APPLYING IN-SITU RECALIBRATION FOR SOUND STRENGTH MEASUREMENTS IN AUDITORIA

Claus Lynge Christensen[1], *Odeon A/S, Denmark*
George Koutsouris[2], *Odeon A/S, Denmark*
Juan Gil[3], *Odeon A/S, Denmark*
Jens-Holger Rindel[4], *Multiconsult, Norway - Odeon A/S, Denmark*

The measurement of the Sound Strength (G) room acoustic parameter requires a calibration of the measuring system, so either the sound power of the source or the sound pressure level at a distance of 10 m in free field is known. Two procedures are described in ISO 3382-1: Calibration in a reverberation chamber and calibration in free field in a large anechoic chamber. Both methods require that all conditions in the measuring chain (amplification gains, sound card, cables etc.) remain fixed during calibration and *in-situ* measurements. This can be impractical mainly because measurements may be performed much later than calibration and at a distant location, and it cannot be guaranteed that the conditions in the measuring chain remain unchanged. In this paper a Two-step calibration procedure is proposed that removes the requirement of fixed conditions of the measuring chain. The procedure adds a second step to the normal reverberation room or free field calibration methods, in order to allow compensation for any shift in octave bands from the calibration to the *in-situ* measurement condition. The procedure is implemented in the ODEON Room Acoustics Software v.13.03 as a Two-step calibration to be applied in connection with the sine-sweep method for room acoustic measurements.

1 INTRODUCTION

The difficulties related to measurement of Sound Strength G have been discussed for some time, especially by Leo Beranek, who claimed that there should be a systematic difference between measurements made by Japanese researchers and other European and American ones.[1,2] The calibration of the sound source for this kind of measurements is not as simple as it appears at first place. One of the most common problems is that the gain in parts of the measuring equipment might have changed from calibration to measurement. Although it is always recommended to save all gain settings of amplifiers, audio interfaces and computer, missing this information is very common, especially if calibration was done long time before the actual measurement. Moreover, changes of gains might be desirable if the user realizes that the gain settings used during calibration are not optimal for the actual measurement. A few proposals have been made recently for assisting practitioners to eliminate factors that can ruin a calibration of the G parameter. A study by Katz[3] suggests using gated versions of the actual hall measurements to calibrate the level of the same measurements, in order to perform an *in-situ* calibration without use of reverberation or anechoic chambers. The method assumes that an window of 5 ms can be applied to separate the direct sound from the reflections in the impulse response. However, due to this assumption only source-receiver configurations with a path difference between direct and reflected sound greater than 1.7 m can be considered. In addition, the 5 ms window might not be sufficient to capture the direct sound at low frequencies. Hak *et al.* have suggested an *in-situ* calibration by placing the microphone at a near-field distance of 1 m.[4] According to their assumption, this distance can offer a sufficiently high level of direct sound, so it is not affected by

[1]email: clc@odeon.dk
[2]email: gk@odeon.dk
[3]email: jg@odeon.dk
[4]email: jhr@odeon.dk

subsequent reflections. Therefore, it should be possible to perform a measurement as in free field. However, a 1 m distance cannot be considered far field, as it is required by the ISO standard 3382-1.[5] Moreover no algorithm is applied to isolate the direct sound from the following reflections, and as a result it is likely to obtain direct sound contaminated by early reflections. Finally, no source is perfectly omni-directional meaning that the proposed method can provide different results at different directions. According to the ISO standard 3382-1[5] measuring positions every 12.5° are needed when measuring the direct sound from a source, to smoothen any irregularities in the radiation pattern. Luykx *et al.*[6] have suggested the use of Barron's theory to determine a reference radius around the source which is not too close to the room walls and still is assumed not to be affected by the direct sound. An average G parameter is derived at the reference radius.

In this paper we introduce the concept of a Two-step calibration method, which combines the procedure for calibrating the source in a reverberation chamber or an anechoic chamber with an *in-situ* correction. In the proposed method, the main calibration is still performed in a well-known environment (reverberation chamber or anechoic chamber) according to the ISO 3382-1,[5] which offers high level of accuracy. However, an *in-situ* correction is applied as a second step, that offers robustness against changes in the measuring equipment. The most important benefits of such a *recalibration* can be summarized as:

1. Correcting for accidental changes in output or input gains for any part of the chain (audio interface, computer volume, loudspeaker amplifier, microphone amplifier).
2. Correcting for intentional changes in output or input gains. For example, if the dynamic range of the recording is not sufficient to derive the desired room acoustic parameters, an increase of gain *in-situ* might be beneficial.
3. Allows the user to change part of the equipment - audio interface, amplifiers, cables or even microphone - provided that the source remains the same.
4. Allows compensation for long term changes of the source itself due to aging, and counteracts drifting of other components involved in the measuring chain (e.g. volume potentiometer).

2 THEORY

2.1 Diffuse-field Calibration

Calibration of an omni-directional source is needed for the estimation of the *Sound Strength* parameter,[5] which quantifies the influence of the room in the perceived loudness and describes the amplification applied to the source by the room. By definition, the Sound Strength G for a specific frequency band, is given as the logarithmic ratio of the sound energy (squared and integrated sound pressure) of the measured impulse response at any position in the room, to that of the sound energy of the impulse response in free field at 10 m distance from the source:

$$G = 10\log_{10}\frac{\int_0^\infty p^2(t)dt}{\int_0^\infty p_{10}^2(t)dt}\,(dB), \tag{1}$$

which can be written as:

$$G = L_{pE} - L_{pE,10}, \tag{2}$$

where $L_{pE} = 10\log_{10}[\frac{1}{T_0}\int_0^\infty \frac{p^2(t)dt}{p_0^2}]$ and $L_{pE,10} = 10\log_{10}[\frac{1}{T_0}\int_0^\infty \frac{p_{10}^2(t)dt}{p_0^2}]$ is the Sound Pressure exposure level (Simply called SPL henceforth) of $p(t)$ and $p_{10}(t)$.

The variables in the equations are as follows:

- $p(t)$: Instantaneous sound pressure of the impulse response at the receiver's position in the room.
- $p_{10}(t)$: Instantaneous sound pressure of the impulse response at 10 m distance from the source in free field.

- p_0: Reference sound pressure, 20 μ Pa.
- T_0: 1 sec.

To calculate Equation 2 the value of $L_{pE,10}$ needs to be known. According to the ISO standard 3382-1,[5] the $L_{pE,10}$ level can be derived either in diffuse-field or in free-field. A number of measuring positions are required for each method. The variance among the measuring positions is generally lower in the diffuse-field method than in the free-field method. This is because in the first method the measurement depends on the sound field in the entire room, while in the second method it solely depends on the direct sound from the source. In this paper we present a new Two-step method based on the diffuse-field calibration. However, the process can also be applied directly for the free-field method.

Using a diffuse field assumption, $L_{pE,10}$ can be calculated from the average SPL, L_{pE}^{RevCh}, inside a reverberation chamber:

$$L_{pE,10} = L_{pE}^{RevCh} + 10\log_{10}\frac{A}{S_0} - L_p^W - 6 \ \ (\mathrm{dB}) \tag{3}$$

In this equation, A is the equivalent absorption area (in m^2) in the room that is calculated from the Sabine's formula (diffuse-field assumption): $A = 0.16V/T$, with V being the volume (in m^3) and T the reverberation time (in sec). In addition, $S_0 = 1\ m^2$. L_p^W equals 31 dB re 1 pW and represents the sound power level of an omni-directional source that results to 0 dB SPL at 10 m distance.

Combining Equation 2 with Equation 3 and replacing L_p^W by 31 dB we obtain:

$$G = L_{pE} - L_{pE}^{RevCh} - 10\log_{10}\frac{A}{S_0} + 37 \ \ (\mathrm{dB}) \tag{4}$$

As mentioned for Equation 2, L_{pE} is the SPL measured *in-situ*. In addition, as can be seen in Equation 4 what matters is the difference in the measured levels in the reverberation chamber and *in-situ* - not the absolute values. Therefore, we do not need to calibrate our measurement to absolute SPL levels (typically 94 dB at 1 kHz calibration[7]) as long as the SPLs are all measured with the same reference. Let us now define:

$$Adj^{RevCh} = -L_{pE}^{RevCh} - 10\log_{10}\frac{A}{S_0} + 37 \ \ (\mathrm{dB}) \tag{5}$$

as the calibration level adjustment. As a result, when measuring *in-situ* the recorded SPL L_{pE} is converted to a G value according to the following formula:

$$G = L_{pE} + Adj^{RevCh} \ \ (\mathrm{dB}). \tag{6}$$

3 METHODOLOGY OF THE TWO-STEP CALIBRATION

The Two-step calibration consists of two steps of measurements. The first step is the calibration in the reverberation chamber as described in 2.1. The second step is a measurement of relative SPL at a known distance and orientation from the source. This second step is performed in the reverberation chamber as well as *in-situ*. Any change in gains per octave indicates a change of gain of one or more components in the measurement chain from reverberation chamber to *in-situ* and is then corrected for. If the difference in the source output for each band is $L_p^{S,Calib} - L_p^{S,Meas}$, the new calibration level adjustment will be:

$$Adj^{inSitu} = Adj^{RevCh} + L_p^{S,Calib} - L_p^{S,Meas} \tag{7}$$

In this case the Sound Strength will be given by:

$$G = L_{pE} + Adj^{inSitu}. \tag{8}$$

If no change in the source output occurs when moving from the reverberation chamber to the *in-situ* environment, then $Adj^{inSitu} = Adj^{RevCh}$, and Equations 8 and 6 provide the same result. In such a case the Two-step calibration becomes equivalent to the method described in Section 2.

3.1 First Step

The first step is performed in the reverberation chamber. The ISO standard 3741[8] provides the guidelines for sound power measurements in a reverberation chamber. A recommended number of source - microphone (receiver) positions is 15. The user should place the microphone at different locations in the room, not too close to the walls or the source. At the end of the process the calibration values for the G parameter are obtained, according to Equation 6.

3.2 Second Step

The second step in the proposed two-step method consists of two sets of measurements: one in the reverberation chamber (*reference* measurement) and one *in-situ*. These are performed at the same distance and direction from the source. The purpose is to derive any difference in the output of the source. Since reflections have to excluded, a gating has to be performed to capture a well-defined part of the direct sound. To ease this process the microphone should be placed as close as possible to the source.

3.2.1 Measurement of Direct Sound - Gating

Figure 1 illustrates the setup used for measuring the direct sound from the source. An electro acoustic source such as a Dodecahedron, which is commonly used in room acoustics measurements, has an imperfect response and directivity pattern. In the frequency domain it produces colouration and in the time domain it produces ringing (smearing in time), rather than a perfect Dirac function. In the first place, the frequency colouration is the reason why we need to make a system calibration per octave band. Ringing, on the other hand, makes it difficult or impossible to isolate the direct sound from reflected sound if measuring the source in a non anechoic environment.

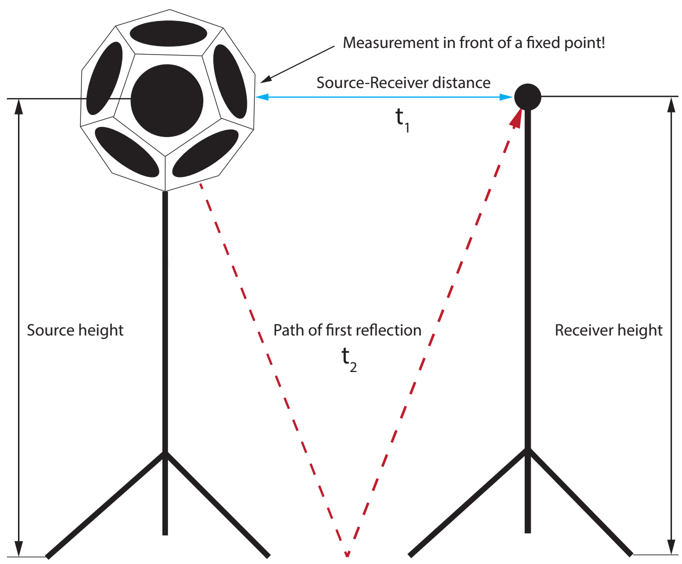


Figure 1: Sketch of source-microphone position during the measurement of the direct sound. A sufficient interval between t_2 and t_1 should be ensured.

Smearing in time depends on the frequency band; at low frequency the smearing is higher, whereas it decreases at higher frequencies. In principle the source rings forever. However, at higher frequencies a significant portion of the direct sound will be included within a time window of a few milliseconds (a millisecond corresponds to approximately 0.344 metres of sound propagation in atmospheric air). To capture a well-defined part of the direct sound from the source, sufficient separation in time between the path of the first reflection and the path from source to receiver is needed. A way to accomplish this is to have Source/Receiver heights equal to 1m and Source-Receiver distance less or equal to 1m.

When performing an *in-situ* correction measurement in an auditorium it will normally be possible to have several metres of distance to walls and ceilings. However the restriction is that the source mounted on a tripod will normally not be further away from the floor than 1 to 1.6 metres. The same restriction applies when performing a measurement of direct sound.

For the second step in the proposed two-step calibration method it is not crucial to measure the entire duration of the direct sound in the reverberation chamber (and *in-situ*). It is only important that:

- A well-defined part of the direct sound (not necessarily the major part) is measured.
- The onset time of the waveform in the lab and *in-situ* is detected in a consistent manner.
- The duration of the time window evaluated is the same in the lab as well as in situ.

For the method to work it is also important that the shape of the waveform included in the gated time window is largely unchanged. If for example strong frequency colouration is introduced in the *in-situ* measurement (e.g. using an equalizer), this produces additional ringing and the wave form can change significantly. In this case the two wave forms are not comparable.

The length of the time window could in principle be specified by the user. However, in ODEON we have chosen to make it of fixed length so the user does not have to enter any geometrical data on source height h_s, receiver height h_r and source-receiver distance d_{sr}. In this way it is also ensured that the same window length is used for both measurements. As a choice we have decided that $d_{sr} \leq 1$ m and $h_s, h_r \geq 1$ m, resulting in a time window of 3.6 milliseconds where the direct sound is not mixed with room reflections. As a safety factor we multiply the time window by 0.9, resulting in a time window length of 3.2 milliseconds to be absolutely sure that no reflections are included. In Figure 1 the time window is represented by $t_2 - t_1$, where t_2 is the time corresponding to the first reflection and t_1 is the time corresponding to the direct sound. Both time values refer to distances measured from the boundary of the source to the microphone.

3.2.2 Response of the System

Some gain changes may be frequency independent, but there may be gain shifts, which do not affect the source response equally in all bands. If a simple broadband gain shift occured, then it would be sufficient to measure the $L_p^{S,Calib}$ and $L_p^{S,Meas}$ levels for one frequency band and apply the correction $L_p^{S,Calib} - L_p^{S,Meas}$ in Equation 7 for the other bands. However, there are cases where the individual bands change differently, thus a broadband shift would not be applicable. A common example is when cables of different length are used in the calibration (reverberation) chamber and *in-situ*. A longer cable provides higher impedance and leads to a drop in the output, while a short cable provides lower impedance, resulting in an increase of the output. This gain difference might not be equal for all frequency bands.

3.2.3 Position the Microphone for Recalibration

The microphone should be placed at a fixed position in front of the source for the *reference* and the *in-situ* measurement. Since the measuring distance is not so large (usually up to 1 m), the source does not act like an omni-directional point source and the waveform may look different in different directions. For that reason it is necessary to keep a fixed distance as well as fixed direction. For example, if a dodecahedron is used, the following steps should be followed:

- Mark one of the loudspeaker units as the reference one.
- Locate the microphone at the same height as the centre of the dodecahedron.
- Ensure that the microphone is pointing exactly towards the reference unit.

When a source, other than a dodecahedron, is used, similar considerations regarding direction and position should be applied.

To obtain a reliable relative SPL measurement with small uncertainty, it may need to be based on multiple measurements. For each measurement the microphone must be repositioned. This means that it is not enough to measure multiple times without repositioning the microphone. A set of multiple measurements should be taken and averaged. It is suggested to obtain the measurements according to the following procedure:

- Remove the microphone away from the current position.
- Reposition the microphone by measuring the distance again.
- A suggested number of measurements is five.

3.2.4 Detecting the Onset Time

As the loudspeaker - such as a dodecahedron - is not capable of producing an ideal Dirac function, the direct sound is smeared in time. Therefore it is not obvious when the direct sound starts. The ISO standard 3382-1[5] suggests on pp. 18-19 that 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 the broad band impulse response. This is called *trigger level* in ODEON, and the default value is 40 dB, when deriving measured room acoustics parameters, such as Reverberation Time T_{30} and ClarityC_{80}.[9] However, a 20 dB *trigger level* has been found to give very reproducible onset time for the *reference* and the *in-situ* measurement, even with poor signal to noise ratio. Using 20 dB rather than 40 dB, one may miss a tiny bit of the direct sound. However, in the Two-step method it is not important that all direct energy is detected. It is just required that the window detected in the *reference* and the *in-situ* impulse response is consistent. To achieve that, ODEON defines the onset time automatically, without allowing the user to perform a manual adjustment, as this could harm the consistency of the detection between the *reference* and the *in-situ* measurement.

3.2.5 Obtaining the Sound Pressure Level per Octave of the Gated Impulse Response

Once the onset time is detected according to Section 3.2.4, gating of the direct sound needs to be performed for a well-defined interval, as described in Section 3.2.1 for each octave band. Measured time intervals may not be precisely derived if calculated directly from the filtered response, because the ringing of the octave filter will continue after the end of the interval and overlap with the first reflections from the room. This can be particularly important for the lower frequency octave bands where the filters are “long” (have long impulse responses). In order to bypass this filter problem, ISO 3382-1[5] suggest the “Window-before-filtering” approach which is the method implemented in ODEON.[10] First, the onset time is estimated from the broad band impulse response. Then, an early broad band time interval, which starts at the onset time and ends desired number of milliseconds later, is isolated. In order to estimate the energy/sound pressure level per octave, octave band filtering is applied for the specified time interval. 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.

3.3 Summary of the Method

Figure 2 illustrates the overall procedure of a Two-step calibration. In the first step the normal diffuse-field calibration is performed by measuring the response at various source-microphone positions inside the reverberation (Rvrb) chamber - usually 15 combinations. For the second step only two small sets of measurements are required at a fixed direction and distance from the source - usually 1 m. Section 3.2.3 provides some suggested guidelines for making the measurements. Initially, a set of measurements is obtained inside the reverberation chamber. All gains should remain fixed and the same as for the first step. Finally, a correction is performed *in-situ*, using another set of measurements

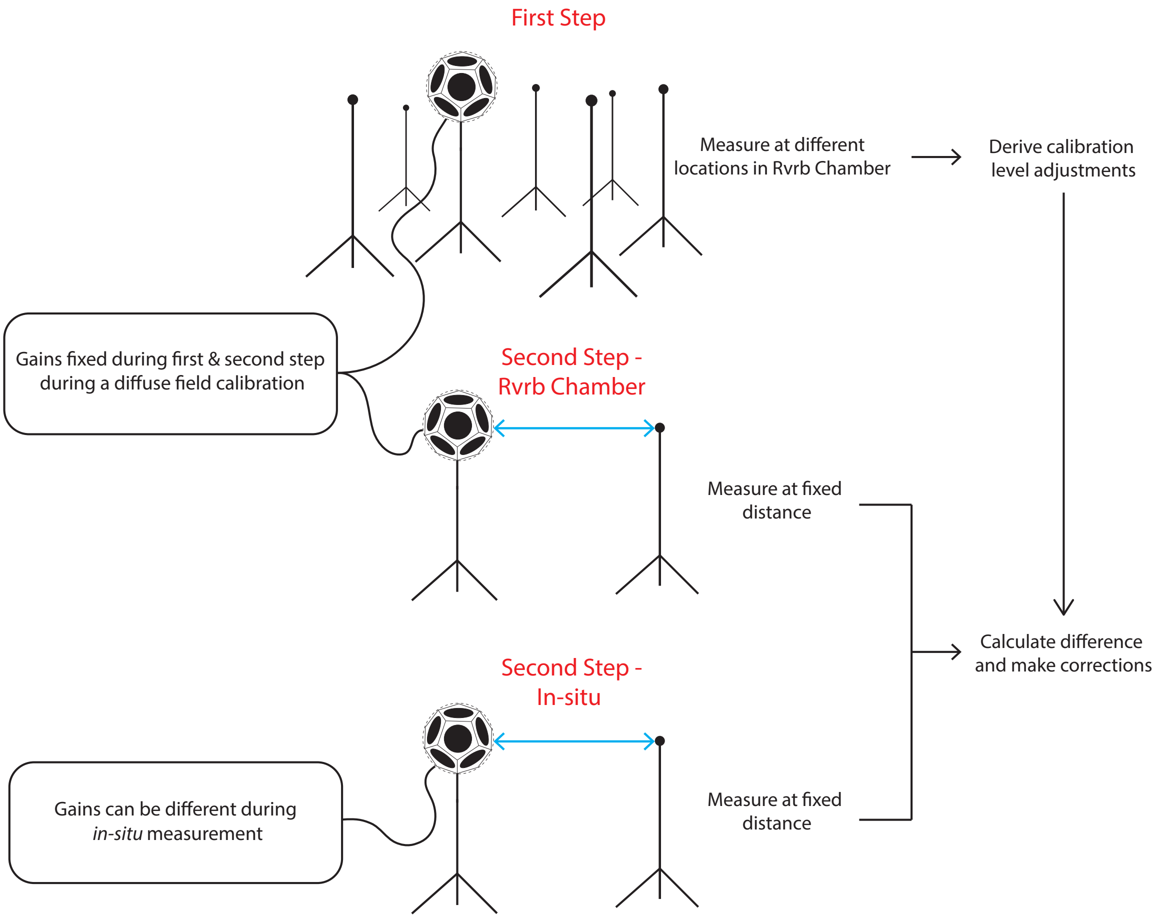


Figure 2: Diagram of the complete Two-step (recalibration) method, based on the diffuse-field assumption.

according to Section 3.2.3. The gains can now differ from the gains used in the reverberation chamber. As a result, a source correction, $L_p^{S,Calib} - L_p^{S,Meas}$, applies and the calibration values Adj^{inSitu} are given by Equation 7.

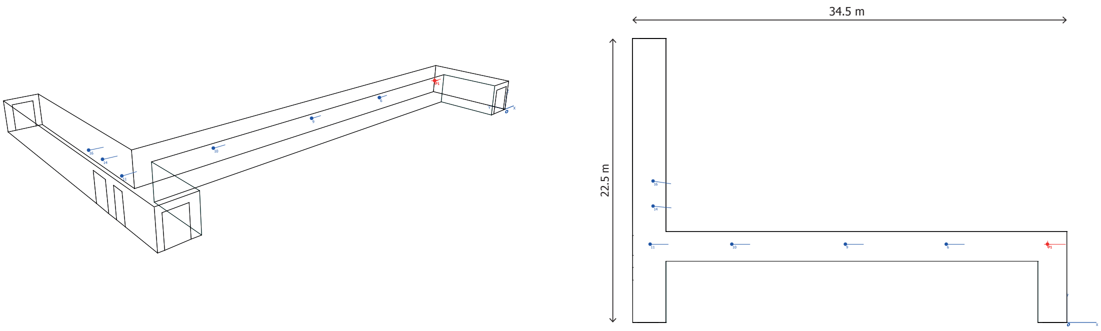
4 EVALUATION OF THE METHOD

The proposed Two-step calibration method was evaluated using a reverberation chamber and the room where G was to be measured. The reverberation chamber 904 in building 354, at the Technical University of Denmark (DTU) was used (Figure 3a). The chamber has a volume of 240 m^3 and a reverberation time of approximately 8 sec at 63 Hz. For performing the *in-situ* correction a basement corridor was used (Figure 3b). An illustration with source-receiver positions is shown in Figure 4. The corridor connects building 352 with buildings 355 and 354 at DTU. It was preferred to use this special room, instead of an auditorium, in order to obtain high variation in G at different locations. The room can be considered a challenging case due to the fact that it has parallel and hard walls, with a highly absorbing ceiling covered by insulated water pipes. This makes the positioning of the microphone critical.

Initially a number of impulse responses were recorded in the reverberation chamber (Figure 3a), for the first step of the recalibration method. All gains in the measuring equipment remained fixed. Afterwards, five measurements were taken at a fixed distance/direction from the source, for use in the second step (Section 3.2). The gains in the reverberation chamber remained fixed for all measurements, but they were increased *in-situ*.

(a) Calibration in Rvrb chamber. **(b)** Measurement *in-situ*.

Figure 3: Photos of the measurement in the reverberation chamber and *in-situ* for the proposed Two-step calibration method.



(a) Perspective view **(b)** Ground plan

Figure 4: The room for the *in-situ* evaluation. It can be considered as a special room with high variation of acoustic parameters at different receiver positions.

4.1 Comparison of the Gated Response in the Reverberation Chamber and *in-situ*

To evaluate the reliability of the gating function described in Section 3.2.1, five measurements were taken inside the reverberation (Rvrb) chamber and another five measurements *in-situ*, at the basement corridor of Figure 4.

For all these measurements the equipment and gain settings remained unchanged. The distance d_{sr} and the direction from the source remained fixed. Initially three microphone distances were evaluated: 0 m, 0.5 m and 1 m. Distance 0 m means that the microphone was almost touching the reference unit of the dodecahedron source. For each distance and set of measurements the average and the standard deviation were calculated. The two closest microphone positions gave a higher standard deviation (STD) for each measurement set, leading to a conclusion that the sound field is less unstable as the microphone approaches the source and more sensitive to errors when following the procedure described in Section 3.2.3. Consequently, it was judged that a distance of 1 m from the source is a more reasonable choice for achieving reproducible results, and it still provides a time window sufficient for the gating of the direct sound (Section 3.2.1).

Table 1 presents the results for each octave band and for $d_{sr} = 1$ m. The difference between the averages of each group is shown in the second last row in the table. In addition to that, the pair of measurements giving the maximum difference in the reverberation chamber and *in-situ* are highlighted in red, and the corresponding difference is shown at the final line of the table.

Table 1: Relative octave band SPL levels (dB) and average for five measurements at 1 m distance and fixed direction from the source, together with their average, inside the reverberation chamber and the basement corridor.

	63	125	250	500	1000	2000	4000	8000
Rvrb Chamber								
Meas. 1	-45.96	-40.07	-35.05	-37.10	-37.94	-31.33	-28.16	-26.94
Meas. 2	-45.99	-40.13	-35.14	-37.24	-38.46	-30.71	-28.34	-26.82
Meas. 3	-46.05	-40.15	-35.12	-37.18	-38.67	-30.58	-28.51	-27.00
Meas. 4	-46.00	-40.18	-35.16	-37.28	-38.96	-30.59	-28.61	-26.89
Meas. 5	-46.03	-40.14	-35.11	-37.18	-38.67	-30.61	-28.49	-26.97
Average	**-46.00**	**-40.14**	**-35.12**	**-37.20**	**-38.53**	**-30.75**	**-28.42**	**-26.92**
STD	**0.03**	**0.04**	**0.04**	**0.06**	**0.34**	**0.29**	**0.16**	**0.06**
In-situ								
Meas. 1	-45.94	-40.16	-35.21	-37.36	-39.30	-30.61	-28.81	-27.12
Meas. 2	-45.79	-40.04	-35.18	-37.34	-39.01	-30.64	-28.71	-27.18
Meas. 3	-46.01	-40.13	-35.16	-37.23	-38.30	-31.00	-28.27	-26.98
Meas. 4	-45.93	-40.11	-35.20	-37.31	-38.68	-30.76	-28.53	-27.13
Meas. 5	-45.84	-40.04	-35.15	-37.27	-38.55	-30.79	-28.38	-26.95
Average	**-45.90**	**-40.10**	**-35.18**	**-37.30**	**-38.76**	**-30.76**	**-28.54**	**-27.07**
STD	**0.08**	**0.05**	**0.02**	**0.05**	**0.35**	**0.14**	**0.20**	**0.09**
Difference between averages	**0.10**	**0.04**	**-0.06**	**-0.10**	**-0.23**	**-0.01**	**-0.12**	**-0.15**
Maximum difference	**0.26**	**0.14**	**0.16**	**0.26**	**1.36**	**0.72**	**0.65**	**0.36**

In most octave bands the difference between the averages is close to zero, proving that the gating algorithm provides stable truncation of the impulse response from the source at a well-defined time. As explained in Section 3.2.1, the goal of the algorithm is not to isolate the whole direct sound in the impulse response, but a well-defined part of it - even shorter than the expected duration. This is done in order to ensure complete truncation of room reflections in all octave bands and especially at low frequencies where smearing occurs, causing overlapping of the direct sound with the following reflections. According to the results in the final line of the table, if only one measurement is performed in the reverberation chamber and *in-situ* the difference (error) can be high. At the 1000 Hz octave band the maximum difference between two measurements is 1.36 dB, which is higher than the Just Noticeable Difference (=1 dB for SPL). On the other hand, averaging over a number of measurements reduces this error. All differences between averages in Table1 are far below 1 JND in all octave bands. It can be suggested that a single measurement can be sufficient if a mounting is attached to the loudspeaker, to ensure that distance and orientation of the measurement point is well defined from lab to in situ measurement. The distance in this case can be rather short (e.g. 10 cm). In this way any errors due to uncertain placement of the microphone can be eliminated.

4.2 Frequency-dependent Change of the System

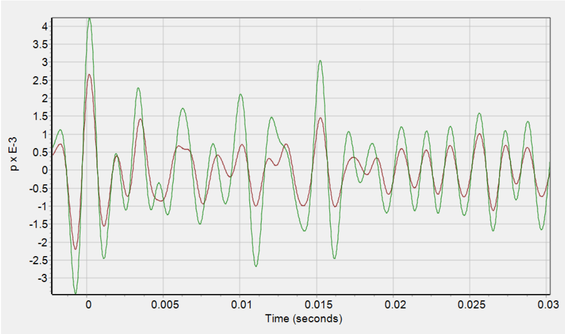
Ideally, a change in the gain settings should affect the output of the source equally for all frequencies. This means that one could calculate the shift in one of the octave bands and apply the same correction for the rest bands. However, it is reasonable to assume that changes in the gain settings can affect the output differently in each octave band. Moreover there might be cases where part of the equipment is modified, leading to altered frequency response. During the measurements *in-situ* the cable connecting the amplifier to the source was found to be too short for the dimensions of the room. So it was finally replaced by a longer one. The longer cable seemed it increased the overall impedance of the measuring circuit. The initial cable had a length of 7 m, while the long cable had a length of 55 m. To evaluate this behavior a similar process to Section 4.1 was followed, this time exclusively inside the basement corridor, with a short and a long cable. Table 2 presents the relative SPL values of the gated impulse response in the two cases. The microphone is placed at 1 m from the source. The bottom line shows a frequency dependent drop ranging between about 3 dB and 4 dB, clearly caused by the impedance added by the cable. In Figure 5 the early part of the impulse response is shown for the octave bands 500 Hz and 1000 Hz, when using a short and a long cable. Careful inspection reveals a few phase shifts, which can be neglected. It can be assumed that the overall waveform does not change significantly and that the change in the cable affects only the amplitude. The rest octave bands showed an even better behaviour, with almost no phase shift in the waveform.

Table 2: Relative SPL levels (dB) of five measurements at 1 m distance and fixed direction from the source, together with their average, using a short and a long cable connecting the source to the amplifier. Gain settings are the same in both cases.

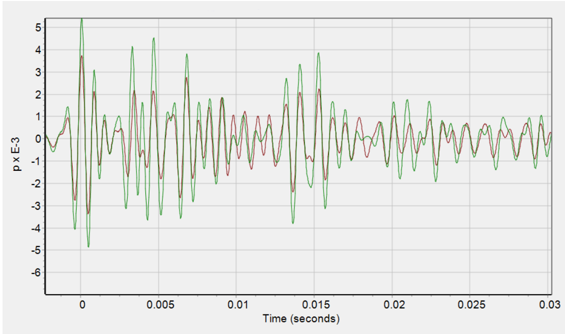
	63	125	250	500	1000	2000	4000	8000
Short cable								
Meas. 1	-37.40	-31.34	-26.48	-28.59	-29.61	-22.52	-19.56	-18.26
Meas. 2	-37.08	-31.19	-26.46	-28.65	-29.70	-22.51	-19.57	-18.20
Meas. 3	-37.28	-31.27	-26.46	-28.61	-29.70	-22.48	-19.58	-18.27
Meas. 4	-37.11	-31.12	-26.33	-28.49	-29.21	-23.55	-19.71	-18.46
Meas. 5	-37.28	-31.22	-26.37	-28.48	-29.26	-23.18	-19.60	-18.52
Average	**-37.23**	**-31.22**	**-26.42**	**-28.56**	**-29.49**	**-22.82**	**-19.6**	**-18.34**
STD	**0.12**	**0.07**	**0.06**	**0.07**	**0.22**	**0.44**	**0.05**	**0.12**
Long cable								
Meas. 1	-39.93	-34.35	-30.00	-32.49	-33.00	-26.74	-22.67	-21.25
Meas. 2	-40.67	-34.78	-30.12	-32.43	-33.01	-26.72	-22.60	-21.10
Meas. 3	-40.87	-34.88	-30.15	-32.43	-33.08	-26.56	-22.59	-21.11
Meas. 4	-40.34	-34.53	-29.99	-32.36	-32.78	-26.88	-22.63	-21.11
Meas. 5	-40.61	-34.77	-30.14	-32.51	-33.10	-26.53	-22.56	-21.04
Average	**-40.47**	**-34.66**	**-30.08**	**-32.45**	**-32.99**	**-26.69**	**-22.61**	**-21.12**
STD	**0.32**	**0.19**	**0.07**	**0.05**	**0.11**	**0.13**	**0.04**	**0.07**
Difference between averages	**-3.24**	**-3.44**	**-3.66**	**-3.89**	**-3.5**	**-3.87**	**-3.01**	**-2.78**

4.3 Evaluation of *G* Parameter Derived in Different Gain Settings

To finally evaluate the reliability of measured G values using the ODEON measurement system with *in-situ* recalibration, two independent set of measurements were carried out using different gain settings inside the basement corridor of Figure 4. The measurements were carried out on two different days and the hole process of assembling equipment, positioning source and microphone at correct coordinates were repeated by two different persons. On the first day the gain of the amplifier was the same as in the reverberation chamber (SG), on the second day the gain of the amplifier was increased (IG), corresponding to a practical scenario where the operator intentionally increases the output level



(a) 500 Hz



(b) 1000 Hz

Figure 5: The early part of impulse response between the source and the microphone at 1m distance when using a short (green) and a long (red) cable.

to sufficiently drive a large room (case 2 in Section 1). The results are presented in Table 3 and 4. For each of the eight octave bands the G parameter for same gain and increased gain is shown at each receiver position. The differences for all positions are calculated and averaged. In addition, one more average value is derived, calculated by the absolute differences. This shows the degree of error in units of Just Noticeable Difference (JND). It can be seen that with the exemption of the lowest octave band (63 Hz) all average differences are well below 1 JND (= 1 dB for SPL and G).

Table 3: Calibrated G values (dB) *in-situ* for same gain (SG) and increased gain (IG) settings. The octave bands from 63 Hz to 500 Hz are shown.

	63 Hz			**125 Hz**			**250 Hz**			**500 Hz**		
	SG	IG	Diff.	SG	IG	Diff.	SG	IG	Diff.	SG	IG	Diff.
Pos1	27.9	31.7	3.8	27.1	27.1	0.0	26.7	25.7	-1.0	24.6	24.8	0.2
Pos2	27.7	30.6	2.9	25.1	25.6	0.5	23.7	23.7	0.0	23.6	23.0	-0.6
Pos3	28.7	31.0	2.3	25.7	25.8	0.1	24.0	23.5	-0.5	22.0	21.2	-0.8
Pos4	29.4	32.8	3.4	26.7	27.9	1.2	25.7	26.2	0.5	21.5	22.1	0.6
Pos5	20.6	20.3	-0.3	18.9	18.8	-0.1	19.5	20	0.5	18.2	18.1	-0.1
Pos6	21.9	21.3	-0.6	20.3	19.9	-0.4	20.0	20.2	0.2	17.8	18.3	0.5
Average			**1.9**			**0.2**			**-0.1**			**0.0**
JND Average			**2.2**			**0.4**			**0.5**			**0.5**

Table 4: Calibrated G values (dB) *in-situ* for same gain (SG) and increased gain (IG) settings. The octave bands from 1000 Hz to 8000 Hz are shown.

	1000 Hz			**2000 Hz**			**4000 Hz**			**8000 Hz**		
	SG	IG	Diff.	SG	IG	Diff.	SG	IG	Diff.	SG	IG	Diff.
Pos1	22.9	23.2	0.3	22.2	23.2	1.0	22.2	22.6	0.4	21.4	22.3	0.9
Pos2	21.5	21.8	0.3	20.1	20.8	0.7	20.2	20.4	0.2	19.0	19.3	0.3
Pos3	20.8	20.9	0.1	19.1	19.5	0.4	19.0	18.8	-0.2	17.0	17.1	0.1
Pos4	20.1	21.2	1.1	18.0	18.7	0.7	18.4	18.6	0.2	16.7	16.9	0.2
Pos5	17.0	16.7	-0.3	15.8	15.6	-0.2	14.9	14.9	0.0	12.5	12.5	0.0
Pos6	16.9	17.0	0.1	14.9	15.3	0.4	14.0	14.1	0.1	11.3	11.9	0.6
Average			**0.3**			**0.5**			**0.1**			**0.4**
JND Average			**0.4**			**0.6**			**0.2**			**0.4**

5 CONCLUSIONS

A Two-step calibration method for measurement of the G parameter has been proposed. The method adds an extra measurement step to the existing calibration methods: diffuse-field and free-field. For this paper the first step is performed in a reverberation chamber (diffuse-field method). The second step is performed both in the reverberation chamber and *in-situ* and allows frequency dependent compensation for common changes to the measurement chain; such as cables, gain of amplifiers, gain settings in the operating system and even drifting of loudspeaker, change of audio interface etc. It is demonstrated that measurements of G can be repeated in a room on different days; with the uncertainty of positioning source and receivers at the original locations, and with a changed gain setting. The method yielded errors smaller than 0.6 JND in the frequency range 125 Hz - 4000 Hz, where values for the second step were based on 5 measurements.

When using just one measurement for the second calibration step the error can be greater than 1 JND. It is suggested that one measurement is sufficient for this step if a mounting device is attached on the loudspeaker ensuring that the second-step reference measurements are always performed at exactly the same distance and orientation. In this research no compensation was taken into account for changes in relative humidity and temperature during the measurements in the reverberation chamber and *in-situ*. Although the environmental conditions were very stable during the measurements for this paper, such factors may affect the results significantly. This is left as a further research for the calibration methods proposed by Odeon A/S.

REFERENCES

1. L.L. Beranek: Concert Halls and Opera Houses - Music, Acoustics and Architecture. 2nd Edition, Springer, New York, 2004. (pp. 512).

2. L.L. Beranek and N. Nishihara: Mean-free-paths in concert halls and chamber music halls and the correct method for calibrating dodecahedral sound sources. J. Acoust. Soc. Am 135 (1) 2014, pp. 223-230.

3. B. F. G. Katz: In situ calibration of the sound strength parameter G, JASA Express Letters, J. Acoust. Soc. Am 138 (2) 2015, EL 167-173.

4. C. C. J. M. Hak, R. H. C. Wenmaekers, J. P. M. Hak, L. C. J. van Luxemburg and A. C. Gade: Sound Strength Calibration Methods, International Congress on Acoustics, ICA, Sydney, Australia. August 2010.

5. ISO 3382-1: Acoustics - Measurement of room acoustic parameters - Part 1: Performance spaces, 2009.

6. M. P. M. Luykx and M. L. S. Vercammen: Evaluation and measuring procedure for strength in sports halls. Proceedings of Euronoise 2012, Prague, pp. 42-47.

7. IEC 60942: Electroacoustics - Sound calibrators, 2003.

8. ISO 3741: Acoustics - Determination of sound power levels and sound energy levels of noise sources using sound pressure - Precision methods for reverberation test rooms, 2010.

9. C. L. Christensen, G. Koutsouris and J. H. Rindel: The ISO 3382 parameters. Can we measure them? Can we simulate them?, International Symposium on Room Acoustics, Toronto, Canada. June 2013.

10. C. L. Christensen, G. Koutsouris and J. Gil: ODEON Room Acoustics Software, Version 13, User's Manual, Denmark, 2015.