

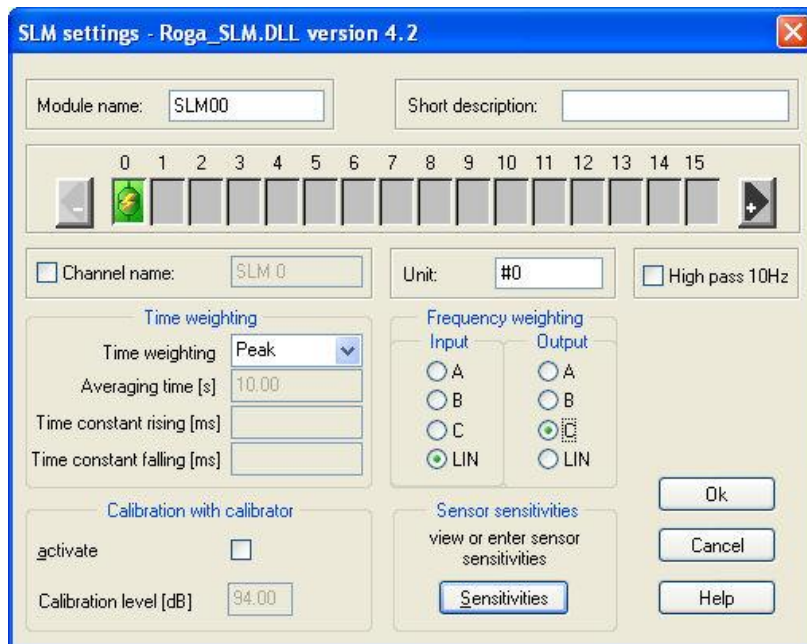
ROGA Instruments SLM and SPM Module Manual

With these DASyLab Add-On modules you may determine sound power levels easy and according to standards. These modules share the following tasks:

The SLM module (Sound Level Measurement) determines a sound pressure level in dB from a time signal (should be a microphone signal in most cases).

The SPM module (Sound Power Measurement) determines the sound power from some sound pressure levels regarding all correction terms necessary.

SLM Module



Inputs

The SLM-module has 1 to 16 inputs, which may be switched on and off by the ,+‘ - and ,-‘ - buttons. The Inputs expect time signals coming from microphone inputs having a scan rate of some kHz. If the Scan rate is too low, the time weightings and frequency weightings cannot be calculated exactly.

With scan rates below 100 Hz a warning message is displayed, because correct time weightings cannot be calculated exactly.

With scan rates below 30 kHz a warning message is displayed, because correct time frequency cannot be calculated exactly.

Outputs

The SLM-module has one output for each input. With at output rate of approximately 20 ms the level in dB of the appertaining input signal is calculated.

Time weightings

The following time weightings may be selected (in dialog in the combo box ,Time weighting' :

FAST	exponential decreasing weighting of past levels with of a time constant of 125 ms
SLOW	exponential decreasing weighting of past levels with of a time constant of 1000 ms
Impulse	exponential decreasing weighting of past levels with of a time constant of 35 ms for increasing and 1500 ms for decreasing levels
Leq	equivalent continuous sound pressure level. Even weighting of the levels in the time window specified (in dialog in input field ,Averaging time [s]' in seconds).
Peak	absolute maximum of instantaneous value of sound pressure.
User defined	if 'user defined' is selected, you may specify the time constants for increasing signals ('Time constant rising') and decreasing signals ('Time constant falling'). I.E. if you specify 125 ms for 'Time constant rising' and 125 ms for 'Time constant falling' the result is the same as the time weighting FAST.

Frequency weightings

The SLM-module is able to calculate frequency weightings A, B, C and LINEAR according to IEC 651.

The accuracy depends on the sampling frequency of the input signal:

Scan rate of input signal	Accuracy grade redeemed
< 30 kHz	Not recommended
30 kHz	Grade 0 up to 5 kHz input signal frequency Grade 1 up to 6,3 kHz input signal frequency
40 kHz .. 80 kHz	Grade 0 up to 12,5 kHz input signal frequency Grade 1 full frequency range
>= 80 kHz	Grade 0 full frequency range

Input frequency weighting

Present frequency weighting of the input signal.

A	frequency weighting A according to IEC 651 & IEC 61672-1:2013
B	frequency weighting B according to IEC 651 & IEC 61672-1:2013
C	frequency weighting C according to IEC 651 & IEC 61672-1:2013
LIN - Z	frequency weighting LINEAR according to IEC 651 & IEC 61672-1:2013

Output frequency weighting

Desired frequency weighting of the sound level. Please note, that not all combinations of input frequency weighting and output frequency weighting are possible.

A	frequency weighting A according to IEC 651 & IEC 61672-1:2013
B	frequency weighting B according to IEC 651 & IEC 61672-1:2013
C	frequency weighting C according to IEC 651 & IEC 61672-1:2013
LIN Z	frequency weighting LINEAR according to IEC 651 & IEC 61672-1:2013

Please note that the dynamic range especially with low frequencies depends on the position of the weighting in the signal flow, whether the frequency weighting is done before or after the ADC (Analog/Digital-Converter).

An example:

You have got a noise signal with portions of 100 dB at 20 Hz and 30 dB at 1 kHz and you need A-weighted level (dB_A), the ADC has a full scale of 60 dB.

1. A-weighting filter before the ADC:

The 20 Hz-Signal is damped by 50,5 dB to 49,5 dB, the 1 kHz Signal remains constant. The sum is below 60 dB and may be acquired correctly by the ADC.

The measurement may be done.

2. A-weighting filter after the ADC:

The 20 Hz-Signal with 100 dB results in overrange for the ADC.

The Measurement cannot be done.

To take the measurement nevertheless, the full scale must be adjusted, so that the ADC can handle 100 dB. The 1 kHz portion with 30 dB-Signal is 70 dB below full scale and will be distorted by background noise. Especially, if you need A-weighting, and there are big portions at low frequencies, a hardware A-weighting before the ADC is strongly recommended.

High Pass 10 Hz

To suppress low frequency noise a high pass filter is provided. It is a two pole butterworth filter with a cut off of 10 Hz. If you check the checkbox, the filter is used, else not.

Calibration

To allow the display of noise levels in dB, the module has to be calibrated.

There are two methods to calibrate the channels of the module

Calibration using a calibrator

Check the checkbox 'activate' in the group box 'Calibration with calibrator', enter the level of your calibrator and start a measurement.

A dialog for monitoring the status of calibration is displayed (SLM calibration'). If more than one SLM-modules are placed on the schematic you have to perform calibration for each of them separately.

If you plug on a calibrator to one of the microphones, the level of this microphone remains constant for a while (Display ,Level is constant xx % with xx of 0 .. 100) and using this level and the level of the calibrator given, the calibration difference is calculated and adjusted (display ,calibration value is taken over' and a calibration value in the column ,new Value'). The calibration for this channel is finished and the calibrator may be plugged onto the next microphone until you get the display, calibration value is taken over' for all channels.

SLM - Calibration

monitoring module: SLM00

Manual input of microphone sensitivities: Please enter microphone sensitivities in mV/Pa in the appropriate fields 'manual input' and click 'Apply manual input'.

Calibration using a calibrator: Please plug the microphones to be calibrated on the calibrator. The microphones receiving the calibration signal will be detected automatically.

If for all channels to be calibrated a calibration value is taken over, please stop the calibration measurement.

Channel	Sensitivity [mV/Pa]		Channel status	Manual input
	Old value	New value		
Channel 0	50.00	--	level is constant (50 %)	
Channel 1	--	--	--	
Channel 2	--	--	--	
Channel 3	--	--	--	
Channel 4	--	--	--	
Channel 5	--	--	--	
Channel 6	--	--	--	
Channel 7	--	--	--	
Channel 8	--	--	--	
Channel 9	--	--	--	
Channel 10	--	--	--	
Channel 11	--	--	--	
Channel 12	--	--	--	
Channel 13	--	--	--	
Channel 14	--	--	--	
Channel 15	--	--	--	

Apply manual input

The order you calibrate the microphones does not matter. The microphone with calibrator plugged on is detected automatically by the constant level.

For microphones, without calibrator the input level varies (display ,level is changing') and calibration is done for these channels.

Direct input of microphone sensitivities

Click the button 'Sensitivities' in the group box 'Sensor sensitivities'. The calibration dialogue will be displayed where you may view and enter the microphone sensitivities.

SLM - Calibration

monitoring module

Manual input of microphone sensitivities: Please enter microphone sensitivities in mV/Pa in the column 'manual input' and click 'Apply manual input'.

If all microphone sensitivities entered are valid, close this dialogue clicking 'OK' else by clicking 'Cancel'.

Channel	Sensitivity [mV/Pa]		Channel status	Manual input
	Old value	New value		
Channel 0	50.00			54,3
Channel 1				
Channel 2				
Channel 3				
Channel 4				
Channel 5				
Channel 6				
Channel 7				
Channel 8				
Channel 9				
Channel 10				
Channel 11				
Channel 12				
Channel 13				
Channel 14				
Channel 15				

Enter the microphone sensitivities in the column 'Manual input' and click 'Apply manual input'.

SLM - Calibration

monitoring module

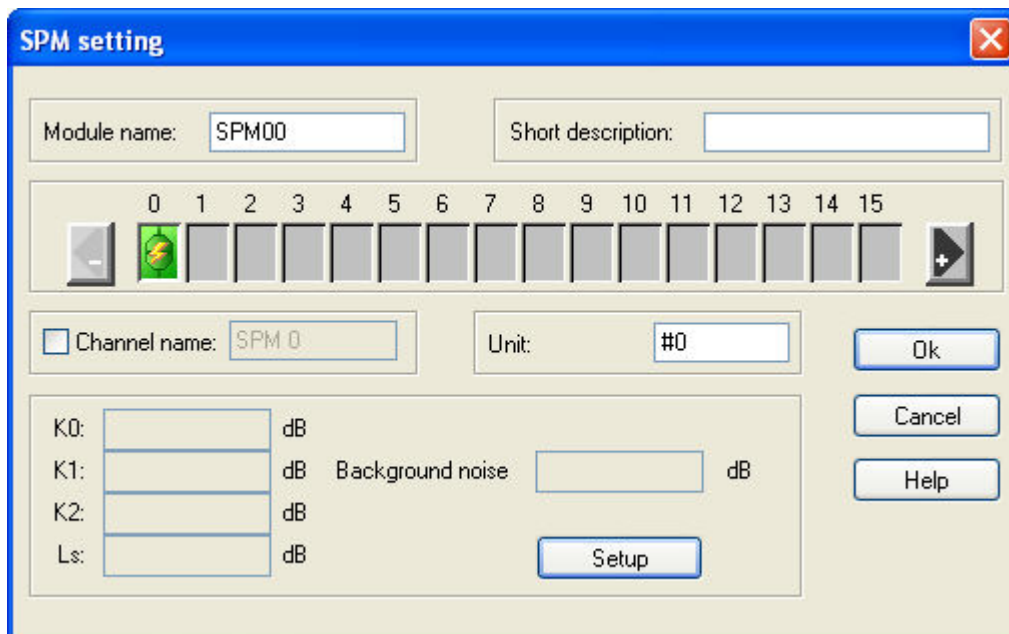
Manual input of microphone sensitivities: Please enter microphone sensitivities in mV/Pa in the column 'manual input' and click 'Apply manual input'.

If all microphone sensitivities entered are valid, close this dialogue clicking 'OK' else by clicking 'Cancel'.

Channel	Sensitivity [mV/Pa]		Channel status	Manual input
	Old value	New value		
Channel 0	50.00			54,3
Channel 1				
Channel 2				
Channel 3				
Channel 4				
Channel 5				
Channel 6				
Channel 7				
Channel 8				
Channel 9				
Channel 10				
Channel 11				
Channel 12				
Channel 13				
Channel 14				
Channel 15				

SPM Module

The SPM-module (Sound Power Measurement) determines the sound power from some sound pressure levels regarding all correction terms necessary.



Inputs

The SPM-module has 1 to 16 inputs which may be switched on and off by the ,+' - and ,-' - buttons. The inputs expect levels in dB (normally coming from SLM-modules).

Output

The SPM module has one output for the sound power level.

Correction terms

For the determination of sound power according to standards, correction terms have to be regarded:

- K0 correction term for barometric pressure and temperature, see DIN 45 635, paragraph 7.1.4.
- K1 correction term for background noise, see DIN 45 635, paragraph 7.1.3.
- K2 correction term for environmental influence, see DIN 45 635, paragraph 7.1.4.
- Ls correction term for the size of the enveloping surface, see DIN 45 635, paragraphs 6.4., 7.2.

Correction term for barometric pressure and temperature K0

- Correction term for barometric pressure and temperature, see DIN 45 635, paragraph 7.1.4.

Enter the temperature in input field ,Temperature' and the barometric pressure in the input field ,Barometric Pressure'. The correction term is displayed in the field ,K0 Setting'.

According to DIN 45 635, for accuracy grade 2 K0 is not necessary, in the standards ISO 374x it is not mentioned at all. Therefore you may select to use K0 or not for calculation (checkbox ,Use K0').

The screenshot shows a 'Settings' dialog box with a tabbed interface. The 'K0' tab is selected. It contains three input fields: 'K0 Setting' (with 'dB' to its right), 'Temperature' (with '°C' to its right), and 'Barometric pressure' (with 'hPa' to its right). A checkbox labeled 'Use K0' is located to the right of the 'Barometric pressure' field. At the bottom right of the dialog are 'OK' and 'Cancel' buttons.

Correction term for background noise K1

Correction term for background noise, see DIN 45 635, paragraph 7.1.3.

Take a measurement with candidate switched off. Then you may declare this sound pressures to be the background noise (button 'Set background noise to last measurement'), or enter the enveloping surface sound pressure level (= sound power level - L_s) of the background noise directly (input field 'Background noise').

Please note, that the measurement of background noise has to be taken with the same frequency weighting as the following measurement.

The actual value of K1 depends on the signal to background noise ratio and is calculated online during measurement. If the energetic sum of information signal and background noise is less than 3 dB above the background noise, correction term K1 cannot be calculated and the output of the module is set to -1000.0 dB.

The screenshot shows a 'Settings' dialog box with a tabbed interface. The 'K1' tab is selected. It contains two input fields: 'K1 Setting' (with 'dB' to its right) and 'Background noise' (with 'dB' to its right). A button labeled 'Set background noise to last measurement' is located to the right of the 'Background noise' field. At the bottom right of the dialog are 'OK' and 'Cancel' buttons.

Correction term for environmental influence K2

Correction term for environmental influence, see DIN 45 635, paragraph 7.1.4. You may specify the environmental influence by two ways:

Direct input

Enter K2 directly in dB in input field ‚K2 Setting‘.

Calculation of K2 via properties of the measuring room

Enter the dimensions (height, width and depth in the input fields ‚Height‘, ‚Width‘ and ‚Depth‘) and the mean absorption coefficient (input field ‚Mean absorption grade‘) or reverberation time of test cage (input field ‚Reverberation time‘) of the test cage.

Please note, that you have to specify the correction term for the size of the enveloping surface L_s prior to evaluation of K2

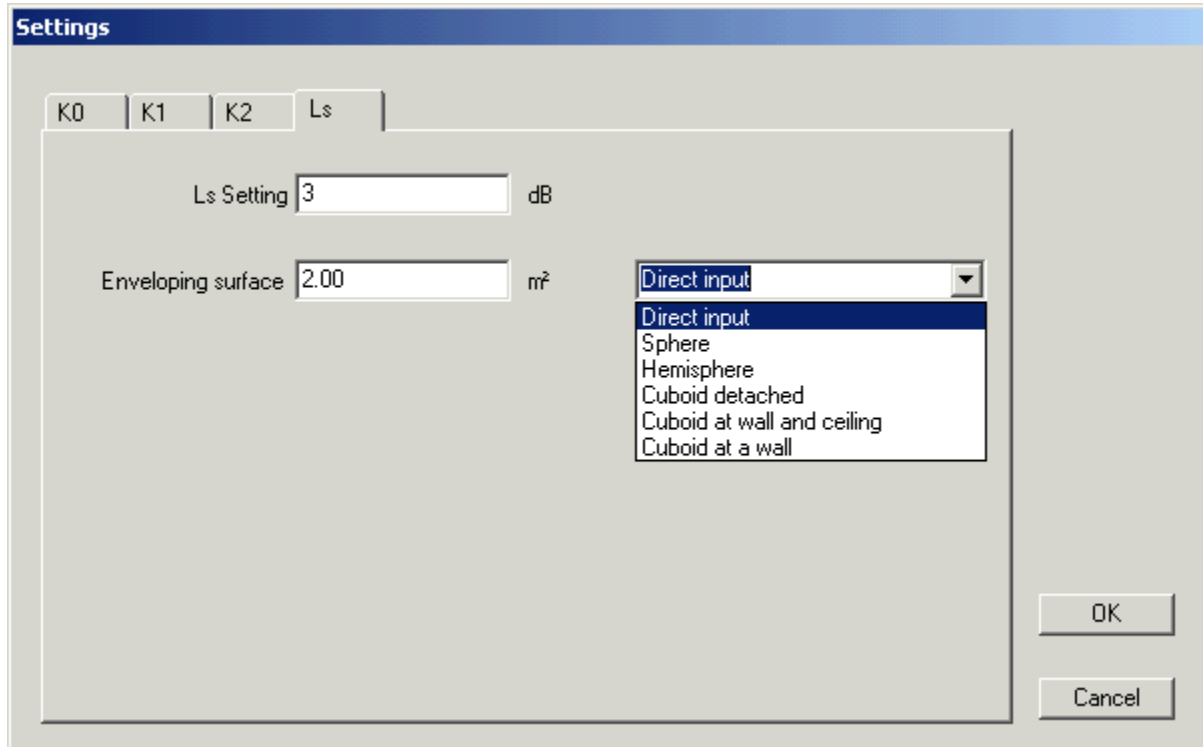
The screenshot shows a software window titled "Settings" with a tabbed interface. The "K2" tab is selected. The window contains several input fields with labels and units:

- K2 Setting**: Input field with unit "dB".
- Mean absorption grade**: Input field with unit "m²".
- Reverberation time**: Input field with unit "s".
- Height**: Input field with unit "m".
- Width**: Input field with unit "m".
- Depth**: Input field with unit "m".

At the bottom right of the dialog box, there are two buttons: "OK" and "Cancel".

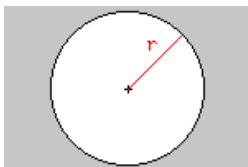
Correction term for the size of the enveloping surface L_s

Correction term for the size of the enveloping surface, see DIN 45 635, paragraphs 6.4., 7.2. You may enter the ratio of the enveloping surface to 1 m² directly in dB (input field ‚ L_s Setting‘) or the enveloping surface in square meters (input field ‚Enveloping surface‘, selection ‚Direct Input‘).



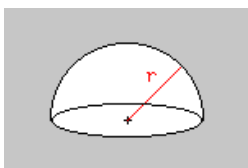
You may also specify the enveloping surface by its shape and dimensions:

Sphere



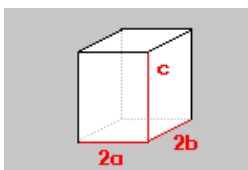
For the calculation the radius must be known.

Hemisphere



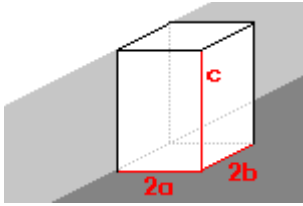
For the calculation the radius must be known

Cuboid detached



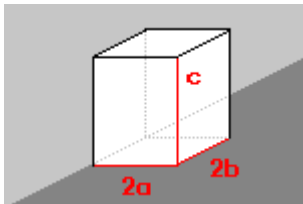
For the calculation the sides 2a, c and 2b must be known.

Cuboid at wall and ceiling



For the calculation the sides 2a, c and 2b must be known.

Cuboid at a wall



For the calculation the sides 2a, c and 2b must be known.

