Abstract

The most popular and traditional method for measuring sound insulation of partitions, as described in ISO 10140, is performed through a sound level meter with a broad band random excitation signal. Additionally, the reverberation time in the receiving room must be measured, frequently through the interrupted noise method. However, alternative ways for evaluating the sound pressure level in source and receiving rooms, and for measuring the reverberation time, are available for decades. Some of these alternatives are based on the measurement of transfer functions through correlation or deconvolution techniques, which supposedly lead to more precise results. This paper presents the foundations for using a deconvolution technique in such a context, and comparisons from experiments performed with the traditional and the deconvolution technique. Such experiments were designed to introduce different background noise conditions and the presence of air flow (which should violate the system's invariant condition for using the deconvolution technique). Results show that the uncertainties associated with the alternative method tend to be slightly lower than those observed from measurements performed with the traditional method. Comparison from a repeatability investigation shows that the alternative method is significantly more precise than the traditional one in regard to this aspect.

Keywords: Sound Insulation, Deconvolution Technique
Measuring sound insulation from partitions through a deconvolution technique

1 Introduction

Correlation and, more recently, deconvolution techniques have become part of standard procedures in room acoustics. One of their most important advantages over other methods for measuring impulse responses is the robustness to noise. An important paper from 2001, by Mueller and Massarani[1] presents not only this particular advantage, but several others. For instance, the possibility to deal with non-linearities from loudspeakers when driven at high levels, and using this to further increase the signal-to-noise ratio.

Besides that, the computational cost for signal processing has become lower, and there are even mobile phone apps which can be used to measure impulse response of rooms with sine sweeps (deconvolution technique) with a reasonable quality. For using these techniques to obtain laboratory precision, the most critical features in the measurement chain are related to sound source, sensor, and the AD/DA converters, just like in any other method, including the "traditional" one.

However, standardized procedures for measuring sound insulation still refer to such techniques as "new" ones, and refer to ISO 18233. Although the procedure for measuring sound insulation of walls through correlation and deconvolution techniques has been presented in journals and conference papers, as Venegas, Nabuco and Massaraní[2] or Hak, Van Hout and Martin[3], we believe that further investigation and reports on the subject are important to consolidate the procedure.

In this paper, a comparison between the traditional method and the one which uses the deconvolution technique with a sine sweep excitation signal is described, for one wall and in-situ approach. Experiments were performed in different conditions: a) regular condition in a university building; b) induced background noise, and c) induced air flow. The last one was an attempt to introduce time variation in the source room conditions, for violating one of the conditions for using deconvolution or correlation techniques. The effect of conditions b) and c) on the resulting transmission loss was evaluated through the reference values obtained in condition a), and through the evaluation of the measurement uncertainty.

A repeatability study was also done, and the resulting uncertainties from both methods were compared.

2 Theoretical background

2.1 Deconvolution Technique for measuring impulse responses

The basic idea of this technique is to excite a system with a known signal, to measure its response, and to separate this response from the system’s characteristic response, that is, the impulse response (in time domain or its Fourier Transform in the frequency domain, the transfer
function). If the system responds linearly to the excitation, and its characteristics do not change during the measurement, this task is relatively simple.

As known, the response of a Linear and Time Invariant (LTI) system to a given excitation signal may be predicted through the convolution integral of such a signal with the system’s impulse response. It is proven through the Fourier Transform, that the convolution operation in such a LTI context is equivalent to a simple multiplication operation in the frequency domain. Therefore, the "deconvolution" operation in time domain may be performed through a simple division operation of complex quantities in the frequency domain:

$$H(f) = \frac{Y(f)}{X(f)},$$

(1)

where $H(f)$ is the system’s transfer function, $X(f)$ is the excitation signal, and $Y(f)$ is the system’s response. If needed, the impulse response is obtained through the inverse Fourier Transform of $H(f)$.

All detailed information and alternatives for processing the signals (performing averages, using filters) in the deconvolution process are described in Mueller and Massarani[1], which is a benchmark on the subject. According to them, the correlation technique, which uses a Hadamard Transform (important in times of low computational capacity) has practically no advantage over deconvolution techniques using sine sweeps and performs worst.

Because deconvolution and correlation techniques can use exactly the same excitation signal, an average process is capable of rejecting part of the background noise (acoustic or electric). Acousticians were pretty much excited when the correlation technique with Maximum Length Sequences showed possibilities to measure rooms even with high background noise conditions. When using deconvolution techniques, non-linearities from loudspeakers may be easily separated from the impulse response because they appear as peaks at their final instants.

2.2 Computing sound pressure level and transmission loss from impulse responses

After obtaining the impulse response, if measured with microphones, the potential sound energy may be computed as:

$$E_{pot_i} = \frac{1}{2\rho c} \int_0^\infty h_i^2(t) dt,$$

(2)

where $h_i(t)$ is the impulse response for the $i$-th source-receiver path, $\rho$ is the density of the medium, and $c$ stands for the velocity of sound in the medium. In practice, the superior integration limit should be substituted by the time instant when the sound decay encounters the background noise. Note that the $i$-th sound-receiver path may be measured either in the source room or between the source and the receiver room.

If the measurement system is calibrated and the electrical sensibility from the microphone is taken into account, the measured impulse response may be scaled to the unit of sound pres-
sure. However, since a ratio from sound energy in the source room to the receiver room must be computed for evaluating transmission loss, once the gain in the measurement system is kept constant, the unit conversion from volts to pascal is not mandatory. The values of $c$ and $\rho$ must not be taken into account for the same reason.

The quantity proportional to the potential energy must be spatially averaged in the source, and in the receiver rooms, respectively:

$$\bar{E} = \frac{1}{n} \sum_{i=1}^{n} \int_{0}^{\infty} h_i^2(t) dt,$$

where $n$ denotes the $i$-th source and receiver pair either in the source room or in the receiver room. Please note that the energies for each signal must be determined, and the average should be taken from the energy values. Averaging the impulse responses will lead to different, and invalid results in many situations.

Because of Parseval’s theorem, $\bar{E}$ may be computed in the frequency domain, more specifically, from the magnitude of the transfer function.

When evaluating energies in 1/1 or 1/3 octave bands, the integration limits in equation (3) should be those of each frequency band. In time domain, the signals must be filtered in frequency bands, and the energy computation must be done for each of them.

For computing the sound transmission loss (sound reduction index) from the potential sound energy obtained from the impulse responses (or from the transfer functions), the procedure is exactly the same as the one used in the traditional method. In in-situ evaluations, a correction term is introduced, and result is the “normalized level difference”[4], which is given by:

$$D_{nT} = 10 \log(\bar{E}) = 10 \log(\bar{E}_{source}) - 10 \log(\bar{E}_{receiver}) + 10 \log \left( \frac{T}{T_0} \right),$$

where $T_0$ is 0.5 s, and $T$ is the reverberation time in the receiving room, which may be evaluated also through the deconvolution technique.

2.3 Uncertainty evaluation

The evaluation of uncertainty may become quite complicated when taking all known information into account. The most simple way of estimating the measurement uncertainty is to take only some statistical parameters into account, such as the standard deviation from the spatial average of the Sound Pressure Level. When evaluated in such a way, they are classified as being of “Type A”, according to the Guide for the expression of uncertainty in measurement[5]. However, better estimation may be obtained when using information from calibration charts of components of the measurement chain or for the whole system. Truncation error, error due to amplitude or frequency resolution, filtering, among others may be evaluated and taken into account, and are classified as of “Type B”. Very interesting studies when measuring sound insulation and room impulse responses may be found in Michalsky, 2011[6] and Dietrich, 2013[7].
In the present study, only uncertainties of type A were evaluated, and the values in equation (4) were considered to be uncorrelated. Therefore, the model used for propagating the uncertainty is given by

$$u_c(y) = \sqrt{\sum_{i=1}^{n} \left( \frac{\partial y}{\partial x_i} u(x_i) \right)^2},$$

(5)

where $u_c(y)$ is the combined uncertainty associated to the output quantity $y$, $x_i$ is the $i$-th input variable (which in our case is the sound pressure level in the source room, sound pressure level in the receiver room, and reverberation time in the receiver room), and $u$ stands for uncertainty.

Equation (5) in the present context turns into

$$u_c(y) = \sqrt{\sigma_{\text{Source}}^2 + \sigma_{\text{Receiver}}^2 + \frac{10}{T \ln(10)} \sigma_T^2},$$

(6)

where $\sigma_{\text{Source}}$ is the standard deviation from sound pressure level in the source room, $\sigma_{\text{Receiver}}$ is the standard deviation in the receiver room, $T$ is the reverberation time, and $\sigma_T$ is the standard deviation from reverberation time in the receiver room.

In the present study, although computed, the expanded uncertainty was not used for the comparisons between different methods, but in a final report, this step should be performed. For doing that, the effective degree of freedom from the measurements must be evaluated, and the correspondent t-distribution factor for the desired confidence interval must be determined and used as a multiplier to the combined uncertainty.

3 Methodology

For comparing the absolute values and the uncertainty for transmission loss obtained through the traditional method and the one which uses impulse responses obtained through a deconvolution technique, a set of two non-qualified reverberation rooms were used. Because the rooms are not qualified according to ISO 10140, the methodology used is the one for evaluating in-situ sound insulation from field measurements, as described in ISO 16283-1.

The source room has a volume of 110 m$^3$, while the receiver room has 20 m$^3$. The source and microphone positions were kept approximately the same in all measurements. Sound pressure level and impulse responses were measured for two source positions, and five microphone positions for each room. Because the receiver room is too small, when measuring the reverberation time, the microphone was placed in three different locations for each of the two different sound source positions. Temperature and humidity were monitored in all experiments.

The power amplifier and the dodecahedral sound source used in both methods are the same. They are part of the same commercial set which is to be used with a commercial sound level meter with software for performing sound insulation measurements. Measurements performed through the traditional method used the entire commercial set, which in turn was set to use
a pink noise as excitation signal. The interrupted noise method is used for measuring the reverberation time by the commercial software.

When measuring impulse responses, the same power amplifier and sound source were used, together with a measuring microphone, pre-amplifier, signal conditioner and a Roland Quad-Capture sound card, connected to a notebook. The sample frequency and the resolution were set to 96 kHz, and 16 bits, respectively. A logarithmic sweep sine of approximately 10 s was used as excitation signal, and the resulting impulse response for each source-receiver pair is an average of 10 measurements.

A open-source Matlab toolbox (ITA-Toolbox[8]), developed by the Institute of Technical Acoustics from Aachen University was used for controlling the measurements and signal processing. Routines for assessing the sound pressures level, the reverberation time, their associated uncertainties, and the combined uncertainties were developed. These routines used specific functions from the ITA-Toolbox for time-windowing and filtering signals, and assessing the reverberation time.

Transmission loss for the adjacent wall was evaluated in the conditions described next.

3.1 Standard condition
In the standard condition, no on purpose perturbation is imposed to the system under test, and therefore is used as a reference for some comparisons. Uncertainties obtained from this experiment are consequence of the methods used, and the regular background noise from outside the rooms.

3.2 Condition with forced background noise
A pink noise from a mobile phone app, driven through a Marshall guitar amplifier was positioned in the receiver room, and set to be 4 dB lower than the test signal sound pressure level transmitted from the source room, in average. This condition creates an unfavorable scene for both methods, and violates what is recommended by ISO 16283-1 for the traditional method.

3.3 Condition with forced ventilation - Time variant system
A fan for residential use was placed in the source room, with the intention of violating the time invariance condition, which is theoretically necessary for using the deconvolution technique. Besides providing air flow, the fan radiates noise, and the effects are combined.

3.4 Repeatability test
In order to evaluate the performance of both methods, exclusively in relation to repeatability of results, 10 measurements (for each method) were performed for only one sound source position, one microphone position in the source room, and one microphone position in the receiver room. In this case, the impulse responses were measured after only one sweep sine period, and not 10 in a row, as in the conditions previously described, so that for each microphone
position, 10 impulse responses were obtained.

4 Results

Figure 1 shows results for the normalized difference level obtained through both methods. They look reasonable, especially for the bands with central frequencies between 125 Hz and 4000 Hz. Within this frequency range, the maximal deviation is around 3 dB, in the frequency band of 160 Hz, but the general behavior follows approximately the same rule. In many frequency bands, they match very well. Of course, this is a first observation. Most important is the fact that, taken their combined uncertainty into account, all results are in the uncertainty range of both methods within this frequency range, with no exception. Note that we have not used the expanded uncertainty for this comparison.

The combined uncertainty for both methods is shown in Figure 2. One observes that the combined uncertainties are in the same range, with a tendency to be smaller above 400 Hz, for the one obtained for the deconvolution technique.

![Figure 1: Results for the normalized difference level in 1/3-octave bands obtained with both methods: Traditional in blue; and through the deconvolution technique, in black.](image)

Figure 3 presents absolute value of differences obtained with induced background noise from the reference results, obtained under the standard condition, in regard to normalized difference level. One observes clearly a general tendency for the method using the deconvolution technique to present results closer to those obtained in the standard condition, even under very unfavorable background noise condition. The combined uncertainty obtained from this experiment is much alike those obtained in the standard condition. Taking the combined uncertainty
into account, one may say that measurements performed with the deconvolution technique for bands up to the one with central frequency equal to 2000 Hz are still valid, while the same is not observed for the traditional method. For the traditional method, results between 500 Hz and 1250 Hz may be regarded as "valid" in a first (and superficial) evaluation.

From the condition with induced air flow, relevant differences from the standard condition for both methods were not observed. Combined uncertainties, especially for measurements performed with the deconvolution technique were slightly greater at higher frequencies.

Large differences appear when analyzing results from the repeatability test, as shown in Figure 4. Here, one of the remarkable advantages from the deconvolution technique may be clearly observed, namely, the large repeatability of results, which is a consequence of using exactly the same excitation signal. Such a feature could not be perceived in Figure 2 because the sound field itself contributes with large uncertainties, related to the lack of diffusion (and strong modal behavior, consequently). Estimated Schroeder frequencies are 340 Hz and 813 Hz for source and receiver rooms, respectively.

5 Conclusions

A method that uses the deconvolution technique for measuring the airborne sound insulation has been investigated under the perspective of ISO 16283-1, which guides in situ evaluations. From the case studied, the method performs similarly to the traditional method under standard, not noisy, conditions. The deconvolution method tends to present measurements uncertain-
Figure 3: Differences obtained when measuring under induced background noise in 1/3-octave bands for both methods, compared to their respective reference measurements.

Figure 4: Results for the combined uncertainty in 1/3-octave bands obtained with both methods from the repeatability test.

...ties which are lower than those obtained when using the traditional one, especially at higher...
frequencies (above the 1/3-octave band of 500 Hz in the case presented). An experiment for testing the influence of higher background noise in the receiving room has shown that the deconvolution method is still able to deliver valid results up to the 2000 Hz band, while the traditional method presents valid results in a more limited frequency range (500 Hz to 1250 Hz 1/3-octave bands). When inducing air flow in the source room, both methods performed well, although it would be expected that this situation would negatively affect results obtained through the deconvolution technique. Very different results were observed when investigating the repeatability of measurements for only one source and microphone positions (in the source and receiver room). In this case, the high capacity of repeating results when using the deconvolution technique has been observed. This advantage will not be so important if source and receiver rooms present strong modal behavior. However, it indicates that the method which uses the deconvolution technique will potentially perform better in laboratory tests as those described in ISO 10140.

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References


