Dr · Jordan · Design

AkuLap

Professional Audio Measurement System

www.dr-jordan-design.de

User Manual for room acoustics ISO 3382

Dr · Jordan · Design

Preface

Modern acoustic measurement systems, such as **AkuLap**, offer a variety of sophisticated methods of measurement. These processes are very complex and require normally a long-term training.
Therefore, AkuLap has automatic measuring methods, which allow measuring reliably many acoustic parameters by non - specialists. The experience of our designers and acousticians has been summarized in this automated system. The goal is to require as little as possible user interaction. At the same time, we can prevent operator errors. For experienced users we have many advanced features even for very special use cases.
The measurement results are written in a clear report. Therefore, you can print this report to easily archive or send them by email.

In this Quick Start Guide, the most important information is summarized. For more information, please refer to the general guidance of AkuLap or the advanced literature.

Warning

This analysis system can generate many synthetic signals. With inappropriate signal levels you can easily damage your equipment (e.g. loudspeakers) or your ear.

Therefore always start with low signal levels and increase the volume carefully.

Wear ear protection.

AkuLap is a professional tool for powerful real-time signal and system analysis.
By using the PC environment, it is not only a cost efficient replacement for classical laboratory equipment. AkuLap offers more powerful features combined with a comfortable user interface. Typical applications are acoustic measurements, room and building acoustics and noise monitoring.
If you run Akulap on a notebook or even a tablet PC, you can easily build a mobile measurement system.

Akulap can use many different front ends. In most cases, you will have the most advanced setup, which consists of a Laptop and a USB measurement microphone.

1 General Description

With this compact measuring system, you can perform a wide range of acoustic measurements in a simple way.

The system consists of a computer and one measuring interface that you connect via USB. Depending on your requirements and you can connect high-precision type approved measurement microphones.

- By using the PC environment you get a large user interface
- Short learning curve
- Measurement results can be quickly and easily exported for documentation as a graphic or table
- Modular system: You can use different vendors manufacturer for measurement microphones.
- Modern computers provide high computing power. The functions are available for advanced analysis. Today measurements are possible, which could not be offered so far in this price range.

1.1 Features

- With the large display, it is very user-friendly compared to classic hand held sound level meters
- Simple measurement of parameters according to ISO 3382.
- Supports measurement via impulse, noise or sinus sweep (LOG Sweep/Chirp)
- Includes wizard for fast and easy measurements. Training phase is very short. Automatic error detection
- Generates reports (HTML / PDF) according to DIN 18041 ISO 3382
- Database manager to organize the measurements
- Average of different measurement positions
- Simple export to Excel
- Measures RT60 with 1/3 octave resolution
- Import and Export of room impulse responses from/to .wav files
- Schroeder plot
- EDT T20 T30
- Clarity and definition
- STI, RASTI and STIPA (IEC 60268-16)

2 Room acoustics

The reverberation time is the most widely parameter to judge on room acoustics. This time (in most cases RT60) measures the duration until the sound level in a room decreases by 60dB after the sound source has been shut down.

2.1.1 Requirements for reverberation measurement

To setup a complete system for room acoustics you will require the following components.

- Akulap with Laptop/Tablet
- Measurement microphone
- Tripod for the microphone
- (Omni directional)-Loudspeaker with amplifier
- Optional sound level calibrator

For measurements of the reverberation time, a sound level calibrator is not required. But this unit with its reference level helps to identify and document e.g. noise sources. In addition, you can verify the measurement chain with one step.

2.1.2 Basic procedure

A measurement requires the following steps:

1. Positioning and setup of loudspeakers and microphone
2. Optional calibration of the microphone
3. Automatic measurement and report generation
4. Repeat the measurement for different positions
5. Average all results
6. Final report generation

All other settings are configured by Akulap itself. Therefore, you will require little training, only.

All measurement results are written into an comprehensive (HTML/PDF) report, which you can print or distribute. It contains in addition to the measurement results, also the original impulse response for later analysis.

2.1.3 Measurement techniques

There are the following methods to measure the reverberation time:

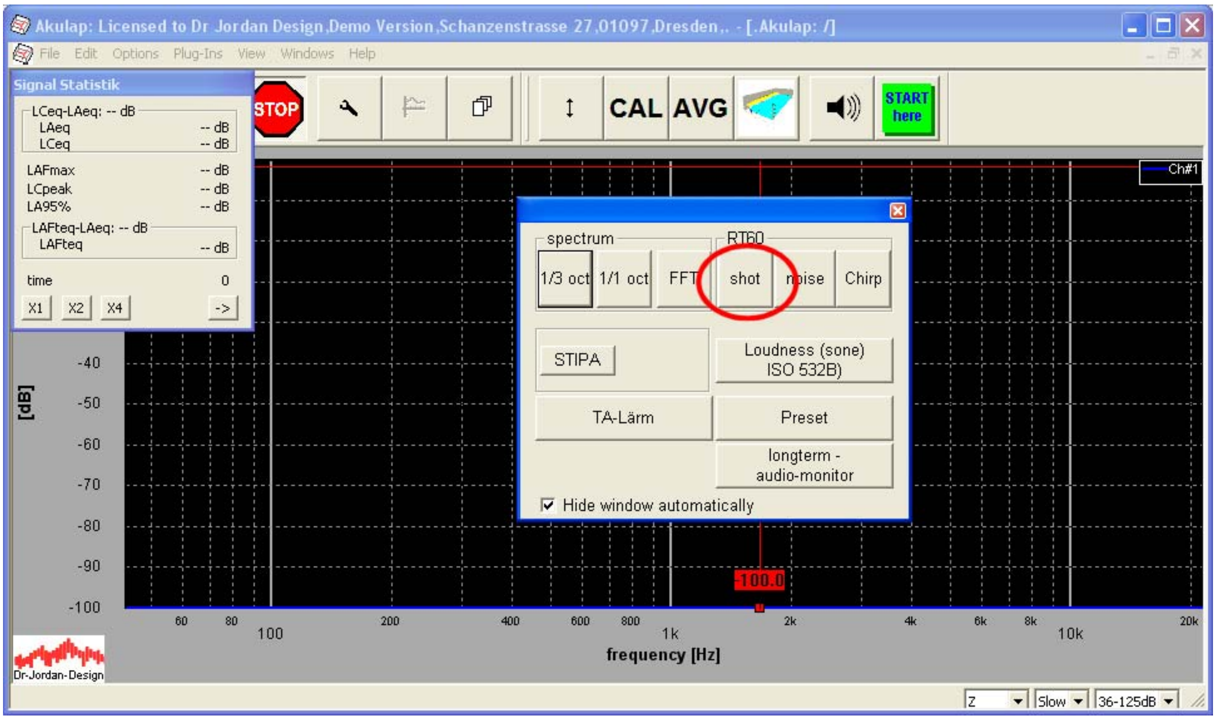
- Excitation with impulsive signals: shots, explosives, hand claps etc.
- Excitation with bursted noise
- Excitation with pseudo-noise sequences Log-Chirps

The direct method with impulsive signal is the easiest technique, because we do not need a loudspeaker. Therefore, we recommend starting with this method. Even simple hand claps are sufficient for small rooms. For advanced measurements, we recommend the Chirp method. Bursted noise should be used only for historical reasons.

3 First steps

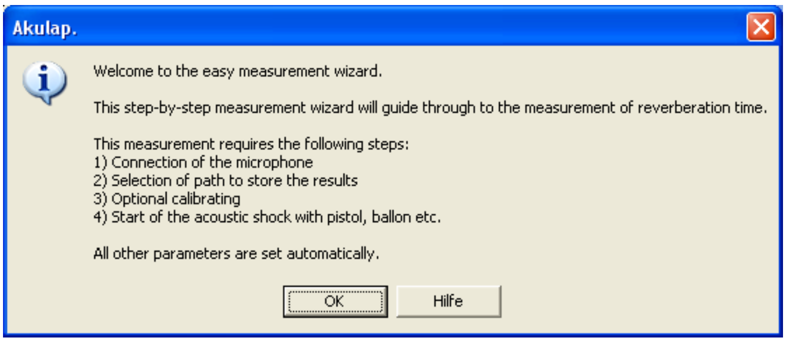
We start with a simple measurement via impulse.

First, place the measurement microphone, connect it to the PC and start Akulap.

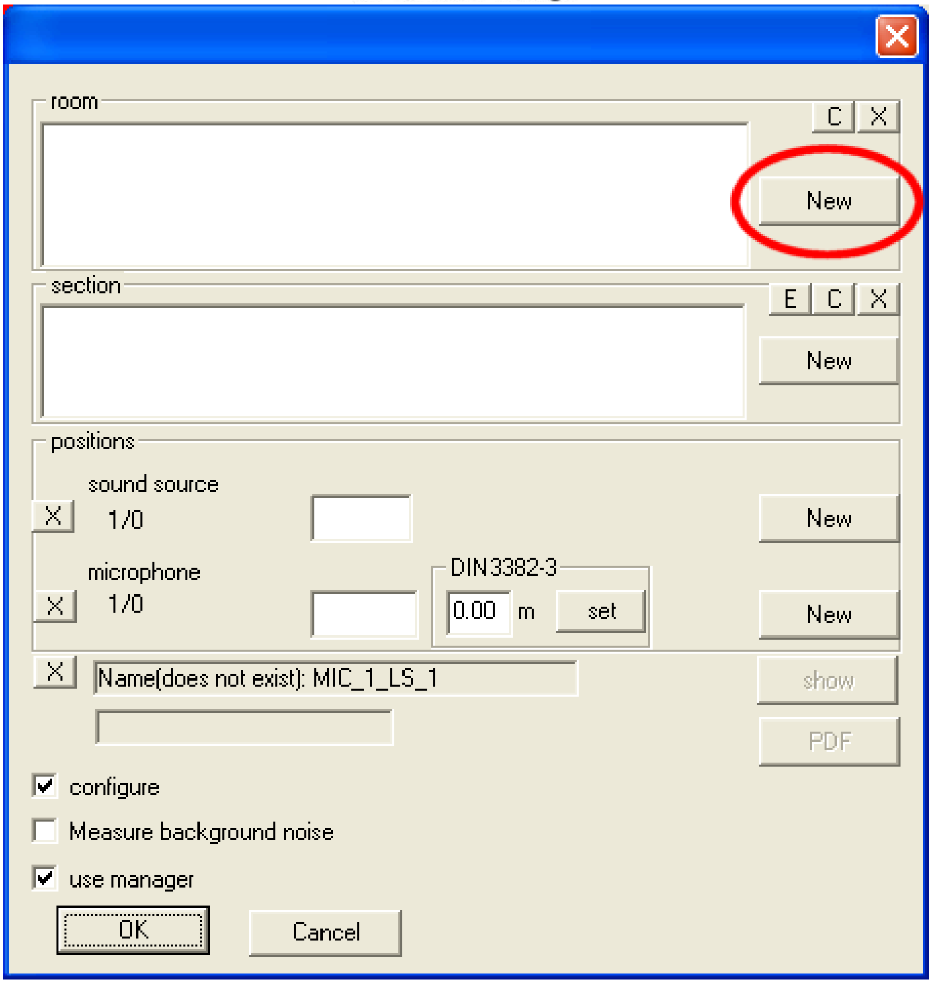


Select SHOT from the preset menu.

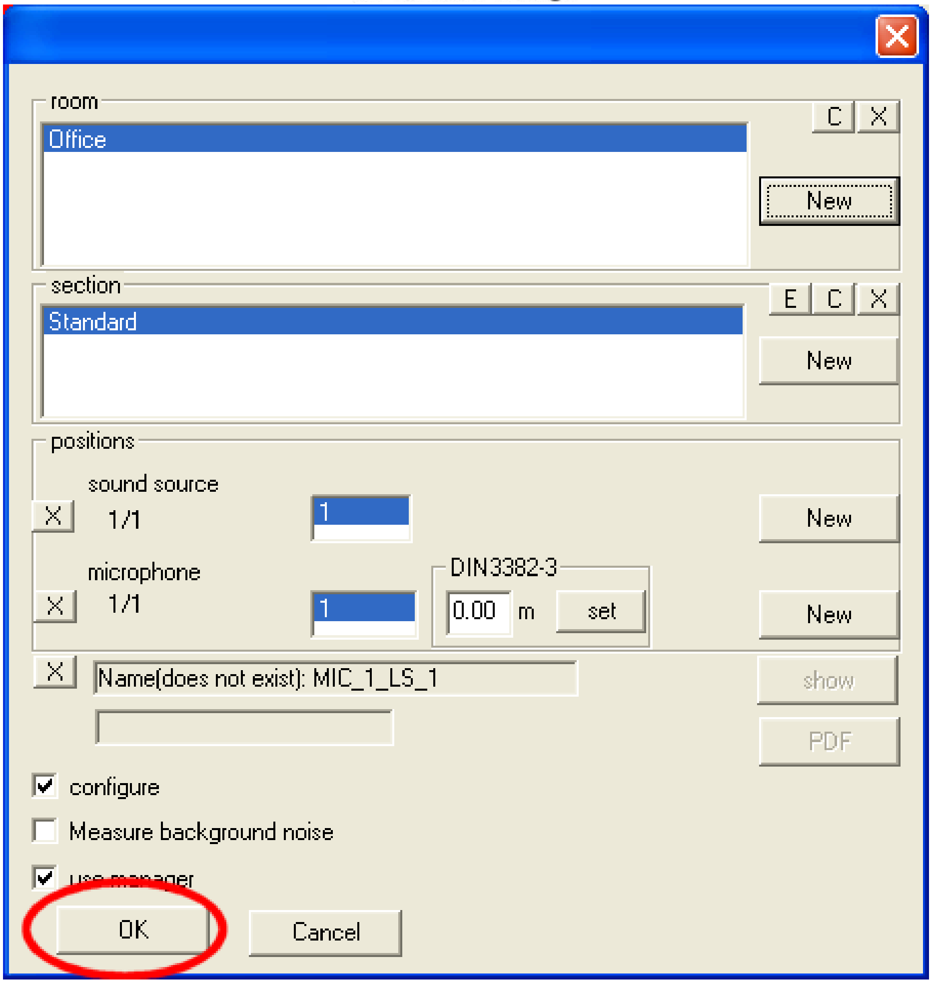
Akulap will show some basic summary of the measurement.



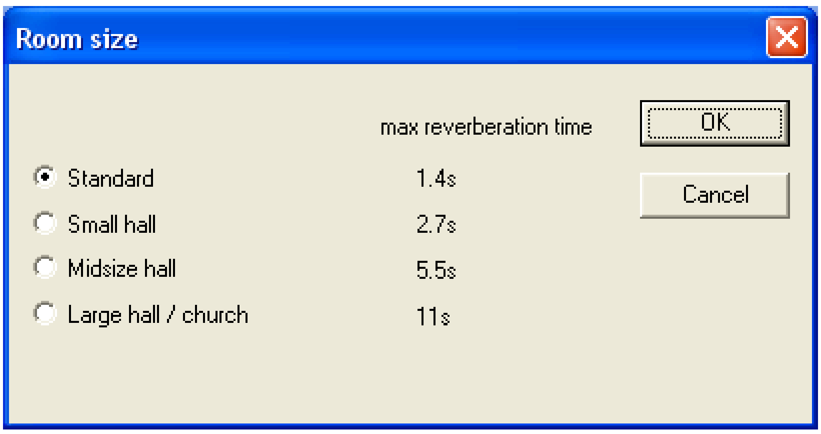
Press OK. The measurement manger will appear.



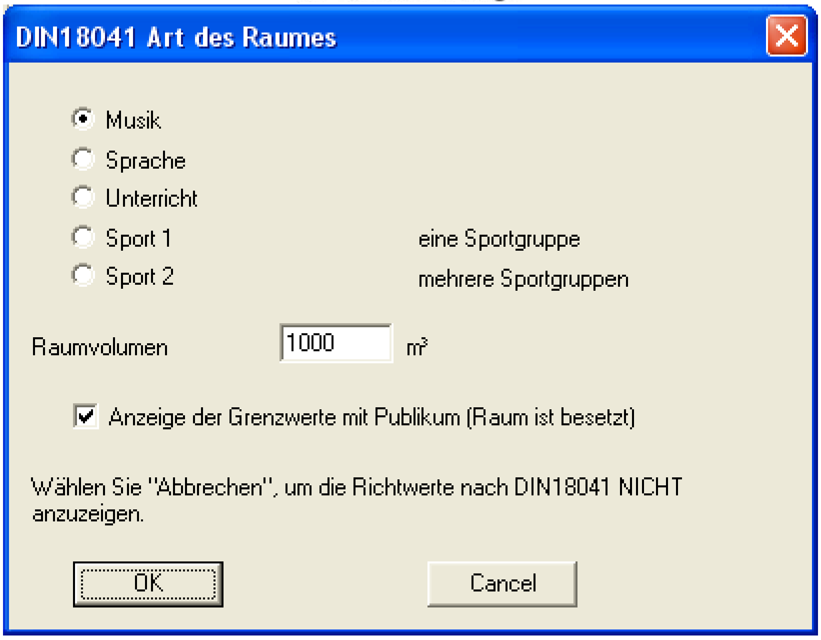
Enter a new room with room/new.



For this first measurement, there is nothing else to configure in this manager window Simply, press OK.

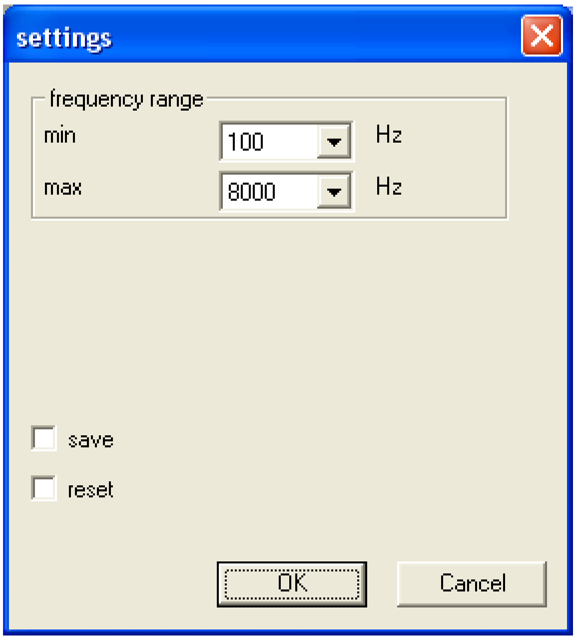


Select a rough estimation for the reverberation time.

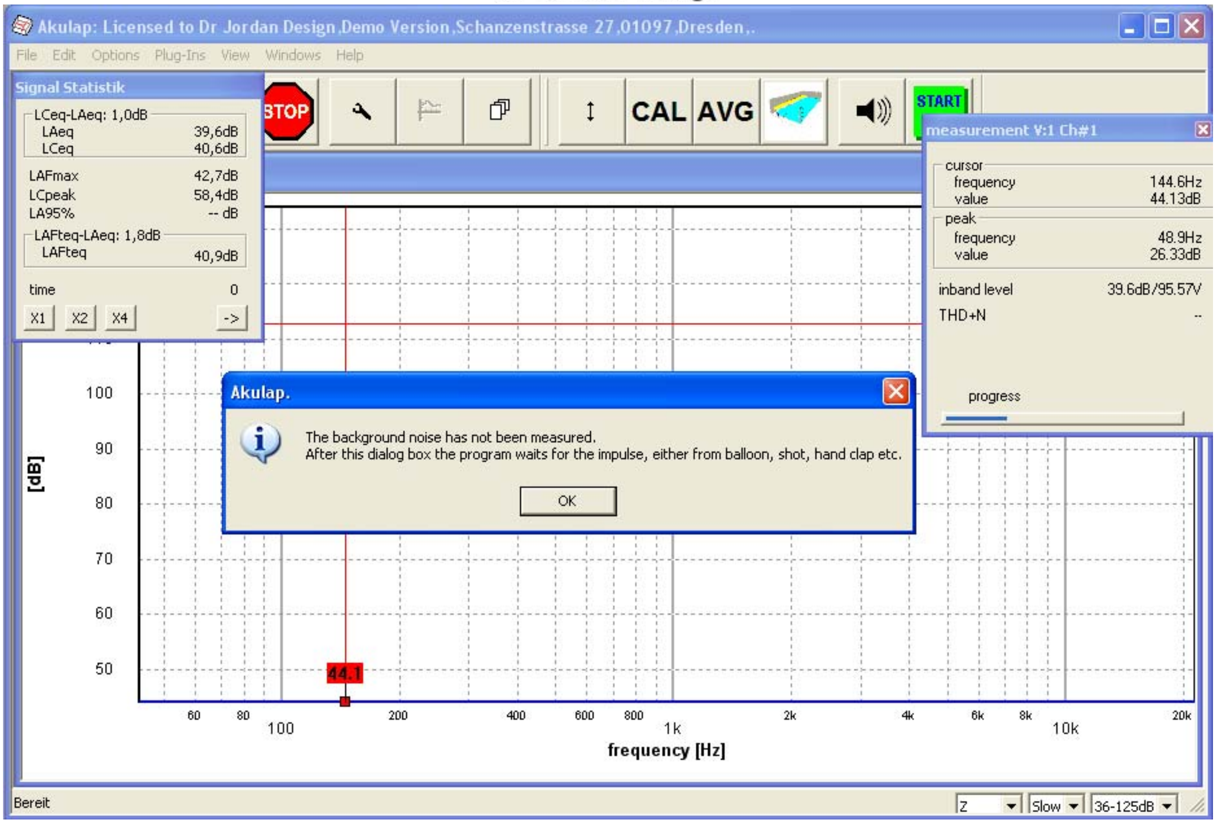


This dialog configure the limits according to the german standard DIN18041. Press cancel to no display any limits for typical rooms.

Enter the frequency limits



Press OK if you modified the settings or cancel to continue.



You should now avoid any ambient noise. Once, you pressed OK, Akulap waits for the trigger signal.

Start your shot, impulse etc. Akulap will start processing immediately. After a few seconds your internet browser will open and shows the measurement results.

Datei Bearbeiten Ansicht Chronik Lesezeichen Extras Hilfe

Measurement of reverberation tim...

file:///C:/Dokumente und Einstellun

Suchen

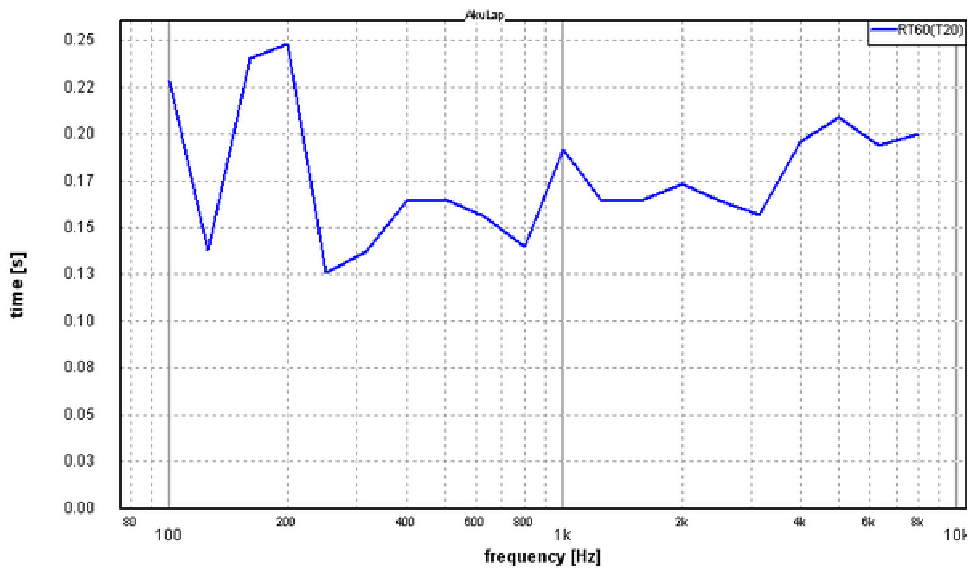
Measurement of reverberation time summary report

room: Office

section: Standard

measurement: MIC_1_LS_1

Saturday, April 18, 2015 17:06:57



reverberation time (83.3Hz to 9600.0Hz) RT60(T20)=0.16s error index=13

Room low-cut frequency (V=1000.0m³)=23.1Hz

For frequencies below this cut-off frequency, the soundfield contains modes. It is not difuse.

For frequencies below this cut-off frequency, the soundfield contains modes. It is not difuse.

definition

At the bottom you will find a link for detailed information and the formatted PDF document.

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Datei Bearbeiten Ansicht Chronik Lesezeichen Extras Hilfe

Measurement of reverberation tim...

file:///C:/Dokumente und Einstellun

Suchen

frequency [Hz]	1000	1250	1600	2000	2500	3150	4000	5000	6300	8000
RT60(T20) [s]	0,19	0,16	0,17	0,17	0,16	0,16	0,20	0,21	0,19	0,20
error index	17	9	66	58	14	49	11	6	2	2

reverberation time in 1/1 octave bands

frequency [Hz]	125	250	500	1000	2000	4000	8000
RT60(T20) [s]	0,24	0,16	0,16	0,16	0,17	0,19	0,20
error index	34	20	27	17	33	4	1

Speech intelligibility STI/RASTI

	125Hz	250Hz	500Hz	1000Hz	2000Hz	4000Hz	8000Hz
0,63Hz	1,00	1,00	1,00	1,00	1,00	1,00	1,00
0,80Hz	1,00	1,00	1,00	1,00	1,00	1,00	1,00
1,00Hz	1,00	1,00	1,00	1,00	1,00	1,00	1,00
1,25Hz	0,99	1,00	1,00	1,00	1,00	1,00	0,99
1,60Hz	0,99	1,00	1,00	1,00	1,00	1,00	0,99
2,00Hz	0,99	0,99	0,99	1,00	1,00	0,99	0,99
2,50Hz	0,98	0,99	0,99	0,99	0,99	0,99	0,98
3,15Hz	0,97	0,99	0,98	0,99	0,99	0,98	0,97
4,00Hz	0,96	0,98	0,97	0,98	0,99	0,98	0,95
5,00Hz	0,94	0,97	0,96	0,97	0,98	0,96	0,93
6,30Hz	0,92	0,95	0,94	0,96	0,97	0,95	0,89
8,00Hz	0,89	0,93	0,91	0,94	0,96	0,93	0,85
10,00Hz	0,85	0,90	0,87	0,91	0,95	0,90	0,79
12,50Hz	0,81	0,86	0,83	0,88	0,93	0,87	0,73
AVG	0,93	0,95	0,94	0,96	0,98	0,95	0,91

Rating:

STI	0,95	excellent
RASTI	0,97	excellent

detailed results
Results as PDF file

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Report reverberation time DIN3382

Information for room and setup

Room location and description

Room capacity

Description wall and ceiling

Number of persons

Description movable items

Temperature and relative humidity

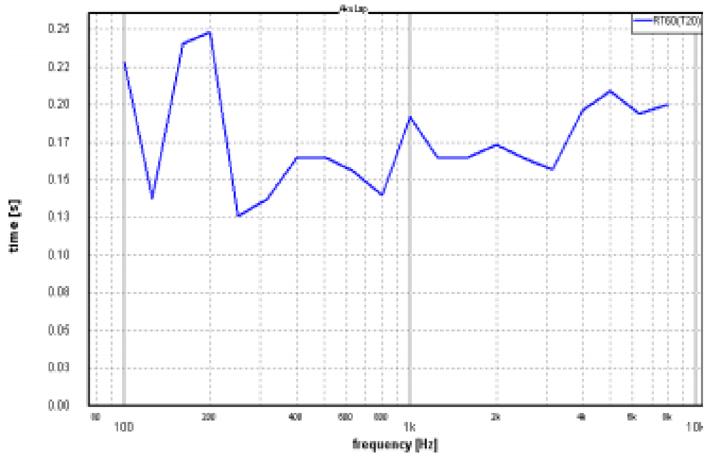
Measurement equipment

Sound signal

Measurement points

reverberation time RT60 in 1/3-octave bands

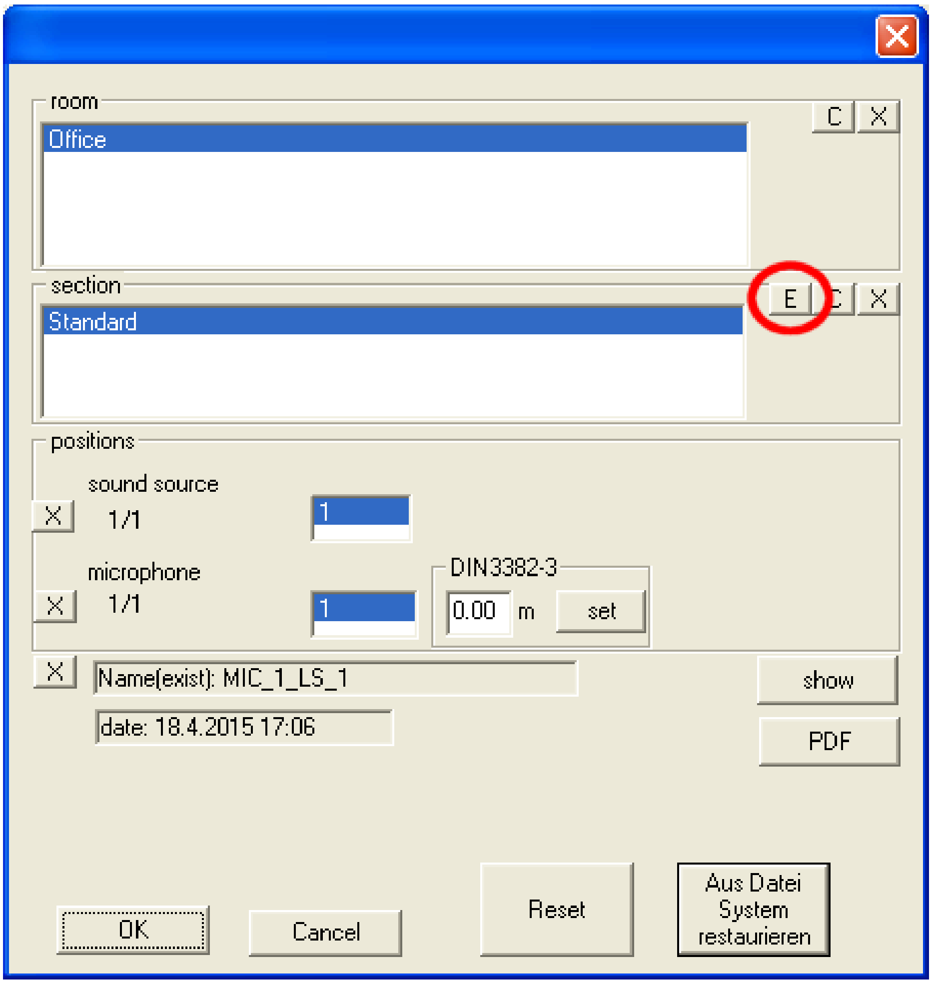
frequency [Hz]	RT60 T20 [s]
100	0.23
125	0.14
160	0.24
200	0.25
250	0.13
315	0.14
400	0.16
500	0.17
630	0.16
800	0.14
1000	0.19
1250	0.16
1600	0.17
2000	0.17
2500	0.16
3150	0.16
4000	0.20
5000	0.21
6300	0.19
8000	0.20

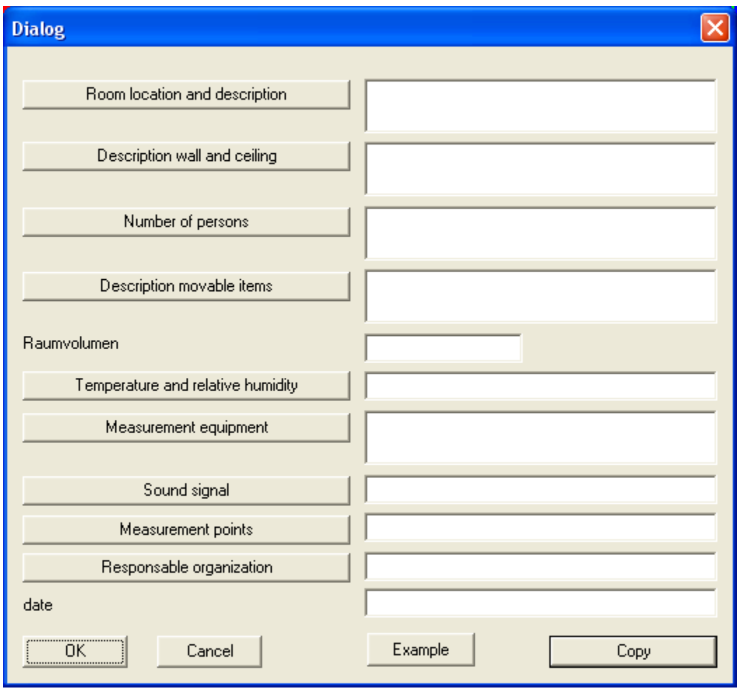


reverberation time wide-band 83Hz to 9600Hz: RT60(T20)=0.16s

The measurement is ready.

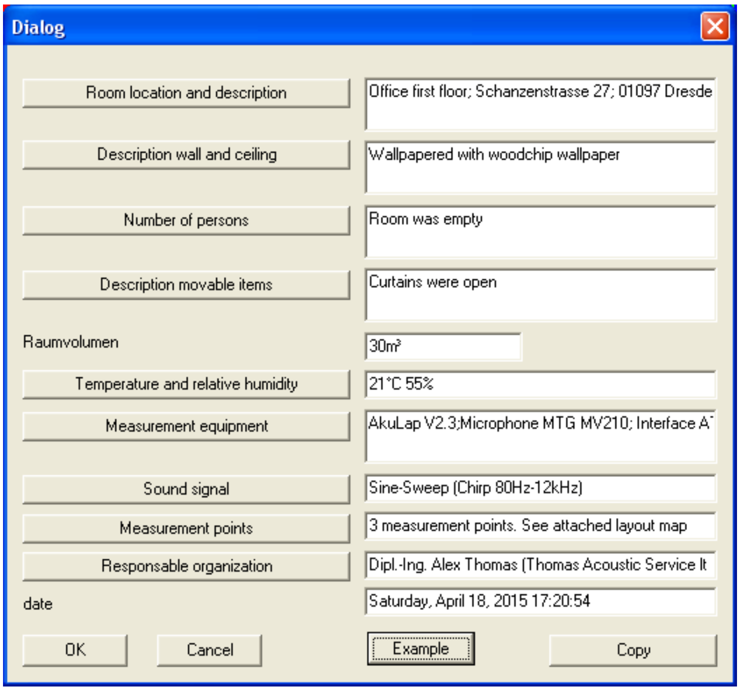
Since, we did not configure any information for the measurement, the info section is blank. To complete this, start the room manager from the menu Plugins->room manager reverb.



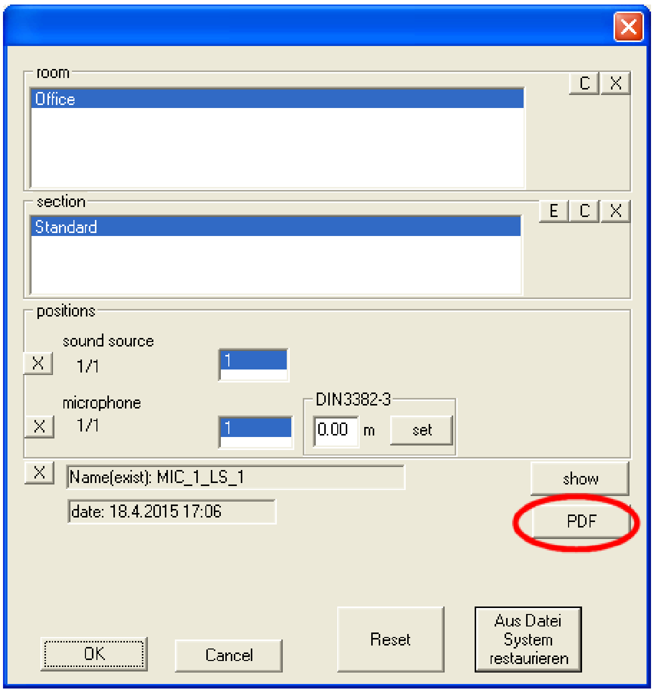


Press the E-button.

Enter all the information.



Press the PDF button to create the PDF document again.



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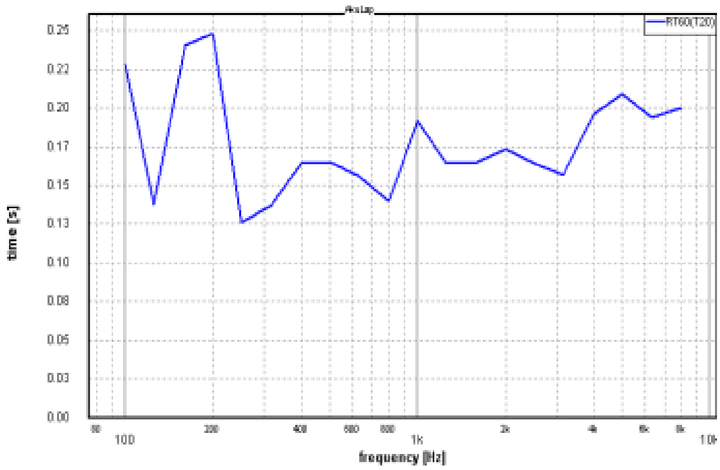
Report reverberation time DIN3382

Information for room and setup

Room location and description	Office first floor; Schanzenstrasse 27; 01097 Dresden
Room capacity	30m³
Description wall and ceiling	Wallpapered with woodchip wallpaper
Number of persons	Room was empty
Description movable items	Curtains were open
Temperature and relative humidity	21°C 55%
Measurement equipment	AkuLap V2.3;Microphone MTG MV210; Interface ATD-2;Loudspeaker Fostex PA578
Sound signal	Sine-Sweep (Chirp 80Hz-12kHz)
Measurement points	3 measurement points. See attached layout map

reverberation time RT60 in 1/3-octave bands

frequency [Hz]	RT60 T20 [s]
100	0.23
125	0.14
160	0.24
200	0.25
250	0.13
315	0.14
400	0.16
500	0.17
630	0.16
800	0.14
1000	0.19
1250	0.16
1600	0.17
2000	0.17
2500	0.16
3150	0.16
4000	0.20
5000	0.21
6300	0.19
8000	0.20



reverberation time wide-band 83Hz to 9600Hz: RT60(T20)=0.16s

Saturday, April 18, 2015 17:20:54 / Dipl.-Ing. Alex Thomas (Thomas Acoustic Service ltd.)

Now, this document contains all the information you entered.

4 Tips and tricks

It is very important to measure the diffuse sound, only. The sound source should be omni directional. The measurement microphone should not capture the direct sound of the loudspeaker. In addition the excitation sound level must be significant higher than the ambient noise.

For a reliable measurement, please note the following hints:

- The ambient level must be as low as possible. Sources are open windows, air conditionings, PCs etc. As long as you are yourself in the measurement room, please be as quiet as possible.
- The loudspeakers must be adequately dimensioned, to reach sufficient sound level. Normally you will need large subwoofers for low frequencies (20Hz-100Hz)
- The sound of source should be omni directional. Frequently, special loudspeakers like dodecahedra are used. But you can also try with several speakers simultaneously. For low frequencies, this is not critical at all, since the wavelength is very long.
- The microphone should record the diffuse sound field only. The microphone should not head directly to the loudspeaker.
- Repeat the measurement at different locations to get the average room reverberation time. Measurements at single positions can vary significantly.

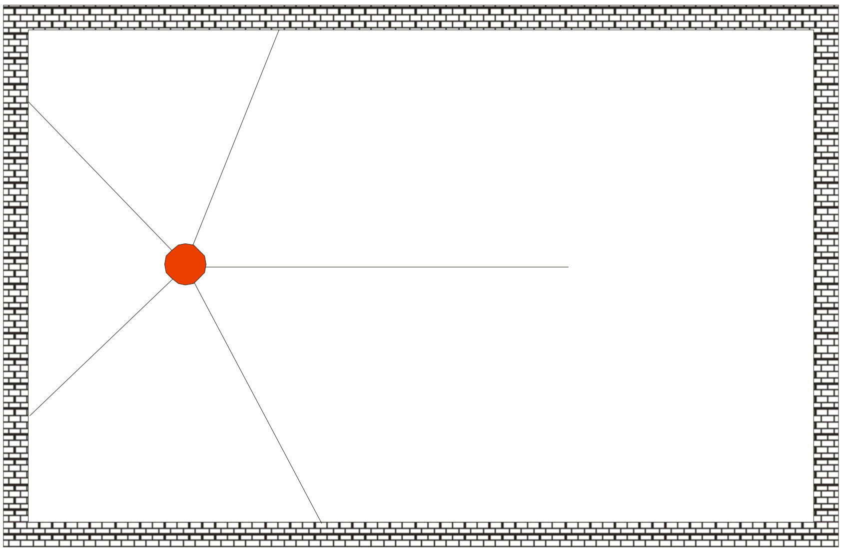
5 Reverberation background information

5.1.1 Definition

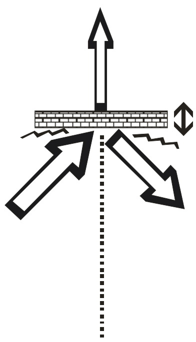
The reverberation time is one of the most important parameters in room acoustics. This parameter indicates the time after the acoustic energy inside a room is below a certain threshold after switching off the sound source. RT60 is widely used, which is defined to be the time after the acoustic energy is below –60dB.

5.1.2 Basic theory

The sound wave starts form its sound source and moves in all directions. A measurement device will notice first the direct sound.

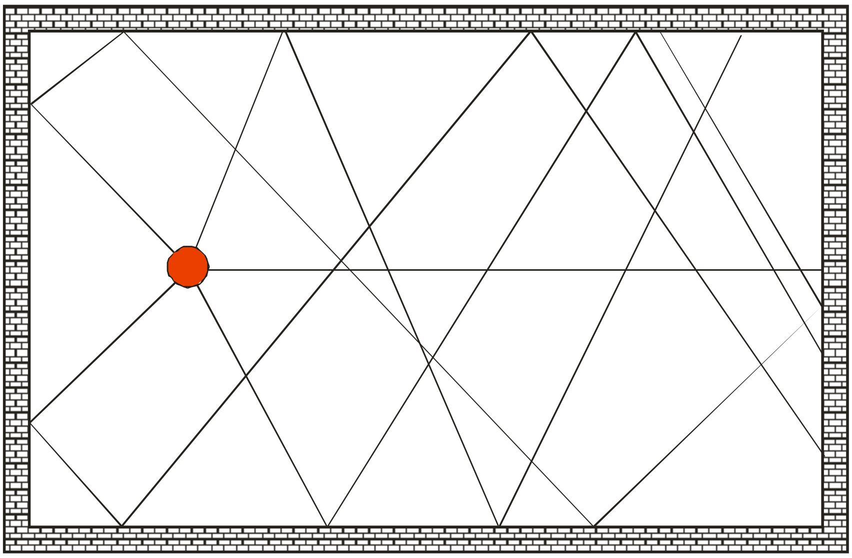


Once it reaches a wall, the wave is reflected changing its level, phase and direction.

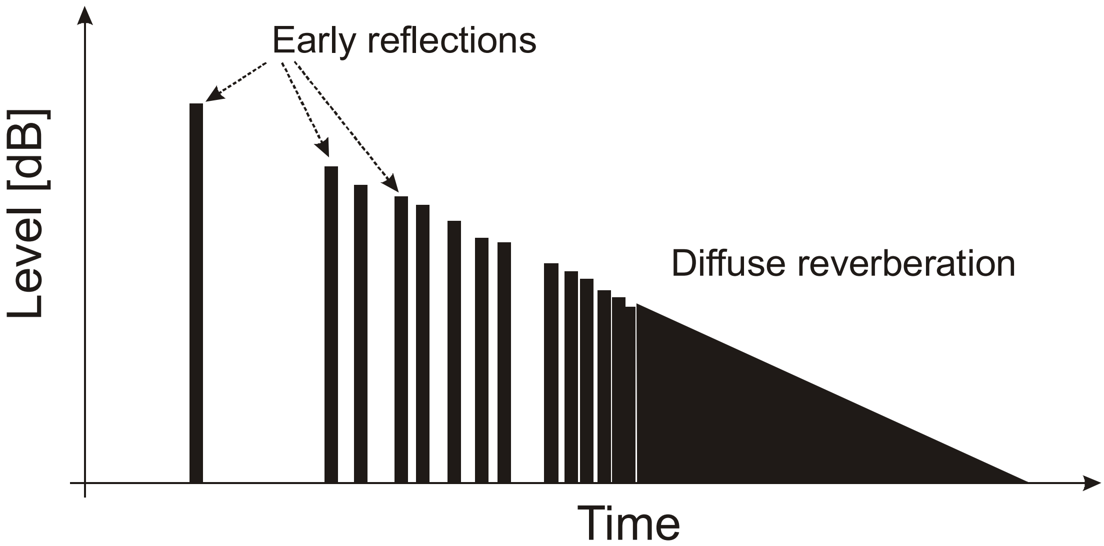


If a sound wave reaches a surface, some part will be reflected according to the law of refraction. Other part will be reflected in a diffuse manner. Other part will be exciting the surface itself (structure-borne sound). Other part crosses the obstacle and will be radiated on the other side. Some part will generate heat. Therefore, the total sound level decreases.

The number of reflecting components increases with the time, while their level decrease. Finally, we reach a diffuse sound field, where the sound arrives virtually from any direction.

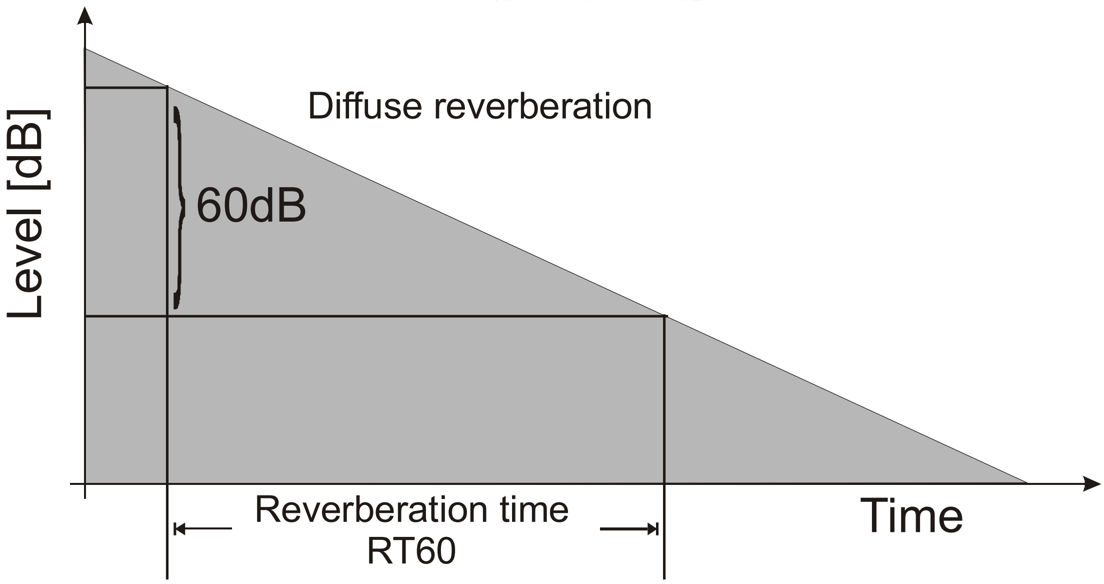


The level decreases exponentially and will be linear in a logarithmic display. The reverberation time is the decay constant.



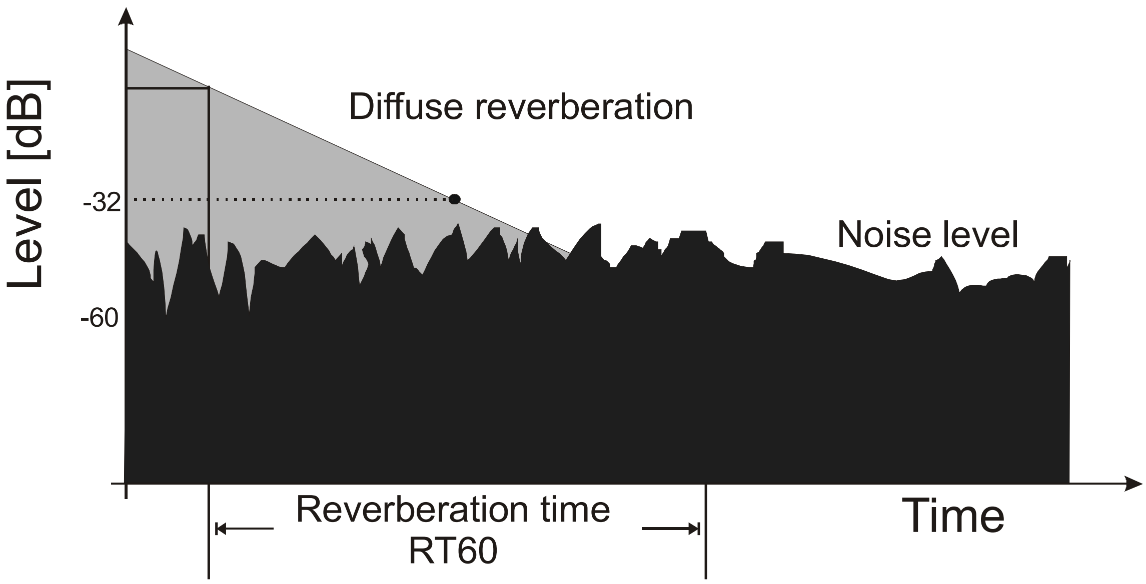
Reflections within 50ms from the main sound wave are not recognized as echoes by the human ear. Instead they improve they increase the overall listening quality.
Echoes are noticed after 100ms for single clicks or more than 1s for music.

If you have a typical diffuse reverberation, the level decays linearly in a logarithmic display. Therefore, it is easy to determine the decay constant. According to DIN/IEC3382 RT60 is the time, until the level decreases by 60dB.



5.1.2.1 Noise Level

In typical rooms we have a noise floor (air conditions, PCs, traffic etc.) from 30 dB to more than 60dB. Therefore we would need very high levels (100dB to 130dB) to measure the decay down to –60dB.



In most cases the decay is measured to a lower value –significantly above the noise level- e.g. –32dB and then extrapolated to –60dB.

5.1.2.2 Critical distance

The distance from a sound source at which the reverberant sound is equal to the direct sound. A very reverberant room has a short critical distance while an acoustically dead room can have a much longer critical distance. Outside rooms with free-field propagation, you will have an infinite critical distance.

5.1.2.3 Equation from Sabine

For many rooms, the reverberation time T can be estimated by the famous equation from Sabine:

$$T = 0.163\frac{V}{A}$$

V is the volume of the room and A is the effective absorbing area.

This equation is valid only, if the absorbing area is small with respect to the total area S. This can be seen from a small example. Let us assume, the walls are absorbing perfectly. This would lead to a reverberation time of 0, because there are no reflecting areas. In this case, the equation from Sabine computes a finite reverberation time, which is wrong.

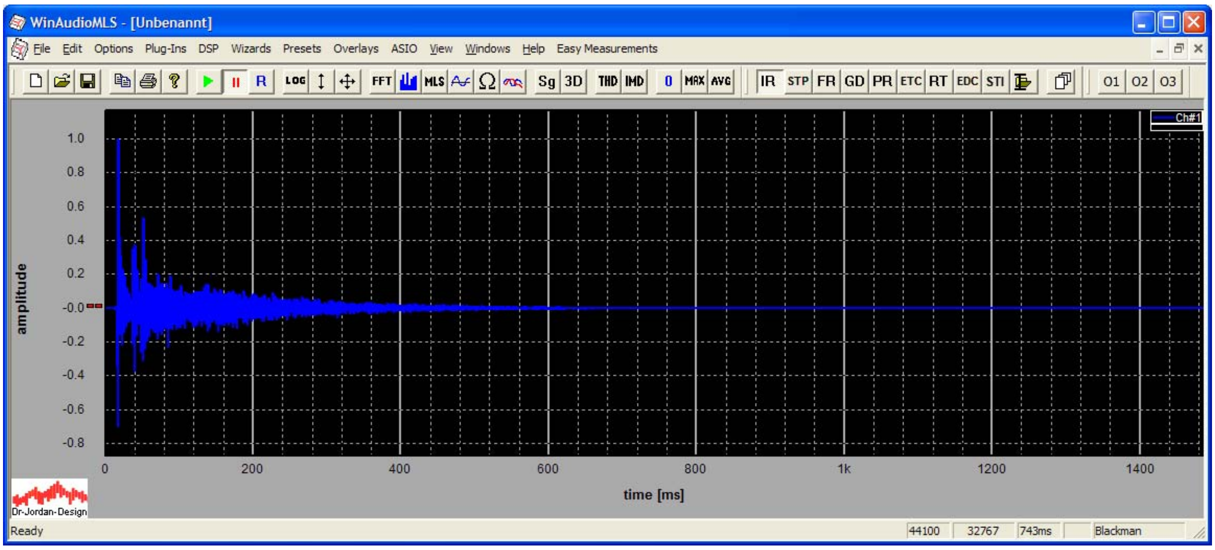
5.1.2.4 Recommended Reverberation Times

Rooms have different requirements for their reverberation characteristics. A classroom should be optimized for maximum speech intelligibility with a short reverberation time, while concert halls for classical music have a much longer reverberation time to improve the overall impression of music.

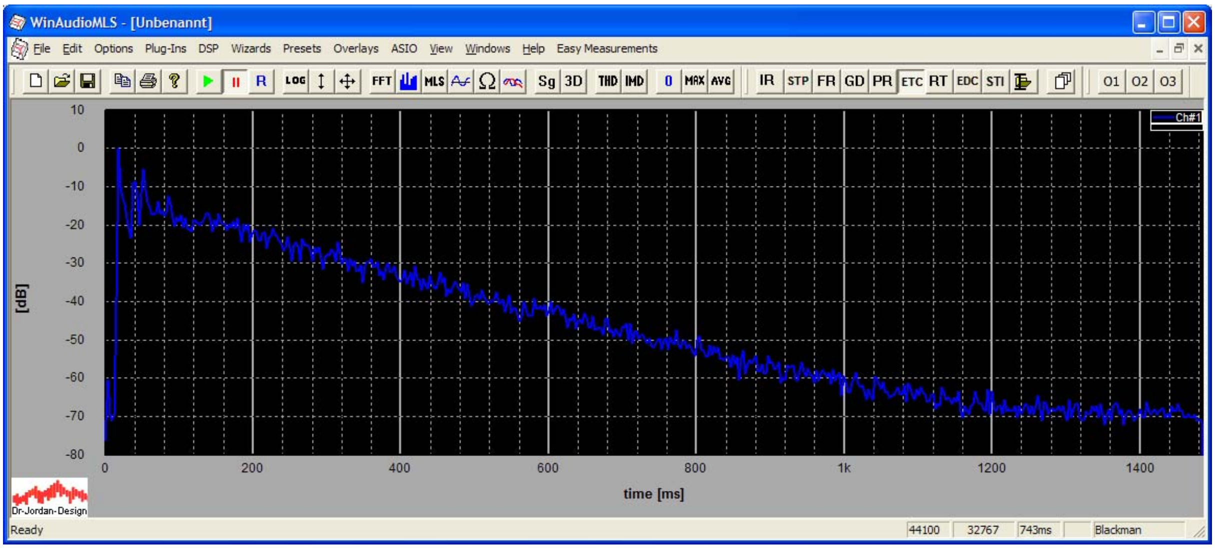
Recording rooms, studios	0.3s
Classrooms	0.6s-0.8s
Office rooms	0.35s-0.55s
Concert halls	~1s-3s

5.1.2.5 Example Room Impulse Response

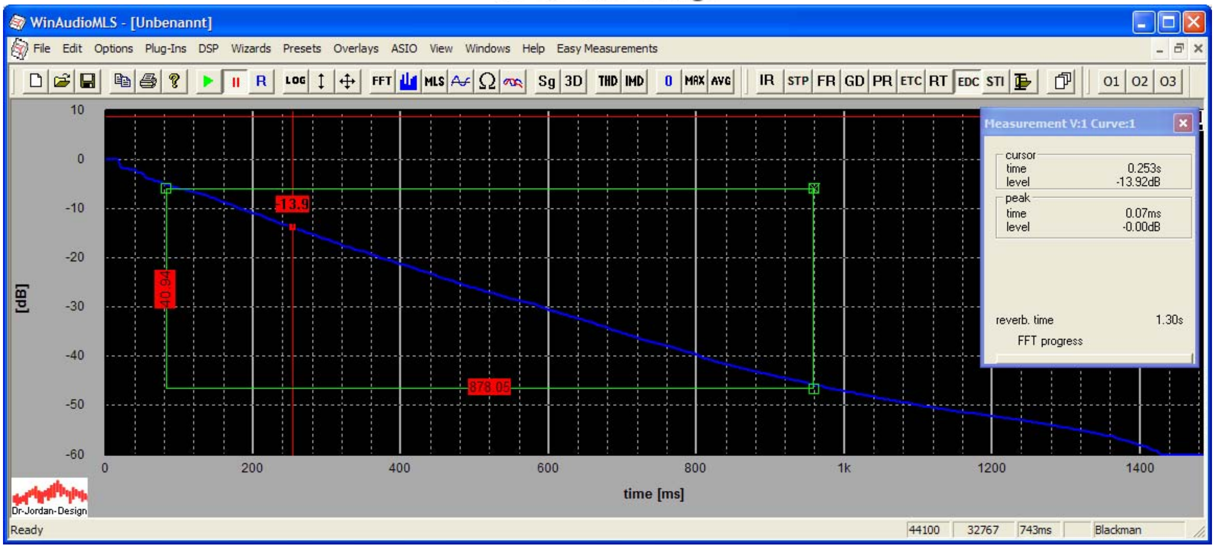
Below you will find an example impulse response, which was measured from a small concert hall.



With such linear display, we can monitor this impulse response up to 600ms. Therefore it is more useful to use a logarithmic display, which is called Energy Time Curve (ETC). We can clearly see the linear decay for the diffuse part.



The most important plot is the Schröder plot (Energy Decay Curve, EDC), which is the basis for reverberation calculation. In this example, we selected a range from –5dB to –45dB as a basis for calculation of RT60. For this example, we get a wide-band reverberation time RT60 of 1.3s.



5.1.3 Measurement

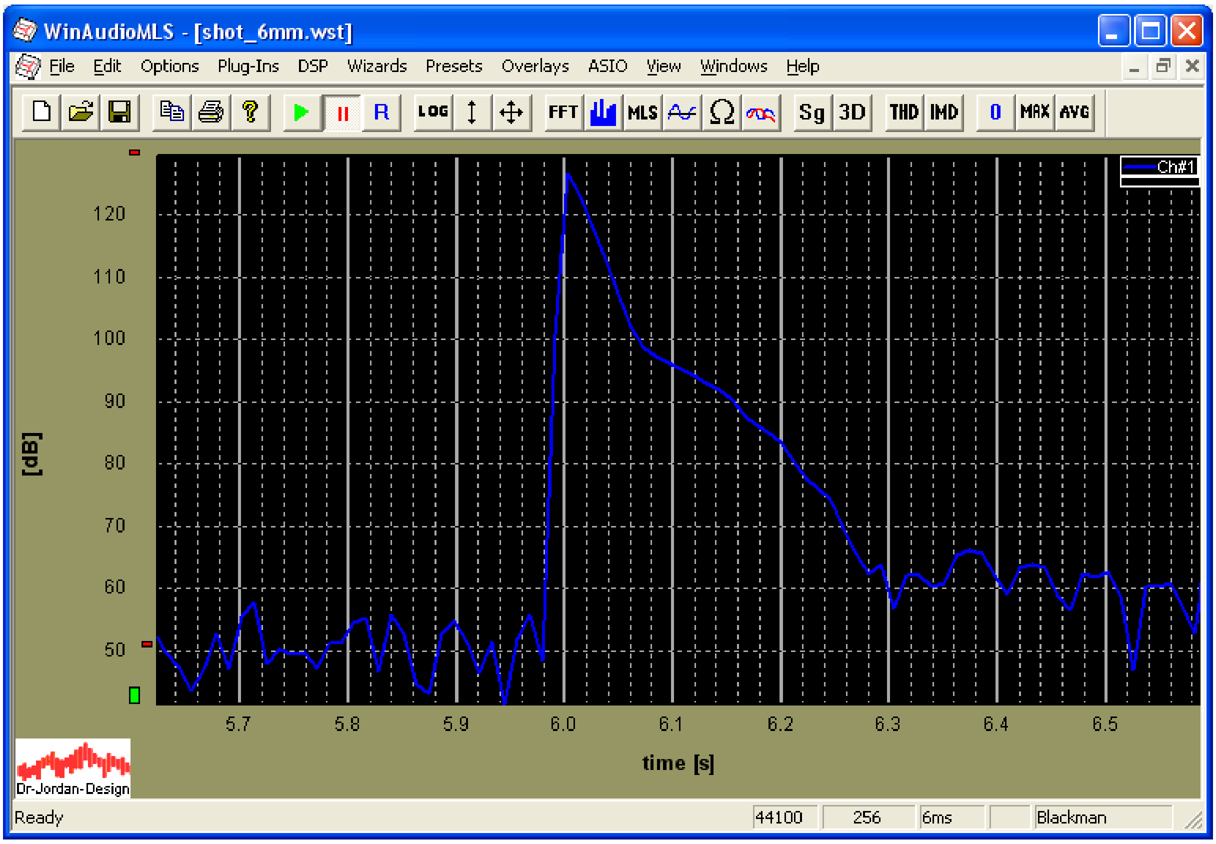
There are four methods to measure the reverberation time:

- Excitation with impulsive signals: shots, explosives, hand claps etc.
- Excitation with bursted noise
- Excitation with pseudo-noise sequences MLS
- Excitation with pseudo-noise sequences Log-Chirps

5.1.3.1 Excitation with impulses

With this method the room is excited with an impulse. The resulting measurement signal at the microphone is directly the room impulse response. Theoretically, the exciting impulse needs to have an infinite level and an infinite short duration. Such a signal is called a Dirac-impulse. In real world such signal does not exist. You can approximate it with explosives, pistols, sparc gaps or balloons filled with explosive gas. It is the goal to excite the room to oscillate at maximum level. Loudspeakers are not suitable for that purpose, because they can generate impulses at low levels, only.
This method is simple and is one of the oldest techniques to measure room impulse responses. On the other hand it is quite dangerous, due to the high signal levels. The microphones must also be able to handle such signal levels. On the other hand, the sound source radiates in all directions from a point source.

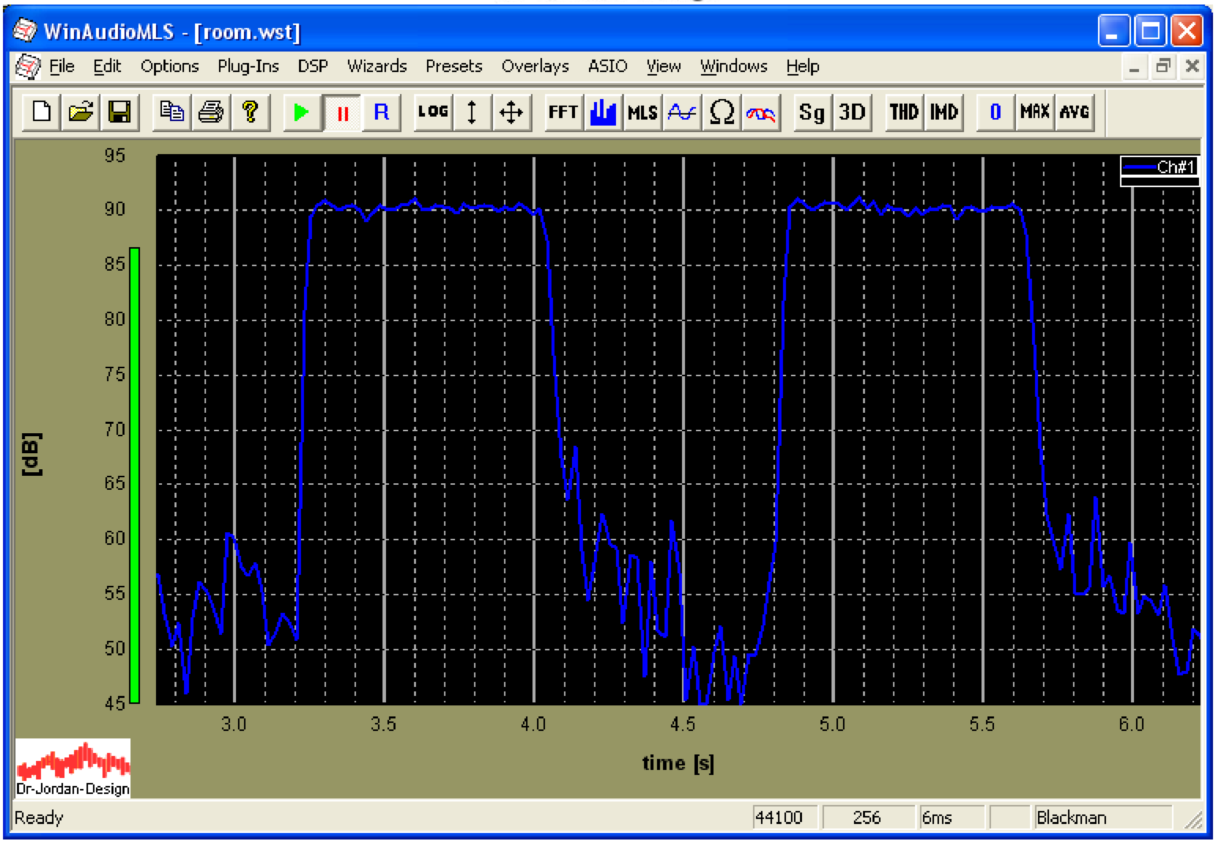
The following example shows the time flow of the sound level after a pistol shot of caliber 6mm. The background noise has a level of 50-60dB. The shot reaches a maximum of 125dB which is limited by typical measurement microphones. The usable dynamic is 60dB. Within 300ms the levels decays by 60db. This is a small room with high attenuation which limits a diffuse sound field.



5.1.3.2 Bursted noise

This method excites the room with noise via loudspeakers and switches of the noise source. Both impulse and noise signal are broadband, covering many frequencies. Single impulses are not suitable for loudspeakers, since they are limited by their mechanics. With noise you can excite the room with much more energy.

The following picture shows the sound level while switching the noise signal. You can generate such signals with the burst signal generator.

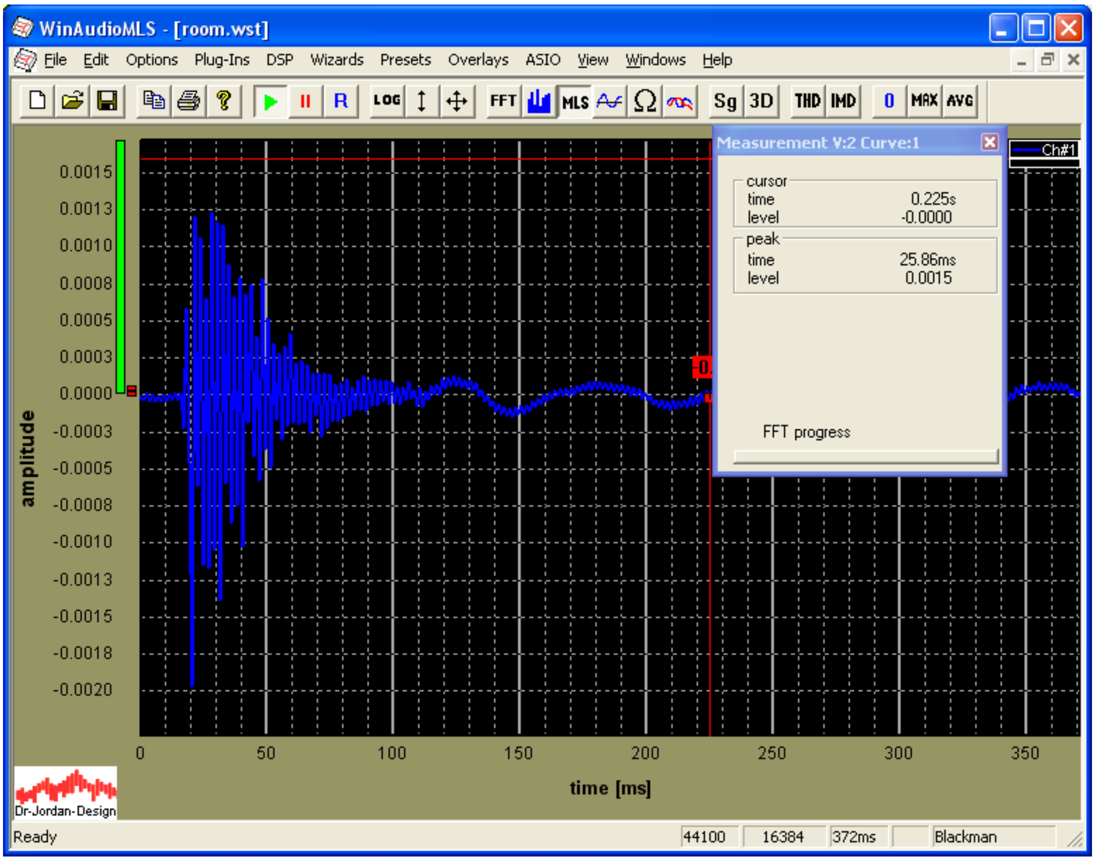


This method is still simple and is implemented by many hand-held devices. Since any other noise has direct impact on the measurement, high sound levels are required. This requires special designed loudspeaker and amplifiers. In addition, this method is only used to measure the reverberation time and not the impulse response.

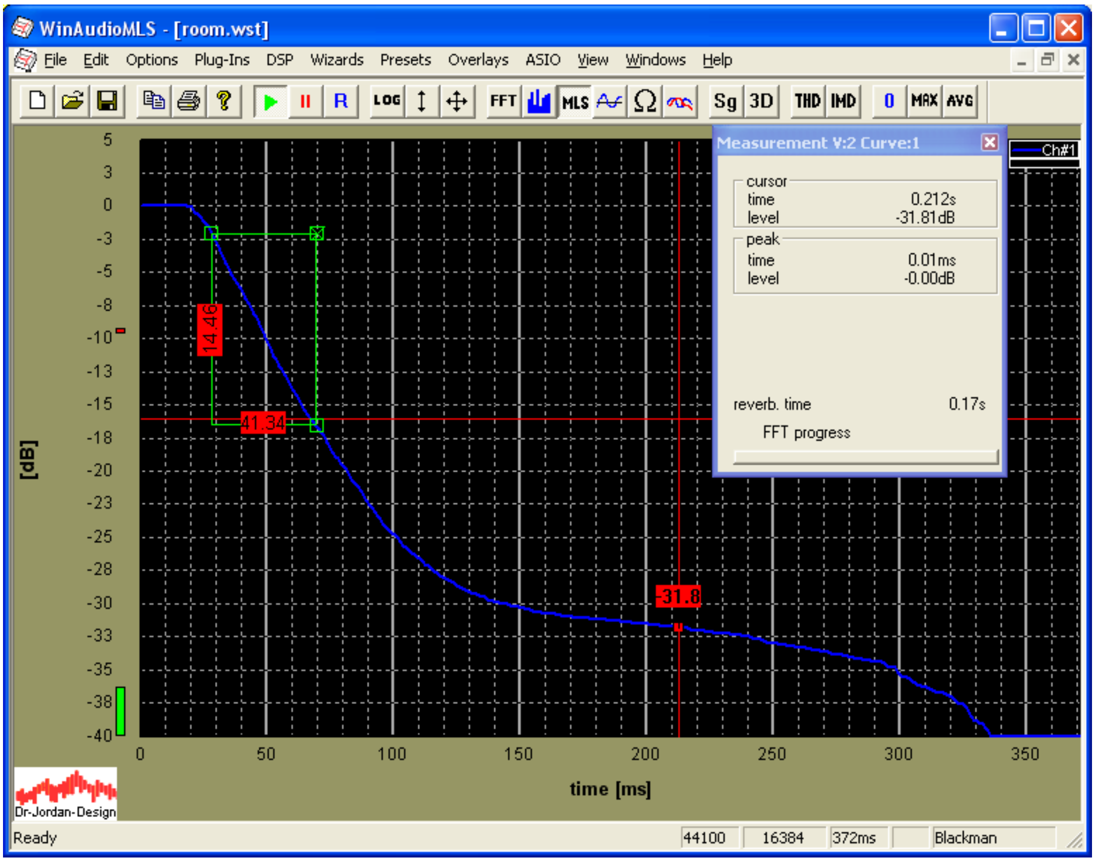
5.1.3.3 Correlation with MLS

This technique uses pseudo-noise sequences, which are radiated via loudspeakers. The special sequences allow suppressing background noise. Therefore you can use lower signal levels. Loudspeaker and amplifiers can be smaller, which is a key factor for mobile measurements. By using mathematics (correlation via Hadamard-transform), we can calculate the impulse response. This impulse response is the base for all other calculation. The reverberation time is based on the Schroeder plot.

Example impulse response



Derived Schroeder-plot with selection rectangle.



5.1.3.4 Correlation with Chirps (Log-Sweep)

Measurements with MLS are widely used, since they allow measuring fast and efficiently. On the other hand there are some disadvantaged, which required more advanced methods.

- Very sensitive to distortion
- Very sensitive to frequency shifts (pitch)
- Equally distributed spectral density
- Distortion effects are difficult to hear

In the area of room acoustics, it is important to have sufficient sound energy compared to the ambient noise. Therefore, loudspeakers are often operated at their power limit, which causes distortion. Such distortion appears with MLS similar to noise. Therefore, the usable headroom is reduced. To reduce such distortion large loudspeakers are used, which are operated in their linear area.
Chirp measurements are much less sensitive to distortion. The size of the loudspeakers can be reduced, which is very attractive for mobile measurements.

MLS signals have a constant spectral density ('white') by design. For real-world measurements, this is not the optimum for loudspeakers, since most of the signal energy is located in the higher frequency area. Let us assume the total amplifier output is 100W in the area of 20Hz to 20kHz. The signal energy in the area between 10kHz and 20kHz is 50W.

47W are located in the range between 500Hz and 10kHz. Only 3W are located in the lower frequencies between 20Hz and 500Hz. This distribution is not suitable for typical loudspeakers, since their power rating reaches the maximum at lower frequencies. The constant spectral density of MLS may easily overload the tweeters. Filter may be used to shape the spectrum, but this technique is complex and has other disadvantages.
Chirp signals decay with 3dB/octave. This is similar to pink noise, although both signals sound completely different. The main signal energy is in the low frequency range which fits the best to typical loudspeakers.
With Chirp signals you can define an lower and upper frequency limit. Signal energy is only generated in the desired range and requires no complex filtering or shaping.

Since MLS sounds like noise, it is very difficult to hear distortions, which may indicate an overload state for your loudspeakers. With Chirp signals you will mostly hear when loudspeakers are close to their limit and reduce the volume. But of course you can easily damage your speaker with Chirp signals. With MLS you will in most cases destroy the tweeter with overheating, with Chirp you can destroy the woofer with mechanical overload. Therefore please adjust the volume carefully.
Unfortunately, at high levels Chirp sequences sound very annoying in contrast to the monotonic MLS.

The measurement procedure is very similar to the MLS-measurement. First, the impulse response is computed. All other parameters are derived from that.