ODEON APPLICATION NOTE
Impulse Response Measurement Calibrations

GK, CLC – revised in October 2020

Contents

1. Scope

Most of the room acoustic parameters provided by ODEON do not require calibration. This is because they depend exclusively on the slope and shape of the energy decay, rather than the actual level. However, *Sound Strength* (G) and *Speech Transmission Index* (STI, STI_{male} and STI_{female}) depend on the level, therefore they require that the ODEON measurement system is calibrated. You can read more about the measurement system in general in the ODEON Manual[1].

In this application note we will explain how to use the two available methods for G and STI calibration: The *Diffuse-field* method and the *Free-field* method. Since both methods are quite sensitive in gain or equipment changes, we have also developed a *two-step* correction method, which simply adds an extra level of security to the process, so that changes in gain or equipment do not ruin the calibration.

2. The G parameter

According to the ISO standard 3382-1, for performance spaces[2], the sound strength G of an omni-directional source, for a specific frequency, is given as the logarithmic ratio of the sound energy (squared and integrated sound pressure) of the measured impulse response to that of the impulse response measured in free field, at 10 m distance from the source:

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

or

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

where $L_{pE} = 10\log_{10}\left[\frac{1}{T_0}\int_0^\infty \frac{p^2\,(t)dt}{p_0^2}\right]$ and $L_{pE,10} = 10\log_{10}\left[\frac{1}{T_0}\int_0^\infty \frac{p_{10}^2(t)dt}{p_0^2}\right]$ are the sound pressure exposure level of $p(t)$ and $p_{10}(t)$ respectively.

The variables in these equations are as follows:

- $p(t)$: Instantaneous sound pressure of the impulse response at the receiver's position.
- $p_{10}(t)$: Instantaneous sound pressure of the impulse response at 10 meters distance from the source in free field.
- p_0: Reference sound pressure, 20µPa.
- T_0: 1 sec averaging time.

According to equation (2), it can be seen that G is a relative parameter, calculated by measuring the absolute values of of L_{pE} and $L_{pE,10}$. However, measuring L_{pE} and $L_{pE,10}$ would only be possible if our system was calibrated with a pistonphone (usually providing 94 dB at 1kHz), so that we could measure absolute SPL values.

In ODEON this is not possible - but even if it was, we should still find a large perfect free-field and locate the source and microphone 10m apart. This is an impractically long distance. Fortunately, we can apply some workarounds in order to calculate the difference between L_{pE} and $L_{pE,10}$ directly, without knowing the absolute values of the individual terms. This technique, which is described in detail in ISO 3382-1, makes use of two well-defined environments: The **reverberation** and **anechoic** chambers. These two environments provide the perfect conditions for the **diffuse-field** and **free-field** method, respectively.

1. Calculation of $L_{pE,10}$ in a Reverberation Chamber

To calculate $L_{pE,10}$ inside a reverberation chamber, we make use of the theory about power estimation in *diffuse-field*. According to this, the power level L_w of an unknown source at a specific frequency can be calculated from the measured average sound pressure level L_{pE}^{RevCh} inside a reverberation chamber:

$$L_w = L_{pE}^{RevCh} - 6 + 10\log_{10}\frac{A}{S_0} \quad \text{dB(re } 10^{-12} \text{ Watt)} \tag{3}$$

where A is the equivalent absorption area in the room and $S_0 = 1\ \mathrm{m}^2$.

The sound pressure level 10 meters away from this unknown source in free field conditions is calculated according to the spherical spread law:

$$\begin{aligned} L_{pE,10} &= L_w - 11 - 10\log_{10}[(10m)^2] \\ &= L_w - 31 \ \text{ dB} \end{aligned} \tag{4}$$

An interesting observation is that a sound source of 31 dB power level provides sound pressure level of 0 dB at 10m.

Substituting equation (3) to (4) leads to:

$$L_{pE,10} = 10\log_{10}\frac{A}{S_0} - 37 + L_{pE}^{RevCh} \quad \text{dB} \tag{5}$$

The absorption area can be calculated from the Sabine's formula (diffuse field assumption): $A = 0.16V/T$, with V being the volume (m^2) and T the reverberation time (sec) of the reverberation chamber. Substituting this and $S_0 = 1\ \mathrm{m}^2$ in equation (5) we get:

$$L_{pE,10} = 10\log_{10}\frac{0.16V}{T} - 37 + L_{pE}^{RevCh} \quad \text{dB} \tag{6}$$

L_{pE}^{RevCh} is the spatial- averaged sound pressure exposure level as measured in the reverberation chamber.

If we subtract both sides of the equation from the term L_{pE} we get:

$$L_{pE} - L_{pE,10} = L_{pE} + 37 - L_{pE}^{RevCh} - 10\log_{10}\frac{0.16V}{T} \quad \text{dB}$$

which provides the main formula for calculation of *G* using a reverberation chamber:

$$\boxed{G = L_{pE} + 37 - L_{pE}^{RevCh} - 10\log_{10}\frac{0.16V}{T} \ \text{dB}} \tag{7}$$

Eq. (7) says that G can be calculated just by measuring the ***relative*** sound pressure levels at the reverberation chamber L_{pE}^{RevCh} and the room under measurement, L_{pE}. The term ***relative*** is used to emphasise that no absolute sound pressure levels are needed. In other words, as long as the equipment stays the same in the reverberation chamber and the room, only the difference between the terms L_{pE} and L_{pE}^{RevCh} is important – not their absolute values. This means that even negative values of L_{pE} or L_{pE}^{RevCh} are still valid for our calibration.

For use in ODEON we can write eq. (7) as:

$$G = L_{pE} + \Delta L^{RevCh} \ \text{dB} \tag{8}$$

where ΔL^{RevCh} is the calibration correction factor that ODEON applies to all L_{pE} measurements:

$$\Delta L^{RevCh} = 37 - L_{pE}^{RevCh} - 10\log_{10}\frac{0.16V}{T} \ \text{dB} \tag{9}$$

2. Calculation of $L_{pE,10}$ in an Anechoic Chamber

To calculate $L_{pE,10}$ inside an anechoic chamber, we make use of the theory about power estimation in *free field*. According to this, the power level L_w of an unknown source at a specific frequency can be calculated from the average sound pressure level L_{pE}^{Anech}, measured at a specific radius around the source. Assuming spherical propagation of sound, the power level is calculated from the following formula:

$$\begin{aligned} L_w &= L_{pE}^{Anech} + 10\log_{10}(4\pi) + 10\log_{10}(d^2) \quad \text{dB(re } 10^{-12} \text{ Watt)} \\ &= L_{pE}^{Anech} + 11 + 10\log_{10}(d^2) \quad \text{dB(re } 10^{-12} \text{ Watt)} \end{aligned} \tag{10}$$

Where d is the distance (radius) from the source (preferably $\geq 3\text{m}$) and L_{pE}^{Anech} is the spatial- averaged sound pressure exposure level at every 12.5° around the source. At 10 meters from the unknown source the sound pressure level becomes:

$$\begin{aligned} L_{pE,10} &= L_w - 11 - 10\log_{10}[(10m)^2] \\ &= L_w - 31 \ \text{dB} \end{aligned} \tag{11}$$

The above implies that if the sound source had a known power level of 31 dB, then the sound pressure level at 10 m would be 0 dB. Substituting equation (10) into (11) leads to:

$$L_{pE,10} = L_{pE}^{Anech} + 20\log_{10}(d) - 20 \ \text{dB} \tag{12}$$

If we subtract both sides of the equation from the term L_{pE} we get:

$$L_{pE} - L_{pE,10} = L_{pE} - L_{pE}^{Anech} - 20\log_{10} + 20 \quad \text{dB}$$

which provides the main formula for calculation of *G* using an anechoic chamber:

$$\boxed{G = L_{pE} - L_{pE}^{Anech} - 20\log_{10}(d) + 20 \quad \text{dB}} \tag{13}$$

Eq. (13) says that G can be calculated just by measuring the ***relative*** sound pressure levels at the at the anechoic room L_{pE}^{Anech} and the room under measurement, L_{pE}. The term ***relative*** is used to emphasise that no absolute sound pressure levels are needed. In other words, as long as the equipment stays the same in the anechoic chamber and the room, only the difference between the terms L_{pE} and L_{pE}^{Anech} is important – not their absolute values. This means that even negative values of L_{pE} or L_{pE}^{Anech} are still valid for our calibration. For use in ODEON we can write eq. (13) as:

$$G = L_{pE} + \Delta L^{Anech} \quad \text{dB} \tag{14}$$

where ΔL^{Anech} is the calibration factor that ODEON applies to all L_{pE} measurements:

$$\Delta L^{Anech} = -L_{pE}^{Anech} - 20\log_{10}(d) + 20 \quad \text{dB} \tag{15}$$

3. General remarks

In simple words, the whole calibration process means that ODEON derives the sound pressure level for a receiver as if the source was an omni-directional source of power level 31 dB/Octave band.
Since version 16 of ODEON, the *G* value is always displayed in measured (and simulated) results. ODEON automatically calculates G and its variations: G_{Early} (taking into account the energy that arrives within 80ms after the direct sound) and G_{Late} (taking into account the energy that arrives later than 80ms, after the direct sound).

Please note: If calibration has not been applied to the measurement files, then the *G* values displayed in ODEON make no sense.

3. Equipment

1. Space

For the *diffuse-field* calibration method a highly reverberant and diffuse room is needed. The ideal type is a *reverberation chamber*. If such a room is not available, a room with hard walls could be acceptable (eg. a big garage or hall). However, the chosen room should be as diffuse as possible meaning that highly symmetric shapes (eg. rectangular) must be avoided and some extra treatment might be required: placing of hard, scattering objects around the walls or the floor. It is a good practice to check always for the degree of linearity of the decay curve in each octave band through one of the *XI* parameters. If *XI* exceeds 10 ‰

this is an indication that the decay is strongly non-linear, because of flutter echo or due to impulsive noise during the measurement. You can read more about the *XI* parameter in the ODEON help file, which can be easily accessed by pressing F1 while working with the measuring system. Another way to check diffusivity practically, is to take a few measurements at random positions and see if the difference between the minimum and the maximum reverberation time does not vary more than 10% at a particular octave band.

For the *free-field* calibration method a very dry room is needed. The ideal type is *an anechoic chamber*. If such a room is not available, a small room with carpets and curtains could be used, since ODEON is able to isolate the direct sound from the subsequent reflections with high success at medium and high frequencies. At low frequencies the isolation is less successful because sound reflections overlap, due to the large wavelengths. A good room for the free-field calibration should have a reverberation time of less than 0.3 sec at different positions.

2. Hardware and Software

The same equipment used in normal sweep-signal measurements will be used for calibration as well. That is:

1) An omni-directional loudspeaker.
2) An omni-directional microphone.
3) Amplifier for the speaker (if a *passive* one is used)
4) Pre-amplifier if a dynamic microphone is used or phantom power – supplied from the audio interface - if a condenser microphone is used.
5) Audio interface (preferably an external sound card and not the built-in into the PC).
6) Lap-top PC.
7) ODEON 13 and later (any edition).
8) Ear protectors (levels in the free-field calibration will most of the time be harmless as there are no reflections in an anechoic chamber, while levels in the diffuse-field calibration can be substantially high).

For a comprehensive overview of the measuring system in ODEON and full guidance to the equipment setup, refer to the User’s manual in Chapter 12.

4. Measurement setup

Start ODEON and open the Options>Program Setup>Measurement Setup window. Keep the settings under Parameters for detection of onset time as they are.For G-parameter calibrations, it is not necessary to set a specific Source Power

Figure 1: In the measurement setup the source power spectrum, the background noise and other parameters can be specified.

Spectrum. ODEON 16 and later will always provide G as a separate parameter, taking automatically into account a source of power spectrum 31 dB/Octave. However, the system needs to be calibrated. For G-parameter, no background noise is taken into account.

Validation exercise: As mentioned already in Sec. 2, the calculation of *G* is based on the assumption that the source power spectrum is 31 dB/Octave band. If you manually insert that power spectrum in Options>Program Setup>Measurement Setup, then the SPL and G values will become identical, because *G* is basically SPL for an omni-directional source of 31 dB/Octave band (Figure 2). In other words, at 10m from the source $L_{pE,10} = 0\ dB$ and therefore from Eq.(2) we have $G = L_{pE}$.

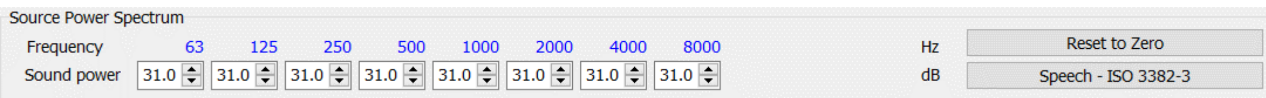


Figure 2: The power spectrum in Options>Program Setup>Measurement Setup affects the *SPL* shown in a measurement file. If the Sound power in manually set to 31.0 dB, the *SPL* becomes identical to the automatically calculated *G*.

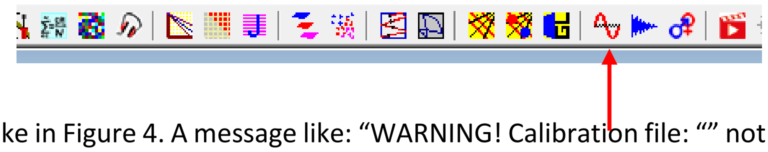
With the setting above the parameters for a receiver become:

SPL	(dB)	23.3	25.5	22.9	25.5	26.9	25.8	23.4	18.6
G	(dB)	23.3	25.5	22.9	25.5	26.9	25.8	23.4	18.6
G(early)	(dB)	23.0	25.0	21.5	22.9	23.4	22.3	21.0	17.4
G(late)	(dB)	12.5	15.8	17.4	22.1	24.2	23.2	19.7	12.4

Figure 3: When the sound power of an omni source is set to 31 dB /Octave power level, then the G value coincides with the SPL value.

1. Set the sweep parameters

Open the Measure impulse response (sinusoidal sweep) interface from the main toolbar:



The interface looks like in Figure 4. A message like: “WARNING! Calibration file: “” not found. No calibration applied.” may be displayed at the bottom of the window but it can be ignored at this phase.

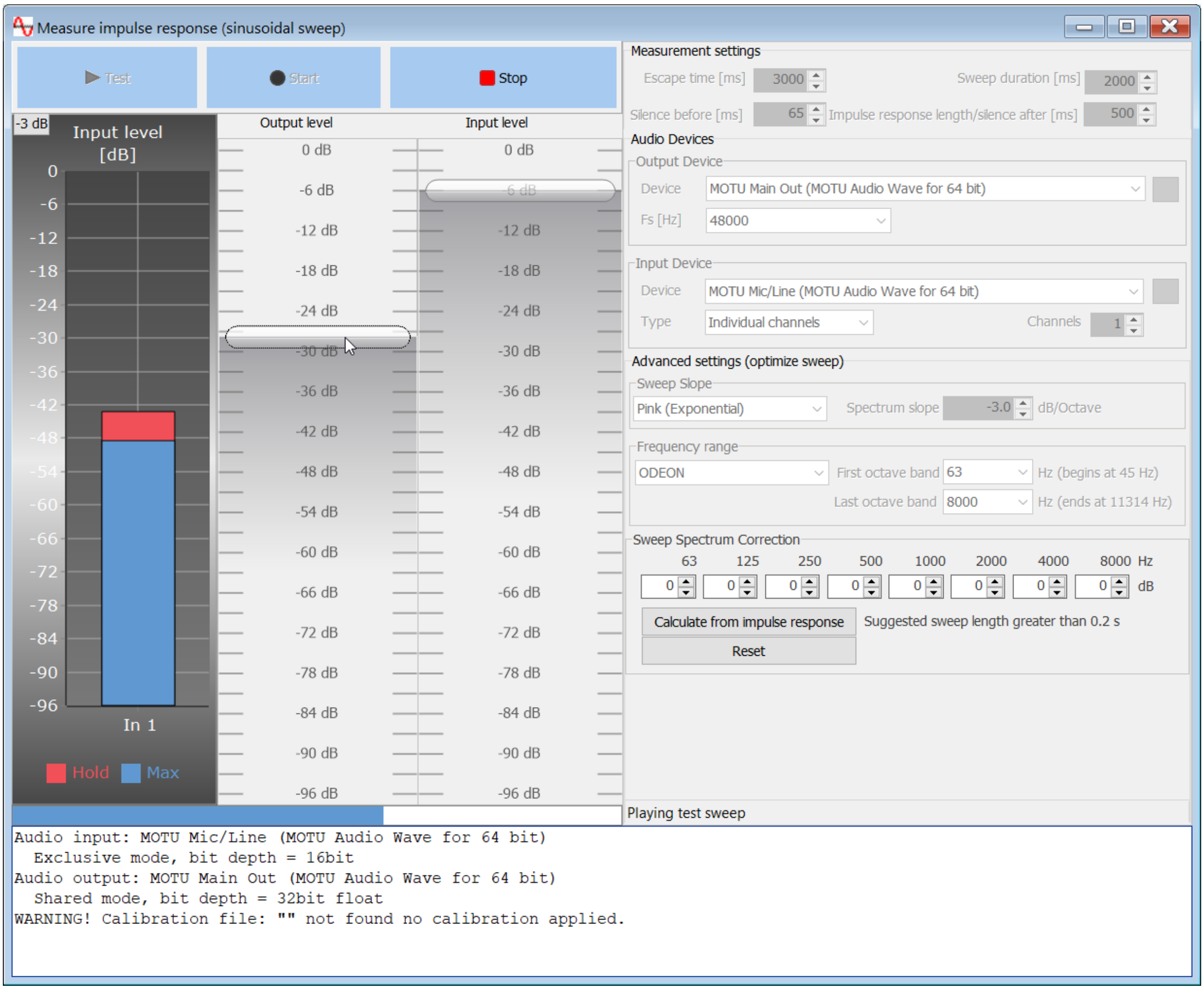


Figure 4: The Measure Impulse response interface, as it looks in ODEON 16.

Measurement settings

In the *Measurement Settings* panel, the sweep duration and the estimated impulse response length can be adjusted, together with other minor parameters. The longer is the sweep, the higher the suppression of the background noise in the room. <u>A 3 dB suppression is achieved for a doubling of the sweep length</u>.

Audio devices

In the *Audio devices* panel the input and output devices should be chosen. According to Section 3, an omni-directional **microphone** and omni-directional **speaker** are needed as input/output devices. These should preferably be connected to the PC via an external high-quality audio interface. In the example of Figure 4, a **MOTU 4 Pre Hybrid** audio interface is connected to the USB port in the PC, while the speaker and microphone are connected to the interface. This means that the input and output devices available in the ODEON drop-down lists, are the MOTU Mic and Out lines respectively.

Advanced settings

In this panel keep the default *Pink* (Exponential) Sweep type, which is the most suitable setting for room acoustic applications. Choose the desired Frequency range from the drop down. The ISO 3382-1 is the most relevant in this application note (125 to 4000 Hz), since the G and STI parameters are defined in this standard. However, you are always free to derive the parameters for the extreme 63 and 8000 Hz bands.

For this application, we completely ignore the Sweep Spectrum Correction panel. The Test button plays the sweep signal without recording any measurement. Click on the Stop button to terminate the test. The sliders adjust the **internal Output/Input levels**. Any other gain adjustments in the measuring equipment (windows mixer, audio interface, amplifiers) belong to **external gains**. To perform an actual measurement, click on the Start button.

2. Adjusting the levels

The most important requirement for a calibrated measurement is that all external level (gain) adjustments must be set to fixed values during the calibration and the measurement. **Only the internal Output/Input levels can be freely changed between recordings for calibration or measurement, since ODEON itself compensates for the adjustment.** During a recording ODEON makes sure the internal Output/input levels are locked.

It is crucial at this point to decide which should be the value of the **external levels** during the measurement in order to drive the room with an adequate signal to noise ratio without overloading. If overloading occurs, ODEON automatically cancels the measurement. The best practice is to maximize the internal levels for most recordings, so that if a few of them lead to overloading, you can freely reduce these internal levels without harming the calibration settings. The most straightforward way of adjusting the external levels is to maximize all values: Windows volume can be set to 100% and amplifier knobs can be turned to the maximum value (Figure 3). Then all settings are easy to remember/replicate during the measurement).

<u>However, one should be careful not to overload and damage the speaker</u>! High output gains are also likely to cause distortion. In addition, the gain of the microphone should be set to a level where the internal noise does not become very high.
<u>Often the amplifiers used are rather powerful for the speakers used so that a much lower gain is sufficient.</u> The corresponding value must then be clearly noted down for reference. Apart from that, no extreme sound power output is needed if the equipment is able to derive a sufficient signal to noise ratio (SPL/Noise parameter inside ODEON) or decay range (eg. 45 dB), in normal room conditions (with moderate ambient noise).

Exclusive mode

The WASAPI audio driver that ODEON uses in the measuring system, allows the *exclusive mode* to be enabled for the input and output devices. This means that:

1) The devices cannot be used by other applications at the same time, something that can guarantee that the measurement is not interrupted – for example by an incoming video call.
2) The Windows levels of the devices are locked to the internal levels in ODEON. In other words, when you move the sliders in the ODEON's Measured Impulse response interface, the windows sliders move as well, at the same amount. For example, if the ODEON sliders are maximized, the window sliders will maximize too, etc.

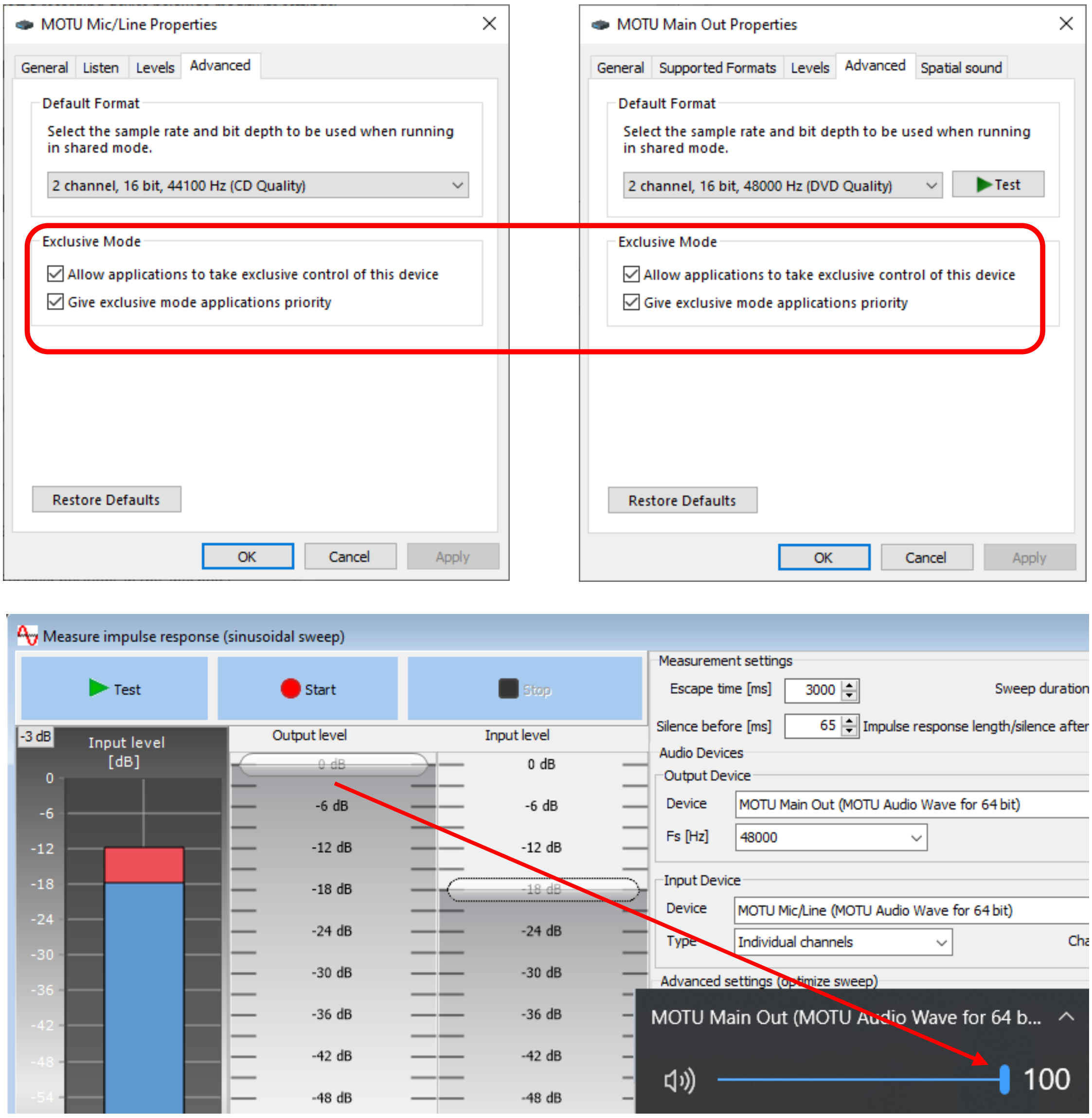
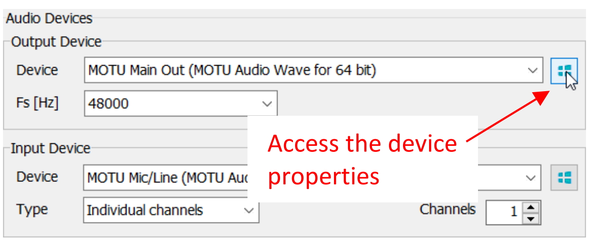


Figure 4: Exclusive mode ensures levels in ODEON and windows move together and no other applications use the audio devices.

Sometimes, exclusive mode can cause conflicts with ODEON and the audio interface, so that several input/output devices cannot be selected from the drop-downs or the level sliders cannot be moved as desired. In such a case, **deactivate Exclusive mode** by unchecking both boxes related to *Exclusive mode* in



the Advanced tab sheets of the devices' properties (see Fig.4). The properties panel can be accessed from within the Measure Impulse Response window in ODEON. Click the button on the right of the Input or Output device.

3. Recording

After fixing the external level adjustments, recordings of impulse responses in the anechoic/dry room can be performed by clicking the Start button in the Measure Impulse Response window. An introductory video on how to obtain measurements in a room can be found on [6]. For a complete guidance, please refer to [1]. The recordings are saved as .WAV files. Some trial measurements should be performed initially by varying the internal levels until an adequate Signal to Noise Ratio (SNR) is achieved. It can be desirable to make the measurement with the highest possible SNR, achievable without overload, but this is not required. **NOTE**: Use ear protectors to prevent hearing damage!

An indication of the quality of the SNR can be obtained by loading the recording on the Load impulse response tool, (under the Tools menu). After each measurement, ODEON loads the impulse response automatically. The SNR is important only in the *diffuse-field* calibration, where a decay range of at least 40 dB is desired. In the *free-field* calibration the impulse response is so short, so that no value of the decay range can be displayed.

5. Making a *Diffuse-field* calibration

In this section we will follow the diffuse-field calibration process step by step, using one of the *reverberation chambers* at the Technical University of Denmark and *Auditorium 21* at the same campus. See more details at the picture description below. You can test the process yourself with the measurement files available in the Measurements\Calibration\Diffuse-field method and Measurements\Calibration\Auditorium 21 folders which come with the ODEON installation.

The diffuse-field calibration should be done ideally in a reverberation chamber. For this application note, the **Large Reverberation Chamber** at the Technical University of Denmark has been used. The volume of the chamber is 245 m^3 and provides a reverberation time of about 8 sec at low frequencies. Reflective panels are placed near the walls at various angles in order to distribute reflections evenly and enhance the diffusivity of the sound field. Even when another type of room is used for the calibration, a diffuse sound field should be achieved, utilizing some scattering surfaces or objects. A perfect, rectangular room with hard walls would lead to flutter echoes (visible in the impulse response) that would violate the assumptions of linear decay, taken for the diffuse-field calibration.

1. Making sure all external levels are fixed

The most important step in the calibration process is to decide on all external gains before starting (input/output volume in soundcard, gains in the amplifiers etc.). Also make sure all your connections in the equipment use the **same cables**, as each cable has its own impedance and this can affect the input/output levels.

Both levels and cables have to remain the same between the Reverberation Chamber and the actual room under measurement.

2. Taking measurements in the *Reverberation Chamber*

Following the guidelines of the previous section we will measure the impulse response at 2 source and 3 receiver positions, which gives 6 combinations. After saving each measurement, ODEON loads the result automatically. An example of a successful impulse response in the reverberation chamber is given in Figure 5, where all parameters have been derived. The **sound strength parameter (*G*)** is has some abstract value, as the system is not calibrated yet. In the *diffuse-field calibration* you should always check that all **Reverberation** and **SPL** values have been derived at all octave bands. A "*" character means that ODEON was unable to derive the corresponding value from the impulse response (eg. due to insufficient signal to noise ratio/ decay range). In such a case, you have to change the sweep parameters, as described in the paragraph 'Set the sweep parameters' in Section 4.

You can find an example of measurements inside the reverberation chamber in the Measurements\Calibration\Diffuse-field method folder.

Measured response - E:\...\1stJuly2015\CalibrationReverbChamber\Dodecahedron\S1R6.wav

Frequency Response | Cepstrum | Spectrogram | Play it!
Raw Impulse response at 1000Hz | Raw decay curve at 1000Hz | Decay curves at 1000Hz T(30) = 4.69 seconds | Decay curves (all bands) | Energy parameters | Parameter bars

Band (Hz)		63	125	250	500	1000	2000	4000	8000
EDT	(s)	7.58	6.40	6.70	5.46	4.38	3.51	2.19	1.40
T(15)	(s)	7.50	7.47	6.87	5.86	4.82	3.59	2.42	1.49
T(20)	(s)	7.71	7.52	6.91	5.88	4.73	3.64	2.46	1.54
XI(T(20))	(‰)	2.1	2.3	0.9	0.4	0.6	0.6	0.7	1.1
T(30)	(s)	7.84	7.41	6.97	5.91	4.69	3.66	2.59	1.63
XI(T(30))	(‰)	0.9	1.1	0.4	0.4	0.5	0.3	1.5	1.7
Curvature(C)	(%)	1.7	-1.6	0.9	0.6	-0.8	0.4	5.3	5.9
Ts	(ms)	547	479	473	390	310	244	143	87
SPL	(dB)	[illegible]	20.2	17.7	22.3	[illegible]	20.5	22.2	24.4
G	(dB)	-8.8	10.8	13.3	8.7	8.2	10.5	8.8	6.6
G(early)	(dB)	12.4	5.5	7.7	[illegible]	[illegible]	[illegible]	5.7	[illegible]
G(late)	(dB)	-11.3	9.3	12.0	7.4	6.8	8.5	5.8	2.1
SPL(Af)	(dB)	-66.0	-36.4	-26.3	-25.6	-22.8	-19.3	-21.3	-25.6
SPL(Direct)	(dB)	-51.1	-31.0	-30.7	-37.3	-37.2	-32.6	-33.0	-36.7
D(50)		0.34	0.24	0.20	0.20	0.20	0.29	0.41	0.51
C(7)	(dB)	-7.4	-6.8	-8.6	-10.7	-10.3	-8.3	-6.2	-5.4
C(50)	(dB)	-2.8	-5.0	-5.9	-6.1	-6.0	-3.9	-1.6	0.1
C(80)	(dB)	-1.0	-3.9	-4.3	-4.6	-4.0	-2.4	0.0	2.5
U(50)	(dB)	-75.4	-57.3	-55.6	-60.4	-60.7	-56.9	-57.1	-58.4
U(80)	(dB)	-74.4	-56.5	-54.3	-59.2	-59.2	-55.9	-56.3	-57.4
MTI(corrected)		*.**	0.00	0.00	0.00	0.00	0.00	0.00	0.00
Echo(Dietsch)		0.78	0.82	0.73	0.64	0.57	0.54	0.49	0.48
Source power	(dB)	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00
Background noise	(dB)	31.00	31.00	31.00	31.00	31.00	31.00	31.00	31.00
L/Noise	(dB)	92.77	108.14	106.36	99.21	*.**	100.63	100.97	95.35
Peak/Noise	(dB)	49.38	72.03	72.29	67.29	*.**	77.39	81.54	78.97
Decay range	(dB)	42.47	60.59	60.27	54.27	56.72	58.23	59.46	56.27

G is not calibrated yet.

SPL(A)	-14.9	dB
SPL(Lin)	-12.5	dB
SPL(C)	-12.8	dB
SPL(A_Direct)	-27.1	dB
STI	0.00	
STI(Female)	0.00	

Figure 5: Room acoustic parameters from a healthy impulse response recording inside the reverberation chamber. A sufficient decay range for each octave band makes sure that all parameters are derived.

3. Performing the calibration

Once all measurements in the *reverberation chamber* have been completed, follow the next steps to perform the calibration:

1) Click on Tools>Measurement Calibration>Diffuse field>One step (requires fixed signal chain). ODEON asks for the volume of the reverberation chamber and then for the impulse responses. Select the files all together using the SHIFT or CTRL key and open them at once.

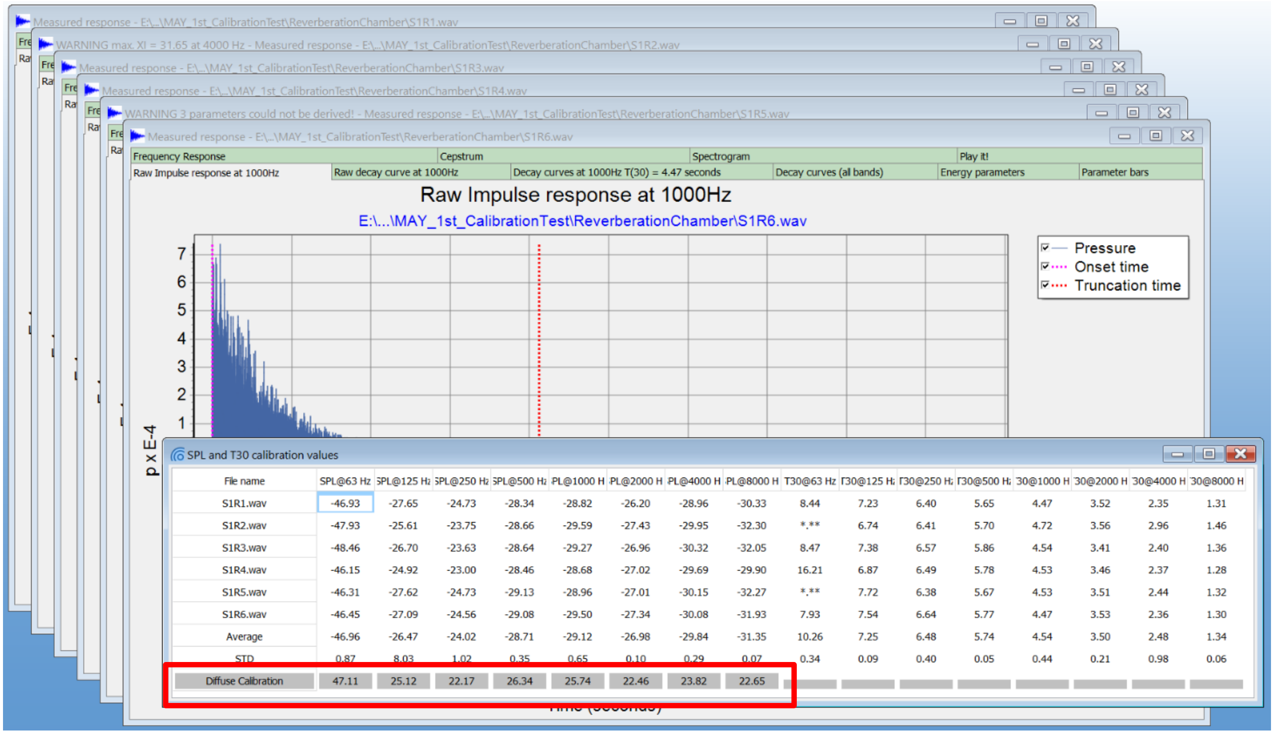
2) ODEON then uses the SPL and T_30 parameters to derive the calibration values for the eight octave bands between 63 Hz and 8000 Hz, as an average from the different positions (

Figure 6). The calibration values ΔL^{RevCh} are displayed at the bottom of table are calculated by eq.(9).

3) After all files all loaded, a dialog pops up and asks for a name of the calibration file. You are prompted to set the file as the active calibration file in the Measurement Setup, which is going to be used with the following measurements.

4) It is possible to change the active calibration file at a later point in the Measurement Setup. However, if the input and output devices are different from those used during the generation of the calibration file, ODEON gives a warning for device mismatch, meaning that all level dependent parameters will not be calibrated.

5) You can now launch the Measurement Setup window to check whether the active calibration file is the correct one.



Calibration values to be added on the SPL measurement, in situ.

Figure 6: Calibration values (factors) derived from 6 measurements inside the Reverberation Chamber. The values are calculated according to eq.(9) and will be added in the SPL measurements in the real room, in order to derive the G values (see eq.(8)).

4. Performing calibrated measurements

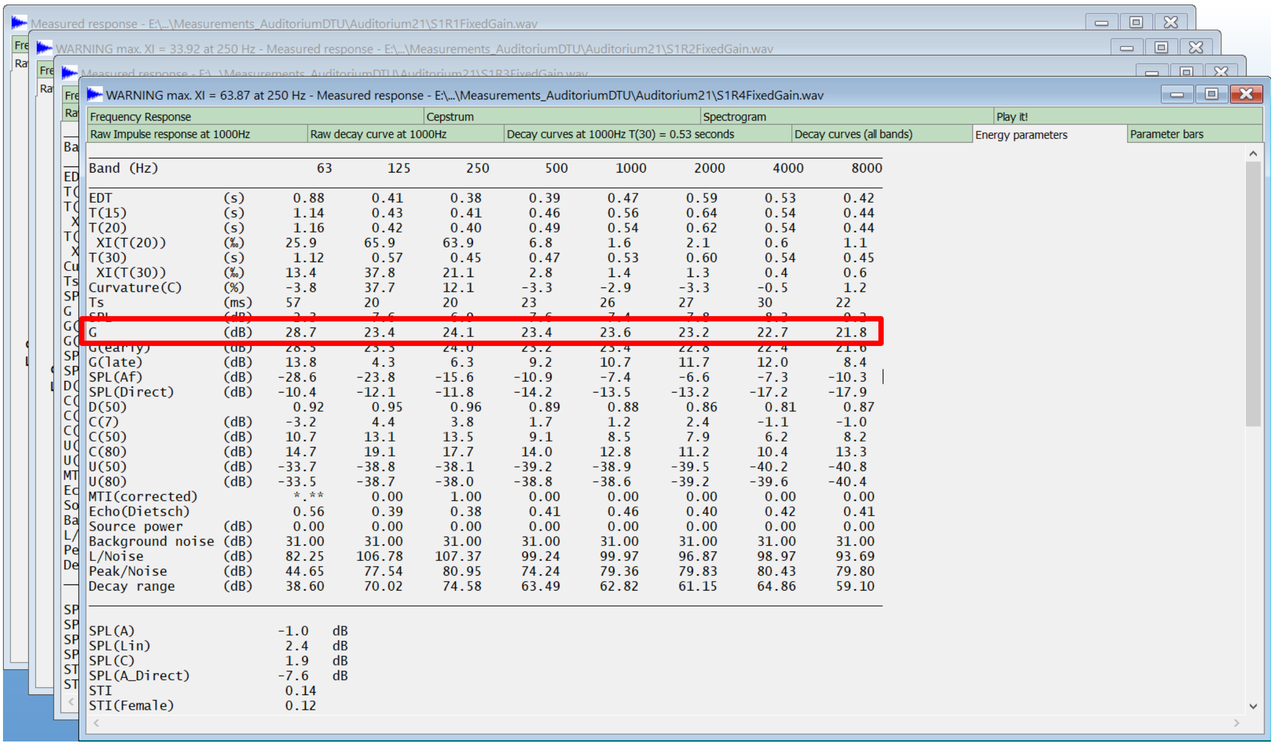
Measurements can be performed with the Measure Impulse Response (Sinusoidal Sweep) tool, as shown in Sec. 4. Calibration will be applied once the recorded Impulse response .WAV files are loaded with the Load impulse response tool, (under the Tools menu). If there is a mismatch between the input/output devices used to derive the calibration file and the ones used for the measurement, a warning is displayed and no calibration is applied.

Assign calibration to existing measurement

It is possible to measure first and calibrate the equipment afterwards as long as the external gains are fixed. When the measurement is finished a calibration file can by assigned by clicking Tools>Measurement calibration>Assign Calibration to Existing Measurements.

An example of calibrated measurement is shown in Figure 7, with the calibration file from

Figure 6. You can test the function on your own with the files in Measurements\Calibration\Auditorium21 folder. For this example use only the files labelled as 'FixedGain'.



Band (Hz)		63	125	250	500	1000	2000	4000	8000
EDT	(s)	0.88	0.41	0.38	0.39	0.47	0.59	0.53	0.42
T(15)	(s)	1.14	0.43	0.41	0.46	0.56	0.64	0.54	0.44
T(20)	(s)	1.16	0.42	0.40	0.49	0.54	0.62	0.54	0.44
XI(T(20))	(‰)	25.9	65.9	63.9	6.8	1.6	2.1	0.6	1.1
T(30)	(s)	1.12	0.57	0.45	0.47	0.53	0.60	0.54	0.45
XI(T(30))	(‰)	13.4	37.8	21.1	2.8	1.4	1.3	0.4	0.6
Curvature(C)	(%)	-3.8	37.7	12.1	-3.3	-2.9	-3.3	-0.5	1.2
Ts	(ms)	57	20	20	23	26	27	30	22
SPL	(dB)	[illegible]	[illegible]	[illegible]	[illegible]	[illegible]	[illegible]	[illegible]	[illegible]
G	(dB)	28.7	23.4	24.1	23.4	23.6	23.2	22.7	21.8
G(early)	(dB)	28.5	23.3	24.0	23.2	23.4	22.8	22.4	21.6
G(late)	(dB)	13.8	4.3	6.3	9.2	10.7	11.7	12.0	8.4
SPL(Af)	(dB)	-28.6	-23.8	-15.6	-10.9	-7.4	-6.6	-7.3	-10.3
SPL(Direct)	(dB)	-10.4	-12.1	-11.8	-14.2	-13.5	-13.2	-17.2	-17.9
D(50)		0.92	0.95	0.96	0.89	0.88	0.86	0.81	0.87
C(7)	(dB)	-3.2	4.4	3.8	1.7	1.2	2.4	-1.1	-1.0
C(50)	(dB)	10.7	13.1	13.5	9.1	8.5	7.9	6.2	8.2
C(80)	(dB)	14.7	19.1	17.7	14.0	12.8	11.2	10.4	13.3
U(50)	(dB)	-33.7	-38.8	-38.1	-39.2	-38.9	-39.5	-40.2	-40.8
U(80)	(dB)	-33.5	-38.7	-38.0	-38.8	-38.6	-39.2	-39.6	-40.4
MTI(corrected)		*.**	0.00	1.00	0.00	0.00	0.00	0.00	0.00
Echo(Dietsch)		0.56	0.39	0.38	0.41	0.46	0.40	0.42	0.41
Source power	(dB)	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00
Background noise	(dB)	31.00	31.00	31.00	31.00	31.00	31.00	31.00	31.00
L/Noise	(dB)	82.25	106.78	107.37	99.24	99.97	96.87	98.97	93.69
Peak/Noise	(dB)	44.65	77.54	80.95	74.24	79.36	79.83	80.43	79.80
Decay range	(dB)	38.60	70.02	74.58	63.49	62.82	61.15	64.86	59.10

SPL(A)	-1.0	dB
SPL(Lin)	2.4	dB
SPL(C)	1.9	dB
SPL(A_Direct)	-7.6	dB
STI	0.14	
STI(Female)	0.12	

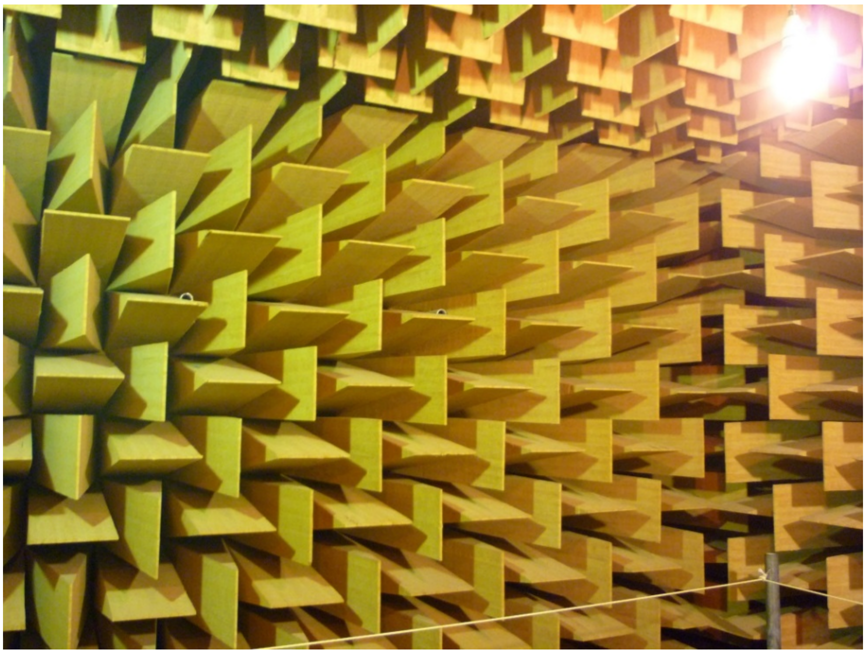
Figure 7: Diffuse-field calibration applied to four impulse response measurements inside Auditorium 21.

Remove calibration from existing measurement

It is also possible to remove calibrations from measurements that are already calibrated. Select Tools>Measurement calibration>Remove Calibration for Existing Measurements to choose the measurement files you want to remove the calibration from and press OK.

6. Making a *free-field* calibration

In this section we will follow the free-field calibration process step by step, using one of the *anechoic chambers* at the Technical University of Denmark, and *Auditorium 21* at the same campus. See more details at the picture description below. You can test the process yourself with the measurement files available in the Measurements\Calibration\free-field method and Measurements\Calibration\Auditorium 21 folders which come with the ODEON installation.



The free-field calibration should be done ideally in an anechoic chamber. For this application note the **small anechoic chamber** at the Technical University of Denmark has been used. The volume of the room is about 60 m^3. Although the method theoretically requires an anechoic room (no reflections from the surfaces), the implementation in ODEON allows the user to use even an ordinary room, but as dry as possible. The main process in a free field calibration is to derive the energy of the direct sound from the source to the receiver. ODEON applies an algorithm which excludes the reflections coming after the direct sound. However, the algorithm works better as these reflections become weaker.

1. Making sure all external levels are fixed

As with the diffuse-field method, it is crucial to decide on all external gains before starting (input/output volume in soundcard, gains in the amplifiers etc.). Also make sure all your connections in the equipment use the same cables, as each cable has its own impedance and this can affect the input/output levels. Both levels and cables have to remain the same between the *Anechoic Chamber* and the actual room under measurement.

2. Taking measurements in the *Anechoic Chamber*

The source and receiver positions are much more critical than in the reverberation chamber. From Eq.(13), it can be seen that the distance *d* between the receiver and the source is an important variable. Ideally the following guidelines must be applied (see Figure 8):

1. The source must be fixed in the middle of the chamber.

2. The receiver (microphone) should be placed ideally at minimum 30 spots around the source, every 12°, for a full 360° circle. For a coarser measurement, 15 spots, every 24° can be acceptable.

3. Make sure no receiver position is closer than 1m to the wall.

4. The distance (radius) *d* between the receiver and the source should be fixed and preferably **greater than 3m**. This distance should be as precise as possible, because the measured sound pressure level can vary significantly with slight changes. A good tip is to use a piece of rope with nooses at the end, tied at the source and microphones stands. Stretching the piece of rope every time you move the microphone around the source will ensure a fixed distance from it. Figure 9 illustrates the geometry for a free-field calibration, according to this principle.

5. Place the source and receiver in a sufficient height to ensure enough separation between the direct sound and the first reflection (see Figure 9). A good guideline is that both source and receiver height is at least 1m above the floor. ODEON will use the heights of source-receiver and distance between them to automatically truncate the impulse response.

Note: When the measurements are taken in a high-quality anechoic room there is no need for truncating the impulse response as there are virtually no reflections. In this case it is advisable to set a high value for the heights (around 10 m) in order to ensure that the whole energy associated with the direct sound is included. Besides, an anechoic room usually has a phantom floor: Source, microphone, people and equipment stand on a metal grid far above the absorptive wedges in the bottom of the anechoic chamber. So practically, there is no floor to create reflections.

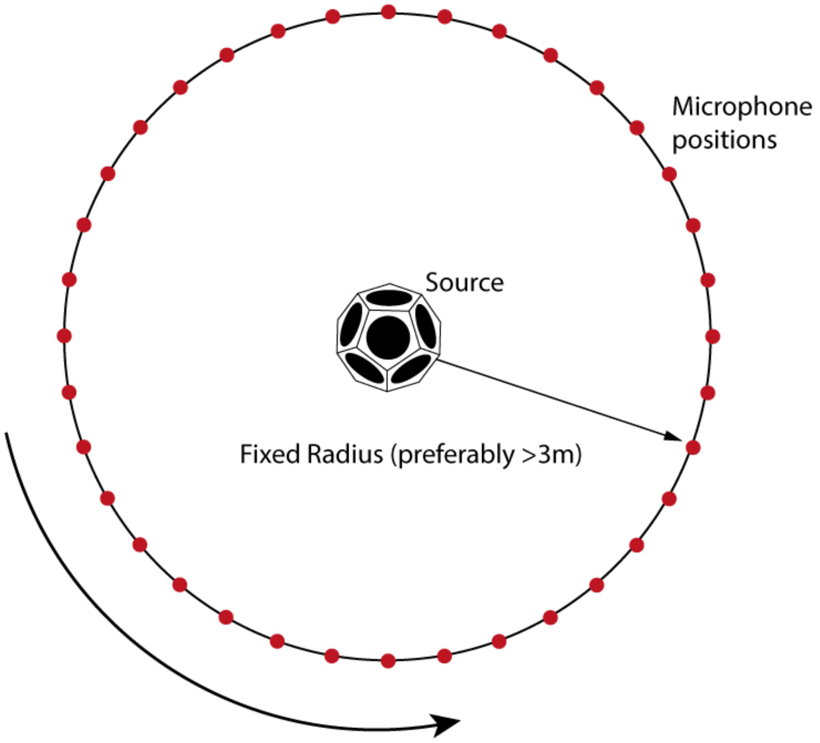


Figure 8: Diagram of measurement positions for free-field calibration. The source should be fixed roughly at the centre of the room and the microphone should be placed at a fixed radius around the source. Ideally 30 positions are required (red spots) but 15 is also an acceptable number. That will correspond to every 12º – 24º. **The main advantage** of this method is that only basic equipment and a piece of rope is necessary. **The main disadvantage** of the method is the large space required.

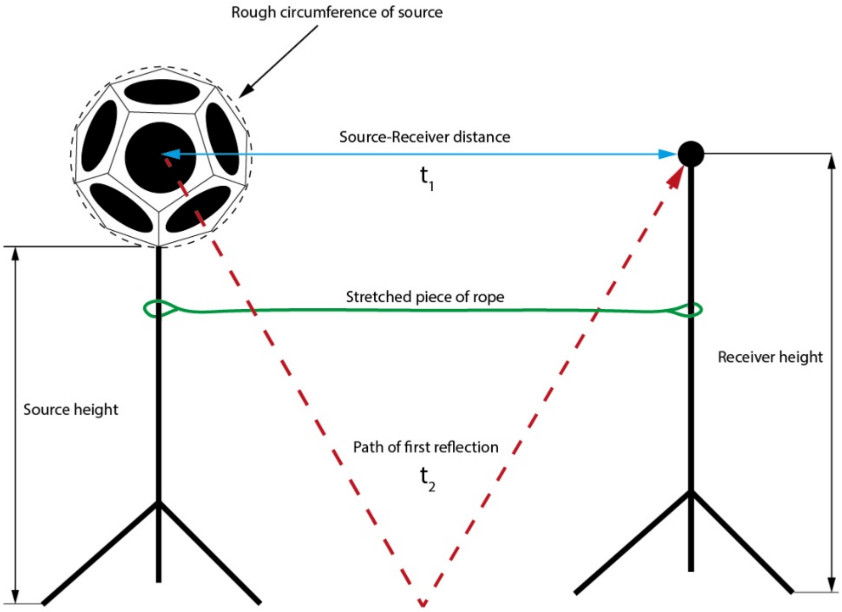


Figure 9: Geometry for impulse response recording in the anechoic/dry room. It is advisable to use a piece of stretched rope with loops to ensure constant distance between source-receiver (preferably <3m). ODEON needs to know the time difference between the arrival of the 1st reflection and the direct sound (**t_2-t_1**). This is calculated by entering the source and receiver heights, as well as the source-receiver distance.

The method described already requires some generous space of at least **3m** radius, plus **1m** from every wall. This can be an impractical requirement. An alternative method is to fix the microphone position and rotate the source instead, on a turntable. It could be argued that the first method is more accurate, as it averages the uneven reflections from the room at the different microphone positions. However, if the free field is evenly distributed inside a well-designed anechoic chamber, the second method requires far less space (see Figure 10).

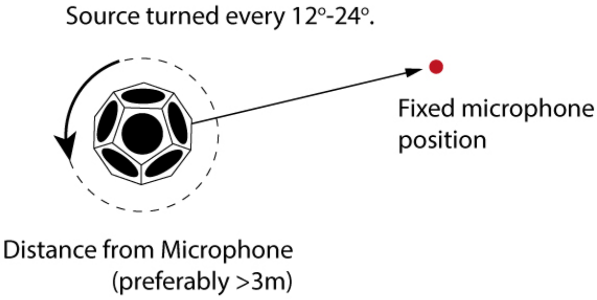


Figure 10: Alternative way of measuring around the source. In this case the microphone is fixed a distance greater than 3m from the centre of the source. The source should be rotated every 12º – 24º, which gives 30 or 15 measurement angles respectively. **The main advantage** of this method is that only about half of the space is required, comparing to the previous method. **The main disadvantage** of the method is the necessity of a turntable or another accurate device to rotate the source at precise angles.

For our example, following one of the two layouts proposed in Figure 8 and Figure 10 we measured the impulse response at 15 spots. The measurements were performed in small anechoic chamber where 3m distance between receiver and source was not possible. Therefore, for this example, which is merely a test of the method, we used **only 1m distance**.

An example of a successful impulse response in the anechoic room is given in Figure 11, where all SPL values have been derived. The sound strength parameter (*G* value) is not derived as the measurements are not calculated yet. It should also be expected that SPL can have very low values (even negative, meaning much lower sound pressure than the reference 20 µPa). In the measurements taken inside the *anechoic chamber*, "*" characters are expected for most of the reverberation or clarity parameters, because any impulse response is truncated too early - at the direct sound. What matters are the values of **SPL**, and you should check that all have been derived.

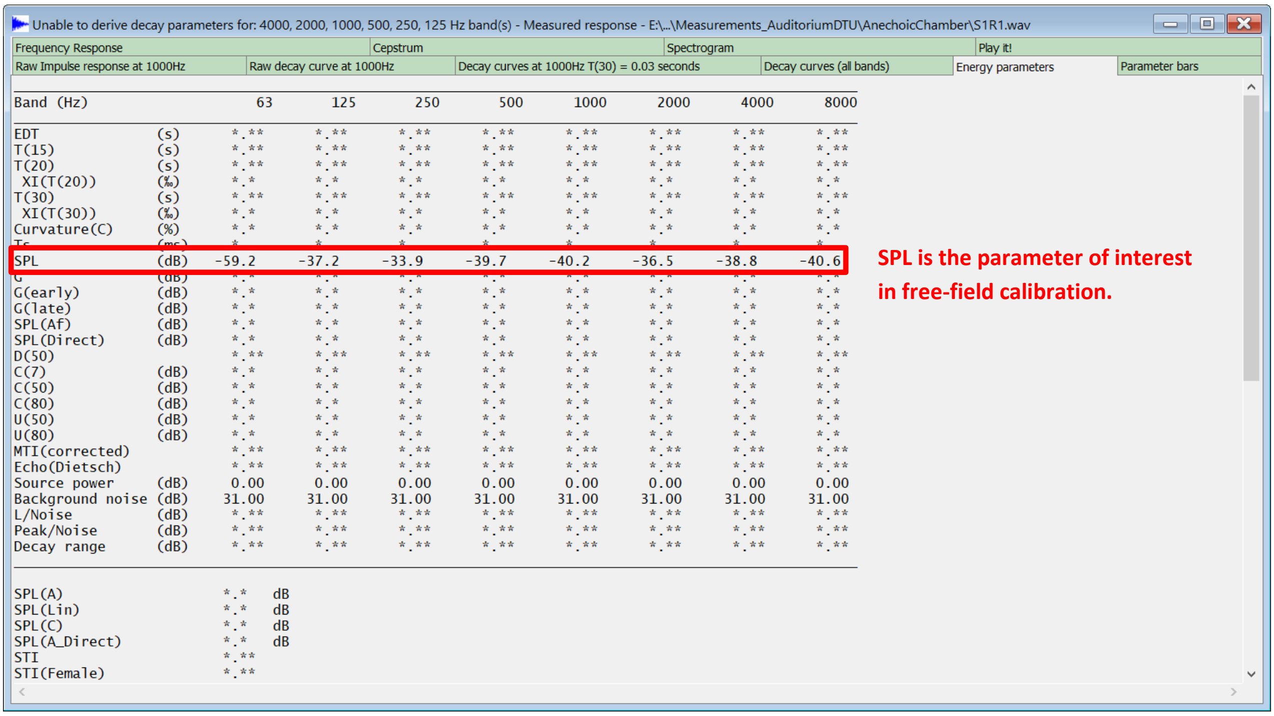


Figure 11: Room acoustic parameters from a healthy impulse response recording inside the anechoic/dry room. Since only the direct sound is taken into account, it should not be expected to obtain all acoustic parameters. The only relevant parameter is SPL.

3. Performing the calibration

Once all measurements inside the anechoic chamber have been completed, follow the next steps to perform the calibration. For this exercise you can use the measured files found in the Measurements\Calibration\free-field method folder.

1) Click on Tools>Measurement Calibration>Free field>One step (requires fixed signal chain). ODEON asks for the source-receiver heights and the distance between them. All geometric values used are shown in the ODEON geometry dialog in Figure 12.

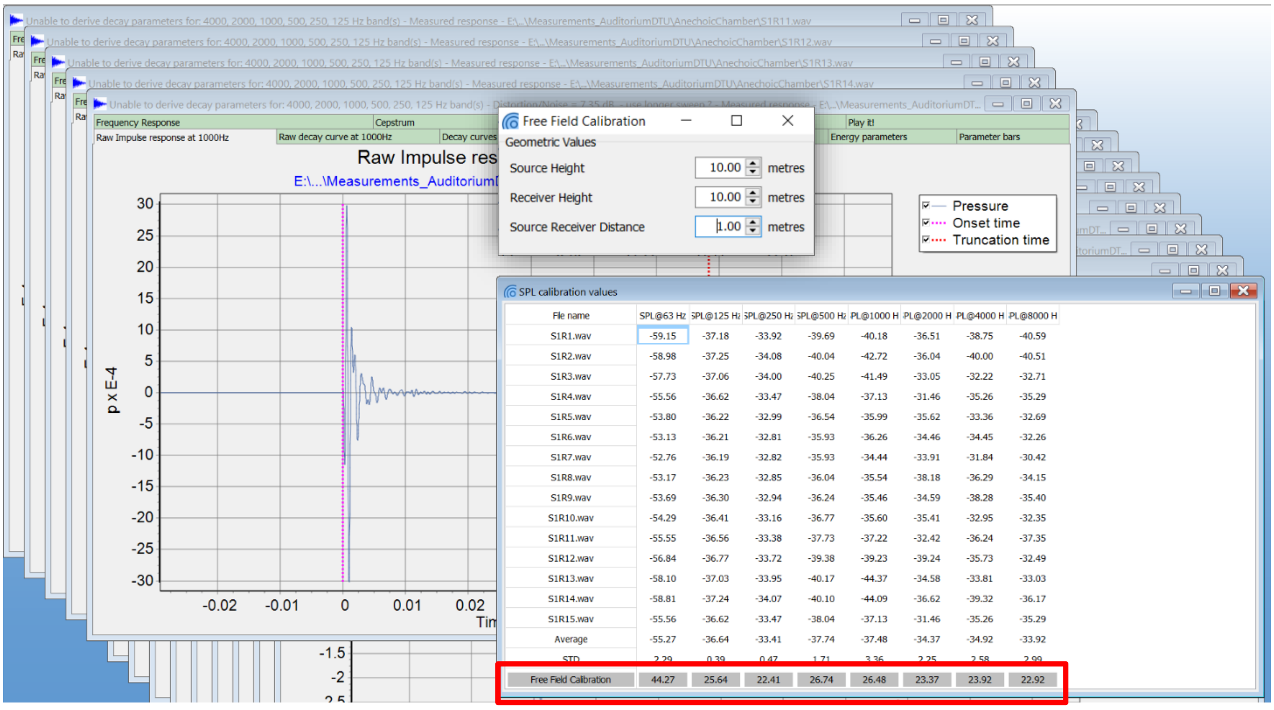
2) Close the geometry dialog window to confirm. ODEON will ask for the impulse responses in the anechoic chamber. Select the files all together using the SHIFT or CTRL key and open them at once.

2) ODEON uses the geometric data of source and receiver to derive the SPL of the direct sound for the eight octave bands between 63 Hz and 8000 Hz, as an average from all measurement positions (Figure 8 or Figure 10). The calibration values ΔL^{Anech} , calculated using eq.(15) are displayed at the bottom of table.

3) After all files are loaded, a dialog pops up and asks for a name of the calibration file. You are prompted to set the file as the active calibration file in the Measurement Setup, which is going to be used with the following measurements.

4) It is possible to change the active calibration file at a later point in the Measurement Setup. However, if the input and output devices are different from those used during the generation of the calibration file, ODEON gives a warning for device mismatch, meaning that all level depended parameters will not be calibrated.

5) You can now launch the Measurement Setup window to check whether the active calibration file is the correct one.



Calibration values to be added on the SPL measurement, in situ.

Figure 12: Calibration values (factors) derived from 15 measurements inside the Anechoic Chamber. The values are calculated according to eq.(15) and will be added in the SPL measurements in the real room, in order to derive the G values (see eq.(14)).

At this point, we can actually see great similarities between the calibration values of Figure 6 and Figure 12. According to Eq.(8) and E.(14) we expect that the final calibration values from both the diffuse-field and the free-field method must be identical, as soon as the conditions are ideal. In our case, several errors contribute to deviations between these values. However, the agreement is convincing. Reasons for possible disagreement are:

1. The reverberation or anechoic chambers are not ideal. This means that the first is not reverberant enough or perfectly diffuse and the second is not dry enough at all positions.

2. Not enough receiver (microphone) positions used.

3. Receiver not equally distant from the source in the anechoic chamber.

4. Instabilities of source output, especially at low frequencies.

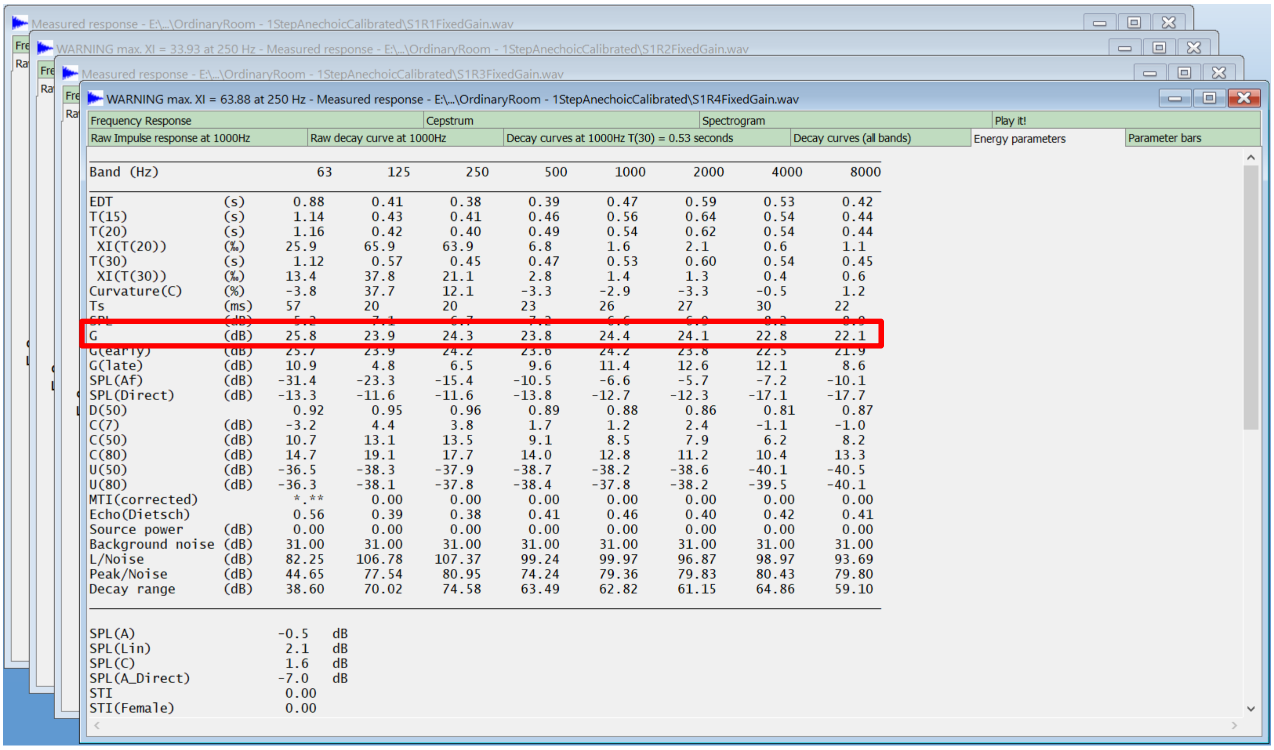
5. Not precise detection of onset and truncation times, especially at the lowest octave bands, because of greater wavelengths.

Note

If the input and output devices are different from those used during the generation of the calibration file ODEON gives a warning for device mismatch, meaning that all level depended parameters will not be calibrated in future measurements. You can now launch the Measurement Setup window to check whether the active calibration file is the correct one.

4. Performing calibrated measurements

At this point the exact same process is followed as in Sec. 5. 4, for the diffuse-calibration measurements. The only difference is that another calibration file is active, which comes from the free-field method. As with the diffuse-field calibration both the Tools>Measurement Calibration>Assign Calibration to Existing Measurements and Tools>Measurement Calibration>Remove Calibration from Existing Measurements functions can be used to assign or remove calibration after the actual measurements have been performed. You can test the functions on your own with the files in Measurements\Calibration\Auditorium21 folder. An example is given below, where the derived G values are close enough to the ones derived using the diffuse-field method in Figure 7.



Band (Hz)		63	125	250	500	1000	2000	4000	8000
EDT	(s)	0.88	0.41	0.38	0.39	0.47	0.59	0.53	0.42
T(15)	(s)	1.14	0.43	0.41	0.46	0.56	0.64	0.54	0.44
T(20)	(s)	1.16	0.42	0.40	0.49	0.54	0.62	0.54	0.44
XI(T(20))	(‰)	25.9	65.9	63.9	6.8	1.6	2.1	0.6	1.1
T(30)	(s)	1.12	0.57	0.45	0.47	0.53	0.60	0.54	0.45
XI(T(30))	(‰)	13.4	37.8	21.1	2.8	1.4	1.3	0.4	0.6
Curvature(C)	(%)	-3.8	37.7	12.1	-3.3	-2.9	-3.3	-0.5	1.2
Ts	(ms)	57	20	20	23	26	27	30	22
SPL	(dB)	5.2	7.1	6.7	7.2	6.6	6.9	8.2	8.9
G	(dB)	25.8	23.9	24.3	23.8	24.4	24.1	22.8	22.1
G(early)	(dB)	25.7	23.9	24.2	23.6	24.2	23.8	22.5	21.9
G(late)	(dB)	10.9	4.8	6.5	9.6	11.4	12.6	12.1	8.6
SPL(Af)	(dB)	-31.4	-23.3	-15.4	-10.5	-6.6	-5.7	-7.2	-10.1
SPL(Direct)	(dB)	-13.3	-11.6	-11.6	-13.8	-12.7	-12.3	-17.1	-17.7
D(50)		0.92	0.95	0.96	0.89	0.88	0.86	0.81	0.87
C(7)	(dB)	-3.2	4.4	3.8	1.7	1.2	2.4	-1.1	-1.0
C(50)	(dB)	10.7	13.1	13.5	9.1	8.5	7.9	6.2	8.2
C(80)	(dB)	14.7	19.1	17.7	14.0	12.8	11.2	10.4	13.3
U(50)	(dB)	-36.5	-38.3	-37.9	-38.7	-38.2	-38.6	-40.1	-40.5
U(80)	(dB)	-36.3	-38.1	-37.8	-38.4	-37.8	-38.2	-39.5	-40.1
MTI(corrected)		*.**	0.00	0.00	0.00	0.00	0.00	0.00	0.00
Echo(Dietsch)		0.56	0.39	0.38	0.41	0.46	0.40	0.42	0.41
Source power	(dB)	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00
Background noise	(dB)	31.00	31.00	31.00	31.00	31.00	31.00	31.00	31.00
L/Noise	(dB)	82.25	106.78	107.37	99.24	99.97	96.87	98.97	93.69
Peak/Noise	(dB)	44.65	77.54	80.95	74.24	79.36	79.83	80.43	79.80
Decay range	(dB)	38.60	70.02	74.58	63.49	62.82	61.15	64.86	59.10

SPL(A)	-0.5	dB
SPL(Lin)	2.1	dB
SPL(C)	1.6	dB
SPL(A_Direct)	-7.0	dB
STI	0.00	
STI(Female)	0.00	

Figure 8: Free-field calibration applied to four impulse response measurements inside Auditorium 21.

7. Two-step calibration

So far, we have seen that external gains in any device in the equipment and various connections have to remain fixed all the way from calibration to the actual measurement. Although it is always recommended to write down all gain settings in the amplifiers, the audio interface and the PC, it is very common to miss this information, especially if calibration was done long before the actual measurement. In this section we introduce the concept of *two-step calibration* which provides extra safety in case external gains in the measuring equipment have been changed accidentally, and high versatility when the user needs to readjust the gains on purpose. The benefits of a *two-step calibration* can be summarized as:

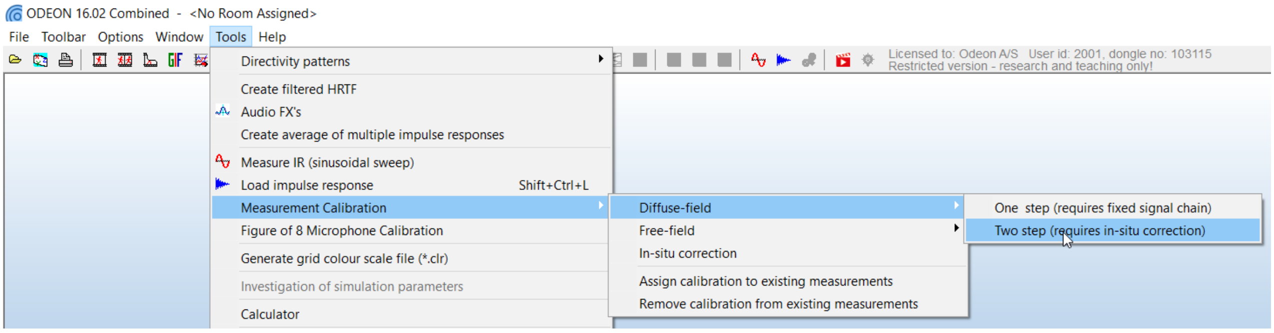
1) Correcting for accidental changes in output or input gains – for any part of the chain (audio interface, PC volume, loudspeaker amplifier, microphone amplifier).
2) Correcting for intentional changes in output or input gains. For example, when the gain settings used during the calibration are not sufficient to drive the room on-site, they need to be increased.
3) Allows the user to change part of the equipment – audio interface, amplifiers – as long as source and microphone stay the same.

1. Method

The *two-step calibration* is essentially one more step added to the basic methods, as they are described in sections 5 and 6. The initial part of a *two-step calibration* is therefore identical with one of the two basic methods. The second step is simply a measurement of the actual output of the source during the calibration procedure.

A Two-step calibration using the *Diffuse-field method* is available at Tools>Measurement Calibration>Diffuse-field>Two-step (requires in-situ correction).

A Two-step calibration using the *Free-field method* is available at Tools>Measurement Calibration>Free-field>Two-step (requires in-situ correction).



The main concept of the *two-step calibration* is that the whole chain of gains in the measuring equipment is considered as a black box and what matters is only the final output from the omni-directional source. Any change in the gains in the measuring equipment is assumed to affect the source output linearly. Therefore, one can measure the *level* by the source at a known distance from the source during the calibration and during the actual measurement and find the exact difference in the gain (if any) between calibration and measurement. Since we want to measure the sound pressure level of the source only, the same process of truncating any reflections as in the Free-field calibration is applied.

As described in Sec. 5 and 6, when the main *diffuse* or *free-field* calibration finishes ODEON derives a calibration value for each octave band and stores it in a calibration file. The following will be described for the Diffuse-field calibration, but can be directly applied to the free-field as well.

The correction value after a diffuse-field calibration is ΔL^{RevCh} and it is given by eq.(9). For the two-step calibration we simply follow two extra steps:

1. While the equipment is still inside the *Reverberation Chamber*, we place the microphone at a specific distance d from the source and we measure the Level, which we will denote by $L_p^{S,Calib}$. This level is now registered, so that any deviation from now on will be trackable.

2. During the actual measurement we place the microphone at the same distance d and preferably in front of the same unit as we did for the calibration. The new SPL is $L_p^{S,Meas}$. This measurement is called *In-situ correction*. The final calibration values will be then:

$$\Delta L_p^{Cal,final} = \Delta L^{RevCh} - (L_p^{S,Meas} - L_p^{S,Calib}) \quad \text{dB}$$

$$\boxed{L_p^{Cal,final} = \Delta L^{RevCh} + L_p^{S,Calib} - L_p^{S,Meas}} \tag{15}$$

Apparently, when all gains are unchanged during the calibration and during the measurement,

$$L_p^{Cal,final} = \Delta L^{RevCh}$$

This condition corresponds to a basic calibration, as described in Sec. 5, where the whole setup and gains remain unchanged from the reverberation chamber to the on-site measurement. Figure 13 shows the concept schematically. A change in the gain in the equipment does not necessarily affect all frequencies in the same amount. In other words, the shift in the frequency response when measuring the $L_p^{S,Calib}$ and the $L_p^{S,Meas}$ might not be linear (see Figure 14). This is why Eq. (15) is calculated eight times, one for each octave band.

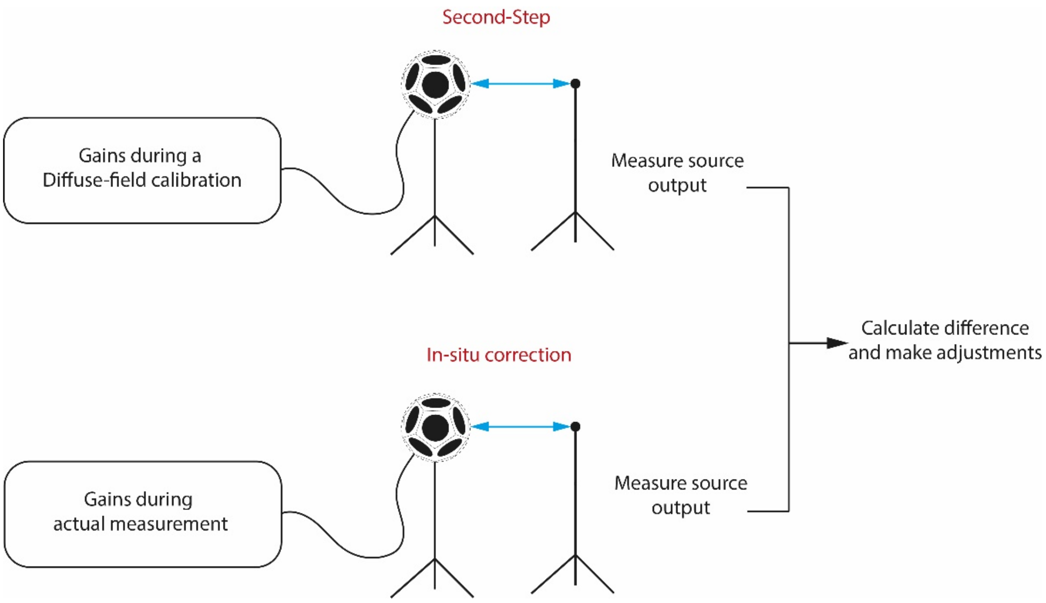


Figure 13: The concept of a Two-step calibration. When all recordings for the main diffuse-field or free-field calibration have been completed, one extra recording is taken still in the same calibration room at a specific distance from the source. The same recording is taken afterwards in the actual room. If any change has been made in the gains of the equipment it will be found as a supplementary adjustment.

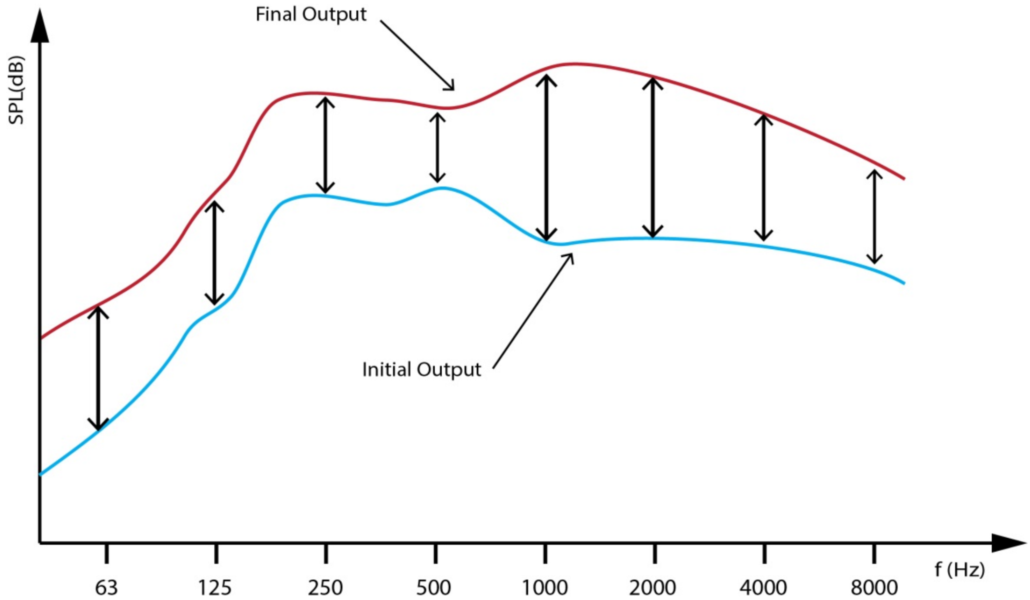


Figure 14: Changing any of the gains in the measurement chain does not necessarily shift the frequency response of the source linearly. All eight octave bands are taken into account separately, with eight correction factor values $L_p^{S,Meas} - L_p^{S,Calib}$ derived.

2. Taking measurements for a two-step calibration

Obtain recordings for the 1st step

First of all, we need to obtain enough recordings for the normal diffuse (or free-field) calibrations, with the guidelines mentioned in Sec. 5 and Sec. 6. For the following we can use the impulse responses included in the folders Measurements\Calibration\diffuse-field method and Measurements\Calibration\Auditorium21.

Obtain recordings for the 2nd step

Still being inside the same room (reverberation or anechoic chamber) we place the microphone close to the source, at a known distance, in front of a known point (putting a sign if needed in order to be able to recognize it later). This will provide us with an impulse response measurement for the second step. The microphone should be placed as close as possible in order to help ODEON distinguish the direct sound from the rest reflections in the room. This is because the whole method relies on reporting the sound pressure level from the source during the calibration and in situ, but without any influence by the room. Figure 15 illustrates the setup. As the microphone approaches the source, the difference between the arrival times of the first reflection and the direct sound, t_2-t_1, increases. This ensures that ODEON will distinguish the direct sound from the first reflection better. In contrast to the free-field calibration method [2], capturing the whole energy of the direct sound is not as crucial. In case the direct sound is truncated too early, only a percentage of its whole energy will be included. However, the same percentage will be included in situ, so the overall difference will be the same. On the other hand, it is rather important to truncate the impulse response soon enough so it does not contain any part of the first reflection, which depends on the associated room, and therefore cannot be controlled.

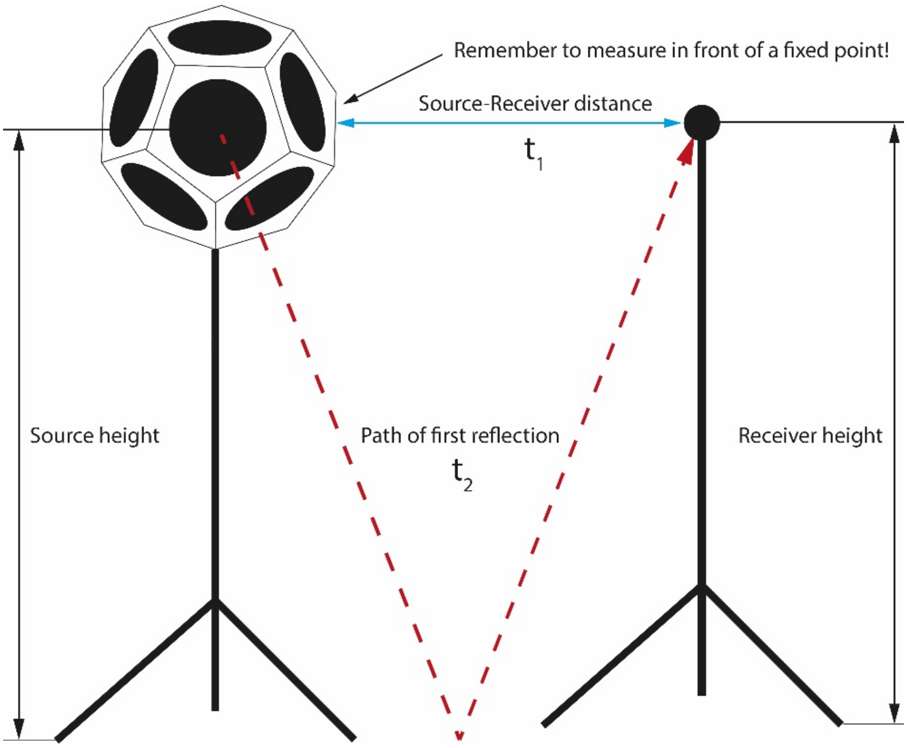


Figure 15: Geometry for impulse response recording for the second step. ODEON needs to have sufficient time difference between the arrival of the 1st reflection and the direct sound (t_2-t_1). This is accomplished by having high source-microphone heights and short distance between source and microphone.

A combination of distances that ensures well-cut of the direct sound is:

- Source height = 1m.
- Receiver height = 1m.
- Source-receiver distance <= 1m.

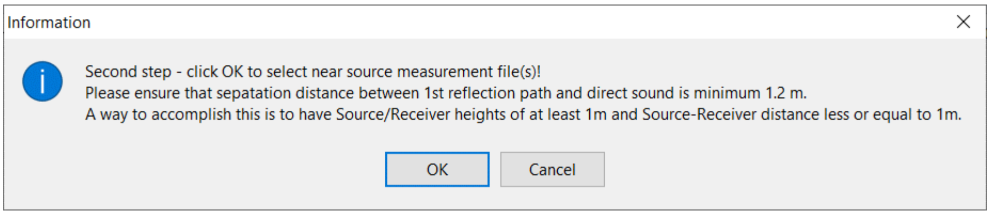
Having all necessary recordings stored, we can begin the actual calibration. The steps will be presented for using the Diffuse-field method, but they can be applied directly for the Free-field method as well.

3. Performing the two-step calibration

1) Choose Tools>Measurement Calibration>Diffuse-field>Two-step (requires in-situ correction). First, this opens the interface for the normal diffuse calibration.

- Enter the volume of the reverberation/diffuse chamber and click OK.
- Select the group of impulse responses recorded in the diffuse chamber, holding down the SHIFT or CTRL button (see [2] and [3] for a recommended number of recordings). It is important to open all impulse responses together – not separately – in order for ODEON to average the results. Press OPEN. ODEON loads all impulse responses and derives the correction values for the diffuse-field calibration.

2) For the second step in the calibration the following warning message is displayed:



This is to remind that the microphone should be as close as possible to the source to help ODEON truncate the impulse response well before the first reflection (which comes from the floor, as the closest surface). Click **OK**.

In the next file menu select the impulse response recording from the 2nd step that was made in front of a known point and known distance from the source.

For the next examples, we have used the impulse response file S1R_D80HR130HS130.wav found inside the folder Measurements\Calibration\diffuse-field method. The impulse response file has been measured at 80 cm in front of the source, at 130 cm height.

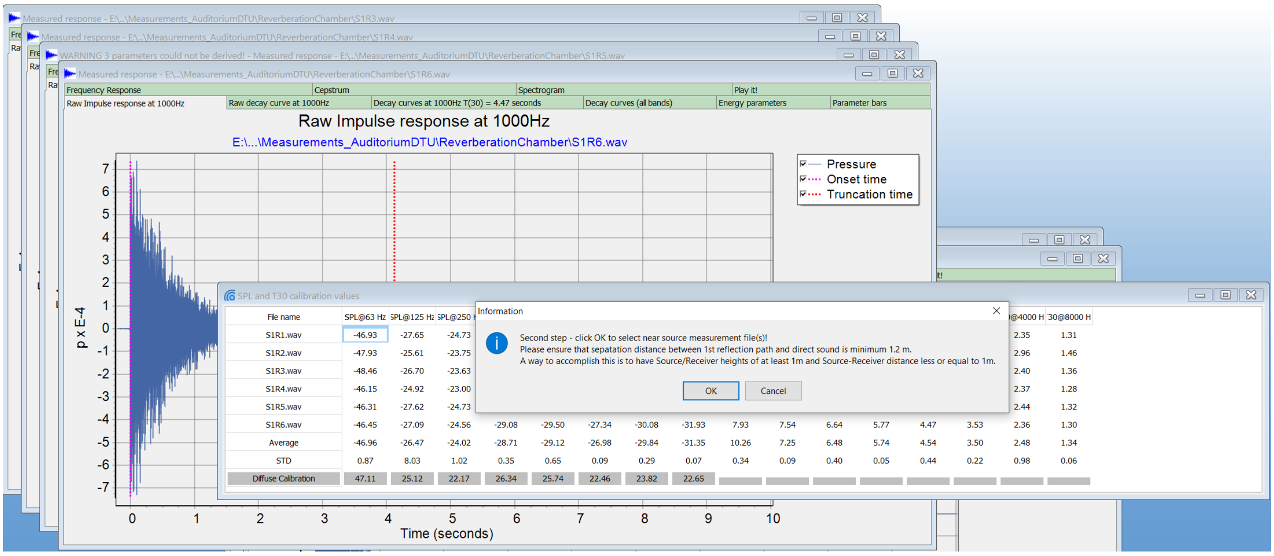


Figure 16: 1st calibration step: Load the recordings made in the diffuse or anechoic chamber. For this application note we used 12 recordings inside the diffuse chamber. After loading these recordings, you can implement the 2nd step by choosing the reference recording at the source.

ODEON asks you to save the file as .2StCalibration. This extension emphasizes the fact that the file cannot be used for calibrated measurements yet. It needs an in-situ correction during the actual measurement in order to be converted to a final calibration file with extension .Calibration. For this application note the file is named CalibrationRevChamb_10Nov2020.2StCalibration.

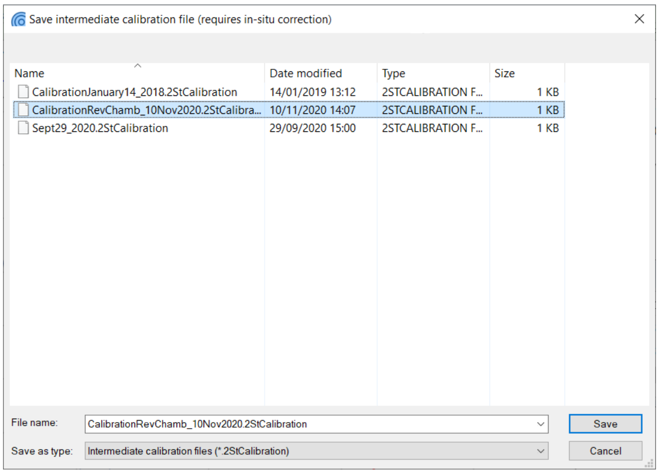


Figure 17: After completing step 1 and 2 you need to save a 2-step calibration file (.2StCalibration). This is only an intermediate calibration file and cannot be used directly for calibrating measurements. An in-situ correction is required.

At the end of the process so far, your ODEON screen should look like the following, with two tables:

- **SPL and T30 Calibration values** from the diffuse-field or free-field main calibration.
- **Source SPL values** reported during the additional 2nd step.

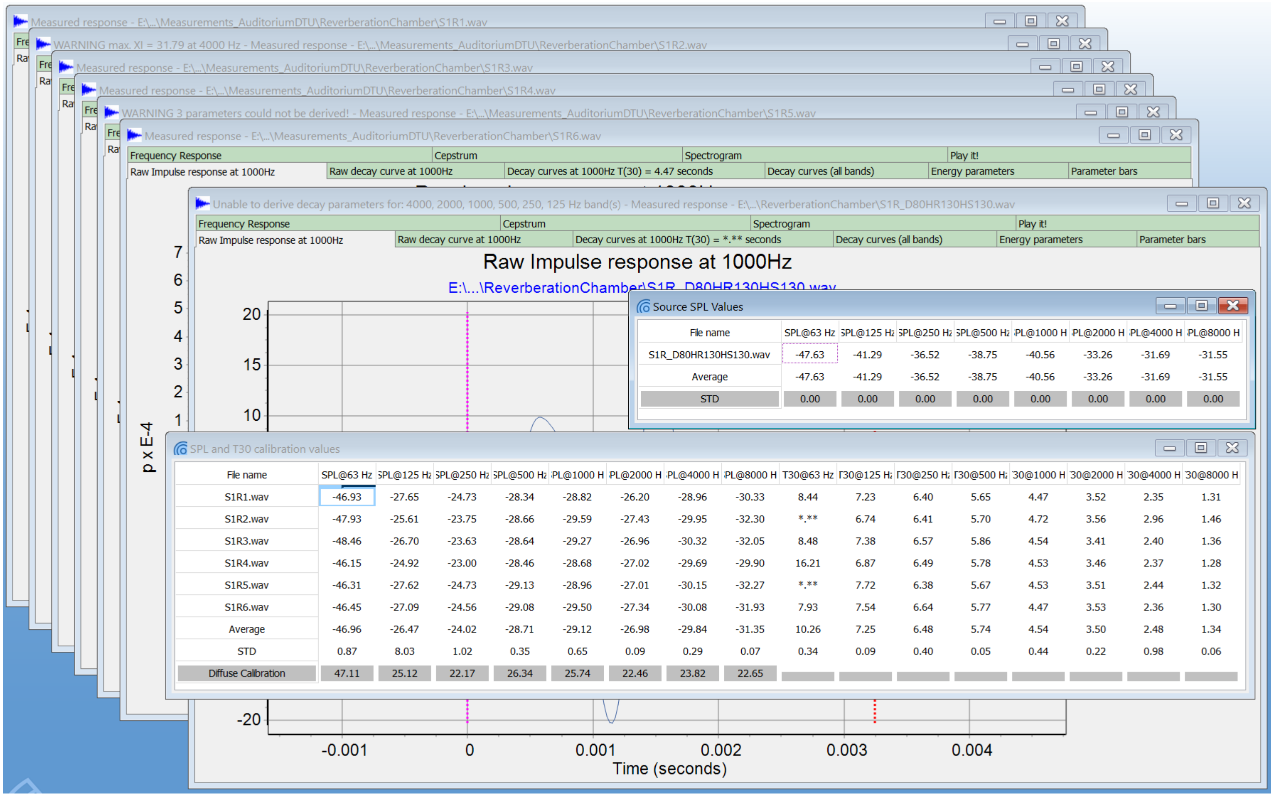


Figure 18: End of two-step calibration. ODEON displays two tables: 1) All recordings made in the diffuse (or anechoic) chamber, with the calibration values. 2) Source output levels at a fixed location from the source (upper small table in the screenshot).

4. In-situ correction

Now the equipment is moved from the reverberation chamber to the actual room under measurement. It is likely that the whole chain of gains in the measuring equipment remains unchanged during this move. In such case the in-situ correction is just trivial, giving zero adjustments for the output. However, if any of the gains have been changed, then the whole calibrated measurement is in danger, since the source output has been changed. The .2StCalibration file is therefore used as a reference file and introduces correction values that compensate for any gain change.

Follow the setup in Figure 15 to obtain an impulse response at exactly the same distance from the source, as during the 2nd calibration step in the reverberation chamber. Remember also that the microphone should be placed in front of the same point (direction) on the source. This is because sources are not perfectly omnidirectional, therefore measuring at a different point might introduce errors in the gain estimation.

Click Tools>Measurement Calibration>In-situ correction. Open the .2StCalibration file that you created during the 2nd calibration step. For this application note the file is called CalibrationRevChamb_10Nov2020.2StCalibration.
A warning message is displayed again for the recommended heights and distance between source and receiver.

From the file menu choose and open the .WAV file that was recorded in front of the source inside the ordinary room. In our case, we use the file S1R_80cmFromSource_ChangedGain.wav inside the folder Measurements\Calibration\Auditorium21. You will be asked to save your new final calibration file which now has an extension .Calibration, indicating that it can be used as any other calibration file for calibrating the following measurements.

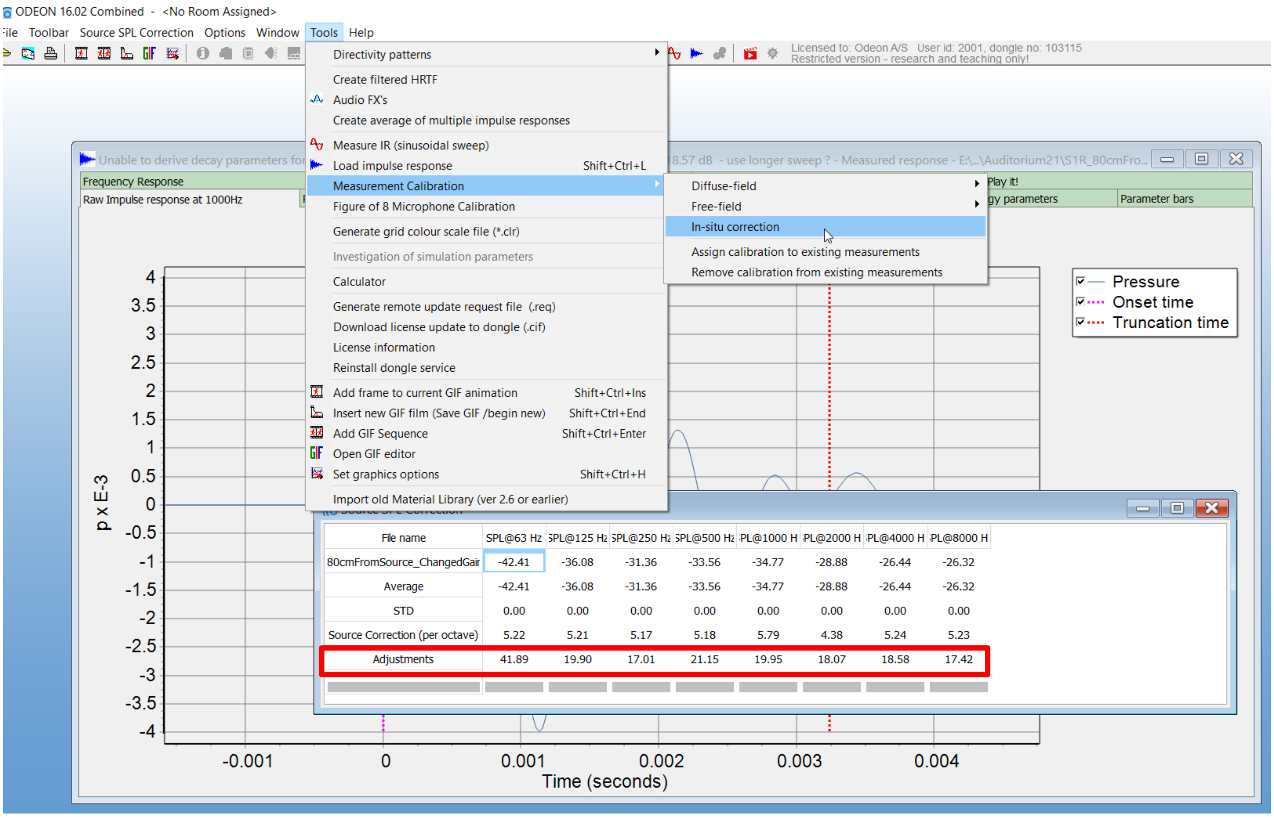


Figure 19: Performing in-situ correction. First, the .2StCalibration file created in Figure 17 is chosen. Second, you need to choose the recording made in-situ in front of the source. If any of the gains in whole measuring chain have been changed, a correction factor will be derived and taken into account for the final calibration.

At this point we can see that there is a source correction (per octave) of about 5 dB across the octave bands. This means that the source output has been increase by approximately this amount. The small differences between octave bands come from the fact that the increase in gain is not entirely linear (see. Figure 14). Since the source output has been increased, less compensation is required now for the calculation of *G*, comparing to the initial diffuse-field calibration. Let's compare Figure 6 with Figure 19. We can see that the Diffuse calibration row in Figure 6 is approximately equal to the sum of the Adjustments row, plus the Source Correction (per octave) in Figure 19.

5. Summary of the steps for a full Two-step calibration

Depending on the time and needs during a calibrated measurement, there are different ways to perform the steps required for a two-step calibration. Remember in all steps you can freely change the internal ODEON gain (in the sweep generator). Restrictions apply only on the external gains in the measuring chain.

Calibration steps first – measurements afterwards

The most straightforward way is to make all calibration settings first and then perform the measurements:

1) Make the impulse response recordings in a reverberation or anechoic chamber, for a *diffuse-field* or *free-field* calibration. This is the 1st step in the Two-step method. **External gains should remain fixed during this time!**
2) Make one more recordings close to the source – at fixed location and fixed distance. This is the 2nd step in the Two-step method. **External gains should remain fixed during this time!**
3) Enter the actual room for measurement. External gains can now be changed to achieve a satisfying signal to noise ratio or might have been accidentally changed.
4) Record an impulse response at exactly the same position as in step 3). This is to monitor any gain differences in the output or input. **From now on external gains should remain fixed again.**
5) Run a Two-step calibration by choosing
   Tools>Measurement Calibration>Diffuse-field>Two-step (requires in-situ correction)
   or Tools>Measurement Calibration>Free-field>Two-step (requires in-situ correction),
   depending on the method you used for the 1st step.
6) Follow the procedure, by loading the necessary information and impulse response recordings, and derive a .2StCalibration file.
7) Run the in-situ correction by choosing Tools>Measurement Calibration>In-situ correction. Load the recording from step 5).
8) Save the final calibration file that should have the extension .Calibration. Make this calibration file active.

Perform measurements. Calibration will be applied on the recorded .WAV files during processing at the Load impulse response tool, (under the Tools menu). The active calibration file will be applied. This is visible in the Options>Program Setup>Measurement Setup. If there is a mismatch between the input/output devices used for the calibration file and the ones used for the measurement, a warning is displayed by ODEON and no calibration is applied.

Measurements first – Calibration afterwards

In ODEON, calibration may be performed after measurements. To do this you can simply choose Tools>Measurement Calibration>Assign Calibration to Existing Measurements to apply the calibration. More analytically, the steps for such a procedure are:

1) Enter the actual room.
2) Perform measurements with an overall output-input combination of gains adequate for a good signal to noise ratio.
3) Make one more recording close to the source (fixed location, fixed distance). This will be the in-situ correction in the Two-step calibration method. **External gains should stay the same as in measurements!**
4) Enter the diffuse-field or free-field lab. **External gains might have been changed at this point!**
5) Make the impulse response recordings necessary for a diffuse-field or free-field calibration. This is the 1st step in the Two-step method. **External gains should remain fixed during this time!**
6) Record an impulse response at exactly the same position as in step 3). This is to monitor any gain differences in the output or input
7) Run a Two-step calibration by choosing Tools>Measurement Calibration>Diffuse-field>Two-step (requires in-situ correction) or Tools>Measurement Calibration>Free-field>Two-step (requires in-situ correction), depending on the method you used for the 1st step.
8) Follow the procedure, by loading the necessary information and impulse response recordings, and derive a .2StCalibration file.
9) Run the in-situ correction by choosing Tools>Measurement Calibration>In-situ correction. Load the recording from step 3).

10) Save the final calibration file that should have the extension .Calibration. You do not have to make this calibration file active.
11) Apply calibration to the measurements from step 2) by choosing Tools>Measurement Calibration>Assign Calibration to Existing Measurements.

8. The STI parameter

All previous sections about the *Diffuse-field*, *Free-field* and *Two-step* methods can be applied for measurements of Speech Transmission Index (STI) as well. The exact same steps are followed, with only two additional settings required in the Measurement Setup.

1. Measurements with the ISO 3382-3 source

STI measurements with the ISO 3382-3 source are suitable especially for open-plan office studies. This type of source is defined in [4] and corresponds to an **Omnidirectional source** that has a speech spectrum, instead of a flat one.

1) Open the Options>Program Setup>Measurement Setup.

2) Choose the Speech-ISO 3382-3 spectrum.

Source Power Spectrum

Frequency	63	125	250	500	1000	2000	4000	8000	Hz	Reset to Zero
Sound power	-	60.9	65.3	69.0	63.0	55.8	49.8	44.5	dB	Speech - ISO 3382-3

3) Specify the Background noise. If a room is already loaded, ODEON will use the values inserted in the Room Setup instead. The background noise is not needed during calibration at the anechoic chamber, but during processing the data from the actual room under evaluation. Therefore, you can obtain the background noise either by measuring it directly in the room with an SPL analyser (not currently possible to do this in ODEON) or by simply estimating it according to literature in similar venues.

Back ground noise spectrum (for RASTI, STI,..U(50)...)

Frequency	63	125	250	500	1000	2000	4000	8000	Hz
Background noise	67.0	60.0	54.0	49.0	46.0	44.0	43.0	42.0	dB

(when a room is assigned, the level of background noise is not specified here but in the room setup)

2. Measurements with an Artificial Mouth

Measurements with the ISO 3382-3 source can be suitable for open-plan office parameters, but they are not as realistic as with an *artificial mouth* source. Such a source is constructed as a box with one loudspeaker driver in front, in order to represent the directivity pattern radiated from a real mouth on a head (see more at [5]). An example of a commercially available *artificial mouth* is the Echo Speech Source, **Type 4720**, by **B&K**.

To perform STI measurements with an *artificial mouth*, the following spectrum needs to be manually inserted in the Options>Program Setup>Measurement Setup.

Source Power Spectrum

Frequency	63	125	250	500	1000	2000	4000	8000	Hz	Reset to Zero
Sound power	62.2	62.2	68.2	70.8	64.5	59.8	54.8	49.5	dB	Speech - ISO 3382-3

Figure 20: Power spectrum used for calibrations with an artificial mouth, that is equivalent to a BB93_NORMAL_NATURAL.SO8 source.

This spectrum is derived from the BB93_NORMAL_NATURAL.SO8 source according to the following process:

1. The sound power spectrum of a BB93_NORMAL_NATURAL.SO8 source is as highlighted in the green box below:

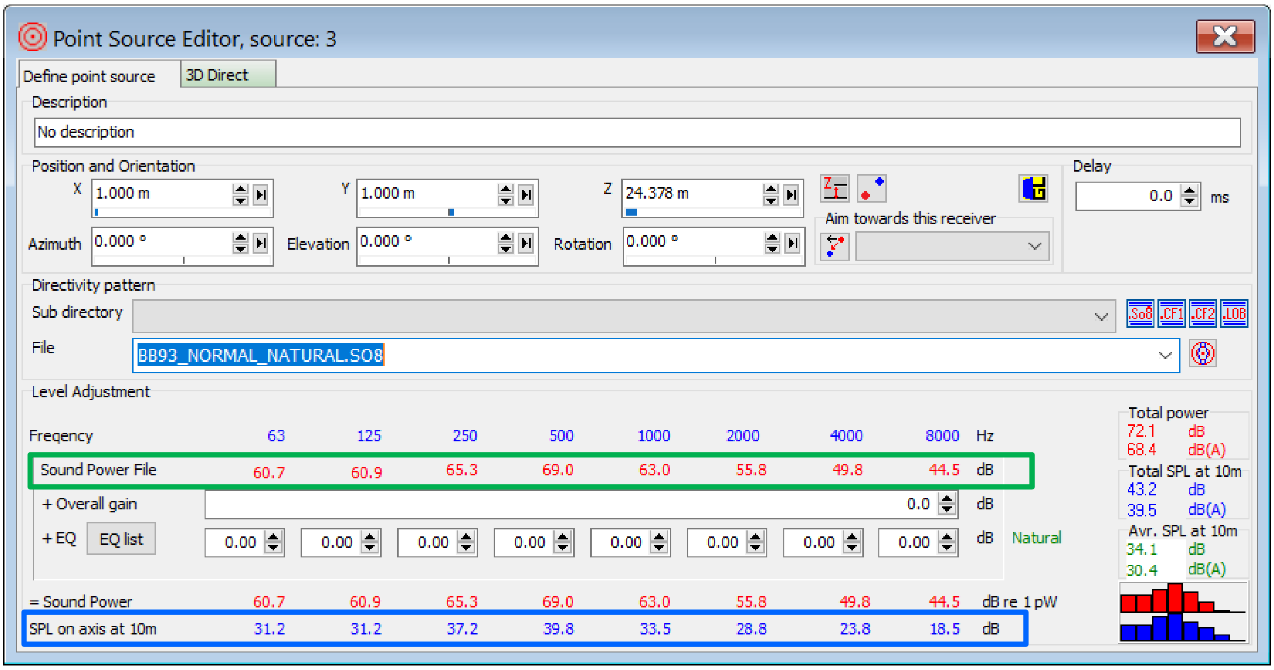
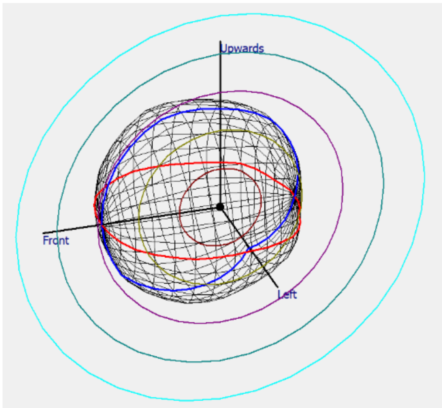


Figure 21: The BB93_NORMAL_NATURAL.SO8 source has a power spectrum as shown in the green box. Since it is not an omnidirectional source, the SPL on axis is different than the SPL on axis by the omnidirectional ISO 3382-3 source.



2. Although the power spectrum is the same with the ISO 3382-3 source, the directivity pattern is not. While the is omnidirectional, the BB93_NORMAL_NATURAL.SO8 has a directivity pattern as shown in the picture on the right. Therefore, the same power spectrum is distributed differently. According to Figure 21, the SPL on axis at 10m is as highlighted in the blue box.
3. To derive the power spectrum of an equivalent omnidirectional source we simply add 31 dB to each band of the SPL at 10m. This is according to the spherical-spread law – as illustrated also in Eq.(4). The final spectrum of such a source is the one used in Figure 21.
4. The equivalent BB93_NORMAL_NATURAL.SO8 omnidirectional source is a bit louder than the ISO 3382-3 by 1.3 to 5 dB.

This process of deriving an equivalent BB93_NORMAL_NATURAL.SO8 omnidirectional source is used only for calibrating the system and not to change the directivity pattern of the source. The actual *artificial mouth* has its own directivity, which is taken into account by its geometry. To perform the calibration of the system, we can only use the *free-field* method by placing the microphone on axis in order to measure the direct sound in front of the *artificial mouth*. Since the artificial mouth is not an omnidirectional source, the *diffuse-field* method cannot be used.

If an anechoic chamber is not available for the *free-field* method, another dry environment can be tested. In this case, extra care has to be taken, so that the microphone is placed on axis at a sufficient distance and height in order for enough separation to occur between the direct sound the first reflection (see Figure 9).

In contrast to calibration with an Omni source (eg. dodecahedron) the distance between the *artificial mouth* and the microphone might not need to be >3m. This is because the loudspeaker driver used is much smaller – **therefore distances >1m can be considered already in far field**.

Based on the source/receiver height and the distance between them, ODEON derives a truncation point for the calculation of SPL. The truncation is done in the **broadband signal**. This is because the octave-band response is already contaminated by filter ringing, which is more visible at low frequencies. Always check that the whole broadband impulse response, is included between the onset and truncation times, defined by the *pink* and *red* dashed vertical lines. In Figure 22: the screenshot on the left shows a broadband impulse response truncated too early so that the response at lower-frequency bands is cropped. This truncation point corresponds to:

Source height: 1m.

Receiver height: 1m.

Distance between Source and Receiver: 1 m.

The screenshot on the right shows a broadband impulse response adequately truncated at 12 ms, with the following geometry details:

Source height: 2m.

Receiver height: 2m.

Distance between Source and Receiver: 1 m.

As can be seen in Figure 22, the broadband impulse response is not a perfect dirac response, as it would be expected when only the direct sound is measured. This is due to the fact that the source in the *artifical mouth* itself cannot move fast enough to produce a perfect dirac response, and due to distortion in the movement. Therefore, depending on the quality of the *artifical mouth* the impulse response ringing might be shorter or longer. The geometry of source and receiver can be adjusted accordingly.

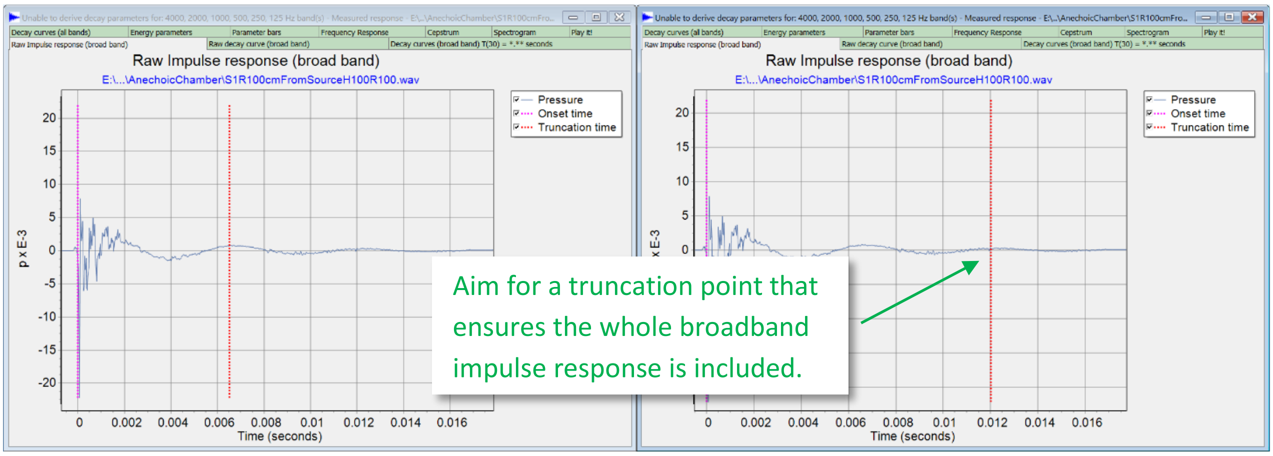


Figure 22: Always check that the broadband impulse response is well included between the Onset and Truncation times by ODEON. If not, increase source-receiver heights and if possible, decreased source-receiver distance (not closer than 1m).

9. Conclusion

The application note describes the steps required for a calibrated measurement of *G* and *STI* using two methods described in the ISO 3382-1 standard for room acoustics: the diffuse-field method (performed in a reverberation chamber) and the free-field method (performed in an anechoic chamber). The sound field for both environments is considered to be fully known, as long as specific assumptions hold (diffuse field and free field respectively). For this reason the sound power of an unknown source can be specified from analytic expressions like eq.(7) and (13) and calibration of the equipment towards a known source like G-ISO 3382-1 and Speech-ISO 3382-3 can be done.

To prevent damaging the calibration by accidental or intentional changes in gain and connections in the equipment, a Two-step method is proposed. This simply measures the source output at a fixed distance so that any further changes can be tracked back.

References

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2. ISO standard 3382-1, Acoustics – Measurement of Room Acoustic Parameters – Part 1: Performance Places.
3. ISO standard 3382-2, Acoustics – Measurement of Room Acoustic Parameters – Part 2: Reverberation Time in Ordinary Rooms.
4. ISO standard 3382-3, Acoustics – Measurement of Room Acoustic Parameters – Part 2: Open-plan offices.
5. IEC standard 60268-16, Sound system equipment – Part 16: Objective rating of speech indelibility by speech transmission index.
6. Claus Lynge Christensen. Impulse Response Measurements, ODEON video tutorials: https://odeon.dk/learn/video-tutorials/measurement-system/.