Room impulse response measurement with a spherical microphone array, application to room and building acoustics

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ABSTRACT

Room impulse response measurement using deterministic signals like sine sweeps is a well-established method to obtain objective parameters that describe the acoustic field in 3-dimensional space. Combined with conventional delay-and-sum beamforming, it becomes a very powerful tool offering precise information about the behaviour of acoustic waves inside or between rooms. More precisely, the use of a transparent array allows us to process the microphone's signals directly by spectral division, a linear deconvolution method, and to analyze the resulting room impulse responses through beamforming. This permits a precise localization of the direct sound and the early reflections over time and space. Additionally, the high signal-to-noise ratio and decorrelation properties offered by the method permit to highlight leakage and airborne sound transmission paths between rooms. Finally, the repeatability of the method allows for a comparison of measurements of various room configurations, for example in the case of acoustic treatment and optimization.

Keywords: 3D Impulse response measurement, spherical array beamforming, room acoustics, airborne sound transmission, reflection, reverberation, localization, insulation.
I-INCE Classification of Subjects Number(s): 74.6, 74.7

1. INTRODUCTION

The measurement and analysis techniques of a sound field's 3D properties have been subject to intensive development and experienced a significant progress over the last years, in application to the acoustic characterization of theatres (1, 2, 3, 4). Mainly, two approaches have emerged. The first one uses sound intensity techniques to obtain the local characteristics of the 3D acoustic field at the listener position, an example of this approach is the remarkable work of Lokky et al. (4). The other approach is based on far-field acoustic attributes (beamforming in time or spherical-harmonics domain) and is primarily specialized to localize sound sources and their reflections in 3-dimensional space (1, 2, 3, 5). In short, this method may be seen as the applicative counterpart of ray-tracing simulation software. The resulting acoustic map is usually superimposed to panoramic pictures or a computer-added-designed model of the place under investigation.

The study presented in this paper is based on a system belonging to the second category. In the first part, a brief description of the measuring method and the signal processing applied will be described, in the second and third parts, application measurements and results will be presented with an emphasis on localization of room impulse response reflections and airborne sound transmission between rooms through a single partition. As a conclusion, advantages and limitations of the method will be pointed out.

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2. SIGNAL PROCESSING AND MEASUREMENT TECHNIQUE

2.1 Signal processing

The schematic 1 represents the measurement principle of the system. As mentioned before, the processing is rather straightforward and can be split into two sections: At the first section, a linear deconvolution is performed between the N signals captured by the microphones and the original signal sent to the room (impulse response measurement as described in ISO-18233 (6)). The N resulting impulse responses are then processed by the beamformer. The algorithm used at this stage can be executed in the time or frequency domain, and advanced algorithms based on cross-correlation and spacial coherence (i.e. CleanSC) can be used to attenuate the artifacts of the beamforming and improve the dynamic ratio of the mapping.

For the beamforming device to be fully functional, the traveling delays between each microphone position of the array and the reflecting surfaces of the three-dimensional space under investigation have to be computed with great precision. In order to achieve this requirement, the geometrical characteristics of the space are recorded using a laser scanner (additionally to the acoustic measurements), and the position and orientation of the microphone array relative to the scan is determined by fitting a selected number of points of the 3D scan to a picture delivered by an optical camera fixed to the array.

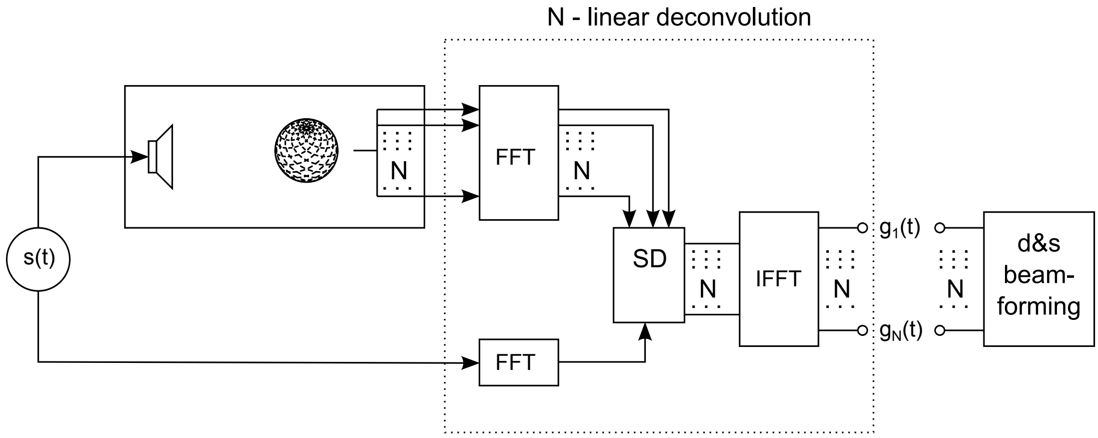

Figure 1 – Measurement principle: the loudspeaker excites the room with a sine-sweep signal which propagates and reaches the N microphones of the array. The linear-deconvolution stage is performed via regularized spectral division (SD) of the Fourier transform (FFT) of the N signals coming from the array by the Fourier transform of the original signal sent to the room. After an inverse-FT (IFFT), the resulting impulse responses are processed by delay-and-sum (d&s) beamforming.

2.2 Hardware and measurement setup

The measurements were performed at the acoustic laboratory of the Institut of Fluid Mechanics and Technical Acoustics Technical University Berlin (ISTA at TU-Berlin). The laboratory is accredited by the *Deutsches Institut für Bautechnik* (DIBt, German Institute for Civil Engineering) according to DIN-4109 (7). It comprises an anechoic and a reverberation chamber, as well as a test transmission suite for small partitions like doors or windows.

A first measurement session was performed in the reverberation chamber inside which, as recommended by the standards, hanging panels have been installed to achieve a high degree of diffusivity. The array, a sphere of 60 cm diameter with 120 microphones, was placed at a height of 1.2 m above the chamber floor and the source was a dodecahedron mounted at approximately 1.4 m above the floor (measurement setup depicted Fig 2). A first series of measurements was performed, then, without moving the source and the array, the floor of the chamber was covered by absorbing material and a new series was performed. This made it possible to perform comparative measurements and to verify if the method could be used to detect the amount of acoustic energy that reaches the floor.

To try out the method on detecting sound leakage through a single partition, a second measurement session was carried out in the test transmission suite for small partitions. A sheet made of glass fiber reinforced plastic (GFRP) was mounted inside an aperture in the wall between source and receiving rooms. The array used was a ring of 75 cm diameter with 48 microphones. This made it possible to perform comparative measurements and to identify and localize potential noise leaks through the partition. In all measurements, a sine sweep of 18 s was used as basis material to excite the object under test and the impulse response was computed by linear deconvolution (sound transmission between rooms measurement also described in ISO 18233(6)).

Figure 2 – Measurement setup: microphone array and dodecahedron in the reverberation chamber.

3. COMPARISON OF MEASUREMENTS

3.1 Early decay (first and secondary order reflections)

Fig. 3 depicts a mapping of the early decay of impulse response (5 to 60 ms after the direct path) inside the reverberation chamber with and without absorbing material. Just after the impulse (transient conditions) the sound field is not diffuse yet, and direct path and early reflections can be isolated spatially and chronologically. The picture on the right reveals a dense and uniform distribution of reflections over the walls, ceiling and hanging diffusers of the chamber. The first order reflections (on the back and side walls and on the top hanging diffuser panel) are identical on both measurements (in time, space, and maximum energy). The picture on the left reveals a much more disparate distribution of reflections: the second and third order reflections that reached the floor have been attenuated by the absorption material before reaching the array (with a much lower energy).

3.2 Reverberation tail

Figure 4 depicts a mapping of the late reverberation inside the chamber (from 70 ms after the direct path) with and without absorbing material. The analysis with the beamforming device reveals a homogeneous energy distribution on the walls of the chamber. If absorbing material is placed on the floor of the chamber, almost no acoustic energy is coming from the floor, as it could be expected, and the early reflections take a much higher significance in the composition of the acoustic field. The measurement with absorbing material also reveals that the reflections at the height of the source are particularly predominant in the late part of the impulse response. This indicates that most of the sound waves propagating horizontally in the chamber did not reach the floor and tend to remain in the horizontal plane. Placing diffusers at the height of the source should resolve this problem.

4. SOUND LEAKAGE DETECTION

The second measurement session took place at the window test bench for airborne sound insulation of windows, doors and small building components. To be able to produce different and comparable conditions, a first measurement was performed, then an aperture was drilled through the sheet and a tube of PVC of 0.86 m (to get a resonance frequency around 2 kHz) was placed in the opening and used as acoustic filter. The microphone array and the source positions remained unchanged. Figure 5 depicts the result of a frequency domain beamforming between 2.4 and 6.4 kHz with and without an opening inside the sheet. The picture on the left (measurement before drilling) reveals a sound leakage at the upper right corner of the sheet. The right picture (after drilling) highlights only the sound path through the opening, and the leakage does not

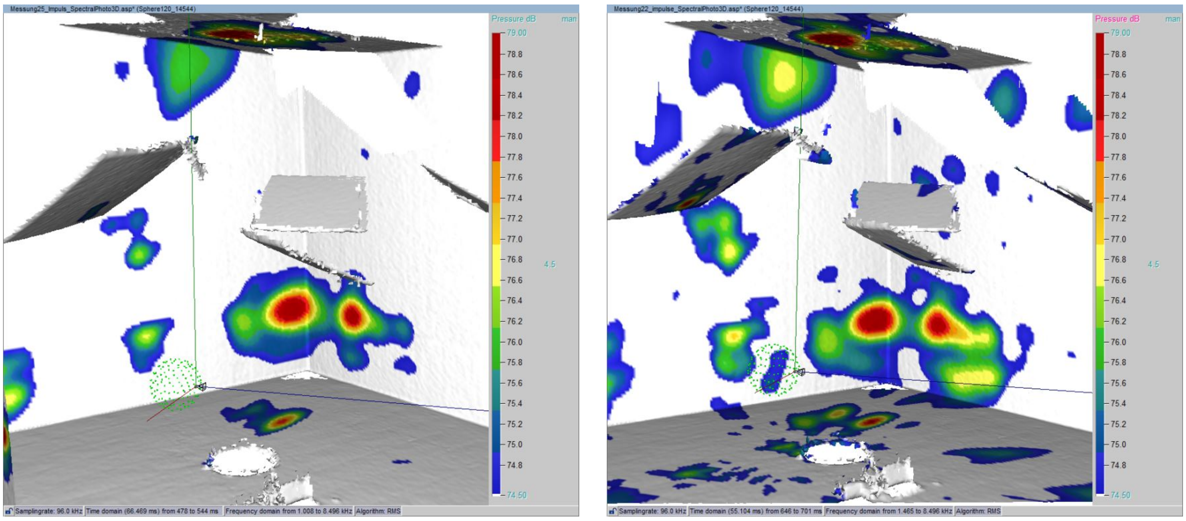

Figure 3 – Mapping of the early decay part of impulse responses (5 to 60 ms after the direct path) with and without absorbing material (left and right images respectively). The color scales of the mappings are set to 4.5 dB (re 20 µPa) dynamic. Frequency domain beamforming from 1.5 to 8.5 kHz.

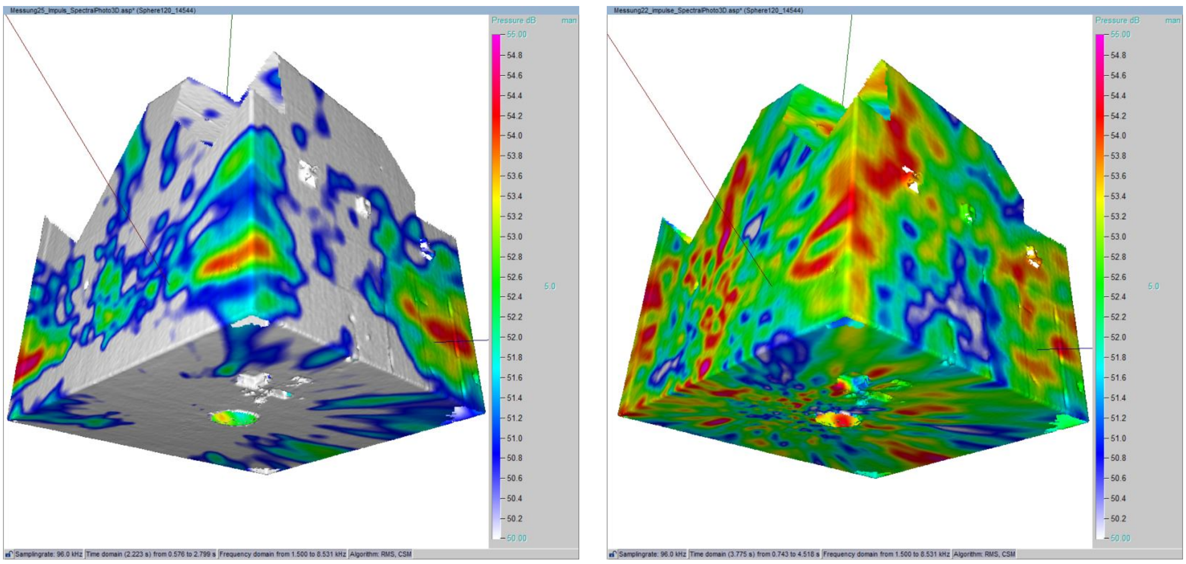

Figure 4 – Mapping of the reverberation tail (from approx 70 ms after the direct sound has reached the microphone array) with and without absorbing material (left and right images respectively). The color scales on the mappings are set to 5 dB (re 20 µPa) dynamic. Frequency domain beamforming from 1.5 to 8.5 kHz.

appear anymore. This is because the amount of acoustic energy traveling through the opening is higher than the dynamic range offered by the beamforming technique.

The pictures reproduced (Fig 6) depict a mapping derived from a single measurement (GFRP sheet before drilling). The only difference lies in the processing of the data: the left picture depicts the mapping of the impulse responses as previously described (section 2.1), whereas the right picture depicts the mapping of the original measured data (the linear deconvolution stage has been omitted in the processing). Important differences in the acoustic energy distribution are revealed: in the right picture, the leakage on the right upper corner almost vanished behind a big spot in the middle of the sheet. This means the plate acts as a non-linear secondary sound source, and most of the non-linearities between the original and the measured signals have been removed at the linear deconvolution stage (8). As a consequence, the acoustic energy propagating to the receiver room through the vibrations of the plate are not present in the impulse response.

This results make the method particularly interesting to differentiate between linear and non-linear sound transmission between rooms and between structure born and airborne sound transmission. Moreover, it is clear that a part of the acoustic energy has been traveling through a nonlinear path (has undergone nonlinear

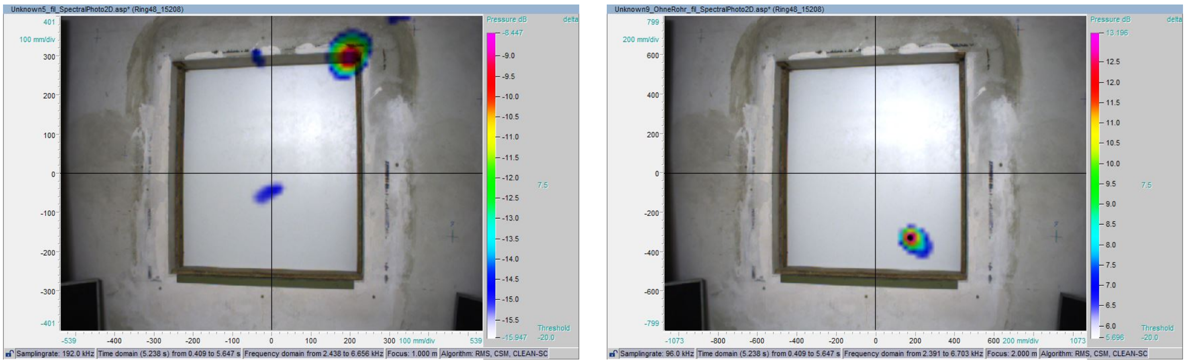

Figure 5 – Measurement of a GFRP plate mounted on the test bench. A leakage can be detected at the right top corner. Frequency domain beamforming from 2.4 to 6.7 kHz

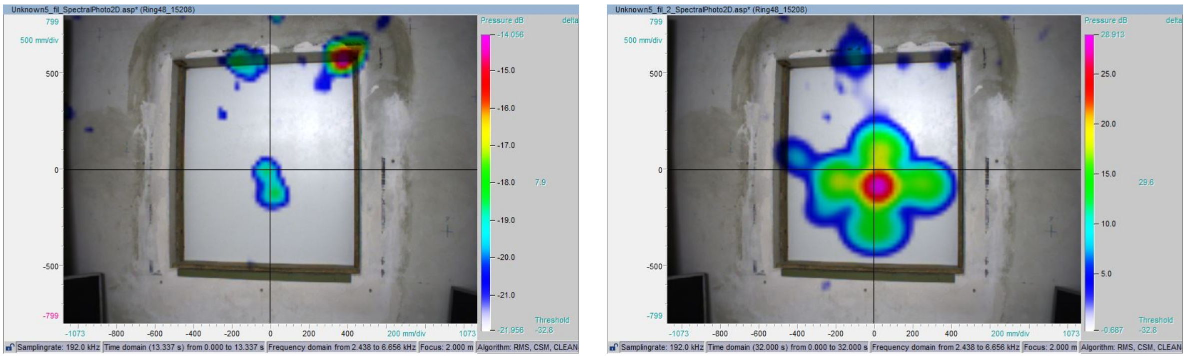

Figure 6 – Frequency domain mapping from 1.8 to 2.4 kHz from the GFRP sheet without out opening. Left: a linear-deconvolution is performed before processing the data by the beamformer; right:the mapping is performed directly from the measured data.

transformations) before entering the receiver room, and this part of the energy will be removed at the deconvolution stage. This means that the computation of the level difference between rooms using linear deconvolution (as defined in ISO18223):

$$D = L_1 - L_2 = 10 \cdot \log \left[\frac{\int_0^\infty h_1(t)^2 \, \mathrm{d}w}{\int_0^\infty h_2(t)^2 \, \mathrm{d}w} \right] dB \quad (1)$$

will give erroneous results. It can be verified simply by measuring the level difference using conventional methods with a sound level meter but this is outside the scope of this study.

5. CONCLUSION

The measurement of room impulse responses using sine-sweep with a microphone array can be used to analyze the acoustic field inside a 3-dimensional space with great precision. This method allows a time as well as a spatial separation of the elements of the impulse response.

The processing at the linear deconvolution stage eliminates almost all harmonic distortions between the original source signal and the recorded one, which makes the method particularly interesting for measurements of high quality room impulse responses. It is also well applicable to detect sound leaks which appear e.g. at the connection between walls and windows and doors or at any air gap between partitions. Most of the energy part traveling through the structure of the partition will be eliminated, and the linear leakage can be highlighted. However, most of the noise nuisance in buildings is traveling through the structures at frequencies lower than the threshold frequency permitted by the beamforming technique. Lowering this frequency can be the subject of further development by combining for example sound pressure and intensity analysis, or by using frequency transposition techniques.

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