BeBeC-2014-06

APPLICATION OF CORRELATION ANALYSES IN 2D AND 3D BEAMFORMING

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ABSTRACT

This article presents an application of beamforming technique using a 3D microphone array and transfer functions measurement with sine sweeps. The calculated room impulse responses (RIRs) can be decomposed into their components with a high time and spatial resolution. Linear spectral deconvolution is used to calculate impulse responses for every channel of a 48-channel spherical microphone array. As an example a conference room was scanned and the reference source and early and late reflections of 1st and 2nd order with 1 ms time resolution were found. The reflections could be precisely located in a 3D model giving completely new insights in sound propagation using a highly reproducible and strictly defined excitation signal.

1 INTRODUCTION

Delay-And-Sum beamforming has manifested itself as a reliable and robust investigation technique for sound source localization over the last 15 years. An important improvement was achieved by using 3D microphone arrays to localize sound sources in 3D with the help of detailed 3D-models [4], [5].

Application of beamforming techniques in the field of room acoustics have been previously presented in [1]. In this paper, the authors use the traditional blasting balloon and alarm pistol as room excitation and the room impulse response (RIR) was directly measured by the microphone array. This direct and fast method has some crucial disadvantages: the pulse excitation generates a non-steady state which does in many cases not correspond to the standard case of application which is to be analyzed and optimized. Moreover these excitations are inherently random regarding excitation energy, room position and spectrum. Doubtless this methods are not convenient to become a new standard method in room acoustics.

The correlation methods that are presented in the following are a big progress because a strictly defined and reproducible excitation signal is used and the RIR is calculated indirectly from the measured data by linear deconvolution.

In our measurement example we show how a RIR can be estimated by the use of a sweep excitation. As we also used a 3D laser scanner to get a detailed room model the RIR can be decomposed in time and space with a very high resolution.

2 LINEAR SPECTRAL DECONVOLUTION

A broadband acoustic excitation $a(t)$ is emitted by a reference sound source. The signal $a(t)$ will be reflected and scattered in the room before it reaches the receiver delayed in time and will be measured as signal $b_i(t)$. So the room can be considered as an acoustical transmission system and the measured result can be described by a convolution of the reference signal $a(t)$ and the impulse response $h_i(t)$ of the room:

$$b_i(t) = a(t) * h_i(t) \tag{1}$$

As we are using arrays of i microphones we measure i signals $b_i(t)$ at i different positions. This leads consequently to i different impulse responses. In the frequency domain this can be written as multiplication of the spectrum of the input signal $A(f)$ and the transfer function (spectrum of the impulse response) $H(f)$.

$$B_i(f) = A(f) \cdot H_i(f) \tag{2}$$

By spectral division the transfer function (and consequently also the impulse response) can be estimated from the reference signal and the measured microphone signal.

$$H_i(f) = \frac{B_i(f)}{A(f)} \tag{3}$$

It is necessary that the frequency range of the reference signal is sufficient. Nevertheless spectral division will produce unstable solutions for real signals that contain noise and distortion. So the implementation of noise suppression and stability limiting is important to yield stable results.

The calculated impulse responses then can be used directly as input data for beamforming in the same way as measured impulse responses. As mentioned above the big advantage of this technique is the possibility to use well defined steady excitation signals. So measurements can be repeated and averaged to increase the signal to noise ratio, or differences can be calculated to illustrate for example the modifications on a surface of a room. As excitation signals non-autocorrelated white noise as well as maximum length sequences (MLS) or sine sweeps can be used. Mueller and Massarani [6] strongly propose the use of sine sweeps because they offer by far the best dynamic range and are relatively tolerant of time variance and totally immune against harmonic distortion. The SNR of a sine sweep measurement is usually 60 dB better in comparison with the generation of a single impulse having the same maximum amplitude [3].

3 PRACTICAL APPLICATION

3.1 Measurement Setup Description

The measurement took place at a conference room with the size of approximately 11 m x 13 m and a height varying between 3.8 m and 2.5 m. The reference source loudspeaker was placed on the stage at 3 m distance to the microphone array. A 3D model of the room was created using a 3D-laser scanner (Fig. 2), which was then used to calculate the 3-dimensional beamforming. We used a 48-channel spherical and acoustically transparent microphone array with a diameter of 35 cm (Fig. 1). The data was recorded for 32 seconds at 192 kHz sampling rate.

As mentioned above we used a logarithmic sine sweep for room excitation. Its length was 30 seconds and its frequency range was 0 kHz to 22 kHz. Fig. 3 shows waveforms of the sinesweep as well as the recorded signal and the calculated impulse responses for every channel.

Figure 1: Spherical 48-channel transparent microphone array used for the RIR estimation.

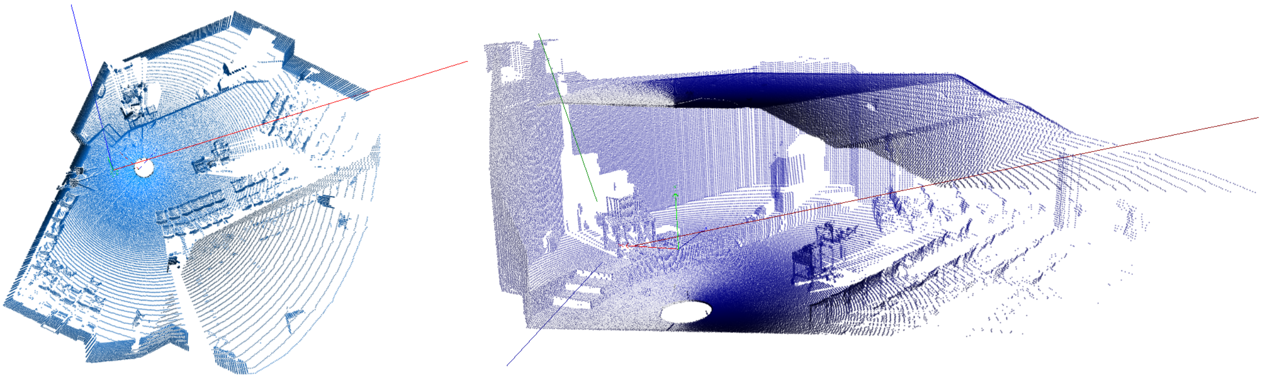
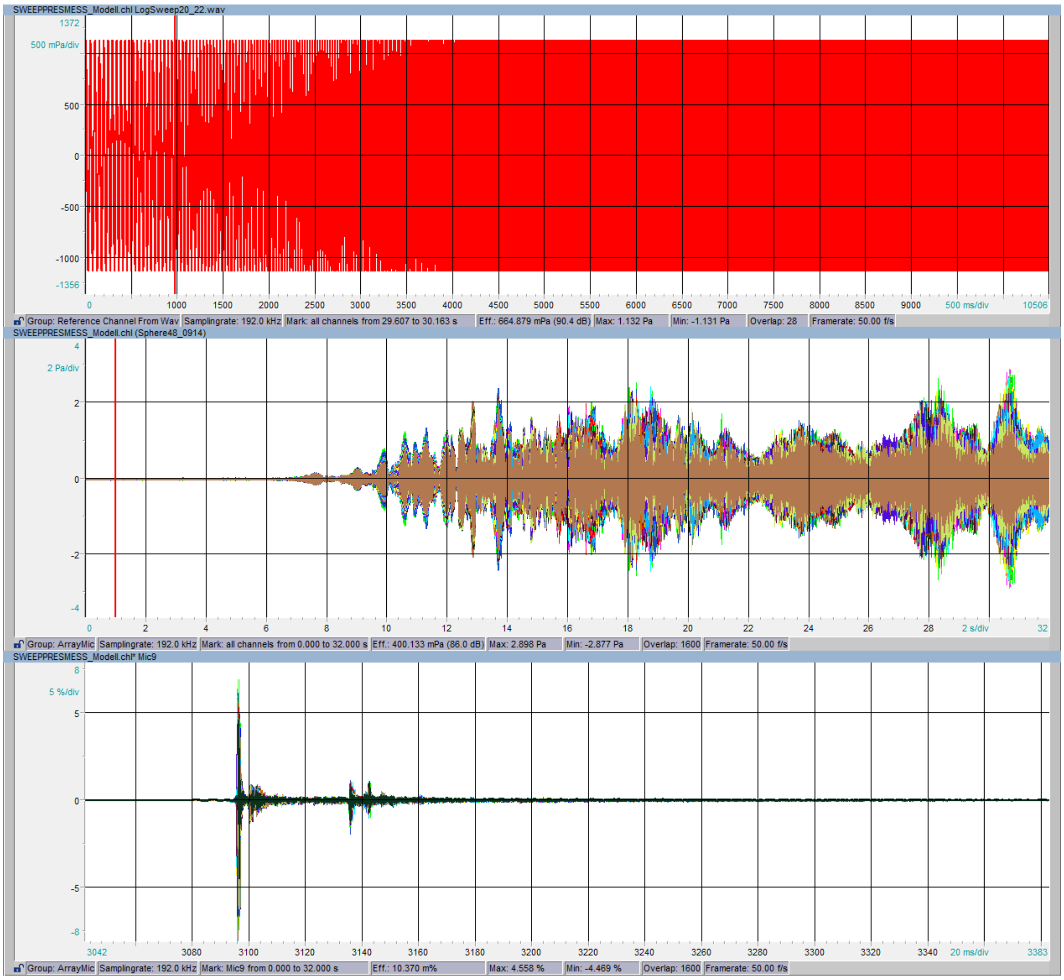


Figure 2: 3D model of the room, with microphone array position (cross section of red and blue line) and speaker on the stage.



Group: Reference Channel From Wav | Samplingrate: 192.0 kHz | Mark: all channels from 29.607 to 30.163 s | Eff.: 664.879 mPa (90.4 dB) | Max: 1.132 Pa | Min: -1.131 Pa | Overlap: 28 | Framerate: 50.00 f/s

Group: ArrayMic | Samplingrate: 192.0 kHz | Mark: all channels from 0.000 to 32.000 s | Eff.: 400.133 mPa (86.0 dB) | Max: 2.898 Pa | Min: -2.877 Pa | Overlap: 1600 | Framerate: 50.00 f/s

Group: ArrayMic | Samplingrate: 192.0 kHz | Mark: Mic9 from 0.000 to 32.000 s | Eff.: 10.370 m% | Max: 4.558 % | Min: -4.469 % | Overlap: 1600 | Framerate: 50.00 f/s

Figure 3: Logarithmic sweep excitation signal (top), measured room response (middle) and calculated impulse response (bottom).

3.2 Results

The calculated impulse responses were localized on the 3D-model. By picking out delay-of-arrival time ranges of just 1 ms, the components of the impulse response can be separated and localized in space. 1 ms corresponds to the maximum runtime delay of the used microphone array with a diameter of 35 cm. To be sure to use only the marked data for the calculation zero-padding technique was applied to all analyses which is described in Ref. [2].

Fig. 4 and Fig. 5 show eight different components of the impulse response localized on the 3d-model. The first peak in the RIR shows the reference speaker as source. (Note the fact that the source is not only mapped to the speaker but also to the wall behind as an effect of the 3d-mapping). It is followed by early reflections from the floor, wall and ceiling delayed by 5 ms to 10 ms. Late reflections from the wall behind the microphone array can be found at delay times of 40 ms, also a second order reflection can be seen at 50 ms.

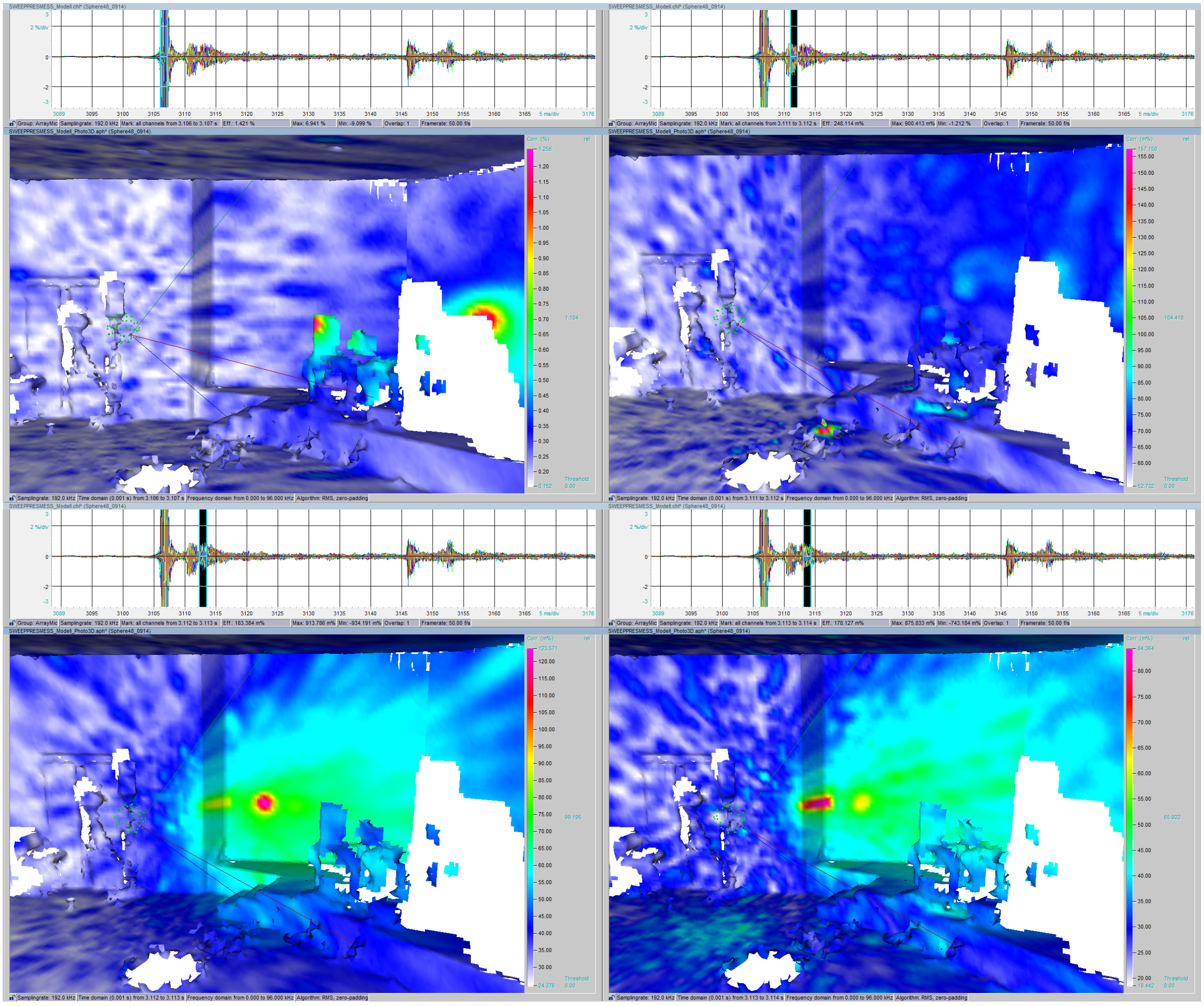


Figure 4: Impulse response of the room showing the source as well as first order reflections at the floor and the wall in the corresponding time of arrival.

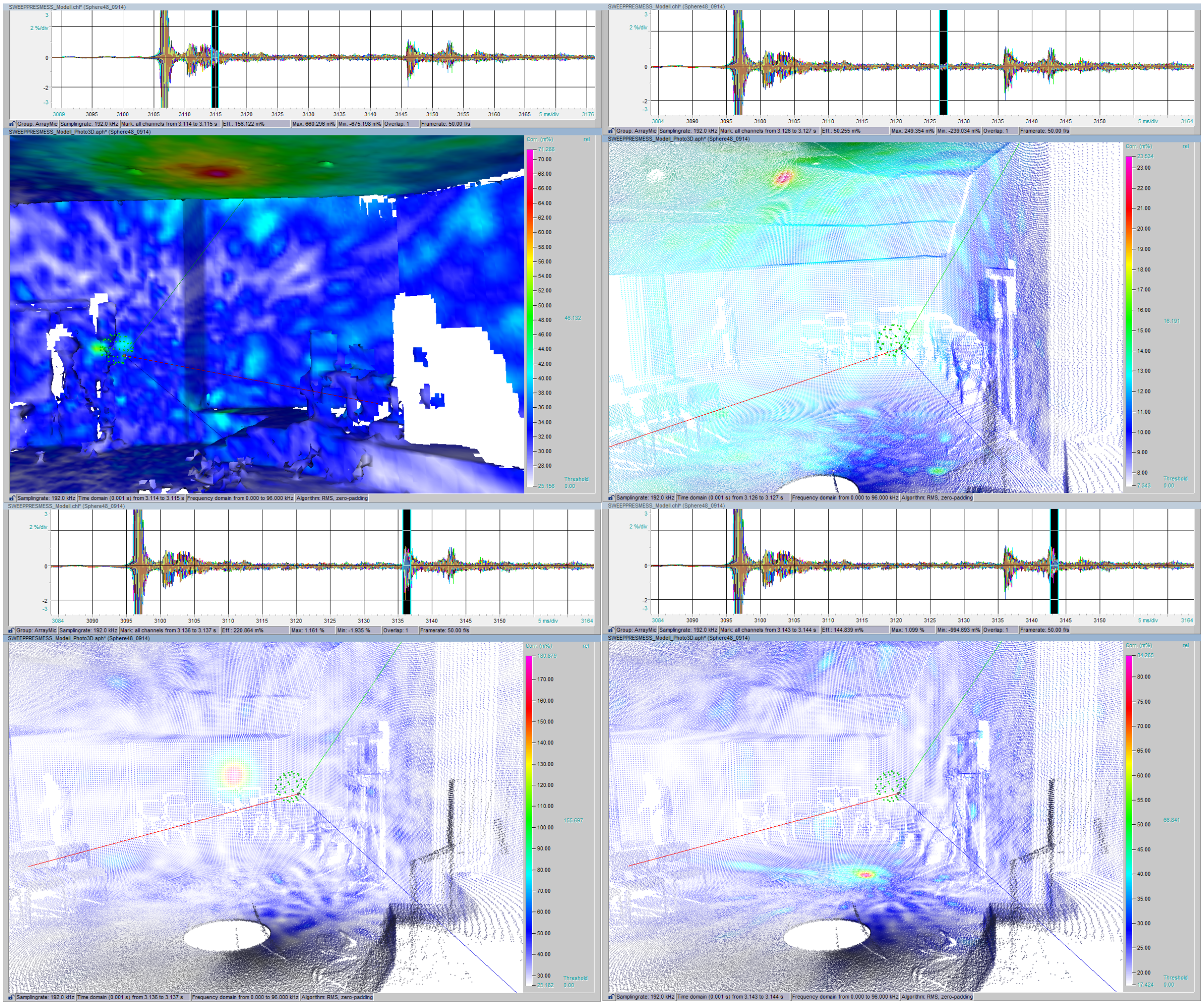


Figure 5: Impulse response of the room showing first order reflections at the ceiling, at the wall and a second order reflection in the corresponding time of arrival.

Of course standard parameters of architectural acoustics can also be calculated from the RIRs. They are given in table 1

Table 1: Architectural Acoustics Parameters, Conference Room.

Parameters	100 Hz	125 Hz	160 Hz	200 Hz	250 Hz	315 Hz	400 Hz
EDT (s)	0.89	0.98	0.94	0.82	0.57	0.88	0.81
T_{20} (s)	1.25	0.86	0.88	0.97	1.2	1.05	1.01
Definition (D_{50} %)	54.42	65.03	49.32	49.3	76.6	71.13	77.54
Clarity (C_{80} dB)	3.45	5.05	3.98	5.71	9.24	6.84	7.77
Center time (s)	0.09	0.08	0.09	0.08	0.07	0.07	0.06

Parameters	500 Hz	630 Hz	800 Hz	1 kHz	1.25 kHz	1.6 kHz	2 kHz
EDT (s)	0.61	0.58	0.54	0.76	0.7	0.67	0.62
T_{20} (s)	0.98	0.92	0.94	1.06	1.09	1.09	1.12
Definition (D_{50} %)	81.31	80.82	83.06	77.28	76.95	80.49	88.45
Clarity (C_{80} dB)	10.09	10.12	10.29	8.6	8.92	9.76	11.19
Center time (s)	0.05	0.06	0.05	0.06	0.06	0.05	0.04

4 CONCLUSIONS

This experiment showed encouraging results. The RIR of the conference rooms could be estimated with a SNR of 79 dB and 1st and 2nd order reflections could be localized in a 3D-model. Of course further progresses have to be made especially concerning the omni directionality of the reference sound source as well as reduction of the noise level. But it has been shown that the combination of 3D-laser scanning technology, beamforming with a 3D-microphone array and RIR estimation by sine sweep excitation can deliver a detailed insight in the time and space dependency of sound propagation mechanisms. This is a very powerful tool, to help acousticians in their work of room and building acoustic optimization. In the future applications of the system will include difference pictures to show the improvement brought by modifications made to a room or to illustrate results of an optimization.

ACKNOWLEDGEMENT

This research work has been funded by the German BMBF under project registration number 03WKP24A.

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