ODEON APPLICATION NOTE
Impulse Response Measurement Calibrations

GK – revised in November 2025

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, inside a *Reverberation* and an *Anechoic* chamber respectively. 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

The sound strength G parameter quantifies the influence of the room in the perceived loudness and describes the amplification applied to the source by the room at a particular receiver position.

According to the ISO standard 3382-1, for performance spaces [2], the sound strength G of an omni-directional source, for a **specific frequency (or band)**, 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), G is calculated as the difference between L_{pE} and $L_{pE,10}$, which have to be measured by a microphone calibrated with a pistonphone (usually providing 94 dB at 1kHz). Such a calibrated measurement provides so-called absolute Sound Pressure Levels (SPL).

In ODEON such absolute measurements are not possible - but even if they were, the measurement of $L_{pE,10}$ would require a large free-field, to position source and microphone 10 m apart. This is an impractically long distance. Fortunately, we can apply a workaround to calculate the difference between L_{pE} and $L_{pE,10}$ relatively, without knowing the absolute values of the individual terms. The simplest version of the technique is described in ISO 3382-1 and utilizes two well-defined environments: A *reverberation* or an *anechoic* chamber. This makes it possible to calculate a relative value of $L_{pE,10}$ in one of the two chambers and subtract it from a relative value of L_{pE} in the measured room, as long as the measuring equipment and all internal levels remain the same.

2.1 Calculation of $L_{pE,10}$ in a Reverberation Chamber (diffuse-field calibration)

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 $S_0 = 1 \text{ m}^2$.

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

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

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:

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

The absorption area can be calculated from the Sabine's formula (diffuse field assumption): $A = 0.16V/T$, with V being the volume (m²) and T the reverberation time (sec) of the reverberation chamber. Substituting this and $S_0 = 1 \text{ m}^2$ in equation (5) we get:

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

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

Substituting $L_{pE,10}$ to formula (2) we get the calculation of G with respect to the measured levels inside the reverberation chamber:

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

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.

2.2 Calculation of $L_{pE,10}$ in an Anechoic Chamber (free-field calibration)

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 $\geq$ 3m) and L_{pE}^{Anech} is the spatial- averaged sound pressure exposure level at every 12.5º around the source. At 10 meters from the unknown source the sound pressure level becomes:

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

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:

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

Similar to the Reverberation chamber method, substituting $L_{pE,10}$ to Eq. (2) gives a formula for G with respect to the measured levels in an anechoic chamber:

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

2.3 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 ODEON 16, 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).

3. General calibration formula

3.1 Single/One-step calibration

Eq. (7) and (13) are very similar in the calculation of G, and for convenience they can be packed in the following generic formula:

$$G = L_{pE} + \Delta L^{Calib} \quad \text{dB} \qquad (14)$$

where:

$\Delta L^{Calib} = -L_{pE}^{RevCh} + 37 - 10\log_{10}\frac{0.16V}{T}$, if using the *Reverberation chamber* method.

or

$\Delta L^{Calib} = -L_{pE}^{Anech} - 20\log_{10}(d) + 20$, if using the *Anechoic chamber* method.

This means that using ODEON, we can calculate a relative level L_{pE} from an impulse response and add a calibration adjustment ΔL^{Calib} to convert it to a G value. This is performed for each octave band separately.

3.2 Two-step calibration

Eq. (14) can be used only when the measuring equipment and all settings have remained fixed between the calibration chamber and the actual measurement room. 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 a long time before the actual measurement. In this section we introduce the concept of *two-step calibration* which provides 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 *two-step calibration* is optional, but it is recommended for its unique benefits:

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 the source and the microphone stay the same.

The two-step calibration consists of a reference impulse response measurement inside the calibration chamber (reverberation or anechoic) and an impulse response measurement in-situ. These two

measurements are performed at the same distance and direction from the source, in order to truncate both impulse responses at the same well-defined part of the direct sound. This part is eventually independent of the room and exclusively dependent on the source. The purpose is to derive any level difference that has occurred in the final output of the source (and equipment), and compensate for that. The process is illustrated in Figure 1.

Eq. (14) can be extended with two extra terms, $L_{pE}^{S,Ref}$, which is the relative level measured at a fixed point (distance and direction) in front of the source inside the calibration chamber and $L_{pE}^{S,Insitu}$ at the same point when the source has been moved to the actual measurement room:

$$G = L_{pE} + \Delta L^{Calib} + \left(L_{pE}^{S,Ref} - L_{pE}^{S,Insitu}\right) \quad \text{dB} \qquad (15)$$

In the *two-step calibration* 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. The change is registered by the difference $L_{pE}^{S,Ref} - L_{pE}^{S,Insitu}$, which is zero if no change has occurred, and Eq.(15) becomes identical to Eq.(14).
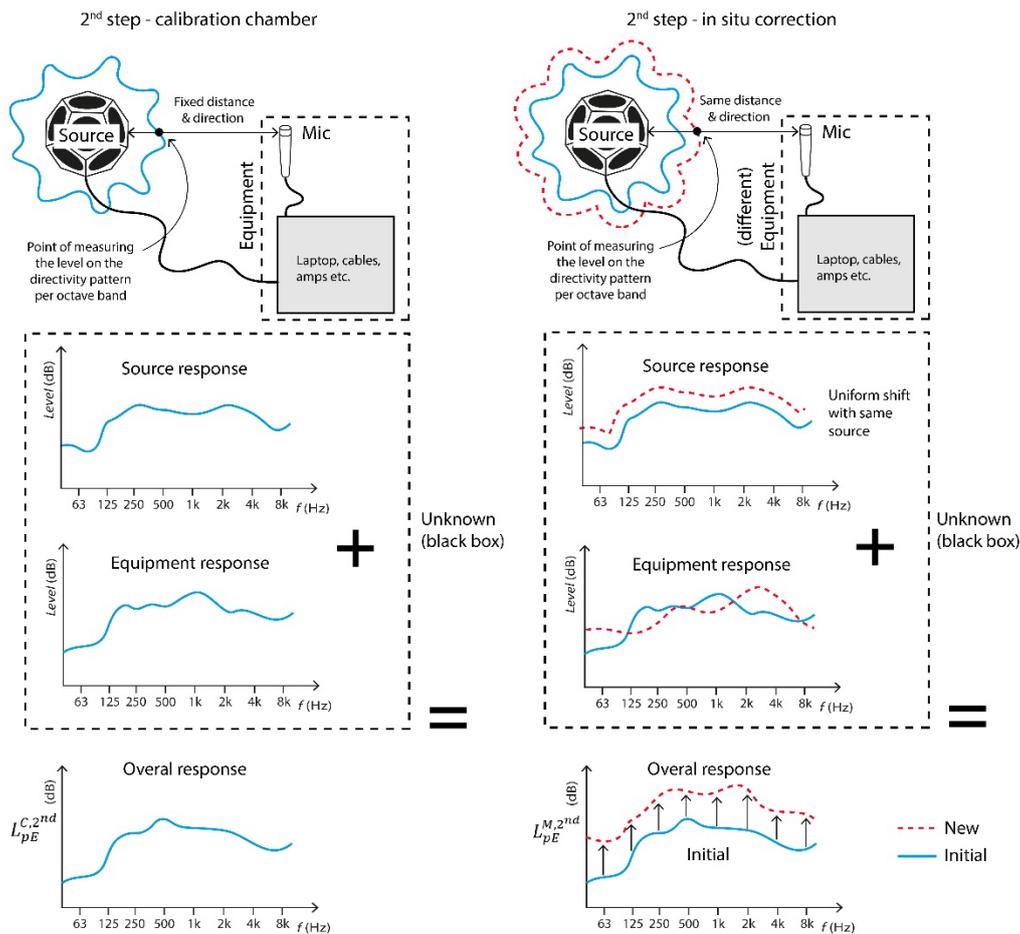

Figure 1: Changing any of the gains in the measurement chain does not necessarily shift the frequency response of the source linearly. All eight octave bands are taken into account separately, with eight correction factor values $L_{p}^{S,Ref} - L_{p}^{S,Insitu}$ derived.

4. Equipment for calibrations

4.1 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 (e.g. a big garage or hall). However, the chosen room should be as diffuse as possible meaning that highly symmetric shapes (e.g. 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 always check for the degree of linearity of the decay curve in each octave band through one of the *XI* parameters. If *XI* exceeds 10 $^o/_{oo}$ 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.

4.2 Hardware and Software

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

4.3 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. The Source Power Spectrum can also remain unchanged. ODEON 16 and later will always provide G as a separate parameter, taking automatically into account a source of power spectrum 31 dB/Octave. By default, ODEON will not show these values if no calibration file has been applied. However, you may check the box at the bottom left to **show uncalibrated parameters**. For G-parameter, no background noise is taken into account.
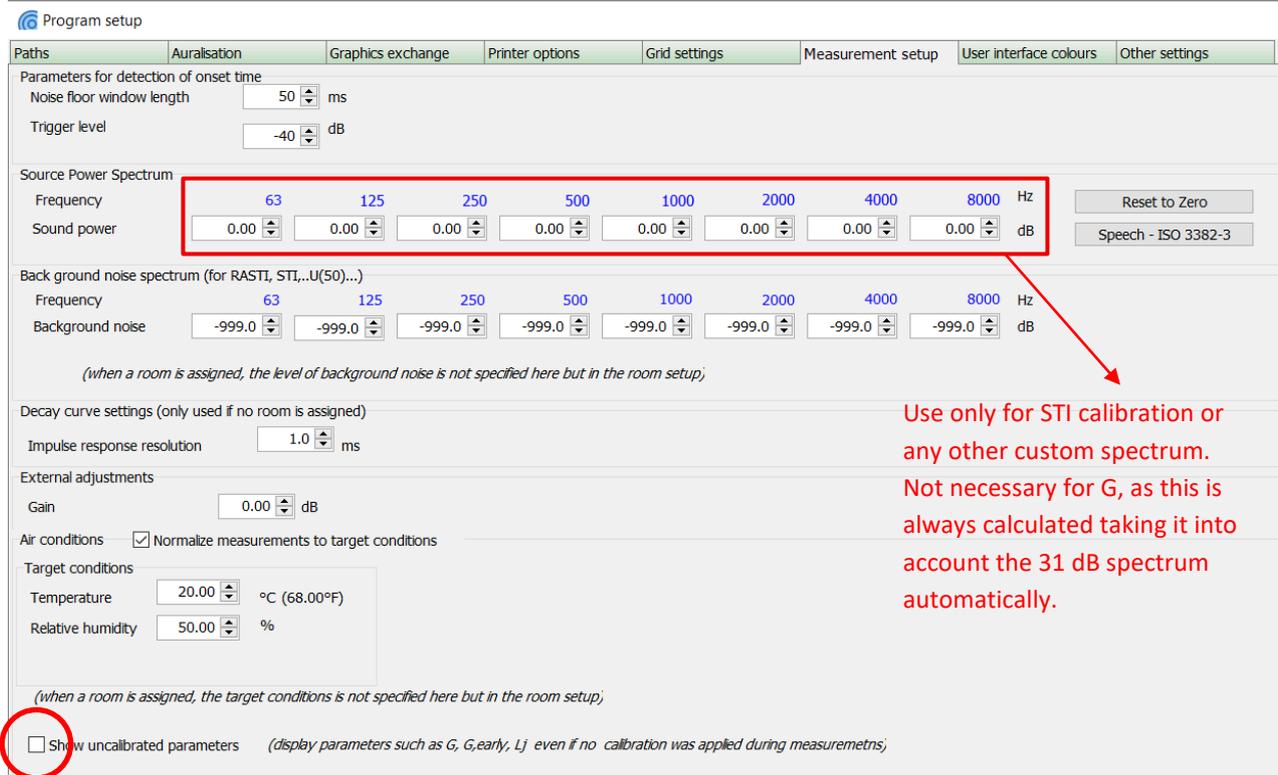
Validation exercise: As mentioned already in Sec. 2, the calculation of *G* is based on the assumption that the source power spectrum is 31 dB/Octave band. If you manually insert that power spectrum in Options>Program Setup>Measurement Setup, then the SPL and G values will become identical, because *G* is basically the SPL for an omni-directional source of 31 dB/Octave band (Figure 2). In other words, at 10m from the source the SPL becomes zero in free field ($L_{pE,10} = 0\ dB$) . 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.

Set the sweep parameters

Open the Measure impulse response (sinusoidal sweep) interface from the main toolbar:
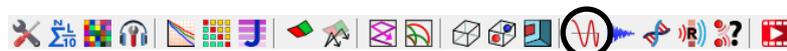
The interface looks like in Figure 4. The following warning message might be displayed at the bottom of the window, but you can ignore this time: "*WARNING! Calibration file: "" not found. No calibration applied*"
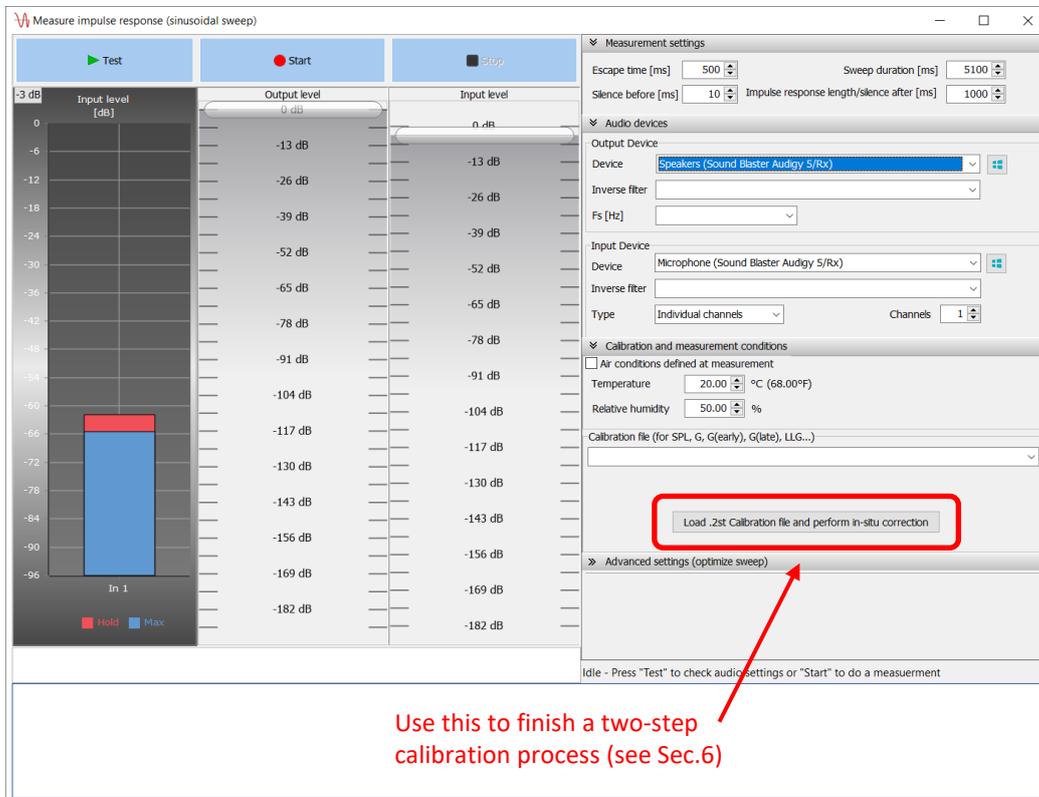

Figure 4: The Measure Impulse response interface, as it looks in ODEON 19.

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.

> **Tip:** To determine the required sweep duration, remember that every doubling of the sweep length results in approximately **3 dB of noise suppression**, with an equal improvement in the usable decay range. For example, if you need **9 dB of overall noise suppression**, you must increase the sweep length by a factor of 8 (!)

Audio devices

In the *Audio devices* panel the input and output devices should be chosen. According to Section 3, an omnidirectional **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.

Calibration and measurement conditions
Air conditions should be specified if the set of measurements is to be compared to another set a later or previous time, with the temperature and humidity of the space possibly having changed.

In this section you can specify the active calibration file for the current measurement or load a two-step (.2st) Calibration file (see Sec.6) to perform an in-situ correction and derive the final calibration file.

Advanced settings (optimize sweep)
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.

Adjusting the levels

The most important requirement for a calibrated measurement is that all external level (gain) adjustments must be set to fixed values for all measurements in each of the calibration and measurement sessions. If a one-step calibration is used (see Sec. 5) the levels have to remain strictly the same in both sessions. If a two-step calibration has been used (see Sec. 6), the levels may differ between the sessions.

> **NOTE:** 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).

> **CAUTION!** High output gains may overload the loudspeaker, causing distortion or even permanent damage. In many cases, the amplifier is more powerful than the speaker, so only a much lower gain is required. This value should always be **clearly documented for reference**.
> Excessive sound power is not necessary, as long as the equipment can achieve a sufficient **signal-to-noise ratio** (SPL/Noise parameter in ODEON) or a usable **decay range** (e.g. 45 dB) under normal room conditions with moderate ambient noise.
> Likewise, the **microphone gain** should be adjusted carefully, ensuring that the internal noise level remains acceptably low.

Exclusive mode

In ODEON 19 exclusive mode (supported from the WASAPI audio driver) is available only for the *input devices*. This is because enabling it for output devices systematically resulted in several conflicts with input devices, so that several combinations could not be selected. **Therefore, from version 19, exclusive mode for output devices is permanently disabled and any setting in the device properties is overridden.** For the input device, exclusive mode is enabled by default and results in the following benefits:

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.

The setting can be activated/deactivated by accessing the device's properties from the button next to Input device dropdown. As it can be seen in the figure below, moving the level slider for the Line-in device moves the Input level slider in ODEON simultaneously.
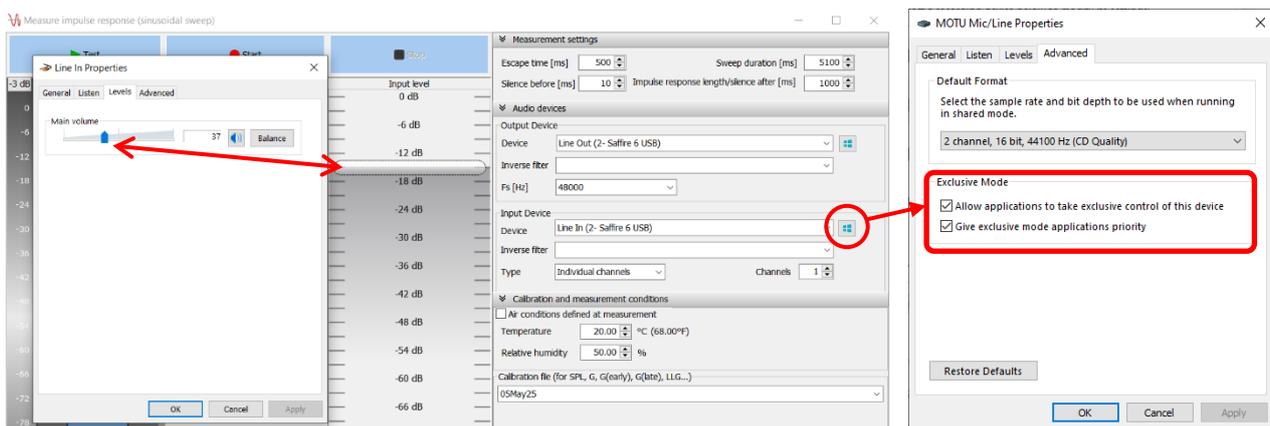

Figure 5: Exclusive mode ensures levels in ODEON and Windows move together and no other applications use the audio devices.

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 this page. 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, especially at high levels!

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 that no value of the decay range can be displayed.

4.4   The calibration interface

Since ODEON 19, both the *diffuse-field* and the *free-field* calibration procedures can be performed in a re-designed user-friendly interface, that offers a thorough overview of the selected files and calibration steps and includes a minimal number of popup dialogs. The new interface can be found at Tools>Measurement Calibration>Diffuse field or Free field.
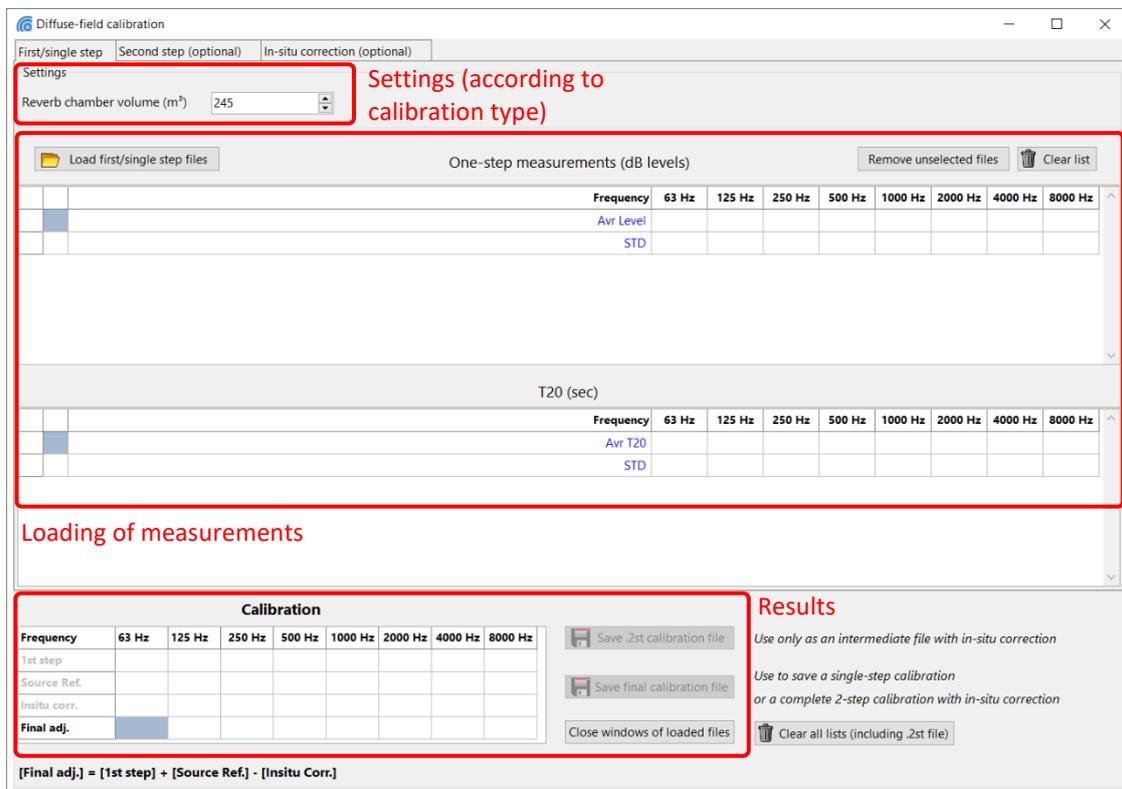

Figure 6: The Calibration interface is separated into three main areas: *Settings*, *Loading of measurements* and *Results and saving*.

The interface consists of three main areas: *Settings* (where we can set the geometric details depending on the type of calibration), *Loading of measurements* area (where all calibration measurements are loaded) and *Results and saving* (where the calculated levels are displayed and can be saved to the final calibration file). The top of the interface is further divided into **three tabsheets**:

1) *First/single step*, which is the main part of the calibration process and can be used without completing the next two parts.
2) *Second step (optional)*, which adds extra versatility to the procedure, by reporting the levels of the source at the moment of calibration.
3) *In-situ correction (optional)*, which works together with the Second step during the actual measurements in-situ. It is possible to perform an *in-situ correction* directly from Tools>Measurement

Calibration>In-situ correction. In this case, the other two tabsheets become invisible, because it is assumed that the first two steps have been completed already.

As opposed to the older implementation of calibration in ODEON, the newer interface offers great versatility in loading the files and processing them: you can load one-step or two-step files in any order, you can add extra files to the already loaded ones and you can deactivate/active specific files in the lists. The steps for deriving both a *Diffuse-field* and the a *Free-field* calibration are similar, what changes is how the calibration measurements are obtained and how they are processed in the interface.

5. Making a Single-step Diffuse-field calibration

In this section we will follow the diffuse-field calibration process with a step-by-step example, using one of the *reverberation chambers* at the Technical University of Denmark and *Auditorium 21* at the same campus. See more details in 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 m³ 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.

5.1 Make sure all external levels are fixed

As a first step, set all external gains (input/output volume in soundcard, amplifiers etc.) to a fixed value. Also make sure not to change any connections in the equipment during the same measurement session, as each cable/adaptor has its own impedance and can affect the input/output levels.
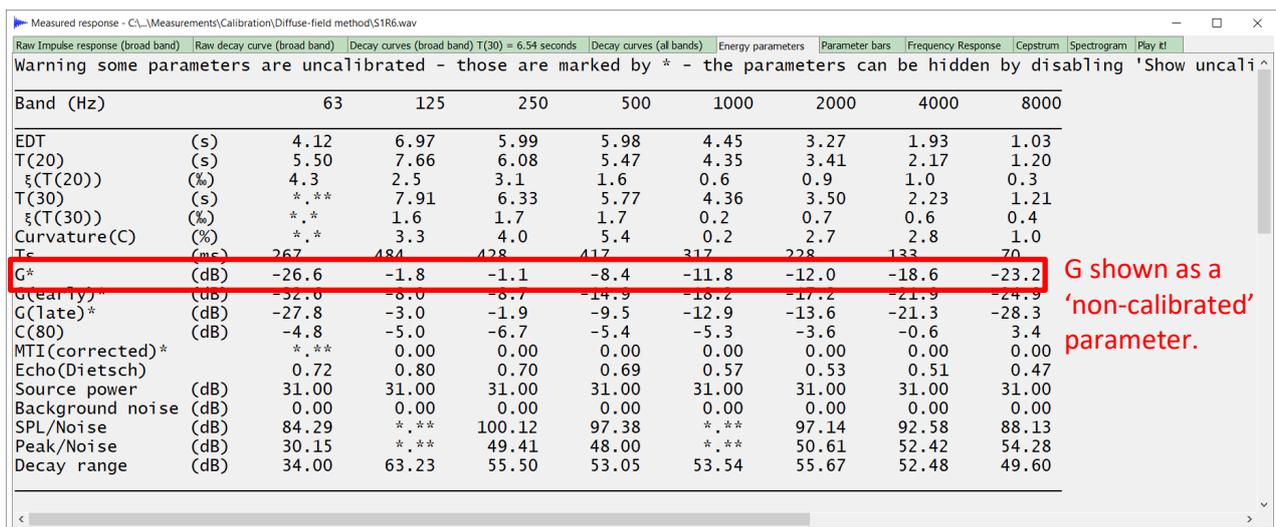
If you are *not* planning to perform a two-step calibration (described in Sec. 8) then make sure that external levels and cables remain fixed both in the Reverberation Chamber and the actual room under measurement.

Remember: While external gains should not be changed, you can always change the internal levels in the ODEON sweep interface 〰 during the same series of measurements, without damaging the calibration process.

5.2 Take 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)** has some abstract value, as the system is not calibrated yet. In the *diffuse-field calibration* you should always check that all **Reverberation, T$_{20}$** 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 (e.g. 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.

> **NOTE:** You can find an example of measurements inside the reverberation chamber in the Measurements\Calibration\Diffuse-field method folder of Odeon's default installation folder.


Figure 7: 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. To be able to see the 'non-calibrated' G values (marked by *), make sure to check the '**Show uncalibrated parameters**' option in the Options>Program setup> Measurement setup.

5.3 Create the calibration file

Once all measurements in the *reverberation chamber* have been completed, follow the next steps to create the calibration file. Here we use the measurements coming with the ODEON installation in the Measurements\Calibration\Diffuse-field method folder:

1) Open the calibration interface by clicking Tools>Measurement Calibration>Diffuse field.
2) For this example, make sure that the Reverb. chamber volume is 245 m^3, which is the default value in the ODEON installation.
3) Click the button Load first/single step files to load the measurements obtained in the previous section. In the dialog that pops up, you may select all files from S1R1 to S1R12 at once, by holding the SHIFT or CTRL key.

Measurement Calibrations
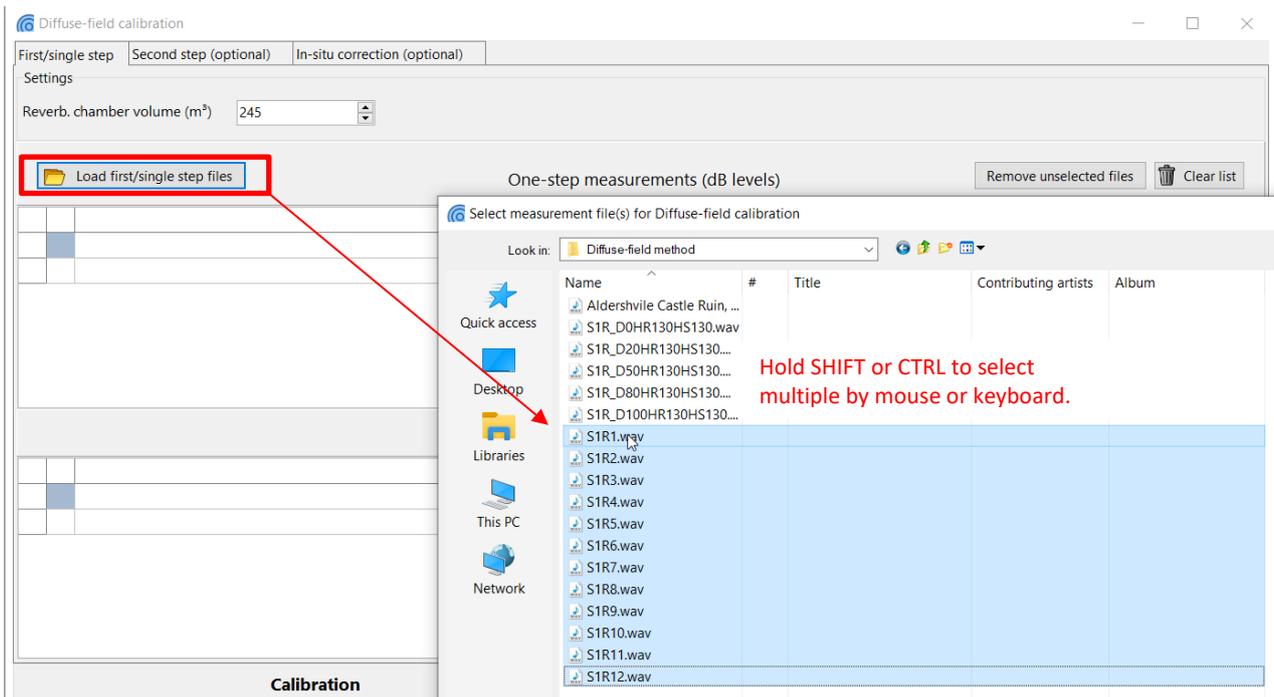
4) The selected files are loaded in separate windows automatically in the usual Load Impulse response interface. Note that the particular impulse responses above are relatively long, so it takes some time to load all of them. The corresponding levels and reverberation times (T_{20}) are displayed in the One-step measurements table, together with their average and Standard Deviation (STD). The calibration values are shown in the Calibration table. According to Eq.(15) the final calibration adjustment is the **sum** of the one-step calibration values plus any possible values from the two-step calibration (source reference and in-situ correction).

5) At this point the *two-step calibration* values are empty, but we can simply save a *one-step/final calibration* file, by clicking the Save calibration file button. This is the simplest version of calibration which however requires that the equipment and gains remain fixed (as described in Sec. 3.1).

6) You will be prompted to set the calibration file as the active one, which is going to be used with the following measurements. It is possible to change the active calibration file at a later point in the Measure impulse response interface 〰 (SHIFT+CTRL+D).

7) You can now launch the Measure impulse response interface to check whether the active calibration file is the correct one.

Note: Always inspect the loaded measurements to ensure that the values are close enough and that there are no outliers per octave band. Aim for a standard deviation (STD) below 1 for most of the bands or up to slightly above 1 for the lowest bands, since these ones are inherently more unstable to measure.
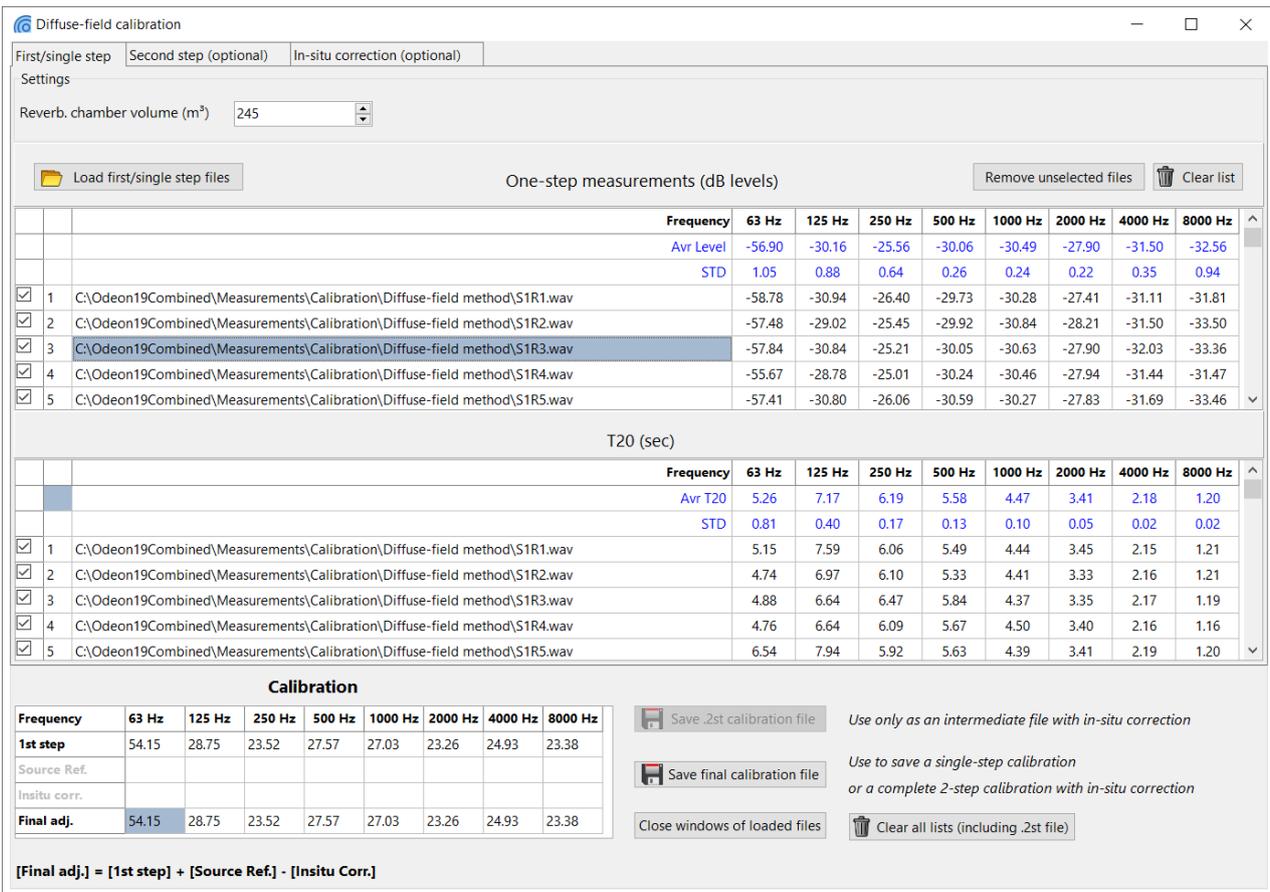
Figure 8: The interface after *one-step* calibration files have been loaded. The relative levels and T$_{20}$ values are displayed, together with the calculated calibration values at the bottom of the window.
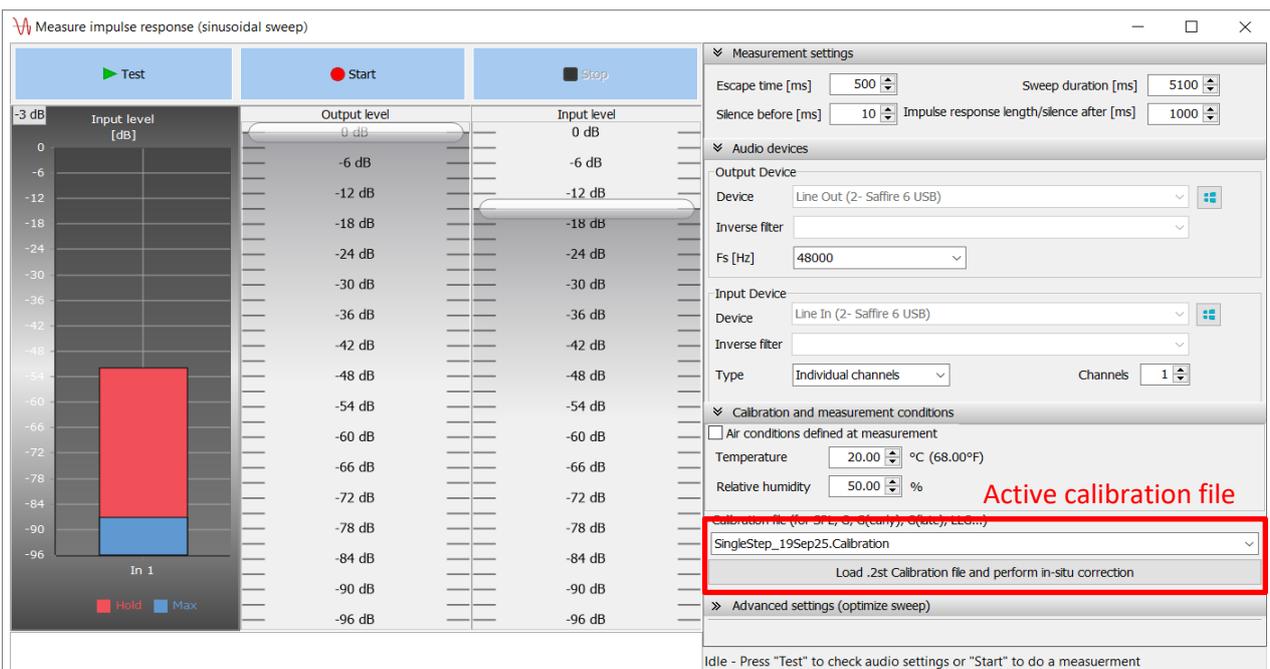

Figure 9: The Measure impulse response interface with the active one-step calibration file.
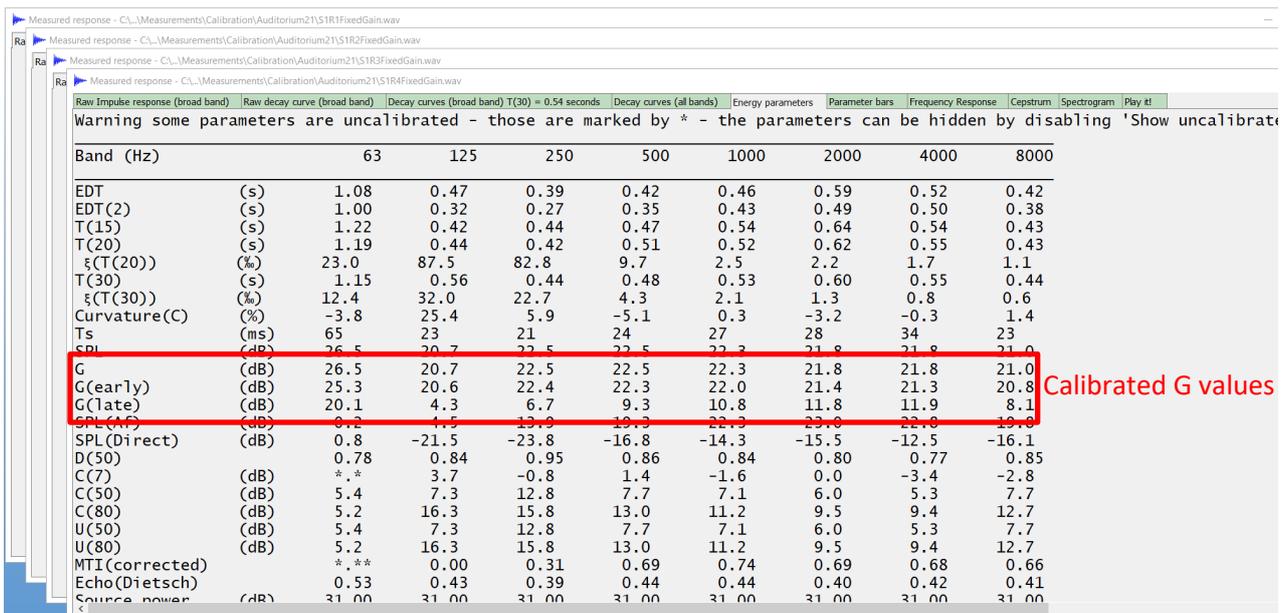
5.4   Performing calibrated measurements

Measurements can be performed with the Measure Impulse Response (Sinusoidal Sweep) tool 〰 (SHIFT+CTRL+D), as shown in Sec. 4. The active calibration file will be applied once the recorded Impulse response .WAV files are loaded with the Load impulse response tool ▬ (SHIFT+CTRL+L). 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 measurements

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. If there is an input/output device mismatch, a warning message will be displayed. The details of input/output devices as well as the name of the calibration file will be listed at the end of the 'Energy parameters' tabsheet in the Measured response interface.

You can test the function yourself with the calibration file from Figure 8 and the measurement files in Measurements\Calibration\Auditorium21 folder. For this example, use only the files labelled as 'FixedGain'. The results are shown in Figure 10.

Figure 10: Diffuse-field calibration applied to four impulse response measurements inside Auditorium 21.

Remove calibration from existing measurements

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. Two-step calibration

6.1 Overview

As mentioned in Sec. 3.2, the one-step calibration can be extended to a two-step calibration, to greatly increase the versatility of our settings and add extra safety in terms of accidental changes in the equipment and gains. According to Eq.(15), the second step consists of registering two measurements of the level at the same distance in front of the source, excluding any reflections from the surrounding room (e.g. the reverberation chamber).

Source reference measurement

After we have finished all the one-step measurements inside the *Reverberation chamber*, we place the microphone at a specific distance d from the source and obtain a new impulse response in order to be able to derive the source reference level $L_p^{S,Ref}$. From now on, any deviation in the source level will be trackable.

> **Note:** It is not mandatory to perform this step in the *reverberation chamber*, since only the direct sound from the source is of interest. However, keeping the setup in the chamber is good practice, as it ensures all levels and equipment settings remain unchanged. Moreover, a large, empty chamber minimizes nearby objects and thus reduces the risk of unwanted reflections.

In-situ correction measurement

During the actual (in-situ) measurements we place the microphone at the same distance d and preferably in front of the same point from the source as we did in step 1. We obtain a new impulse response to derive the in-situ correction $L_p^{S,Insitu}$.

Figure 11 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 $L_p^{S,Ref}$ and the $L_p^{S,Insitu}$ might not be linear (see Figure 1). This is why Eq. (15) is calculated independently for each octave band.

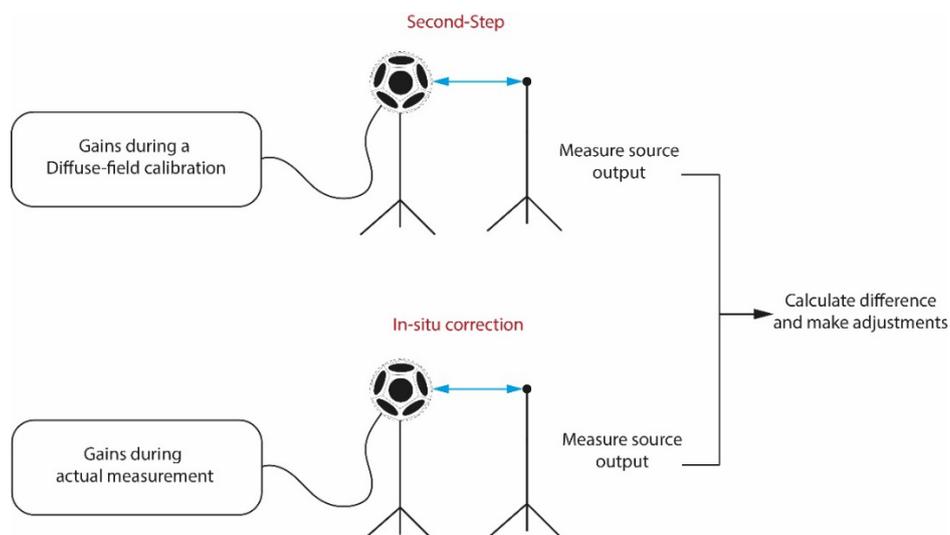**Figure 11:** The concept of a Two-step calibration. When all recordings for the main diffuse-field or free-field calibration have been completed, one extra recording is taken still in the same calibration room at a specific distance from the source. The same recording is taken afterwards in the actual room. If any change has been made in the gains of the equipment it will be found as a supplementary adjustment.

6.2 Best practices for obtaining the 2nd-step measurements

It is important to understand that in the 2nd-step measurements (both the *reference* and *in-situ*), we need to distinguish the direct sound from the remaining reflections in the room. Therefore, the microphone should be placed as close as possible to the source. Figure 11 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. 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.
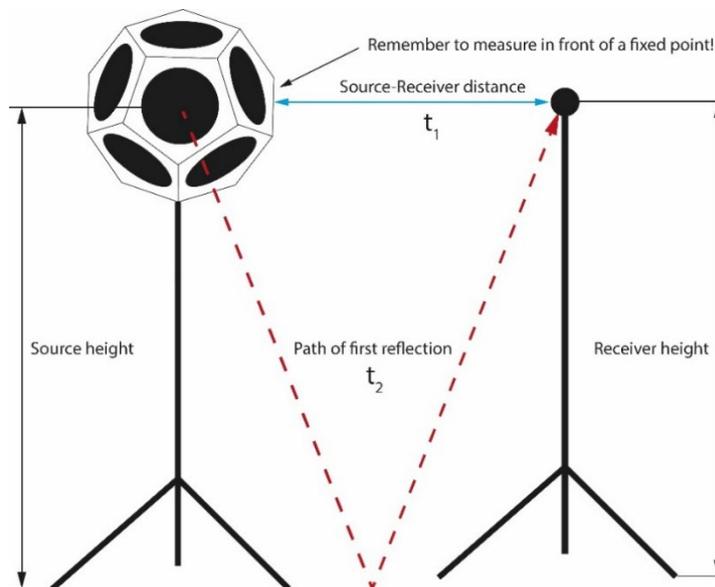
Figure 12: Geometry for impulse response recording for the second step. ODEON needs to have sufficient time difference between the arrival of the 1st reflection and the direct sound (t_2-t_1). This is accomplished by having high source-microphone heights and short distance between source and microphone.

The following values can ensure isolation of the direct sound and are default in the ODEON Calibration interface.

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

Note: Although only one measurement in front of the source is enough, an average over several measurements will minimize any errors that occur from inaccurate placement of the microphone. **Either of the following is recommended:**
1) Place the microphone again at the same distance from the fixed point on the source and repeat the measurement. Do this for at least 4 times.
2) If the source is completely omni-directional at the horizontal plane (e.g. Odeon Omni) place the microphone at a fixed radius around the source and repeat for at least 4 different angles.

6.3   Inserting the values in the Calibration interface

We can now go back to the example from Sec. 5.2 and the interface as shown in Figure 6 (with the First-step files loaded). We will load the following two files:

1) **In the *Second step (optional)* tabsheet:** Source reference file S1R_D80HR130HS130.wav from the folder Measurements\Calibration\Diffuse-field method. Use the Load second step (source ref.) files button. An information message will appear to remind that the same point for measurement should be used both for the s*econd step* and the *in-situ correction*. Click OK to continue.

2) **In the In-situ correction (*optional)* tabsheet:** In-situ correction file S1R_80cmFromSource_ChangedGain.wav from the folder Measurements\Calibration\Auditorium21. Use the Load in-situ corr. files button. An information message will appear again to remind that the same point for measurement should be used both for the s*econd step* and the *in-situ correction*. Click OK to continue.

The interface with all the data should now look like in Figure 13. Note that the 'Calibration' table summarizes all the adjustments needed for Eq. (15), i.e. the *1st step* adjustments ΔL^{Calib}, the *source reference measurements* $L_{pE}^{S,Ref}$ and the average in-situ correction measurements $L_{pE}^{S,Insitu}$. If more than one files have been loaded, the average is used. The Sum row at the end is therefore given as below:

$$Sum = \Delta L^{Calib} + \ L_{pE}^{S,Ref} - L_{pE}^{S,Insitu} \quad \text{dB} \tag{16}$$

6.4   Saving – loading calibration files

You can now click on the Save calibration file button to save the full calibration file that takes into account the *1st step* values as well as the *source reference* and *in-situ* corrections (*Final adj.* line in the Calibration table in Figure 13). You may also save a .2st calibration file which includes both the *1st step* values and the *source reference*, but not the in-situ correction. In this case you can save all the calibration progress so far in the reverberation chamber and load the .2st calibration file together with an in-situ correction measurement.

At this point we can see that the *in-situ correction* measurement was approximately **5 dB higher** than the *source reference* for each octave band*.* This corresponds to an increase in the output gain by the same amount. It is not known precisely where this increase took place within the measuring chain (e.g. Windows volume, audio interface potentiometer) but this information is not important. What matters is only the final output measured by the microphone at the same distance in front of the source (Figure 10).

Application note
Measurement Calibrations
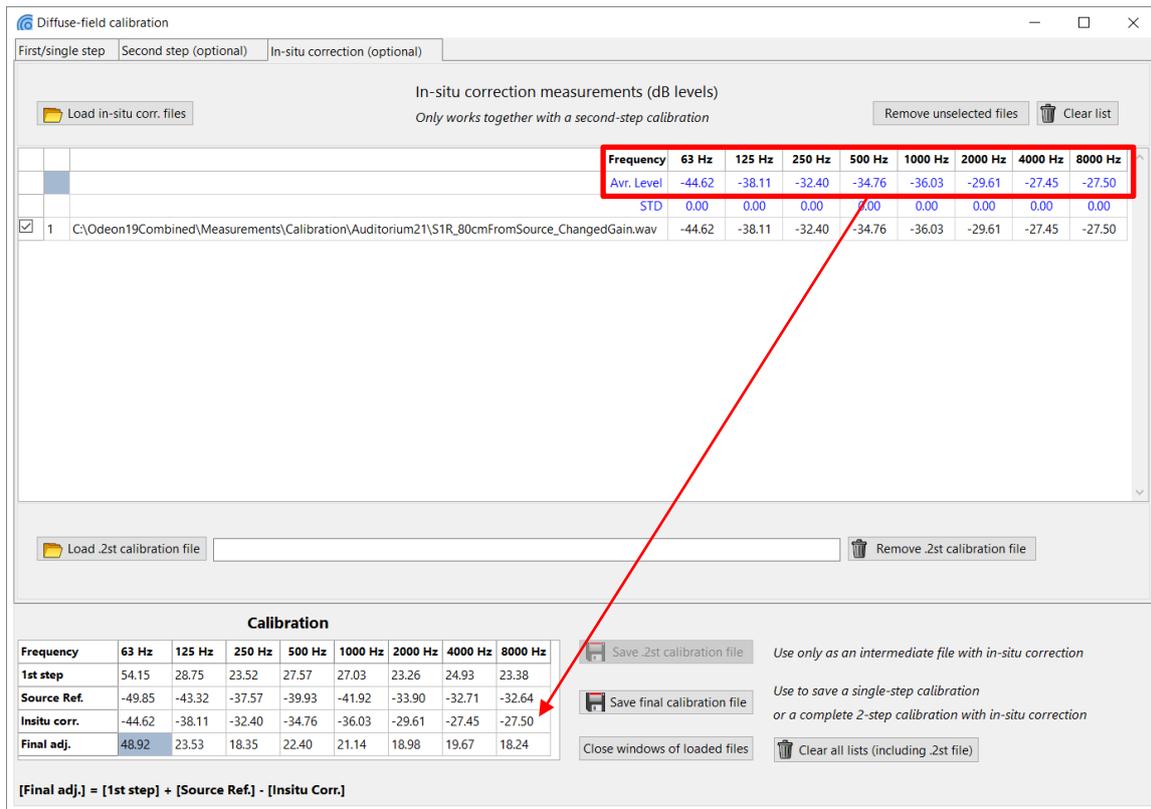
Figure 13 The interface after *two-step* calibration files have been loaded. 1st step calibration values, source reference as well as in-situ correction have been derived. The 'Sum' row at the end of the 'Calibration' table gives the final values from Eq. (16). The negative sign for the in-situ correction has been included already in the corresponding row.

6.5   Summary of the workflow for Single-step and Two-step calibration

The following workflow chart summarizes the steps needed for a *Single* or *Two-step calibration*, as they have been presented so far. Both types of calibration lead eventually to the same file for use with measurements. The chart is based on the *Diffuse-field* method, but it can be readily applied to the *Free-field* method as well.

Calibration-measurement workflow
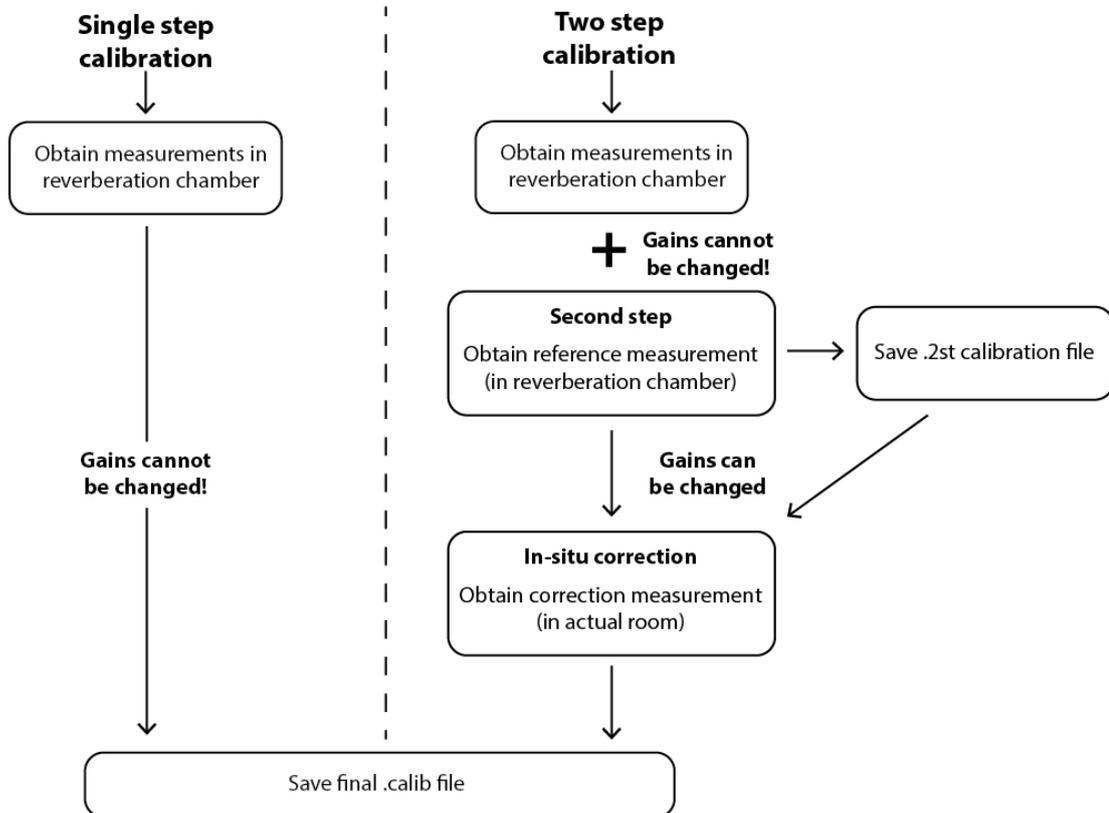
Figure 14 Workflow for derivation of a final calibration file using either a single step or two-step approach. The two-step method involves a few more measurements but it guarantees more reliable results. **Note in which parts of the workflow gains should remain unchanged and in which parts they can be changed**.

Note: In all steps you can freely change the internal gain in the ODEON's sweep generator without damaging the calibration. The restrictions apply only on the external gains in the measuring chain.

In addition to the workflow of Figure 14, there are two main possibilities in applying calibration to measurements. The most intuitive and straightforward way is to obtain a calibration file first and perform measurements afterwards with that file active. However, it also possible to apply a calibration file to existing measurements. Figure 15 shows the two possibilities.

Order of calibration & measurements

Posibility 1

Final .calib file

↓

Set calibration file active

+

Make measurements

Posibility 2

Make measurements

+

Final .calib file

↓

Apply calibration to existing measurements

Figure 15 The final calibration file can be used either before or after performing the measurements.

Possibility 1: Obtain calibration file first - Measure afterwards

As mentioned, this is the most typical and intuitive approach. The following actions are based on the complete two-step calibration. **For a single-step calibration, you only need to perform actions 1, 3 and 12**.
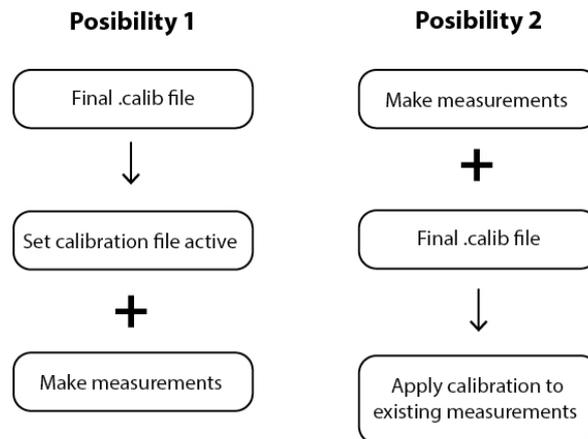
1) **Reverberation chamber measurements (First step):** Record impulse responses in the reverberation chamber. *Keep all external gains fixed.*
2) **Source reference measurement(s) (Second step):** Record one or more additional impulse response(s) close to the source at a fixed position (location & distance or radius). *Use exactly the same external gains as in Step 1.*
3) Open the Tools>Measurement Calibration>Diffuse Field interface and load the *First step* files into the First/Single step tabsheet, using the corresponding button.
4) Load the *Source reference* files into the Second step (optional) tabsheet.
5) Save a .2st calibration file for later use during in-situ measurements. You may now close the interface.
6) Move the equipment into the actual measurement room.
7) *External gains may now be adjusted* (or might have changed accidentally). Adjust them to achieve a good signal-to-noise ratio, then *keep them fixed for all remaining measurements*.
8) **In-situ correction measurement(s):** Record one or more impulse response(s) at exactly the same position used in Step 2. This will capture any gain changes in the output or input chain.
9) Open the calibration interface by choosing Tools>Measurement Calibration>In-situ correction or by clicking the Load .2st Calibration file and perform in-situ correction button in the Measure impulse response window 〰. In this mode, the first two tabsheets (Figure 13) are disabled.
10) When prompted, load the previously saved .2st calibration file file.
11) Load the in-situ correction measurement(s) obtained in Step 8, using the corresponding button.
12) Save the final calibration file using the corresponding button in the interface. You will be prompted to set it as the active file.

Note: If there is a mismatch between the input/output devices used in the final calibration file and the ones used for the measurement, a warning is displayed by ODEON and no calibration is applied.

The active file will be shown in the _{Measure impulse response} _(since sweep) interface 〰. You may also select another calibration file stored in the list. This will ensure that calibration will be applied to any measurements of G and STI.
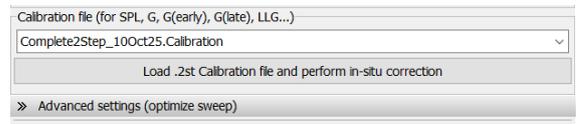
If there is any mismatch between the set input/output audio devices and the active calibration file, these will be highlighted in red colour. Note that sometimes the audio devices are physically the same, but they have been assigned different names due to changes of drivers, Windows language etc. If you are confident that the mismatch is because of a false system interpretation, you can click _{Reassign devices to calibration file} to resolve the mismatch. In the example below, the names of the mismatched devices are almost identical and can be safely forced to be matched.
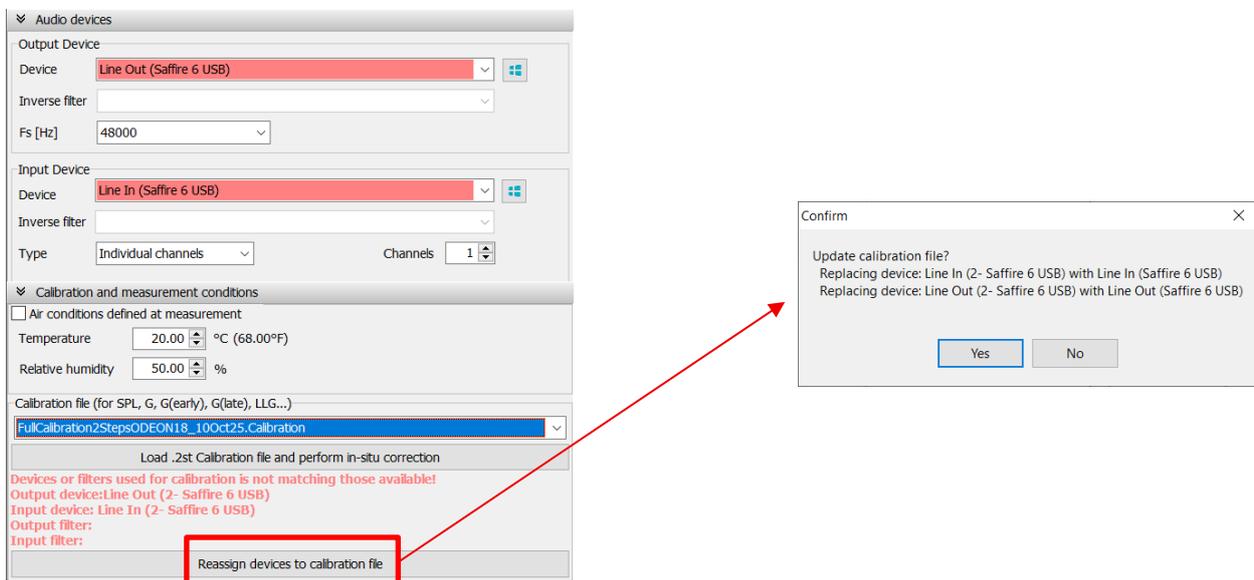
Possibility 2: Measure first – Apply calibration file afterwards

There might be time/planning reasons for which calibration should be performed after measurements. The following actions are based on the complete two-step calibration and are somewhat reversed, with the *in-situ* correction performed *before* the source reference measurements. **For a single-step calibration, you only need to perform actions 4, 6, 9 and 10**.

Room measurements (main set): Record impulse responses in the actual room with an overall input/output gain set for a high SNR, without clipping. *Keep all external gains fixed.*

1) **In-situ correction measurement(s):** Record one or more additional impulse response(s) close to the source at a fixed position (location & distance or radius). *Use exactly the same external gains as in Step 1.*

2) After completing all room measurements, relocate the setup to the reverberation chamber any time afterwards.

25

3) Adjust external gains as needed for the chamber environment, then *keep them fixed from this point onward.*

4) **Reverberation chamber measurements (1st step):** Record impulse responses in the reverberation chamber using the chamber gain settings from Step 4.

5) **Source reference measurement (2nd step):** Make one or more additional recordings at the same position used in Step 2.

6) Open the Tools>Measurement Calibration>Diffuse Field interface and load the *First step* files into the First/Single step tabsheet, using the corresponding button.

7) Load the *Source reference* file(s) into the Second step (optional) tabsheet.

8) Load the *In-situ correction* file(s)into the In-situ correction (optional) tabsheet. There is no need to create a separate .2st calibration file, since the in-situ correction was already obtained during the room measurement session.

9) Save the final calibration file by clicking the corresponding button at the bottom of the interface. You will be prompted to set it as the active file; since the actual measurements were performed earlier, you can select **No**.

10) Apply the saved calibration file to your measurements via Tools>Measurement Calibration>Assign Calibration to Existing Measurements.

Using Odeon Omni as a pre-calibrated source

Odeon Omni comes with a .2st calibration file which can be loaded directly into the calibration interface, so that only an in-situ correction has to be performed. In this way there is no need to access a special chamber. The pre-calibration file comes with the default ODEON installation, but it can be also found on this page. Make sure to store it in the Program data>Odeon>Calibrations folder.

> **Note:** At the moment **it is recommended to use a cable** when you perform calibrated measurements with Odeon Omni. The Bluetooth connection can occasionally become unstable, resulting in some variation of the levels in the final impulse response. This does not affect the rest of the Room Acoustic Parameters, but it can affect G and STI which depend strictly on the actual level of the impulse response.

To use the file, follow these steps:

1) Make one or more *in-situ* correction measurements(s) in the actual room, with a microphone-source distance/radius of **70 cm**. It is recommended to measure at least 4 to 8 points around the source, in the horizontal plane, to reduce any fluctuations in the results.

2) Open the Tools>Measurement Calibration>In-situ correction interface and load the file: DiffuseCalib_Average3Omnis.2StCalibration. A series of *1st Step* and *Source Ref.* values will appear directly in the 'Calibration' table, but they will remain inactive until *in-situ* correction measurements have been inserted.

3) Load the *in-situ* correction file(s), measured at **70 cm** radius. Now the 'Calibration' is complete.

4) Save the final calibration file using the corresponding button next to the 'Calibration' table. You may set it as the active one in the subsequent dialog.

7. Making a Single-step 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 photo description below. Apart from the differences specific to the anechoic chamber measurements, all remaining steps for deriving and applying the calibration are identical to those of the Diffuse-field method, for both the **single-step** and **two-step** calibration procedures (Sec. 5 and 6). 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.
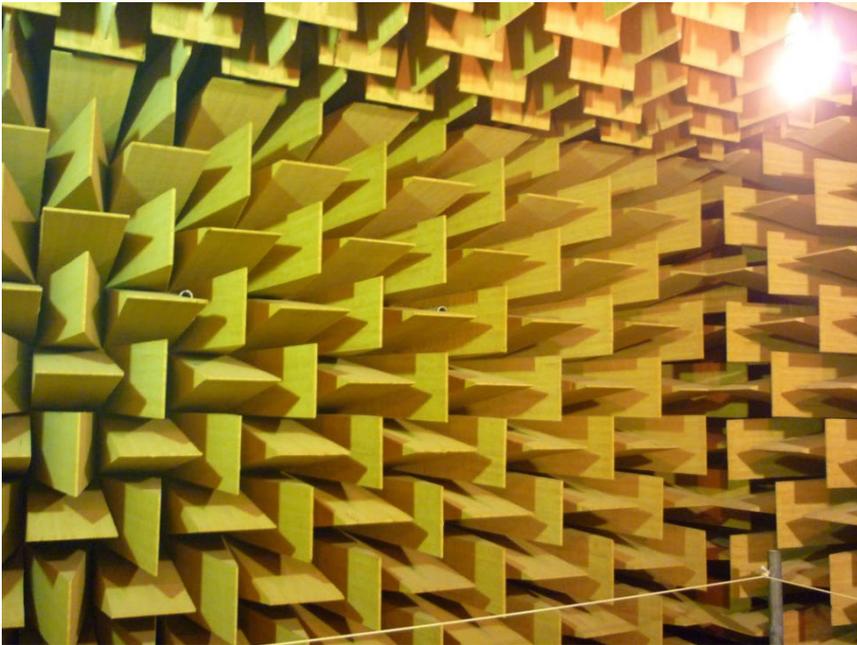
The free-field calibration should be done ideally in an anechoic chamber. For this application note, the **small anechoic chamber** at the Technical University of Denmark has been used. The volume of the room is about 60 m^3. Although the method theoretically requires an anechoic room (no reflections 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.

7.1 Make sure all external levels are fixed

As with the *diffuse-field method*, set all external gains (input/output volume in soundcard, amplifiers etc.) to a fixed value. Also make sure not to change any connections in the equipment during the same measurement session, as each cable/adaptor has its own impedance and can affect the input/output levels.

If you are *not* planning to perform a two-step calibration (described in Sec. 6), make sure that external levels and cables remain fixed both in the Anechoic chamber and the actual room measurements.

> **Remember:** While external gains should not be changed, you can always change the internal levels in the ODEON sweep interface ⋁⋁ during the same series of measurements, without damaging the calibration process.

7.2 Make measurements in the Anechoic Chamber

Here, 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 and should be maintained as precisely as possible throughout the measurements in the anechoic chamber. There are practically two possible setups to obtain these measurements.

Fixed source – Moving microphone

The main advantage of this method is that only conventional source and microphone stands are needed, together with a piece of rope. However, a large available area is required in the anechoic chamber:

1) The source must be fixed in the middle of the chamber or at least 4 m from the absorption wedges. This is to provide space for at **least 3 m distance** d (radius) between the microphone and the source (see Figure 16), plus at least 1 m between the microphone and the wedges.
2) The microphone (receiver) should be placed ideally at **minimum 30 spots around the source**, every 12°, for a full 360° circle. For a lower-precision measurement, 15 spots (every 24°) can be acceptable.
3) Ensure that the distance between source and receiver is as accurate as possible, as even small deviations can significantly affect the measured sound pressure level.
A practical method is to use a rope with loops at each end, attached to the source and microphone stands. By stretching the rope each time you reposition the microphone, you can maintain a consistent and repeatable distance from the source (see Figure 17).

Note: In a high-quality anechoic room there is no need for truncating the impulse response, since 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.
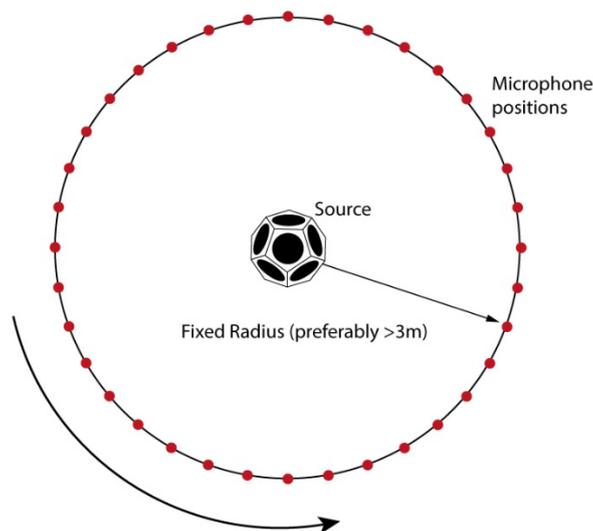

Figure 16: Diagram of measurement positions for free-field calibration. The source should be fixed roughly at the centre of the room and the microphone should be placed at a fixed radius around the source. Ideally 30 positions are required (red spots) but 15 is also an acceptable number. That will correspond to every 12° – 24°.
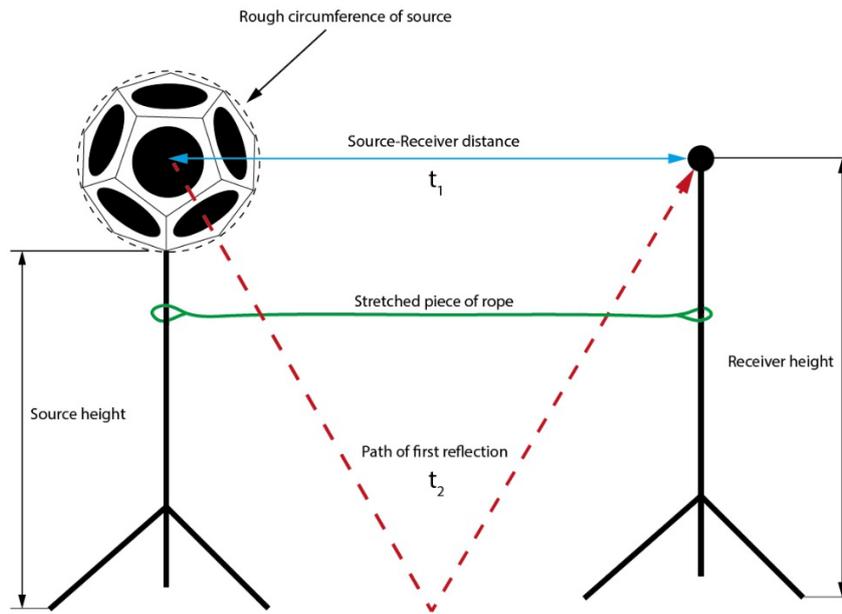
Figure 17: Geometry for impulse response recording in the anechoic/dry room. It is advisable to use a piece of stretched rope with loops to ensure constant distance between source-receiver (preferably <3m). ODEON needs to know the time difference between the arrival of the 1st reflection and the direct sound (t_2-t_1). This is calculated by entering the source and receiver heights, as well as the source-receiver distance.

Fixed microphone – rotating source

The above method requires sufficient space - ideally a radius of at least **3 m around the source**, plus a minimum clearance of **1 m from each wall**. This can be an impractical requirement. An alternative method is to fix the microphone position and rotate the source on a turntable instead. It could be argued that the first method is more accurate, as it averages 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 present method requires far less space (see Figure 18).
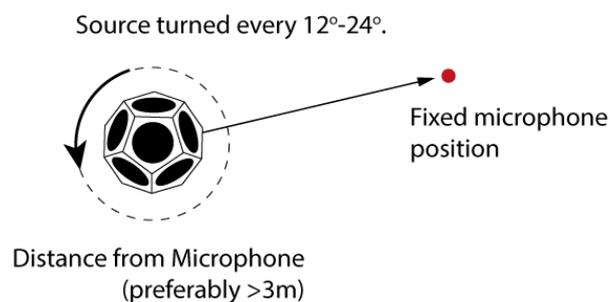
Figure 18: Alternative way of measuring around the source. In this case the microphone is fixed a distance greater than 3m from the centre of the source. The source should be rotated every 12º – 24º, which gives 30 or 15 measurement angles respectively. **The main advantage** of this method is that only 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 setups proposed in Figure 16 and Figure 18, we measured the impulse response at 15 spots. The measurements were performed in a small anechoic chamber where 3 m

distance between receiver and source was not possible. Therefore, for this example, which is merely a test of the method, we used **only 1 m distance**.

An example of a successful impulse response in the anechoic room is given in Figure 19, where all uncalibrated SPL values have been derived. These values can be very low - often even negative - indicating sound pressure levels well below the reference value of 20 μPa. However, this is not a problem, as it simply reflects very low absolute pressure levels rather than an error in measurement. Uncalibrated values are indicated with a "*" next to them. In measurements taken inside the *anechoic chamber*, "*.**" characters might be seen for some room acoustic parameters, because the impulse response is typically too short. However, the only parameter of interest at this point is **SPL***.


Figure 19: Room acoustic parameters from a healthy impulse response recording inside the anechoic/dry room. Since the impulse response is too short, some acoustic parameters might not be derived. However, **the only relevant parameter is SPL***.

7.3   Create the calibration file

Once all measurements in the *anechoic chamber* have been completed, follow the next steps to create the calibration file. Here we use the measurements coming with the ODEON installation in the Measurements\Calibration\Free-field method folder:

1) Open the calibration interface by clicking Tools>Measurement Calibration>Free field.
2) Set the **source/receiver heights at 10 m** and **source-receiver distance at 1 m**.
3) Click the button Load first/single-step files to load the measurements obtained in the previous section. In the dialog that pops up, you may select all files from S1R1 to S1R15 at once, by holding the SHIFT or CTRL key.
4) The selected files are loaded in separate Measured response windows automatically. The corresponding levels are displayed in the One-step measurements table, together with their average and Standard Deviation (STD). The calibration values are shown in the Calibration table. According to Eq.(15), the final calibration adjustment is the **sum** of the one-step calibration values plus any possible values from the two-step calibration (source reference and in-situ correction). Figure 20 shows the interface with the calculated parameters.
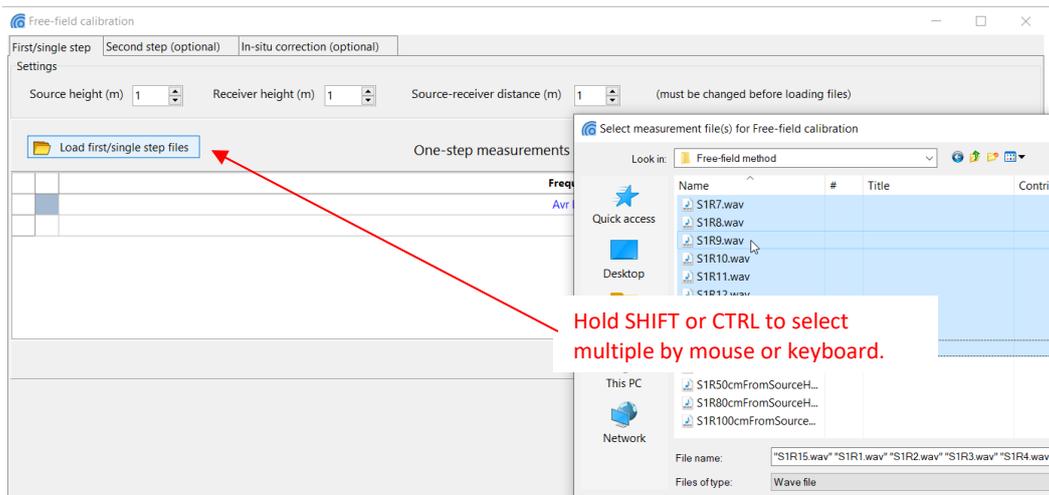
5) At this point only *First step* values are shown, meaning that we can simply save a *Single-step calibration* file, by clicking the Save calibration file button. This is the simplest version of calibration which however requires that the equipment and gains remain fixed (as described in Sec. 3.1).
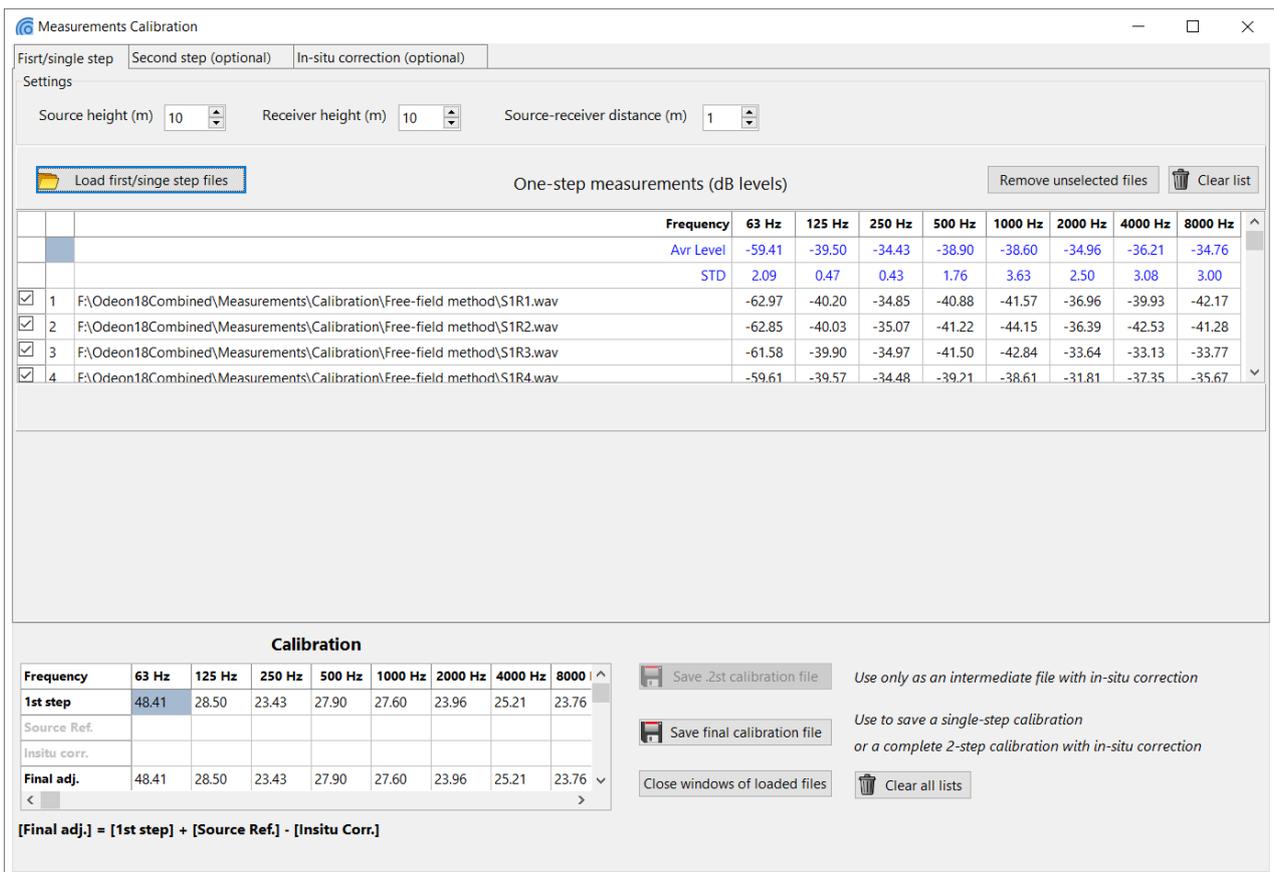

Figure 20: The interface after *one-step* calibration files have been loaded. The relative levels are displayed, together with the calculated calibration values at the bottom of the window.

6) You will be prompted to set the calibration file as the active one, which is going to be used with the following measurements. It is possible to change the active calibration file at a later point in the Measure impulse response interface 〜 (SHIFT+CTRL+D). 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.

You can now launch the Measure impulse response interface to check whether the active calibration file is the correct one (see Figure 8 for a similar screenshot from the *Diffuse-field calibration*).

7.4 Performing calibrated measurements

At this stage, the same procedure as described in Sec. 5.4 for using a *Diffuse-field* calibration file should be followed. The only difference is that another calibration file is active, which comes from the *Free-field* method. 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 a calibration after the actual measurements have been performed.

You can test the functions yourself with the calibration file from Figure 20 and the measurement files in Measurements\Calibration\Auditorium21 folder. For this example, use only the files labelled as 'FixedGain'. Figure 21 presents a comparison of the same measurement (S1R4FixedGain.wav in Auditorium21) calibrated with both the Diffuse and the Free-field methods. The final G parameters are very close to each other (within 1 dB Just Noticeable Difference – JND), apart from the lowest band (63 Hz), where the difference is much higher.

From the standard deviation (STD) values of the derived levels in Figure 8 and Figure 20, it can be observed that the *Free-field* measurements are more unstable at 63 Hz (with STD values above 2 dB), whereas the *Diffuse-field* measurements exhibit greater stability (STD values only slightly above 1 dB). This indicates that the *Diffuse-field* method yields a more reliable calibration than the *Free-field* method.

Several factors contribute to this difference, including the increased difficulty in accurately filtering low frequencies - due to their longer wavelengths and the resulting spectral blending (smearing), as well as the presence of octave-band filter ringing. These artefacts are more pronounced in the *Free-field* method, where only the direct sound from the source to the microphone is captured, without the averaging effects of diffuse reflections.

Measured response - C:\...\Measurements\Calibration\Auditorium21\S1R4FixedGain.wav

		Raw Impulse response (broad band)	Raw decay curve (broad band)	Decay curves (broad band) T(30) = 0.54 seconds			Decay curves (all bands)	Energy parameters	Parameter bars
T(15)	(s)	1.22	0.42	0.44	0.47	0.54	0.64	0.54	0.43
T(20)	(s)	1.19	0.44	0.42	0.51	0.52	0.62	0.55	0.43
ξ(T(20))	(‰)	23.0	87.5	82.8	9.7	2.5	2.2	1.7	1.1
T(30)	(s)	1.15	0.56	0.44	0.48	0.53	0.60	0.55	0.44
ξ(T(30))	(‰)	12.4	32.0	22.7	4.3	2.1	1		
Curvature(C)	(%)	−3.8	25.4	5.9	−5.1	0.3	−3	Calibrated G values, using	
Ts	(ms)	65	23	21	24	27	28	the **Diffuse-field** method	
SPL	(dB)	−4.5	−10.3	−8.5	−8.5	−8.7	−9		
G	(dB)	26.5	20.7	22.5	22.5	22.3	21.8	21.8	21.0
G(early)	(dB)	25.3	20.6	22.4	22.3	22.0	21.4	21.3	20.8
G(late)	(dB)	20.1	4.3	6.7	9.3	10.8	11.8	11.9	8.1
SPL(Af)	(dB)	−30.8	−26.5	−17.1	−11.7	−8.7	−8.0	−8.2	−11.2

Measured response - C:\...\Measurements\Calibration\Auditorium21\S1R4FixedGain.wav

		Raw Impulse response (broad band)	Raw decay curve (broad band)	Decay curves (broad band) T(30) = 0.54 seconds			Decay curves (all bands)	Energy parameters	Parameter bars
T(15)	(s)	1.22	0.42	0.44	0.47	0.54	0.64	0.54	0.43
T(20)	(s)	1.19	0.44	0.42	0.51	0.52	0.62	0.55	0.43
ξ(T(20))	(‰)	23.0	87.5	82.8	9.7	2.5	2.2	1.7	1.1
T(30)	(s)	1.15	0.56	0.44	0.48	0.53	0.60	0.55	0.44
ξ(T(30))	(‰)	12.4	32.0	22.7	4.3	2.1	1		
Curvature(C)	(%)	−3.8	25.4	5.9	−5.1	0.3	−3	Calibrated G values, using	
Ts	(ms)	65	23	21	24	27	28	the **Free-field** method	
SPL	(dB)	−10.3	−10.6	−8.6	−8.2	−8.1	−8		
G	(dB)	20.7	20.4	22.4	22.8	22.9	22.5	22.1	21.4
G(early)	(dB)	19.6	20.3	22.3	22.6	22.6	22.1	21.6	21.1
G(late)	(dB)	14.3	4.0	6.6	9.7	11.4	12.5	12.2	8.5
SPL(Af)	(dB)	−36.5	−26.8	−17.2	−11.4	−8.1	−7.3	−8.0	−10.8

Figure 21: G values derived for the same impulse response measurements inside Auditorium 21, using both the **Diffuse-field** and **Free-field** calibration methods.

8. Calibration for the STI parameter

As with simulations (see the Application note on Calculation of Speech Transmission Index in Rooms), measurement of the Speech Transmission Index (STI) parameter also requires accurate knowledge of the levels, similar to G measurements. All procedures described in the previous sections - *Diffuse-field*, *Free-field*, and *Two-step* methods - are applicable to STI measurements as well, with only the following two additional settings required under Options>Program setup>Measurement Setup:

8.1 Measurements with the ISO 3382-3 source

The ISO 3382-3 source is defined in [4] and corresponds to an **Omnidirectional source** that has a speech spectrum, instead of a flat one. It is primarily intended for *open-plan office* measurements, but it can also be used for any other STI measurement, since it can be applied directly to the same physical omnidirectional source employed for G measurements.

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	−99.00	60.90	65.30	69.00	63.00	55.80	49.80	44.50	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.

Background 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)
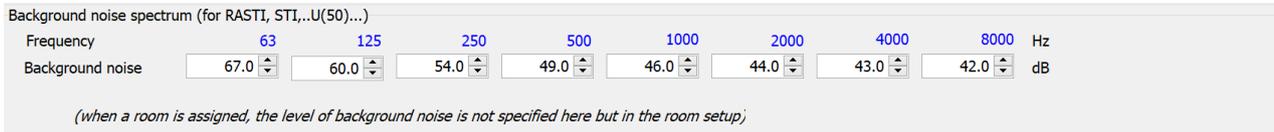
8.2 Measurements with an Artificial Mouth

(Works only with a *Free-field* calibration)

Measurements with the ISO 3382-3 source can be suitable for open-plan office parameters and rough STI measurements, 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	62.20	62.20	68.20	70.80	64.50	59.80	54.80	49.50	dB		Speech - ISO 3382-3

Figure 22: Power spectrum used for calibrations with an artificial mouth, that is equivalent to a BB93_NORMAL_NATURAL.SO8 source.

Derivation of *artificial mouth* spectrum

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 23: 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 of the BB93_NORMAL_NATURAL.SO8 is the same as the ISO 3382-3 source, the directivity pattern is not omnidirectional, but it looks like in the picture on the right. Therefore, the same power spectrum is distributed differently.
3. According to Figure 23, the SPL on axis at 10m is as highlighted in the blue box. To derive the power spectrum of an equivalent omnidirectional source we 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 22.
4. The equivalent BB93_NORMAL_NATURAL.SO8 omnidirectional source is a bit louder than the ISO 3382-3 by 1.3 to 5 dB.
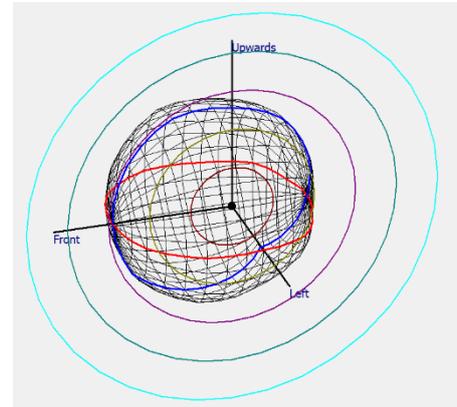
This process of deriving an equivalent BB93_NORMAL_NATURAL.SO8 omnidirectional source is used only for calibrating the system and not to change the directivity pattern of the source. The actual *artificial mouth* has its own directivity, which is taken into account by its geometry.

Perform the calibration with an *artificial mouth*

To perform the calibration of this 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.

> **Tip:** 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 12).

In contrast to calibration with an omnidirectional source (e.g. dodecahedron), the distance between the *artificial mouth* and the microphone does 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 on 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 24, 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 24, 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.
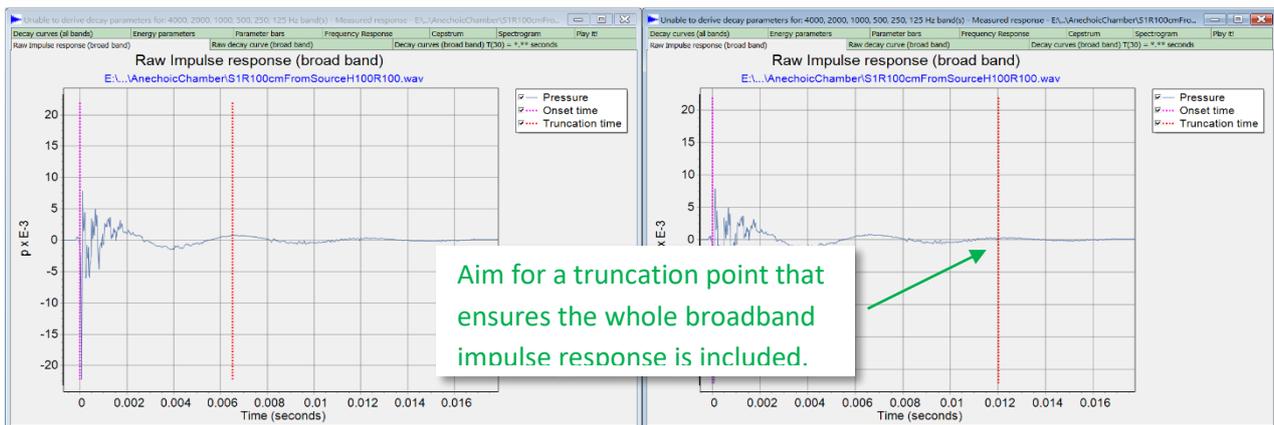

Aim for a truncation point that ensures the whole broadband impulse response is included.

Figure 24: 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. 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 accomplished.

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

10. References

1. C.L. Christensen & G. Koutsouris. ODEON Room Acoustics Software, manual, version 19, Odeon A/S, Denmark 2025 (https://odeon.dk/download/Version19/OdeonManual.pdf).
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.