Model for assessing the influence of an omnidirectional source's directivity in room acoustics measurements

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Introduction

The acoustical characterization of an environment by means of measurements is the first step towards a full comprehension of the effects that sound provokes during its propagation. The measurement chain of an acoustical domain is generally a quite complex system that starts with an excitation signal and ends with the determination of the impulse response. The measurement procedures are described both in the ISO 3382 [1] and ISO 18233 [2] standards. The presence of many sub-systems affects the ongoing signal and causes an alteration of the input at the end of each step. As a result, the impulse response as well as the parameters that from it are derived, will present a perturbation that needs to be quantified through the definition of an uncertainty [3]. The present work will deal with a specific part of the transmission path that alters the sound flow, which is the influence of the omnidirectional sources directivity on the final measurement.

Omnidirectional Sources

An omnidirectional source is an electroacoustic sound propagation system, which consists of multiple-driver loudspeakers positioned in such a way to reproduce a quasi-spherical sound field. A perfect omnidirectionality cannot be achieved for several reasons, such as interference between radiation from discrete drivers, inherent directivities of individual drivers (depending on geometric features), diffraction effects and surrounding enclosures. The design of omnidirectional sources is possible until the frequency exceeds a certain value (cut-off frequency). Then, the wavelength decreases to the order of the driver's diameter and the single driver radiates with a sharp and non-spherical directional pattern.

Measurements

A typical omnidirectional speaker for practical application was evaluated. Measurements were conducted both in the anechoic chamber of the Institute of Technical Acoustics in Aachen and in the assembly Hall (Aula) of RWTH Aachen University by Witew and Behler, within a study on uncertainties in room acoustics [4]. The measurement equipment in the anechoic chamber mainly included a custom made measuring apparatus. The loudspeaker was placed on a turntable whereas a free field microphone was fixed to a pivot arm in the acoustic far field at a distance of 2m from the acoustic centre of the loudspeaker. After each measurement the turntable and the microphone arm where moved in 5 degree steps so that the upper hemisphere of the directivity of the loudspeaker was measured. Once one hemisphere was measured, a composite 1368 set of complex frequency response functions was compiled to enable the characterization of the omnidirectional source's radiation for the investigated loudspeaker. In Figure 1, the radiation diagram of the loudspeaker for 4 kHz is given, showing a very irregular and inhomogeneous radiation pattern.

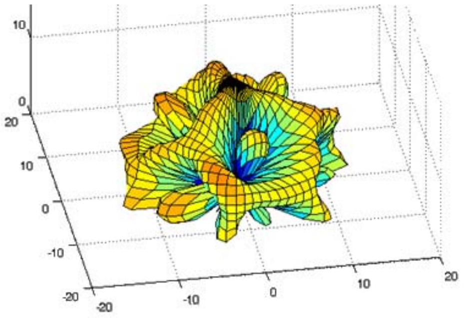

Figure 1: Radiation diagram at 3 kHz for an omnidirectional source

The assembly hall of RWTH Aachen University is a lecture room with dimensions of 22x28x10 m (WxLxH), and volume of V = 5500 m^3. In accordance with ISO 3382, six microphone positions were chosen, five of which located on the main parquet and one on the rear balcony. For each of the six microphone positions, 36 measurements were conducted. After one measurement, the turntable was rotated by an angle of 10°.

Quantification of Omnidirectionality

In order to study the radiated field from the measured source, a method based on area-weighted spatial standard deviation of radiated levels over a free field measurement sphere was chosen. This method, alternative to the one included in the ISO 3382, was formally proposed by Leishman [5]. It calculates an area-weighted spatial standard deviation of either normalized or non-normalized frequency response function levels (from a complete set of 1368 narrowband measurements) as:

$$\sigma_{AWL} = \sqrt{\frac{\sum_{m=0}^{M-1}\sum_{n=0}^{N-1} S_{m,n}\left[L_{m,n}(f) - \langle L_{m,n}(f)\rangle_S\right]^2}{\sum_{m=0}^{M-1}\sum_{n=0}^{N-1} S_{m,n}}}, \qquad (1)$$

where $\langle L_{m,n}(f)\rangle_S$ is the arithmetic area-weighted spatial average of levels, and $S_{m,n}$ is the area weighting factor (i.e., the effective sampling area per microphone on the measurement sphere), obtained by surface integration over appropriate sections of the measurement sphere. The integers m and n indicate the 5° increments in angles θ (polar) and ϕ (azimuth). In Figure 2 a graphical representation of the simplified sampling concept is shown. The nineteen measurement points for a single value of n (single vertical plane) are highlighted, together with a sampling area example.

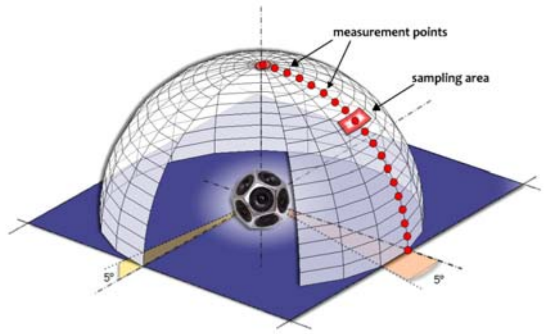

Figure 2: Simplified sampling scheme on the measurement hemisphere. Points in red show the position of the microphones.

Points in red correspond to the position of the microphones. Figure 3 shows the narrowband standard deviation (std) plot for the measured dodecahedron source. Measurements confirm that the omnidirectional source emits a quasi-spherical sound field until the cut-off frequency occurs (almost at 1 kHz). Over this value, the radiation pattern diverges significantly from an ideal spherical shape.

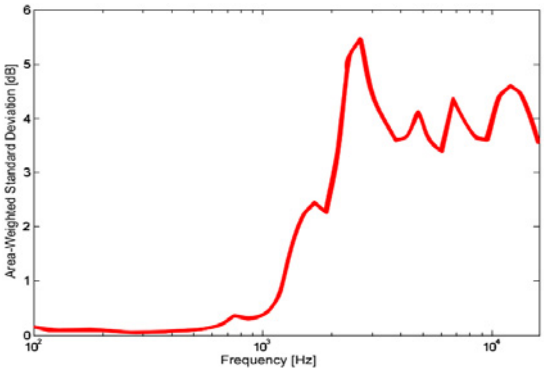

Figure 3: Narrowband area-weighted spatial standard deviations for the dodecahedron source under test.

For each frequency band, it was possible to obtain the distributions for the whole set of measurements. The typology of the distributions over 1 kHz were proved to be non-Gaussian (Figure 4), thus suggesting the idea of using Monte Carlo Simulations.

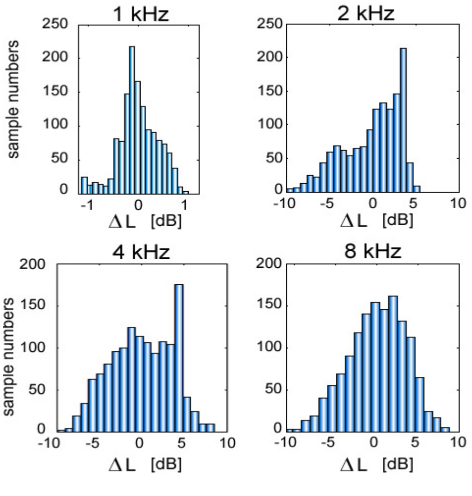

Figure 4: Histograms of the mean deviation of levels for three frequency bands respect to the sample numbers.

The Model

The idea is to model the radiation of the sound from loudspeaker by means of a geometrical model, based both on the Image Sources Method (ISM) [6] and Radiosity model [7]. The temporal distribution of reflections and sound decay can be expressed by the following equation, which is often used in statistical reverberation theory:

$$E(t) \approx E_0 e^{(-\bar{n}\alpha)t} = E_0 e^{-13.8\cdot\frac{t}{T}} \qquad (2)$$

where n the total average of reflections which a ray with a given direction undergoes per second is [8]. This formula is valid under several hypotheses: no medium absorption occurs ($m = 0$); the room is a cube with side l ($S=l^2$, $V=l^3$); α is small compared with unit, Sabine's formula is valid. Eq. (2) describes the behaviour of the sound field as it is supposed to be at the output of one omnidirectional source. The usefulness of this equation results both in its compactness and its dependence on the reverberation time, T. Therefore, several echograms related to different rooms can simply be produced by changing the reverberation time.

In order to completely describe the propagation of sound in space, the model has to account for the diffuse sound field, thus allowing the total sound field (i.e. the echogram profile) to be split into its specular and scattered components. Totally diffuse reflections from a wall can be described by Lambert's law [8]. However, in practical situations only a certain fraction s will be reflected in a diffuse manner while the remaining fraction 1-s will be specularly reflected [7]. Therefore, a mixed specular-diffuse scattering model better satisfies the requirements of the present study. The exponential decay curve can be thought as made up of a specular and a diffuse contribution. The specular part of reflections, while playing a primary role for the first orders of reflections, rapidly decreases as the orders increase, as shown in Figure 5.

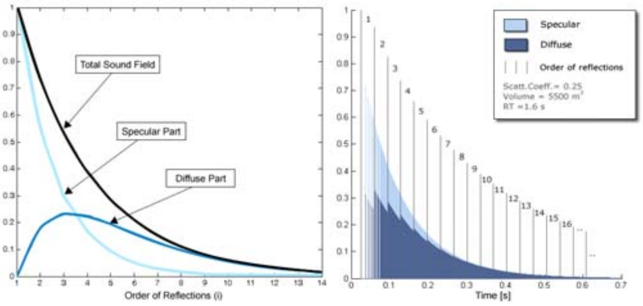

Figure 5: Conversion of specular into diffuse sound energy during subsequent reflections in equal time interval and per each reflection (right).

This happens because after the first reflections, diffuse reflections progressively occur and the conversion of specular sound energy into diffuse sound is irreversible. In other words, for reflection some of the energy of an incident sound ray is converted into non-specular sound, but reflection of non-specular energy will never result in the formation of a single sound ray.

Plots of Figure 5 were obtained by splitting the components of the sound field according to the following equation [9]:

$$E_{tot}(t)=E_{spec}+E_{scat}=E_0\left[e^{\bar{n}[\ln(1-\alpha)(1-s)]t}+e^{\bar{n}[\ln(1-\alpha)s]t}\right] \quad (3)$$

where s is the so called scattering coefficient, which is defined as the ratio of the non-specularly reflected sound energy to the totally reflected energy [10]. The directivity of each measurement point of an omnidirectional source will be modelled with this method. Moreover, Monte Carlo Simulations, programmed for evaluating the uncertainty of the dodecahedral source, will implement this profile, suitably modified for taking into account the random distribution of levels assumed by each point.

Monte Carlo Simulations (MCS)

MCS fit the purpose of this research, where the distribution of the model output of an omnidirectional source's directivity, with many unknown distributions as input, is investigated. This situation complies with the law of propagation of uncertainty where, though the input quantities present irregular distributions, the output can be reasonably characterized by its expectation value and uncertainty. The model is based on the idea that each reflection of an echogram corresponds to a discrete direction of the loudspeaker (Figure 6).

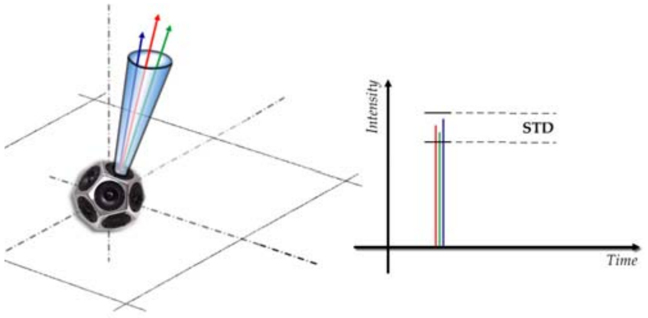

Figure 6: Each discrete direction of the loudspeaker, comprehensive of its deviation from the mean, corresponds to a discrete reflection of the echogram.

Those directions present a deviation from a mean value which has been estimated in terms of level distributions. In order to have a global view on how these deviations affect the final measurements (i.e. the impulse response), a random value, extracted from the distributions of the sampled data (which is the deviation from the mean of the sound pressure level, $\Delta L_{m,n}$), is algebraically added to each of the reflections of one echogram. In particular, each direction of the loudspeaker is characterized by a std, (eq. (1)), and a variance of the sample mean, given by

$$\sigma_{SM}^2=\frac{\sigma_{AWL}^2}{n} \quad (4)$$

This equation follows the central limit theorem [11]. The proper value for n should be the number of population samples (1368 in this case). Problems arise from different weighting that each point has on the resulting sound power captured by a hypothetical receiver. As the distance between the source and one point increases, the significance of its contribution should be lower, thus suggesting the choice of a different value for n instead of 1368. Different values have been used in the simulation, since a theoretical framework for this parameter has not been found yet. Final results will show that n is an important tuning element of the model.

Recalling eq. (3), it is reasonable to apply the std to the specular part of the sound field and the variance of the mean to the diffuse part. This operation leads to the generation of an echogram where reflections are not regularly decaying at an exponential rate; instead, they present a variable deviation, according to the directivity distribution of the loudspeaker. Each generated echogram will present a unique sequence of levels, thus imitating the statistical variation that physical processes usually assume in nature. By iterating this core algorithm, a "synthetic" representation of the behaviour of one omnidirectional source is obtained.

Model implementation

Starting from a set of data as obtained from measurements in anechoic room, the simulation allows to obtain (i) a statistical characterization of the loudspeaker in terms of the Leishman method, (ii) a set of synthesized impulse responses, modelled both according to the physical characteristics of an environment and to the sample distributions of the loudspeakers, (iii) and finally the std due to the deviation of loudspeaker's directivity. The final comparison is realized between the output of the model environment and output of the real environment, both of them being a normalized std.

The model environment follows two paths: the first generates a simulated impulse response, while the second is a massive data processing starting with the acquisition of the anechoic data of loudspeaker's directivity and ending with the distributions for each frequency band. These paths then merge together in the MCS, which pick samples randomly from the distributions and add them to each reflection taken from the synthetic echogram. After some iteration, this process produces the std in the impulse response, which depends only on the inhomogeneous directivity of the omnidirectional source. Figure 7 further exemplifies the structure of the algorithm.

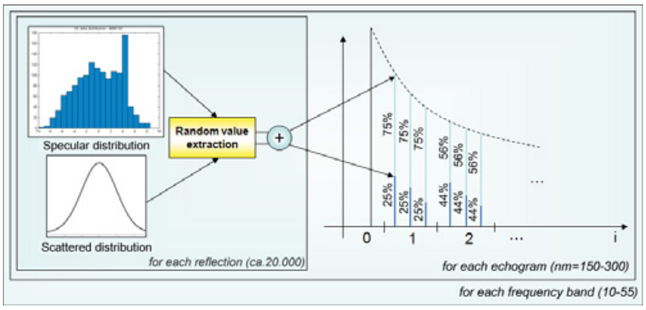

Figure 7: Intuitive illustration of the MCS core.

Results

The behaviour of the assembly Hall of RWTH Aachen University was assessed starting from the impulse responses as collected in six measurement points. For each point, 36 measurements corresponding to the rotation of 10 degrees of the omnidirectional source were recorded. This set of data was later processed in order to obtain the std of the measurements divided by the mean energy.

Similarly to the real environment, the std of the model against the mean energy has been obtained through a computer simulation (250 iterations of MCS for a 2 kHz frequency).

Results from the real environment and from MCS are given in Figure 8. Comparisons are on the basis of std normalized with respect to the mean energy. The main curve profile is similar. Both curves present one peak below 10 ms and have a decreasing decay. The main differences lie in the amplitude dimension, in the speed of decrement and on the amount of details, which are less in the model. The curves produced by the model have a quasi-flat behaviour in the last tail. It is worth mentioning that the variations which the model curves undergo mainly follow a change in *n*. In fact, the two curves on the right differs only for *n*, which is respectively one (lower curve) and ten.

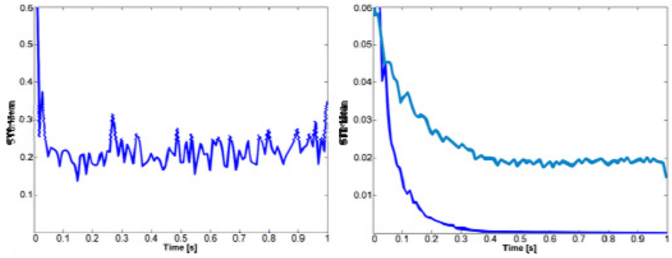

Figure 8: Comparison between the std to mean energy ratio of the real (left) and model environment (right) for a frequency band of 2kHz. The two curves in the right figure correspond to a different value for n, one (lower curve) and ten (upper curve) respectively.

Conclusions

A model to quantify the influences of an omnidirectional source's directivity in room acoustics has been proposed. Starting from measurements conducted on omnidirectional sources in anechoic chamber, the model predicts the amount of the deviation, i.e. the uncertainty, introduced by this element into the measurement chain. Although in its first formulation, results show that the model is performing well. The variations detected by measurements in a real environment, as depending exclusively by the omnidirectional source, seem to be properly quantified.

Mismatching may be improved by increasing the complexity of the model. Upgrades include the removal of some simplifications within the theoretical framework (i.e. the assumption of a cubical room, the mean attenuation neglected), as well as the analysis of different room geometries to be tested. Moreover, attention should be paid to a correct estimation of the initial part of the echogram, specifically the gap of time that elapses between the direct sound emission, the first reflection and the following reflections. Finally, the definition of the distribution for the diffuse part of the sound field may need further investigations. An important tuning element is represented by the denominator of the variance of the sample mean, that is the number of elements that influence the scattered part of the sound field (*n*). In conclusion, the model proposed here is a noteworthy starting point for creating a more complex framework, which could be able to estimate beforehand the perturbations introduced by the room acoustic measurement chain.

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