



Measurement of 3D Room Impulse Responses with a Spherical Microphone Array

Jean-Jacques Embrechts

Department of Electrical Engineering and Computer Science/Acoustic lab, University of Liège, Sart-Tilman B28, 4000 Liège 1 (Belgium).

Summary

Directional room impulse responses (DRIRs) are composed of the sound contributions reaching a given location in the room from a well-defined direction in space. DRIRs can be useful in many applications, such as the evaluation of spatial room acoustics parameters, the detection of unwanted specular reflections or the 3D auralization of acoustic spaces. A spherical array containing 16 microphones has been realized to measure DRIRs. The logarithmic sinesweep technique is first applied to measure 16 impulse responses, one for each microphone. A spherical harmonics (SH) decomposition of the sound field is then obtained. Spatial aliasing, placement errors and the ‘white noise gain’ (WNG) have been analysed to define the useful bandwidth of this measure, i.e. [250Hz – 4kHz]. The coefficients of the SH decomposition are then processed by some beamforming methods, in order to compute the DRIR in any direction around the spherical array. Time and 3D space representations can be generated. The results obtained in some rooms are illustrated in this paper: it is shown that the combination of the ‘delay-and-sum’ and ‘minimum-variance distortionless response’ beamforming methods is particularly well suited for the analysis of DRIRs.

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1. Introduction

Room acoustics studies often involve the measurement of room impulse responses (RIR) [1]: the RIR is defined as the response of an acoustic space at a given measurement position if a sound impulse is emitted at another position by an omnidirectional point source. On the other hand, the directional (or spatial) room impulse response (DRIR) is a *subset* of the RIR, including only the contributions that reach the receiving position from a well-defined direction in space or from a set of directions included in a more or less extended solid angle [2]. DRIRs can be obtained by using a directional microphone at the receiving position or a suitable microphone array.

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Room acoustics computer programs are nowadays able to compute DRIRs. In the ray tracing method

for example, it is only necessary to collect the sound rays’ contributions into different solid angles, according to their direction of incidence at each receiving position. However, applications of *measured* DRIRs in room acoustics are not frequent, probably because they require specific instruments to acquire and process the directional information. Some examples can be found in [3,4, 7, 8]. In this paper, we describe some applications of a spherical array containing 16 microphones, which has been specifically designed for the measurement of DRIRs.

2. The spherical array of microphones

This spherical array of microphones has been designed by H. Feron [5]. Figure 1 gives a picture of this equipment.

Most characteristics were chosen to optimize the properties of the array, taking into account the material available, as for example the acquisition facilities. For practical reasons, the size must also be limited since this measuring equipment is

intended to be used on-site and must be handy and portable.



Figure 1. The spherical array with some of its numbered microphones [5].

The following characteristics have been chosen:

- rigid sphere with a radius of 10cm,
- 16 omnidirectional microphones,
- *nearly uniform* distribution of the microphones on the sphere [6]: this allows for a 3rd order spherical harmonics decomposition of the sound field with 16 microphones,
- two soundcards (eight inputs each),
- recording in *wav* format and signal (post-) processing in *Matlab*.

The radius of the sphere has been determined after consideration of three limitations: spatial aliasing, which increases with frequency, the uncertainties related to the microphones' placement on the sphere (phase errors) and the errors due to noise. Finally, a radius of 10cm was chosen, giving a useful bandwidth of [250–4000] Hz.

3. Beamforming

The sound pressure field $P(k, r, \Omega)$ existing on the spherical antenna of radius r can be described by a series of spherical harmonics functions Y_n^m (see equation 1). In this equation, Ω represents the pair of angular values φ and δ , defined in figure 2.

$$P(k, r, \Omega) = \sum_{n=0}^{\infty} \sum_{m=-n}^n P_{nm}(k, r) Y_n^m(\Omega) \quad (1)$$

If the spherical harmonics coefficients $P_{nm}(k, r)$ are known, then the 3D sound field can be completely reconstructed on the spherical antenna. However, the coefficients P_{nm} can only be recovered for a limited number of values of the index n : this determines the maximum order of the array.

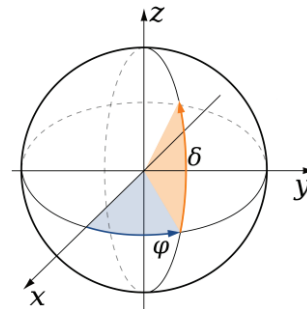


Figure 2. Spherical coordinates φ and δ [5].

In practice, the coefficients P_{nm} are computed through a suitable linear combination of the signals measured by the 16 microphones [7]: if Ω_j is the position on the sphere of microphone number j ,

$$P_{nm}(k, r) = \sum_{j=1}^{16} \alpha_j P(k, r, \Omega_j) Y_n^{m*}(\Omega_j) \quad (2)$$

The coefficients α_j of this linear combination are defined by the *nearly uniform* distribution of the microphones on the sphere [5,6].

The next step of the signal processing is the *beamforming*: the part of the sound field corresponding to a particular direction of incidence (or *look-up* direction Ω_L) is obtained by applying a set of weights W_{nm} to the spherical harmonics coefficients:

$$Q(k, \Omega_L) = \sum_{n,m} P_{nm}(k, r) W_{nm}^*(k, \Omega_L) \quad (3)$$

These weights depend on the frequency (k), on the look-up direction and on the method of beamforming. It is shown for example in [7] that if the sound field on the rigid sphere is composed of an infinite number of plane waves, then the amplitude density in a particular direction Ω_L is given by equation 3 with well-defined weight functions (method of *plane-wave decomposition*).

In this work, the following beamforming methods have been applied instead:

- the delay-and-sum (DAS),
- the minimum-variance distortionless response (MVDR) described in [8].

The DAS method consists in applying a different delay (or phase shift) to each output of the individual microphones such that the signals become in-phase for a plane wave coming from the look-up direction. This leads to a particular expression for the weight functions in equation 3. On the other hand, the MVDR method belongs to the class of *optimal* beamforming techniques, which try to define the weight functions that maximize the signal to noise ratio and perhaps other properties of the array. For this class of methods, the weights depend on the sound field being analysed (adaptive beamforming).

4. Measurement of DRIRs

4.1. Principle

Figure 3 illustrates the general layout used for the measurement of the DRIRs in a room. A logarithmic sine sweep signal is generated by the source, which gives 16 sweeps recorded by the microphones. These sweeps are then de-convolved (post-processing) to obtain 16 impulse responses [9]. Just note that the sweep is generated between 250 Hz and 4 kHz which are the limits of the useful bandwidth of the array. The computed impulse responses are therefore filtered in this frequency band.

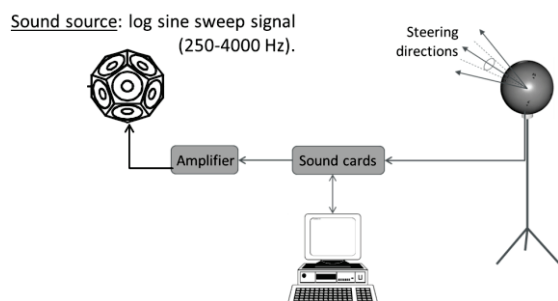


Figure 3. General layout for the measurement of directional impulse responses [5].

In equation 2, $P(k,r,\Omega_j)$ therefore represents one particular impulse response (for microphone number j) and P_{nm} are the corresponding spherical harmonics coefficients. A DRIR is then defined by choosing a particular *look-up* direction and applying equation 3. Moreover, by *steering* several directions around the receiving position, one can obtain 2D (one angle is varied) or 3D (two angles) representations of the DRIRs.

4.2. Measurement in a long corridor

Figure 4 shows a view of this particular indoor space, which has been chosen to illustrate the detection of flutter echoes along the axial direction of the corridor.

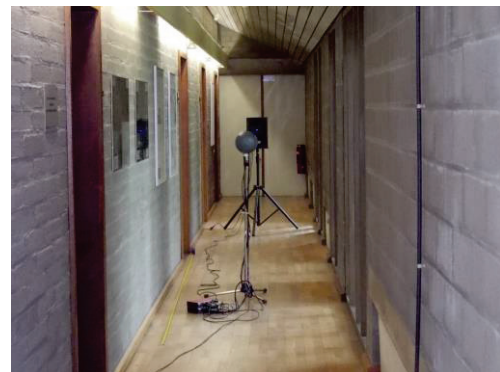


Figure 4. Measurement of DRIRs in a long corridor.

The corridor is a long (quasi-) parallelepipedic room which is closed at both ends by two heavy wooden doors. The total length between those two reflecting surfaces is 27.8m.

Figure 5 shows the ‘omnidirectional’ room impulse response measured at a given receiving position in the corridor. Strong specular reflections already appear, especially after 200ms. Of course, no information about their direction of arrival can be obtained from this graph. The same figure 5 also shows the DRIR in the direction 180° , which is the direction opposite to the loudspeaker if you look at figure 4. On this graph, the flutter echo is clearly apparent: strong reflections are detected at 93.7ms (first reflection on the rear door), 255.6ms, 417.6ms and 579.3ms. All these reflections are separated by 162ms which corresponds approximately to the time of a back and forth between both terminal doors (55.6m). Figure 5 shows that the beamforming acts as a spatial filter to emphasize the presence of sound contributions along a particular direction of incidence.

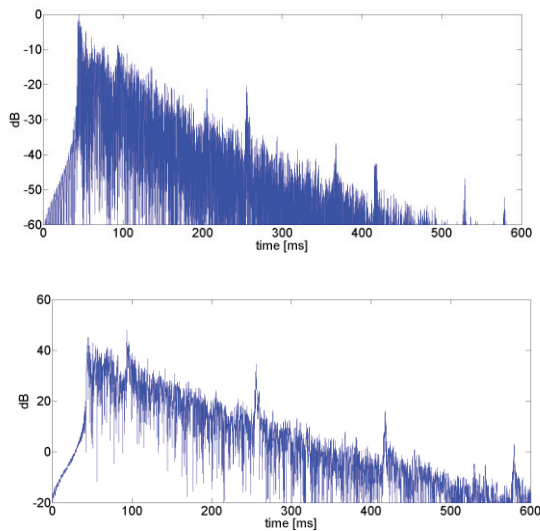


Figure 5. Omnidirectional RIR (up) and DRIR in the $\varphi=180^\circ$ direction (down), measured at the same position in the long corridor. Note that the reference for the decibel scale is different for both figures.

The DRIR in figure 5 has been obtained by the delay-and-sum (DAS) algorithm. It has been shown in [5] that this beamforming was particularly efficient for the measurement of DRIRs because of the time delay compensation that is applied at the very beginning of the method. This leads to a good compromise between the time and space resolutions of the sound contributions (reflections).

It could also be interesting to isolate a particular reflection and locate it in the 3D space. If this contribution significantly emerges from the RIR, this can be done quite efficiently with the MVDR algorithm. First, a short time window must be defined in the impulse response, which contains this particular reflection. Then, a directivity analysis restricted to this time window is performed by the beamforming MVDR, which gives the result shown in figure 6: in this case, the strong reflection detected between 254 and 256ms is located in the horizontal plane ($\delta=0^\circ$) and in the 180° direction in this plane.

Just note that the MVDR algorithm is applied at a specific frequency (1623 Hz) for which the measurement accuracy is optimal [5].

4.3. Measurement in a small studio

Figure 7 shows a top view of this studio. Its horizontal dimensions are approximately 7m and 5m. The distance between the (parallel) floor and

ceiling is 2.93m. The studio is equipped with absorbing materials (in grey) and diffusors (in brown). The DRIRs have been measured in this studio for six loudspeakers' positions identified by the rectangular boxes, but only the lower right position in the figure (FAR3) will be analyzed in this paper.

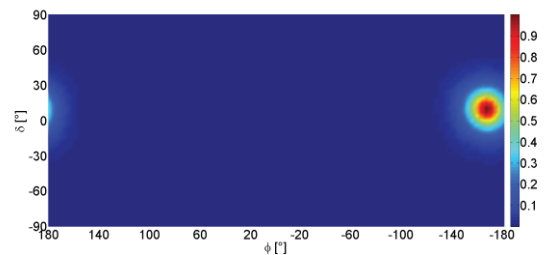


Figure 6. Application of the MVDR algorithm to the RIR of figure 5, after time windowing between 254 and 256ms.

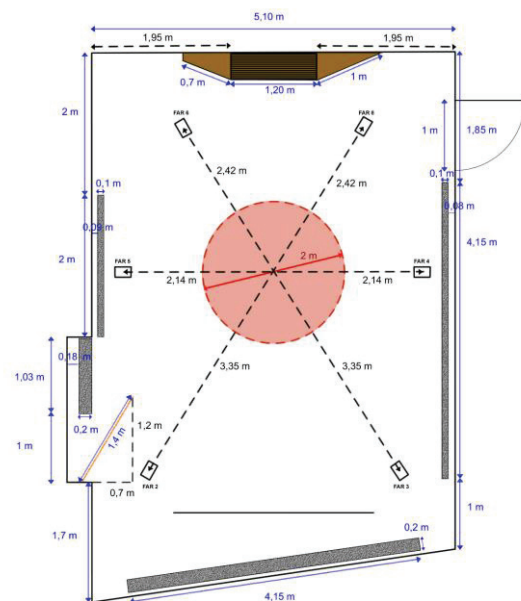


Figure 7. Plane view of the studio.

The spherical antenna is placed at the center of the circular zone (figure 7), at the same height as the loudspeakers, i.e. 1.5m above the floor.

Figure 8 shows the 200 first milliseconds of the impulse response, indicating that some early reflections are present and can influence the acoustics of this studio. The omnidirectional RIR as well as the DRIR in the direction of the source ($\varphi=-140^\circ$) are presented.

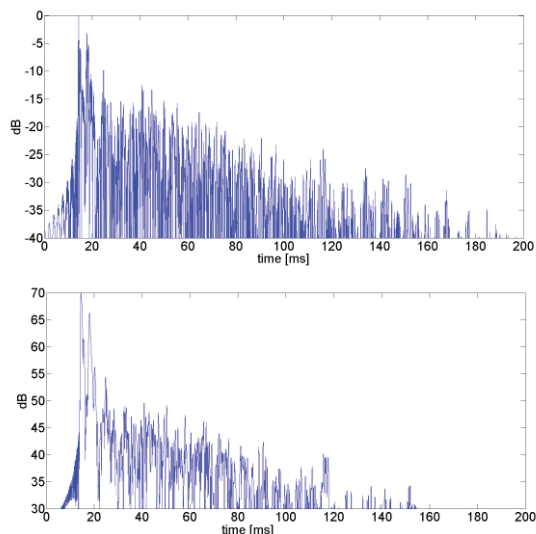


Figure 8. Omnidirectional RIR (up) and DRIR in the direction $\varphi = -140^\circ$ (down), measured at the same position in the studio. Note that the reference for the decibel scale is different for both figures.

In order to test the validity of the DRIR measurements, the first- and second-order image sources have been computed and some of them have their position localized in table 1. In this small room, the early reflections reach the receiving position with very similar delays: it is therefore difficult to identify them separately.

Table I. Real source and 1st order image sources in the small studio.

Source identification	Distance to receptor (m)	Delay (ms)	Azimuth φ ($^\circ$)
Real source	3.13	9	-144
Floor	4.23	12	-144
Ceiling	4.34	13	-144
Right wall	3.77	11	-132
Left wall	7.35	22	-110

However, the application of the beamforming MVDR to time windowed parts of the impulse response can again be tested in this respect. Figure 9 shows the contribution of the direct sound, at time $t=15$ ms. Also, figure 10 clearly shows the presence of the first reflection on the floor, about 3ms after the direct sound ($\delta \sim -30^\circ$), and the first

reflection on the ceiling ($\delta \sim +30^\circ$), about 4 to 5ms after the direct sound.

These examples illustrate the power of the MVDR algorithm, even if it is applied to very short time windows. Of course, deviations of some milliseconds can be observed between theoretical delays and their real counterparts.

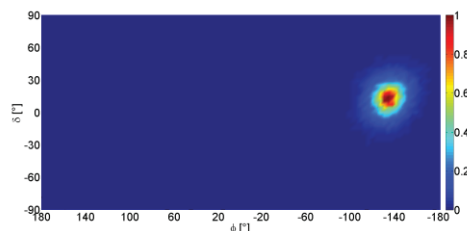


Figure 9. Application of the MVDR algorithm to the RIR of figure 8, after time windowing between 14 and 16ms (real source).

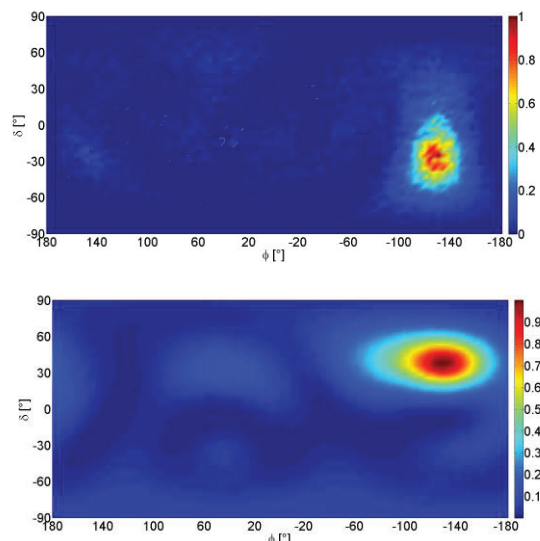


Figure 10. Same as figure 9, with a time window between 17 and 18ms (up, first reflection on the floor) and between 19 and 20ms (down, first reflection on the ceiling).

4.4. Measurement in a theatre

Figure 11 shows a view of the microphone array in this theatre. A few results are shown in the following.

The omnidirectional RIR measured at a given position is shown in figure 12. We will focus our attention on two particular contributions: the direct sound (around 50ms) and the reflections around 90ms.

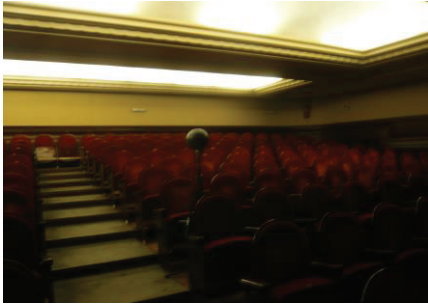


Figure 11. Measurement of DRIRs in a theatre.

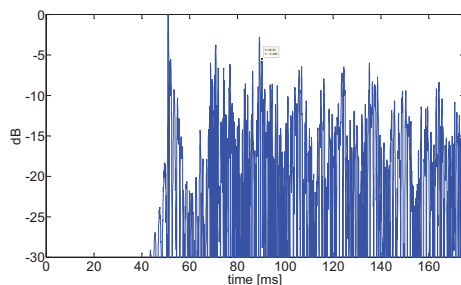


Figure 12. Omnidirectional RIR measured at a given position in the theatre.

Figure 13 shows that the localisation of the direct sound is again quite accurate, since this contribution is well isolated in the RIR.

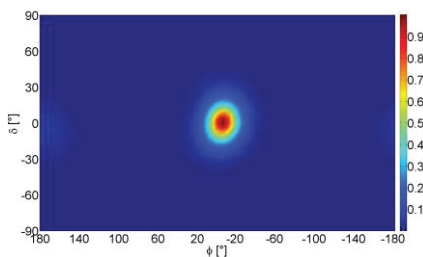


Figure 13. Application of the MVDR algorithm to the RIR of figure 12, after time windowing between 50 and 52ms.

On the other hand, figure 14 shows the result of applying two different time windows around 90ms: two very different representations are obtained, probably because several important sound contributions are included in the time window of 2ms. As these contributions are not enough separated from one another on the time axis, this complicates the interpretation of the DRIR diagram.

5. Conclusion

This paper has shown some applications of a spherical microphone array for the measurement

of DRIRs. The combination of the DAS and MVDR beamforming methods is particularly appreciated for the identification and localisation of early reflections in the room. However, it is also shown that this task of identification becomes difficult if a sound contribution is not enough isolated and/or dominant in the RIR.

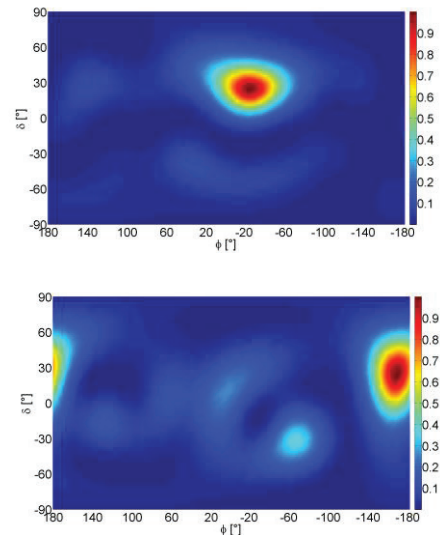


Figure 14. Same as figure 13 around 90ms, with a time window of 1ms (up) and 2ms (down).

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