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Sound level meter directional response measurement in a simulated free-field

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Sound level meter directional response has to be measured in free-field, a condition that is usually met by using an anechoic chamber. However, using one is not always a possible option for cost or availability reasons. This paper describes an automated method that uses a structure built to provide a simulated free-field and a time-selective technique to remove the room reflections from the frequency response, thus enabling the directional response of an instrument to be measured without an anechoic chamber. The effects of different sources of uncertainty – external noise, microphone positioning, temperature, etc. - on the obtained results are discussed. Finally, the proposed method is then compared to the classical toneburst method for validation.

Introduction

Traditionally, free-field directional responses of Sound Level Meters (SLM) are obtained in anechoic chamber by sending single-frequency impulses, measuring the magnitude of the received signal and repeating this for every frequency and angle of interest. This process can be both time and money consuming especially if a laboratory doesn't have its own anechoic chamber.

Being able to measure free-field responses of instruments is important for SLM manufacturers as IEC 61672-1 [1] specifies limits on the influence of the case on the measured sound pressure level for various angular and frequency sectors.

This work presents an automated method which is both faster and doesn't require an anechoic chamber. It makes use of swept-sine excitation signals and signal processing to remove room reflections (ground, walls, etc.) while keeping them from the device under test, thus creating a *simulated free-field*.

The objective is to show that this method can be used to measure SLM directional responses with similar accuracy to the classical toneburst method.

1 Simulated free-field

1.1 Principle

A free-field describes an environment where only the direct sound from a source is measured. Although measuring in an ideal free-field is a condition that is almost impossible to meet, it can be artificially achieved in anechoic rooms by covering every surfaces with material that will absorb sound reflections. It can also be obtained in an ordinary room, given that the test layout is such that the first reflections will come long after the direct sound so they can be filtered; such an environment is known as a *simulated free-field*.

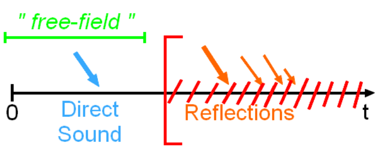

Figure 1: Signal received by the microphone - any sound waves arriving after the direct sound is filtered.

From a signal processing point of view, this means that a time window will be applied on the recorded signal that encompasses the direct sound and excludes any further reflections (see Figure 1).

The geometry of a simulated free-field is an ellipsoid where the loudspeaker and the microphone are its focal points. Its dimensions of depends on the location of the first reflective surface – typically the ground. Figure 2 illustrates this geometry.

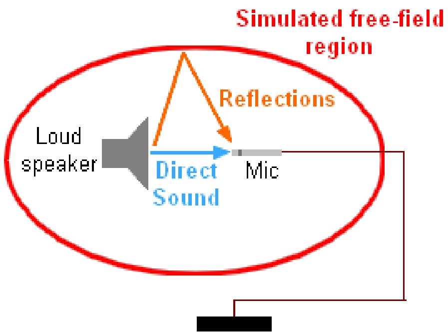

Figure 2: Principle of a simulated free-field - the path of the first reflection is longer than the path of the direct sound. Sound waves with a shorter path are considered to propagate within a free-field.

The layout of the test system is defined by two parameters: the lowest frequency to test and the desired delay before the first reflections reach the microphone.

To be usable with the classical method, the test system need to have the ability to record a full wavelength at the lowest frequency f_{min} . Equation 1 expresses the direct path's length as a function of f_{min} :

$$D_p \geq \lambda_{f_{min}} \Leftrightarrow D_p \geq \frac{c}{f_{min}} \qquad (1)$$

where:

- D_p is the direct path of sound from the loudspeaker to the microphone (m);
- c is the speed of sound in air (340.3 m.s^{-1});

The other parameter to set is the path (or delay) of the first reflections – the perimeter path P of the ellipsoid. To meet the first condition, they have to travel at least twice the path of the direct sound (equation 2):

$$P > 2.D_p \qquad (2)$$

The larger P is, the easier it will be to separate the test signal from the reflections.

1.2 Application

According to the IEC 61672-1 standard [1], the directional response of a sound level meter must be provided from 250 Hz up to 12.5 kHz for class 1 instruments. Figure 3 shows the length of the simulated free-field for $P = 2.125\ D_p$:

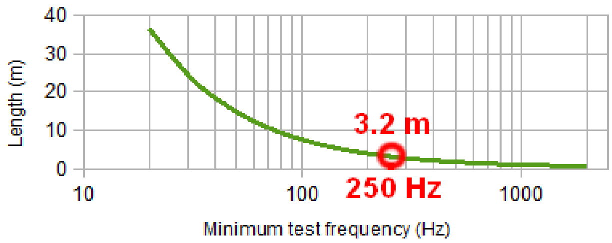

Figure 3: Simulated free-field length *P* against the minimum test frequency.

For practical reasons, the minimum frequency was kept at 250 Hz. A metallic structure was built to create a free-field meeting those conditions: it is a 3.2m long cubic frame (see Figure 4 below). The loudspeaker was mounted on a wooden plane and hung to the structure 1.6m away from the floor. The instrument under test was then mounted on a rod connected to a software-controlled turntable, allowing the whole measurement process to be automated. The rod needs to be long enough to avoid adding early reflections, meaning that the bent part has to be outside the simulated free-field (see Figure 2).

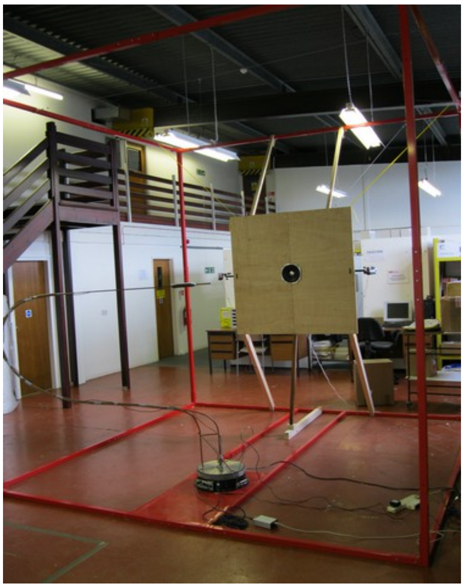

Figure 4: Metallic frame (in red) enclosing the simulated free-field.

With this configuration, the first obstacle is the ground whose reflections will reach the microphone 4.5 ms after the direct sound.

The positioning of the microphone was done using two lasers pointing at the microphone focal point of the simulated free-field (see Figure 5). Extreme care was taken to position correctly the microphone across different test set-ups as it was experimentally found to be an important source of uncertainty.

Figure 5: Accurate microphone positioning using lasers.

2 Signal processing

To obtain the frequency response, the swept-sine method was used, as many have emphasized its advantages to other popular methods ([2] to [6]). The most noticeable ones are the high signal-to-noise ratio (SNR) achieved and the removal of room reflections and harmonic distortions (THD). Also, [7] indicates that labs using this method seem to obtain lower spread in results.

The principle is to create an excitation signal *x(t)* that is played through the loudspeaker and recorded by the Device Under Test – or DUT - as *y(t)*. With *h(t)* being the DUT's impulse response, the system can be described by equation 3:

$$x(t) * h(t) = y(t) \tag{3}$$

In this work, *x(t)* is a logarithmic sweep (sine wave whose frequency exponentially increases with time) that covers the whole frequency range. As described in [1], such a signal can be crafted in the frequency domain to have an emphasis on the weak frequencies of the loudspeaker while maintaining a constant time envelope, thus allowing it to be played back using the maximum dynamic range of the sound device.

A Fast Fourier Transform (FFT) is performed on the excitation signal, and then inverted to obtain a deconvolution spectrum $X^{-1}(f)$. A high-pass filter at 20 Hz is applied on that spectrum to remove the influence of low frequencies at a later stage.

The recorded signal is then isolated from *y(t)* by cross-correlating it with the input signal and applying a first time window to remove any sound that occurred before or after the sweep. The FFT of this signal gives the recorded signal spectrum *Y(f)*. Equation 4 shows how the transfer-function *H(f)* of the system DUT + reflections if obtained:

$$H(f) = Y(f).X^{-1}(f) \tag{4}$$

This operation is also called a *spectral division*. The Inverse Fast Fourier Transform (IFFT) of the transfer function gives the system's Impulse Response. This is where reflections and distortions can be easily seen and removed by applying a second time window around the direct sound, as shown in Figure 6.

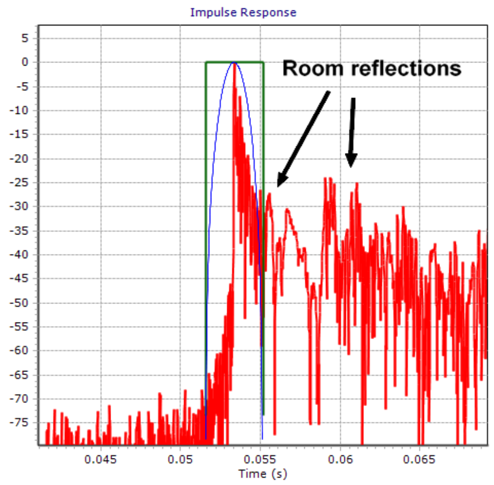

Figure 6: Impulse response windowing and removal of room reflections.

Soares describes in [2] how various window functions (Blackman-Harris, Hamming, Kaiser, etc.) affect the end results. Great care should be taken when selecting the width of this window: it needs to be broad enough to include all the direct sound and the reflections from the instrument's case, but also sufficiently narrow to reject those from the room.

Finally, the FFT of the windowed Impulse Response is the desired DUT's transfer-function.

The block-diagram on Figure 7 shows the procedure this procedure:

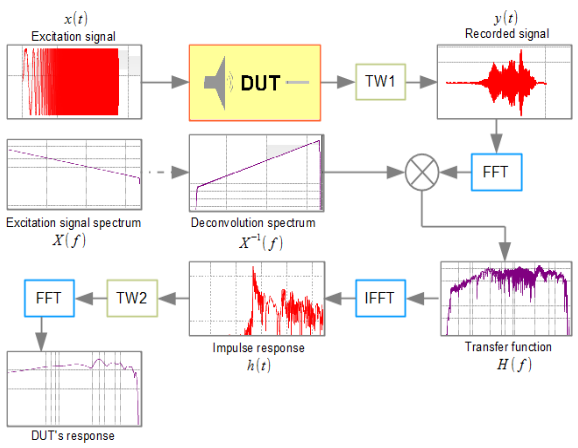

Figure 7: Procedure to measure a transfer-function with sweeps.

3 Test procedure

The whole process was made automatic to reduce the "human error" factor by using a computer with a custom software controlling the loudspeaker and the turntable. The basic steps of measurement are shown on Figure 8.

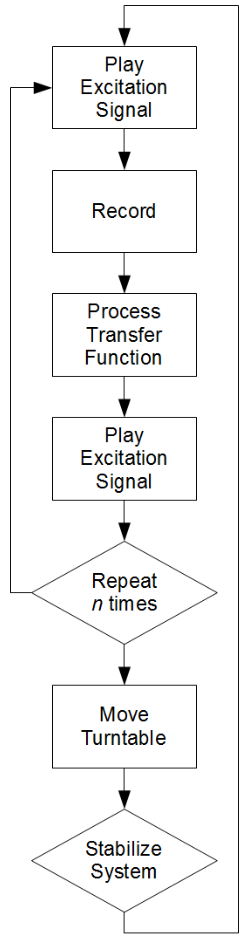

Figure 8: Procedure for recording a directional response.

The excitation signal used was a logarithmic swept-sine from 20Hz to 20kHz of 2^{19} samples long at a sample rate of 48 kHz (~5.5 sec). The number of samples has to be a power of 2 so that a FFT can be performed directly on the signal without extra padding.

The system was experimentally found to be robust to ambient noise - footsteps, quiet speech, distant cars, etc. - when the excitation signal was played with a sufficient SNR, typically 40dB. As a consequence, the transfer-function at each angle was obtained using 2 coherent averages only.

Results were stored both as fractional octave bands as described in IEC 61672-2 [8], and as 24th-octave bands for a finer analysis.

With 10° angular steps, 2 measurements per angle and 30 seconds of stabilization time between each step, a complete directional response is obtained in 33 minutes.

The two following set-ups were used: a rod with a Cirrus optimus sound level meter fitted on, and another one with its Cirrus MK:224 microphone and pre-amplifier only.

The influence of the case of the sound level meter was finally obtained by subtracting the responses of the two above set-ups, as shown in equation 5:

$$L_{case}(f) = L_{mic+case}(f) - L_{mic}(f) \qquad (5)$$

where:

- L_{case} *(f)* is the influence level of the tested SLM.
- $L_{mic+case}$ *(f)* is the response of the first test set-up.
- L_{mic} *(f)* is the response of the second test set-up.

Figure 9 shows an example of directional response of a sound level meter processed with this method.

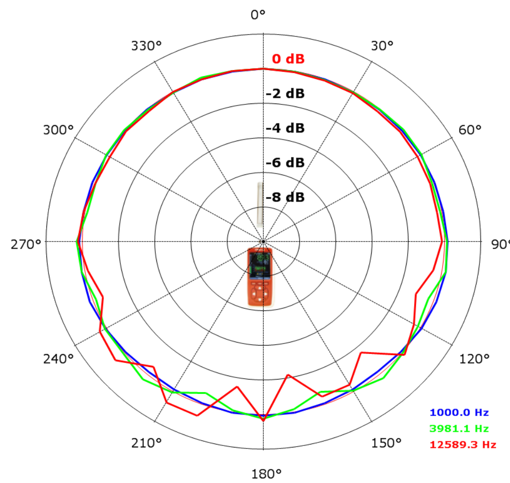

Figure 9: Case influence of a Cirrus optimus sound level meter at 1kHz, 4kHz and 12.5kHz.

4 Validation of the proposed method

Some parameters contributes to the difference that can be observed between obtained results. The sources of uncertainty are the following:

- $\mathbf{u_{rounding}}$: Rounding of the results (0.01 dB).
- $\mathbf{u_{cal}}$: Microphone calibration.
- $\mathbf{u_{pos}}$: Microphone position and angle to the loudspeaker.
- $\mathbf{u_{repeat}}$: Standard deviation between replications of a measurement (Repeatability).
- $\mathbf{u_{repro}}$: Reproducibility of the experiment.
- $\mathbf{u_{env}}$: Environmental conditions (temperature, air pressure, humidity).

As previously mentioned, the system was found robust to ambient noise, which results in a very low standard deviation between consecutive measurements at the same angle.

The reproducibility uncertainty was evaluated by re-setting the microphone's angle and position to the loudspeaker, the gain of the different amplification stages. It was found to be the most contributing factor to the method uncertainty of measurement, which is due to the fact that it requires human actions and allows for “human error”.

Environmental conditions also bring a non negligible uncertainty because it isn't possible to control their variations within this open metallic structure.

The combined uncertainty $\mathbf{u_c^2}$ of the proposed method was then obtained by propagating the individual sources of uncertainty according to equation 6 as described in the ISO/IEC GUM [9] and can be seen and Figure 10:

$$u_c^2 = \sum_{i=1}^{N} \left[\frac{\partial f}{\partial x_i} \right] u^2(x_i) \qquad (6)$$

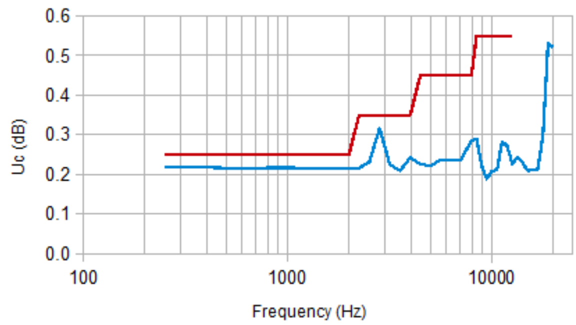

Figure 10: Uncertainty of measurement of the proposed method (blue) and its maximum tolerance within ± 30° to the reference direction (red).

The peaks at 4 kHz and 19 kHz are mainly due to the non-uniformity of the loudspeaker sound field at those frequencies. The consequence is that a slight offset in the position of the microphone against the acoustic centre of the loudspeaker may introduce some differences in the results when resetting the system.

For the validation of this method, directional responses from the same set-up were obtained by using both this method and the classical toneburst method which consists of sending a single frequency sine wave through the loudspeaker, recording the magnitude of the response before reflections and then switching to the next frequency. The case reflections data is then obtained by comparing the difference in magnitude between the two previously described set-ups. This method can achieve a really high SNR, but also takes much longer and has a lower frequency resolution as pointed out by [2].

The criterion of the normalized error E_n described in the ISO Guide 43 [10] was used. It compares the difference between the obtained results of both methods to the combination of their uncertainties (equation 7).

$$E_n = \frac{|L_{pm} - L_{ref}|}{\sqrt{U_{pm}^2 + U_{ref}^2}} \quad (7)$$

where:

- L_{pm} is the result obtained with the proposed method.
- L_{ref} is the result obtained with the reference method (toneburst method).
- U_{pm} is the expanded uncertainty of the proposed method.
- U_{ref} is the expanded uncertainty of the reference method.

If E_n is lower or equal to 1, the proposed method can be validated. Figure 11 presents the estimated normalized error:

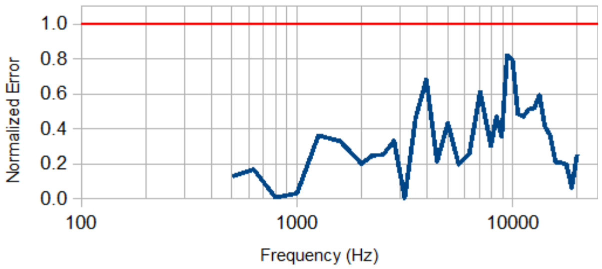

Figure 11: Normalized error E_n of the proposed method against the classical method

The software used for the classical method doesn't measure frequency responses below 500 Hz, so only the frequencies above were verified: the normalized error is lower than 1 at all frequencies, so the proposed method can be validated.

Conclusion

This work presented an effective automated method to measure directional response without an anechoic room was presented in this work. Comparison with the classical method for measuring transfer function showed that the results obtained have a similar accuracy. Some enhancements could be made to reduce the reproducibility uncertainty by automating more parts of the process, thus lowering the human error factor.

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[9] ISO/IEC Guide 98-3:2008, *Guide to the expression of uncertainty in measurement*.

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