Progress in sound reflection measurements on noise barriers in situ

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In situ measuring sound reflection from noise barriers is a very difficult task; it is addressed in the European technical specification CEN/TS 1793-5, but some open problems still remain. In the frame of the EU funded QUIESST project, working package 3, several new improvements have been introduced to get a better and more robust measurement method. They include: using a square 9-microphone array not rigidly connected to the loudspeaker, multichannel acquisition, optimized alignment of free-field and global impulse responses including fractional step shifts and least squares estimation of the best relative position, corrections for geometrical divergence and sound source directivity. Overall, these improvements led to a new definition of the reflection index. In this paper the essential of these improvements is presented and some exemplary results are shown.

1 INTRODUCTION

It is important to know the sound absorbing performance of noise barriers and claddings installed alongside roads and railways. Laboratory measurements of sound absorption are done in reverberant conditions which don't correspond to the vast majority of real situations, where the sound field is not diffuse. Also, there is the need to check noise reducing devices after their installation, i.e. *in situ*. For this purpose, the former ADRIENNE project developed a method for measuring sound reflection from noise barriers or claddings *in situ*, standardized in the European technical specification CEN/TS 1793-5[1]; for the first time the very difficult task of measuring sound reflection in situ became possible[2], but some open problems still remain: the low frequency limit of the measurement is often greater than 100 Hz when the rotating microphone-loudspeaker assembly is turned in the lowest positions[1,2], the proposed correction for geometrical

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divergence is quite unusual, the effect of sound source directivity is not taken into account. Therefore in the frame of the QUIESST[3,4,5] project (2010-2012), working package 3, a completely revised test method, applicable to flat and non-flat products, has been defined. Results are expressed as a function of frequency, in the one-third octave bands between 100 Hz and 5 kHz[1].

2 GENERAL PRINCIPLE

The single microphone of CEN/TS 1793-5 is substituted with nine microphones arranged in a 3x3 square grid (0,80 x 0,80 m); multichannel acquisition can be exploited. The array is placed between the sound source (loudspeaker) and the device under test. The sound source emits a transient sound wave that travels past the microphone array position to the device under test and is then reflected on it (see figure 1). The microphones receive both the direct sound travelling from the sound source to the device under test and the reflected sound (including scattering). A free-field measurement, taken with the same source and microphone configuration but far from any reflecting object, is then subtracted from the previous one to isolate the reflected component. The ratio of the power spectra of the direct and the reflected components, gives the basis for calculating the sound reflection index, averaged on the nine microphones:

$$RI_j = \frac{1}{n_j} \sum_{k=1}^{n_j} \left[\frac{\int_{\Delta f_j} \left| F\left[h_{r,k}(t) \cdot w_{r,k}(t) \right] \right|^2 df}{\int_{\Delta f_j} \left| F\left[h_{i,k}(t) \cdot w_{i,k}(t) \right] \right|^2 df} \cdot C_{geo,k} \cdot C_{dir,k}\left(\Delta f_j\right) \cdot C_{gain,k}\left(\Delta f_g\right) \right], \tag{1}$$

where:

- $h_{i,k}(t)$ is the incident reference component of the free-field impulse response at the k-th measurement point (microphone);
- $h_{r,k}(t)$ is the reflected component of the impulse response taken in front of the sample under test at the k-th measurement point (microphone) after the signal subtraction;
- $w_{i,k}(t)$ is the time window (Adrienne shape[1]) for the incident reference component of the free-field impulse response at the k-th measurement point (microphone);
- $w_{r,k}(t)$ is the time window (Adrienne shape[1]) for the reflected component at the k-th measurement point (microphone);
- F is the symbol of the Fourier transform;
- j is the index of the one-third octave frequency bands (between 100 Hz and 5 kHz);
- Δf_j is the width of the j-th one-third octave frequency band;
- k is the microphone number according to figure 1 (k = 1, ..., 9);
- n_j is the number of microphone positions on which to average ($n_j \geq 6$).

In this formulation three newly defined corrective factors are included, $C_{geo,k}$, $C_{dir,k}(\Delta f_j)$ and $C_{gain,k}(\Delta f_g)$, to be discussed later on (see point 2.3).

2.1 The Improved Low Frequency Limit

In order to get valid measurement results down to the 100 Hz one-third frequency band when measuring on noise reducing devices having a height ≥ 4 m, the Adrienne temporal window is used as follows:

- with a total length of 7,9 ms to process the impulse responses coming from microphones 1 to 6; the *RI* values obtained over the six microphones shall be averaged to get the final *RI* values in the one-third frequency bands having centre frequency 100 Hz, 125 Hz and 160 Hz;
- with a total length of 6,0 ms to process the impulse responses coming from microphones 1 to 9; the RI values obtained over the nine microphones shall be averaged to get the final *RI* values in the one-third frequency bands having centre frequency from 200 Hz to 5 kHz.

These specifications, together with the prescribed distances (loudspeaker and microphone array height equal to half the barrier height; loudspeaker to barrier (reference plane) distance 1,50 m; microphones to barrier (reference plane) distance 0,25 m) guarantee that ground reflections are excluded from the analysis window of the impulse response component reflected on the device under test.

When the device under test doesn't have the minimum dimensions for the results be valid on the full frequency range (height < 4 m), the Adrienne temporal window shall have a reduced length so as to exclude ground reflections from the reflected component of the impulse responses; different window lengths for microphones 1 to 3, microphones 4 to 6 and microphones 1 to 9 may be used.

2.2 The Improved Signal Subtraction Technique

In principle, the signal subtraction technique requires the loudspeaker and microphones relative position be kept constant in order to get a perfect alignment between the impulse responses measured in front of the device under test and in the free field for the same microphone. This may be very difficult when on site, due to placement of the equipment on an irregular terrain, small movements of the loudspeaker cone or the microphones when displacing the equipment, variations in the response of the measuring equipment due to temperature or electrical deviations occurring between the free field and the reflected measurements, etc. Therefore it is necessary that, before performing the signal subtraction, the free field signal is corrected for a small shift relative to the impulse response in front of the device under test at each microphone. Since in general the actual time shift is not equal to a multiple of the temporal sample size Δt, step wise shifting of one or more data points is inadequate.

An accurate alignment can be done as follows (see also Ref. [7]); it allows the placement of the microphone array without a rigid connection to the loudspeaker; the unavoidable misalignments between the impulse responses measured in front of the device under test and in the free field for the same microphone may be compensated until then they are ≤ 5 cm.

1. For each microphone position, an impulse response measured in front of the device under test and one measured in the free field with nominally the same geometry are compared.
2. The free field impulse response is repeatedly shifted with a small "moving step" $\Delta\tau$ (which is a fraction of the temporal step Δt between the discrete points of the acquired data, see below).
3. The sum of the squared differences between the free field impulse response and the impulse response measured in front of the device under test is calculated in a limited interval around the first and main peak of the impulse response measured in front of the device under test.

4. The operations in 2 and 3 are repeated until the minimum of the sum in 3 is found (least squares); the number n of moving steps $\Delta\tau$ needed to get this least square minimum is recorded.
5. The free field impulse response is finally shifted with the temporal step $n\Delta\tau$ found in 4 and its amplitude is adjusted so that the amplitude of its first (and main) peak is exactly the same of the first (and main) peak of the impulse response measured in front of the device under test.
6. The shifted and amplitude adjusted free field impulse response is subtracted from the impulse response measured in front of the device under test.

The shifted and amplitude adjusted free field impulse response used in step 6 above is discarded after the subtraction; the free field impulse response used to calculate the reflection index according to Eqn. (1) is the original, unchanged one.

In order to shift the free field impulse response in n moving steps, $n\Delta\tau$, with $\Delta\tau$ considerably smaller than the temporal step Δt between the discrete points of the acquired data, the following procedure is applied.

a. The free field impulse response is Fourier transformed in the frequency domain and its phase is changed by multiplying it with a frequency dependent factor $exp(i2\pi fn\Delta\tau)$.

b. The resulting phase corrected Fourier transform is inverse transformed to generate the shifted free field impulse response in the time domain which then can be used for signal subtraction.

Figure 2 shows an example: the direct component is quite completely cancelled.

As the goal of the operation is to remove the incident component of the impulse response (the "direct sound"), leaving only the reflected one, the signal subtraction effectiveness can be measured by the decibel level reduction in the incident component from the measurement to the result of the signal subtraction. Specifically, following Robinson and Xiang[6], the sum of the energy within 0,5 ms of either side of the first and main peak of direct sound can be compared before and after subtraction to find the effective reduction. This defines the reduction factor R_{sub}:

$$R_{sub} = 10\lg\left[\frac{\displaystyle\int_{t_{p,k}-0,5ms}^{t_{p,k}+0,5ms} \left|h_{i,k,FF}(t)\right|^2 dt}{\displaystyle\int_{t_{p,k}-0,5ms}^{t_{p,k}+0,5ms} \left|h_{i,k,RES}(t)\right|^2 dt}\right] \text{dB}, \qquad (2)$$

where:

$h_{i,k,FF}(t)$ is the incident reference component of the free-field impulse response at the k-th measurement point ***as measured***;

$h_{i,k,RES}(t)$ is the residual incident component of the impulse response taken in front of the sample under test at the k-th measurement point ***after the signal subtraction***;

$t_{p,k}$ is the time instant where the first peak of the incident component of the impulse response at the k-th measurement point is located (before the signal subtraction);

the other symbols are as previously defined.

A reduction factor R_{sub} equal to the peak to noise ratio of the measurement can be considered a complete subtraction, since this would leave nothing in the area of the direct sound except the background noise.

2.3 The New Correction Factors

$C_{geo,k}$ is a geometrical divergence correction factor taking into account the path length difference between the direct and reflected waves at the k-th measurement point:

$$C_{geo,k} = \left(\frac{d_{r,k}}{d_{i,k}} \right)^2 , \tag{3}$$

where $d_{i,k}$ is the distance from the front panel of the loudspeaker to the k-th microphone and $d_{r,k}$ is the distance from the front panel of the loudspeaker to the device under test (reference plane) and back to the k-th microphone following specular reflection. It is worth noting that for non flat complex devices it is difficult to predict the exact travel path of each wave, considering also non specular scattering; therefore the geometrical divergence correction factors are calculated on the basis of specular reflection on an ideal flat reflecting surface.

$C_{dir,k}(\Delta f_j)$ is a correction factor used to compensate the difference of sound source directivity, at the k-th measurement point, due to the different incidence angles of direct and reflected waves on the microphones. In principle this factor must be measured only once for a given sound source and makes the measurements independent from the particular sound source used. It is given by:

$$C_{dir,k}(\Delta f_j) = \frac{\int_{\Delta f_j} \left| F\left[h_{i,k}(t,\alpha_k) \cdot w_{i,k}(t) \right] \right|^2 df}{\int_{\Delta f_j} \left| F\left[h_{i,k}(t,\beta_k) \cdot w_{i,k}(t) \right] \right|^2 df} , \tag{4}$$

where:

- α_k is the angle between the line connecting the centre of the front panel of the loudspeaker to microphone 5 and the line connecting the centre of the front panel of the loudspeaker to microphone k (see figure 3);
- β_k is the angle between the line connecting the centre of the front panel of the loudspeaker to microphone 5 and the line connecting the centre of the front panel of the loudspeaker to the specular reflection path to microphone k (see figure 3);
- $h_{i,k}(t,\alpha_k)$ is the incident reference component of the free-field impulse response at the k-th measurement point;
- $h_{i,k}(t,\beta_k)$ is the incident reference component of the free-field impulse response at a point laying on the specular reflection path for microphone k and at distance $d_{i,k}$ from the centre of the front panel of the loudspeaker;

the other symbols are as previously defined.

Figure 4 shows the correction factors measured for a Zircon loudspeaker.

$C_{gain,k}(\Delta f_g)$ is a correction factor used to compensate a gain mismatch (if any) of the amplification settings between the “free-field” and “barrier” measuring equipment. This factor can also be used as a validation criterion to reveal an unwanted change in the relative distance between the sound source and the microphone grid. It is defined as the amplitude ratio of the spectra of the “barrier” and “free-field” impulse responses anechoic parts.

3 SOME EXEMPLARY RESULTS

The above outlined method has been verified by 8 independent laboratories on 13 samples installed on 2 test sites in Grenoble (France) and Valladolid (Spain). Overall, the test has been conducted following the procedure for an inter-laboratory test in order to be able to get the repeatability and reproducibility of the method. At the time of writing the measurement values are under statistical analysis; the results will be available after the conclusion of the QUIESST project in November 2012; it can be anticipated that the repeatability should be quite good for an in situ method[8]. At the time of writing some measurement results can be showed and commented.

Figure 5 shows a complex, strongly non flat and sound absorbing, sample on the Valladolid test site; figure 6 shows the values of the reflection index obtained by the eight laboratories. Figure 7 shows a gently non flat and sound absorbing sample (wood chips and concrete) on the Grenoble test site; figure 8 shows the values of the reflection index obtained by the eight laboratories. In both cases the agreement among the different laboratories may be judged fairly good, considering also that some laboratories did this kind of measurement for the first time, using different equipments, under different weather conditions.

4 CONCLUSIONS

In the frame of the QUIESST project several new improvements have been introduced in the measurement of reflection index of noise barriers and claddings; overall, these improvements led to a new, better and more robust measurement method. The use of a square 9-microphone array, not rigidly connected to the loudspeaker, and multichannel acquisition make easier on site measurements; the optimized signal subtraction technique gives nearly zero residuals, which can be quantitatively estimated by the reduction factor R_{sub}. The correction factors for geometrical divergence and sound source directivity give *RI* values physically meaningful and independent of the sound source used. The final inter-laboratory test conducted by eight laboratories on two test sites validated the method and the ongoing statistical analysis will give the repeatability and reproducibility values.

5 ACKNOWLEDGEMENT

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6 REFERENCES

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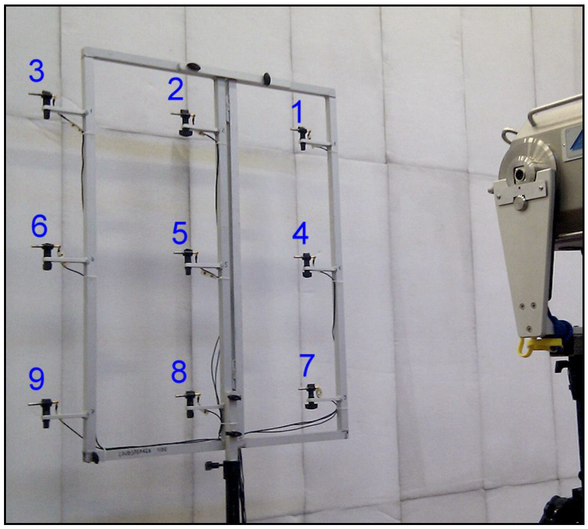

Fig. 1 – Microphone grid and sound source in front of a test absorptive surface.

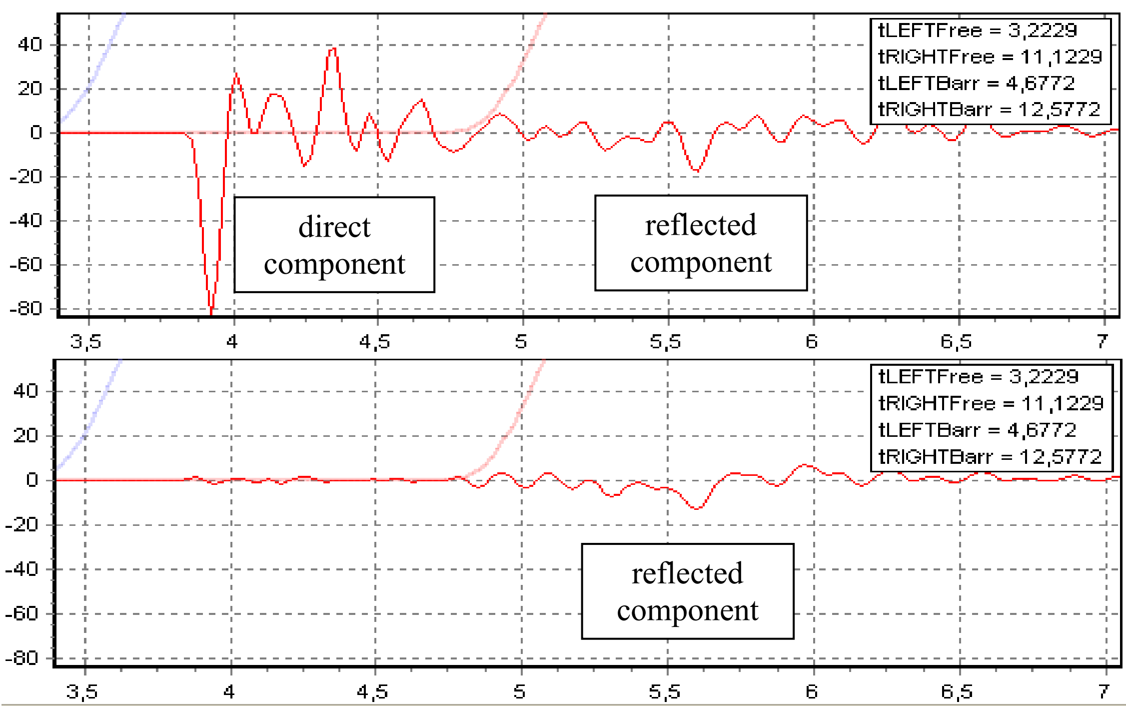

Fig. 2 – Top: impulse response taken in front of a flat reflective barrier. Bottom: the same impulse response after the signal subtraction.

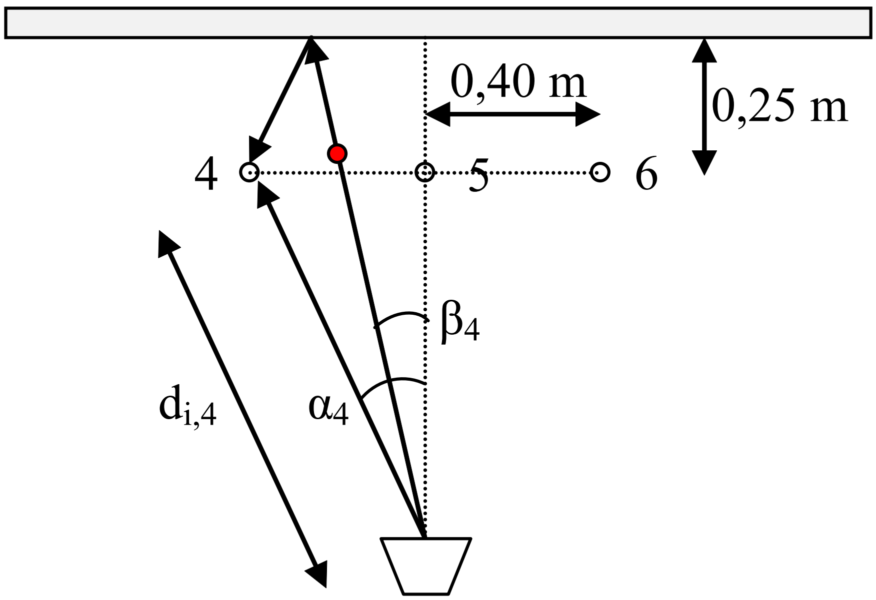

Fig. 3 – (Not to scale) Sketch showing microphone positions 4, 5, and 6 (white circles), the angles α_4 and β_4 for microphone 4 and the point, at a distance $d_{i,4}$ from the loudspeaker centre plate, where measurements to get the correction factor $C_{dir,4}$ are done (red circle).

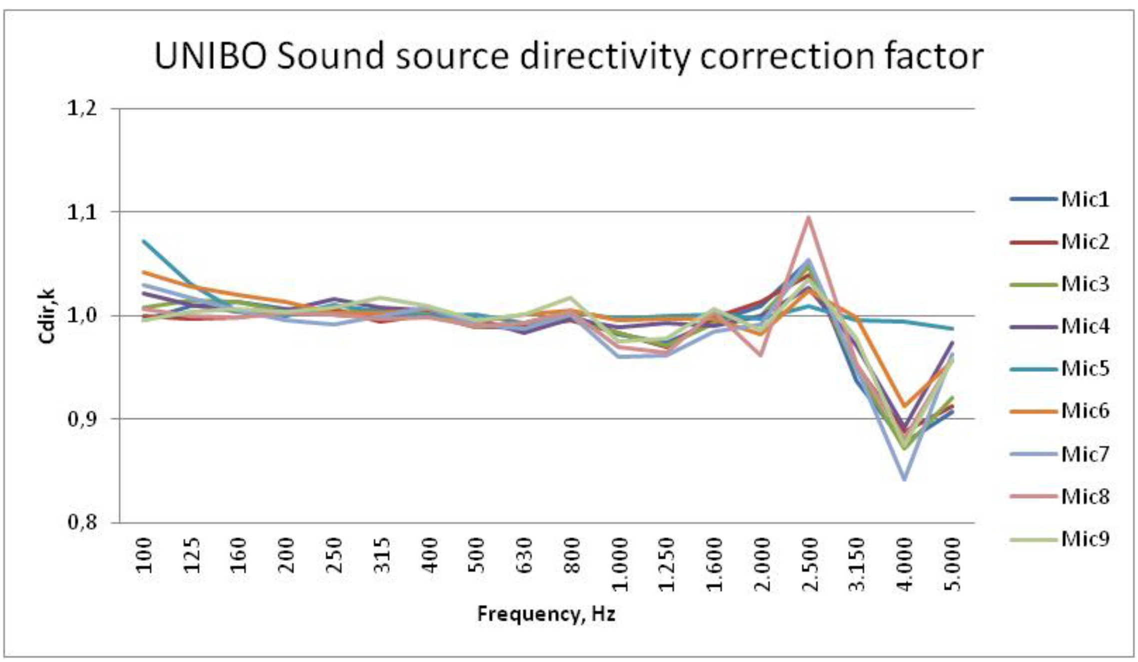

Fig. 4 – Correction factors for the sound source directivity. Zircon loudspeaker and 9 microphones on the QUIESST measurement array.

Fig. 5 – Sample 1 (strongly non flat, sound absorbing) on the Valladolid test site.

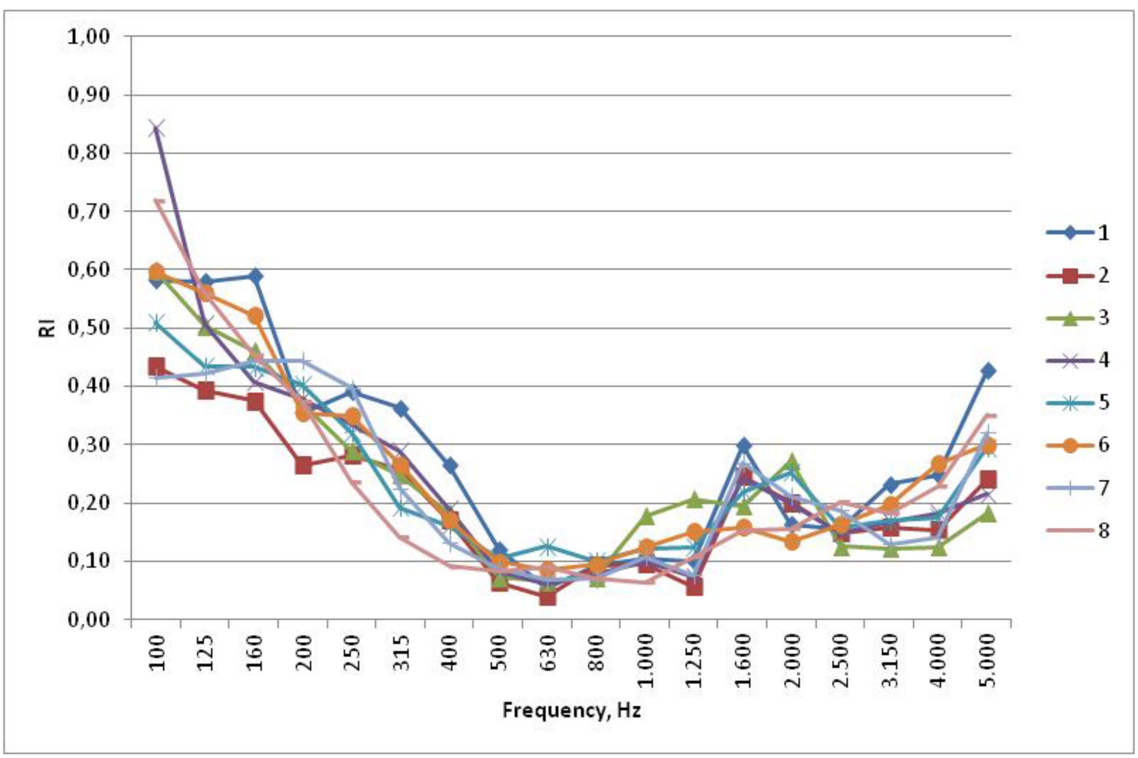

Fig. 6 – Reflection index values, according to Eqn. (1), measured by eight independent laboratories on sample 1 (se figure 5) on the Valladolid test site.

Fig. 7 – Sample 3 (gently non flat, sound absorbing) on the Grenoble test site.

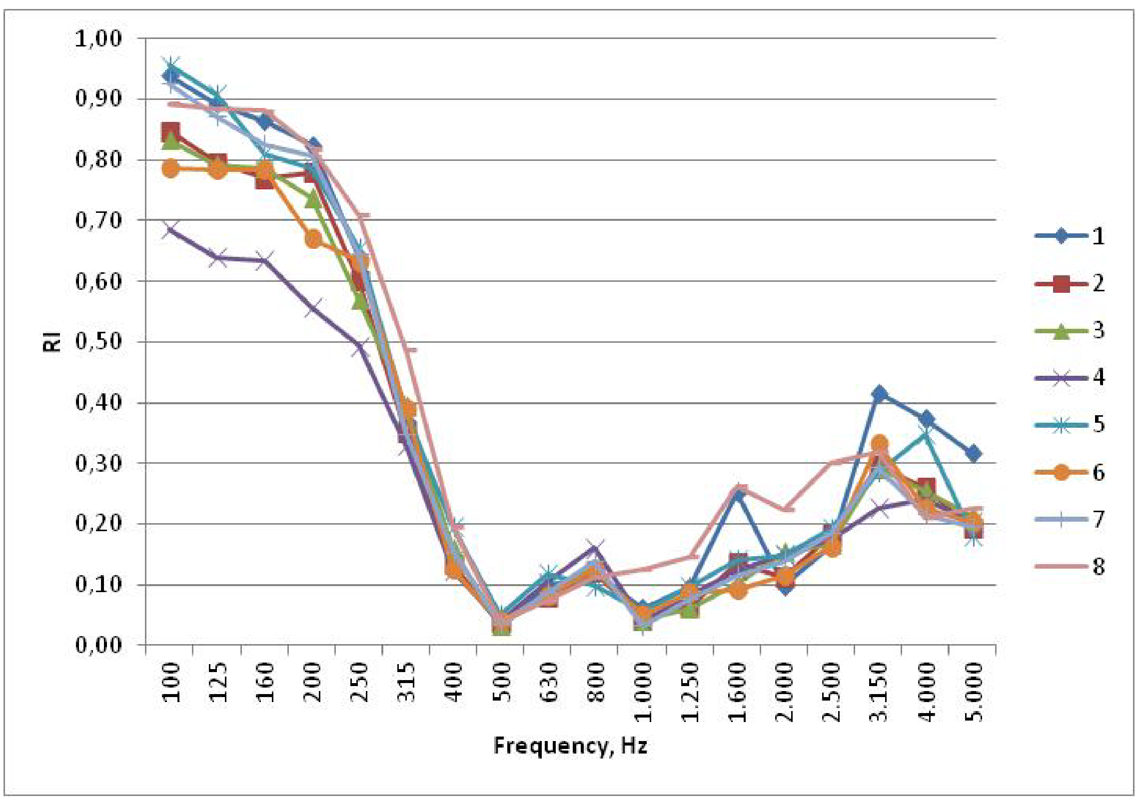

Fig. 8 – Reflection index values, according to Eqn. (1), measured by eight independent laboratories on sample 3 (see figure 7) on the Grenoble test site.