AURALISATION OF A SYMPHONY ORCHESTRA WITH ODEON – THE CHAIN FROM MUSICAL INSTRUMENTS TO THE EARDRUMS

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ABSTRACT

Auralisation is a very complicated technique, the quality of which depends on every single part in a long chain, starting with the anechoic recording of the sound to be used as input, continuing through the room acoustic simulation that connects the source and the receiver, and ending in the presentation of the sound to the listening person.

The auralisation of a symphony orchestra is an even bigger challenge because of the large number of sound sources distributed over a considerable area. Recently, a number of high quality anechoic recordings of symphonic music aiming at the multi-source auralisation technique have been available. So, the aim of this paper is to give an overview of the auralisation chain with particular reference to the multi-source auralisation of a symphony orchestra.

1. INTRODUCTION

Auralisation is a technique that has developed to a state that allows many useful applications in research and in room acoustic design. The acoustic quality in the state-of-the-art auralisation may be sufficient if the purpose of auralisation is to demonstrate the difference between alternative room acoustic designs that give different reverberation times, or to demonstrate the difference between basic room shapes like rectangular, fan shape, circular shape etc.

Auralisation can also be an important tool for evaluation of acoustical qualities and defects in room designs. Actually, the ability to demonstrate room acoustical defects is by far the most important requirement for auralisation as a design tool; it is little helpful to auralise the qualities, if there is no warning for possible defects.

Some of the calculation methods in Odeon version 9.2 for obtaining the best possible room impulse response and the application for the multi-source auralisation are explained in this paper.

2. THE AURALISATION CHAIN

Auralisation is the result of a long chain of modelling, assumptions, approximations, and calculations. Logically, the chain can be considered from the source to the receiver:

- **Source**
  - Anechoic recording, microphone position(s)
  - Directivity, resolution in frequency and space
  - The multi-channel method
  - The multi-source method
- **Room**
  - Room geometry modelling
  - Sound reflections, scattering
  - Early and late reflections, 3D incidence
- **Receiver**
  - HRTF (head related transfer function)
  - Generation of BRIR (binaural room impulse response) or surround sound impulse response
  - Reproduction method (headphone / loudspeakers)
  - Level of reproduction, calibration

Being a long chain, it is obvious that the quality of the auralisation is mostly influenced by the weakest link in the chain. Since the introduction of auralisation a lot of efforts have been devoted to the accurate modelling of the receiver, with emphasis on localization, the front-back confusion etc. The importance of modelling the source directivity has been studied recently [1] and a new multi-channel method has been suggested as an attempt to improve this part of the chain [2].

However, when auralisation is used as a tool for room design the accurate room modelling is by far the most important part of the chain. If the auralisation results are presented through loudspeakers instead of headphones, what is often the case, there is obviously no need for the HRTFs, and the accurate receiver modelling may be the least important part of the simulation. However, the level of reproduction should always be considered with great care.

3. SOURCE MODELLING

For room acoustic modelling the single musical instruments are usually modelled as point sources, with or without a frequency dependent directional characteristic. Considering the interaction of musical instruments and the acoustics of a room, it has been shown to be important to take the directivity into account, at least for some instruments like the clarinet, the trumpet and the French horn [1].

As the directional characteristics of musical instruments may change rapidly from one tone to the next, it is a very rough approximation to apply octave band averages of the directivity. As a possible solution to this problem a better, but also much more complicated, method has been suggested, namely the multi-channel method [2].

Considering a group of musical instruments like a symphony orchestra, the normal method has been to apply a nearly anechoic recording of an orchestra to one or two point sources on the stage in the room model. Obviously, this is a very rough acoustical approximation to the large and complicated sound source of a real orchestra. Recently it has been possible to overcome this problem by the multi-source method [3, 4] using synchronised, anechoic recordings of each single instrument, which can then be combined into a full orchestra. With this method it has been possible to improve significantly the quality of auralisation with an orchestra and it has been possible to include important parameters like apparent source width (ASW), balance between instruments, and influence of reflecting surfaces near the orchestra.

4. ROOM MODELLING

4.1. Room geometry

The creation of the room model and especially the amount of geometrical details has an impact on the realism and quality of the acoustic simulations. Today it is often possible to have access to a very detailed 3D model, which may be created by the architect. However, it is questionable which degree of detail is necessary for a good acoustical simulation; too many details may even make the model less useful for the acoustic purpose [5]. For the import of models, e.g. in the dxf interchange format, Odeon has a very efficient import function, which automatically can make geometrical checks and an intelligent reduction of the number of surfaces, see the example in Fig. 1.

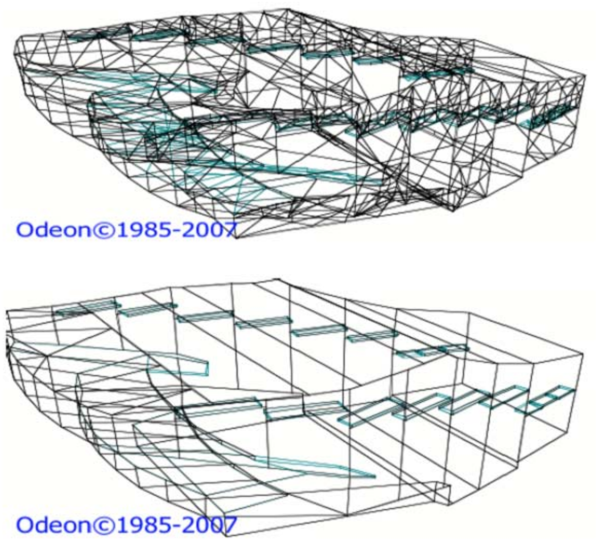

Figure 1: *Example of room geometry imported from a CAD model. Top: Original with 1362 surfaces. Below: After the Odeon Glue Surfaces algorithm is applied, reduced to 209 surfaces.*

It should be noted that the two models in Fig. 1 have the same degree of room geometrical detail, but the subdivision of many significant areas is removed in the lower model.

4.2. Ray tracing and Vector Based Scattering

When the surfaces in the room model have been assigned material properties (absorption and scattering coefficients) and the sources have bee defined, the calculation of the sound propagation can begin. There are several steps in this calculation, the first one being the ray tracing, i.e. to follow a large number of rays emitted from the source in all directions. Typically several thousands of rays are used from each source, and each ray can be followed through several hundred reflections.

In this process the scattering from the surfaces is very important. For a realistic simulation of the reflections of sound waves, even from a very smooth surface, it is necessary to consider scattering effects; even if the scattering coefficient is very small, it should never be set to zero. A scattering coefficient $s = 0$ represents the ideal specular reflection, whereas a value of $s = 1$ represents diffuse reflection, see Fig. 2.

In some early room acoustic models the "Monte-Carlo method" was applied, meaning that with a scattering coefficient of e.g. $s = 0.01$, statistically only 1% of the reflections were diffuse and 99% of the reflections were specular. However, this is a very inefficient method that requires an extremely large number of rays to perform well. In stead the vector based scattering method was developed for Odeon [6]. With this method every reflection is scattered more or less, depending on the mid-frequency scattering coefficient. The specular direction is a vector with the weight (1-s) and the diffuse direction is random and is given the weight s; then the actual direction of the reflected ray is given by the weighted sum of the two vectors, see Fig. 3.

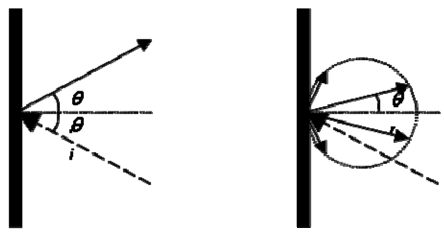

Figure 2: Ideal sound reflection. *Left: Snell's law of specular reflection (s = 0). Right: Lambert's law of diffuse reflection (s = 1).*

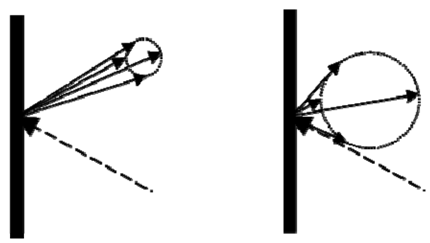

Figure 3: Vector based scattering. *Left: Examples of possible reflection with s = 0,1. Right: Examples of possible reflection with s = 0,9.*

4.3. Angle dependent reflection

The absorption coefficients of the surfaces are used to calculate the attenuation of the reflected sound in octave bands. Although the available absorption coefficients of the materials are usually determined by a standard measurement method in a diffuse sound field, an approximate reflection model has been developed for Odeon that takes into account the area of the reflecting surface as well as the angle of incidence [7]. In the case of a soft material and grazing incidence the phase can change sign.

4.4. Reflection Based Scattering

Due to the wavelength of sound being comparable to typical dimensions of reflecting surfaces in rooms, the scattering effects are very important for a room acoustic simulation model.

The scattering effects can be divided into two; the scattering due to the surface roughness, and the scattering due to edges and finite dimensions of the reflecting surface [6]. Both can be described by a frequency dependent scattering coefficient, s_s and s_d.

The surface scattering coefficient can be measured (at random incidence) according to the method laid down in ISO 17497-1 [8]. It typically increases with the frequency as shown in Fig. 4. Thus, in Odeon the user only needs to assign a single, mid-frequency value for surfaces with high roughness; then the frequency dependent scattering as in Fig. 4 is assumed automatically.

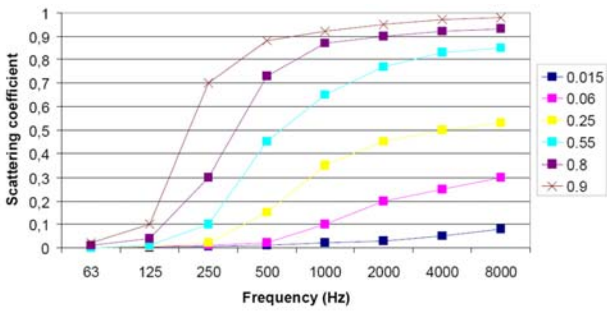

Figure 4: *Frequency dependent scattering coefficients for surfaces with different roughness. The mid-frequency scattering coefficients are indicated in the right side of the graph.*

The diffraction scattering coefficient depends on the surface dimensions, the distance from surface to source and receiver, and the distance of the geometrical reflection point to the nearest edge of the surface. It is also frequency dependent, being higher at low frequencies.

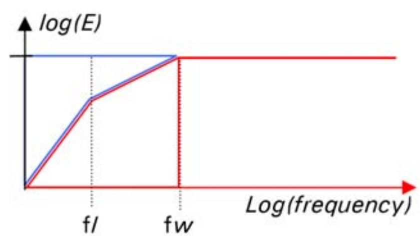

Figure 5: *The reflected energy (in direction of the specular reflection) from a suspended surface. The frequencies f_w and f_l depend on the width and length of the surface, together with angle of incidence and distance to source and receiver.*

In Odeon the combined scattering effect is called the reflection based scattering coefficient s_r and it is calculated from:
where s_d is the scattering coefficient due to diffraction and s_s is the scattering coefficient due to surface roughness.

Based on the theoretical models in [9, 10], the scattering coefficient due to diffraction is calculated from:
where K_w and K_l are attenuation factors calculated from the width and length, respectively, of the reflecting surface. The reflected energy without scattering due to diffraction is proportional to the product of the attenuation factors, which are characterised by the limiting frequencies, f_w and f_l, see Fig. 5. In addition, s_e is an edge scattering coefficient depending on the distance of the reflection point from the nearest edge of the surface [9]. It follows from the diffraction theory that the scattering increases with the distance from surface to source and receiver.

An example of how the reflection based scattering works is shown in Fig. 6. If a relatively small surface (e.g. a table) reflects sound from a source close to the surface, there is very little scattering, but when the source is far away from the surface the scattering is high, in accordance with the diffraction theory.

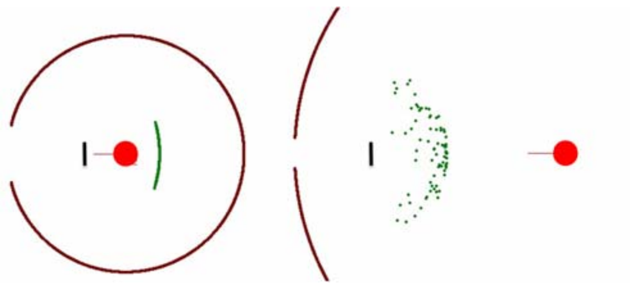

Figure 6: *Reflection based scattering from a small surface. Left: source close to surface. Right: Source far from surface.*

5. THE IMPULSE RESPONSE

5.1. Calculating the reflections in a receiver point

When the ray tracing is finished the next step is to calculate the impulse response in the receiver point. Early reflections (e.g. up to 2nd or 3rd order) can be treated by the image source method, but the big question is how to calculate the late reflections to get reliable results without excessively long calculation times.

In the early ray tracing models the receiver could either be a sphere, big enough to catch the rays, or the receiver was a point that could be catched by a cone around the ray. However, none of the methods are reliable in a room model with complicated geometry. Instead a new method was developed in Odeon for the late reflections, as described below.

For the calculation of late reflections, the rays are treated as carriers of energy. Each time a ray hits a surface, a secondary source is generated at the collision point, see Fig. 7 upper part. The energy of the secondary source is the total energy of the primary source divided by the number of rays and multiplied by the reflection coefficients of the surfaces involved in the ray's history up to that point. Each secondary source is considered to radiate into a hemisphere with a directivity that may take the frequency dependent scattering of the surface and the angle of incidence into account. The contributions from the secondary sources are collected from the receiver position, see Fig. 7 lower part. The time of arrival of a reflection is determined by the sum of the path lengths from the primary source to the secondary source via intermediate reflecting surfaces plus the distance from the secondary source to the receiver. A visibility test is made to ensure that a secondary source only contributes a reflection if it is visible from the receiver. Thus the late reflections are specific to a certain receiver position and it is possible to take shielding and convex room shapes into account.

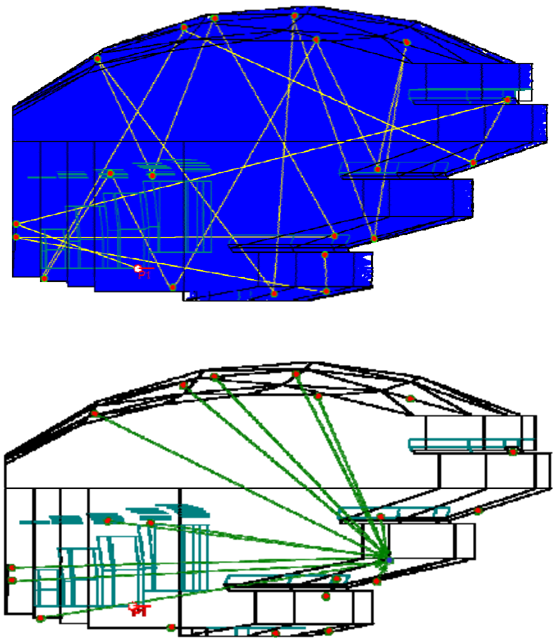

Figure 7: *Principle of late reflections in Odeon. Above: Secondary sources created on the surfaces by one of the rays in the ray tracing. Below: A receiver point collects contributions from the visible secondary sources.*

5.2. Auralisation

As a part of the point response calculation, impulse responses for auralisation are created, either Binaural Room Impulse Responses (BRIR) for binaural auralisation over headphones or multi channel impulse responses for auralisation over a surround sound system. Finally auralisation output is also available in B-Format for further processing. No matter the final method for presentation, impulse responses are composed from information on each of the reflections arriving at the receiver; the amplitude of the reflection in nine octave bands from 63 Hz to 16 kHz and the angles of incidence [11]. To obtain a satisfactory quality of auralisation, usually a reflection density of 50-100 reflections per millisecond is needed, thus a typical impulse response of 2 seconds duration is based on at least 100.000 reflections. If the sound field is far from a diffuse field and a majority of the reflections carry very little of the total energy, higher density will be needed.

5.3. Binaural auralisation

The key to binaural auralisation is the Head Related Transfer Function (HRTF). Each of the reflections is filtered through a set of transfer functions depending on the angle of incidence (both azimuth and elevation). In Odeon the HRTFs can be imported, so it is possible to use individual HRTFs or e.g. the HRTFs available from CIPIC [12] or MIT [13].

The “Achilles' heel” to binaural simulation and reproduction is that the HRTF must be individualized for optimum performance, i.e. it must be measured on the individual for whom the auralisation is to be synthesized. Also the transfer function of the headphones used for reproduction must be compensated for. Without individual HRTF the out-of-head experience tends to be disappointing and frontal localization almost impossible.

In practice it is possible to compensate for the headphones used, but it is unrealistic to measure the HRTF of each listener and to create simulations for each individual. Therefore a facility has been created that can enhance the localization obtained from a set HRTF. The method combines several methods that have been suggested over the last decade or so, one of them being an enhancement of the dips and notches in the frequency responses of the HRFT [14]. The result is enhanced localisation cues from non-individualized HRTFs and this greatly improves the out-of-head experience as well as frontal localisation without any noticeable colouration.

5.4. Surround sound auralisation

Surround sound is obtained through the Ambisonics technology [15-18]. Each reflection is added to a 3D energy histogram at a given sampling frequency storing the total energy per octave as well as the intensity per octave as a vector. Finally, the histogram is encoded into a B-Format impulse response, each sample being added with random sign of the sound pressure. This is decoded into a number of impulse responses corresponding to the number of speakers to be used for reproduction.

In Odeon the user may specify the positions of the loudspeakers (minimum 2 and maximum 50 satellites and one optional subwoofer) and the decoding for that particular loudspeaker rig will be optimized (whether loudspeakers are arranged in a circle, a rectangle or whatever). In most cases the output selected is probably a ‘standard setup’ such as a 2.0, 2.1, 4.1, 5.1, or 7.1 system. In those cases the wave files produced by Odeon can be reproduced directly or played back using the Window Media Player (assuming that an appropriate sound card and loudspeaker system is present).

5.5. Convolution

No matter which of methods of auralisation is used, the next step in the auralisation chain is convolution. If the method is binaural, the 2-channel BRIR is convolved with a mono signal producing a stereo output, which can be played back on a set of headphones. For a 5.1 Surround sound impulse response, six convolutions are carried out producing a 6-channel wave file to be reproduced over a surround system.

5.6. Mixing auralisations

In order to produce auralisations with several sources and signals, the convolved signals can be mixed into one single file, which may for example include all the 'voices' of an orchestra auralised with their correct location in the 3D space. Again the number of channels in the output depends on the auralisation method selected. In Odeon the point response simulation, convolver and mixer are integrated into one user interface, allowing a complete orchestra simulation to be carried out in one single operation. The mixer allows a delay and level adjustment to be added to each track and having the mixer integrated in the same user interface, it is automatically secured that tracks are only allowed to be combined if they are compatible, i.e. having the same receiver-position and –orientation.

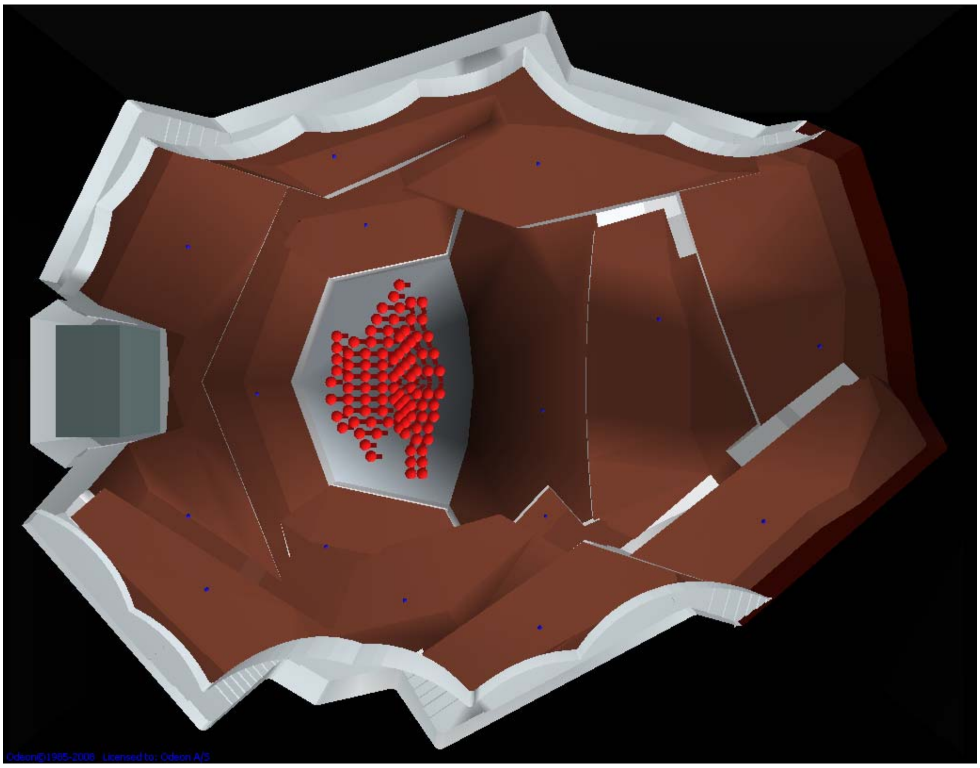

Figure 8: *A view from above into the Odeon model of the DR Concert Hall. The red spots indicate source positions and the small blue spots are listener positions that have been selected for simulations.*

6. A NEW CONCERT HALL WITH A SIMULATED ORCHESTRA

6.1. The new DR Concert Hall, Copenhagen

The new Concert Hall of the Danish Radio in Copenhagen, Denmark has been chosen as an example for this paper. The concert hall has been designed by the architect Jean Nouvel and Mr. Toyota from Nagata Acoustics has been the acoustic advisor.

A view into the Odeon model of the hall is seen in Fig. 8. The principle of the design is the same as in the Berlin New Philharmonic, i.e. with the orchestra near the centre and the audience placed in different terraces surrounding the orchestra. Above the stage is a large suspended reflector. The hall was inaugurated in January 2009, and it has a capacity of 1800 in the audience. The volume is 28.000 m^3 and the reverberation time is designed to be around 2.4 s at 500 Hz with a full audience. Another view from one of the listener positions is seen in Fig. 9.

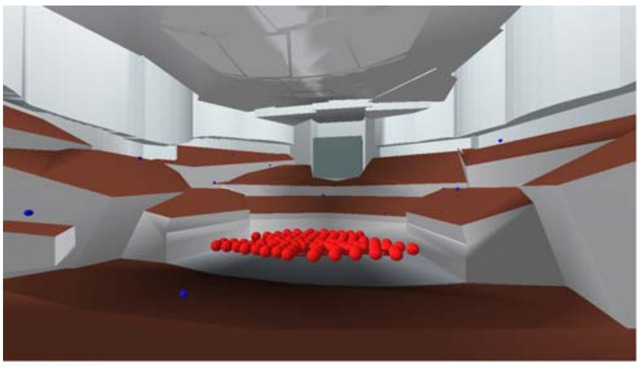

Figure 9: *A view into the Odeon model of the DR Concert Hal from one of the selected listener positions.*

6.2. The Orchestra setup for a Mahler symphony

The auralisations described here were with a part from the 1st Symphony of Gustav Mahler [11]. An overview of the set-up is seen in Fig. 8. In total 95 musicians were simulated and 39 individual anechoic recordings were applied, see Table 1.

Table 1: *Musical instruments used for the auralisation of Mahler's 1st Symphony*

Instrument	Number of sources	Number of Recordings	Directional characteristic
1st violin	16	2	Violin
2nd violin	14	2	Violin
Viola	12	1	Violin
Cello	10	1	Omni
Double bass	8	1	Omni
Flute	4	2	Omni
Oboe	4	4	B-Clarinet
Clarinet	5	4	B-Clarinet
Bassoon	3	3	B-Clarinet
French horn	7	7	French horn
Trumpet	4	4	Trumpet
Trombone	3	3	Trumpet
Tuba	1	1	Omni
Percussion	4	4	Omni

As for each group of the string section, only one or two recordings were used and played simultaneously from a large number of sources. The 1st and 2nd flute were playing the same (*unisono*), and so were the 3rd and 4th flute. Concerning the clarinets, there are four different parts, but in the score the composer has requested at least two musicians to play the 4th clarinet (the E Flat clarinet), so there are in total five clarinet players. Only four different directional characteristics were used as approximations for the most directional instruments, and an omni directional source was used for the less directional instruments, see Table 1.

7. CONCLUSION

With the multi-source auralisation of a symphony orchestra it has become possible to evaluate by listening some of the more subtle acoustic qualities of a concert hall, like apparent source width and balance between groups of instruments. However, as described above every part of the auralisation chain is important for obtaining a result with sufficiently high quality.

It remains to be mentioned, that for this application of auralisation it is an unsolved problem how to separate the influence of the music being played and the ears listening from the acoustical performance of the hall.

8. ACKNOWLEDGMENTS

The Odeon model of the DR Concert Hall was kindly made available for these orchestra auralisations by Dr. Anders Chr. Gade, and the use of the model for the orchestra auralisations was kindly accepted by Mr. Toyota. The anechoic recordings of the Mahler Symphony were made by Jukka Pätynen, Ville Pulkki, and Tapio Lokki from Helsinki University of Technology with musicians from various Finnish orchestras [19]. The directional characteristics were measured by Felipe Otondo [1].

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