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THE DENORMS ROUND ROBIN TEST: MEASUREMENT PROCEDURE AND POST-PROCESSING OF TIME DATA

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The DENORMS Round Robin Test (RRT) is intended to study and improve the techniques used for the determination of the sound absorption coefficient of materials, with particular focus on the low-frequency range and on measurements in reverberation rooms. It is well known that the main reason why it is difficult to extend the frequency range of interest below 100 Hz is the low modal density. The modal behavior of a room is dependent on its geometry and this is one of the reasons why a discrepancy of the results coming from different laboratories can be found even if the same material is tested. This paper describes the measurement procedure developed to allow uniform acquisition and post-processing of acoustic response data of laboratories participating in the RRT, with and without absorbing materials inside. The tests and the post-processing operations performed on the measured data are also discussed in the paper.

Keywords: low frequency sound absorption, time reversal mirror, reverberation time, modal behavior, standing waves

1. Introduction

The sound absorption coefficient of building materials is one of the main parameters of interest for the acoustic designers. This coefficient is frequency-dependent and it is usually determined either with measurements in reverberation room or with measurements in impedance tube [1]. The latter technique can be applied only to small samples, it is very sensitive to boundary conditions, especially when used to determine the transmission loss of stiff samples that do not behave like equivalent fluids, and it is meant for the determination of the normal incidence sound absorption coefficient. On the other hand, measurements in reverberation room allow to assess the sound absorption coefficient for diffuse incidence on samples having an area of at least 10 m^2, and, because of the diffuse incident excitation, it is more representative of the actual behavior of the material in real operating conditions. Therefore, the two methods usually provide dissimilar results for the same type of material [2]. Moreover, the distribution of the sound pressure within a room in presence of standing waves, which are typical of the low frequency range, considerably varies from point to point. In this condition, the statistical theory (and the Sabine formula for the sound absorption area) can no longer be applied, unless a sufficient number of modes is grouped and the sound pressure is frequency-averaged over them. The transition from modal behavior to uniformly diffused field is given by the so-called Schroeder's frequency.

It is difficult to find results below 100 Hz on reports from reverberation rooms measurements. Because frequencies below this band are usually not considered, the procedures and measurement equipment which are commonly used are not suitable to properly excite such low frequencies. However, if performance evaluations are required in this range, the measurement procedure must be somewhat adjusted and particular care must be paid to the employed excitation signals.

The DENORMS Round Robin Test (RRT) on the low-frequency sound absorption of materials in reverberation rooms and in impedance tubes focuses on the collection and post-processing of time series data recorded by various laboratories on the same specimens. The scope of the RRT is to identify and study the differences among data coming from the participating laboratories, making sure that the measurements have been performed and post-processed according to consistent procedures. With this aim, general recommendations about data acquisition have been provided and a protocol has been proposed to all the RRT participants in order to make the extraction of the necessary information uniform. A script has been prepared in Matlab® that could be tailored to the specific characteristics of the time series provided by each laboratory. In this article, an overview of the proposed procedure is described taking as an example the data collected in the Laboratory of Acoustics at KU Leuven.

2. Description of the procedure

2.1 Measuring reverberation time

Under certain conditions, systems like rooms, auditoria [3] or even musical instruments [4] can be assumed linear time-invariant, for which the simple diagram in Fig. 1 is valid.

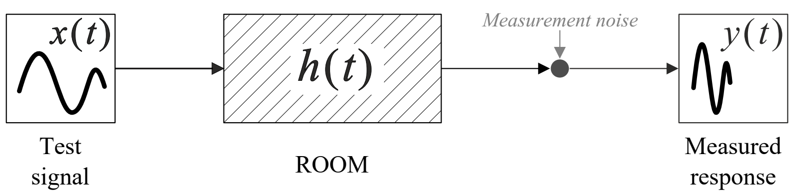


Figure 1: Input/output of linear time-invariant system (room).

In Fig. 1, $x(t)$ represents the input signal, $y(t)$ represents the output signal and $h(t)$ is the impulse response of the room. The output is related to the input by the impulse response through the formula:

$$y(t) = x(t) \otimes h(t) \tag{1}$$

with the symbol $\otimes$ indicating the convolution operation. The impulse response $h(t)$ is related to the transfer function $H(f)$ by the relation $H(f) = \mathcal{F}\{h(t)\}$, where $\mathcal{F}\{h(t)\}$ indicates the Fourier transform. It is worth remarking that a transfer function is always defined between a "source position" and a "recording position". For this reason, it is important to make multiple measurements and compute many reverberation time values from which a spatial average on the room can be derived.

The impulse response can be measured by exciting the system through an impulse, the energy contained being the area under the squared magnitude of the time signal. However, this is not always possible or convenient; namely, if the room is large, a great amount of energy is needed, which can be injected either through a sharp impulse with amplitude tending to infinity, or with a broader, smaller-amplitude impulse. The first choice would be suitable to properly excite the high frequency bands, but it would introduce non negligible non linearities; to the contrary, a broader, smaller-amplitude impulse is not suitable to collect information about the high frequency range. The problem of properly covering the frequency range of interest is common to other acoustic measurements, such as vibro-acoustic excitation of beams or panels by impedance hammer [5, 6], where tips of different materials are used to generate impulse responses of different durations.

An alternative to impulse signals is provided by random signals, such as white or pink noise. The main drawback of this technique is of technical nature: depending on the signal all the frequency bands of interest should be excited with equal power or energy, thus a huge amount of amplification is needed. Moreover, random signals are often prone to be distorted by the loudspeaker. However, it is a relatively simple method to collect the minimum set of information required by the scopes of the RRT.

A third possibility is to use deterministic signals. Among the several available options, the time reversal mirror is one of the most efficient techniques. It consists in numerically generating an exponential sine sweep (ESS) and a particular signal, called inverse filter (IF), which is created so that its convolution with the ESS results in a Dirac delta function $\delta(t)$. If the ESS signal is reproduced and recorded inside a room, the result of the convolution between the recorded ESS and the IF is a delta function followed by all the reflections carried out by the reflecting surfaces. A similar technique is used also to characterize the sound absorption of road barriers [7].

In the light of the above considerations, the RRT procedure establishes two different methods to measure the reverberation time, which each laboratory could choose between:

- Time Reversal Mirror (TRM) of an exponential sine sweep
- Interrupted Pink Noise (IPN)

In particular, the use of TRM technique was recommended in the procedure because it automatically rejects non-linearities due to distortions in the reproduction system, it has a high signal-to-noise ratio [8] and it returns more valuable information than the IPN technique since it provides the impulse response of the room [9]. Techniques based on impulse signals, like gunshot or balloon pop, have not been considered suitable for the tests because characterized by poor frequency range coverage and scarce reproducibility.

The quantities to be measured based on the previous analysis are:

- Early Decay Time (EDT), that is, a reverberation time relative to the initial 10 dB of decay
- Reverberation time T10 (decay from -5 to -15 dB)
- Reverberation time T20 (decay from -5 to -25 dB)
- Reverberation time T30 (decay from -5 to -35 dB)

The measurements are initially performed in empty reverberation room, then they are repeated with a prescribed set of materials delivered to the participant laboratories in succession [10].

2.2 Generation of the excitation signal

If the TRM technique is used, an ESS signal is suggested with a duration of at least 30 s. For large rooms and when very low sound absorption is expected, the duration of the time sweep can be increased up to 60 s. According to Farina [9], the ESS must be numerically generated using the formula:

$$x(t) = \sin\left[\frac{2\pi f_1 T}{\ln\left(\frac{f_2}{f_1}\right)}\left(e^{\frac{t}{T}\ln\left(\frac{f_2}{f_1}\right)} - 1\right)\right] \tag{2}$$

where f_1 and f_2 are the starting and ending frequencies of the sweep and T is the overall duration of the time signal. The frequency range of interest for the DENORMS Round Robin Test has been set from f_1 = 20 Hz to f_2 = 5 kHz, so the acoustic system used to generate the test signal must be able to reproduce the audio file with high fidelity in this frequency range. The IF is then generated in two steps:

1. the numerically-generated ESS is mirrored in time;
2. a proper amplitude modulation is applied to account for the fact that the ESS has not a flat spectrum but rather a pink spectrum. The mirrored ESS is therefore multiplied by $10^{w(t)/20}$, where $w(t)$ is a linear function of time t starting from 0 dB and ending to $-6 \cdot \log_2(\omega_2/\omega_1)$, where ω_1 and ω_2 are the start and end angular frequencies.

With IPN, the only required step is to generate a pink noise signal, possibly discarding very low frequencies, lasting long enough to make the sound energy density within the room uniform. When the room has been saturated, the signal is abruptly interrupted and the decay is recorded.

Both with TRM and with IPN, the type of loudspeaker used must be able to cover the frequency range of interest, thus an appropriate woofer and/or a midrange loudspeaker is suggested.

2.3 Recording and preparation of the time series

In order to obtain an average reverberation time, it is suggested that the tests be performed with different source positions, and that the input signal is recorded in multiple positions of the microphones, placed at different heights so to uniformly cover the volume of the room. All the microphones signals should be recorded with a system having a sampling frequency of, at least, 44100 Hz and a 16 bit resolution. If available, another microphone can be placed in front of the sound source to directly record its emission.

When the TRM technique is used, the recorded ESS signals are convolved with the IF computed from the excitation signal in order to obtain the impulse response of the room (Fig. 2).

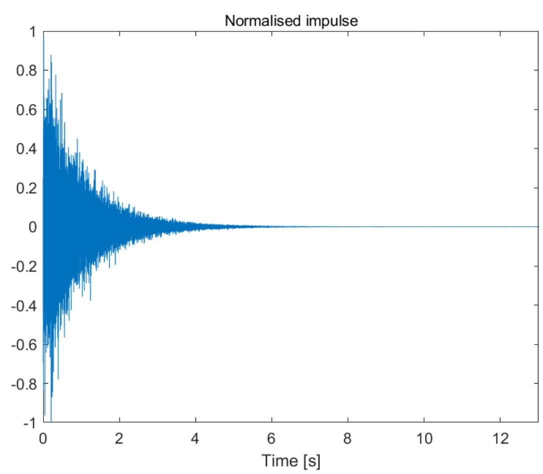


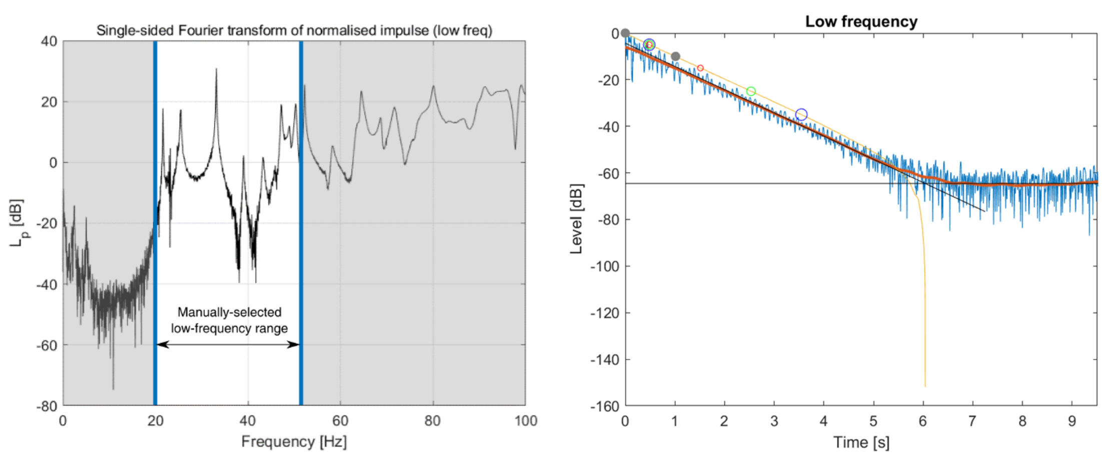
Figure 2: Example of impulse response computed using the TRM technique.

When IPN is used as an excitation signal, the portion of the recording corresponding to the decay after the source has been turned off can be post-processed directly.

2.4 Filtering of the decay

Depending on the adopted measurement technique, the time signals are filtered in one-third octave bands from 20 Hz to 5 kHz. A dedicated low-frequency analysis is made by applying a Butterworth filter to a user-defined low-frequency range, delimited by a lower frequency and a higher frequency directly selected on a graphic input window that shows the FFT spectrum of the signal, as in Fig. 3(a). This low-frequency range should be selected so to include at least 5 modes.

This extended band can, for example, bring out the beat phenomenon, which occurs at low frequency when the spacing between two adjacent modes is below 10 Hz and is not always visible with standard one-third octave filters (see for instance Fig. 3(b)). In [11], a FEM model is made of the same room whose measurement results are discussed in Section 3. The beating phenomena that follow from the experimental data was also reproduced by means of this FEM model. The blue series in Fig. 3(b) represents the moving average over 16 ms of the normalized sound pressure level measured by a microphone, the orange curve being its downsampled, smoothed version. The yellow curve represents the Schroeder's backwards integration of the normalized sound pressure level from the end of the decay (time t) and the start of the decay (time 0). This operation is required by ISO 3382-2 standard to obtain reverberation time from an impulse response. The determination of the integration limit t can be especially problematic, and the work of Lundeby et al. [12] has been taken as the main reference to automatize this task. In particular, the background noise is calculated based on the final 10% of the decay curve, while the initial portion of the curve must be processed with a series of smoothing and piecewise segmentation operations; to estimate the decay slope, a dynamic range of 10 dB is normally evaluated starting 10 dB above the noise level.



(a) (b)

Figure 3: (a) FFT of the time signal with manually-selected low-frequency range, where the modal behavior of the room is clearly visible, and (b) corresponding decay generated from the low-frequency filtered signal.

Figure 4 shows the filtered decay curves for the post-processing of an impulse response relative to the one-third octave bands centered at 50 Hz, 100 Hz, 200 Hz, 400 Hz, 800 Hz and 1600 Hz. In Fig. 4, the colored bullets indicate the initial and ending integration times employed to compute EDT (filled gray circles), T10 (red circles), T20 (green circles) and T30 (blue circles). The computation time to obtain, from a single ESS, the 22 decay curves corresponding to the one-third octave bands from 50 Hz to 5 kHz and the low-frequency decay curve was about 120 s with a sixth generation Intel® Core$^{\text{TM}}$ i7 processor. In order to retain the possibility to analyze the individual results and reconstruct anomalous outcomes although in an automatic post-processing, the script saves images of all the decay curves and a log file listing all the errors encountered in the process.

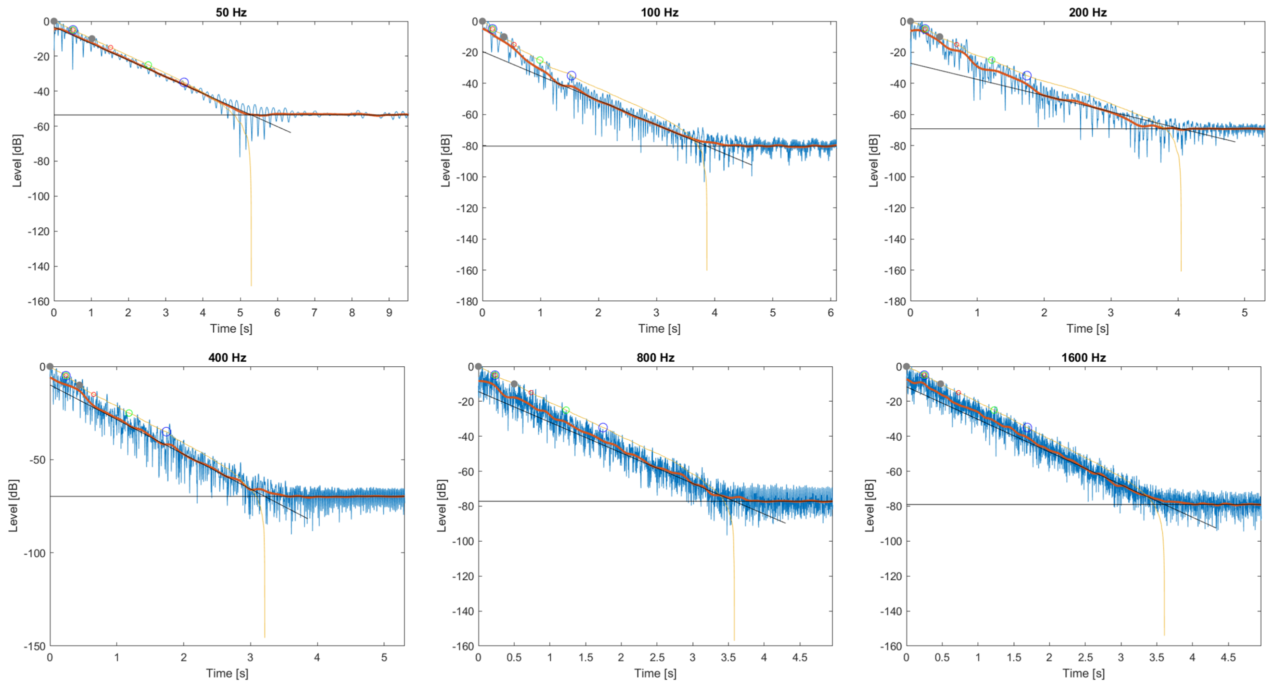


Figure 4: Example of decay curves generated for 50 Hz to 1600 Hz one-third octave bands.

3. The tests at the Laboratory of Acoustics – KU Leuven

3.1 Description of the room and of the materials

As an example of the application of the procedure to a real case, this section shows the results of the post-processing of the data collected at the reverberation room of the Laboratory of Acoustics, Department of Physics, KU Leuven. The tests were carried out on three types of specimens [10], all having a surface of 10 m^2, and in two different conditions of the test room: with diffusers and without diffusers. The configurations adopted during the tests are reported in Table 1.

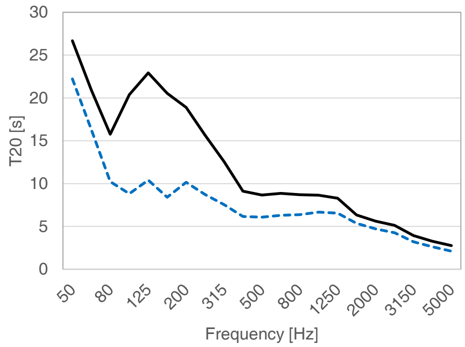
Table 1: Description of the measurements in the Laboratory of Acoustics at KU Leuven.

	Thickness [mm]		
Test	ECOPHON Industry Modus S	Fantoni 4For	Fantoni 4Akustik
a) Empty room	-	-	-
b) Glasswool	200	-	-
c) Glasswool + drilled wood panel	200	16	-
d) Glasswool + milled wood panel	200	-	16

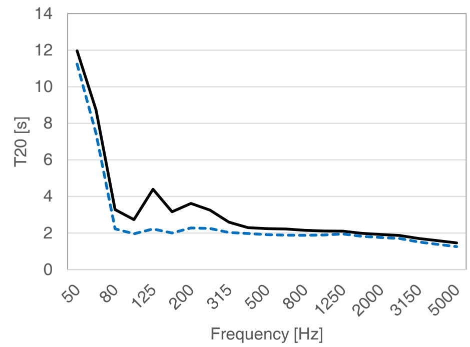
The measurements were carried out in March 2018 using 6 source positions. During the tests, temperature and humidity have been monitored since these parameter have a strong influence on the air absorption [13]. An OROS OR36 analyzer was used for generating and recording the signal. The loudspeaker used during the tests was a Brüel & Kjær Type 4295 sound source. The sound pressure level was recorded in 7 positions by BSWA and Brüel & Kjær 1/2” microphones placed at different heights inside the room and connected to the OROS OR36 analyzer. All the signals were recorded and exported in .mat format for the next post-processing stages. The total number of measurements performed is around 300.

3.2 Results

The results of the measurements are summarized in Fig. 5. It can be noted that the presence of the diffusers in the room influences the reverberation time, reducing it. Such behavior could hardly be anticipated, and it can be attributed both to the reduction of the mean free path of the acoustic waves [14] and to the low frequency dissipation of energy due to the structure of the diffusers. Further investigations will be made in the future.



(a)



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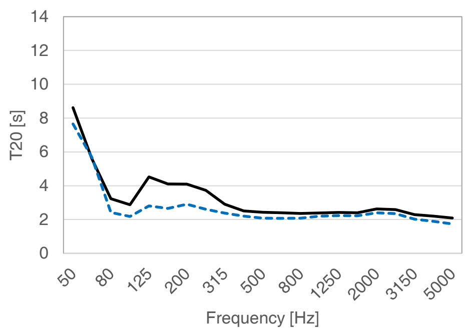

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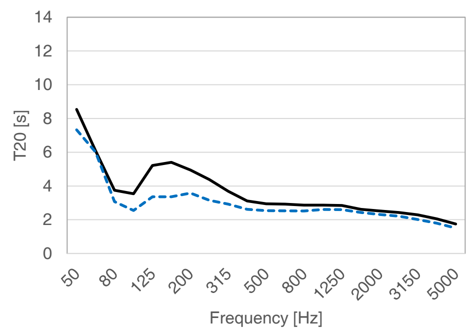

(d)

Figure 5: T20 measured at the Laboratory of Acoustics, KU Leuven, with (dashed line) and without (solid line) diffusers: (a) empty room; (b) glasswool; (c) glasswool + drilled wood panel; (d) glasswool + milled wood panel.

4. Conclusions

This paper presents the procedure developed for the acquisition and post-processing of time series data in the framework of the DENORMS Round Robin Test (RRT) on the low-frequency sound absorption of materials in reverberation room and in impedance tube. Some general prescriptions have been provided to promote the collection of high-quality data and to make their collation easier, but the choice of the preferred measurement technique was up to the specific laboratory. The recommended technique is the time reversal mirror with an exponential sine sweep, which has proven to be reliable and robust for the purposes of the RRT. A code has been developed and tailored to the characteristics of the data collected by each laboratory. The application of the post-processing code to the data collected in the reverberation room of the Laboratory of Acoustics at KU Leuven is presented. So far, the script has been successfully applied to the data coming from five laboratories that used either TRM or IPN techniques, and further five laboratories will provide time series data in the next months. Complete, fully post-processed data are expected to be a valid base of investigation for the analysis of data in the low-frequency range.

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