



Stage acoustics and sound exposure in performance and rehearsal spaces

Methods for physical measurements

Remy Wenmaekers

/ Department of the Built Environment

bouwstenen 233

Stage acoustics and sound exposure in performance and rehearsal spaces for orchestras

Methods for physical measurements

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Summary

Title

Stage acoustics and sound exposure in performance and rehearsal spaces for orchestras: methods for physical measurements

Introduction

Good acoustics in the concert hall is important for the audience to be able to fully appreciate music. Additionally, good acoustics is necessary for symphony orchestra members to be able to hear each other and to optimise their performance. This research is focussed on the so called ‘stage acoustics’ in performance spaces and its judgement by physical measurements. Such measurements are used by acousticians during the design of a performance space or rehearsal room to control or check its acoustic quality for unamplified orchestral music. Physical measures have been introduced by other researchers that are linked to perceptual attributes such as ‘the ease of ensemble’ and ‘the amount of reverberance’. The goal of current research is to further develop the procedures for physical measurements on stage. A special interest is the relation between the outcome of the measurements and the stage and building design. Another acoustic aspect related to the performance of music is the exposure of musicians to high sound levels. Sound levels are often that high that musicians risk damaging their ears. The topic sound exposure is also dealt with in the current research.

Problem

So far, the correlation between the physical parameters that are commonly used and their perceptual counterparts has shown to be weak. Researchers have not always been able to confirm the findings by others. The question is whether this is caused by shortcomings in perceptual testing methods/strategies or by shortcomings in physical parameters or measurement methods. On one hand, it was proven to be hard to get reliable and reproducible musicians’ opinions. On the other hand, many researchers have doubts whether the methods used to measure physical stage acoustic parameters are sufficiently accurate and appropriate. Most researchers have performed studies that involved both perceptual evaluations and physical measurements. However, the number of halls investigated in such studies has proven to be too limited to find significant indicators for the acoustic quality of the stage. Researchers must join forces and collect data that can be analysed with more statistical power. Before starting to collect this new data, it is necessary to know which research methods are most suitable.

Methods and results

The current research focusses on defining optimal measurement methods to obtain stage acoustic parameters from a metrological viewpoint. Intentionally, there is no attempt made to verify or find new optimal ranges for physical parameter values. The definition of the common stage acoustic parameters ST_{early} and ST_{late} have been extended and optimised. Acoustic measurements have been performed in a wide variety of performance spaces and rehearsal rooms to test the extended parameters. Besides, the calibration process of the omnidirectional sound source used has been studied by measurements in the acoustic laboratory. The uncertainty of various calibration methods has been established leading to guidelines for calibrating the sound source. Another factor of uncertainty is the influence of the orchestra members on the acoustics on stage. So far, it has been common to measure on unoccupied stages under the assumption that the chairs and music stands sufficiently represent the obstruction of sound by the orchestra members. To test this assumption, extensive measurements have been performed using an orchestra of dressed mannequins on five different stages, orchestra pits and rehearsal rooms. Results showed that in most cases, parameter values are significantly influenced by the orchestra and a measurement protocol has been established that can be used with a real orchestra on stage. Furthermore, the influence of the directivity of the musical instruments and human sound perception has been studied by comparing measurements performed with omnidirectional and directional transducers. This showed that most current stage acoustic parameters are only moderately sensitive to the direction of sound on real stages.

A special interest is the application of the physical parameters to evaluate sound exposure. A prediction model has been developed that can calculate sound levels at the ears of musicians under different (acoustic) conditions. The model has been validated by measurements in a symphony orchestra and the model was proven to be sufficiently accurate. Next, the model was used to study the impact of common measures to control sound levels in the orchestra such as available space, screens and acoustic treatment. It was shown that realistic control measures are not effective enough to obtain significant sound exposure reductions for most instruments except for the low strings section. Among others, this is caused by the contribution of the own instrument to the sound levels in the orchestra for most instrument types. So far, hearing protection devices such as ear plugs are the only effective solution to preserve the hearing of musicians.

Outcome

The work presented in this dissertation contributes to an understanding of how stage acoustic parameters are measured most accurately and how they can be used to study sound levels on stage. Guidelines are presented that will assist researchers in future studies and engineers in designing or judging stages and rehearsal rooms.

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Chapter 1

Introduction

1.1 Stage acoustics and sound exposure

For over 100 years, acousticians have been focusing on the acoustics in a performance space from an audience point of view, judging reverberance, clarity and spaciousness, as summarized by Barron [1] and Beranek [2]. In the past four decades, also the importance of good acoustic conditions on stage, referred to as stage acoustics, has been recognized by musicians as well as many acousticians [3-15]. Gade [4] identified three acoustic factors related to the musical performance that seem most important for musicians on stage: hearing yourself without the need to force the instrument (support), hearing other musicians (ease of ensemble) and hearing the hall (reverberance), illustrated in Figure 1.1. The acoustics on stage influence the appreciation of these factors by acoustic properties such as reverberation time, level of reflected sound, time delays and frequency balance. Subsequently, architectural properties of the hall, such as room volume, shape, dimensions and materials, influence the stage acoustics properties.

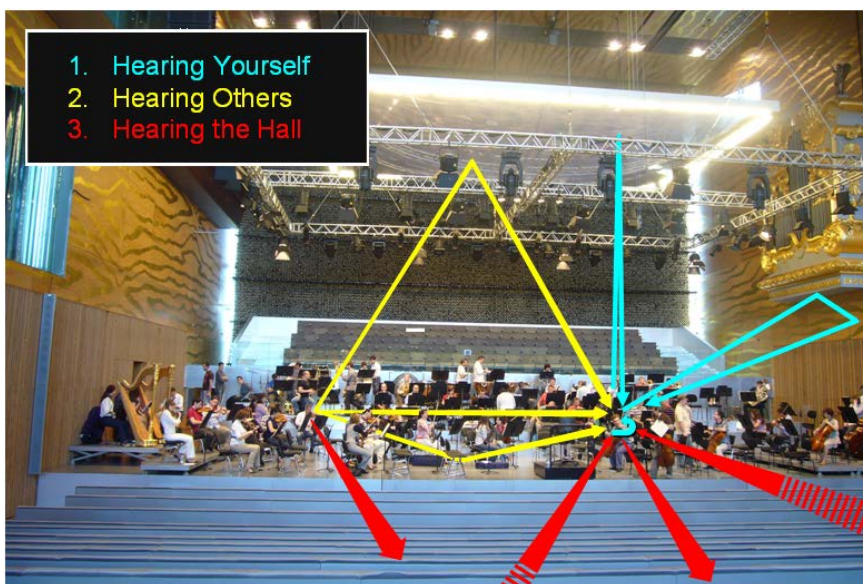


Figure 1.1: Sound paths on the stage of concert hall Casa da Musica (Porto, Portugal).

The following examples illustrate how perceptual, acoustical and architectural aspects are related: Sufficient room volume is needed for the room to be reverberant enough when the room is occupied by sound absorbing seats and persons. With a given reverberation time, the room volume determines the strength, related to the perception of loudness. Both reverberation time and strength are important for the perception of reverberance [49]. Surfaces close to the stage, such as side walls, back wall and ceiling panels reflect the sound back to the stage with a short delay. The level and delay of this ‘early’ reflected sound influences the perception of support and ease of ensemble. Also, direct sound is beneficial for hearing each other, which is influenced by risers and by the distances between players.

Acoustic conditions are also important for the musicians in relation to sound pressure levels which are often above 85 dB(A) in the orchestra and can cause hearing damage [16-20]. Safety and health agencies are working with musicians, conductors and orchestra directors on controlling the sound exposure¹ of members of professional orchestras [21, 22]. To limit the sound levels, measures are proposed like the introduction of screens and very high risers within the orchestra or even ear plugs [20, 23]. A study in 2012 in the orchestra pit of Het Muziektheater Amsterdam, see Figure 1.2, showed the importance of the stage² environment on the occurring sound levels [24]. A comparison was made between the sound levels within the orchestra for two different acoustic configurations of the orchestra pit while playing the same musical piece. Differences up to 3 dB (i.e. sound energy doubling) were found caused by the different acoustic configurations. The configuration with reduced sound levels was also appreciated more in terms of the ability to hear each other. This shows that stage acoustics is important from a musical as well as an occupational health point of view.



Figure 1.2: Sound exposure measured in the orchestra pit of Het Muziektheater Amsterdam [24].

¹ “Sound exposure” is the common term for “noise exposure” when dealing with music

² In this thesis, an orchestra pit or rehearsal room is also considered to have ‘stage acoustics’

1.2 Examples of stage acoustics research

Laboratory experiments have been conducted to investigate the properties of the acoustic sound field that are valued by musicians. An important example of such an experiment, performed by Gade [25], is shown in Figure 1.3. Musicians were positioned in anechoic rooms and recorded with directional microphones. A virtual sound field was created using electronic delays and a reverberation room which was played back to the musicians via loudspeakers. This way, the musicians could play together in real time while the sound field was varied by the researchers. The audibility of reflections for solo players was tested and the threshold levels were found to be different for violin players compared to flute players. In general, a high level of early reflections and reverberation was preferred. The further musicians are sitting apart, the shorter the delay and the higher level of early reflected sound had to be.

The experiments were conducted using artificial reflections delayed by 23, 43 and 83 ms relative to time of emission. From these experiments it was concluded that for early reflected sound a time window of approximately 20 to 100 ms relative to the own direct sound is relevant for support and ensemble playing. The sound arriving after 100 ms was suggested to be relevant for the loudness of the late reverberation.

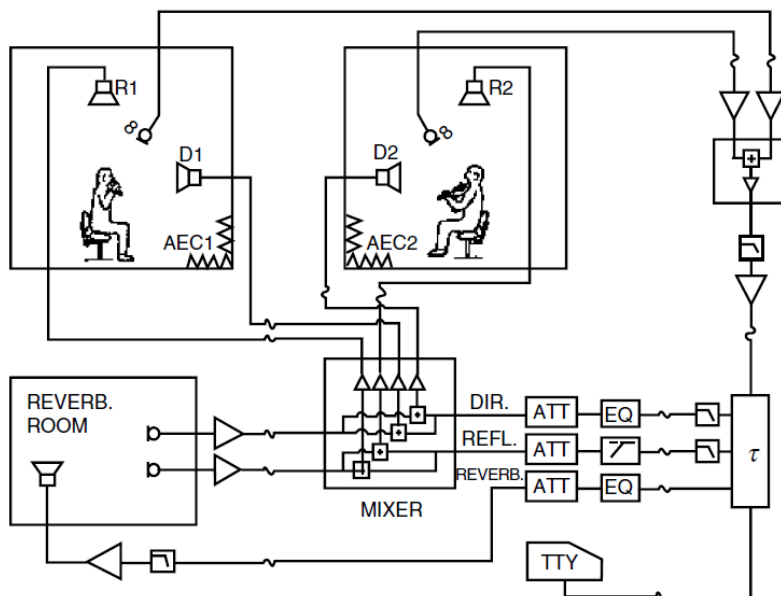


Figure 1.3: Diagram of arrangement for simulation of ensemble conditions in a large orchestra [25].

A maximum delay of 100 ms means that early reflection path lengths should not be longer than approximately 34 m. Considering that a musician is sitting 1 m away from the stage wall, the maximum width of stage should therefore be $17 + 1 = 18$ m. Figure 1.4 illustrates the time intervals of sound paths on a typical 18 m wide and 12 m deep stage, and a canopy or ceiling at 10 m height. The sound sources are assumed to be synchronised. The direct sound of ‘yourself’ arrives at the musician within $t=1$ ms (blue dot) and most 1st order early reflections of ‘yourself’ arrive within 100 ms (blue dashed line).

The direct sound of ‘others’ can arrive up to 60 ms after the ‘departure’ of the sound from the sound source, depending on the mutual distance up to 20 m (yellow circles). When the mutual distance between musicians increases, the time window within which their 1st order reflection can arrive from stage surroundings narrows.

Thus, most early 1st order reflected sound from ‘own’ and ‘others’ arrives within 100 ms after the departure of the sound (blue and yellow dashed lines) and most late reflected sound coming back from the hall arrives after 100 ms. This indicates that a stage with common dimensions indeed facilitates the need for 1st order reflections to be early enough.

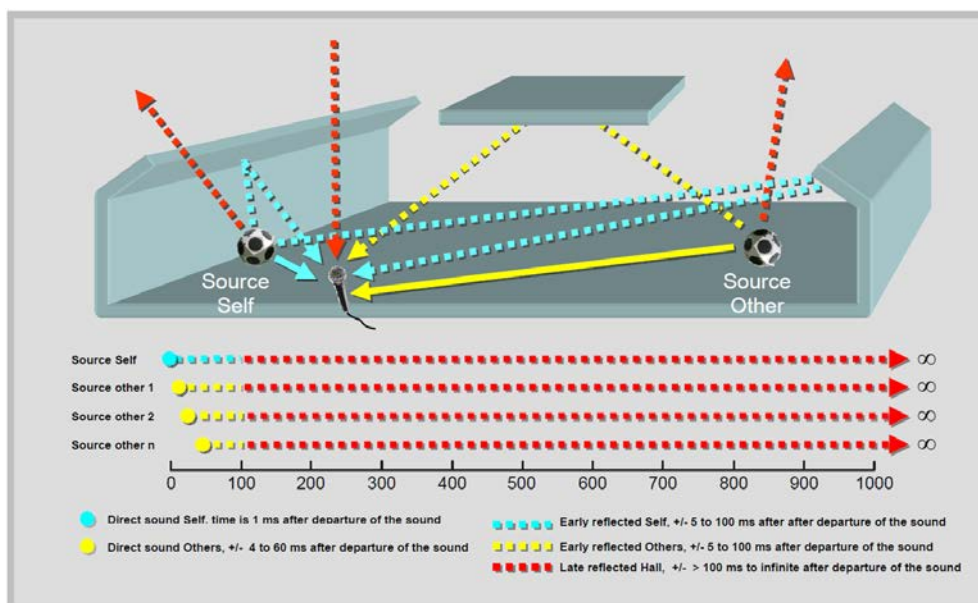


Figure 1.4: Typical time intervals of sound paths on a stage.

Besides the delay of reflected sound, its strength also is considered. Acoustic parameters have been developed that aim to measure the strength of reflected sound. The most commonly used stage acoustic parameters are Early Support or ST_{early} , that relates to ‘Ensemble conditions’ and Late Support or ST_{late} that relates to ‘Perceived reverberance’. At 1 m source to receiver distance, they measure the ratio in dB of the sound energy of early reflected sound (arriving 20 to 100 ms after the direct sound) or late reflected sound (arriving 100 to 1000 ms after the direct sound) and the sound energy of the direct sound. See equations 1.1 and 1.2, where p is the pressure in the impulse response. The parameters are a result of studies by Gade [25-27] and are also mentioned in the ISO 3382-1 standard [28]. Other parameters, which will be discussed in chapter 2, mostly follow a similar concept.

$$ST_{early} = 10 \lg \left(\frac{\int_{20}^{100} p^2(t) dt}{\int_0^{10} p^2(t) dt} \right) \quad [1.1], \quad ST_{late} = 10 \lg \left(\frac{\int_{100}^{1000} p^2(t) dt}{\int_0^{10} p^2(t) dt} \right) \quad [1.2]$$

A number of surveys were performed by Gade whereby musicians were asked to judge the acoustic conditions on stage in various halls. An example of results from one study is shown in Figure 1.5, where 12 different configurations of one stage were compared. A relation was found between the average ‘Ease of Ensemble’ and the acoustic parameter ST_{early} (even though individual musicians often disagreed with each other). On average, musicians preferred the configurations with the highest values for ST_{early} . Based on all measured data from various concert halls, the optimal range was estimated to be -13 to -11 dB for ST_{early} .

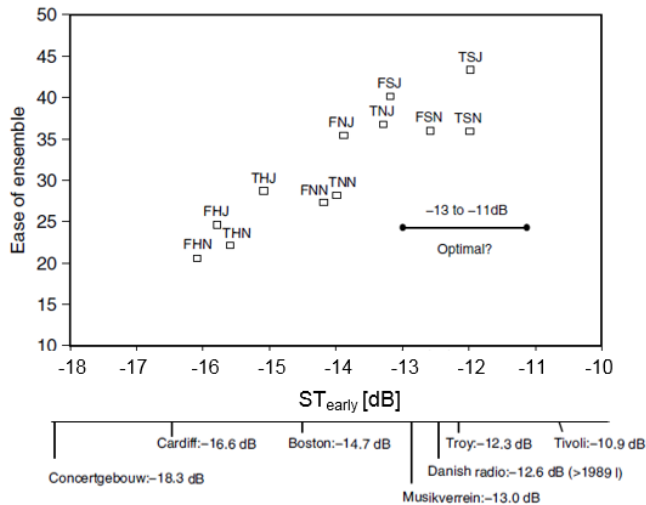


Figure 1.5: Ease of ensemble versus ST_{early} from experiment in the old Danish Radio Concert Hall (squares in the graph) and other halls (below graph) [29].

1.3 Current design guidelines for stages

Research has inspired acousticians in developing guidelines for the design of stages. Current guidelines are summarised in this section in detail to give insight in the relevance of research on stage acoustics and also because they are not treated elsewhere in this thesis.

Appropriate acoustics for the listener in the hall, such as proper reverberation, is an essential starting point for the design of performance spaces to (also) satisfy the needs of performers. In a next step, the stage acoustic design focusses on the stage and its surrounding surfaces. One of the first research-based guidelines come from Marshall et al. [30] in 1978:

- Early reflections of sound between musicians are essential;
- There is a minimum and maximum delay for early reflections to be useful;
- Horizontal reflections are preferred over vertical reflections;
- High frequency components in reflections are most important;
- Reflections from different instrument groups should be balanced;
- The reduction of sound due to the presence of the orchestra should be compensated.

The question is how to translate such desires into architectural solutions. In 1980, Allen [31] recognised many common design features of concert hall stages that could provide musicians with proper acoustic conditions. A recessed stage at the back of a shoebox hall would naturally form walls and balconies reflecting early sound back to the orchestra. The typical exposed stage surrounded by audience usually contained side reflecting walls by elevating the audience on the back and sides of the stage. Ceiling reflectors had mostly been applied to improve weak spots in the audience area, but could potentially be used to provide reflected sound from musician to musician (even though Marshall pointed out that lateral reflections were preferred). A relatively small performance area would avoid long delays of direct and early reflected sound (as the conductor Sir Maldom Sargent suggested to “never give musicians all the space they want” [31]). Another common feature was the use of racked platforms or risers, beneficial for both the audience and musicians to improve clarity.

Gade provides an extensive overview of important design considerations for stages in section 3.8 of Barron’s book *Auditorium Acoustics and Architectural Design* [32] from 1993 (updated in 2010). He notes that floor space tends to have increased over time, with only 130 m² in the Vienna Musikvereinssaal while newer halls sometimes offering up to 250 m². Gade suggests that 200 m² of total area is sufficient for a 100 piece orchestra and larger stages need variable walls to adapt their size (such as the Gulbenkian hall by Allen [33] and studied by Barron [34]). On too large stages, musicians tend to spread out leading to delay problems or move too close to the stage edge away from reflecting surfaces. To further improve direct sound paths, detailed requirements for risers are given and a semi-circular shape is preferred by the strings players.

Gade suggests that the width of the stage should not exceed 18 m resulting in a depth up to 12 m. Dammerud [11] also concludes that relatively narrow stages are preferred (out of the 6 stages he analysed, the 2 narrowest stages are 16.1 and 17.6 m, while the two widest stages are 19.5 and 20.8 m). Dammerud mentions that stages should not be too deep either,

because the French horns need a reflective surface behind them. In contrast, a questionnaire among musicians by Meyer [35] showed that they preferred a width of at least 18 m, which is highly contradictory. Gade mentions that the stage side walls should be oriented such that sound is reflected back to the orchestra. This is confirmed by Dammerud and he additionally suggests that the side walls near the loud instruments should be fan shaped or absorptive to avoid the sound of loud instruments to be excessive on stage. Adding absorption behind loud instruments is also mentioned by Kahle to be liked by musicians [26]. To provide horizontally reflected sound, most authors have mentioned the use of tilted upper wall parts [33, 26, 11], up to 2.5-3 m height [11].

Additionally, ceiling reflectors might be necessary in halls with high ceilings. There is little agreement on the size, amount and height of such reflectors. According to Gade [32], 8 to 10 m height is an appropriate range. For instance, an array of reflectors at 5 m was found unsatisfactory by a large orchestra [36]. Meyer suggests that from an acoustic point of view the height should be between 6 and 10 m. But when asked to musicians, Meyer found that they prefer a height of 10 m and more. Such conditions are in line with Dammerud who found that high ceilings seem to be preferred based on questionnaires with musicians who judged different stages. His explanation for this is that sound from the loud instruments in the back is reflected via the ceiling or reflector towards the soft instruments in the front and vice versa. The advantage for the wind and brass players is that they can hear the strings better, but at the same time wind and brass instruments are too loud for the strings. Meyer suggested that a tilted reflector above the wind and brass instruments would take advantage of their directivity in the frontal direction and solve this problem.

Architectural guidelines are seldom a direct result of experimental studies. Often, researchers focus on the understanding of musicians' preferences related to acoustical aspects such as delay and level of reflections or the value of acoustical parameters (with the exception of Dammerud who found that architectural parameters are better predictors for preference). It is also worth mentioning that some researchers used computer modelling to investigate the relation between geometry and the acoustic parameters [37, 38], however focussing on recital stages rather than large stages for a symphony orchestra.

Ueno concludes that there is sufficient proof that both early and late levels of reflected sound are important [39], but optimal ranges for parameters related to such levels are not conclusive. For instance, in 1989, Gade suggested an optimal range for ST_{early} of -12 +/-1 dB, which could not be confirmed by Dammerud [11]. In 2007, Gade suggested a lower value of ST_{early} of -14 dB in the Springer Handbook of Acoustics [40] as a guideline for concert hall stages. This is within the range mentioned by Beranek which is -12 to -15 dB [41]. For ST_{late} , Gade found that the preferred halls had values between -12 and -13 dB, while Dammerud found optimal values between -17 and -15 dB.³ However, no guidelines have been mentioned in literature for ST_{late} or any other stage acoustic parameter besides ST_{early} .

³ In chapter 7 and 8 of this thesis an average preferred range is used for ST_{late} of -14 +/- 1 dB which included Gade's lower limit and Dammerud's upper limit.

1.4 Problems related to stage acoustic research

The need for relevant and reliable stage acoustic parameters was emphasized in the keynote lecture by Vorländer titled “What do we know in room acoustics?” presented at the international conference on acoustics Forum Acusticum 2011 [42]. Much research has been performed to compare perceptual evaluations to results of measured physical parameters on various stages, for instance Gade [25-27], Dammerud [11], Giovannini [10], Luxemburg et al. [43]. Other type of research has recently been done by Schärer Kalkandjiev and Weinzierl [44] who analysed performance attributes from recordings of actual performances and compared them to the physical parameters. Unfortunately, results from these studies have shown that the correlations between the musician’s judgement of stage environments or musicians’ performance and the available physical parameters are often rather weak or even non-existent. The question is whether this is caused by shortcomings in perceptual testing methods or strategies or by shortcomings in physical parameters or measurement methods.

On one hand, it was proven to be hard to get reliable and reproducible musicians’ opinions [25-27, 11, 10, 43]. This is because even if musicians observe differences in acoustic conditions they find it difficult to describe them in (different) acoustical terms. Besides, the preferences of acoustic conditions are very personal and instrument dependent. An excellent overview of the state of the art on this topic has most recently been documented by Schärer Kalkandjiev [45].

On the other hand, many researchers have doubts whether the standardised method used to measure physical stage acoustic parameters is sufficiently accurate and appropriate [10, 11, 33-40]. Two of the most recent studies that (also) dealt with this topic were performed by Dammerud [15] in 2010 and by Guthrie [46] in 2014. Dammerud covers a broad scope of topics related to stage acoustics in his thesis.

Most researchers studying stage acoustics have performed studies that involved both perceptual evaluations and physical measurements. As Gade concludes [29], the number of halls investigated in such studies (including his own) has proven to be too limited to find significant indicators for the acoustic quality of the stage. Researchers must join forces and collect data that can be analysed with more statistical power. Hopefully, stronger evidence will be found for relations between musicians’ perception of acoustics on stage and physical parameters that are used to predict them.

Before starting to collect this new data, it is necessary to know which research methods are most suitable. The current research focusses on defining optimal measurement methods to be used in the physical measurement from a metrological viewpoint. Perceptual aspects of stage acoustics are not dealt with in detail. Intentionally, there is no attempt made to verify or find new optimal ranges for physical parameter values. A special interest is the application of physical parameters such as ST_{early} and ST_{late} to evaluate sound exposure.

1.5 Problem statements

Based on the existing literature, a number of problems have been identified, which will be covered in the current thesis. Measurement uncertainties caused by the following metrological issues need further exploration:

- 1 The current choices for the time intervals in the available parameter formulas varies for different researches. Besides, researchers have a desire to study acoustic parameters as a function of distance, not captured in the commonly used ST parameters that are measured at 1 m only. It is unknown what impact the choice of time interval limits has on the accuracy and sensitivity of the parameters for judging and comparing stages.
- 2a The directivity of the commonly used sound source for stage acoustic measurements is not fully omnidirectional. The directionality affects most energy ratio parameters when measuring acoustic parameters relatively close to the sound source [47]. Its effect on stage acoustic parameters in particular is unknown.
- 2b Some parameters use the direct sound measured on stage to determine the sound power of the sound source (ST_{early} and ST_{late} following ISO 3382-1), others use various types of sound power measurements in the laboratory as a reference (ST_{early} , ST_{late} following Gade [27] and G_{early} , G_{7-50} , G_{late} [11]). The measurement uncertainty of these methods is unknown.

And, the following influences of the musicians and/or orchestra are not taken into account in the current methods:

- 3 All parameters are commonly determined on an unoccupied stage, preferably with chairs and stands. Research by O’Keefe [48] and Dammerud [11] suggests that the attenuation of sound of the orchestra on stage should be taken into account. The impact of this attenuation on stage acoustic parameter values has not been investigated on real stages.
- 4 All parameters are commonly determined by using a (nearly) omnidirectional sound source and microphone. However, most musical instruments are highly directional above the 500 Hz octave band [6]. Also, musicians make use of directional hearing by their ears. Possibly, directional transducers are more relevant for stage acoustic measurements.
- 5 The symphony orchestra consists of more than 80 instruments playing in various groups [6]. Current single or average parameter results from stage acoustic measurements do not take into account the number of instruments and the properties of those instruments like sound power and frequency spectrum [29].
- 6 No research has considered to use stage acoustic parameters for judging sound levels on stage which might be important in relation to the sound exposure of musicians.

1.6 Outline

The above mentioned problems have been addressed in this thesis and the results are presented in the next chapters as six self-contained studies. New insights from each study have been used in the subsequent studies. The studies have already been published in peer reviewed journals or have been submitted. Each chapter starts with an abstract and an introduction of the background, theory and literature.

Chapter 2 (*problem 1*) starts with a literature study on the time intervals used in stage acoustic parameters based on energy ratios. It contains an extensive review of all previous work on this topic. After a discussion of the architectural and musical relevance of the time intervals, an optimisation and extension of the most commonly used parameters, i.e. ST_{early} and ST_{late} , is proposed. These parameters can be measured as a function of distance. After that, the impact of using the various parameters is investigated by measurements on concert hall stages.

Chapter 3 (*problem 2a and 2b*) deals with fundamental metrological issues related to measurements with a dodecahedron as an omnidirectional sound source. Various methods for determining its sound power are investigated based on literature and measurements. The uncertainty of these methods is determined and used to find the most appropriate calibration method. Additionally, the directivity of various dodecahedron sound sources has been compared and a new method of using rotational averages is validated.

Chapter 4 (*problem 3*) contains a study on the impact of the orchestra members on the measured acoustic parameters. The optimised methods described in chapter 2 and 3 are used in this study. A mannequin orchestra has been used as a substitute for a real orchestra, which was necessary to be able to perform a large number of measurements. Acoustic parameters are studied for five different stages and orchestra pits in occupied and unoccupied conditions. A method is presented and validated to measure stage acoustic parameters with a real orchestra on stage.

Chapter 5 (*problem 4*) is a further investigation of measurements on occupied stages and pits with mannequins, this time using directional transducers. A single loudspeaker was used with a directivity similar to a trumpet and a head and torso simulator was used with microphones placed in the ears of a dummy. The sensitivity of various stage acoustic parameters to directivity is presented.

Chapter 6 (*problem 5*) is very different from the other chapters as a model is proposed for calculating sound levels in the orchestra. Measurement data is used as an input for the model such as instrument directivity, anechoic music and the proposed stage acoustic parameters. The aim is to calculate the possible impact of acoustic conditions on the sound level at each musician's position.

Chapter 7 (*problem 5/6*) is a continuation of the work on the sound level prediction model. Model input data are fine-tuned based on the work as presented in chapter 4. Next, a validation of the model is presented based on measurements with an orchestra in three different halls. The possibilities of acoustic measures to reduce sound levels in the orchestra are explored.

The thesis ends with a summary of the most important conclusions and recommendations.

Errors or uncertainties have been judged based on a just noticeable difference of 2 dB for the *ST* parameters based on personal communication with Gade. It should be noted that no experimental proof exists for this estimated JND. If the JND would turn out to be very different, than some conclusions regarding the impact of measurement uncertainty presented in this thesis may have to be reconsidered.

1.7 Contribution of others

The chapters that have been published as journal or conference papers are co-authored by colleagues and master students. It should be noted that all written material was produced by the current author and at least 90% of the work originates from the current author.

1.8 References

- [1] M. Barron: Auditorium Acoustics and Architectural Design. London: Spon Press, 1993.
- [2] L.L. Beranek: Concert Halls and Opera Houses: Music, Acoustics, and Architecture. 2nd edn. New York: Springer-Verlag, 2004.
- [3] A.H. Marshall, D. Gottlob, H. Alrutz: Acoustical conditions preferred for ensemble. J. Acoustic. Soc. Am. 64 (1978) 1437-1442.
- [4] A.C. Gade: Musicians' ideas about room acoustic qualities. Tech. rept. No. 31. Technical University of Denmark, 1981.
- [5] G.M. Naylor: The achievement of ensemble. Appl. Acoust. 23 (1988) 109-120.
- [6] J. Meyer: The Sound of the Orchestra. J. Audio Eng. Soc. 41 (1993) 203-213.
- [7] J.S. Bradley: Some effects of orchestra shells. J. Acoust. Soc. Am. 100 (1996) 889-898.
- [8] K. Ueno, H. Tachibana: Cognitive modeling of musicians' perception in concert halls. Acoust. Sci.& Tech. 26(2) (2005) 156-161.
- [9] J.Y. Jeon, M. Barron: Evaluation of stage acoustics in Seoul Arts Center Concert Hall by measuring stage support. J. Acoust. Soc. Am. 117(1) (2005) 232-239.
- [10] M. Giovannini: Measurement and analysis methodologies for acoustical characterisation of spaces for classical music performance. PhD Thesis, Politecnico di Torino, 2007.
- [11] J.J. Dammerud: Stage acoustics for symphony orchestras in concert halls. PhD thesis, Department of Architecture and Civil Engineering, University of Bath, 2010.
- [12] E.W.M. van den Braak, C.C.J.M. Hak, H.J. Martin, L.C.J. van Luxemburg: Influence of stage risers on stage acoustics. Proceedings Forum Acusticum 2005, Budapest, pp. 2421-2426.
- [13] E.W.M. van den Braak, L.C.J. van Luxemburg: New (stage) parameter for conductor's acoustics? Proceedings of the 155th ASA conference, Paris, 2008.

- [14] E.W.M. van den Braak, C.C.J.M. Hak, L.C.J. van Luxemburg: New (stage) parameter for conductor's acoustics? - part 2: further investigation on LQ7-40. Proceedings of Internoise 2009 Ottawa, Canada, 2009.
- [15] L.C.J. van Luxemburg, R.H.C. Wenmaekers, C.C.J.M. Hak, M. Kivits: Stage acoustics further development of parameter LQ7-40. Proceedings of Internoise 2010 Lisbon, Portugal.
- [16] E. Jansson, A. Axelsson, F. Lindgren, K. Karlsson, T. Olaussen: Do musicians of the symphony orchestra become deaf? Pages 62-74 of: Acoustics for Choir and Orchestra, 1986.
- [17] K.R. Kähäri, A. Axelsson, P.A. Hellström, G. Zachau: Hearing assessment of classical orchestral musicians. Scand. Audiol., 30 (2001) 13-23.
- [18] J. Lee, A. Behar, H. Kunov, W. Wong. Musicians' noise exposure in orchestra pit. Appl. Acoust. 66 (2005) 919-931.
- [19] J. H. Schmidt: Hearing changes in classical musicians and risk factors. PhD thesis, Faculty of Health Sciences, University of Southern Denmark, Odense, 2011.
- [20] E.J.M. Janssen, H.W. Helleman, W.A. Dreschler, J.A.P.M. de Laat: Noise induced hearing loss and other hearing complaints among musicians of symphony orchestras. Int. Arch. Occup. Environ. Health, 82 (2009) 153-164.
- [21] Arboconvenant Podiumkunsten. Fysieke en psychosociale belasting, schadelijk versterkt geluid vroegtijdige re-integratie. Arbo, 's Gravenhage 15 mei 2002.
- [22] M. Coronel: Gezond Musiceren. Brochure, Stichting Arbopodium, 2007.
- [23] A. Behar, W. Wong, H. Kunov: Risk of Hearing Loss in Orchestra Musicians: Review of the Literature. Medical Problems of Performing Artists, Dec 2006, 164-168.
- [24] Level Acoustics: Onderzoek naar mogelijkheden tot reductie van de geluidbelasting van de musici in Het Muziektheater A'dam. Report LA.91204.R02, 21 March 2012.
- [25] A.C. Gade: Investigations of Musicians' Room Acoustic Conditions in Concert Halls. Part I: Methods and Laboratory Experiments. Acustica 69 (1989) 193-203.
- [26] A.C. Gade: Investigations of Musicians' Room Acoustic Conditions in Concert Halls. Part II: Field Experiments and Synthesis of Results. Acustica 69 (1989) 249-262.
- [27] A.C. Gade: Practical aspects of room acoustic measurements on orchestra platforms. 14th ICA Beijing F3-5, 1992.
- [28] ISO 3382-1: Acoustics - Measurement of room acoustic parameters – Part 1: Performance rooms. International Organization for Standardization, 2006.
- [29] A.C. Gade: Acoustics for symphony orchestras; status after three decades of experimental research. Proc. of International symposium on room acoustics, IRSA 2010.
- [30] A.H. Marshall, D. Gottlob, H. Alrutz: Acoustical conditions preferred for ensemble. J. Acoustic. Soc. Am. 64 (1978) 1437-1442.
- [31] W. A. Allen: Music stage design. Journal of Sound and Vibration, 69(1) (1980) 143-147.
- [32] M. Barron: Auditorium Acoustics and Architectural Design. E & FN SPON, London (1993), pp. 56-61.
- [33] W. A. Allen: The Gulbenkian Great Hall, Lisbon, I: Acoustic design. Journal of Sound and Vibration, 59(4) (1978) 469-479.
- [34] M. Barron: The Gulbenkian Great Hall, Lisbon, II: an acoustic study of a concert hall with variable stage. Journal of Sound and Vibration, 59(4) (1978) 481-502.
- [35] J. Meyer: Trends in concert hall design - experiences of the last 50 years. Proceedings of Forum Acusticum Aalborg, 2011.

- [36] E. Kahle: Acoustic feedback for performers on stage-return from experience. Proceedings of ISMRA 2016.
- [37] W.Chiang, Y.Shu: Acoustical design of stages with large plane surfaces in rectangular recital halls. Appl. Acoust. 64 No. 9, pp. 863-884, 2003.
- [38] I. Karagiannis, A. Sotiropoulou: Computer simulation in exploring stage acoustics design. Proceedings of ICSV Athens (2016).
- [39] K. Ueno, H. Tachibana: A consideration on acoustic properties on concert-hall stages. Building Acoustics, 18(3-4) (2011) 221-235.
- [40] T. Rossing, ed.: Springer handbook of acoustics. Springer Science & Business Media, 2015, pg. 301-350.
- [41] L. Beranek: Concert halls and opera houses: music, acoustics, and architecture. Springer Science & Business Media, 2012.
- [42] M. Vorländer: What do we know in room acoustics? Proceedings of Forum Acusticum 2011, Aalborg.
- [43] L.C.J. van Luxemburg, P.H. Heijnen, M. Kivits, C.C.J.M. Hak: Stage acoustics experiments on 7 stages of concert halls in the Netherlands. Proceedings of Internoise 2009 Ottawa, Canada.
- [44] Z. Schärer Kalkandjiev, S. Weinzierl: The influence of room acoustics on solo music performance: An empirical case study. Acta Acustica unit with Acustica, 99(3) (2013) 433-441.
- [45] Z. Schärer Kalkandjiev: The Influence of Room Acoustics on Solo Music Performances. An Empirical Investigation. PhD thesis, Audio Communication Group, University of Berlin, 2015.
- [46] A. Guthrie: Stage acoustics for musicians: a multidimensional approach using 3D ambisonic technology. PhD Thesis, Rensselaer Polytechnic Institute, Troy, New York, 2014.
- [47] R.S. Martín, I. Witew, M. Arana, M. Vorländer: Influence of the source orientation on the measurement of acoustic parameters. Acta. Acust. united Ac., Vol. 93(3) (2007) 387-397.
- [48] J. O' Keefe: Reflection Density and Attenuation on Stages and Rehearsal Halls. Proceedings of the International Symposium on Room Acoustics, ISRA, 2010.
- [49] D. Lee, D. Cabrera, W.L. Martens: The effect of loudness on the reverberance of music: Reverberance prediction using loudness models. The Journal of the Acoustical Society of America, 131(2) (2012) 1194-1205.

Chapter 2

Stage acoustic parameters

Original Title: On Measurements of Stage Acoustic Parameters: Time Interval Limits and various Source-Receiver Distances
Authors: R.H.C. Wenmaekers, C.C.J.M. Hak, L.C.J. van Luxemburg
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Abstract

The most widely recognized objective stage acoustic parameters are the Early Support (ST_{early}) and Late Support (ST_{late}). In these parameters the early and late reflected sound energy is measured within a certain time interval. Different time interval limits have been proposed for the stage acoustic parameters but there is no agreement on the preferable limits. There is a growing interest to measure stage acoustic parameters at various source to receiver distances. In this chapter the influence of perceptual and architectural parameters, synchronicity, source to receiver relationship and the measurement system on stage acoustic parameters is discussed. Based on existing and new insights an optimization and extension of the ST parameters is proposed such that they can be measured at any distance between source and receiver using the most appropriate time interval limits. Theoretical assumptions were checked and confirmed based on systematic analyses of measured results for different concert hall stages with various conditions and various source to receiver distances.

2.1 Introduction

Stage acoustics is concerned with the experience and appreciation of the acoustics by performers on stage in concert halls, opera houses, theatres and other venues for performing arts. Important acoustical factors for performers are the hearing of their own instrument, the hearing of others' instruments and the hearing of the acoustic response of the hall. The balance between these factors is important for playing ensemble [1, 2, 3, 4, 5]. Various objective parameters have been introduced to describe stage acoustics, all based on acoustic properties that can be found in room impulse responses. On stage, the room impulse response (RIR) contains the direct sound and sound reflected from the stage surroundings and the hall. Most objective parameters measure the amount of sound energy within a certain time interval by integrating over the squared sound pressure of a RIR. The choice of the time limits of the time interval may be based on a perceptual or architectural relevance, but might be influenced or limited by the used measurement methods.

Researchers have proposed different time limits; it is clear that there is no agreement on these time interval limits. So far, no studies have investigated the effect of the choice of time limits on measured stage parameters. In this chapter, the available parameters and their metrological issues are discussed in section 2.2. In section 2.3, the perceptual, architectural and orchestral principles of time delays on stage are explored. In section 2.4 it will be discussed which time intervals and measurement conditions seem most appropriate resulting in a proposal for optimisation and extension of currently used stage acoustic parameters. In section 2.5, theoretical assumptions are checked using a large set of measurement data of different concert halls. The stage acoustics of the measured halls is evaluated through the optimised and extended stage acoustic parameters in section 2.6. Finally, the results are discussed in 2.7 and the main conclusions are presented in section 2.8.

2.2 Stage acoustic parameters

The most widely recognized objective stage acoustic measures are those as proposed by Gade. These measures are based on laboratory and field experiments as described in [3, 4]. Experiences with these parameters have been discussed by Gade in [6, 7]. Two parameters have been included in the annex of the standard ISO 3382-1 [8] on the 'Measurement of room acoustic parameters - Performance spaces' since 1997: Early Support (ST_{early}) and Late Support (ST_{late}). Other parameters like ST_2 or ST_{total} , Clarity Stage (CS) and Early Ensemble Level (EEL) have not been included in the standard. In the following subsection the development of the various parameters will be treated. This development clearly illustrates the issues and discussions related to time intervals on stages.

2.2.1 Early Support

The ST_{early} was intended as a measure to describe the assistance of early reflections to the hearing of the own instrument. At 1 m distance, it measures the difference between the

reflected sound level within the 20-100 ms time interval after the arrival of the direct sound and the sound level of the direct sound plus floor reflection, measured in the time interval 0-10 ms of the RIR [8].

Originally, the lower time limit of the time interval 20 to 100 ms was 10 ms instead of 20 ms [2]. However, the lower limit was changed to 20 ms because it was difficult to isolate the direct sound within 10 ms in the low octave bands with the available measurement techniques using octave band filtered sweeps [9]. As a consequence, in the current ST_{early} parameter a gap exists between 10 and 20 ms where sound energy is not taken into account. Therefore, in practice, it is recommended to place the transducers at a minimum distance of 4 m from any stage boundary of interest to avoid sound arriving within the 10 to 20 ms time interval [6, 10].

Two different upper time limits 100 ms and 200 ms were used in two different parameters $ST1$ and $ST2$ respectively. The name $ST1$ was replaced by ST_{early} in 1992 to avoid confusion with another parameter Speech Transmission Index STI [6]. $ST2$ was replaced by ST_{total} with a numerator time interval of 20 to 1,000 ms ($ST2$ is sometimes mistaken by researchers for ST_{late} [11]). $ST2$ was intended to describe the amount of support from the room. However, it was not included in the ISO standard. According to [7], it has never been investigated whether the 100 ms upper time limit for ST_{early} is the optimum choice.

2.2.2 Late Support

The parameter Late Support (ST_{late}) was intended as a measure to describe the perception of reverberance. At 1 m distance, it measures the difference between the reflected sound level in the time interval 100-1,000 ms after the arrival of direct sound and the sound level of the direct sound plus floor reflection, measured in the time interval 0-10 ms of the RIR [8].

Originally, the clarity C_{80} parameter was used at 1 m distance to measure late support, also known as Clarity Stage (CS), measuring the ratio of sound energy arriving within the time interval 0-80 ms and after 80 ms to infinity [8]. In the CS, it is assumed that the amount of energy before 80 ms at 1 m distance is dominated by the direct sound [4]. The energy after 10 ms was omitted in the numerator, the upper boundary increased from 80 to 100 ms and numerator and denominator switched to match the ST_{early} definition resulting in the currently used ST_{late} .

At first, time to infinity was chosen as an upper time limit instead of 1,000 ms [3, 4, 5]. However, in 1992 the upper time limit was fixed to 1,000 ms, comparable to the C_{80} parameter definition that was used in [5], although Reichardt et al. [12] originally proposed infinity. At that time, the 1,000 ms upper limit was chosen in order to save calculation time. According to [9], the influence is negligible in typical acoustic conditions of a concert hall.

The ratio between ST_{early} and ST_{late} seems to be useful for describing the degree of masking of ensemble information by late reflections [6].

2.2.3 Early Ensemble Level

The parameter Early Ensemble Level (*EEL*) was intended as a measure to describe the ease of hearing others [3]. It measures the difference between the direct and reflected sound level within the time interval 0-80 ms after the emission of the sound and the sound level of the direct sound plus floor reflection, measured in the time interval 0-10 ms of the RIR at 1 m distance. An interesting feature of this parameter is its sensitivity to the effect of the time delay that occurs between musicians sitting at a distance from each other. The temporal window for measured arriving direct and reflected sound narrows when the source and receiver are further apart. As a result, *EEL* can only be measured at a distance between 2 and 27 m.

It is striking that an upper time limit of 80 ms was used in the *EEL* instead of the 100 ms in the *ST* parameters. However, this may have been different for the sake of comparison with the clarity C_{80} [9]. To the knowledge of the authors, it has never been investigated whether the 80 ms upper integration limit is the optimum choice for the *EEL*.

After comparison of the parameters with measured subjective parameters by questionnaires ST_{early} appeared to correlate better with the ‘hearing of others’ or the ‘ease of ensemble’ than *EEL*, which was originally intended for this use [4]. This paradox is not yet fully explained [7]. Nevertheless, ST_{early} is recommended to be used in relation to ‘ensemble i.e. ease of hearing other members of an orchestra’ and *EEL* was not used in further analyses or added to the standard [3, 4, 8].

2.2.4 Measurement conditions

The RIR for deriving the *ST* parameters should be obtained at 1 m distance between an omnidirectional sound source and receiver, chosen as a distance to be comparable to the distance from the performer’s ear to his own instrument. The sound source distance is measured from the physical centre of the sound source, described in the ISO standard as the ‘acoustic centre’. Both the sound source and the microphone height from the floor should be either 1.0 m or 1.5 m [8]. However, in an earlier version of the standard from 2006 the transducers’ height could vary between 1.0 and 1.5 m. No suggestions are made for the choice of transducer height. Nevertheless, Gade recommends using a 1.0 m transducer height because it represents the acoustic centre of most musical instruments best and because it may give a more realistic effect of eventual attenuation of furniture and seats [9].

It is recommended to perform at least 3 measurements at different positions on stage and the positions should be reported [8]. No comments are made on the orientation of the microphone relative to the sound source. In the concept of the *ST* parameters, the sound source represents the instrument and the microphone represents the musicians’ ears. In most cases the instrument is in front of the musicians who are facing the conductor. This suggests that the sound source and microphone should be put in a line crossing the conductors’ position, where the source is in between the conductors’ position and the microphone. In

most research in which *ST* parameters were measured, the source and receiver are represented as a single location on stage without reporting source to receiver orientation [3, 10].

As mentioned earlier, all transducers should at least be kept 4 m away from any stage boundaries. Also, when the orchestra is not present during measurements, as is often the case, chairs and music stands must be present. However, chairs and stands should be removed in a radius of 2 m around the transducers to avoid reflections arriving within the time interval 0 to 10 ms [6, 8]. It is striking that between 2 and 4 m distance around the transducers seats and stands are allowed, possibly causing reflections in the 10-20 ms time interval.

2.2.5 The reference sound level

In the concept of the *ST* parameters and *EEL*, the direct sound of the sound source at 1 m distance is used as a reference, sometimes denoted $E_e(\text{dir})$ where ‘e’ stands for emission (which is especially important for *EEL*). In current practice, the direct sound and floor reflection are measured within the time interval 0-10 ms [8]. In case of transducers heights of 1.0 m the floor reflection will arrive 3.6 ms after the direct sound. For the 250 Hz octave band, which is most critical, these two components will be smeared out over the whole time interval 0-10 ms [6]. When the transducers are set at a height of 1.5 m the floor reflection will arrive 6.3 ms after the direct sound.

2.2.6 Variations on the support parameters

Some research presented at conferences has discussed the measurement method for the direct sound reference $E_e(\text{dir})$. It was suggested that the floor reflection should be absorbed by sound absorbing material [13], but an average difference was found of less than 0.5 dB. Also it was suggested to use a floor-reflection-free time interval of 0-5 ms [14]. One other issue is how to deal with the directional characteristics of the standard omnidirectional sound source at high frequencies. It should be noted that researchers have used different types of omnidirectional sound sources. For instance, in the 80’s Gade used an icosahedron sound source with a 500 mm diameter containing 20 loudspeakers, while nowadays the dodecahedron sound source is commonly used with a smaller diameter of approximately 350 mm containing only 12 loudspeakers. This may result in less smooth directivity patterns especially at high frequencies. If the sound source is not adequately omnidirectional, Gade [6] suggested that the source should be rotated such that always the same directivity maximum is pointing towards the microphone. For this position, the deviation from the sound level at 1 m distance derived from a sound power measurement in the laboratory should be determined and used as a correction. A comparable option was suggested by Dammerud et al. [15], who suggested to use the sound power calibration methods as described in [8]. However, these methods exclude the influence of the floor reflection. Hak et al. [16] showed that for a single measurement at 1 m distance and a 1.5 m transducer height a maximum absolute error of 1.5 dB, 3.5 dB and 3.5 dB can be made in measured sound level for octave bands 1, 2 and 4 kHz respectively using a common standard omnidirectional sound source

on stage. Maximum possible level deviations are reduced to below 0.5 dB for all frequency bands when averaging over 5, 7 or 8 measurements of equal angular stepwise rotation. Furthermore, it was shown that the difference between the average value derived from three different precision G calibration methods and an (in situ) on stage sound power measurement at 1 m distance is -0.8, 0.0 and +0.7 dB for the low, mid and high frequency range respectively when using 8 impulse responses while rotating the sound source in 45 degree steps [17, see [chapter 3](#) for the continuation of this study]. These differences are similar to the accuracy of the three precision methods themselves. This suggests that the $E_e(\text{dir})$ component can be measured on stage with 0.5 dB accuracy, when using an average of 5, 7 or 8 measurements of equal angular stepwise rotation, without removing the floor reflection or performing off-site calibrations. The influence of directivity deviation is only investigated for the total sound level but has not yet been investigated for room acoustical parameters.

Another series of papers have discussed possible variations in time intervals for the ST parameters. It was suggested by Chiang et al. to extend the ST_{early} time interval from 20-100 ms to 7-100 ms [18] or 5-80 ms [19]. This way, ST_{early} could be used to perform measurements closer to the stage boundaries. These parameters were denoted $ED100$ and $ED80$ respectively. Comparable to the $ED80$, the $LD80$ was suggested extending the ST_{late} time interval from 100-1,000 ms to 80 ms to infinity. In a similar way, the Late Sound Strength G_l was suggested by Dammerud [15], where the time interval 80 ms to infinity is measured at 1 m distance relative to the reference sound level at 10 m in the free field. When using 1,000 ms as a upper time limit, the impulse response should not contain noise within the 0-1,000 ms interval, but this is not easily controlled. Using time to infinity instead of 1,000 ms as an upper time limit is perceptually clearer. However, in both cases the noise tail of the impulse response can be of influence on the result. The relationship between the impulse response and the noise tail (the decay range) can be described by the Impulse to Noise Ratio (INR) [20]. An INR of at least 45 dB for a measured RIR on stage is recommended by the authors, where time to infinity is defined as the time of the cross point between the decay curve and the noise floor of the impulse response [21].

Also, parameters similar to the EEL have been proposed in some conference papers. Braak & Van Luxemburg [22] proposed a parameter, denoted LQ_{7-40} , which is measured where source and receiver positions represent different musicians on stage. The parameter measures the difference between the early reflected sound level in the time interval 7-40 ms and the late reflected sound level in the time interval 40 to infinity. It is different from the EEL concept because the direct sound is omitted from the time interval. Also, the time interval is not dependent on the source to receiver distance like the EEL . The masking of ensemble information by late reflections is measured, comparable to the ratio between ST_{early} and ST_{late} . No reference is made to the sound level at 1 m distance.

Vercammen and Lautenbach [23] and Dammerud et al. [15] used a similar approach omitting the direct sound from the measure using an interval 5-80 ms and 7-50 ms respectively. Both use a reference sound level at 10 m in the free field using the G and therefore are denoted G_{5-80} and G_{7-50} . Ueno and Tachibana [24] also note in their paper that,

in the case of measuring *EEL*, the direct sound should be omitted in order to evaluate the early reflections. Finally, Dammerud et al. [15] also proposed the Early Sound Strength G_e where a time interval of 0-80 ms is used without omitting the direct sound. It is stated that the G_e and G_{7-50} measured with source receiver distances larger than 1 m are highly influenced by the presence of the orchestra [25].

2.2.7 Other stage acoustic parameters

Other parameters have been used that are not based on energy ratio's like the Early Decay Time (*EDT*) which was originally introduced for describing auditorium acoustics [26] and the Modulation Transfer Function (*MTF*) as used in the Speech Transmission Index (*STI*) [27]. However, these parameters do not directly rely on time intervals and are therefore not further reviewed in this chapter.

2.3 Relevant time intervals on stage

In general, there are three different aspects that seem important to objectively describe the acoustic conditions on stage: the direct sound, the early reflections and the late reflections. In the previous section it was shown that variations have been made in the stage parameters by different researchers. It is clear that there is no agreement on the time interval limits in the stage acoustic parameter formulas. In this section, the influence of five different aspects on the choice of time interval will be investigated.

2.3.1 Perceptual parameters

The main goal for defining an objective stage acoustic parameter is that it will correlate with the subjective experience by the musicians. Various studies have been performed where musicians were tested under controlled conditions in a laboratory, aiming to find the important aspects of the room impulse response that relate to ensemble playing. The results of these studies have been summarized by Gade [7]. Researchers have focused on the temporal characteristics like the usefulness or annoyance of delayed reflections, the frequency properties like bandwidth and frequency balance (timbre) and the level properties like audibility and masking. The studies have been performed with single players or small ensembles playing together. Although it may not be suitable to directly apply or extrapolate the results from these studies to a full orchestra situation, it gives information on what is important on stage. Marshall et al. [28] suggested as a first design guide that preferred early reflection delays should lay between 17 and 35 ms relative to the arrival of direct sound, based on experiments with delayed reflections at 10, 20, 40 and 80 ms where a trio was asked to judge their ease of playing ensemble. Gade [3, 4] also concluded that early reflections are an important factor for ensemble playing, based on experiments with musicians in different sound fields with a mix of three reflections at 23, 43 and 83 ms relative to the time of emission. He found that when the direct sound from the other is masked (by reverberation or the sound of other players), it cannot be fully compensated by strong but further delayed early

reflections. Also, it was concluded that meaningful reflections for ensemble playing can arrive up to 100 ms and reflections up to 200 ms after the direct sound can provide support for soloists as long as they have the correct (moderate) sound level. Another interesting factor is that strong early reflections may also come too early between 5 and 20 ms and cause unfavourable coloration effects [29].

2.3.2 Architectural parameters

A second goal when defining an objective stage acoustic parameter is the applicability as a tool in optimizing stage environments through experiments with measurements or designing stages through simulations. To be able to do so, the parameter should be sensitive to changes in architectural parameters like stage dimensions, hall dimensions and surface properties. To investigate the relation of stage dimensions to the stage acoustic parameters time intervals, an inventory is made of maximum delayed first order reflections from stage surroundings. A list of stage dimensions is used from Dammerud [30]. In his study the average stage-dimensions of 22 purpose built concert halls are summarized. Table 2.1 shows the minimum, average and maximum dimensions of these 22 stages. The ceiling height has either been determined from the physical ceiling or from a canopy or overhead reflector above the stage.

Table 2.1. Stage dimensions (min, avg and max from 22 stages)

	min	average	max
Width [m]	15.5	22.0	35.0
Depth [m]	8.0	11.0	14.5
Height [m]	7.5	13.0	22.0

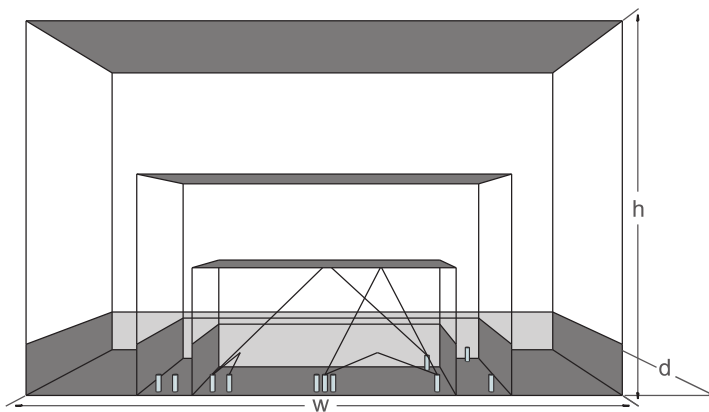


Figure 2.1. Three stage scenarios with minimum, average and maximum stage size. The source and receiver positions are represented by white columns of 1 m height.

To find the possible maximum delay in reflections based on the dimensions in Table 2.1 three different rectangular geometric scenarios have been studied:

1. minimum stage size $w \times d = 15.5 \times 8 \text{ m}^2$ + orchestra size $w \times d = 13.5 \times 6 \text{ m}^2$;
2. average stage size $w \times d = 22 \times 11 \text{ m}^2$ + orchestra size $w \times d = 20 \times 9 \text{ m}^2$;
3. maximum stage size $w \times d = 35 \times 14.5 \text{ m}^2$ + orchestra size $w \times d = 20 \times 9 \text{ m}^2$.

With the ‘maximum stage size’, it is assumed that the orchestra is not using the whole stage.

Furthermore five different source-receiver (S-R) positions are used to find the maximum possible delays:

1. front of stage on the side with S-R distance 1 m;
2. front of stage in the middle with S-R distance 1 m;
3. front of stage on the side to front of stage on the opposite side with S-R distance equal to orchestra width;
4. front of stage in the middle to front of stage on the side with S-R distance equal to half the orchestra width;
5. front of stage on the side to back of stage on the opposite side with S-R distance equal to $(\text{orchestra width}^2 \times \text{orchestra depth}^2)^{1/2}$.

The three scenarios and their source and receiver positions are illustrated in Figure 2.1 with some examples of ceiling and back wall reflection paths.

Table 2.2. Delay in ms relative to the emission of the sound for every possible sound path directly or 1st order reflected via sidewall, back wall or ceiling for every combination of stage size and source-receiver.

	s-r distance > 1m								
S/R pos	front side to side			front mid to side			front side to back side		
Stage size	min [ms]	avg [ms]	max [ms]	min [ms]	avg [ms]	max [ms]	min [ms]	avg [ms]	max [ms]
Direct	39	58	58	20	29	29	43	64	64
Sidewall	45	64	102	65	93	131	48	70	105
Backwall	57	83	98	45	65	84	46	66	78
Ceiling	55	91	136	43	76	126	57	95	138
All max	57	91	136	65	93	131	57	95	138

	s-r distance 1m						all cases max		
S/R pos	front side			front middle					
Stage size	min [ms]	avg [ms]	max [ms]	min [ms]	avg [ms]	max [ms]	min [ms]	avg [ms]	max [ms]
Direct	3	3	3	3	3	3	43	64	64
Sidewall	82	120	157	45	64	102	82	120	157
Backwall	41	58	79	41	58	79	57	83	98
Ceiling	38	70	122	38	70	122	57	95	138
All max	82	120	157	45	70	122	82	120	157

The maximum delay in ms relative to the emission of the sound has been determined for every possible direct sound path or 1st order reflected via sidewall, back wall or ceiling for every combination of stage size and source-receiver, see Table 2.2. It is shown that for an average stage size, the maximum delay of a first order reflection is 120 ms at 1 m S-R distance and 95 ms at larger S-R distances (O’Keefe [14] concluded that the boundary between the early discrete reflection zone and the late diffuse reflection zone ranges from just over 100 ms to just under 400 ms for 12 stages).

In all cases, the largest possible delay for a 1st order reflection is found for the shortest S-R distance and the maximum delay does not increase when the direct delay increases. This implies that in general the time interval between the arrival of the direct sound and the maximum 1st order reflection narrows when the S-R distance increases. For the average stage, the temporal window of possibly arriving 1st order reflections after the direct sound arrival is 31 ms at the largest S-R distance of 22 m diagonally crossing the stage.

2.3.3 Synchronicity

For the average stage, the delay in direct sound between different instruments rises up to 64 ms while the 1st order reflected sound arrives up to 120 ms, see Table 2.2. The effect of these delays have already been described as early as 1826 by Chladni [31]:

‘Der Raum, welchen ein Orchester einnimmt, darf [auch] nicht grösser seyn, als nöthig ist, weil sonst der (bey eines mässigen Temperatur der Luft etwa 1040 bis 1060 Fuss in einer Sekunde durchlaufende) Schall nicht schnell genug von einem Ende des Orchesters zum andern gelangen, und jeder Mitspieler die entfernten zu spät hören würde, so da als kein genaues Uebereintreffen im Takte Statt finden könnte, selbst, wenn man den Takt noch so laut schlagen, oder ihn wohl gar, wie bey den Alten üblich war, mit hölzernen oder eisernen Taktschuhen stampfen wollte.’

Experiments by Gade [3] confirmed that with direct sound delays larger than 20 ms, corresponding to a 7 m mutual distance, the ease of ensemble decreased based on studies with six different sound fields. Besides acoustical delays, the musicians in the orchestra itself can intentionally play ahead or delayed. It is often stated that brass and percussion players sitting at the back of the stage play ahead of the conductors’ baton [3]. This way, they aim to let the direct sound of the different instruments sections arrive simultaneously at the stage front line and consequently the listeners’ positions behind the conductor in the hall. On the average sized stage, this effect could impose an extra delay up to 26 ms for the direct sound from the front row players being audible in the back of the stage after their own instruments’ sound. Therefore, it may be relevant for reflections to arrive more early for sound travelling from the front to the back of the stage.

2.3.4 Source to receiver relationship

Traditionally, when describing room acoustic parameters in a concert hall the whole orchestra is represented by a single omnidirectional sound source which is placed on stage at a minimum of 3 consecutive positions [8]. The listener is represented by a single omnidirectional microphone which is placed in the audience area at a minimum of 6 to 10 consecutive positions. This implies that usually less source positions than receiver positions are used and the receiver positions may be assessed per source position (see Figure 2.2a).

In stage acoustic measurements the number of source and receiver positions can be equal, because one can use the same location as a source as well as a receiver position. The distance between the source and receiver can represent the distance between the own instrument and the musicians' ear, like the 1 m distance in the ST_{early} and ST_{late} parameters or the mutual distance between two different players like in the EEL parameter. In the latter case, one could assess the transfer of sound from all different sound source positions towards a single receiver, as illustrated in Figure 2.2b.

The support by reflections is important for every combination of two musicians at various distances. Especially at larger S-R distances musicians may only rely on reflected sound for mutual hearing. Therefore, the transfer of sound on stage should be considered at various source-receiver distances.

One might be tempted to use more receiver positions than source positions because one can easily use multiple microphones to record the impulse responses simultaneously, while the sound source needs to be operated one at a time. Nevertheless, on stage, the transfer of sound from every position to the other seems equally important and using equal source and receiver positions on stage seems obvious. Although, in theory, the transfer of sound between both omnidirectional source and receiver is reciprocal.

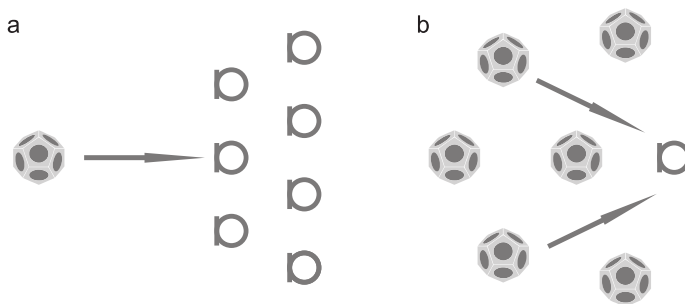


Figure 2.2. (a) single source towards multiple receivers (b) multiple sources towards single receiver.

Another aspect of the source to receiver relationship is the delay of the arrival of the direct sound after emission of the sound from the source. When for instance judging the clarity of the 'orchestra' at receiver positions in the audience area, the objective parameter C_{80} is used where the time interval of 0 to 80 ms and 80 ms to infinity is relative to the time of arrival of the direct sound. The C_{80} is determined for every source at a single receiver position without

taking into account the delay of the direct sound arrival between the different source positions. This may be correct because the orchestra itself corrects for this delay in order that the direct sound arrives simultaneously in the audience area behind the conductor (see section 2.3.3). However, when the receiver is positioned in the orchestra itself, it might be necessary to compensate for the delay in arrival in the direct sound when studying the sound energy within a certain temporal window.

The architectural parameter analysis showed that maximum arrival times of early reflections from stage surroundings are approx. 91 to 120 ms relative to the time of emission for an average stage size. The above mentioned source to receiver model based on synchronized sources is illustrated in Figure 2.3. It clearly shows that for larger S-R distances the direct sound is delayed whilst the time interval of possible arriving 1st order reflection after that direct sound narrows.

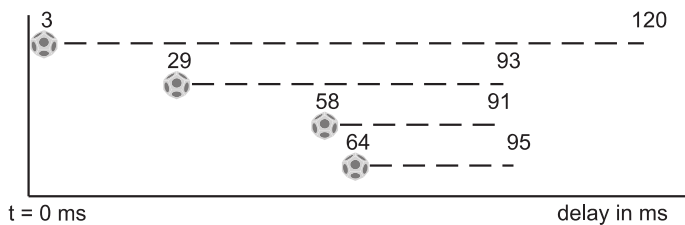


Figure 2.3. Arrival of direct sound (marked by source symbols) and reflections (dashed lines) from self and 3 others relative to the time of emissions at S-R 1 m distance position front side on average sized stage for synchronized sources.

2.3.5 Measurement system influence

In the stage parameter definitions the energy from the direct sound + floor reflection and the following reflections needs to be separated. Also it has been proposed to make the separation between the direct sound and the floor reflection, see section 2.2.6. However, here the limits of the measurement systems are reached and the physical disadvantage of low frequencies must be taken into account. Important factors are the signal response of the used sound source and the filter characteristics, which cause a smearing of the measured signal. This means that energy from adjacent reflections may overlap and the sound energy of each reflection cannot be isolated without the influence of these adjacent reflections. To avoid this, a certain period of time must be kept between the last and following reflection around the time interval limit. The minimum required distance between equally strong reflections was determined for a B&K 4292 omni directional sound source and Dirac room acoustic measurement software, where the RIR is first divided in the time domain and then filtered through band-pass filters as recommended by ISO 3382-1 [8]. It was found that a minimum distance between neighbouring reflections of 17, 12 and 9 ms is required to reduce the influence to ≤ 0.1 , ≤ 0.5 and ≤ 1.0 dB respectively (for the separate octave bands 250, 500, 1,000 and 2,000 Hz). This suggests, that the reference level of direct sound + floor reflection at 1 m S-R distance (delay 4 ms at 1 m height) in the 250-2,000 Hz octave bands should be

measured without reflections coming earlier than $4+17 = 21$ ms after the direct sound for ≤ 0.1 dB influence. Also, it seems to be impossible to separate the direct sound and floor reflection accurately using a single measurement. The sound level of a single reflection at $4+9 = 13$ ms can be measured with an influence ≤ 1 dB, and when multiple reflections are measured in a wider temporal window the influence will likely decrease. At larger S-R distances the direct sound and floor reflection will merge and the sound level of a single reflection at 9 ms can be measured with an influence ≤ 1 dB.

When measuring sound levels, it is also necessary to take into account the size of the wavelength. When measuring the sound level of a single reflection, EN 1793 [32] recommends to use a time window of at least one corresponding wavelength. For the 250 Hz octave band with lower edge frequency at 176 Hz this suggests that the time window should at least be 6 ms. However, to effectively capture reflections on stage the time window should be much more than 6 ms.

2.4 Summary and proposal for extended ST parameters and measurement conditions

Perceptual studies by several researchers have shown that early reflections arriving before approx. 100 ms after emission of the sound are likely to be relevant for playing ensemble on concert hall stages for various source-receiver distances. Reflections arriving after approx. 100 ms can be described as late reflections which are necessary to provide a certain amount of reverberance. A new architectural analysis of common stage dimensions reveals that with an average stage size most 1st order reflections will arrive within 100 ms after emission of the sound. Also, it shows that in general the time interval between the arrival of the direct sound and the maximum delayed 1st order reflection from the stage boundary narrows when the S-R distance increases. The source to receiver relationship study shows that the approach of the *EEL*, where the time interval is relative to the time of sound emission instead of the time of direct sound arrival, seems also valid in relation to the likelihood of arriving 1st order reflections from the stage boundary under the assumption that the emission of sound by the different sound sources is synchronised.

However, to be able to investigate the impact of early reflected sound energy at various distances, the direct sound should be omitted from the time interval (as used by the *EEL*). It was found that the direct sound must be omitted by using a time interval starting at 9 to 13 ms after the arrival of the direct sound to sufficiently reduce its influence to ≤ 1.0 dB. To be able to omit the direct sound, the source and receiver positions must be kept at least 2 m from any stage boundary, seating and stands. However, to be able to measure an accurate reference level at 1 m containing the direct sound and floor reflection only, no reflections may arrive before 21 ms (influence reflected sound ≤ 0.1 dB). This implies that for the reference measurement at 1 m distance, source and receiver positions must be kept at least 4 m from any stage boundary, seating and stands.

Based on these insights, it is proposed to extend the commonly used ST parameters by introducing a variable time point ‘103-delay’ that takes into account the delay of direct sound by increased distance, see equation 2.1 and 2.2, where the ‘delay’ is the S-R distance divided by the speed of sound. This way, the parameters can be measured at S-R distances up to 25 m, considering a time interval width of 30 ms as an acceptable minimum. The time interval of early reflected sound starts at 10 ms instead of 20 ms to be able to measure closer to the stage boundaries up to 2 m. Infinity is used instead of 1,000 ms as an upper time limit for the late reflections because it is conceptually clearer. At 1 m distance, the parameters are similar to the ISO 3382-1:2009 parameters.

$$ST_{early,d} = 10 \lg \left(\frac{\int_{10}^{103-\text{delay}} p_d^2(t) dt}{\int_0^{10} p_{1m}^2(t) dt} \right) \quad (2.1)$$

$$ST_{late,d} = 10 \lg \left(\frac{\int_{103-\text{delay}}^{\infty} p_d^2(t) dt}{\int_0^{10} p_{1m}^2(t) dt} \right) \quad (2.2)$$

Where,

- $ST_{early,d}$ = Early Support at distance d [dB]
- $ST_{late,d}$ = Late Support at distance d [dB]
- p_d = Sound pressure measured at distance d [Pa]
- p_{1m} = Sound pressure measured at 1 m distance in the free field [Pa]
- delay = S-R distance divided by the speed of sound [ms]

Furthermore, following ISO 3382-1, it is recommended to use 1.0 m transducer heights and to perform the measurements on a stage occupied with chairs and stands (see [chapter 4](#) for up to date insights on this topic). The extended ST parameters should be determined as an average over 5 measurements, while rotating a dodecahedron loudspeaker in steps of 72 degrees.⁴ A decay range INR of at least 45 dB is recommended for all measured RIR's on stage.

It is expected that $ST_{early,d}$ is a valuable parameter to investigate the contribution of early reflections to ensemble playing with increasing source-receiver distance. It is likely that

⁴ In case of judging separate octave band results only.

stages exists where ST_{early} measured at 1 m S-R distance is relatively high, suggesting good support from early reflections for the own instrument, while $ST_{early,d}$ at 10 m S-R distance is relatively low, suggesting poor support from early reflections for sound from the other players. In such cases, ST_{early} measured at 1 m S-R distance only may not be sufficient in describing stage support.

In contrary, it is expected that $ST_{late,d}$ is not dependent on the S-R distance. In a theoretical diffuse sound field, the sound level is not dependent on the S-R distance outside the critical distance. By the $ST_{late,d}$ the energy of the late part of the impulse response is measured using a time interval relative to the time of sound emission. This implies that, if reflections after 103 ms are considered as diffuse reflections, the (average) energy from these reflections may be equal for any S-R distance.

However, it must be investigated whether $ST_{early,d}$ is a reasonable indicator for the subjective impression of ensemble playing at various distances. Such an investigation should also reveal preferred values for $ST_{early,d}$. So far, the research presented in this chapter is limited to the metrological aspects of stage acoustic parameters and subjective evaluations are considered as future work. In the next section, the impact of using the variable time interval instead of using fixed time intervals as described in section 2.6 is investigated for measurements on various concert hall stages. Besides that, the advantage of the $ST_{early,d}$ and $ST_{late,d}$ is investigated through evaluation of the different concert hall stages.

2.5 The effect of the time interval choice in measured stage acoustic parameters

2.5.1 Method

The effect of the time interval choice in stage acoustic parameters on measured results has been studied for 8 different concert hall stages with various conditions. On all stages impulse response measurements have been performed using a comparable measurement method and equal source-receiver layout for optimal comparison. From these impulse responses the sound level L_{a-b} has been calculated using Dirac 5 for different time intervals $a-b$ at various distances d relative to the sound level of the direct sound + floor reflection at 1 m distance in the 0-10 ms interval corresponding to the ST parameters, see equation 2.3. All values have been arithmetically averaged over 250 to 2,000 Hz octave bands.

$$L_{a-b} = 10 \lg \left(\frac{\int_a^b p_d^2(t) dt}{\int_0^{10} p_{1m}^2(t) dt} \right) \quad (2.3)$$

Based on the findings as reported in the previous sections there are two time boundaries which connect the three important time intervals direct, early reflected and late reflected sound. The time points 5, 7, 10 and 20 ms have been used by [19, 23], [18, 15, 22], [3, 4] and [3, 4] respectively to describe the transition time point between the direct sound and the early reflections which will be denoted 'x'. The time points 40, 50, 80 and 100 ms have been used by [22], [15], [3, 4, 19] and [3, 4, 18] respectively to describe the transition time point between the early reflections and late reflections and will be denoted 'y'. Other relevant time points are 0 ms, which is defined as the arrival time of the direct sound. Finally, the variable time point denoted 'var' will be used which is defined as the time point of 103 ms minus the delay between the time of sound emission from the source and the time of direct sound arrival at the receiver as proposed in the extended ST parameters in section 2.4.

An overview of the concert hall stages that have been used is given in Table 2.3. During the 2009 tour by the Dutch Student Orchestra NSO, stage acoustic measurements have been performed in 7 Dutch halls (halls A to G). Additional measurements have been performed in one of these halls after refurbishment where the seats were replaced and a canopy was installed (hall E+). Also, measurements were performed during a refurbishment of hall C at the time that no seats were installed in the audience area (hall C-). Finally, measurements were performed in a concert hall, where a canopy was installed (hall H+) that could be lifted to the ceiling (hall H). The stages were empty during the measurements (except hall A where this was not possible). All seats and stands were removed to save time during the measurements and to be able to use the exact same source and receiver positions on all stages.

Table 2.3. Concert hall properties

hall	stage width (m)	stage depth (m)	stage height (m)	stage area (m ²)	Canopy/ Reflectors/ Shell	Chairs on stage	Seats in the hall**	RT _{unocc}	Hall Volume [m ³]
A*	7 - 20	4 - 12	15.5	150	No	Yes	2000	2.3	19,000
B	16.4	11.2	13	240	Shell+Refl	No	1450	2.0	11,500
C	18.0	11.5	21	210	Reflectors	No	1250, old	2.0	14,400
C-	18.0	11.5	21	210	Reflectors	No	No seats	4.0	14,400
D	20.4	13.7	17	260	Reflectors	No	1900	2.0	15,500
E	17.4	11.7	15	210	No	No	2200, old	2.4	27,700
E+	17.4	11.7	10.4	210	+Canopy	No	2200, new	2.9	27,700
F	21.6	17.5	8.5	390	Large shell	No	900	1.7	12,000
G	17.5	12.6	18	220	No	No	1050	2.3	16,500
H	22.0	10.5	16.0	231	No	No	1400	2.4	17,500
H+	22.0	10.5	9.1	231	+Canopy	No	1400	2.4	17,500

A measurement layout with fixed source and receiver positions was developed to match the positioning of instrument groups in a symphonic orchestra, illustrated in Figure 2.4. The source and receiver positions are put on a rectangular grid of 1.5 m by 1.8 m with a 2 m border around the outside positions to avoid reflections arriving before 10 ms. The positions represent the following instrument sections or persons: 1: timpani or percussion, 2: woodwinds left, 3: horns, 4: 1st violins, 5: woodwinds right, 6: brass, 7: celli, 8: 2nd violins, 9: viola, 10: conductor, 11: 1st violins back row or harp, 12: double basses. The size of the layout for halls A to G was 14.8 m deep and 11.5 m wide, so all positions 1 to 10 would fit. Positions 11 and 12 were later added to the layout for the measurements on stage H which is wider, however in this hall position 1 could not be used because of the limited stage depth.

Because of the symmetry of the layout and of most halls and stages, only 6 positions need to be both source and receiver positions and 6 positions are only receivers. Also, for source position 1 only half of the receivers have to be measured. Measurements are performed at 1 m distance for all source positions. In total, this results in $5 \times 12 + 1 \times 7 = 67$ S-R pairs. In the NSO project only source positions 1 to 4 were used to save time (36 S-R pairs). The 1 m distance measurement was performed with one microphone in front of the sound source and a second microphone and the right side (seen from the audience area). For optimal comparison of all stages A-H the positions 11 and 12 are not used in this research, resulting in a mutual S-R distance between 1 and 10.6 m and on average 5.3 m. It was found that the S-R distance was kept within 0.15 m accuracy.

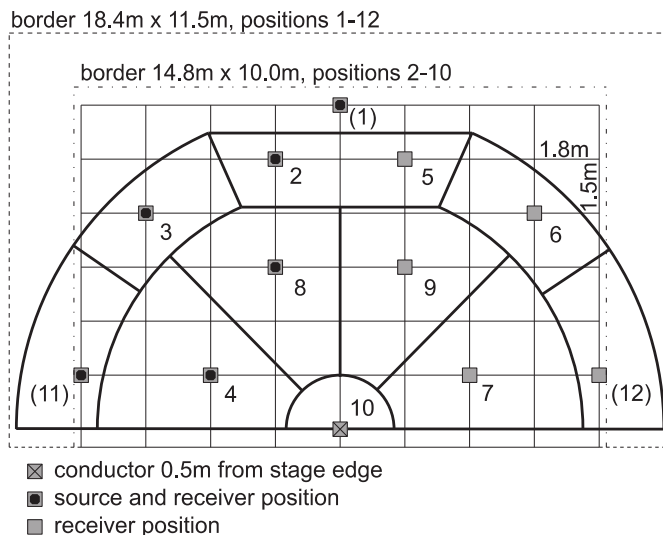


Figure 2.4. Measurement positions layout

For every S-R pair, 4 impulse responses have been measured using room acoustic measurement software Dirac and a 5 s exponential sweep signal while rotating the B&K type

4292 sound source in equal steps of 90 degrees. The rotation of the sound source is done to correct for directivity deviations, which is a problem when measuring close to a dodecahedron shaped sound source (recent research has shown that actually 5, 7 or 8 equal angular steps result in a considerable reduction of uncertainty [16]). Care was taken that the Impulse to Noise Ratio [20] of all measured impulse responses was > 45 dB for all frequency bands of interest. All transducers were put at a height of 1.3 m in accordance with ISO 3382-1:2006. Except for stage A, no chairs were on the stages that could have influenced the 1 m distance measurement. The different relative sound pressure level measurement at 1 m distance per microphone was taken into account.

2.5.2 Direct sound and early reflected sound at 1 m S-R distance

First, the influence of the direct sound and floor reflection is investigated on results of the various proposed early reflected sound time intervals measured at 1 m distance using equation 2.3. Figure 2.5 shows the average value for all possible variations L_{x-100} including L_{0-100} for every sound source position and receiver position at 1 m distance from that source position. Due to the possible influence of the chairs on stage, hall A is excluded in the average. At the source position S4 which is furthest away from any stage boundary the average difference between L_{10-100} and L_{20-100} is less than 0.3 dB. Similar results were found at S1, S2 and S3 for large stages E and F, but on average the difference increases up to 2.7 dB for S1, which is most often in close proximity to stage walls. This suggests that the time interval 10-20 ms is sensitive to very early arriving reflections and it confirms that these very early reflections can be measured without considerable influence from the direct sound and floor reflection as was concluded in section 2.3.5. The average difference between L_{7-100} and L_{10-100} is 1.0 dB for all source positions while the average difference between L_{5-100} and L_{10-100} is 4.9 dB. It is very likely that in the L_{5-100} and L_{7-100} (part of) the energy from the direct sound and/or floor reflection is included in the measurement at 1 m distance. As a result, the later reflections are partly masked by the direct sound. However, the results confirm that the time interval to capture the early reflected sound needs to start at 10 ms to be able to take into account reflections from close by stage boundaries (between 2 and 4 m) distance at all positions.

2.5.3 Direct sound and early reflected sound at all source-receiver distances

Secondly, the influence of the direct sound and floor reflection on results of the various proposed time intervals for early reflected sound measured at various distances is investigated. To be able to do so, trend lines of parameter values over distance for all possible variations L_{x-40} including L_{0-40} have been calculated for every hall. An example of the relation between the individual parameter values and the trend line are shown in Figure 2.6. The trend line is determined using a least squares fitting of a logarithmic relation $y = a \lg(x) + b$. For every trend line the correlation coefficient R^2 is calculated, which describes the fraction of variation in y that is explained by its relationship with x .

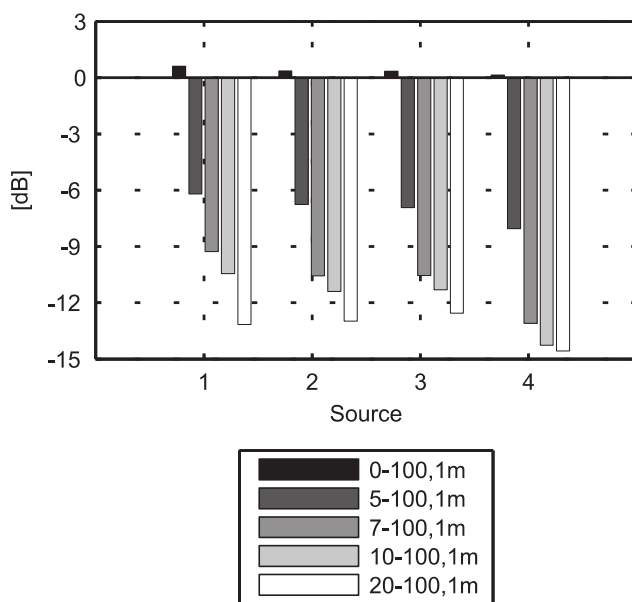


Figure 2.5. Halls B-H+ average value for all possible variations L_{x-100} and L_{0-100} per source position

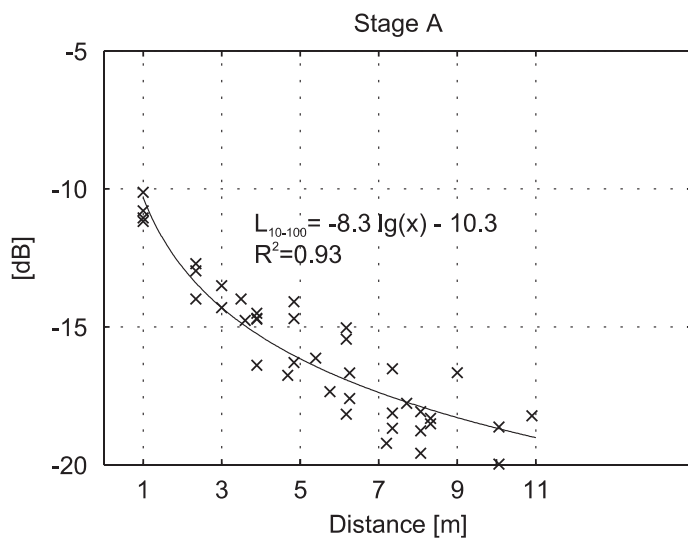
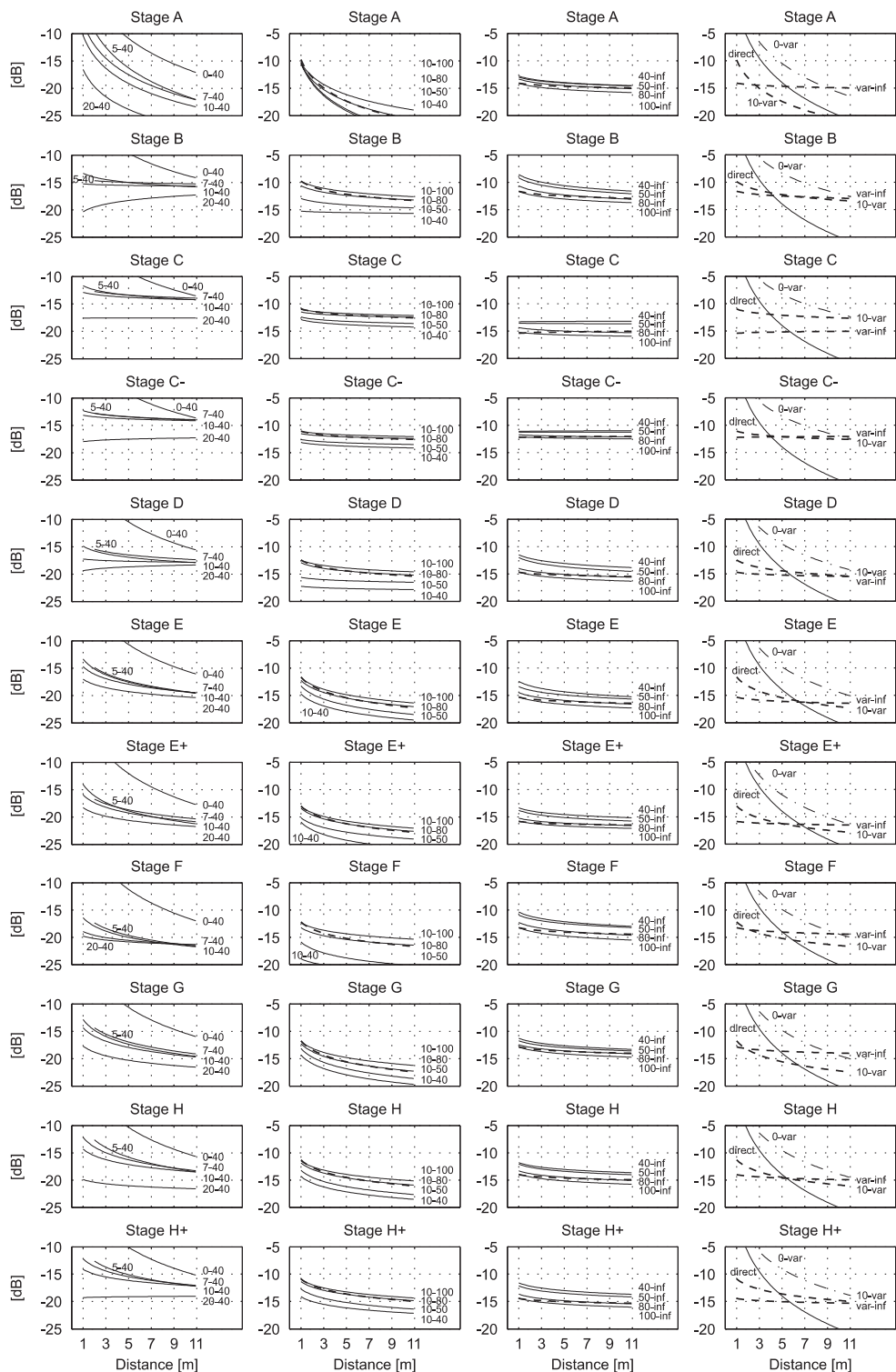


Figure 2.6. Example of the relation between the individual L_{10-100} parameter values and the trend line for stage A.

An upper time limit of 40 ms is chosen to make the parameters as sensitive as possible to changes in the lower time limit. The results of these trend lines are presented in the first column of graphs in Figure 2.7. The average correlation between the parameter value and the distance is very strong for L_{0-40} ($R^2 = 0.95$) in all halls, strong for L_{x-40} in hall A ($R^2 = 0.80$), moderate for L_{x-40} in halls D-H ($R^2 = 0.20$) and weak for L_{x-40} hall B, C and C- ($R^2 = 0.03$). The trend line for L_{5-40} is calculated without the 1 m distance measurements, because their results deviate too much from the trend after 1 m. The results for L_{0-40} ms for all halls are almost equal and it can be seen that per hall the parameters L_{5-40} , L_{7-40} and L_{10-40} are very similar and only differ slightly closer to the sound source. But in most cases L_{20-40} is considerably lower, except for the largest stage F. It appears that on stage F little sound energy arrives in the time interval 5-20 ms. This suggest that the increase of L_{5-40} and L_{7-40} close to the source may be caused by influence from the direct sound and floor reflection, which is also the case for some other stages. However, on all other stages it appears to be necessary to include at least the 10-20 ms time interval to measure all early reflected sound (although this may influence the parameter value less when using a higher upper limit than 40 ms). Also, the results confirm that it is necessary to omit the direct sound and floor reflection to be able to measure differences between different halls as was suggested by many other researchers, see section 2.2.6.

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Figure 2.7: variations in time intervals of L_{a-b} per distance for 11 stages; 1st column: variation in lower limit for $x-40$ ms; 2nd column: variation in upper limit for $10-y$ ms where dashed line has $y = 103$ -delay; 3rd column variation in lower limit for $y-inf$ ms where dashed line has $y = 103$ -delay; 4th column: comparison of $10-var$ and $var-inf$ to $0-var$ and theoretical direct sound decay.



2.5.4 Early reflected sound at all source-receiver distances

Based on the previous results the influence of the upper time limit for early reflected sound is further investigated using a 10 ms lower time limit. The trend lines of parameter values over distance for all possible variations L_{10-y} including L_{10-var} have been calculated for every hall, see the second column of graphs in Figure 2.7. The parameter L_{10-var} is drawn as a dashed line without a label. The average correlation between all parameter values and the distance is very strong for hall A ($R^2 = 0.86$) and very weak for hall C ($R^2 = 0.04$). For the other halls, the average correlation increases with the size of the temporal window: $R^2 = 0.15, 0.28, 0.39, 0.41$ for $L_{10-40}, L_{10-50}, L_{10-80}, L_{10-100}$ respectively. This implies that, although the trend lines for all parameters seem to be more or less parallel, the deviations of the individual measurements from the logarithmic trend line are smaller with increasing size of the temporal window. In the parameter L_{10-var} the temporal window size is made dependent on the S-R distance using the variable upper time limit '103-delay'. Although the L_{10-var} trend line appears almost equal to the L_{10-80} trend line, the correlation to a logarithmic fitted trend line increases from moderate for L_{10-80} ($R^2 = 0.39$) to strong for L_{10-var} ($R^2 = 0.49$).

In Figure 2.8a, the deviation of the average value per hall from the average value for all halls is presented for all possible variations L_{10-y} and L_{10-var} for all S-R combinations. Also, the average deviation of L_{10-100} at 1 m distance is included. It is shown that the difference in ranking between the halls is small for all parameters measured over various distances. However, the difference between L_{10-100} at 1 m distance per hall is smaller and only hall A and hall B are more than 1 dB different from the hall average value. It is shown that the time interval choice is not critical for ranking the stages when performing measurements with various S-R distances, except for hall F with a relatively large stage. A different ranking is found between halls with various S-R distances compared to 1 m distance only⁵.

2.5.5 Late reflected sound at all source-receiver distances

First, the impact of using 'time to infinity' instead of 1,000 ms is investigated at 1 m S-R distance. For all halls, the difference between $L_{100-1000}$ and $L_{100-inf}$ was calculated for the 250 to 2,000 Hz octave bands. It was found that for all halls with a reverberation time < 2.5 s and for all separate octave bands, the difference is < 0.02 dB with all files having an INR > 45 dB. Two exceptions are halls E+ and C- where the difference is < 0.05 dB and < 0.17 dB respectively. It can be concluded that the difference between using 1,000 ms and 'inf' is negligible when the noise in the RIR is sufficiently reduced (INR > 45 dB).

Using the same approach as in section 2.5.4 the influence of the lower time limit of the late reflected sound level is investigated. The trend lines of parameter values over distance for all possible variations L_{y-inf} including $L_{var-inf}$ have been calculated for every hall, see the third column of graphs in Figure 2.7. The parameter $L_{var-inf}$ is drawn as a dashed line without

⁵ Based on a later analyses on data from more stages/rooms it was found that Stage A was an outlier negatively influencing this conclusion. Results for most stages show a high correlation between $ST_{early,d}$ and ST_{early} , see annex section 2.9, which indicates that $ST_{early,d}$ does not necessarily lead to a better discrimination between stages.

a label. It is clear that the differences between the different parameter trend lines per stage are small and that the trend lines seem parallel. The $L_{var-inf}$ ($R^2 = 0.24$) shows weaker correlation with distance than the other parameters L_{y-inf} ($R^2 = 0.50$ for all L_{y-inf} parameters), except for stage C, where $L_{var-inf}$ shows a slight increase per distance and the other parameters L_{y-inf} are almost flat ($R^2 < 0.10$).

In Figure 2.8b, the deviation of the average value per hall from the average value for all halls are presented for all possible variations L_{y-inf} and $L_{var-inf}$ for all S-R combinations. Also, the average deviation of $L_{100-inf}$ at 1 m distance is included. It is shown that the difference in ranking between the halls is again small for all parameters measured over various distances as well as the 1 m. Also, it is shown that the time interval choice is not critical for ranking the stages and results from measurements at 1 m distance are similar to results with various S-R distances.

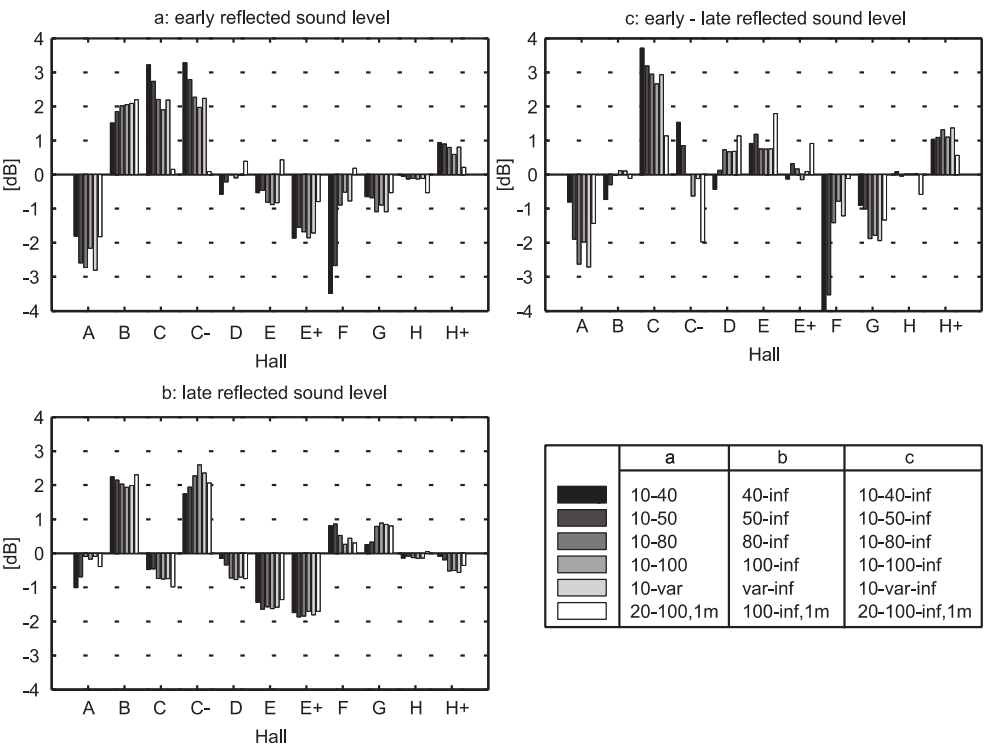


Figure 2.8. Average value deviation for (a) the early reflected sound level, (b) the late reflected sound level, (c) the difference between early and late reflected sound level. The reference value (0 dB) is the average value of all halls.

2.5.6 Balance between Early and Late reflected sound at all source-receiver distances

To study the balance between early and late reflected sound level, the difference between L_{10-y} and L_{y-inf} was calculated denoted $L_{10-y-inf}$. In Figure 2.8c, the deviation of the average value per hall from the average value for all halls are presented for all possible variations $L_{10-y-inf}$ and $L_{10-var-inf}$ for all S-R combinations. Also, the average of $L_{10-100-inf}$ at 1 m distance is included. It appears that the ranking between the halls more strongly depends on the choice of time limit 'y'. This is mainly caused by the influence of hall F on the results. When hall F is omitted from the data, the influence of the time interval is comparable to the early and late reflected sound level separately.

2.6 Evaluation of different stages

Although the time interval choice seems to be not very critical for comparing average stage values, the highest correlation to a logarithmic trend line was found for the early reflected sound level using L_{10-var} , which is equal to the $ST_{early,d}$ as defined in section 2.4. This implies that the use of a variable time interval is most suitable to describe the decay of early reflected sound over distance. On the contrary, in theory the late sound level was expected to vary little over distance as in a diffuse sound field. The lowest correlation to a logarithmic trend line for the late reflected sound level is found for $L_{var-inf}$, which is equal to the $ST_{late,d}$ as defined in section 2.4. This suggest that a variable time interval is also most suitable to describe the late reflected sound level. To further study the relation between early and late reflected sound level L_{10-var} and $L_{var-inf}$ are presented in the fourth column of graphs in Figure 2.7. Besides that, the early reflected sound level including direct sound and floor reflection is presented denoted L_{0-var} (which is similar to the original *EEL*) and the theoretical direct sound level decay based on a 6 dB decrease per distance doubling for a monopole.

It is shown that the L_{0-var} is comparable for all halls, especially closer to the sound source, while the L_{10-var} and $L_{var-inf}$ are clearly different per hall. On stage A, F and G the late reflected sound level is higher than the early reflected sound level at S-R distance > 2 m. For stage A and G this can be explained by the mainly absorptive stage surroundings existing of sloped seating areas. Stage F has fewer early arriving reflected sound energy as a result of the large dimensions compared to the other stages. On stages B, D, E, and H the early and late sound level is more or less equal, although on stage B both early and late sound is louder compared to stages D, E and H. The designs of stages D, E and H are similar, having a reflective wall surrounding the stage without any horizontal reflectors above the stage (the reflectors in hall D are tilted and reflect the sound from the stage towards the audience area). The overall increase in reflected sound level in hall B is probably caused by the smaller volume of the space compared to the halls D, E and H and the shell shaped stage. Stage C appears to be a clear outlier where the early reflected sound level is at least 3 dB higher than the late sound for all distances. This might be explained by the highly sound diffusing walls and downwards tilted reflective balcony-edge surrounding the stage combined with the two reflectors above

the stage. Together, these applications seem to provide a large amount of early reflected sound energy on stage. The difference between stage C and C- illustrates that the early reflected sound does not change, when the acoustics of the hall changes, while the late reflected sound increases by approx. 3 dB. In hall H, the addition of the canopy increases the early reflected sound level by approx. 1 dB while the late reflected sound decreases approx. 0.5 dB, so effectively increasing the early to late difference by 1.5 dB. These effects are not found in hall E with and without a canopy, which might be caused by this canopy being zigzag shaped and partly acoustically transparent.

Interviews with several professional musicians and conductors showed that there seems to be a general agreement that hall A and F have a very poor reputation in terms of stage acoustics while stages B and C have a very good reputation. Judgments on other halls' stages are less distinct. The measurement results are well in line with these judgments as it is generally accepted that stages with high early reflected sound energy are favoured over stages with low early reflected sound energy. However, in these particular cases it can be concluded that ST_{early} at 1 m distance does not fully predict this outcome while $ST_{early,d}$ assessed over various distances does. It is notable that even though ST_{late} is relatively high in hall B, possibly causing masking part of the direct and early reflected sound, this stage is still one of the musicians' favourites.

2.7 Discussion

This chapter has provided a detailed analysis of relevant metrological issues concerning stage acoustic measurements resulting in a proposal to optimise and extend the existing ST_{early} and ST_{late} parameters so they can be measured at various source-receiver distances, see section 2.4. While this study has provided several new insights, it is also important to mention the limitations of this study:

- The study did not investigate whether $ST_{early,d}$ is a reasonable indicator for the subjective impression of ensemble playing at various distances.
- The study did not establish preferred values for $ST_{early,d}$ at various distances.
- The study checked the theoretical assumptions based on measurements on empty stages without chairs and stands. Theoretical assumptions were only checked for concert hall stages and results may only count for stages close to the surveyed ones.

In spite of these shortcomings, the present study provides an optimal time interval to measure early and late reflected sound energy at various source-receiver distances on stage based on existing and new insights. To the best of our knowledge it is for the first time, that it has been demonstrated that the amount of early reflected sound energy on stage is distance dependent and correlates strongly to a logarithmic trend line using a variable time interval, while the amount of late reflected sound energy is not clearly dependent on the distance. Also, a variable time interval has not been reported in previous studies.

Future work should focus on:

- Exploring the influence of chairs, risers, stands, screens and persons on stage on the parameter results (see [chapter 4](#)).
- Exploring the influence of the actual instrument directivity [33] compared to the omnidirectional sound source directivity (see [chapter 5](#)).
- Exploring the impact of separate architectural applications like stage walls, diffusers and reflectors on the parameter results.
- Finally, it should be investigated whether $ST_{early,d}$ correlates with the subjective impression of ensemble playing and whether the balance between $ST_{early,d}$ and $ST_{late,d}$ may be a relevant descriptor of the early reflections masking by reverberation at all distances on actual stages.

2.8 Conclusions

Different time interval limits have been proposed for the stage acoustic parameters but there is no agreement on the preferable limits. Also, there is a growing interest to measure stage acoustic parameters at various source to receiver distances. In this research, a detailed analysis of relevant metrological issues concerning stage acoustic measurements has led to the optimised and extended $ST_{early,d}$ and $ST_{late,d}$ parameters that can be measured at various source-receiver distances using a variable time interval of ‘103-delay’. Theoretical assumptions were checked and confirmed based on systematic analyses of measured results for different concert hall stages with various conditions and various source to receiver distances. It can be concluded that different time interval limits did not result in a different ranking of the measured stages.

In the following annex section 2.9 and annex chapter 9, more measurement data of (other) performance spaces is presented, most of which were collected after publishing current chapter as a journal paper. The larger data set reveals that early reflected sound levels measured at 1 m and averaged over measurements at various distances are highly correlated. This explains why, against Gade’s expectation, ST_{early} was found to be a significant predictor for the ease of playing ensemble with other players, while it was intended as a measure to describe the hearing of the own instrument. This means that, for a general judgement of ‘ease of playing ensemble’ on stage, it is likely sufficient to measure the ST parameters at 1 m distance. For detailed analyses of stage acoustics, measurements at further distances are additionally valuable.

Unfortunately, the results presented in this chapter have been obtained on unoccupied stages without audience and without proper sound source calibration. In future, it would be recommended to investigate stages in both occupied and unoccupied conditions to check if effects or trends found still exist after people enter the space.

2.9 Annex: Summary of additional measurements

2.9.1 Measurements

The previous sections 2.1 to 2.8 have been published as a journal paper. In this section 2.9, the measurement results in concert halls are further analysed accompanied with those from other spaces such as orchestra pits, rehearsal rooms and outdoor theatres that have been collected after publishing the journal paper. The detailed description of the measurements of these spaces can be found in Chapter 9, Annex.

In the concert halls, T_{20} varies between 1.6 for a multipurpose hall with 12,000 m³ to 2.9 s for a rather large concert hall of 28,000 m³. In the rehearsal rooms T_{20} varied between 0.7 and 1.4 s with room volumes varying between 650 and 3,000 m³, where the largest room happened to have the shortest reverberation time.

An overview of the ranges found for averaged $ST_{early,d}$ and $ST_{late,d}$ are given in Table 2.4. The early and late reflected sound levels have different ranges depending on the room type.

Table 2.4: Range of distance averaged $ST_{early,d}$ and $ST_{late,d}$ found in various room types. CC is the covered part of the pit, OC/OO is at least one position in the open part of the pit, * average values without including the outliers.

	N	$ST_{early,d}$ [dB]	range	$ST_{late,d}$ [dB]	range
Concert Halls	8	-17 to -12	5 dB	-16 to -12	4 dB
Orchestra pits	5	-9 to -4 (CC)	5 dB	-21 to -13 (CC)	8 dB
		-13 to -8 (OC/OO)	5 dB	-21 to -14 (OC/OO)	7 dB
Rehearsal rooms	7	-14 to -6	8 dB	-21 to -10	11 dB
Open air theatres*	2	-22 to -13	11 dB	-28 to -18	10 dB
All indoors	20	-17 to -4	13 dB	-21 to -10	11 dB
All	22	-22 to -4	18 dB	-28 to -10	18 dB

The smallest ranges are found in the 8 concert hall stages, which only vary in average $ST_{early,d}$ and $ST_{late,d}$ by 5 and 4 dB respectively. If the parameter values indeed correlate with the judgement of musicians on concert hall stages and assuming that there is a preference among the stages measured, than the 5 dB difference among stages (which is 2.5 times the estimated JND of 2 dB) must already cause a substantial difference in acoustic conditions. However, for $ST_{early,d}$ and $ST_{late,d}$ this correlation is yet to be proven and it has not yet been confirmed either for the ‘original’ ST_{early} and ST_{late} parameters [15].

It is interesting to evaluate how valuable $ST_{early,d}$ and $ST_{late,d}$ measurements at various distances are compared to the 1 m measurements as used in the ‘original’ ST_{early} and ST_{late} parameters. Figure 2.9 compares $ST_{early,d}$ values measured at 1 m only and all $ST_{early,d}$ values averaged over all distances. It is clear that, for both early and late support, a strong correlation exists between measurements at 1 m and those averaged over all distances (with the exception of Stage A with the semi-circular shape). The data shows a difference in $ST_{early,d}$ of 1.8 dB/+/-0.8 dB between the averages at 1 m and averages over all distances. $ST_{late,d}$ is almost identical at 1 m and at all distances with a difference of only 0.3/+/-0.4 dB.

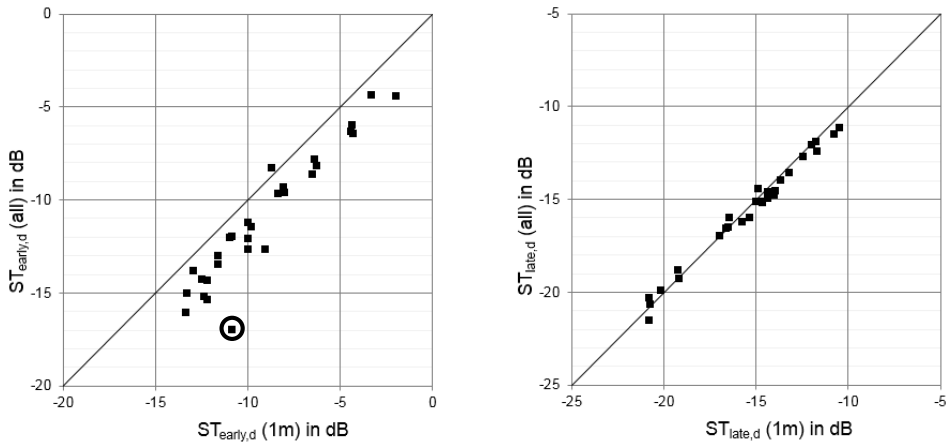


Figure 2.9: Comparison for ST parameters at 1 m versus all distances, in concert halls, orchestra pits and rehearsal rooms. Left: Comparison of $ST_{early,d}$ measured at 1 m distance and $ST_{early,d}$ averaged over all distances; difference 1.8 dB/+/-0.8 dB, excluding the outlier marked with a circle (Stage A). Right: Comparison of $ST_{late,d}$ measured at 1 m distance and $ST_{late,d}$ averaged over all distances; difference 0.3 dB/+/-0.4 dB.

For comparing averages of (empty) stages, it seems that it might be sufficient to only measure at 1 m distance as is common by measuring the original ST_{early} and ST_{late} . Measurements taken at various distances have shown to be additionally valuable for detecting outliers and trends. $ST_{early,d}$ has shown to follow a logarithmic trend line that decays over distance (see figure 2.6). Variation along this trend line occurs due to the complex reflection patterns that are position dependent. This variation should therefore not be considered as measurement error or uncertainty. In some cases, outliers from the ‘cloud’ have been detected that reveal effects, such as the lower support at the edge of the stage (for instance concert hall stage E, see section 9.1), the different trends in the open and covered part of the pit (section 9.2), and the focusing effects in the circular shaped open air theatres (section 9.4). These effects would have gone unnoticed with measurements at 1 m only and confirm the need of measurements at various distances. Also, such effects cannot be predicted by trend lines.

2.9.2 Predictors

From the rehearsal rooms, it was concluded that ‘stage volume’ can be a predictor for the amount of early reflected sound (measured by $ST_{early,d}$), see section 9.3.5. In the left panel of Figure 2.10, the relation between ‘stage volume’ and $ST_{early,d}$ is also shown for the concert halls (see section 9.1) and the orchestra pits (see section 9.2) and data measured by Gade [5]. For the orchestra pits, a similar trend is found and $ST_{early,d}$ decreases with increasing volume of the pit. For the concert halls, the relation between the ‘stage volume’ and $ST_{early,d}$ exists but with a large variation, likely because the largest part of the ‘stage volume’ is open to the hall and other properties than stage volume also determine its value.

For the rehearsal rooms, Barron’s revised theory was used to predict $ST_{late,d}$. The right panel of Figure 2.10 shows the comparison of $ST_{late,d}$ measured and predicted for the rehearsal rooms, the concert halls and data by Gade [5]. A 1.2 dB offset was applied to partly correct for the error in calibration of the sound source in-situ instead of in the laboratory (see chapter 3). It seems that the average absolute error between measured and predicted values is larger for concert halls stages (3.1 dB) than for rehearsal rooms (1.4 dB). $ST_{late,d}$ appears to be systematically underestimated by theory in case of the concert halls. Repeated measurements on Stage C, using a measurement equipment calibrated in a reverberation room instead of in-situ, showed an error of only 0.2 dB (see section 4.3.4) instead of 2.8 dB presented here. This indicates that in the course of this research project new knowledge might have led to more accurate measurements. Therefore, Barron’s revised theory might be more accurate as a predictor for $ST_{late,d}$ than shown in Figure 2.10.

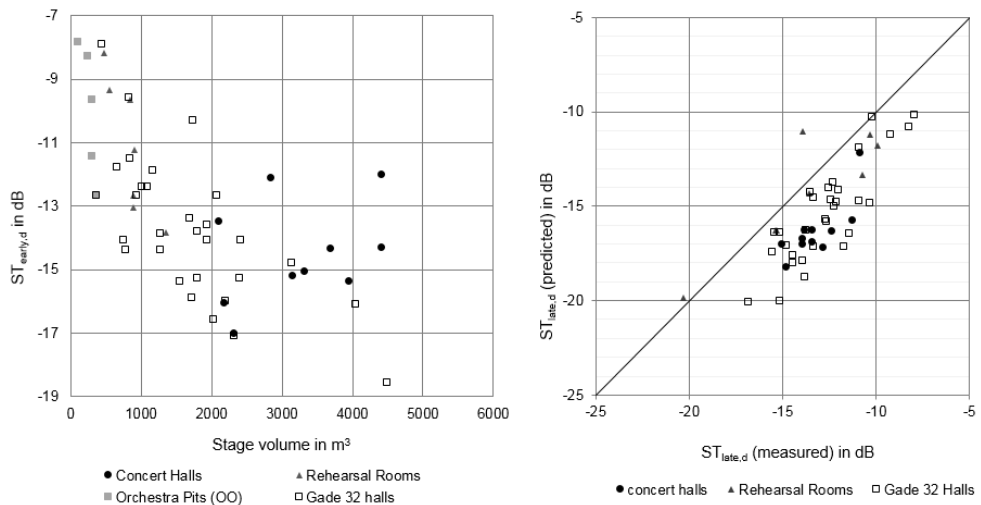


Figure 2.10: Predictors for ST parameters. Left: $ST_{early,d}$ as a function of ‘stage volume’. Right: $ST_{late,d}$ measured (+1.2 dB to correct for the systematic calibration error) versus Barron’s revised theory; average error 3.1 dB for concert halls and 1.4 dB for rehearsal rooms. Gade 32 halls from [5], ST1- 1.8 dB is used instead of $ST_{early,d}$ and $-CS+1.2$ is used instead of ST_{late} (CS=Clarity Stage).

2.10 References

- [1] A.C. Gade: Musicians' ideas about room acoustic qualities. The Acoustics Laboratory, Tech. Univ. of Denmark, Report No.31, 1981.
- [2] A.C. Gade: Acoustics of the orchestra platform from the musicians' point of view. Proceedings of Acoustics for Choir and Orchestra, Royal Swedish Academy of Music No. 52, 1986.
- [3] A.C. Gade: Investigations of musicians' room acoustic conditions in concert halls. I. Methods and laboratory experiments. *Acustica* 69 (1989) 193-203.
- [4] A.C. Gade: Investigations of musicians' room acoustic conditions in concert halls. II. Field experiments and synthesis of results. *Acustica* 69 (1989) 249-261.
- [5] A.C. Gade: Acoustical survey of eleven European concert halls - a basis for discussion of halls in Denmark. The Acoustics Laboratory, Tech. Univ. of Denmark, Report No.44, 1989.
- [6] A.C. Gade: Practical aspects of room acoustic measurements on orchestra platforms. Proceedings of 14th ICA, Beijing, 1992.
- [7] A.C. Gade: Acoustics for symphony orchestras; status after three decades of experimental research. *Building Acoustics*, volume 18, number 3, 4 (2011), 181-206.
- [8] ISO 3382-1-2009: Acoustics -- Measurement of room acoustic parameters -- Part 1: Performance spaces, International Organisation for Standardisation (ISO), Geneva (CH), 2009.
- [9] A.C. Gade: obtained from personal communication with A.C. Gade in 2012.
- [10] J.Y. Jeon, M. Barron: Evaluation of stage acoustics in Seoul Arts Center Concert Hall by measuring stage support. *J. Acoust. Soc. Am.* 117 (2005), 232-239.
- [11] S. Ternström, D. Cabrera, P. Davis: Self-to-other ratios measured in an opera chorus in performance. *J. Acoust. Soc. Am.* 118(2005), 3903-3911.
- [12] W. Reichardt, O. Abdel Alim, W. Schmidt: Definition und Messgrundlage eines objektiven Maßes zur Ermittlung der Grenze zwischen brauchbarer und unbrauchbarer Durchsichtigkeit bei Musikdarbietung. *Acustica* 32 (1975) 126-137.
- [13] J. O'Keefe: On the sensitivity of stage acoustics measurements. *J. Acoust. Soc. Am.*, Sabine Centennial Proceedings pp 223-226, 1994.
- [14] J. O'Keefe: Reflection density and attenuation on stages and rehearsal halls. Proceedings of ISRA, Melbourne, 2010.
- [15] J.J. Dammerud, M. Barron, E. Kahle: Objective assessment of acoustic conditions for symphony orchestras. *Building Acoustics* 18 (2011), 207-219.
- [16] C.C.J.M. Hak, R.H.C. Wenmaekers, J.P.M. Hak, L.C.J. van Luxemburg: The source directivity of a dodecahedron sound source determined by stepwise rotation. Proceedings of Forum Acusticum Aalborg, 2011.
- [17] C.C.J.M. Hak, R.H.C. Wenmaekers, J.P.M. Hak, L.C.J. van Luxemburg, A.C. Gade: Sound strength calibration methods. International Congress on Acoustics, Proceedings of ICA Sydney, 2010.
- [18] W.Chiang, S. Chen, C. Huang: Subjective assessment of stage acoustics for solo and chamber music performances. *Acta. Acust. united Ac.* 89 (2003) 848-856.
- [19] W.Chiang, Y.Shu: Acoustical design of stages with large plane surfaces in rectangular recital halls. *Appl. Acoust.* 64, No. 9 (2003) 863-884.
- [20] C.C.J.M. Hak, J.P.M. Hak, R.H.C. Wenmaekers: INR as an estimator for the decay range of room acoustic impulse responses. Proceedings of the 124th AES conference, Amsterdam, 2008.

- [21] C.C.J.M. Hak, H. Vertegaal: What exactly is time infinity for acoustical parameters? Proceedings of the 16th ICSV, 2009.
- [22] E.W.M. van den Braak, L.C.J. van Luxemburg: New (stage) parameter for conductors acoustics. Proceedings of Acoustics08, Paris 2008
- [23] M. Vercammen, M. Lautenbach: The renovation of De Doelen Concert Hall. Proceedings of the NAG-DAGA, Rotterdam, 2009.
- [24] K. Ueno, H. Tachibana: A Consideration on Acoustic Properties on Concert-Hall Stages. Building Acoustics, volume 18, number 3, 4 (2011), 221-236.
- [25] J.J. Dammerud, M. Barron: Attenuation of direct sound and the contributions of early reflections within symphony orchestras. J. Acoust. Soc. Am. 128 (2010) 1755-1765.
- [26] V.L. Jordan: Room acoustics and architectural acoustics development in recent years. Appl. Acoust. 2 (1969) 59-81.
- [27] G.M. Naylor: Modulation transfer and ensemble music performance. Acustica 65 (1988) 127-137.
- [28] A. H. Marshall, D. Gottlob, H. Alrutz: Acoustical conditions preferred for ensemble. J. Acoust. Soc. Am. 64 (1978) 1437-1442.
- [29] T. Halmrast: Orchestral timbre: Comb-filter coloration from reflections. J. Sound Vib. 232 (2000) 53-69.
- [30] J.J. Dammerud: Stage acoustics for orchestras in concert halls. PhD Thesis, University of Bath, 2009.
- [31] E.F.F. Chladni: Ueber vortheilhafte Einrichtung eines Locals für gute Wirkung des Schalles, Allgemein Musikalische Zeitung, No 35, 1826.
- [32] EN 1793-1997: Identifier Road traffic noise reducing devices. Test method for determining the acoustic performance. Intrinsic characteristics of sound absorption. Brussels, 1997.
- [33] J. Pätynen, T. Lokki: Directivities of symphony orchestra instruments. Acta. Acust. united Ac. 96 (2010) 138-167.

Chapter 3

The sound power as a reference

Original title: The sound power as a reference for Sound Strength (G), Speech Level (L) and Support (ST): uncertainty of laboratory and in-situ calibration
Authors: R.H.C. Wenmaekers, C.C.J.M. Hak
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Abstract

Some room acoustic parameters require the sound power of the sound source to be known. The Sound Strength G uses the free field sound pressure level at 10 m distance as a reference value. Speech intelligibility parameters like the A-weighted Speech Level, $L_{p,A,S,4m}$, and the Speech Transmission Index, STI , can require an absolute source level defined at 1 m distance from the sound source. The Early and Late Support parameters use the sound level at 1 m distance as a reference level. In this chapter, all proposed methods to obtain the sound power level for room acoustic applications are investigated, using various omnidirectional sound sources with a dodecahedron shape containing 12 loudspeakers. It is shown that, for octave bands 250 to 8,000 Hz, the sound power can be determined with 0.8 dB uncertainty when using precision methods (diffuse field, intensity or free field). Alternative laboratory calibration methods that only measure in a single plane of the sound source, show deviations up to 2 dB per octave band. Eleven different stepwise rotational averages, used in such a single plane free field method, have been investigated. It can be concluded, that the uncertainty is reduced only when using 12.5 degree steps (ISO 3382-1) and when using equal-angular rotations with 5 or 7 steps. Furthermore, the uncertainty of in-situ calibration has been investigated. A comparison of results from different researchers shows that a correction factor should be applied to correct the in-situ calibration for its deviation from the laboratory calibration. For each calibration method the uncertainty is presented. Results show that some methods might be sufficiently accurate to be able to measure single number ratings for G , ST , $L_{p,A,S,4m}$ and STI with an uncertainty in the order of magnitude of 1 JND, provided that no other measurement errors are introduced in the measurement chain.

3.1 Introduction

Various room acoustic parameters have been introduced, some of which have been included in the ISO standard 3382 on the measurement of room acoustic parameters [1]. Many room acoustic parameters are defined in such a way, that the sound power of the sound source is not relevant. Energy decay related parameters (EDT , T_{20} and T_{30}), energy related parameters (C_{80} , D_{50} , T_s), lateral energy measures (J_{LF} , J_{LFC}), spatial impression parameters ($IACC$) and the spatial decay parameter ($D_{2,s}$) all make use of a relative definition.

However, some room acoustic parameters do require knowledge of the sound power of the sound source. Level parameters (G , L_J , and $L_{p,A,S,4m}$) and stage support parameters (ST_{early} and ST_{late}) measure the amount of energy of a room impulse response within a certain time interval and/or direction, and compare it to the energy of the impulse response from the same sound source measured in the free field at a certain distance. Also, when determining speech intelligibility parameters, like STI , the signal to noise ratio is determined based on a ratio of absolute sound pressures (following ISO 16268-16 [2], only if background noise is taken into account, and following ISO 3382-3 [3], for all measurements in open plan offices). Following ISO 3382, all of these room acoustic parameters should be determined using an omnidirectional sound source. A certain deviation from omni directionality is allowed and the deviation limits are given in the standard.

Lundeby et al. [4] asked 8 different teams to determine room acoustic parameters for 10 source-receiver combinations in a 1,800 m³ auditorium. He found a standard deviation for the level parameter Sound Strength G of only 0.2 to 0.3 dB for the octave bands 125, 1,000 and 4,000 Hz. These results would suggest that G can be measured with very high accuracy, even when using a variety of loudspeaker types (a single loudspeaker in a box, a cube with six loudspeakers and a dodecahedron with twelve loudspeakers). Unfortunately, in their paper it is not explained how the sound sources were calibrated. In theory, the relation between the sound power of a point source and its sound pressure level (SPL) in the free field is clear. However, in earlier research it was shown that different methods to obtain the free field SPL may yield different results with deviations more than 2 dB for individual octave bands [5]. The low standard deviations found by Lundeby et al. [4] between 8 different measurement teams are therefore highly questionable.

The deviations in sound power determination are problematic as one would like to measure most room acoustic parameters within the limits of their Just Noticeable Difference (JND), which can be as low as 1 dB for some level parameters. It is necessary to find out when measurement methods are sufficiently accurate and which methods should be avoided. In this chapter, the difference between various calibration methods, and the impact of simplifications within these methods, are investigated. This is done by using available data from literature and, to close some knowledge gaps or to check whether the findings from literature can be reproduced, by performing additional measurements or analysing shared data from other researchers. In some cases, results are only available from one single research(er) using one single loudspeaker type, which is often dodecahedrally shaped. An

extensive amount of measurements necessary to significantly identify all uncertainty contributions in each measurement method was not executed, but the main conclusions are based on results from multiple investigations involving various researchers and sound sources. The overview presented in this chapter provides valuable insight in the merits of sound source calibration for room acoustic purposes and some major problems have been identified.

In this chapter, the background on the most relevant room acoustic parameters and common calibration methods is discussed in section 3.2. In section 3.3, the differences of various laboratory measurement methods, concerning the sound power as a reference for room acoustic parameters, are discussed. Findings from literature and new research have been combined to investigate uncertainties. In section 3.4, problems with the directivity of the sound source are discussed, which is particularly important for the calibration methods that only measure in the horizontal plane of the sound source while taking rotational averages. In section 3.5, in-situ measurement methods are discussed, including (floor) interference effects and variations over different positions. In section 3.6, the chapter will conclude with a summary and conclusions on the uncertainty that can be expected from using the available methods.

3.2 Background

3.2.1 Room acoustic parameters and sound power

The sound strength G is used to investigate the sound distribution in a concert hall or to compare the sound levels between different concert halls. Originally, Lehmann introduced the *Stärkemass* or ‘Strength Index’, which was defined as the difference in *SPL* in the hall caused by an omnidirectional sound source on the stage, and the sound power level L_w of the same sound source [6]. Later on, G was defined as the *SPL* at a listener position in the hall, with reference to the *SPL* at 10 m distance from the same sound source in a free field. The late lateral sound energy level L_J is similar to G in its definition of the reference level. Here, the late sound energy arriving after 80 ms at a figure-of-eight microphone is compared to the total sound energy at 10 m distance at an omnidirectional microphone in the free field. For G the 10 m distance was chosen by Barron and Lee for reasons of convenience because this choice will often result in measured values of the sound level L_p close to the reference value $L_{p,10m}$ and hence values of G close to 0 dB [7]. G can either be determined using stationary noise or using impulse responses [1], where the sound pressure exposure level L_{pE} is determined (up to the time point that the energy decay, or Impulse to Noise Ratio (*INR*) [8], is 30 dB or lower). The single number rating for G is calculated from the average of the 500 and 1,000 Hz octave band, defined by ISO 3382-1 as G_M , and the Just Noticeable Difference (*JND*) is 1 dB. For L_J the single number rating is calculated from the average of the 125 to 1,000 Hz octave band, denoted by $L_{J,avg}$. The *JND* for L_J is unknown.

$L_{p,A,S,4m}$ is the A-weighted *SPL* of normal speech at 4 m distance from an omnidirectional sound source.⁶ $L_{p,A,S,4m}$ is used together with $D_{2,S}$, the decay of sound per doubling of the distance, to describe to decay of speech sound in an open plan office. A standardised spectral level $L_{p,S,1m}(f)$ defined at 1 m distance is used to describe the sound power of the normal speaker [3]. After A-weighting the speech spectrum, the sound level in dB(A) is predominantly dependent on the level in the 500 and 1,000 Hz octave bands. The distance of 4 m is chosen as a nominal distance where the far field starts [9]. In a similar way, the absolute *SPL* of speech is used in the calculation of the Speech Transmission Index (*STI*), where the *SPL* of the received speech $L_{p,S}(f)$ and the *SPL* due to background noise $L_{eq,N}(f)$ is used to determine the signal to noise ratio (*SNR*). In both $L_{p,A,S,4m}$ and *STI* parameters, the sound power of the sound source must be known to be able to apply the standardised spectrum and level. The JND for the *STI* is 0.03 [10]. The impact of reverberation on *STI* is small in open plan offices, as shown by Wenmaekers et al. [11]. In a worst case scenario, where reverberation does not affect the *STI* calculation and the influence of background noise is fully dominant, it is possible to translate the JND of the *STI* as a JND of the *SNR* in dB. In this way, the maximum possible error in Signal level (or speech level $L_{p,S}$) due to calibration uncertainty can be investigated. In this chapter, the individual errors $\Delta L_{p,S}(f)$ in dB over the octave bands 125 to 8,000 Hz are translated into a modulation transfer index *MTI*(*f*) by $(\Delta L_{p,S}(f) + 15)/30$. The *MTI*(*f*) is weighted in accordance with IEC 60268-16 and the *STI* is calculated. Then, to arrive at a weighted single-number rating error, $\Delta L_{p,S,single}$ is calculated as $(STI \times 30) - 15$ dB. A difference of 0.03 in *STI* appears at approximately 1 dB difference in $\Delta L_{p,S,single}$.

The stage acoustic parameters Early Support, ST_{early} , and Late Support, ST_{late} , are used to investigate the ensemble conditions and perceived reverberance by musicians on concert hall stages. Both parameters are commonly determined at 1 m distance from an omnidirectional sound source and measure the early or late reflected sound energy relative to the direct sound energy of the sound source [12, 13]. The extended parameters $ST_{early,d}$ and $ST_{late,d}$ have been introduced that can be used to study the transfer of reflected sound energy over the stage at various distances [14 or [chapter 2](#)]. The 1 m distance reference is chosen as it is comparable to the distance of the musical instrument to the players' ears [12] (even though other researchers suggested that this distance is smaller in many cases [15, 16]). The direct *SPL* at 1 m distance was intended to be the free field sound level, as explained by Gade [17]. The method described in ISO 3382 uses a reference level measured in-situ. The single number rating for the ST parameters is the average of the four octave bands 250 to 2,000 Hz and the estimated JND for the ST parameters is 2 dB [18].

An overview of the parameters mentioned in this section is presented in Table 3.1. The different frequency ranges are used in the different parameters because they relate to different perceptual aspects. In this chapter, the frequency ranges as suggested by ISO 3382 are used.

⁶ Although less relevant in the context of current thesis, speech parameters were also investigated.

Table 3.1. Overview of room acoustic quantities that depend on a reference level.

Acoustic Quantity	Parameter	Reference	JND ⁷	Single number frequency averaging
Sound Strength	$\bar{G}$	$L_{p,10m}$ or $L_{pE,10m}$	1 dB	500-1,000 Hz
Late Lateral Sound Energy Level	L_J	$L_{p,10m}$ or $L_{pE,10m}$	not known	125-1,000 Hz
A-weighted speech level at 4 m	$L_{p,A,S,4m}$	L_w or $L_{p,1m}$	not known	A-weighted level*
Speech Transmission Index	STI	L_w or $L_{p,1m}$	0.03 STI (=1 dB in $\Delta L_{p,S,single}$)	Weighted over 125 - 8,000 Hz
Early Support	ST_{early} or $ST_{early,d}$	$L_{p,1m}$ or $L_{pE,1m}$	2 dB **	250-2,000 Hz
Late Support	ST_{late} or $ST_{late,d}$	$L_{p,1m}$ or $L_{pE,1m}$	2 dB **	250-2,000 Hz

* because the 500-1,000 Hz bands are dominant after A-weighting the speech spectrum, the 500 and 1,000 Hz bands average level will be used as indicator for $L_{p,A,S,4m}$.

** estimated by A.C. Gade.

3.2.2 Calibration methods in room acoustics

A large number of methods for the determination of sound power levels of sound sources is described in the ISO 3740 series [19], ranging from reverberation room methods to intensity or pressure level measurements in a (hemi) free field. Depending on the desired accuracy, different methods can be applied. For the determination of the sound power level of an omnidirectional sound source used for room acoustic measurements, ISO 3382-1 suggests two different calibration methods. ISO 3382-3 refers to this same part of the standard and suggests determining the sound power with at least ‘engineering accuracy’.

Following the first method mentioned in ISO 3382-1, one directly measures the *SPL* at 10 m distance to the sound source in an anechoic room or if only a smaller anechoic room is available, one measures at a distance of at least 3 m (to avoid being in the near field of the loudspeaker) and translates the measured *SPL* to a 10 m distance. In accordance with ISO 3382-1, the energy mean sound level needs to be determined from 29 measurements at every 12.5° step rotation of the sound source. It is remarkable that the sound power of the omnidirectional sound source, which most often has a dodecahedral shape containing 12 loudspeakers, is determined from measurements in one single horizontal plane only, see Figure 3.1. In contrast, the methods in the ISO 3740 series suggest measuring over an equally spaced grid around the sound source. In ISO 3382-1, no scientific study is quoted that motivates the 12.5° rotation method, nor is its accuracy discussed. Among others, this method has been applied by Barron and Lee [7], Aretz and Orłowski [20] and Dammerud [21]. In this chapter, this method will be denoted the ‘single plane free field method’.

⁷ It should be noted that the JND’s mentioned are based on experiments with broadband signals. In this thesis, the JND of all separate octave band values is assumed to be equal to the ‘broadband JND’, which is pessimistic.

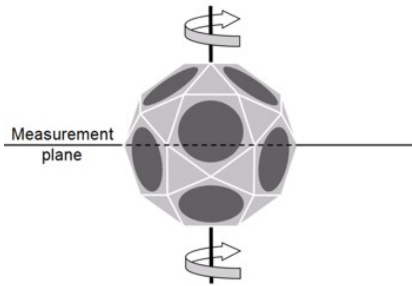


Figure 3.1. Side view of vertical rotation axis and horizontal measurement plane of a dodecahedron loudspeaker, used in the ‘single plane free field method’ for calibrating omnidirectional sound sources.

The second method described in ISO 3382-1 uses the precision method for reverberation rooms in accordance with ISO 3741 [22]. When using stationary noise, the sound power is determined following the full standard’s procedure. When using impulse responses, a system calibration can be performed in the reverberation room without measuring the actual sound power. During a system calibration, the sound exposure level L_{pE} is measured in the reverberation room, then corrected for the amount of absorption in the room and used as a reference for any future sound exposure level measurements (comparing relative levels). According to ISO 3741:1999, in octave bands, the uncertainty in terms of the standard deviation of reproducibility is equal to or less than 2.5 dB for 125 Hz, 1.5 dB for 250 Hz, 1 dB from 500 Hz to 4,000 Hz and 2 dB for 8,000 Hz (in the 2010 edition of ISO 3741 the octave band values are no longer mentioned). Among others, this method has been applied by Gade [23], Virjonen et al. [9] and Beranek [24].

Essentially, the Support parameters ST_{early} and ST_{late} use a third method to determine the sound power level of the sound source, described in ISO 3382-1. A reference level is determined in-situ from the impulse response measured at 1 m distance using a 0-10 ms window. Within this 0-10 ms time interval, the floor reflection is included which introduces an error in the determination of the direct sound. Also, errors due to loudspeaker directivity are being introduced because of the use of only one single measurement. Gade [17] suggests to compensate for these errors by first determining the SPL at 1 m distance based on a sound power laboratory measurement. After that, the measured SPL at 1 m distance is determined in-situ on a reflective surface and for a certain fixed angle or aiming point relative to the omnidirectional sound source. The difference between these two values could be applied to the ST_{early} and ST_{late} as a fixed correction factor for future in-situ measurements. After performing this procedure once, there would no longer be the need for calibrating the sound source in the laboratory (in the short term). It should be noted that this correction procedure is not mentioned in ISO 3382-1.

Similar in-situ methods have been used by various researchers to measure G . Beranek [24] notes that all researchers mentioned in his paper used an in-situ calibration measuring the sound pressure level at 1 m distance, except for the Takenaka R&D institute who used a reverberation room calibration. Dammerud and Barron [25] used a reference microphone at

1 m during their scale model measurements, as the used spark source could not reproduce the same output power for every measurement. They extracted the direct sound by windowing over the 0 to 3.5 ms interval of the impulse response. San Martin et al. [26] used a similar approach on concert hall stages by windowing over the 0 to 5 ms interval. An overview of all mentioned calibration methods is presented in Table 3.2.

Table 3.2. Overview of calibration methods using for room acoustic parameters.

Calibration method	Standard	Procedure	Averaging
1. single plane free field method (ISO 3382-1, Annex A)	none	<i>SPL</i> measured in an anechoic room at any distance > 3m and translated to the <i>SPL</i> at 10 m distance	Average taken from 29 rotations in the horizontal plane with steps of 12.5°
2. Reverberation Room (ISO 3382-1, Annex A)	ISO 3471	<i>SPL</i> and <i>T</i> measured in a reverberation room of at least 200 m ³ for octave bands ≥ 125 Hz	<i>SPL</i> measured over 30 s for 6 microphone positions (if standard deviation is below 1.5 dB)
3A. In-situ on stage original (ISO 3382-1, Annex C)	none	The direct <i>SPL/SEL</i> measured in-situ from the measured impulse response using a 0-10 ms window at 1 m source-receiver distance	Single measurement at random angle relative to the omnidirectional sound source
3B. In-situ on stage update [17]	ISO 3471	idem 3A, with additional correction for 1 m distance <i>SPL</i> derived from a sound power measurement in a reverberation room	Single measurement at fixed angle relative to the omnidirectional sound source
3C. In-situ on stage by others [24,25]	none	The direct <i>SPL/SEL</i> measured in-situ from the measured impulse response using a 0-3.5 or 0-5 ms window at 1 m source-receiver distance	Single measurement at random fixed angle relative to the omnidirectional sound source

3.2.3 Signal processing in-situ measurements

In case of source and receiver heights of 1.0 m, the floor reflection will arrive 3.6 ms after the direct sound, and for the 250 Hz octave band, which is the most critical in the 250-2,000 Hz frequency range, these two components will be smeared out over the whole time interval 0–10 ms due to the band filtering. So, it seems to be impossible to separate the direct sound and floor reflection by windowing. In [section 2.3.5](#), it was mentioned that, to reduce the error in determining the sound level of the direct sound separately from a signal with direct sound and a single equally loud reflection to ≤ 0.1 , ≤ 0.5 and ≤ 1.0 dB, a minimum distance between direct sound and reflection of 17, 12 and 9 ms is required respectively for the separate octave bands 250, 500, 1,000 and 2,000 Hz (see [section 3.7](#) for the type of signal processing used)

[14]. When measuring sound levels, it is also necessary to take into account the size of the wavelength and it is recommended to use a time window of at least one corresponding wavelength. For the 250 Hz octave band with lower edge frequency at 176 Hz this suggests that the time window should at least be 6 ms and for the 125 Hz octave band with lower edge frequency at 88 Hz at least 11 ms. So, it seems that a minimum time window of 11 ms is needed for highly accurate measurements starting from the 125 Hz octave band, while room reflections should not arrive before 17 ms. This again shows that the floor reflection cannot be excluded.

Other elements of signal processing exist that might cause variation of results from different researchers. Among others, the determination of the impulse response starting point can be done in various ways, which might influence results from in-situ calibrations. Also, as pointed out by Lundebjerg et al., the method of filtering can be of influence on results of level calculations. Uncertainty introduced by signal processing is mostly relevant for judging reproducibility and less relevant for judging repeatability. Here, we assume that the uncertainty contribution of the signal processing is part of the whole measurement procedure and is therefore included in the analysis. Detailed analysis of uncertainty due to signal processing is outside the scope of our work, but would be interesting for further research.

3.3 Laboratory calibration measurements

3.3.1 Precision methods

The uncertainties in the determination of the sound power (or reference *SPL*) of sound sources have been studied under laboratory conditions by Vorländer and Raabe [27]. They organised a round robin among 8 laboratories during which the sound power of a B&K 4204 reference sound source was measured in hemi-anechoic and in reverberant conditions. The uncertainty due to the use of different types of signal processing was included in the research. They concluded that, in general, both methods yield almost exactly the same results, except in the frequency ranges below 100 Hz and above 10 kHz. The maximum standard deviation σ and reproducibility limit R (probability level of 95% using a coverage factor of 2.8) of both methods are 1.0 dB and 2.8 dB in the 125 Hz octave band and 0.3 dB and 0.8 dB in the 250 to 8,000 Hz octave bands. The reproducibility is similar to the values suggested by ISO 3741:1999.

To the knowledge of the authors, no literature exists about the stability of dodecahedron loudspeakers, which is known to be a critical design element for the type of sound sources. Sound power measurements (at 120 dB sound power level) performed by the authors in a 90 m³ reverberation room, repeated 10 times over four years using a dodecahedron loudspeaker B&K 4292, a B&K 2734 amplifier and a B&K 4189 microphone on a rotating boom, showed a standard deviation of 0.3 dB in the separate 250 to 8,000 Hz octave bands (the 125 Hz octave band could not be evaluated as the room volume was too small). The standard deviation of our measurement with the dodecahedron sound source is similar to the standard

deviation found by Vorländer and Raabe for the reference sound source. It shows that, even over a four year period, the stability of a dodecahedron loudspeaker, in this case a B&K 4292, can be as high as a reference sound source type B&K 4204, provided all other used measurement equipment is highly stable too.

Furthermore, the difference between two types of precision calibrations has been investigated for the dodecahedron loudspeaker type B&K 4292, to investigate whether the difference would fall within the reproducibility limits as found by Vorländer and Raabe and mentioned in ISO 3741:1999. The sound power was determined in a 200 m³ reverberation room in accordance with ISO 3741, see Figure 3.2a, and in an anechoic room via a sound intensity measurement using a sweep scan method following ISO 9614-3 [28], see Figure 3.2b (see section 3.7 for details). Note that the intensity measurement used by us is different from the sound level measurement used by Vorländer and Raabe, although both are executed in a (hemi)anechoic room. Our measured results for the direct reverberation method and anechoic intensity method are presented in Figure 3.3a as average sound power levels relative to the average result of both measurement methods. The maximum difference between L_w for the methods individually and their average L_w is at most 0.3 dB over the 250 – 4,000 Hz octave bands with a standard deviation of 0.5 dB over the individual receiver positions or surfaces. In the 125 and 8,000 Hz octave bands, the maximum difference is larger, up to +/- 1.0 dB with a standard deviation up to 1.0 dB. For the 125 – 4,000 Hz octave bands, the difference in our results are well within the reproducibility limits of $R = 0.8$ dB at 250 – 8,000 Hz and $R = 2.8$ dB at 125 Hz as found by Vorländer and Raabe. We can conclude that the difference in results from our two different measurements can be attributed to each test methods' precision.

3.3.2 Full sphere versus single plane

Besides the reverberation room precision method, ISO 3382-1 recommends performing a sound power measurement in an anechoic room, where the average level is determined from 12.5° rotation of the sound source, denoted here by the 'single plane free field method'. Unlike the intensity measurement, where a full sphere around the source is measured, in the ISO 3382-1 method, only the horizontal plane is measured (see Figure 3.1). To find out if an error is introduced by measuring in a single plane instead of measuring around a full sphere, we performed measurements using the single plane free field method at 1 m and 7 m distance from the physical centre of the omnidirectional dodecahedron sound source in an anechoic room. We compared the results to the intensity measurement presented earlier, which takes into account all directions. A noise signal was recorded at both distances simultaneously while rotating the sound source using a turntable, see Figure 3.2c (see section 3.7 for details). The deviation in measured sound power at both distances from the results of the intensity measurement is presented in Figure 3.3b. It is shown that the results for the two measured distances follow a similar trend. The average over all octave bands is equal for both distances.

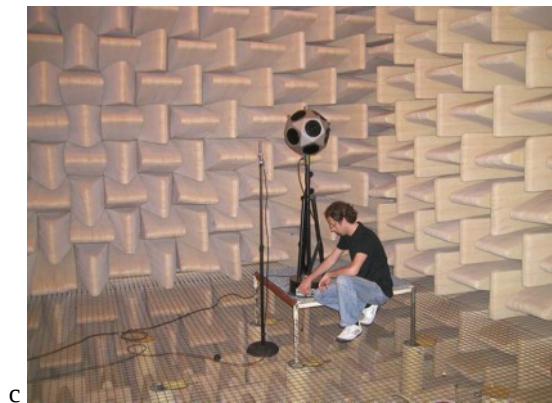


Figure 3.2. (a) Measurement setup in the reverberation room (using a reference source); (b) Measurement setup in the anechoic room using the intensity probe (a metal grid was used to define the scanning surface); (c) Measurement setup in the anechoic room using an omnidirectional microphone and a turntable (single plane measurement).

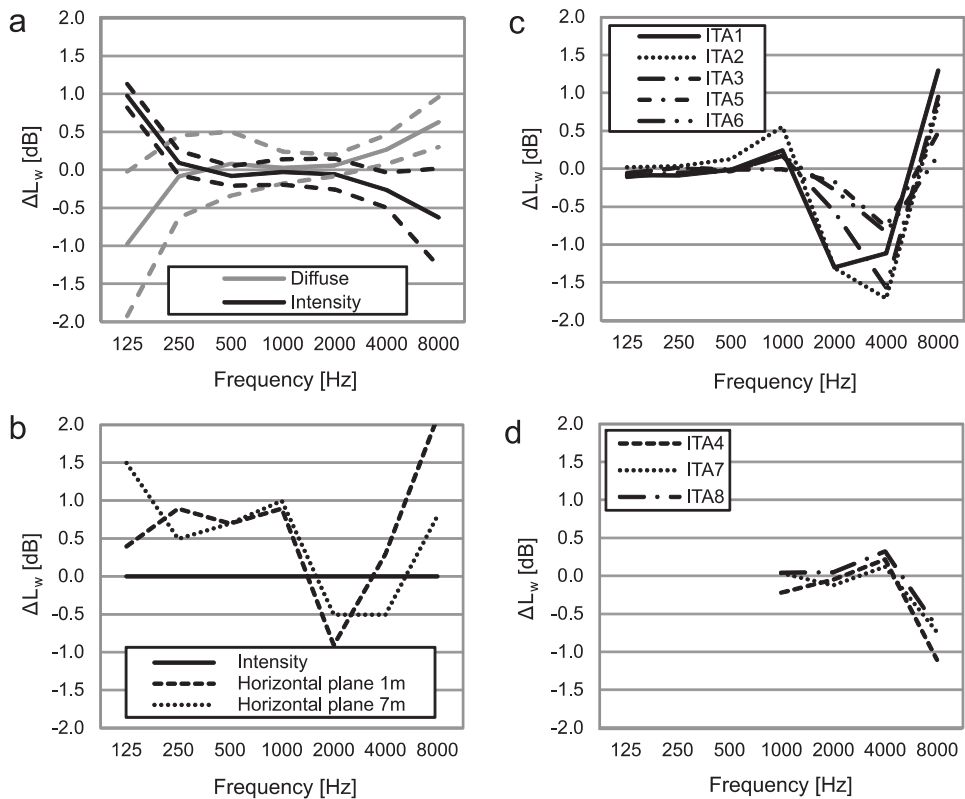


Figure 3.3. (a) Comparison of the sound power levels, measured in diffuse field using an omnidirectional microphone and anechoic conditions using an intensity probe, relative to the mean. (b) Comparison of the sound power level measured at 1 m and 7 m distance in the horizontal plane in an anechoic room, relative to the sound power level measured using an intensity probe over the full sphere in an anechoic room. (c) The deviation between the full sphere average and single plane average for various moderate size dodecahedron loudspeakers (with a section of 300-400 mm). (d) The deviation between the full sphere average and single plane average for various small size dodecahedron loudspeakers (diameter approximately 100 mm) that are used in a three way loudspeaker setup.

The deviation of the results using the single plane free field method from the results from the intensity measurement is above 0.5 dB up to 2.1 dB for almost all octave bands, see Figure 3.3b. Surprisingly, for the frequency range below 500 Hz where the dodecahedron sound source is expected to be fully omnidirectional, the deviation is 0.8 dB on average. Note that the expected reproducibility for precision calibration methods is 0.8 dB for 250 and 500 Hz. This means that the deviation up to 500 Hz, that we found comparing the intensity method and the single plane free field method, can be attributed to the overall precision of the measurement method. However, the larger deviation above 500 Hz might be caused by the simplification of measuring only a single plane instead of measuring a full sphere.

To investigate whether the high frequency deviation between the full sphere and single plane measurement found for the B&K 4292, presented in Figure 3.3b, is representative for dodecahedron loudspeakers in general, directivity data for various dodecahedron loudspeakers have been analysed which were measured by the Institut für Technische Akustik (ITA), RWTH Aachen (sources denoted ITA1 to ITA4 are described in [29]). The directivity was measured with a 5 degree grid over half the sphere, using a similar setup as described in Leishmann et al. [30], Section II. The emitted sound power, averaged over the full sphere, is determined by weighting over the surface area per grid point as described in Leishmann et al. [30], Section IV-B, equations 3 to 6. The emitted sound power, averaged over a single plane, was derived from the 72 measurements in the horizontal plane in the base of the half sphere. The calculated deviations between the full sphere average and single plane average are presented in Figure 3.3c for various moderate size dodecahedron loudspeakers (with a diameter of 300-450 mm) and in Figure 3.3d for various small size dodecahedron loudspeakers (diameter approximately 100 mm) that are used in a three way loudspeaker setup for the 2 kHz octave band and higher. For the moderate size dodecahedrons, a similar frequency trend is found as presented in Figure 3.3b. Now, we can conclude that the single plane method introduces a systematic error with a slight variation over different sound sources with similar size.

3.3.3 Discussion on laboratory calibration measurements

The uncertainty of the precision calibration methods, to determine the sound power of a highly stable omnidirectional sound source, is within 0.8 dB for the octave bands 250 – 8,000 Hz. However, in the 125 octave band, the deviation is found to be > 1.0 dB which is larger than the JND in case of *G*. The absolute deviation between a full sphere average (any precision method) and the single plane average (as suggested by ISO 3382-1) is significant for frequency bands 2,000 – 8,000 Hz with errors up to 2.1 dB for various measured dodecahedron loudspeakers. This systematic deviation can be attributed to the geometrical simplification in the measurement method.

To determine the total uncertainty for each method, models are defined that show how the different identified sources of uncertainty propagate through the measurement procedure. Because the number of data points in the Vorländer and Raabe study and our studies are limited, it is not possible to determine its distribution. However, following Vorländer and Raabe, we have no reason to doubt that data would not be normally distributed. The standard deviation is expanded with a coverage factor of 2 multiplied by a factor $\sqrt{2}$, recommended for cases with a small amount of data. This way, we can expect to arrive at a level of confidence of approximately 95%.

The uncertainty of the precision method and single plane free field method can be expressed by

$$U_{95\%,precision} = \pm 2.8 \sqrt{\sigma_{precision}^2}, \quad (3.1)$$

and

$$U_{95\%,plane} = offset_{plane} \pm 2.8 \sqrt{\sigma_{plane}^2 + \sigma_{precision}^2}, \quad (3.2)$$

where $U_{95\%,precision}$ is the uncertainty of the precision method at 95% confidence level in dB (coverage factor 2.8), $\sigma_{precision}$ is the maximum standard deviation of the precision methods in dB taken from Vorländer and Raabe [27], $U_{95\%,plane}$ is the uncertainty of the single plane free field method at 95% confidence level in dB, and $offset_{plane}$ and σ_{plane} is the average systematic error and standard deviation due to geometrical simplification for 6 moderate size dodecahedron loudspeakers in dB.

Table 3.3 shows the values for the parameters used in equation 3.1 and 3.2 and the calculated uncertainty $U_{95\%,precision}$ for the precision method and $U_{95\%,plane}$ for the single plane free field method. It seems that for the room acoustic parameters mentioned in section 3.2 and presented in Table 3.1, determining all separate octave band values within an JND limit of 1 dB (uncertainty at 95% confidence level) is not possible when using either the precision methods or the single plane free field method for calibration. For the single number ratings G , $L_{p,A,S,4m}$, ST_{early} and ST_{late} , a calibration can be performed with an uncertainty ≤ 1 JND using either a precision calibration method or the single plane free field method. For the STI , the uncertainty is slightly larger than the JND.

Table 3.3. Uncertainty factors and calculated uncertainty for the precision method and single plane free field method

	Octave band with mid frequencies [Hz]							Single number ratings average*			
	125	250	500	1000	2000	4000	8000	125-1000	250-2000	500-1000	STI
$\sigma_{precision}$	1	0.3	0.3	0.3	0.3	0.3	0.3	0.5**	0.3	0.3	0.4**
$offset_{plane}$	0	0	0	0.2	-0.9	-1.1	0.8	0	-0.2	0.1	-0.3
σ_{plane}	0.1	0.1	0.1	0.2	0.6	0.5	0.4	0.1	0.1	0.1	0.1
$U_{95\%,precision}$	+/- 2.8	+/- 0.8	+/- 0.8	+/- 0.8	+/- 0.8	+/- 0.8	+/- 0.8	+/- 1.4	+/- 0.8	+/- 0.8	+/- 1.1
$U_{95\%,plane}^{***}$	0 +/- 2.8	0 +/- 0.9	0 +/- 0.9	0.2 +/- 1.0	-0.9 +/- 1.9	-1.1 +/- 1.6	0.8 +/- 1.4	0 +/- 1.4	-0.2 +/- 0.9	0.1 +/- 0.9	-0.3 +/- 1.2
$U_{95\%,plane}$ lower limit	-2.8	-0.9	-0.9	-0.8	-2.8	-2.7	-0.6	-1.4	-1.1	-0.8	-1.5
$U_{95\%,plane}$ upper limit	2.8	0.9	0.9	1.2	1.0	0.5	2.2	1.4	0.7	1.0	0.9

* under the assumption that the deviation in different octave bands is correlated

** estimated by the average over the octave bands

*** offset and deviation at 95% confidence level and 2.8 coverage factor

3.4 Uncertainties due to directivity

In the previous section, some directivity characteristics of the dodecahedron sound source were already revealed in the difference between a sound power measurement averaged over the full sphere and averaged over a rotation in a single plane only. In this section, the impact of sound source directivity on room acoustical measurements, and sound source calibration in particular, is investigated.

3.4.1 Background on polyhedron loudspeakers and directivity

Recently, the properties of omnidirectional sound sources have been investigated in more depth. Leishmann et al. [30] compared the directivities of various regular polyhedron loudspeakers (RPL). They conclude that none of the investigated RPL's are being consistently exceptional in omnidirectional behaviour above their 'cut-off frequency', the frequency above which the directivity increases rapidly (see [30] for the definition of 'cut-off frequency'). Among others, the reduction in omnidirectional behaviour is caused by the spread of the individual loudspeakers relative to the wavelength and by the break up phenomenon on the individual loudspeakers' cone. The tetrahedron, with four loudspeakers, showed the best performance in the 4 kHz band. The dodecahedron was also found to be a good choice, among other things due to its highest cut-off frequency (1,463 Hz for a 14.6 cm radius) and the most uniform radiation in the 2 kHz octave band. An advantage of the dodecahedron shape, is that the increase of the number of loudspeakers will result in a higher (possible) sound power. It is striking, that the icosahedron shape with 20 loudspeakers, the type that was used by Gade [23], did not show a more uniform radiation than any other RPL with a lower amount of loudspeakers.

Lundeby et al.[4] raised the problem of the effect of the directivity of the dodecahedron and the cube loudspeaker on measured room acoustic parameters. They indicate that, for 18 rotation steps of 20°, a standard deviation of almost 0.4 dB in G can be found at the 4 kHz octave band. Actually, San Martín et al. [31] found that, for single measurements in a concert hall at 18 m distance, deviations in all room acoustic parameters, except T_{30} , were above the JND for random orientations of two different sound sources at octave bands with mid-frequencies above 1 kHz. Also, for 24 measurements taken in the horizontal plane (see Figure 3.1), over a 120° area in steps of 5° using both sound sources, the standard deviation at 2 and 4 kHz is larger than one JND for G . San Martín et al. conclude that the uncertainty is not sufficiently reduced when an average value is taken over 3 rotations, as suggested by ISO 3382-1 for field measurements (mentioned authors state that it is common practice to average within a 120° area in steps of 40°). As mentioned before, for sound source calibration, ISO 3382-1 recommends using 29 steps of 12.5°.

The possibility of averaging over multiple rotations with equally sized steps (for instance 4 steps of 90 degrees or 6 steps of 30 degrees) has been investigated by the authors, and our first results were presented in [32]. The maximum sound level deviation from the sound level determined from a full rotation average was determined for various equal-angular step

averages. For every possible single rotation (= 1 step), and for every possible multiple rotation averages over equal-angular steps up to a number of 8 steps with any random initial aiming point, the maximum deviation in sound level from the full rotation average was determined at various source-to-receiver distances in a concert hall. An even distribution of the steps over the full rotation was chosen to take into account small differences in sound power by the different loudspeakers (instead of rotating within a single 120° area). It was concluded that, when choosing any of the 1, 2, 3, 4 or 6 equal-angular rotation averages, the maximum deviation was not significantly reduced. Surprisingly, when using an average over 5, 7 or 8 equal-angular rotations, the maximum possible deviation from the full rotation average could be dramatically reduced, far below the JND up to the 4 kHz octave band for all source-to-receiver distances. The dodecahedron loudspeaker has rings of 3 or 6 loudspeakers distributed along the axis of rotation. Possibly, the number of 5 and 7 equal-angular rotations both work because these are prime numbers resulting in a non-symmetrical distribution of measurement points. (These results are only based on one condition, a random concert hall. To confirm the validity of the equal-angular averaging method under different conditions, results will also be presented at the end of this section for anechoic conditions using multiple sources).

Investigations were continued by Martelotta [33], who looked at the possibilities of finding an optimal choice of (relatively small) angles for a two or three step rotation average with any random starting point. For his study, he measured room impulse responses for every 5° step over 120°, similar to San Martín et al. [31]. He concludes that two measurements, spaced by 60°, or three measurements, spaced by 30°, result in the lowest standard deviation from the average compared to single measurements (using any of the discrete 5° steps as the initial aiming point) in room acoustical parameters G , C_{50} , LF and EDT . For a G measurement at 10 m distance in a 1,200 m³ auditorium, the standard deviation, averaged over 2 and 4 kHz, was reduced from 0.67 dB for a single measurement to 0.25 dB and 0.31 dB for the two or three step rotation average.

3.4.2 Comparison study of single plane rotational averaging methods

Table 3.4 gives an overview of findings on deviations due to directivity and source rotation. It can be concluded that, for judging level related parameters in separate octave bands ≥ 2 kHz and at a single position in a hall, the standard deviation due to source directivity can be larger than the JND. An even larger uncertainty can be expected when measuring the sound level close to the sound source, which is relevant for this work. However, there is no agreement on which type of rotation average should be used to find a sufficiently reliable average over the horizontal plane (while ignoring the fact that a single plane average is not the same as a full sphere average). To investigate the various rotational averaging methods, impulse response measurements have been performed by the authors in an anechoic room at 7 m distance using a dodecahedron loudspeaker B&K Type 4292 (see section 3.7 for more details). For every rotation of 5°, an impulse response was determined

resulting in 72 measurements. Then, for all averaging methods presented in Table 3.4, every possible average in sound level was determined (for instance, in case of a ‘3 rotation average within a 120° area in steps of 40°’, the average could be determined 24 times) and its absolute deviation from the average over all 72 measurements is calculated. Linear interpolation was used when needing angles in between the 5° step. Over all these possible absolute deviations within a certain rotational average method, the standard deviation has been determined for each separate octave band from 500 to 8,000 Hz.

Table 3.4. Overview of findings on directivity and source rotation.

Authors	Orientation and averaging	Conclusions	Distance	Denoted
Lundebj et al. [4]	18 rotation steps of 20°	σ of almost 0.4 dB at the 4 kHz	auditorium, distance not stated	-
San Martín et al.[31]	2 sources with random orientation	absolute deviation in all room acoustic parameters (except T30) above the JND for bands > 1kHz	concert hall at 18 m distance	-
San Martín et al.[31]	24 single measurements over an 120° area in steps of 5°	σ over rotation results at 2 and 4 kHz above JND	concert hall at 18 m distance	‘single’
San Martín et al.[31]	3 rotation average within a 120° area in steps of 40°	σ not significantly reduced	concert hall at 18 m distance	‘0-40-80’
ISO 3382-1 [1]	29 rotation average in steps of 12.5°	deviation unknown	>3 m in anechoic room	‘29x12.5’
Hak et al. [32]	2 to 8 equal-angular rotation average	maximum deviation reduced far below the JND for all octave bands by averaging over 5, 7 or 8 equal-angular rotations	concert hall at 1, 5 and 18 m distance	‘2EA’ to ‘8EA’
Martelotta [33]	2 or 3 rotation average within 120° area	optimal choice is 60° spacing for 2 rotation average or 2x30° spacing for 3 rotation average.	auditorium at 10 m distance	‘0-60’ ‘0-30-60’

Results are presented for the individual octave bands in Figure 3.4 for the B&K 4292. The standard deviation increases with frequency up and until 2 kHz, while variations in the 2, 4 and 8 kHz bands are of the same order of magnitude. To confirm that the results found for the B&K 4292 are representative for dodecahedron loudspeakers in general, the ITA directivity data for various dodecahedron loudspeakers as presented in section 3.3, has been analysed and the average standard deviations for the 2, 4 and 8 kHz octave band are presented in Figure 3.5a for the moderate size dodecahedron loudspeakers and in Figure 3.5b for the small size dodecahedron loudspeakers. The standard deviation of 72 repeated measurements without rotation was found to be below 0.03 dB, proving that the deviations found are indeed caused by directivity variations.

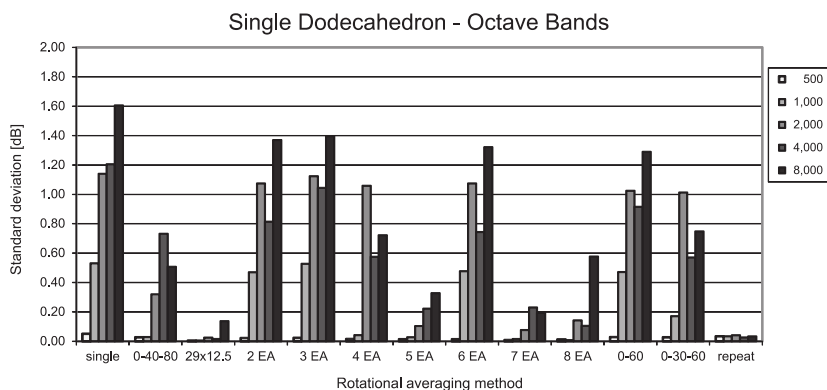


Figure 3.4. Standard deviation of the sound level by stepwise rotations, for a B&K 4292 measured at 7 m distance in an anechoic room, in single octave bands from 500 to 8,000 Hz for various averaging methods (see Table 3.4 for explanation of methods).

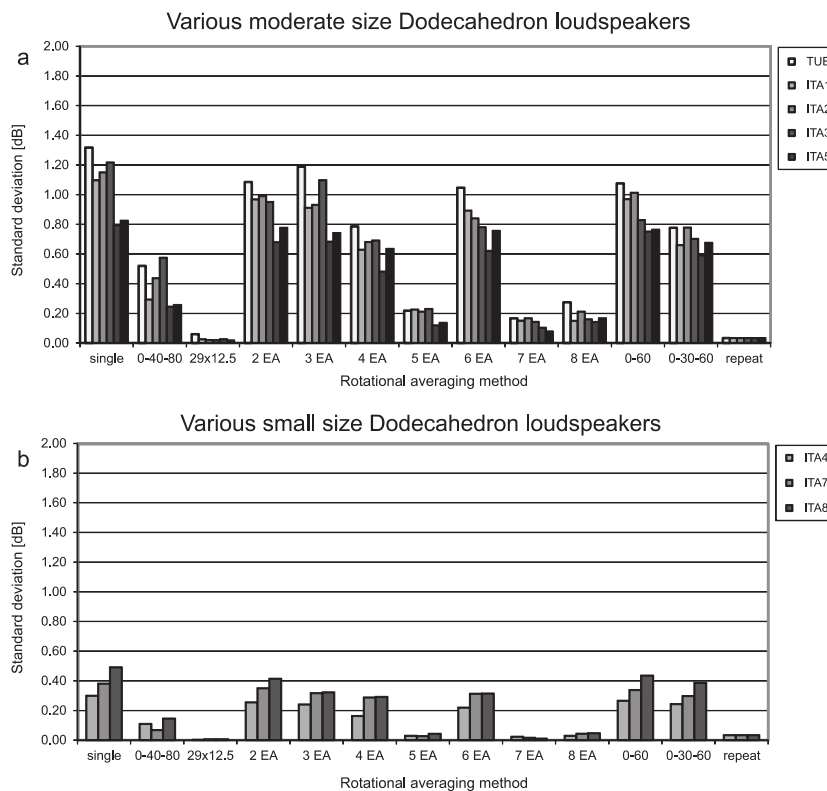


Figure 3.5. Average standard deviation of the sound level by stepwise rotations for 2, 4 and 8 kHz for various loudspeakers: (a) moderate size dodecahedron loudspeakers (b) small size dodecahedron loudspeakers.

3.4.3 Discussion on uncertainties due to directivity

From our results of deviation in sound level for the various single plane averaging methods as discussed in the previous subsection, it can be concluded that an accurate estimation of the single plane average in single octave bands above 500 Hz is not possible using a single random measurement. The deviation from a full single plane average is only reduced significantly for the ISO 3382-1 method ('29x12.5') and the equal-angular rotations using 5 and 7 steps ('5EA' and '7EA'). For the 5 steps equal-angular rotation, the standard deviation introduced is 0 dB in the octave bands up until 1 kHz and 0.1, 0.2 and 0.3 dB at 2, 4, and 8 kHz respectively. It should be noted that, even though the result of a multiple step rotation average may approach the result of a single plane average, results can still deviate up to 2 dB from a full sphere average as mentioned in section 3.3. For laboratory calibration purposes, it is therefore recommended to either measure a full sphere average if one is interested in accurate separate octave band measurements, or measure as many steps as possible (with 29 steps σ is below 0.1 dB) to determine the single plane average if one is only interested in the single number ratings mentioned in section 3.3.3. However, as we will discuss in the next section, a single plane rotational average with less averaging steps (only 5) can be a practical tool when calibrating in the field.

3.5 Field calibration measurements

The possibility of performing calibrations in the field has been investigated. The deviation of in-situ calibration compared to laboratory calibration has been reported by various researchers.

3.5.1 Interference by the floor reflection

Gade stated in the appendix of his report on 'Acoustical Survey of 11 European Concert halls' [23] that at first, he used an in-situ calibration at 1 m distance on stage to be able to measure G , while applying a time window to filter out the direct sound together with the floor reflection. He corrected the calibration level based on the theoretical effect of the floor reflection. Afterwards, when comparing the in-situ calibration to a reverberation room calibration, he concluded that the theoretical correction did not predict the actual deviation accurately. As Gade noted, the floor reflection does not increase the sound level equally in all frequency bands. Due to comb filtering, some frequencies will be amplified while others are cancelled out. Figure 3.6 shows the difference in SPL between the average of two on stage measurements and a measurement in an anechoic room in one-third-octave bands and full octave bands, performed by the current authors at 1.5 m transducer heights. The effect of interference can be clearly observed up to 500 Hz. Above 500 Hz, an average increase in SPL is found of 1.1 dB. The interference effect is not visible in the full octave band data.

Beranek [34] also reports the difference between an in-situ measurement using stationary noise measured with two microphone positions on either side of a dodecahedron sound

source, and a calibration performed in a reverberation room, see Table 3.5. Hak et al. [5] showed results for various cases, among which are measurements at 1 m distance on stage of two concert hall stages, see Figure 3.7. A full rotation average was taken using stationary noise and an 8 equal-angular step rotation average was taken using impulse responses. It should be noted that in the latter, actually a system calibration was performed and the source to receiver distance was determined from the impulse response, causing a 15 cm deviation in distance determination. More in-situ measurements were compared to a calibration in an anechoic room by Dammerud [35] for G_{0-10} using 29 averages of 12.4° steps in the anechoic room and 4 single measurements at 1 m distance on 8 different stages. A single measurement was taken per position with the same rotation of the sound source relative to the microphone for all measurements, as suggested by Gade [17].

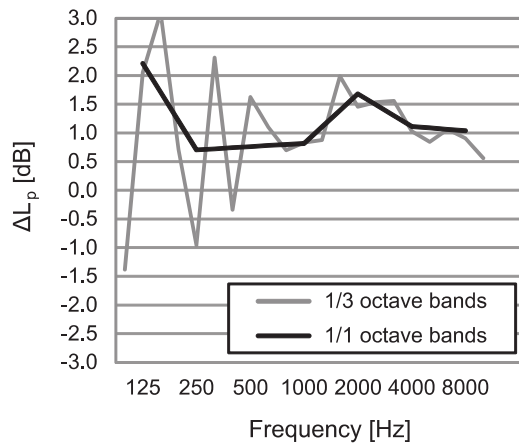


Figure 3.6. Effect of floor interference: difference in *SPL* between the average of two on stage measurements and a measurement in an anechoic room. Results averaged over a full rotation in the horizontal plane while producing a noise signal.



Figure 3.7: In-situ measurement setup on a concert hall stage with a source-receiver distance of 1 m.

Table 3.5. Overview of errors found by using an in-situ calibrations.

ΔL_w (error in L_w by in-situ calibration)	Int. [ms]	h [m]	d [m]	Octave band with mid frequencies [Hz]						Average	
				125	250	500	1k	2k	4k	500-1k	250-2k
1. Theoretical [23]	0-10	1.2	1	-4	1.7	-0.2	0.6	0.4	0	0.2	0.6
2. Reverberation Room vs Stage [23]	0-10	1.2	1	0	2.5	1.1	0.5	0	0	0.8	1
3. Reverberation Room vs Stage [34]	0-inf	1.5	1	2.7	2.2	1.7	1.1	0.9	0.4	1.4	1.5
4. Precision vs Stage, Noise [5]	0-inf	1.5	1	2.9	1.2	1	1.3	0.1	0.3	1.2	0.9
5. Precision vs Stage, System cal. [5]	0-inf	1.5	1*	-1.5	0	0.1	-0.1	0.6	0.8	0	0.2
6. Anechoic vs 8 Stages [35]	0-10	1.0	1	-2.2	2.7	0.3	1.1	1	0.7	0.7	1.3
Average over 2, 3, 4 and 6 (offset _{insitu})				**	2.2	1.0	1.0	0.5	0.4	1.0	1.2
Standard deviation over 2, 3, 4 and 6 (σ_{insitu})				2.4	0.7	0.6	0.3	0.5	0.3	0.3	0.3

*The physical distance was 1.0 m, but the distance determined from the impulse response and used in the system calibration was 0.85 m.

** The measured results have a varying sign, therefore the average is not a valid offset.

The various deviations found are summarised in Table 3.5 together with an average offset and standard deviation for the measurements that used a correct source-receiver distance. All researchers used a 0-10 ms time window on a stage with objects or surfaces beyond 4 m distance, windowing out room reflections and reducing the maximum error by windowing to 0.1 dB (see section 3.2.2). It is likely that the floor reflection is an important cause for these errors. The variation in the measurements might be explained by directivity problems of the

sound source, by different types of omnidirectional sound sources and by different transducer heights. Other possible causes of error might be microphone misplacements and other unknown variations due to moving the sound source and microphone. These possible causes of error will be discussed in the next two subsections.

3.5.2 Microphone misplacement

The error due to transducers ‘misplacement’ was investigated by Gehe in a master thesis [36]. He found that the error in G_{0-10} , due to his ± 1 cm error in sound source to microphone distance, would be negligible. Earlier, Gade [17] already concluded that a distance error of up to 30 cm, would lead to an error in the ST parameters of maximum 1 dB. Obviously, such a large error in distance is never made when performing measurements accurately. To investigate the actual deviation in measured sound level due to misplacement of the microphone, the current authors measured the sound level using impulse responses, windowed for 0-10 ms, on a concert hall stage at a distance d of 0.950, 0.975, 1.000, 1.025 and 1.050 m distance in steps of 2.5 cm (see Annex A for more details). To reduce the error due to directivity deviation, a 5 equal-angular stepwise rotation average was used. The transducers height was varied from 1.0 m, 1.2 m to 1.5 m. Figure 3.8 shows the results for a 1.2 m transducer height together with theoretical values based on the inverse square law $20\lg(d)$. It is clear that the measured results deviate from the theoretical results, possibly due to the floor reflection, but the results are close to the theoretical values for the 250 to 1,000 Hz octave bands. These results show that accurate placement of the transducers is necessary for direct field calibrations. When the microphone placement is done within 1 cm, the maximum error can be expected to be below 0.1 dB and can therefore be neglected.

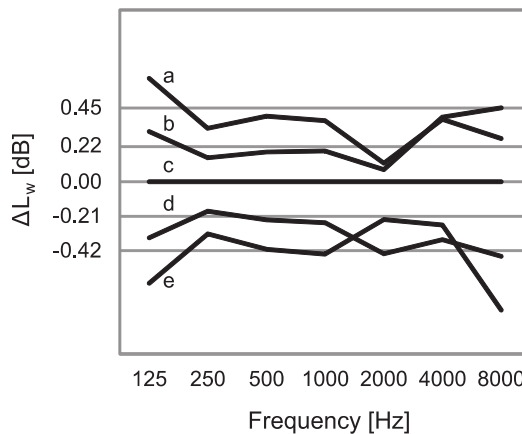


Figure 3.8. Sound level per frequency measured on a concert hall stage at a distance of a: 0.950, b: 0.975, c: 1.000, d: 1.025 and e: 1.050 m distance, normalised to the 1.000 distance measurement, together with theoretical values derived using the inverse square law: 0.45 dB, 0.22 dB, 0 dB, -0.21 dB and -0.42 dB respectively.

3.5.3 Variation over different positions and the influence of objects

Gehe [36] also looked at the variation in measured ST reference level at different positions on stage. For 13 measurements spread over a stage, the standard deviation in G_{0-10} was only $\Delta\sigma_{250-2000} = 0.2$ dB. The variation of G_{0-10} over various positions was also determined for the data from Dammerud [35] by the current authors. The average standard deviation in G_{0-10} over only 4 measurements is found to be $\sigma_{250-2000} = 0.1$ dB for 2 stages without risers (in this context, risers are devices to create height differences on the stage), and $\sigma_{250-2000} = 0.2$ dB for 6 stages with risers. In both cases, such low standard deviations are achieved, as long as the exact same orientation of the dodecahedron sound source towards the microphone is used, and measurements on risers are performed with both transducers on the same riser (and same height). We performed a similar study for various sets of our own measurement data of stage acoustic measurements using a dodecahedron loudspeaker. During the measurements, no special attention was given to aim the loudspeaker towards the microphone accurately, because a 5 equal-angular step rotation average was used as a method to improve accuracy, see section 3.4. On one stage, the effect of chairs on the measured deviation was investigated.

Table 3.6. Overview of standard deviation of in-situ calibrations at different positions on stage.

	Int. [ms]	h [m]	d [m]	Octave band with mid frequencies [Hz]						Average	
				125	250	500	1000	2000	4000	500- 1000	250- 2000
Gehe [36]: with risers (13 positions, rotation aimed)	0-10	1.0	1.0								0.2
Dammerud [35]: no risers (4 positions, rotation aimed)	0-10	1.0	1.0	0.2	0.2	0.1	0.1	0.3	0.2	0.1	0.1
Dammerud [35]: with risers (4 positions, rotation aimed)	0-10	1.0	1.0	0.5	0.5	0.3	0.2	0.3	0.4	0.3	0.2
concert hall, empty stage (4 positions, rotation random)	0-10	1.35	1.0	0.1	0.1	0.1	0.4	2.4	1.6	0.3	0.6
concert hall, empty stage (4 positions, 5 step average)	0-10	1.35	1.0	0.1	0.1	0.0	0.1	0.2	0.6	0.1	0.1
concert hall, stage with chairs (4 positions, 5 step average)	0-10	1.35	1.0	0.5	0.3	0.2	0.8	1.0	0.8	0.5	0.5

In Table 3.6, all available results of σ per octave band are presented. A maximum $\sigma = 0.6$ dB per octave band and $\sigma = 0.2$ dB for single number ratings is reached on stages without risers, when aiming the sound source to the microphone, or taking a 5 equal angular step rotation average. Risers introduce a slight increase in uncertainty with a $\Delta\sigma_{250-2000} = 0.1$ dB, while chairs in (very) close proximity to the transducers increase the uncertainty only

moderately by $\Delta\sigma_{250-2000} = 0.4$ dB. As expected, single measurements show a larger deviation at frequencies above 500 Hz when the omnidirectional sound source is not aimed at the microphone correctly: a single measurement results in an $\sigma_{250-2000}$ of 0.6 dB while a 5 equal angular step rotation average results in an average $\sigma_{250-2000}$ of 0.1 dB. Adding chairs in close proximity to the transducers results in a $\sigma_{250-2000}$ of 0.5 dB.

3.5.4 Discussion on field calibration methods

It can be concluded, that the difference in laboratory and field calibration methods found by different researchers varies considerably. The variance in the measurements by different authors cannot be explained by the error due to time windowing (maximum error 0.1 dB) and microphone misplacement (maximum error 0.1 dB with 1 cm accuracy). Performing multiple in-situ measurements in various halls using the same equipment results in a standard deviation of 0.1 - 0.6 dB per octave band (on a flat stage floor), which includes the time windowing uncertainty, microphone misplacement uncertainty and the uncertainty due to the source aiming for single measurements or the horizontal plane averaging. It is likely that the largest part of the deviations found between laboratory and field calibration methods can be attributed to the floor reflection. This means that a correction for the floor reflection is the most important step to reduce the error. The correction should be determined by comparing the in-situ sound power to a laboratory sound power using precision methods.

To determine the total uncertainty for the in-situ methods, a model is defined that shows how the different identified sources of uncertainty propagate through the measurement procedure. While including a coverage factor of 2.8, the in-situ method can be expressed by an offset and an average standard deviation by

$$U_{95\%,insitu} = offset_{insitu} \pm 2.8\sqrt{\sigma_{insitu}^2 + \sigma_{precision}^2 + \sigma_{repositioning}^2 + \sigma_{rotation}^2} \quad (3.3)$$

and with laboratory correction,

$$U_{95\%,insitu,corrected} = 2.8\sqrt{\sigma_{precision}^2 + \sigma_{repositioning}^2 + \sigma_{rotation}^2}, \quad (3.4)$$

where $U_{95\%,insitu}$ is the uncertainty of the in-situ method at 95% confidence level in dB, $offset_{insitu}$ and σ_{insitu} is the average systematic error and standard deviation due to in-situ calibration for 4 different researchers, $\sigma_{precision}$ is the standard deviation of the precision methods, $\sigma_{repositioning}$ is the standard deviation of 4 repeated measurements over different stages and $\sigma_{rotation}$ is the standard deviation of the rotational averaging method (with a fixed aiming point $\sigma_{rotation}$ is assumed to be 0).

The calculated uncertainty for the in-situ method, without and with laboratory correction, is shown in Table VII in case of using a 5 equal-angular step rotation average. It is clear that actual in-situ calibrations should be avoided for accurate level measurements because the uncertainty of individual octave band values and single numbers ratings for G and ST parameters is (much) larger than the parameters' JND. However, the in-situ method with laboratory correction as proposed by Gade, where a fixed correction value is used for the difference in laboratory and field calibration (see section 3.2.2), seems promising in terms of

being both a reasonably accurate and practical method for the single number ratings. Aiming the sound source to the microphone, or taking a 5 step rotation average without aiming, both are an effective method to control the deviation in measured reference level between different positions on stage as long as the microphone is placed at the correct distance within 1 cm. The introduction of risers does not appear as a concern for measuring the reference level, but, chairs close to the transducers should be avoided. To avoid the introduction of any other uncertainties, the signal processing should be kept identical after determining the fixed correction. It should be noted, that the in-situ method was not tested for calibration inside office spaces and therefore no results are presented for STI.

Table 3.7. Uncertainty factors and calculated uncertainty for in-situ method, without and with laboratory correction using precision methods.

	Octave band with mid frequencies [Hz]						Single number ratings average*	
	125	250	500	1000	2000	4000	250-2000	500-1000
$offset_{insitu}$	0	2.2	1.0	1.0	0.5	0.4	1.2**	1.0**
σ_{insitu}	2.4	0.7	0.6	0.3	0.5	0.3	0.3	0.3
$\sigma_{precision}$	1	0.3	0.3	0.3	0.3	0.3	0.3	0.3
$\sigma_{repositioning}$	0.2	0.2	0.1	0.1	0.3	0.2	0.1	0.1
$\sigma_{rotation}$	0	0	0	0	0.1	0.2	0	0
$U_{95\%,insitu}^{***}$	0 +/-7.3	2.2 +/-2.2	1 +/-1.9	1 +/-1.2	0.5 +/-1.9	0.4 +/-1.4	1.2 +/-1.2	1 +/-1.2
$U_{95\%,plane}$ lower limit	-7.3	0	-0.9	-0.2	-1.4	-1	0	-0.2
$U_{95\%,plane}$ upper limit	7.3	4.4	2.9	2.2	2.4	1.8	2.4	2.2
$U_{95\%,insitu-corrected}^{****}$	+/-2.9	+/-1	+/-0.9	+/-0.9	+/-1.2	+/-1.2	+/-0.9	+/-0.9

* the 125-1,000 Hz and STI average were not calculated due to a lack of sufficient data

** these values are corrected and are misprinted in the published paper

*** offset and deviation at 95% confidence level and 2.8 coverage factor

**** deviation at 95% confidence level and 2.8 coverage factor

3.6 Conclusion

The results from above-mentioned existing and new research illustrate the concern about the accuracy of calibrating the dodecahedron loudspeaker as an omnidirectional sound source for room acoustical measurements of Sound Strength (G), Speech Level (L) and Support (ST) type of parameters. When measuring the sound power produced by a dodecahedron loudspeaker as an omnidirectional sound source, the effect of its directivity cannot be neglected and many other errors can be introduced, especially when calibrating in-situ. We can discriminate between three groups of calibration methods, each with a different uncertainty for different frequency bands and for the single number ratings for G , L_J , ST , $L_{p,A,S,4m}$ and STI . For each method and parameter, the uncertainties are summarized in Table 3.8.

3.6.1 Precision methods

A laboratory calibration using precision methods like the reverberation room or intensity method results in an uncertainty of 2.8 dB at 125 Hz and 0.8 dB in the frequency range 250-8,000 Hz. The most practical method to achieve a full sphere average is probably the reverberation room method as mentioned in ISO 3382-1, as was also recently suggested by Beranek and Nishihara [37]. For single number ratings, the uncertainty varies between 0.8 and 1.4 dB. The precision methods can be sufficiently accurate to be able to calibrate the sound source for measuring the room acoustic parameters G , ST and $L_{p,A,S,4m}$ because their JND's are larger than the uncertainty, except for the separate 125 Hz octave band. Single number ratings for parameters that are sensitive for the large uncertainty at the 125 Hz octave, like the STI and L_I have an uncertainty > 1 dB. For measuring STI , the uncertainty of the precision method is slightly larger than the JND. This holds for the worst case scenario where the room acoustics has no influence (anechoic room). In practice the uncertainty due to calibration for measuring STI will be (just) below the JND.

3.6.2 Single plane free field method

The method mentioned in ISO 3382-1, where the sound power is determined from 12.5 degree steps while rotating in the horizontal plane, is conceptually clear but it results in a systematic error. Above 1,000 Hz, we showed deviations up to 2 dB per octave band between the single plane free field method and a full sphere measurement based on measurements of 9 different dodecahedron loudspeakers. It is clear that the deviation can be significantly improved by reducing the dodecahedron's diameter, but a sound power measurement should always cover the full sphere if one is interested in separate octave bands. For measuring single number ratings for G , ST , and STI , the uncertainty of a single plane average is on average 33% more than the precision methods.

3.6.3 In-situ measurement

We have shown that an in-situ calibration at 1 m distance is not accurate with an uncertainty of ± 7.3 dB for separate bands and ± 2.4 dB for single number ratings G and ST . The systematic deviation between the in-situ measurement and a laboratory calibration can be corrected if the error is known for the particular sound source and transducers' height. This can be useful in circumstances where the actual sound power of the source is difficult to reproduce for each different measurement condition. The correction between the laboratory calibration and in-situ calibration can be determined using a full rotation average (using a 12.5° steps average resulting in 29 measurements or using an average over 5 equal-angular rotation steps) or by determining the correction between the laboratory calibration and the in-situ measurement at 1 m for a fixed sound source aiming point and fixed height (as suggested by Gade [17]). Results from various researchers have shown that the single number ratings of measured results using both the rotational average or fixed aiming point method can be reproduced with 0.3 dB standard deviation over different measurement points

on a single stage, with and without risers, and over stages in different halls. However, the in-situ method including laboratory correction should only be used for measuring single number ratings for G and ST and it should not be used in case one is interested in separate octave bands.

Table 3.8. Summary of measurement uncertainty at 95% confidence level of different calibration methods for moderate size dodecahedron loudspeakers

Type of calibration	Uncertainty				
Frequency	octave bands*	L_J	ST	G , $L_{p,A,S,4}$ m ,	STI^{**}
JND	1-2 dB	-	2 dB	1 dB	1 dB
Precision method ISO 3740 reverberation room or intensity probe over sphere or free field over sphere	$\leq \pm 2.8$	± 1.4	± 0.8	± 0.8	± 1.1
Single plane free field method - as suggested by ISO 3382-3 - using a full rotation average or 29 steps of 12.5 degrees.	$\leq \pm 2.8$	0 $\pm 1.4^{***}$	-0.2 $\pm 0.9^{***}$	0.1 $\pm 0.9^{***}$	-0.3 $\pm 1.2^{***}$
In-situ measurement: - on a stage floor without risers or chairs - 0-10 ms time window - microphone placement accuracy 1 cm - a fixed source to microphone aiming point or 5 equal-angular rotation average	$\leq \pm 7.3$	-	1.2 $\pm 1.2^{***}$	1 $\pm 1.2^{***}$	-
In-situ corrected As in-situ measurement with additional correction for difference between laboratory precision measurement and in-situ measurement	$\leq \pm 2.9$	-	± 0.9	± 0.9	-

* the maximum uncertainty over all octave bands is shown, which is dominated by the 125 Hz octave band. For uncertainties per octave band, see Table 3.3 and Table 3.7.

** The uncertainty presented for STI holds for the worst case scenario when the room acoustics has no influence. The uncertainty due to calibration for measuring STI might be lower than presented.

*** the two figures represent the offset and the random deviation

3.6.4 Conclusion

Based on the JND for single number ratings, an uncertainty <1.0 dB would be desired for measuring G and STI and an uncertainty <2.0 dB for measuring ST_{early} of ST_{late} . It is clear that even the most accurate calibration processes described in this chapter have uncertainties that are in the same order of magnitude of the parameters' JND. This might seem as if these calibration processes are sufficiently accurate. However, it should not be overseen that performing an accurate sound power calibration is just the first step in taking an accurate measurement of the room acoustic parameters. For instance, the directivity of the

dodecahedron sound source will still influence the measurement result with errors well above 1.0 dB [4, 30, 31, 32]. With current available measurement methods, it still might not be possible for room acoustical parameters that use the sound power as a reference level, to be measured accurately enough.

3.7 Appendix

3.7.1 Measurement equipment

For our measurements the same measurement set was used. The power amplifier had a built-in white noise generator. For every measurement this noise generator was set to exactly the same value, so the sound power level of the omnidirectional sound source was always the same. The sound source was a 12 loudspeaker omnidirectional sound source (dodecahedron) with a diameter of approx. 40 cm. The measurement equipment consisted of the following components:

- sound source: omnidirectional (B&K Type 4292), directivity in compliance with ISO 3382;
- signals: stationary white noise and an exponentially swept sine;
- input/output: USB audio device (Acoustics Engineering - Triton);
- power amplifier: (Acoustics Engineering - Amphion);
- turntable: 80 s for one rotation (B&K Type 2305);
- microphone: ½" omnidirectional ICP (B&K Type 4189);
- sound intensity probe: (B&K Type 3520);
- software: DIRAC (B&K Type 7841).

3.7.2 Diffuse-field measurement conditions:

The diffuse-field measurements were carried out in the reverberation room (200 m³) of the Faculty of Applied Sciences of the Delft University of Technology. All sound power measurements were done according to ISO 3741. According to this standard two source positions and four microphone positions were used for the direct method. For the system calibration measurements (impulse response measurement using e-sweeps and deconvolution) two sound source positions and three microphone positions were used. For each situation the measurement results were averaged over the microphone and sound source positions.

3.7.3 Sound Intensity measurements:

The sound intensity measurements were carried out in the anechoic room of the Faculty of Applied Sciences of the Delft University of Technology. The measurements were performed according to ISO 9614-3 using a metal mesh cube with dimensions 1.05 x 1.05 x 1.05 m³ and a mesh size of 15 x 15 cm². Using the sweep scan method all individual surfaces were scanned in two directions and averaged to one intensity measurement. The sound

intensity of the bottom surface was determined by turning the omnidirectional sound source upside down. The total sound power level is obtained by summing the 6 separate sound intensity results.

3.7.4 Free-field measurement conditions:

The free-field measurements were also carried out in the anechoic room of the Faculty of Applied Sciences of the Delft University of Technology. In this room a measurement is performed at 1 m distance (near field) and 7 m distance from the centre of the omnidirectional sound source. For both distances full rotational measurements were performed using stationary white noise while rotating the sound source around its vertical axis using a turntable with a rotation speed of 360°/80s. For the stepwise rotational measurements, exponential sweep signal were used to determine an impulse response.

3.7.5 In situ measurement conditions:

The measurements in situ were carried out on the stage of the symphonic concert hall and the chamber music hall of the Frits Philips Muziekcentrum Eindhoven. The (near-field) measurements are performed at a distance of 1 m from the centre of the omnidirectional sound source. During the measurements the stages were unoccupied.

3.7.6 Signal processing

The impulse response is calculated through deconvolution of the room's response to a stimulus signal with the stimulus signal itself. The result is a time domain signal which is filtered using band-pass filters as recommended by ISO 3382-1 and using the reverse filtering technique. The filters are IEC 61260-compliant full and third octave frequency band filters. To enable accurate measurements of very short reverberation times, time reversed filtering is used. According to ISO 3382-1 the impulse response starting point is determined from the broadband impulse responses and defined as the point where the signal level first rises above 20 dB below the maximum level. However, in all free field measurements presented in this chapter, the source to receiver distance was measured by the distance between the physical centre of the dodecahedron sound source and the microphone (without using information from the impulse response).

3.8 References

- [1] ISO 3382: Acoustics – Measurement of room acoustic parameters. International Organisation for Standardisation (ISO). Geneva (CH), 2009.
- [2] IEC 60268-16: 2003: Sound system equipment – Part 16: Objective rating of speech intelligibility by speech transmission index. International Electrotechnical Commission, Geneva (CH), 2003.
- [3] ISO 3382-3:2012: Acoustics – Measurement of room acoustic parameters – Part 3: Open plan spaces. International Organisation for Standardisation (ISO), Geneva (CH), 2012.

- [4] A. Lundeby, T.E. Vigran, H. Bietz, M. Vorländer: Uncertainties of measurements in room acoustics. *Acustica* 81 (1995) 344-355.
- [5] C.C.J.M. Hak, R.H.C. Wenmaekers, J.P.M. Hak, L.C.J. van Luxemburg, A.C. Gade: Sound strength calibration methods. *Proceedings of ICA*, Sydney, 2010.
- [6] P. Lehmann: Über die Ermittlung raumakustischer Kriterien und deren Zusammenhang mit subjektiven Beurteilungen der Hörsamkeit. PhD Thesis, Technical University Berlin, 1976.
- [7] M. Barron, L. J. Lee: Energy relations in concert auditoriums. I. *J. Acoust. Soc. Am.* 84 (1988) 618-628.
- [8] C.C.J.M. Hak, R.H.C. Wenmaekers, L.C.J. van Luxemburg: Measuring room impulse responses: impact of the decay range on derived room acoustic parameters. *Acta. Acust. united Ac.* 98 (2012) 907-915.
- [9] P. Virjonen, J. Keränen, V. Hongisto: Determination of acoustical conditions in open-plan offices: proposal for new measurement method and target values. *Acta. Acust. united Ac.* 95 (2009) 279-290.
- [10] J.S. Bradley, R. Reich, S.G. Norcross: A just noticeable difference in C50 for speech. *Applied Acoustics* 58 (1999), 99-108.
- [11] R.H.C. Wenmaekers, N.H.A.M. van Hout, L.C.J. van Luxemburg, C.C.J.M. Hak: The effect of room acoustics on the measured speech privacy in two typical European open plan offices. *Proceedings of Internoise*, Ottawa, 2009.
- [12] A.C. Gade: Investigations of musicians' room acoustic conditions in concert halls. I. Methods and laboratory experiments. *Acustica* 69 (1989) 193-203.
- [13] A.C. Gade: Investigations of musicians' room acoustic conditions in concert halls. II. Field experiments and synthesis of results. *Acustica* 69 (1989) 249-261.
- [14] R.H.C. Wenmaekers, C.C.J.M. Hak, L.C.J. van Luxemburg: On measurements of stage acoustic parameters - time interval limits and various source-receiver distances. *Acta. Acust. united Ac.* 98 (2012) 776-789.
- [15] K. Ueno, K. Kato, K. Kawai: Effect of room acoustics on musicians' performance. Part I: Experimental investigation with a conceptual model. *Acta. Acust. united Ac.* 96 (2010) 505-515.
- [16] R.H.C. Wenmaekers, C.C.J.M. Hak, H.P.J.C. de Vos: Binaural sound exposure by the direct sound of the own musical instrument. *Proceedings of ISRA 2013*, Toronto (2013).
- [17] A.C. Gade: Practical aspects of room acoustic measurements on orchestra platforms. *Proceedings of 14th ICA*, Beijing, 1992.
- [18] A. C. Gade: Estimated value obtained from a personal conversation with A.C. Gade in 2009.
- [19] ISO 3740:2000: Acoustics - Determination of sound power levels of noise sources - Guidelines for the use of basic standards, International Organisation for Standardisation (ISO). Geneva (CH), 2000.
- [20] M. Aretz, R. Orłowski: Sound strength and reverberation time in small concert halls. *Appl. Acoustics* 70 (2009) 1099-1110.
- [21] J.J. Dammerud: Stage acoustics for orchestras in concert halls. PhD Thesis, University of Bath, 2009.
- [22] ISO 3741:1999: Acoustics - Determination of sound power levels of noise sources using sound pressure - Precision methods for reverberation rooms. International Organisation for Standardisation (ISO), Geneva (CH), 1999.

- [23] A.C. Gade: Acoustical survey of eleven European concert halls. Report No.44, Technical University of Denmark, 1989.
- [24] L. Beranek: Subjective rank-orderings and acoustical measurements for fifty-eight concert halls. *Acta. Acust. united Ac.* 89 (2003) 494-508.
- [25] J.J. Dammerud, M. Barron: Attenuation of direct sound and the contributions of early reflections within symphony orchestras. *J. Acoust. Soc. Am.* 128 (2010) 1755-1765.
- [26] R. San Martín, M. Arana, J. Machín, A. Arregui: Impulse source versus dodecahedral loudspeaker for measuring parameters derived from the impulse response in room acoustics. *J. Acoust. Soc. Am.* 134 (2013) 275-284.
- [27] M. Vorländer, G. Raabe: Calibration of reference sound sources. *Acustica* 81 (1995) 247-263.
- [28] ISO 9614-3: 2002: Acoustics - Determination of sound power levels of noise sources using sound Intensity - Part:3 Precision method for measurement by scanning. International Organisation for Standardisation (ISO), Geneva (CH), 2002.
- [29] I.B. Witew and G.K. Behler: Uncertainties in measurement of single number parameters in room acoustics. *Proc. Forum Acusticum*, Budapest, Hungary, 2005.
- [30] T.W. Leishman, S. Rollins, H.M. Smith: An experimental evaluation of regular polyhedron loudspeakers as omnidirectional sources of sound. *J. Acoust. Soc. Am.* 120 (2006) 1411-1422.
- [31] R. San Martín, I.B. Witew, M. Arana, M. Vorländer: Influence of the source orientation on the measurement of acoustic parameters. *Acta. Acust. united Ac.* 93 (2007) 387-397.
- [32] C.C.J.M. Hak, R.H.C. Wenmaekers, J.P.M. Hak, L.C.J. van Luxemburg: The source directivity of a dodecahedron sound source determined by stepwise rotation. *Proceedings of Forum Acusticum*, Aalborg, 2011.
- [33] F. Martellotta: Optimizing stepwise rotation of dodecahedron sound source to improve the accuracy of room acoustic measures. *J. Acoust. Soc. Am.* 134 (2013) 2037-2048.
- [34] L. Beranek: *Concert halls and opera houses: Music, acoustics, and architecture*. Springer Verlag, New York, 2004.
- [35] J.J. Dammerud: The ST measures without the standard reference level. White paper available online, june 2013, www.stageac.wordpress.com/articles.
- [36] C.A. Gehe: Måling av ST på scenen i konsertsaler. Master Thesis, Norges Teknisk-naturvitenskapelige Universitet NTNU, Norway, 2013.
- [37] L. Beranek, N. Nishihara: Mean-free-paths in concert and chamber music halls and the correct method for calibrating dodecahedral sound sources. *J. Acoust. Soc. Am.* 135 (2014) 223-230.

Chapter 4

Orchestra members on stage

Original title: How orchestra members influence stage acoustic parameters on five different concert hall stages and orchestra pits
Authors: R.H.C. Wenmaekers, C.C.J.M. Hak, M.C.J. Hornikx
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Abstract

Stage acoustic parameters aim to quantify the amount of sound energy reflected by the stage and hall boundaries and the energy decay over time. In this research, the effect of orchestra presence on parameter values is investigated. The orchestra is simulated by dressed mannequins, which have been compared with humans with respect to acoustic properties. Impulse response measurements were performed in a concert hall, a theatre, a rehearsal room and in two orchestra pits. We measured on the empty stage floors, the stage floors with music stands and chairs only, and the floors occupied by the mannequin orchestra. Results show that the direct and reflected sound levels and the energy decay are significantly affected by the orchestra compared to an empty stage or a stage with chairs and stands only. Both the direct sound and early reflected sound levels are reduced by the orchestra with the distance. The late reflected sound level is reduced considerably more than can be expected based on Barron's revised theory. It can be concluded that measurements on a stage without the orchestra being present results in significant differences. A practical method is presented to perform a 'musician friendly' stage acoustic measurement with a real orchestra.

4.1 Introduction

In concert hall and theatre design, it is important to consider the acoustics as perceived by the performers as well as by the audience. Much research in the field of stage acoustics has focused on defining physical parameters to predict musicians' perception of the acoustics. The Early and Late Support parameters ST_{early} and ST_{late} were introduced by Gade [1] and modified to $ST_{early,d}$ and $ST_{late,d}$ by Wenmaekers et al. [2, or [chapter 2](#)]. These ST parameters measure the amount of sound energy arriving early and the sound energy arriving late on stage. Other suggestions for parameters are the Early Decay Time, *EDT*, Early Ensemble Level [1], *EEL*, and the Early Sound Strength [3], or G_{early} . In these other parameters, the direct and early reflected sound energy are included. Extensive information on the various (other) parameters that have been used in stage acoustics can be found in [section 2.2](#) [2].

However, there is still a lack of agreement on what stage designs are preferred by musicians and what physical parameters are of importance [5]. Studies using questionnaires with orchestra members judging concert halls did not consistently reveal a correlation between perceptual attributes and physical parameters [5]. For instance, Dammerud et al. [3] conclude that acoustic parameters that measure early reflected sound on stage have poor subjective relevance while parameters judging late reflected sound can be significant predictors of musicians' preference. One of the various possible causes for a lack of correlation might be related to uncertainties in the physical measurement, for instance when excluding the effect of absorption and scattering by orchestra members on the parameter values [3]. This is the subject of investigation in the current chapter. It is often suggested that including source and receiver directivity might improve the subjective relevance of early reflected sound parameters [3,4,6]. In the current chapter, this topic is considered to be important future research (see [chapter 5](#)) and only omnidirectional parameters are investigated.

Dammerud and Barron (in this chapter denoted D&B) investigated the influence of orchestra members on the direct sound transfer in a hemi-anechoic scale model [7]. Results showed a significant attenuation of the sound in the 0-50 ms time interval for the 1 and 2 kHz octave band, which increases by the distance between source and receiver. In the lower frequency bands up to 500 Hz, no attenuation was observed. These findings confirmed the results of earlier small scale experiments by Krokstad et al. [8] and Skålevik [9]. Chairs, stands and orchestra members do not only cause a direct sound attenuation. D&B also note that the energy in the impulse response smears out over time, and 'within orchestra reflections' can also increase the level of sound in the orchestra [7]. This suggests that the direct sound attenuation by the orchestra members might be compensated by the orchestra reflections to some extent.

The direct sound path through the orchestra is not the only path affected by orchestra members. Further scale model and computer model studies by Dammerud [10] indicate that reflected sound levels in a hall are significantly influenced by the presence of the orchestra on stage. Results are based on different stage enclosures in a generic concert hall shape.

Dammerud concluded that parameters that involve late reflected sound (ST_{late} and G_{late}) and, to some extent, parameters taken at 1 m source to receiver distance (ST_{early}) are moderately reduced by the orchestra. However, parameters that are taken with varying source to receiver distances and involving early reflected sound (G_{early} , G_{7-50}) showed considerable reductions. To the best of our knowledge, no study on real scale has investigated the effect of orchestra members on the sound propagation over a sound path where energy is reflected from room boundaries.

The goal of this chapter is to better understand the effect of orchestra members, when present on stage, on the amount of direct, early and late reflected sound. Measurements have been performed on 5 stages and orchestra pits in unoccupied conditions and with a full scale orchestra. The orchestra has been simulated by using dressed mannequins. In this chapter, we show that the reduction in sound passing directly through the ‘real’ orchestra is considerably higher than measured on scale by D&B [7]. The average reduction in early reflected sound level due to the orchestra members is small at short source to receiver distances (r), but the reduction increases considerably with increasing r . In section 4.2, the measurement methods are explained and the acoustic properties of mannequins are compared to those of humans. In section 4.3, results are presented from the stage acoustic measurements on the stages and in orchestra pits. The results are discussed in section 4.4. In section 4.5, a practical example is presented of a measurement method that can be used to measure a stage occupied by a real orchestra.

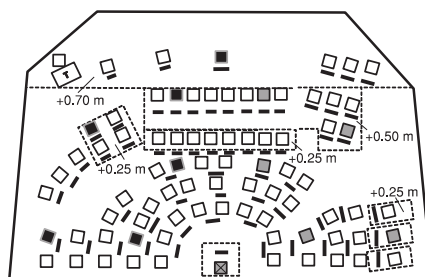
4.2 Methods

In this section, a description of the various halls is given together with source and receiver positions. The measurement setup is described, and the used acoustic parameters are explained. In addition, results from two studies on sound absorption and sound propagation attenuation of the mannequins are presented.

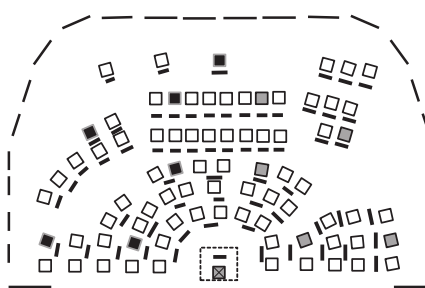
4.2.1 Halls and positions

Acoustic measurements were performed on a concert hall stage, a theatre stage, a rehearsal room stage, and in two orchestra pits. Figure 4.1 shows pictures of the different venues with the mannequin orchestra on stage or in the pit. On the concert hall stage, part of the orchestra members were positioned on risers. In case of the theatre stage, measurements were taken with and without the wooden reflective elements around the orchestra. Figure 4.1 also shows schematic overviews of the positioning of chairs and loudspeaker and microphone positions. For the concert hall and theatre stage, the dimensions of the source and receiver grid are equal to the grid shown below in Figure 4.2. For the orchestra pits and rehearsal room, the source–receiver grid was scaled so it would fit in the available space and positions would correspond to possible seating locations of musicians. Complementary to the positions shown in Figure 4.2, a receiver position at 1 m distance was used at each source position placed behind the source in a line towards the conductor position (10).

Concert Hall stage (CH)



Theatre stage (TH)



Rehearsal room stage (RH)

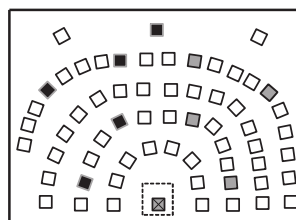
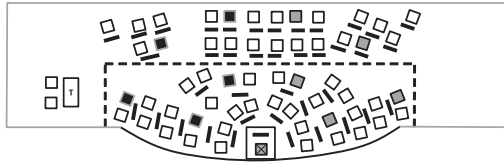


Figure 4.1, part 1. Concert hall stage of Muziekgebouw Eindhoven (CH); Theatre stage of Parktheater Eindhoven (TH); Stage of the wind orchestra rehearsal room MFC Berg aan de Maas (RH). Positions of chairs, stands and mannequins per room, together with a schematic representation of the measurement grid. Filled chairs transducer positions.

Theatre Orchestra Pit (THop)



Opera House Orchestra Pit (OHop)

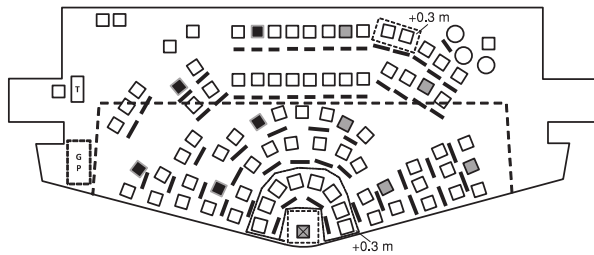


Figure 4.1, part 2. Orchestra pit of theatre Parktheater Eindhoven (THop); Orchestra pit of opera house Nationale Opera and Ballet Amsterdam (OHop). Positions of chairs, stands and mannequins per room, together with a schematic representation of the measurement grid. Filled chairs transducer positions.

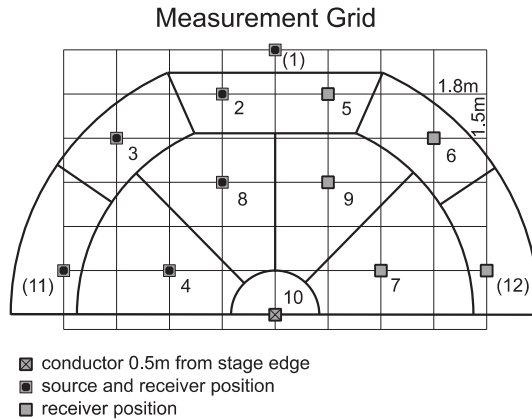


Figure 4.2. Positions of chairs, stands and mannequins per room, together with a schematic representation of the measurement grid. Filled chairs transducer positions.

4.2.2 Measurement conditions

Measurements were performed under three different conditions: (1) empty stage, (2) stage with chairs and stands and (3) stage with chairs, stands and orchestra members (mannequins). On the concert hall stage, theatre stage and in the opera house pit, an 80-piece mannequin orchestra was present. In the theatre pit and wind orchestra rehearsal room, a 60-piece mannequin orchestra was present. Music stands were present in most halls, indicated by the short bold lines in the drawings in Figure 4.1, with the exception of the rehearsal room where no stands were available.

4.2.3 Measurement setup

Impulse responses were measured using an omnidirectional sound source (B&K Type 4292), a single omnidirectional microphone (B&K Type 4189), an Amphion amplifier (AE) and measurement software Dirac 6.0 (B&K Type 7841). The transducer height was 1 m. Although impulse response measurements from multiple rotations of the source would have been preferred to suppress source directivity effects that arise at higher frequencies, this was not possible due to time constraints, and a single measurement for every source-receiver combination was taken. A fixed source aiming point was used perpendicular to the stage edge. Using measurement data from a previous study on sound source calibration [11, or [chapter 3](#)] we determined that, compared to a full rotational average with 72 measurements,

the uncertainty (probability level of 95% using a coverage factor of 2.8) of a single IR measurement is:

- 2.8 dB for the direct sound (0-10 ms), 1.2 dB for the early reflected sound (10-100 ms) and 0.4 dB for the late reflected sound (100-1,000 ms) for the separate octave bands up to 2 kHz.
- 0.8 dB for the direct sound (0-10 ms), 0.3 dB for the early reflected sound (10-100 ms) and 0.1 dB for the late reflected sound (100-1,000 ms) for the average of the octave bands 250 to 2,000 Hz.

Gade recommends to remove objects in a radius of 2 m around the transducers when performing stage acoustic measurements and keep at least 4 m distance from side walls [12]. At first, this is important to be able to accurately window out the direct sound without interference of the early reflected sound, and to always include the earliest wall reflection in the measurement. Later on, Gade recommended to use the free field sound pressure at 1 m distance from the sound source as a reference [12], which is different from the definition in ISO 3382-1 [13]. Following Gade's recommendation, in the current study a sound power measurement was performed in a reverberation room to determine the reference level by $L_{\text{ref}} = L_w - 11$ dB. Even now the reference level is not determined on stage, it would still be necessary to remove objects around transducers and keep a distance from the side wall to include all reflections within the early energy time window. However, during the first set of measurements in the rehearsal room, it was found that removing objects from a 2 m radius around both a source and receiver position resulted in too many orchestra members being removed from the stage. It is clear that this would limit the possible outcome of this study. Therefore, it was decided not to remove any objects during measurements apart from the chairs and/or mannequins on the sound source position and one the microphone position. The uncertainty related to this problem is discussed in section 4.3.2.

4.2.4 Stage acoustic parameters

Four different acoustic parameters were selected for the investigation: an early and late decay parameter, i.e. EDT and T_{20} , and an early and late sound level parameter, i.e. ST_{early} and ST_{late} . The EDT measures the reverberation time over the 0 to -10 dB drop of the Schroeder curve, and T_{20} is determined over the -5 to -25 dB drop. Both EDT and T_{20} should be measured at distances larger than 1 m from the source [1]. While T_{20} ignores the influence of direct sound, the EDT is sensitive to the direct sound in the measurement, which can be strong close to the sound source. $EDTF$, the ratio of EDT for 250 and 500 Hz relative to 1 and 2 kHz, was a promising predictor for the evaluation of *Timbre* [1]. The Early and Late Support parameters ST_{early} and ST_{late} are used as introduced by Gade [1], as well as those extended and modified by Wenmaekers et al., $ST_{\text{early},d}$ and $ST_{\text{late},d}$ [see section 2.4]. In the published paper this section also contains a summary of the main findings from chapter 2, but this would lead to unnecessary repetition here.

4.2.5 Impulse response quality

All measurements presented in the results section had a decay range INR > 40 dB for the separate 250 to 2,000 Hz octave bands. For an accurate calculation of ST_{early} and ST_{late} parameters at 1 m distance (error ≤ 0.2 dB), the decay range INR of the impulse response should be at least 28 and 39 dB respectively [14]. It is assumed that the requirement for G is valid for $ST_{early,d}$ and $ST_{late,d}$ at further distances, which is an INR ≥ 28 dB for an error ≤ 0.1 dB (the higher requirement at 1 m distance is caused by the strong direct sound peak in such a measurement which increases the INR). With the achieved decay range INR > 40 dB, the reverberation time parameters EDT and T_{20} can be determined with errors ≤ 0.2 JND and ≤ 0.7 JND respectively (equivalent to 1% and 4%).

4.2.6 Comparison of sound absorption

Kath [15] has shown that the sound absorption of people is mostly dependent on the clothing. Men wearing a suit absorb sound up to two times more than women wearing a summer dress, and persons in bathing suits absorb very little: a total absorption less than 0.2 up to 2 kHz per person. To investigate the validity of using mannequins instead of humans for the experiments in the various halls, sound absorption measurements were performed. Eight male participants wore normal clothing with long trousers and a thin jacket. Eight female participants wore normal clothing with arms and legs covered. The participants sat on chairs with a foam back and foam seating, which are similar to chairs that are normally used by an orchestra. Mannequins were chosen to be made out of fibre glass (9 kg per mannequin) because they absorb little sound, similar to humans without clothes. For clothing, fleece jumpsuits (3 mm thick, 200 g/m²) with long sleeves were chosen with a hood to simulate hair. The sound absorption measurement was performed in a 90 m³ reverberation room with an 8 m² surface area surrounded by a wooden perimeter following ISO 354 [16]. Even though it is recommended to use a larger room volume and sample size, for the sake of comparing objects this setup was judged sufficient. Two configurations were tested with varying area per seat: 1 and 2 m² per chair; see Figure 4.3. The configuration with 1 m² per chair was tested with male and female participants separately, see Figure 4.4 for an example of the setup. A configuration with 2 m² per chair was tested with a mix of 50% men and 50% women.

Figure 4.5 shows the measurement results expressed as the total sound absorption A per person or chair as a function of frequency. In the third octave bands 400 to 5,000 Hz a significant increase in sound absorption is observed when the chairs are occupied. Compared to real men only, the sound absorption of the mannequins is almost equal while the sound absorption of real women is 18% lower on average for the third octave bands 400 to 5,000 Hz. This can be explained by the fact that the women were smaller than the men and wore thinner/less clothing. Compared to the mixed compositions of men and women, the sound absorption of mannequins wearing jumpsuits is on average 9% higher in this frequency range.

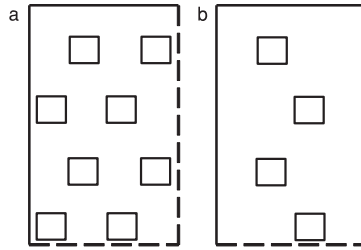


Figure 4.3. Sound absorption measurement setup in the reverberation room: (a) 1.0 m² per chair with 8 chairs; (b) 2.0 m² per chair with 4 chairs.



Figure 4.4. Sound absorption measurement setup in the reverberation room with 1 m² per chair occupied by 8 women.

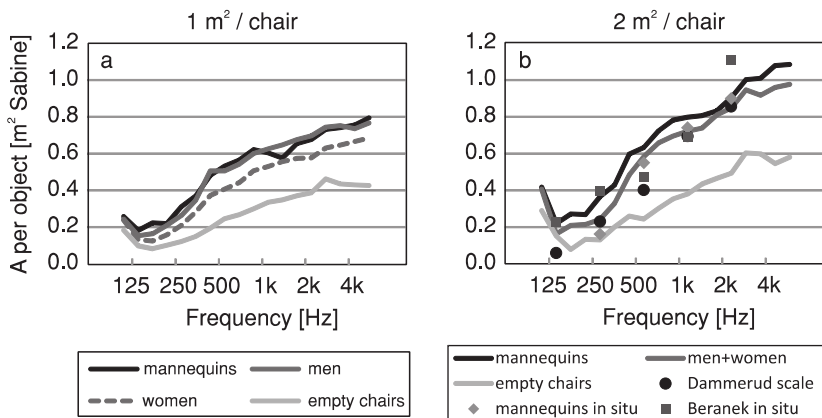


Figure 4.5. The total sound absorption per empty chair, per chair with mannequins and per chair with individuals for two configurations: Left: 1 m² per chair; Right: 2 m² per chair. The right graph shows reference values for sound absorption per orchestra member found in literature [7,17] and in-situ values based on our measurement on the concert hall stage with and without the mannequin orchestra (using Sabine's equation).

Because the jumpsuits turned out to be indispensable for handling and protecting the mannequins during measurements and transport, it was decided not to reduce the clothing for the mannequins that were to simulate female orchestra members. Figure 4.5 also shows reference values for sound absorption per orchestra member found in literature [7,17] and in-situ values based on our measurement in the concert hall. Differences per octave band exist with a maximum of 0.25 and the measured absorption of the dressed mannequins averaged over all octave bands is 0.12 larger than the average value of all references. The latter agrees with our mannequins having a larger sound absorption than real men and women mixed by 0.09. However, the absolute absorption values should be interpreted with care because the volume of the measurement room does not meet the volume of the standards.

4.2.7 Comparison of sound propagation attenuation

In addition to sound absorption, the attenuation of sound passing through a group of mannequins (without music stands) was investigated. A picture of the experimental setup in a sports hall is shown in Figure 4.6 and diagrams of the setup are shown in Figure 4.7. Two configurations of 1 m² and 2 m² per chair were made in a checkerboard pattern. Different types of sound paths have been investigated: front to back (FB), left to right (LR), and diagonal (DIA). While some paths are blocked by chairs, other paths are unobstructed. Impulse response measurements have been performed while rotating the sound source in steps of 72 degrees. To study the attenuation of the direct sound by humans or mannequins on chairs, the difference in G_{0-20} is calculated between the empty floor condition and the condition with occupied chairs.

Figure 4.8 shows the sound attenuation for chairs occupied by humans (men and women) and mannequins. For a dense group of 1 m² per occupied chair, little variation is observed between different sound paths and attenuation tends to increase with frequency with a dip at 1,000 Hz. This dip is caused by the highest constructive interference for 1,000 Hz by the floor reflection at 6 m distance in the empty floor case, which is attenuated by the objects. Sound paths with an open line of sight show similar attenuation to blocked lines of sight.

For the more spacious grouping of 2 m² per occupied chair, larger differences are shown between sound paths. Surprisingly, only the open sound path going from front to back stands out for being different, even showing a slight increase in sound level, while the open sound path going from left to right is more similar to other sound paths.

In general, more attenuation occurs when the chairs are occupied by mannequins instead of humans. For the 1 m² and 2 m² per chair configuration, the average absolute difference for the 500 to 4,000 Hz octave bands is 1.1 dB and 0.9 dB respectively. Possibly, this difference is caused by the reference group of humans consisting of a mix of 50% men and 50% women having lower sound absorption properties than the mannequins as reported above.



Figure 4.6. The attenuation of direct sound through a group of humans and a group of mannequins has been measured in a sports hall for validation. The picture shows the setup with the group of mannequins.

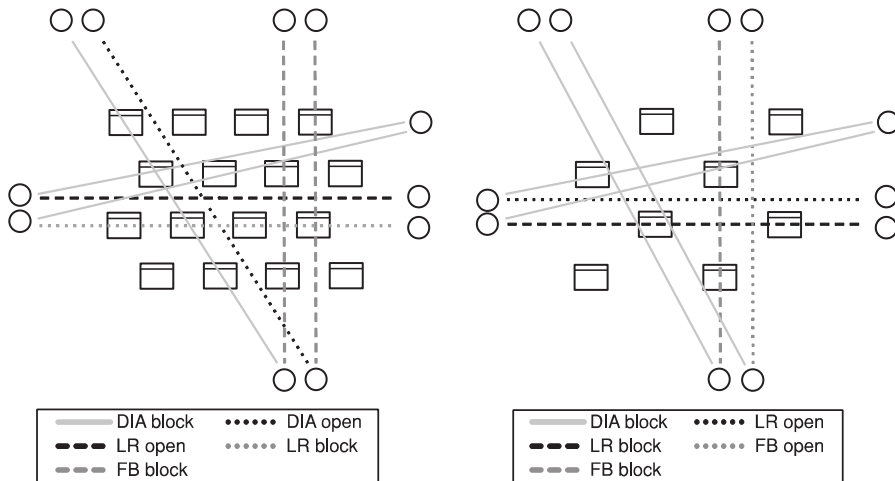


Figure 4.7. Measurement positions and location of seats for the sound attenuation validation study. Source receiver distance 6-7 m. Left: 1 m² per chair. Right: 2 m² per chair.

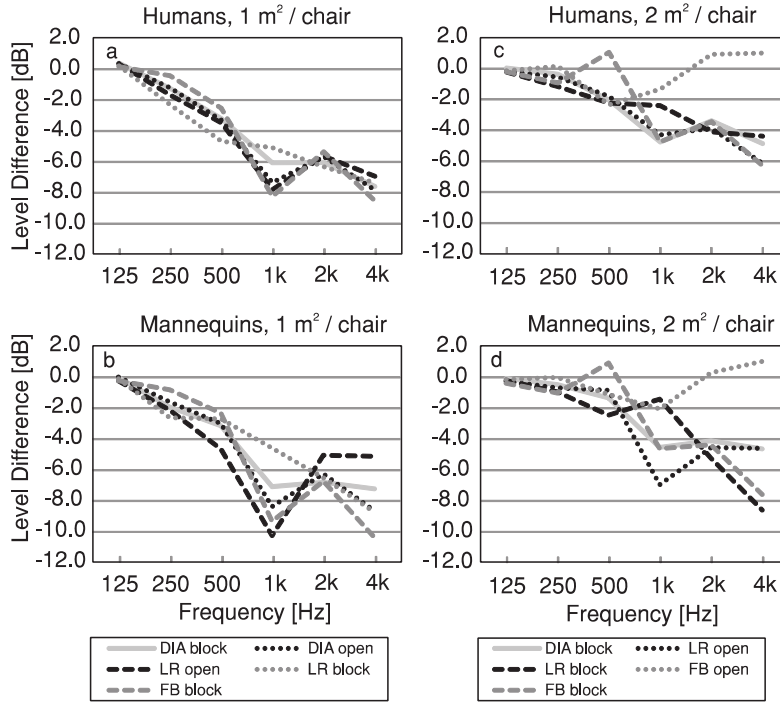


Figure 4.8. The level difference between the direct sound measured with an empty floor and the direct sound measured (a) 1 m² per human on chair (b) 1 m² per mannequin on chair (c) 2 m² per human on chair and (d) 2 m² per mannequin on chair.

4.2.8 Comparison conclusion

We can conclude that the mannequins with fleece jumpsuits are a sufficiently accurate substitute for real male humans: their sound absorption is almost equal to real men and the attenuation of sound, averaged over various directions over 6 m distance through a group of mannequins deviates equal to or less than 1.1 dB from real men. The use of these mannequins instead of real humans can be seen as the scenario with most attenuation: in case of a men-only orchestra. If a mannequin orchestra is needed to be reproduced for experiments, it is likely that similar rigid plastic mannequins with orchestra-like clothing will have proper absorption and sound propagation attenuation. If normal clothing can be used instead of jumpsuits, it would be possible to discriminate between men and women having different absorption properties. However, it is recommended to check the sound absorption in a reverberation room.

4.3 Results stage experiments

In this section, results are presented obtained from the experiments performed on the stages and in the orchestra pits. First, the direct sound attenuation will be presented. After that, the results for the stage acoustic parameters are presented. Note that the discussion of the results is found in section 4.4.

4.3.1 Direct sound results

It can be expected that most sound energy from the direct sound and floor reflection is captured within a time window of 10 ms after the actual direct sound arrival. Additionally, D&B [7] showed that ‘within orchestra reflections’ are visible in the impulse response at least up to 25 ms. The direct sound is denoted L_d , the direct sound including floor reflection is denoted L_{df} and the direct sound including floor reflection passing through the orchestra is denoted L_{dfo} . Impulse responses were measured on the full scale theatre stage without the wooden reflective elements as visible in Figure 4.1, but with stage curtains hung at 2.8 m distance from the nearest transducer position. Due to the curtain reflections, we can only investigate L_{dfo} for the time window 0-16 ms. The original impulse responses measured in the scale model study by D&B [18] have been re-analysed for these time windows. The difference between the sound level within the 0-25 ms and the 0-16 ms interval of only 0.3+/-0.2 dB found is negligible. This means that our data can be compared to the data presented in D&B [18].

Figure 4.9 shows our results for $L_{df}-L_d$ and $L_{dfo}-L_d$ for each measured combination of source and receiver on the theatre stage as a function of r for the octave bands 125 Hz to 4,000 Hz. The direct sound L_d is derived from the sound power measurement in the reverberation room by $L_d=L_w-10\lg(4\pi r^2)$. A line is presented in the graphs as well that is based on the analytical empty stage model for $L_{df}-L_d$ as used by D&B [7].

The variation for $L_{dfo}-L_d$ can partly be explained by the ‘object density’ in the direct sound path. The lowest sound reduction is found in the areas that are more open, typically positions on the same side at the edge of the orchestra (Edge). An average amount of reduction is found for sound paths that run diagonally or sideways (Diag/Side). The largest sound reduction is found for paths going from the front to back in the middle section (Front-Back). The results for $L_{dfo}-L_d$ are presented in Figure 4.10 together with one trend line that corresponds with our data and one based on D&B [7], Table II. Our sound paths along the edge are plotted together with their trend line B, which was a line without obstructions. The diagonal and sideways sound paths are compared to their trend line A and the front-back sound paths to their trend line C.

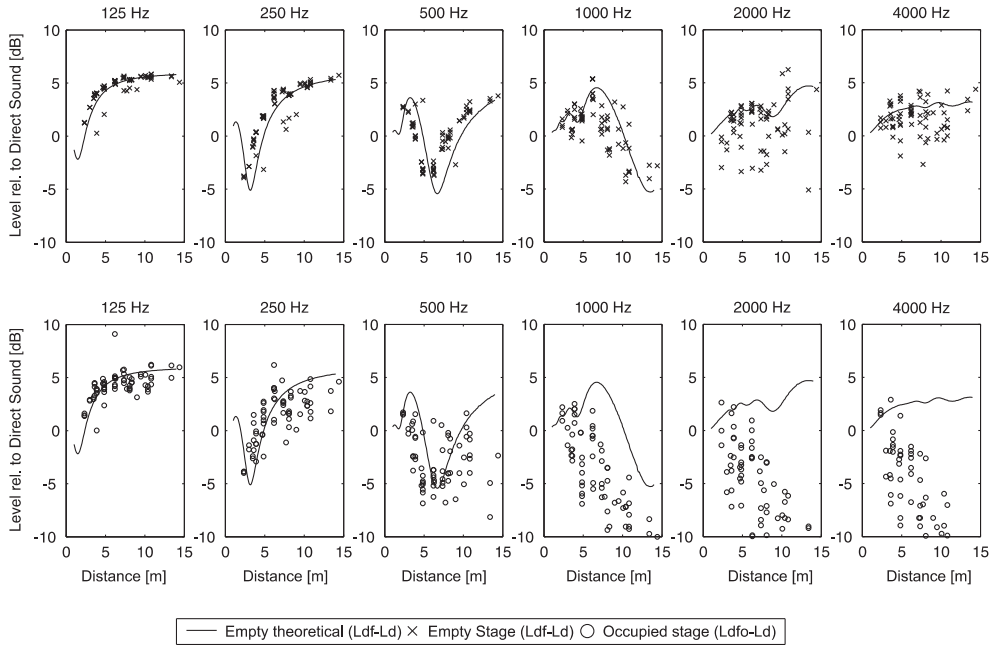


Figure 4.9. Individual data points for $L_{df}-L_d$ in upper graphs and $L_{dfo}-L_d$ in lower graphs as a function of r . The lines represent theoretical values for $L_{df}-L_d$ in all graphs.

4.3.2 Early reflected sound threshold

To obtain threshold values, at above which energy from room reflections can be considered to be stronger than the energy reflected from objects on stage, we investigated $ST_{early,d}$ for the theatre stage without wooden reflecting elements. At short source receiver distances of 1-3 m, the maximum value for $ST_{early,d}$ is -16 dB, with a 2 to 3 dB increase in the occupied condition compared to the empty stage. For 3-5 m distance, $ST_{early,d}$ was hardly affected by objects' reflections with a maximum value $ST_{early,d} = -18$ dB. Beyond 5 m, $ST_{early,d}$ was reduced by the objects on stage and a value of $ST_{early,d} = -20$ dB was not exceeded.

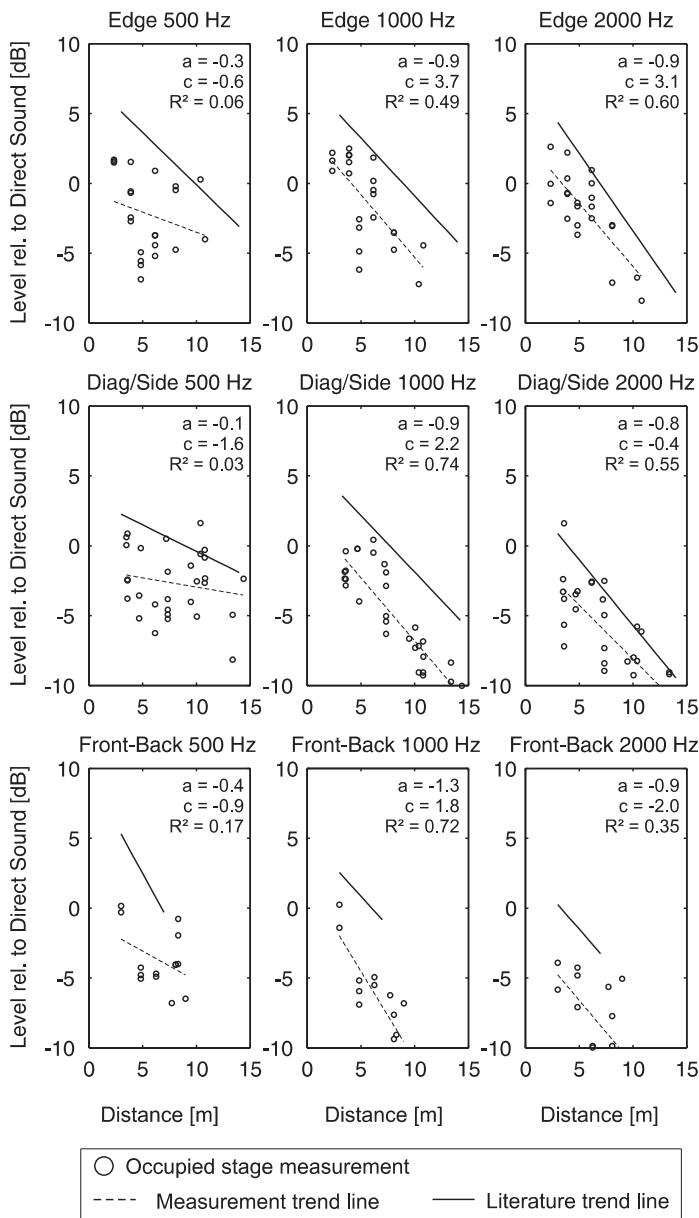


Figure 4.10. Individual data points for $L_{dfo}-L_d$ as a function of source-receiver distance for the octave bands 500, 1,000 and 2,000 Hz. Edge = sound path along the edge of the orchestra, Diag/Side = sound path diagonal or sideways, Front-Back = sound path from front to back in the middle of the orchestra. Dashed lines are linear trend lines for $L_{dfo}-L_d$ based on the measured data. The coefficients a and c for the trend line with equation $a \times r + c$ are given for each line together with the correlation coefficient R^2 . The solid lines are trend lines as suggested by Dammerud and Barron [7].

4.3.3 Early sound results

In Figure 4.11, absolute results are presented for $ST_{early,d}$ and EDT as a function of r . For the stages, every graph shows results per source position. For each orchestra pit a grouping is used which has shown to be typical for orchestra pits [19, see [section 9.2](#)]: both positions in the open part (O-O); just one of both positions in the open or covered part (O-C); and both positions in the covered part (C-C).

Besides EDT per distance, the position average T_{20} is presented as lines because it showed few variation over distance. To investigate the overall effect of the chairs, stands and mannequins on $ST_{early,d}$, Figure 4.12 shows the differences in $ST_{early,d}$ for all measurements on all stages and pits as a function of r .

For the measurements at 1 m distance, averaged over the 5 stages, the $ST_{early,d}$ was higher than the original ST_{early} by 2 dB in the empty state and by 3 dB in the occupied state (similar to earlier findings as mentioned in [section 4.2.4](#)). For ST_{early} at 1 m distance the absolute difference between the empty and occupied state is 1.6+/-1.4 dB and for $ST_{early,d}$ at 1 m distance 0.9+/-0.7 dB.

4.3.4 Late sound results

The distance averaged $ST_{late,d}$ did hardly differ from ST_{late} measured at 1 m distance by more than 1 dB and $ST_{late,d}$ is chosen for further analyses. Figure 4.13 shows the average $ST_{late,d}$ per stage for all positions and per orchestra pit for the three groups O-O, O-C and C-C. One might expect that the late arriving energy can be described as being part of the diffuse sound field. Following Barron's revised theory [20], ST_{late} or $ST_{late,d}$ can be predicted by the room volume V and reverberation time T :

$$ST_{late} = 10 \lg \left[\frac{31200T}{V} e^{-(13.82 \times 0.103/T)} \right] - 20 \quad (3)$$

Alternatively, it can be written as:

$$ST_{late} = 10 \lg(312T/V) - 6.2/T . \quad (4)$$

For the empty stages with a well-defined room volume (CH and RH), $ST_{late,d}$ is predicted using Eq. (3) with 0.1 and 0.3 dB deviation from the measurements for the CH and the RH respectively, which suggests that the revised theory holds for the empty stage. For the concert hall (CH) and rehearsal room (RH), we investigated whether the difference in $ST_{late,d}$ could be explained by a measured 5% reduction in T . Based on the reduction in T , the addition of the chairs, stands and mannequins would only lead to -0.3 dB and -0.5 dB difference, while the measured difference is -1.7 dB and -4.1 dB.

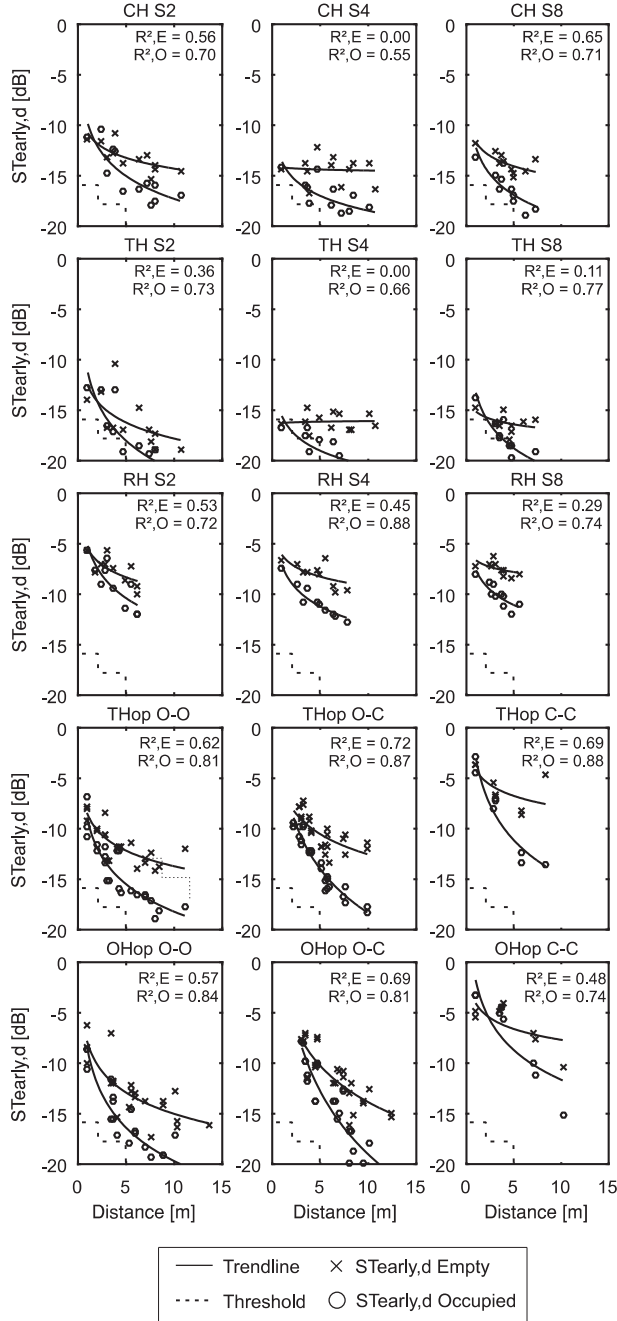


Figure 4.11, part 1. Individual data points for $ST_{early,d}$ as a function of source receiver distance r for the empty and occupied stages and pits (250-2,000 Hz average). The dashed line in the $ST_{early,d}$ figures, including the x-axis above 5 m, represents the threshold. S2, S4, S8 = source position 2, 4 and 8, O-O = open-open, O-C = open-covered, C-C = covered-covered. R^2,E and R^2,O = correlation coefficient, empty and occupied.

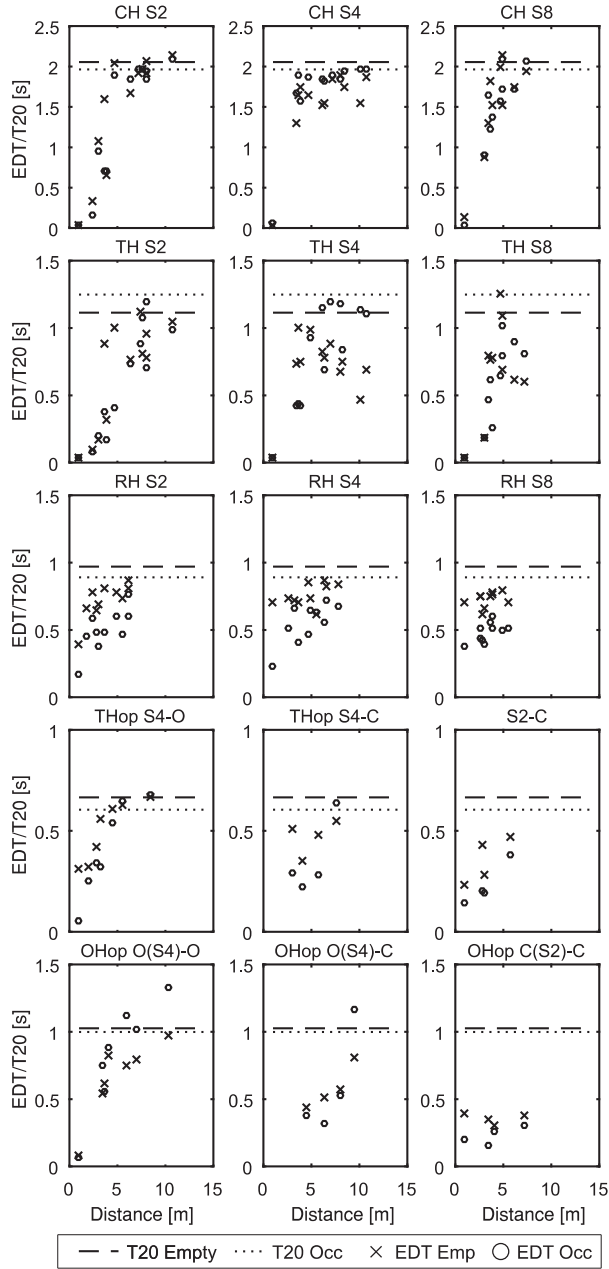


Figure 4.11, part 2. Individual data points for EDT as a function of source receiver distance r for the empty and occupied stages and pits (250-2,000 Hz average). The dashed lines in the EDT figures represent the distance averaged reverberation time T_{20} , for both unoccupied and occupied stage. S2, S4, S8 = source position 2, 4 and 8, O-O = open-open, O-C = open-covered, C-C = covered-covered. R^2_E and R^2_O = correlation coefficient, empty and occupied.

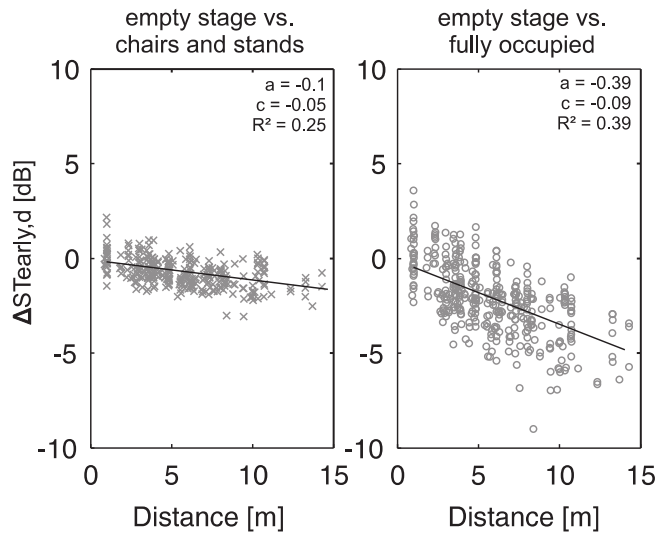


Figure 4.12. Difference in $ST_{early,d}$ as a function of source-receiver distance for all measurements of all stages. Left: empty floor minus floor with chairs and stands. Right: empty floor minus floor occupied with mannequins.

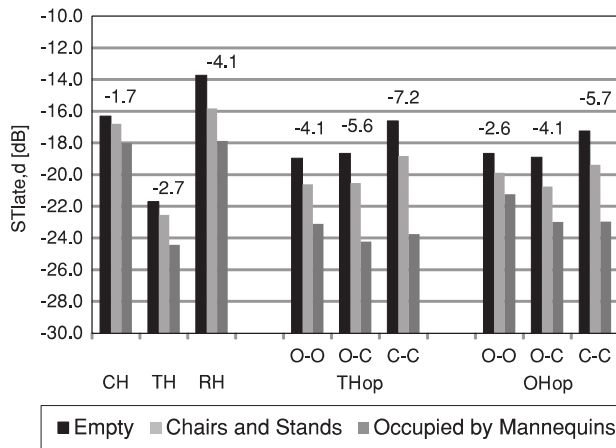


Figure 4.13. Average $ST_{late,d}$ per stage and pit for the empty condition, the condition with chairs and stands, and the condition with chairs, stands and mannequins. Numbers in the graph represent the difference between empty floor and fully occupied conditions.

4.4 Discussion

4.4.1 Direct sound

The interference of the direct sound with the floor reflection on the empty stage floor corresponds well with theory for the lower frequencies up until the 500 Hz octave band, see Figure 4.9, although the theoretical curve is shifted by approximately 1 m. Small changes of 0.1 m in geometrical properties used in the analytical model can cause such a shift. At 1,000 Hz and above, measured values deviate from theory resulting in both higher and lower values for $L_{df}-L_d$. Most of the variation can be explained by the directivity of the sound source at these frequencies (uncertainty of ± 2.8 dB, see section 4.2.3).

The direct sound level is reduced at most positions when the stage is fully occupied. This consistency shows that the reduction in sound level by absorption and scattering is larger than the increase of sound level by ‘within orchestra reflections’. In the frequency bands 500, 1,000 and 2,000 Hz, $L_{df}-L_d$ is consistently lower in our study compared to D&B with a constant shift over all distances varying from -3 to -6 dB. This shift brings the reduction of sound close to the source nearer to 0 dB compared to the trend lines by D&B. The uncertainty introduced by the directivity of the sound source can explain the variation, but cannot explain our values being consistently lower. Consistent deviations could be caused by uncertainty in the sound power of the source. However, D&B [7] explain in their paper that, after calibration, measured G_{df} corresponded well with theory and our calibration procedure has an uncertainty of only 0.8 dB [21]. Also, the measured sound absorption of our mannequins and their scale model orchestra are in the same order of magnitude at frequencies 1,000 and 2,000 Hz, see Figure 4.5. Considering all these factors, we must conclude that the attenuation of sound passing through our full scale orchestra is indeed 3 to 6 dB more at each distance than through the scaled orchestra by D&B.

4.4.2 Early sound

The early reflected sound level, measured by $ST_{early,d}$, tends to decrease as a function of r for most source positions or source-receiver groups in both unoccupied and occupied conditions, see Figure 4.11. In case the source is near a stage wall, the path length of the reflection from that wall increases with the distance to the receiver r and as a result, the sound energy decreases with r for that particular reflection. Because this wall reflection is dominant in the parameter, the total early reflected sound energy will decrease with r . The attenuation by the orchestra also increases when reflection path lengths through the orchestra increase. For source positions close to walls, and for relatively small stages and pits, this explains the general tendency that the early reflected sound is reduced by the orchestra increasingly as a function of r .

When the sound source is located in the middle of the orchestra, $ST_{early,d}$ on the unoccupied concert hall and theatre stages at S4 show no correlation with r . Surprisingly, $ST_{early,d}$ does correlate reasonably well with the r for S4 and S8 when the orchestra is included (for CH, R^2

is 0.55 and 0.71, and for TH, R^2 is 0.66 and 0.77). At these positions and stages, $ST_{early,d}$ is close to the measurement threshold. In that case, measured $ST_{early,d}$ increases by the presence of the orchestra at short distance due to ‘within orchestra reflections’ and decreases at further distance. This is likely the reason why the overall measured $ST_{early,d}$ also decreases as a function of r on these stages at source positions in the middle of the orchestra.

The EDT shows an increase as a function of distance for both unoccupied and occupied conditions, but the trend is not as clear as for $ST_{early,d}$. On some stages/pits, EDT approaches T_{20} at larger distances and EDT is 0 s at 1 m distance because of the strong direct sound component. On the large stages (CH and TH), the difference in EDT between empty and occupied state can both be positive and negative. In the smaller ‘rooms’ (RH stage, THop and the covered part of OHop), EDT is systematically reduced by the presence of the orchestra similar to its reduced T_{20} . Possibly, this reduction is consistent because both the early and late sound field contains of a larger number of reflections in a small space and direct sound is less dominant. In contrary, EDT increases when the orchestra is present in the open area of the large orchestra pit (OHop), which might indicate that the direct sound is more reduced than the early reflected sound is. It could be expected that the EDT frequency balance, $EDTF$, would increase when the mannequin orchestra is on stage because of the largest attenuation in the high frequencies. However, no significant stage-average differences for $EDTF$ were found and the average standard deviation of differences in $EDTF$ per stage was high ($\sigma = 0.4$ [-]).

The difference between $ST_{early,d}$ in occupied and unoccupied state is presented in Figure 4.12 for all single measurements. The average distance dependent reduction of $ST_{early,d}$ due to the orchestra as a function of r is not significantly different per stage or pit. A linear trend can be observed with a low correlation and the largest variance in the condition with the orchestra present. No significant difference was found for source receiver combinations along the edge of the stage and combinations in the middle of the stage. The variation might therefore be attributed to the irregular pattern of the objects combined with the complexity of multiple reflection paths. The reduction in dB/m by chairs, stands and mannequins is 4 times larger than the reduction by chairs and stands only. This shows that unoccupied chairs with stands cannot simulate a stage being occupied by a full orchestra; one might as well measure on an empty stage which is more convenient.

Because the variation of $ST_{early,d}$ and EDT over distance is different per stage, judging the amount of early reflected sound on stage based on solely 1 m distance measurements can limit the judgement of the stage’s performance. However, as Dammerud [10] concluded, an advantage of measuring close to the source is that ST parameters are least influenced by the presence of the orchestra. When comparing the original ST_{early} to $ST_{early,d}$ measured at 1 m, it seems that $ST_{early,d}$ is least influenced. When considering a JND of 2 dB for the ST parameters, as estimated by Gade, the influence of the orchestra could be neglected when measuring $ST_{early,d}$ at 1 m distance but could be relevant for measurements at larger distances. For original ST_{early} , the influence of the orchestra is just above 2 dB. For EDT , most of the measured results were influenced by the orchestra above the JND of 5% and an occupied

stage might be necessary for valid measurement. Changes in *EDT* due to occupation are not consistent and more research is necessary to determine the reliability of this parameter for the use on stage.

4.4.3 Late sound

Most stages show a reduction in the average reverberation time T_{20} of 5-10% after putting the orchestra on stage as can be expected [17,22]. One exception is the theatre stage (TH) at which the T_{20} increased. Possibly, placing the mannequins in the more reverberant side stage (behind the stage curtains) for storage had a larger effect on the average T_{20} of these coupled spaces.

The late reflected sound, measured by $ST_{late,d}$, is reduced by the presence of the orchestra much more than Barron's revised theory predicts, see Figure 4.13. It seems that part of the sound energy is absorbed and screened in propagation paths through the orchestra. The amount of reduction seems to depend on the 'virtual stage volume' and its connection to the hall. The smaller and less exposed the stage to the hall is, the more the late reflected sound is reduced by the orchestra. The following can be observed:

- when comparing the concert hall and theatre: while both stages were occupied by the same orchestra setup, having equal absorption properties, $ST_{late,d}$ is influenced more by the orchestra in the non-reflective theatre stage compared to the reflective concert hall stage;
- when comparing the two orchestra pits: the open/uncovered area is smaller in THop compared to OHop, resulting in a larger reduction in $ST_{late,d}$ when occupied by the orchestra. Also, the orchestra has a larger impact on $ST_{late,d}$ for positions under the stage overhang compared to positions in the open area.

4.5 Practical example

For most survey measurements, using a mannequin orchestra would be too time-consuming and expensive.⁸ Therefore, we investigated how a measurement can be performed with a real (professional) orchestra in the concert hall (CH) in a short time span with a signal that is comfortable for musicians. Three sources and six receivers were used at the locations as suggested by Gade [1,5], see Figure 4.14. To avoid cables, six recorders (TASCAM DR-40) were used with six omnidirectional microphones (DPA 4060). The sound source (B&K Type 4292) and amplifier (AE Amphion) with an external battery pack were portable. Another recorder played back a 5.46 s MLS signal repeated twice preceded by a voice countdown. The sound level of the MLS signal was set just below 90 dB(A) at the nearest seating distance.

The measurement took place at the start of a rehearsal and all transducers were in place before the entrance of the orchestra members. The musicians were informed in advance via

⁸ A 10 minute measurement would likely cost less than €1,000 if the orchestra needs to be paid. This is much less expensive than the €5,000 for rental and transport of the mannequins with clothing.

an online website and video. After playing the signal once on stage, all musicians accepted to participate without hearing protection. With the help of four persons, it took 6 minutes to play the signal at all 3 source positions. The musicians were asked to hold their instruments in still playing position while being silent for 20 s during each measurement. The actual rehearsal started four minutes after removing all equipment while marking the transducer positions with tape. During a short rehearsal break the distances between tapes were measured to obtain the exact r .

In the laboratory, a sound power calibration was performed in a reverberation room. Impulse responses were obtained using software Dirac 6.0 (B&K Type 7841) which corrects for device clock speed errors (necessary for an asynchronous measurement). The impulse response decay ranges (INR's) at one m exceeded 38 dB with an average of 46 dB and at larger distances the INR exceeded 29 dB with an average of 34 dB. Based on these INRs, it is expected that $ST_{early,d}$ and $ST_{late,d}$ are calculated within 0.2 dB error, which is lower than the uncertainty for not rotating the source. The measurement results of $ST_{early,d}$ are presented in Figure 4.15 together with earlier measurements with the mannequin orchestra. The trend lines of both occupied measurements only differ by 0.5 dB on average and a similar variation along the trend line can be observed. The average result for $ST_{late,d}$ of both measurements differed only by 0.2 dB. Based on these results, we can conclude that 'musician friendly' stage acoustic measurements can be done within 10 minutes time.

4.6 Conclusion and recommendation

The direct, early and late sound on stage, measured by $ST_{early,d}$, $ST_{late,d}$, EDT and T_{20} at various distances, is significantly influenced by the orchestra. Original ST_{early} (with a 20-100 ms time window) varies just above the estimated JND of 2 dB while $ST_{early,d}$ at 1 m distance (with a 10-100 ms time window) varies just below the JND. The late reflected sound level in most stages and pits, measured by ST_{late} or $ST_{late,d}$, is reduced above the estimated JND and considerably more than can be expected based on Barron's revised theory.

It is clear that significantly different values of stage acoustic parameters are found on occupied stages compared to unoccupied stages, even at 1 m distance. Chairs and stands on stage, as suggested in ISO 3382-1, do not substitute a real orchestra; one might as well measure on an empty stage. For extensive research a mannequin orchestra has shown to be an accurate but time-consuming method. Survey measurements can be done with a real orchestra within 10 minutes with results showing reasonable agreement with those by the mannequins. Further research should focus on:

- Correlation between perceptual attributes and parameter values when being measured on occupied stages and/or distances above 1 m.
- The importance of directivity of the musical instruments and musicians' ears (see [chapter 5](#)).
- Methods to correct parameters measured on empty stages or model values for occupied stages.

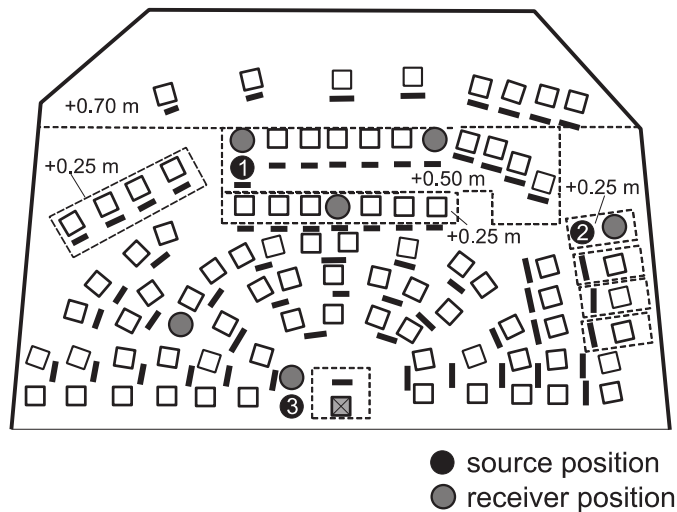


Figure 4.14. Source and receiver positions used for the measurements with the real orchestra, as suggested by Gade [1,5]. Note that source position 2 would preferably have been located closer to the centre of the stage.

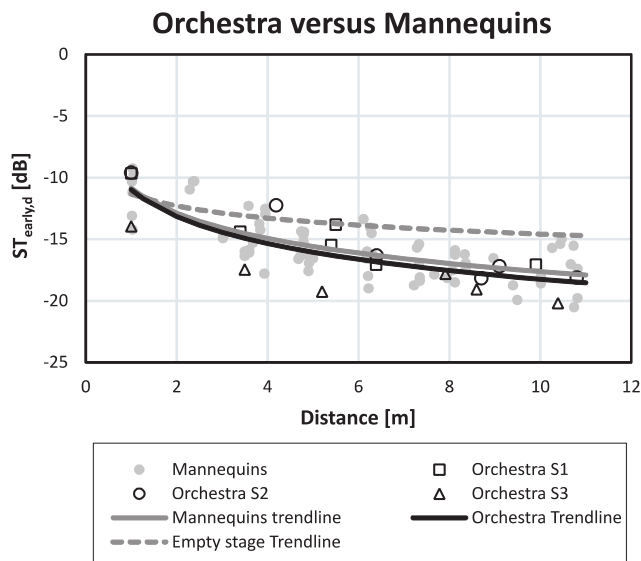


Figure 4.15. Individual data points for $ST_{early,d}$ as a function of source receiver distance for the same concert hall stage occupied by the real orchestra and the mannequin orchestra. The lines represent the logarithmic trend line of all measurements with the real orchestra and the mannequin orchestra.

4.7 References

- [1] A.C. Gade: Investigations of musicians' room acoustic conditions in concert halls. *Acustica* 69 (1989) 193-203 and 249-261.
- [2] R.H.C. Wenmaekers, C.C.J.M. Hak, L.C.J. van Luxemburg: On measurements of stage acoustic parameters - time interval limits and various source-receiver distances. *Acta. Acust. united Ac.* 98 (2012) 776-789.
- [3] J.J. Dammerud, M. Barron, E. Kahle: Objective Assessment of Acoustic Conditions for Symphony Orchestras. *Building Acoustics* 18 (2011) 207-219.
- [4] J. Meyer, E.C. Blassoni de Serra. On the masking effect of instrumental musicians, (original title: Zum Verdeckungseffekt bei Instrumentalmusikern). *Acustica* 46 (1980) 130-140.
- [5] A. C. Gade: Subjective and objective measures of relevance for the description of acoustics conditions on orchestra stages. International Symposium on Room Acoustics, Toronto, Canada 2013.
- [6] A. Guthrie: Stage acoustics for musicians: a multidimensional approach using 3d ambisonic technology. PhD Thesis, Rensselaer Polytechnic Institute, Troy, New York, 2014.
- [7] J.J. Dammerud, M. Barron: Attenuation of direct sound and the contributions of early reflections within symphony orchestras. *J. Acoust. Soc. Am.* 128 (2010) 1755-1765.
- [8] A. Krokstad, J. Vindspoll, R. Sæther: Orkesterpodium, samspill og solo Orchestra platform, ensemble and solo. The Laboratory of Acoustics, Technical University of Trondheim, Norway, 1980.
- [9] M. Skålevik: Sound transmission between musicians in a symphony orchestra on a concert hall stage. Proceedings of the 19th ICA, Madrid 2007.
- [10] J.J. Dammerud: Stage acoustics for orchestras in concert halls. PhD Thesis, University of Bath, 2009.
- [11] R.H.C. Wenmaekers, C.C.J.M. Hak: The sound power as a reference for Sound Strength (G), Speech Level (L) and Support (ST): uncertainty of laboratory and in-situ calibration. *Acta Acust. Acust.* 101 (2015) 892-907.
- [12] A.C. Gade: Practical aspects of room acoustic measurements on orchestra platforms. Proceedings of 14th ICA, Beijing, 1992.
- [13] ISO 3382-1:2009: Acoustics – Measurement of room acoustic parameters – Part 1: Performance spaces. International Organisation for Standardisation (ISO), Geneva 2009.
- [14] C.C.J.M. Hak, R.H.C. Wenmaekers, L.C.J. van Luxemburg: Measuring room impulse responses: Impact of the decay range on derived room acoustic parameters. *Acta. Acust. united Ac.* 98 (2012) 907-915.
- [15] U. Kath: Der Einfluß der Bekleidung auf die Schallabsorption von Einzelpersonen. *Acustica* 17 (1966) 234-237.
- [16] ISO 354:2003: Acoustics -- Measurement of sound absorption in a reverberation room. International Organisation for Standardisation (ISO), Geneva (2003).
- [17] L. Beranek, T. Hidaka: Sound absorption in concert halls by seats, occupied and unoccupied, and by the hall's interior surfaces. *J. Acoust. Soc. Am.* 104 (1998) 3169-3177.
- [18] The measurement data was obtained from the website <https://stageac.wordpress.com/>, date 4 June 2015.
- [19] R.H.C. Wenmaekers, C.C.J.M. Hak: Early and late support measured over various distances: The covered versus open part of the orchestra pit. *Building Acoustics* 20 (2013) 313-322.

- [20] M. Barron: Theory and measurement of early, late and total sound levels in rooms. J. Acoust. Soc. Am. 137 (2015) 3087-3098.
- [21] M. Vorländer, G. Raabe: Calibration of reference sound sources. Acustica 81 (1995) 247-263.
- [22] H.S. Jong, J.Y. Jeon: Acoustic characterization of on-stage performers in performing spaces. Appl. Acoust. 114 (2016) 159-170.

Chapter 5

Source and receiver directivity

Original title: Sensitivity of stage acoustic parameters to source and receiver directivity: measurements on three stages and in two orchestra pits
Authors: R.H.C. Wenmaekers, C.C.J.M. Hak., M.C.J. Hornikx, A.G. Kohlrausch
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Abstract

Stage acoustic parameters are commonly determined in concert halls using omnidirectional transducers, but might be more valid when using directional transducers. In this chapter, the sensitivity of stage acoustic parameters to source and receiver directivity has been investigated by measurements on three stages and in two orchestra pits. A single loudspeaker was used with a directivity similar to a trumpet, aimed in 12 evenly spread directions. As a directional receiver, a head and torso simulator (HATS) was used with its viewing direction towards the conductor position. Measurements were also taken with (nearly) omnidirectional transducers. i.e. a dodecahedron loudspeaker and a single microphone. The investigated stage acoustic parameters measuring reverberation time and reflected sound levels were sensitive to the directivity of the measurement transducers. The parameters dealing with early sound, EDT and $ST_{early,d}$, are more sensitive than the parameters dealing with late sound, T_{20} and $ST_{late,d}$. When comparing results measured with a head and torso simulator to results measured with an omnidirectional microphone, the EDT tends to be lower and the $ST_{early,d}$ higher for the ear directed towards the sound source. The results of measurement using the directional source show that EDT and T_{20} have lowest values and $ST_{early,d}$ and $ST_{late,d}$ highest values, when the sound source is directed towards the closest surfaces that cause a first order reflection towards the receiver. Further research is necessary to determine whether the differences in parameter values would lead to noticeable differences.

5.1 Introduction

Omnidirectional directivity of the sound source and receiver is used in the definition of many physical parameters in room acoustics describing energy decays, energy ratios, energy levels [1] or modulation reductions [2]. Exceptions are those that describe apparent source width and listener envelopment [1] which use directional transducers such as a figure-of-8 microphone or head and torso simulator (HATS) to simulate the directional hearing of humans. Omnidirectional parameters are also used to describe acoustic conditions for the orchestra on stages of performance spaces [3, 4]. Directional properties of the musical instrument and the listener are not taken into account by these measures while possibly being relevant for the musicians on stage, as indicated by research recently presented by Dammerud [5] and Guthrie [6]. In the current chapter, it is investigated whether measurements with directional transducers, that simulate the characteristics of an instrument and a listener, lead to significantly different stage acoustic parameters than when using omnidirectional transducers.

5.1.1 Background

For measuring room acoustic parameters, the dodecahedron loudspeaker is the most commonly used sound source, containing 12 drivers equally spread over a full sphere to approach omnidirectional directivity. Polyhedron loudspeakers are only omnidirectional below their ‘cut-off frequency’, which is determined by the radius of the sound source [7]. Leishman et al. found that the dodecahedral shape is the optimal choice for being the best possible combination of reasonable omnidirectional radiation and sufficient sound power. Due to the spherical radiation pattern room acoustic measures vary for different source rotations [8]. The error due to the directive behaviour at frequencies above 1,000 Hz can partially be compensated by averaging over multiple source rotations [9, or [section 3.4](#)]. Another type of practical omnidirectional source uses two drivers, face-to-face, with loudspeaker cabinets shaped as a cone [10]. This type of source is more omnidirectional but less powerful, which makes it less suitable for measurements in large rooms like performance spaces.

Orchestral instruments are well known for their directive projection of sound: brass instruments have a strong directivity in the direction of the bell while string instruments project sound perpendicular to their radiating body. For woodwind instruments, a complex directivity is found caused by the combined radiation of the open holes and the end of the tube. The directional properties of musical instruments have been investigated by Meyer [11] and by Pätynen and Lokki [12]. It is clear that their directivity is complex, depending on the tone played and varying among playing styles and different instruments of the same type [13]. Besides, the position of the acoustic centre of the source may vary per note played and per frequency, as shown by Shabtai and Vorländer [14]. As a result, it is difficult to simulate directivities of the various musical instruments by (single) loudspeakers. A possible solution to this problem is to perform measurements with individual loudspeakers in a spherical array

and synthesizing any directivity from these responses using spherical harmonic decomposition, as investigated by Pasqual [15] and Pollow et al. [16]. However, a high spherical discretisation of loudspeaker positions is necessary to be able to simulate complex directivities at high frequencies, which is especially important for auralisation purposes. The large number of measurements per position makes the measurement procedure time consuming: a single measurement position can cost hours even when using fully automated measurement equipment. In the scarce amount of time available in concert halls when performing measurements, this procedure is not yet feasible for extensive room acoustic investigations in performance spaces.

Microphones can be omnidirectional up to high frequencies because of their limited size. By contrast, humans make use of directional hearing and the human auditory system contains a set of complex mechanisms utilizing this directivity, for instance to localise sounds [17]. The directivity of the (outer) ears in the head are captured in the Head Related Transfer Function (HRTF). Individualised HRTFs are a necessary input for binaural reproduction of virtual acoustic environments for optimal immersion and localisation [18]. A HATS with a typical head and ear shape can be used to take into account the directivity of the ears in measurements [1]. In room acoustics, the HATS is used when measuring the Inter-Aural Cross Correlation. Besides, the HATS has been used to study the impact of a playback room acoustics on recorded room acoustics [19]. The HATS is not commonly used for measuring the typical ‘omnidirectional’ room acoustic parameters, but measurements did show a clear directivity of the hall for various room acoustic parameters measured with a HATS [20].

In room acoustic research, the directive properties of the instrument and the listener are considered in auralisations of measured sound fields [21] or simulated sound fields [22]. Also in stage acoustics, such auralisations have been used for listening tests to investigate the musician’s response to variations in stage acoustic conditions under laboratory conditions [23, 6, 24]. Some research has involved analyses of the direction of the arrival of sound using spherical microphone arrays in rooms [25] and concert halls [26]. Such methods can be used to analyse the spatial composition of the impulse response, showing direction and arrival times of reflections and the degree of isotropy of the late reverberant field. Pätynen et al. [26] claim that such 2D/3D visualizations “reveal considerably more information in an intuitive manner” than the ISO 3382-1 parameters. However, the results from their 2D/3D visualisations are difficult to quantify unambiguously and, as a result, findings cannot be easily related to perceptual aspects or compared to measurement results from others. Pätynen et al. [26] conclude that their methods “can be potentially used as a basis for novel objective indicators of the quality of concert hall acoustics”. The 3D visualisation tools seem promising, but their value in room evaluation and design is still to be established.

In stage acoustics, some researchers investigated the influence of directivity on ISO 3382-1 parameters. Schrärer and Weinzierl [24] calculated binaural room impulse responses using a geometrical acoustic model with the directivity of musical instruments included. The performance of musicians was investigated while playing their instrument in a real time auralisation of modelled concert halls. Measured performance attributes, e.g. *Tempo* and

Dynamic Strength, were compared to calculated results of room acoustic parameters, e.g. reverberation time T and late support ST_{late} . The acoustic parameters were derived from the impulse responses including the source directivity. Even though results from this study showed great differences in preferences by musicians, parameters that measure reverberation, early and late energy seemed to influence some performance attributes. However, it is uncertain how the choice of using actual source directivity has affected the parameter results and no validation of the method is discussed in the paper. While Schrärer and Weinzierl [24] took into account source directivity, Guthrie [6] investigated the direction of the sound field at the receiver position by using a spherical microphone array on stage. Acoustic parameters were measured for various sound field directions (front/rear, up/down and left/right/sides). Evaluations from listening tests by musicians indicated a correlation between the perception of *Spatial Support* and the Top/Sides ratio of the measured stage acoustic parameters values (where Top refers to the sound coming from above). This correlation agrees with the finding by Dammerud [5] that musicians might prefer a certain ratio of stage height and stage width.

5.1.2 Current research

The perceptual studies mentioned indicate that stage acoustic parameters that include directivity might be valuable to judge ensemble conditions on stage, even though correlations with perceptual attributes or musicians' evaluations are often weak. The goal of the current research is to explore the differences in 'omnidirectional' and 'directional' stage acoustic parameter values using measurements performed on three stages and two orchestra pits. In section 5.2, the methodology is further explained. In section 5.3, results are presented and discussed. This chapter ends with a conclusion.

5.2 Method

Impulse responses (IRs) have been measured on various stages using a loudspeaker with 12 aiming directions and a dodecahedron using all loudspeakers simultaneously. The single directional loudspeaker has a directivity similar to that of a trumpet. So, the directional measurements simulate the hypothetical case where a trumpet-like sound source is aimed in various directions. As a recording device, both an omnidirectional microphone and a HATS have been used at many different positions on stage. From the measured impulse responses, the variation in stage acoustic parameters in all these possible transfer paths is investigated.

5.2.1 Measurement setup

Impulse responses were measured using different setups. First, an omnidirectional sound source (B&K Type 4292) was combined with a single omnidirectional microphone (B&K Type 4189). In this chapter, the term omnidirectional is abbreviated by 'omni'. Secondly, a dodecahedral loudspeaker with 12 individually connectable loudspeakers in separate compartments was used, denoted the 'directional source' (made by ITA Aachen [27]), together with a head and torso simulator (HATS, B&K Type 4128C). The 12 separate

binaural IRs are pressure-based summed to simulate a third type of setup: an omni sound source combined with the HATS. 1 monaural and 12 binaural impulse responses were measured per combination of source and receiver position with a 2.73 s exponential sweep signal.

The trumpet has one of the most extreme directivities among musical instruments and it is expected that stage acoustic parameters would be most influenced by a source with a trumpet directivity. By coincidence, a single loudspeaker of the directional source has a directivity pattern similar to a trumpet. Its directivity was measured in an anechoic room and compared to the directivity of a trumpet [12]. Figure 5.1 shows that there is a good resemblance for the 250, 1,000 and 4,000 Hz octave bands.

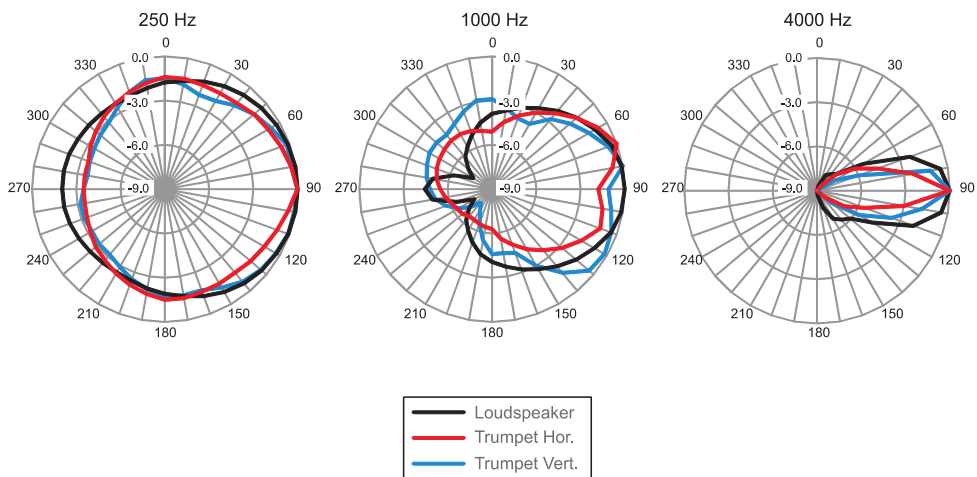


Figure 5.1. The directivity of a single loudspeaker of the directive source compared to a trumpet for the octave bands 250, 1,000 and 4,000 Hz. The trumpet directivity is shown for the horizontal and vertical plane which are slightly different. For the loudspeaker, these are identical.

A system calibration of each measurement setup was performed in a reverberation room using the same system settings. This system calibration is used to derive the reference level for sound strength G at 10 m and stage support ST at 1 m, in this case based on a relative sound power. Because the sound power is measured in diffuse field conditions, it is possible to compare the parameter results for the two different source and receiver setups.

5.2.2 Measurement conditions

Measurements took place on stages of a concert hall (CH), a theatre (TH) and a rehearsal room (RH), and in two different orchestra pits in a theatre (THop) and an opera house (OHop). The stages were occupied by an orchestra of dressed mannequins. A source and receiver grid was used that was adapted to each stage size. For the concert hall, the

measurement grid of 6 source and 12 receiver positions is shown in Figure 5.2. The transducers were placed 1 metre above the floor. The sound source was positioned with the same loudspeaker pointing at right-angles to the stage front for each position. The HATS was oriented with its viewing direction towards the conductor position. All 72 combinations of source and receiver position as shown in the measurement grid (Figure 5.2) have been measured on the CH and TH stages. A scaled grid without positions 11 and 12 was used for the RH stage. For the orchestra pits THop and OHop, position 1 was not used and the grid was scaled to fit the pits' dimensions. The measurement conditions and the mannequin orchestra are explained in detail in [section 4.2](#) [28]. For every source and receiver combination, the 12 separate loudspeakers of the directive source were measured using the HATS as a receiver and a single measurement was taken with the omni source and microphone. In total, 293 omni impulse responses and 3,516 binaural impulse responses were measured. No measurements at 1 metre distance were taken because it was expected that, due to its size, the directional source is not a realistic replacement for a trumpet when the source-receiver distance is small.

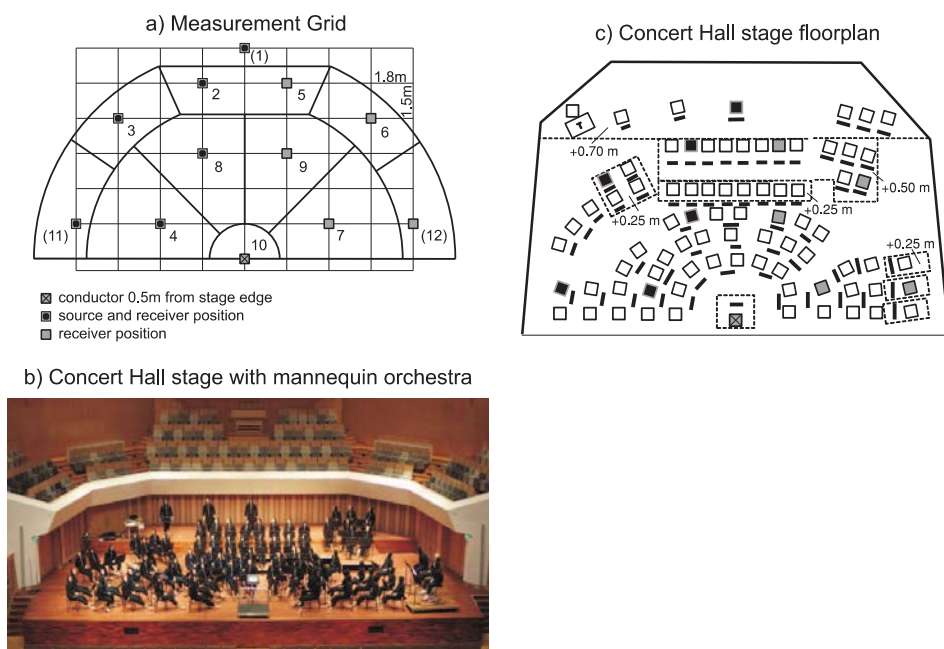


Figure 5.2. a: schematic source and receiver grid; b: orchestra setup on the concert hall stage; c: schematic orchestra setup.

5.2.3 Acoustic parameters

Four different stage acoustic parameters were used for the investigation that are measured with various source and receiver distance: an early and late decay parameter, i.e. EDT and T_{20} , and an early and late sound level parameter, i.e. $ST_{early,d}$ and $ST_{late,d}$. The T_{20} measures the reverberation time over the -5 to -25 dB drop of the backwards integrated energy curve, and EDT is determined over the 0 to -10 dB drop [1]. The EDT is sensitive to the direct sound in the measurement, which can be strong when measuring close to the sound source on stage, while T_{20} ignores the influence of direct sound. The $ST_{early,d}$ and $ST_{late,d}$ based on work by Gade [3] modified in [29, see chapter 2], measure the early and late reflected sound level on stage at various distances relative to the direct sound at 1 m distance. More background information on these parameters can be found in [section 2.4](#) [29] and [section 4.2.4](#) [28]. The $ST_{early,d}$ excludes the direct sound from the measurement because earlier research has shown that parameters that do include direct sound, e.g. G_{early} or EEL , varied considerably less among different stages because the direct sound is too dominant [29, or [section 2.6](#)]. In most cases, the acoustic parameters EDT , T_{20} , $ST_{early,d}$ and $ST_{late,d}$ are directly calculated from the measured impulse responses using software Dirac 6.0. Apart from the ‘directional only’ and the ‘omni only’ measurement setup, the third setup is calculated by pressure summing the impulse responses of the 12 loudspeakers of the directional source.

5.2.4 Measurement uncertainty

95% of all measurements with all transducers presented in the results section had a decay range or impulse response to noise ratio $INR > 40$ dB for the separate 250 to 4,000 Hz octave bands which is sufficient to determine the decay parameters within a 5% error and energy parameters within a 0.2 dB error [30]. The decay range of the 125 Hz octave band was considerable lower for the single loudspeaker IR’s and is therefore excluded from the research.

The directional source contains 12 separate loudspeakers. The sound power of each driver was measured in the reverberation room. It was found that, for the separate octave bands 250-2,000 Hz, the deviation in sound power level of a single loudspeaker compared to the average of all loudspeakers is < 0.5 dB. For the 4,000 Hz octave band, the maximum deviation is approximately 1.0 dB. These deviations are in the same order of magnitude as the uncertainty of a sound power measurement of 0.8 dB [31]. Therefore, in this work, the sound power of each loudspeaker is treated to be equal.

5.3 Results and discussion

Using the measured impulse responses, the impact of source and receiver directivity on the acoustic parameters is evaluated. In the first subsection, the difference between the omni microphone and the HATS is presented for a sound source with omni directivity. In the second subsection, the directional source will be compared to an omni source with the HATS as a receiver.

5.3.1 Omni source with omni and HATS receiver

Figure 5.3a and 5.3b show measured results of EDT and $ST_{early,d}$ respectively for the concert hall (CH) and the opera house orchestra pit (OHop). Source position 3 is chosen as an example, which is located under the stage overhang in case of the orchestra pit. On every receiver position, the middle value of three represents the omni microphone, flanked by the left and right ear of the HATS on that same position.

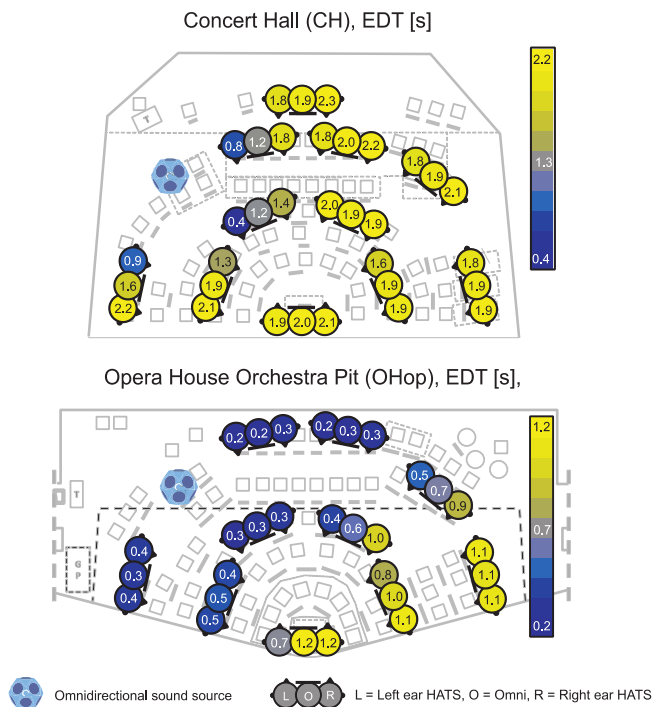


Figure 5.3a. Comparison of acoustic parameter EDT for the omni microphone (omni) and the HATS (left ear and right ear) using an omni sound source. Source position 3 and all receiver position on the concert hall stage (CH) and in the opera house orchestra pit (OHop). Average results for the 250 to 2,000 Hz. For OHop, the dashed line indicates the edge of the stage overhang. Note that the colours are scaled differently for CH and OHop.

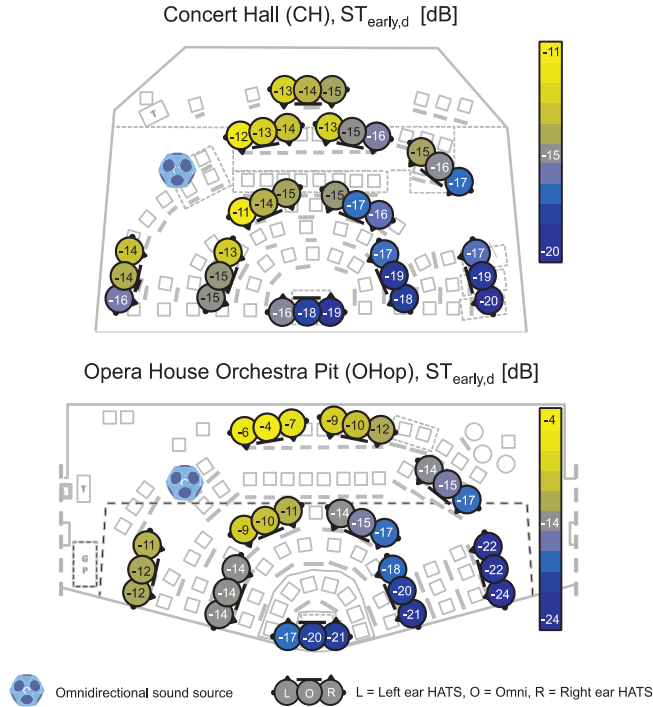


Figure 5.3b. Comparison of acoustic parameter $ST_{early,d}$ for the omni microphone (omni) and the HATS (left ear and right ear) using an omni sound source. Source position 3 and all receiver position on the concert hall stage (CH) and in the opera house orchestra pit (OHop). Average results for the 250 to 2,000 Hz. For OHop, the dashed line indicates the edge of the stage overhang. Note that the colours are scaled differently for CH and OHop.

In general, it can be observed that the measured values are distance dependent. In the case of EDT , values are lower when the receiver is closer to the sound source. At further distances values approach the reverberation time T_{20} (For CH, $T_{20} = 2.0$ s and for OHop, $T_{20} = 1.2$ s for the same frequencies). In CH, the EDT is close to T_{20} for most positions, while in OHop the EDT is close to T_{20} only at the furthest positions and EDT is approaching 0 s near the source. Such low values suggest that the ‘room decay’ starts almost 10 dB lower than the direct and very early sound. In case of the $ST_{early,d}$, values are higher when the receiver is closer to the sound source. The decay over distance is stronger in OHop compared to CH due to the stage overhang [32]. Focussing on the difference between the omni microphone and the two ears of the HATS the following can be observed. EDT is often lower, and $ST_{early,d}$ is often higher, when the ear is directed towards the sound source compared to the omni microphone parameter value. The opposite occurs for the ear directed away from the source: EDT is often higher, and $ST_{early,d}$ often lower for that ear compared to the omni microphone. Even a moderate rotation of the ear towards or away from the source gives this effect.

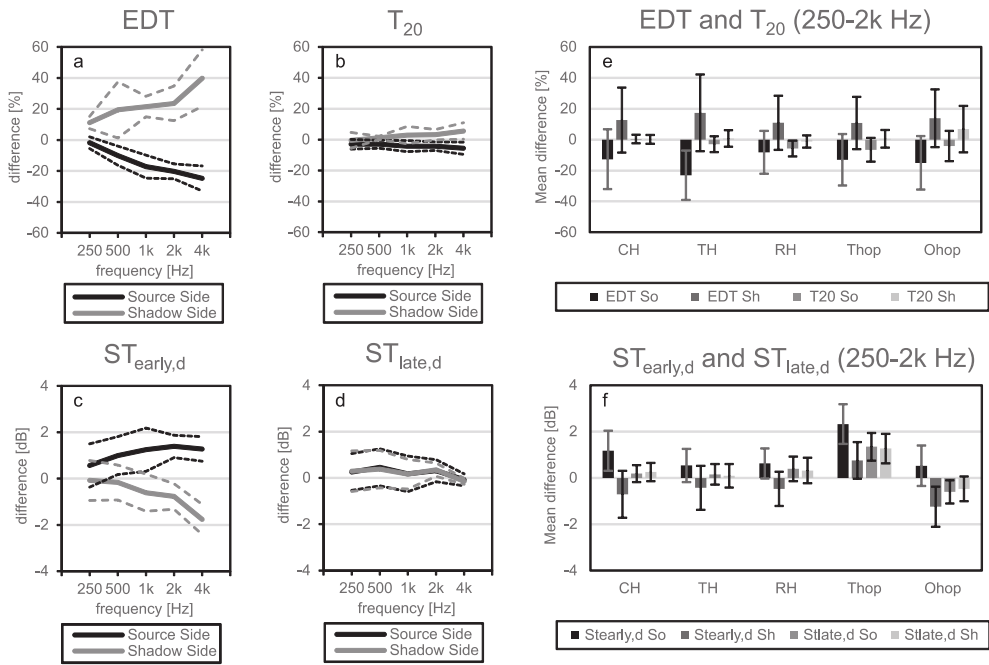


Figure 5.4. Difference in acoustic parameters EDT , T_{20} , $ST_{early,d}$ and $ST_{late,d}$ for an omni microphone (omni) and a HATS (Source side ear, denoted So, and Shadow side ear, denoted Sh) using an omni sound source. a-d: average results as a function of frequency for all source and receiver positions for all stages and pits (solid lines), with standard deviation over the five stage/pit averaged results (dashed lines). e-f: average results for the 250 to 2,000 Hz average for all source and receiver positions per stage and pit (bars), with standard deviation over the microphone positions per stage/pit (error bars). Per stage the order of bars is the same as given in the legend.

To investigate the difference in results between HATS and omni microphone for all 5 stages, the average deviation between results from using the HATS and the omni microphone is computed. For each source and receiver combination, it is determined which ear is directed towards the sound source (denoted ‘Source Side’ or ‘So’) and directed away from the sound source (denoted ‘Shadow Side’ or ‘Sh’). Figure 5.4 shows the average results for each acoustic parameter as a function of frequency and for a broadband average (250-2,000 Hz, as suggested by Gade [3]). The frequency dependent curves show the deviation averaged for all stages/pits with the standard deviation over the 5 stages/pits averages. The deviation increases with frequency, which can be expected based on the increasing directivity of the ear with frequency. A difference between the ‘Source Side’ ear and ‘Shadow Side’ ear is found for the ‘early’ parameters EDT and $ST_{early,d}$. The average single number ratings per stage/pit also show a lower EDT and higher $ST_{early,d}$ at the source side ear and vice versa for the shadow side ear in most cases with a difference of 1 to 2 standard deviations. It is clear

that the early parameters are more sensitive to receiver directivity than the late parameters. The average deviation is up to 40% for EDT and 2 dB for $ST_{early,d}$ for the separate octave bands and $\leq 20\%$ for EDT and ≤ 1 dB for $ST_{early,d}$ for the single number ratings. For the ‘late’ parameters T_{20} and $ST_{late,d}$, the deviation is $\leq 5\%$ and ≤ 0.5 dB respectively for the octave bands averaged over all stages/pits and for the single number averages per stage/pit.

One exception is the theatre orchestra pit (THop), where both ears show a positive deviation for $ST_{early,d}$ and $ST_{late,d}$, and the deviation of average $ST_{late,d}$ is above 1 dB. This effect is not observed in the decay parameters EDT and T_{20} . A possible explanation could be a variation in the output of one of the sound sources or in the sensitivity of the microphones, due to the measurements per location being done spread over 6 months and the sound power calibration was done after this period. To double check this exception, 5 measurement positions were repeated in TH and THop on a single day using one omni sound source, an omni microphone and the HATS. This time, the results for TH and THop were similar, in contrast to the data presented in Figure 5.4, which indicated that an unknown error must have been introduced during the original THop measurement.

All measurement data consistently show the trend that EDT is lower and $ST_{early,d}$ higher when the ear is directed towards the sound source compared to the other ear. For all stages and pits, the omni microphone value is in between the value measured at the two ears. A likely explanation for EDT being affected by the directivity of the ears in such a way is that the orientation of the source side ear results in a higher direct/reflected ratio causing a lower EDT . However, the relative increase in direct sound cannot explain $ST_{early,d}$ being higher at the source side ear because the direct sound is not included in the parameter definition. The explanation can be found in the typical geometry of a stage with a reflective back wall, two side walls and without a ‘front wall’, combined with the fact that the orchestra members direct their backs towards these walls by looking at the conductor. As a result, the 1st order reflections which influence $ST_{early,d}$ most always arrive at the source side ear first, resulting in a higher $ST_{early,d}$ compared to an omni microphone. The geometry of the orchestra pit, being sunken underneath the stage, is different from a stage as it includes a front wall behind the conductor. In this case, 1st order reflections can also arrive first at the ear on the shadow side relative to the sound source. Still, measured $ST_{early,d}$ tends to be higher at the source side ear in the orchestra pits. Likely, more reflected sound is coming from the back because multiple reflections occur in the covered part under the frontstage overhang. The parameters based on late sound, T_{20} and $ST_{late,d}$, are hardly affected by receiver directivity indicating that the sound field is diffuse in the late part of the impulse response.

5.3.2 Directional source with HATS receiver

A sound source with a directivity similar to a trumpet has been used to investigate the sensitivity of the acoustic parameters to directivity in combination with the HATS as a receiver. The variation in parameter values for the 12 different directions of the loudspeakers are compared to the results from the pressure summated IR’s (simulating an omnidirectional

source) as a reference. For the sake of simplification, the parameter values are averaged for the two ears of the HATS (although in retrospect an omni microphone would have been more appropriate for this comparison).

Figure 5.5 shows the maximum and minimum deviation of the parameter values for the directive source relative to the results from the simulated omni source. The presentation of results is similar to that of Figure 5.4. Again, the ‘early’ parameters EDT and $ST_{early,d}$ show a larger sensitivity to directivity than the ‘late’ parameters T_{20} and $ST_{late,d}$ and all parameters show an increase in deviation with frequency. The minima and maxima are quite symmetrical along the x-axis. For the separate octave bands, the average deviation over the stages and pits is $\leq 80\%$ for EDT , $\leq 20\%$ for T_{20} , ≤ 4 dB for $ST_{early,d}$ and ≤ 2 dB for $ST_{late,d}$. The average deviation in single number ratings (250-2,000 Hz) for the various stages and pits is up to 40% for EDT , 15% for T_{20} , 2 dB for $ST_{early,d}$ and 1 dB for $ST_{late,d}$. The deviations due to source directivity are larger than those found in the comparison of the HATS’ ears to the omni microphone in section 5.3.1. This is because in case of the directive versus omni source comparison, maximum and minimum deviations of 12 angles are presented, while in case of the HATS versus omni receiver only two angles relative to the source are used that have a varying direction. In contrast to the results for the HATS versus omni receiver results, the parameters describing late sound, T_{20} and $ST_{late,d}$, also show a small dependency on directivity of the sound source.

Even though the stages and pits have different geometries and contain different materials, the average influence of the source directivity on the parameters seems similar in each room. The question is whether the directions at which these minimum and maximum values occur are also equal for the stages/pits. For four source receiver combinations the variation in $ST_{early,d}$ over the 12 possible source direction has been investigated for each stage/pit (similar trends were found for EDT with opposite sign). Figure 5.6 explains how the 12 loudspeakers, equally distributed over a dodecahedron shape, can be visualised in a schematic way seen from the top of the sphere. The measurement results for $ST_{early,d}$ for the high frequencies average (2 and 4 kHz) are presented in Figure 5.7 using this visualisation. Positive values indicate more early reflected sound energy at the ears in case the sound source is directed towards that side than the omnidirectional source. Different trends are found for the different source-receiver combinations and for the different stages and pits. As can be expected, the highest $ST_{early,d}$ values are found when the source is directed towards the nearest reflective surface(s) causing a first order reflection. In case of the concert hall and theatre, these surfaces are the back and side walls. In the rehearsal room, directing the source towards the low ceiling yields the highest values in three out of four positions. On the contrary, in the orchestra pits, the least sound is reflected from above when source and receiver are in the open area (S4R8 and S4R7). In all stages/pits, values are generally lower when the sound source is directed towards the conductor or stage edge, which would be the common aiming direction for a trumpet. The upwards directions are extreme for a trumpet player, but instead are representative for other brass instruments such as the tuba.

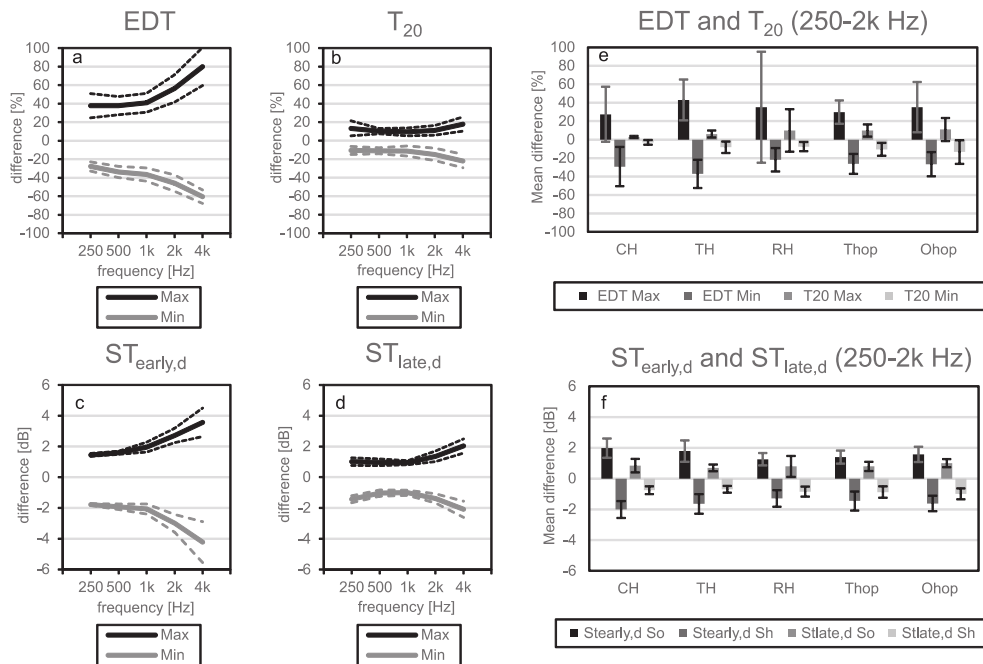


Figure 5.5. Difference in acoustic parameters EDT , T_{20} , $ST_{early,d}$ and $ST_{late,d}$ for a HATS (average results of two ear) using an directional sound source (12 aiming directions). a-d: position averaged maximum and minimum results of 12 loudspeakers as a function of frequency for all source and receiver positions for all stages and pits (solid lines), with standard deviation over the stage/pit averaged maximum and minimum results (dashed lines). e-f: position averaged maximum and minimum results for the 250 to 2,000 Hz average for all source and receiver positions per stage and pit (bars), with standard deviation over the stage/pit results (error bars). Per stage the order of bars is the same as given in the legend.

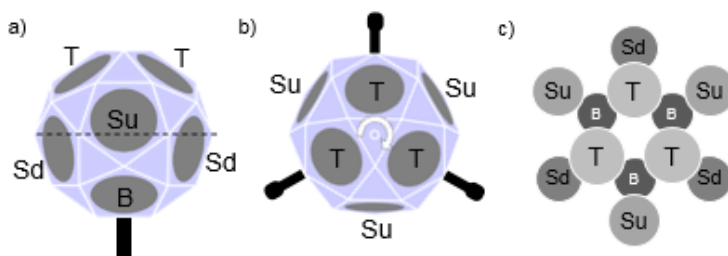


Figure 5.6. a) Side view of the dodecahedron with loudspeaker positions, b) Top view of the dodecahedron with loudspeaker positions, c) Schematic top view with loudspeaker positions. T: top position; Su: side upwards position; Sd: side downwards position; B: bottom position.

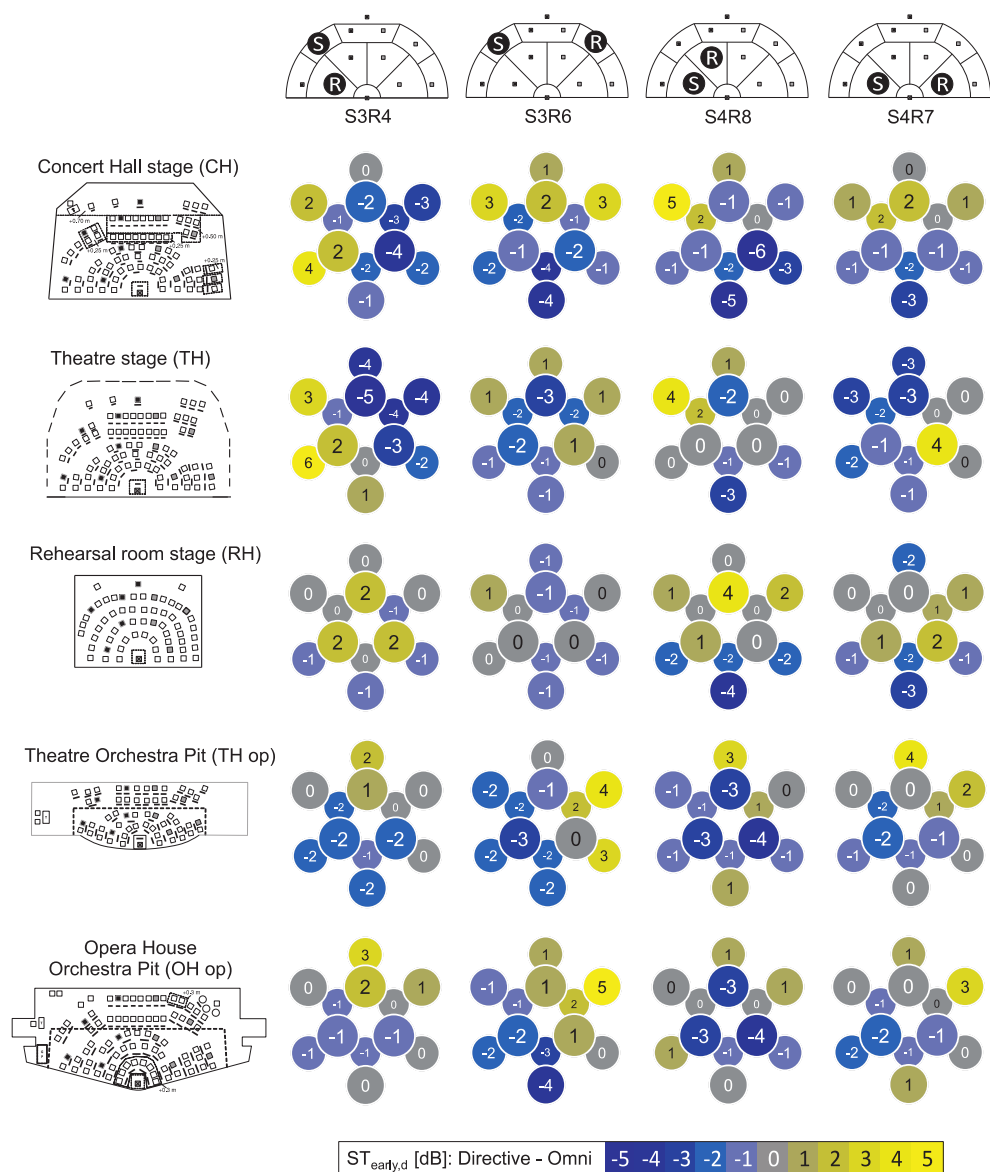


Figure 5.7. Variation of acoustic parameter $ST_{early,d}$ for the directional sound source with 12 aiming directions and the average results of both HATS' ears. Four different source-receiver combinations per different stages/pits. Average results for the 2,000 and 4,000 Hz octave band.

5.4 Conclusions

The sensitivity of stage acoustic parameters to source and receiver directivity has been investigated by measurements on three stages and in two orchestra pits. A single loudspeaker was used with a directivity similar to a trumpet and aimed at 12 evenly spread directions on 6 positions per stage or pit. As a directional receiver, a head and torso simulator (HATS) was used at 12 receiver positions on stage with its viewing direction towards the conductor's position. Besides, measurements were taken with (nearly) omnidirectional transducers, i.e. a dodecahedron loudspeaker and a single microphone. It should be noted that results are limited to instruments with a strong directivity such as the trumpet and that the dodecahedron loudspeaker is not perfectly omnidirectional.

The investigated stage acoustic parameters, reverberation time and support, have shown to be sensitive to the directivity of the measurement transducers. The parameters dealing with early sound, EDT and $ST_{early,d}$, are more sensitive than the parameters dealing with late sound, T_{20} and $ST_{late,d}$. For the single number ratings, the deviations are summarized in Table 5.1.

Table 5.1: Average absolute values of deviations found for parameter results when using directional instead of omnidirectional transducers. Single number rating 250 to 2,000 Hz based on the averages of various source and receiver combinations on five different stages and orchestra pits. Condition A compares the HATS with an omnidirectional microphone while using an omnidirectional source. Condition B compares a directional source with an omnidirectional source while using the HATS as a receiver. Values refer to both positive and negative deviations. For T_{20} the JND is assumed to be similar as for EDT . The JND for $ST_{early,d}$ and $ST_{late,d}$ has been estimated to be 2 dB.

	Source	Receiver	EDT	T_{20}	$ST_{early,d}$	$ST_{late,d}$
A	Omni source	HATS source side or shadow side ear versus omni microphone	< 20%	< 5%	< 1 dB*	< 0.5 dB*
B	Directional source maximum deviation of 12 directions versus omni source	HATS two ears average	< 40%	< 15%	< 2 dB	< 1 dB
	Just Noticeable Difference (JND)		5%	5%	2 dB	2 dB

* excluding the Theatre Orchestra Pit (THop)

When comparing results measured with a head and torso simulator (HATS) to results measured with an omnidirectional microphone (condition A in Table 5.1), the EDT tends to be lower and the $ST_{early,d}$ higher for the ear directed towards the sound source. The opposite effect occurs for the other ear on the shadow side. For EDT , the increase in direct sound relative to reflected sound due to the higher directivity is the reason that values are lower at the source side ear. For the early reflected sound level, measured by $ST_{early,d}$, the typical

geometry of a stage and setup of the orchestra members (with their back towards the (most) reflective side of the stage or pit) cause levels to be higher at the source side ear. A similar effect occurs when comparing results measured with a directive source to results measured with an omnidirectional source (condition B in Table 5.1). EDT and T_{20} show the lowest values and $ST_{early,d}$ and $ST_{late,d}$ highest values, when the sound source is directed towards the closest surfaces that cause a 1st order reflection towards the receiver.

The just noticeable difference (JND) can be used as an indicator to judge the perceptual relevance of these findings (although the JNDs were determined for omnidirectional parameters). Considering the reverberation time parameter EDT , with a JND of 5%, it is clear that ignoring the directivity of the sound source or receiver will lead to a stage or pit average difference of EDT up to 8 times larger than the JND. This also raises the question whether the EDT is a reliable parameter for measurements relatively close to the sound source. The reverberation time T_{20} is much less sensitive to transducer directivity. For the extended and modified stage support parameters $ST_{early,d}$ and $ST_{late,d}$, the JND is unknown but estimated by Gade to be 2 dB. The differences found are lower than the estimated JND when measuring stage/pit average single number ratings for these ST parameters with directive transducers instead of omnidirectional transducers. Only at frequencies ≥ 4 kHz are deviations above 2 dB.

It can be concluded that the influence of source and receiver directivity on stage acoustic parameters is limited, except for EDT measurements. Therefore, it might be sufficient to judge these parameters by measurements using omnidirectional transducers. Perceptual evaluations could provide more certainty about this conclusion. The parameter values show an asymmetry between the two ears, with the largest interaural difference for the Early Decay Time which is a predictor for the perception of reverberance. It can be expected that the brain combines the different signals at the two ears to one perception. Possibly, an omnidirectional acoustic parameter is a reasonable predictor for this combined perception. However, more research is necessary to find out whether for instance perceived reverberance is close to the average of the individual ears, closest to the most reverberant side or the least reverberant side. It would also be interesting to find out whether the JNDs are different for binaurally presented acoustic conditions.

5.5 References

- [1] ISO, Geneva, Switzerland, ISO 3382-1: Acoustics – Measurement of room acoustic parameters – Part 1: Performance spaces. 2009.
- [2] IEC, Geneva, Switzerland, IEC 60268–16: Sound System Equipment – Part 16: Objective Rating of Speech Intelligibility by Speech Transmission Index. 2003.
- [3] A.C. Gade: Investigations of musicians’ room acoustic conditions in concert halls. *Acustica* 69 (1989) 193-203, 249-261.
- [4] G.M. Naylor: Modulation transfer and ensemble music performance. *Acustica* 65 (1988) 127-137.

- [5] J.J. Dammerud: Stage acoustics for orchestras in concert halls. PhD Thesis. Bath: University of Bath; 2009. Source: <https://stageac.wordpress.com/phd/>
- [6] A. Guthrie: Stage acoustics for musicians: a multidimensional approach using 3d ambisonic technology. PhD Thesis. New York: Rensselaer Polytechnic Institute; 2014.
- [7] T.W. Leishman, S. Rollins, H.M. Smith: An experimental evaluation of regular polyhedron loudspeakers as omnidirectional sources of sound. *J. Acoust. Soc. Am.* 120 (2006) 1411-1422.
- [8] R.S. Martín, I. Witew, M. Arana, M. Vorländer: Influence of the source orientation on the measurement of acoustic parameters. *Acta. Acust. united Ac.* 93(3) (2007) 387-397.
- [9] R.H.C. Wenmaekers, C.C.J.M. Hak: The sound power as a reference for Sound Strength (G), Speech Level (L) and Support (ST): uncertainty of laboratory and in-situ calibration. *Acta. Acust. united Ac.* 101 (2015) 892-907.
- [10] R. Kruse, A. Haeussler, S. van de Par: An omnidirectional loudspeaker based on a ring-radiator. *Appl. Acoust.* 74(12) (2013) 1374-1377.
- [11] J. Meyer: *Acoustics and the performance of music*. 5th ed. New York: Springer; 2009.
- [12] J. Pätynen, T. Lokki: Directivities of symphony orchestra instruments. *Acta. Acust. united Ac.* 96 (2010) 138-167.
- [13] F. Otondo, J.H. Rindel: The influence of the directivity of musical instruments in a room. *Acta. Acust. united Ac.* 90 (2004) 1178-1184.
- [14] N.R. Shabtai, M. Vorländer: Acoustic centering of sources with high-order radiation patterns. *J. Acoust. Soc. Am.* 137(4) (2015) 1947-1961.
- [15] A.M. Pasqual: Sound directivity control in a 3-D space by a compact spherical loudspeaker array. PhD thesis. Brazil: State University of Campinas, Faculty of Mechanical Engineering; 2010.
- [16] M. Follow, J. Klein, P. Dietrich, M. Vorländer: Including directivity patterns in room acoustical measurements. *Proceedings of Meetings on Acoustics ICA*, 2013.
- [17] A.D. Brown, G.C. Stecker, D.J. Tollin: The precedence effect in sound localization. *J. Assoc. Res. Otolaryngol.* 16 (2014) 1-28.
- [18] H. Møller: Fundamentals of binaural technology. *Appl. Acoust.* 36(5) (1992) 171-218.
- [19] C.C.J.M. Hak, R.H.C. Wenmaekers: The Impact of Sound Control Room Acoustics on the Perceived Acoustics of a Diffuse Field Recording. *WSEAS Transactions on Signal Processing* 4(6) (2010).
- [20] C.C.J.M. Hak, M. van Haaren, R.H.C. Wenmaekers, L.C.J. van Luxemburg: Measuring room acoustic parameters using a head and torso simulator instead of an omnidirectional microphone. *Proc Intersound 2009 Ottawa, Canada*; 2009.
- [21] H. Tahvanainen, J. Patynen, T. Lokki: Studies on the perception of bass in four concert halls. *Psychomusicology: Music, Mind and Brain* 25(3) (2015) 294-305.
- [22] L.M. Wang, M.C. Vigeant: Evaluations of output from room acoustic computer modeling and auralization due to different sound source directionalities. *Appl. Acoust.* 69 (2008) 1281-1293.
- [23] K. Ueno, K. Kato, K. Kawai: Effect of room acoustics on musicians' performance. Part I: experimental investigation with a conceptual model. *Acta. Acust. united Ac.* 96 (2010) 505-515.
- [24] Z. Schärer Kalkandjiev, S. Weinzierl: The influence of room acoustics on solo music performance: an experimental study. *Psychomusicology Music, Mind and Brain* 25(3) (2015) 195-207.

- [25] B.N. Gover, J.G. Ryan, M.R. Stinson: Measurements of directional properties of reverberant sound fields in rooms using a spherical microphone array. *J. Acoust. Soc. Am.* 116 (2004) 2138-2148.
- [26] J. Pätynen, S. Tervo, T. Lokki: Analysis of concert hall acoustics via visualisations of time-frequency and spatiotemporal responses. *J. Acoust. Soc. Am.* 133 (2013) 842-857.
- [27] G.K. Behler, M. Pollow: Variable source-directivity using dodecahedron-loudspeakers. *J. Acoust. Soc. Am.* 123(5) (2008) 3614-3614.
- [28] R.H.C. Wenmaekers, C.C.J.M. Hak: How orchestra members attenuate direct, early reflected and late reflected sound on five different concert stages and orchestra pits. *J. Acoust. Soc. Am.* 140(6) (2016) 4437-4448.
- [29] R.H.C. Wenmaekers, C.C.J.M. Hak, L.C.J. van Luxemburg: On measurements of stage acoustic parameters - time interval limits and various source-receiver distances. *Acta. Acust. united Ac.* 98 (2012) 776-789.
- [30] C.C.J.M. Hak, R.H.C. Wenmaekers, L.C.J. van Luxemburg: Measuring room impulse responses: impact of the decay range on derived room acoustic parameters. *Acta. Acust. united Ac.* 98 (2012) 907-915.
- [31] M. Vorländer, G. Raabe: Calibration of reference sound sources. *Acustica* 81 (1995) 247-263.
- [32] R.H.C. Wenmaekers, C.C.J.M. Hak: Early and Late Support measured over various distances: The covered versus open part of the orchestra pit. *Building Acoustics* 20(4) (2013) 313-322.

Chapter 6

Sound level model for symphony orchestras

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Authors: R.H.C. Wenmaekers, C.C.J.M. Hak
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Abstract

Musicians in a symphony orchestra rely on the direct and reflected sound on a concert hall stage to be able to hear each other. Besides ensemble conditions, members and directors of symphony orchestras are concerned about the noise levels musicians are exposed to. However, the actual contribution of the different parts of the sound field cannot be derived from sound level measurements in orchestras. In this chapter, a prediction model is presented that can be used to investigate the distribution of the direct, early reflected and late reflected sound from all musicians to the total sound level at a single musician's position. It is shown that the contributions of each different aspect to the total sound level are in the same order of magnitude. In some cases, the direct sound dominates, while in other cases, the early or late reflected sound does. Considerable variations in sound levels are found between a concert hall, rehearsal room and orchestra pit, due to the difference in room acoustical properties. An example is presented of calculated sound levels for a violin's position in the orchestra for the three halls. The results from the example show that the model has potential for studying the influence of architectural as well as acoustical aspects on the sound levels that occur in a symphonic orchestra, both from a health and musical point of view.

6.1 Introduction

Sound levels and loudness play an important role in the performance of and listening to music [1]. The conductor and musicians in an orchestra search for the appropriate sound level balance between different instruments. This is not only important for the listeners in the audience area. A good balance is also crucial for the musicians themselves to be able play together easily [2], for instance to control timing [3]. For more information about ensemble playing and stage acoustics, an extensive literature review can be found in Gade [4].

Besides ensemble conditions, members and directors of symphony orchestras are concerned about the noise levels musicians are exposed to. In accordance with European Directive 2003/10/EC [5], professional musicians should be protected from sound levels that may cause hearing damage. The results of earlier investigations have shown that the sound levels within an orchestra can cause hearing loss [6]. Also, research has shown that the sound level will vary between different musicians playing different instruments and musical pieces, and between musicians being positioned differently on various stages [7].

Factors that determine the sound levels can be musically based: the piece and its interpretation by the conductor and musicians; or acoustically based: the impact of the stage and hall reflections. Schärer Kalkandjiev and Weinzierl [8] have shown in a case study that a highly experienced solo cello player responded to a louder acoustical environment (higher Sound Strength or G) by reducing his power, while a lack of reverberation (not necessarily a lower G) encouraged the musician to play more powerful. In terms of sound exposure, this results in an interesting contradiction: by playing in a louder (acoustically more amplifying) and more reverberant environment, the output power is reduced by the musician. What if the own instruments' direct sound is causing most of the exposure? Then, the total exposure might be lower in a louder acoustic environment.

To further investigate the balance between direct sound and reflected sound, in this chapter, we will focus on the acoustical aspects of sound levels. Sound reflected from the stage boundaries arrives relatively early after the direct sound. The (much) later reflected sound is mostly caused by reflecting surfaces in the audience area. To some extent, the amount of early and late reflected sound can be controlled separately by the individual design of the stage and design of the hall. Therefore, it is interesting to study the separate contributions of direct, early and late sound to the total sound exposure of the musicians.

The contribution of each instrument and each acoustical aspect to the total sound level cannot be separately determined from sound (pressure) level measurements [9]. As a solution to this problem, a sound level prediction model is presented in this chapter where the direct, early reflected and late reflected sound energy transfer is calculated using the sound power and directivity of symphony orchestra instruments together with stage acoustic parameters Early and Late Support. The goal of the model is to study trends of the contributions of the different instruments and different acoustical aspects to the total sound level received by every musician in the orchestra. The possibilities and limitations of the model will be discussed, and first results will be presented based on state of the art knowledge. However,

the trends calculated by the current model should be compared to subjectively critical limits (however scarcely available) while considering the uncertainties and limitations of the model. We hope that our findings, derived from an architectural acoustics point of view, may inspire other fields of research concerning the understanding of sound exposure and of musical performance.

In this chapter, the possibilities of the model will be described in the section 6.2, together with discussions of its limitations. In the section 6.3, an example of the necessary input data will be presented. In the section 6.4, a study is presented to validate (part of) the model. In section 6.5, a case study is presented that show the possibilities of the model to give insights in the sound levels within a symphony orchestra, both from a health as well as from a musical point of view. This chapter concludes with a discussion on further development of the model and a conclusion. Some parts have been updated after publication in the journal to avoid confusion with findings presented in [chapter 4](#) and [chapter 5](#).

6.2 Model

A visual representation of the sound level prediction model, together with the relevant parameters, is presented in Figure 6.1. The various parameters will be explained in the following subsections (a schematic overview of the model is also shown in chapter 7, figure 7.2).

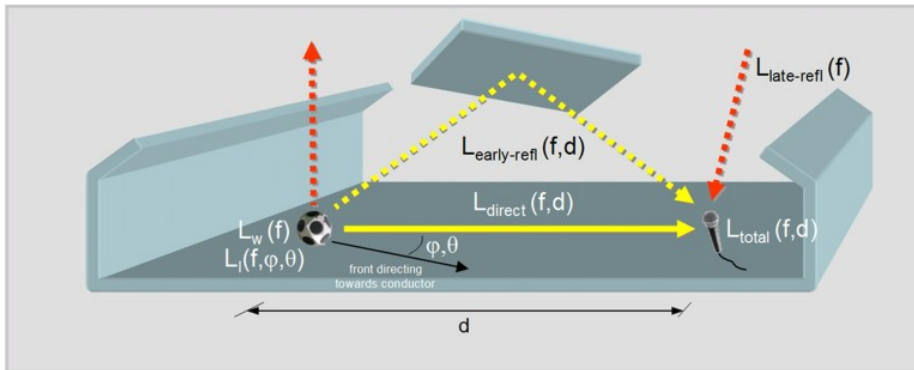


Figure 6.1. Summary of the sound level prediction model from one source to one receiver on a concert hall stage. The direct sound level, L_{direct} , is calculated based on measured instrument directivity and distance. The early and late reflected sound level, $L_{early-refl}$ and $L_{late-refl}$ respectively, are calculated based on measured room acoustic parameters.

6.2.1 Sound path

The transfer of sound from a sound source to a receiver in a room can be fully described by the room impulse response, which can either be measured or predicted. In this model, the impulse response is divided into three typical room acoustical aspects to study the balance between them: the direct sound, the early reflected sound and the late reflected sound. The

direct sound path is of interest to study the influence of distance, instrument directivity and the attenuation of sound by objects. The early reflected sound is generally considered to be meaningful for ensemble playing on stage while the late reflected sound may contribute to a sense of feedback from the hall. However, late reflected sound can also be detrimental for ensemble playing as it can mask direct and early reflected sound [4]. The direct sound is calculated analytically using the directivity of the sound source and receiver, together with an empirical model to include the effect of the obstruction by other orchestra members into the model. The early and late reflected sound energy is estimated from measured room impulse responses using an omnidirectional sound source and receiver on an empty stage.

In this chapter, available measured data from empty stages will be used, as these were available at the time of writing this chapter. However, the impulse response should ideally be determined under occupied conditions (see [chapter 4](#)), which will be used in the updated model that will be presented in the next [chapter 7](#). Directivity is not considered for the calculation of early and late reflected sound levels. As was shown in [chapter 5](#) (written later), the influence of instrument and receiver directivity is small for reflected sound (< 2 dB) and therefore negligible.

6.2.2 Sound source directivity

In general, a sound source can be described by the sound intensity $L_I(f, \varphi, \theta, d)$ with independent parameters frequency (f), orientation (elevation φ and azimuth θ) and distance (d). A musical instrument cannot easily be defined by these parameters, because the spectrum and directivity may change per note and playing style. When assessing sound levels, however, one is often interested in an average value over time. It may then be legitimate to use time average values for a musical instrument's directivity. In this model, measured average values of sound intensity per angle $L_I(f, \varphi, \theta)$, for common orchestral instruments at free field distance, are used from Pätynen & Lokki (2010), obtained from averaging over several tones within the instruments tonal range.

The relative sound intensity $L_I(f, \varphi, \theta)$ of the instruments is made available by Pätynen & Lokki [10], for the octave bands 125 Hz to 16 kHz in accordance with the common loudspeaker format (CLF) with steps of 10 degrees. The CLF format is defined by 36 arcs running from the front to the back of the sound source. This results in a finer grid in the front direction where the directivity of a loudspeaker is usually the highest. In contrast, in this model the CLF output is calculated without rotating the base of the original coordinate system 90 degrees downwards on the transverse axis [11], see Figure 6.2. This way, the angle between a source and receiver is easily defined by elevation φ and azimuth θ . This will reduce the fineness of the grid in front of the sound source, however, as the CLF data points have been interpolated from only 20 microphone measurements, the information is still an interpolation from an even coarser grid. The front orientation of the directivity is defined as the frontal viewing direction of the musician.

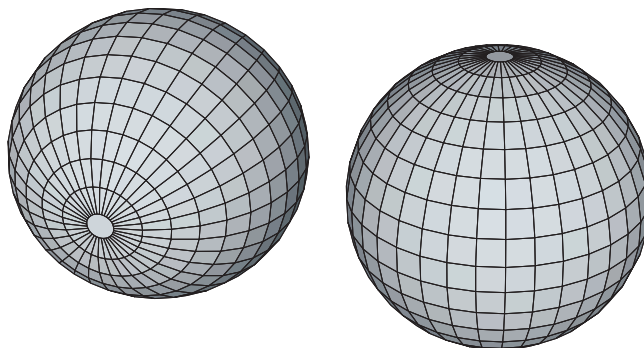


Figure 6.2. The CLF format is defined by 36 arcs running from the front to the back of the sound source, see left Figure. In this model the CLF output is calculated without rotating the base of the original coordinate system 90 degrees downwards on the transverse axis, see right figure. This way, the angle between a source and receiver is easily defined by elevation φ and azimuth θ .

6.2.3 Direct sound of other musicians' instruments

The direct sound path depends on the source-to-receiver distance and orientation of the source relative to the receiver, assuming that both musicians face the conductor. Besides that, the attenuation of the orchestra as obstruction between the source and receiver is included from a model based on values of $\Delta L(f)$ measured in a scale model by Dammerud and Barron [12] (in the next chapter, an updated model is described that uses our real scale measured values as presented in [section 4.3.1](#)). In terms of assessing sound exposure in accordance with ISO 9612, a measurement should be done positioning an omnidirectional microphone on the shoulder at the most exposed ear. However, Schmidt et al. [7] concluded that positioning small omnidirectional microphones in front of the musicians ear canal is a more appropriate method to measure sound exposure of musicians, because it better represents the exposure from the own musical instrument (for instance, it is often not possible to put a microphone on a violins player's left shoulder) and it is not always possible to predict the most exposed ear. In the model presented in this chapter, a similar method is used where the sound level is calculated for the two ears while taking into account the transfer function of the receiver's head (HRTF). The HRTFs were measured for the 0-degree elevation while rotating in 10 degree steps in an anechoic room using omnidirectional DPA 4060 miniature condenser microphones in front of the ear canal of a B&K type 4128C head and torso simulator, see Figure 6.3. The HRTFs were calibrated using the sound pressure level of the same microphones in the free field at centre position of the head.

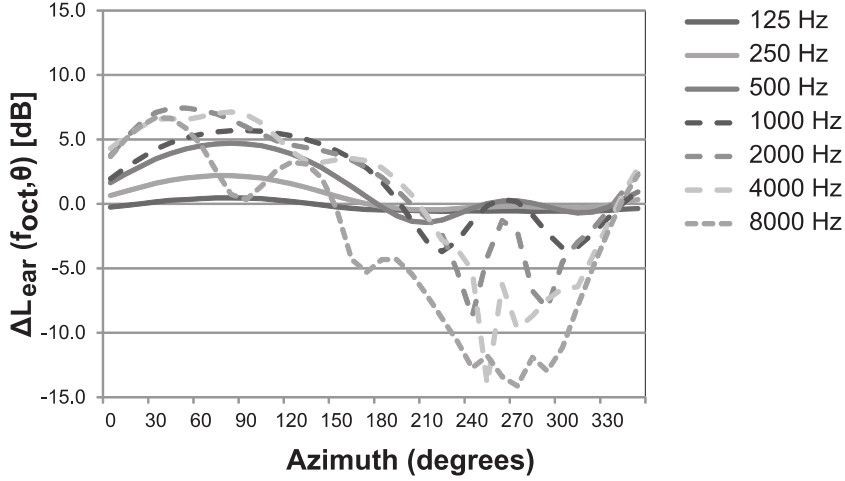


Figure 6.3. The HRTFs for the 0-degree elevation for the right ear, measured in an anechoic room using omnidirectional DPA 4060 miniature condenser microphones in front of the ear canal of a B&K type 4128C head and torso simulator. The HRTFs are calibrated using the sound level of the same microphones in the free field at the of centre head's position.

L_{direct} is calculated as

$$L_{direct}(f, d) = L_{eq;1m}(f, \varphi, \theta) - 20 \lg(d) + \Delta L_{orch}(f, d) + \Delta L_{ear}(f, \theta) \quad (1)$$

And

$$\Delta L_{orch}(f, d) = a(f) \cdot d + c(f) \quad (2)$$

where $L_{eq;1m}(f, \varphi, \theta)$ is the sound level in dB at 1 m distance per octave band 125 to 8,000 Hz at elevation φ and azimuth θ in degrees estimated from measured values of sound intensity $L_I(f, \varphi, \theta)$ and $L_{eq;1m;front}(f)$ derived from the frontal anechoic recordings of every instrument; d is the source receiver distance in m; $\Delta L_{orch}(f, d)$ is the attenuation by the orchestra in dB estimated from scale model measurements using an attenuation factor 'a' in dB loss per m through the orchestra and a constant 'c' in dB for the overall shift of attenuation due to the effect of the floor and orchestra reflections, see Table II in Dammerud and Barron [12] (in the next chapter, our real scale measured values are used as presented in [section 4.3.1](#)), flat floor path A; and $\Delta L_{ear}(f, \theta)$ is the HRTF expressed as a difference in level between an omnidirectional microphone in front of the ear canal and the same microphone in the free field at the of centre head's position.

6.2.4 Direct sound of the own instrument

The direct sound level of the own instrument also needs to be modelled. Pätynen & Lokki, [10] determined the sound intensity per angle $L_I(f, \varphi, \theta)$ per instrument using the musician's head in the centre. In our model, however, a reference point on the musical instrument itself needs to be regarded as the 'point source' or 'acoustic centre' for the direct sound of the own instrument (even though this 'point' can vary depending on notes played and playing style). The chosen reference points are explained in section 6.4. Hereby, the actual distance between the instrument and the reference microphone is taken into account, see Figure 6.4. For each individual ear, the direct sound of the own instrument is calculated as

$$L_{direct;own}(f, d) = L_{eq;microphone}(f, \varphi, \theta) - 20 \lg \left(\frac{d_{instrument_to_ear}}{d_{microphone_to_instrument}(\varphi, \theta)} \right) \quad (3)$$

where $L_{eq,microphone}$ is the sound level at the microphone on a straight line crossing the ear and the reference point on the musical instrument; $d_{instrument_to_ear}$ is the distance between the reference point on the musical instrument and the ear; and $d_{microphone_to_instrument}$ is the distance between the microphone position and the reference point on the musical instrument.

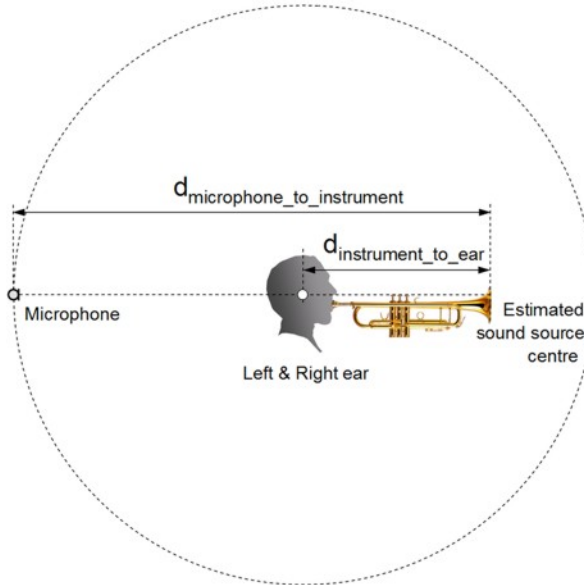


Figure 6.4. The direct sound of the own instrument is calculated at the ears by applying the inverse square law using the sound pressure level measured on a line from the instrument through the ear to the measurement radius of the directivity measurement where the head is in the middle, see equation 3.

The musician's ear is in the near field of the musical instrument within less than 1 m distance and the far-field rules, like the inverse square law, might not be sufficiently reliable. Therefore, a validation study has been performed based on new measurements in an anechoic room with several musicians, which will be presented in section 6.4 of this chapter. Note that the HRTF is not included in the prediction of the own instruments' sound level, because it was found that the agreement between measurement and model was best without HRTF.

It should be noted that bone-conducted sound might also contribute to the own instrument's sound level, however, this is not taken into account by the model.

6.2.5 Sound power of each instrument

The sound power $L_w(f)$ is obtained from anechoic recordings of different musical fragments made available by Pätynen et al. [13] for each instrument. The equivalent sound pressure level $L_{eq;front}(f)$ in dB is determined for every recording for the frontal microphone per instrument per musical piece. The silent parts in the recordings have not been removed, so results can be interpreted as a sound exposure level for the particular piece of music (although one could also use smaller time samples). An absolute sound level calibration was made available by Pätynen et al.. The sound power $L_w(f)$ is calculated as

$$L_w(f) = L_{eq;front} + 20\lg(d) + 10\lg\left(\sum_{n=1}^N S_i 10^{\frac{L_i(f,\varphi,\theta)}{10}}\right) \quad (4)$$

where $L_{eq;front}$ is the equivalent sound pressure level in dB per frequency band in Hz in front of the musician (microphone no. 6, see Table II in Pätynen et al. [13]); S_i is the partial surface in m² per angle of the directivity data on a sphere of 1 m radius ($N = 614$); $L_i(f,\varphi,\theta)$ is the relative sound intensity at elevation φ and azimuth θ in degrees ($N = 614$), the musician's viewing direction is defined as 0 dB; and d is the microphone distance in m ($d = 2.3$ m). It should be noted that, by using a fixed sound power, the interaction between the acoustics and the musicians in the room is not taken into account.

6.2.6 Early and late reflected sound

After the direct sound, sound arrives via reflections from the stage and hall boundaries. From laboratory studies using simulated sound fields, Gade [2] concluded that reflections, arriving up until approximately 100 ms after the 'orchestra onset', may support the musicians in playing ensemble. In [section 2.3.2](#) [14] it was shown that, for an average size stage, first order reflections indeed arrive within a maximum delay of 100 ms after the 'orchestra onset'. When a simultaneous onset of all instruments is assumed, the direct sound of the other player's instrument will arrive delayed, for instance, the sound travelling over 10 m distance will arrive 29 ms later. For the early reflections of this instrument to be 'supportive', they should arrive within 100 - 29 = 71 ms after the arrival of the direct sound of that instrument. So, the time interval width between the arrival of the direct sound and the supportive early

reflections depends on the source-to-receiver distance. To measure the amount of energy of this supportive early reflected sound, taking into account the distance-dependent time interval, the Early Support at various distances can be used, denoted $ST_{early,d}$, as introduced by Gade [2] and modified by Wenmaekers et al. [14, or [chapter 2](#)], calculated as

$$ST_{early;d} = 10 \lg \left(\frac{\int_{10}^{103-delay} p_d^2(t) dt}{\int_0^{10} p_{1m}^2(t) dt} \right) \quad (5)$$

where $ST_{early,d}$ is the Early Support at distance d in m; p_d is the sound pressure measured at distance d ; p_{1m} is the sound pressure measured at 1 m distance; and delay is the source-receiver distance divided by the speed of sound.

In [section 2.5.4](#) [14] it was shown that in most cases the Early Support depends on the source-to-receiver distance. Using the Early Support parameter, the source-to-receiver distance d and the sound power of the sound source or instrument L_w , the early reflected sound level $L_{early-refl}$, is calculated as

$$L_{early-refl}(f, d) = L_w(f) + ST_{early;d}(f, d) - 11 \quad (6)$$

Late reflections describe the amount of reverberation. For the orchestra members, it is important to receive ‘feedback’ from the hall to get an impression of what the audience is hearing [2]. However, late reverberation can also mask the useful direct and early reflected sound. All reflections arriving 100 ms after the ‘orchestra onset’ are considered late reflections. Again, the time interval start, relative to the arrival of the direct sound, is dependent on the distance. To measure the amount of energy of late reflected sound, the Late Support at various distances can be used, denoted $ST_{late,d}$, as introduced by Gade [2] and modified by Wenmaekers et al. [14, or [chapter 2](#)], calculated as

$$ST_{late;d} = 10 \lg \left(\frac{\int_{10}^{\infty} p_d^2(t) dt}{\int_0^{10} p_{1m}^2(t) dt} \right). \quad (7)$$

In contrast to the $ST_{early,d}$, in [section 2.5.5](#) [14] it was shown that $ST_{late,d}$ is not dependent on the source-to-receiver distance, so a fixed value over the stage is valid. Similar to the early reflected sound level $L_{early-refl}$, the late reflected sound level $L_{late-refl}$ is determined from the

sound power L_w of the instrument and the Late Support at various distances d denoted $ST_{late,d}$ as

$$L_{late-refl}(f) = L_w(f) + ST_{late;d}(f) - 11 \quad (8)$$

It has been suggested to use Sound Strength G as a parameter for describing the early and late reflected sound on stage [15], instead of using the ST parameters. Both types of parameters are defined in such a way, that the sound pressure level of the impulse response (within a certain time interval) is compared to a reference sound pressure level: the direct sound measured in the free field at a given distance. The reference sound pressure level for G is determined at 10 m distance and for the ST parameters at 1 m distance. It should be noted that the reference level of the ST parameters is intended to be derived from a laboratory sound power measurement [16] and should not be derived from an in situ measurement on stage where the floor reflection is present as is suggested by ISO 3382-1. Due to the difference in reference level, this results in a 20 dB difference between the parameters when the same time interval is used: for instance G_{10-100} and $ST_{early,d}$ at 1 m distance, both using a 10-100 ms time window. The 1 m reference, which is more or less equal to the distance between the own instrument and the ears, is conceptually clear for judging stage acoustic conditions and therefore the ST parameters are superior over G parameters which are using an arbitrary reference distance of 10 m. The choice of appropriate time intervals for judging early and late reflected sound levels, either using the ST or G definition, has been thoroughly investigated, see [chapter 2](#). It was concluded that a time window defined relative to the departure of the direct sound, which has been introduced in the extended support parameters $ST_{early,d}$ and $ST_{late,d}$, is conceptually superior over a time window defined relative to the arrival of the direct sound at the receiver position, which is commonly used in G parameters like G_{late} .

The direction of arrival of the reflected sound might be important for both predicting sound levels and judging ensemble conditions. As explained in section 6.3.2, varying (receiver) sensitivity with respect to the spatial sensitivity of the hearing system and the sound source directivity are not taken into account by the model for reflected sound. This means that in the prediction of the sound level of the reflected sound, the model assumes that the energy is transmitted and received equally spread over all directions. In theory, the ‘omnidirectional to omnidirectional transmission’ might be valid under diffuse sound field conditions, which might be a valid approximation for the late reflected sound on a concert hall stage. The early reflected sound level depends more on discrete reflections and the ‘omnidirectional to omnidirectional transmission’ might be less accurate, see [chapter 5](#).

6.2.7 Measurement conditions

The $ST_{early,d}$ and $ST_{late,d}$ are measured on a grid of measurement positions in the orchestra area, on stage or in a rehearsal room. Omnidirectional transducers are used, preferably at 1

m height, kept at least 2 m away from any boundaries or objects. Measurements were taken at empty stages (although occupied conditions are recommended, see [chapter 4](#)). To reduce the measurement uncertainty of single $ST_{early,d}$ and $ST_{late,d}$ measurements, an average value over 5 stepwise rotations of the omnidirectional sound source was used[17, or [section 3.4](#)] and impulse responses with a decay range INR [18] of at least 45 dB. The reference level of ST, the direct sound at 1 m distance, is derived from a sound power measurement in a reverberation room, as suggested by Gade [16, also discussed in [chapter 3](#)]. More background information on the measurement procedure can be found in [section 2.5.1](#) [14].

6.3 Input and output

6.3.1 Early and Late Support for various halls

$ST_{early,d}$ and $ST_{late,d}$ have been measured by the authors in various concert halls [14, or [section 2.4](#)], in orchestra pits [19, or [section 9.2](#)] and in rehearsal rooms [20, or [section 9.3](#)]. A logarithmic trend line can be calculated from various results of $ST_{early,d}$, per octave band for different source-to-receiver positions per hall. The trend line has the form ‘ $a \lg(d) + b$ ’, where ‘ a ’ and ‘ b ’ are constants and ‘ d ’ is the source-to-receiver distance. In Table 6.1, the 250 to 2,000 Hz octave bands averaged constants to describe the $ST_{early,d}$ trend line, and the single number $ST_{late,d}$ are presented for a concert hall stage, an orchestra pit and a rehearsal room. Also, the average reverberation time T_{mid} (500 and 1,000 Hz) is given. In Figure 6.5, the trend lines for $ST_{early,d}$ are presented. For the orchestra pit, three different trends are given, each having a distinctive shape for different types of source and receiver positions: both positions in the open part; both positions in the covered part; and just one of both positions in the open or covered part [19, or [section 9.2](#)]. In the calculations that will be presented in section 6.5, the single number values for $ST_{early,d}$ and $ST_{late,d}$ have been used for each individual octave band.

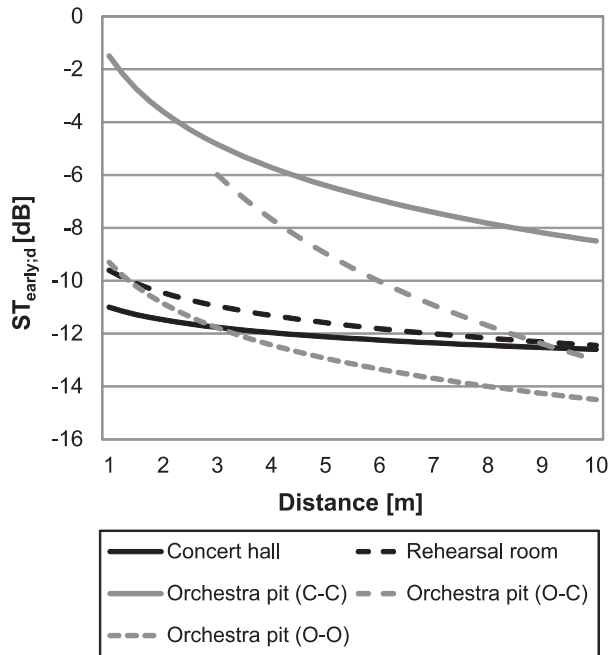


Figure 6.5. Trend lines of $ST_{early,d}$, as a function of distance measured in a concert hall ($R^2 = 0.04$, this typical concert hall stage has almost no decrease of $ST_{early,d}$ over distance), rehearsal room ($R^2 = 0.45$) and orchestra pit: O-O, both positions in the open part ($R^2 = 0.74$); C-C, both positions in the covered part ($R^2 = 0.88$); and O-C, just one of both positions in the open or covered part ($R^2 = 0.85$).

Table 6.1. $ST_{early,d}$, $ST_{late,d}$ and T_{mid} measured in a concert hall, orchestra pit and rehearsal room.

Hall	$ST_{early,d} = a \lg(d) + b$ [dB]		$ST_{late,d}$ [dB]	T_{mid} [s]
	a	b	average	Average
Concert hall stage 14,400 m ³	-1.6	-11.0	-15.2	2.0
Rehearsal room 2,500 m ³	-2.8	-9.6	-11.8	1.2
Orchestra pit (Covered-Covered)	-7.0	-1.5	-15.2	1.3
Orchestra pit (Open-Covered)	-13.4	+0.4	-17.3	1.3
Orchestra pit (Open-Open)	-5.2	-9.3	-16.0	1.3

6.3.2 Orchestral layout

Based on the typical Mahler Symphony 1 orchestration and the typical American orchestra layout [21], an orchestra setup is selected for the model with all musicians positioned on a rectangular grid, see Figure 6.6. The receiving musician 8 (violin) is used in this chapter as an example. In case of the concert hall stage, musicians 56-58 and 63-74 are elevated by 0.3 m and musicians 59-62 and 81-82 are elevated by 0.6 m to simulate risers. In the case of an orchestra pit, it is assumed that all woodwind, brass, horn and percussion players are positioned in the covered part of the pit and all strings sections are positioned in the open part of the pit. A distance of 1.3 m (side to side) and 1.6 m (back to back) was assumed between the musicians on the grid, resulting in a 2.1 m² area per musician (similar to Dammerud and Barron [12] who used 2.3 m² per musician).

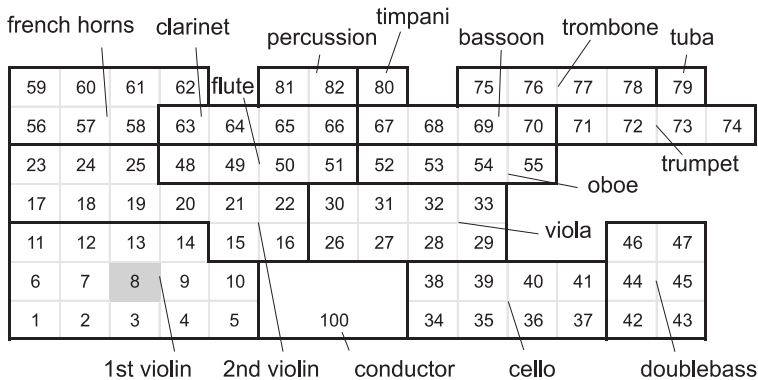


Figure 6.6: Generic orchestra setup for Mahler Symphony 1. Strings: 1-14: 1st violin, 15-25: 2nd violin, 26-33: viola, 34-41: violoncello, 42-47: double bass. Woodwinds: 48-51: flute, 52-55: oboe, 63-66: clarinet, 67-70: bassoon. Brass: 71-74: trumpet, 75-78: trombone, 79: tuba. Separate instruments: 56-62: French horn, 80: timpani, 81-82: percussion.

6.3.3 Sound power of musical pieces

To assess spectral and level differences between instruments the sound power L_w is needed, which is derived from the calibrated anechoic recordings of musical pieces by Pätynen et al. [13]. Separate instrument recordings were made of different orchestral excerpts of music. From the recordings of the Mahler Symphony no. 1 sample (first 2:12 min of the fourth part) and Bruckner Symphony no. 8 sample (1:27 min) the equivalent sound power levels have been determined using equation 4. Figure 6.7 shows the A-weighted sound power level per instrument per musical piece, as an equivalent sound power level over the musical sample. Large differences between the two pieces occur only at the violin sections. Because of relatively small differences between the two pieces and because only the Mahler piece has a percussion part, Mahler will be used for calculations in this chapter.

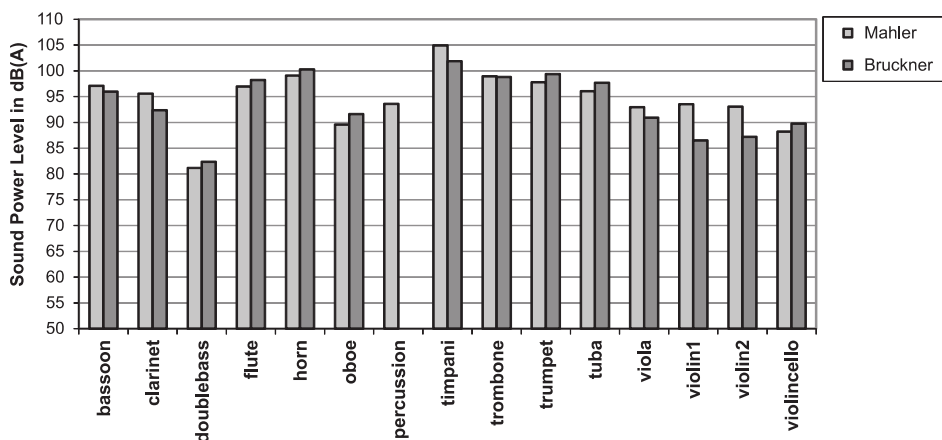


Figure 6.7. Average A-weighted sound power level per instrument for different musical pieces Mahler Symphony no. 1 sample (2:12 min) and Bruckner Symphony no. 8 sample (1:27 min).

6.3.4 Output

The output of the model gives values for $L_{direct}(f)$, $L_{early-refl}(f)$ and $L_{late-refl}(f)$ for every combination of source and receiver position. Also, values for $L_{total}(f)$ can be determined, which is the energetic summation of $L_{direct}(f)$, $L_{early-refl}(f)$ and $L_{late-refl}(f)$. For a symphony orchestra of 100 musicians this results in $100 \times 100 = 10,000$ values for every parameter and frequency band. The sound energy of sound sources with equal instrument and musical parts are energetically summed to study grouping effects.

Human ears are highly sophisticated sound receivers, with varying sensitivity with respect to frequency. The varying sensitivity is introduced in this model by A or C weighting the sound level in accordance with IEC 61672. The A-weighting is commonly used when assessing relatively high sound exposure levels, even though it represents the low isophone curve with 40 dB at 1,000 Hz. The C weighting, representing the isophone curve with 100 dB at 1,000 Hz, is more appropriate when assessing loudness in symphony orchestras, as sound levels are often 80 dB up to 100 dB. Besides weighted levels, the model has also been used to consider separate octave bands 125 to 8,000 Hz.

6.4 Validation

6.4.1 Measurement setup

The direct binaural sound level of the own instrument has been measured in an anechoic room with two DPA 4060 miniature condenser microphones fixed in front of the musician's ears, see Figure 6.8. Also, a B&K type 4189-A-021 microphone was positioned at 2 m distance from the musicians' ears at equal height, see Figure 6.8, denoted Ref. This microphone is used to determine the reference sound level in front of the musician $L_{eq;microphone}$

(f), see equation 3. The sound pressure levels measured using the DPA microphones were corrected to match the flat frequency response of the B&K microphone, based on a comparison study of the microphones in a diffuse field (reverberation room) and direct field (anechoic room). However, it should be noted that at high frequencies for both microphones, the sound levels measured close to the ear are 2.5, 4.5 and 2.0 dB higher in the octave bands 2, 4 and 8 kHz, respectively, than in the free field. These differences are considered to be caused by the sound field, and not by the type of microphone, so no additional correction is made.

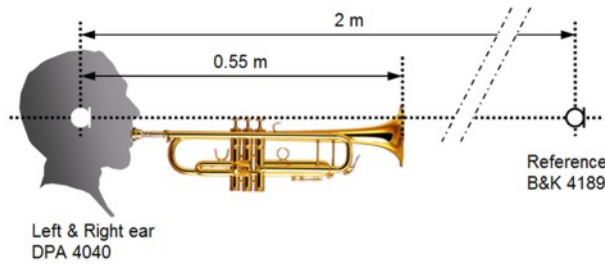


Figure 6.8. Side view of the microphone setup. Two DPA 4040 microphones are positioned in front of the musician's ears. The distance from the centre of the musician's head to the reference microphone B&K 4189 is 2 m. The distance between the centre of the sound source (the bell of the trumpet) to the musician's ears is 0.55 m.

6.4.2 Procedure

Various musical instruments were investigated in the research: flute (2x), piccolo (2x), trumpet, flugelhorn, bass trombone, trombone and violin. Every musician was asked to play C major scales in the native playing range of the instrument over two octaves up and down, with altered articulation (staccato and legato) and musical dynamics (piano and forte). All tones were played with constant speed. While playing, calibrated recordings have been made using the three microphones. The average sound pressure level was determined for the whole recording session. Afterwards, the background noise level of the measurement system was determined. In this research, sound levels are only presented if they are at least 10 dB above the background noise level.

6.4.3 Distance between instrument and ears

Part of the model's input in equation 3 are the geometrical parameters elevation φ , azimuth θ and distance between the instrument and the musicians left and right ear. The applied angles of elevation φ and azimuth θ are illustrated in Figure 6.9. The values determined for the musicians in this research are presented in Table 6.2. For the flute, the geometrical parameters were determined relative to half of the tube length at 40 cm to the left ear and 20 cm to the right ear and, for the piccolo, at 28 cm to the left ear and 15 cm to

the right ear. For the trombone player’s right ear, the geometrical parameters were determined relative to the bell, slightly on the left at 50 cm. The trombone player’s left ear is in close proximity to the tubes on the shoulder, which also radiates sound, so the geometrical parameters were determined relative to 30 cm. For both ears of the trumpet/flugelhorn player, the geometrical parameters were determined relative to the bell in front of the player at 55 cm. For the violin, the geometrical parameters were determined relative to the bridge, more or less in the middle of the soundboard at 20 cm to the left ear and 25 cm to the right ear. The neck of the violin was pointing towards 330 degrees azimuth.

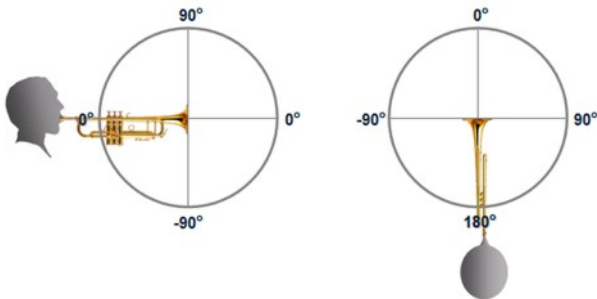


Figure 6.9. Side view showing Elevation θ (left) and top view showing Azimuth ϕ (right) used in to describe the angle between the sound source centre (in this example the bell of the trumpet) and the musicians’ ears.

Table 6.2. Geometrical parameters for the angle and distance between instrument and ears.

Instrument	Elevation θ [°]	Azimuth ϕ [°]	D1* [m]	D2** [m]
Flute (L)	10	260	0.30	0.40
Flute (R)	10	250	0.30	0.20
Piccolo (L)	10	260	0.16	0.28
Piccolo (R)	10	250	0.16	0.15
Trumpet/ Flugelhorn (L&R)	0	180	0.55	0.55
(Bass) Trombone (L)	10	180	0.40	0.30
(Bass) Trombone (R)	10	150	0.40	0.50
Violin (L)	30	180	0.20	0.20
Violin (R)	30	135	0.20	0.25

* Distance between middle of the head and the reference point on the musical instrument

** Distance between each ear and the reference point on the musical instrument

(see Figure 6.4 for an graphical explanation of the geometry)

6.4.4 Results from binaural measurements

Table 6.3 shows the measured level difference in dB between the left and right ear per instrument. The results are presented per octave band and A-weighted (broadband). For reference, the absolute A-weighted sound level is also presented, which shows that the direct sound of the instrument is above 90 dB(A) in most cases, and even up to 100 dB(A) in one case (remember that this is averaged over playing scales both in Piano and Forte). The A-weighted sound level difference at the two ears varies from -3.4 to -7.4 dB for the flutes and piccolos positioned on the right side of the head. For the trumpet and flugelhorn, a +0.7 dB and -1.7 dB A-weighted difference is found, respectively, caused by the bells being slightly off center to the left for the trumpet and off center to the right for the flugelhorn. A striking +4.7 and +4.9 dB A-weighted level difference is found for the trombones, with differences of +11 to +14 in the high frequency bands. For the violin, an A-weighted level difference is found of +2.3 dB, which was expected to be (much) higher.

Table 6.3. Measured interaural level difference per instrument, measured in an anechoic room.

Instrument	125	250	500	1000	2000	4000	8000	A-weighted		
	L-R	L-R	L-R	L-R	L-R	L-R	L-R	L	R	L-R
Flute 1		-2	-6	-7	-1	+1		86	93	-6.4
Piccolo 1			-4	-10	-1	-9	-6	90	93	-3.4
Flute 2		-3	-6	-8	-4	-3		86	93	-7.4
Piccolo 2			-5	-13	-2	-8	-9	92	97	-4.3
Trumpet	+1	+1	+1	-1	+3	+3	+4	97	96	+0.7
Flugelhorn	0	0	0	-2	-1	-2		98	100	-1.7
Bass trombone	+3	+3	+4	+6	+13	+12		96	91	+4.9
Trombone	+2	+3	+3	+6	+11	+14		97	92	+4.7
Violin	-1	0	+1	+4	0	0	+4	92	90	+2.3

Interaural level differences (ILD) have been reported earlier by Meyer [22], measured in an anechoic room, and Schmidt [7], measured in a rehearsal room. Schmidt reported a level difference of -7.4 dB for the flute and -6.7 for the piccolo. For the trumpet, values of 0 dB and +1.4 dB were reported by Meyer and Schmidt respectively, and for the trombone +3 dB and +3.8 dB. These values are (more or less) similar to what was found in this study. It is striking though, that Meyer and Schmidt found a level difference for the violin of +10 dB and +5.3 dB respectively, which is much higher than the +2.3 dB that was found in this study. But, it should be noted that the playing style of the violin player might have a large influence on the ILD. Using equation 3, we estimated that the ILD is +2.3 dB when the neck of the violin is pointing towards 330 degrees azimuth and the ILD is estimated to be + 8 dB when the neck of the violin is pointing towards 270 degrees azimuth. This might explain the differences between the various studies.

6.4.5 Model calculations of levels from the musician's own instrument

Using the sound level measured in front of the musician at 2 m distance, the directivity per angle from Pätynen & Lokki, and the geometrical parameters as presented in Table 6.2, the sound level at the musicians' ears have been estimated by equation 3. The results are compared to the actually measured sound levels. The difference between the measured and estimated values are presented in Table 6.4 for the left ear and Table 6.5 for the right ear. The differences are presented per octave band and A-weighted. For reference, the absolute A-weighted sound level is also presented. In the column to the right, the A-weighted difference between measured and estimated is presented.

Table 6.4. The difference in interaural level difference between measured and estimated, left ear.

Instrument	125	250	500	1000	2000	4000	8000	A-weighted		
	L	L	L	L	L	L	L	Meas	Estim	M-E
Flute 1		3	9	-2	-1	0		86	87	-0,7
Piccolo 1			-1	-2	3	-7	-5	90	88	1,7
Flute 2		6	7	-4	-5	-1		86	89	-3,7
Piccolo 2			0	-2	4	-4	-5	92	89	3,3
Trumpet	0	-1	3	-6	3	2	0	97	99	-2,2
Flugelhorn	0	0	4	-3	0	0		98	100	-2,1
Bass trombone	-3	-4	-1	-2	4	-1		96	97	-1,4
Trombone	-3	-2	-1	0	5	2		97	97	-0,4
Violin		2	8	-2	-1	-1	8	92	93	-0,9

Table 6.5. The difference in interaural level difference between measured and estimated, right ear.

Instrument	125	250	500	1000	2000	4000	8000	A-weighted		
	R	R	R	R	R	R	R	Meas	Estim	M-E
Flute 1		-1	9	0	-5	-6		93	93	-0,1
Piccolo 1			-3	2	-1	-3	-4	93	93	-0,3
Flute 2		2	6	-3	-7	-3		93	95	-2,2
Piccolo 2			0	5	1	-2	-2	97	94	2,2
Trumpet	-1	-2	2	-4	0	-1	-3	96	99	-2,8
Flugelhorn	0	0	5	-1	0	3		100	100	-0,5
Bass trombone	-2	-3	-2	-4	-5	-8		91	93	-2,7
Trombone	-1	-2	-1	-3	-2	-6		92	94	-1,5
Violin		4	7	-4	1	3	6	90	91	-1,0

The comparison of the measured and estimated binaural sound levels shows that, for the individual frequency bands, errors are found up to 9 dB. The mean absolute error is approximately 3 dB for the individual frequency bands. For most instruments, both positive and negative errors occur over the frequency range, except for the trombones at the right ear that show only negative errors. When looking at the A-weighted errors, for the left ear, the model overestimates the A-weighted level by 0.7 to 3.7 dB. For the right ear, the model overestimates the A-weighted level by values between 0.1 and 2.8 dB. Exceptions are the piccolos that are underestimated by 1.7 to 3.3 dB.

6.4.6 Conclusions on the validation study

Looking at the individual frequency bands, we can conclude that the model is accurate within a mean absolute deviation of 3 dB, with maxima up to 9 dB. This relatively large deviation can be caused by the fact that the musician's ear is so close to the musical instrument, that the instrument cannot be considered as a point source. Also, the directivity that is used in the model was obtained from different instruments playing a different repertoire. Additional measurements with multiple musicians and instruments could produce more uniform results. However, considering the model's purpose being to investigate the different contributions of many different aspects to the sound exposure of musicians within an orchestra, we can conclude that the prediction of the direct A-weighted sound exposure of the own instrument at the musicians ear can be done within 1.7 dB(A) average deviation with a maximum deviation of 3.7 dB(A).

6.5 Case study

Using the symphony orchestra sound level prediction model, as presented in section 6.2, and the input data, as presented in section 6.3, a calculation has been made for the violin position 8 (see Figure 6.10) as an example. For every instrument group, the direct, early reflected and late reflected sound level as received at the violin player's left ear, and the total sound level received per group at both ears, is calculated. It should be noted that the equivalent sound level is calculated over the whole musical sample, resulting in a time-average balance. From a musical point of view, this may only show an average sound level balance, not the actual balance in a certain shorter piece like a note, measure or phrase.

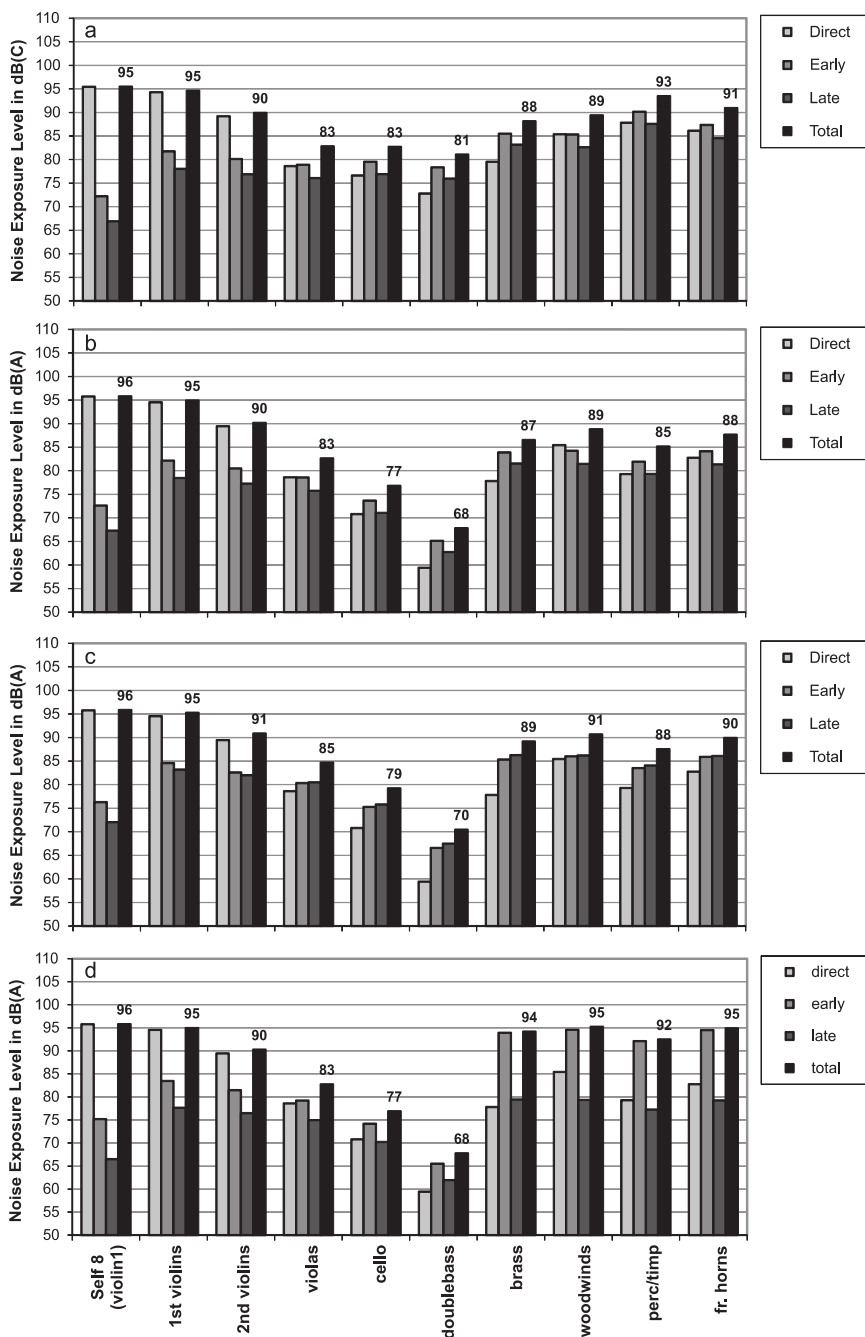


Figure 6.10. Sound level balance in dB for the violin position 8 (left ear) per instrument group, for the direct sound (direct), early reflected sound (early), late reflected sound (late), and the summation of direct, early and late (total). a: Concert Hall, C-weighted; b: Concert Hall, A-weighted; c: Rehearsal Room, A-weighted; d: Orchestra Pit, A-weighted.

In Figure 6.10, the results for the violin's position calculation are presented in four bar graphs, the first graphs 'a' and 'b' both represent the sound levels for the concert hall stage, where graph 'a' shows C-weighted values and graph 'b' shows A-weighted values. It is shown, that the C-weighted levels are 3 to 13 dB higher than the A-weighted levels for the instrument groups with more low frequency energy: cello, double bass, timpani and horns. For the instrument groups nearby position 8, the 1st and 2nd violins, the direct sound level dominates over the reflected sound level. For instruments sitting further away, the contributions of direct, early and late sound are within a range of 5 dB and reflected sound tends to dominate over direct sound. The total A-weighted levels, commonly used for judging sound exposure, show that the direct sound of the own instrument and the own group are more or less equal, with 96 and 95 dB(A) respectively. All other instrument groups, except for the cello and double bass, are 5 to 12 dB(A) lower in total level, thus contributing moderately to the sound exposure at the violin's position.

The 2nd to 4th bar graph in Figure 6.10, graphs 'b', 'c' and 'd', show a comparison of A-weighted sound levels for the concert hall (b), rehearsal room (c) and orchestra pit (d), see Table 6.1 for their properties. In this case, only the commonly used A-weighting is chosen for presentation to be able to evaluate the sound exposure. The late reflected sound level is 3.4 dB higher in the rehearsal room for all instruments, resulting in approximately 2 dB higher total levels from instruments sitting further away. The sound exposure at the violins position, caused by all instruments, increases by 0.8 dB(A). In the orchestra pit, a dramatic increase is shown for the early reflected sound coming from the players in the covered part of the pit causing an increase in total level up to 5 dB(A). The late reflected sound level is (much) lower.

The total sound level (self and all others) at the violin position is 100.1 dB(A) for the concert hall, 100.8 dB(A) for the rehearsal room and 102.8 dB(A) in the orchestra pit. A difference of almost 3 dB between on stage and orchestra pit conditions is calculated, meaning a doubling of the noise dose.⁹

In Figure 6.11, the sound level in dB(A) is presented as an average within each instrument group. The total sound level of the own instrument within the group is shown together with the received direct, early, late and total sound summed for all others. In general, sound levels are relatively high for this particular musical piece. Total equivalent sound levels are calculated ranging from 94 to 101 dB(A), even if one's own instrument is not producing levels above 85 dB(A), like the cello, double bass and the conductor. For the high strings, the own instrument sound is equally loud as the direct sound of all others. For the brass, woodwinds, timpani and French horns, the direct sound of all others appears to be the loudest component. The differences between the right and left ear are small, 1 to 2 dB(A). Because the violin players direct their left ear towards the orchestra, direct sound from all others is 1.5

⁹ Note that this is a purely theoretical case of equal repertoire and playing style. In reality, the repertoire in the orchestra pit will be different and musicians might play softer to avoid overpowering the singers.

dB(A) higher at the left ear. In a similar way, for the cello and double bass, the right ear receives up to 4 dB(A) more direct sound from others.

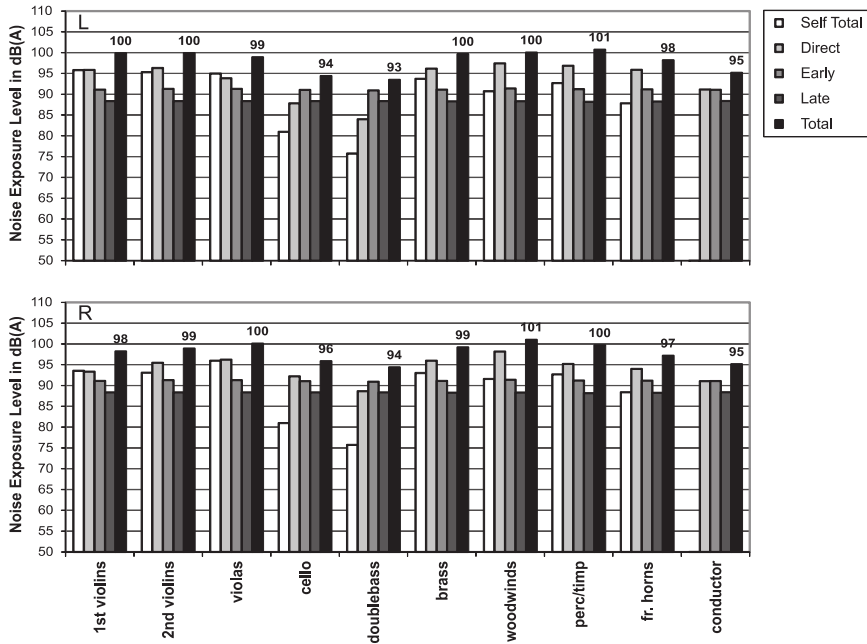


Figure 6.11. Average sound levels in dB(A) per instrument group on a concert hall stage for the left ear (L) and right ear (R), for the total own instrument sound (Self Total), the direct sound of all others (Direct), early reflected sound of all others (Early), late reflected sound of all others (Late), and the summation of self, and direct, early and late of all others (Total).

In accordance with European Directive 2003/10/EC [5], one is required to wear hearing protection above equivalent sound levels $L_{A,eq} \geq 85$ dB(A). The measured results from the individual players in the anechoic room, as presented in section 6.4, and the calculated results for the full orchestra, as presented in this section, show equivalent sound levels well above 85 dB(A), up to 100 dB(A). Peak levels could not be assessed using the model. Due to the large dynamics of music, the sound exposure should be evaluated over a larger period than one musical sample of 2 minutes. Nevertheless, if one would like to keep the daily noise dose below 85 dB(A), and the $L_{A,eq}$ is 100 dB(A) for this particular Mahler Symphony no. 1 sample (2:12 min), one should rehearse this excerpt not more than 7 times a day without wearing ear plugs, while remaining practically silent during the rest of the day. This example confirms, that sound exposure control for musicians can be quite a challenge.

6.6 Discussion

The presented work is a result of an ongoing study on the development of a model to estimate the sound levels within an orchestra. It is shown that the model has much potential for studying the influence of architectural as well as acoustical aspects on the sound levels that occur in a symphony orchestra. It is also important to mention the limitations of this model:

- The directivity of the instruments is not taken into account in the (measured) early and late reflected sound level. In a similar way, the directivity of the listener's ears have not been taken into account. (This turned out to be reasonable, see [chapter 5](#)).
- The attenuation by the orchestra is not taken into account in the measured room acoustical parameters which may result in an overestimation of the early and late reflected sound. (This turned out to be necessary, see [chapter 4](#)).
- The outcome of the model could not be validated based on actual sound level measurements. (This is dealt with in [chapter 7](#)).
- The model does not take into account how orchestra musicians might adjust their playing levels as a function of what (which levels) they hear from their own instrument/group and/or other parts of the orchestra.

In spite of these shortcomings, it is shown that the model has the potential to give valuable insights in the sound level distribution of different instruments in a symphonic orchestra. The study of the sound levels has shown to be interesting from a health point of view. To effectively control sound exposure of musicians in an orchestra, it is important to understand the contributions of the various aspects that determine the total exposure. The introduced prediction model can give insight into all these aspects, provided that the uncertainties are further reduced. In the future, it would be interesting to use the model to study the impact of screens between musicians and different orchestra setups on sound exposure. Also, other types of stage environments could be analysed and the impact of architectural elements on within-orchestra sound levels could be investigated.

The study of these sound levels has also shown to be interesting from a musical point of view. The results might be useful to study the effect of orchestra setup together with room acoustics on ensemble playing for different pieces of music. For instance, it would be interesting to investigate whether the predicted sound level balance matches the perceived loudness balance by orchestral musicians under various conditions.

6.7 Conclusion

In this chapter, a model is presented to study the direct, early reflected and late reflected sound distribution in symphony orchestras. The model for the direct sound level of the own instrument has been validated by measurements, showing an average deviation of 1.7 dB(A). The results from the measurements and model revealed that, in some cases, like the violin player, the positioning of the instrument by the musician can have a larger influence on the interaural level differences. Interaural level differences reach up to 13 dB for separate octave bands and 7.4 dB for A-weighted sound levels. This confirms that when judging musicians sound exposure, the sound levels should be studied close to each ear separately, instead of only using one position on the musicians shoulder as suggested by ISO 9612.

The distribution of the direct, early reflected and late reflected sound levels have been studied for a concert hall stage, a rehearsal room and an orchestra pit using the model. Results indicate that the contribution of each aspect (own direct sound, others' direct sound, early and late reflected sound), to the total sound level at the receiving musician's position can be in the same order of magnitude. This shows that actual sound level measurements in an orchestra could never reveal the impact of the various aspects individually, and a model, as presented in this chapter, is indispensable.

As an example, the sound level distribution has been investigated for a sample of music from Mahler Symphony no. 1 at a 1st violin's position in the orchestra. The balance between direct and reflected sound depends on the distance to the various instrument groups. The difference in acoustics of the concert hall stage and the rehearsal room is expressed by the increase in the late reflected sound level in the rehearsal room by 3.4 dB, which results in a 0.8 dB(A) higher calculated total sound exposure at the violin's position. The difference in acoustics of the concert hall and the orchestra pit is expressed by the increase in the early reflected sound level, especially for the instruments positioned in the covered part of the pit. At the violin's position, a difference in total sound level of 2.7 dB(A) between on stage and orchestra pit conditions is calculated, meaning a doubling of the noise dose.

The results from the example show that the model has potential for studying the influence of architectural as well as acoustical aspects on the sound levels that occur in a symphonic orchestra.

6.8 References

- [1] L. Beranek: The sound strength parameter G and its importance in evaluating and planning the acoustics of halls for music. *J. of the Acoust. Soc. of Am.* 129 (5) (2011) 3020-3026.
- [2] A.C. Gade: Investigations of musicians' room acoustic conditions in concert halls. *Acustica*, 69 (1989) 193-203 and 249-261.
- [3] L. Nowicki, W. Prinz, M. Grosjean, B.H. Repp, P.E. Keller: Mutual adaptive timing in interpersonal action coordination. *Psychomusicology: Music, Mind, and Brain*, vol 23(1) (2013) 6-20.
- [4] A.C. Gade: Acoustics for symphony orchestras; status after three decades of experimental research. *Building Acoustics*, 18 (2011) 181-206.
- [5] Directive 2003/10/EC of the European Parliament and of the Council of 6 February 2003 on the minimum health and safety requirements regarding the exposure of workers to the risks arising from physical agents (noise). *Official Journal of the European Union* L42, 38-44.
- [6] E.J.M. Janssen, H.W. Helleman, W.A. Dreschler, J.A.P.M. de Laat: Noise induced hearing loss and other hearing complaints among musicians of symphony orchestras. *Int. Arch. of Occ. and Env. Health*, 82 (2009) 153-164.
- [7] J.H. Schmidt: Hearing changes in classical musicians and risk factors. PHD Thesis, Faculty of Health Sciences, University of Southern Denmark, 2011.
- [8] Z. Schärer Kalkandjiev, S. Weinzierl: The influence of room acoustics on solo music performance: an empirical case study. *Acta. Acust. united Ac.* 98 (2013) 433-441.
- [9] M.A. Rodrigues, M.A. Freitas, M.P. Neves, M.V. Silva: Evaluation of the noise exposure of symphonic orchestra musicians. *Noise and Health*, Vol 16 (2014) 40-46.
- [10] J. Pätynen, T. Lokki: Directivities of symphony orchestra instruments. *Acta. Acust. united Ac.*, 96 (2010) 138-167.
- [11] J. Pätynen: Directivities of orchestra instruments for auralisation. *Proceedings of the EAA Symposium on Auralization*, Espoo, Finland, 15-17 June 2009.
- [12] J.J. Dammerud, M. Barron: Attenuation of direct sound and the contributions of early reflections within symphony orchestras. *J. Acoust. Soc. Am.* 128 (4) (2010) 1755-1765.
- [13] J. Pätynen, V. Pulkki, T. Lokki: Anechoic recording system for symphony Orchestra. *Acta. Acust. united Ac.*, 94 (2008) 856-865.
- [14] R.H.C. Wenmaekers, C.C.J.M. Hak, L.C.J. van Luxemburg: On measurements of stage acoustic parameters - time interval limits and various source-receiver distances. *Acta. Acust. united Ac.*, 98 (2012) 776-789.
- [15] J.J. Dammerud, M. Barron, E. Kahle: Objective assessment of acoustic conditions for symphony orchestras. *Proceedings of the International Symposium on Room Acoustics, ISRA 2010*, Melbourne, Australia.
- [16] A.C. Gade: Practical aspects of room acoustic measurements on orchestra platforms. *Proceedings of 14th ICA*, Beijing, 1992.
- [17] C.C.J.M. Hak, R.H.C. Wenmaekers, J. Hak, L.C.J. van Luxemburg: The source directivity of a dodecahedron sound source determined by stepwise rotation. *Proceedings of Forum Acusticum 2011*, Aalborg, Denmark.

- [18] C.C.J.M. Hak, R.H.C. Wenmaekers, L.C.J. van Luxemburg: Measuring room impulse responses: impact of the decay range on derived room acoustic parameters, *Acta. Acust. united Ac.*, 98 (2012) 907-915.
- [19] R.H.C. Wenmaekers, C.C.J.M. Hak: Early and late support measured over various distances: the covered versus open part of the orchestra pit. *Building Acoustics*, 4 (2013) 313-321.
- [20] R.H.C. Wenmaekers, L.J.W. Schmitz, C.C.J.M. Hak: Early and late support over various distances - rehearsal rooms for wind orchestras, *Proceedings of Forum Acusticum*, Krakow, 2014.
- [21] J. Meyer: *Acoustics and the performance of music*, 5th ed., New York, Springer, 2009.
- [22] J. Meyer: Die Problematik des gegenseitigen Hörens bei Musikern. *Proceedings of the 12th Tonmeistertagung*, München, 1981.

Chapter 7

Orchestral musicians' sound exposure

Original title: Why orchestral musicians are bound to wear earplugs: about the ineffectiveness of physical measures to reduce sound exposure

Authors: R.H.C. Wenmaekers, B. Nicolai, M.C.J. Hornikx, A.G. Kohlrausch
Submitted

Abstract

Symphony orchestra musicians are exposed to noise levels that put them at risk of developing hearing damage. This study evaluates the potential effectivity of common control measures used in orchestras on open stages with a typical symphonic setup. A validated acoustic prediction model is used that calculates binaural sound exposure levels at the ears of all musicians in the orchestra. The model calculates the equivalent sound levels for a performance of the first 2 minutes of the 4th movement of Mahler's 1st symphony, which can be considered representative for loud orchestral music. Calculated results indicate that risers, available space and screens at typical positions do not significantly influence sound exposure. A hypothetical scenario with surround screens shows that, even when shielding all direct sound from others, sound exposure is reduced moderately with the largest effect on players in loud sections. In contrast, a dramatic change in room acoustic conditions only leads to considerable reductions for soft players. It can be concluded that significant reductions are only reached with extreme measures that are unrealistic. It seems impossible for the studied physical measures to be effective enough to replace hearing protection devices such as ear plugs.

7.1 Introduction

Most classical musicians are regularly exposed to a daily noise exposure level above 80 dB(A) and therefore risk hearing damage [1]. The relation between noise exposure and hearing damage has been investigated for orchestral musicians in the past 25 years [2]. Musicians appear to perform better in hearing tests than a comparable general population, possibly due to a selection bias [3]. Nevertheless, musicians do have a work-related risk of developing hearing disorders such as hearing loss [3], tinnitus and presbycusis [4, 5]. Researchers have not always found alarming hearing damage among classical musicians [6]. Still, most researchers do advise that sound exposure levels should be reduced [3, 5, 7, 8] if only because any unnecessary hearing damage is undesirable [9]. The current chapter is mostly concerned with the reduction of equivalent sound levels and the following literature review will focus on research dealing with causes and possible solutions.

A number of researchers have measured the sound exposure of musicians which could reveal the effect of the physical environment on sound levels. However, it is difficult to compare results across studies because time averaging methods vary [10]. Researchers used dosimeters attached to the shoulder or microphones on stands, while [11] promotes the use of miniature microphones attached near both ears of the musicians. Also, factors such as changing seating position and different repertoire make the interpretation of results difficult. For instance, O'Brien et al. [12] found that the long-term exposure in a symphony orchestra varied between three different venues by 0 to 4 dB, depending on the studied musician. The mean exposure was slightly lower in the rehearsal room compared to the concert hall and higher in the orchestra pit. In the orchestra pit, a generally more intense repertoire was played that could explain the higher sound levels. The lower levels in the rehearsal room are explained by breaks with speech that reduce the equivalent sound levels. According to Gade [13], musicians might play more passionately and thus louder during performances which could also explain lower sound levels during rehearsals. Finally, the three venues likely also vary in room acoustic conditions possibly increasing sound levels in the rehearsal room and orchestra pit. O'Brien et al. [14] also investigated $L_{A,eq}$ during individual rehearsal (averaged over 20 min.), which were higher than $L_{A,eq}$ in the orchestra (averaged over many performances) by +3 to +5 dB for high strings and up to +7 dB for flute and brass. The higher sound levels during individual rehearsal are likely caused by more intense playing with fewer breaks compared to orchestral rehearsal or performance. Exceptions are the cello and contrabass who receive -4 and -9 dB lower levels, respectively, because these instruments are less powerful in the mid to high frequencies compared to other instruments in the orchestra. These results suggest that most musicians would be better off in terms of sound exposure per unit of time playing in the orchestra instead of practicing individually, which is highly counterintuitive. To investigate the contribution of the own instrument to the sound levels in the orchestra, Schmidt et al. [11] compared active and inactive periods. It was shown that most instrument groups, except low strings, have a significantly higher exposure when playing. It was concluded that, even in the orchestra, musicians are primarily exposed to their

own instrument. If the own instrument is contributing most to the total sound exposure, then rotating positions in the same section would not be an effective measure to control sound exposure. Indeed, Schmidt et al. [11] did not find any statistical difference in exposure among musicians within the same group.

Other studies have been dedicated to the effect of screens that are used to shield musicians from loud instruments. Gade et al. [15] sent questionnaires to 46 opera houses. 23% of the orchestras used screens in the orchestra pit and screens were only used if there was sufficient space (2 m² per musician). Camp and Horstman [16] measured the effect of a free standing plastic screen (unknown dimensions) between two neighbouring positions in an orchestra pit using a loudspeaker and a single microphone and found reductions of -1, +2, -8, -9, -13 and -15 dB for the mid pure tone frequencies of the octave bands 125 to 4,000 Hz. The authors of the current work repeated this experiment with a floating 12 mm wooden panel with dimensions 1 x 1 m². A head and torso simulator was positioned with its ears at 30 cm distance from the middle of the screen and a directional sound source on the other side at the same distance. Impulse responses were measured with and without the screen and the difference in sound pressure level was determined in octave bands. Good agreement was found with the measured data from Camp and Horstman for most octave bands except for the 500 Hz band that showed considerable smaller values: -3 dB instead of -8 dB. According to [17], a screen with absorptive material that is wrapped around the back of the head shows similar reductions while additionally avoids sound being reflected back to the source. The screens on stands are often found ineffective because musicians have to sit uncomfortably close to them [18, 19]. As an alternative, O'Brien et al. [19] experimented with a tall shielding barrier. The measured attenuation by the screen was 3 to 4 dB(A) with two trumpet players at 1.5 m distance to the screen and the microphone on the other side at 0.5 m from the screen. Results from these studies should be interpreted with care because they do not include the sound of other players that can reduce the effect of screens. In the study by Libera and Mace [18], both higher and lower sound levels were observed after introducing screens while the whole orchestra was playing in a small rehearsal room. Martinez [20] measured reductions of only 1 dB(A) on both sides of a large 10 m wide and 2.3 m high barrier between the strings and the other sections, also in a rehearsal room. In contrast, O'Brien et al. [19] measured a reduction of 4 to 6 dB(A) while the whole orchestra was playing in a pit when introducing the barrier screen between a cello player and the trumpets.

Chasin [21] mentions the possible effect of risers on the reduction of sound exposure. He measured a reduction in sound pressure level of 5 to 7 dB in the high frequencies at positions in front of the trumpets when placing trumpet players on risers. This reduction is achieved because of the strong directionality of the trumpet at high frequencies. However, Eaton and Gillis [22] point out that the mid-frequency components (say, 500-2000 Hz) of the trumpet are most powerful compared to the higher frequency bands. As a result of a lower directivity at mid frequencies, the actual reduction of overall sound exposure by placing trumpets on risers might be much lower than suggested by Chasin.

Researchers have suggested room acoustic guidelines to control sound levels, focussing on orchestra pits and rehearsal rooms instead of concert hall stages. Tennhardt & Winkler [23] give detailed suggestions for rehearsal room design. They noted that, in contradiction to what was believed earlier, the goal should not be to reproduce the same acoustic conditions as in the performance venue. They suggest a reverberation time (T) as low as 0.8 to 1.1 s which would allow smaller rooms without increased sound levels compared to concert halls. Gade et al. [15] found that most orchestra pits had no high frequency absorption materials inside and 70% had a wooden finish. They suggest that absorption in the pit should indeed be avoided because it would lead to reduced support and musicians would play louder if they cannot hear themselves or colleagues. For rehearsal rooms, Gade [13] suggested that there might be less need to play powerful due to more support in smaller rooms. However, this can only lead to an improvement if the increase in sound levels due to the smaller room is less than the reduction in sound power produced. Therefore, Gade [13] suggests to introduce at least 8 m² of equivalent sound absorption area, A , per musician in symphony orchestra rehearsal rooms, leading to a room volume (V) between 5.000 and 10.000 m³ for a 100-person symphony orchestra. Some researchers have used the stage acoustic parameters ST_{early} or ST_{late} to derive recommendations for the room volume of rehearsal rooms. ST_{early} describes the dB ratio of early reflected sound energy between 20 and 100 ms and the direct sound energy, both measured at 1 m distance for the sound source. Similarly, ST_{late} describes the dB ratio of late reflected sound energy after 100 ms and the direct sound energy. Pompoli et al. [24] suggested the amount of early reflected sound, measured by the stage acoustic parameter ST_{early} , as a guideline to control sound levels in rehearsal rooms. Considering a suitable value for ST_{early} of -12 dB (+/-2 dB), rehearsal rooms require a V between 750 and 2,500 m³. Additionally, Wenmaekers et al. [25, see section 9.3] used the late reflected sound level (ST_{late}) to obtain requirements for V and T , resulting in $V \geq 2,000$ m³ for $T \geq 1.0$ s. Another guideline is presented in the Norwegian standard NS-8178 “Acoustic criteria for rehearsal and performance spaces”, see Rindel [26]. A recommended range of V and T is given for powerful acoustical music with a maximum V of 3,000 m³. In an informative section of the standard, a prediction model is presented that calculates the sound levels in the room for a given ensemble to obtain guidelines for room volume. In contrast, Lautenbach and Vercammen [27] suggest that large rooms ($V > 8,000$ m³) are highly preferred over relatively small rooms ($V = 4,000$ m³). Figure 7.1 summarizes all mentioned requirements. It is clear that the mentioned strategies lead to different requirements for V and T . With a reverberation time of 1.1 s, four guidelines overlap at a volume of approximately 2,000 m³. Gade’s guideline results in a much large volume (4,500 m³) when the reverberation time would be 1.1 s. Only for reverberation times around 1.7 s, Gade’s guideline and the guideline based on a ST_{late} of -13 dB are similar.

If measures at the source are not sufficient to reduce the daily noise exposure level below 85(A), individual hearing protection must be worn [1]. Moulded ear plugs can easily reduce sound levels by 20 dB or more. However, several studies have shown that musicians are

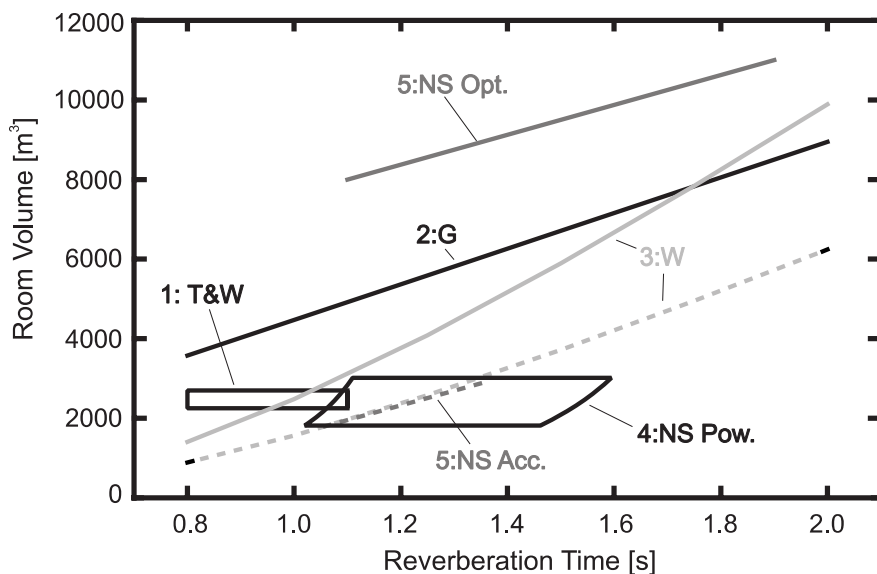


Figure 7.1. Summary of guidelines suggested in literature for rehearsal rooms with an orchestra comprising 90 musicians:

- 1: Tennhardt and Winkler [23], 25-30 m³ per musician and $T = 0.8-1.1$ s and $V_{\min} = 2,000$ m³;
- 2: Gade [13], $A = 8$ m² per musician;
- 3: Wenmaekers et al. [25], $ST_{\text{late}} = -13$ to -15 dB, predicted by $10 \log (312T/V) - 6/T$;
- 4: NS-8178 [26], T and V range for powerful music, $V_{\min} = 1,800$ m³ and $V_{\max} = 3,000$ m³;
- 5: NS-8178 [26], prediction model $L_{p,A,\text{diff}} = L_{w,A} + G - 31$ dB, symphony orchestra $L_{w,A} = 115$ dB, G is 6 dB for optimal conditions ($L_{p,A,\text{diff}} = 90$ dB at forte) and G is 11 dB for acceptable conditions ($L_{p,A,\text{diff}} = 95$ dB at forte).

reluctant to wear ear plugs or other hearing protection devices (HPDs) [28, 29, 30], and mostly wear them when hearing problems already exist [3]. The main reasons are that HPDs hinder the own performance and make it difficult to hear others play [26]. Nevertheless, there are reports of successful ear plug use by musicians [31]. If HPDs are used, they are worn during group or orchestral rehearsal and very rarely during individual rehearsal because ‘musicians feel their own instrument is not noisy, but it is the neighbouring instruments that cause the problems’ [28]. They might even think that they themselves are worse off than their neighbours causing the noise: ‘It is a logical assumption that the players directly in front of the trumpets are exposed to a much higher level than the trumpet players themselves; however, this was routinely not the case.’ [12].

It is clear that there is a desire to control sound levels in orchestras preferably by using physical measures. However, sound exposure measurements have not been conclusive on their effectiveness. Small scale experiments with screens or barriers are either too optimistic or too specific to lead to general conclusions. Various acoustic guidelines lead to different solutions and their effect on sound exposure has not been validated. In the current chapter,

the prediction model as presented in the previous chapter is used to investigate the effectiveness of the following common physical measures that aim at controlling sound levels in orchestras:

- Increasing the distance between musicians
- Changing the height of risers
- Using absorptive screens
- Changing the acoustic properties of the room
- Rotating the position of musicians

In the next section 7.2 the modelling method will be briefly summarised accompanied by a comparison to measurements. In section 7.3, the model is used to study the possible control measures. The chapter ends with a conclusion.

7.2 Method

7.2.1 Sound level distribution model

A sound level distribution model for symphony orchestras is used as presented by Wenmaekers and Hak [32, see [chapter 6](#)], updated and programmed in Matlab by Nicolai [33]. A schematic overview of the model is given in Figure 7.2. The model will be briefly summarised here. See [chapter 6](#) [32] for details and equations and the thesis [33] for more background information.

The model calculates direct sound, early and late reflected sound separately, using anechoic recordings to obtain the sound power of each instrument. The binaural direct sound level is calculated analytically. At the source, the directivity characteristics of each instrument and the geometry of the orchestra members' seating positions are used to determine the directional sound power. The interference of the floor reflection for low frequencies and the attenuation by the orchestra members at high frequencies are taken into account based on measured data [30]. At the receiver position, directional weighting is applied for the two ears using a Head Related Transfer Function (HRTF) measured with microphones in front of the ears of a dummy head. A special contribution is the modelling of the direct sound of the own instrument. The distances and angles between the estimated acoustic source centre of each instrument and each ear have been measured to be used in the model.

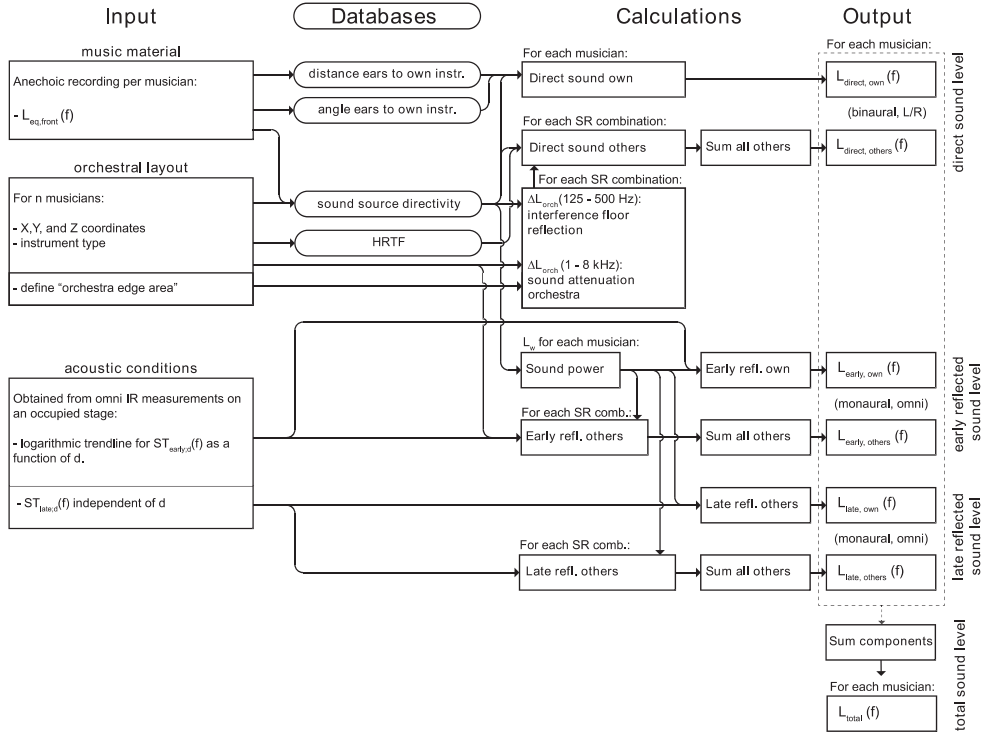


Figure 7.2. Overview of the sound level prediction model by Wenmaekers and Hak [32, see [chapter 6](#)], updated and programmed in Matlab by Nicolai [33]. Figure 3.2 in [33].

Secondly, the monaural sound level from reflected sound wave contributions is calculated based on impulse response measurements on occupied stages at various distances using omnidirectional transducers [34, see [chapter 4](#)]. Stage acoustic parameters $ST_{early,d}$ (distance dependent) and $ST_{late,d}$ (fixed values) are calculated which use a time point of 103 ms relative to the time of emission to separate early from late reflections [35, 36 see [chapter 2](#)]. The ST parameters compare the reflected sound energy to the sound power measured in a reverberation room [37, see [chapter 3](#)]. The directivity of the instrument is not taken into account for the calculation of the sound level of reflected sound, which has shown to have a negligible influence ($< 2 \text{ dB}$) in the frequency bands with the largest contribution to the A-weighted sound level (500-1000 Hz) [38, see [chapter 5](#)]. In the model, the sound power of the instruments is combined with the measured ST parameters to obtain the absolute early and late reflected sound levels for every combination of two musicians. The monaural sound energies are summed with the binaural direct sound energy at each separate ear to obtain the total sound level per ear. Equivalent sound levels can be obtained for the whole Mahler piece or shorter time intervals can be studied such as the running average of L_A .

To estimate the effect of screens, the musician to musician direct sound is attenuated by the reduction of a single screen (-1, +2, -3, -9, -13 and -15 dB for the octave bands 125 to 4,000 Hz, see section I). The model does not take into account distance dependent screening, which is reasonable because the receiver is very close to the screen (0.3 m) compared to the source at larger distance. This simplification results in an overestimation of the screening effect at larger mutual distance, but because these levels are less dominant in the total level it is expected to have a small influence on total levels. It is assumed that the screen has no influence on reflected sound levels, which is a reasonable assumption in the case of sound absorbing screens.

7.2.2 Measurements

Available anechoic recordings of each separate instrument from the first 2 minutes of the 4th movement of Mahler's 1st symphony titled 'Stürmisch bewegt' are used as an input for the model [39]. The piece is a typical example of a loud passage with all instruments simultaneously and alternatingly playing. Measurements have been performed with a symphony orchestra that played the same music to validate the model. Besides, scales were measured played by individual players and whole instrument groups. Ten musicians that volunteered to play individually, see Figure 7.3, were equipped with binaural DPA 4060 microphones positioned 1–2 cm lateral to the entrance of the ear canal of both the left and right ear using custom-made ear holders. Both the music and a calibration tone were recorded with a TASCAM DR-40. The digital signal processing was performed in MATLAB 2014b. The expected accuracy of the sound level measurements is ± 2 dB, which approximates the tolerance of a Type 2 sound level meter in the 500 and 1,000 Hz octave bands where sound levels in the orchestra have shown to be dominant [40].

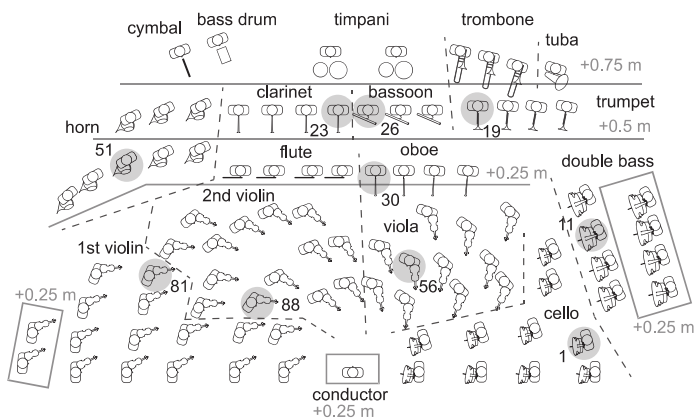


Figure 7.3. Floor plan of the orchestra in MGE with the positioning the 10 musicians that were measured (grey circles). The numbers are used to define the sound source in the model. Figure 4.3 in [33].

Measurements were undertaken in three different venues during a rehearsal on stage: a concert hall denoted ‘MGE’ ($T_{500-1k}=1.8$ s, $V=14,400$ m³), a theatre with an orchestra shell denoted ‘VTM’ ($T_{500-1k}=1.3$ s, $V=12,000$ m³) and a theatre with an orchestra shell and electroacoustic enhancement denoted ‘PDB’ ($T_{500-1k}=1.6$ s, $V=13,000$ m³). The 95-person orchestra occupied approximately 200 m² in each hall. Surprisingly, after performing all measurements it turned out that the rooms had almost equal values related to early and late reflected sound, namely $ST_{early,d} = -15.5 \pm 0.7$ dB and $ST_{late,d} = -17.9 \pm 0.3$, respectively.

7.2.3 Comparison for the scales

Figure 7.4 shows the calculated and measured $L_{A,eq}$ as a result of the 1st violin player (no. 81) playing a scale. The $L_{A,eq}$ distribution is shown over the positions in the orchestra for both ears together with the interaural level difference (ILD). The sound power of the individual player is predicted using the measurement at his right ear. As can be seen in the scatter plots, the model and measurements show a similar large difference between $L_{A,eq}$ at the player’s ear and those of the other musicians with a slight decay over distance for the other musicians.

The scale was played individually by the ten musicians in 3 different halls. The mean absolute deviation (MAD) between calculated and modelled $L_{A,eq}$, averaged over the 9 inactive players, varies between 1 and 4 dB for the different instruments and halls. The best agreement between measurement and model is observed for the viola, French horn and clarinet with $MAD < 2$ dB and standard deviation (SD) < 1.5 dB over the 9 receiver positions for all 10 source positions in 3 venues. The 1st violin, 2nd violin, bassoon and trumpet show a slightly larger $MAD < 2.5$ dB with $SD < 2$ dB. The cello, double bass and oboe show a poorer agreement with the model, with a maximum MAD of 3.8 dB for double bass in PDB.

In the model, it is assumed that all players with the same part play equally loud. For the scale experiment, each musician in a group is modelled using the sound power of the individual player. The modelled group results are compared to the measurements with groups. The overall results are similar to the situation with scales played individually, with $MAD < 4$ dB in most cases. Two exceptions are the 1st violin and the double bass groups with a MAD up to 8 dB, which suggests that the individual player’s colleagues in these groups played much louder than the individual player did. These exceptions demonstrate a limitation of the experiment; it seems difficult for other players to reproduce the individual player’s strength for a single scale. Extensive results of the scale experiment can be found in the appendix of [33].

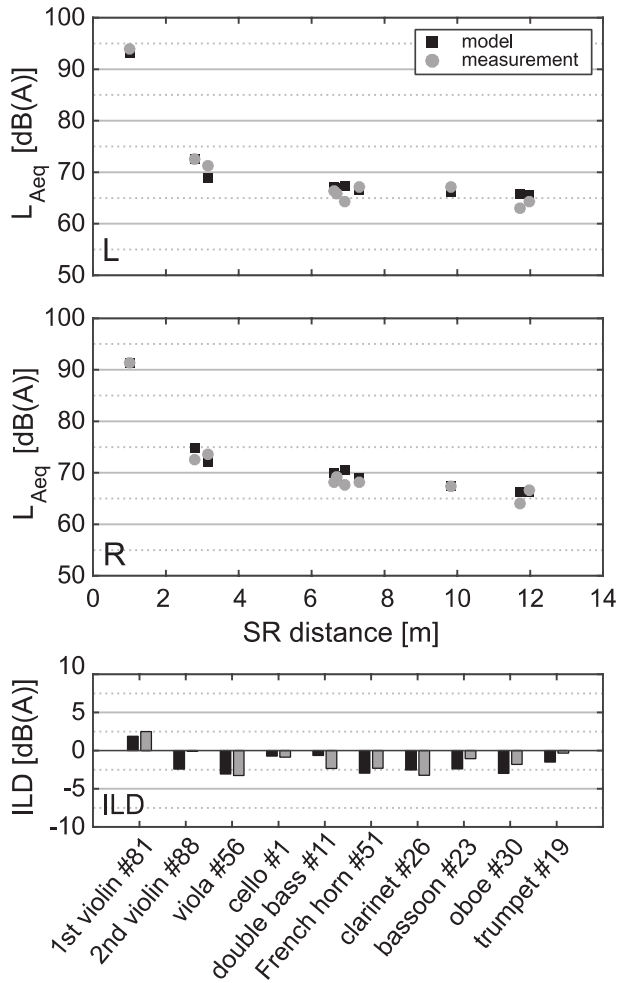


Figure 7.4. Modelled and measured sound exposure level $L_{A,eq}$ as a result of a scale played individually by the 1st violin player 81 in MGE. Model and measurement results for the left ear (L), right ear (R) as a function of SR distance and ILD per musician. Figure 5.4 in [33].

7.2.4 Comparison for the symphony

The 85 bars of the Mahler piece have been divided into 46 short excerpts. An example of the $L_{A,eq}$ per excerpt for the measured and modelled cello and trumpet is shown in Figure 7.5. The MAD in $L_{A,eq}$ has been analysed at the 10 receiver positions in the orchestra for each hall using all excerpts (in total $46 \times 10 \times 3$). The MAD per excerpt per hall (46×3), averaged over the 10 receiver positions, is within a range of 1 to 6 dB. The majority of the positions

show a $MAD < 3$ dB for both ears. The ILD has a deviation between model and measurement lower than 2 dB in most cases. In general, the deviations per excerpt are consistent for the three venues. This indicates that specific passages of the orchestra's performance might consistently be interpreted differently from the player(s) in the anechoic room. Extensive results of the symphony experiment can be found in the appendix of [33].

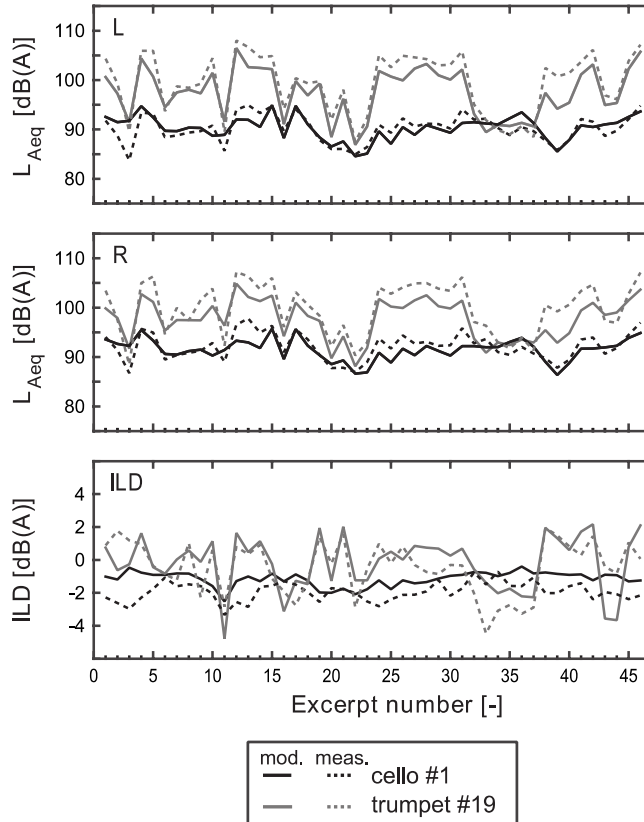


Figure 7.5. $L_{A,eq}$ and ILD as a function of 45 excerpts for the 85 bars of the Mahler piece played in MGE. Modelled (solid) and measured (dashed) results for the cello (black) and trumpet (grey). Figure 5.10 in [33].

In order to get an overview, the $L_{A,eq}$ per musician for the complete Mahler piece is presented in Figure 7.6. Absolute values show that the lower measured $L_{A,eq}$ at the cello and contrabass are well predicted by the model. For 65% of the 60 microphone positions in total, the $L_{A,eq}$ difference between the calculated and measured values is below the expected accuracy of the measurement (± 2 dB). The model does not structurally over- or

underestimate the sound exposure. An interesting outlier is the French horn section in MGE. Listening to the recordings reveals that they played with much more expression in this hall compared to other halls. This likely caused the 6 dB higher $L_{A,eq}$ measured at the French horn player's ears and 3-5 dB higher levels at the right ear of close others. The prediction for ILD is mostly in line with the measurements, with the exception of the 2nd violin player whose left ear was relatively close to the instrument.

For the modelled results, both the total and the separate own instrument $L_{A,eq}$ are presented in Figure 7.6. As expected, the contribution of the own instrument is low for the cello and contrabass. For the other players, the own instrument's $L_{A,eq}$ is within 10 dB from the total $L_{A,eq}$. This indicates that both the own instrument and the other instruments influence the total sound exposure level when averaged over active and inactive periods.

The modelled results are very similar for the three different halls. The early and late reflected sound levels were almost equal and influence the model output by not more than 0.5 dB. The different orchestral layouts have a predicted influence below 1.5 dB. The larger differences found in the measurements of different venues suggest that factors not included in the model must have influenced the sound levels more. Nevertheless, in many cases the model is sufficiently accurate to predict the absolute sound levels within 2 dB with a maximum deviation of 6 dB. In the next section the model will be applied to estimate the effectiveness of sound exposure control measures. The model was validated for the prediction of absolute sound levels in the three different rooms but not validated for the configurations that are calculated in the next section. As we will see, the calculated differences are often smaller than 2 dB(A) and the question is whether such small differences can be measured significantly. A repeated measurement of the sound pressure level at positions within the orchestra, playing a 3 minute excerpt twice for the same conditions with a 15 minute coffee break in between, showed differences between 0 and 0.3 dB(A). Even though this repeatability might be low enough to measure significant sound level differences between different stage conditions, this was not further investigated in current research. Therefore, for the model, it is assumed that if it predicts differences between configurations larger than 2 dB the change in sound exposure can be considered significant, because this is the overall precision of a measurement.

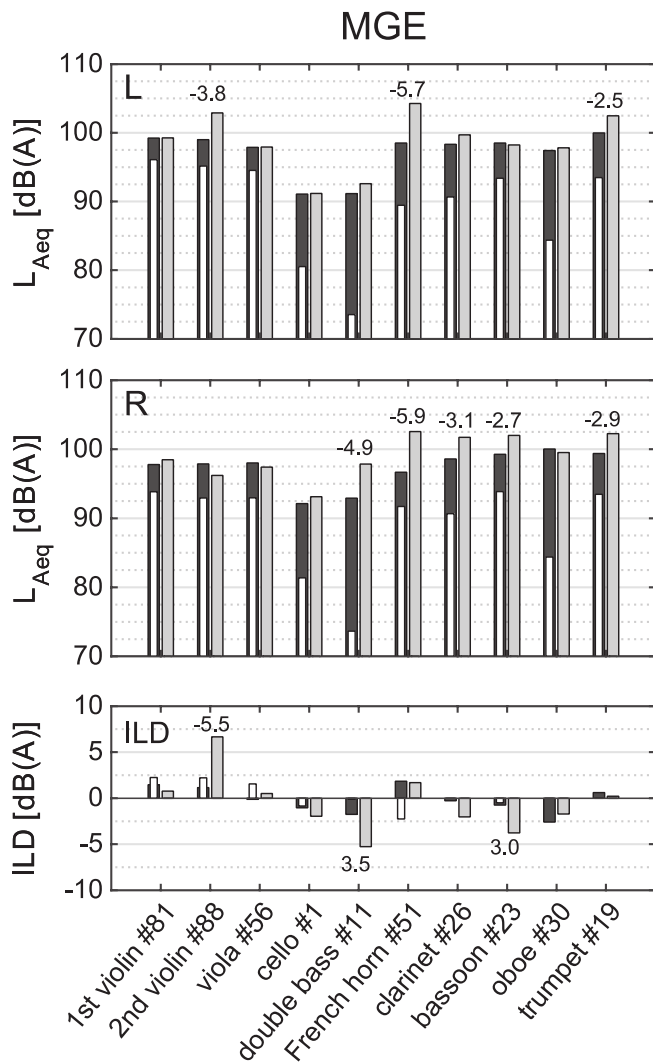


Figure 7.6a. Modelled and measured sound exposure level $L_{A,eq}$ for the left ear (L), right ear (R) and interaural level difference (ILD) as a result of the Mahler piece modelled using anechoic recordings (Pätynen et al., 2008) and played in **MGE**. For the modelled results, both the total $L_{A,eq}$ (dark grey) and the own instrument $L_{A,eq}$ (white bar) are presented. Numbers indicated differences between modelled and measured results larger than 2 dB, which is the expected accuracy of a sound exposure measurement. Figure 5.14 in [33].

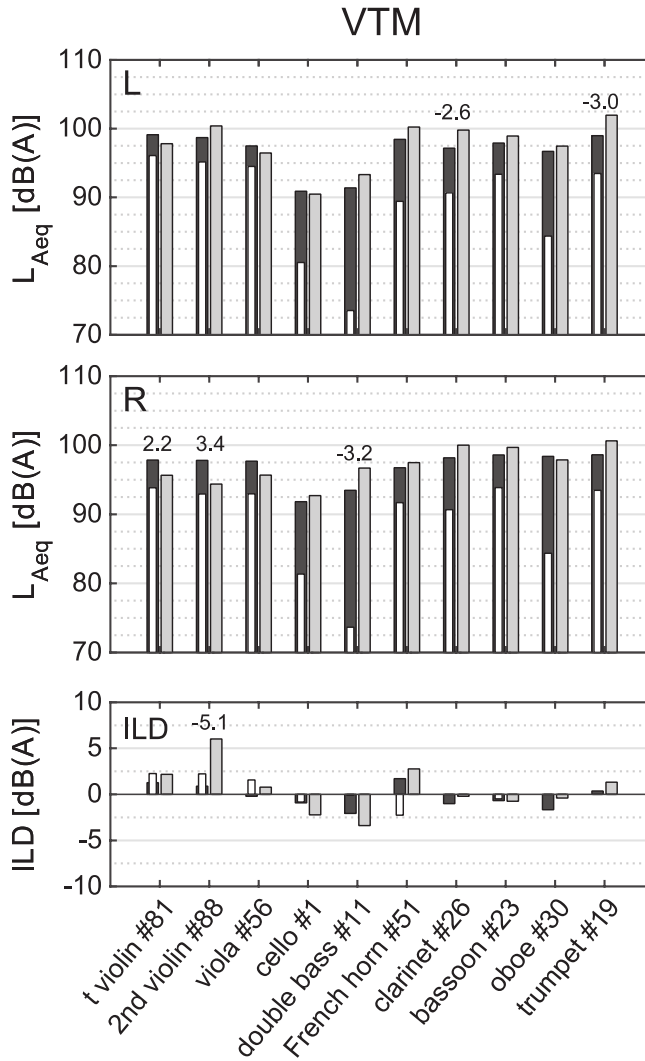


Figure 7.6b. Modelled and measured sound exposure level L_{Aeq} for the left ear (L), right ear (R) and interaural level difference (ILD) as a result of the Mahler piece modelled using anechoic recordings (Pätynen et al., 2008) and played in **VTM**. For the modelled results, both the total L_{Aeq} (dark grey) and the own instrument L_{Aeq} (white bar) are presented. Numbers indicated differences between modelled and measured results larger than 2 dB, which is the expected accuracy of a sound exposure measurement. Figure 5.14 in [33].

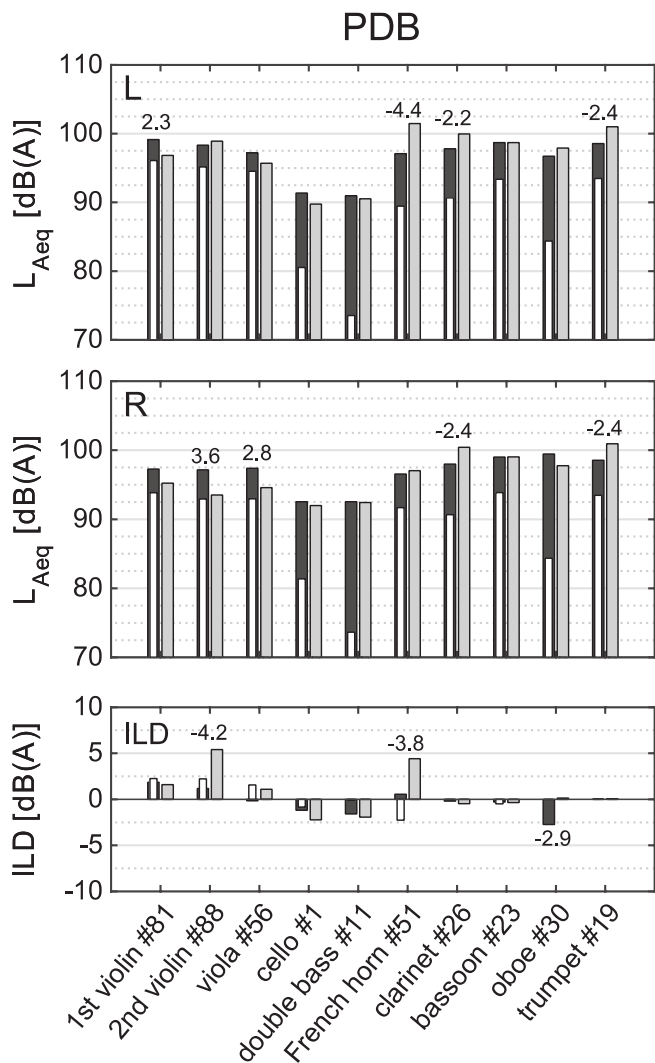


Figure 7.6c. Modelled and measured sound exposure level L_{Aeq} for the left ear (L), right ear (R) and interaural level difference (ILD) as a result of the Mahler piece modelled using anechoic recordings (Pätynen et al., 2008) and played in **PDB**. For the modelled results, both the total L_{Aeq} (dark grey) and the own instrument L_{Aeq} (white bar) are presented. Numbers indicated differences between modelled and measured results larger than 2 dB, which is the expected accuracy of a sound exposure measurement. Figure 5.14 in [33].

7.3 Effectiveness of control measures

7.3.1 Configurations

The effectiveness of a number of control measures has been investigated using the model. As a reference configuration, the concert hall model MGE is used including its orchestra layout (approximately 2 m²/musician) with risers (0.25, 0.5 and 0.75 m), see Figure 7.3, and $ST_{early,d} = -16$ dB and $ST_{late,d} = -18$ dB in occupied conditions. The following configurations were tested:

- To investigate the effect of available space, the orchestra layout is scaled from 2 m²/musician to 1.5 and 2.5 m²/musician.
- The effect of risers is investigated by multiplying the height by a factor of 2 and 4 and by removing the risers as a whole. This leads to a maximum height of 3 m of the brass and percussion in the last row.
- A 1 x 1 m² screen is considered that is positioned at 0.3 m behind a musician. Equal screen attenuation is assumed for those musicians in the shadow of the screen. Besides, to estimate the maximum possible reduction by screens an extreme hypothetical case is modelled as if the receivers are fully surrounded by such screens. The acoustic screening by music stands is included in the attenuation factor for sound passing through the orchestra [34] and not modelled separately.
- The acoustic properties of the room are changed by increasing the early and late reflected sound levels ($ST_{early,d}$ and $ST_{late,d}$) by +6 dB, and –6 dB to simulate relatively large acoustic interventions (+ 6 dB corresponds to a small rehearsal room < 1,000 m³ and -6 dB to a dry room such as a drama theatre [34].
- The effect of rotating the position of musicians is investigated for the three spacing conditions by evaluating the variation among players within in the same strings section.

The control measures were not tested for an orchestra pit environment because the model was not validated under these conditions. The effectiveness of control measures in an orchestra pit might be different from that on an open stage.

7.3.2 Results and discussion

Table 7.1 shows the calculated change in sound exposure for the Mahler excerpt at ten musicians' positions in the orchestra for the different stage configurations. Results are rounded to one decimal place to be able to appreciate the small differences between musicians and between configurations. This does not necessarily reflect the accuracy of the model, which has not been validated for predicting differences in sound pressure level.

Calculated results indicate that the riser height influences the sound exposure levels by less than 0.5 dB in most cases. Changing the distance between players also has a limited influence on the calculated sound exposure with an average of +/- 0.6 dB (positive or negative sign for decrease or increase of space). Similar to the effect of risers, mostly the direct sound

level is influenced by changing the available space. The result is a maximum reduction of only 1 dB when the 95 musicians occupy 250 m² instead of 200 m², which in practice would be a substantial increase of space.

The impact of single screens behind musicians is low, in most cases sound exposure is reduced by less than 1 dB. Only the violins' left ears and the trumpet player's ears show a higher reduction up to 1.5 dB. Drawing comparisons of results to other studies is difficult because conditions are different, such as orchestra setup, room conditions and repertoire etc. Nevertheless, our values are in line with findings by Martinez [20] showing only 1-dB reduction by a large barrier in the middle of the orchestra. The 4 to 6-dB reduction measured by O'Brien et al. [19] at a cello player in front of trumpet players might be an optimistic case where the loudest players sit behind the softest player. In the modelled orchestra, low string players sit at a larger distance from brass players reducing the impact of screens.

The calculated results for the extreme case with screens completely surrounding the musician (at only 0.3 m distance) should be interpreted with care, because the scenario is not realistic. The highest sound reduction by such screens is calculated for the woodwind and brass players with an average of 3 dB. This can be explained by the fact that their close neighbours are in total louder than the player's own instrument. Less sound reduction is calculated for the high string players with an average of 2 dB. Their own direct sound is louder compared to their close neighbours' total, which is especially the case for the left ear near the string instrument. An asymmetrical reduction is also calculated for the low strings with 1 dB on average. In this case, the louder instruments (brass) are sitting on the right side which makes screens 1 dB more effective on this side.

While surrounding screens would be most effective near the brass and woodwinds, changing room acoustic conditions is most effective for those who play softest, namely the low strings. This is because their own and their neighbours' direct sound level are relatively weak compared to the sound level of early and late reflections. Decreasing the early and late reflected sound levels by 6 dB results in an average reduction of 3 dB (left ear) and 2 dB (right ear) for the low strings. At all other instrument positions the reduction is below 1 dB.

Table 7.1. Calculated sound exposure $L_{A,eq}$ difference in dB for variations in riser height, musicians spacing, screening and room acoustic conditions. The reference condition shows absolute values. The other conditions show values relative to the reference condition. It should be noted that the case with screens surround is a hypothetical (unrealistic) scenario. Bold values show significant results larger than +/- 2 dB.

- Large table on next page -

		reference model	Riser height x 2	Riser height x 4	Risers removed	1.5 m ² /musician	2.5 m ² /musician	screens behind	screens surround	ST _{early,d} & ST _{late,d} -6 dB	ST _{early,d} & ST _{late,d} +6 dB
Conf.	Space per mus.	2.0	2.0	2.0	2.0	1.5	2.5	2.0	2.0	2.0	2.0
	Riser step height	0.25	0.5	1.0	0	0.25	0.25	0.25	0.25	0.25	0.25
	Screens	No	No	No	No	No	No	Yes	Yes	No	No
	ST _{early,d} & ST _{late,d}	Ref.	Ref.	Ref.	Ref.	Ref.	Ref.	Ref.	Ref.	-6 dB	+6 dB
Left	1st violin #81	99.2	0.0	-0.1	0.1	0.4	-0.5	-1.1	-1.7	-0.4	1.3
	2nd violin #88	99.0	0.0	-0.1	0.0	0.5	-0.6	-1.3	-2.1	-0.4	1.4
	viola #56	97.9	0.0	-0.1	0.0	0.5	-0.5	-0.4	-1.5	-0.6	1.7
	violin cello #1	91.1	0.0	0.0	0.2	0.4	-0.4	-0.1	-0.3	-2.8	4.6
	double bass #11	91.2	0.0	-0.2	0.1	0.6	-0.5	-0.2	-0.6	-3.0	4.7
	French horn #51	98.5	0.0	-0.1	0.1	0.6	-0.7	-0.4	-2.7	-0.5	1.5
	clarinet #26	98.3	0.7	0.2	0.2	0.7	-0.6	-0.6	-2.5	-0.6	1.9
	bassoon #23	98.5	0.7	0.5	0.2	0.8	-0.5	-0.5	-2.1	-0.6	1.8
	oboe #30	97.4	-0.3	-0.6	-0.2	0.9	-0.9	-1.0	-3.4	-0.7	2.1
	trumpet #19	100.0	0.0	-0.5	-0.1	0.5	-0.5	-1.1	-3.5	-0.4	1.3
Right	1st violin	97.8	0.0	0.0	0.0	0.5	-0.4	-0.2	-2.0	-0.6	1.7
	2nd violin #88	97.9	0.0	0.0	0.0	0.5	-0.6	-0.3	-2.5	-0.6	1.7
	viola #56	98.0	0.0	-0.2	0.1	0.7	-0.7	-0.8	-2.4	-0.6	1.7
	violin cello #1	92.1	0.1	0.0	0.3	0.8	-0.7	-0.7	-1.0	-2.0	3.9
	double bass #11	92.9	-0.1	-0.4	0.2	0.9	-0.9	-1.1	-1.8	-1.7	3.7
	French horn #51	96.7	0.1	0.0	0.1	0.5	-0.5	-0.1	-1.4	-0.7	2.1
	clarinet #26	98.6	0.5	-0.1	0.0	0.7	-0.7	-0.3	-3.2	-0.6	1.8
	bassoon #23	99.3	0.6	0.1	0.1	0.6	-0.5	-0.4	-2.6	-0.5	1.6
	oboe #30	100.0	-0.1	-0.2	-0.1	0.6	-1.1	-0.3	-5.4	-0.4	1.3
	trumpet #19	99.4	0.1	-0.6	0.2	0.7	-0.6	-1.5	-2.7	-0.5	1.5
ILD	1st violin #81	1.5	0.0	0.0	0.0	-0.1	0.0	-0.9	0.3	0.2	-0.4
	2nd violin #88	1.1	0.0	0.0	0.0	0.0	0.0	-1.0	0.4	0.1	-0.3
	viola #56	-0.1	0.0	0.1	0.0	-0.3	0.2	0.3	0.9	0.0	0.0
	violin cello #1	-1.0	0.0	0.0	-0.2	-0.4	0.2	0.6	0.7	-0.8	0.7
	double bass #11	-1.8	0.1	0.2	0.0	-0.3	0.3	0.9	1.2	-1.2	1.1
	French horn #51	1.8	0.0	-0.1	0.0	0.1	-0.2	-0.3	-1.3	0.3	-0.6
	clarinet #26	-0.3	0.2	0.3	0.2	-0.1	0.1	-0.3	0.6	0.0	0.1
	bassoon #23	-0.8	0.2	0.4	0.1	0.1	0.0	-0.1	0.6	-0.1	0.2
	oboe #30	-2.6	-0.1	-0.4	-0.1	0.3	0.2	-0.6	1.9	-0.3	0.8
	trumpet #19	0.6	0.0	0.1	-0.3	-0.2	0.2	0.3	-0.8	0.1	-0.2

This shows that, with this common seating configuration, the direct sound levels are dominant at the ears of players of loud(er) instruments. Increasing the reflected sound levels by +6 dB (as in a small rehearsal room) results in a larger difference in sound exposure than reducing reflected sound levels -6 dB for both soft and loud instruments. For all instruments, except for low strings, the increase in total exposure is between 1.3 and 2.1 dB, which means that some musicians experience a significant increase. This shows that changing acoustic conditions only moderately affects sound exposure. Only the low strings find their exposure rise significantly by 3.7 to 4.7 dB.

In most cases, the interaural level differences (ILDs) are not significantly changed by noise control measures. An exception form the cases with screens and instruments that receive much louder direct sound from their neighbours compared to their own instrument, which are the horn player sitting in the middle of the horn section and the oboe player sitting next to the flutes.

Table 7.2. Calculated range in sound exposure $L_{A,eq}$ in dB over positions within the string player groups for different spacing: 1.5, 2.0 and 2.5 m² per musician. The configuration is the current concert hall condition with 0.25 m riser step height and no screens. The 1st and 2nd violins are first analysed per different part and then averaged.

	1.5 m ² /musician			2 m ² /musician (ref)			2.5 m ² /musician		
Instrument	Left	Right	ILD	Left	Right	ILD	Left	Right	ILD
1st violins	1.5	2.2	2.1	1.3	1.8	1.9	1.2	1.5	1.6
2nd violins	1.4	1.1	1.0	1.2	1.0	0.9	1.0	0.9	0.7
viola	1.6	1.1	0.8	1.4	1.1	0.7	1.1	1.0	0.6
cello	2.4	4.2	1.9	2.2	3.7	1.8	1.9	3.3	1.7
double bass	2.0	3.3	1.3	1.8	3.0	1.2	1.5	2.6	1.1

In order to show the effectiveness of rotating players' positions in their own instrument section, Table 7.2 presents the range in sound exposure over positions within the five strings sections. The range varies between 1.0 and 4.2 dB and is smaller when the space per musician increases. The 1st violin players with the highest exposure are located in the middle of the 1st and 2nd violin sections combined; not those positioned closer to the horns as one might expect. For the other string sections, positions closest to other louder sections are indeed exposed most. The amount of variation among high string players is below 2 dB for most cases and the effectiveness of rotating those musicians seems limited, which confirms the findings by Schmidt et al. [11]. Only for the low string players, the exposure of the right ear varies significantly when rotating.

7.4 Summary and conclusion

The effectiveness of common control measures to reduce sound exposure of orchestral musicians has been investigated using an acoustic prediction model. The model calculates the equivalent sound levels for a performance of the first 2 minutes of Mahler's 1st symphony part 4, which can be considered representative for loud orchestral music. Validation measurements have shown that the calculation model is able to predict the $L_{A,eq}$ within 2 dB deviation for 65% of the investigated microphone positions, with a maximum deviation of 6 dB. It should be noted that the model does not take into account a possible different playing style under different conditions that might affect sound levels.

The 2 dB accuracy of a sound exposure measurement has been used as a limit above which control measures are judged being significantly effective. With the same amount of musicians and equal room acoustic conditions, the available space and the height of risers have only a small effect (< 1 dB) on the sound exposure of musicians in the orchestra. A slightly larger effect is obtained by introducing a sound absorbing screen closely behind a musician sitting near (many) loud players, leading to a maximum decrease of 1.5 dB. The reduction by screens that would completely surround a musician (however not realistic) varies between 0.5 dB for low strings and on average 3.4 dB for players in loud instrument sections such as brass and woodwinds. In contradiction to the effect of screens, changing the room acoustic conditions, by extremely reducing the early and late reflected sound levels, only has a significant effect for the low strings, ranging from 3.7 to 4.7 dB. For most other musicians, changing room acoustic conditions would not lead to changes in sound exposure of more than 2 dB. Similarly, rotating positions has the largest effect on exposure of low string players by 1.8-3.7 dB for typical concert hall acoustic conditions. For high string players, rotating positions has an effect less than 2 dB in most cases.

In general, it can be concluded that the calculated effectiveness of common control measures to sound exposure of musicians playing in a symphony orchestra is within a limited range of 0.5 to 5 dB and in many cases below the measurement accuracy of 2 dB. Extreme unrealistic measures are necessary to achieve the highest reductions. This conclusion confirms results from a number of studies who also found limited reduction in sound exposure, or no reduction at all, after introducing physical measures. A reduction of 3 dB could improve conditions for musicians as much as shortening their rehearsal time by a half, which means that physical measures should not be completely neglected. However, it seems impossible for physical measures to be effective enough to replace hearing protection devices such as ear plugs that can easily attenuate 20 dB or more.

Earlier research has shown that higher equivalent sound levels occur during individual rehearsal compared to group rehearsal. Therefore, most musicians are better off playing in the orchestra than rehearsing at home. Still, many musicians predominantly focus on taking measures when playing in the orchestra. It seems that professional musicians, but also active amateur musicians, playing current modern powerful instruments and music have no other choice than to protect their ears with ear plugs under all circumstances if they wish to avoid the risk of developing hearing damage.

7.5 Recommendations for future research

The current study has focused on equivalent sound levels while disregarding peak sound levels which cannot be studied using the prediction model. The peak sound levels measured in symphony orchestras are mostly below the risk limit of 140 dB(C) described in the EU Directive [1, 12]. Nevertheless, peak sound levels are relatively high and should be considered in the evaluation of exposure to orchestral sound. The impact of control measures, such as screens, on peak sound levels needs further investigation. The current research could also be extended by studying the effectiveness of control measures when playing different repertoire (if available in anechoic format) and with different orchestra setups. Besides, validation measurements would be valuable to check our findings based on calculations. Such measurements would require a high precision in terms of reproducibility of the power output by actual musicians.

The effect of control measures on sound exposure could also be studied for smaller ensembles, such as chamber music ensembles, wind bands, jazz or percussion ensembles. The relative influence of the diffuse sound field sound level might be less dominant in a small ensemble because they comprise fewer players at larger distance, possibly in a room with more sound absorption. In such a scenario, screens are potentially more effective than has been shown in the present study. Nevertheless, the exposure of the own instrument will always play an important role and most instruments expose the own player considerably. This means that, even if barriers could be more effective to block the sound of others in smaller ensembles, ear plugs would always be necessary to protect the ears from the own instrument's exposure.

The model does not take into account how orchestra musicians might adjust their playing levels as a function of what (which levels) they hear from their own instrument, group or other parts of the orchestra. Also, a higher level of reflected sound energy could result in a softer playing style, as was shown for a cello soloist [41], which reduces the direct sound exposure of the own instrument. There might be a positive effect of higher reflection levels in terms of lower total exposure levels for musicians with loud instruments. More research is necessary to investigate if and how musicians adapt to their acoustic environment in an orchestra.

The conclusions in the present study should be interpreted with care for conditions different from open stages or (large) rehearsal rooms. For instance, in an orchestra pit the

effectiveness of control measures will likely be different. Nevertheless, the contribution of the sound of the own instrument to the total exposure will be substantial under all conditions. And, sound levels in orchestra pits are usually higher than in ‘open rooms’, see [13], most likely also when using control measures. Therefore, the main conclusion also applies for orchestral musicians playing in orchestra pits: ear plugs are inevitable for protecting musicians’ ears.

7.6 References

- [1] European Parliament and the Council of the European Union: Directive 2003/10/ec of the European parliament and of the council of 6 February 2003. Official Journal of the European Union L42 (2003) 38-44.
- [2] A. Behar, W. Wong, H. Kunov: Risk of hearing loss in orchestra musicians. *Medical Problems of Performing Artists* 4 (2006) 164-168.
- [3] J.H. Schmidt, E.R. Pedersen, H.M. Paarup, J. Christensen-Dalsgaard, T. Andersen, T. Poulsen, J. Bælum: Hearing loss in relation to sound exposure of professional symphony orchestra musicians. *Ear & Hearing* 35(4) (2014) 448-460.
- [4] H. Laitinen, T. Poulsen: Questionnaire investigation of musicians’ use of hearing protectors, self-reported hearing disorders, and their experience of their working environment. *International Journal of Audiology* 47(4) (2008) 160-168.
- [5] E.J.M. Jansen, H.W. Helleman, W.A. Dreschler, J.A.P.M. de Laat: Noise induced hearing loss and other hearing complaints among musicians of symphony orchestras. *International Archives of Occupational and Environmental Health* 82(2) (2009) 153-164.
- [6] K.R. Kähäri, A. Axelsson, P.A. Hellström, G. Zachau: Hearing assessment of classical orchestral musicians. *Scandinavian Audiology* 30(1) (2001) 13-23.
- [7] H.M. Laitinen, E.M. Toppila, P.S. Olkinuora, K. Kuisma: Sound exposure among the Finnish National Opera personnel. *Applied Occupational and Environmental Hygiene* 18(3) (2003) 177-182.
- [8] I. O’Brien, T. Driscoll, W. Williams, B. Ackermann: A clinical trial of active hearing protection for orchestral musicians. *Journal of Occupational and Environmental Hygiene* 11(7) (2014) 450-459.
- [9] J.D. Royster, L.H. Royster, M.C. Killion: Sound exposures and hearing thresholds of symphony orchestra musicians. *J. Acoust. Soc. Am.* 89(6) (1991) 2793-2803.
- [10] M.A. Rodrigues, M.A. Freitas, M.P. Neves, M.V. Silva: Evaluation of the noise exposure of symphonic orchestra musicians. *Noise & Health* 16(68) (2014) 40.
- [11] J.H. Schmidt, E.R. Pedersen, P.M. Juhl, J. Christensen-Dalsgaard, T.D. Andersen, T. Poulsen, J. Bælum: Sound exposure of symphony orchestra musicians. *Annals of Occupational Hygiene* 55(8) (2011) 893-905.
- [12] I. O’Brien, W. Wilson, A. Bradley: Nature of orchestral noise. *J. Acoust. Soc. Am.* 124(2) (2008) 926-939.
- [13] A.C. Gade: Sound levels in rehearsal and medium sized concert halls; are they too loud for the musicians?” *J. Acoust. Soc. Am.* 131(4) (2012) 3243-3243.
- [14] I. O’Brien, T. Driscoll, B. Ackermann: Sound exposure of professional orchestral musicians during solitary practice. *J. Acoust. Soc. Am.* 134(4) (2013) 2748-2754.

- [15] A.C. Gade, J. Kapenekas, B.T. Andersson, J.I. Gustafsson: Acoustical problems in orchestra pits; causes and possible solutions. In 17th International Congress Acoustics Roma (2001).
- [16] J.E. Camp, S.W. Horstman: Musician sound exposure during performance of Wagner's Ring Cycle. *Medical Problems of Performing Artists* 7(2) (1992) 37-39.
- [17] W. Williams, G. Stewart: Noise exposure reduction for orchestral musicians. *Acoustics Australia* 39(2) (2011) 73-74.
- [18] R. Libera, S. Mace: Shielding sound: A study on the effectiveness of acoustic shields. *Journal of Band Research* 45(2) (2010) 23-41.
- [19] I. O'Brien, J. Wood, B. Ackermann: Assessment of an acoustic screen used for sound exposure management in a professional orchestra. *Acoustics Australia* 41(2), 146 (2013).
- [20] C.Z. Martinez: The effect of acoustic screens on orchestra musicians. Bachelor Thesis, Hochschule für Musik Detmold (2012).
- [21] M. Chasin: *Hear the music: Hearing loss prevention for musicians*. Canada: Author.
- [22] S. Eaton, H. Gillis: Review of orchestra musicians' hearing loss risks," *Canadian Acoustics* 30(2) (2002) 5-12.
- [23] H.P. Tennhardt, H. Winkler: Studies on the acoustical planning of orchestral practice rooms. *Acta Acust. Acust.* 81(4) (1995) 293-299.
- [24] R. Pompoli, N. Prodi, A. Farnetani: A note on the acoustics of orchestra rehearsal rooms. *Building Acoustics* 19(1) (2012) 1-11.
- [25] R.H.C. Wenmaekers, L.J. Schmitz, C.C.J.M. Hak: Early and late support over various distances-rehearsal rooms for wind orchestras. *Proceedings of Forum Acusticum Krakow* (2014).
- [26] J.H. Rindel: New Norwegian standard on the acoustics of rooms for music rehearsal and performance. *Proceedings of Forum Acusticum Krakow* (2014).
- [27] M. Lautenbach, M. Vercammen: Stage acoustic requirements extended to orchestra rehearsal rooms. *Proceedings of ISMRA 2016*.
- [28] H. Laitinen: Factors affecting the use of hearing protectors among classical music players. *Noise & Health* 7(26) (2005) 21-29.
- [29] I. O'Brien, B.J. Ackermann, T. Driscoll: Hearing and hearing conservation practices among Australia's professional orchestral musicians. *Noise & Health* 16(70) (2014) 189-195.
- [30] K. Chesky, M. Pair, E. Yoshimura, S. Landford: An evaluation of musician earplugs with college music students. *International Journal of Audiology* 48:9 (2009) 661-670.
- [31] E.F. Beach, I. O'Brien: In their own words: interviews with musicians reveal the advantages and disadvantages of wearing earplugs. *Medical Problems of Performing Artists* 32(2) (2017) 101-110.
- [32] R.H.C. Wenmaekers, C.C.J.M. Hak: A sound level distribution model for symphony orchestras: Possibilities and limitations. *Psychomusicology: Music, Mind, and Brain* 25(3) (2015) 219-233.
- [33] B. Nicolai: The influence of stage acoustics on sound exposure of symphony orchestra musicians. Master Thesis. Eindhoven University of Technology (2016). Available via www.researchgate.net/profile/Remy_Wenmaekers.
- [34] R.H.C. Wenmaekers, C.C.J.M. Hak, M.C.J. Hornikx: How orchestra members attenuate direct, early reflected and late reflected sound on five different concert stages and orchestra pits. *J. Acoust. Soc. Am.* 140(6) (2016) 4437-4448.

- [35] A.C. Gade: Investigations of musicians' room acoustic conditions in concert halls. *Acustica* 69 (1989) 193-203 & 249-261.
- [36] R.H.C. Wenmaekers, C.C.J.M. Hak, L.C.J. van Luxemburg: On measurements of stage acoustic parameters: time interval limits and various source-receiver distances. *Acta Acust. Acust.* 98(5) (2012) 776-789.
- [37] R.H.C. Wenmaekers, C.C.J.M. Hak: The sound power as a reference for sound strength (G), speech level (L) and support (ST): Uncertainty of laboratory and in-situ calibration. *Acta Acust. Acust.* 101(5) (2015) 892-907.
- [38] R.H.C. Wenmaekers, C.C.J.M. Hak, M.C.J. Hornikx, A.G. Kohlrausch: Sensitivity of stage acoustic parameters to source and receiver directivity: Measurements on three stages and in two orchestra pits. *Appl. Acoust.* 123 (2017) 20-28.
- [39] J. Pätynen, V. Pulkki, T. Lokki: Anechoic recording system for symphony orchestra. *Acta Acust. Acust.* 94(6) (2008) 856-865.
- [40] IEC 61672-1. Electroacoustics-Sound level meters-Part 1: Specifications (2002).
- [41] Z. Schärer Kalkandjiev, S. Weinzierl: The influence of room acoustics on solo music performance: An empirical case study," *Acta Acust. Acust.* 99(3) (2013) 433-441.

Conclusions and recommendations

8.1 Conclusions about stage acoustic measurements

The work presented in this dissertation contributes to an understanding of how stage acoustic parameters are measured most accurately and how they can be used to study sound levels on stage. The goal was to break the vicious circle where unknown uncertainties in measured physical parameters might have led to the fact that researchers have hardly been able to find or confirm relations between perceptual evaluations and physical parameters dealing with stage acoustics. In the research presented in chapters 2 to 5, a number of often criticized points have been investigated.

Errors or uncertainties have been judged based on a just noticeable difference of 2 dB for the ST parameters based on personal communication with Gade. It should be noted that no experimental proof exists for this estimated JND. If the JND would turn out to be very different, than some conclusions regarding the impact of measurement uncertainty presented in this thesis may have to be reconsidered.

Many acousticians assume that the level of early and late reflected sound is an important criterion for acoustic conditions on stage, even though it is not yet fully proven. If one wishes to measure these levels, the concept of Gide's support (ST) parameters should still be considered state of the art. Current research has provided new insights in how such parameters should be measured. The following can be concluded:

- The time limits used in ST_{early} and ST_{late} are reasonable. Stages with suitable dimensions and properties¹⁰ are not ranked differently with other time limits (see [section 2.5.4](#)).
- To measure reflected sound levels over distance the time window should be modified using a variable time point '103-delay' (delay direct sound relative to time of emission in ms) to separate early sound from late sound (see [section 2.4](#)). Measurements of ST over distance, denoted $ST_{early,d}$ and $ST_{late,d}$, have shown to be valuable for detecting outlier stage positions and acoustical artefacts. The perceptual relevance of measurements taken over distance (such as $ST_{early,d}$) is unknown.

¹⁰ When excluding stage A which has no reflecting surfaces near the stage and stage H with extreme dimensions.

- The distance averaged $ST_{early,d}$ correlates strongly with ST_{early} measured at 1 m, making it a redundant parameter for comparing stages using a single value based on an average over positions. It should be noted that this conclusion is based on measurements on empty stages and $ST_{early,d}$ might correlate less well with ST_{early} on occupied stages compared to empty stages. The correlation between distance averaged $ST_{early,d}$ and ST_{early} measured at 1 m might explain the paradox mentioned by Gade why ST_{early} at 1 m distance correlates to the perception of hearing others at a larger distance.
- The sound power of the dodecahedral sound source should be determined in the laboratory, preferably in a reverberation room. An in-situ calibration at 1 m distance on stage, as mentioned in ISO 3382-1, should be avoided because it has an uncertainty above the estimated JND (see [section 3.6](#)).
- For single number ratings averaged over a number of positions on stage a single measurement with random orientation of the dodecahedral sound source is sufficiently accurate ($<$ estimated JND). For evaluating frequency dependent results on single positions an average of 5 equal angular rotations of the omnidirectional sound source will improve accuracy (see [section 3.4](#)).
- The decay range of the impulse response should be sufficient. An impulse to noise ratio $INR \geq 35$ dB is necessary assuming one would also like to measure T_{20} accurately (see [section 4.2.5](#)).
- ST_{early} measured on an empty stage only moderately differs from that measured on an occupied stage with a variation just over the estimated JND. ST_{late} is often reduced above the estimated JND when the stage is occupied (see [section 4.3.4](#)).
- For all parameters measured at distances ≥ 1 m, the stage should be occupied by an orchestra. A ‘musician-friendly’ measurement protocol has been applied successfully that could be executed within 10 minutes of rehearsal time (see [section 4.5](#)).
- If the orchestra is not present, one might as well measure on a stage without chairs and stands because they hardly affect the sound field on stage (see [Figure 4.12](#)).
- A height of 1 m is reasonable for most musical instruments played by seated musicians. Therefore, the source and receiver height could be fixed to 1 m. The 1.5 m height as suggested in ISO 3382-1 should be avoided.
- If the sound power of the sound source is measured in the laboratory, it is not necessary to remove objects close to transducers (see [section 4.3.2](#)).

- The source and receiver positions used by Gade result in a good overall average over stage positions (see [Figure 4.15](#)). In the case of 1 m distance measurements, the microphone could be behind the sound source in a line towards the conductor to represent the actual positions of instrument and musician.
- Source and receiver directivity have an influence on $ST_{early,d}$ and $ST_{late,d}$ with a value just below the estimated JND. It might therefore be sufficient to judge these parameters by measurements using omnidirectional transducers (see [section 5.4](#)).
- When comparing results measured with a head and torso simulator to results measured with an omnidirectional microphone $ST_{early,d}$ tends to be higher for the ear directed towards the sound source. $ST_{early,d}$ and $ST_{late,d}$ show highest values, when a directive sound source is pointing towards the closest surfaces that cause a 1st order reflection towards the receiver (see [section 5.4](#)).

Conclusions about other types of stage acoustic parameters:

- Parameters that include direct sound, such as EEL and G_{early} , are only moderately sensitive to changes in acoustic conditions on stage. If used, they should be measured at source-receiver distances well above 1 m (see [section 2.5](#)).
- The EDT is highly dependent on the direct/diffuse ratio and measurements at 1 m result in EDT values near 0 s. If used, EDT should be judged at source-receiver distances well above 1 m (see [section 9.1](#)). EDT is highly sensitive to source and receiver directivity (see [section 5.3](#)).

8.2 Recommendations for stage design and its evaluation

Following the mentioned guidelines for determining the physical parameters will not automatically lead to a sensible evaluation of all stages or other spaces that are used by orchestras. Rooms that have properties that make them less suitable for a symphony orchestra can happen to have proper ST values. This was confirmed during our validation study of the sound level distribution model (see [section 7.2.2](#)). Three very different stages in very different halls had nearly equal ST parameter values. According to the orchestra's management, the orchestra had a clear preference for the concert hall stage and disliked the two theatres. The theatres had a reverberation time which was too low and one had artificial reverberation. One theatre stage had too large stage dimensions causing audible echoes. This example shows that a certain value of $ST_{early(d)}$ and $ST_{late(d)}$ does not guarantee good acoustic conditions for the orchestra. Perhaps it is not that surprising that Dammerud mainly found correlations between architectural features and stage preference. Possibly, these stages were already too different in terms of basic properties and comparison solely on ST parameters is difficult.

If the basic properties are reasonably optimal then the ST parameters may assist in fine-tuning the stage design and a correlation between musicians' preference and ST values can be expected. An example of such case is the experiment that Gade (see [section 1.2](#)) performed in the Danish Radio Concert Hall where 12 different configurations of the same stage were compared (a stage with good basic properties). Under these conditions a higher value of ST_{early} resulted in a more preferred condition. However, if one compares a random selection of stages, the number of variables is too large and musicians might not be judging ST values but other (acoustic) properties. It might be necessary to only compare stages that fall into a certain category, for instance with a similar room type or stage size. Then a preferred ST for that stage type could be determined and used as a guideline. Other room types could be isolated from concert halls, such as theatres, rehearsal rooms, orchestra pit and even outdoor stages.

In literature, there is a good agreement among acousticians that certain basic properties are essential for good stage acoustics (see [section 1.3](#)). Figure 8.1 shows the concert stage that is preferred by the musicians in Dammerud's study (The Anvil Basingstoke) and stage C that had the best balance of early and late reflected sound (Muziekgebouw Eindhoven). Both stages fulfil the requirements mentioned in the guidelines: moderate stage dimensions, large tilted upper side walls, risers and a high ceiling (the ceiling reflectors in these halls project sound to the audience). Possibly, this is a rather 'safe' stage design that is generally liked by musicians. However, these concert halls are of moderate size with room volumes around 14,000 m³.

Looking at large concert halls with room volumes of 25,000 m³ or more, see Figure 8.2, one also recognises the architectural features mentioned in the guidelines such as side walls with tilted upper parts, a semi-circular riser system and ceiling reflectors. However, the stages in these large concert halls are wider than 18 m and deeper than 12 m resulting in a floor area

above 250 m². The reflectors are positioned high above the stage, only moderately influencing early reflected sound levels. There is a risk that time delays are too long and early reflected sound levels are too low on these large stages. Because these concert halls are relatively large, such acoustic conditions might correspond with the expectation of the musician. However, it is questionable whether different stage acoustic conditions are indeed desired in a different room acoustic or architectural context.



Figure 8.1: Left: The Anvil, Basingstoke (1994). Right: Muziekgebouw Eindhoven (1992).

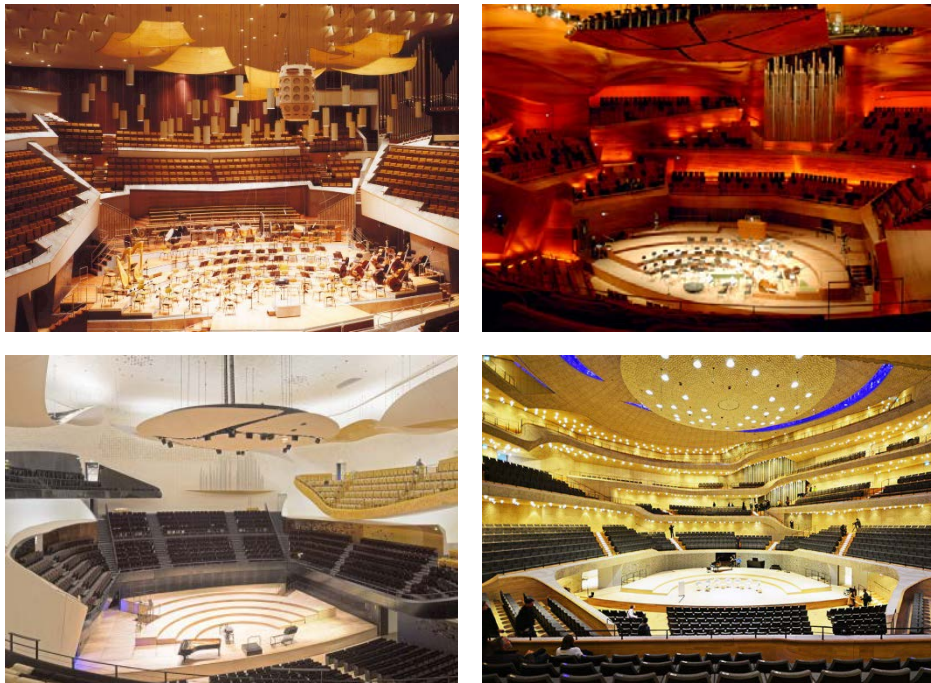


Figure 8.2: Top left: Berlin Philharmonie (1963), Top right: Danish Radio Copenhagen (2009), Bottom left: Philharmonie Paris (2015), Bottom right: Elbphilharmonie Hamburg (2017).

8.3 Conclusions about sound level predictions

In this study, a sound level prediction model has been developed for the symphony orchestra (see [chapter 6](#)). Direct sound levels have been calculated analytically while including source and receiver directivity and the obstruction of orchestra members. A specific method to estimate the sound level of the own instrument was validated. The levels of reflected sound have been estimated using measured stage acoustic parameters $ST_{early,d}$ and $ST_{late,d}$ combined with the total sound power of each instruments. Available anechoic recordings of separate instrument parts and the geometry of the orchestra have been used as an input for the model. The output of the model is the sound level at each musician's ear and the contribution of the own and other instruments to that sound level. Besides, the contribution of the direct, early and late reflected sound can be computed. Results indicate that the contribution of each aspect (own direct sound, others' direct sound, early and late reflected sound), to the total sound level at the receiving musician's position can be in the same order of magnitude. This shows that actual sound level measurements in an orchestra could never reveal the impact of the various aspects individually and the model is indispensable. A validation study has shown that in many cases the model is sufficiently accurate to predict absolute A weighted sound levels within 2 dB with a maximum deviation of 6 dB (see [section 7.2](#)), even though a possible adjustment of musicians to their acoustic environment is not taken into account.

8.4 Recommendations concerning sound exposure

While acousticians usually try to avoid large distances among players and encourage reflected sound to improve ensemble conditions, Meyer found that musicians tend to prefer more space and less reflection if asked directly. Possibly, this originates from the general idea that sound levels in symphony orchestras are high because of the proximity of other players or because of the amplification of sound by reverberation or reflection. In contrast to what is often expected by musicians, our model calculations have shown that more available space or very high risers do not lead to a significant reduction in sound exposure of musicians (see [section 7.3](#)). Reducing early or late reflected sound by less reflective surface area, added sound absorption or increased room size only has a significant effect on sound exposure of the two softest instruments, namely the cello and contrabass. Actually, they are the only musicians that are exposed to higher sound levels in the symphony orchestra than during their individual rehearsing. This would suggest that there is no ‘acoustic reason’ to complain about sound levels in the orchestra for most musicians if they feel comfortable without any hearing protection when playing alone.

Still, it is perfectly reasonable why musicians sitting near loud instruments feel like their hearing is damaged because of others. Often screens are placed in front of loud players in an attempt to reduce sound levels. However, our calculations have shown that screens are not significantly effective in reducing sound exposure (see [section 7.3](#)). This is because, for most instrument players except the low strings, both the own instruments’ sound and the sound reflected from the room boundaries have a large contribution to the total sound exposure. Using a screen might be more comfortable but screens cannot protect the ears from getting damaged. If screens are used, than they should be sound absorbing to avoid sound being reflected to the player behind. Currently, no solutions are available that can protect musicians’ ears other than personal hearing devices such as ear plugs and a well-balanced work load. Ear plugs should both be worn during individual rehearsal and in most cases during group rehearsal or performances of moderately to loud pieces.

From a room acoustic point of view, the results from the calculation model are unexpected as well. A number of researchers have suggested guidelines for orchestral rehearsal rooms in an attempt to avoid such smaller rooms being too loud, which is a common complaint among musicians. Still, calculations show that total sound exposure levels are not significantly increased for most instrument players when they play in a smaller room with higher early and late reflected sound levels compared to a large concert hall (see [section 7.3](#)). This is highly counterintuitive and it does seem reasonable to design rooms with moderate amplification of sound, if only for the sake of proper conditions for ensemble playing. Rehearsal rooms do not have to be as reverberant as performance spaces because more detailed listening is desired. In a room volume that is smaller than that of a performance space, a too strong amplification by the room can be avoided by a reverberation time that is low enough.

8.5 Recommendations for future research

The writer of this thesis has no other choice than to underline Gade's conclusion that after decades of research on stage acoustics "...there is still lacking sufficient, experimental verification regarding which properties of the sound fields and architectural features of halls govern the subjective experiences of orchestra musicians." It seems that it takes acoustic data from many more performance and rehearsal spaces and perceptual data from many more orchestral musicians to find out what are (average) optimal conditions. The studies presented in this thesis will assist researchers in performing accurate and sensible acoustic measurements. However, looking at the limited amount of performance spaces included in studies so far, it is necessary to develop new research methods that can substantially increase the amount of included stages and participating musicians.

The alternative to field measurements is the use of simulated sound fields presented in anechoic rooms, which have been used to study the musician's response to acoustic conditions during solo performances or in small ensembles. Acoustic conditions can be changed relatively easily, but only after extensive and careful preparation of the simulations. Still, the number of rooms and musicians involved in such studies are limited because organising and performing the laboratory experiments is at least as time consuming as performing field studies. And, if one wishes to study ensemble conditions of large orchestras, it is necessary to somehow include a full orchestra in the simulations. It would be worth exploring this possibility, but only if a large number of musicians and rooms can be investigated.

The simulated sound fields used in studies are often generated using geometrical room acoustic models. Such models are also used by acousticians during the design process to optimise the room design. However, the accuracy of the geometrical models is often not high enough: absolute values of (stage) acoustic parameters are not predicted within 1 JND without 'tuning' the model to measurements (which is an ambiguous method). Therefore, it is uncertain if auralisations on stage using these models are sufficiently realistic. More research is needed to improve modelling methods, which might be additionally challenging for stages because of the close proximity to the sound source(s) and the obstruction by musicians.

Chapter 9

Annex: Measurements in performance and rehearsal spaces

The main focus of the research has been to optimise and develop methods to physically measure stage acoustics. During the course of that research, many different types of stages have been measured to test the measurement methods, but also to study their acoustic properties. They consist of concert hall stages, theatre stages, orchestra pits, rehearsal rooms and even open air theatres. The aim of this chapter is to find relations between room dimensions, shapes and materials and the early and late sound levels on stages, which is interesting from an architectural point of view. The results of these studies have been presented by the author at conferences, some based on work by master students supervised by the author.

In section 9.1, detailed results are presented from the concert hall measurements as discussed earlier in [chapter 2](#). In section 9.2, a study is presented about orchestra pits. Section 9.3 deals with rehearsal rooms for Dutch wind orchestras and section 9.4 described the results from a research expedition to ancient open air theatres in Greece. For all stages discussed in this chapter, the $ST_{early,d}$ and $ST_{late,d}$ are used as the main parameters to study the acoustic conditions, as proposed in [section 2.4](#), equations 1 and 2. For the concert hall data in section 9.2 reverberation time parameters have also been included such as EDT and T_{20} . An overall discussion about the possible relevance of the $ST_{early,d}$ and $ST_{late,d}$ parameters using the measurements in this annex has already been presented in [section 2.9](#).

NOTE: It should be noted that in most cases the reference level (0-10 ms) was derived from measurements at 1 m distance on site (except for the ancient theatres). The method of using sound power measurements in the laboratory, as introduced in [chapter 3](#), was not common at the time of most measurements. As a result, the uncertainty in these measurements is 1.2+/-1.2 dB (an average offset of -1.2 dB for parameter results), see Table 3.7 in [section 3.5.4](#). However, the balance between early and late reflected sound is not affected by the type of sound power calibration. Also, it should be noted that all stages and venues were unoccupied during these measurements, contrary to what is recommended based on later research as presented in [chapter 4](#).

9.1 Concert halls

9.1.1 Introduction

In many traditional shoebox halls, the stage is positioned to the back of the hall, while in many modern halls the audience is also seated behind the stage or on its sides. Especially in the surround halls, the stage has to be carefully integrated into the total hall to ensure visibility of the orchestra for as many seats as possible. Still, many concert halls have a (sunken) stage surrounded by side walls and a back wall and only the stage front is opened to the audience. Such walls are common elements that provide early reflections for the orchestra. They vary from straight walls to walls with tilted or scattering elements. Even though rectangular shaped stages exist, it is common to use fan shaped stages to improve the projection of the sound from the orchestra towards the audience and to avoid the loud instruments in the back to be too loud on stage. The ceiling is usually high while in some halls floating reflectors are hung over the stage to provide vertical early reflections. Furthermore, it is common to position the woodwinds, brass and percussion on risers to improve communication with the conductor and to avoid blocking of the sound of these groups by the strings sections in front. A detailed study on stage design has been performed by Fichera, see her master thesis [1].

9.1.2 Method

In 2009, at the start of this research project on stage acoustics, a large set of measurements was collected in 7 Dutch concert venues by two master students, Kivits and Heijnen [2]. Additional measurements were performed by the author in Casa da Musica, Porto. Impulse responses were measured using omnidirectional transducers on a grid that was equal on all stages, see Figure 9.1. The stages were empty without chairs. The measured impulse responses have been used to study the effect of time intervals on early and late reflected sound levels. See [chapter 2](#) for more details on the used methods.

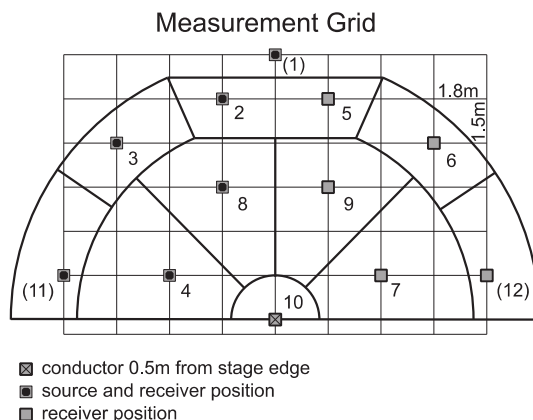


Figure 9.1: Schematic representation of the measurement grid used on the concert hall stages. Additionally, measurements were taken at 1 m distance from the sound source.

9.1.3 Results and discussion

In this section, the measurement results of stage acoustic parameters related to reverberation time, EDT and T_{20} , early reflected sound levels, ST_{early} and $ST_{early,d}$, and late reflected sound levels, ST_{late} and $ST_{late,d}$, are presented in Table 9.1. Table 9.1 also contains architectural data for each hall such as dimensions, as suggested by Dammerud et al. [3], see Figure 9.2. The coefficient ‘a’ and ‘b’ of the logarithmic trend lines $a \lg(d) + b$ for $ST_{early,d}$ and the average values for $ST_{late,d}$ are given for each concert hall stage in Table 9.2. The results are discussed per individual hall in Table 9.1. An overall discussion on relations between architectural and acoustical parameters is presented in section 2.9.

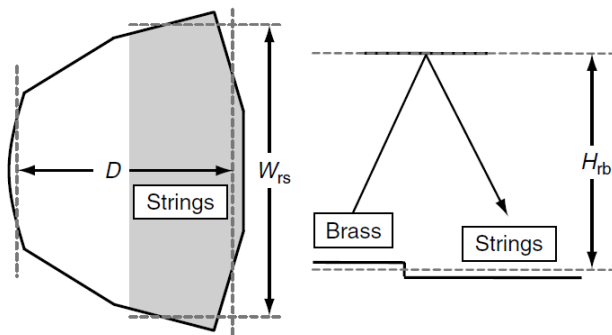
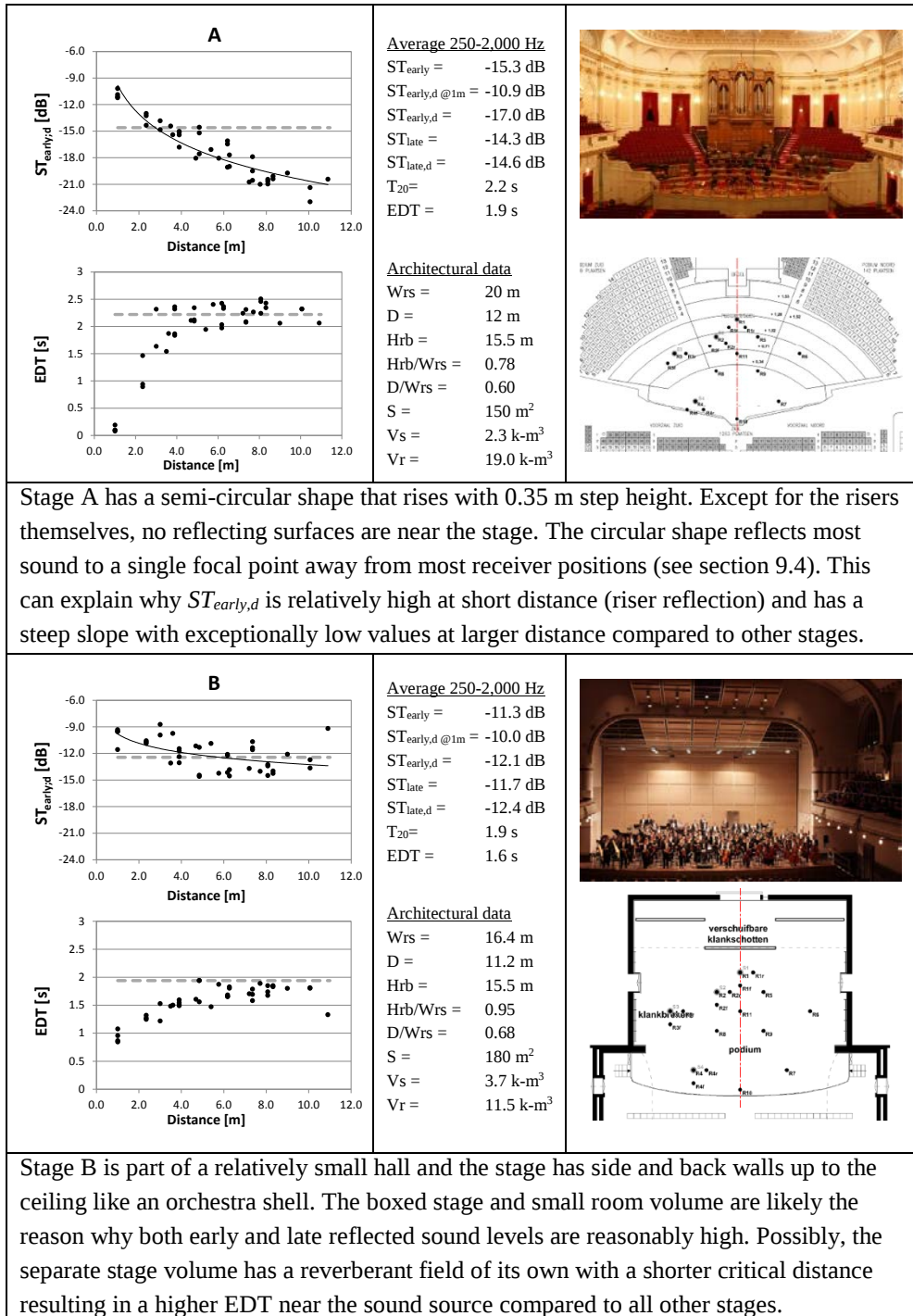


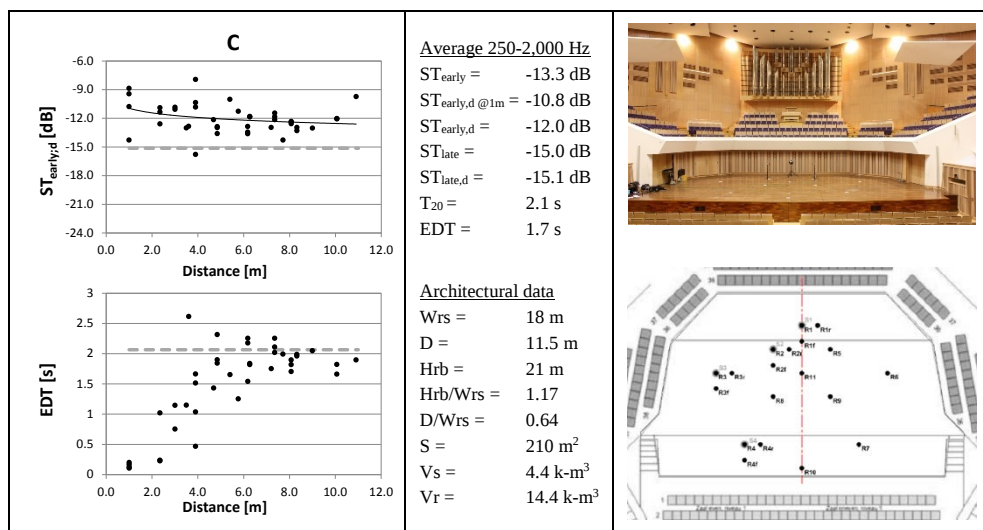
Figure 9.2: Plan and long section of a generic stage showing the method for obtaining the proposed architectural measures [3]. W_{rs} = width reflecting surfaces strings, D = distance between the back end of the stage accessible to the orchestra (left side of the figure) and the average stage front, H_{rb} = height reflecting surfaces brass.

Table 9.1: Left: $ST_{early,d}$ and EDT as a function of distance (dots represent individual results) and the average value for $ST_{late,d}$ and T_{20} (grey dashed lines). Middle table: Average values for ST_{early} and ST_{late} measured at 1 m distance and $ST_{early,d}$, $ST_{late,d}$, T_{20} and EDT averaged over all measured distances. All parameter values are determined for the average of 250 to 2,000 Hz. Architectural data (see Figure 9.2): W_{rs} = width reflecting surfaces for strings, D = distance between the back end of the stage accessible to the orchestra and the average stage front, H_{rb} = height reflecting surfaces brass, S = stage area in m^2 , V_s = stage volume ($H_{rb} \times S$), V_r = room volume in $k \cdot m^3$ ($m^3 \cdot 1000$). Right: photograph and floor plan of the concert hall stage.

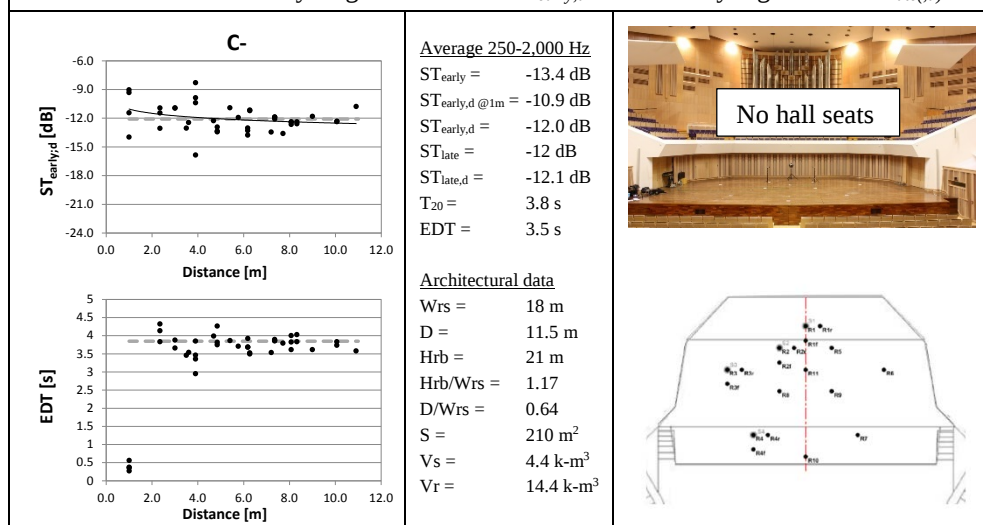
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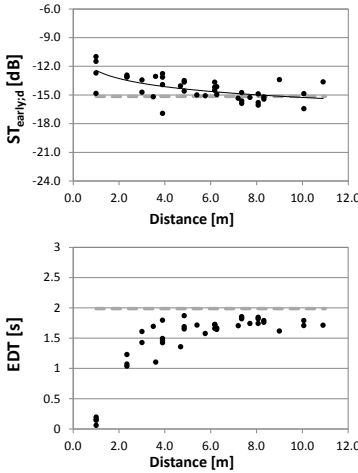
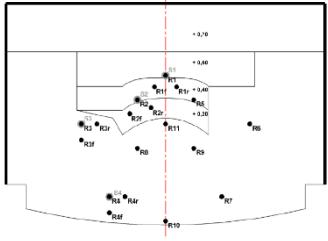


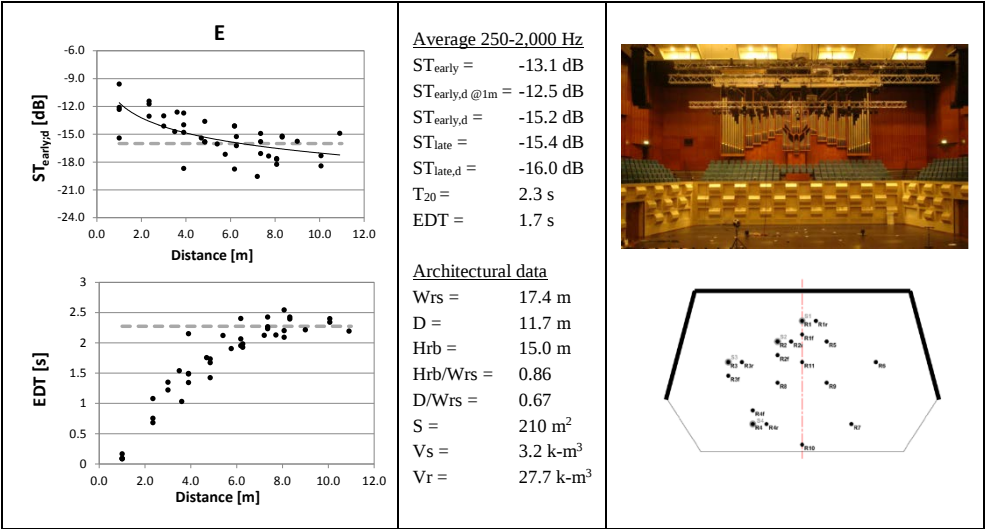


Stage C has a sound diffusing back wall with a tilted reflective upper edge. Two freely hanging reflectors project the sound from left to right on the front of the stage and seating area. These features allow for a similar amount of early reflected sound ($ST_{early,d}$) as stage B. $ST_{early,d}$ and EDT is more spread over distance than on other stages, possibly because there are specular reflections from the tilted upper wall that are unobstructed by musicians. This is the only stage which has a $ST_{early,d}$ considerably higher than $ST_{late,d}$.

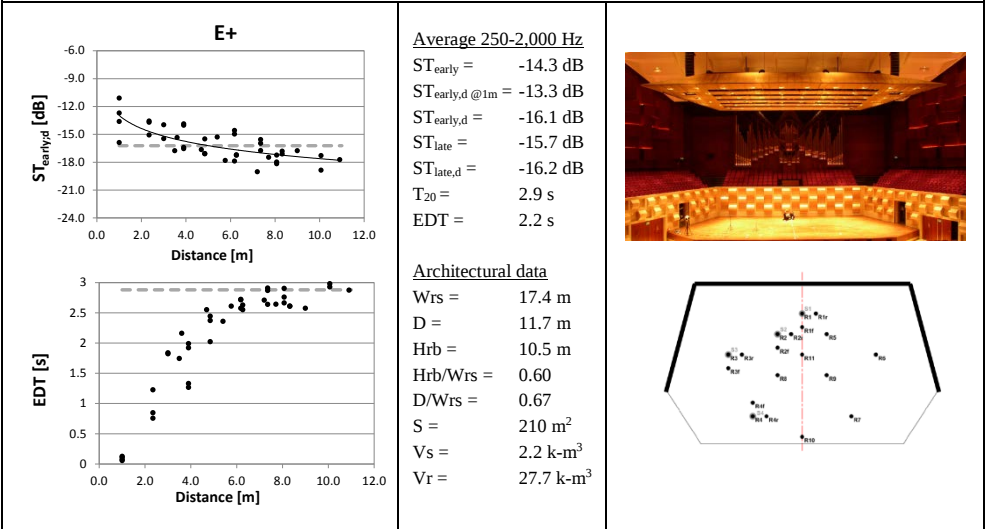


Stage C was also measured when all the seats were taken out of the concert hall for refurbishment. As a result, T_{20} doubled and ST_{late} increased by 3 dB, which is less than the 4.7 dB estimated using Barron's revised theory (See section 9.5). $ST_{early,d}$ gave almost exactly the same results as the situation with seats, which confirms that this parameter is indeed not dependent on late reflections from the audience area. The EDT, also describing early sound, changed drastically and is close to T_{20} at distances above 1 m.

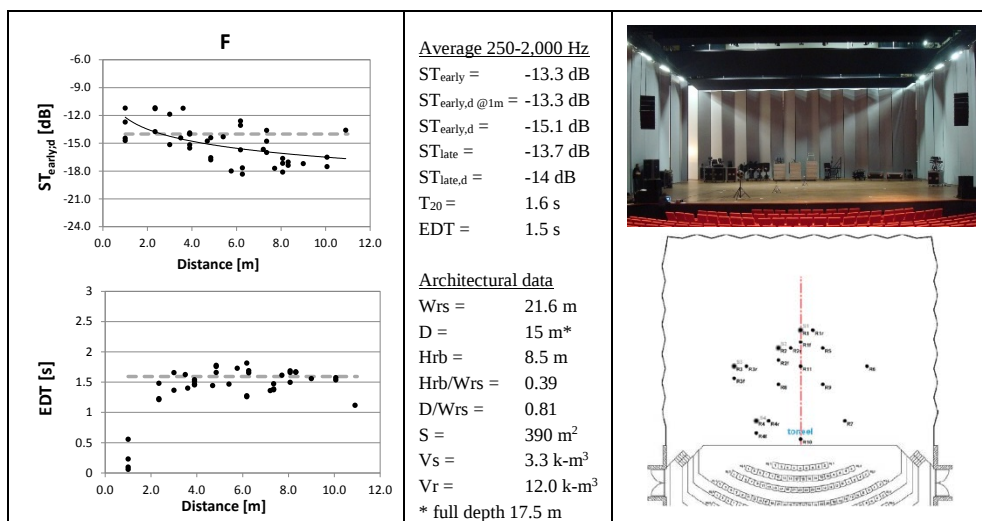
D 	<u>Average 250-2,000 Hz</u> ST_{early} = -13.1 dB $ST_{early,d @ 1m}$ = -12.5 dB $ST_{early,d}$ = -14.3 dB ST_{late} = -14.7 dB $ST_{late,d}$ = -15.2 dB T_{20} = 2 s EDT = 1.6 s <u>Architectural data</u> Wrs = 20.4 m D = 13.7 m Hrb = 17.0 m Hrb/Wrs = 0.83 D/Wrs = 0.67 S = 260 m² Vs = 4.4 k-m³ Vr = 15.5 k-m³	 
Stage D has a rectangular shape with diffusing elements on the upper part of the walls. Ceiling reflectors project the sound towards the audience area. $ST_{early,d}$ is 2 dB lower compared to stages B and C, which can be explained by the larger size of the stage and the lack of downwards aiming reflecting or diffusing surfaces. $ST_{late,d}$ is nearly equal to the value on stage C, probably because the room volume and reverberation time are nearly the same. - The table continues on the next page with hall E and E+ -		



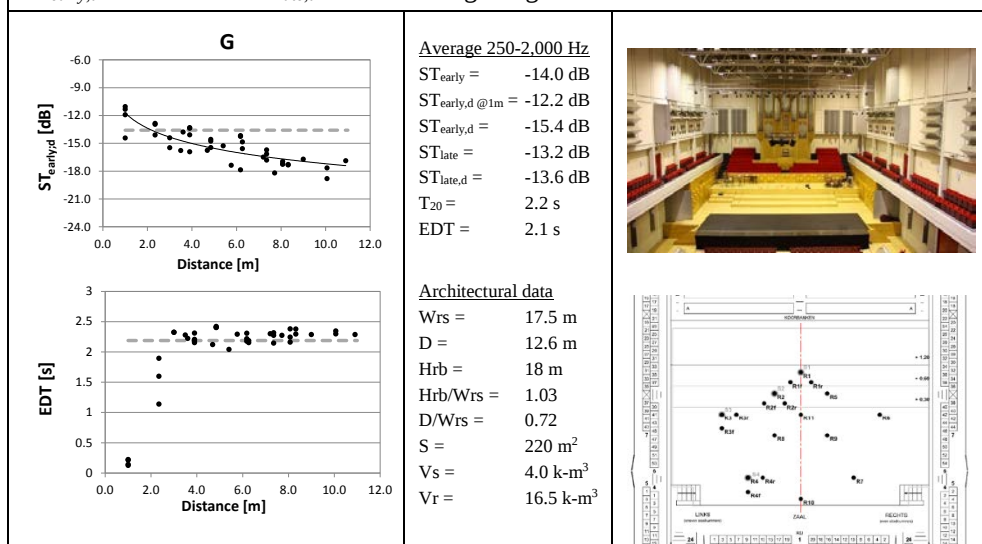
Stage E has dimensions similar to stage C but with side walls that scatter sound in the vertical direction (instead of horizontally). $ST_{early,d}$ is 3 dB lower than stage C, possibly because it has no tilted upper wall parts and ceiling reflectors. $ST_{late,d}$ is relatively low caused by the almost double room volume compared to stage C. EDT approaches T_{20} only after 6 m, probably due to a relatively long critical distance of the hall.



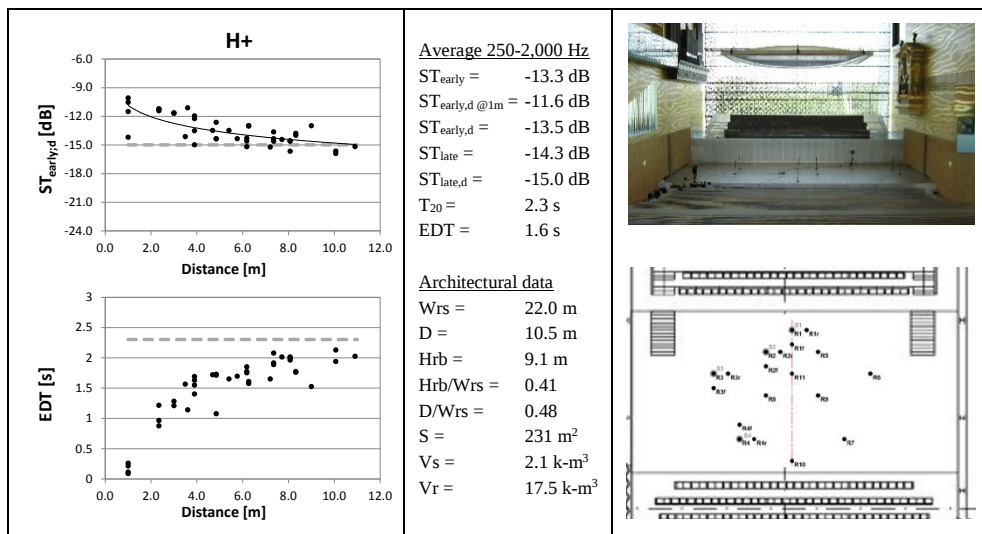
The concert hall was renovated and its seats were replaced, resulting in a longer reverberation time. A semi-transparent canopy was placed over stage E, to improve the early support for weaker instruments at the front of the stage. The measurements show that less negative outliers exist in $ST_{early,d}$ after placing the canopy, which confirm that the mentioned goal was reached. However, the average amount of early and late reflected sound did not increase. Also, $ST_{late,d}$ did not increase even though T_{20} is 0.6 s higher.



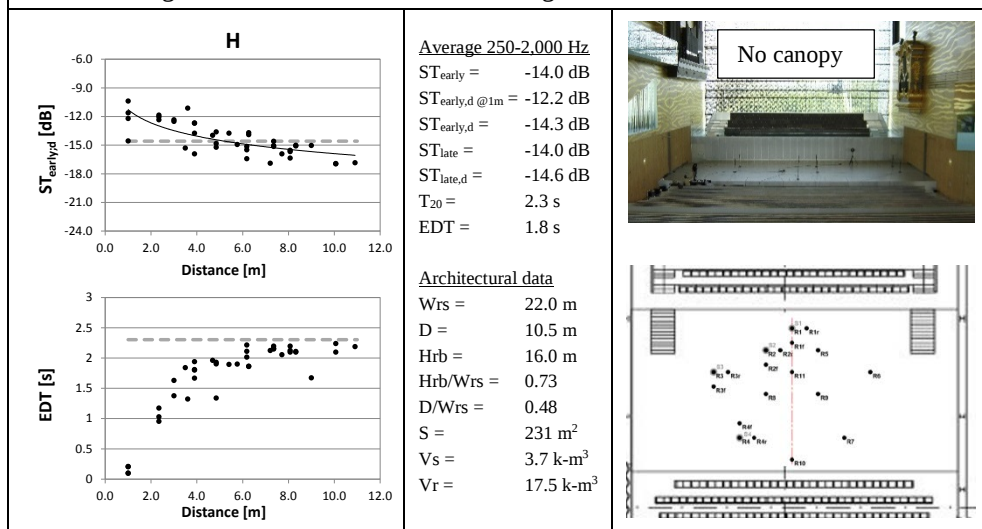
Stage F has tall zigzag shape walls and is relatively wide and very deep. As a result, early reflected sound has a relatively long delay which is heard as an echo. Ceiling reflectors aim the sound towards the audience. The low reverberation time combined with a low room volume happens to result in an $ST_{late,d}$ comparable to other stages, but $ST_{early,d}$ is lower than $ST_{late,d}$ due to the large stage size.



Stage G has a low back wall and mostly parallel side walls. Only near the front stage, the side wall elements reflect the sound away from the orchestra. This stage has similar results for $ST_{early,d}$ and $ST_{late,d}$ as stage F. $ST_{early,d}$ is relatively low and EDT is already equal to T_{20} above 3 m, indicating that critical distance is short. Possibly this is caused by the relatively few reflecting elements near the stage.



Stage H is relatively wide and not deep. Its walls are clad with QRD diffusors similar to stage C, but walls are not tilted in the upper parts. A special feature is the curved canopy above the stage that directs sound back to the stage.



The same stage was also measured with the canopy lifted to the ceiling. As a result, average $ST_{early,d}$ decreases by 1.2 dB and $ST_{late,d}$ increases by 0.4 dB. In addition, removing the canopy had an effect on EDT that increases with 0.2 s on average. In the original configuration with the canopy lowered $ST_{early,d}$ is above $ST_{late,d}$ up to 5 m distance similar to stage C, but $ST_{early,d}$ decays more over distance on stage H. It seems that the canopy compensates for the lack of tilted side wall reflections resulting in a reasonably high $ST_{early,d}$. However, the stage is relatively wide similar to stage F with the risk of side wall reflections arriving relatively late and musicians sitting too far apart.

Table 9.2: The coefficient ‘a’ and ‘b’ of the logarithmic trend lines $a \lg(d) + b$ for $ST_{early,d}$ and the arithmetic average values for $ST_{late,d}$ are given for each concert hall stage. The correlation coefficient R^2 is given for each trend line. *Most ST parameter values are likely approximately 1.2 dB too low because an in-situ calibration was used.

Hall	$ST_{early,d} = a \lg(d) + b$ [dB]			$ST_{late,d}$ [dB]
	a	b*	R^2	Average*
A	-10.8	-9.8	0.87	-14.5
B	-3.4	-9.8	0.32	-12.5
C	-1.6	-11.0	0.09	-15.2
C-	-1.5	-11.0	0.08	-12.1
D	-2.8	-12.4	0.40	-15.2
E	-5.4	-11.6	0.47	-16.0
E+	-4.6	-13.0	0.57	-16.3
F	-4.3	-12.2	0.34	-14.0
G	-5.5	-11.7	0.69	-13.6
H	-4.6	-11.3	0.58	-14.6
H+	-3.9	-10.9	0.54	-15.0

9.1.4 Conclusion

The 8 concert halls discussed have stage designs with various dimensions and different acoustic solutions. For instance, stage surface area S varies between 150 and 390 m², stage volume V_s varies between 2,100 and 4,400 m³ and room volume V_r varies between 11,500 and 28,000 m³. Most of these room properties vary by a factor of 2 or more, which are substantial differences from an architectural point of view. However, it seems that such differences only moderately influence acoustic parameter results. For instance, $ST_{early,d}$ varies by 5 dB (between -17 and -12 dB) and $ST_{late,d}$ varies by 4 dB. The range is larger than the estimated JND of 2 dB, but the question is whether musicians are able to detect differences among many of the stages.

The uncertainty of the measurement is approximately 1 dB when using laboratory sound power calibration and approximately 2 dB when using in-situ calibration. This suggests that the parameters can be measured accurately enough to find significant differences. In a design process, they also have to be predicted accurately enough, which is a topic that is discussed in section 2.9.2.

In the next subsections, other room types are investigated to study the sensitivity of the acoustic parameters for more extreme differences in architectural features.

9.2 Orchestra pits

9.2.1 Introduction

A typical modern orchestra pit is partly covered by the stage floor and partly opened to the auditorium. Because of that, the orchestra pit can be an acoustically challenging environment for orchestra musicians. Typically, the louder instruments (brass instruments, wind instruments and percussion) may be positioned in the covered part of the pit. The instruments that generally play less loud (violins, celli and double basses) may be positioned in the open part of the pit. As a result, the sound of the different instrumental sections is projected differently towards the audience in the auditorium compared to the soloists and choir on stage. Also, the ensemble playing of the musicians is different in an orchestra pit compared to a fully open stage setting in a concert hall, partly due to the difference in the acoustic environment. To investigate the difference in acoustic conditions, stage acoustic parameters have been measured in an orchestra pit, while varying its size. The results are compared to measurements in concert halls. A detailed study on orchestra pits has been performed by Angela van der Heide, which is recommended for further reading [4].

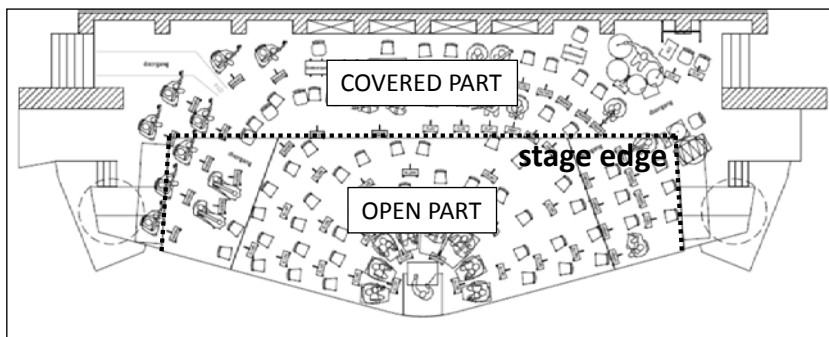


Figure 9.3a: orchestra setup in the normal size orchestra pit.

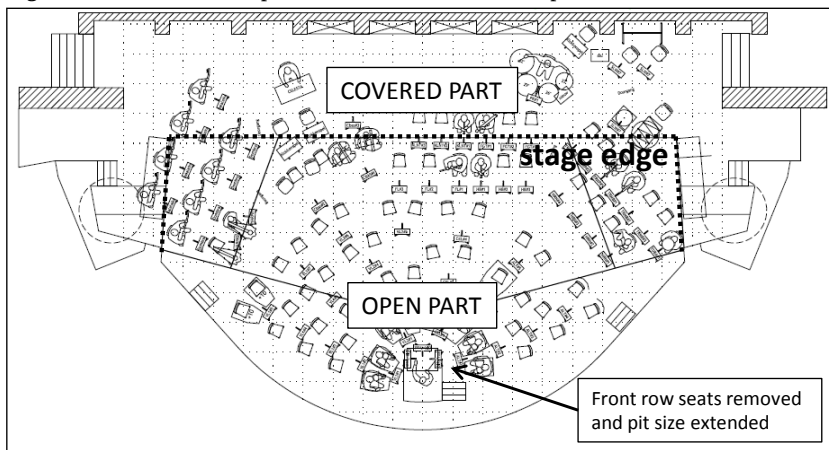


Figure 9.3b: orchestra setup in the large size orchestra pit.

Figure 9.3a shows a plan of a typical orchestra setup in the normal size orchestra pit of Het Muziektheater Amsterdam, which was investigated in this research. The transition between the open and covered parts is indicated by a dashed line. In an attempt to reduce the sound exposure of the musicians in this pit, and to improve ensemble conditions, the impact of increasing the pit size was studied. This was done by taking out several rows of audience seats and moving the whole orchestra towards the audience, placing as many musicians in the open part of the pit as possible, see Figure 9.3b.

9.2.2 Method

Figure 9.4 shows the plan of the orchestra pit in the normal size configuration. In the larger size configuration, the position S2, R4 and R6 are moved to also be in the middle of the open area, while the positions under the covered part remain unchanged. For all source positions S1 to S3, additional measurements were performed at 1 m distance at the front and right side of the loudspeaker (front towards the audience).

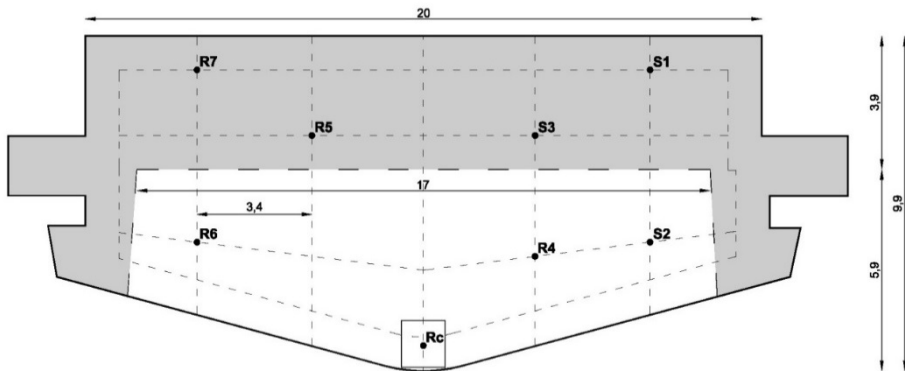


Figure 9.4: Plan of the normal size orchestra pit with source positions S1-S3 and receiver positions R1-R3 (equal to S1-R1) and receiver positions R4-R7 and Rc [4].

It should be noted that the ST parameters should normally not be measured while placing the transducers closer than 2 m from any room boundary. This is particularly important to avoid exclusion of early reflections arriving before 10 ms in the measurement interval 10 ms to 103 ms-delay. However, in the case of an orchestra pit, it is impossible to fulfil this condition because of the close proximity of the ceiling and back wall. This exclusion will probably have resulted in an underestimation of the amount of early reflected sound energy.

The measurement method is similar to that used during the measurements in the concert halls (see [5] for more details). During the measurements, the orchestra pit was empty, except for the larger size condition where seats and percussion instruments were stored along the back wall of the pit that may have caused additional absorption or scattering. Also, during both measurement sessions, different stage sceneries for opera were present.

9.2.3 Results

Figure 9.5 shows the measurement results for $ST_{early,d}$ and $ST_{late,d}$ as a function of distance for the normal and large size orchestra pit of Het Muziektheater Amsterdam. In every graph, results are divided into three groups: both positions in the covered part of the pit; just one of both positions in the open or covered part of the pit; both positions in the open part of the pit. In both the normal and large size pit, the individual values for $ST_{early,d}$ of each group show a strong correlation to a distinctive trend line. For the $ST_{late,d}$ no clear distinction is found for the three groups and no correlation exists for the values as a function of the distance.

For both pit sizes, at a distance close to the source, the $ST_{early,d}$ is only a few dB below the direct sound level at 1 m distance in the covered part of the pit, while in the open part $ST_{early,d}$ is approximately 10 dB lower. The trend lines of $ST_{early,d}$ over distance seem to run more or less parallel for the groups ‘covered-covered’ and ‘open-open’. The trend lines ‘open-covered’ are close to the group ‘covered-covered’ at short distance and close to the group ‘open-open’ at larger distance. Furthermore, we can conclude that, due to the enlargement of the orchestra pit, the trend lines have tilted: $ST_{early,d}$ is almost unchanged close to the source, but decreased by 3 to 5 dB at distances beyond 10 m. The average $ST_{late,d}$ is 2.8 dB lower in the larger orchestra pit configuration.

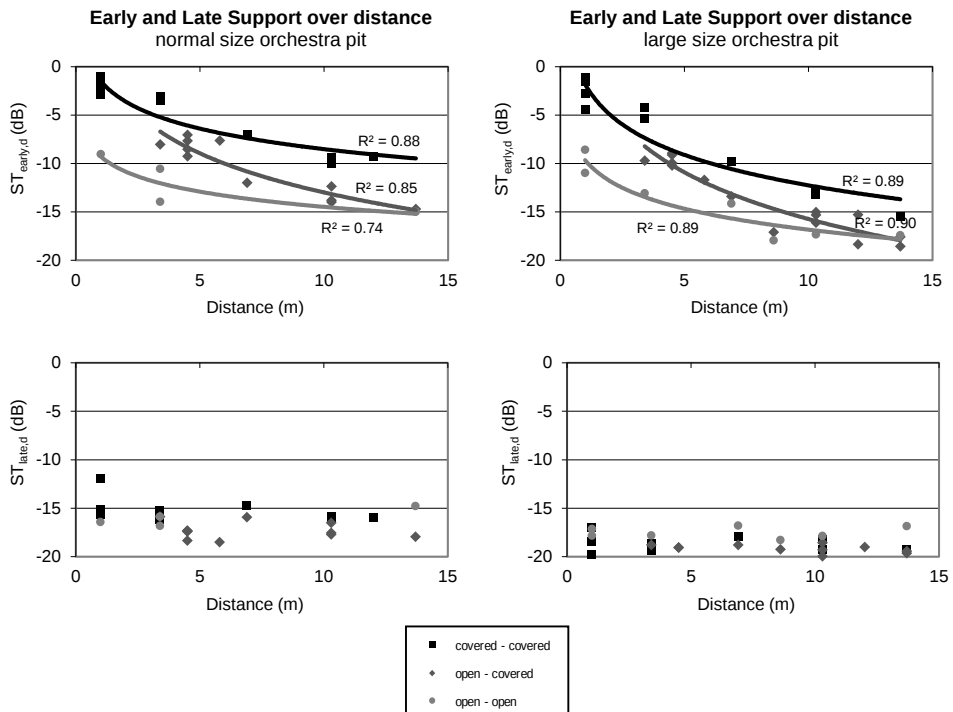


Figure 9.5: $ST_{early,d}$ (upper graphs) with logarithmic trend lines and $ST_{late,d}$ (lower graphs) over distance for the normal size orchestra pit (left graphs) and large size orchestra pit (right graphs).

9.2.4 Comparison to concert hall stages

To investigate the possible meaning of the measurement results in the orchestra pit, a comparison is made by the measurements on concert hall stages presented in section 9.1. In the left graph in Figure 9.6, the results for the orchestra pit are compared to reference values from the concert halls. When comparing $ST_{early,d}$ it can be seen that the amount of early reflected sound in the open part of the pit is more or less similar to a concert hall stage with a high $ST_{early,d}$. However, when the source, receiver or both are in the covered part of the pit, the amount of early reflected sound level is considerably higher, especially at a shorter S-R distance. Compared to the stage with a high $ST_{early,d}$ the investigated orchestra pit shows a larger variation per distance, the most for scenarios C-C and O-C.

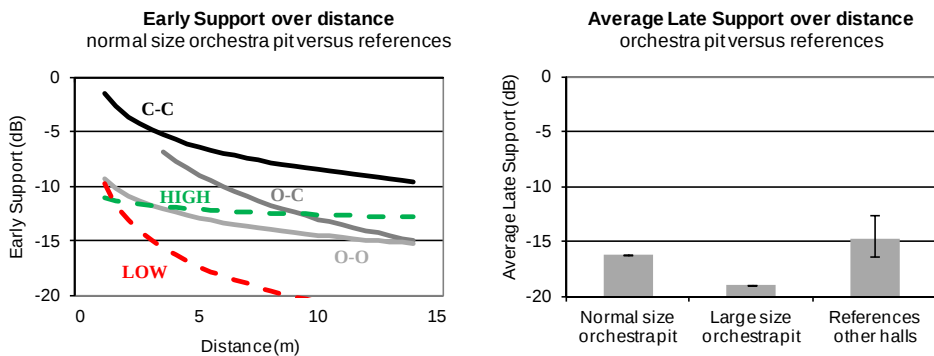


Figure 9.6: Comparison of measurement results for the orchestra pit and various concert halls. Left: Trend lines Early Support over distance for the normal size orchestra pit (as from Figure 9.5 left), for the S-R groups covered-covered (C-C), open-covered (O-C) and open-open (O-O). Besides, trend lines are presented for concert hall C with a 'HIGH' $ST_{early,d}$ and for concert hall A with a 'LOW' $ST_{early,d}$.

Right: Average Late Support for the normal size orchestra pit, large size orchestra pit and the average value and range for 10 concert hall stage configurations (stage C- is excluded).

In the right graph in Figure 9.6, the average $ST_{late,d}$ for both orchestra pit configurations are compared to the average value and for the different concert hall stages of section 9.1. It can be seen that, in general, less late reflected sound arrives in the orchestra pit than on typical concert hall stages. This can be explained by the lower T in the opera house of 1.3 s compared to the concert halls with an average T of 2 s. When the orchestra pit is enlarged, the coupling of stage and pit is stronger and the average $ST_{late,d}$ decreases by 2.8 dB. Actually, it could have been expected that, when the pit has a larger opening, more sound can be reflected back into the pit. It seems that this is not the case. The higher $ST_{late,d}$ in the normal size orchestra pit might be explained by the 'late reverberation' that may build up in the more closed volume of the (empty) pit.

9.2.5 Discussion

Results for only one orchestra pit were presented in this section. The three different sound path trends have also been observed in measurements in another orchestra pit, see annex 6.2.7 and [5], but the impact of enlarging the orchestra pit was not tested in other pits.

The measured stage acoustic parameters can be used to study the sound exposure of musicians, see [Chapter 6](#). The measurement results for $ST_{early,d}$ show a high early reflected sound level close to the sound source in the covered part of the pit. On concert hall stages, the early reflected sound level close to the sound source is often at least 10 dB lower than the direct sound at 1 m distance. For this orchestra pit, however, the early reflected sound level is in the same order of magnitude as the direct sound at 1 m distance ($ST_{early,d}$ close to 0 dB). So, it can be expected that the sound exposure due to instruments in close proximity will be higher in the pit compared to the stage, up to 2 to 3 dB. When the pit is enlarged, the $ST_{early,d}$ close to the sound source appears to be almost unchanged. So, it can be expected that, only when more musicians take place in the open part of the pit (where $ST_{early,d}$ is much lower) the sound exposure can be reduced by enlarging the orchestra pit.

For practical reasons the measurements were performed in an empty unoccupied orchestra pit. In 2014, after performing the current study, measurements have been repeated in the normal size orchestra pit while being occupied by the mannequin orchestra as presented in [Chapter 4](#). It was found that both $ST_{early,d}$ and $ST_{late,d}$ are reduced by the orchestra, but the trends described in this section were unchanged and conclusions are still valid.

9.2.6 Conclusion

In this section, the modified and extended Early and Late Support parameters have been used to investigate the ‘stage acoustics’ of an orchestra pit. By doing so, for the first time, it has been shown that three distinct groups of sound paths can be discriminated: both positions in the open part; both positions in the covered part; and just one of both positions in the open or covered part. In pits, higher early reflected sound levels are measured compared to concert hall stages, with $ST_{early,d}$ between -9 and -4 dB in the covered part. In contrast, lower late reflected sound levels are found compared to concert stages with $ST_{late,d}$ as low as -21 dB.

9.2.7 Appendix

For the sake of completion, the $ST_{early,d}$ and $ST_{late,d}$ of the orchestra pits that were investigated by Van der Heide [4] in 2010 have been calculated from the original impulse responses, see Figure 9.7. More background on the pits can be found in her thesis. In particular, the orchestra pit ‘STS’ with diffusors or curtains against the back wall shows the three typical trends as described in Figure 9.5. Orchestra pit ‘PAR’ shows an opposite trend for the ‘open-covered’ positions. No explanation was found for this result. The orchestra pit ‘VES’ is very small and therefore shows few differences between results for open or covered areas.

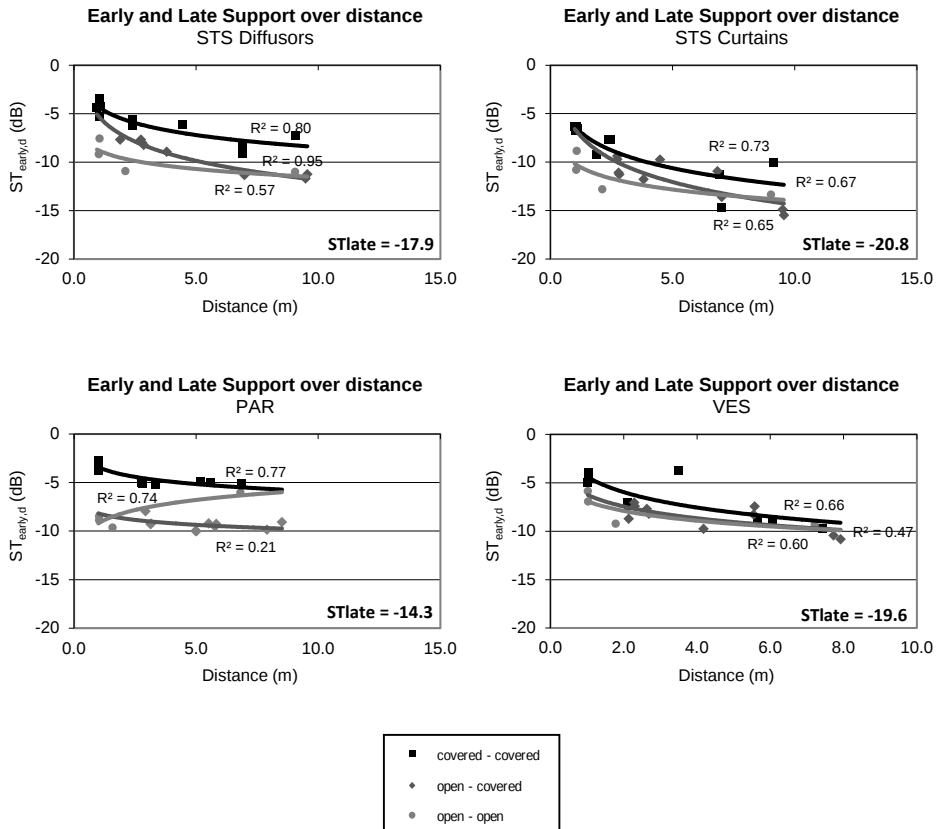


Figure 9.7: Early and late support for other orchestra pits.

9.3 Rehearsal rooms for wind orchestras

9.3.1 Introduction

In the Netherlands, many musicians play in a non-professional wind orchestra. Especially in the southern part of the country, almost every village has its own wind orchestra and many larger towns have multiple orchestras. The wind orchestras are divided into three groups: the concert band or ‘harmonie’, with a mix of woodwind and brass instruments; and the fanfare and brass band, both only having brass instruments. These bands are accompanied by a percussion group, which in most cases also form a separate full percussion orchestra. In total, over 2400 orchestras and ensembles are member of the Royal Dutch Music Association (KNMO).

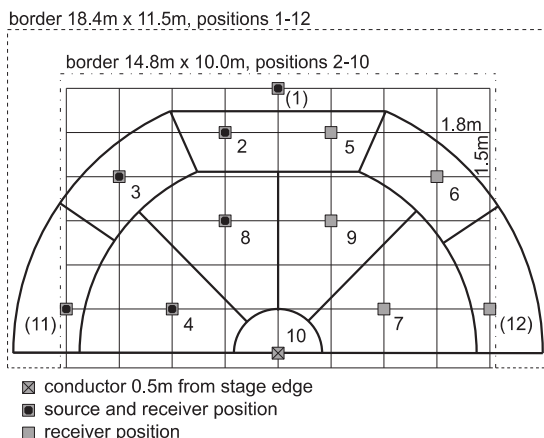
Most of these orchestras have a weekly rehearsal in their local hall (in Dutch: the ‘harmoniezaal’ or ‘fanfarezaal’), which is a dedicated room for the orchestra to rehearse and perform. Typically, the wind and percussion orchestra share the same rehearsal and performance space. Often, these halls also act as a community center resulting in multifunctional demands for the use.

Most of the halls are not purpose build for the orchestra only, and, as the building budget is often limited, they are being built by local architects and contractors, often members of the same community. Only recently, acousticians are (sometimes) involved in the design process. However, little is known about the acoustic characteristics of these halls and the demands of these orchestras for such halls. In this section, 7 rehearsal rooms have been investigated using stage acoustic measurement methods. A detailed study on rehearsal rooms has been performed by Schmitz, which is recommended for further reading [6].

9.3.2 Method

Impulse response measurements have been performed on a grid of source and receiver positions over ‘the orchestra area’. In the research on stages of symphony orchestras a grid with fixed dimensions was used, see Figure 9.8a. However, it was found that in most rooms used by wind orchestras, the available space is much smaller (120 m² instead of 200 m² on average). The various sections in the wind orchestra are distributed in a similar way as in the symphonic orchestra. Therefore, the same distribution of positions was used in this research, but with dimensions scaled down, see Figure 9.8b. More detailed information about the used methods is found in [7].

a



b

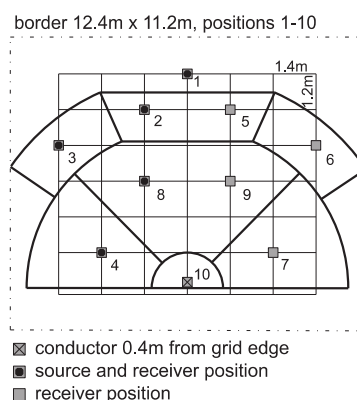


Figure 9.8: Measurement positions (a: symphony orchestra, b: wind orchestra)

9.3.3 Modelling

To investigate the contribution of the various reflecting surfaces to the total early reflected sound level within the 10-103 ms time window (equal to $ST_{early,d}$), the Image Source Method (ISM) has been used. It was found that, to be able to investigate the sound level in the time window up to 103 ms after departure of the sound, only the 1st and 2nd order reflections need to be taken into account. For every 1st and 2nd order reflection, the sound path length from source to receiver 'd' in m is calculated. Then, the sound level of each reflection is determined using the inverse square law $-20\lg(d)$, while taking into account the level reduction due to sound absorption by a factor $-10\lg(1/(1-\alpha))$. The direct sound and first floor reflection were discarded as they arrive within the 0-10 ms time window. An average value over the 250-2,000 Hz octave bands is considered. Because of the relatively large frequency range, the energies of different sound paths were summed. The exception to this rule are the 2nd order reflected sound paths that arrive twice with the same path length (like the floor-ceiling path and ceiling-floor path); here the sound energy was considered to arrive in phase, adding 3 dB extra in sound level. Instead of using the positions of the grid in Figure 9.8b, a source-receiver (S-R) pair in the center of the orchestra was considered. The S-R distance was increased by moving S and R outwards over the center line between positions 1 to 10.

Seven different rooms have been used for the investigation, each having different room dimensions and different locations of sound absorbing materials. Table 9.3 shows an overview of the room dimensions. For the non-rectangular rooms, a range of dimensions is given, indicated by a '/'. Room MK was measured twice: once with a 1.5 m high screen in front of the percussion (position S1/R1), denoted 'MKs' and once without the screen, denoted

‘MKn’. The stage volume is computed by assuming a maximum depth of 11 m multiplied by the average width and height of the stage or the room at the position of the orchestra.

The material properties used for the ISM are given in Table 9.4. Note that, in case the sound absorbing material on the wall is above 1.5 m, the low absorption coefficient is used of the reflective part of the wall in the ISM. The floor is fully reflective (a thin carpet in some of the rooms was neglected). The shape and material properties of the different rooms are further illustrated in Figures 9a to 9g. In the middle column, a 3D figure of each room is presented and material properties are indicated. The position of the measurement grid is illustrated using a red rectangle and figures are oriented in such a way that the conductor’s position is in the southwest (bottom-left) side of the room.

Table 9.3. Rehearsal room dimensions: w=width, d= depth, h=height, F=floor area, V = volume

	<i>w [m]</i>	<i>d [m]</i>	<i>h [m]</i>	<i>F [m²]</i>	<i>V_{room} [m³]</i>	<i>V_{stage} [m³]</i>
BK	14.5	11.5	4	165	650	500
ML	12.5	18	4	225	900	550
BM	12	30	3/6.7	360	1,920	470
HZ	11.5/16.5	29	5.5/5.9	400	2,250	850
MK	14	29	6	350	2,400	890
BZ	15/20.5	24	2.1/7	430	2,500	908
HB	13.8	26.5	8.9	380	3,000	1,350

Table 9.4. ISM 250-2,000 Hz absorption coefficients. F=front wall, B=back wall, L/R=side walls, C=ceiling

	<i>F</i>	<i>B</i>	<i>L</i>	<i>R</i>	<i>C</i>
BK	0.20	0.20	0.20	0.20	0.05
ML	0.10	0.10	0.10	0.10	0.50
BM	0.10	0.10	0.10	0.10	0.05
HZ	0.05	0.05	0.05	0.05	0.05
MK	0.05	0.20	0.40	0.40	0.10
BZ	0.30	0.70	0.05	0.05	0.25
HB	0.30	0.05	0.05	0.05	0.90

9.3.4 Results

For each room, the results for the measurements are presented in Figures 8.9a to 8.9g. Per room, the graph on the left illustrates the $ST_{early,d}$ as a function of distance for the individual S-R positions (dots show the 5-rotation average) and a logarithmic trend line is shown over

all measurement S-R combinations (black line). The ISM prediction results for $ST_{early,d}$ are presented as a red dashed line. Besides that, the position averaged $ST_{early,d}$, ST_{late} and the reverberation time T_{mid} (T_{30} averaged for 500 and 1,000 Hz) are presented. The results for each room are briefly discussed in the figure captions.

In general, results for $ST_{early,d}$ show that (much) more early reflected sound energy is present in most of these rehearsal rooms compared to concert hall stages. A similar increase was found for measurements in the covered part of orchestra pits, however, even the smallest rehearsal room still has a lower $ST_{early,d}$ than the covered part of the orchestra pit. In most rooms, there is a limited decay of $ST_{early,d}$ over distance, which results in an almost equal distribution of early sound energy over the orchestra. Additionally, the ST_{late} shows that the amount of late reflected sound energy in these rehearsal rooms ($ST_{late} = -21$ to -10) can be similar to that on concert hall stages ($ST_{late} = -16$ to -12), even though the reverberation time is lower.

The ISM prediction results also show a moderate decay in $ST_{early,d}$ over distance, but, the decay over distance does not appear to be logarithmic but almost straight (sometimes even slightly bell shaped). In general, the predicted $ST_{early,d}$ is lower than the measured value, which might be due to the scattering of surfaces that is not included in the ISM prediction. Looking further into details, the ISM calculations reveal that the reduction of early reflected sound levels due to sound absorption is less than one might expect. For instance, the sound absorbing ceiling with $\alpha = 0.5$ in room ML, only reduces the 1st and 2nd order ceiling reflections by 3 dB. This amount of reduction is also achieved when almost doubling the source to ceiling distance. This explains why the $ST_{early,d}$ of room ML and HZ is very similar, even though the room volume is almost 2.5 times larger.

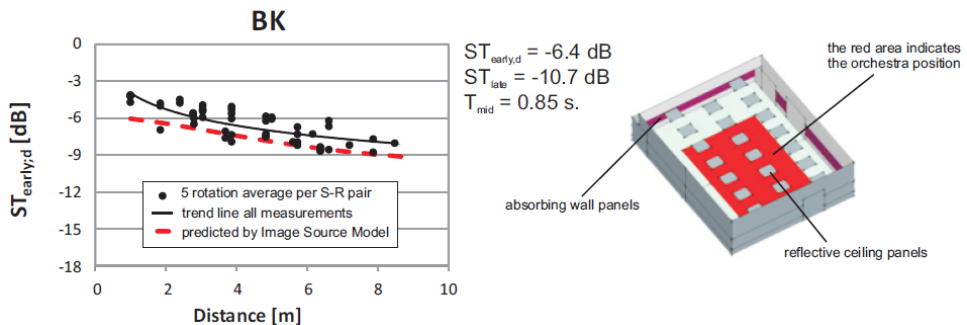


Figure 9.9a: BK is a 650 m³ temporary rehearsal room. It has a reflective ceiling at 4 m and 20 concave reflectors at 3 m height. Sound absorbing panels made out of perforated board are attached to 25% of the sidewalls surface. The $ST_{early,d}$ is high due to the 1st order ceiling and 2nd order ceiling to floor and wall to wall reflections. The steep trend in $ST_{early,d}$ is well predicted but 2 dB lower as measured.

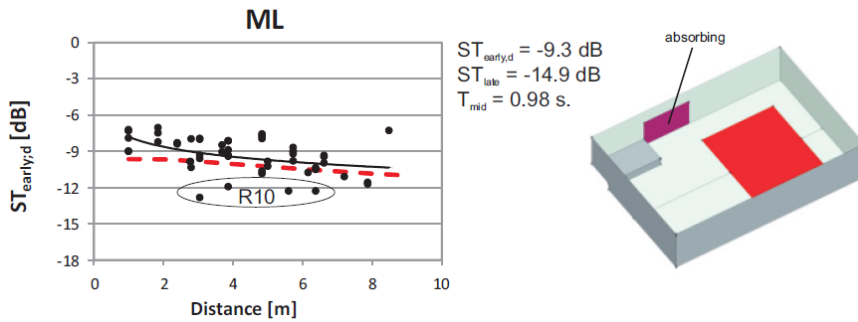


Figure 9.9b: ML is a 900 m³ typical rehearsal room that is also used as a community center. It has a lowered ceiling with absorptive and reflective tiles at 4 m height and reflective walls. The $ST_{early,d}$ is lower than in room BK having the same ceiling height but now partly absorbing. The trend in $ST_{early,d}$ is well predicted. The four outliers are results for position R10 at 1.9 m height in the middle of the room.

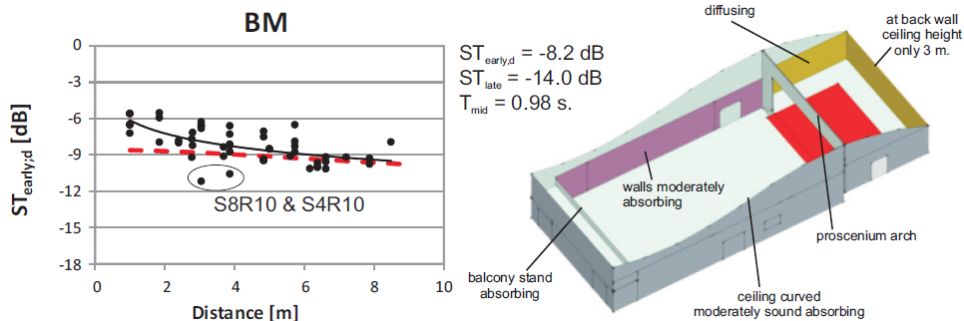


Figure 9.9c: BM is a 1,920 m³ multifunctional hall with a 100 m² stage with zigzag shaped stage walls (indicated in the figure by the yellow surface). The walls and ceiling in the hall are covered with open laths (moderately sound absorbing). The stage is a 450 m³ ‘box’ opened to the hall. The $ST_{early,d}$ is relatively high due to the small ‘stage volume’. The trend in $ST_{early,d}$ is well predicted above 4 m S-R distance. The outliers are results for position R10 (just outside the stage on a platform).

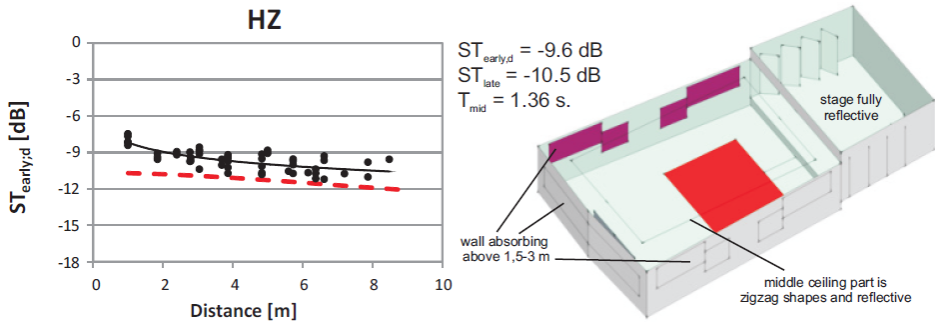


Figure 9.9d: HZ is a 2,250 m³ multifunctional hall with a stage. The orchestra prefers to rehearse in the lower section of the hall, having a reflective ceiling at 5.5 m which is diffusive. The walls are reflective at ear level and absorptive above. The $ST_{early,d}$ is almost equal to room ML. The predictions show that the energy reflected by the 5.5 m reflective ceiling is comparable to the 4.0 m ceiling in room ML which is 50% absorbing.

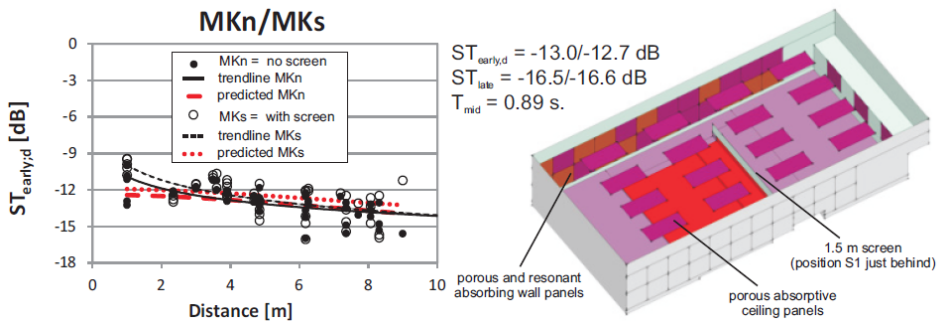


Figure 9.9e: MK is a 2,400 m³ rehearsal room in a large industrial hall. Sound absorbing panels have been applied to the side walls (50% porous and 50% resonant) and are hung from the ceiling. A 1.5 m high removable screen can be put at the back of the orchestra to shield the percussion. The $ST_{early,d}$ is about -12 dB which is within the preferred range. Adding the screen results in a 0.5 dB increase, which is also predicted well.

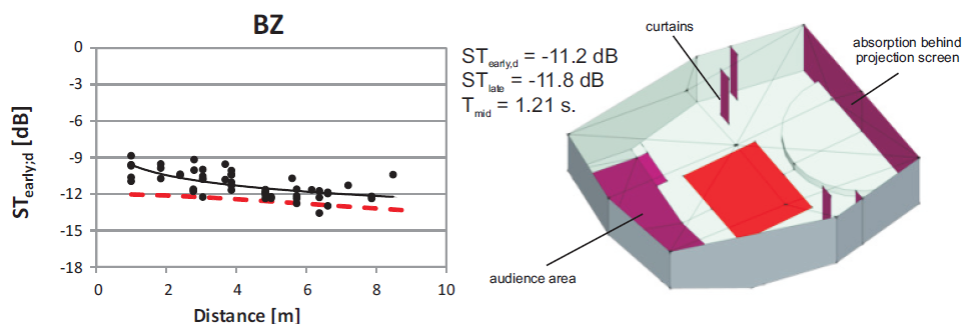


Figure 9.9f: BZ is a 2,500 m³ auditorium with inclined seating area. The orchestra prefers to rehearse in the middle of room. The back wall is sound absorbing, while the ceiling is egg-shaped with reflective and scattering panels. The $ST_{early,d}$ is about -11 dB which is within the preferred range. Similar to MK, in BZ the reflected sound level is equally dependent on the ceiling and side wall reflections.

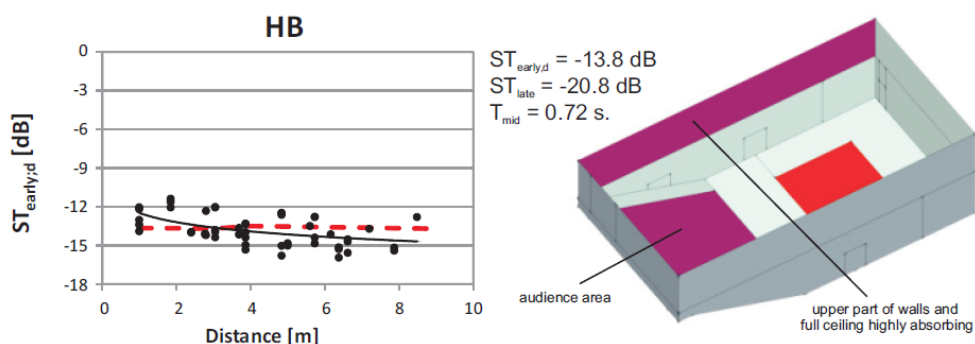


Figure 9.9g: HB is a 3,000 m³ multifunctional auditorium with inclining seating area. The lower parts of the side walls and the full back wall are reflective, while the full ceiling and upper side-walls are highly sound absorbing. The $ST_{early,d}$ is about -14 dB which is just below the preferred range for concert halls. The predictions show that, even though the T_{mid} is only 0.7 s. which is much lower than common for a concert hall, the walls provide the enough early reflected sound energy.

9.3.5 Discussion

In Figure 9.10, the $ST_{early,d}$ is presented as a function of room volume using blue diamonds. For the larger spaces above 1,500 m³, there appears to be a clear trend. However, the smaller rooms, BK and ML, appear to be exceptions. Based on the ISM predictions it can be concluded that the surfaces nearest to the orchestra determine the value of $ST_{early,d}$, which are often the ceiling and side walls. Therefore, it might be more appropriate to consider the ‘stage volume’ instead of the room volume in deep rooms. In Figure 9.10, the $ST_{early,d}$ is also presented as a function of ‘stage volume’ using red dots. Now, a trend can be observed for all measured rooms. Nevertheless, it appears that, for a 900 m³ stage volume, the dimensions, presence of a back wall and the material properties can still make a difference up to 3 dB in $ST_{early,d}$. Even with a larger room volume, the $ST_{early,d}$ can be very high due to a small stage volume (for instance room BM with a 1,920 m³ room volume and a 500 m³ stage volume, see Figure 9.9c).

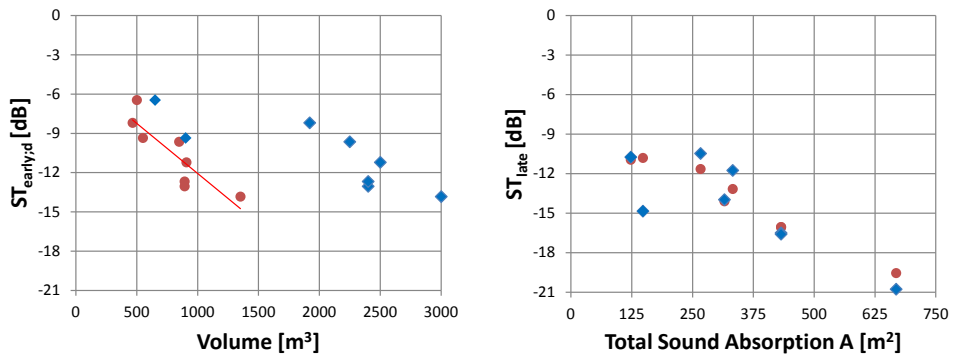


Figure 9.10: Left, $ST_{early,d}$ as a function of room volume (blue) and ‘stage volume’ (red, linear trend line $R^2 = 0.75$), Right, ST_{late} as a function of the total sound absorption, measured (blue) and predicted (red), see [section 4.3.4](#).

In Figure 9.10, right graph, the Late Support ST_{late} is presented as a function of the total amount of sound absorption A, derived from V and T_{30} using Sabine’s equation ($A = 0.161 T/V$). Besides, Barron’s revised theory is used to predict ST_{late} ($10\log(312T/V) - 6/T$) from an exponential decay. It is shown that, in 6 out of 7 cases, ST_{late} is predicted within 1.5 dB error (The single outlier is room ML and no reason has been found for this larger error). It is clear that a relation exists between the amount of sound absorption and the amount of late reflected sound energy (as also suggested by Barron’s revised theory).

9.3.6 Conclusion

Stage acoustic parameters have been studied for 7 different rooms, used for rehearsal and performances by wind orchestras. Both measurements and predictions were used to investigate the impact of sound absorption and room dimensions. In general, the Early Support over distance, $ST_{early,d}$, could be predicted within 2 dB error using the Image Source Method and 1st and 2nd order reflection only. This shows that the $ST_{early,d}$ is highly dependent on discrete sound reflections, of which its sound level depends on the distance between the sound source or receiver to the room surfaces and, to some extent, the surface properties like sound absorption and diffusion. The results suggest that, with a stage volume of at least 900 m³ a $ST_{early,d}$ value can be achieved between -13 and -11 dB, which is mentioned by Gade as a possible optimal range for ST_{early} . On smaller stages, it might be almost impossible to reduce $ST_{early,(d)}$ below -11 dB. With a good ‘stage design’ as a starting point, the rest of the rehearsal room could be designed to achieve appropriate late reverberation. Now, an assumption is made that a good orchestra surrounding can be designed in any room with a room volume larger than the appropriate ‘stage volume’ (this may lead to a ‘room in a room’ design). With Reverberation Time and Late Support available as design parameters, and the room volume possibly being dependent on other room requirements, one could look for an optimal balance between reverberation and late reflection sound level. Figure 9.11 shows the ST_{late} as a function of room volume for various reverberation times, based on Barron’s revised theory. From this graph, the reverberation time can be determined that might fit the available room volume for a full symphony orchestra to achieve the desired ST_{late} , see Table 9.5.

Table 9.5: optimal reverberation time for rehearsal rooms based on volume and $ST_{late} = -14$ dB.

V_{room}	1,000	2,000	3,000	4,000	5,000	6,000
T	0.75	1	1.25	1.35	1.5	1.75

It is clear that a room volume below 2,000 m³ is not a good choice for an orchestra rehearsal room, as reverberation must be very low to avoid a loud room. For a wind orchestra rehearsal room, a moderate reverberation might be desired, possibly with a maximum of 1.35 s (it is uncommon to perform in reverberant concert halls). This would ask for a 4,000 m³ rehearsal room to avoid the room being too loud. For a symphony orchestra, that may desire more reverberation, possibly up to 1.75 s, a 6,000 m³ volume would be needed for a rehearsal room. It is clear that the 7 measured rehearsal rooms do not fulfil any of such requirements.

It should be noted that all results in this section have been determined for rooms without the absorption and diffusion of the orchestra members on stage. As was shown later, see [chapter 4](#), the orchestra has the largest effect on acoustic parameters in the smallest rooms.

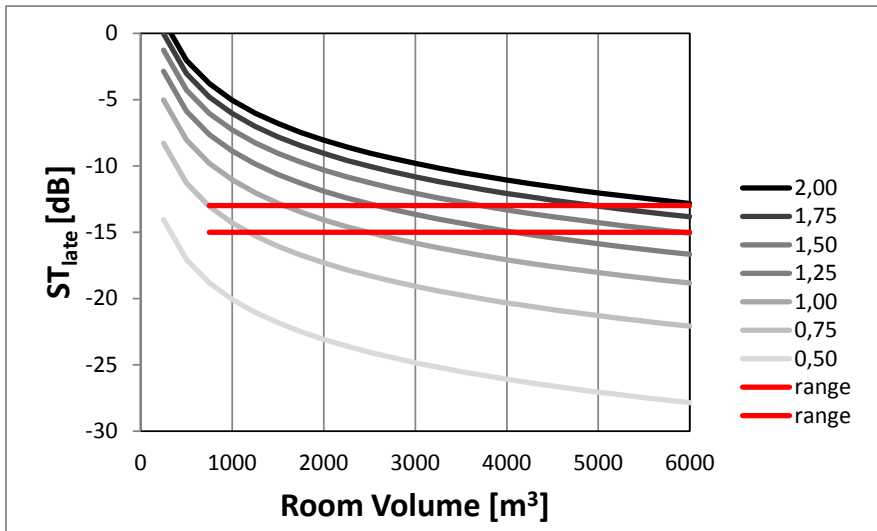


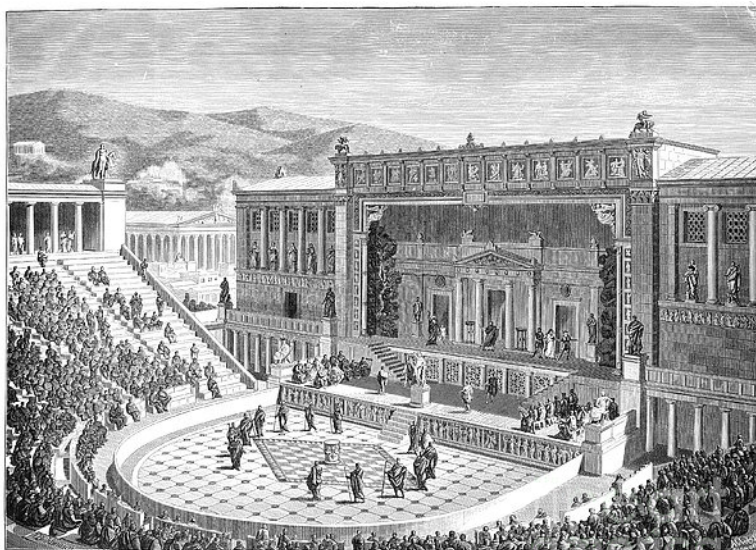
Figure 9.11: ST_{late} as a function of room volume for various reverberation times. Possible optimal range of -15 to -13 as mentioned by Gade.

9.4 Open air theatres in Greece

This section has been presented as part 4 of a series of papers on acoustic measurements in Greek open air theatres done by 6 master students supervised by Constant Hak and the author. Results have been presented at ICSV 23, 2016 in Athens [8,9,10,11].

9.4.1 Introduction

The stages of ancient theatres can be characterized by their circular shape called the ‘*Orchestra*’, in front of a rectangular elevated stage (‘*Proskenion*’) with a highly decorated back wall (‘*Skene*’) [12]. During performances, a group of musicians would be located in the circular shape, while actors were positioned on the elevated stage. The musicians could consist of singers and dancers (‘*chorus*’) sometimes accompanied by instruments like strings, flutes and percussion [13]. Only some clues are available to the preferred positioning of the musicians. For instance, in the Dithyramb plays, dedicated to the god Dionysus, a chorus consisting of up to 50 singers would be dancing in a circular shape [14]. However, these plays did not involve any actors and developed in Greece before the time of the theatres. In the theatre the exact centre of the ‘orchestra’ was commonly occupied by an altar, which would be used during ceremonies and not necessary during plays. It is likely that the chorus has been dancing around this centre in the theatre too. Figure 9.12 shows a sketch by J. Buhlmann [15] depicting a scene of a play taking place at the theatre of Dionysos in Athens. In the sketch a circular 12 person chorus and the altar in the middle of the ‘orchestra’ can be recognised. The Proskenion is occupied by a number of actors at random positions.



THEATRE OF DIONYSOS AT ATHENS.
Restored from recent excavations.

Figure 9.12: Theatre of Dionysos Athens, sketch by J. Buhlmann [15].

It can be assumed that in the ancient theatres, acoustic support must have been beneficial for performers similar to orchestral musicians in concert halls. As part of the Ancient Acoustics project [8, 9, 10], stage acoustic measurements have been carried out in the Odeon of Herodes Atticus and the theatre of Argos (Figure 9.13). Acoustic measurements have also been performed in the theatre of Epidauros, but due to time constraints and large crowds no stage acoustic measurements could be performed there.

The goal of this explorative study is to investigate the behaviour of the early and late reflected sound level at the ‘orchestra’ (circle) and on stage in their current state. Comparing the two ‘stages’ is particularly interesting, because in Argos the back wall is missing while in Odeon Herodes Atticus the back wall still exists. The Argos theatre has a severely damaged seating area while at the Odeon Herodes Atticus the seats have been renovated. During our measurements at Odeon Herodes Atticus, no stage floor was present.

The absolute value of the parameters might have limited value because both the audience area and stage were unoccupied during the measurements. It is not the goal to compare the acoustic conditions found in ancient theatres with those as preferred in modern concert halls, but to study their trends in relation to their geometry.

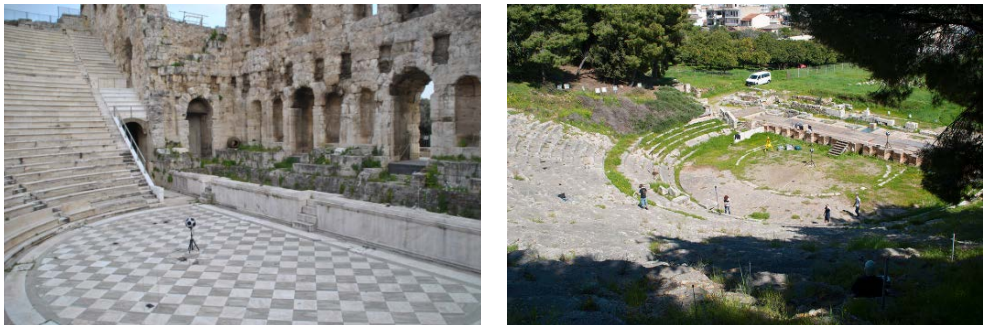


Figure 9.13: Odeon of Herodes Atticus (left) and the theatre of Argos (right).

It is well known that the typical concentric shape of the audience area gives a focused reflected sound to the stage. In most theatres, the center of the round shape of the seating area lies in the middle of the ‘orchestra’. When source and receiver are both at this center, for instance when clapping ones hands, a typical ‘intimate chirp’ sound, caused by focusing and a time-step increasing sequence of ‘staircase reflections’ can be observed [16]. Focusing also occurs when source and receiver are positioned point-symmetrically along the center of a circle.

An example is presented in Figure 9.14, which shows a reflection path analyses of a reconstructed Roman theatre [17]. A position on stage and a position in the ‘orchestra’ lie on the opposite side of the same circle with the ‘orchestra’s center in the middle. Sound ‘rays’ are reflected from both lower parts of the audience staircase and from the ‘Diazoma’ (horizontal passage between several rows of seats) via the canopy above the stage (not shown in the picture). The 3D model is constructed using flat partial surfaces, only showing one

‘ray’ hit per surface. In reality, a sound wave hitting a perfect circular shape will likely result in strong focusing of sound.

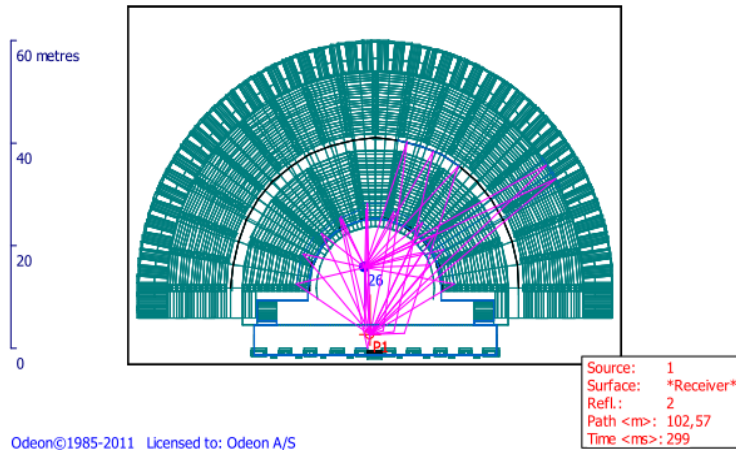


Figure 9.14: Early reflections from the stage position to a receiver in the orchestra. [17].

9.4.2 Measurement positions

A schematic plan with the numbering of the measurement positions is shown in Figure 9.15. In the Odeon of Herodes Atticus, see Figure 9.16, positions were selected on two half circles at 3 and 6 m from the centre of the ‘orchestra’, with 10 positions per half a circle. Source positions were selected in the middle of the ‘orchestra’ (S1) and on the half circles in forward direction (S2 and S3) and sideward direction (S4 and S5). Besides, 12 receiver positions were selected on the edge of the remainder of the original stage with a mutual distance of 2.5 m. Source positions S6 and S7 were on the stage edge, parallel to source positions S1 and S5. In the theatre of Argos, see Figure 9.13, the stage back wall is missing. Therefore, no source and receiver positions were selected on the stage edge. The positions in the ‘orchestra’ are equal to the positions used in the Odeon of Herodes Atticus.

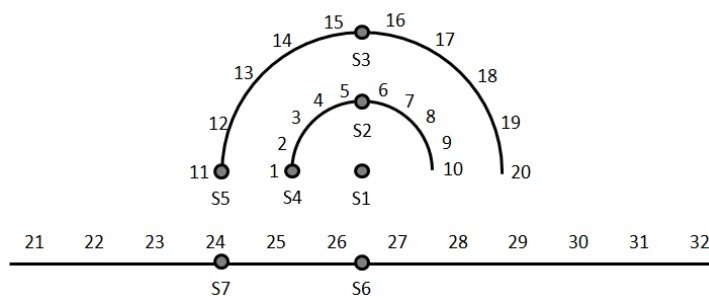


Figure 9.15: Source – receiver numbering.

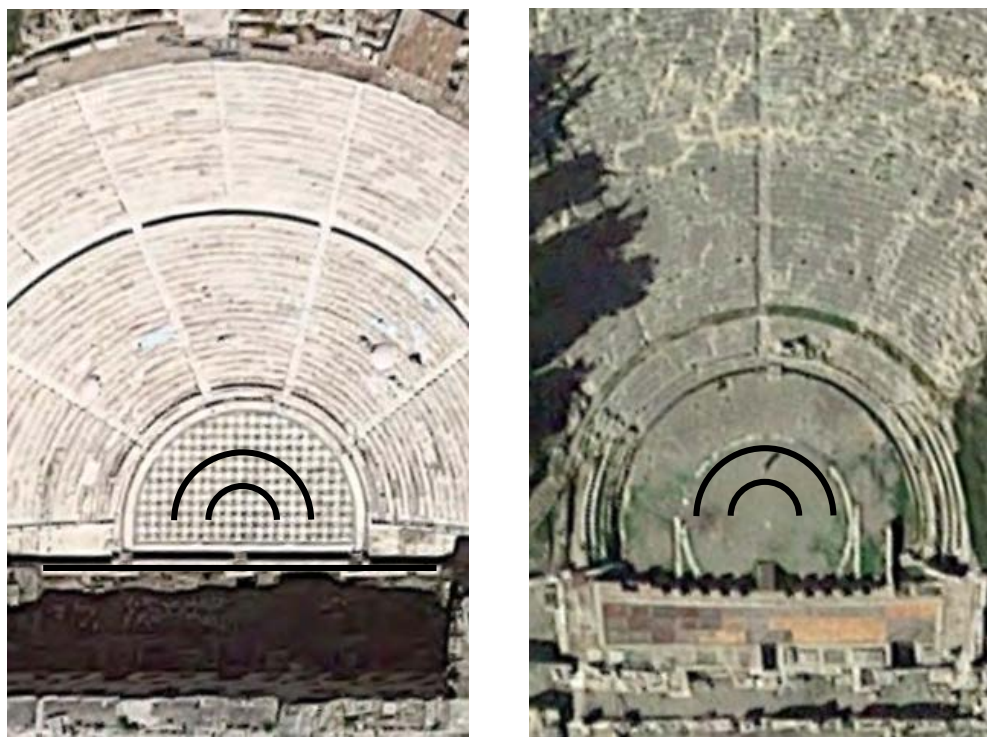


Figure 9.16: Schematic positions in the Odeon of Herodes Atticus (left) and the theatre of Argos (right).

9.4.3 Results

Figures 17a-c show the results for $ST_{early,d}$ and $ST_{late,d}$ as a function of source-receiver distance. For the Odeon of Herodes Atticus, Figure 9.17a only shows results for both source and receiver position at the ‘orchestra’ in front of the stage, while Figure 9.17b only shows results for positions with either source in the ‘orchestra’ and receiver on the stage edge or vice versa. For the theatre of Argos, only positions were measured at the ‘orchestra’, shown in Figure 9.17c. $ST_{early,d}$ measured in concert halls decays over distance and $ST_{late,d}$ is not dependent on distance. A similar trend can be observed in the current data, expressed by clouds of data points. However, in the ancient theatres, clear outliers from these clouds are found with a higher reflected early or late sound level compared to the ‘cloud’. These are indicated in the graphs by circles or lines.

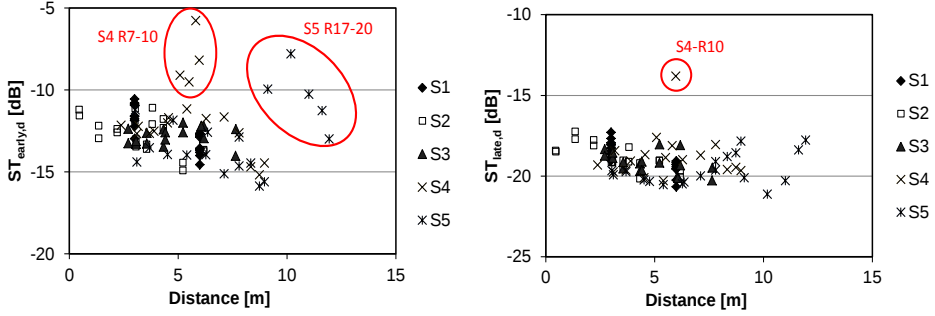


Figure 9.17a: $ST_{early,d}$ and $ST_{late,d}$ in Odeon of Herodes Atticus, positions in the ‘orchestra’.

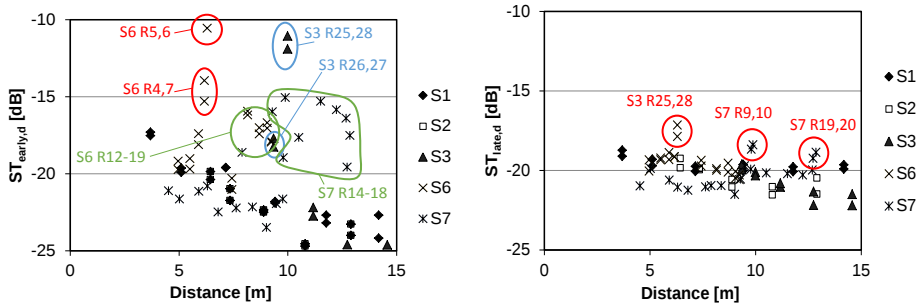


Figure 9.17b: $ST_{early,d}$ and $ST_{late,d}$ in Odeon of Herodes Atticus, positions from ‘orchestra’ to stage.

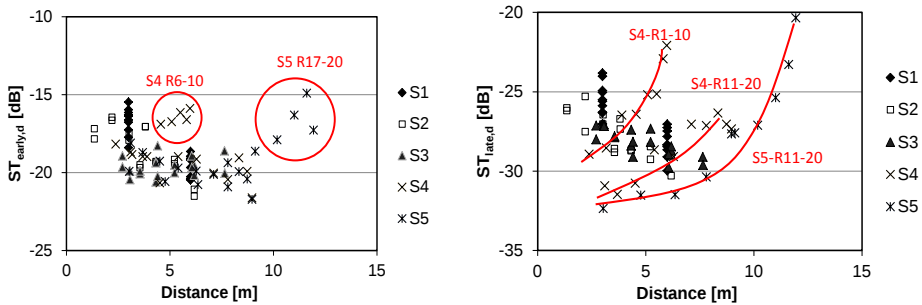


Figure 9.17c: $ST_{early,d}$ and $ST_{late,d}$ in theatre of Argos, positions in the ‘orchestra’.
The trend lines indicate results that correspond to a group of source or receiver positions.

In each theatre, S4R10 is one of the positions that clearly stands out from the ‘cloud’ by a stronger early and late reflected sound level. For further investigation of these positions, the ETCs (Energy Time Curves) for each theatre are presented in Figure 9.18. For comparison, the ETC of position S1R15 which has the same source-receiver distance as S4R10 (6 m), but without the focussing effect, can be observed.

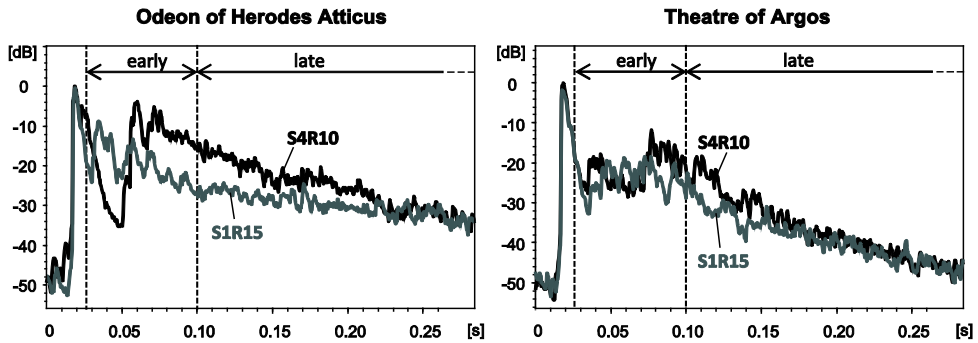


Figure 9.18: Smoothed broadband Energy Time Curves for positions S4R10 with focussing and S1R15 without focussing for both theatres. The direct sound is equally loud in each graph (equal distance).

9.4.4 Discussion

The results for the early and late reflected sound level, measured by the support parameters $ST_{early,d}$ and $ST_{late,d}$, show clear outliers for certain positions, see Figure 9.17. Over 5 dB higher levels are found when the source and receiver are positioned point-symmetrically along the center of the ‘orchestra’ compared to a non-symmetrical source-receiver positions. Examples of such positions are S4R10 versus S5R20 within the ‘orchestra’ and positions S6R5 versus S7R17 with one position in the ‘orchestra’ and one position on the stage edge. When listening to the measured impulse responses, the typical ‘chirp sound’ can be heard, which one also hears when clapping ones hands in the centre of the ‘orchestra’. The ETCs presented in Figure 9.18 illustrate that indeed multiple delayed reflections arrive at the receiver position (the decay curve is lifted), both before and after 103 ms after the emission of the sound (103 ms separates early from late in the ST parameters). The reflections are stronger than those in a different measured impulse response not being in the focal point. In the Odeon of Herodod Atticus, these reflections can be observed up to 225 ms. This delay corresponds to a path length of 77 m, which is approximately twice the distance from the ‘orchestra’ centre to the last row of the seats. This shows that even the last seating row causes focussing effects. In Argos, the increase in sound level is observed in the ETC up until 150 ms, which indicates that reflections from the severely damaged seating beyond 25 m do no longer cause focussed reflections.

An increased sound level is not limited to the exact focussing point. The results for $ST_{late,d}$, measured in Argos at positions S4R1 to S4R10 and S5R11 to S5R20, show that the reflected sound level gradually increases when positions are closer to the focal point. The reflected sound level is also stronger in positions close to the focal point measured in the Odeon of Herodus Attica, for example S7R14 to S7R18 and S6R12 to S6R19. It seems that this is only the case for positions on the same circle with the ‘orchestra’ centre in the middle. This trend is not observed for the positions S3R21 to S3R32 on a straight line: only positions S3R25 to S3R28 which are more or less on the same circle show an increased level caused by focussing.

At the positions where focussing occurs, the typical ‘chirp’ sound can be heard clearly. In a clapping experiment with two persons positioned point-symmetrically along the center of the circle, the effect is only heard by the listener (receiver) and not by the person clapping and listening (both source and receiver). Therefore, it will likely be unnoticed by most visitors of amphitheatres.

Literature suggests that the ‘chorus’, a group of singing and dancing individuals, might have formed circles around the centre of the ‘orchestra’. In this formation, one will hear strong focussed sound coming from other persons in the circle, especially from those standing on the opposite side. The temporal filtering, heard as a ‘chirp’ when clapping ones hand, will cause the voice to sound unnatural. The ‘chirp sound’ is quite audible and is heard by the many tourists visiting the theatres nowadays, and must have been heard by the performers in earlier times. It might even have had an effect on their performance, but whether the extra ‘support’ was beneficial is uncertain, because the ‘chirping’ can also cause unwanted colouration and poor balance among different performers. Besides, the question is whether the effect still exists when the theatres are occupied.

9.5 References

- [1] I. Fichera: How architectural design of stages influences the acoustic parameters. Master Thesis, Eindhoven University of Technology (2016).
- [2] P. Heijnen, M. Kivits: Podiumakoestiek – Een onderzoek naar de objectieve metingen en subjectieve belevingen. Master project report, Eindhoven University of Technology (2009).
- [3] J.J. Dammerud, M. Barron, E. Kahle: Objective assessment of acoustic conditions on concert hall stages – limitations and new strategies. Proc. of International symposium on room acoustics, IRSA (2010).
- [4] A.H.M. van der Heide: The acoustics of orchestra pits, a case study: Het Muziektheater Amsterdam. Master Thesis, Unit BPS, Eindhoven University of Technology (2011).
- [5] R.H.C. Wenmaekers, C.C.J.M. Hak: Early and late support measured over various distances: The covered versus open part of the orchestra pit. *Building Acoustics* 20 (2013) 313-322.
- [6] L. Schmitz: Music Rehearsal Room Acoustics: Ranking the ensemble conditions of music rooms intended for rehearsal using rhythmic sounds of indefinite pitch. Master Thesis, Eindhoven University of Technology (2014).
- [7] R.H.C. Wenmaekers, L.J.W. Schmitz, C.C.J.M. Hak: Early and late support over various distances: rehearsal rooms for wind orchestras. 7th Forum Acusticum, 07-12 September 2014, Cracow, Poland (2014).
- [8] C. Hak, N. Hoekstra, B. Nicolai, R. Wenmaekers: Project ancient acoustics part 1 of 4: A method for accurate impulse response measurements in large open air theatres. *Proceedings of ICSV Athens* (2016).
- [9] N. Hoekstra, B. Nicolai, B. Peeters, C. Hak, R. Wenmaekers: Project ancient acoustics part 2 of 4: Large-scale acoustical measurements in the Odeon of Herodes Atticus and the theatres of Epidauros and Argos. *Proceedings of ICSV Athens* (2016).
- [10] C. van Loenen, M. van der Wilt, A. Diakoumis, K. Hooymans, R. Wenmaekers, C. Hak: Project ancient acoustics part 3 of 4: Influence of geometrical and material assumptions on ray-based acoustic simulations of two ancient theatres. *Proceedings of ICSV Athens* (2016).
- [11] R. Wenmaekers, B. Nicolai, N. Hoekstra, C. Hak: Project ancient acoustics part 4 of 4: Stage acoustics measured in the Odeon of Herodes Atticus and the theatre of Argos. *Proceedings of ICSV Athens* (2016).
- [12] S.L. Vassilantonopoulos, J.N. Mourjopoulos: A study of ancient Greek and Roman theatre acoustics. *Acta Acoustics united with Acustica* 89 (2003) 123-130.
- [13] J.H. Rindel, A.C. Gade, M. Lisa: The virtual reconstruction of the ancient roman concert hall in Aphrodisiac, Turkey. *Proceedings of the Institute of Acoustics*, Vol 28, Pt.2, 316-323 (2006).
- [14] Wikipedia, see website <https://en.wikipedia.org/wiki/Dithyramb>
- [15] J.C. Ridpath. *Ridpath's History of the World - Volume 2*, New York, Merrill & Baker (1897).
- [16] N.F. Declerq, C.S.A. Dekeyser: Acoustic diffraction effects at the Hellenistic amphitheater of Epidauros: Seat rows responsible for the marvellous acoustics. *J. Acoust. Soc. Am.* 121 (4) (2007) 2011-2022.
- [17] J.H. Rindel: Roman Theatres and Revival of Their Acoustics in the ERATO project. *Acta Acoustics united with Acustica* 99 (2013) 21-29.

Chapter 10

Biography and list of publications

Remy Wenmaekers was born on the 3rd of July in 1982 in Stein, the Netherlands. In 2008, he graduated with honours as a Master of Science in Architecture, Building and Planning, with a specialization in Building Physics (track “Physics of the Built Environment”). Since 2006, he is acoustic consultant, researcher and teacher at the company Level Acoustics & Vibration (LA&V) housed in the acoustics laboratory of the department of the Built Environment at Eindhoven University of Technology (TU/e). Its mission is to transfer knowledge by participating in innovative building designs and applied research and by organising master classes. Remy has worked on various building projects developing solutions to optimise the room acoustics and the insulation of sound. He applies various laboratory and field measurements and prediction modelling methods. He is teacher by giving lectures, organising workshops/courses and supervising students at TU/e and other universities.

Examples of projects where Remy Wenmaekers was involved are:

- office buildings (e.g. ministry building Rijnstraat 8 The Hague, architect OMA)
- auditoria (e.g. RIVM Utrecht, Felix Claus Dick van Wageningen Architecten)
- concert venues (e.g. Theatre de Vest, Alkmaar, architect TenBrasWestinga)
- outdoor sports venues (e.g. noise barrier for Raceway Venray)
- master classes on room and building acoustics (e.g. Eddy Gerretsen, Anders Gade)
- lectures and workshops (e.g. room acoustics for Hogeschool van Amsterdam)
- measurement expeditions with students (e.g. Amphitheatres Greece)

Working at Level Acoustics & Vibration, Remy shares results from research related to projects at scientific conferences. Preliminary research on the concert hall stage of Casa da Musica by Renz van Luxemburg, the founder of LA&V, was the basis for a PhD research proposal on the topic of stage acoustics, which was accepted by TU/e in 2010. Remy published his first journal paper on stage acoustics in 2012 while working for LA&V. In 2013 Remy acquired funding at NWO for a 50% part time appointment at TU/e for four years to be able to dedicate more time on the research project. The final result of this project is current thesis containing 6 published/submitted 1st author journal papers on stage acoustics. Besides, he (co)authored 4 published journal papers, 31 conference papers and 8 professional publications on various topics in acoustics.

10.1 Peer reviewed journal papers

10.1.1 1st Author on stage acoustics

Wenmaekers, R.H.C., Nicolai, B., Hornikx, M.C.J. & Kohlrausch A.G. (2017). Reduction of orchestral musicians' sound exposure by modifying riser height, available space, screening and acoustic conditions. Submitted.

Wenmaekers, R.H.C., Hak, C.C.J.M. Hornikx, M.C.J. & Kohlrausch A.G. (2017). Sensitivity of stage acoustic parameters to source and receiver directivity: Measurements on three stages and in two orchestra pits. *Applied Acoustics* 123, 20–28.

Wenmaekers, R.H.C., Hak, C.C.J.M. & Hornikx, M.C.J. (2016). How orchestra members influence stage acoustic parameters on five different concert hall stages and orchestra pits. *Journal of the Acoustical Society of America*, 140(6), 4437-4448.

Wenmaekers, R.H.C. & Hak, C. C. I. M. (2015). The sound power as a reference for sound strength (G), speech level (L) and support (ST): uncertainty of laboratory and in-situ calibration. *Acta Acustica united with Acustica*, 101(5), 892-907.

Wenmaekers, R.H.C. & Hak, C.C.J.M. (2015). A sound level distribution model for symphony orchestras: possibilities and limitations. *Psychomusicology: Music, Mind & Brain*, 25(3), 219-239.

Wenmaekers, R.H.C., Hak, C.C.J.M., and Luxemburg, L.C.J. van (2012). On measurements of stage acoustic parameters: Time interval limits and various source-receiver distances. *Acta Acustica united with Acustica*, 98, 776–789.

10.1.2 1st Author or co-author on other topics

Wenmaekers, R.H.C., Hak, C.C.J.M. & Hornikx, M.C.J. (2014). The effective air absorption coefficient for predicting reverberation time in full octave bands. *Journal of the Acoustical Society of America*, 136, 3063-3071.

Hornikx, M.C.J., Hak, C.C.J.M. & **Wenmaekers, R.H.C.** (2015). Acoustic modelling of sports halls, two case studies. *Journal of Building Performance Simulation*, 8(1), 26-38.

Hak, C.C.J.M., **Wenmaekers, R.H.C.**, and Luxemburg, L.C.J. van (2012). Measuring room impulse responses: Impact of the decay range on derived room acoustic parameters. *Acta Acustica united with Acustica*, accepted for publication 2 september 2012.

Hak, C.C.J.M., **Wenmaekers, R.H.C.** (2010), The Impact of Sound Control Room Acoustics on the Perceived Acoustics of a Diffuse Field Recording, *Wseas transactions on signal processing*, Issue 4, Volume 6, October 2010.

10.2 Conference papers

10.2.1 Stage acoustics or sound exposure musicians

Wenmaekers, R.H.C., Hak, C.C.J.M. Hornikx, M.C.J. & Kohlrausch A.G. (2017). Methods to measure stage acoustic parameters: guidelines for future research. ASA/Forum Acusticum Boston.

Wenmaekers, R.H.C., Hak, C.C.J.M. Hornikx, M.C.J. & Kohlrausch A.G. (2017). Symphony orchestra musicians: reduction of sound exposure by physical measures. EFAS Interlaken.

Wenmaekers, R.H.C. & Hak, C.C.J.M. (2015). Noise exposure of musicians the own instruments sound compared to the sound from others. Euronoise 2015, May 31 - June 3, 2015, Maastricht, The Netherlands (pp. 1-4).

Wenmaekers, R.H.C. & Hak, C.C.J.M. (2015). How a full orchestra of dummies attenuates direct and reflected sound. Auditorium Acoustics 2015 Paris (pp. 1-8).

Wenmaekers, R.H.C., Schmitz, L.J.W. & Hak, C.C.J.M. (2014). Early and late support over various distances: rehearsal rooms for wind orchestras. 7th Forum Acusticum, 07-12 September 2014, Cracow, Poland (pp. 1-7).

Wenmaekers, R.H.C., Hak, C.C.J.M. & Vos, de, H.P.J.C. (2013). Binaural sound exposure by the direct sound of the own musical instrument. Proceedings of the International Symposium on Room Acoustics 2013, 9-11 June 2013, Toronto, Canada (pp. 1-9).

Wenmaekers, R.H.C. & Hak, C.C.J.M. (2013). Early and late support measured over various distances : the covered versus open part of the orchestra pit. Proceedings of the International Symposium on Room Acoustics (ISRA 2013), 9-11 June 2013, Toronto, Canada (pp. 1-8).

Wenmaekers, R.H.C., Hak, C.C.J.M. & Luxemburg, van, L.C.J. (2011). The influence of Room Acoustic Aspects on the Noise Exposure of Symphonic Orchestra Musicians. 10th International Congress on Noise as a Public Health Problem (pp. 1-8). (pt.3, No. 33).

Wenmaekers, R.H.C., Hak, C.C.J.M. & Luxemburg, van, L.C.J. (2011). A Model for the prediction of Sound Levels within a Symphonic Orchestra based on measured Sound Strength. Proceedings of Forum Acusticum 2011 Aalborg (pp. 1-6).

Luxemburg, van, L.C.J., **Wenmaekers, R.H.C.,** Hak, C.C.J.M. & Kivits, M. (2010). Stage acoustics, further development of parameter LQ7-40. Proceedings of Internoise 2010 (Noise and sustainability), Lisbon, Portugal, June 2010 (pp. 1-8).

Wenmaekers, R.H.C., Hak, C.C.J.M. & Luxemburg, van, L.C.J. (2010). De invloed van het orkest op de akoestiek van de podiumomgeving. NAG dag 24 november 2010.

10.2.2 Ancient amphitheatres (project Ancient Acoustics)

Hak, C.C.J.M., Hoekstra, N., Nicolai, B. & **Wenmaekers, R.H.C.** (2016). Project Ancient Acoustics Part 1 of 4 : a method for accurate impulse response measurements in large open air theatres. 23rd ICSV, 10-14 July 2016, Athens, Greece (pp. 1-8).

Hoekstra, N., Nicolai, B., Peeters, B.P.A., Hak, C.C.J.M. & **Wenmaekers, R.H.C.** (2016). Project Ancient Acoustics Part 2 of 4 : large-scale acoustical measurements in the Odeon of Herodes Atticus and the theatres of Epidaurus and Argos. 23rd ICSV, 10-14 July 2016, Athens, Greece (pp. 1-8).

van Loenen, R.C., van der Wilt, M., Diakoumis, A., **Wenmaekers, R.H.C.** & Hak, C.C.J.M. (2016). Project Ancient Acoustics Part 3 of 4 : influence of geometrical and material assumptions on ray-based acoustic simulations of two ancient theatres. 23rd ICSV, 10-14 July 2016, Athens, Greece (pp. 1-8).

Wenmaekers, R.H.C., Nicolai, B., Hoekstra, N. & Hak, C.C.J.M. (2016). Project Ancient Acoustics Part 4 of 4 : stage acoustics measured in the odeon of herodes atticus and the theatre of argos. 23rd ICSV, 10-14 July 2016, Athens, Greece (pp. 1-8).

10.2.3 Concert hall measurement methods

Hak, C.C.J.M. & **Wenmaekers, R.H.C.** (2015). Measuring impulse responses in a fully occupied concert hall. Auditorium Acoustics 2015 Paris (pp. 1-9).

Wenmaekers, R.H.C. & Hak, C.C.J.M. (2014). Reference level in ISO 3382 parameters: G, ST, L and STI. Proceedings of DAGA, 40. Jahrestagung für Akustik, 10-13 Maerz 2014, Oldenburg, Deutschland (pp. 1-2).

Hak, C.C.J.M., **Wenmaekers, R.H.C.**, Hak, J.P.M. & van Luxemburg, A.P.D. (2011). The Source Directivity of a Dodecahedron Sound Source determined by Stepwise Rotation. Proceedings of Forum Acusticum, Aalborg, 27 June-1 July 2011 (pp. 1875-1879).

Hak, C.C.J.M., **Wenmaekers, R.H.C.**, Hak, J.P.M., Luxemburg, van, L.C.J. & Gade, A.C. (2010). Sound Strength Calibration Methods. Proceedings of 20th International Congress on Acoustics, ICA 2010 Sydney: Australian Acoustical Society.

Hak, C.C.J.M., Haaren, van, M.A.J., **Wenmaekers, R.H.C.** & Luxemburg, van, L.C.J. (2009). Measuring room acoustic parameters using a head and torso simulator instead of an omnidirectional microphone. Proceedings of Internoise 2009 Ottawa, Canada.

Hak, C.C.J.M., Hak, J.P.M. & **Wenmaekers, R.H.C.** (2008). INR as an Estimator for the Decay Range of Room Acoustic Impulse Responses. Proceedings of the 124th AES conference, Amsterdam 2008 (pp. 7434).

10.2.4 Recording room acoustics

Hak, C.C.J.M. & **Wenmaekers, R.H.C.** (2015). Room in room acoustics: the influence of the direct/diffuse sound field ratio in a listening room on played back recorded acoustics. Euronoise 2015, May 31 - June 3, 2015, Maastricht, The Netherlands (pp. 1-5).

Hak, C.C.J.M. & **Wenmaekers, R.H.C.** (2013). Room in room acoustics: using convolutions to find the impact of a listening room on recording acoustics. Proceedings of the International Symposium on Room Acoustics (ISRA 2013), 9-11 June 2013, Toronto, Canada (pp. 1-9).

Wenmaekers, R.H.C. & Hak, C.C.J.M. (2009). A comparison of the variable acoustics of two foley studios for sound effects recording. Proceedings of the International Conference on Acoustics including the 35th German Annual Conference on Acoustics (NAG/DAGA 2009), 23-26 March 2009, Rotterdam, The Netherlands.

Hak, C.C.J.M. & **Wenmaekers, R.H.C.** (2008). The Effect of room acoustics on the perceived acoustics of reproduced sound. Proceedings of the 37th International Congress on Noise Control Engineering (pp. 8). Shanghai-China: Internoise 2008.

10.2.5 Office acoustics

Wenmaekers, R.H.C. & Hak, C.C.J.M. (2015). Spatial decay rate of speech in open plan offices: the use of D2,S and Lp,A,S,4m as building requirements. Euronoise 2015, May 31 - June 3, 2015, Maastricht, The Netherlands (pp. 1-6).

Wenmaekers, R.H.C., Hout, van, N.H.A.M., Luxemburg, van, L.C.J. & Hak, C.C.J.M. (2009). The effect of room acoustics on the measured speech privacy in two typical European open plan offices. Proceedings of Internoise 2009 Ottawa, Canada.

10.2.6 Other topics

Hak, C.C.J.M., Hout, van, N.H.A.M., **Wenmaekers, R.H.C.** & Hornikx, M.C.J. (2013). Sound field in shoebox-shaped sports halls with non-uniformly distributed sound absorption: a case study. Proceedings of the AIA-DAGA 2013, 18-21 March 2013, Merano, Italy.

Wenmaekers, R.H.C., Hout, van, N.H.A.M., Luxemburg, van, L.C.J. & Rindel, J.H. (2010). Workshop on Advanced Room Acoustic Prediction Modelling. Proceedings of 20th International Congress on Acoustics, ICA 2010 Sydney: Australian Acoustical Society.

Wenmaekers, R.H.C., van der Aa, B.A., Pronk, A.D.C., Couthinho, A. & Luxemburg, van, L.C.J. (2009). The Sound Insulation of Water. NAG/DAGA 2009 (pp. 1339-1341).

Wenmaekers, R.H.C., Hak, C.C.J.M., Martin, H.J. & Luxemburg, van, L.C.J. (2008). Air Absorption Error in Room Acoustical Modeling. Proceedings of the 155th ASA conference, Paris 2008 (pp. 3129-3134).

10.3 Professional publications

Wenmaekers, R.H.C. & Hak, C.C.J.M. (2017). Akoestiek meten met mensen in de zaal. *Natuurkunde*, 2017.

Hak, C.C.J.M., **Wenmaekers, R.H.C.**, van der Wilt, M., Hoekstra, N., Diakoumis, A., van Loenen, R.C., Peeters, B.P.A. & Nicolai, B. (2016). Akoestiek van openluchttheaters in kaart gebracht. *Bouwfysica*, 2016(3).

Wenmaekers, R.H.C. & de Vos, H.P.J.C. (2016). Akoestiek en gehoorbelasting bij orkestmusici. *Geluid*, 2016(4), 6-9.

Hak, C.C.J.M., **Wenmaekers, R.H.C.** & Luxemburg, van, L.C.J. (2012). Geavanceerde impulsresponsie-meettechnieken: toepassingen bij muziek- en spraakoverdracht. *Bouwfysica*, 2012(3):9.

Postma, B.N.J., Luxemburg, van, L.C.J., Hak, C.C.J.M. & **Wenmaekers, R.H.C.** (2012). Zaalakoestisch ontwerpen ten tijde van de tweede helft 19e eeuw. *Bouwfysica*, 2012(3):1.

Wenmaekers, R.H.C., Hak, C.C.J.M., Hout, van, N.H.A.M., Heide, van der, A.H.M. & Luxemburg, van, L.C.J. (2012). De 'podiumakoestiek' van de orkestbak van het muziektheater Amsterdam. *Bouwfysica*, 2012(3), 13-17.

Weerd, van der, J., **Wenmaekers, R.H.C.** & Wit, de, M.H. (2007). De zonsverduistering in Turkije op 29 maart 2006 : meting versus beleving. *Bouwfysica*, 18(1), 14-18.

Weerd, van der, J., **Wenmaekers, R.H.C.** & Wit, de, M.H. (2007). De zonsverduistering in Turkije van 29 maart 2006. Het licht tijdens het verloop van de verduistering. *Zenit*, 34(1), 14-19.

10.4 Other achievements

23 April 2013:	Funding received for the research project at NWO
17 May 2013:	Article in Brabants Dagblad “Geluid concertzaal onder de loep”.
31 May 2014:	Presentation during music festival “Muziek op de Dommel”.
9 July 2014:	Article in Eindhovens Dagblad “Dummy-orkest in het Muziekgebouw”.
9 July 2014:	Radio interview on NPO Radio 6.
11 July 2014:	Radio interview on NPO Radio 4.
20 Oct 2014:	Organisation of the event “Architectural Acoustics” and presentation during the Dutch Design Week 2014 in Eindhoven with a 1,000 person audience. Publications in various magazines and two radio interviews.
14 jan. 2015:	Radio interview on NPO Radio 5 “De kennis van nu”.
28 Febr. 2015:	Article in Het Parool “Doof van de subtielste symfonieën”.
March 2016:	Measurement expedition “Ancient Acoustics” to three ancient theaters in Greece
10 april 2015:	Presentation during the Fysicadag at TU/e with a 300 person audience.
24 Oct. 2015:	Presentation at the event “Ancient Acoustics” during the Dutch Design Week 2015 in Eindhoven with a 100 person audience.
12 Nov. 2015:	Invited lecture during a Dutch AES meeting.
Juli 2016:	Presentation of four conference papers about project ‘Ancient Acoustics’ at the 23th International Congress on Sound & Vibration in Greece.
2010 - 2017:	Supervisor of bachelor and master projects and thesis: - Angela van der Heide - Rick de Vos - Lennart Schmitz - Amanda Nitidara (Indonesia) - Wouter Wittebol - Bareld Nicolai - Ilaria Fichera (Italy) - Johannes Wagner (Germany)
2010 - 2017:	Reviews for Journals: - Journal of Building Performance Simulation - Journal of Audio Engineering Society - Journal of the Acoustic Society of America - Journal of Building and Environment - Journal of Building Acoustics - Journal of Noise Control Engineering Journal

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Paloma Cecilia Guzman Molina

Stage acoustics and sound exposure in performance and rehearsal spaces for orchestras: Methods for physical measurements

Good acoustics in the concert hall is important for the audience to be able to fully appreciate music. Additionally, good acoustics is necessary for symphony orchestra members to be able to hear each other and to optimise their performance. This research is focussed on the so called 'stage acoustics' in performance and rehearsal spaces and its judgement by physical measurements. Such measurements are used by acousticians during the design of a performance space or rehearsal room to control or check its acoustic quality for unamplified orchestral music. The goal of current research is to further develop the procedures for physical measurements on stage. A special interest is the relation between the outcome of the measurements and the stage and building design. Another acoustic aspect related to the performance of music is the exposure of musicians to high sound levels. Sound levels are often that high that musicians risk damaging their ears. The topic sound exposure is also dealt with in the current research.

The work presented in this dissertation contributes to an understanding of how stage acoustic parameters are measured most accurately and how they can be used to study sound levels on stage. Guidelines are presented that will assist researchers in future studies and engineers in designing or judging stages and rehearsal rooms.

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