique incidence, especially for high frequencies [10–12]. Therefore, a sound source with omnidirectional radiation properties and a short impulse response is needed. This is a demand which is difficult to fulfill.

In this paper, the approach presented by Cobo et al. [13] for the inverse filtering of Maximum Length Sequences (MLS) is used to improve measurements of the sound absorption properties according to CEN/TS 1793-5 [1] by making use of an omnidirectional sound source. For obtaining the impulse response, the MLS technique is used as suggested by CEN/TS 1793-5, the principle is however also applicable to other impulse response measurement methods (e.g. sine sweep, logarithmic sine sweep, etc.). In a second step, the obtained impulse responses are used to measure the sound absorption properties of a test wall without the need of performing an additional free-field measurement.

2. Influence of the sound source

Measurements were performed using two different sound sources, one a JBL 2123H speaker in a self-built cabinet, the other a B&K OmniSource 4295 [2]. These two loudspeakers were chosen due to their different sound radiation directivity properties (Fig. 1). It has to be stressed that the B&K OmniSource 4295 does not comply with all performance requirements as described in CEN/TS 1793-5 because of its long impulse response function, as will be shown later.

The different directivity properties of the sound sources are mainly due to their different cabinet design. For practical reasons, the JBL 2123H loudspeaker is housed in a simple box made of chip board, whereas the B&K OmniSource 4295 is specifically designed to possess omnidirectional radiation properties [2]. Part of this special design is a cone mounted in front of the loudspeaker membrane with a length of approx. 35 cm and an aperture diameter of approx. 4 cm. Whereas this cone is one of the main reasons for the omni-directionality of the sound source, its impulse response function is severely influenced by this design resulting in an IR with a length of over 7 ms (Fig. 2).

When analyzing the reason for this long impulse response, a contour plot of the Fourier transform of the free-field impulse response is examined. Fig. 3 shows the frequency response, Fig. 4 a spectrogram of the impulse response functions of the two

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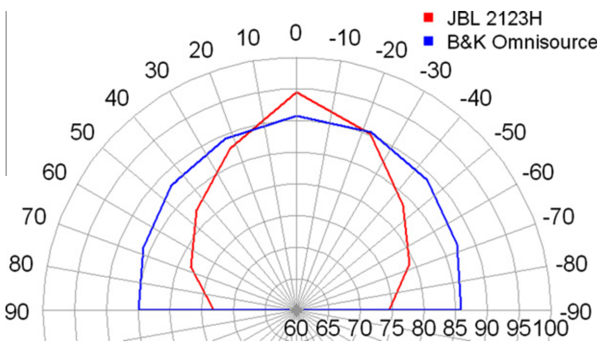

Fig. 1. Sound source directivity – overall sound pressure levels. More detailed data can be found in [2].

loudspeakers. In Fig. 5 the frequency is plotted against a function of the time window length applied to the impulse response. The time window here consists of a left-sided Blackman window with a length of 0.15 ms, a section with value 1 of variable length (which is varied on the ordinate), and a right-sided Blackman window of length 0.15 ms. One can clearly see the arrival of the impulse response after approx. 3.6 ms, which corresponds to a distance between loudspeaker membrane and microphone of 1.25 m at an ambient air temperature of 25 °C. After another ~2.1 ms, strong rippling starts in the signal for the B&K OmniSource 4295. These 2.1 ms correspond to another 0.72 m of sound propagation, which is the double length of the cone mounted in front of the loudspeaker membrane. As can be seen from this, the primary impulse travels towards the front of the cone, is reflected there and starts to build a standing wave inside the cone according to the acoustic transmission line equations for a half-sided open tube. Also, after approx. 13 ms another strong rippling effect starts due to the ground reflection entering the time window.

This visibility of periodic ripples is also an advantage of plotting the spectrum versus cumulative window length over a standard spectrogram. Where a spectrogram only shows additional energy arising in the signal, this periodic structure of the ripples is a clear sign of an occurring echo. As can be seen from Fig. 5, at 12 resp. 13 ms such rippling starts indicating the arrival of an echo at the microphone. In this specific case, these are the components reflected by the ground (annotation: the tripods of the two loudspeaker-microphone assemblies differ in height of approx. 0.1 m thus resulting in slightly different travel times of the signal to the ground and back to the microphone).

A second way to determine the delay of these components is to examine the frequencies of the occurring ripples. As a signal containing a pure echo:

$$y(t) = x(t) + ax(t+\tau) \tag{1}$$

results in a frequency distribution

$$|y(f)| = \sqrt{1 + a^2 + 2a\cos(2\pi f\tau)}|X(f)| \tag{2}$$

with τ the time delay between the arrival of the direct and the reflected signal, the analysis of the occurring local maxima in the frequency distribution also leads to this τ. When analyzing the local maxima of these ripples for the B&K OmniSource 4295 in Fig. 5 that establish after approx. 5.7 ms, a periodic structure can be seen indicating such a pure echo. Taking the maxima from the slice of the contour plot at 5.7 ms and between 1.5 kHz and 5.5 kHz into account, one obtains a τ of approx. 2.26 ms and therefore a spatial delay of 0.77 m, which again corresponds to the double length of the cone in front of the loudspeaker membrane.

2.1. Measurement setup

In order to investigate the influence of the sound source on the measured sound absorption properties, a simplified experimental setup based on CEN/TS 1793-5 was used. A plane concrete wall was chosen as test sample in a large laboratory hall with dimensions exceeding 5 × 5 m, providing assumed sound absorption properties of $\alpha \approx 0$ over the frequency range of interest (100 Hz–5 kHz).

For clarity reasons, and also with respect to the shortest time-difference between the arrival of the direct and reflected part of the impulse response in this configuration, only the central microphone position (No. 5) is presented. In this setup, the microphone is mounted 0.25 m, the loudspeaker 1.5 m away from the test wall and aligned so as to produce normal sound incidence.

For this configuration, the reflectivity τ of the test wall is calculated for both sound sources according to:

$$r = \frac{|F((h_{reflected}(t) - h_{free-field}(t)) \cdot w_{reflected}(t) \cdot t)|^2}{|F(h_{free-field(t)} \cdot w_{free-field}(t) \cdot t)|^2} \tag{3}$$

where h denotes the impulse responses for the free-field and reflection measurements, w the Adrienne window, t the multiplication with the time elapsed since emission of the impulse as a propagation correction of the wave, and F the symbol of the Fourier transform. This propagation correction implies the assumption of spherical wave propagation. As t is interpreted as a variable in (3),

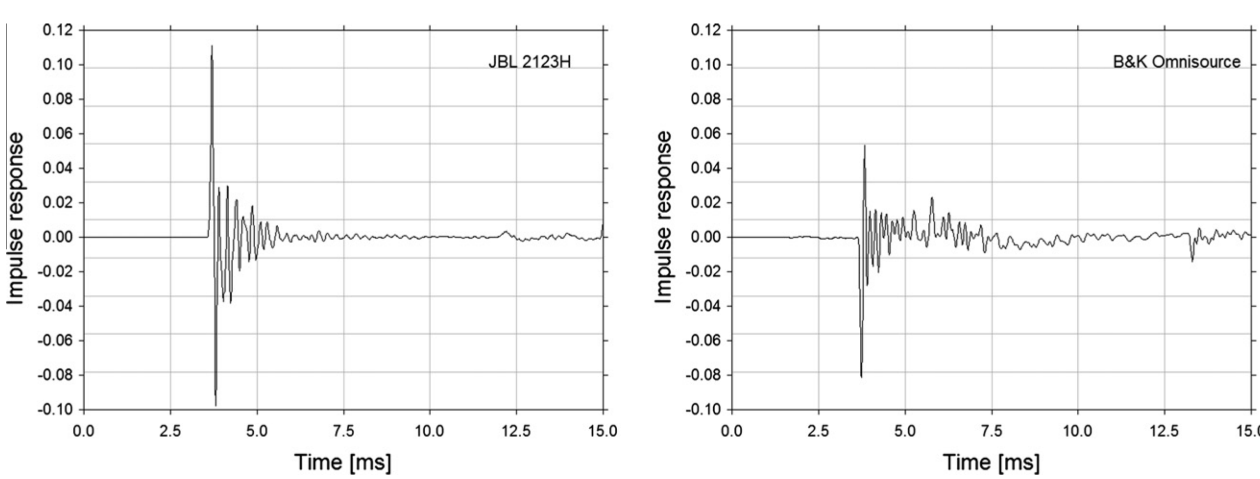

Fig. 2. Impulse response of both sound sources: the peaks after 12.5 ms are ground reflections.