Detection of events by means of plane wave decomposition analysis and cross-correlation technniques using a circular array of microphones

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Abstract

This article deals with the study and analysis of room acoustics through a process of sound field decomposition sampled with microphones circular arrays. The spatial characteristics of the sound field inside a room can be meaningfully described by means of microphone array processing techniques. In this context, the set of impulse responses sampled by a microphone array can be seen as an image made of acoustic planewave footprints. Due to the circular geometry of the microphone array, these footprints have a cosine-like shape that can be fully described as a function of the direction of arrival (DOA) of the impinging plane wave. Plane-wave decomposition (PWD) technique using microphone arrays have been shown to be a very useful tool within the applied acoustics community for their multiple applications in room acoustics analysis and synthesis. This paper conducts an analysis methodology based on the above method to analyze the sound field inside a selected set of real rooms having a well-defined purpose. Through the development of a cardioid microphones circular array as well as the implementation of a wave detection algorithm in echograms based on a cross-correlation method, the acoustics of a room (reflections, absorptions,etc.) is obtained, analyzed and compared following to extrapolate some conclusions about its features, performance and quality.

Keywords: Circular microphone arrays, plane wave decomposition, cross-correlation method, sound field analysis.

1. Introduction

Room acoustics is an old scientific domain but the acoustic behavior of a room considering all its spatial factors is not straightforward. This problem is known since antiquity: Greeks and romans started to look the acoustic qualities in their architectural constructions mainly based on purely practical designs. Amphitheaters and legendary concert halls with amazing acoustic behavior have already been built years ago. Apart from human perception, no precise assistive technology for acoustic analysis was available at that time. In comparison, high resolution room acoustic measurement technology is quite a young branch of acoustics, as modern technology is required to extract the information carried within acoustic waves, which is not detectable by humans. Only within the past few years have real applications in acoustics become into the focus of interest, since sufficiently powerful processors and audio hardware have gotten more affordable. The application of these methods in room acoustics is therefore a new and interesting topic. Later, it was physicist Sabine with his statistical model for measuring the reverberation time of rooms who would lead to the modern analysis of architectural acoustics. Assuming that a room can be considered as an acoustic transmission system, the impulse response provides a correct description of the changes when an acoustic signal travels from one point to another. Therefore, the experimental determination of impulse responses is a fundamental task in the acoustics of a room. Some spatial parameters (e.g. apparent source width or listener sur-

For a circular array, the Kirchhoff-Helmholtz and Rayleigh integrals are used to later extrapolate the sound field using cylindrical harmonics.

rounded) involve directionality and are measured with slightly more complex microphone arrangements or dummy head microphones. However, these methods are optimized for human perception only and are not applicable for a physical description or reconstruction of the incident wave field, neither for the high definition extraction of directional acoustic properties and other methods have to be applied for these purposes [1-3]. Sound pressure and velocity at all points inside an area of interest describe a wave field and comprise all of the information on the corresponding room influences and properties in a highly general manner. Knowledge about the complete wave field can deliver more information than a single-point omnidirectional or stereo measurement optimized than human perception can. As this knowledge comprises a complete general physical description, it can be seen as a superset of all possible measuring methods. Most of these can be derived during post-processing by extrapolation to virtual measurement positions. Ever since the very first steps were made in measurement, technology has become much more accurate and state-of-the-art measurement equipment is often processor based and contains complex methods of digital signal processing [4,5].

There are currently multiple and advanced analysis methods based on the study of room impulse responses that have contributed to new developments and improvements in the acoustic field [6-8]. For a long time, acoustic measurements have been performed using a single microphone, disregarding very important information related to the spatial characteristics of the sound field within the room. The latest recently proposed methods used microphone arrays to capture more accurately the sound field with all its spatial characteristics. Microphone arrays are widely employed in many acoustic signal processing tasks, such as speech enhancement, source localization or echo cancellation. A very interesting application of microphone arrays is the analysis of the spatial characteristics of a sound field, which allows acoustic designers to systematically investigate the benefits and drawbacks of the reflection phenomena occurring inside a given room [9,10]. In fact, the measurement of simultaneous impulse responses at different spatial positions allows acousticians for a complete analysis of the sound field, including the estimation of room geometries, the detection of main reflections and the identification of other phenomena in sound propagation [11]. A number of analysis methods have been developed in the last years to investigate the impact of acoustic reflections within a hall [12,13]. Some of these methods are based on the measurement of multiple impulse responses with circular microphone arrays, which provide full azimuth coverage [14]. In this paper, plane-wave decomposition (PWD) technique for measuring spatial sound characteristics by means of circular arrays is presented. Furthermore, it has provided added advantages for room acoustics analysis and design performance than classical approaches [15-18]. For a circular array, the Kirchhoff-Helmholtz and Rayleigh integrals are used to later extrapolate the sound field using cylindrical harmonics [19]. This tool can be regarded as making an acoustic snapshot of each room in such a way that any process could be virtually recreate in it, such as reproduction sounds, technical analysis or listening from anywhere in the room.

In summary, the objective of this paper is to conduct an analysis methodology based on plane wave decomposition where the results obtained are analyzed to identify the most significant reflections of rooms, extracting information about the dispersion and distribution of energy by means of different plane-wave detection methods. The most relevant is based on a cross-correlation algorithm where besides, results are associated with the analyzed room geometry in order to verify the proper work of this method.

The paper is structured as follows. Section II briefly describes the fundamentals of circular microphone arrays and multi-trace impulse responses. Section IV presents our proposed approach PWD. Sections V and VI describe the experiments conducted in real rooms by this method and it corresponding results, respectively. Finally, Section VII summarizes the main points of this work.

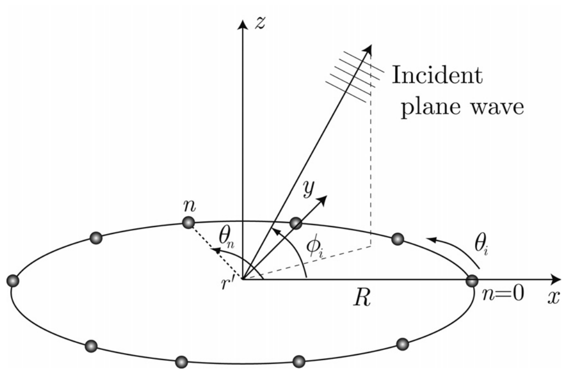
2. Plane waves and circular arrays

A microphone array is a set of spatially distributed microphones that is used to record and process sound signals in a meaningful way. The simultaneous processing of the signals acquired by the array can be used to enhance a signal of interest, locate the source of the signal or to gather information about the acoustic environment. Although microphone arrays can have any shape, this paper is focused on microphone arrays having a circular geometry because Uniform Circular microphone Arrays (UCAs) are known for their full-azimuth spatial properties and their suitable geometry for modal array processing [15], [18], [19].

2.1 Circular Microphone Array Geometry

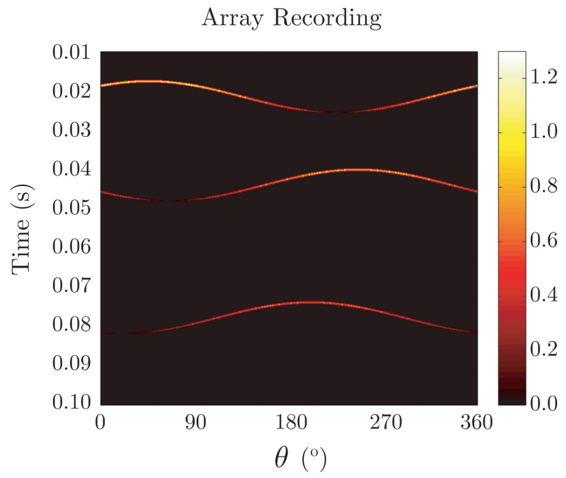
The UCA geometry with N elements and radius R, positioned in the horizontal (x,y) plane used throughout this paper is shown in Figure 1.

Plane waves are considered incident on the array with azimuth angle θ_i, elevation angle ϕ_i and propagation speed c ($c \approx 343$ m/s). A microphone position on the array is specified by an azimuth angle $\theta_n = (n-1)\frac{2\pi}{N}$, with n=1,..., N. The time instant at which the plane wave arrives at the center of the array is known as intercept time and is denoted as i. The arrival times of a plane wave impinging on the circular array as a function of the microphone angle are described by a cosine-shaped curve. The



■ **Figure 1.** *Geometry of plane waves incident on a circular array in the horizontal plane.*

crest of the curve occurs for the microphone having an angle closest to Θ_i, since this is the first microphone to notice the presence of the wave. An example of di fferent plane-waves (with directions 45°, 245° and 200°) arriving with different intercept times (22 ms, 44 ms and 78 ms) is shown in Figure 2, where an array with R =1.5 m and N= 288 has been employed in the simulation.



■ **Figure 2.** *Simulated multi-trace impulse response with 3 plane waves coming from i = {45 , 245 , 200 }.*

For a given sampling frequency f_s, these curves can be mathematically expressed as:

$$t_n(\tau_i,\Theta_i)=\left[f_s\tau_i - f_s\frac{R}{c}\cos(\Theta_n-\Theta_i)\right]$$

where $t_n(\tau_i,\Theta_i)$ represents the time instant (in samples) at which a plane-wave with DOA Θ_i and intercept time τ_i arrives at the n -th microphone.

Spatial Aliasing

As above mentioned, an UCA is a sampled aperture by means of a finite-number of sensors. Analogous to the Nyquist frequency in temporal sampling, we have a restriction on minimum spatial sampling rate. The distance can be calculated as

The flexibility of the plane wave decomposition lies in the fact that the sound field can be calculated at any position, a feature which is desired for any purpose of acoustic analysis and room auralization.

$$d=2R\sin\left(\frac{\pi}{N}\right)$$

The above distance determines the spatial aliasing frequency, which is given by

$$f_{al}=\frac{c}{2d}$$

2.2 Multi-trace Impulse Response

The way acoustic waves behave in a space can be fully described by taking impulse response measurements, which provide a description of the changes sueffered by an acoustic signal when it travels from one point to another in a room [6,7]. When measuring simultaneously N impulse responses, a time-space matrix can be constructed by storing in azimuth order all the acquired impulse responses in the array:

$$\mathbf{H}=[\mathbf{h}_1,\mathbf{h}_2,\ldots,\mathbf{h}_N]$$

where $\mathbf{h}_n=[h_n(1),\ldots h_n(1)]^T$ is the L-length impulse response acquired at microphone n from a test measuring source. The elements of $\mathbf{H}$ are denoted as $H(t,n)$, where t and n denote a given time sample and microphone, respectively. This matrix allows for a complete observation of the room behavior by plotting together all the acquired responses.

3. Plane-wave decomposition

This section briefly describes the concepts of the application of plane wave decomposition. For a complete description, please refer to [14], [18].

3.1 Cylindrical Harmonics

Considering a circular array of equally spaced dots of radius R, centered at r', both pressure and the normal component of velocity are stored as $p(\Theta,t)$ and $v_n(\Theta,t)$ for each azimuth angle . With this data set, the reconstruction of the sound field can be made by means of integral equations of Kirchho -Helmholtz in cylindrical coordinates [20-22]. However, for circular arrays it is only possible to reconstruct the wave field inside the circle with these equations. This has been demonstrated by means of works where both incoming and outgoing wave fields contained strong artifacts inside a circle, which were due to the fact that these fields had sources and wells in the origin, which were not there for the original sound field. By taking the sum of both incoming and outgoing field, the sources and wells cancelled each other out, resulting in a proper reconstruction inside the circle as well [23], [24].

In order to obtain the impulse responses at any point in the room, a pre-processing based on the decomposition into cylindrical harmonics is needed, which is given by $M^{(1)}(k_\theta,\omega)$ and $M^{(2)}(k_\theta,\omega)$:

$$M^{(1)}=\frac{H'^{(2)}_{k_\theta}(kR)P(k_\theta,\omega)-H^{(2)}_{k_\theta}(kR)j\rho c V_n(k_\theta,\omega)}{H'^{(1)}_{k_\theta}(kR)H'^{(2)}_{k_\theta}(kR)-H^{(2)}_{k_\theta}(kR)H'^{(1)}_{k_\theta}(kR)}$$

$$M^{(1)}=\frac{H'^{(1)}_{k_\theta}(kR)P(k_\theta,\omega)-H^{(2)}_{k_\theta}(kR)j\rho c V_n(k_\theta,\omega)}{H'^{(2)}_{k_\theta}(kR)H'^{(1)}_{k_\theta}(kR)-H^{(1)}_{k_\theta}(kR)H'^{(2)}_{k_\theta}(kR)}$$

where k is the wavenumber, ρ is the density of the medium, Hand Hare the Hankel functions and their derivatives of order k_θ and kind 1 and 2 and $M^{(1)}(k_\theta,\omega)$ and $M^{(2)}(k_\theta,\omega)$are the decomposition of the incoming and outgoing sound field in cylindrical harmonics, respectively, with reference to the circular array. $P(k_\theta,\omega)$ and $V_n(k_\theta,\omega)$are the spatial and temporal transforms of the pressure and normal velocity values obtained in the array ($p(\theta,t)$ and $v_n(\theta,t)$). Besides, for sources located outside the circle, it is possible to consider a single set of expansion coefficients that $M=\frac{1}{2\pi}\left[M^{(1)}+M^{(2)}\right]$ can be calculated as follows:

$$M(k_\theta,\omega)=\frac{\frac{1}{2}\left[P(k_\theta,\omega)+\rho c V_n(k_\theta,\omega)\right]}{H^{(1)}_{k_\theta}(kR)+H^{(2)}_{k_\theta}(kR)-jH'^{(1)}_{k_\theta}(kR)-jH'^{(2)}_{k_\theta}(kR)}$$

3.2 Plane-Wave Decomposition

Knowing that from auralization purposes, only the incoming component is required, the plane wave decomposition of the sound field in terms of cylindrical harmonics becomes [14]:

$$S(\theta,\omega)=\frac{1}{2\pi}\sum_{k_\theta} e^{-jk_\theta\frac{\pi}{2}} M(k_\theta,\omega)e^{jk_\theta\theta}$$

Hence, the flexibility of the plane wave decomposition lies in the fact that the sound field can be calculated at any position, a feature which is desired for any purpose of acoustic analysis and room auralization [1].

4. Proposed approach

Next, we describe a method to detect plane waves on above representations by means of an alternative detection method which is based on cross-correlating a circularly-shifted cosine-shaped mask with a binarized version of the original image.

In addition to the multi-trace impulse response which time-space matrix is formed by all the acquired impulse responses in the array (see Figure 2), there are two main representations that appear when plotting the set of impulse measurements with UCAs. One useful representation of the sound field is the frequency-space decomposition which is obtained after applying the PWD method and the sound field can be easily interpreted as a summation of plane waves having different directions of arrival, Figure 3 (a). Another meaninful representation to be taken into account is the time-space decomposition representation where the different plane waves can be identified as sharp peaks corresponding to their original intercept times and azimuth directions, Figure 3 (b).

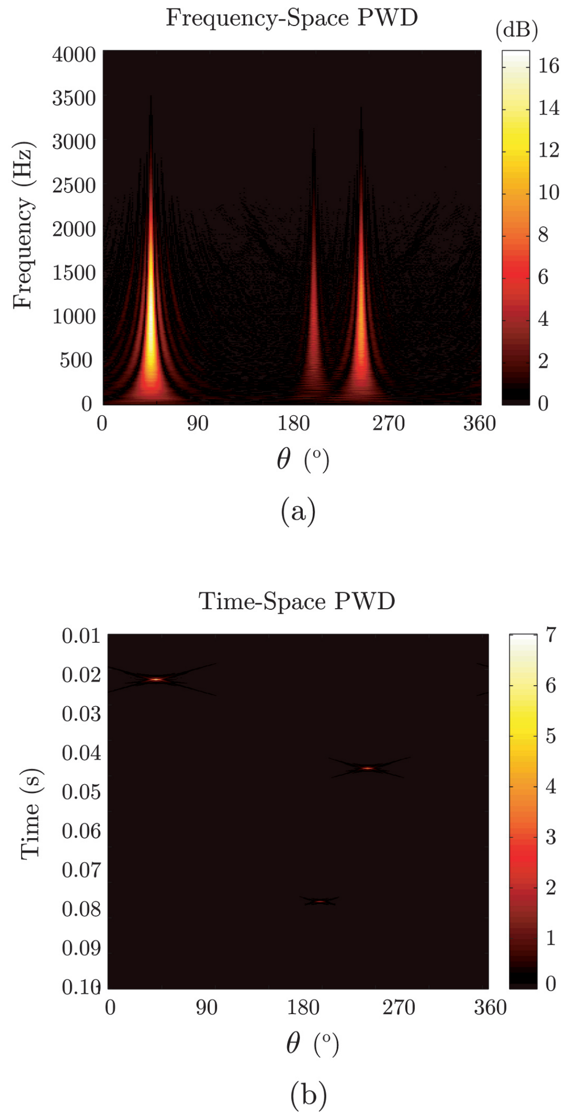


Figure 3. *Frequency-Space PWD (a) and Time-Space PWD (b) representations for the above simulated example with 3 plane waves coming from i = {45 , 245 , 200 }.*

The detection of main reflections from the measurements carried out in Section V, has also been performed by applying other different wave-detection methods as "Manual Detection" and "PWD Detection" that are explained below.

4.1 Manual Detection

Manually selected plane-waves obtained by visual analysis. To make the selection, five different subjects familiarized with image signal processing were asked to identify visually the peaks they found more representative.

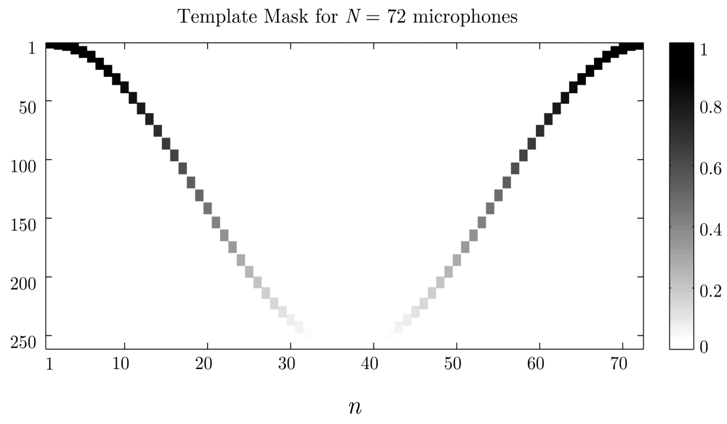


Figure 4. *Template Correlation Mask.*

4.2 Plane Wave detection method

The correlation mask must match the properties of the curves found in the multi-trace impulse response, as given by Equation (1). To this end, the specific parameters of the array are used to build a cosine-shaped mask adapted to the processing parameters. Moreover, since cardioid microphones are used, the amplitude registered by each microphone can be modeled as

$$A_n(\theta_i)=0.5+0.5\cos(\theta_n-\theta_i)$$

Therefore, the template correlation mask $\mathbf{M}$ will have dimensions $\left(\left\lceil f_s \frac{2R}{c} \right\rceil, N\right)$ and will be filled as follows:

$$M(t,n)=\begin{cases} A_n(0) \text{ if } t=\left\lceil f_s \frac{R}{c}(1-\cos(\theta_n)\right\rceil+1 \\ 0 \text{ elsewhere} \end{cases}$$

An example correlation mask for a microphone array with $N = 72$ microphones is shown in Figure 4.

Local maxima in the resulting cross-correlation must be obtained by following another thresholding step.

4.3 PWD detection

Wave events are automatically detected by applying amplitude thresholding and region selection over the PWD representation. The region labelling is performed for eight-connected neighboring pixels and those regions with a considerable number of components are selected. The final selected values are those corresponding to the maximum values of the multi-trace impulse response in the surviving regions.

5. Experimental set-up

This section presents the application of the above shape recognition methods to multitrace impulse responses measured in real rooms located at Castilla-La Mancha University. First, a description of the analyzed rooms and the experimental set-up is provided. Next, the obtained multi-trace impulse responses are processed to detect the main room reflections.

5.1 Test Rooms

1. Room 1 -Meeting Room. This room was chosen for being a conventional mid size meeting room having

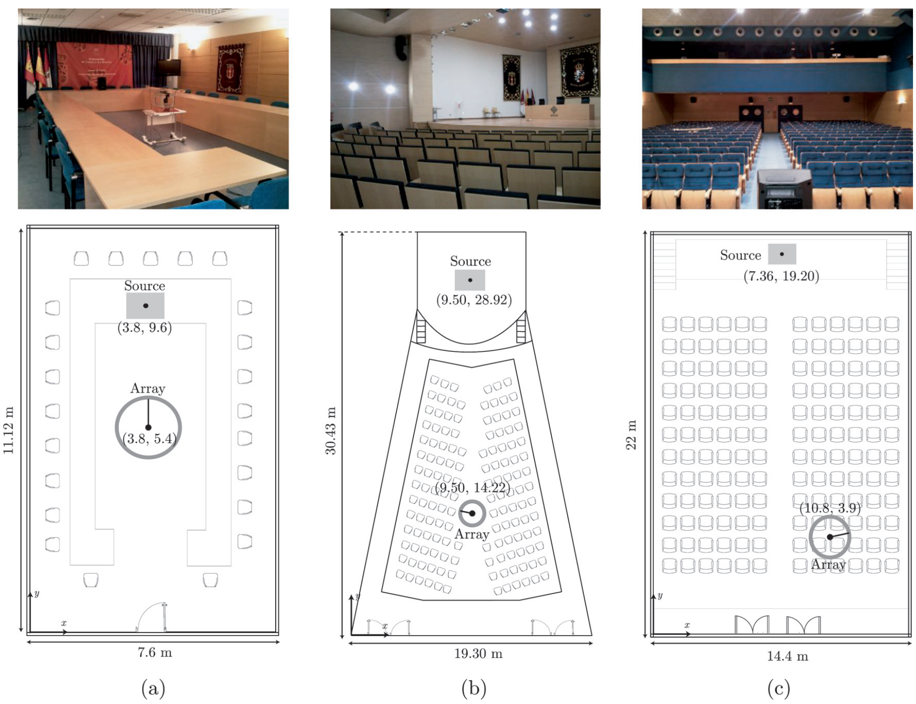


Figure 5. *Photographs and floor plans of the three measured rooms. (a) Room 1 -Meeting Room. (b) Room 2 -Auditorium. (c) Room 3 -Conference Hall.*

Wave events are automatically detected by applying amplitude thresholding and region selection over the PWD representation.

Measurements have been performed using a single microphone, disregarding very important information related to the spatial characteristics of the sound field within the room.

representative acoustic properties as shown in Figure 5 (a). Two walls are covered by wood panels and courtains and the other two are made in plaster. The measured reverberation time for the 1000 Hz band was approximately 0.65 s with the room empty.

2. Room 2 –Auditorium. The floor plan and dimensions of this room are shown in Figure 5 (b). The measured reverberation time for the 1000 Hz band was approximately 1.41 s. Moreover, its shape is significantly different from Room II since it has a trapezoidal-shaped plant.

3. Room 3 -Conference Hall. It has a classical shoebox shape with the exception of a gallery built at half the height of the room, intended for audiovisual equipment control. The dimensions and floor plan are shown in Figure 5 (c). The measured reverberation time was 1.23 s. The wall materials are wood and marble.

5.2 System implementation

The measurements were conducted with the same test loudspeaker positioned at the center of the stage. The array was positioned at the audience area. All the rooms were kept empty and unaltered when performing the measurements. The array was composed of two condenser cardioid microphones attached to the end of a 2 m long rod on a circular sliding turntable. The array measurements were conducted by carrying out automatically repeated captures for all the required microphone positions uniformly distributed over a circle of 2 m diameter placed within the listening area. Maximum Length Sequences (MLS) [25] were used as the sources of excitation (sampling frequency f_S = 44100 Hz). The source test signal was controlled by means of a laptop computer with an M-Audio Fast Track Pro audio interface and processed with an adhoc Matlab program. Since the room conditions (temperature, humidity, etc) did not change significantly during the measurements series, the results can be assumed to be the same as in the situation where all the impulses responses are measured simultaneously with a full array of N = 72 microphones.

5.3 Measurements and Plane-Wave Detection

The different multi-trace impulse responses, and their corresponding template crosscorrelation applied on time-space and frequency-space decompositions are shown in Figure 6. The first column displays the impulse responses recorded for all neighboring receiver positions along the circular array where in spite of the fact that the sound field presents a complex structure due to interference and diffraction, many reflection events can still be discriminated. PWD representations (columns 3 and 4) provide high energy compaction at localized plane-wave reflections. Frequency-space PWD representations already give a general impression of the spatial energy distribution at the measuring point. As shown in the fourth column of Figure 6, the energy mostly concentrates on the direct-sound direction (angular location of the first cosine maximum). The evolution with time of the spatial properties of the sound can be observed both in multi-trace impulse responses (columns 1 and 2) and time-space PWDs (column 3). Both first and late reflections can be identified in these representations, showing how the density of reflected waves coming from multiple directions increases significantly with time.

Regarding the detection of main reflections from the measured data, it has been performed by applying the di erent wave-detection methods explained in the Section IV:

Manual Detection

The white squares in the first column of Figure 6 denote the final selected reflections, which are the ones that were commonly selected by, at least, 4 subjects. Despite not being a very accurate ground truth, these manually selected values will be used to compare the detection performance of the automatic detection methods.

Correlation-Based Detection

Wave events are automatically detected by following the image processing procedure described in Section V. The detected reflections are shown as white circles in the second column of Figure 6.

PWD Detection

Similarly, the detected reflections are shown as white circles in the third column of Figure 6.

6. Results

The performance of the proposed method is summarized in Table I, showing the number of correctly detected wave-events (Correct), the number of false positives (FP), the number of false negatives (FN), the Mean Absolute Error in the time axis (MAEt) and Mean Absolute Error in the angle axis (MAE). The performance of the correlation-based (CB) method over the multi-trace impulse response and the PWD-based detection (PWD) was studied by comparing the number of false negatives (undetected reflections) and false positives (badly detected reflections) with respect to the manually selected events (Manual). A given event is assumed to be correctly detected when the distance to the closest manually selected event is less than 10 in angle and 100 samples in time. Moreover, for the correctly detected reflections, the mean absolute errors over the time axis (MAEt) and over the angular axis (MAE) were computed.

In general the correlation-based method provides lower deviation in the angle dimension in comparison with the PWD-based method, which is a very important issue in the application at hand. Furthermore CB is capable of de-

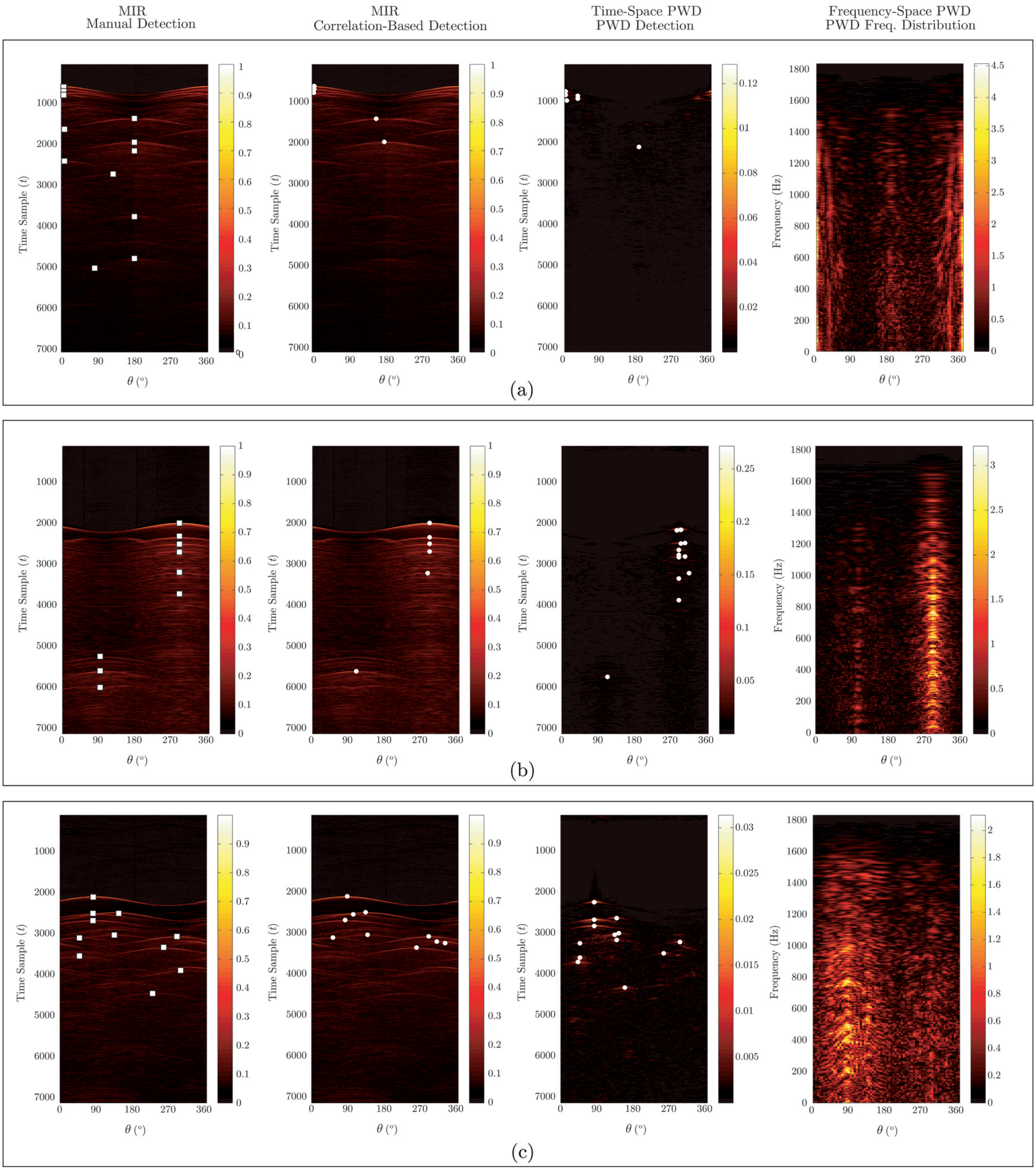


Figure 6. *Manual detection (column 1), proposed correlation-based approach (column 2), time-space PWD thresholding (column 3) and frequency-space PWD representations (column 4). (a) Room 1. (b) Room 2. (c) Room 3.*

tecting automatically approximately 60% of events and PWD method the 62.5%. However, the number of correctly detected waves, false positives and false negatives is very dependent on the analyzed room. In fact, the rate of false positives for all the rooms is quite smaller when using CB (6.3%) and although PWD seems to provide more detected reflections, many of them should be discarded (34.4%). In fact, further work is needed to analyze the effects of the different parameters involved in the processing, the robustness with different types of rooms, the mean performance over larger datasets and the use of more suitable models for characterizing plane-wave footprints with different height.

7. Conclusion

In this paper, the objective has been to verify that a decomposition of the data into cylindrical harmonics and a wave-detection method based in cross correlation are capable of identifying and separating plane wave events in impulse response measurements with a circular array. These techniques have revealed the spatial coherence of neighboring responses, leading to a far better insight in the complex wave fields in enclosed spaces than the analysis of individual impulse responses. Furthermore, these approaches have enabled perceptual evaluation of the sound field in a volume of the hall, instead of at one

Application of microphone arrays is the analysis of the spatial characteristics of a sound field, which allows acoustic designers to systematically investigate the benefits and drawbacks of the reflection phenomena occurring inside a given room.

local position, without the use of headphones, i.e., with natural temporal and spatial cues. This may yield a major step forward in room acoustic consultancy practice.

On the other hand, an image processing technique based in cross correlation method has been analyzed in the context of sound field analysis using UCAs. Sound field representations have been shown to describe accurately the spatial characteristics of sound at a listener position in three real rooms. Although detecting every plane-wave reflection accurately is very difficult, it was shown that this method is capable of identifying many of the most meaningful echoes automatically in these representations, providing a relatively small error both in time and angle.

Acknowledgments

The Spanish Ministry of Science and Innovation supported this work under the projects TEC2012-37945-C02-01 and TEC2012-37945-C02-02.

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Biographies

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Ana M. Torres received the M.Sc. in Telecommunications Engineer from Polytechnic University of Valencia, Spain, in 2003. From 2004 to 2006 she worked within a multinational automotion company and she participated as researcher at the the Institute of Telecommunications and Multimedia Application (iTEAM) working in the field of audio signal processing since end of 2012 when she achieved her PhD from the Universidad Politécnica de Valencia.

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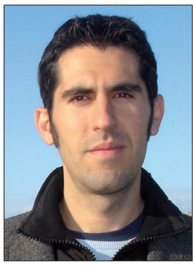

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