3-DIMENSIONAL ROOM IMPULSE RESPONSE MEASUREMENTS IN CRITICAL LISTENING SPACES.

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1   INTRODUCTION.

Most of the knowledge and the existing objective acoustic measures defined to describe the sound field in large spaces are commonly evoked to predict the acoustic performance of small rooms. It is now becoming evident that this practice leads to flawed concepts.

This document aims to investigate the importance of directionality of the energy as it decays in a room. An attempt to define a new method to measure the effect of three dimensional reflection density across time is proposed. This method is based on the analysis of the impulse response in three dimensions using B-Format signals.

2   DIRECTIONAL MEASURES.

The temporal characterization of sound decay is based on ratios of impulse response vs. total signal integration[1]. Examples of these metrics are Clarity Index or Early to Late Sound Index (C_{80}), Definition or Deutlichkeit (D_{50}), Early Decay Time (*EDT*), Centre Time, Total Sound Level or Strength (*G*). The Lateral Fraction (*LEF*) uses a combination of an omnidirectional and a figure of eight microphone[2]. The Interaural Cross Correlation[4] *IACC* uses a binaural dummy head microphone in order to compare two pressure signals. The Listener Envelopment *LEV* measures the perception of the level of surround of the sound that is coming from the reverberant field[5].

It is now becoming more apparent that the total energy of the decay may be the key to obtain a sense of envelopment in small rooms[6]. The sound-fields in small rooms need to be characterized with more detail in terms of lateral reflections and level of diffuseness or otherwise[8]. The results that may emerge from the mentioned measures still do not show clearly which can be a desirable acoustic quality standard. The reason may be that they do not take into account such a fast process of sound decay and its spatial and temporal reflection's distribution.

The analysis of directionality of reflections is not new. The principal attempts have been made in controlled acoustic environments such as anechoic chambers but not for critical listening rooms, which usually are non-diffuse spaces due to the non-uniform absorption distribution. Microphone arrays have been used for mapping the energy using time delay differences instead of the coincident approach[9]. Some developments to preserve the acoustical fingerprint of concert halls had been done using B-Format[10], which was an extension of a method proposed by Gerzon[11]. This method focuses on making complete characterisation of impulse responses. This approach may be the basis for developing further attempts to extract directional behaviour of decay.

Thiele was the first to characterize directionality of reflections[12]. Yamazaki and Itow[13], Sekiguchi, Kimura and Hanyuu[14], Okubo et al[15] and Gover, Ryan and Stinson[16] achieved the same outcome but by using a time difference approach, which involves the use of omnidirectional microphone arrays. Dimoulas et al[17] utilised a B-Format hybrid approach to estimate the exact location of a source by using two Soundfield microphones. This was achieved with a combination of coincident microphone and the delay difference technique of triangulation of the source.

3   BACKGROUND THEORY.

3.1 Small Room Acoustics.

The concept of a small room does not directly meet the assumptions for the well known Sabine's theory for reverberation[18]. Its average *RT* at mid frequencies is usually bellow 0.4 s and the existence of true diffuse field conditions in such small volumes is not found. The density of

reflections is not enough to create a diffuse reverberant decay such as the ones found in big rooms[1,21]. Depending on the acoustic treatment in the room, absorption from each surface will lead to a non-uniform flow of acoustic energy across the room[19,22]. The lack of diffusion in such small spaces, especially at low frequencies gives problems of sound colouration. Moreover there is evidence of some complex flow of energy such as vortex modes in 3 dimensions, and its implications for acoustic quality are not completely known[23].

3.2 Room measurements.

Modern acoustic measurements are based on the extraction of information from the transfer function of the room. Rooms can be considered as a three-dimensional filter and its Impulse Response reveals its acoustic properties. Traditionally, acoustic measurements are done with one omnidirectional microphone, which captures the impulse response of the room with a pressure transducer. The directionality of the reflections and the flow of acoustic energy are therefore not obtained.

3.3 Soundfield microphone and the B-Format.

Craven and Gerzon invented the Sound field microphone[24]. This device, originally intended for recording, is able to capture the directionality of a sound field by using a minimum number of coincident microphones. From a single unified format it may encode Mono, Stereo, Surround and Ambisonic signals. It consists of a quasi coincident microphone array that is designed to capture sound in a virtual point inside the array by using four closely positioned capsules in a tetrahedral arrangement. It produces three orthogonal figure of 8 polar responses centred at the origin and a fourth signal that has an omni-directional polar response[25]. Due to the nature of its construction, it is possible to derive Blumlein configurations with virtual pairs of orthogonal signals (X, Y) (X, Z) (Y, Z). These can be used to extract direction of incoming signal on each corresponding plane.

4 DESCRIPTION OF THE TECHNICAL APARATUS.

The measurement technique proposed here retrieves the pressure gradient in the 3 axial coordinates at the listening position by using a Soundfield microphone. For the data presented, acoustic measurements were made in two different rooms with speakers located at fixed angles from the microphone. In the first room, a typical 5.1 monitor configuration was used. In the second room, 3 loudspeaker positions were measured (0°, 45° and 90° see figure 2). A comparison of sound-fields generated from a speaker radiating at 0° in both rooms is presented in section 6.

By analysing the energy and polarity of signals X and Y it is possible to extract the directions of the incoming sound, that is, the direct sound from the speaker and it reflected versions from the room boundaries. The measurements of impulse responses were obtained by using a Soundfield Microphone Model SPS 422B connected to its dedicated Rackmount Control Unit-1U. A fixed microphone gain of -30 dB was chosen. A Focusrite sound card model Saffire PRO 26 I/O with multiple inputs was used. The balanced B-Format output signals W, X, Y, Z from the Rack unit were fed to the sound card inputs and from there to a laptop via a firewire cable. To excite the room, a single Genelec 8040 monitor speaker was used for each position. The selection of the speaker obviously affects the responses measured as shall be discussed later. Nevertheless, one of the novelties that this method attempts to introduce is that of measuring room conditions with the speakers that will be used in the room in the correct. Notwithstanding, if one is assessing the acoustic conditions of the room for purposes other than critical monitoring with the loudspeakers under test, the technique will be somewhat inadequate.

WinMLS software running 4 simultaneous acquisition channels was used for the acoustic analysis. The excitation signal chosen was *Linear Swept Sine* wave as this method is known to maximize the Signal to Noise Ratio (*SNR*) and is more immune to distortion artefacts due to its lower crest factor[27,29]. Depending on the expected *RT* value for the room, the length of the Sweep should be selected accordingly to allow a full measurement of the response (allowing for background noise and extrapolation of T20 or T30). The four impulse responses were obtained by performing simultaneous *RT* measurements by selecting the 4 simultaneous channel acquisition mode. The channels were routed to the sound card as X=1 Y= 2, Z=3 and W= 4. The signal has been acquired

at 16 bits, 48 kHz sample rate. Four channels impulse responses were obtained in B-format. This was imported into Matlab for analysis.

5   DATA ANALYSIS

5.1   Determining the Direction of Energy

The impulse responses to be analysed consist of 3 velocity signals (*X*, *Y*, *Z*) and one pressure signal (W). Each of the impulse responses contains the decaying energy in the room. However, the velocity signals contain this information weighted in terms of direction of arrival of reflections. Reflection direction may thus be extracted from the relative ratios between the amplitudes of these signals.

The algorithm developed identifies the main reflections from peak values of the impulse responses by scanning the signal in *X* and *Y* vectors and then using the 'findpeaks' function to define time windows where a full reflection occurs. The analysis preserves the polarity information (+ or -) and amplitude of the peaks. Each window is comprised of a specific peak corresponding to a reflection. The limits of the window are determined from each reflection peak to the next successive peak.

The algorithm matches the peaks in each of the analysis vectors in order to extract the required amplitude and phase values to determine reflection direction.

Due to the fact that the early reflections are coherent signals with the direct sound, this algorithm can be used to find the second time window that contains the first early reflection, second early reflection and so on. As it enters the reverberant tail, then the coherence between reflection and direct sound is diminished and as such, an analysis of the energy is more adequate. The results presented here, analyse the entirety of the signal in terms of its amplitude level, hence the method for checking reflection coherence has not yet been included in this study.

This approach is very sensitive to the calibration of all signal output from the microphone. If no such calibration has been achieved then the results cannot be relied upon in terms of exact direction of arrival of energy. The *W* channel may be useful to determine a calibration method since it is omni-directional. The other 3 velocity signals should be set such that a source at 45º azimuth and elevation angles equates to a similar output on any channel. Care must be taken when treating the correct magnitude of the *W* signal due to the fact that it need to be corrected by a factor of +3 dB according to the following linear scale expression[33]:

$$W_{corrected} = \sqrt{2}\left(W_{measured}\right) \qquad \{\text{mV}\} \tag{1}$$

The direction of arrival of the energy is defined as opposite to the direction of the particle velocity *u(x,y,z)*. Therefore, the magnitude (*M*) and directions of the incoming reflections ($\theta_{arrival}$) that impinge the Soundfield microphone are calculated with the opposite direction of the orthogonal components of each X,Y and Z vectors[34]:

$$M = \left|\vec{M}\right| = \sqrt{\vec{X}^2 + \vec{Y}^2} \qquad \{\text{mV}\}, \tag{2}$$

$$\theta_{arrival} = \left(\frac{180º}{\pi\,\text{rad}}\right)\tan^{-1}\left(\frac{-\vec{Y}}{-\vec{X}}\right) \qquad \{\,º\,\} \tag{3}$$

5.2   Results

The data is analysed directly through the B-Format impulse responses. Unfortunately, depending on the sample misalignment introduced by the measurement system a number of systematic errors may occur in the analuysis. The following pilot test was run to measure the accuracy of the incoming angle found of the direct sound for each loudspeaker position. Ten measurements were taken per each position in both rooms under test. The Blumlein technique is applied using *X* and *Y*

signals. The horizontal plane only is analysed given its importance for the sense of spaciousness.
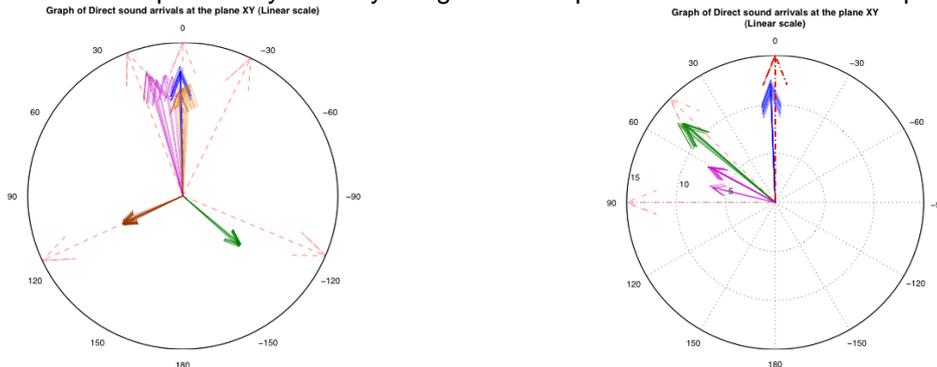
Figure 1: Repeatability of the Soundfield Microphone in Blue room 2 (Left graph) and Live room (Right graph). Left Front speaker and 90⁰ position (Purple), Centre Front (Blue), Right Front (Light Brown), Left Rear (Dark Brown), Right Rear and 45⁰ position (Green), Real directions (dashed lines)

The Repeatability test shows that the method delivers a substantial variability in some directions. For the case of the left side, the Left Rear speaker was the best performer and Left Front speaker was the worst of all. For the case of the right side, the 90⁰ position shows the least consistent localization error. In terms of localization of direct sound the Right Rear speaker and Right Front speaker showed deviation of angle within a maximum variation of ±30⁰. These results motivate the development of an intelligent algorithm for determining the exact direction of the incoming reflections. This is currently being investigated.

6   THE ANALYSIS OF TWO PRACTICAL CASES.

The following analysis was performed in two small rooms: a) A small control room denominated Blue Room 2, which its dimensions are 4.16 X 2.75 X 2.87 m (V=32.1 m³, Mean Free Path = 2.20 m,). This room exhibits non-diffuse decay and has a small *RT* from the order of 0.17 s @ 1 kHz. b) A middle size room denominated Live room, with dimensions of 10.48 X 5.77 X 3.89 m (V=217.2 m³, Mean Free Path = 3.93 m), has a medium *RT* in the order of 0.8 s @ 1 kHz. Both rooms have carpeted floor and absorbent ceiling but the control room has limited areas with reflective walls.

Figure 2: Photos of the Blue Control Room 2 (left) and P.A. room at with speaker at 45⁰(right).
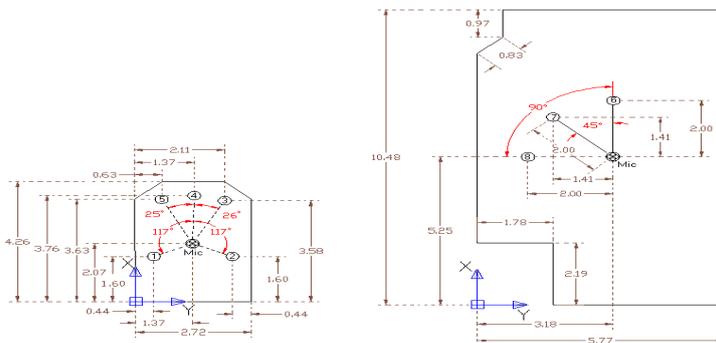
Figure 3: Layout of the speaker placement in Blue Control Room 2 (left) and P.A. room (right).

The purpose of the analysis presented here is to expose the nature of the reflections and how they change over time across the horizontal plane. Eight source positions were measured and compared with the real source of direction. One case, for the centre speaker in each room, is discussed here.

Details for each speaker position relative to the microphone are presented in the following table:

Table 1 - Loudspeaker positions for both rooms

Position ref. to Microphone	Distance (m)	Angle (º)
1.- Left Rear speaker	1.034	117º
2.- Right Rear speaker	1.029	-117º
3.- Right Front speaker	1.682	-26º
4.- Centre Front speaker	1.693	0º
5.- Left Front speaker	1.726	25º
6.- Centre Front speaker	2.000	0º
7.- Left Front speaker	2.000	45º
8.- Left Side speaker	2.000	90º

The following graphs present a distribution of reflected energy in the room throughout its decay. The data presented corresponds to a histogram of direction of reflections for a given time window indicated in the captions. Two rooms are compared: A small, highly damped control room - the Blue room (left); and a large lightly damped live room (right). Speaker positions 4 and 6 from Table 1 are indicated. The direct sound has been omitted for clearer representation.
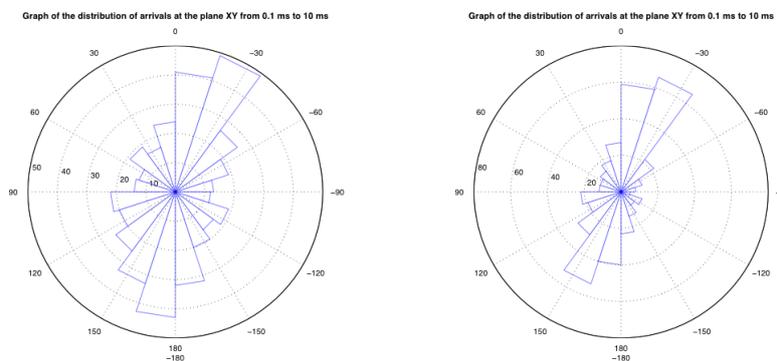
Figure 4: The distribution of early reflections that occur during the first 10 ms after the direct sound.

Blue room (left) and Live room (right).

It is clear that energy concentrates along the main front-back axis of the room during this period, perhaps as expected, since most of the energy is being generated along this axis from the centre speaker. It is interesting to see that the early reflections in the Live room are more directional due to the nature of this position in the room, which only excite the axial direction of the sound. Despite of the exponential increase of reflections density it is still possible to track some of the early reflections and extract the angle of incoming direction. The smaller Blue room has a higher density of reflected energy arriving from the sides since these walls are much closer in this case.
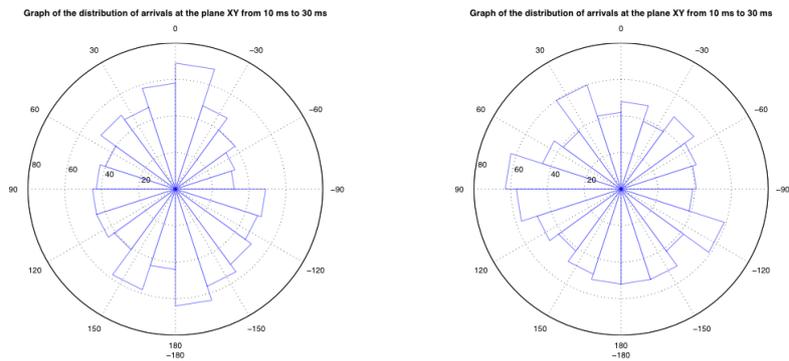
Figure 5: Directional energy density 10ms to 30ms. Blue room (left) and Live room (right).

In this stage there is clear evidence of the spread of reflections in both rooms. A lateralization of the soundfield appears stronger in the Live room which shows a somewhat better distribution of energy across all directions in the plane. The Blue room appears to 'polarise' energy towards the front and back directions.
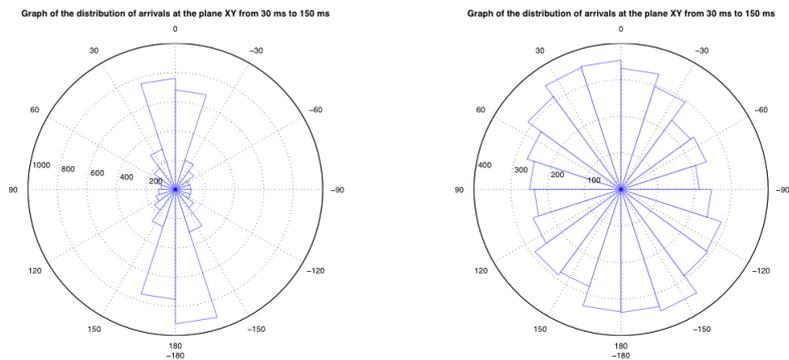
Figure 6: The early reverberant tail from 30 to 150 ms. Blue room (left) and Live room (right).

At this stage, it is clear that the density of reflections has increased substantially by reading the histogram values. Also quite clear is the difference in directional energy spread across the sound field for each room. In the Blue room the reflections are beginning to be grouped around the front and the back of the receiver with a strong polarisation and higher reflection density than the Live room along this axis. Directional energy density in The Live room is closer to homogeneous for all directions, an indication of a diffuse field.
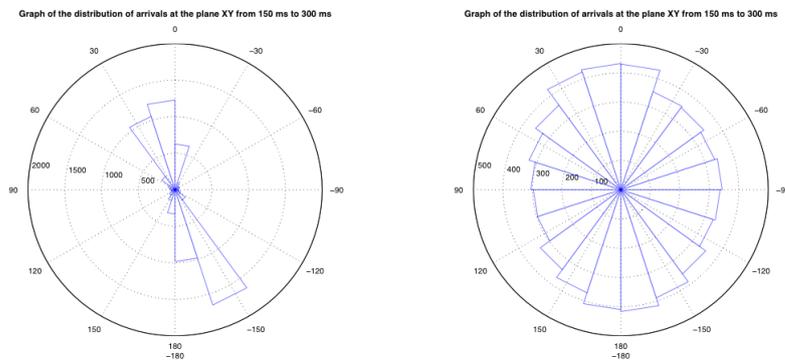
Figure 7: The late reverberant tail from 150 to 300 ms. Blue room (left) and Live room (right).

At this particular stage of the decay, the differences between the two rooms are evident.

Figures 5,6 and 7 clearly show that the highly damped room concentrates most of its reflected energy along the axis of excitation, whereas the larger lightly damped room decays into a sound field which approaches diffuse conditions.

7   CONCLUSIONS

It has been shown that by using a single Soundfield microphone in B-Format it is possible to extract and analyse temporal and spatial distribution of reflections from the sound decay of small rooms. The method is suitable for extraction of direction of energy at the early part of the impulse response. Once the reflections become incoherent with respect to the direct sound, the analysis cannot extract reliable results, but it is still plausible to study the diffuness of the decay by analysing the spatial distribution of reflections across time due to fact that the maximum deviation error of the reflections is still within one third of a quadrant.

Using the present system it is clear that a differentiation between the sound-field of each room under the measurement conditions is possible. A note of caution to the likely changes expected if the source is omni-directional rather than a conventional loudspeaker.

The Method proposed, addresses temporal and spatial issues of the sound decay in small rooms when analysing its reproduction quality. A room can now be analysed in terms of the time it takes for the sound field generated from a given source within it to become diffuse or otherwise. This may well be correlated to changes in subjective perception between rooms and this is the subject of further investigation by the authors.

A complete movie of the decay can be generated and tailored with variable time window frames and may be adapted to show more detail of several parameters if it is required. For more details it can be accessed at http://www.hud.ac.uk/sengjrp.htm

Further work will be done in refining the measurement system, testing the accuracy of incoming reflection angle.

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