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Self-calibrating method for sound reflection index measurements

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A self-calibrating method for determining *in situ* the sound reflection index of materials in general, and sound barriers in particular, is proposed. The method exploits the availability of direct sound information in the transfer function, avoiding the need of a free field measurement. Effects on the performance of the approach of differences in acoustic shadowing between the direct and reflected waves, of geometrical inaccuracies, of loudspeaker directivity, and of numerical windowing effects on the spectrum of the reflection coefficient, are addressed.

1 Introduction

Measurement methods for obtaining the sound absorption properties of materials and surfaces of structures such as sound barriers can be divided into two groups: (1) laboratory measurement methods and (2) *in situ* measurement techniques. Probably the best known laboratory method for material absorption characterization is the one-dimensional impedance tube (or 'Kundt tube') method where the surface impedance is obtained under normal incident angles by positioning sound pressure maxima and minima in a standing wave (Kuttruf 2000). This principle has been explored and used in improvements of "impedance tube methods" and can be found in different variations, e.g. the methods proposed by Dickinson (1970), Seybert and Ross (1977), Chung (1991), Jones (1997), Fahy (1984), Dunlop (1986) and others. Several of these methods have been also accepted in legal standards, such as ISO 10534 or ISO 13472. Tube methods have many advantages. They only require relatively inexpensive instrumentation and only put modest demands on laboratory space. However, they are only appropriate for samples with limited size and their use is limited to the determination of sound absorption properties of porous materials.

A second approach among laboratory methods is the diffuse field method, which is designed to determine the random incident absorption coefficient. In this method, which is typically performed in reverberant room, the reverberation time of the room is determined before and after the placement of a sample. The method allows for the reliable determination of absorption properties of larger samples. The limitations of this method are related to imperfections of the required diffuse field, caused by eigenmodes at low frequencies due to the room confinement. Also sample edge effects contribute to the uncertainty on such measurements, by making the apparent sound absorption higher than in reality. (Kosten 1960, Kuhl 1983, Fuchs, 2000).

The class of so-called free-field measurement methods is quite popular, not only for use in a laboratory but also *in situ*. In this approach, the impulse response of a reflecting surface (e.g. of a wall structure, a material under test, or a sound barrier) is measured by transmitting sound from loudspeaker in front of the surface, and monitoring of the sound waves by a microphone in between the loudspeaker and the surface. The impulse response then contains, in order of arrival, the direct sound wave, the wall reflection, the floor reflection, and later reflections from objects in the neighbourhood. Some of the used approaches assume sound waves as plane waves; others consider a spherical wave approximation. Well known are techniques that use a wave field analysis. 'Direct' methods typically work with plane wave models or the mirror source models and allow calculation of different acoustic quantities or transfer functions from measurements performed in front of the sample (Barry 1974, Kurze, 1968, Cops and Lauriks, 1985, Allard 1985), intensity methods (Suzuki 1996) and acoustical holography (Boeckx 2003). 'Indirect' or 'inverse' methods are based on measurement of phase gradients (Leguis and Nicolas 1987) or level differences (Sabatier 1993).

Finally, a large part of free field methods is based on windowing and separation of the incident and reflected wave in time domain. Among these, there are tone burst methods (Cops and Myncke, 1973), broadband deconvolution methods using deterministic broadband signals (Garai 1993), methods for shortening loudspeaker's impulse responses, so called "impulse sharpening techniques (Wilms, 1991), and others.

An alternative way to select the reflected wave contribution from a wall reflection impulse response is done by subtracting the direct sound, which has been determined in a separate free field measurement. Several researchers have been working on the improvement of this method. Mommertz (1998) did an extensive study on measurements of inhomogeneous surfaces, perforated ceilings including the uncertainty of measurement due to loudspeaker directivity and influence of spherical wave fronts that might cause in grazing incidence angles negative absorption coefficients. Improvements of the method for low frequencies down to 100Hz ware achieved by Karjalainen and Tikander (2001)

In relation to practical applications and measurements in situ, such as porous road surfaces and inhomogeneous or non-flat noise barriers, a large amount of work has been conducted by Garai 1998, Clairbois 1998, Anfosso-Ledee 2000, Berengier 2002, Massarani 2003, De Geetere 2004.

In the period 1995-1997 the so-called 'Adrienne method' for determining sound absorption properties of noise barriers was developed in the framework of a EU research project (EU project Adriene 1998). In this method, which is averaging the sound reflection index over a range of angles, a microphone is connected to the loudspeaker in order to maintain the absolute distance between the source and microphone the same for all incident angles, which is very convenient for the precise subtraction of the direct wave arrival determined in a separate free-field measurement, from the wall reflection impulse response.

In the Adrienne method the lowest assessed frequency is 100Hz for normal incidence, and up to 500 Hz as the angle of incidence is increased till 40degrees. Reflection index (RI) is calculated for third octave bands 200-5000 Hz. The Adrienne approach also proposes a single number rating of the overall sound absorption by parameter the DL_R (CEN/TS 1793-5).

Ongoing research efforts in relation to the improvement of measurement methods of sound absorption and sound transmission of noise barriers are performed in the framework of the European research project Quiesst (EU project Quiesst). A new method under test proposes a sound reflection determination based on a simultaneous and therefore very fast measurement with 3x3 microphones placed at the square mesh of 80x80 cm with 40cm interspacing (Fig.1). In this method the loudspeaker is not anymore connected with one microphone like in Adrienne method. As a result, the relative positioning between the microphone mesh and the loudspeaker (distance and grid orientation) during the measurement of the wall reflection becomes more complicated. In particular the subtraction of the direct sound from the measured impulse response is not so straightforward as in Adrienne method, since the repeatability of the relative positioning between the wall reflection measurement and the free-field measurement takes a lot of care and iteration. On the other hand, due to the freedom to perform free-field measurements at different loudspeaker-grid distances, the new method allows in principle to calibrate to a precise degree geometrical diffraction and loudspeaker directivity effects.

In this article, we discuss in detail some issues related to the new Quiesst method. We propose a pragmatic self-calibrating approach for determining the sound reflection index by exploiting to a maximum extent the direct sound information in the impulse response of the wall reflection measurement, thus avoiding the need for maintaining the loudspeaker amplification and microphone sensitivity settings between the free field measurement, and the wall reflection measurement, which happen at different times.

2 Self-calibrating approach for loudspeaker amplification and microphone sensitivity settings

The pressure reflection spectrum of a sound barrier is defined as the ratio between the spectra of the reflected and incoming sound. In the procedure under evaluation in the Quiesst project, the incoming sound is measured doing a separate free field (FF) measurement far away from any obstacle (or in a (semi) anechoic room) with the microphone grid centre at 125 cm from the loudspeaker membrane centre, the same distance as the one used in the measurement with the sound barrier (SB) of interest at 150 cm from the loudspeaker membrane centre (‘ the SB measurement’), and thus at 25 cm from the microphone grid centre, the microphone grid aligned parallel with the plane of the wall, and perpendicular to the normal to the loudspeaker membrane. The measured FF and SB impulse response signals can be written as Eq. 1:

$$S_{FF,j}(t) = A_{LS}\left(S_{LS,125,\alpha_j}(t)\otimes M_{\alpha_j}(t)\right)$$

$$S_{SB,j}(t) = A'_{LS}\left(S_{LS,125,\alpha_j}(t)\otimes M'_{\alpha_j}(t)\right)+\ldots$$

$$A'_{LS}\left(S_{LS,175,\beta_j}(t)\otimes R_{\beta_j}(t)\otimes M'_{\beta_j}(t)\right)$$

where $\otimes$ is the convolution operator, j refers to the index (1 to 9, counted from top to bottom and from left to right) of the respective microphones and their position with respect to the loudspeaker, $S_{FF,j}(t)$ is the measured FF signal, $S_{SB,j}(t)$ is the measured SB signal, A_{LS} [m.s^{-1}.V^{-1}] is the voltage to membrane velocity conversion factor of the loudspeaker, $S_{LS,dj,\alpha j}(t)$ [Pa.m^{-1}.s] is the response function of the loudspeaker, which is depending on the angle (α_j [deg]) between the arriving wave vector and the normal to the grid and wall, and on the detection distance (d_j [m]), due to geometrical divergence and to the not isotropic loudspeaker directivity. $M_{\alpha j}(t)$ [V.Pa^{-1}] and $M_{\beta j}(t)$ [V.Pa^{-1}] are the impulse responses of the microphone positioned under an angle α_j [deg] and β_j[deg] respectively, both angles taken with respect to the normal to the wall. $R_{\beta j}(t)$ [Pa.Pa^{-1}] is the dimensionless pressure reflection impulse response of the sound barrier for an angle of incidence β_j.

The hyphened symbols can be different from the not-hyphened ones, due to the possibility that the analogue or digital settings for loudspeaker amplification and the microphone sensitivities during the SB measurement might not be exactly the same as the ones during the FF measurement.

By Fourier transformation to frequency domain, the relations above can be transformed to Eq. 2:

$$S_{FF,j}(f) = A_{LS}\left(S_{LS,125,\alpha_j}(f)\ M_{\alpha_j}(f)\right)$$

$$S_{SB,j}(f) = A'_{LS}\left(S_{LS,125,\alpha_j}(f)\ M'_{\alpha_j}(f)\right)+\ldots$$

$$A'_{LS}\left(S_{LS,175,\beta_j}(f)\ R_{\beta_j}(f)\ M'_{\beta_j}(f)\right)$$

with f [Hz] the frequency.

Provided a suitable orientation of the microphones, the microphone responses are to a good approximation independent on the angle of incidence, and of the frequency, so that the following simplification can be made in Eq. 3:

$$M_{\alpha_j}(f) = M_{\beta_j}(f) \equiv M_j$$

with M_j a frequency and angle independent sensitivity factor.

The reflection index can then be obtained by windowing in time domain the reflected sound (Eq.4):

$$S_{SB,j}(t) = A'_{LS}\left(S_{LS,125,\alpha_j}(t)\otimes M'_{\alpha_j}(t)\right)+\ldots$$

$$A'_{LS}\left(S_{LS,175,\beta_j}(t)\otimes R_{\beta_j}(t)\otimes M'_{\beta_j}(t)\right)$$

↓ windowing around reflected wave arrival

$$S_{SB,R,j}(t) = A'_{LS}\left(S_{LS,175,\beta_j}(t)\otimes R_{\beta_j}(t)\otimes M'_{\beta_j}(t)\right)$$

Applying the above simplification, Fourier transforming this signal to frequency domain, and normalizing the result with the FF spectrum, we obtain (Eq. 5):

$$\left.\begin{aligned} S_{FF,j}(f) &= A_{LS}\left(S_{LS,125,\alpha_j}(f)\ M_j\right) \\ S_{SB,R,j}(f) &= A'_{LS}\left(S_{LS,175,\beta_j}(f)\ R_{\beta_j}(f)\ M'_j\right) \end{aligned}\right\}$$

$$\Rightarrow \frac{S_{SB,R,j}(f)}{S_{FF,j}(f)} = \frac{A'_{LS}\left(S_{LS,175,\beta_j}(f)\ R_{\beta_j}(f)\ M'_j\right)}{A_{LS}\left(S_{LS,125,\alpha_j}(f)\ M_j\right)} = \frac{A'_{LS}}{A_{LS}}\frac{M'_j}{M_j}\frac{S_{LS,175,\beta_j}(f)}{S_{LS,125,\alpha_j}(f)}R_{\beta_j}(f)$$

As a consequence, the pressure reflection spectrum can be written as (Eq. 6):

$$R_{\alpha_j,\beta_j}(f) = \left(\frac{A'_{LS}}{A_{LS}}\frac{M'_j}{M_j}\frac{S'_{LS,175,\beta_j}(f)}{S_{LS,125,\alpha_j}(f)}\right)^{-1}\left(\frac{S_{SB,R,j}(f)}{S_{FF,j}(f)}\right)^{-1}$$

$$\equiv \left(\phi_j\frac{S'_{LS,175,\beta_j}(f)}{S_{LS,125,\alpha_j}(f)}\right)^{-1}\left(\frac{S_{SB,R,j}(f)}{S_{FF,j}(f)}\right)^{-1}$$

For obtaining the directivity related factor

$$\frac{S'_{LS,175,\beta_j}(f)}{S_{LS,125,\alpha_j}(f)}$$

from two free-field measurements with the microphone grid at respectively 125 cm and 175 cm from the loudspeaker we refer to the procedure proposed in the Quiesst project. Here we focus on the determination of the (frequency and angle independent) loudspeaker amplification and microphone sensitivity related factor

$$\phi_j = \frac{A'_{LS}}{A_{LS}}\frac{M'_j}{M_j}$$

Due to manipulations with the equipment, or due to wanted or unwanted changes in analogue or digital electronic settings, it is sometimes difficult to maintain or accurately assess changes in the amplification of the loudspeaker and in the sensitivity of eh individual microphones when changing the measurement configuration from FF to SB or *vice versa*. As a result, the factors ϕ_j can differ from unity. If large differences are encountered, then this can be seen as a warning that some measurement system settings have substantially changed, and it is recommendable to double check the measurements. However, for small deviations, one can recover the ϕ_j values as follows.

One selects the direct sound from the SB signal as follows (Eq. 9)

$$S_{SB,j}(t) = A'_{LS}\left(S_{LS,125,\alpha_j}(t)\otimes M'_{\alpha_j}(t)\right)+\ldots$$
$$A'_{LS}\left(S_{LS,175,\beta_j}(t)\otimes R_{\beta_j}(t)\otimes M'_{\beta_j}(t)\right)$$

$\downarrow$ windowing around direct wave arrival

$$S_{SB,D,j}(t) = A'_{LS}\left(S_{LS,125,\alpha_j}(t)\otimes M'_{\alpha_j}(t)\right)$$

This windowed signal is the same as the one which is used in the subtraction procedure. Here we use it as follows. First we consider the signal in frequency domain, and make the reasonable assumption that the microphone response is to a good approximation angle and frequency independent (Eq. 10):

$$S_{SB,D,j}(f) = A'_{LS}\,S_{LS,125,\alpha_j}(f)M'_j$$
$$S_{FF,j}(f) = A_{LS}S_{LS,125,\alpha_j}(f)M_j$$

It is now obvious that the factor ϕ_j can be obtained by taking the ratio between the spectrum obtained from windowing the direct signal from the SB signal, $S_{FF,j}(f)$, and the spectrum of the FF signal, $S_{SB,D,j}(f)$ (Eq. 11):

$$\frac{S_{SB,D,j}(f)}{S_{FF,j}(f)} = \frac{A'_{LS}\,S_{LS,125,\alpha_j}(f)M'_j}{A_{LS}S_{LS,125,\alpha_j}(f)M_j} = \frac{A'_{LS}}{A_{LS}}\frac{M'_j}{M_j} = \phi_j$$

According to the reasoning above, the value for ϕ_j as obtained from the ratio of the two spectra as described above should be frequency independent. Indeed, deviations of ϕ_j from unity can be typically described to deviations in the global loudspeaker amplification or microphone sensitivity settings. On the other hand, due to uncertainties in the measurements, e.g. related to signal windowing effects, affecting mainly low frequencies, or to effects of microphone shadowing, affecting mainly high frequencies, some deviations from flat behaviour can be expected at those low and high frequencies, while negligible in the frequency range between 500Hz and 2000Hz. This is why we propose to derive ϕ_j (for every microphone j) as follows from numerical spectra for the SB-D and FF data (Eq. 12):

$$\phi_j = \frac{\sum_{f_n=500-2000Hz}\left|S_{SB,D,j}(f_n)\right|}{\sum_{f_n=500-2000Hz}\left|S_{FF,j}(f_n)\right|}$$

with f_n the linearly spaced measurement frequencies .

The thus obtained value for ϕ_j can then be inserted in Eq. 6 to obtain an 'amplification and sensitivity corrected' reflection coefficient. The algorithm is schematically depicted in Figure 1.

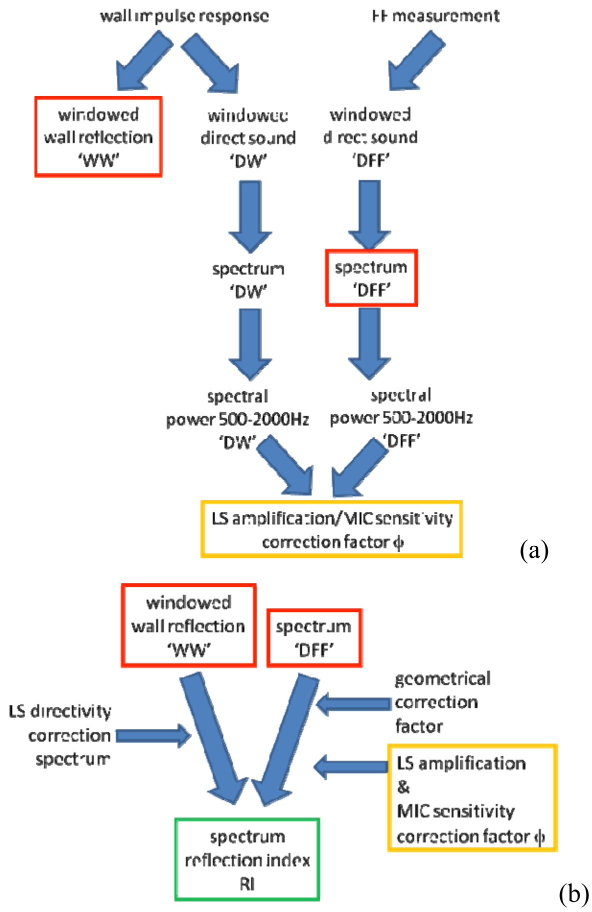

Figure 1: (a) Scheme of data processing in order to obtain the loudspeaker (LS) amplification factor/microphone (MIC) sensitivity factor ϕ from the windowed and filtered wall reflection signal and the free-field signal. (b) Together with the geometrical correction factor, which takes into account the orientation and coordinates of the microphone with respect to the loudspeaker, the factor ϕ can be used to extract the reflection index RI from the spectrum of the windowed wall reflection signal (WW) and the free-field reference measurement at 175 cm (DFF).

For the sake of completeness, we mention that in the above, 'Adrienne' window functions were used with equal length 0.7 ms for the FF and direct sound (sufficiently short to totally exclude contributions from reflected wave), and the classical length 5.18 ms for the wall reflections.

It is worth to note that a measure for the average amplitude ratio between the direct sound wave in the signal $S_{SB,j}(t)$ obtained in the configuration with the wall, and the free-field signal $S_{FF,j}(t)$, also becomes available during the subtraction procedure, during which the optimum subtraction of the direct signal contribution from the wall measurement impulse response is found by relatively time shifting the two signals and optimizing their amplitude ratio until their difference in a time window around the direct wave arrival becomes minimum. The thus obtained amplitude ratio is indeed a measure for ϕ_j. We have found that ϕ_j values obtained from Eq. 12 on one hand, and from the subtraction optimization on the other hand, are consistent with fluctuations of about 2%. Both approaches appear thus can be considered as equivalent.

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