PRODUCT DATA

DIRAC Room Acoustics Software Type 7841

DIRAC PC software is used for measuring a wide range of room acoustical parameters. Based on the measurement and analysis of impulse responses, DIRAC supports a variety of measurement configurations. For accurate measurements according to the ISO 3382 standard, you can use internally or externally generated stimulus signals through a loudspeaker sound source.

Survey measurements are easily carried out using a small impulsive sound source, such as a blank pistol or even a balloon.

Speech intelligibility measurements can be carried out in compliance with the IEC 60268–16 standard through an artificial mouth-directional loudspeaker sound source, such as Echo Speech Source Type 4720 or through direct injection into a sound system, taking into account the impact of background noise.

DIRAC is a valuable tool not only for field and laboratory acoustics engineers, but also for researchers and educational institutions. To learn more about DIRAC and its applications, please visit www.bksv.com/DIRAC.

Uses and Features

Uses

- Measure the acoustics of any room, hall or space
- Measure speech intelligibility
- Check acoustics before and after modification
- Compare acoustics of different rooms
- Scale model measurements
- Test and validate sound systems
- Research and education on acoustics
- Troubleshooting room acoustics

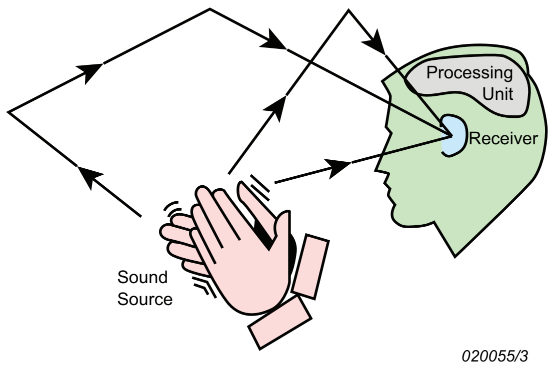
Features

- Dual-channel room acoustics
- Compliant with ISO 3382 (room acoustics), ISO 18233 (analysis methods) and IEC 60268–16 (speech intelligibility)
- Quick and easy measurements using an optional calibrated speech source (Echo Speech Source Type 4720)
- Time reverse filtering to measure short reverberation times
- Impulse response editing with unlimited undo
- Auto Measure for large rooms
- Predefined and user-defined parameters
- Auralisation of any sound played in a room, using the room’s impulse response
- Comparisons and statistics of results

Brüel & Kjær

About DIRAC

Fig. 1
Basic principle of impulse response measurement



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 gives you an impression of whether music would sound pleasant or speech would be intelligible in this room. DIRAC uses this principle as the basis for measuring the acoustical properties of a system through impulse responses.

Impulse Responses

The mathematical impulse or *Dirac delta function*, named after the theoretical physicist Paul A.M. Dirac, is infinitely short and has unit energy. A system's response to such an impulse contains all the information on the system and, as such, is convenient for analysis and storage. DIRAC measures and saves acoustical impulse responses and calculates acoustical parameters from impulse responses.

Other Excitation Signals

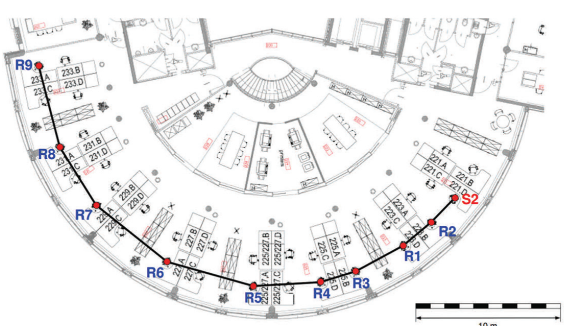
Through deconvolution, DIRAC can also calculate the impulse response using other excitation signals, thereby enabling the use of loudspeaker sound sources. These sources feature a better directivity, frequency spectrum and reproducibility than impulsive sound sources, and offer increased dynamic range through the method of deconvolution. Examples of suitable non-impulsive excitation signals are the MLS (Maximum Length Sequence) signal, the sweep or swept sine (sine with frequency increasing linearly or exponentially with time), white noise and pink noise.

Use Case

In this use case, a recently constructed office building has been fitted with an open-plan office, and DIRAC is used to verify the acoustic performance of the room in accordance with the ISO 3382–3 standard.

The measurements require a calibrated sound source, and DIRAC contains the tools to perform such a calibration in a diffuse-field (reverberation chamber) or in a free-field setting. The measurements start with a standard level calibration of the microphone.

Fig. 2
DIRAC measurement in open-plan office. Impulse responses are measured at different workstations. All ISO 3382–3-relevant room acoustic parameters can be calculated from these parameters



The loudspeaker source is positioned at one of the workstations (S2) and a sound level meter is moved, in turn, to each measurement position (R1–R9). DIRAC, running on a laptop, switches on the speaker test signal and instantly calculates the speech intelligibility indices from the sound level meter microphone signal.

The new intermittent stimulus makes it possible to get all four required measurement quantities from a single measurement at each position.

Comparisons of speech intelligibility indices at various seating positions and statistical analysis of the results are available in DIRAC.

Fig. 3
Measurement system: Excitation signal from PC

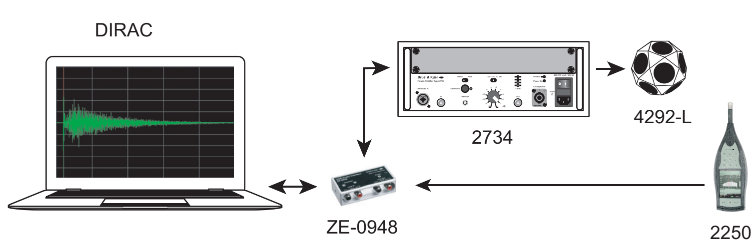
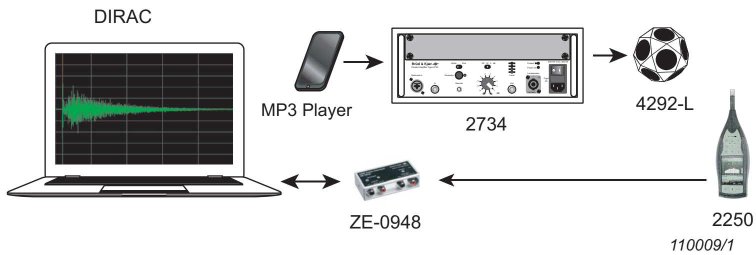


Fig. 2 illustrates the typical use of DIRAC to measure reverberation time and speech intelligibility in an open-plan office. The system components are shown in Fig. 3 and include the recommended USB Audio Interface ZE-0948 (see Sound Device on page 4).

If PC and sound source are far apart, the excitation signal may be played back from a CD or MP3 player (Fig. 4).

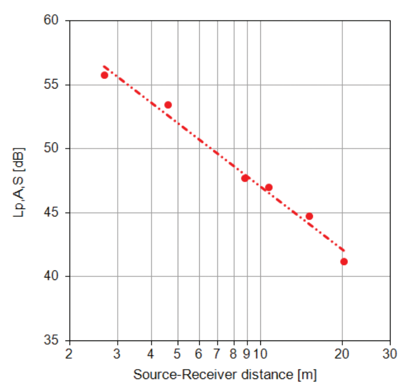
Fig. 4
Measurement system: Excitation signal from MP3 player



A CD with suitable stimulus signals is included with DIRAC for this purpose. Power Amplifier Type 2734-B is available with a wireless audio system, which makes it easy to place and move the power amplifier and speaker.

Parameters may be plotted versus frequency or versus source-receiver distance, as in Fig. 5, where the measured A-weighted sound pressure levels of a standard speech signal are graphed against the logarithmic source-receiver distance. From the regression line through these values DIRAC can calculate $L_{p,A,S,4}$, the A-weighted speech level at 4 m, and $D_{2,S}$, the spatial decay rate of speech.

Fig. 5
Regression line through the speech sound levels, indicating the spacial decay of sound



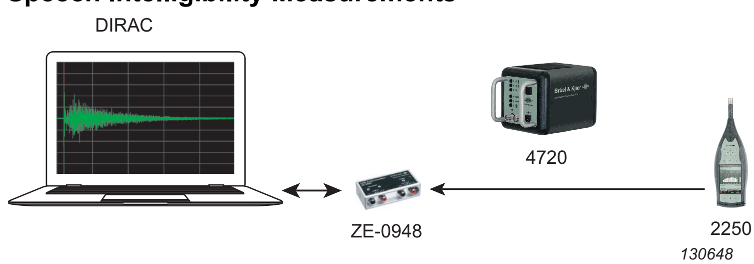
A similar graph can be generated for the Speech Transmission Index (STI) with a regression line to calculate the distraction distance r_D and the privacy distance r_P.

All parameter values detailed in ISO 3382–3 can be viewed in DIRAC and exported at once for use in a report.

In combination with the single measurement per position, this makes for a very fast and easy solution to the problem of the plan office measurements.

Speech Intelligibility Measurements

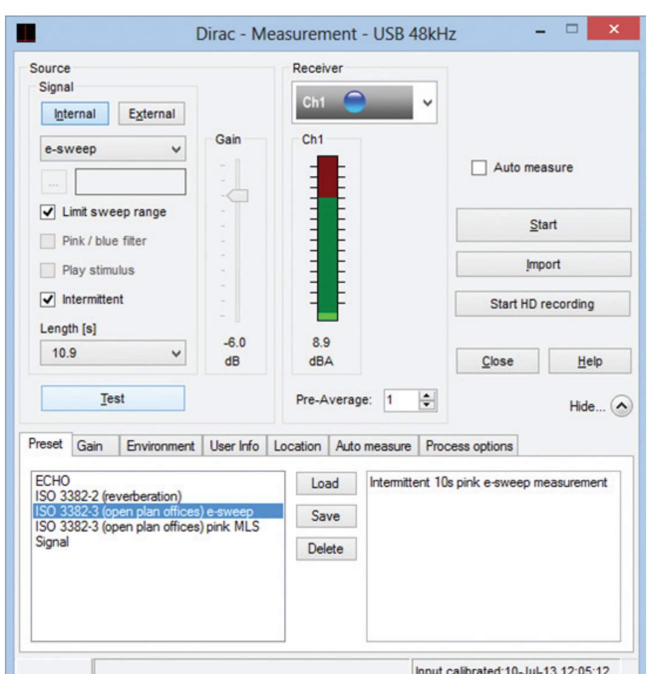
Fig. 6
Measurement system: Excitation signal from Echo Speech Source Type 4720



For speech intelligibility measurements in accordance with IEC 60268–16, Echo Speech Source Type 4720 is recommended.

User Interface

Fig. 7
Typical DIRAC setup, with a 10.9 second e-sweep excitation in test mode.



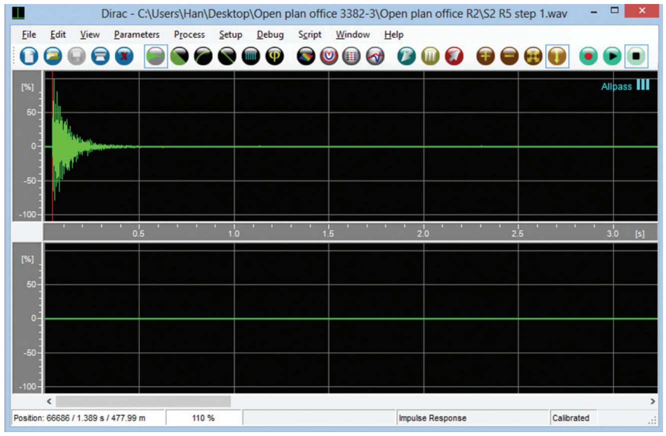
The DIRAC user interface is shown in Fig. 7, ready for measurement. Measurement presets make it very easy to set up a measurement.

In this example, the setup for an ISO 3382–3 measurement is chosen. excitation signal, e-sweep is selected due to its superior rejection of background noise and distortion.

Background noise is suppressed in exchange for longer excitation time – 3 dB improvement for each doubling of excitation time. In this case, a reverberation time of around 1 s is expected, and a sweep time of 10.9 s will improve the impulse response-to-noise ratio (INR) by over 10 dB compared to traditional measurement methods.

Pressing the Test button runs the sweep signal repetitively for adjusting speaker output level and microphone input level.

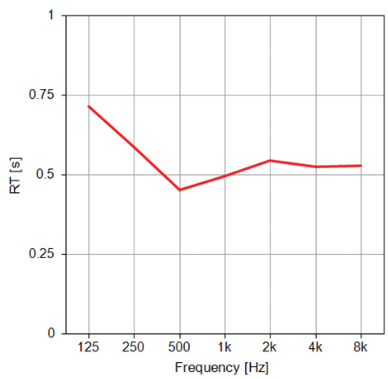
Fig. 8
The measured impulse response – for saving, inspection, listening and calculation of all room acoustic parameters



Pressing the Start button will run the measurement and calculation, and present the impulse response (Fig. 8). This should have a clean "tail" with no background noise. Listening to the impulse response played back through the speaker will verify this and reveal any audible echoes in the room.

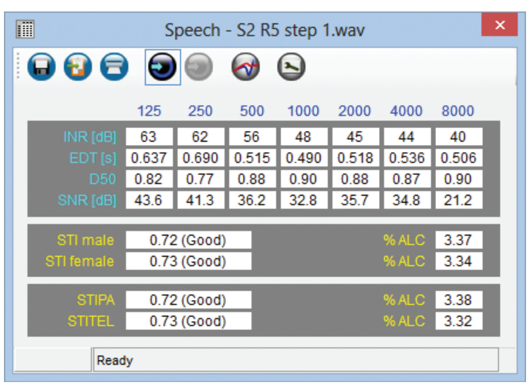
Results

Fig. 9
Reverberation Time spectrum showing values that are long for an office environment



From the impulse response, any room acoustic parameter can be calculated. Selecting "Reverberation" calculations shows the RT spectrum (Fig. 9). It is relatively high at 125 Hz, which means the sound is likely "booming" and not very suitable for an open-plan office.

Fig. 10
"Speech" calculation, with "Good" results



In "Speech" calculations, Fig. 10 shows "Good" intelligibility. The maximum possible score is 1.00, and the score of around 0.7 is due to the reverberation time of the room (the high signal-to-noise ratio (SNR) values indicate little background noise). The influence of lower SNR values may be simulated by manually entering speech and noise levels. Usually a number of speaker and listener positions are investigated to ensure that the spread in results is within specification and to identify possible bad spots. Parameter graphs are available to visualise and document this, as in Fig. 25 through Fig. 31.

Required Hardware

The minimum hardware required to use DIRAC comprises a PC with a sound device, an impulsive sound source, such as a blank pistol, and a microphone connected to the actual sound device line input. Each of these three components can be varied, depending on the type of measurement to be performed.

Sound Device

Fig. 11
USB Audio Interface ZE-0948 is recommended for line level interface to microphone and speaker systems. ZE-0948 comes with DIRAC as Type 7841-100 (see Ordering Information)

Typical sound device functions, used by DIRAC, are line input, line output and gain controls. In case of a notebook or laptop PC, sound device functions are integrated or can be used via a USB device. DIRAC determines sound device properties using a sound device loopback test, where the sound device output is connected to the input. The test determines the optimum sample format checks the linearity, synchronicity and frequency response. In this way, the software becomes independent of the sound device and the input and output gain can be easily controlled from within DIRAC. A dedicated sound device (Fig. 11) is recommended. USB Audio Interface ZE-0948 (the size of a matchbox) has dual-channel line inputs and outputs, and input overload indication. It connects to a USB socket for power and interface and complies with standard USB class specifications and plug-and-play architecture, allowing instant use with no user-installed drivers.

At high sound pressure levels, the signal from the microphone may be sufficient to perform impulse response measurements when fed directly into the sound device line input; however, additional amplification is usually required. In this case, a sound level meter with a line output could be used. For a list of recommended models, see Ordering Information, page 12.

Sound Sources

Fig. 12
The omnidirectional sound source OmniPower Type 4292-L is recommended for ISO 3382 compliant measurements

Instead of using an impulsive sound source, a loudspeaker sound source can be used. To measure room acoustical parameters in compliance with the ISO 3382 standard, an omnidirectional sound source such as the OmniPower™ Type 4292-L should be used.

OmniPower sound source features:
- Omnidirectional (12-loudspeaker dodecahedral configuration)
- Lightweight (8 kg)
- Maximum sound power of 122 dB re 1 pW (100 – 3150 Hz)
- Conforming with DIN 52210, ISO 140 and ISO 3382 standards

You can use the Echo Speech Source Type 4720 to simulate a real speaker in speech intelligibility measurements according to the IEC 60268–16 standard. The speech source was developed specifically for use with DIRAC and delivers fully calibrated signals for various measurement conditions.

Fig. 13
Echo Speech Source Type 4720 is the preferred source for speech intelligibility measurements

Echo Speech Source features:
- Fully calibrated sound levels and spectra
- Reference speech signal at 1 m: 60 dB(A)
- Five built-in stimulus signals
- Mounts on common tripods
- External input and output
- Compact and lightweight (10×10×17 cm, 1 kg)

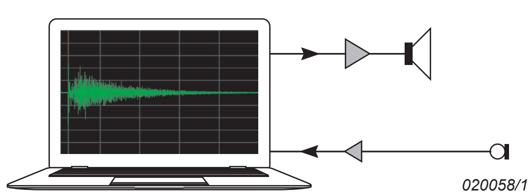
You can also measure speech intelligibility through a sound reinforcement system using that system's loudspeakers and obtaining the excitation signal directly from DIRAC or an external device such as Echo Speech Source or a CD or MP3 player. A stimuli CD with a range of excitation signals comes with DIRAC.

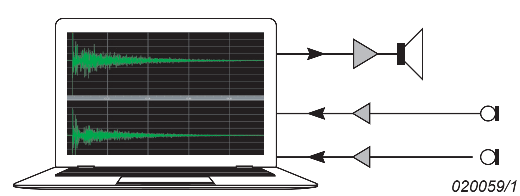
Measurement Methods

DIRAC supports several impulse response measuring methods, which are related to the sound source. Which method is used, depends on the situation.

The Internal MLS, lin-Sweep, or e-Sweep methods are accurate but require a connection between the PC and a loudspeaker sound source or some other system.

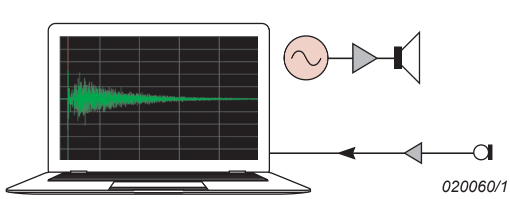
Fig. 14
Internal stimulus. DIRAC produces the excitation signal

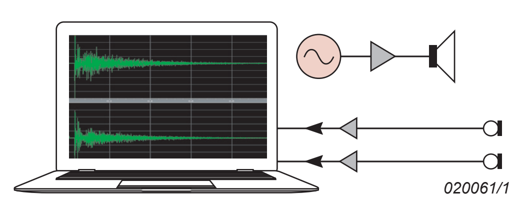




The External MLS, lin-Sweep, or e-Sweep methods do not require a connection between the PC and a sound source or other system, which is convenient for long distances. An external device is required, such as a CD or MP3 player. This is illustrated in Fig. 4, Fig. 6 and Fig. 15.

Fig. 15
External stimulus. Use the Stimulus CD supplied with DIRAC, or record your own stimuli from DIRAC's output





The External Noise method allows the use of any broadband continuous signal source, such as noise or music, but the method is less accurate, and only one measurement channel is available.

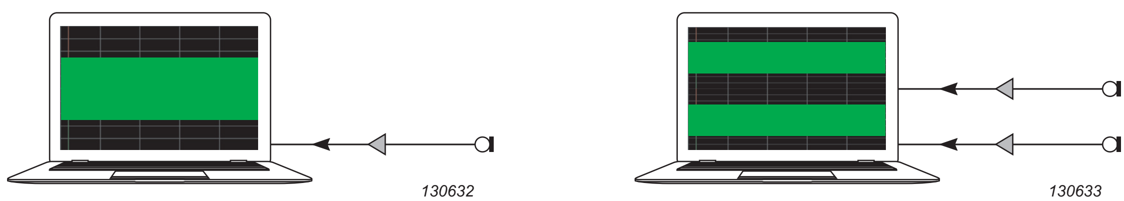
Fig. 16
External Noise: excitation by broadband signal, such as noise or music

The External Impulse method allows the use of small lightweight sound sources, such as balloons or blank pistols, but is less accurate.

Fig. 17
External Impulse: excitation by impulsive signal, such as from blank pistol or paper bag

The Signal method is used for sound level and sound directivity measurements.

Fig. 18
Signal measurement: DIRAC can also handle plain signal measurements without a specific excitation



Practical Examples

Fig. 19 shows some examples of practical measurement setups to calculate spatial parameters.

Fig. 19
Left: *Reverberation time and energy ratio survey measurement*
Right: *Measuring inter-aural cross correlation (IACC)*

Acoustical Parameters

DIRAC can calculate a set of acoustical parameters, from single- or dual-channel impulse responses, depending on the receiver type used during the measurement. You can select from six different receiver types (Table 1). The full list of available acoustical parameters is given in Specifications on page 11.

Table 1
Relation between receiver type selected and parameters to be calculated

Receiver Type	Button	Parameters
Single Omnidirectional Microphone		All but spaciousness and I
Switchable Omni-bidirectional Microphone		All but IACC and I
Dual Omnidirectional Microphone		All but LF, LFC, IACC and I
Omnidirectional + Bidirectional Microphone		All but spaciousness and I
Head Simulator		IACC, $IACC_x$
Intensity Microphone Probe		All but IACC

Calibration

DIRAC supports a sound-device loopback test, which enables optimal operation and user control of the sound device from within DIRAC, and two different kinds of calibration:

1. System calibration – enables the measurement of the sound strength (G) and related parameters, and improves the accuracy of lateral fraction (LF and LFC) measurements if needed.

System calibration is best performed in a reverberation chamber, but it can also be performed in a free-field environment, for instance, in situ on a large stage.

2. Input level calibration – enables measurement of absolute sound pressure (or voltage) levels and related parameters ($L_{eq,I}$).

For ISO 3382–3 measurements, both calibrations are required.

Results

Impulse Response Views

DIRAC can display an impulse response in several ways. The Energy-Time Curve shows the average energy progression or highlights the energy peaks, the Forward Integration Curve shows the cumulative energy progression, and the Decay Curve displays the backwards integrated energy progression.

In a time domain view you can select any part of the impulse response, and then edit, listen to or view details of the selected interval.

Fig. 20
Time domain views: original impulse response and Energy-Time Curve from a single-channel measurement

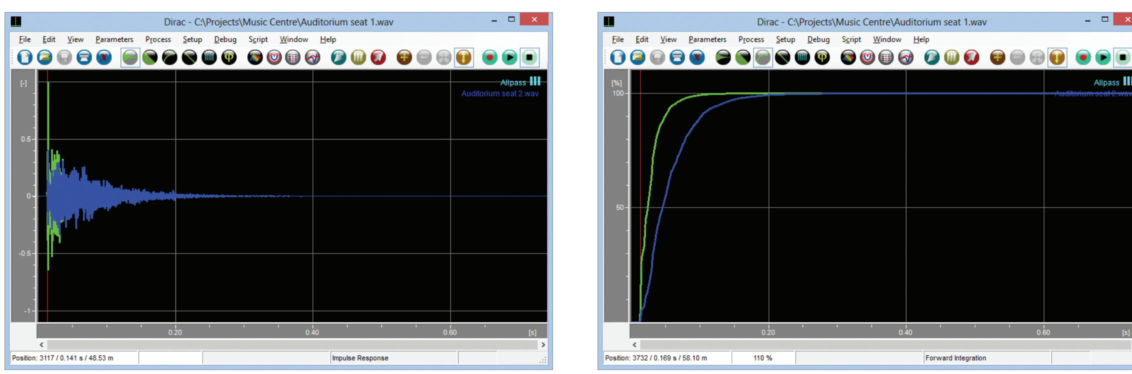


Fig. 21
Time domain views: comparing two single-channel impulse responses and their Forward Integration Curves

Several frequency spectrum views allow convenient magnitude and phase analysis in the frequency domain.

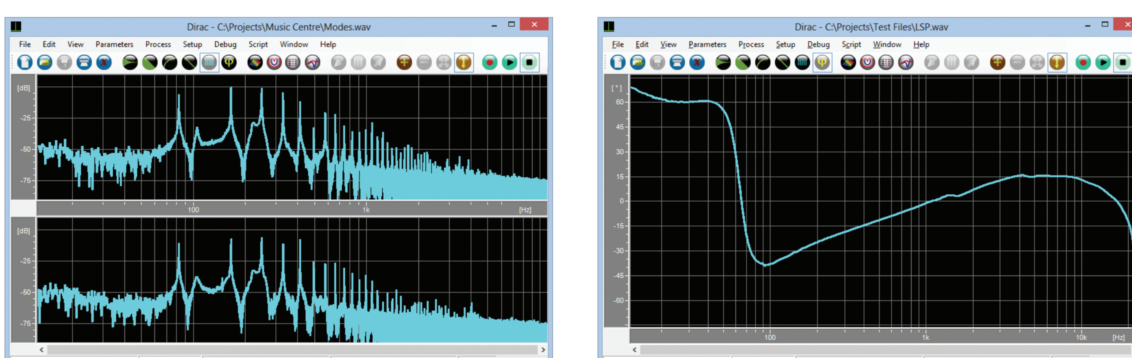


Fig. 22
Frequency domain views: linear FFT and phase spectra

Energy-Time-Frequency Plots

To give a clear view of the spectral progress of an impulse response, DIRAC features several types of energy-time-frequency plots, such as the CSD (Cumulative Spectral Decay) and the spectrogram.

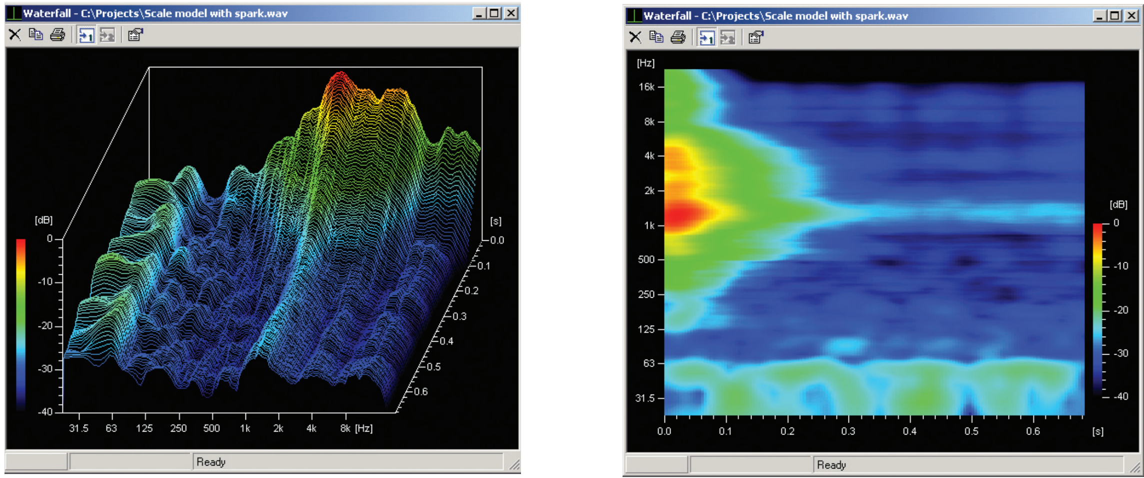


Fig. 23
Energy-Time-Frequency plots: waterfall plot and spectrogram

Directivity Plot

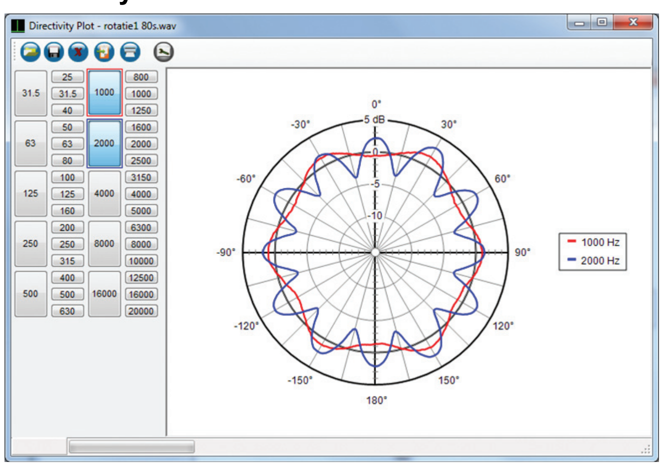


Fig. 24
Directivity plot: The directivity of a sound source in two different frequency bands

The directivity of a sound source can be investigated by playing a pink-noise signal through the source and recording the resulting output while rotating the device. The recorded signal can then be graphed in a directivity plot. The directivity plot can also be used to investigate the directivity of acoustical parameters such as *Strength* or *Clarity*.

Parameter Tables and Graphs

Acoustical parameters, derived from the impulse responses, can be displayed in table format or graphically. Measurements can be grouped, and over each group of files you can calculate averages, minima, maxima, and standard deviations of the measured acoustical parameters. Grouped files and their setup can be saved as a Project. The results can be viewed on screen or copied and pasted into a report. You can also calculate and save, in a single run, a user-defined set of parameters over a project.

Fig. 25
Parameter tables can be customised, for example, to show specific related parameters

ISO3382 - Chamber Music Hall S1R1.wav

		125	250	500	1000	2000	4000
T20 [s]		1.788	1.411	1.224	1.206	1.225	1.132
T30 [s]		1.644	1.362	1.212	1.237	1.283	1.158
G [dB]		14.84	11.48	12.40	11.30	10.23	10.23
Ts [ms]		108.5	65.6	60.9	61.2	74.8	76.5
C80 [dB]		0.98	4.85	5.04	4.80	2.82	2.99
D50 [-]		0.37	0.63	0.56	0.64	0.56	0.52
LF [-]		0.18	0.07	0.07	0.18	0.18	0.17

Fig. 26
Parameter graph showing magnitude spectrum

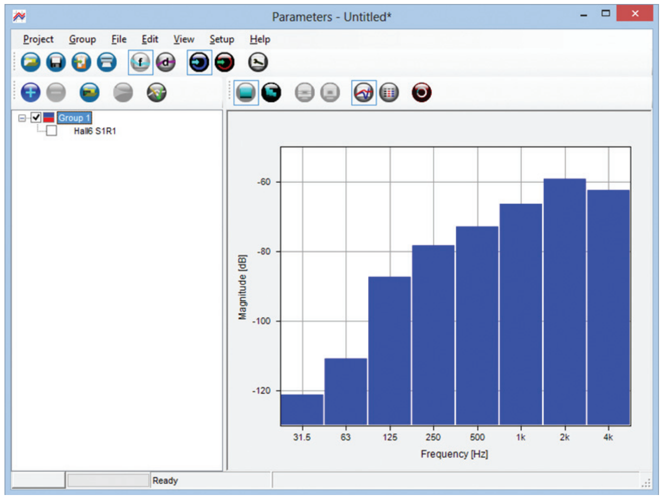


Fig. 27
Parameter graph showing D_{50} average and standard deviation over four receiver positions

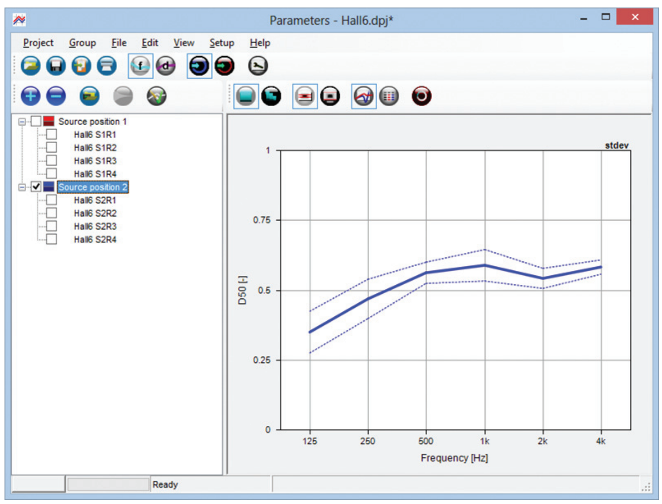


Fig. 28
Parameter graph showing D_{50} average over four receiver positions, for two different source positions

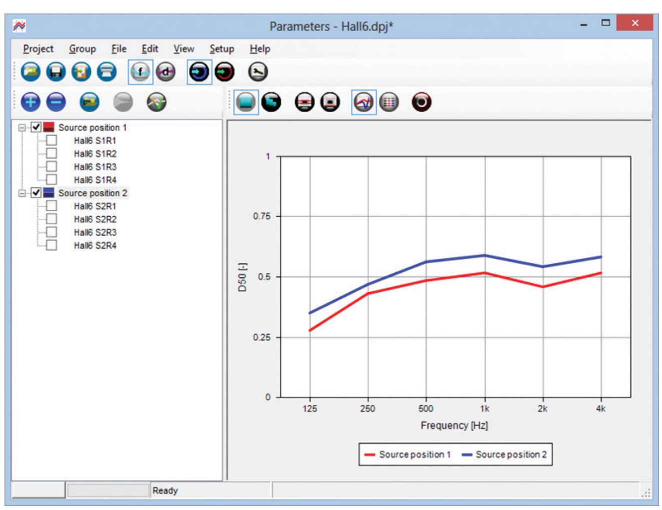


Fig. 29
Parameter graph showing STI for different receiver positions plotted against the source-receiver distance

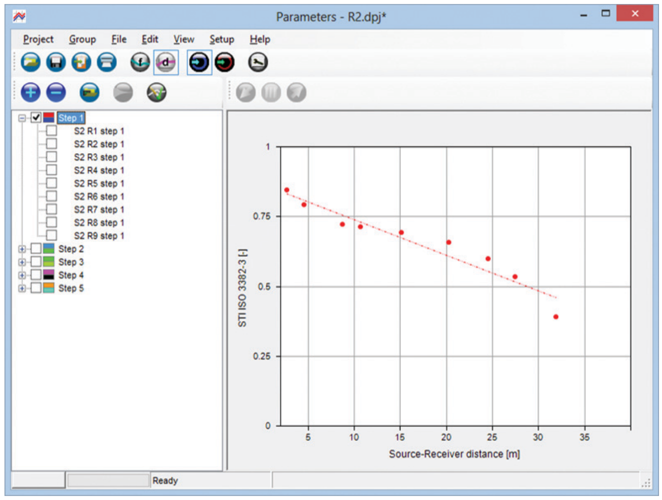
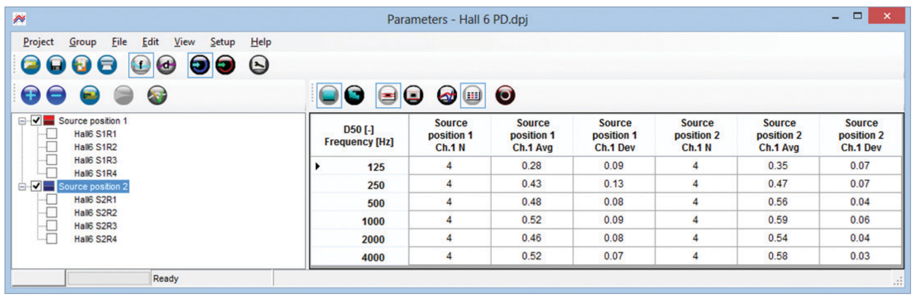
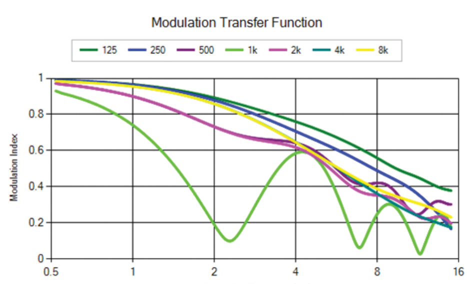


Fig. 30
D_{50} table: Average and standard deviation over four measurement positions for two source position groups and two channels per measurement. For each frequency, the number of usable results is given



D50 [-] Frequency [Hz]	Source position 1 Ch.1 N	Source position 1 Ch.1 Avg	Source position 1 Ch.1 Dev	Source position 2 Ch.1 N	Source position 2 Ch.1 Avg	Source position 2 Ch.1 Dev
125	4	0.28	0.09	4	0.35	0.07
250	4	0.43	0.13	4	0.47	0.07
500	4	0.48	0.08	4	0.56	0.04
1000	4	0.52	0.09	4	0.59	0.06
2000	4	0.46	0.08	4	0.54	0.04
4000	4	0.52	0.07	4	0.58	0.03

Fig. 31
MTF graph showing MTI as a function of the modulation frequency



The MTF graph can be used to investigeate the causes for low Modulation Transmission Index (MTI) values. A constant MTF over F indicates background noise, a continuously decreasing MTF indicates reverberation and an MTF first decreasing and then increasing with F indicates an echo (Fig. 31).

Other Applications

Fig. 32
Measurement in a scale model of a rever-beration chamber, using a miniature omnidirectional sound source

Scale Model Measurement

To predict the acoustics of, for instance, a concert hall that is being designed but not yet realised, you can measure impulse responses in a scaled down model of the hall. After DIRAC has converted the scale model impulse responses to real-world impulse responses, you can analyse them in the usual way. For scale model measurements using an impulsive source, DIRAC can post-average the impulse responses with automatic alignment.

To hear in advance how, for instance, a trumpet will sound in the real hall, in DIRAC you can convolve a dry trumpet recording with the converted impulse responses.

Specifications – DIRAC Room Acoustics Software Type 7841

STANDARDS

Conforms with the following:
IEC 61260: 1/1-octave and 1/3-octave Bands Class 0
ISO 3382: Acoustics – Measurement of the reverberation time of rooms with reference to other acoustical parameters
IEC 60268–16: Sound system equipment – Part 16: Objective rating of speech intelligibility by speech transmission index
ISO 18233: Application of new measurement methods in building and room acoustics
ISO/DIS 13472–1: Acoustics – Measurement of sound absorption properties of road surfaces in situ – Part 1: Extended surface method
NF S 31–089: Acoustique – Code d'essai pour la détermination de caractéristiques intrinsèques des écrans installés in situ
EN 1793–5: Road traffic noise reducing devices – Test method for determining the acoustic performance – Part 5: Intrinsic characteristics – In situ values of sound reflection and airborne sound insulation
EN 1793–6: Road traffic noise reducing devices – Test method for determining the acoustic performance – Part 6: Intrinsic characteristics – In situ values of airborne sound insulation under direct sound field conditions

OPERATION

The software is a true 32-bit Windows® program, operated using buttons and/or menus and shortcut keys

HELP AND USER LANGUAGE

Concise context-sensitive help is available throughout the program in English

MEASURING METHODS

Internal or External: MLS, lin-Sweep, e-Sweep, Noise, Impulse and User-defined
Intermittent stimulus (IR + background noise)
Echo Speech Source
Signal measurements
Stimulus lengths: 0.34 – 350 s
Pre-average: 1 – 999 times
Filters: None, Pink + Blue stimulus (optional)
Measurements can be executed automatically

RECEIVER TYPES

Single omnidirectional, dual omnidirectional, switched omni-bidirectional, omnidirectional and bidirectional, artificial head, sound intensity probe

FREQUENCY RANGE

10 1/1-octave bands from 31.5 Hz to 16 kHz
30 1/3-octave bands from 20 Hz to 20 kHz

CALCULATED PARAMETERS

- Early Decay Time, EDT
- Reverberation Times, T_{10}, T_{20}, T_{30}
- Reverberation Time (user-defined decay range), T_X
- Reverberation Time (from best decay sections), RT
- Bass Ratio (based on reverberation time), BR(RT)
- Impulse response-to-Noise Ratio, INR
- Signal-to-Noise Ratio, SNR
- Treble Ratio (based on reverberation time), TR(RT)
- Peak-to-Noise Ratio, PNR
- Strength (Level relative to 10 m free-field), G
- Strength over user-defined interval, G_{XY}
- Early Strength, G_{80}
- Late Strength, GL
- Relative Strength, G_{rel}
- Magnitude Spectrum
- Magnitude Spectrum Pink (–3 dB/octave offset)
- Equivalent Sound Level, L_{eq}
- Equivalent (A- and C-Weighted) Sound Level, L_{Aeq}, L_{Ceq}
- Minimum Sound Level (F/S and A-/C-/Z-weighting), L_{min}
- Maximum Sound Level (F/S and A-/C-/Z-weighting), L_{max}
- Peak Sound Level (F/S and A-/C-/Z-weighting), L_{peak}
- Percentile Sound Level (F/S and A-/C-/Z-weighting), L_N
- Bass Ratio (based on level), BR(L)
- Treble Ratio (based on level), TR(L)
- Sound Intensity, I
- Level Difference, D
- Centre Time, T_S
- Clarities, C_{30}, C_{50}, C_{80}
- Clarity (user-defined integration interval), C_X
- Definition (Deutlichkeit), D_{50}
- Definition (Deutlichkeit, user-defined integration interval), D_X
- Hallmass, H
- Energy Ratio, ER
- Echo Criterion (for music and speech), EC_{music}, EC_{speech}
- Echo Criterion (user defined), EC_{user}
- Early Lateral Energy Fractions, LF, LFC
- Inter-Aural Cross-correlation Coefficient, $IACC_{80}$
- Inter-Aural Cross-correlation Coefficient (user-defined integration interval), $IACC_X$
- Early Lateral Sound Level, GEL
- Late Lateral Sound Level, LG
- Early Support, ST_{early}
- Late Support, ST_{late}
- Total Support, ST_{total}
- User-defined Support, ST
- Modulation Transfer Index, MTI
- Speech Transmission Index for ISO 3382–3, STI
- Speech Transmission Index (male and female), STI
- STI for PA Systems, STIPA
- Room Acoustics STI, RASTI
- STI for TELecommunication Systems, STITEL
- Percentage Loss of Consonants, % ALC
- A-weighted SPL of Speech, $L_{p,A,S}$
- A-weighted SPL of Speech at 4 m, $L_{p,A,S,4}$
- Spatial Decay Rate of Speech, $D_{2,S}$
- Distraction Distance, r_D
- Privacy Distance, r_P
- Reflection Index, RI
- Sound Insulation Index, SI
- Sound Power Reflection Factor, Q_W

POST-PROCESSING

All parameters can be viewed in table and/or graph format
Parameters are graphed versus frequency or source-receiver distance
Measurements can be grouped, and over each group the average, standard deviation, minimum and maximum can be calculated
The calculated results of multiple groups can be displayed in a single graph or table
Groups can be saved in project files

CALIBRATION

Sound Device Loopback Test: For optimum sample format selection and linearity, synchronicity and frequency response checks
System Calibration: In diffuse or direct sound fields, for measurement of Strength G and related parameters
Input Level Calibration: For sound level measurements and speech intelligibility measurements that have to be evaluated for various background noise conditions

REVERBERATION TIME RANGE

1/1-octave bands: 0.002 – 100 s (1 kHz)
1/3-octave bands: 0.006 – 100 s (1 kHz)
Minimum reverberation times inversely proportional to frequency

SCALE MODEL

Scaling Factors: Adjustable between 0.01 and 100
Frequency Range: 80 kHz (1/3-octave band), at 192 kHz sample frequency

AURALIZATION

The impulse response sample rate is adjusted automatically to match that of the anechoic sound fragment. The sound source frequency characteristic can be compensated for to avoid sound coloring

IMPULSE RESPONSE VIEWS AND PLOTS

Impulse Response, Energy-Time Curve, Forward Integration Curve, Decay Curve, magnitude frequency spectrum, phase frequency spectrum, CSD plot, waterfall plot, spectrogram. Overlay view of second impulse response. Directivity plot

PRINT AND EXPORT

Graphs and tables can be exported via the clipboard, or printed. All results can be printed or exported in ASCII (text) format for further processing in other programs, or exported in Odeon format. Calculated results of multiple parameters can be saved for an entire project

SUPPORTED FILE FORMATS

Wave (.WAV) 8-/16-/24-/32-bit integer. 32-/64-bit float. 1 – 2 channels
Raw (.PCM) 8-/16-bit integer, 32-bit float. 1 – 2 channels
Text (.TXT) 32-bit float. 1 – 2 channels
MLSSA (.TIM) 32-bit float. 1 channel

COMPUTER SYSTEM REQUIREMENTS

Operating Systems: Windows® 8, Windows® 7 or Windows Vista®
Microsoft .NET Framework 4.5
CPU: Minimum 1 GHz
RAM: 2 GB
Free Disk Space: Minimum 500 MB
Auxiliary Hardware: CD-ROM drive, 16- to 32-bit graphics display/adaptor, mouse or other pointing device
Sound Device: 2 channels, full duplex, 22.05, 44.1, 48, 96 or 192 kHz sample rate, support for CoreAudio/WASAPI or ASIO

Ordering Information

Type 7841 DIRAC Room Acoustics Software
includes:
- Software on CD ROM
- Stimuli Signals on CD ROM
- HASP Key
- Loopback Cable AO-0593*

OPTIONAL ACCESSORIES

ZE-0948 USB Audio Interface
includes:
- Leather pouch KE-0456
- USB Cable Assembly AO-0708
- 2 × RCA Phono Cable AO-0707
- 4 × RCA Phono to BNC Adaptor JP-0071
- 1 × RCA Phono to 1/4″ Jack Adaptor JP-0072

Type 4720 Echo Speech Source
includes:
- Installation software
- Power Supply ZG-0864

Type 2238	Integrating Sound Level Meter
Type 2239	Integrating Sound Level Meter
Type 2250	Hand-held Analyzer
Type 2270	Hand-held Analyzer
Type 2260	Precision Sound Level Analyzer
Type 7841-X-100	Upgrade to latest version of Type 7841

* Cannot be used with the USB Audio Interface ZE-0948. Use Cable AO-0707 (included with USB Audio Interface ZE-0948)

AO-0585	Cable from 2238/2239 AC Output to Sound Device Input (3.5 mm jack plug), 3 m
AO-0586-D-030	Cable from 2250 or 2260 Aux. Output to Sound Device Input (3.5 mm jack plug), 3 m (10 ft)
AO-0586-D-100	Cable from 2250 or 2260 Aux. Output to Sound Device Input (3.5 mm jack plug), 10 m (33.3 ft)
AO-0586-D-150	Cable from 2250 or 2260 Aux. Output to Sound Device Input (3.5 mm jack plug), 15 m (50 ft)
AO-0592	Cable to extend AO-0585 or AO-0586 (3.5 mm jack plug female/male), 10 m (33.3 ft)
AO-0592-V	Extension Cable, 3.5 mm jack plug, customer-specified length

NOTE: For information on sound sources, please see Sound Sources on www.bksv.com.

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