

AcouCheck 2

**Room Acoustic Analysis using
Notebook and Tablet Computers**

User's manual

July 2017

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1. Introduction

The 'AcouCheck 2' software turns any notebook or tablet computer with an attached microphone into an easy-to-use measuring device for room acoustic parameters, reverberation times and others.

The software is primarily designed for education and information purpose, aiming at measurements without specialized equipment and using simple and standard gear instead. The software guides through the measurement and analysis process and illustrates the functionality of measurements.

Using simple equipment the software does not claim to provide reference quality measurements. But results should be sufficient to evaluate the quality of communication rooms or classrooms in schools.

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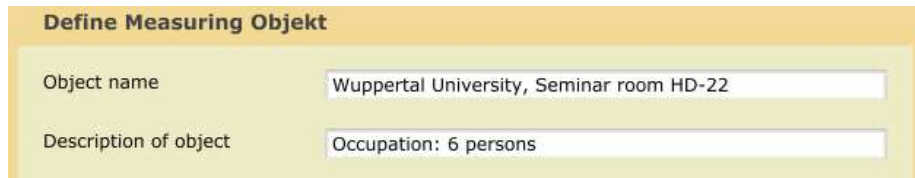
2. A Beginner's Guide

The user interface of the AcouCheck software is designed to lead through a series of measurements including evaluation. The tabs on top switch between pages showing the steps of the process.

2.1. Preparing a Series of Measurement

The Measuring Object

A series of measurement is always related to one room, that is not changed between the individual measurements. Assuming a seminar room named 'HD-22' at Wuppertal University with 6 persons inside, our measuring object is described as in Figure 2.1.



The screenshot shows a yellow dialog box titled "Define Measuring Objekt". Inside, there are two text input fields. The first field is labeled "Object name" and contains the text "Wuppertal University, Seminar room HD-22". The second field is labeled "Description of object" and contains the text "Occupation: 6 persons".

Figure 2.1 Measuring object definition

Preparing Measurements

The Computer

A notebook or tablet computer running the AcouCheck software on one of these operating systems¹:

- Windows (version XP-SP3 or higher)
- OS-X (version 10.6. or higher)
- iOS (version 7 or higher)
- Android (version 4.03. or higher)

¹ For installation and operation on different platforms see Appendix A.7

OBS: A microphone input with attached microphone or an internal microphone is required for running AcouCheck.

The Microphone for Measurements

A suitable microphone is required for the measurements. The microphone signal must be available to software running on the computer. This can be achieved using one of the following options:

- As a low cost solution simple electret microphone connected to an USB audio interface can be used (cost starting about 10 Euro each). Most USB audio interfaces will be recognized by the computer within a minute after connecting the USB plug. When recognized, the interface can be selected by the software.
- Usually notebook and tablet computers supply an internal microphone or an integrated microphone input can be used with an external microphone attached. But these options require some care: In the signal paths of many notebook computers a signal manipulation is applied. This may be useful for speech applications, but it must (!) be switched off for measurements. In any doubt about manipulations it is preferable to use the external solution above.
- Highest quality results, but also at highest costs, can be achieved using a professional measurement microphone or an omnidirectional studio microphone and a professional audio interface. Connection of the microphone is done by robust XLR connectors. The 'Phantom power' required for professional condenser-microphones is supplied by the audio interface. Price ranges of each the microphone and the audio interface are starting about 100 Euro und reach up to far above 1000 Euro for premium devices.

Impulse Sound Source

The easiest way to measure reverberation is using an impulse as sound source. Impulses can be generated by different means:

- A simple impulse source is a strong hand clap. It is essential to use 'hollow hands' to achieve a sonorous sound. A strong hand clap is suitable only for smaller rooms up to classroom size, provided the background noise is low.
- If a louder source is required, this can be achieved by bursting an air bubble. This can be a balloon touched with a needle, or an about 10 cm sized packaging bubble held tight (not too tight) in one fist and then smashed by the other flat hand.
- In very large room also bangers or alarm pistols can be used. Specific regulations must be considered in these cases where applicable.

Ear protection

When using loud sound sources an ear protection is required. This can be done over ears (earmuffs) or in the ear canals. Please have in mind that a very loud impulse sound can damage inner ears immediately (blast trauma).

OBS! Never produce an impulse or any other loud sound close to any persons ears! This also applies to hand claps!

Positions of sound sources and microphone in a room

Reverberation time is a statistic measure of a room and thus results of individual measurements are spreading. Thus a number of measurements are required to determine reverberation time.

According to DIN 18041 a minimum of six measurements shall be taken at different positions of sound sources and microphone. Furthermore it is advisable to take more measurements, e.g. 10 – 12.

When choosing 4 source positions and 6 microphone positions this results in 24 combinations, more than needed.

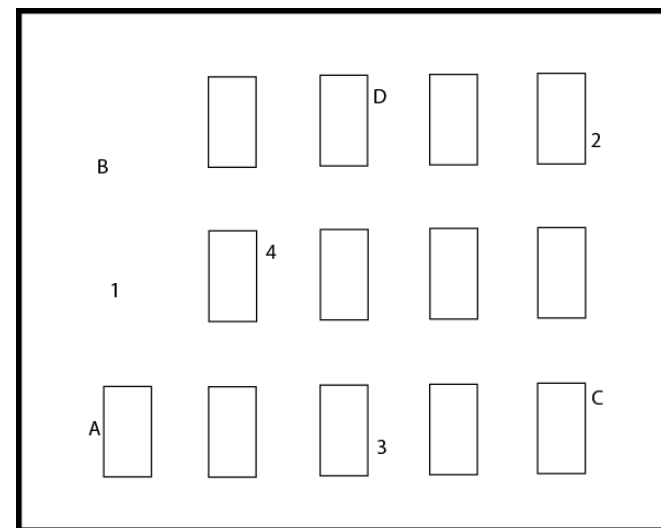


Figure 2.2 Sketch of a classroom with positions of sound sources- (A, B, C, D) and microphone (1, 2, 3, 4).

Typical speaker and listener positions in a room can be selected as sound source and microphone positions. Positions shall be spread around the room and keep some distance to walls (> ca. 1 m). In each measurement selecting a distance between source and microphone in the range from 1/3 to 2/3 of the room diagonal is a good choice.

Silence

Any noise occurring during a measurement will degrade the quality of results. Before measuring noise sources shall be removed, windows and doors shall be closed and noise from outside shall be avoided. During each 5 second periods of recording no noise except the measuring signal shall be present.

Preparing the Microphone

Connecting the Microphone

If an external audio interface is used, connect this device to the computer (usually via USB). Connect the microphone to the microphone input. If more than one input is available, use the 1st or 'left' input. If a professional microphone is used and phantom power is required, switch it on. Then give a minute or two for automatic recognition of the audio interface.

Select and Check the Audio Input

After starting the AcouCheck software, on page 1 the octave levels shall be visible. Tap slightly on the microphone to check the signal.

If there is no or very low signal, do at least one of the following:

- go to 'Audio setting' and select another audio input (not possible on most tablet computers).
- Check in Windows or OSX settings if the microphone input is on and if the microphone level is not set too low.

Setting the microphone level

When the microphone signal is visible in the display check for appropriate levels. In figure 2.3 examples of correct levels of the background noise (left) and of a test impulse (right) are shown. The noise shall only be visible in the green range. Impulse peaks (the small bars) shall reach the orange range in all used octaves. Peaks may also reach the red range, but should not hit the top.

If levels are too high or too low, try these options of correcting:

- If levels are too high or too low in general use the Windows or OSX settings to modify the microphone level. If a professional audio interface is used, usually the microphone gain can be modified.

- If the level difference between impulse and background noise is too small, either the noise must be reduced or a louder impulse source is required (or both).
- Of course both options can be combined – 'intelligent testing' is demanded.
- Tablet computers operated by iOS or Android usually do not offer a microphone level setting option. Instead the microphone signal is limited at high levels, resulting in a dynamic distortion of the signal. Therefore it is strongly recommended to explore the dynamic behaviour by test measurements on these computers.²

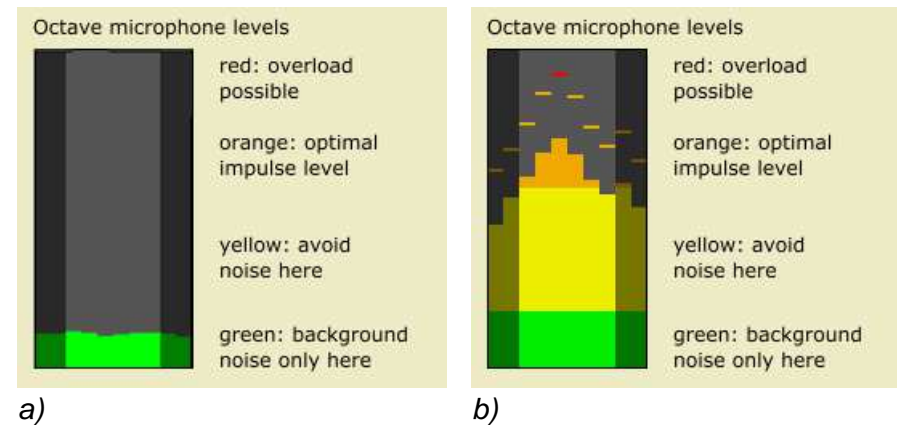


Figure 2.3 Microphone levels are shown for all ten audible octaves, from left, dark coloured 31 and 63 Hz, then six used octave bands, (125, 250, 500 Hz, 1, 2, 4 kHz), rightmost and dark 8 and 16 kHz. a) shows the background noise in the 'green range', b) shows a correctly levelled impulse, all peak markers reach the orange range

² see also the chapter 'Nonlinear and Time-variant Signal Paths' in Appendix A 4.

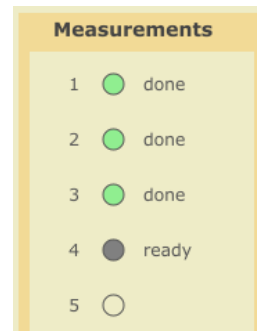
2.2. Taking Measurements

Starting a series of measurements

The AcouCheck software allows to take 10 to 12 individual measurements (depending on the computer type) in each series. It is recommended to make use of all twelve measurement slots to achieve a maximum of accuracy. In case of single measurements turn out to be outliers of bad quality, these can be excluded from further processing.

Individual- measurements

Measurements are taken in a numbered series. The upcoming measurement shows a status 'ready'. Each finished measurement is the labelled 'done'. In figure 2.4 measurements 1-3 are 'done' and measurement 4 is 'ready':



The active measurement

Initially the 'oscilloscope' shows the running microphone signal (see figure 2.5 a.).

The 'impulse record' mode is started automatically if an incoming impulse exceeds the internal trigger threshold and this is notified by the red 'recording' light (see figure 2.5 b). After 5 seconds recording is stopped and the internal evaluation process is started, indicated by a yellow light. When the yellow light is off, the active measurement and evaluation is done. The measured room impulse response is stored in the background (see Appendix 6).

The button 'File' allows to reload an impulse from an earlier measurement for re-processing instead of measuring. The button 'HPF' (high pass filter) reduces low frequency noise. The button 'Sensf' lowers the impulse detection threshold.

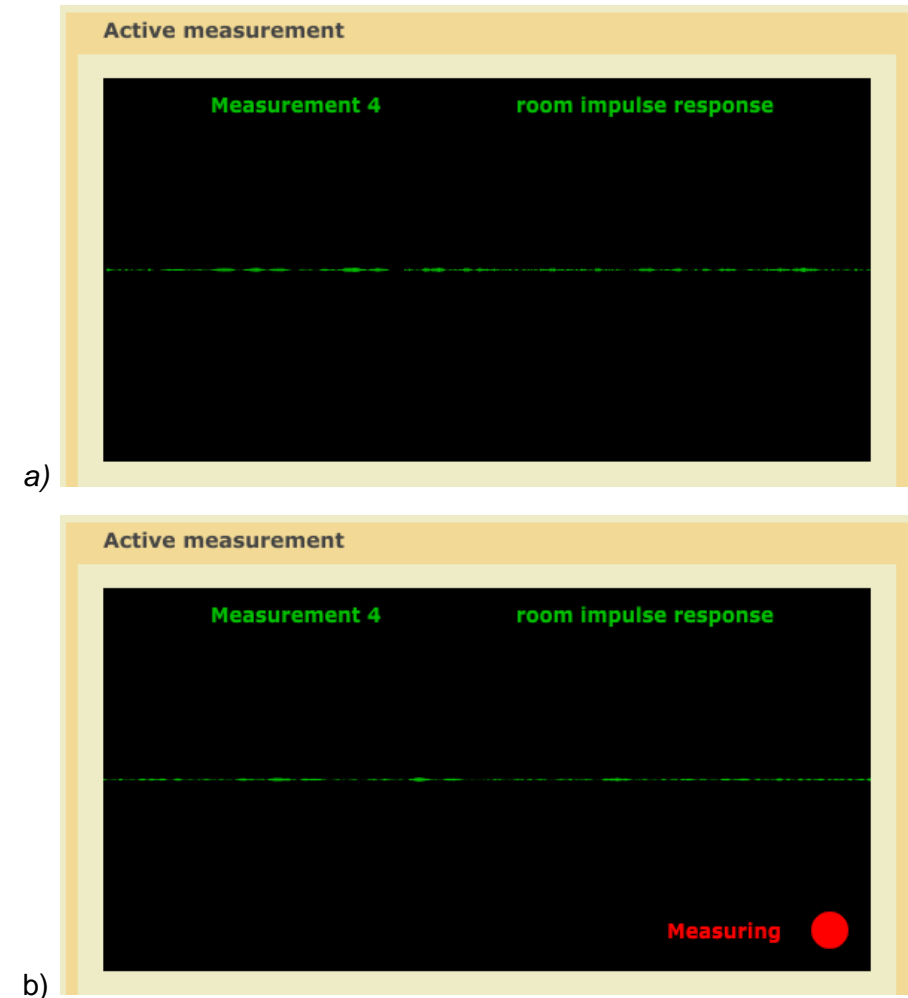


Figure 2.5 Measuring a room impulse response
a) a snapshot while waiting for an impulse
b) during the five second recording

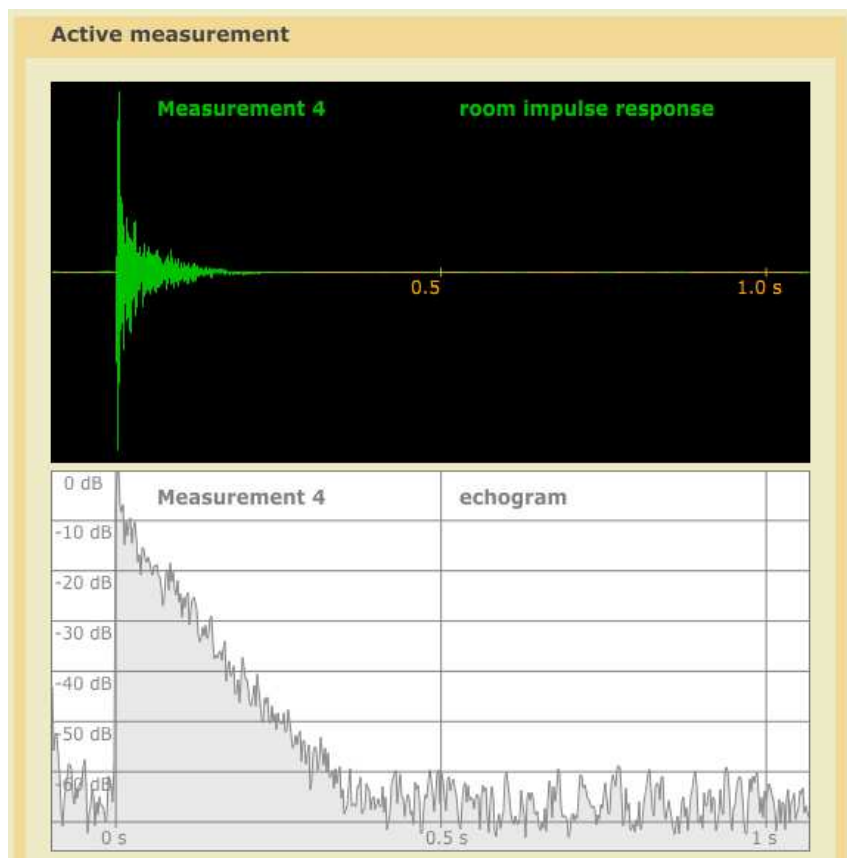


Figure 2.6 Measured room-impulse response (top) and corresponding echogram (bottom) from a living room showing a reverberation time about 0.4 s

In Figures 2.6 and 2.7 two examples of measured impulses are shown as impulse oscillograms (top) and as echograms (bottom). The echograms show reverberation decays in a logarithmic [dB] scale and thus allows to estimate the measurement quality, visible as an approximate linearity of decay.

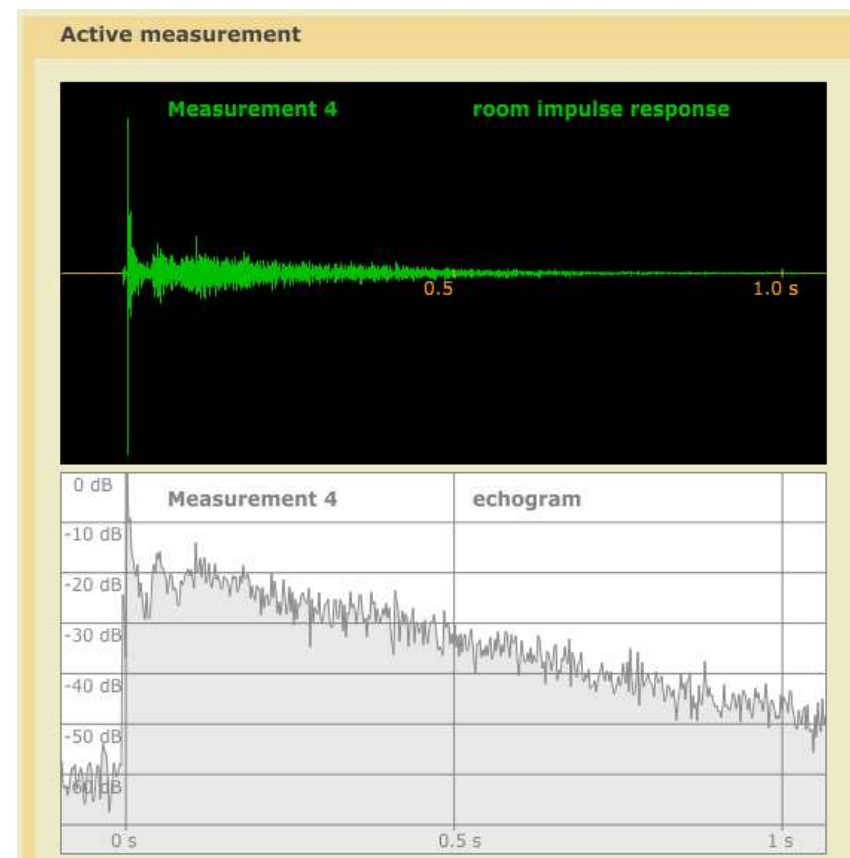


Figure 2.7 Measured room-impulse response (top) and corresponding echogram (bottom) from a concert hall showing a reverberation time about 2.0 s.

At this point either another measurement can be initialized by the 'next' button or the current measurement can be repeated (e.g. if distorted by a noise event).

2.3. Evaluating Measurements

The 'Evaluation' page shows an intermediate step of the evaluation process.

The measured room impulse response covers the full recorded spectrum. A first processing step is splitting the spectrum into octave bands using a set of filters as shown in figure 2.8.

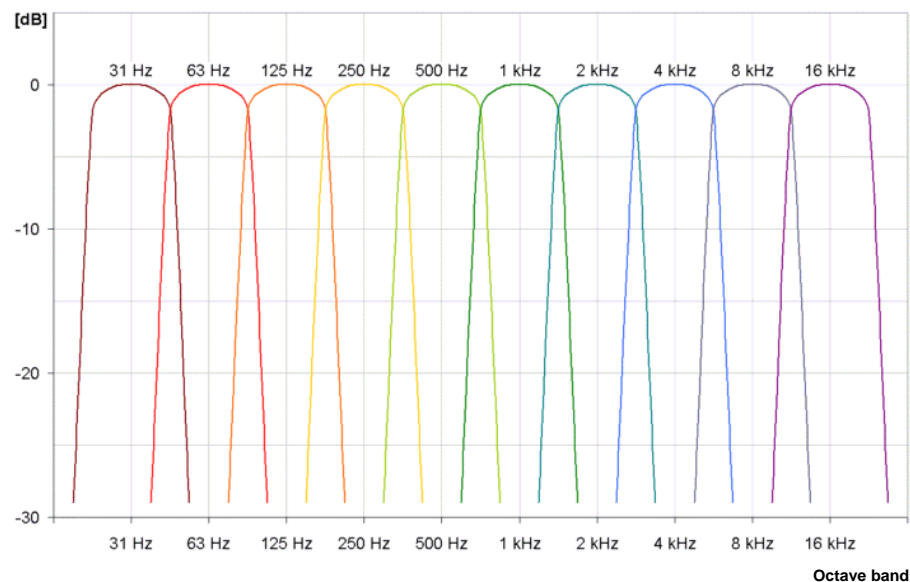


Figure 2.8 Filter functions of all 10 standard-octave filters in the human audible range. Six median octave bands from 125 Hz to 4 kHz are used in AcouCheck's standard mode. In 'advanced mode' the 63 Hz and 8 kHz octave bands are added..

Each of the octave filtered impulse responses is transformed by the function

$$f(x) = 10 * \log_{10} (x^2) \quad [\text{dB}]$$

into logarithmic values shown in echogram plots (see figure 2.9.)

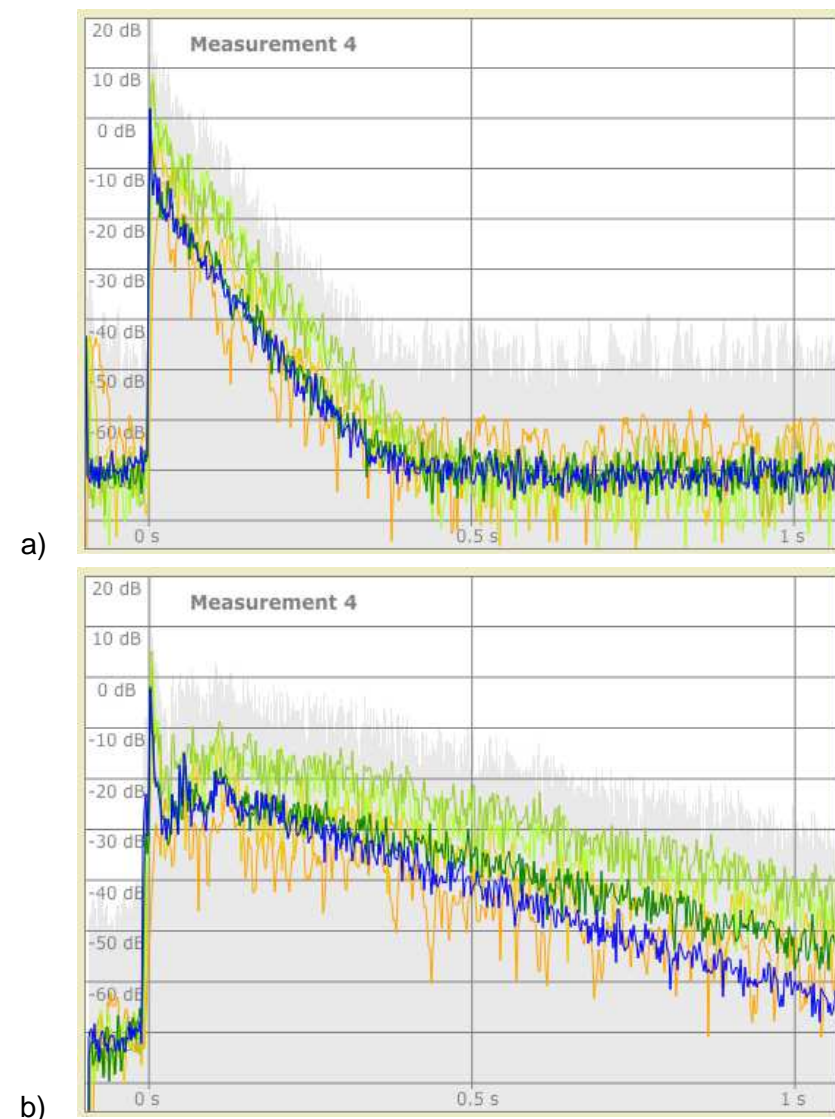


Figure 2.9 Octave echogram plots show the temporal reverberation decay in a logarithmic [dB] scale, Two examples, a) a living room (reverberation time ~ 0.4 s) b) a concert hall (reverberation time ~ 2 s)

Right after the direct sound impulse echogram plots show a continuous decay superposed by random-like fluctuations, until the background noise floor is reached. From the decays the reverberation times can be read as inverse slope values multiplied by a constant factor. Visually a quick decay represents a short reverberation time, in contrast a slow decay leads to a long reverberation time.

In figure 2.9b the slightly shorter reverberation times at high frequencies (blue lines) the slightly longer reverberation times at mid and lower frequencies (green and yellow lines) are noticeable.

From figure 2.9 it can also be found that the low frequency decays (orange lines) are less uniform than higher frequency decays. Thus reverberation time measurements at low frequencies often are less precise than at higher frequencies.

Major reasons may be:

- The sound energy fraction per octave is comparatively small at low frequencies in a short impulse acceleration signal. This can result in a poor signal-to-noise ratio.
Particularly when handclaps are used for acceleration, it is essential that the joined hands form a closed cavity after each clap
- At low frequencies the eigenfrequency density per octave is lower than at higher frequencies. Thus individual eigenfrequencies become more noticeable in their spatial sound field patterns. Thus results of measurements at different room positions usually scatter more than at higher octaves..

Looking at single octave decays

An octave solo view is activated when the same octave button is clicked twice (slower than double-click). In this view (see figure 2.10) also the 'backward integral' used for the slope calculation is shown and it is particularly suitable for visual quality estimation.

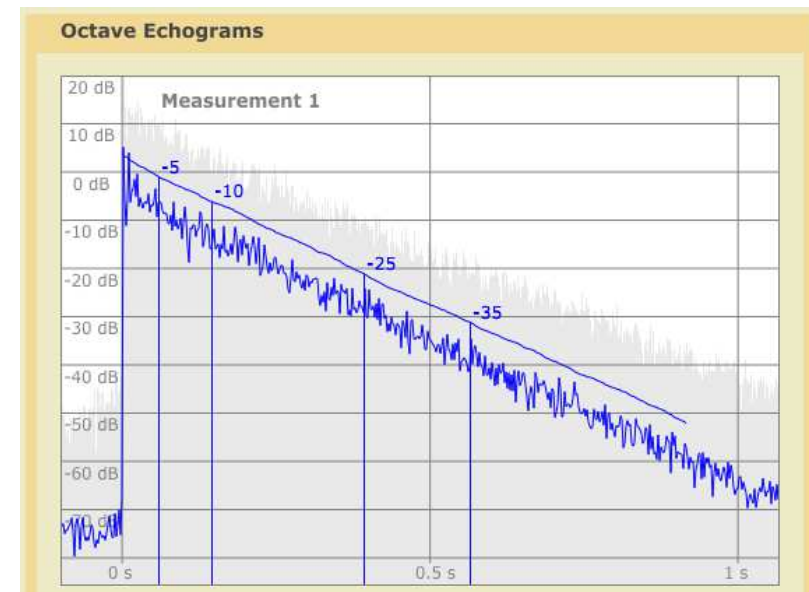


Figure 2.10 Single octave echogram and 'backward integral' of a good quality measurement.

2.4. Assessing measurement results

On page '4. Assessment' six octave bands measurement results from 125 Hz to 4 kHz are shown. In figure 2.11 results from a single reverberation time measurement in a concert hall are shown as a line. The shade behind the line indicates an estimate of the uncertainty of measurement. Mostly this uncertainty decreases from lower to higher frequencies..

In figure 2.12 measured reverberation times from six measurements taken at different sound source and microphone positions in a lecture hall are superposed. The mean values are added as a bold line. The individual measurements agree within a narrow range slightly increasing towards low frequencies. This increase can be observed in many measurements, sometimes even more pronounced.

Excluding faulty measurement

Particularly if background noise is present during a measurement or a sound source is insufficient result may become faulty. Faulty measurements are distinguishable either by their uncertainty being considerably larger than shown in figure 2.11 or they show themselves as outliers (or both). These measurement results can be excluded from further assessment and calculations by unchecking their 'includes'.

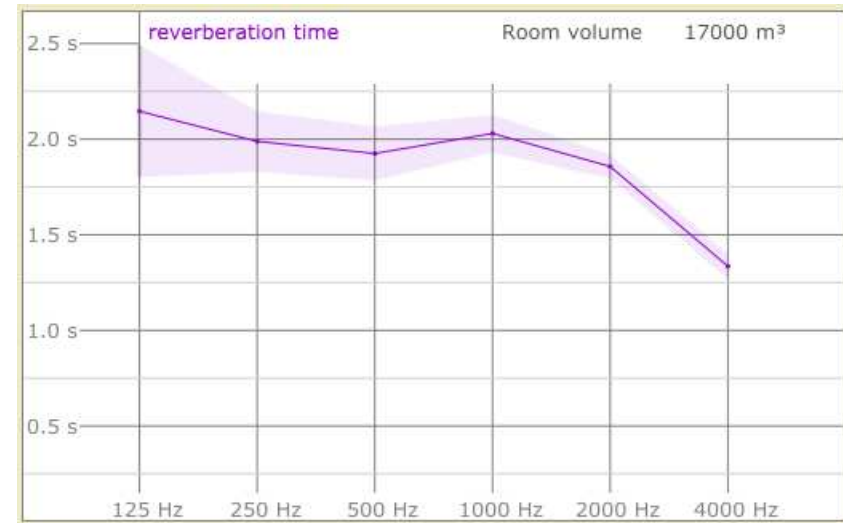


Figure 2.11 Reverberation times of a concert hall – results of a single measurement

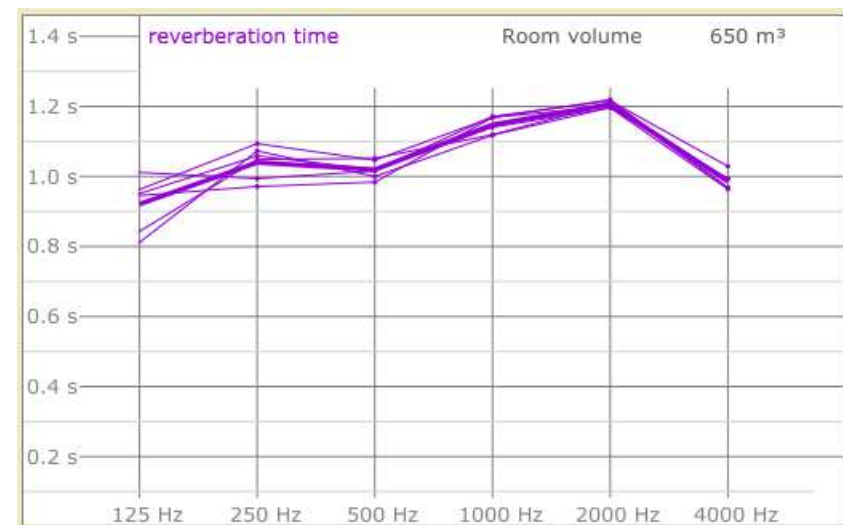


Figure 2.12 Reverberation times of a lecture hall – six individual measurements and mean values

Comparing to standards

In German standard DIN 18041 good practise reverberation times are recommended for different types of occupation. The recommendations are depending on the net room volume. After entering the room volume and selecting a criterion (occupation type) the recommended reverberation time range is shaded in light green with a dark green center line.

Selectable occupation types cover music, speech, education and sports from the latest DIN 18041 (2016). Additional options include older DIN 18041 versions and the 'EBU listening room' recommendation (as an example of acoustically very 'dry' rooms).

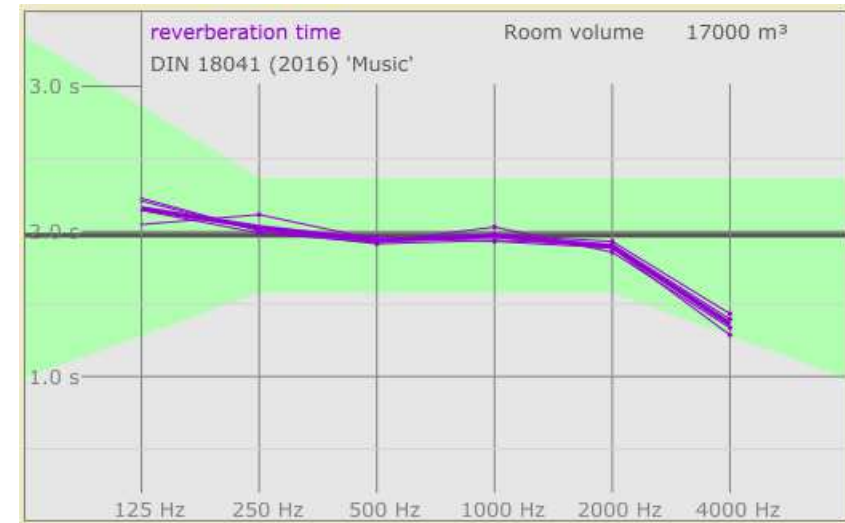


Figure 2.13 Several reverberation time measured in a concert hall and DIN 'music' recommendation

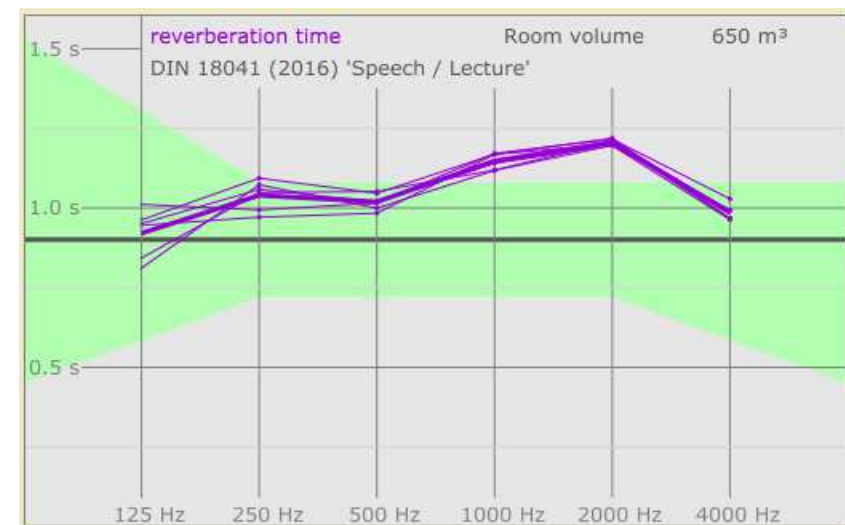


Figure 2.14 Reverberation times measured in an unoccupied lecture hall and DIN 'speech' recommendation.

2.5. Predictive calculations

On page '5. Calculate' effects of additional sound absorbers in a measured room can be precalculated. Additional absorbers may be persons, furniture, absorber elements or even modified surface absorption coefficients. Up to eight different absorber types can be 'added' to a room by calculation.

Selecting an absorber type is started by clicking / touching an 'add absorber' button. This immediately switches to the absorber database page (see chapter 2.7). On this page an absorber can be selected from the database. The button 'apply' confirms the selection and switches back to the calculation page. The button 'back' switches back without selecting an absorber type..

After selecting an absorber type the effective amount of absorption can be set by the up and down buttons. When using the mouse pointer or a pen on a touch screen the value can also be set by horizontal dragging in the numerical field above the up and down buttons.

As soon as any additional absorbers are added to the room by calculation a new green line shows the calculated reverberation times including the additional absorption.

The calculation is done separately for each octave using Sabine's reverberation formula:

$$T = 0.163 \ V / A \quad [s]$$

OBS: the constant value 0.163 [s/m] is only valid, if metric (SI) units are used for all variables (A [m²], V [m³], T [s])

1st step: The room's effective absorption area is calculated from the measured reverberation time and the room volume using the inverse Sabine's formula :

$$A_{\text{effective}} = 0.163 \ V / T_{\text{measured}}$$

2nd step: The additional equivalent absorption area is added.:

$$A_{\text{calculated}} = A_{\text{effective}} + A_{\text{additional}}$$

The additional equivalent absorption area adds together the effects of all additional absorbers, for instance in case of two absorber types and several persons:

$$A_{\text{additional}} = S_1 \cdot \alpha_1 + S_2 \cdot \alpha_2 + n_P \cdot A_P$$

where S_1, S_2 are geometric absorber areas,
 α_1, α_2 are sound absorption coefficients,
 n_P is the number of persons and
 A_P is the equivalent sound absorption area of a single person.

Finally Sabine's formula is used to calculate 'new' reverberation times from the calculated absorption areas:

$$T_{\text{calculated}} = 0.163 \ V / A_{\text{calculated}}$$

These calculated reverberation times are prognostic. Their quality depends not only on the measurements, but also on the assumed absorber data. The more realistic the assumptions, the better the prognosis will be.

Adding persons

In most cases reverberation measurements are performed in unoccupied rooms, while reverberation times of occupied rooms are of more practical interest. In these cases an 'unoccupied' can be transformed to a virtually occupied measurement by the calculation introduced before. This can readily be done for people standing on the ground or sitting on acoustically hard seats as exemplified in figure 2.15 a).

In contrast, if upholstered seats are 'occupied' by calculation a simple addition of the people's absorption leads to invalid results, because the seat absorption is 'switched off' at least partially (see figure 2.15b). In these cases purpose-made absorption data are required quantifying the difference in absorption between unoccupied and occupied seats.

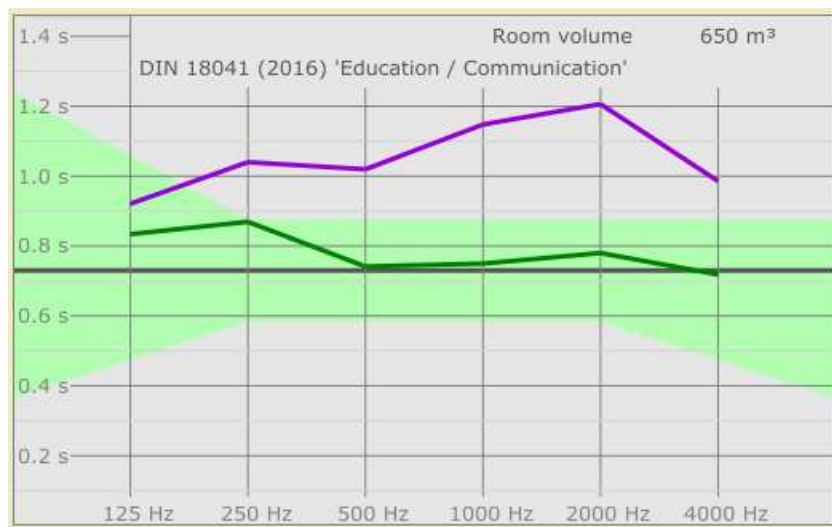


Figure 2.15 a Calculated effect of additional persons in the room
Reverberation times of a lecture hall, measured when unoccupied (violet) and calculated for 100 Personen on wooden seats (green)

The absorption of a person can be more or less, depending on:

- their body sizes (adults show more absorption than children)
- their clothing (typical winter clothing is more absorptive than thin summer clothing)
- standing persons show more absorption than sitting persons
- the distance between individuals – at low distance persons obstruct each other partially and thus reduce their absorptive effect.

So, for different occupations different person absorption data sets may be appropriate. In case of uncertainty it may be worth to calculate variants with a minimum and with a maximum assumed absorption using different data sets.



Figure 2.15 b Calculated effect of additional persons in the room
For a measured concert hall with upholstered seats the corresponding calculation is not valid (see text).

Adding absorbers – improving rooms

In numerous cases of room acoustic analysis reverberation times turn out to be too long. In these cases additional absorbers for improvement can be 'tested' by calculation⁴.

In figure 2.16 improvement options for a conference room are shown. In actual state (needle-felt floor, occupied by 9 persons) reverberation times are acceptable short at high frequencies, but far too long at lower frequencies.

In figure 2.16 a) two different types of plate absorbers (absorption maxima at 125 Hz and at 250 Hz respectively, each 18 m²) are added to the side walls. This measure, which almost exhausts the available spare area, improves the room considerably at low frequencies and approaches the DIN 'speech' recommendation.

If 100 mm of porous broadband absorbers are mounted instead of the plate absorbers, much shorter reverberation times are resulting and the tighter DIN 'education' recommendation is met almost completely (see figure 2.16 b).

Anyway it should be considered if the selected absorbers are adequate for the specific room and if the required surface is available.

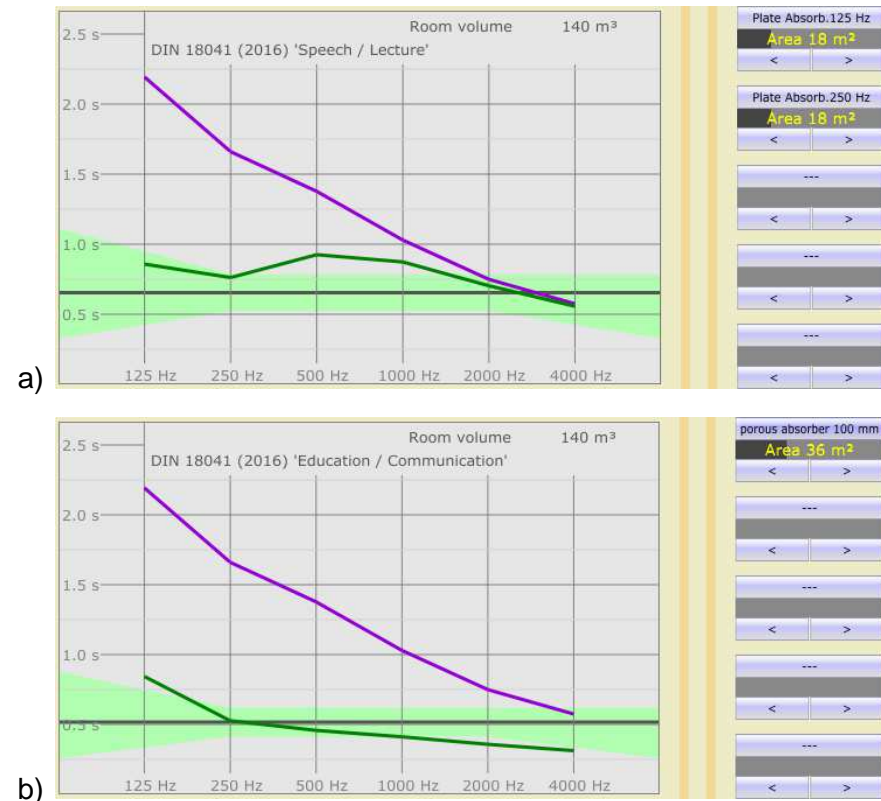


Figure 2.16 Example of improvement calculations for a conference room showing insufficient low frequency absorption.

- a) The effect of two different plate absorber types at the side walls, compared to the DIN recommendation 'speech'
- b) The effect of broadband absorbers at both side walls compared to the DIN recommendation 'education'

2.6. Documentation of results

On page '6. Documentation' a text summary of the actual series of measurements can be created. When the button 'Create (txt)' is used, the summary is created in simple text format. The document can be edited, copied to the clipboard or saved to a text file (Windows and OSX only).

Reverberation times in the summary are supplemented by a quality indicator (0..9). These quality indicators can be read as follows:

- 9 good or very good quality
- 8 moderately good quality
- 7 still acceptable quality
- 6 inferior quality, better not to use
- 5 or less: bad quality – do not use !

If there is any doubt in measurement quality or if the quality indicator lower than 8, it is recommended to look closer at these measurements, e.g. by a critical view on the backward integral on software page 3 (clicking twice on an octave button with 1 sec. between, not a doubleclick).

Alternatively a semicolon separated table of the results can be created using the button 'Create (csv)'. Results in this format can be imported into spreadsheet software for further processing.

2.7. The Sound Absorber Data Base

Page 6. 'Absorber' shows a browser with the data base tree on the left. Once an absorber is selected its characteristics will be shown on the right.

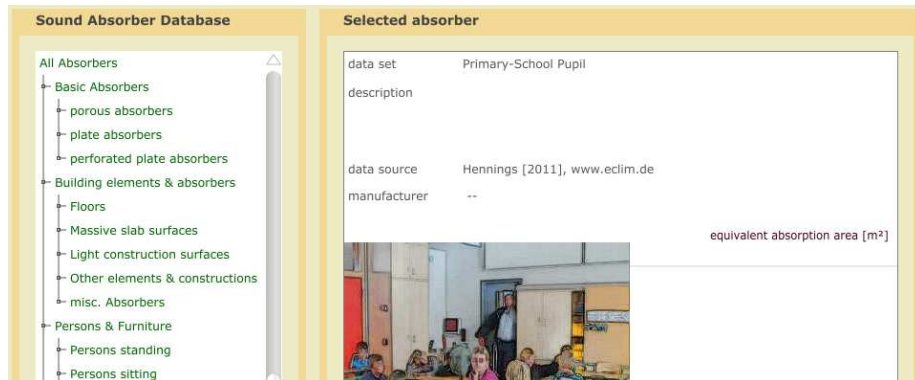


Figure 2.17 The sound absorber browser with the selection tree on the left and the selected absorber on the right.

The absorber database can be browsed using the keyboard navigation keys (↑, ↓, ←, →) or using the mouse. Each absorber's data are shown immediately when selected. The navigation tree is divided into several branches:

- a 'Basic' selection of absorber types.
- a number of different construction types and their surfaces
- various types of persons and furniture
- several particular absorber types
- absorbers from different manufacturers

Each absorber data set consists of:

- The absorber name
- A short description of the absorber
- The absorber data source, if known
- The manufacturer of commercially available absorbers
- Octave absorption coefficients, if the absorber type is areal measured, or equivalent absorption areas in case of persons or absorber elements. All absorption data are shown graphically.
- An image of the absorber if available.

Part of the data are 'practical sound absorption coefficient' (α_p) according to ISO 11654 (rounded to steps of 0.05 and limited to a maximum of 1.0). Absorption coefficients not complying with ISO 11654 may exceed 1.0 depending on conditions of their measurements.

However, an absorption coefficient above 1.0 means that more sound energy than available is absorbed, if interpreted directly. As this contradicts to basic laws of physics, such values are well done to be treated with caution. A simple although coarse precaution is limiting values to 1.0 at maximum. In AcouCheck this can be done by activating the button 'max 1.0'. More about interpretation of absorber data can be found in chapter 3.4 of the 'Advanced Topics' part in this manual.

3. Advanced Topics

This part of the manual addresses advanced users being familiar with basic room acoustic measurements as outlined in the previous chapter '2. Beginner's Guide'.

OBS: It is required to activate the '**advanced mode**' on page 1 of the software. Otherwise advanced features of the software are not accessible.

Additional features in 'advanced mode' operation include:

- Sine sweep accelerated measurements
- additional measures (T_{20} , T_{30} , EDT, Rad⁵, T_S , C_{80} , C_{50} , D_{50})
- additional octave bands are included (63 Hz and 8 kHz).
- Variants in predictive calculations

3.1. Using 'Sine-Sweeps' for Measurements

Advantages of 'Sine-Sweep' measurements

Impulse accelerated measurements are easily corrupted by unwanted noise. This is because impulses, although being rather loud, contain little sound energy due to their very short duration.

A very short impulse is suitable for measurement due to its spectrum containing all audible frequencies at the same time. The 'trick' of using a sine sweep is to transmit the different spectral parts successively. Thus at a similar signal level a much larger amount of energy can be radiated, increasing with the duration of the sweep.

⁵ 'Rad' is used to abbreviate the 'reverberation radius' or 'critical distance'.

After receiving the measured signal is 'convolved' mathematically with a reciprocal sweep, and thus collapsed to a room impulse response similar to an impulse accelerated measurement. As an advantage, sine sweep measured room impulse responses are much less sensitive to noise.

Also, results of sine sweep accelerated measurements are exactly reproducible if sound source and microphone are unchanged (except minor deviations due to person movements or background noise). In contrast results of impulse accelerated measurements are scattered, as in general impulses cannot be reproduced exactly

The Procedure of 'Sine-Sweep' Measurements

The Measuring Setup

Obviously a loudspeaker and an appropriate amplifier are required to radiate the sinus sweep. For accuracy of measured results a high level of the sweep signal is favorable, but signal distortions must be avoided. Distortions can be caused by overdriving the amplifier, but also loudspeakers cause distortion when operated at high levels. To avoid these both amplifier and loudspeaker should be operated considerably below their power limits.

Rules for the selection of sound source and microphone positions are similar to impulse accelerated measurements (see chapter 2.1)

In professional reverberation measurements often dodecahedron loudspeakers are used to achieve a uniform angular sound radiation.

But also directed loudspeakers of good quality can be used for measurements..

A high level (SPL) of the measuring signal is advantageous to improve the signal-to-noise ratio (SNR) of measured results, but any distortion caused by overdri-ving the amplifier or by high loud-speaker level can affect the mea-suring quality and thus should be avoided.

OBS! If a professional audio interface is used for sine sweep mea-surements, it is absolutely necessary to deactivate a 'monitor' option, if available. Otherwise measurements will be invalid and feedback noise may occur (see also Appendix. 2).

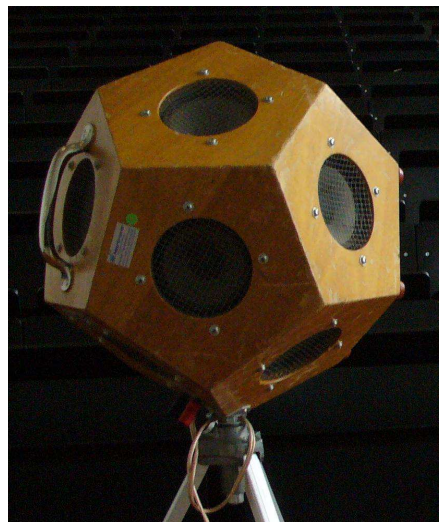


Figure 3.1 An aged dodecahe-dron loudspeaker still used at Wuppertal University..

3.2. Various Reverberation Time 'Types'

Originally the reverberation time is defined as the time for a 60 dB decay from the initial level of a sound event - an impulse or after switching off a constant noise source. A reverberation time measured over the full decay of 60 dB is called T_{60} . This requires a high initial 'signal-to-noise ratio' (SNR) of more than 60 dB because for the measurement the signal must exceed the background noise. In many measuring situations this requirement is not fulfilled and thus T_{60} can not be measured.

To overcome this limitation the self-similarity of exponential decay is used. This means that e.g. exponential decays from 0 dB to -30 dB and from -30 dB to -60 dB are of equal length. Thus a doubled exponential 30 dB decay time is equal to T_{60} . Practically the decay time from -5 dB to -35 dB is measured and its double is called T_{30} . Similarly the tripled decay time from -5 dB to -25 dB is called T_{20} .

In addition a weighted mean value of T_{20} and T_{30} is shown as T_{mix} . Weighting is done by the measuring quality so that T_{mix} can be assumed to be almost a 'best possible' measurement in a given situation. It should be mentioned that T_{mix} is primarily intended for beginners in room acoustics and does not refer to any standard.

Finally the 'early decay time' (EDT) is measured as the decay time from 0 dB to -10 dB, multiplied by six. In contrast to T_{20} and T_{30} often showing similar values, EDT can give quite different results.

3.3. Clarity and Transmission Quality Measures

The reverberation time is a statistical measure of the decay of sound energy in a room, and thus it is a property of the whole room. Individual measurements are samples scattering around a mean value called the 'reverberation':

In contrast measures describing the transmission quality from a source (speaker, musician, instrument, etc.) at a specific source location to a receiver (listener, microphone) at a specific receiver position.

These measures are related to the fact, that time variations of speech and music in loudness and spectrum should be well recognizable at the listeners' positions.

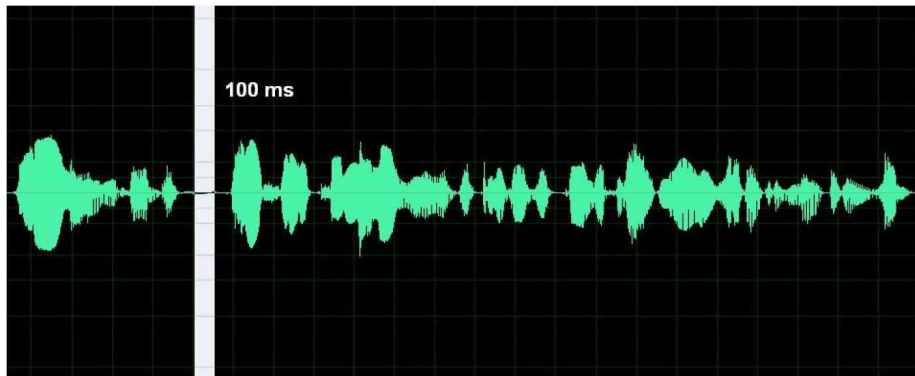


Figure 3.2 An oscillogram of speech

The oscillogram in fig. 3.2 shows about 4.5 seconds of speech. For scaling a short time segment of 100 ms is coloured white. It can be seen that the typical lengths of short articulation elements vary in the range from 50 to 200 ms. To achieve a good speech intelligibility successive articulation elements shall be blended at least possible on the transmission path from the speaker to the listener.

The amount of 'blending' is quantified by 'clarity and transmission quality measures'. The **Deutlichkeit** D_{50} counts signal parts delayed up to 50 ms relative to the direct signal positively, and all later parts negatively. It is defined as follows

$$D_{50} = W_{0..50 \text{ ms}} / W_{0..\infty \text{ ms}}$$

where $W_{0..50 \text{ ms}}$ is the 'early' sound energy from 0 to 50 ms and $W_{0..\infty \text{ ms}}$ is the total sound energy.

D_{50} is dimensionless and ranges from 0 to 1. The higher its value the better and clearer is the speech transmission.

The **Deutlichkeitsmaß** C_{50} is equivalent to D_{50} but expressed in the logarithmic [dB] scale. C_{50} is defined as:

$$C_{50} = 10 \log_{10} (W_{0..50 \text{ ms}} / W_{50..\infty \text{ ms}}) \text{ [dB]}$$

where $W_{0..50 \text{ ms}}$ is the 'early' sound energy from 0 to 50 ms and $W_{50..\infty \text{ ms}}$ is the 'late' sound energy after 50 ms.

The range of C_{50} is not limited. Speech transmission quality increases with higher values of C_{50} .

C_{50} and D_{50} are related by:

$$C_{50} = 10 \log_{10} (D_{50} / (1 - D_{50})) \text{ [dB]}$$

As a high transmission quality example in an acoustically well treated classroom D_{50} values above 0.75 and C_{50} values of +5 dB and more can be observed.

Similar to speech the clarity of music transmission can be quantified by the **Klarheitsmaß** C_{80} , defined correspondingly, but the separation between 'early' and 'late' sound energy is set at 80 ms.

$$C_{80} = 10 \log_{10} (W_{0..80 \text{ ms}} / W_{80..\infty \text{ ms}}) \text{ [dB]}$$

where $W_{0..80 \text{ ms}}$ is the 'early' sound energy from 0 to 80 ms and $W_{80..\infty \text{ ms}}$ is the 'late' sound energy after 80 ms.

As C_{50} also higher C_{80} values indicate a distinct clarity. In the *Teatro alla Scala* in Milano C_{80} values above 0 dB can be observed, a benefit for text intelligibility in the opera. In contrast the *Großer Musikvereinsaal* at Vienna, a world class concert hall for classical and romantic symphonic music, shows C_{80} values up to -5 dB, similar to the *Concertgebouw* at Amsterdam.⁶

A common issue of D_{50} , C_{50} and C_{80} is their hard limit between positively and negatively weighted energy parts. This may result in artefacts when parts of the signal flip between positive and negative weighting due to minor changes in geometry. This does not correspond to any audible effect.

The **Center Time** T_s uses the time itself as a weighting function instead of the step functions of the previous measures, and thus there is no risk to produce artefacts. The Center Time calculates the temporal shift of the 'center of energy' of the sound signal at the receiver with respect to the direct sound.

$$T_s = \frac{\int_0^\infty t W'(t) dt}{\int_0^\infty W'(t) dt}$$

Smaller values of the Center Time indicate a better transmission quality and more clarity. A very good speech transmission quality corresponds to values below about 50 ms.

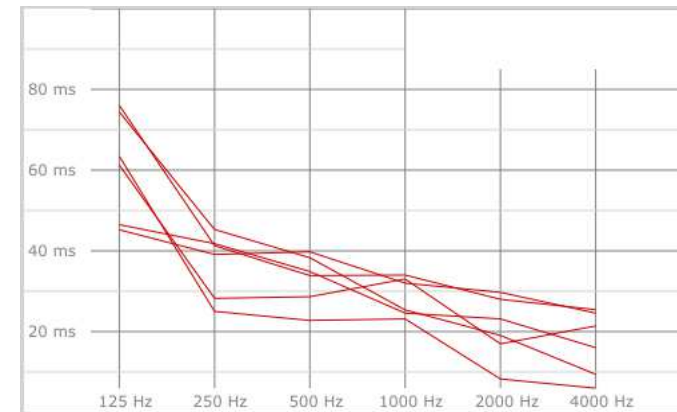


Figure 3.3 Measured octave Center Times of different transmission paths in a classroom

Clarity measurements (C_{80} , C_{50} , D_{50} und T_s) can be based on octaves or even third octaves. As an example in figure 3.3 octave center times measured for six different transmission paths in a fully occupied primary school classroom showing very good transmission quality.

Averaging measured clarity results from different signal paths is not adequate, because clarity measures refer to individual signal paths and not to a room (as reverberation times do).

⁶ It is told that H. v. Karajan added to his praise of the Musikvereinssaal: "... successive notes tend to merge into each other ..." (cited after Beranek). This remark indicates that the wanted effect of envelopment by music in a concert hall may be correlated with a slightly reduced clarity.

3.4. Sound Absorber Data and Planning

Calculations of additional absorber effects (software page 5. Calculation) base on Sabine's formula. In simplified form the formula can be , written as:

$$T_{\text{sab}} = 0.163 V / A$$

where V is the room volume in [m³]

and $A = \sum S_i \alpha_i$

is the equivalent sound absorbing area in [m²]

Each part (i) of A is defined by two values, the geometric area S_i and the sound absorption coefficient α_i (range: 0 .. 1).

Sound absorption coefficients of numerous surface types are tabulated in the literature. Usually absorption coefficients are measured in a similar way: Effects of absorber samples on reverberation times in a reverberation chamber are interpreted.

Often Measurements are done in third octave bands. Octave band values are mean values of three third octave band values. 'Practical' sound absorption coefficients α_p according to ISO 11654 imply also that octave values are rounded to steps of 0.05 and values are limited to a maximum of 1.0. Reasons causing (physically invalid) measured values above 1.0 are discussed in the next chapter.

Interpreting Sound Absorption Coefficients

The definition of the sound absorption coefficient implicitly assumes an absorber surface of infinite size. In real measurements this assumption is approximately fulfilled, if the dimension of the measured absorber is large compared to the sound wavelength. This can be fulfilled

filled at high frequencies. E. g. in the 4 kHz octave band wavelengths range from 6 to 12 cm, small against absorber sizes of a few meters.

At low frequencies relations are quite different: In the 125 Hz octave band wavelengths approximately range from 2 m to 4 m and thus are of similar size or even bigger than common absorbers. In such cases the absorber's effective size is remarkably larger than its geometric size. At each edge the absorber appears to be about a quarter wavelength bigger than its geometric dimension. As an example, an absorber of 2 m x 2 m in (geometric) size will appear as almost doubled in effective size if the wavelength is 2 m (frequency = 170 Hz). This also has the effect that a longish shape of an absorber can increase its effect at unchanged geometric surface, e.g. if the example absorber is transformed to a geometric size of 0.5 m x 8 m.

These effects are often neglected when measuring absorption coefficients in reverberation rooms and when using the measured absorption coefficients for planning a room. Practically the geometric effects mean that measured absorption coefficients are only valid for applications where the geometry and size of the absorbers are similar to the measurement setup (which can be read from well made measurement reports). In all other cases there is a (more or less unknown) uncertainty in planning results. To overcome this uncertainty it may be necessary to check the achieved result by measurement after completion of the room and by adjustment if required.

Room Edge Effects

Besides the absorber shape effects also room edge effects can play a big role, especially at low frequencies. Each normal wave component reflected at a massive slab results in a standing wave with zero velocity at the surface and increasing velocity up to a quarter wavelength distance. In porous absorbers this micro movement in front of a massive slab is reduced by friction. Wherever two slabs are connected in

an edge, this effect occurs in two dimensions, as there are two normal directions. This means that the absorption effect can approximately be doubled in the vicinity to room edges. The 'vicinity' is defined by the first velocity maximum at a quarter wavelength distance from the reflective surfaces. And so this effect practically only plays a role at low frequencies, because the 'vicinity' is too small at high frequencies.

This effect means that low frequency absorbers are more efficient when mounted close to room edges. Specific 'edge absorbers' make extensive use of the effect.

Both edge effects and absorber shape effects partly decouple the absorption effectiveness from geometric size of absorbers and thus introduce uncertainty into the numeric planning.

This uncertainty in calculation could be overcome if the total absorptive effect is separated into a pure surface dependent part and additional length dependent parts for absorber edge and room edge effects. With this separation any absorber geometry could be calculated adequately and more precisely than at present.⁷

A Procedure to Assure Quality

As a result of the addressed effects, and also of uncertainties in material data, deviations between in-advance calculations and built results can be expected, even if planning has been well done

To achieve an approved room acoustic quality it is worthwhile to measure acoustic parameters after interior fitting and to correct if necessary. In case of a number of similar rooms, like classrooms in a school, it can reduce effort and cost to equip and measure one room in advance and to apply the revised acoustic design to all other rooms after having optimised the tested room

⁷ In the absorber database of AcouCheck there are a few (experimental) 'Equivalent absorption width' datasets from edge absorbers that can be used in prognostic calculations.

Appendix

A.1. Didactic Context

There are many opportunities to integrate the AcouCheck software into university courses, seminars, informative meetings and more. How this is done, depends among others on the target audience and their precognition, the learning target, the number of lessons and last but not least on the lecturer's didactic preference.

As a single example the situation at Wuppertal University is shortly summarized here: Architecture covers a wide range of topics, and thus only a small number of lessons is scheduled for teaching room acoustics. At Wuppertal students of architecture get a total of 15 lessons covering all aspects of architectural acoustics during their bachelor studies. Approximately one third of these lessons is available for room acoustics. In consequence of these facts the introduction to general acoustics and to room acoustics is kept short. Aural samples and illustrative experiments are integrated in order to clarify connections between human aural experience and physical description. After this introduction the AcouCheck software is used as a tool to 'throw' the students into practical work. Doing so, it is essential to link this work to acoustic basics and thus to give the students a chance to understand what they are doing.

Students are analyzing seminar rooms or similar using the software in small groups of mostly three. From experience it has been learnt that part of the student groups can do this autonomously, other groups need some assistance from their tutors. It could also be learnt that almost all of the students did this task with commitment.

The basic acoustics course is completed by a visit and guidance to a regional concert hall as the 'climax event'.

A.2. A Short Introduction to Microphones

The most important component in acoustic measurements is the microphone. Usually microphones need a preamplifier to increase their low output voltages. For connecting to a computer an appropriate audio interface is required to convert the analogue microphone signal to the digital computer format. This chapter focuses on different microphone types and on interfacing to computers.

Microphone-Types

Beginning at the best transmission quality – condenser microphones. Since the first condenser microphones by Georg Neumann around 1930, music and speech can be transmitted and recorded in a high quality. Today condenser microphones are mostly used for measurements and for studio and live recordings of music.

In the 1960s electret microphones were developed as a simplified type of condenser microphones, today's widest spread microphone type. Dynamic microphones combining robustness and quite good transmission quality play a big role on stages.

Measuring microphones

Measuring microphones are mostly characterized by their small diaphragm and capsule diameters, often widening towards the output connector. The small diameter results in an almost spherical directional pattern at all frequencies and in minor influence on the sound field. Corresponding to their naming measuring microphones are first class for any room acoustic measurements. Nevertheless room acoustic measurements can also be done using other microphone types, possibly resulting in a reduced quality of the measurements.

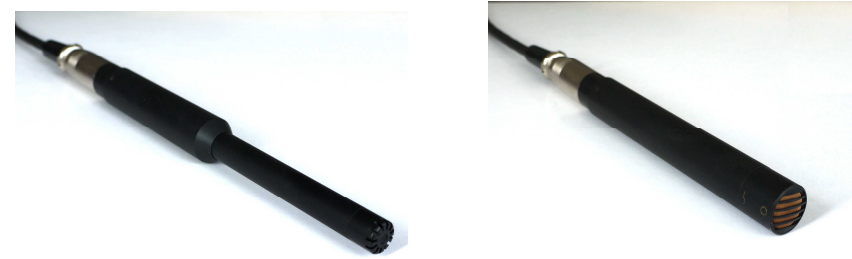


Figure A.1 A measuring microphone (left) and studio condenser microphone showing omni or spherical directivity (right).

Studio condensor microphones

Studio condenser microphones are equal to measuring microphones in transmission quality, but they are designed primarily for high quality music and speech recording. Studio microphones are categorized by two features: directivity pattern and diaphragm size.

In contrast to measuring microphones with their omnidirectional characteristics studio microphones often show unidirectional characteristics as cardioid or super-cardioid. A special pattern is 'figure-of-8' with equal sensitivity from front and from back, but completely insensitive from sides. There are also microphone with switchable or continuously variable directional pattern.

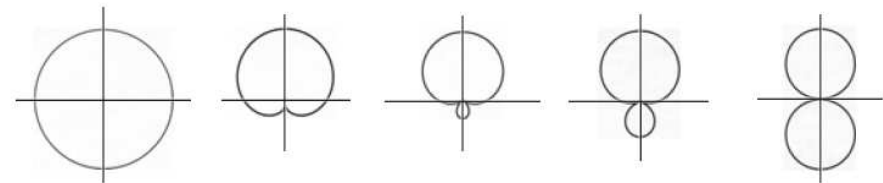


Figure A.2 Directional patterns of microphones (front side to top, from left to right: omni (spherical), cardioid, super cardioid, hyper cardioid, figure-of-8).

Large diaphragm condenser microphones ($\varnothing > 25$ mm) are suitable for recording individual voices or instruments. They show a more frequency dependent directional pattern than small diaphragm condenser microphones (\varnothing typically 10-15 mm). Omni directional small diaphragm microphones often show a very smooth frequency response and can be considered as almost equivalent to measuring microphones for room acoustic measurements. With a few restrictions all studio microphones are suitable for room acoustic measurements, the less directed the better. 'Extreme' directivities like 'figure-of-8' should be avoided in measurements.

Electret-microphones

Electret microphone capsules can be manufactured in small sizes and at very low cost and thus they are part of diverse electronic devices like cellular phones, headsets, computers mobile recorders and many more. High quality electret capsules are also built into some (lower cost) measuring and studio microphones. For room acoustic measurements electret microphone capsules are the second best choice after 'pure' measuring microphones and omni studio condenser microphones. Usually their frequency responses are pretty flat (if not degraded by any internal filters). Due to their small sizes the self generated noise is often higher than from other microphone types.

Dynamic microphones

Dynamic microphones are mainly used to pick up instruments or voices on stage, where they benefit from their mechanical and environmental robustness. Their transmission performance is fairly good but does not compete with condenser microphones. Dynamic microphones can also be used for room acoustic measurements. Similar to condenser microphones they are better suited if less directional. Most dynamic microphones show cardioid or super cardioid directivity. Dynamic omni microphones are rather seldom.

Preamplifiers and audio interfacing

Connection of a microphone to a computer is usually done by a combined microphone preamplifier and analog-to-digital conversion, called 'audio interface' or 'soundcard'. In most PCs a soundcard is integrated. External audio interfaces are usually connected via USB, but also FireWire, Thunderbolt or other connections can be used.

Microphone connection in consumer devices

Microphone preamplifier and microphone must be compatible. Most consumer electret microphones require a supply voltage (typical 5 V) from the preamplifier. Common connector wirings for electret microphones and consumer audio interfaces are shown in Figure A.3.

Microphone connection in professional equipment

In professional stage and studio equipment different connection types are used. Audio signals go through 'symmetric' cables on two separate conductors with inverse polarity. Mass and shield conductors are not connected to signal lines. Thus long cables can be used at low noise caused by external electromagnetic fields.

Most audio interfaces supply the 'phantom power' (48 V DC, + pole on both signal lines and – pole on shield) required by professional condenser microphones. Professional microphones mostly use robust XLR type connectors.

Professional audio interfaces often include a 'monitor' option looping the microphone signal through to the line or speaker outputs. When used for sine sweep measurements, it is absolutely necessary to deactivate this option. Otherwise measurements will be invalid and feedback noise may occur.

Consumer and professional connections are mechanically and electrically incompatible to each other. Practically this means that both

microphone and preamplifier must be of the same 'device family', both 'professional' or both 'consumer'.

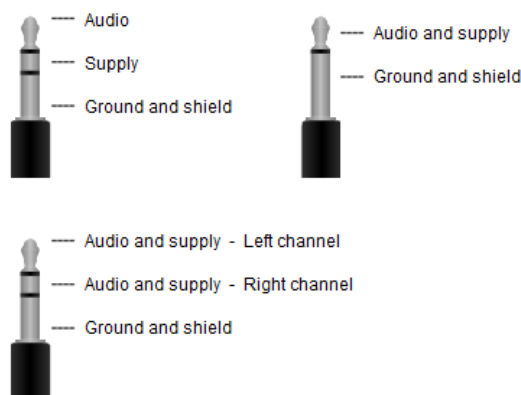


Figure A.3 Electret microphone connectors. Most consumer microphones soundcards and audio interfaces use a 3.5 mm TRS (3 pole) jack with separate audio and power poles (top left). If a TS (2 pole) jack is used (top right), audio and supply share one pole and require an adaptor for TRS connections. Stereo microphones always use the shared poles (bottom)



Figure A.4 Two examples of audio interfaces: A simple 10 Euro consumer USB interface (left), and a highest quality professional interface at 100 times the cost (right, photo: RME)

A.3. The Sound Absorber Database

The Structure of the Database

The sound absorber database is divided into two parts⁸:

The basic part of the database is fix and integrated into the software. It contains a number of 'showcase' absorbers, absorption data from different constructions, persons and furniture and several more 'exotic' absorbers.

The second part of the database is variable and is particularly used for absorber data from manufacturers. These data are automatically downloaded from the internet and stored locally. At each time the computer is connected to the internet while AcouCheck is started an automatic check for new data is done in the background and local data will be updated if required.

The variable part of the database is locally stored in the subdirectory 'absdata' of the 'AcouCheck data location' (see Appendix A 6). It is not recommended to modify the locally stored data manually, as this may result in malfunction and in unwanted behaviour of the software.

⁸ The AcouCheck 2 data base format is not compatible with data from the software version 1.0. The 'old' data cannot be used with AcouCheck 2.

A.4. Measurement Accuracy Considerations

There is no absolute reference for reverberation measurements like a prototype reverberation chamber, analogous to the former prototype meter bar in Paris (or newer length references). Instead measuring algorithms are defined, mainly in ISO 3382. If these algorithms are applied digitally to a sampled measuring signal, the accuracy of reverberation times and other time related measures is reduced to the sampling clock accuracy. As computer clocks usually are quartz locked, time related errors can be assumed as very low.

Algorithm Tests

In any digital signal processing the possibility of software errors must be considered. Such errors can be unremarkable and not easily detectable in a complex software. One precaution is that the software needs frequent plausibility tests of its function and its results during the development. This will probably detect malfunctions, but algorithmic errors may pass these tests.

Algorithmic accuracy tests can be performed by comparison to other established measuring systems, that can be considered as well tested 'quasi standards'. If more than one reference is used for comparisons, the risk of remaining undetected errors is even lower.

Comparison of Reverberation -Algorithms

Algorithms of the AcouCheck-software have been compared to the algorithms of three common room acoustic software applications, each basing on ISO 3382.

Each of the applications including AcouCheck were used to evaluate identical impulse responses from several rooms. No pair of two applications agreed exactly (< 0.01 s) in all samples. But in most cases results from all of the four applications agreed within 0.02 to 0.05 s.

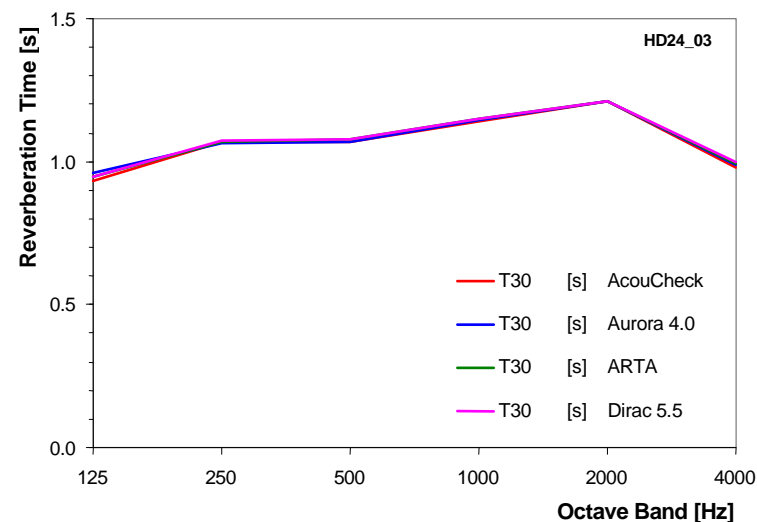


Figure A.5 An example from the algorithm comparison of AcouCheck and three other software applications.

Sporadic deviations up to 0.1 s from all of the others was observed in results of each application. These results indicate that ISO 3382 algorithms have been well implemented in all of the four applications. Small deviations are not surprising, because ISO 3382 includes a few algorithmic alternatives.

Further Algorithm Tests

Clarity and transmission quality measures basing on the relation between early and late sound energy (T_s , C_{80} , C_{50} , D_{50}) do not evaluate the decay slope as reverberation times do. Instead sound energy multiplied by a time dependent weighting function is integrated. In T_s the time itself is used as weighting function, whereas in C_{80} , C_{50} and D_{50} a step function is applied, with a step at 80 ms or at 50 ms respectively. The time scale of the weighting functions starts at the time of the direct sound in the room impulse response.

This leads to a critical point, because the zero time can easily be set to the steep rise of the direct signal in the unfiltered impulse response, but after octave filtering the 'direct' peak is smoothed especially in the lower octaves, so that any zero point setting becomes fuzzy and unstable. In addition the filters cause a signal delay, increasing with decreasing frequency. A practical solution is to set the zero time from the unfiltered signal and to add an individual time shift in each octave to compensate for the filter delay.

During the development of AcouCheck this was implemented by measuring center times (Ts) of 1 sample impulses (almost a Dirac impulse) which should result in 0 ms in all octaves. In a first step filter delays were not compensated, and measured results showed the octave filter delays in each octave. These measured filter delays were integrated as delay compensations into the software code. Check measurements resulted in residual deviations below 2 ms at lower octaves and far below 1 ms at higher octaves.

Acceleration Signal Effects

Using a sine sweep as acceleration and then convolving the received signal with an inverse sweep gives a good approach to an ideal measurement using a high energy Dirac impulse. The effective acceleration is only 1 sample long ($< 25 \mu\text{s}$), but broadened by the response functions of the used equipment (amplifier, loudspeaker and microphone). Results are reproducible exactly, apart from noise events during the measurements.

In contrast impulse accelerations usually are broader in time and scatter in their shapes and in their directivities. As a consequence results cannot be expected to be reproducible exactly.

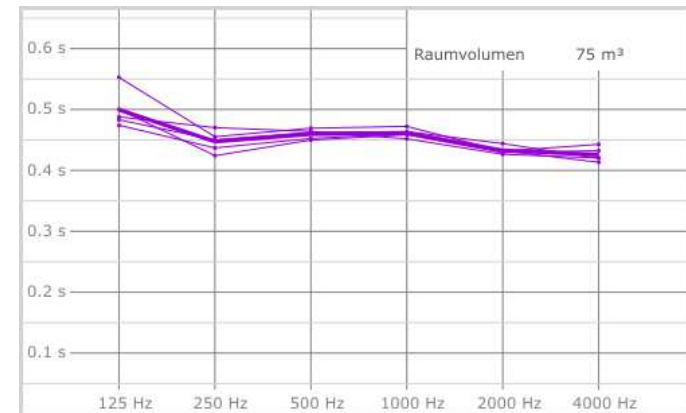


Figure A.6 Five repeated measurements using strong handclap accelerations as an example of reproducibility.

As can be seen in figure A.6, results of repeated impulse accelerated measurements scatter around their mean values. The example shows results spreading about 0.02 s at higher octaves and about 0.03 s at lower octaves, and one spike 0.05 s at 125 Hz. In this case reproducibility is better than the expectable scattering of measured results from different positions of sound source and microphone. Thus lacking reproducibility is no dominant source of errors in this case. This quality level of impulse accelerated measurements requires a very low background noise level and sufficient impulse energy in all measured octaves. Practically in case of any doubt about reproducibility a series of test measurements similar to figure A.6 can give information about the quality to be expected.

Nonlinear and Time-variant Signal Paths

Measuring reverberation times and other room acoustic parameters requires a linear and time invariant signal path of the measurement system. When using professional audio equipment with no effects inserted this condition is fulfilled in the range from the noise level up to clipping.

Internal microphone signal paths of numerous notebook-computers include signal manipulations in their standard settings, e.g. to improve speech transmission for VOIP⁹. These signal manipulations introduce distortions to room acoustic measurements and prevent valid results and thus need to be switched off for any measurements.

In contrast to notebook-and desktop computers, iOS and Android driven tablet computers do not offer an option to adjust the microphone gain-correctly for the input signal. Instead at high levels the signal is limited to avoid clipping of the AD conversion, introducing a nonlinear and time invariant signal manipulation. These devices can only be used for measurements below the limiter threshold. In figure 7 an echogram from a room impulse response recorded twice in parallel on two devices is shown:

- a) professional equipment showing a wide linear dynamic range,
- b) the internal signal path of a tablet computer, where the dynamic compression can be well observed. On this specific device the delayed release results in a reasonably linear decay from -5 dB down to -35 dB, and thus T_{20} and T_{30} can be measured at an acceptable error level. Other room acoustic parameters derived from this impulse response are invalid.

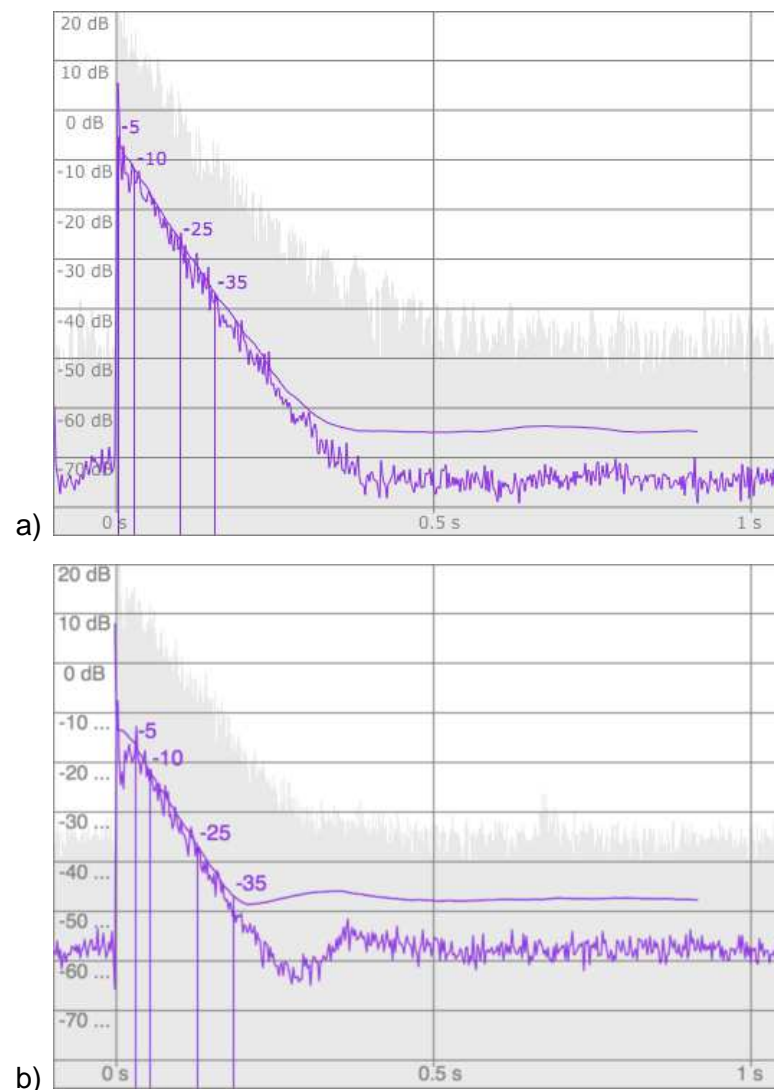


Figure A.7 An octave echogram of a room impulse response
a) recorded by professional audio equipment.
b) recorded via the internal signal path of a tablet-computer.

⁹ Voice over IP

A.5. Literature and Planning Tools

Room Acoustic Literature (a small selection)

W. Fasold, E. Veres: Schallschutz und Raumakustik in der Praxis;
2nd Ed, Berlin, 2003 (3rd Ed. scheduled Dec. 2107)

W. Fasold, E. Sonntag, H. Winkler: Bau- und Raumakustik;
Köln, 1987 (out of print, contains numerous absorber-data)

H. Kuttruff: Room Acoustics; London 2006

L. Beranek: Concert Halls and Opera Houses;
2nd Ed., New York, 2004

Planning Tools and Standards (a small selection)

ARTA: Room acoustic / audio measuring software; www.artalabs.hr

Aurora: Room acoustic measuring software; www.angelofarina.it

CATT-Acoustic: Room acoustic simulation software. www.catt.se

DIN 18041: German standard on room acoustics.

DIN EN ISO 3382: Messung von Parametern der Raumakustik.

DIN EN ISO 11654: Schallabsorber für die Anwendung in Gebäuden.

Dirac: Room acoustic measuring software;
www.acoustics-engineering.com

ODEON: Room acoustic simulation software. www.odeon.dk

PTB – Soud absorber-database; see www.ptb.de

Zorba Sound absorber calculation software. www.rahe-kraft.de

A.6. File Systems and Locations

The Location AcouCheck-Data

All data used by AcouCheck are located in one place of the computer's storage (hard disk or flash memory) called 'AcouCheck data location'. The location depends on the operating system.

On Windows computers the data location is identical to the program location. Thus AcouCheck is a 'portable' software and can also be used from an USB stick.

On OS-X computers data are located at '<user>/Library/ACM2'.

On iOS devices a subdirectory 'ACM2' will be created at an internal location not suitable for external access.

On Android devices a subdirectory 'ACM2' will be created in the directory called 'SD card'.

Saving of Room Impulse Responses

At first run of AcouCheck a sub-directory 'imp' of the 'AcouCheck data location' is created. Each time a new series of measurements is started (when the first measurement is finished) a subdirectory of 'imp' is created and named according to the format 'JJJJMMTT_hhmm'. At this location the impulse responses of the individual measurements are stored automatically in the background. The impulse responses are always named 'impulse01.flac' to 'impulse12.flac'. As an example, if a series of measurements is started at date 2016-10-16 and time 9:55 h (time in 24h format, no AM and PM), the corresponding subdirectory will be named '20161016_0955'

Starting at version 1.3x, AcouCheck writes room impulse responses formatted as 'FLAC' files (free lossless audio compression). Room impulse responses written as 'FLAC' are reduced in size by a factor of

four or more compared to the same impulse written in an uncompressed format (e.g. 'WAV')¹⁰¹¹¹²

All stored room impulse responses can be reloaded on AcouCheck page 2 for later evaluation. On Windows computers also old 'wav' formatted room impulse responses can be loaded.

The Location of the Sound Absorber-Database

The variable part of the database (see Appendix A.3) is locally stored in the subdirectory 'absdata' of the 'AcouCheck data location'. It is not recommended to modify the locally stored data manually, as this may result in malfunction and in unwanted behaviour of the software.

On iOS devices a subdirectory 'ACM2' will be created in the software sandbox and is not accessible from outside.

On Android devices a subdirectory 'ACM2' will be created in the directory 'SD card'.

¹⁰ Impulse data from the 1st program version were written as uncompressed 'WAV' files. These files can still be read by the latest Windows versions for re-processing.

¹¹ In lossless compression no information at all is lost. Compression is achieved only by reducing redundancy. This is in contrast to lossy compression type like 'mp3', where inaudible or even hardly audible parts of the audio signal are omitted. This is why lossy compression is not suitable for measurement signals.

¹² The degree of compression of FLAC (and other lossless compression types) is signal dependent and thus FLAC impulse response files vary in size although all impulses from AcouCheck are of the same length.

A.7. System Requirements and Installation

System Requirements

Die Software AcouCheck ist für Notebook- und Tablet-PCs mit verschiedenen Betriebssystemen geeignet:

- Windows (version XP-SP3 or higher)
- OS-X (version 10.6. or higher)
- iOS (version 7 or higher)
- Android (version 4.03 or higher).

The computer requires an internal or external microphone connected to an appropriate microphone input. Details about microphones and audio interfacing is found in Appendix A.2.

Installation

The installation procedure depends on the operating system:

Windows

Installation is simply done by unpacking the delivered Zip-file to any preferred location on the computer's hard disk or flash drive. It is recommended to use a directory <preferred location>\AcouCheck. If an older version (version number 1.3x or higher) exists on the computer, it may be replaced. For convenience a link may be created in the start menu or on the computer desktop. For portable use AcouCheck can also be installed on an USB stick.

When a large number (hundreds) of measurements have been done it may be worthwhile to delete unused impulse files from time to time.

Uninstalling is just done by deleting the file 'AcouCheck.exe' and the directories 'imp' and 'absdata' including all their content.

OS-X

Installation on OS-X computers is done as usual by starting the installation file 'AcouCheckXXX.pkg', where 'XXX' denotes the program version.

iOS

Installation on iOS devices is only possible from the app store¹³. Installation directly from a file is impossible¹⁴.

Android

Installation on Android devices is done by starting the installation package 'AcouCheckXXX.apk', where 'XXX' denotes the program version. The installation requires three steps:

1. In Android settings → security → unknown sources
'Allow installation from unknown sources' ON.
2. Starting the file 'AcouCheckXXX.apk' - the installation is done
3. In Android settings → security → unknown sources
'Allow installation from unknown sources' OFF.

First Start

After installation AcouCheck needs a first program start while the computer is connected to the internet (important: the internet connection must exist before AcouCheck is started). In this procedure the manufacturer supplied part of the absorber database is downloaded and stored locally on the computer. Depending on the download speed this may require up to several minutes. Now the AcouCheck software is ready for operation (offline or online). Each time when con-

¹³ The iOS version of AcouCheck is scheduled of 3rd quarter 2017.

¹⁴ Except the system has been freed from its limitations by a 'jail break'. This procedure is risky and requires technical knowledge.

nected to the internet on start AcouCheck is checking the absorber database in the background and updates if required.

Start Parameter Control

Some software settings of the PC version can be controlled by start parameters if required.

The user interface language can be set independently of the system language by:

<code>lang=DE</code>	switches the user interface language to German
<code>lang=EN</code>	switches the user interface language to English

The aspect ratio of the program window can be set by:

<code>ratio=4:3</code>	sets the aspect ratio to 4 : 3
<code>ratio=3:2</code>	sets the aspect ratio to 3 : 2
<code>ratio=16:10</code>	sets the aspect ratio to 16 : 10
<code>ratio=16:9</code>	sets the aspect ratio to 16 : 9

The width of the program window can be set by:

<code>width=1200</code>	sets the width to 1200 pixels (as an example).
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A.8. Licensing, Copyright, Disclaimer

The software AcouCheck may be freely used for education and for information. It may not be used commercially. It may not be used as a planning tool in commercial projects. It may not be used commercially as a consulting tool.

The software is delivered AS IS. The software is always used at the user's own responsibility and the user's own risk.

It is explicitly stated that acoustic measurements and acoustic calculations may give inexact results due to their methods and due to imprecise input data.

In the software code of AcouCheck the software library Juce (licensed from J. Storer) and free FFT routines by Laurent de Soras are used.

A.9. Project Information

Project Team:

Prof. Dr.-Ing. Karsten Voss, Bergische Universität Wuppertal (D),
Fakultät Architektur und Bauingenieurwesen - btga,
www.btga.uni-wuppertal.de (Projekt coordination)

Dr. Detlef Hennings, freelance scientist, developer and lecturer,
Köln (D), www.eclim.de (Concept and design, software-
development, didactic tests)

Dipl.-Ing. Edwin Rotzal, Bergische Universität Wuppertal, Fakultät
Architektur und Bauingenieurwesen, www.btga.uni-wuppertal.de
(Reference measurements, technical tests)

Advisory Board:

Dipl.-Ing. Sebastian Goossens, Institut für Rundfunktechnik (IRT),
D-80939 München, www.irt.de

Karlheinz Stegmaier, M.A., Akustikbüro Krämer & Stegmaier,
D-10553 Berlin, www.akustik-berlin.de

Dipl.-Ing. Jörg Wuttke, Jörg Wuttke Consultancy,
D-76327 Pfinztal, www.ingwu.de

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www.forschungsinitiative-zukunft-bau.de

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Lahnau Akustik, D-35633 Lahnau-Dorlar, www.lahnau-akustik.de

Silentrooms, D 60439 Frankfurt, www.silentrooms.de

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Akustik & Raum, CH-4600 Olten, www.akustik-raum.ch

Caparol, D-64372 Ober-Ramstadt, www.caparol.de

OWAconsult, D-63916 Amorbach, www.owaconsult.com

