Room impulse response measurement and delay-and-sum beamforming, application to room and building acoustics.

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Summary
Room impulse response measurements using deterministic signals like sine sweeps is a well-established method used 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 behavior of acoustic waves inside or between rooms. The use of a transparent array allows us to process directly the signal captured by the microphones by deconvolution, and the resulting room impulse responses by the beamformer. This permits a precise localization of the direct sound and the early reflections over time and space. Additionally, the high signal-to-noise ratio offered by the method permits 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.

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1. Introduction

The measurement and analysis techniques of a sound field's 3D properties in application to the acoustic characterization of theatres [1, 2, 3, 4] have been subject to intensive development and experienced a significant progress over the last years. Especially, 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 dedicated 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 on panoramic pictures or 3d-models of the object 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 measurement 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.

2. Signal processing and measurement technique

2.1. Signal processing

Schematic 1 represents the measurement principle of the system. As mentioned before, the processing is rather straightforward and can be split into two sections: the first section is based on the impulse response measurement as described in ISO-18233 [6]: a linear deconvolution is performed in the frequency domain (equation 1) between the N signals captured by the microphones $y_i(t)$ and the original signal $x(t)$ sent to the room).

$$h_i(t) = IFFT\left[\frac{FFT(y_i(t))}{FFT(x(t))}\right] \qquad (1)$$

The second section is a conventional delay-and-sum beamforming with the N resulting impulse responses

($h_i(t)$). 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 range 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, 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.

3. Comparison of measurements

3.1. Hardware and measurement setup

A first series of 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 carried out in the reverberation chamber inside which, as recommended by the standards, hanging panels were 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, a Nor276 Dodecahedron loudspeaker (Norsonic), was 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.

3.2. Early decay (first and secondary order reflections)

Fig. 3 depicts a mapping of the early decay of the impulse response (5 to 60 ms after the direct path) inside the reverberation chamber without absorbing material. Immediately 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 analysis reveals a dense and uniform distribution of reflections over the walls, ceiling and floor of the chamber.

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

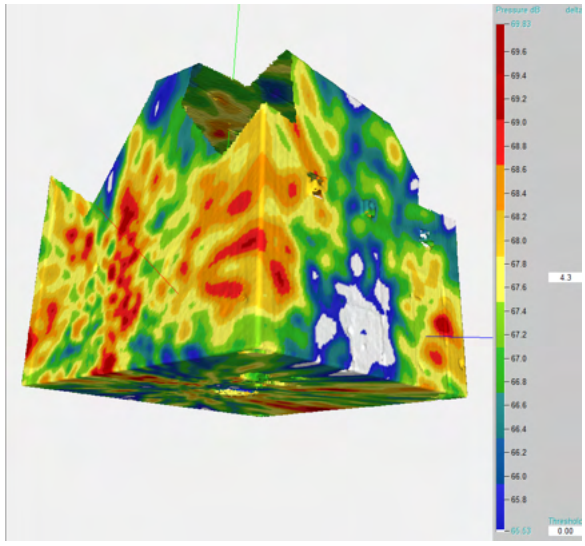


Figure 3. Mapping of the reverberation tail (from approx 70 ms after the direct sound has reached the microphone array) without absorbing material

3.3. Reverberation tail

Figure 5 depicts a mapping of the late reverberation inside the chamber (from 2 s after the direct path) 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, the reflections at the height of the source appear particularly predominant in the late part of the impulse response (Figure 4). 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.

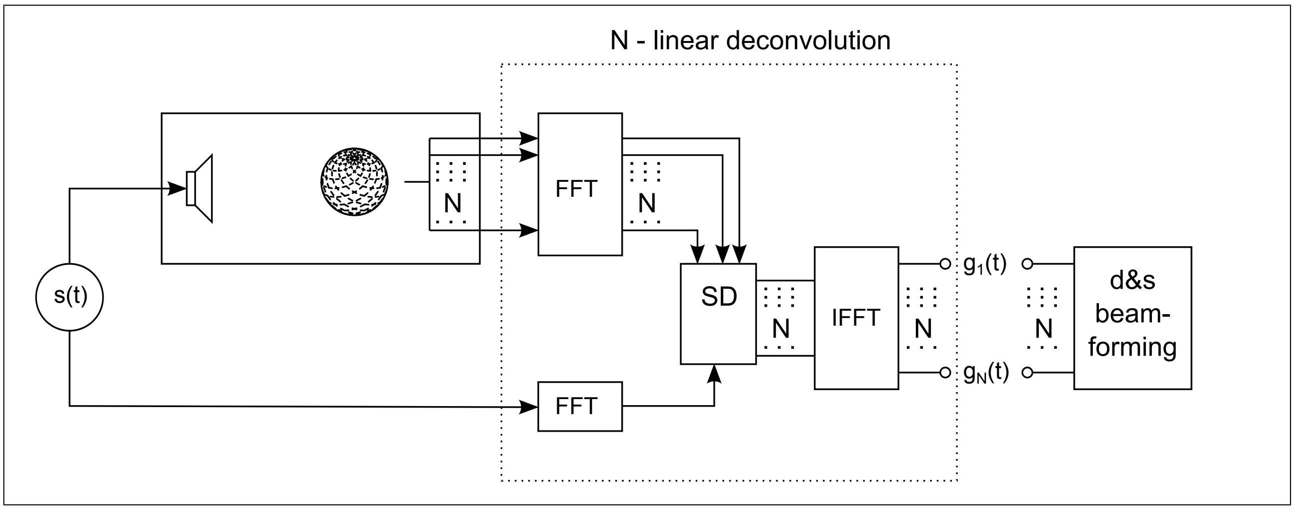


Figure 1. Measurement principle: the loudspeaker excites the room with a sine-sweep signal which propagates and reaches the N microphones of the array. The linear-deconvolution stage is performed via regularized spectral division (SD) of the Fourier transform (FFT) of the N signals coming from the array 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.

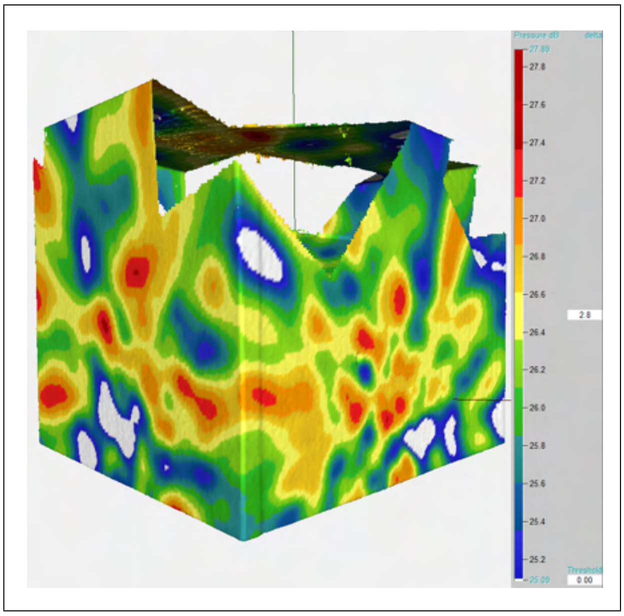


Figure 4. Mapping of the reverberation tail (from approx 2 s after the direct sound has reached the microphone array) without absorbing material. The color scales on the mappings are set to 2.9 dB (re 20 µPa) dynamic. Frequency domain beamforming from 2.5 to 5.3 kHz.

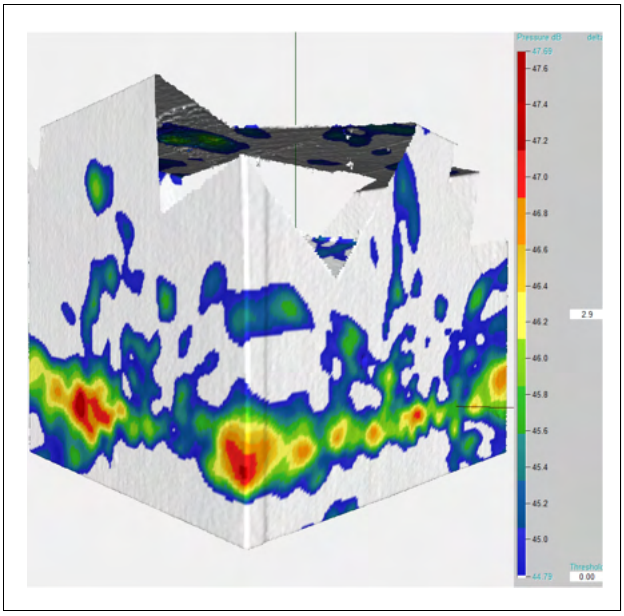


Figure 5. Mapping of the reverberation tail (from approx 2 s after the direct sound has reached the microphone array) with absorbing material. The color scales on the mappings are set to 2.9 dB (re 20 µPa) dynamic. Frequency domain beamforming from 2.5 to 5.3 kHz.

4. Room for Reception

4.1. Measurement setup

A second measurement session was carried out to analyze the acoustic characteristics of two rooms (named *Arnold* and *Conrad*). To allow measurement comparison, they have been selected of identical size and configuration, only separated by a movable wall. Only one room (*Conrad*) has been treated acoustically to allow measurements comparison.

The rooms are used for meetings, congresses, symposiums, weddings and other events. The multipurpose character of the rooms poses a challenge when the time comes to decide how to judiciously place acoustic treatment in the rooms. Reaching a good acoustic transmission between a stage with speakers or a small group of musicians and an audience as well as, at the same time, maintaining good acoustic conditions even if a large number of guests want to communicate with each other. It was decided to keep intact as much as

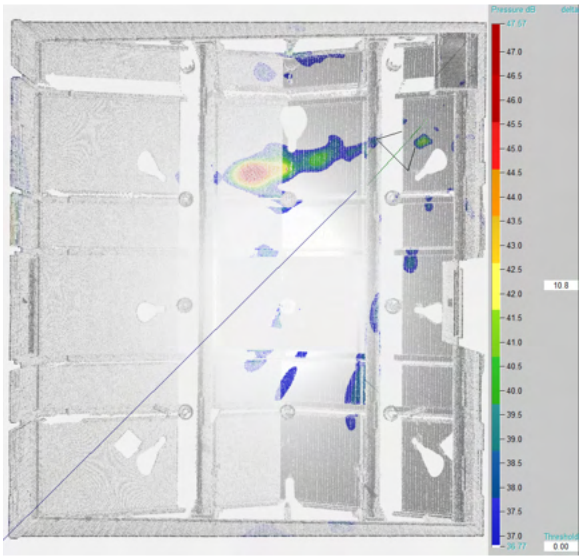


Figure 6. Ceiling of the Room *Arnold*, Mapping of the Early reflections (50 ms after the direct path) (Conrad - with absorbent material.)

possible the early reflections and solely treat the late reverberation. Thus, preserving the energy transmission between the stage and the audience while reducing the overall noise level by shortening the reverberation time.

Localizing the early reflections between the stage and the audience has been done with a series of measurements using the Acoustic Camera. Figure 6 depicts a mapping of the early reflections (50 ms after the direct path) of room *Arnold* (no treatment).

4.2. Results

4.2.1. Reverberation Time

The reverberation time of both rooms has been computed from the impulse responses measured with the Acoustic Camera. The results reveal a notable decrease in reverberation time, especially in lower and middle frequencies (Figure 7). Additional measurements have been performed with a sound level meter with type approval (Nor131 from Norsonic) for verification. The reverberation time was measured in four measurement points in each room. The sound source used was a blank pistol. The results obtained with the SLM and the Acoustic Camera are remarkably alike. The results obtained with the SLM show a lower absolute error in higher frequencies than in lower frequencies. Nevertheless, the absolute error of the measurements done is less than 2% in those measured in room *Arnold* and around 6.5% in room *Conrad*.

It is to be noted that the results obtained from the Acoustic Camera show a dramatic increase at the frequencies above 8kHz and should be considered erroneous. This was due to the limited frequency bandwidth of the Audio board used during the measurement.

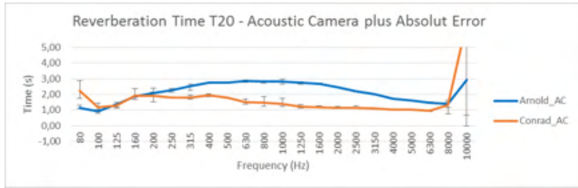


Figure 7. Reverberation Time in both rooms, with the absolute error for each third-octave band - Acoustic Camera

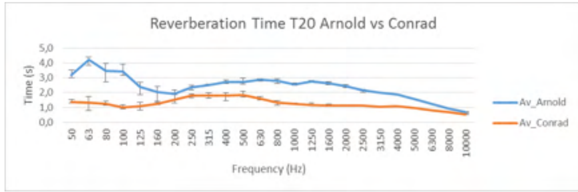


Figure 8. Reverberation Time in both rooms, with the absolute error for each third-octave band - Sound Level Meter

Table I. Parameters recommended by the ISO 3382, measured in each room.

	$Arnold_{Av}$	$Conrad_{Av}$
EDT (s)	2.45	1.15
Definition (%)	24.99	44.73
C80 (dB)	-2.41	1.92
C50 (dB)	-4.78	-0.92
Center Time (s)	0.22	0.43

4.2.2. Additional acoustic parameters

The Acoustic Camera offers the calculation of additional room acoustic parameters recommended by the standard ISO-3382 [8]. Table I shows the improvement of the speech parameters (Clarity and Definition) after the application of the acoustic treatment inside room *Conrad*.

4.3. Leakages and Apparent Sound Reduction Index

Leakages often stem from imperfections made during the design or the construction phase of a building. Leakage tests are usually complicated and time consuming, but using a beamformer helps simplifying this process to a single measurement and the results can be quickly and clearly displayed over an acoustic map. Figure 9 depicts the result of a full-range frequency analysis. The results highlight the left extremity of the movable partitioning wall as well as a door in

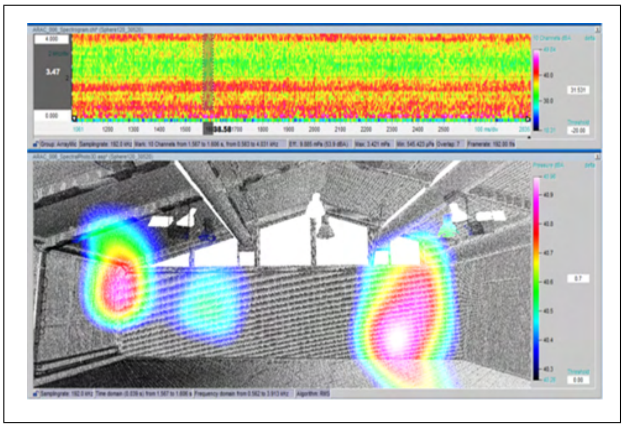
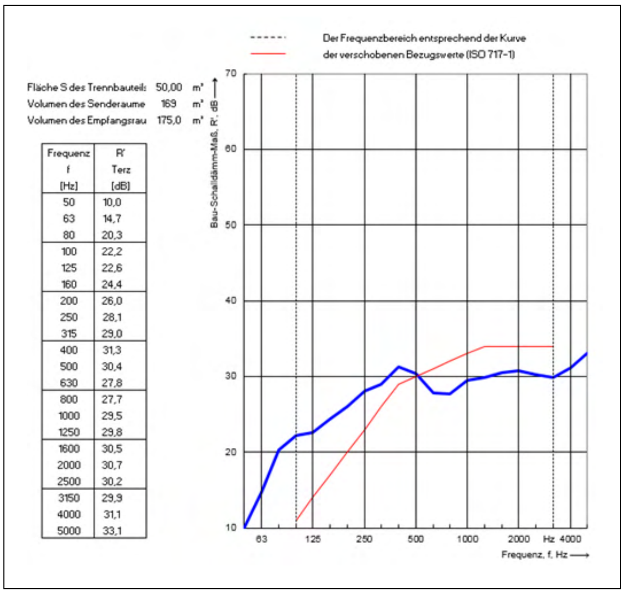


Figure 9. Leakage detection, wide frequency band.



Frequenz f [Hz]	R' Terz [dB]
50	10,0
63	14,7
80	20,3
100	22,2
125	22,6
160	24,4
200	26,0
250	28,1
315	29,0
400	31,3
500	30,4
630	27,8
800	27,7
1000	29,5
1250	29,8
1600	30,5
2000	30,7
2500	30,2
3150	29,9
4000	31,1
5000	33,1

Figure 10. Apparent Sound Reduction Index report of the movable wall.

the wall (Figure9 spot on the left and right hand-side respectively) as the weakest elements.

As shown in the results of the Apparent Sound Reduction Index depicted figure 10, R' decreases abruptly in the frequency range between 600Hz and 800Hz. When analyzing these frequencies in detail, the spectral decomposition (spectrogram) shows a higher energy level, and the result of the beamformer highlights the end of a ventilation tube above the movable wall (Figure 11).

5. CONCLUSIONS

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.

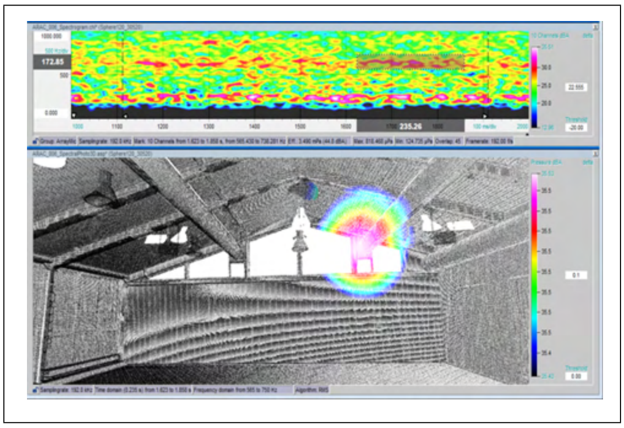


Figure 11. Leakage detection, detail frequencies between 400Hz - 800Hz.

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 with a high signal-to-noise ratio. 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. 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. The use of Minimum Length Sequences instead of sine sweep may also be an interesting research direction as their computation method (the Hadamard transform) is based on the correlation algorithm (in contrary to convolution) and is rejecting all background and extraneous noise occurring during the measurement.

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