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.

Wireless and multi-channel features enable fast measurement setup and execution.

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

- Measuring the acoustics of any room, hall or space
- Speech intelligibility measurements
- Checking the acoustics before and after modification
- Comparison of acoustics of different rooms
- Scale model measurements
- Test and validation of sound systems
- Research and education on acoustics
- Troubleshooting room acoustics

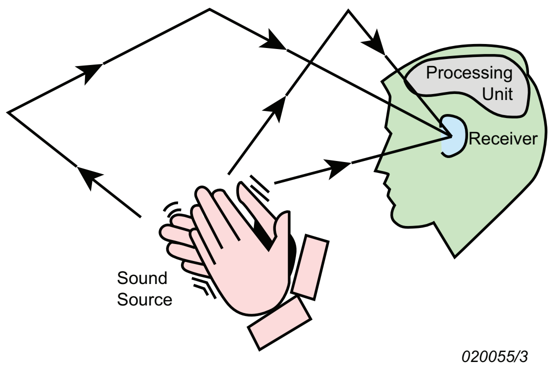
Features

- Multi-channel room acoustics
- Compliant with ISO 3382 (room acoustics), ISO 18233 (analysis methods) and IEC 60268-16 (speech intelligibility)
- Wireless measurements using HBK 2255 Sound Level Meter and HBK 2755 Smart Power Amplifier
- 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
- Auralization of any sound played in a room, using the room's impulse response
- Comparisons and statistics of results

About DIRAC

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.

Fig. 1   Basic principle of impulse response measurement



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 linear, time-invariant 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

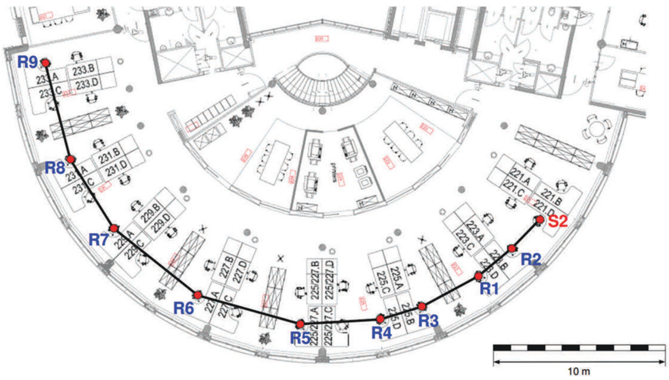
Through the method of deconvolution, DIRAC can also calculate the impulse response using other excitation signals, thereby enabling the use of loudspeaker sound sources. These sources feature a better directivity, frequency spectrum and reproducibility than impulsive sound sources, and offer increased dynamic range through 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 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(s).

Fig. 2   DIRAC measurement in an 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 (SLM) 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. Multiple (wireless) sound level meters can significantly speed up the measurement process.

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 analyses of the results are available in DIRAC.

Fig. 3   Measurement system: Excitation signal from PC

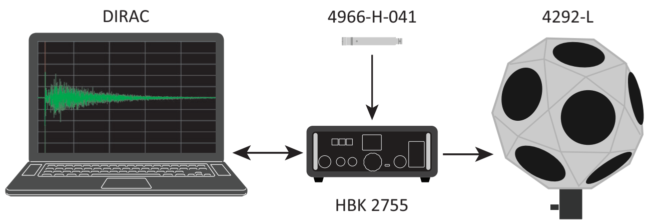
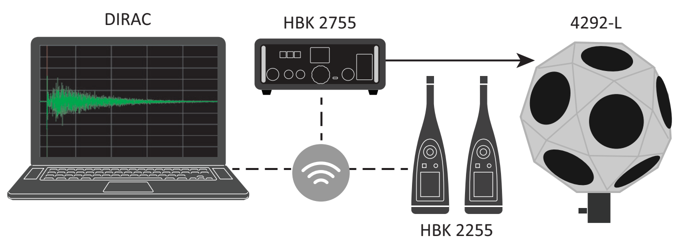


Fig. 2 illustrates the typical use of DIRAC to measure reverberation time and speech intelligibility in an open-plan office. The system components are shown in Fig. 3 and include HBK 2755 Smart Power Amplifier which can be used as a sound device (see The Sound Device on page 4).

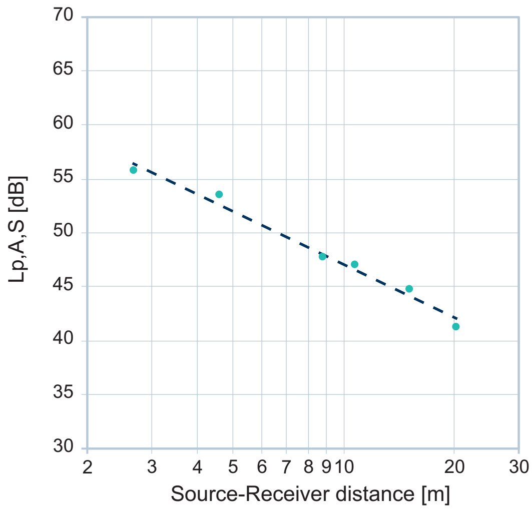
Fig. 4   Measurement system: Wireless setup



If the PC is far from the source or the microphones, a wireless setup becomes useful. Fig. 4 shows a wireless setup where the PC, sound device (HBK 2755) and one or more sound level meters communicate via Wi-Fi®.

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 calculates the A-weighted speech level at 4 m (Lp,A,S,4) and the spatial decay rate of speech ($D_{2,S}$).

Fig. 5 Regression line through the speech sound levels, indicating the spatial 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 comfort distance (r_C).

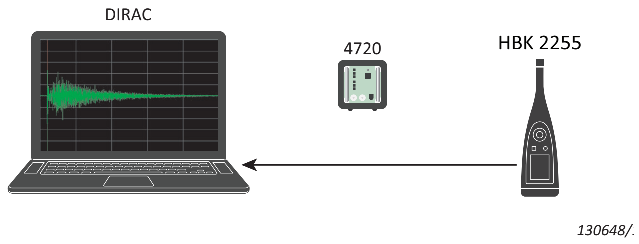
All parameter values detailed in ISO 3382-3 can be viewed in DIRAC and exported at once for use in a report.

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

Speech Intelligibility Measurements

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

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



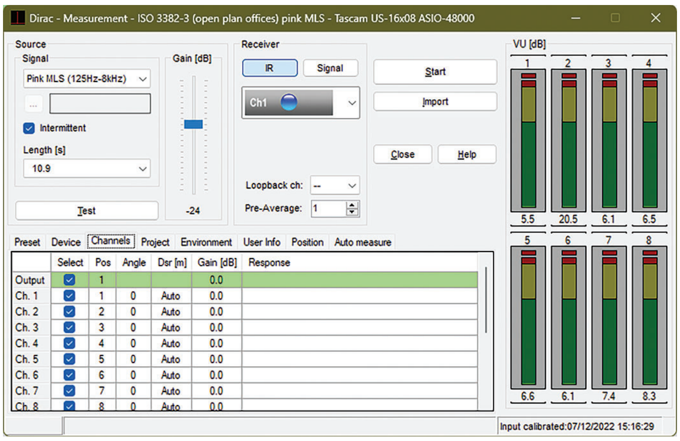
User Interface

Setup

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

In this example, the setup for an ISO 3382-3 measurement is chosen. The intermittent pink MLS (maximum length sequence) excitation signal is selected.

Fig. 7 Typical DIRAC setup, with a 10.9 second MLS excitation



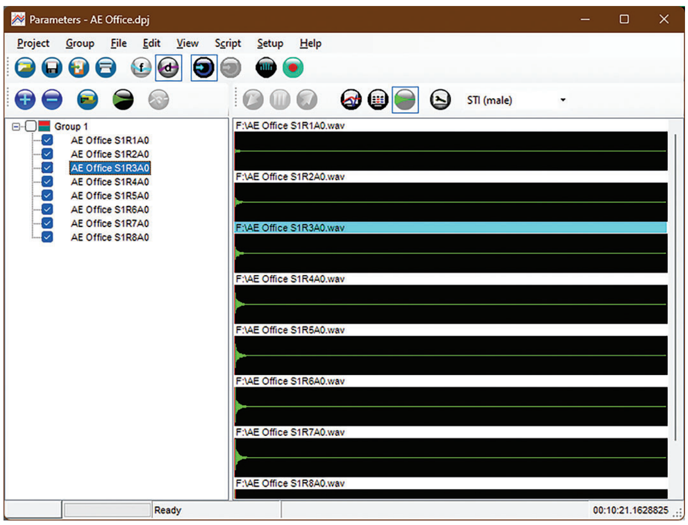
Longer excitation times can be used to suppress background noise. Each doubling of excitation time results in a 3 dB reduction in background noise. In this case, a reverberation time of around 1 second is expected, and an excitation time of 10.9 seconds will improve the impulse response-to-noise ratio (INR) by over 10 dB compared to traditional measurement methods.

Testing

Pressing the Test button runs the MLS signal continuously for adjusting speaker output level and microphone input level.

Pressing the Start button will run the measurement and calculation, and present the impulse responses (Fig. 8). These 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.

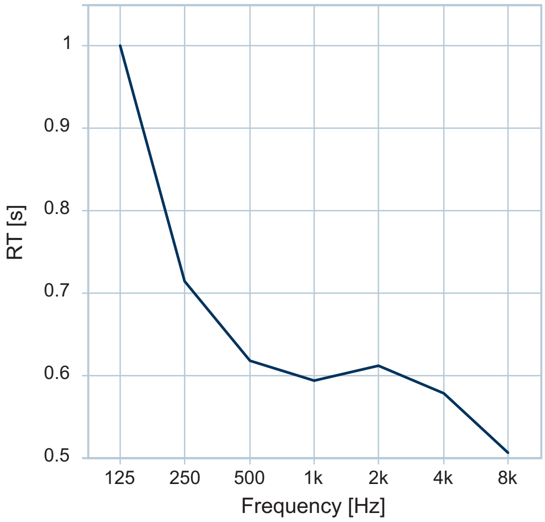
Fig. 8 The measured impulse response – for saving, inspection, listening and calculation of all room acoustic parameters



Results

From the impulse response, any room acoustic parameter can be calculated. Selecting Reverberation calculations shows the averaged 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. 9 Reverberation time spectrum showing values that are long for an office environment



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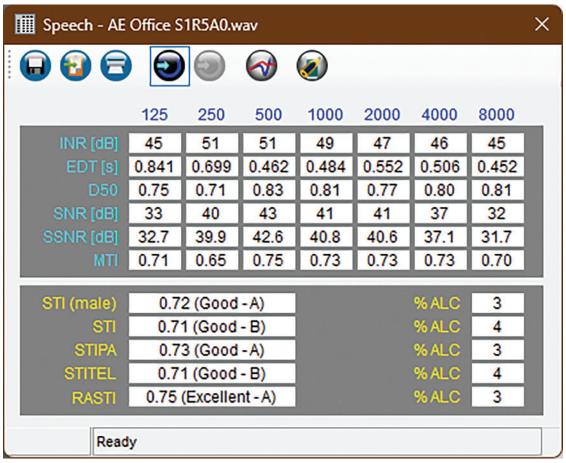
In Speech calculations, Fig. 10 shows 'Good' intelligibility. The maximum possible score is 1.00. The score of around 0.7 is due to the reverberation time of the room (the high signal-to-noise ratio (SNR) values indicate little background noise). The influence of lower SNR values may be simulated by 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 problem spots. Parameter graphs are available to visualize and document this, as in Fig. 20 through Fig. 26.

Fig. 10 Speech calculation, with 'Good' results



	125	250	500	1000	2000	4000	8000
INR [dB]	45	51	51	49	47	46	45
EDT [s]	0.841	0.699	0.462	0.484	0.552	0.506	0.452
D50	0.75	0.71	0.83	0.81	0.77	0.80	0.81
SNR [dB]	33	40	43	41	41	37	32
SSNR [dB]	32.7	39.9	42.6	40.8	40.6	37.1	31.7
MTI	0.71	0.65	0.75	0.73	0.73	0.73	0.70

STI (male)	0.72 (Good - A)	% ALC	3
STI	0.71 (Good - B)	% ALC	4
STIPA	0.73 (Good - A)	% ALC	3
STITEL	0.71 (Good - B)	% ALC	4
RASTI	0.75 (Excellent - A)	% ALC	3

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 sound device line input. Each of these three components can be varied, depending on the type of measurement to be performed.

The Sound Device

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.

A dedicated sound device 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" on page 10.

HBK 2755 Smart Power Amplifier doubles as a sound device that is optimal for DIRAC. With its line, IEPE and P48 inputs, it supports most microphones and it easily drives the OmniPower™ Type 4292-L for insulation measurements.

Fig. 11 USB Audio Interface ZE-0948 is recommended for line level interface to microphone and speaker systems

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Fig. 12 HBK 2755

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The Sound Sources

A loudspeaker sound source can be used instead of an impulsive sound source. The ISO 3382 standard requires an omnidirectional sound source, such as OmniPower Type 4292-L, to measure room acoustical parameters.

Type 4292-L 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.

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 digital audio player. A range of excitation signals comes with DIRAC.

Fig. 13 *The omnidirectional sound source OmniPower Type 4292-L is recommended for ISO 3382 compliant measurements*

Fig. 14 *Echo Speech Source Type 4720 is the preferred source for speech intelligibility measurements*

Measurement Methods

DIRAC supports several impulse response measuring methods that differ by stimulus type, signal source and receiver type.

Stimuli

The available stimuli cover all possible measurement scenarios.

All stimuli can be used in an intermittent mode which includes a background noise measurement.

STIMULUS	USE
MLS	General purpose
Pink MLS	General purpose, improved low-frequency response
Pink MLS (125 Hz to 8 kHz)	ISO 3382-3 and speech intelligibility measurements
Exponential Sweep (125 Hz to 4 kHz)	ISO 3382-1, -2 room acoustics measurements
Exponential Sweep (20 Hz to 20 kHz)	General purpose
Exponential Sweep (20 Hz to f_S/2)	Scale model measurements
Linear Sweep	Electronic systems measurements
User Defined	Stimulus contained in a user-supplied .wav file
Impulse	Measurements using an alarm pistol or whip, for example
ECHO Speech Source	Stimulus generated by Echo Speech Source Type 4720

Signal Sources

The stimulus can either be generated by DIRAC itself and played back through an attached sound device, or it can be generated externally, for instance from a CD player, Echo Speech Source or HBK 2755 Smart Power Amplifier.

When the stimulus playback and the microphone input are both using the same device, with the same sample clock, as is the case for most USB devices, then the measurement is called synchronous. Synchronous measurements always use a DIRAC internally generated signal, and have the advantage that the source-receiver distance can be measured automatically.

Receiver Types

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 which are described below.

RECEIVER TYPE	BUTTON	PARAMETERS
Single omnidirectional microphone	Ch1	All but spaciousness and I
Switchable omni-bidirectional microphone	Ch1	All but IACC and I
Dual omnidirectional microphone	Ch1 Ch2	All but LF, LFC, IACC and I
Omnidirectional and bidirectional microphone	Ch1 Ch2	All but spaciousness and I
Head simulator	Ch1 Ch2	IACC, $IACC_x$
Intensity microphone probe	Ch1 Ch2	All but IACC

Calibration

DIRAC supports three 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 ($L_{eq,I}$).
3. Source-receiver distance calibration: Allows the measurement of the source-receiver distance for synchronous setups, and is usually performed using a loop-back cable.

For ISO 3382-3 measurements, all three 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.

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

Fig. 15 *Time domain views: Original impulse response and energy-time curve from a single-channel measurement*

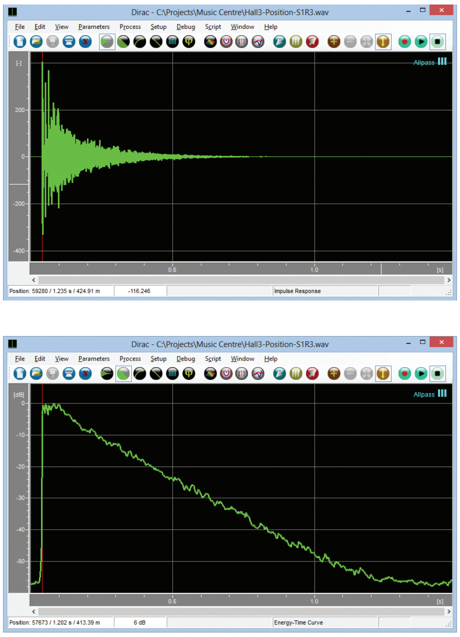


Fig. 16 *Time domain views: Comparing two single-channel impulse responses and their forward integration curves*

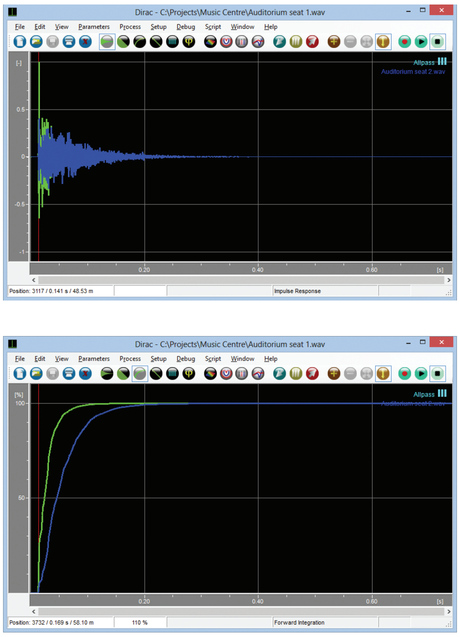
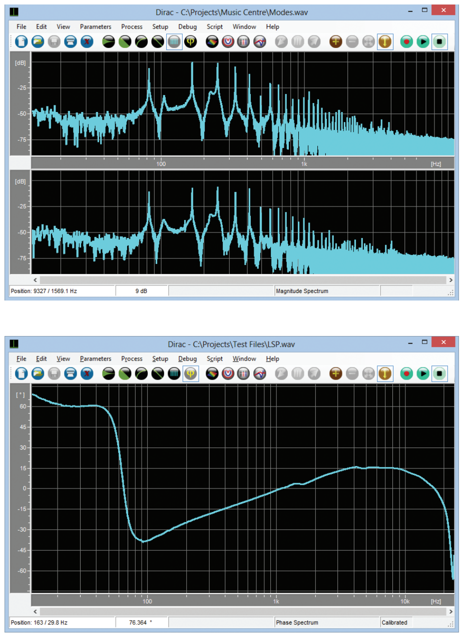


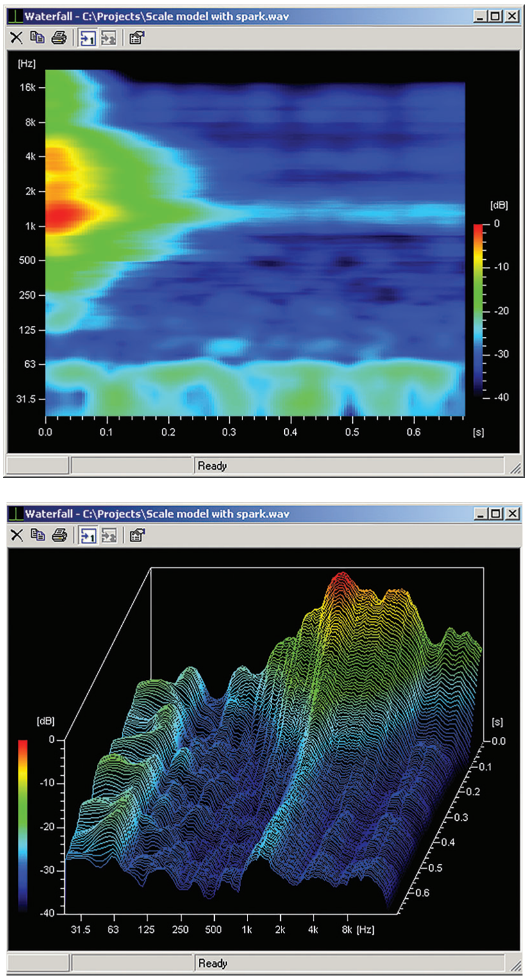
Fig. 17 *Frequency domain views: Linear FFT and phase spectra*



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.

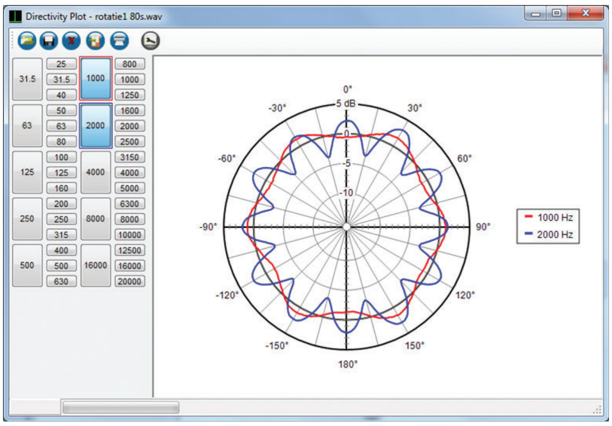
Fig. 18 Energy-time-frequency plots: Spectrogram and waterfall



Directivity Plot

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*.

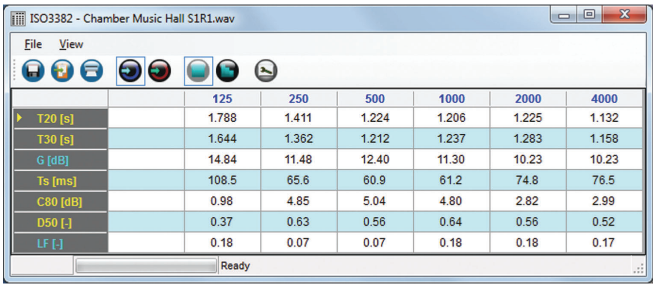
Fig. 19 Directivity plot: The directivity of a sound source in two different frequency bands



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. 20 Parameter tables can be customized, 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. 21 Parameter graph showing magnitude spectrum

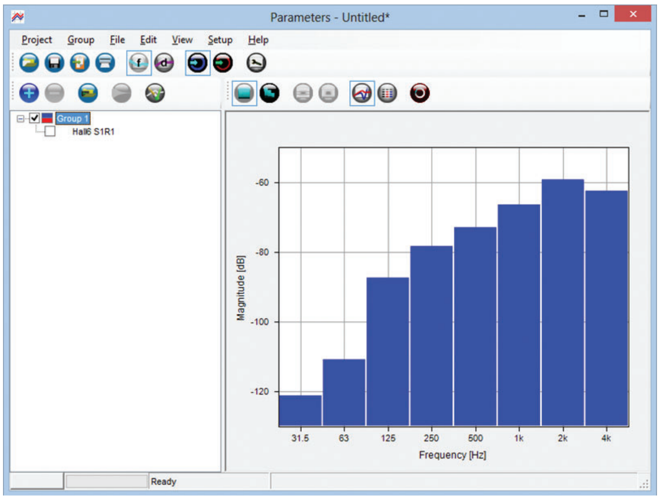


Fig. 22 Parameter graph showing D_{50} average and standard deviation over four receiver positions

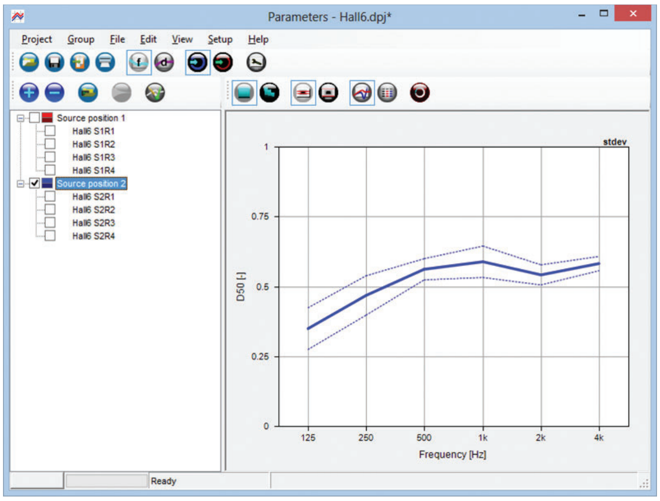


Fig. 23 *Parameter graph showing D_{50} average over four receiver positions, for two different source positions*

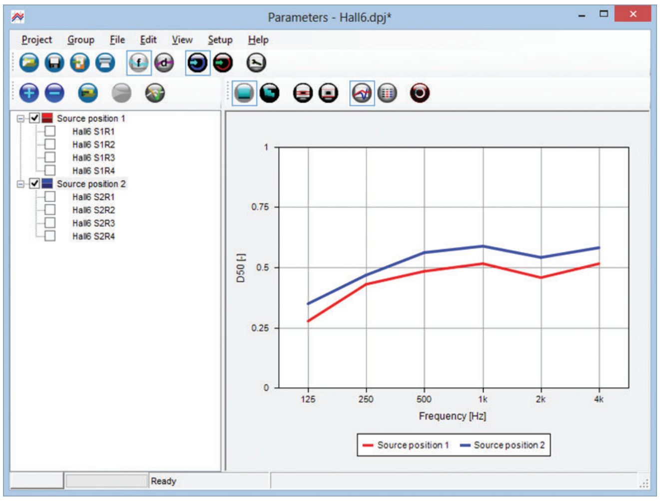


Fig. 24 *Parameter graph showing STI for different receiver positions plotted against the source-receiver distance*

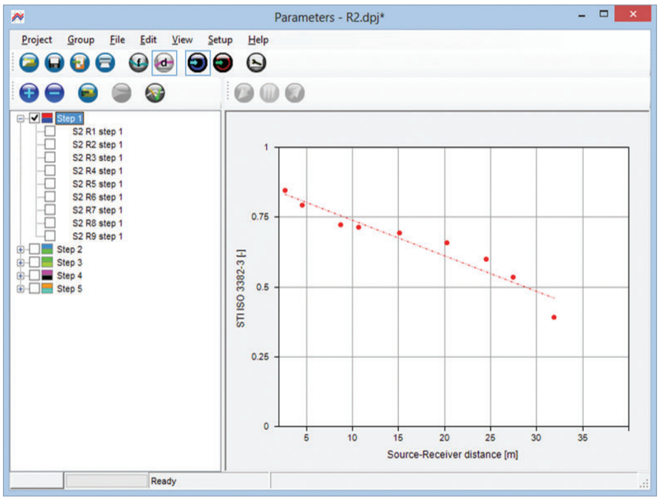
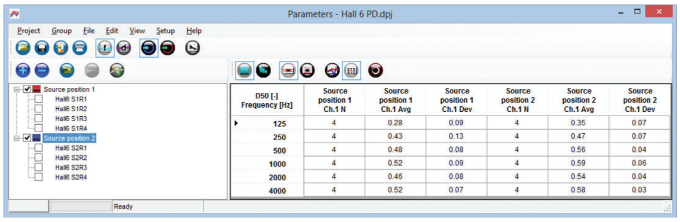


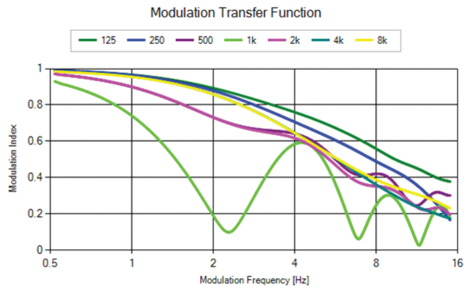
Fig. 25 *D_{50} table: Average and standard deviation over four measurement positions for two source position groups and two channels per measurement. For each frequency, the number of usable results is given*



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

The MTF graph can be used to investigate the causes for low modulation transmission index (MTI) values. A constant MTI over frequency indicates background noise, a continuously decreasing MTF indicates reverberation and an MTI first decreasing and then increasing with frequency indicates an echo (Fig. 26).

Fig. 26 *MTF graph showing MTI as a function of the modulation frequency*



Real-time Analysis

The RTA window, which can be opened for each input channel, gives a real-time view of the signal spectrum. The signal spectrum can be displayed in different modes (octave, third-octave and continuous) and allows a comparison with a reference spectrum.

Other Applications

Scale Model Measurement

To predict the acoustics of, for instance, a concert hall that is being designed but not yet realized, 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.

Fig. 27 *Measurement in a scale model of a reverberation chamber, using a miniature omni-directional sound source*

STANDARDS

Conforms with the following:
IEC 61260: Electroacoustics – Octave-band and Fractional-octave-band Filters
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
EN 1793: Road traffic noise reducing devices – Test method for determining the acoustic performance – Parts 4, 5 and 6

OPERATION

The software is 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

IR (impulse response) and signal measurements
Internal or External: MLS, e-sweep, lin-sweep, 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
Pre-trigger: 0 to 1000 ms
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 and treble ratio (based on level), BR(L) and 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, late and total support, ST_{early}, ST_{late} and ST_{total}
- User-defined support, ST
- Modulation transfer index, MTI
- Speech transmission index (male and female), STI
- User-defined speech transmission index, STI_{user}
- 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, $L_{p,A,S,4}$
- A-weighted background noise level, $L_{p,A,B}$
- Spatial decay rate of speech, $D_{2,S}$
- Comfort distance, r_C
- Distraction distance, r_D
- Privacy distance, r_P
- Reflection index, RI
- Sound insulation index, SI
- Sound power reflection factor, Q_W
- Reduction factor, R_{sub}
- Harmonic distortion, THD and THD+N
- Spurious-free dynamic range, SFDR
- Effective number of bits, ENOB
- Intermodulation distortion, IMD
- Crest factor, CF

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

Source-receiver Distance Calibration: Measures the sound propagation delay between signal output and input or loudspeaker and microwave
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

REAL-TIME ANALYSIS

SPL graph display in octaves, third-octaves or continuous. Octaves and third-octaves also in table format.
Spectrum Freeze, Max-hold and Reference graphs.

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 colouring

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 software 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
- Multi-channel Wave file import
- 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® 11, 10, 8, 7 or Windows Vista®
CPU: Minimum 1 GHz
RAM: 2 GB
Free Disk Space: Minimum 500 MB
Auxiliary Hardware: Mouse or other pointing device
Sound Device: N channels, full duplex, 22.05, 44.1, 48, 65, 96 or 192 kHz sample rate, support for Core Audio/WASAPI or ASIO

Ordering Information

Type 7841-X* **DIRAC Room Acoustics Software**
Type 7841-XU* **Upgrade from DIRAC Version 6 to Version 7**
Software delivered on a removable USB storage device

OPTIONAL ACCESSORIES
2255-N-S HBK 2255 Sound Level Meter with Noise Partner†

BZ-7451 Analysis-quality Audio Licence for HBK 2255

HBK 2755 Smart Power Amplifier

Type 4292-L OmniPower Sound Source
which includes:
- KE-0462: Carrying Bag for Type 4292-L
- UA-1690: Tripod

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

ZE-0948 USB Audio Interface
which includes:
- Leather Pouch KE-0456
- USB Cable Assembly AO-0708
- 2 × RCA Phono Cable AO-0707
- 2 × RCA Phono to BNC Adaptor JP-0071
- 1 × RCA Phono to ¼″ Jack Adaptor JP-0072

Type 4955-H-041 ½″ Free-field Microphone including Preamplifier Type 1706

Configuration Examples

Connect one or more HBK 2255 Sound Level Meters as wireless microphones (only requires BZ-7451 installed on HBK 2255) to create a wireless DIRAC configuration.

SIMPLE CONFIGURATION
2255-N-S HBK 2255 Sound Level Meter with Noise Partner
FW-2255-000 Standard firmware for HBK 2255 (general, type-approved)
BZ-7451 Analysis-quality Audio Licence for HBK 2255
Type 7841-N DIRAC Room Acoustics Software, node-locked

COMPLETE ROOM ACOUSTICS MEASUREMENT SYSTEM
For users who also perform sound insulation measurements, add BZ-7451 and DIRAC to a building acoustics kit to form a complete building and room acoustics measurement system.
2255-B-K01 HBK 2255 Building Acoustics Kit
which includes:
- HBK 2255 Sound Level Meter with Building Acoustics Partner
- HBK 2755 Smart Power Amplifier
- OmniPower Sound Source Type 4292-L
- Convenient carrying bags

BZ-7451 Analysis-quality Audio Licence for HBK 2255
Type 7841-N DIRAC Room Acoustics Software, node-locked

Please note: For information on sound sources, please go to www.bksv.com.

* Where ‘X’ = ‘N’ for node-locked or ‘F’ for floating license
† Recommended firmware: FW-2255-000, general type-approved firmware (standard)

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