SIMULATIONS, MEASUREMENTS AND AURALISATIONS IN ARCHITECTURAL ACOUSTICS

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Room acoustic computer modelling has become an important tool in the acoustical design of rooms, and also the range of applications has increased in recent years. Also the room acoustic measurement technique has developed significantly in recent years, e.g. by new methods in ISO 18233. Considering computer modelling as a simulated measurement means that there is a close connection to the measurement methods, particularly as laid down in the ISO 3382 series that covers performance spaces, open plan offices and ordinary rooms. With these new standards the number of room acoustic parameters has grown, so in addition to the traditional reverberation time there is today a rather long list of more specialised parameters. The parameters are used for the design specifications, for the simulations during the design, and finally for the verification measurements. In some projects with special acoustical demands the use of auralisation in the design phase has become a useful supplement to the calculated parameters. In this paper the advantages and weaknesses of room acoustic measurements compared to simulations are discussed, and the state-of the-art methods as implemented in the ODEON room acoustics software are briefly presented with some examples. The measured impulse response is often used as a true reference of a real room impulse response and geometrical acoustic simulations are considered to be only a crude representation of it. However, both approaches have their own challenges and limitations. Geometrical acoustic models do not include wave phenomena, such as interferences and diffraction, as they simplify sound propagation by rays. The advantages of acoustic simulations with such models include a perfectly omnidirectional and impulsive sound source, no distortion problems, full control of the background noise, and a well-defined onset time of the impulse response. On the other hand, impulse response measurements include wave phenomena, but they do have their own weaknesses, which may cause significant errors in the derivation of the ISO-3382 room acoustic parameters. Due to the presence of background noise in the measured impulse response it is difficult to evaluate which part of the impulse response is valid. In addition, the directivity of the sound source used for measurements often has strong lobes at high frequencies and distortion artefacts may cause errors in the derived results. In this paper simulated and measured parameters are compared in a number of well documented cases and the various sources of errors are discussed. It is concluded that doing room acoustic measurements correctly may be more difficult than it appears at first glance, and both measurements and simulations require high level acoustical qualifications by the operator.

1. Introduction

The measured impulse response is often used as a true reference of a real room impulse response and geometrical acoustic simulations are considered to be only a crude representation of it. However, both approaches have their own challenges and limitations. Geometrical acoustic models do not include wave phenomena, such as interferences and diffraction, as they simplify sound propagation by rays. The advantages of acoustic simulations with such models include a perfectly omni-directional and impulsive sound source, no distortion problems, full control of the background noise, and a well-defined onset time of the impulse response. On the other hand, impulse response measurements include wave phenomena, but they do have their own weaknesses, which may cause significant errors in the derivation of the room acoustic parameters. When there are differences between measured and simulated room acoustic parameters it is not obvious which one is the most reliable.

2. Room acoustic parameters

The room acoustic parameters described in the international standard ISO 3382-1 [1] are the reference for objective evaluation of acoustics in rooms from impulse responses. Evaluation of some of the ISO 3382-1 parameters for performance spaces is an important part of an acoustic report for a new or existing hall. The parameters can be derived either by measuring the acoustic impulse responses of existing rooms or by means of simulation, e.g. with some of the available geometrical acoustics algorithms. Both measurements and simulations have their own strengths and limitations. In any case it is not the question whether to simulate or measure the parameters, indeed we need both. If the room does not exist yet, simulations are useful in order to predict and optimize the acoustics, and when the same room has been built measurements are useful for documentation. When an existing room is to be refurbished, measurement of acoustic parameters in the room is an invaluable input in order to objectively evaluate the acoustics under existing conditions and as input to the simulation process, so that the initial simulation model can be calibrated to best mimic the existing conditions before starting to simulate changes. Precision of measurements and simulations are equally important – indeed making decisions based on imprecise measurement results or calibrating a simulation model to fit imprecise measurement data is just as bad as imperfect simulations. This has been one of our major motivations for implementing robust measurement facilities into the ODEON Room Acoustics Software, which is not too sensitive to user interaction or measurement conditions.

Impulse response measurements are important for the analysis of the acoustics in any kind of room, small or large, simple or complex. An impulse response is simply the response of a room to a Dirac function emitted as a sound signal from a source. In principle more than one source can be used for the impulse response excitation, but for ISO 3382-1 measurements only one omni-directional source should be used. The ISO 3382-1 standard gives the framework for measurement of room acoustic parameters, but lacks detail on the requirements needed for derivation of certain room acoustic parameters as discussed by Hak *et al.* [2]. One of the major problems is the truncation of the impulse response at the correct time. Any recording of a room impulse response is likely to have a degree of background noise, due to the ambient noise in the room and/or to the noise of the measuring equipment. This background noise is visible at the cease of the impulse response and needs to be left out of the analysis. Otherwise the real energy decay in the room might be misinterpreted, often leading to longer reverberation times. The truncation according to ISO 3382-1 can be done manually, without any guidelines given. This can be a source of serious errors, if not performed carefully for the different octave bands considered.

Another important aspect in the post-processing of an impulse response is correct detection of the onset time, i.e., the arrival of direct sound from the source to the receiver – this is tricky as the real life sound source will not produce a perfect Dirac function. Careless post-processing can result in large differences between measured and simulated results for parameters such as clarity C_{80} [1, eq. (A.10)], which is the ratio of energy in the impulse response before and after a time limit of 80 ms:

$$C_{80} = 10 \cdot \lg \frac{\int_0^{80\,\mathrm{ms}} p^2(t)\,\mathrm{d}t}{\int_{80\,\mathrm{ms}}^{\infty} p^2(t)\,\mathrm{d}t} \quad (\mathrm{dB}) \tag{1}$$

where $p(t)$ is the sound pressure as a function of time t in the impulse response.

In this paper a selection of the most important ISO 3382-1 parameters is investigated in terms of *measurements* and *simulations*. The differences are discussed and their significance is concluded within the frame of the corresponding Just Noticeable Difference – JND. **Table 1** shows the parameters used in the present study, together with the respective JND. Both *measurements* and *simulations* are carried out with the ODEON Room Acoustics Software, version 12.1.

Table 1. Room Acoustic Parameters investigated in this paper. All parameters are derived by formulas given in the ISO 3382-1 standard [1].

ISO 3382 Parameter	Symbol	Subjective Limen
Early Decay Time	*EDT* [s]	5%
Reverberation Time (20 dB range)	T_{20} [s]	5%
Reverberation Time (30 dB range)	T_{30} [s]	5%
Clarity (50 ms)	C_{50} [dB]	1 dB
Clarity (80 ms)	C_{80} [dB]	1 dB
Definition	D_{50}	0.05
Centre Time	T_s [s]	10 ms
Sound Strength	G [dB]	1 dB

3. Comparison of measurement and simulation technology

In contrast to impulse response simulations, measurements may be considered accurate in a broader frequency range due to the actual representation of wave phenomena (interaction due to phase shifts, diffraction etc.). Input data such as *absorption* and *scattering* coefficients are inherent and the room geometry is fully included by definition. On the other hand, a group of limitations, such as imperfect omni-directional sources, presence of background noise and distortion due to the loudspeaker and the filtering required impose errors in the final results. **Table 2** summarizes the facts associated with existing measurement and simulation processes. The main issues for measurements are those related to the sound source and the background noise. For the simulations the most important issues are the uncertainly of material data and the approximation of the wave phenomena.

Table 2. Facts associated to measurements and simulations.

Facts	Measurements	Simulations
Room geometry	Fully included by definition	Approximated
Alteration of room geometry	Difficult	Easy
Wave phenomena (phase information, diffraction)	Fully included – inherent in the real sound field	Approximated with varying accuracy
Wall properties	Fully included – inherent in the real room	Absorption - scattering coefficients have to be measured or estimated, with limited accuracy
Air absorption (a function of temperature and humidity)	Fully included but may vary significantly in different measurements	Calculated, but very accurate
Source directivity	Not perfect: Lobes at high frequencies	Perfectly omni-directional
Dynamic range of source	Insufficient at very low and very high frequencies. Distortion at high levels	Unlimited dynamic range at all frequencies. No distortion
Calibration of source	Special procedure needed for the strength parameter, *G*	Perfect per definition
Background Noise	Limits the dynamic range, compensation necessary	Not present
Microphone directivity	Omnidirectional microphone. Some parameters require figure-of eight pattern or a dummy head	All directivities available
Results in octave-bands	Filtering is required, which alters the original signal	Results are derived directly in different bands - no alteration due to filtering
Onset time of impulse response	Critical, especially at low frequencies	Perfect per definition
Reproducibility	Not perfect: Depends heavily on the source	Can be perfect, depending on the algorithm
Influence of operator	Knowledge and experience important	Knowledge and experience very important

4. Simulating the room impulse response

Simulations in room acoustics are well known to provide fast and effortless estimation for the ISO 3382 parameters. They are mainly based on geometrical acoustic algorithms which simplify the wave phenomena to fundamental geometrical tasks. Phase information is generally excluded, so that the results can be considered valid for frequencies above Schroeder's limiting frequency [3]: $f_S = 2000\sqrt{(T/V)}$ Hz, where T is the reverberation time in seconds and V is the volume in m^3. Below this limit the modes in a room are very distinct and prominent, but cannot be accurately predicted, due to the lack of phase information. On the other hand, above f_S a high modal overlap is present, so that wave effects due to phase can be neglected without significant loss of information for the acoustic field. Despite their simplified approach, geometrical acoustic simulations are invaluable for predicting the ISO 3382 parameters in a wide variety of rooms, from offices and music studios to auditoriums and concert halls. Even though simulations offer a simplified approach of a real-world sound field they still have a number of advantages over measurements: The source is perfectly omni-directional, there are no problems with distortion, there is no background noise so the dynamic range is infinite at all frequencies, no filtering is required and the results are reproducible if the stochastic nature of the algorithm used is eliminated (deterministic ray tracing [4]).

4.1 Modelling the room and acoustic properties of materials

The basis for simulating the impulse response is the digital model of the room. This implies that the geometry of the room is simplified, sometimes to make a very rough room model only representing the main shape of the room, and in other cases being a rather close approximation, if created directly from the architect's 3D model. However, because of the wavelength of audible sound, the degree of geometrical detail in the room model is generally not the main source of uncertainty in the simulations. The acoustical data representing the materials, i.e. the *absorption coefficients* (α) and the *scattering coefficients* (s) are often more important for the uncertainty. The available data for a well-defined highly absorbing material, which has been tested in the laboratory, come with a significant uncertainty (see **Table 3** and **Table 4**).

Table 3. Uncertainty of measured absorption coefficients.

Frequency, Hz	125	250	500	1000	2000	4000
Type A mounting, α(mean)	0,26	0,85	1,11	1,07	1.02	1,03
Standard deviation	0,070	0,051	0,030	0,040	0,046	0,047
Type E-400 mounting, α (mean)	0,64	0,78	0,98	1,06	1,06	1,06
Standard deviation	0,107	0,053	0,038	0,032	0,035	0,047
Average std.dev.	0,088	0,052	0,034	0,036	0,040	0,047
95% confidence range	± 0,18	± 0,10	± 0,07	± 0,07	± 0,08	± 0,09

Table 4. Estimated uncertainty of measured scattering coefficients

Frequency, Hz	125	250	500	1000	2000	4000
Assumed α_s	0,25	0,25	0,25	0,25	0,25	0,25
Assumed α_{spec}	0,29	0,33	0,59	0,74	0,83	0,87
s (example)	0,05	0,10	0,45	0,65	0,77	0,83
Standard deviation, δ_s	0,04	0,04	0,04	0,04	0,04	0,07
95% confidence range	± 0,08	± 0,08	± 0,08	± 0,08	± 0,08	± 0,14

The standard deviation on absorption coefficients is the *Inter-laboratory reproducibility* from a Round Robin in 2002 organized by ASTM [5] with 16 participating laboratories. Two different test samples were applied, a 51 mm thick glass fibre panel, which was either laid directly on the floor (Type A mounting) or suspended 400 mm from a rigid surface (Type E-400 mounting). The mean value and the standard deviation between the 16 laboratory results are given in Table 3.

Looking at the 1 kHz octave band as an example, the absorption coefficient reported from a laboratory test has a 95% confidence range of ± 0.07, which means that with 95% probability the true value is within this range. In other words, there is a 5% risk that the true absorption coefficient deviates more than 0.07 from the measured value. At 125 Hz the 95% confidence range is even higher: ± 0.18. This clearly shows that the absorption data represents a significant source of uncertainty in any room acoustic calculation, including the traditional use of Sabine's equation.

The uncertainty on the scattering coefficient is also worth noting, although the influence on the uncertainty of the calculation results may be less dramatic as for the absorption. The standard deviation on scattering coefficients has been calculated here using equation (A5) found in ISO 17497-1 [6] and applying data on the *Intra-laboratory repeatability* on the measurement of absorption coefficients also reported in [5]. For the purpose of the calculations a typical set of scattering coefficients have been applied, having $s = 0.50$ at the mid-frequencies (between 500 and 1000 Hz).

Looking at the influence of scattering on the simulated room acoustic parameters, high scattering coefficients above 0.40 tend to give approximately the same results. However, low scattering coefficients in the range from 0.00 to 0.10 can have a very strong influence on the calculation results, and thus should always be regarded carefully. In fact, it is recommended to look at the scattering coefficients in a logarithmic scale; for example the following steps in scattering coefficient are approximately of equal importance: 0.40 – 0.20 – 0.10 – 0.05 – 0.025 – 0.0125. Finally, the quality of a simulation result is influenced by the knowledge and experience of the user. This is particularly important in relation to the input data for the materials.

4.2 Calculation of the impulse response

Although geometrical models for room acoustic simulation can be a fairly complicated matter, it is much easier to derive ISO 3382-1 parameters from such a simulation than it is from a real impulse response measurement: 1) the onset time of the impulse response is well defined from geometry, 2) there is no need for digital filtering which may blur octave band results in the time domain and 3) background noise is not a problem. Two types of parameters shall be described shortly. *Decay parameters* such as T_{30} and *time interval parameters* such as C_{80}.

ODEON makes use of hybrid calculation methods which is based on a combination of the *image source method* and a special *ray radiosity* method in order to predict arrival times of reflections at a receiver and the strength of reflections in octave bands [4]. The calculation methods are energy based, so adding the octave band energy to a time histogram forms directly the squared impulse response which is needed in order to derive parameters such as T_{30} and C_{80}, without the need for any digital filtering. The length of the impulse response predicted is usually limited by maximum path length for which the rays are traced. The early part of the response (early reflections) is determined by a list of image sources up to a certain transition order, typically 2nd order. For higher order reflections a Fibonacci-spiral shooting of rays is initiated, resulting in a large number of reflection points, distributed on the surfaces of the room. Each point is replaced by a *secondary source*, which radiates sound according to the relative strength and delay of the corresponding reflection. An algorithm called *reflection and vector based scattering* uses as input data the scattering coefficient of the surface, the distance between the present and the previous reflection points, as well as the angle of incidence, to produce a unique directivity pattern for the secondary source [4]. Once all image and secondary sources have been detected, the energy information they carry can be collected from all visible receivers in the room, effectively leading to a squared impulse response.

4.3 Deriving decay parameters

Decay parameters, such as T_{30}, can be derived from the *squared impulse response*. The ISO 3382-1 standard describes that T_{30} can be derived in the following way: The decay curve is the "graphical representation of the sound pressure level in a room as a function of time after the sound source has stopped" (interrupted noise assumed). The decay curve can also be derived from an impulse response measurement using Schroeder's backwards integration [7]. This backwards integrated decay curve, derived from an impulse response, corresponds to the decay curve obtained from the decay of interrupted noise – taking the average of curves from an infinite number of measurements:

$$E(t) = \int_t^\infty p^2(\tau)\,\mathrm{d}\tau = \int_\infty^t p^2(\tau)\,\mathrm{d}(-\tau) \tag{2}$$

where $p(\tau)$ is the sound pressure as a function of time in the impulse response.

One problem with the backwards integration is that some energy is not included in the real impulse response due to its finite length t_1. The problem can be corrected by estimating the energy that is lost due to the truncation. This amount of energy can be added as an optional constant C, so Eq. (2) changes to:

$$E(t) = \int_{t_1}^{t} p^2(\tau)\,\mathrm{d}(-\tau) + C \qquad \text{where} \quad t_1 > t \tag{3}$$

If the curve is not corrected for truncation, the estimated decay time may be too short.

In **Figure 1** is shown a simulated decay curve at 1000 Hz. The two blue curves are the squared impulse response in dB and the backwards integrated curve, respectively. The black curve is the backwards integrated curve which has been corrected for truncation. In order to derive a decay parameter, the appropriate range of the backwards integrated and corrected decay curve is evaluated and a least-squares fitted line is computed for the range. For T_{20} the range is from 5 dB to 25 dB below the steady state level and for T_{30} the range is from 5 dB to 35 dB below the steady state level. The slope of the fitted line gives the decay rate, d in dB per second, from which the reverberation time is calculated e.g. as $T_{30} = 60/d$.

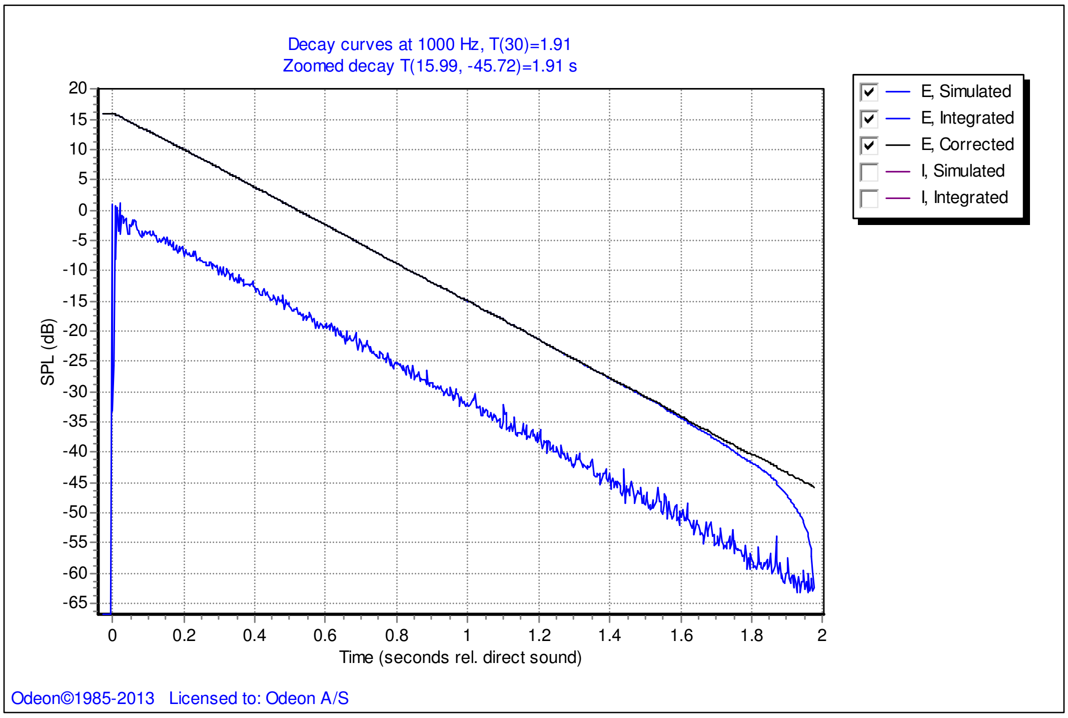

Figure 1. Example of simulated squared impulse response and integrated decay curve with and without correction for truncation.

4.4 Deriving time interval parameters

Parameters such as C_{80} make use of the energy arriving at the receiver in specific time intervals, relative to the direct sound. In the case of C_{80} the time intervals are from 0 to 80 ms and from 80 ms to infinity after the arrival of direct sound (see Eq. (1). In order to make a decent prediction of C_{80} it is important that the onset time is well defined. When the source is visible from the receiver this is not a problem as the onset time can be derived from source and receiver position and even in slightly coupled spaces this may be precise enough.

Measured time interval parameters may not be precisely derived if calculated directly from the filtered response, because filters create delay and smear the response in time. This can be particularly significant for the lower frequency octave bands where the filters are “long”. In order to bypass this filter problem, ISO 3382-1 suggests the "Window-before-filtering" approach which is the method implemented in ODEON. First the onset time is estimated from the broad band impulse response. In order to estimate the energy arriving for example during the first 80 ms, the response is gated from the onset time up to 80 ms and octave band filtered afterwards. This creates a filtered response which is longer than the original broad band response in order to include the filter tail. Then the energy of the gated filtered response is calculated including the tail of the filter, taking into account most of the smeared energy. Note that the C_{80} parameter may not make sense in a space where receiver and source positions are strongly decoupled as the build-up of the impulse response may take considerably longer than the 80 milliseconds.

4.5 Accuracy of a simulation due to number of rays and transition order

In order to derive a measure for the accuracy of the simulations, the global average deviation from measured results is considered. As the room acoustic parameters have different units (e.g. sec., dB, %) the deviation between measured and simulated results is expressed in terms of JND. Then the global average of deviations from measurements is calculated like this:

$$Error = \frac{\sum_{n=1}^{N_{AP}} \sum_{i=1}^{N_{Freq}} \sum_{j=1}^{N_{Pos}} \frac{\left| AP_{measured}(n,i,j) - AP_{simulated}(n,i,j) \right|}{JND(n)}}{N_{AP} \cdot N_{Freq} \cdot N_{Pos}} \tag{4}$$

where

$AP_{measured}$ is the measured value of acoustic parameter n at frequency i and position j,
$AP_{simulated}$ is the simulated value of acoustic parameter n at frequency i and position j,
$JND(n)$ is the subjective limen (just noticeable difference) of acoustic parameter n,
N_{AP} is the number of acoustic parameters (5),
N_{Freq} is the number of frequency bands (3),
N_{Pos} is the number of source-receiver positions (10).

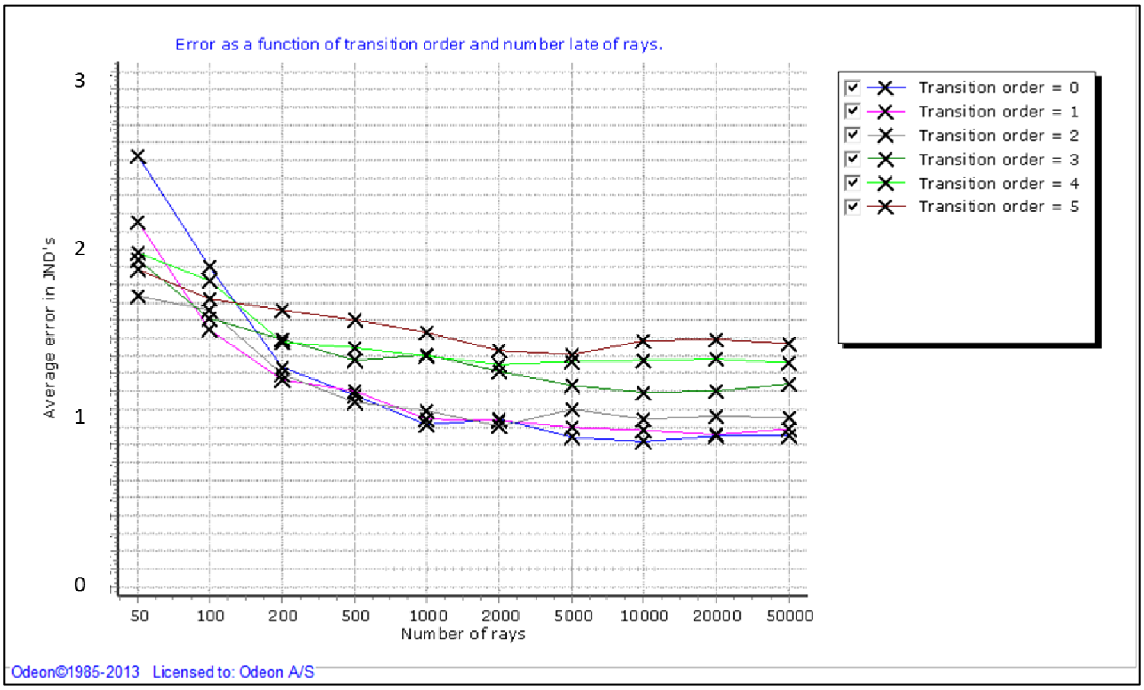

Figure 2. Accuracy of calculations in the auditorium as a function of transition order TO and number of rays. The displayed parameter is the global average of deviations from measurements in units of JND for 5 parameters, 3 octave bands and 10 source-receiver positions.

Figure 2 displays the results of the global average error for the auditorium, which is described in more detail later. Five acoustic parameters were considered: EDT, T_{30}, D_{50}, C_{80}, and T_S. The corresponding JND values are listed in **Table 1**. Three octave bands were used; 500, 1000 and 2000 Hz. Two source positions were combined with five microphone positions, i.e. in total 10, see **Figure 7.** The results show, that very good agreement is obtained with 5000 rays and transition orders 0, 1 or 2. The global average error is then around 1 JND. However, it is obvious that higher transition orders should be avoided. It is also seen that the results will not improve if more rays are used.

4.6 Auralisation, how to explain room acoustics with sound

The room impulse response obtained from a simulation contains information about direction of incidence of each sound reflection in the 3D space. This means that the impulse response can be transferred to a so-called binaural room impulse response (BRIR) by means of a head related transfer function (HRTF). An example is shown in **Figure 3**. By convolving the BRIR with a sound recording (preferably from an anechoic environment), the resulting two-channel sound file, when presented through headphones, can give the impression of listening in the chosen receiver position the simulated (virtual) room. This is *auralisation*.

Auralisation can be used as a tool during the design process, and this can be particularly useful in order to avoid acoustical defects in the room design [8]. The technique has been further developed to the advanced *multi-channel*, *multi-source* auralisation, which may produce a highly realistic simulation of an orchestra in a concert hall [9].

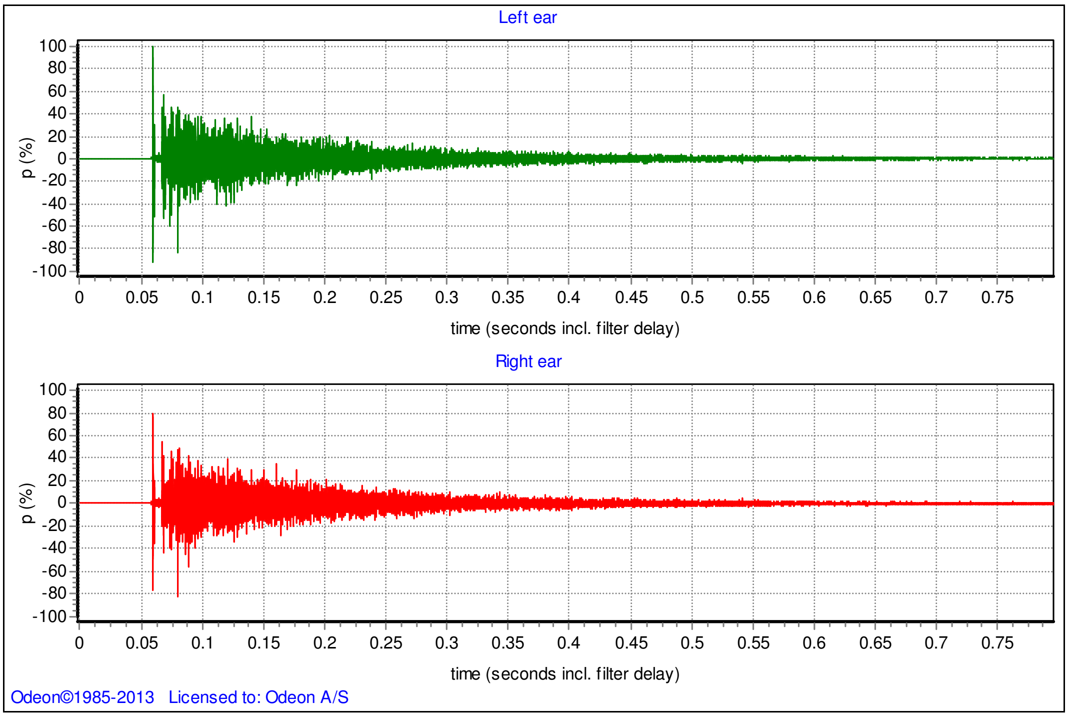

Figure 3. Example of calculated binaural room impulse response (BRIR) that can be used for auralisation. Upper part is for the left ear and lower part for the right ear. In this example the BRIR is 1500 ms long, but zoomed to show the first 80 ms.

5. Measuring the room impulse response

An impulse response can be obtained directly by recording the response to hand-clapping, popping of a balloon/paper-bag, a gunshot or even a hard footstep. As a modern alternative an impulse response can be obtained indirectly by producing a Maximum Length Sequence (MLS) or a sweep signal using an electro acoustic source. The latter methods stretches the impulse (Dirac function) in time and the measured response is deconvolved in order to form the impulse response. Using time stretched excitation, a substantial amount of energy is emitted from an electro acoustic source with limited maximum acoustic output, allowing superior signal to noise ratio. Reproducibility is also easier to control with electro-acoustic stimuli, due to uniform radiation.

Among the many available measurement methods the preferred one today is the swept sine method using a rather long exponential sweep from very low to very high frequencies [10, 11]. This method can produce impulse responses with very good dynamic range and the harmonic distortion by the loudspeaker is separated from the true impulse response, since it will appear at negative arrival times, i.e. before the onset of the impulse response [11]. Still there can be some influence from non-harmonic distortion [12, 13], so a high quality loudspeaker and power amplifier is important.

5.1 Capturing the impulse response

The sound source is a critical part of the measuring chain. For the measurement of the ISO 3382 room acoustic parameters the source must be as omni-directional as possible. The most common choice is a *dodecahedron* source, i.e. a source with 12 loudspeaker units pointing in different directions. The directivity pattern for such a source is reasonably omni-directional at low and mid frequencies, but at 2000 and 4000 Hz the directivity is not perfect (typical variations between max. and min. are 5 – 7 dB).

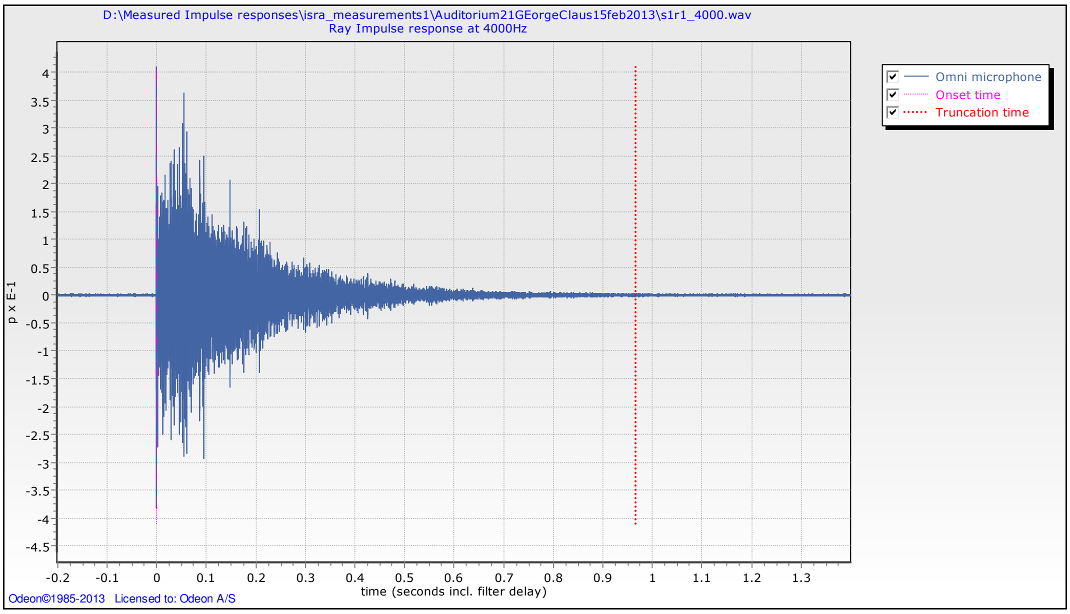

Figure 4. Impulse response obtained with the sweep method in ODEON. Combination S1-R1 at 4000 Hz .in the auditorium described later.

5.2 Filtering the impulse response

The octave-band filters typically used in the processing of room impulse responses are 2nd order Butterworth filters in accordance with the IEC 61260 [14]. These analogue filters can be implemented using digital infinite impulse response (IIR) filters. ODEON uses such type of filters and defines a finite *effective length*, allowing 99.9% of the energy in the tail of the filtered impulse response to be included. The filtering process introduces unwanted transient effects in the beginning of the response, which cease after about one effective length of the filter.

A *reverse* filtering algorithm is applied for decay analysis so that all the transients are repositioned at the tail of the impulse response. ODEON automatically excludes this transient tail when processing the impulse response. The *reverse* method has also the advantage of eliminating the *stretching* of the filtered signal, which occurs due to the delay of the filter itself. This *stretching* effectively leads to energy smearing, altering the slope of the decay curve. After processing the signal with *reverse* filtering, an extra *forward* filtering is applied, allowing for suppression of phase distortion. This combination of *reverse-forward* filtering in the decay analysis is used for the calculation of *decay parameters*, such as T_{30}. For the *time interval parameters* only *forward* filtering is applied for each gated window. The smearing of the energy is precluded by taking into account the effective length of the filter at the end of each window, as extra impulse response time.

5.3 Noise floor and truncation of the impulse response

When measuring an impulse response the dynamic range is limited by background noise which may influence all parameters that can be derived from the impulse response significantly if its level isn't very low or compensated for. At some time after the onset time the impulse response will decay to the level of the noise floor and the rest of the recorded response is not valid – this time we denote *truncation time*. The Truncation time is unique to each band of interest. The energy of noise arriving after the Truncation time should be excluded from analysis; however energy before it is also influenced by noise. Most of the impulse response recordings, whether recorded directly or obtained using the sweep method, come with a noticeable noise tail, due to the ambient background noise and noise of the transmission line involved (PC sound card, cables and microphone). This noise tail should be removed before deriving the decay curve and the ISO 3382 room acoustic parameters. Lundeby *et al.* [15] have proposed an algorithm for detecting the noise floor and truncating the recording at the cross-point between the pure impulse response and the noise floor. The cross-point is estimated by an iteration process of impulse response smoothing and regression line fitting. The ODEON measurement system utilizes a modification of this method in order to estimate the appropriate truncation time for each octave-band.

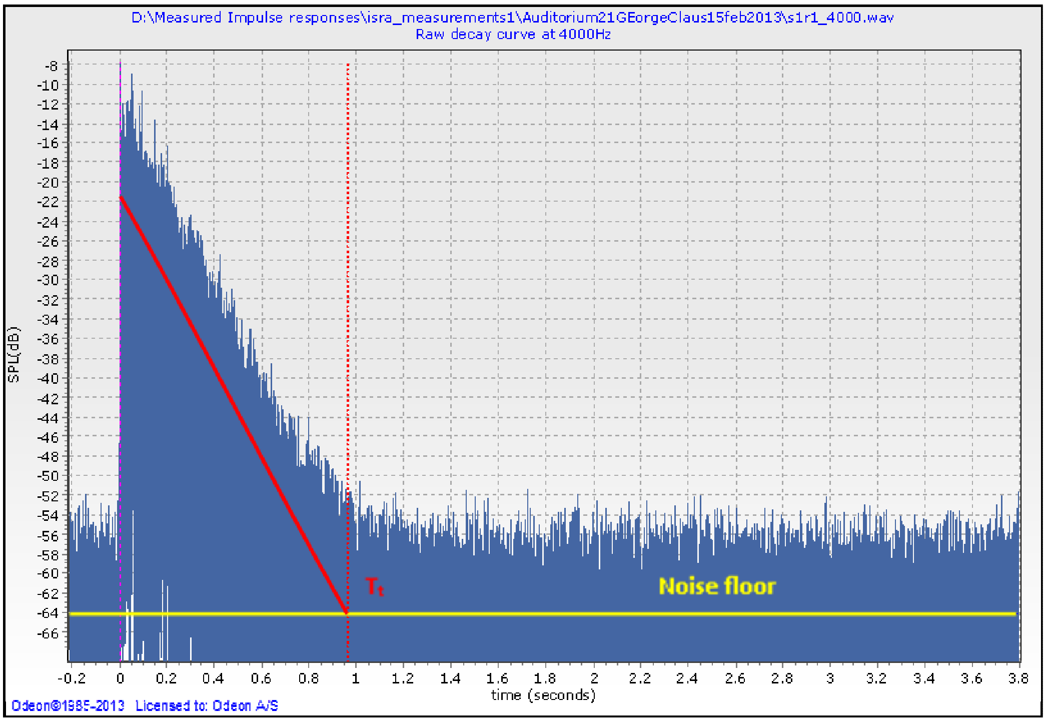

Figure 5: Example of squared impulse response with indication of noise floor and truncation time T_t.

Still the background noise is present in the backwards integrated decay curve in the range between the onset time and the truncation time and this will result in an over-estimation of the decay time when the energy contained in the noise floor is not negligible. However this may also be compensated for if the level of the noise floor is well estimated. In addition to the tail correction it is suggested that the background noise floor excluding the truncated tail can be subtracted from the valid part of the squared impulse response.

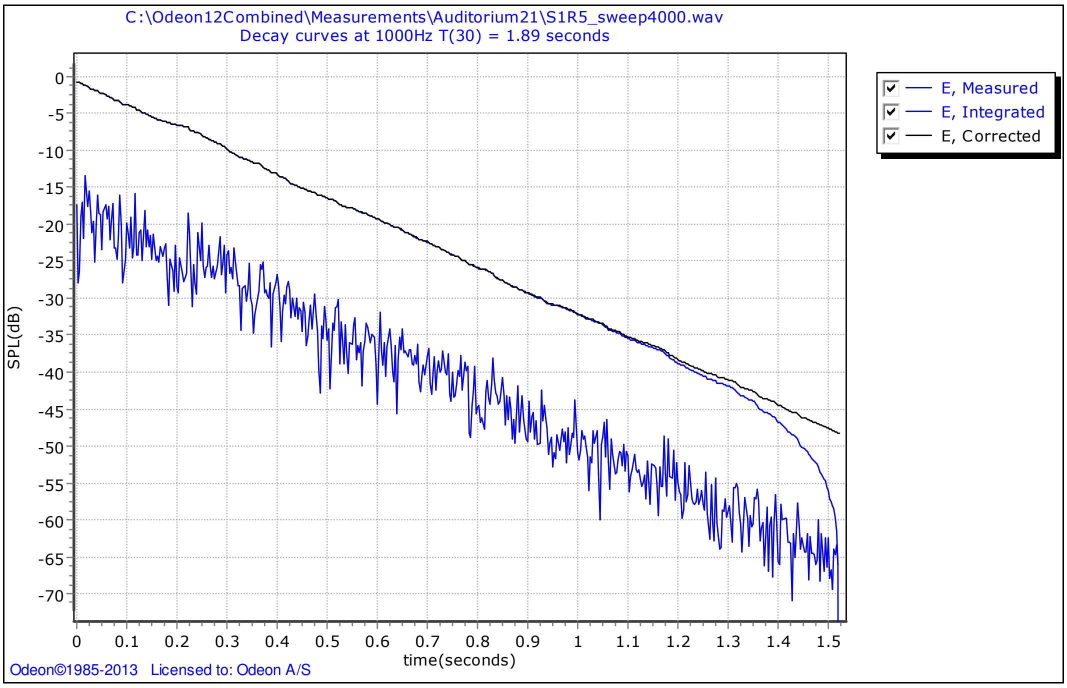

Figure 6. Example of measured squared impulse response curves and integrated decay curves.

6. Example, measuring and simulating an auditorium

6.1 Description of the room

The room which is used as an example for both simulations and measurements is Auditorium 21 located at the Technical University of Denmark. The volume is approximately 1160 m^3 and it has a capacity around 200 people.

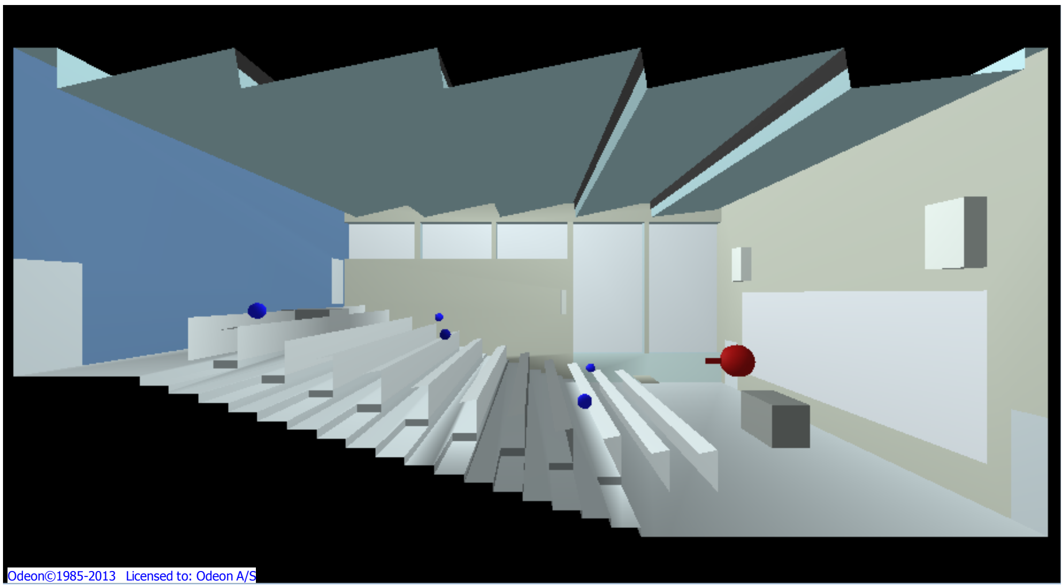

Figure 7. View into the 3D model of the auditorium. Blue spots mark the receiver positions and the red spot is one of the source positions.

The materials are mainly wood panels, glass, gypsum board and hard rows of chairs. The model shown in Figure 7 shows the absorption characteristics of the surfaces by the use of Acoustic Colours, a method introduced in ODEON in 2001 [16].

6.2 Measurements with varying the sweep length

For the combination P1-R5 the different impulse responses were obtained with sweep lengths of 0.5, 1, 2, 4, 8, 16 and 32 sec in order to evaluate whether the signal to noise ratio (S/N) increases by 3 dB per doubling of sweep length, as expected, and to evaluate the impact on derived values of T_{30}. These measurements were performed at very low level in order to obtain a wide span of S/N levels in the recorded impulse responses. The values of D_{50} and C_{80} only showed small differences with increasing sweep lengths. T_{30} did show some changes with increasing sweep lengths. Three approaches for deriving T_{30} were tested: 1) T_{30} derived directly from the backwards integrated curve with no corrections, 2) T_{30} derived from the curve with correction for truncation according to Eq. (3) and finally 3) T_{30} derived from the curve with correction for truncation, as well as correction for noise floor in the valid part of the impulse response.

In **Figure 8** it can be seen that for long sweep lengths/high dynamic range all three methods agree that T_{30} is 1.89 *s*. When T_{30} is derived without compensation for truncation of the impulse response, the values derived are too high. If compensating for the truncation of the impulse response only, T_{30} tends to be too long. However, when the backwards integrated curve is compensated for background noise, the result is more stable even for rather short sweep lengths and closer to the “correct” value. This is the method implemented in ODEON 12.1.

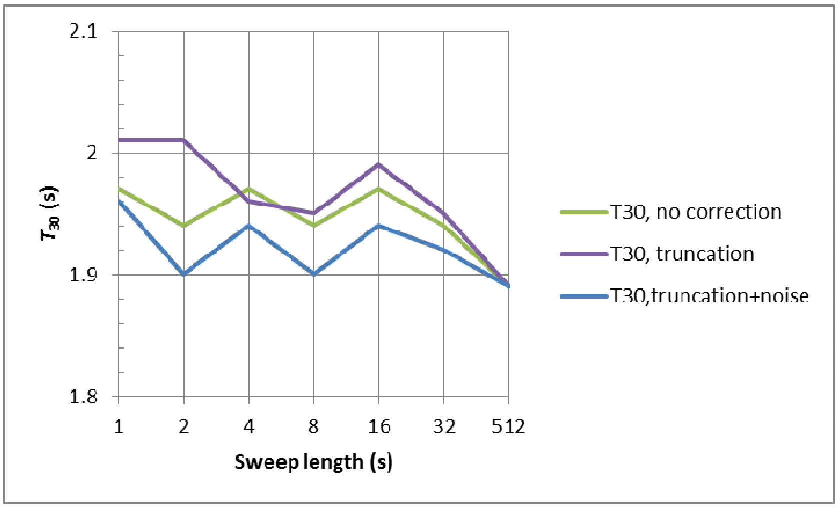

Figure 8. $T_{30,\ 1000\ Hz}$ derived from measured impulse responses with increasing sweep lengths and with and without correction for impulse response truncation and noise floor.

6.3 Comparison of measured and simulated results

In **Figure 9** measured and simulated values of EDT, T_{30}, SPL (the G value with a source power level of 31 dB), T_S, C_{80} and D_{50} are displayed for source position P1 and five receiver positions at 1000 Hz. It should be noted that the absorption data in the model were adjusted in order to get close agreement in T_{30}, results. But it is interesting to look at the other room acoustic parameters.

The agreement between measured and simulated parameters at 1000 Hz is within 0.5 JND for most parameters, which is very satisfactory. The difference between measured and simulated EDT varies from 0.01 to 0.07 seconds with an average deviation of 0.52 JND. It is interesting to see that both measured and simulated values of EDT (1.98 and 1.96 seconds) are marginally higher that T_{30} (1.89 and 1.91 seconds) so this undesired feature of the room is detected in simulations as well as in the real room. Values of measured and simulated SPL, Ts, C_{80} and D_{50} are all in good agreement - and measured and simulated values agree on the variation with position.

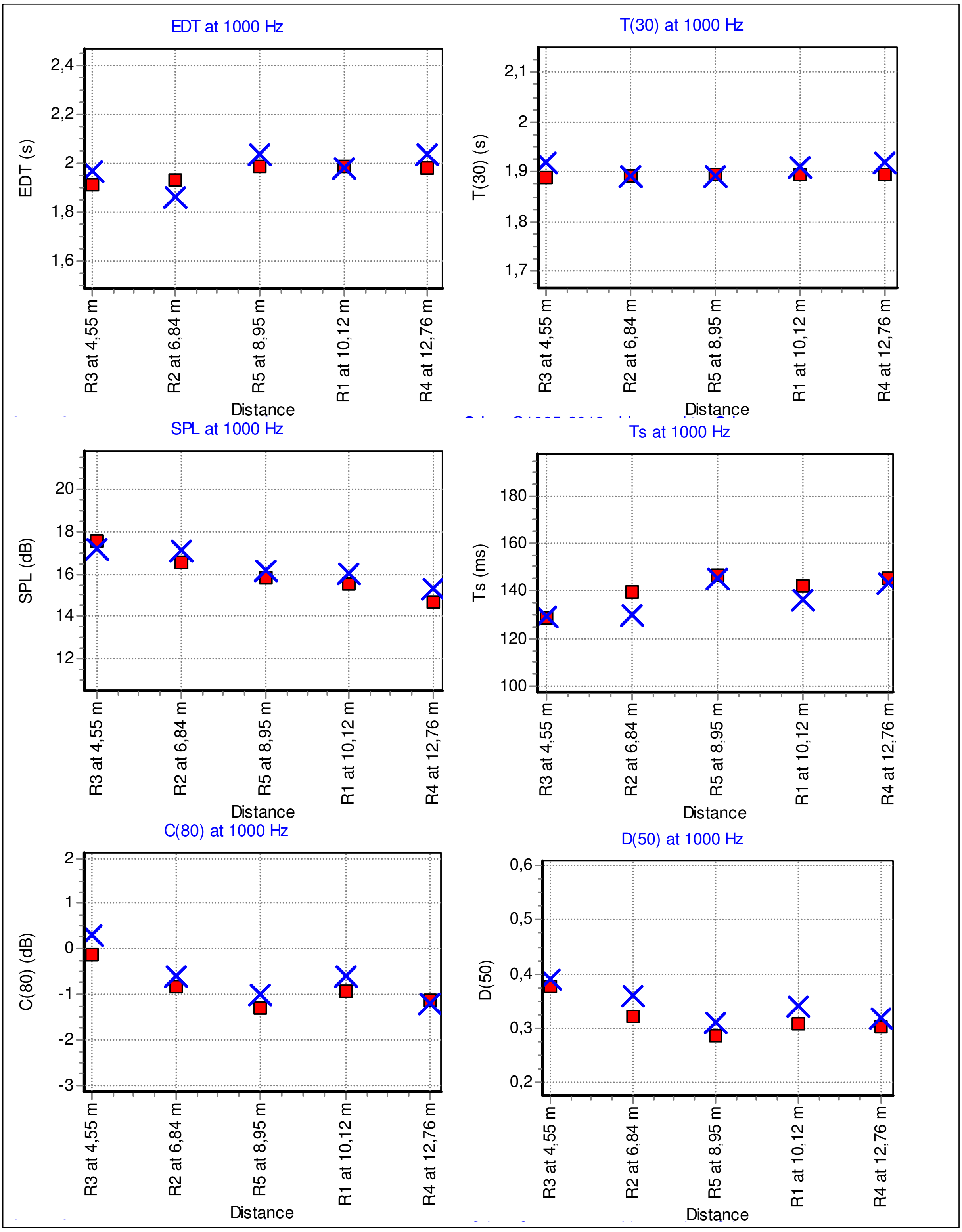

Figure 9: Simulated and measured room acoustic parameters for the five receivers in auditorium 21. Simulated parameters displayed with *red squares* and measurements with *blue crosses*.

6.4 Uncertainty of receiver position

In practice it is not possible to position the microphone (nor the source) at an exact position when conducting room acoustic measurements. So, if reproducing the measurement at a later time slightly different results in terms of ISO3382-1 parameters should be expected. When a person is sitting in the auditorium the position will not be exact either. In order to give an idea of the uncertainty of measured parameters if the receiver position is not exact, measurements in a region close to receiver position 5 in the middle of the audience area were repeated with position offsets 30 cm right, 30 cm left, 15 cm front, 15 cm back, 10 cm up, and 10 cm down – a total of 7 positions including the original position. The graph below (**Figure 10**) shows statistics for the 7 positions for the parameter C_{80}, which is chosen as an example. Measured as well as simulated results are included for comparison. As can be seen the simulated values in receiver positions that are close to each other only show minor deviations, much less than deviations between the measured results, probably because phase is not included in the simulation model.

At 1000 Hz *none* of parameters have a standard deviation larger than 0.7 when normalized to Just Noticeable Differences (JND) and even at 125 Hz all parameters except SPL have a standard deviation less than 1.2 JND's.

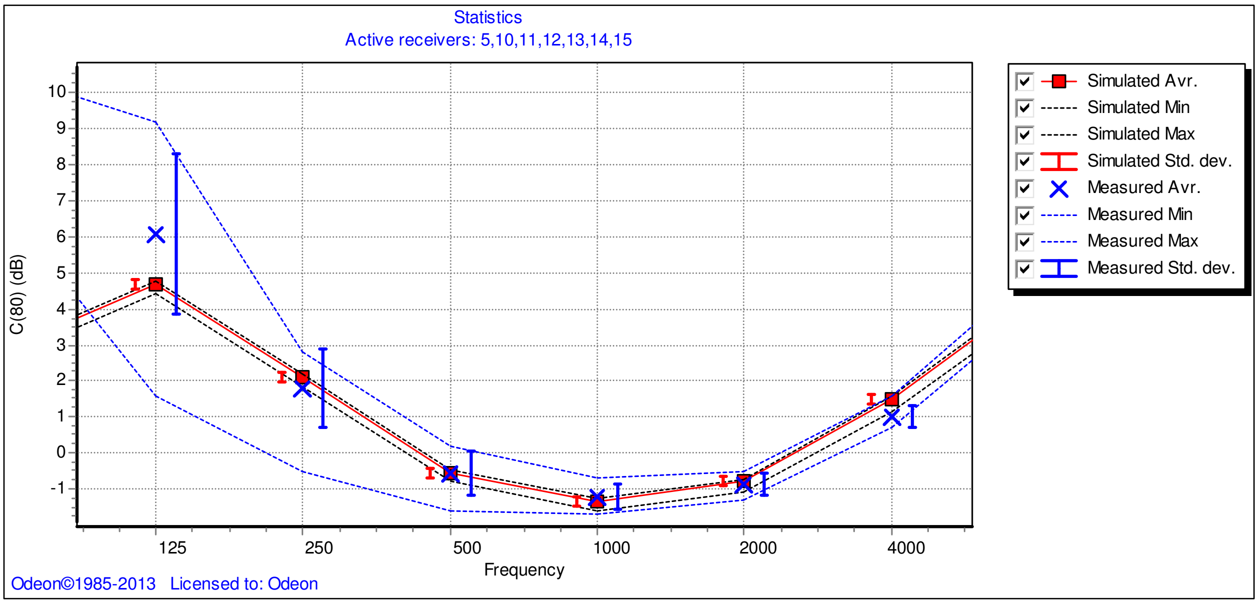

Figure 10: Uncertainty due to receiver position, simulated and measured variation of the Clarity C80 in seven positions close to receiver 5 in the auditorium.

7. Conclusion

Both simulations and measurements have strengths and weaknesses. The most important reasons for uncertainty in simulations are the input data for absorption and scattering of the surfaces and the rough approximations of wave phenomena like diffraction and scattering.

The main reasons for unreliable measurements are due to the sound source; the dynamic range is limited and the loudspeaker may create distortion that have an unwanted influence on the measurements. At high frequencies (2 kHz and above) the commonly used dodecahedron source has a directivity that is far from omnidirectional. For some parameters like C_{80} the correct setting of the onset time in the impulse response is critical. Truncation of impulse responses and background noise in impulse responses may lead to systematic errors on T_{30} if not compensated for. However, if compensating for both errors, correct results may be achieved even with moderate signal to noise ratios.

Deviation was not larger than 0.7 JND for measurements of any of the ISO 3382-1 parameters tested at 1000 Hz when using 7 different positions within a volume of (w, l, h) = (0.6, 0.3, 0.2) metres around a central position in an auditorium – this indicates that the results can be reproduced even if the receiver position is not exact. For simulation results in ODEON it seems that small deviations at the position are negligible.

It is possible to simulate and measure accurately the room acoustic parameters according to ISO 3382-1 if care is taken in the implementation and use of simulation and measurement algorithms. In the auditorium example used here there is close agreement between measured and simulated values.

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