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The ISO 3382 parameters: Can we simulate them? Can we measure them?

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ABSTRACT

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 is valid. In addition, the sound source used for measurements often has strong lobes at high frequencies and cannot produce an ideal Dirac function. For this reason and due to distortion products by the octave-band filtering process detection of the arrival time of the direct sound from a measured impulse response is of questionable accuracy. 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.

1 INTRODUCTION

The room acoustic parameters described in the ISO 3382 standard are the reference for objective evaluation of acoustics in rooms from impulse responses. Evaluation of some of the ISO3382-1 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[1]. 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 acoustics 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 measurements one and only one omni-directional source should be used. The ISO 3382 standard give the framework for measurement of room acoustic parameters, but lack detail on the requirements needed for derivation of certain room acoustic parameters[2]. One of the major problems is the correct 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 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 at 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} [3]:

$$C_{80} = 10 \times log_{10} \frac{\int_{0}^{80\ ms} p^2(t)dt}{\int_{80\ ms}^{\infty} p^2(t)dt} \tag{1}$$

where $p(t)$ is the sound pressure as a function of time t (pressure 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 standard[3].

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

2 IMPULSE RESPONSE SIMULATIONS

Simulations in room acoustics are well known to provide fast and effortless estimation for the ISO 3382 parameters. They are mainly based on geometrical acoustics 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[4]: $f_{Sch} = 2000\sqrt{T_{60}/V}$ [Hz], where T_{60} is the reverberation time in the room in seconds and V is its 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_{Sch} 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[5]).

2.1 Modelling the Room – Uncertainty of Input Data

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 2** and **Table 3**).

Table 2: 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 3: 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[6] 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 2**.

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[7] and applying data on the *Intra-laboratory repeatability* on the measurement of absorption coefficients also reported in[6]. 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.

2.2 Calculation of the impulse response

Although geometrical models for room acoustics simulation can be a fairly complicated matter, it is much easier to derive ISO 3382 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[5]. 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 transition order, typically up to 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[5]. 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 an energy (squared) impulse response.

2.3 Deriving decay parameters

Decay parameters, such as T_{30}, can be derived from the *squared impulse response*. The ISO 3382 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[8]. This backwards integrated decay curve, derived from an impulse response, corresponds to the decay curve obtained from the decay of interrupted noise – if taking the average of curves from an infinite number of measurements[3]:

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

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 in Eq.(2):

$$E(t) = \int_{t_1}^{t} p^2(\tau)d(-\tau) + C \quad where\ t < t_1 \tag{3}$$

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

In **Figure 1** a simulated decay curve is shown at 1000 Hz. The blue curves are the squared impulse response in dB and the backwards integrated curve. 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 $T_{30} = 60/d$.

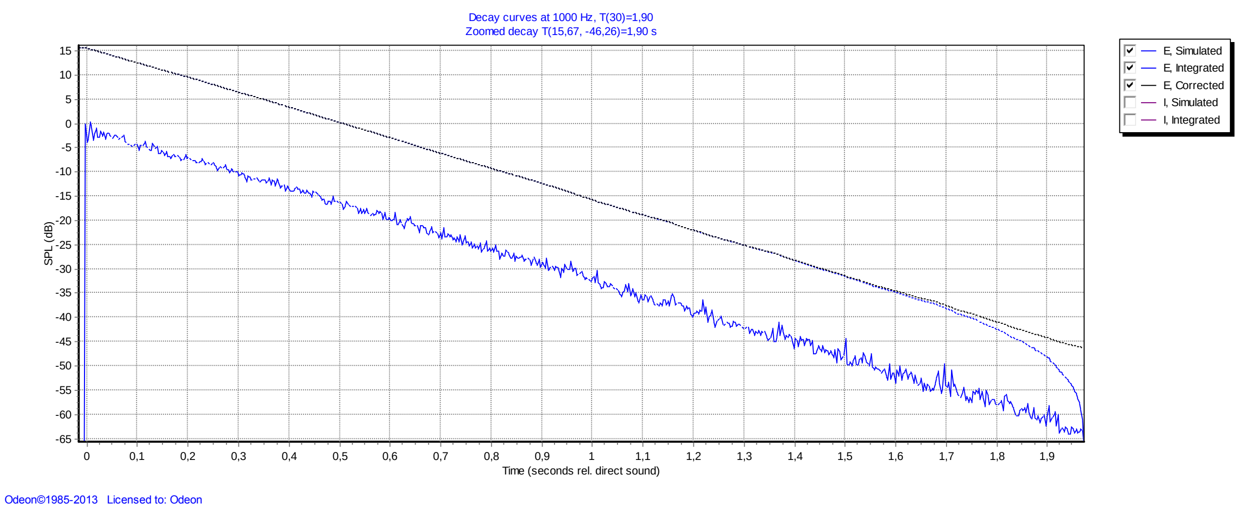

Figure 1: Example of simulated squared impulse response curves and integrated decay curves.

2.4 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} (Eq.(1)) the time intervals are from 0 to 80 ms and from 80 ms to infinity, after the arrival of direct sound. 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 suggest 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 first 80 ms of the broad band response arriving after the onset time is gated 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 counted 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.

3 IMPULSE RESPONSE MEASUREMENTS

In contrast to impulse response simulations (Sec.2), measurements may be considered accurate in a broader frequency area, 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 filtering required impose errors in the final results. **Table 4** 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 material data and the approximation of the wave phenomena.

Table 4: 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 measured or assumed, with limited accuracy

Air absorption (a function of temperature and humidity)	Fully included but may vary significantly in different measurements	Calculated from theoretical formulas, 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 G parameter	Perfect per definition
Background Noise	Present and limits the dynamic range	Not present
Microphone directivity	Omnidirectional microphone Some parameters require figure-of eight pattern or a dummy head	All directivities available
Results in different octave-bands	Filtering is required, which alters the original signal	Results are derived directly in different bands - no alteration due to filtering.
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

3.1 Measuring an 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[9]. This method can produce impulse responses with very good dynamic range and minimized harmonic distortion by the loudspeaker. Still there can be some influence from non-harmonic distortion[10,11]. **Figure 2** shows an example of squared impulse response with indication of estimated noise floor and truncation time.

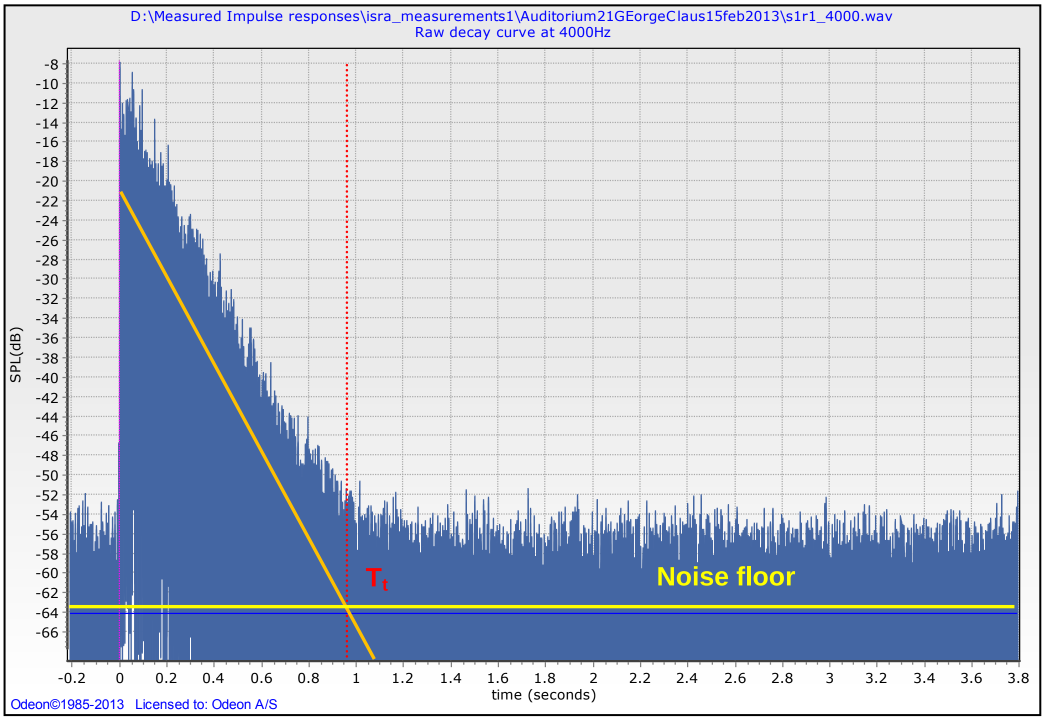

Figure 2: An impulse response with background noise and correct truncation (cross) point T_t.

3.2 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[12]. 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 (Sec.2.3) so that all the transients are re-positioned 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* (Sec.2.4) 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.

3.3 Detecting the Onset Time

An ideal impulse response, according to geometrical acoustics[1], would consist of an ensemble of Dirac functions with appropriate delays and strengths. The ideal direct sound from a source should be a perfect Dirac function too, arriving at a time equivalent to the distance between the source and the receiver. However in reality the direct sound and all the other reflections are not perfect Dirac functions. In fact, each reflection consists of an onset (before the peak value), a peak and some decay. As the frequency gets lower, the decay of a reflection may overlap with the onset of a subsequent reflection. In the derivation of many of the ISO 3382 parameters it is vital to capture the amount of early energy correctly. ODEON uses advanced algorithms for successfully detecting the energy from the direct sound and discriminating it from the following reflections. For every impulse response the detected onset time is indicated in the display. According to the ISO 3382-1 standard[3, pp. 18-19] the onset of the direct sound and of the whole impulse response should be *at least 20 dB* below the peak of the direct sound. In ODEON this value is called *Trigger level* and by default is set to 40 dB. This number is automatically moderated if the trigger point is getting too close to the noise floor. The onset time is detected from the broad band impulse response as suggested in the above reference.

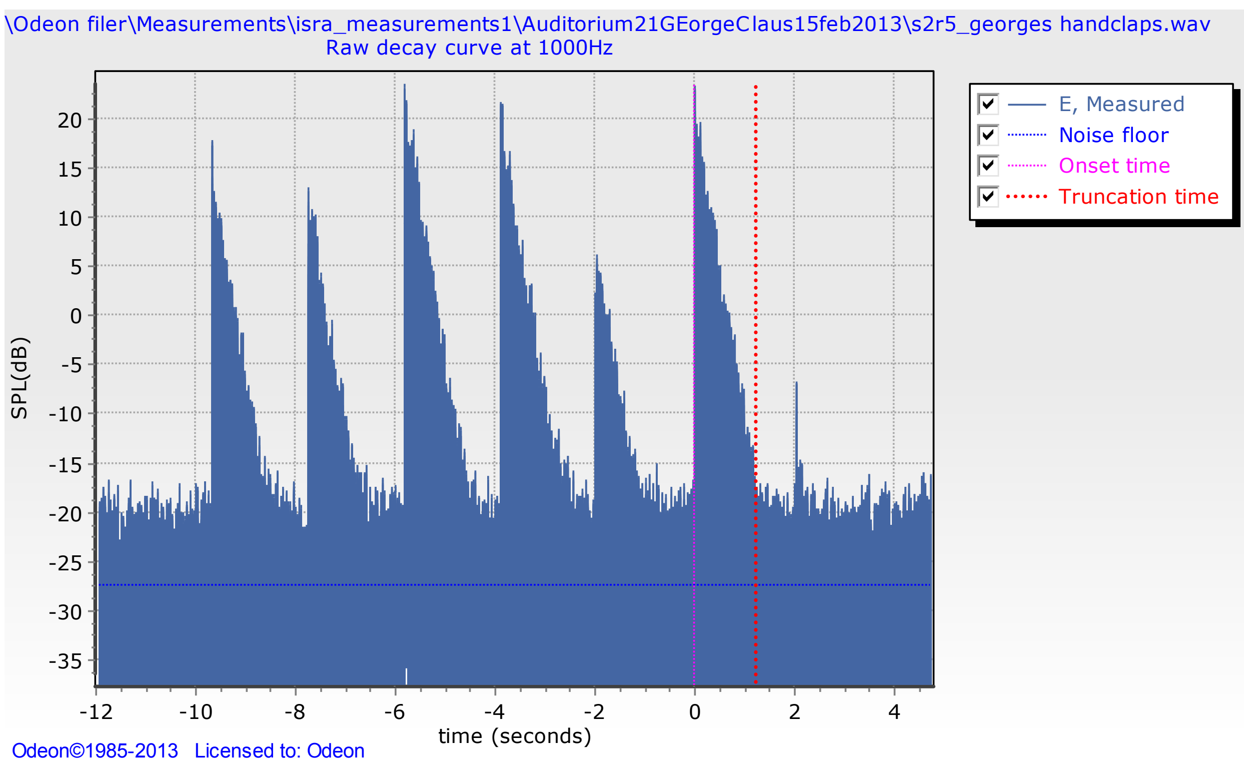

Figure 3: Display of onset time and truncation time at 1000 Hz for a series of handclaps. The recording contains multiple impulse responses – ODEON chooses the one with the maximum broad band peak level.

In the ODEON software both the trigger level and a number of other parameters can be set manually in order to provide the flexibility to the user to control the onset time. However the default value of -40 dB for the trigger level seems to work well. A short time interval, the Noise floor window length, can be specified for the detection of noise floor before the onset time. This is used for separating multiple impulse response from each other or indeed to exclude small amounts of impulsive noise - the default value of 10 ms for the Noise floor window length seems to work well too.

3.4 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.* 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[13]. 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.

3.5 Decay Curve and Noise Correction

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. Apart from the tail correction we suggest that the estimated background noise floor excluding the truncated tail (Sec.3.4) can be subtracted from the valid part of the backwards integrated decay curve.

4 INVESTIGATIONS ON MEASUREMENTS

Two different rooms are studied in this investigation an auditorium and *Hagia Sophia*, a large mosque/cathedral in Istanbul. The first room falls within the category of rooms that are typically investigated using the ISO 3382-1 standard, where the latter is a large space with many coupled volumes where it is a bigger challenge to make simulations and measurements of the ISO 3382-1 parameters agree.

The auditorium was *Auditorium 21* at the Technical University of Denmark which is used for lectures. This auditorium was chosen because it was accessible for measurements and not for its excellent acoustics conditions. A model of the room consisting of 198 surfaces was modelled in Trimble Sketchup[a]. We normally model the audience area with a limited level of detail e.g. as a plane with skirts as this area is normally highly absorbing and scattering. In this room however in the unoccupied state the audience area consists of hard wooden chairs and desks. It was found that a model with a more detail audience area gave better agreements between simulations and measurements so the more detailed version of the audience area was included in this study. Initial materials were selected according to a visual inspection of the room and absorption coefficients were fine tuned in order to match average measured and simulated T_{30} values.

In this auditorium we examine:

- The influence of different signal to noise ratios on the obtained measurement results by using different sweep lengths.
- Whether meaningful measurement results can be obtained using different excitation signals (sweep, hand claps, popping balloons).
- The sensitivity to uncertainty of exact receiver position on measured and simulated parameters.
- Measured and simulated parameters for a number of receiver positions.

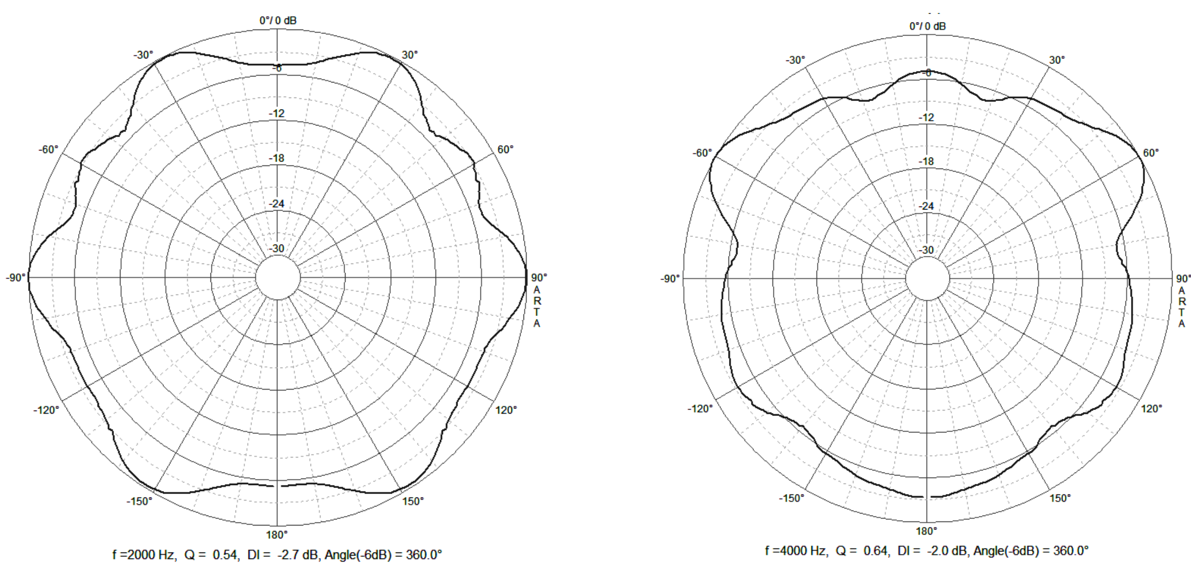

Figure 4: Horizontal directivity pattern of the *Dodecahedron Loudspeaker D12*, by LANGE Loudspeakers, at 2000 and 4000 Hz.

For the impulse response series obtained with ODEON a *dodecahedron* omni-directional source was employed. **Figure 4** shows the directivity pattern for such a source by LANGE Loudspeakers[b]. It can be seen that at 2000 and 4000 Hz the source is not perfectly omni-directional.

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[a] http://www.sketchup.com

[b] http://www.langeloudspeakers.com

Impulse response recordings were made and processed with ODEON 12.1 for 2 sources (P) and 5 receiver (R) positions, giving a total of 10 combinations in accordance with the ISO 3382-1 standard[3]. The following study is focused only on the results from source one (P1). A perspective view of the auditorium, together with a ground plan and the source-receiver positions is displayed in **Figure 5**. The impulse responses were recorded with an exponential sweep signal. **Figure 6** shows an example of the impulse response for the combination S1-P1 at 4000 Hz. Room acoustic parameters described in ISO 3382-1[3] where derived.

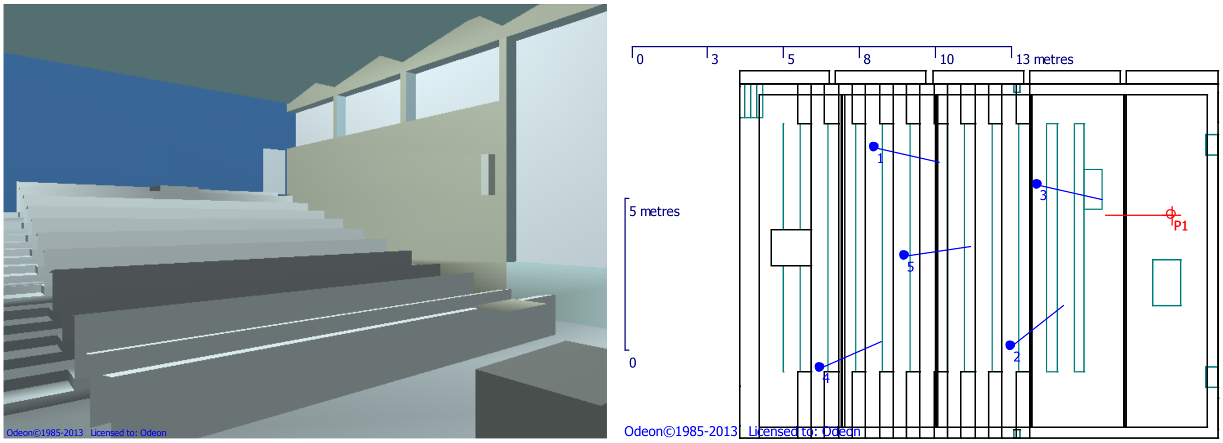


Figure 5: Model of the *auditorium 21* at the Technical University of Denmark, as it appears inside ODEON. *Left*: Three-dimensional rendering of Auditorium 21 with colors corresponding to the reflectance of the surfaces[14]. *Right*: Source and receiver positions inside *auditorium 21*.

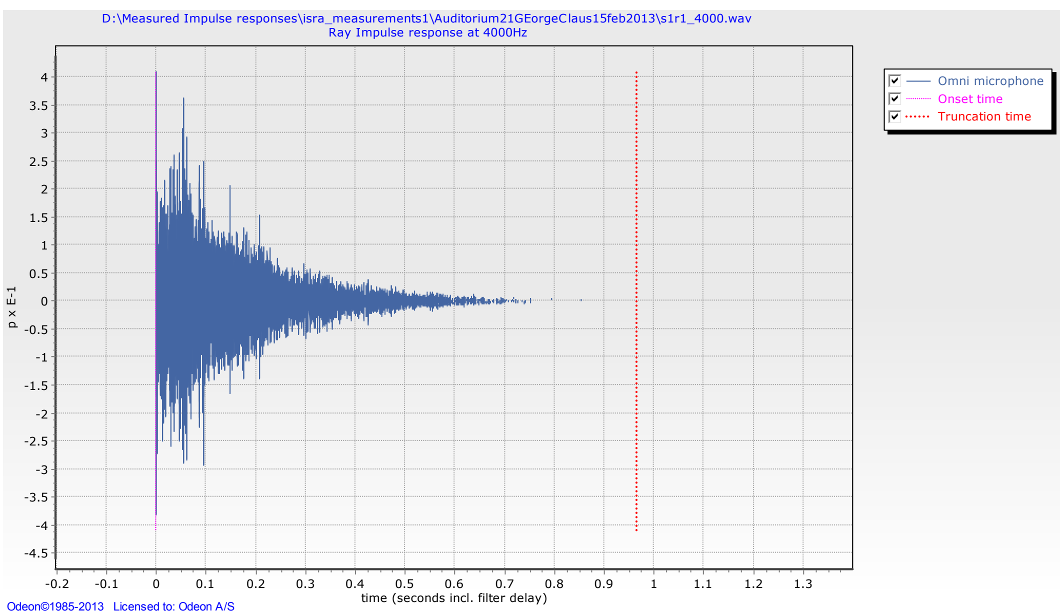


Figure 6: Typical impulse response obtained with the sweep method in ODEON. Combination S1-R1 at 4000 Hz.

4.1 Varying 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. Thus, are not displayed. 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, as described in 3.5.

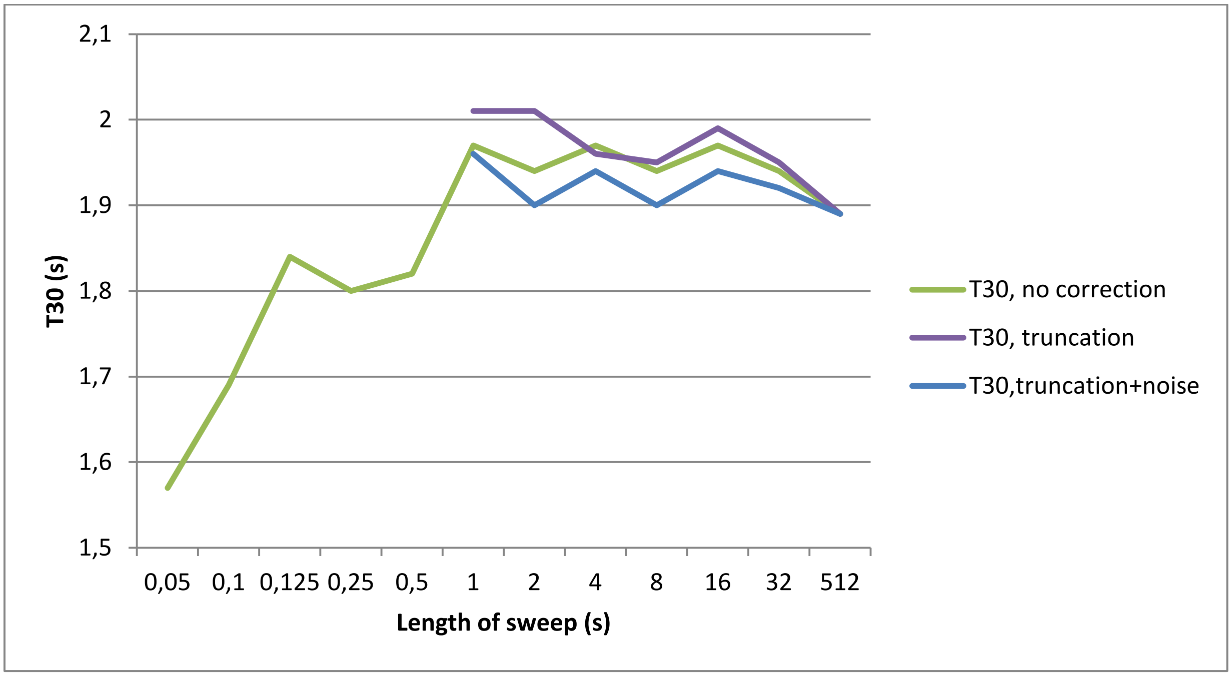


Figure 7: $T_{30,\ 1000\ Hz}$ derived from measured impulse responses with increasing sweep lengths and with and without correction for impulse response truncation and noise floor.

In **Figure 7** 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 without a doubt too low for short sweep lengths. For the inexperienced this can be problematic as the uncorrected curve does provide a (wrong) result for T_{30} where as corrected curves will not allow deriving T_{30}. If compensating for the truncation of the impulse response only then T_{30} tends to become too long. When the backwards integrated curve is compensated for background noise, the result stabilizes much faster at the “correct” value. This is the method implemented in ODEON 12.1.

In **Figure 8** the dynamic range - SPL/Noise, corresponding to the sweep length used in **Figure 7** is given. The SPL/Noise is displayed for any impulse response analyzed in ODEON and it is an indicator whether sufficient S/N was present during the impulse response measurement. The S/N can be increased either by using a higher output on the loudspeaker, by increasing the sweep length or by using better equipment (cables, sound card etc.). As can be seen the graph forms an almost straight line indicating an increase in S/N by 3 dB per doubling of sweep length, which is expected from the sweep-method theory[15].

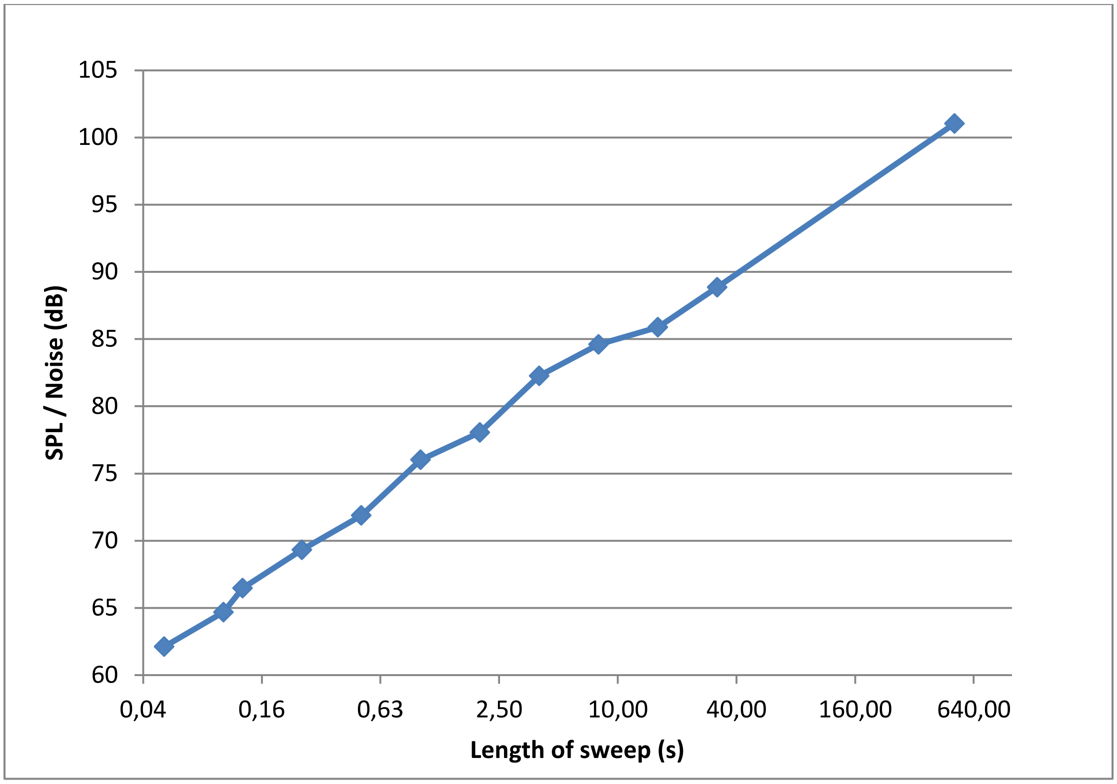


Figure 8: Measured Signal to Noise ratio with varying length of sweep.

4.2 Uncertainty of 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 was 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 graphs below (**Figure 9** to **Figure 14**) show statistics for the 7 positions for each of the parameters EDT, T_{30}, Ts, D_{50} and C_{80}. 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. However, as will be seen later, the simulated results do indeed differ between the significantly spaced receiver positions 1-5.

The T_{30} value in **Figure 10** shows a standard deviation of 0,02 seconds (0.2 JND) at 1000 Hz, increasing to 0.09 seconds at 125 Hz so it only varies little with position and should be easy to reproduce even if position is not exact. This agrees with the idea that T_{30} should be a global value that relates to the room not to a local position. All other parameters do show larger deviations and with some increases in deviations towards lower frequencies. The standard deviation for the other parameters are however not as big as one may think when visually inspecting the graphs. 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 (the G value obtained with a source power of 31 dB) have a standard deviation less than 1.2 JND's.

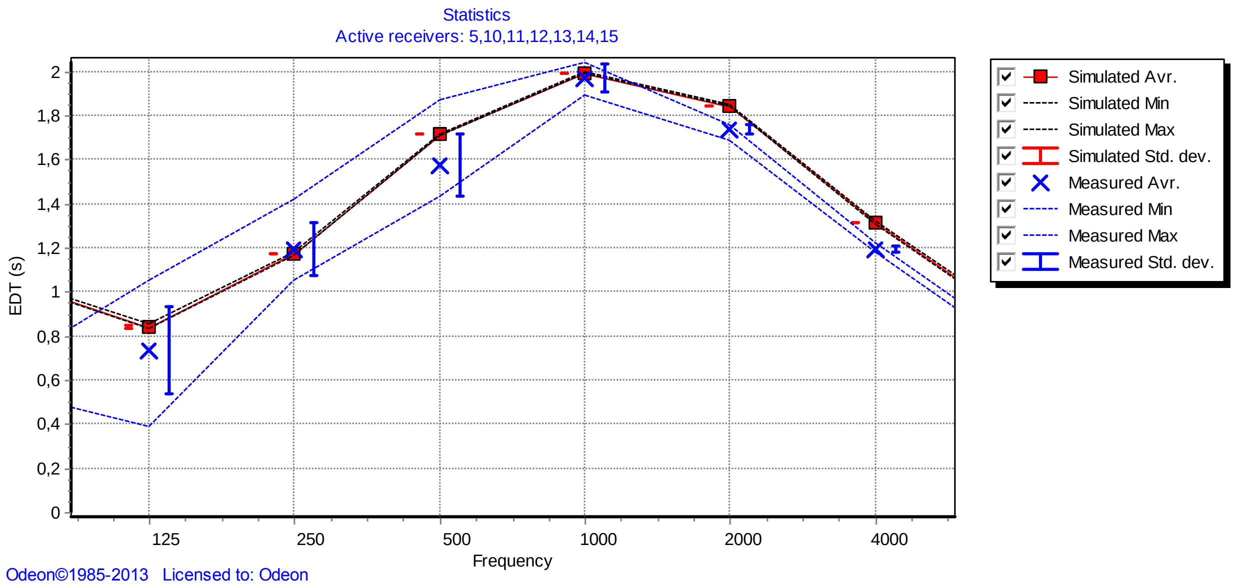


Figure 9: Uncertainty of position. Statistics on early decay time, EDT.

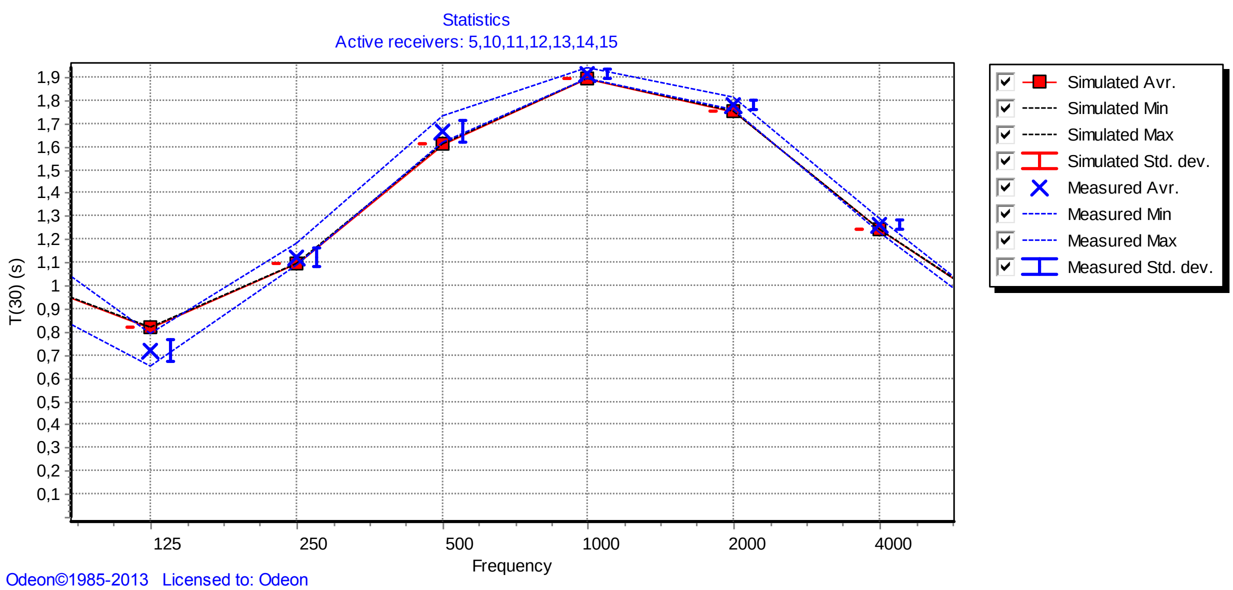


Figure 10: Uncertainty of position. Statistics on reverberation time, T_{30}.

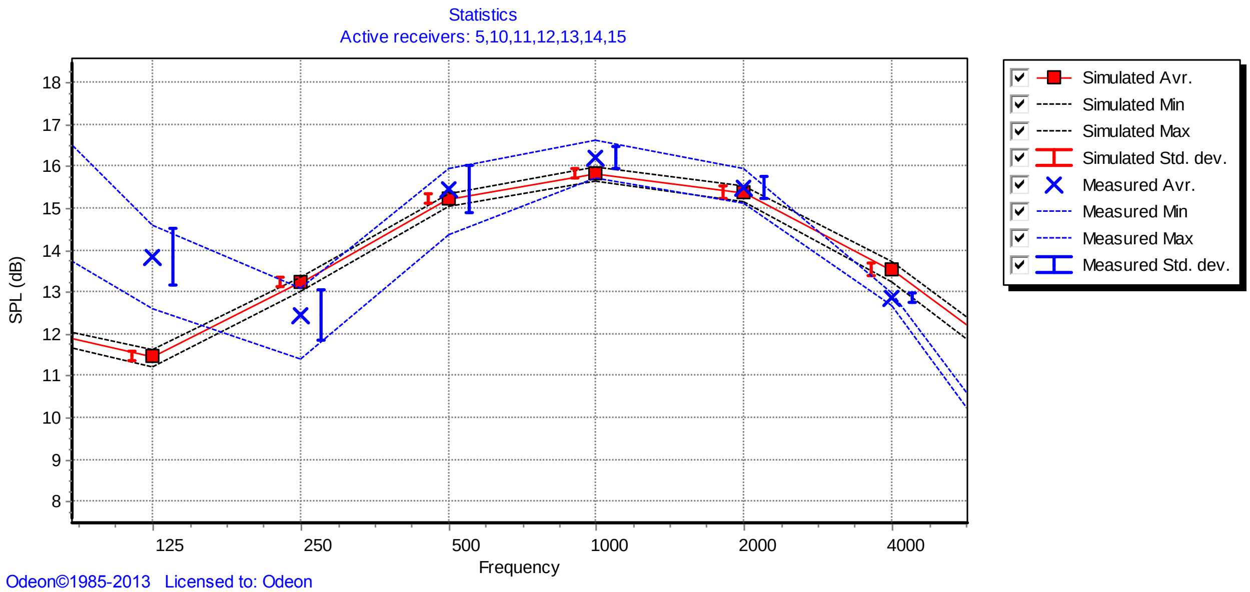


Figure 11: Uncertainty of position. Statistics on sound strength, SPL.

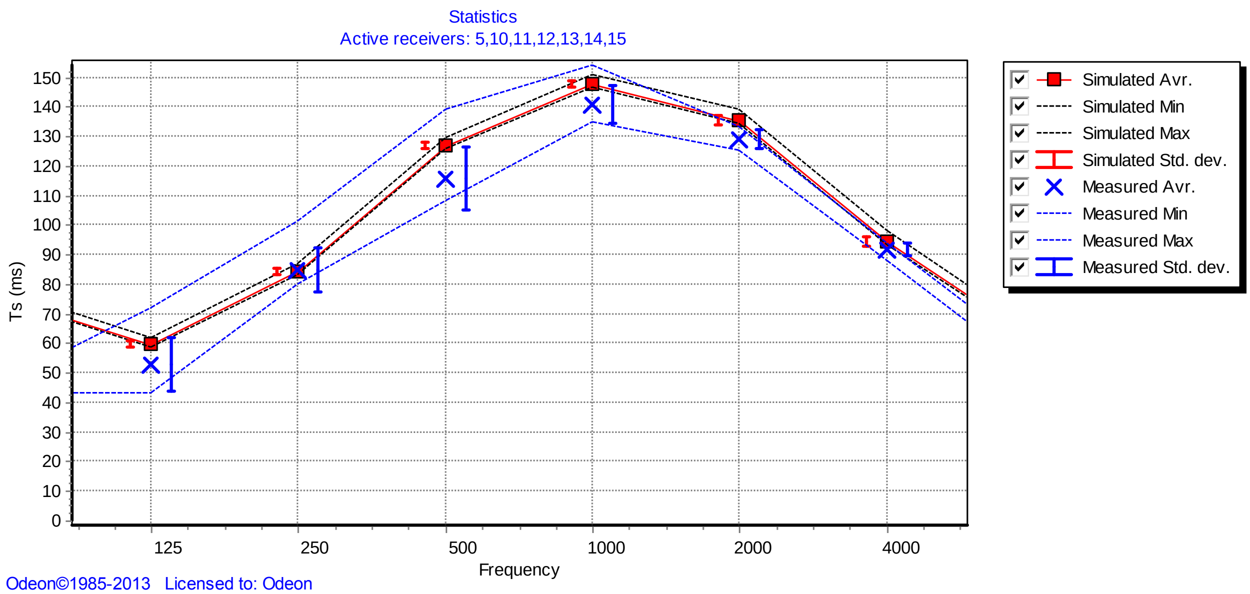


Figure 12: Uncertainty of position. Statistics on gravity time, Ts.

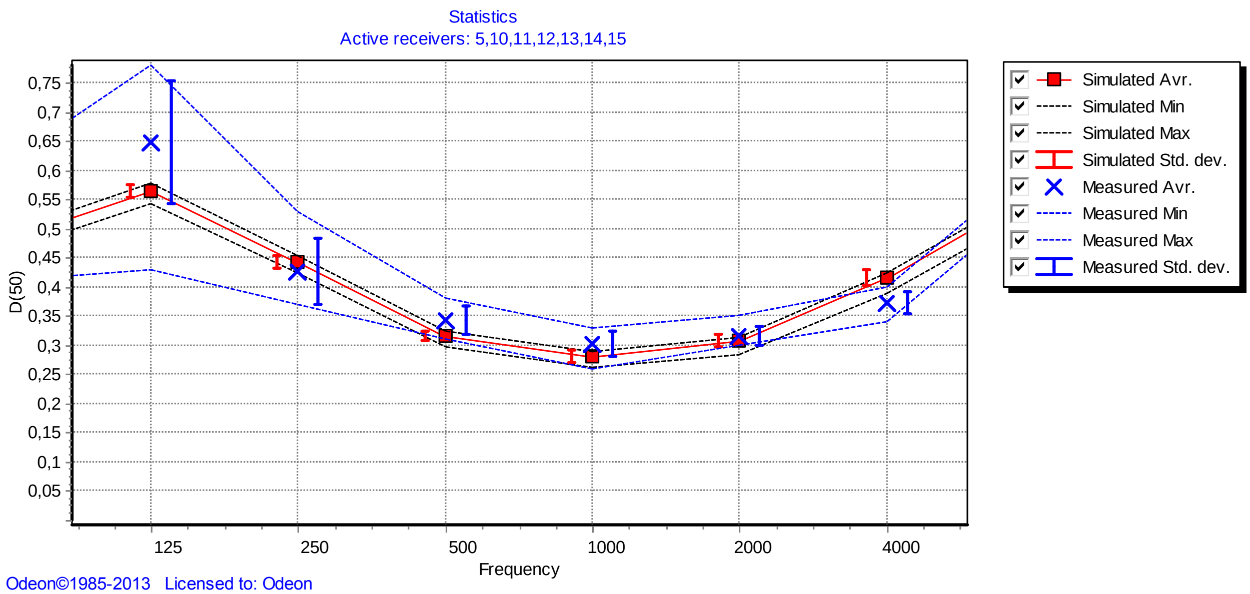


Figure 13: Uncertainty of position. Statistics on gravity time, D_{50}.

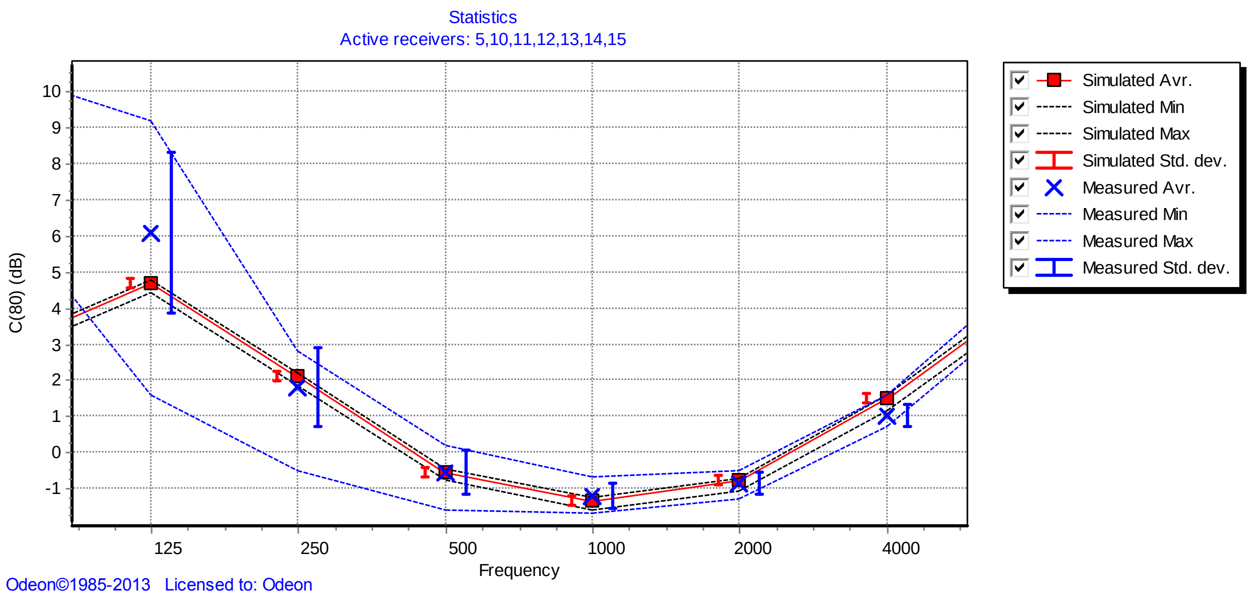


Figure 14: Uncertainty of position. Statistics on gravity time, C_{80}.

4.3 Different excitation stimuli

For the best quality of measurements the sweep method is recommended. However for surveys, it may be convenient to use a simpler approach e.g. to record and analyse hand claps, popping of balloons/paper bags etc. To get an idea how this works when the recorded measurements are analysed in ODEON we recorded a series of hand claps (**Figure 3**), popping of three balloons and as a reference an impulse response obtained from a 4 seconds long sweep with sufficient signal to noise ratio.

As can be seen in **Table 5** the T_{30} values obtained with balloons are in fine agreement with the sweep method. For the rest of the parameters there are some, though not large, differences from the values obtained with the sweep signal, probably mainly because the source (a person popping a balloon) is far from being an omni-directional source. There was some deviation between the T_{30} value obtained by hand clap and with the sweep method – it is very likely that this is due to insufficient S/N – in fact ODEON was not able to estimate the level of the noise floor which is why the line displaying the noise floor in (**Figure 3**) is dotted – thus the T_{30} value is expected to be too long as concluded in section 4.1 – the volume of auditorium 21 is to large to be excited with this stimuli.

Table 5: Values of eight room acoustic parameters at 1000 Hz obtained by impulse responses from different stimuli: Hand-clap, popping a balloon and sweep signal (reference measurement).

	Hand Clap	**Balloon 1**	**Balloon 2**	**Balloon 3**	**Sweep 4 *s***
EDT [s]	2.00	1.7	2.00	1.73	1.87
T_{20} [s]	2.01	1.92	1.87	1.92	1.80
T_{30} [s]	1.98	1.87	1.89	1.89	1.88
C_{50} [dB]	-3.8	-3.6	-3.7	-5.1	-2.0
C_{80} [dB]	-1.4	-0.6	-1.1	-1.7	-0.30
D_{50}	0.30	0.30	0.30	0.24	0.39
T_s [s]	145	124	143	140	127

5 COMPARISON OF SIMULATED AND MEASURED RESULTS

5.1 Auditorium 21

In **Figure 15** measured and simulated values of EDT, T_{30}, SPL (the G value with a source power of 31 dB), T_s, C_{80} and D_{50} are displayed for P1 and five receiver positions at 1000 Hz. In addition an extra frequency band (125 Hz) is displayed for C_{80}.

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. As for T_{30}, the difference between measured and simulated values varies from 0.01 to 0.03 seconds with an average deviation of 0.16 JND (it should be noted that the model was calibrated to match the average T_{30}). 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 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. There is not space for graphs displaying the rest of the octave bands from 125 to 4000 Hz, but except for the 125 and 250 Hz band, which shows deviations above 1 JND, there seems to be a fine agreement here too.

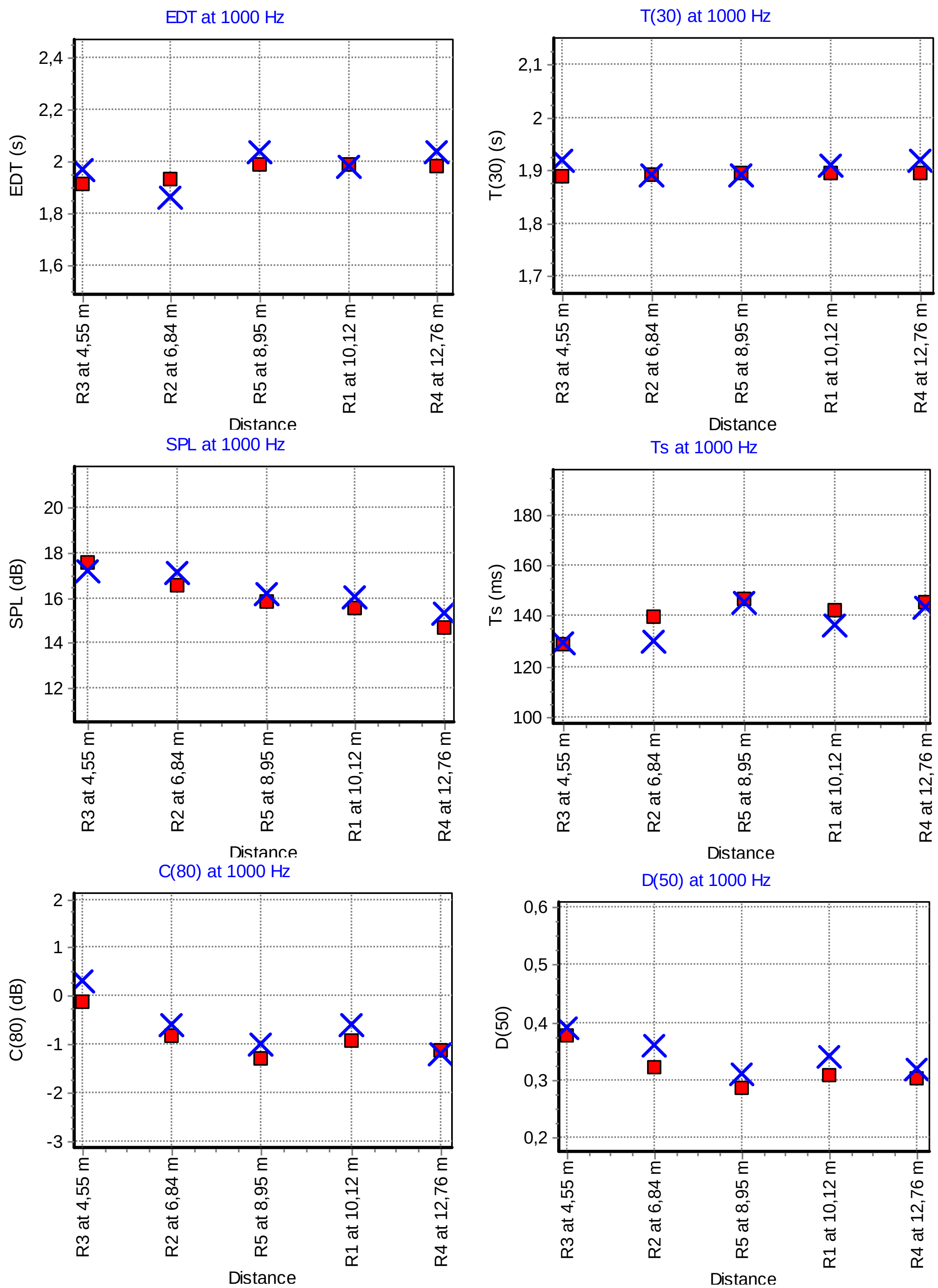


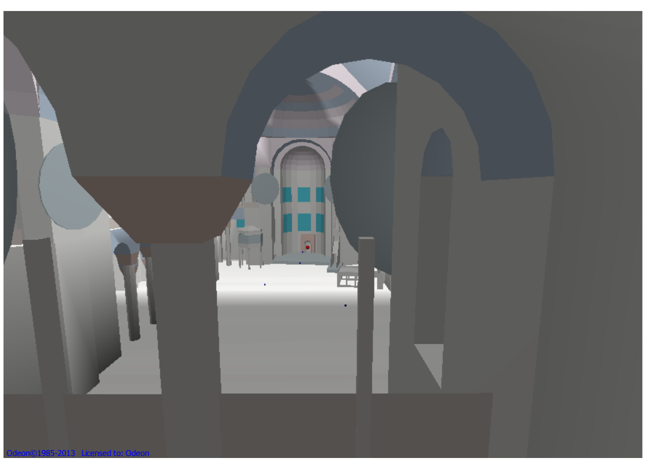
Figure 15: Simulated and measured room acoustic parameters for the five receivers in auditorium 21. Simulated parameters displayed with *red square* and measurements with *blue cross*.

5.2 Hagia Sophia

The second room studied is Hagia Sophia a large mosque/cathedral in Istanbul. This room has a length of almost 100 metres, a height of approximately 54 metres, a total volume around 200000 m^3 containing many coupled volumes and a reverberation time in the mid-frequency range close to 10 seconds. Hagia Sofia may be considered a performance space but not the typical auditorium for which ISO 3382-1 is probably intended. Therefore it is reasonable to expect that it is more difficult to find agreement between measured and simulated ISO 3382-1 parameters in this type of room and some of the parameters may have limited use.

Hagia Sophia was studied as part of the CAHRISMA project (Conservation of the Acoustical Heritage by the Revival and Identification of the Sinan's Mosques' Acoustics) a three-year project financed by the EU. In the CAHRISMA project room acoustics modeling and measurements were conducted on a number of mosques and byzantine churches according to ISO 3382. The modeling was done in the ODEON software and measurements were done with some MATLAB code that was created during the project. The measured impulse responses are still available so we have analyzed them with our current measurement system – though we do not have the calibration data for evaluation of the G parameter.

In the CAHRISMA project there was fine agreement between measured and simulated values of reverberation parameters and SPL however parameters such as Ts, D_{50}, and C_{80} were in pretty bad agreement for many positions. Some of the reason probably being that detection of onset time in measurements was not reliable and that parameters such as C_{80} were derived directly from the octave band filtered impulse response without "Window-before-filtering" as recommended by the current ISO 3382-1 – some of the C_{80} values were off by more than 12 dB! It must be noted that 6 of the 11 measurement positions presented here were located in distant galleries without any sight to the source and that a parameter like C_{80} is probably not a good indicator in such locations – still it is comforting for the user of simulation and measurement software that the output of simulation and measurements are in agreement.



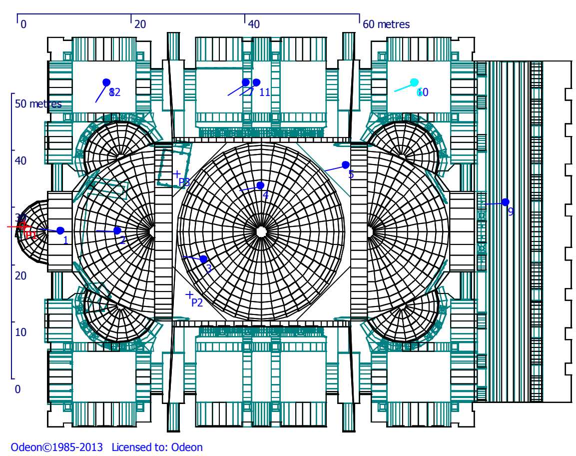


Figure 16: Left: Three-dimensional rendering of Hagia Sophia with colors corresponding to the reflectance of the surfaces [14]. Right: Ground-plan of Hagia Sophia.

In **Figure 16** measured and simulated values of EDT, T_{30}, T_s, C_{80} and D_{50} are displayed for P1 and 11 receiver positions at 1000 Hz. In addition an extra frequency band (125 Hz) is displayed for C_{80}.

The difference between measured and simulated parameters at 1000 Hz are larger in terms of JND's for Hagia Sofia than was the case for Auditorium 21 however the span for some of the parameters are also much larger than is the case for a normal auditorium.

For EDT the difference between measured and simulated values differs between 0.01 and 3.49 JND's. The biggest deviation is found at the receiver point where source and receiver are very close to each other which is 16 meters in this large room! Indeed the first 8 dB of the measured backwards integrated curve is very steep because it is influenced by direct sound. The average deviation between measured and simulated EDT for the 11 positions is 1.15 JND.

As for T_{30}, the difference between measured and simulated values differs between 0.11 and 0.54 JND's with an average difference of 0.22 JND's. Both simulations and measurements shows that T_{30} is almost a constant throughout the whole volume although the space consists many coupled volumes and some of the receivers are located at very remote positions.

The measured and simulated values of Ts in the 11 positions follow the same trends with position and vary substantially in the volume from 233 to 975 milliseconds. Using the JND of 10 ms for Ts as given in ISO 3382-1 would suggest that there are large deviations between measured and simulated values of Ts however this value is highly questionable for a room with this volume and reverberation time – maybe a relative value of 5 % would be more meaningful.

The D_{50} parameter is within 1.19 JND for eight of the eleven positions. The three last positions are within 2.28 JND's.

C_{80} has a deviation ranging from 0.47 to 4.66 JND's with an average deviation of 2.27. One may say that this deviation is very high however considering that C_{80} in the 11 positions ranges from -18.8 to 4.8 dB it shows a much higher fluctuation than in a normal auditorium. In any case one should be careful using parameters such as C_{80} for evaluation of C_{80} in coupled spaces such as Hagia Sofia as the buildup of the squared impulse response by far exceeds the 80 milliseconds time limit used for the C_{80} parameter.

There is not space for graphs displaying the rest of the octave bands from 125 to 4000 Hz, but the agreement between measured and simulated values are fairly similar to those of the 1000 Hz band.

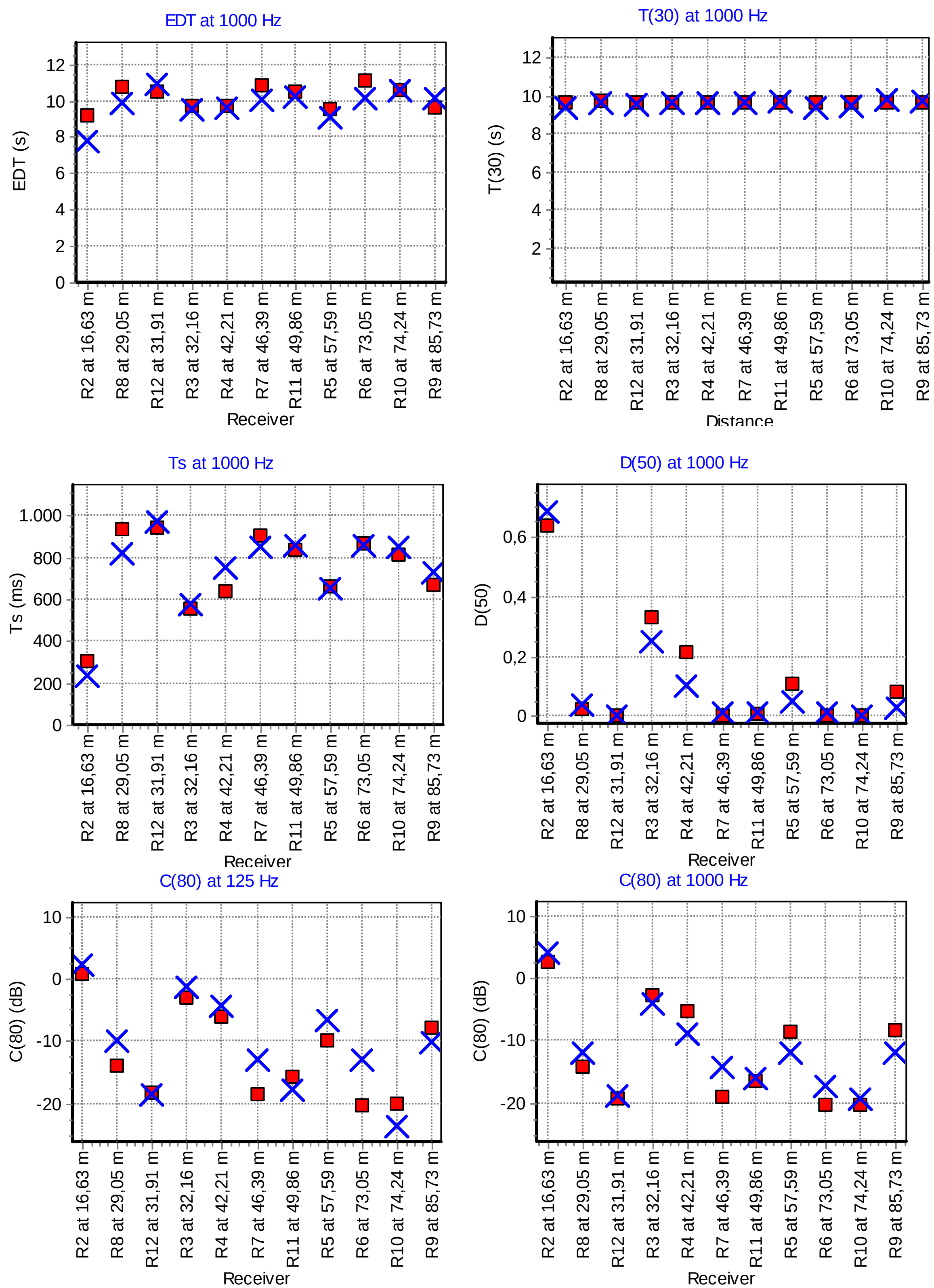


Figure 17: Room acoustic parameters simulated and measured in *Hagia Sophia* at 1000 Hz. Simulated parameters displayed with *red square* and measurements with *blue cross*.

6 CONCLUSIONS

It is possible to make reasonable measurement of T_{30} using alternative excitation signals such as hand claps or popping of a balloon if sufficient signal to noise ratio can be obtained. The other ISO 3382-1 parameters may have limited accuracy probably because the source is not perfect omni-directional.

Truncation of impulse responses and background noise in impulse responses may lead systematic errors on T_{30} if not compensated for. If compensating for both errors correct results may be achieved with moderate signal to noise levels.

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 at a central position in an auditorium – this indicates that parameters can be reproduced even if receiver position is not exact. For simulations in ODEON small deviations in position seems negligible.

It is possible to simulate and measure room acoustics parameter according to ISO 3382-1 if care is taken in implementation and use of simulation and measurement algorithms. Indeed the examples used in this paper show reasonable agreement. In the auditorium example there is fine agreement between measured and simulated values and even in the large cathedral Hagia Sofia trends with receiver position agree well between simulations and measurements.

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