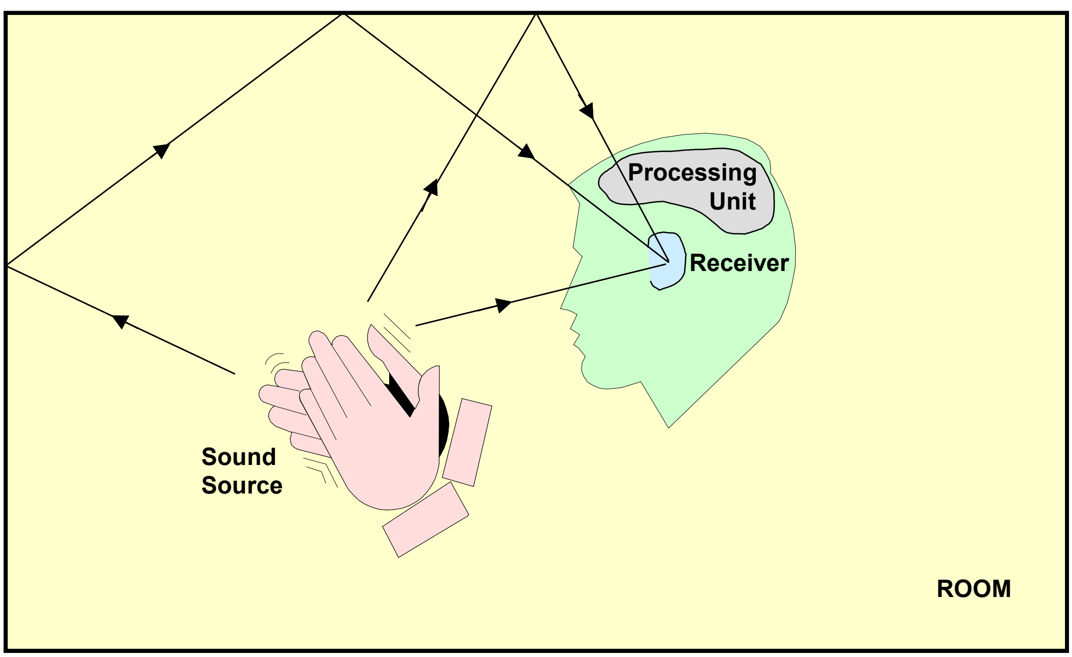
Acoustics Engineering

TN001

Technical Note

MEASURING IMPULSE RESPONSES USING DIRAC

An Introduction



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Part I: Theoretical Aspects

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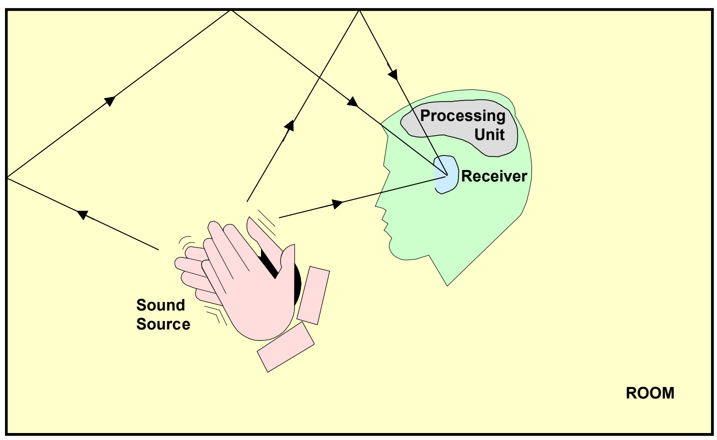
1 Impulse Responses

To investigate the acoustical properties of a room, you can clap your hands and listen to the response of the room. Although it may not be easy to describe accurately what you hear, this method evidently gives you an impression of whether in this room music would sound pleasant or speech would be intelligible. This principle is the basis for describing a system through one or more of its impulse responses.



Basic impulse response measurement system.

In fact, judging the room's acoustical response to clapping hands is a particular case of an impulse response based measurement model, in the sense that the hands as a source, the ears as receivers and the brain as processing unit do not fulfil the following requirements:

1. Normally, the source and receiver positions should not be located close to each other, but reflect real positions. The source could for instance be located somewhere on the stage, and the microphone somewhere in the listening area.
2. The source should have a well-defined and reproducible sound power radiation as a function of direction and frequency.
3. The ideal acoustical impulse is defined as an impulsive signal of infinitely short duration, infinitely high power, and unit energy. However, the spreading effect in time of an acoustical system and the limited relevant frequency bandwidth allows this requirement for impulsive sources to be relaxed to produce a signal with a duration short enough to have negligible impact on the response.
4. The receiver should have a well-defined sensitivity as a function of direction and frequency.
5. It should be possible to store measured impulse responses for later reference.

6. The acoustical parameters are derived from impulse responses by means of well-defined algorithms, usually through a computer.

All of these requirements 1 through 6 aim at getting the same results and conclusions from different engineers, using different measurement equipment, but measuring the same acoustical system. How to meet these requirements will be described in more detail in the next chapters.

Basically, the acoustical properties of a practical[1] system can be fully and accurately described through a set of impulse responses, if measured under the proper conditions. Any acoustical parameter can then be derived from this set of impulse responses. Acoustical parameters may reflect material properties or subjective perceptive measures, and will also be described in the next chapters.

2 Source Signals

Requirements 2 and 3, mentioned in the previous paragraph, are not easy to combine in a practical impulsive sound source. A loudspeaker sound source has good properties with regard to the reproducibility, spectrum and directivity, but is not able to produce very short, high level impulsive signals. On the other hand, blank pistols, balloons and other mechanical means can produce high-level impulsive signals with a wide frequency range, but lack controllable directivity and good reproducibility.

Fortunately, an impulse response can also be calculated from the response to another signal, under the condition that this signal is well defined and has sufficient energy at each relevant frequency. By choosing a signal that has its energy spread in time, it can be handled by a loudspeaker sound source.

An example is white noise, which *on average* has a flat frequency spectrum. Through a mathematical operation on the response signal and the white noise excitation signal (deconvolution), the impulse response can be extracted. Due to the randomness of the excitation signal, the extracted impulse response shows residual noise, which is reduced as the measurement time is increased.

Another example is the Maximum Length Sequence (MLS) signal. This is a periodic, pseudorandom white noise signal, having the desirable property that its frequency spectrum over one period is as flat as the spectrum of an ideal impulse. The extracted impulse response is therefore not polluted by any noise due to the excitation signal.

[1] This stems from the fact that the response to any kind of signal of a ‘practical’ (linear and time invariant) system can be derived from the impulse response. Any particular signal can be considered as a sequence of impulses, each with its own time delay and weighting factor. Adding up the impulse responses with the same time delays and weighting factors, results in the system’s response to that particular signal.

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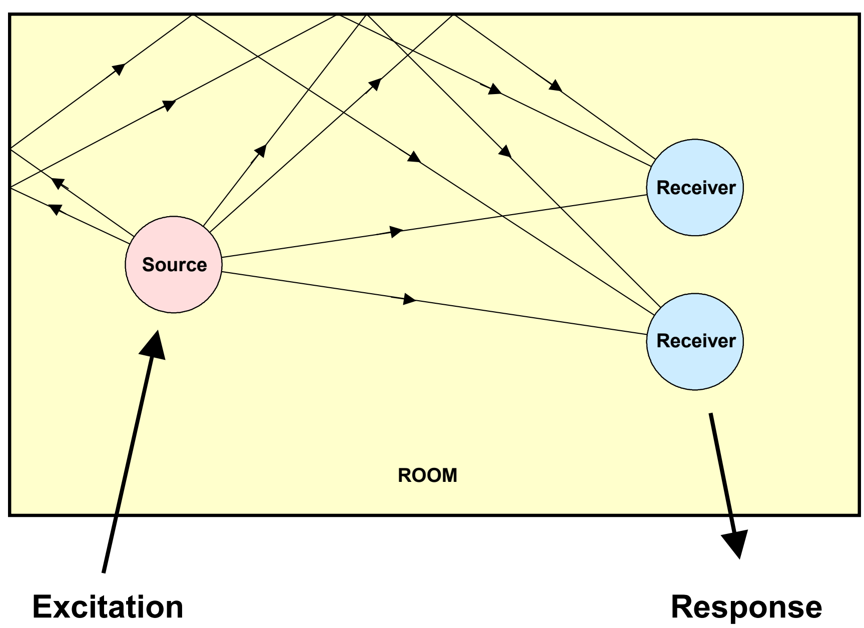
The third example is a swept sine, with a frequency sweeping over the desired frequency range. Again, this is not a random signal, and the extracted impulse response is free from noise contributed from the excitation signal. In practice, the sweep signal allows a higher power output level than an MLS signal, due to its lower crest factor through the transmission system.

Despite the advantages of loudspeaker sound sources, the much smaller and lighter impulsive sources are often preferred for survey or engineering measurements.

On the next page, an overview is given of the different types of impulse response measurement methods, based on the type of room excitation signal used.



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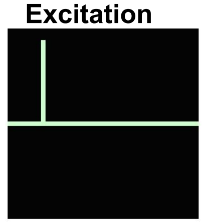
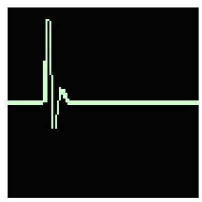
Excitation **Response**

Ideal impulse (Dirac function)
- Infinitely short
- Infinite power
- Unit energy
- Omnidirectional (room)
- Mouth-directional (speech)



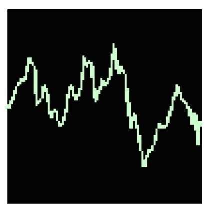
Exact impulse response
- Only theoretical
- No additional processing necessary

Approximated impulse
- Shotgun, balloon, clapping hands, spark gap, whip, paper bag, etc.
- Poorly defined directivity
- Poor reproducibility



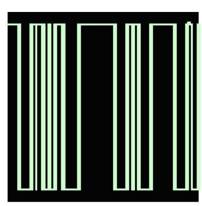
Appr. impulse response
- No additional processing necessary

Random signal
- White/pink noise, music, etc.
- To be considered as weighted sum of delayed impulses with random weighting factors



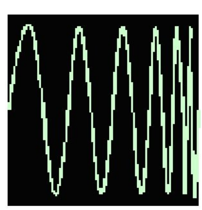
Noise response
- To be deconvolved with source signal to obtain impulse response
- Residual noise

MLS signal
- Maximum Length Sequence
- Pseudorandom noise
- To be considered as weighted sum of delayed impulses with weighting factors +1 and –1



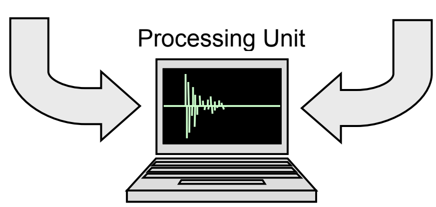
MLS response
- To be deconvolved with source signal to obtain impulse response

Sweep signal
- Periodically swept sine
- To be considered as weighted sum of delayed impulses with defined weighting factors



Sweep response
- To be deconvolved with source signal to obtain impulse response

Processing Unit



Impulse Response

3 Parameters Derived From Impulse Responses

3.1 Reverberation times

The reverberation time is a measure of how fast sound energy decays. For rooms, the ISO 3382 standard [1] prescribes the use of an omnidirectional sound source and an omnidirectional microphone to measure the reverberation time. The source is placed for instance on stage, and the microphone is placed at successive listening positions.

The early decay time EDT gives information on the direct sound decay and is derived from the decay section of the impulse response between 0 dB and –10 dB below the initial level.

The reverberation times T_{20} or T_{30} give information on the diffuse sound decay and are derived from the decay section of the impulse response between –5 dB and –25 or –35 dB respectively below the initial level.

3.2 Spaciousness

Spaciousness parameters relate to the spatial impression and are measured by signal differentiation between the two ears, hence requiring 2 impulse responses per listening position. In each case, according to ISO 3382 the measurement is carried out using an omnidirectional sound source, which is placed for instance on stage.

The early lateral energy fraction LF is the lateral fraction of the total energy arriving at the receiver position within the first 80 ms. For this parameter, the impulse responses are measured using an omnidirectional microphone and a bidirectional microphone with the direction of maximum sensitivity pointed towards the side walls of the room.

The early lateral energy fraction LFC is also the lateral fraction of the total energy arriving within the first 80 ms, but calculated in a different way. For this parameter, the impulse responses are measured using an intensity microphone probe, with the line through the microphones pointing towards the sidewalls.

The inter-aural cross correlation coefficient IACC is a measure of how much the left ear and right ear signals correlate. $IACC_{0,80ms}$ involves only the early reflections, $IACC_{80ms,\infty}$ involves only the reverberant sound, and $IACC_{0,\infty}$ involves both. For this parameter, the impulse responses are measured through a measurement head, looking into the source direction.

3.3 Energy Ratios

These parameters relate to a.o. perceived definition or speech intelligibility. For rooms, the ISO 3382 standard prescribes the use of an omnidirectional sound source and an omnidirectional microphone to measure energy ratios. The source is placed for instance on stage, and the microphone is placed at successive listening positions.

The centre time T_S is the time of the centre of gravity of the squared impulse response.

The clarity C_{80} is the early-to-late arriving sound energy ratio, where “early” means “during the first 80 ms” and “late” means “after the first 80 ms”.

The definition D_{50} is the early-to-total arriving sound energy ratio, where “early” means “during the first 50 ms”. D_{50} is also called “Deutlichkeit”.

3.4 Levels

The sound strength G is a measure for the overall transferred energy from the source to the receiver. For rooms, the ISO 3382 standard prescribes the use of an omnidirectional sound source and an omnidirectional microphone to measure G. The source is placed for instance on stage, and the microphone is placed at successive listening positions.

3.5 Speech intelligibility

To measure the speech intelligibility, several measuring configurations may apply. In case the background noise is of no concern, and the source signal is injected directly into a sound reinforcement system, no separate sound source is needed. In case a real talker should be emulated, a small loudspeaker source should be used, with the directivity characteristics as prescribed by the IEC 60268-16 standard [2], and at the normal speaker position. In case the background noise is significant, the source signal should have a standard voice spectrum, according to IEC 60268-16, and the output level should be calibrated. The receiving microphone is always omnidirectional.

The speech transmission index STI is the basic measure for the speech intelligibility, using the response signal over all octave frequency bands from 125 Hz through 8 kHz.

The room acoustics speech transmission index RASTI is a simplified version of the STI, and may be used instead of STI under typical conditions of a practical room.

The speech transmission index for public address systems STIPA is a simplified version of the STI, and may be used instead of STI under typical conditions of PA systems.

The speech transmission index for telecommunication systems STITEL is a simplified version of the STI, and may be used instead of STI under typical conditions of a telecommunication channel.

The percentage articulation loss of consonants % ALC can be approximately derived from STI, RASTI, STIPA or STITEL.

3.6 Stage parameters

Stage parameters are measures relating to the supporting sound feedback to the performers on the stage. An omnidirectional source is placed on stage, and the microphone is placed at a distance of 1 meter from the source.

ST_{early} is the late-to-early arriving sound energy ratio, where “late” means “between 20 and 100 ms” and “early” means “during the first 10 ms”.

ST_{late} is the late-to-early arriving sound energy ratio, where “late” means “after 100 ms” and “early” means “during the first 10 ms”.

ST_{total} is the late-to-early arriving sound energy ratio, where “late” means “after 20 ms” and “early” means “during the first 10 ms”.

3.7 Specials

Some parameters require a special setup.

An example is the measurement of the parameters in a scale model of a real room. To deal with the ultrasonic frequencies, a special sound source and microphone are required. A suitable sound source is a spark gap.

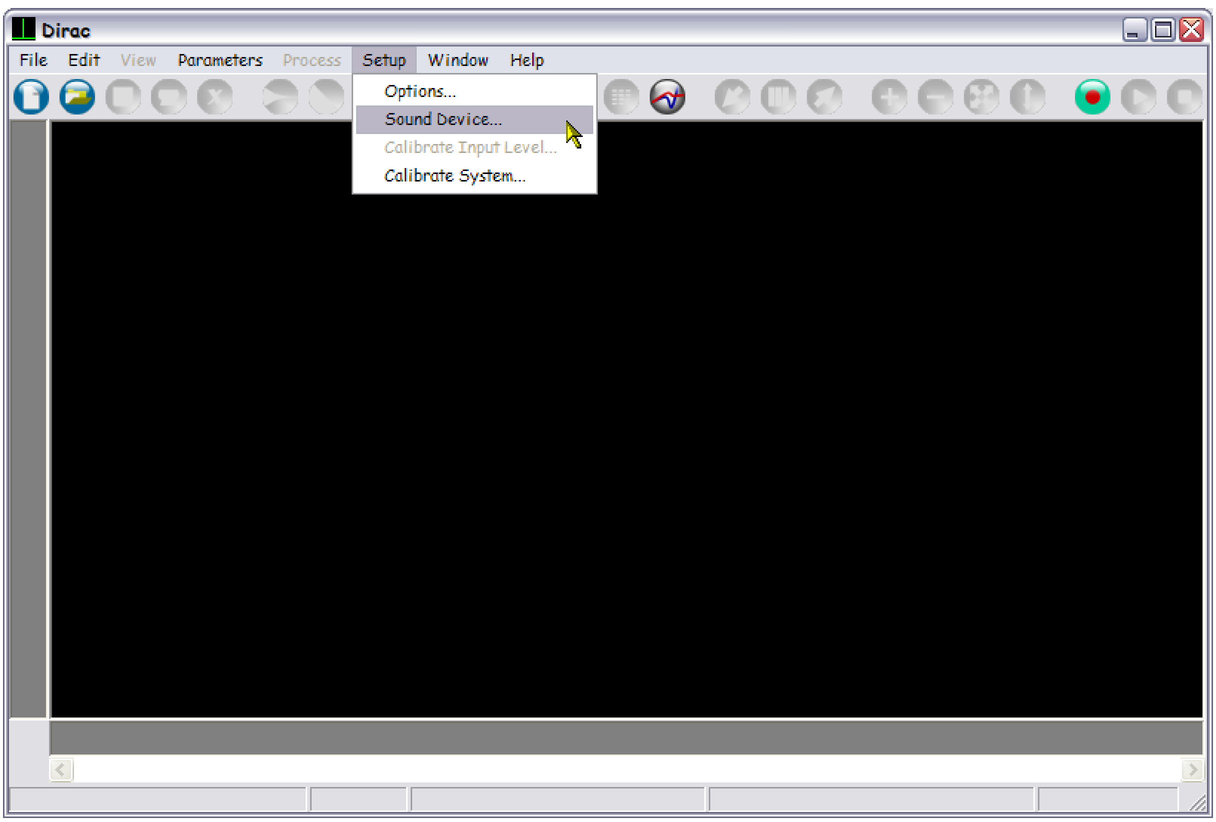
Another example is the in situ measurement of the absorption or reflection. This requires a special loudspeaker-microphone probe.

Part II: Practical Measurements Using Dirac

4 Sound devices

Dirac aims to work with any Windows compatible sound device. However, not all sound devices present the same controls for setting playback and recording volume, and other options such as 3D sound and treble and bass controls.

In order to correctly control a sound device, Dirac needs to execute a sound device calibration procedure. To start this procedure, you should first open the **Setup** menu and select **Sound Device...**.



Initially, Dirac will use the default sound input and output lines used by Windows. In the **Sound Device Setup** dialog you can select the device you wish to use for the actual measurements. For optimum performance and full control of the sound device by Dirac, you should perform the calibration by clicking the **Calibrate** button and following the ensuing instructions.

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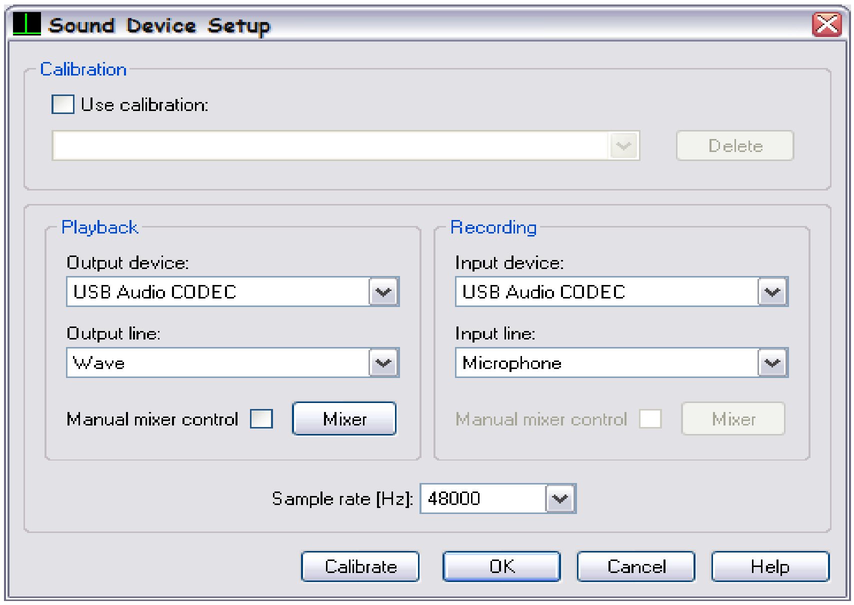
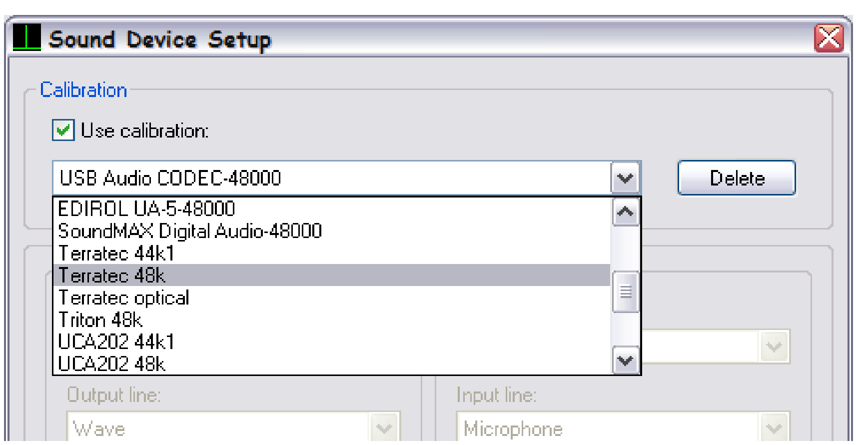


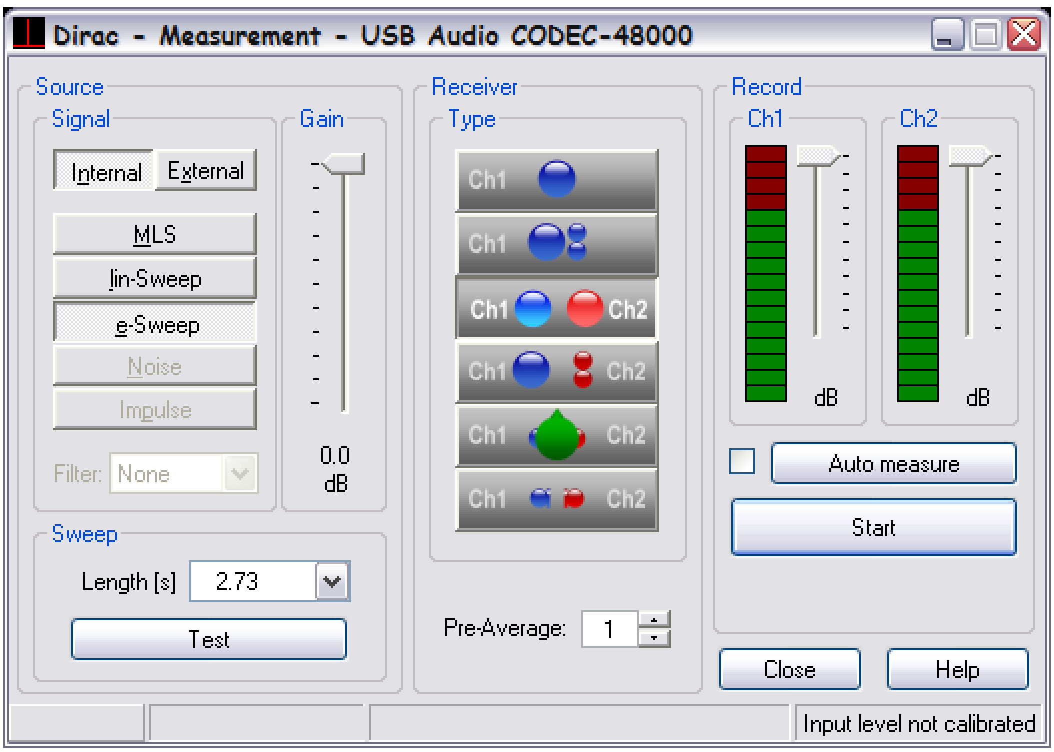
During the calibration procedure, Dirac finds the optimum settings for all relevant volume controls, and the optimum DAC levels. Also, the frequency response of the sound device is measured and, if needed, used to compensate the measurements.

Any number of calibrations for different sound devices and/or sample rates can be executed. The calibration that is used during measurements can be selected in the **Sound Device Setup** dialog.



5 Measuring the impulse response

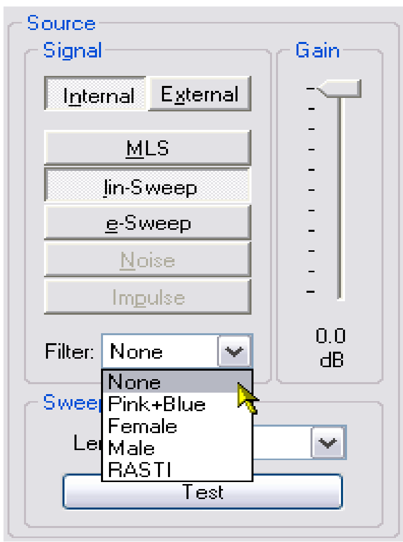
Once the soundcard is calibrated, a measurement can be started from the measurement window, which can be opened by clicking the **Measure** button on the toolbar.



The measurement window consists of 3 sections: Source, Receiver and Record.

In the **Source** section, the excitation that is used is set up. Dirac itself can generate the excitation signal (**Internal MLS, lin-Sweep or e-Sweep),** or it can be supplied by an **External MLS**, **lin-Sweep**, **e-Sweep, Noise** or **Impulse** source.

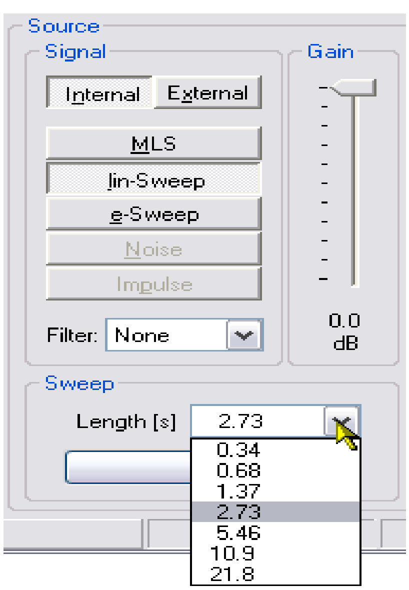
The **Internal** excitation signals **MLS** and **lin-Sweep**, can be filtered before being sent to the output, by selecting the appropriate filter from the **Filter** drop down list box.



When **None** is selected, the signal will not be filtered, and the output spectrum will be white. When **Pink+Blue** is selected, the signal is pink filtered at the output, resulting in a pink (1/f) excitation spectrum. The subsequent recording is blue filtered, compensating for the initial pink filtering. A white noise signal has less energy in its low frequency bands, making accurate measurements at these frequencies difficult due to the relatively poor S/N ratio. Pink filtering results in equal energy in all frequency bands, and is especially useful when the signal must be played over a loudspeaker system containing tweeters that may be damaged by a white noise signal of enough strength to get acceptable results at low frequencies.

The **Male** and **Female** filters are used for STI measurements. When **Female** is selected, the signal spectrum is shaped corresponding to a female voice, according to IEC 60268-16. When **Male** is selected a spectrum that is shaped like a male voice is sent to the output.

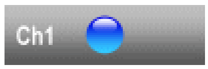
The **RASTI** filter results in a spectrum that is shaped according to a RASTI source as described in IEC 60268-16.



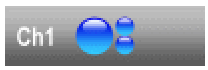
The length of the generated signals can be set through the signal **Length** drop down list box.

Once the excitation is set up, the **Test** button can be used to send the source signal to the output without starting a recording. This allows you to set the gains in the Measurement dialog and when needed the (external) gains on the microphone and power amplifiers.

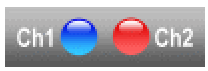
In the **Receiver** section you can choose from 6 different receiver types.



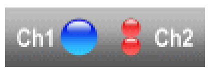
Single omnidirectional microphone



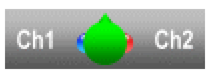
Switchable omni-bi-directional microphone



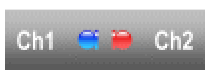
Dual omnidirectional microphone



Omnidirectional + bidirectional microphone

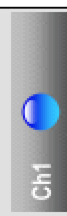
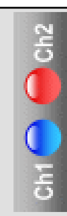
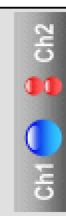
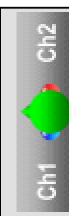
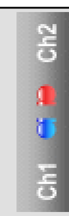


Head simulator



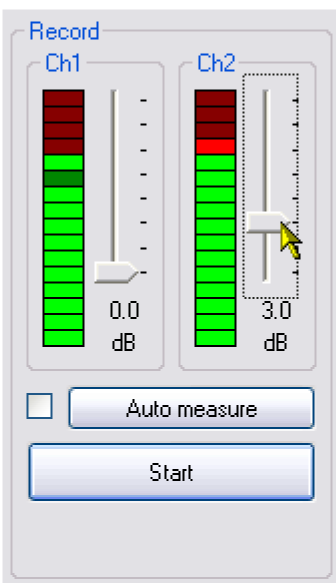
Intensity microphone probe

The choice depends on the parameters you want to measure, according to the following table.

						
SNR, INR	√	√	√	√	√	√
G, G_{rel}, L_{eq}, L_{Aeq}, L_{Ceq}, BR(L)	√	√	√	√		√
EDT, T_{10}, T_{20}, T_{30}, T_X, RT, BR(RT)	√	√	√	√		√
T_S, C_{30}, C_{50}, C_{80}, C_X, D_{50}, D_X, H	√	√	√	√		√
LF		√		√		
LFC						√
$IACC_{80}$, $IACC_X$					√	
ST_{early}, ST_{late}, ST_{total}	√	√	√	√		√
STI, RASTI, STITEL, STIPA, %ALC	√	√	√	√		√
RI, Qw	√	√	√	√		√
Magnitude, Magnitude Pink	√	√	√	√	√	√

With MLS and sweep signals it is possible to improve the decay range of the impulse response (or Impulse response to Noise Ratio INR) through a process called pre-averaging.

For this the source signal is repeated a number of times (as set by the **Pre-Average** number), and the recorded signals are averaged. Every doubling of the number of pre-averages should theoretically result in a 3 dB increase of the INR. In practice the increase is less, but still significant. Note that this improvement of the INR only works for (uncorrelated) environmental noise, and not for (correlated) noise caused by non-linearity and/or time variance of the system being measured.



The third section of the measurement window (**Record**) is used to check the recorded signal strength and to start a recording. The recording level can be adjusted using the **Ch1** and **Ch2** gain sliders. Each element of the level meters represents a 3 dB signal level range. The gains should be set such that the level meters reach the topmost green levels. This allows for a 12 dB headroom.

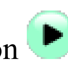
When you press Start, the actual recording will start and the measured impulse response is displayed.

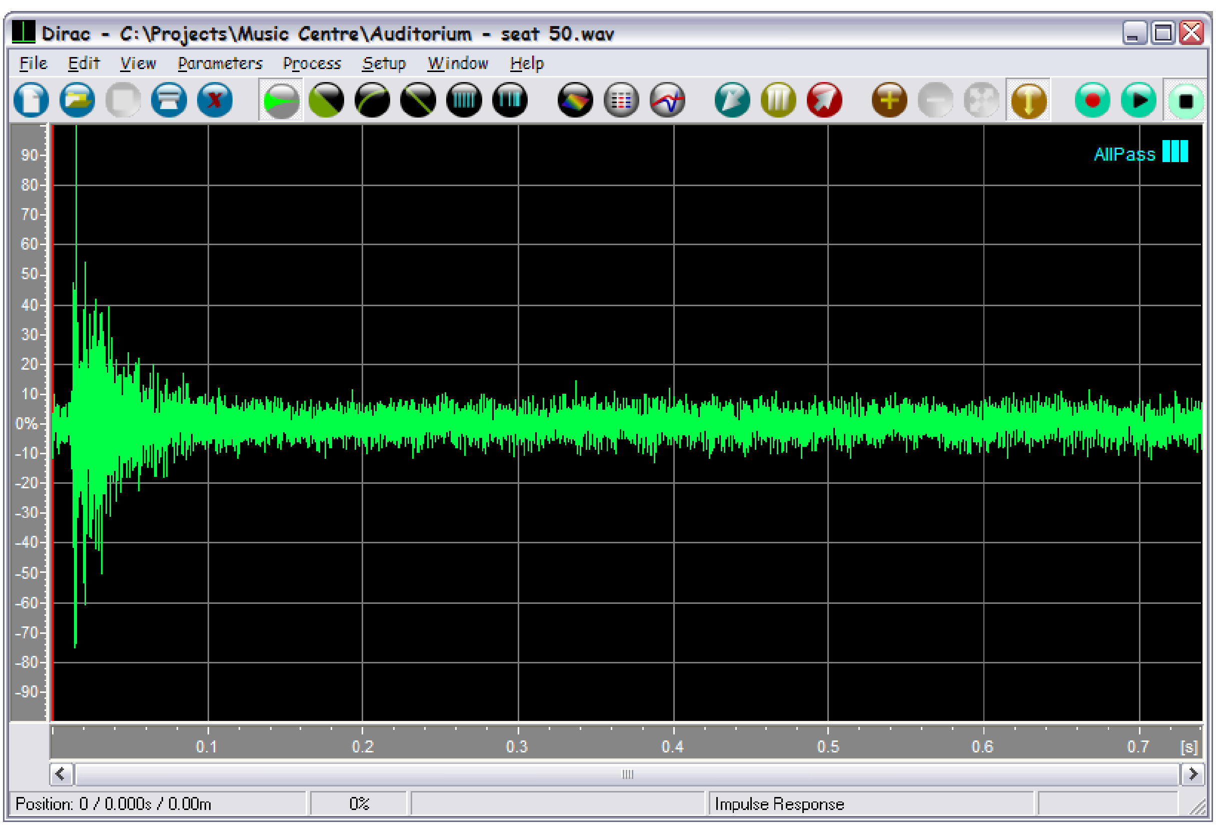
The simplest way of measuring an impulse response is often the use of an impulsive source signal such as a clap of your hands, a balloon or a blank pistol. By selecting **External Impulse**, the response to an impulsive signal is simply recorded without any subsequent processing.

6 Judging the impulse response quality

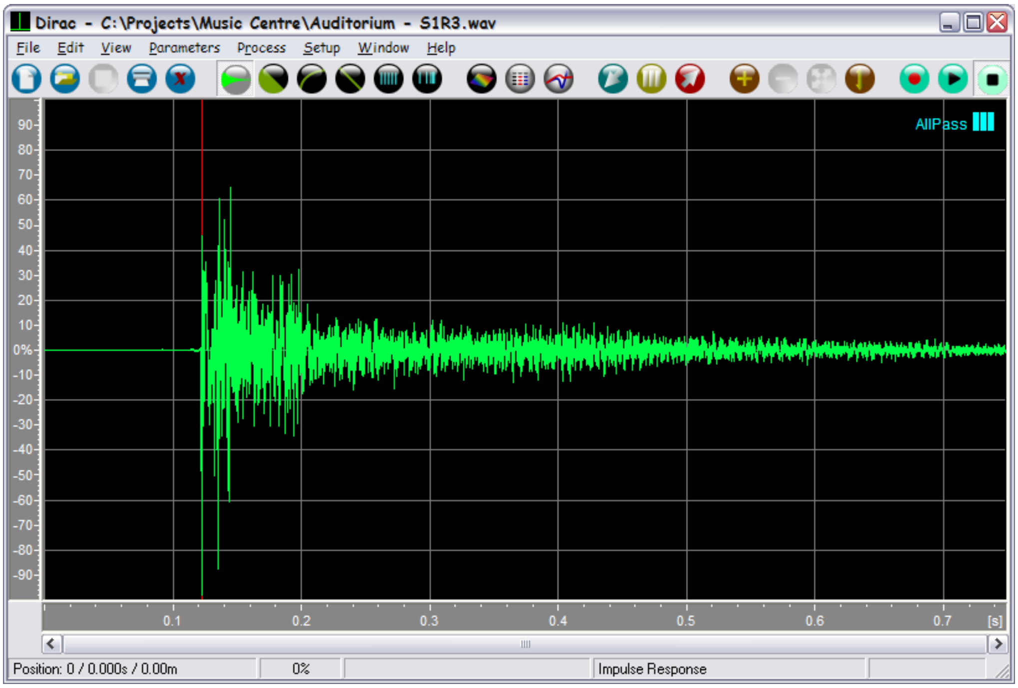
When performing a series of measurements, it is important to verify that they are of sufficient quality. There are a number of methods to quickly judge the quality of a measurement. First you should look at the image of the impulse response. It should have an essentially flat tail, and be fully contained within the measurement interval.

You can also look at the Impulse response to Noise Ratio INR, which gives you the ratio between the maximum level of the impulse response and the noise level in the tail of the response. According to ISO 3382, the INR should be better than 35 dB to determine the T_{20} and better than 45 dB for the T_{30}.

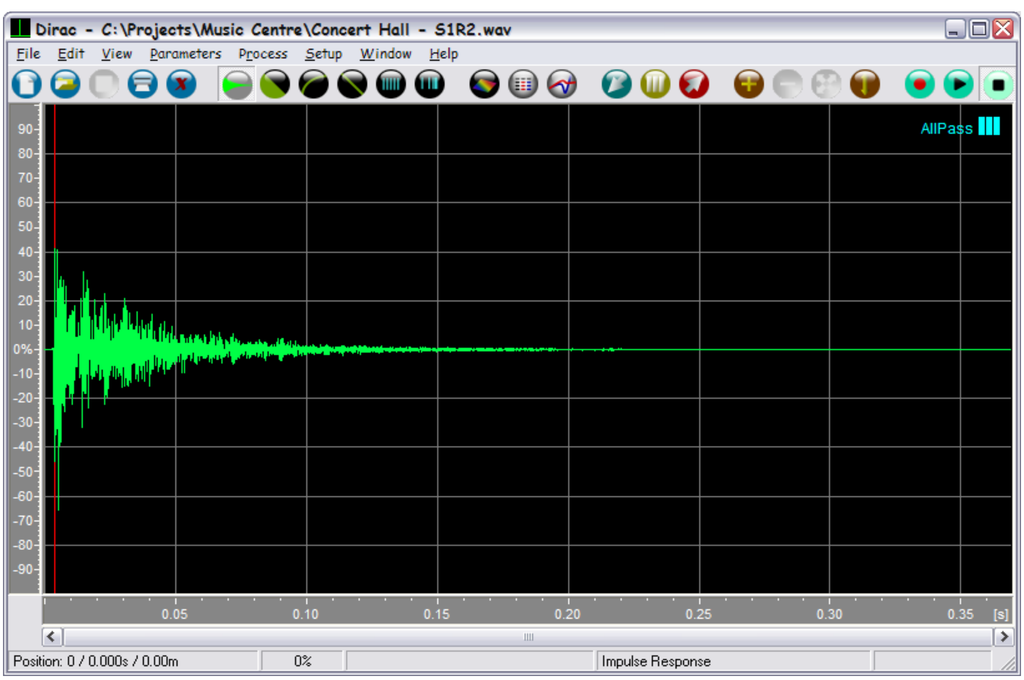
Finally you can simply listen to the impulse response, by pressing the **Play** button  on the toolbar. This will often reveal problems that are not evident from a visual inspection.



An impulse response containing too much noise.



An impulse response that is too long for the capture time that was used.



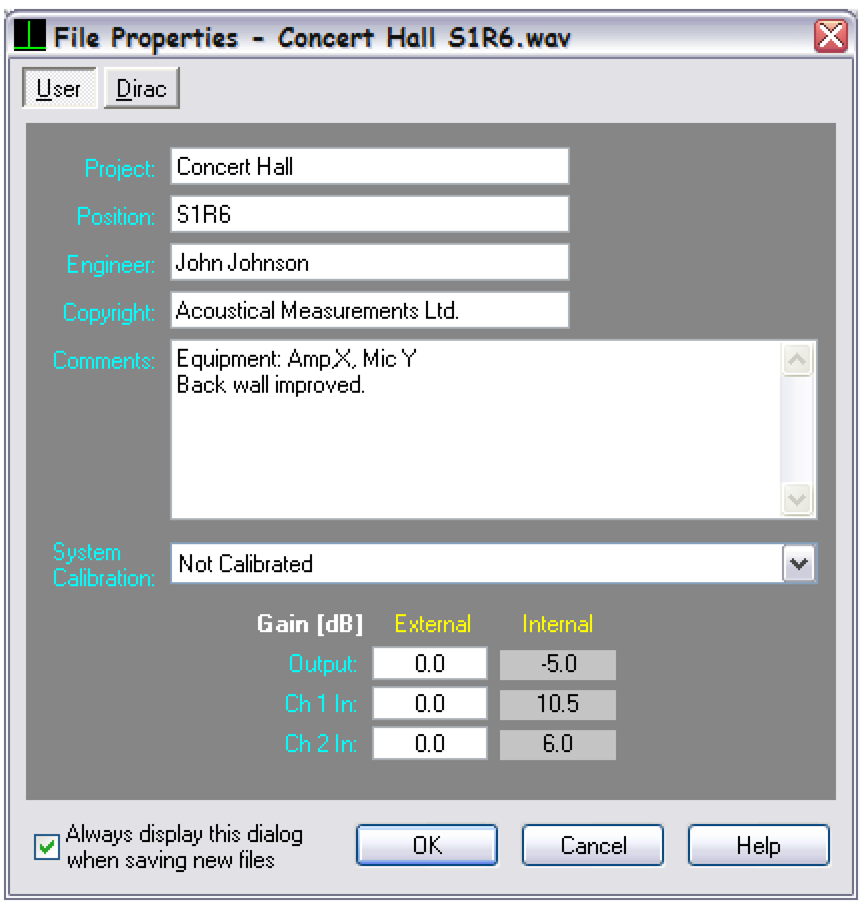
OK!

7 Storing impulse responses

Dirac stores impulse responses in standard wave files, having the extension .wav. In addition to the time domain impulse response, the files stored by Dirac contain extra information that is relevant to the correct interpretation of the measurement.

The user can enter some of this information as the file is being saved.

When saving a file, Dirac will suggest a filename based on the project name and the measurement position (as entered in the dialog below) followed by a sequence number.



8 Viewing the impulse response

8.1 Time domain views

Dirac presents several different views of the impulse response. These views can be accessed quickly using the view mode buttons on the toolbar:
The most elementary view is the time domain view that is presented with a new measurement or when an existing impulse response file is opened. This view helps in judging the quality of the impulse response and allows you to investigate individual reflections. The other time domain views present the forward integration view, the energy-time curve and the decay envelope, which is the noise compensated, backward integrated squared impulse response. This curve serves as a starting point for reverberation time calculations.

It is possible to look at the time domain impulse response within a third- or full-octave frequency band by using the filter buttons:

You can also zoom in on individual features of the response using the zoom buttons:

8.2 Frequency domain views

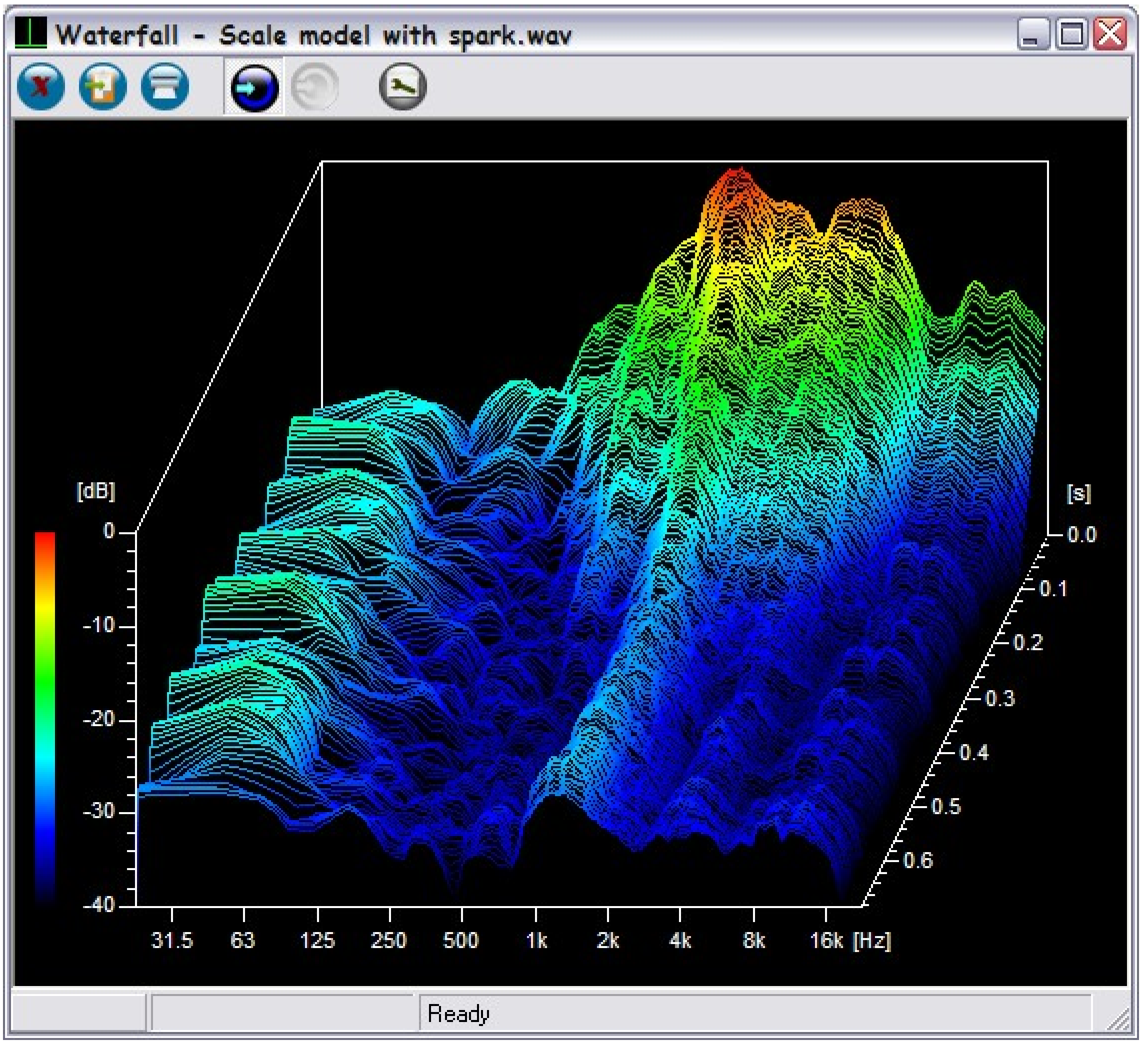
Dirac presents line spectra with a linear or logarithmic time axis. The line spectra can optionally be third-octave smoothed. The frequency views are quickly accessible through the spectrum view mode buttons on the toolbar:

8.3 Overlays

Detailed comparisons between two measurements can be made in both the time- and frequency-domains using overlays (accessible through the **View** menu). Both signals can be aligned automatically based on a crosscorrelation.

8.4 Combined Time-Frequency views

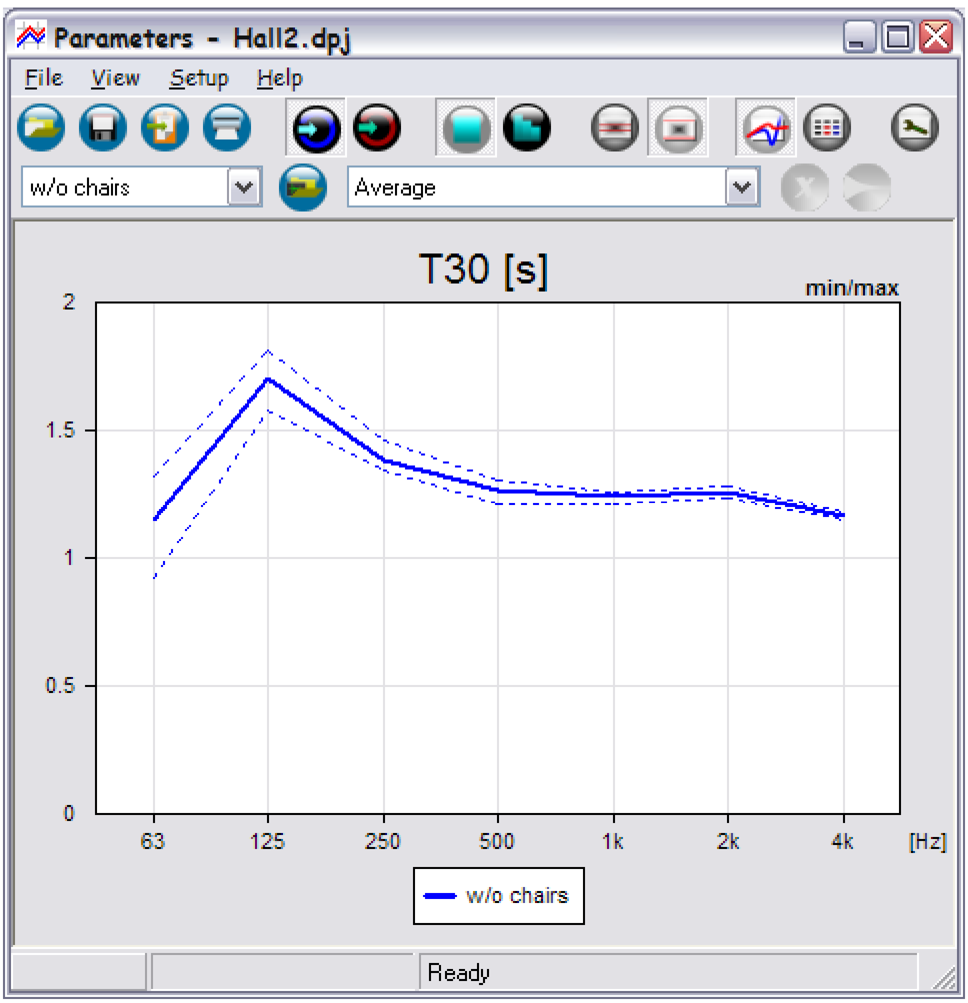
At times (e.g. to investigate modes or a resonance) it is important to see how the spectrum of the impulse response changes with time. To this end, Dirac presents combined time-frequency views (waterfall plot and spectrogram). This view can be accessed through the **Energy Time Frequency Plot** button on the toolbar. An example of this view is displayed below.



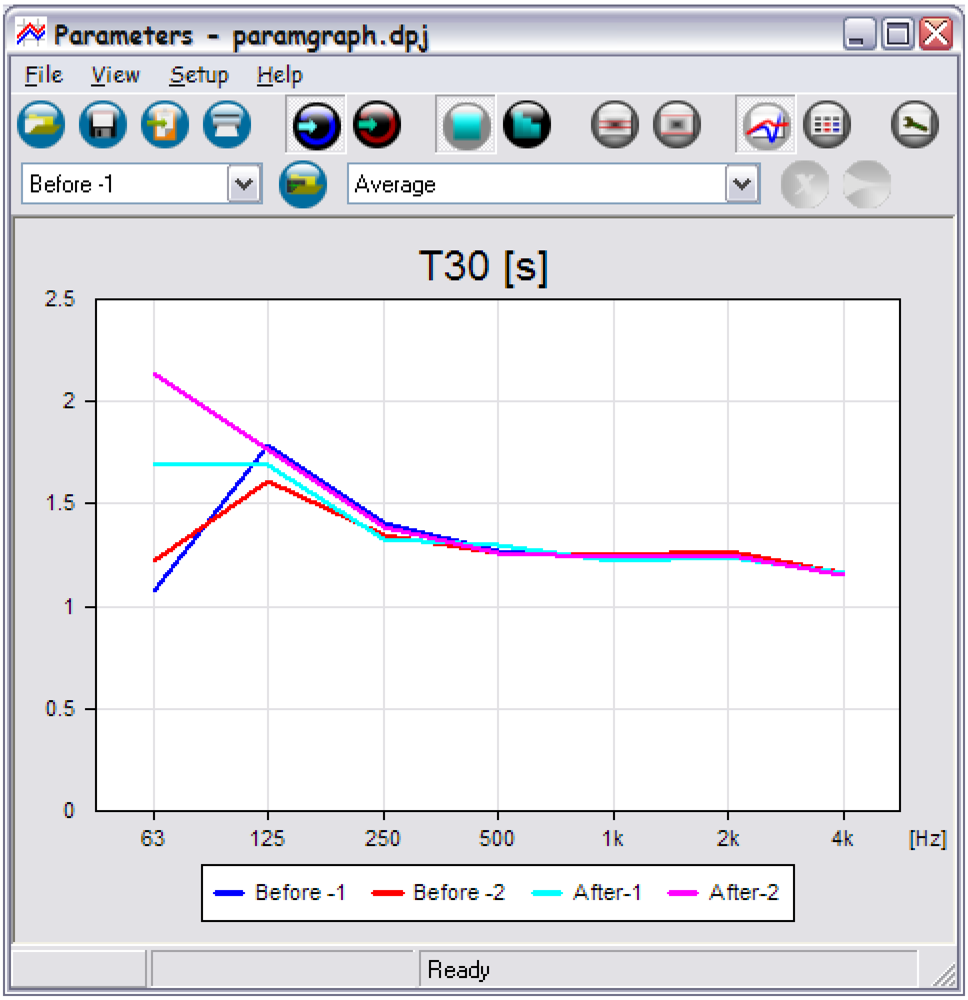
The ETF properties dialog that can be accessed using the **Properties** button  in the toolbar provides many options to modify the way in which the data is displayed.

9 Viewing the parameters

Many acoustical parameters can be calculated from impulse responses. Parameters can be viewed as a graph, as shown below, by clicking the **Graph** button on the toolbar. When multiple files are selected into the Graph window, the average and the spread will be displayed. Files can also be grouped, and multiple graphs can be displayed simultaneously.



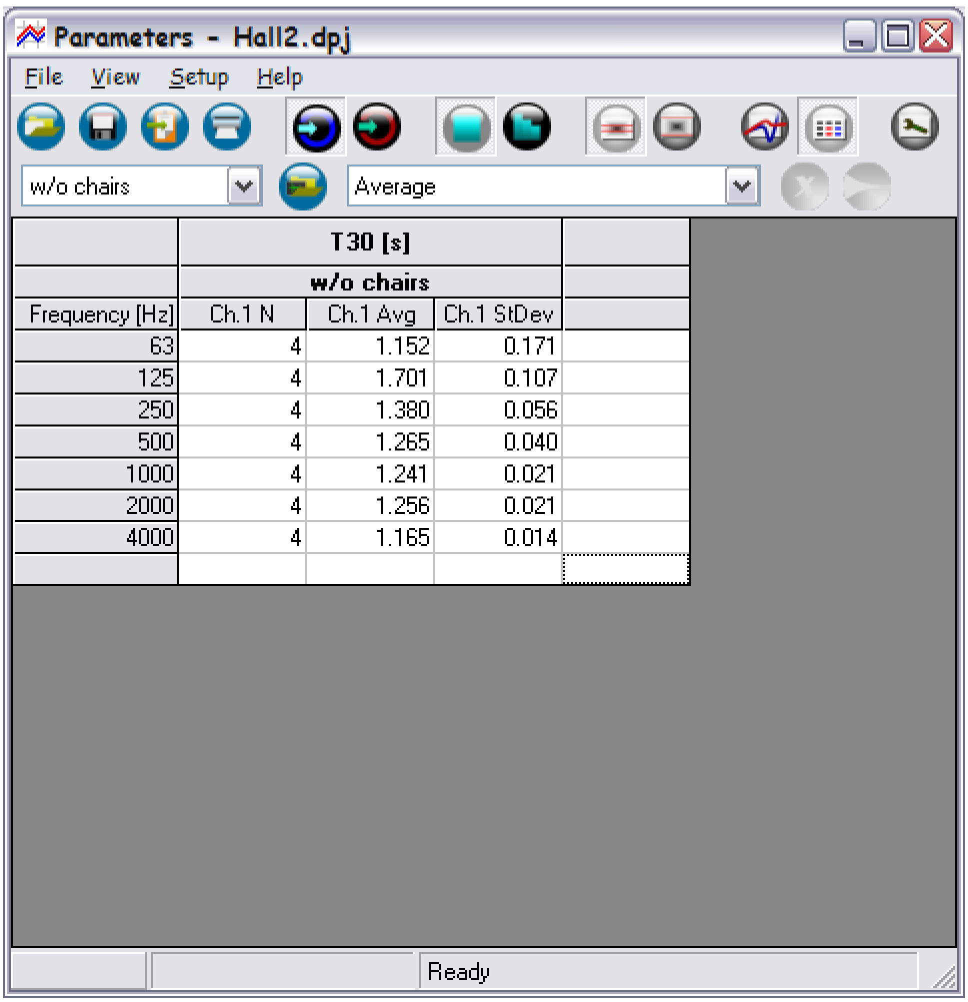
Average and spread of multiple files.



Multiple file groups.

All graphs can be printed, copied to the clipboard, and saved to a bitmap file for inclusion in a report.

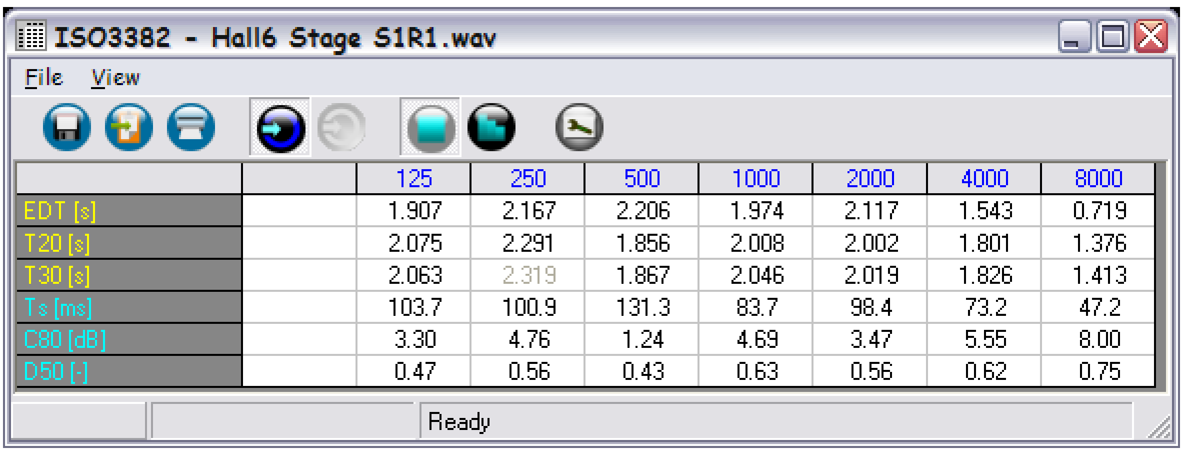
Parameters can also be displayed in table format. This can be done by clicking the **Table** button in the Graph window:



	T30 [s]			
	w/o chairs			
Frequency [Hz]	Ch.1 N	Ch.1 Avg	Ch.1 StDev	
63	4	1.152	0.171	
125	4	1.701	0.107	
250	4	1.380	0.056	
500	4	1.265	0.040	
1000	4	1.241	0.021	
2000	4	1.256	0.021	
4000	4	1.165	0.014	

As with the graphs, tables can be printed, copied onto the clipboard, and saved to a text file that can be opened in a spreadsheet program.

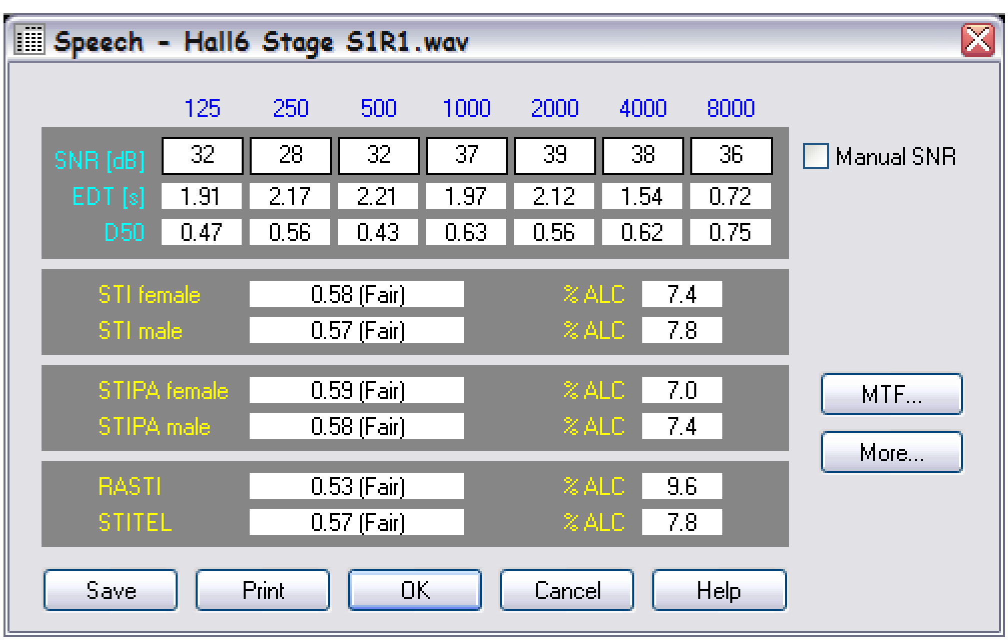
Another way to display tables is through the Parameter menu. The ISO 3382 table, for instance, lists all parameters that are defined in the ISO 3382 standard.


ISO3382 - Hall6 Stage S1R1.wav
File View

		125	250	500	1000	2000	4000	8000
EDT [s]		1.907	2.167	2.206	1.974	2.117	1.543	0.719
T20 [s]		2.075	2.291	1.856	2.008	2.002	1.801	1.376
T30 [s]		2.063	2.319	1.867	2.046	2.019	1.826	1.413
Ts [ms]		103.7	100.9	131.3	83.7	98.4	73.2	47.2
C80 [dB]		3.30	4.76	1.24	4.69	3.47	5.55	8.00
D50 [-]		0.47	0.56	0.43	0.63	0.56	0.62	0.75

Ready

As another example, the speech table lists a number of parameters that define the speech intelligibility.


Speech - Hall6 Stage S1R1.wav

	125	250	500	1000	2000	4000	8000
SNR [dB]	32	28	32	37	39	38	36
EDT [s]	1.91	2.17	2.21	1.97	2.12	1.54	0.72
D50	0.47	0.56	0.43	0.63	0.56	0.62	0.75

Manual SNR

STI female	0.58 (Fair)	% ALC	7.4
STI male	0.57 (Fair)	% ALC	7.8
STIPA female	0.59 (Fair)	% ALC	7.0
STIPA male	0.58 (Fair)	% ALC	7.4
RASTI	0.53 (Fair)	% ALC	9.6
STITEL	0.57 (Fair)	% ALC	7.8

10 References

[1] **ISO 3382** (1997) Acoustics - Measurement of reverberation time in auditoria.

[2] **IEC 60268-16** Sound system equipment. Part 16: Objective rating of speech intelligibility by speech transmission index.

Acoustics Engineering develops systems for the prediction and measurement of acoustical parameters, resulting in user-friendly tools that enable you to perform fast and accurate acoustical measurements and calculations.

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