Plane Wave Decomposition and Beamforming for Directional Spatial Sound Localization

Muhammad Imran and Jin Yong Jeon*
Department of Architectural Engineering
Hanyang University
222-Wangsimni-ro, Seoul, Korea
mimranh1@gmail.com, jyjeon@hanyang.ac.kr

Akhtar Hussain
Centers of Excellence in Science and Technology (CESAT)
Islamabad, Pakistan
akhussain@hotmail.com

***Abstract*— Sound field decomposition using spherical microphone arrays has become a focus of interest for spatio-temporal analysis of the sound field in room acoustics, spatial audio rendering applications, and an estimation of the direction and localization of acoustic sources. The existing methods to localize acoustics sources and describe the sound field normally depend on omnidirectional sound field capturing by utilizing a few pairs of microphones and employing traditional correlation techniques. Therefore, these methods yield an approximate estimation of an overall acoustic analysis and parameters estimation of the captured sound field.**

In this paper, we performed sound field decomposition and beamforming to analyze the spatio-temporal structures of the captured impulse responses (IRs) using a 32-channel spherical microphone array in an anechoic chamber hall and in a real auditorium. We propose an approach based on a combination of plane wave decomposition (PWD), using spherical harmonics (SH) analysis, and an adaptive beamforming technique, as weighting filters for the PWD. Among adaptive beamforming techniques minimum variance distortionless response (MVDR) is investigated and used to compute the directional behavior of sound fields in the form of directional impulse responses (DIRs), in order to explain the influence of surrounding space on the propagation of the direct sound and several reflections. Furthermore, simultaneous localization of multiple sources and the variation in their temporal behavior is investigated and presented. The comparison between the results of computed directions of incidence of sound and the directions known beforehand showed a good agreement with each other inferring an effective application of spherical microphone arrays to sound field analysis.

Keywords—Plane wave decomposition; Beamforming, Spatio-temporal analysis; Directional impulse responses

I. INTRODUCTION

Sound field analysis and synthesis to acquire standardized acoustics parameters, estimate the directional characteristics of sound field, and locate the sound sources has been intensively investigated during the past few decades. The acoustical parameters and the sound propagation in real environments play an important role in analyzing the acoustical behavior of the sound field. The propagation of sound in an enclosed space is influenced by the reflections from surrounding, therefore, the analysis of such reflections is important in order to characterize acoustical properties. Recently, microphone arrays based processing has been investigated for sound field synthesis in order to emulate spatial and temporal behavior of the propagation of sound in spaces. Different microphone array configurations have been studied and proposed for acoustical analysis, such as, direction of arrival (DOA) estimation and localization of the source, using correlation among the pairs of microphones. However, the traditional methods to estimate the directional characteristics and locating the sound source typically depend on omnidirectional room impulse response (RIRs) measurements, therefore, provide only an approximation of spatial and temporal characteristics of acoustic propagation. Such methods result in limited spatial resolution and limited capability for localization. Further, they do not entirely incorporate the directional aspects of the captured RIRs, consequently, spatial information becomes merged with non-directional RIRs.

In this paper, we propose a sound field synthesis method based on plane wave decomposition and adaptive beamforming using spherical microphone array. Plane wave decomposition enables spatio-temporal characterization of captured RIR, whereas, adaptive beamforming synthesizes the directional impulse responses (DIRs) in order to analyze the directional behavior and localization of the sound sources. The use of spherical microphone array is suggested in order to realize a combination of PWD and adaptive beamformer, such as, optimal MVDR beamformer, that is appropriate for the reverberant environments and provides high resolution in terms of direction finding. The proposed method realizes a PWD to achieve spatial decomposition of the sound source and its corresponding reflections. From PWD, DOA of the direct sound and several reflections is estimated and decomposed sound field is visualized in space and time by 3D graphical representation. The MVDR is applied in PWD as beamformer weighting function to steer at a particular direction in order to collect the response of signal for that direction. The measurements were carried out in an anechoic chamber for the verification of developed algorithmic chain. The algorithms are designed and developed in the Matlab for processing and computations of the methodology.

II. SPHERICAL HARMONICS AND SOUND FIELD DECOMPOSITION

In this section the theoretical background of the spherical harmonics (SH) and spherical Fourier transform is presented that is the base for sound field synthesis. The Kirchhoff Helmholtz integral describes the sound pressure inside a source free volume when the sound pressure and velocity on its surface are known. By computing the sound field on a real or imaginary surface, the

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rest can be determined mathematically, enabling plane wave decomposition and beamforming. The Laplace spherical harmonics are defined for different orders and modes and provide combined solutions to the angular part of the linear wave equation. The orders and corresponding degrees of the SH define the accuracy and detail of the resolution of the sound field captured by a microphone arrays. By selecting higher SH orders, a higher directional selectivity of the array is obtained.

In a spherical microphone array system, it is assumed that a plane wave impinges on the surface of the sphere exciting an infinitely small node on its surface. This is known as perfect spatial sampling that results from the summation of infinite SH orders and corresponding modes to generate unlimited directional gain. However, in practice, it not possible to sum over all SH orders (i.e., up to infinity). Therefore, the SH summation is restricted to a certain order 'N' that leads to a limited attainable gain and spatial resolution.

A function that is square integrable at the surface of a unit sphere, its spherical Fourier transform and the inverse transform are given as follows [1].

$$f_{nm} = \int_0^{2\pi} \int_0^{\pi} f(\theta, \varphi)\, Y_n^{m^*}(\theta, \varphi) sin\theta d\theta d\varphi \quad (1)$$

$$f(\theta, \varphi) = \sum_{n=0}^{\infty} \sum_{m=-n}^{n} f_{nm}\, Y_n^m(\theta, \varphi) \quad (2)$$

$$Y_n^m(\theta, \varphi) = \sqrt{\frac{(2n+1)}{4\pi} \frac{(n-m)!}{n+m}}\, P_n^m(cos\theta) e^{im\varphi} \quad (3)$$

Where, Y_n^m are spherical harmonics of order n, and mode m, P_n^m are the Legendre functions of order n, and mode m, θ and φ describe the azimuth and elevation angles respectively.

III. Methodology

Accurate estimation, detection and localization of the acoustic source are big challenges. Sound encounters a frequency range involving three orders of magnitude from 20 Hz to 20 kHz. Different physical laws, such as diffraction at low frequencies, scattering at high frequencies, and specular and diffusive reflections, have to be considered and applied to sound field modeling [1]. To estimate the behavior of sound, its directional information and spatio-temporal analysis are very important. In our approach it is supported by dynamically adapted sound field capturing and plane wave decomposition using spherical harmonics. On one hand, the decomposed sound field is processed for visualization after treatment with an inverse spatial transform, and on the other hand, it is resynthesized into DIRs computed by an MVDR beamformer. Fig. 1 shows the computational methodology for the overall process.

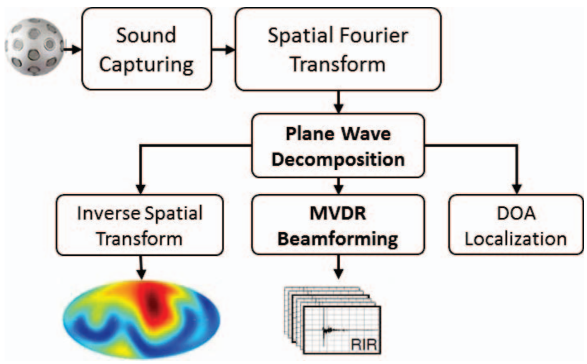

Fig. 1. Computational flow chat of methodology

A. Microphone Arrays and Sound Capturing System

Commonly, microphones are arranged in the form of an array on a real or imaginary spherical surface, taking advantage of its rotational symmetry. Some of the advantages of spherical surface geometries are easy mathematical handling and physical design [4]. Therefore, a spatially distributed array of microphones, known as a spherical microphone array, is used in this study to capture the sound field.

Theoretically, continuous sound field properties are required on the spherical surface. Array configurations are categorized into many types depending on some critical parameters such as measurement radius, surface of the sphere, and the number of microphones and their arrangement [5]. One example is a rigid reflecting spherical surface with evenly embedded microphones that applies the sound velocity to be zero on its surface. For such configurations the development of mathematical modal and computations are easy to handle. The configuration factors also affect the frequency range [6]. Design of the microphone array involves finding the best tradeoff among different factors. At low frequencies, a large radius of sphere is required whereas at high frequencies more microphones are needed for wideband arrays. Our microphone array, the Eigenmike, yields values at discrete spatial sampling points with a rigid spherical surface of diameter of 8.4 cm embedded with 32 evenly distributed microphones. The Eigenmike is designed to resolve 5 KHz to 6 KHz frequencies with a low contribution of spatial aliasing [7].

The process for sound capturing is describe in the following paragraph and the methodology is shown in Fig. 1. During the measurement process, impulse response is initially captured and subsequently subjected to offline post-processing to improve the signal to noise ratio. The actual practical approach for capturing the RIR is presented in Fig. 2, as proposed in [8].

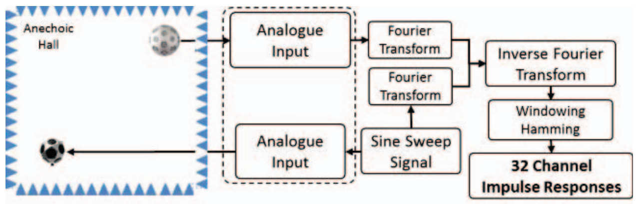

Fig. 2. Simultaneous acquisition of room impulse

B. Plane Wave Decomposition and Adaptive Beamforming

In this section the mathematical model of PWD and beamformer is presented and the process for computation is described. Plane wave decomposition applies the Fourier transform to the captured RIRs and to put them into the frequency domain. As a subsequent step, the spatial Fourier transform converts the frequency domain RIRs into spherical wave spectrum domain. From the spatial Fourier transform, the sound field is decomposed into spherical harmonics. The radial part of the solution to the wave equation in spherical coordinates is given by a combination of spherical Bessel and Hankel functions [9]. The radial part is compensated during the array processing by modal radial filters, considering the respective configuration of selected microphone arrays and is discussed later in this section.

The mathematical procedure for plane wave decomposition is described below in a summarized form, however, the theoretical detail is available in [1], [10] and [11]. As a first step, the spatial Fourier coefficients are generated that consider only the angular solutions of the wave equation given in terms of Legendre polynomials and exponential functions. The latter can be merged into a single expression represented by the surface spherical harmonics as given in (3). The spatial Fourier transform, optimized for sound field analysis, is used to compute the Spatial Fourier coefficients using SH by considering the array grid weights and sound pressure captured in the form of RIRs as given in (4).

$$P_{nm} = \sum_{n=0}^{N} \beta_s p_s Y_n^{m^*}(\vartheta, \varphi) \tag{4}$$

Here, Y_n^m are spherical harmonics of the order 'n' and mode 'm', 'β_s' are the grid weights and 'p_s' are the complex sound pressures.

Secondly, the radial part for the wave equation is incorporated including frequency dependencies. The radius and the configuration of the microphone array are considered that results in coefficients correspond to a real plane wave solution. A radial function denoted by '$b_n(\frac{\omega}{c}r)$' is incorporated, where 'ω' represents the angular frequency, 'c' the speed of sound and 'r' the array radius. The spherical configurations of the array and microphone types are critical parameters in the formulation of the radial function. Because the microphone array used in this study is a rigid spherical body embedded with uniformly distributed cardioid microphones, the modified radial function is as given in (5) [9].

$$b_n\left(\frac{\omega}{c}r\right) = 4\pi i^n \left(j_n\left(\frac{\omega}{c}r\right) - \frac{j_n'\left(\frac{\omega}{c}r\right)}{h_n^{(2)'}\left(\frac{\omega}{c}r\right)} h_n^{(2)}\left(\frac{\omega}{c}r\right) \right) \tag{5}$$

In this radial function the spherical Bessel functions $j_n'\left(\frac{\omega}{c}r\right)$ and spherical Hankel functions $h_n^{(2)'}\left(\frac{\omega}{c}r\right)$ provide solutions in spherical coordinates. Hence, incorporating the radial functions of a plane wave for the direction $(\vartheta_\omega, \varphi_\omega)$, the coefficient are computed as follows.

$$P_{nm} = Y_n^{m^*}(\vartheta_\omega, \varphi_\omega) b_n\left(\frac{\omega}{c}r\right) \tag{6}$$

In the final step of plane wave decomposition the radial structure of the sound field is compensated. Therefore, the frequency domain radial filter coefficients are computed as follows [12].

$$d_n(kr) = \frac{1}{b_n(kr)} \tag{7}$$

The complex-valued decomposed plane wave signal is computed as a results according to the (8).

$$Y = \frac{4}{(N+1)^2} \sum_{n=0}^{N} \sum_{m=-n}^{n} Y_n^m(\vartheta_l, \varphi_l) P_{nm} d_n\, c_n \tag{8}$$

Here, Y_n^m are spherical harmonics of the order 'n' and degree (mode) m and P_{nm} are the spatial Fourier coefficients.

To visualize the three dimensional room responses, reflection analysis is conducted based on plane wave decomposition. Important factors affecting the reflection analysis are the temporal windowing of the RIR and the spherical steering grid. An iteration process is applied in the algorithm to compute the plane wave decomposition that depends on the steering grid density. In this approach, the time-windowed iterative plane wave decomposition provides a spatio-temporal intensity distribution matrix [13], [14]. The results produced by the algorithm are the times, levels, and directions of the reflections. Algorithms have been developed and implemented to deliver plane wave outputs for all the given values of angles, termed the array look directions, at the same time.

A spatial-filter based beamformer, as described above using PWD beamforming, for narrowband signals can be reasonably well characterized by a single frequency. However, the spatial beamformer could not yield the same beamforming patterns for all frequency bands limiting its utility for broadband frequency response. The noise from the side lobes of the beamformer direction will not be attenuated uniformly over its entire spectrum, resulting in some disturbing artifacts in the array output [15]. Therefore, a response-invariant broadband beamforming technique must be implemented. In other words, a spatio-temporal filter is applied to the array outputs. To implement such filter, we had to consider an adaptive beamforming techniques, such as, MVDR beamforming filters instead of a simple spatial filter. MVDR filters resolve the issues arising from effects of side lobes, noise attenuation along the steering directions, and fixed array structures. Thus they are widely used in adaptive beamformer methods in which the basic idea is to choose the coefficients of filters to minimize the output power and the constraints without affecting the desired signal.

In this study, therefore, we use the MVDR beamforming weights for its certain advantages. In the MVDR Beamformer a finite impulse response filter is commonly applied to constrain the desired signal using unity gain while minimizing the filter output energy. This technique was developed for direction of arrival estimation from a desired signal during array processing. The optimal filters are given by (9) [18].

$$F^T(k) = \frac{W^H(k) S^{-1}(k)}{W^H(k) S^{-1}(k) W(k)} \tag{9}$$

Here S^{-1} is the matrix inverse. For microphone arrays with $\Delta d \leq \lambda/2$, these filters establish a super directivity beam that shows large noise sensitivity. These filters are reshaped by using a stability factor $\beta(k)$ [18].

$$F^T(k) = \frac{W^H(k)(S^{-1}(k) + \beta(k)\boldsymbol{I})^{-1}}{W^H(k)(S^{-1}(k) + \beta(k)\boldsymbol{I})^{-1} W(k)} \tag{10}$$

'$\boldsymbol{I}$' is the identity matric, and $\beta(k)\boldsymbol{I}$ is the relative level of sensor noise for each microphone. As a result of applying MVDR filtering weight in PWD output plane wave, we can estimate the directional characteristics of RIRs by computing DIRs for the directions of interest, providing the grid of steering angles to the beamformer.

IV. EXPERIMENTAL STUDY

An experimental evaluation and implementation of the proposed methodology is required in order to verify the developed algorithm for sound field decomposition and beamforming. Therefore, two measurement setups have been proposed to physically validate the algorithmic processing chain. For the verification process the measurements are conducted in an anechoic chamber hall, while for the experimental evaluation in real environment, the measurements are performed in an auditorium. The experimental study includes capturing the impulse responses with the Eigenmike microphone array, evaluating the PWD and MVDR beamformer, and finally representing and evaluating the accuracy of results. In addition, the results are compared with a measured dataset known beforehand.

A. *Verification of the methodology*

The measurements setup for verification was established in a full scale anechoic chamber hall in well controlled conditions at Daewoo Acoustics Labs, South Korea. The Eigenmike array was implemented and investigated experimentally. A high power omnidirectional dodecahedron was installed as sound source on the supportive meshed floor at different positions around the spherical microphone array. The schematic of the placement of three dodecahedron speakers at three different position around the array is shown in Fig. 3. The position of the source-1, source-2 and source-3 are 315°, 0° and 45° in azimuth respectively. The elevations for all sources are kept constant at 90°.

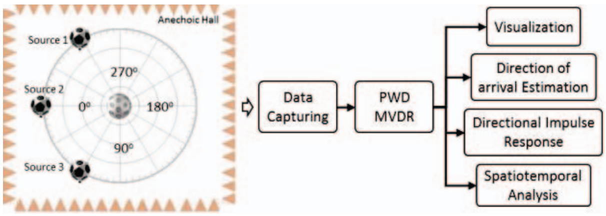

Fig. 3. Schematics for measurement setup in anechoic chamber

1) *Procedure:*

The Dodecahedrons were driven by a sine sweep input signal generated after calibrating it to a common sound pressure level. The measurements were repeated three times for each individual position of the source to confirm the repeatability in capturing system. Impulse responses of the source signals are captured at each elements of the array at a sampling rate of 44.1 KHz, using the techniques introduced in [8] and shown in Fig. 2. The origin of each selected position of the source was considered from the center of the array, indicating the listener's positon.

A Discrete Fourier Transform (DFT) was applied instead of Fourier transform to the captured impulse responses in order to transform them into frequency domain. In this way they deliver complex valued pressures and combine the discrete sampling in both the temporal and spectral domain. Spatial Fourier transform analysis was performed that yields spatial Fourier coefficients as a subsequent input to the plan wave decomposition. The radial filters were applied as discussed in section B. The plane wave decomposition was resolved to spherical harmonics of the order six (N=6) using (8). Generally, the (8) outputs decomposed sound field for the whole steering vector of angles simultaneously. In (8) the term c_n represent the complex float vector or matrix of beamshaping coefficients that are kr-dependent. The PWD perform a plane wave decomposition but in our proposed method it is switched to an MVDR-beamformer using the c_n coefficients as MVDR weighting coefficients represented in (10). For the subsequent analysis, a center frequency of 2 kHz is perceived that lies in the optimum frequency range for this array configuration. The corresponding theory of sound field analysis is presented in [1], [4], and [12].

2) *Verification Results*

a) Localization: In this section we present the spatio-temporal analysis and visualization of measurements from the anechoic chamber. The sound sources are placed around the array as depicted in Fig. 3. All the three sources are placed at same distance from the center of array. A signal generator algorithm developed in Matlab is used to adjust the sound pressure levels (SPLs) and delay times for each source. Therefore, multiple signals with different arrival directions, SPLs and arrival times were generated in precise and controllable manners. We have performed measurement in two different ways in order to analyze the strength of PWD and performance of the MVDR weighting filters. As a first measurement, plane waves impinging to Eigenmike that arrive at different times generated by three sources one by one. The sources have the same level but different time delays as: the source-1 have 315° position and delivers sound at 0 ms, source-2 have 0° position and delivers sound at 15 ms and the source-3 have 45° position and delivers sound at 30 ms. This analysis show the performance of PWD algorithm to analyze the spatiotemporal structures of the impulse responses on different time frames. Fig. 4 shows results from the first measurement setup, by plane wave decomposition on different time windows and shows that the sound sources are separated in space and time. Fig 4(a) shows the results of source-1 placed at 315° azimuth angle with delay time of 0 msec, Fig 4(b) shows the results of source-2 placed at 0° azimuth angle with delay time of 15 msec and Fig 4(c) shows the results of source-3 placed at 45° azimuth angle with delay time of 30 msec. All the sources were kept at 90° elevation angle. Second measurement shows two plane waves of different SPLs generated form source-1 and source-3 impinging on Eigenmike arrays. These sources are different in sound pressure level and at different positions while the plane waves from them, arrive simultaneously at the array. The recorded IRs are processed through PWD, and MVDR beamformer is applied at two steering angles in order to localize the simultaneous arrival of two sources. The results of the steered power response of MVDR is shown in Fig. 5. In this analysis we showed the MVDR strength to separate the two sources and compute the localization. Fig. 5 results from the second measurements by MVDR Weighting filters shows that two sound sources are detected and localized simultaneously places at 315° and 45° azimuth angle respectively. The plane waves from both sources have different Sound Pressure Levels, arrived at Eigenmike with 0 ms delay

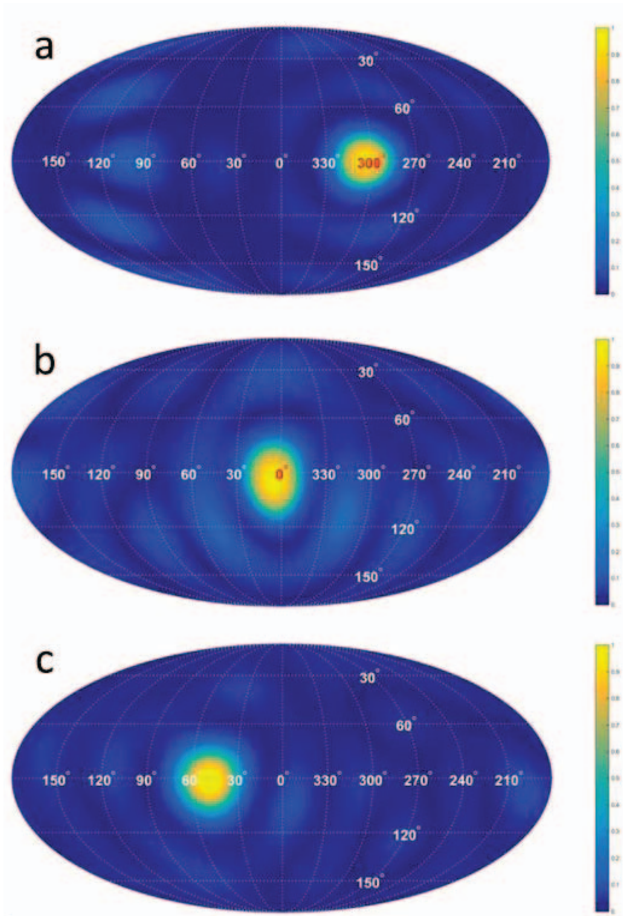

Fig. 4. Result from the first measurements by plane wave decomposition on different time windows

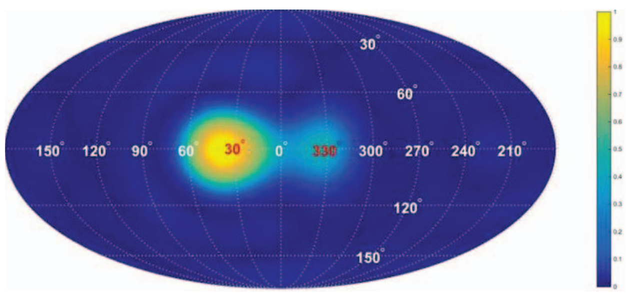

Fig. 5. Result from the second measurements by MVDR Weighting filters

b) Direction of arrival estimation (DOA): A comparison of the expected locations of incoming sound, as computed in the first measurement process, and the measured positions of the three sources placed at different location in Fig. 3 are presented in table 1.

TABLE I. ERROR ESTIMATION FOR THE EXPECTED AND THE COMPUTED DIRECTIONS OF SOUND SOURCE

SOURCE	Expected Location $(\theta_e^o, \varphi_e^o)$	Computed Location $(\theta_c^o, \varphi_c^o)$	Error estimation $\Delta\theta^0$	Error estimation $\Delta\varphi^0$
Source-1	315°, 90°	314°, 92°	-1°	2°
Source-2	0°, 90°	2°, 93°	2°	3°
Source-3	45°, 90°	48°, 92°	3°	2°

Table 1 shows the accuracy of the detection of the source locations in the anechoic chamber. $\Delta\theta^0$ and $\Delta\varphi^0$ represent errors in the azimuth and elevation angles respectively, and the expected locations were obtained by measurements. From table I, it can be seen that the results of the error estimation for the expected and the computed locations are found within the range of -1° to 3°.

B. Auditorium Measurements

The second part of the experimental study intended at estimating the performance of the array and proposed methodology in a real auditorium in order to analyze the sound propagation, detection and localization of direct sound and several reflections. Therefore, the measurements were executed in 433-seat Sejong Chamber Hall (SCH) located in Seoul, South Korea and is described by [19]. It uses reverse-fan shaped and tilted side walls designed with saw-tooth-type diffusers (1-D corrugations) to provide diffusive reflections. The size of the hall is 17m (width) by 7.6 m (Height) by 19 m (Length) with 3,200 m^3 Volume. The mean reverberation time (RT) for this hall is 1.18 seconds.

1) Measurement Procedure: The measurements were carried out for three positions R04, R12 and R20 and the results for only R04 position are represented in this study. An omnidirectional dodecahedron positioned at the center of the stage in the front portion while the microphone array was placed at the three positions during the measurements as show in Fig. 6. Both the source and the microphone array were placed sufficient above the floor taking into account the height of instruments and audience ear positions. The measurement

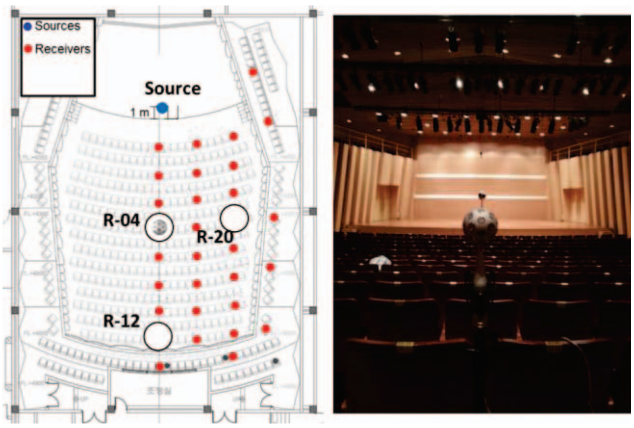

Fig. 6. Source and receiver positions on the floor plan (left), Reverse-fan shaped 433 seat Sejong Chamber Hall (SCH) located in Seoul South Korea (right)

procedure was followed as mentioned in section B and the picture of the measurement setup is shown in Fig. 6.

As shown in Fig. 7, the reference coordinate systems for each measurement position to visualize the processed sound field in a 3D graphical from was chosen in such a way that it considers the center of the sphere as the origin. The line directing backward form center of array as the base line of azimuth angle. The elevation angle is measured from the line stretching to the ceiling, the z-axis [12].

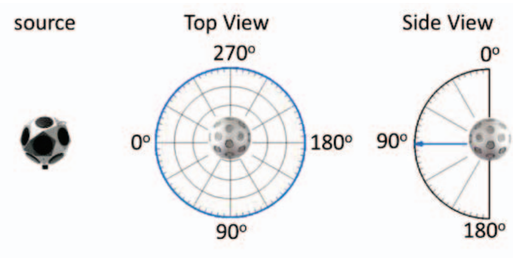

Fig. 7. The reference coordinate systems for spherical microphone array

2) Experimental results and discussion: Plane wave decomposition was employed with spherical harmonics of order six (N=6) while the analysis was performed at the optimum frequency of 2KHz to attain the best presentation of the visualization of sound pressure level [12]. Fig. 8 shows the measured response of array at 2KHz without time windowing for first 80msec data of captured IR. The figure explain spatial distribution of overall sound pressure level of the sound field impinging at the microphone array arriving from all directions and certain dominant peaks appearing from the direct sound and early reflections. From the Fig. 8, it is observed that most of the sound field is scattered from the right side of the hall, front celing (Stage) and left side floor of the hall.

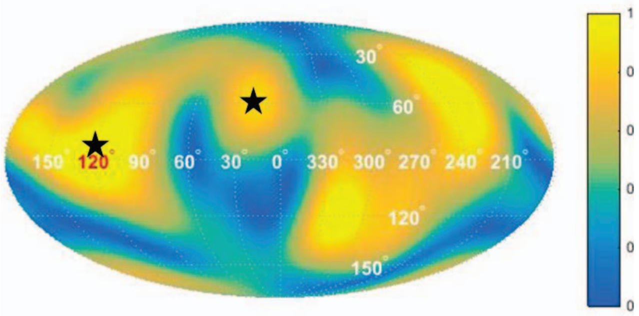

Fig. 8. The first 80 msec data of impulse response for plane wave decomposition captured at the position R04 at SCH

To localize the direct sound and individual reflection's arriving directions a spatio-temporal analysis was performed in order to analyze the influence of the walls scattering elements of the space. One way to compute the directions of the direct sound and the succeeding reflections is the approximation of plane wave decomposition from temporal segmentation of captured data. However, time windowing segmentation of PWD can only predict the direction for all possible reflections.

In this study we proposed MVDR Weighting coefficients described in (10) and utilized in (8) for beamforming at certain steering direction to generate the directional impulse response for the direction of interest. Two steering directions are selected as shown by stars in Fig. 8 at different azimuth and elevation angles, for synthesizing the directional impulse responses. The resultant DIRs are shown in Fig. 9.

DIRs can be used to get several room acoustics indicators such as the directional reverberation times. These impulse responses also provides an estimate of the arrival direction and the location of the surfaces of the hall responsible for reflections and the behavior of the sound propagation from that direction.

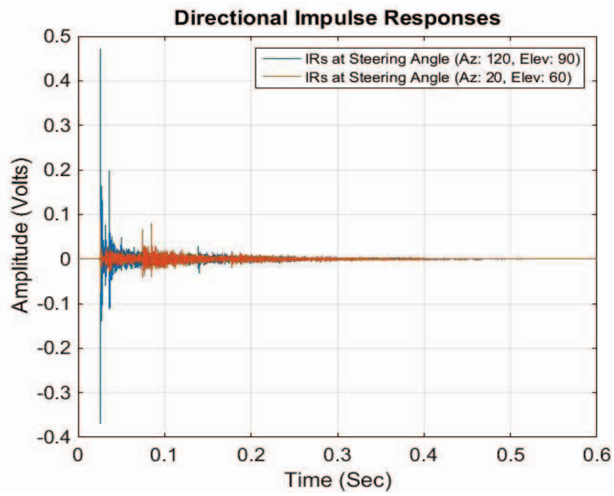

Fig. 9. The first 80 msec data of impulse response for plane wave decomposition captured at the position R04 at SCH. The sound pressure level is normalized

V. Conclusion

In this research, we presented an experimental implementation and validation of algorithms developed for plane wave decomposition and MVDR beamforming using a spherical microphone array that is supported by available theory. Spherical microphone array measurements for spherical harmonics of orders six (N=6) were studied and implemented. The direct sound and early reflections were detected by the array using impulse responses with reasonable accuracy, therefore showing a prospective for a useful acoustics measurement tool. The detection and localization of reflections were improved by applying time windowing to RIRs before subjecting to the plane wave decomposition. The comparison between the results of computed directions of incidence of sound and the directions known beforehand showed a good agreement with each other inferring an effective application of spherical microphone arrays to sound field analysis. The directional impulse responses were synthesized based on MVDR beamformer. In addition, MVDR beamformer is used for multiple source localization for difference steering directions with reasonable accuracy. From DIRs various directional room acoustic parameters can be further computed such as directional reverberation, directional clarity and central time.

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