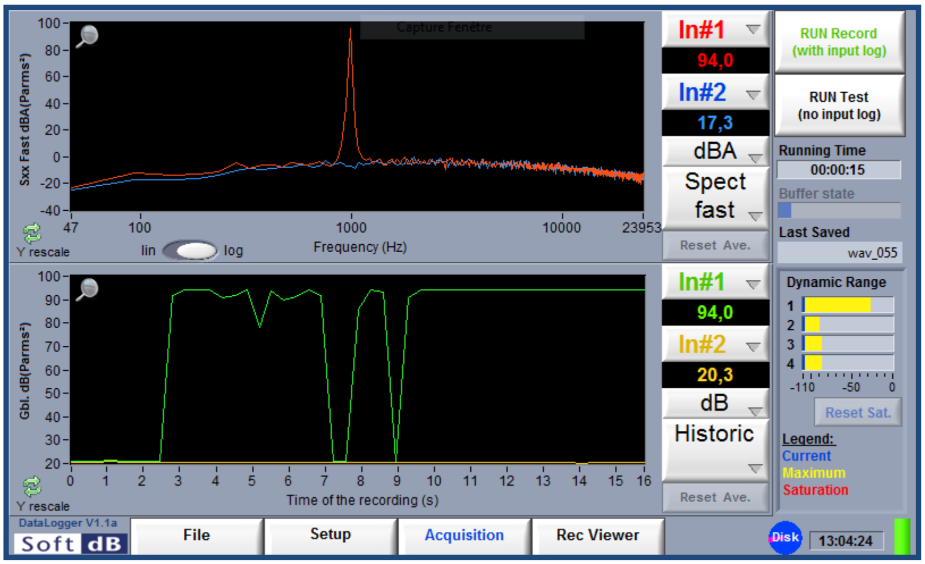
Opus Suite

Data Logger Module

User Guide – v1.2
2014-01-21



Compatible Hardware:

Alto

Concerto

Conductor

Soft dB Inc.
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Quebec (Quebec) Canada G1S 3G3
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Soft dB
WWW.SOFTDB.COM

CONTENTS

1 Introduction

Congratulations on your purchase of the ***Opus Suite Data Logger module***.

The ***Opus Software Suite*** is a sound and vibration software that contains several measurement modules:

- SLM 4-ch module : 4-channels, Class 1 (IEC 61672 and ANSI S1.43)
- SLM & 3Vib module : 1 SLM channel (same as SLM 4-ch module) and 3 vibration channels (ISO 8041 and ISO 2631)
- **Data Logger module**
- Building Acoustics Suite
  - Sound Transmission (ASTM E 336/ISO 140-4)
  - Impact Insulation (ASTM E 1007/ISO 140-7)
  - Room noise (ANSI/ASA S12.2-2008)
  - Reverberation Time (ISO 3382)
  - Speech Privacy (ASTM E 2638 and ASTM E 1130)
- Building Vibration module (DIN 45669-1 and ANSI S2.46)
- Intensity module (IEC 1043)
- Hammer Impact module
- Power Transformer Suite

The ***Opus Suite*** is intended to run on a ***Concerto.*** The software can also be installed on a Conductor unit or on any PC if using an Alto unit. Moreover, some post-processing functions are available on a PC even if no compatible unit is detected.

The current user's manual presents the **Data Logger Module**.

General Specifications

4 analog inputs	Real-time acquisition and recording into .wav file format Sampling rate from 4 kHz to 48 kHz 2 input dynamic ranges: +/-1.5 V and +/-6 V[1] 24 bit resolution, 100 dB signal-to-noise ratio 3 input types: Direct DC, Direct AC and ICP
2 analog output	Multi-Tone, White Noise and Pink Noise generator mixable on the 2 outputs Output dynamic range: +/-2 V
Real-time display	Time Signal, Time History, FFT spectrum (instant, exponential fast, exponential slow and average). 1 or 2 graphs display with 2 plots per graph.

[1] In a Concerto unit, the range of inputs 1 and 2 are selectable but the range of inputs 3 and 4 are fixed at +/-1.5 V.

2 Compatible Hardware

Every hardware option has an embedded state of the art Soft dB SR-MK3 DSP board allowing real-time and precise measurement with very low energy consumption.

Concerto

Handy, lightweight, fully rugged military tablet
All in one instrument
WLAN communication allows using the Concerto as a monitoring station with remote access.
http://www.softdb.com/en/acoustic/products/concerto.php

Alto

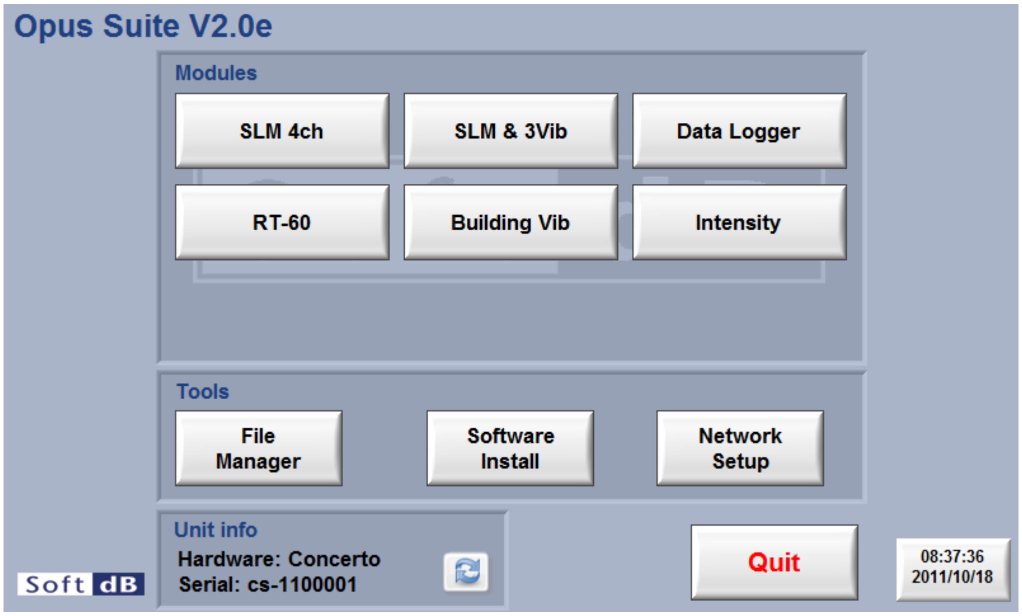
6 or 4 24-Bit asynchronous inputs and 2 outputs
Compact, low-consumption, and flexible
Needs to be connected to a PC.
Competitive price.
http://www.softdb.com/en/acoustic/products/alto.php

Conductor

Rugged platform for acoustics and vibration measurements.
Mainly used for the I-Track sound intensity mapping system.
http://www.softdb.com/en/acoustic/products/conductor.php

3 Opus Environment

The ***Concerto*** unit comes equipped with the **Opus** Environment. This environment acts as a main interface that gives access to the different modules and tools.



Modules	The modules buttons will launch the associated module. When a module is opened, a license verification check is done. If no license is found for that module, a message will indicate the limitations.
File Manager	The File Manger button will launch the File Manager Utility (see section 10, p. 50)
Software Install	The Software Install button will launch a browser from which an Opus software installer can be launched.
Network Setup	The Network Setup button will close the Opus software and access Windows. Then, the network can be set through Windows.
Unit info Hardware: Concerto Serial: cs-1100001	The Unit info gives the information about the hardware type (Concerto, Alto or Conductor) and the serial number of the unit. The refresh button allows resetting the connection with the acquisition board (useful with an Alto unit).
08:37:36 2011/10/18	The Clock indicator displays the time and date on the unit. To change time, simply click on the indicator to display a dialog window.

<table>
<tr>
<td>
</td>
<td>The Quit button will quit the application differently according to the hardware used.
Concerto hardware:
Hold 5 sec to shut down the unit.
Press and release to enter standby mode.
Alto or Conductor hardware:
Press and release to close the application and return to Windows.</td>
</tr>
</table>

4 Software Mode

The Data Logger software will run in a different mode depending of the hardware it is running on. Depending of hardware used, the look and functionalities will be slightly different at launch. The following table enumerates the differences. Unless otherwise noted, this document relates to the software used with the Concerto hardware.

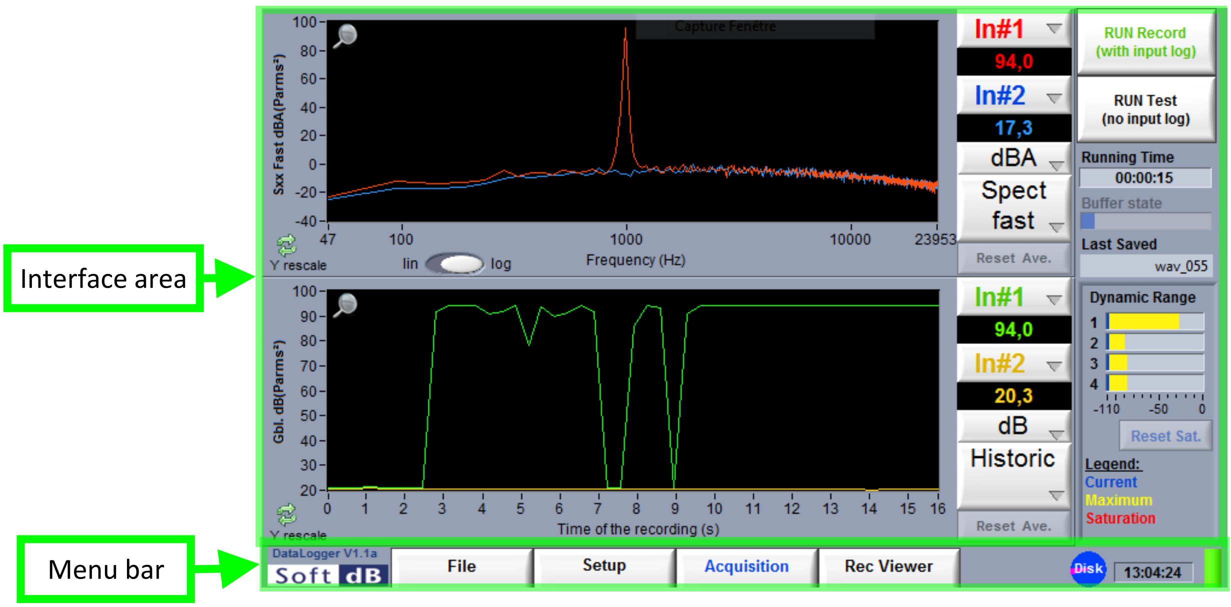
	Concerto hardware	**Other hardware: Conductor, Alto**
Panel Size	800x480 pixels	800x600 pixels
Position of the menu bar	Bottom of panel	Top of panel
Number of inputs managed	4	6
Touch screen functions	Yes	No
Output generation	Embedded generator	Embedded generator or wave file
Post-Processing	Limited (Rec Viewer & Playback)2	Full

[2] Post-Processing functions are disabled on the Concerto hardware because the unit is intended for acquisition. The advanced post-analysis of the acquired data should take place on a PC.

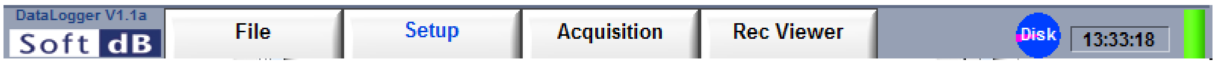
5 Main Interface

The main interface is separated in two areas: the Interface area and the menu bar at the bottom.

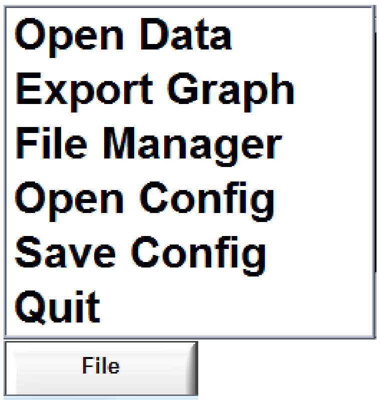
The menu bar mainly contains the controls that allow the user to navigate through the application (see section 5.1, p.10 for details). The interface area will host the three interfaces: **Setup** (section 0, p.11), **Acquisition** (section 7, p.17) and **Rec Viewer** (section 8, p.23)**.** At launch, the application opens with the **Setup Interface** (**Acquisition Interface** shown below).



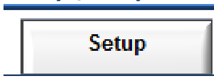
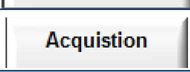
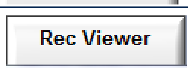
5.1 Menu Bar



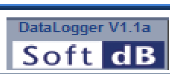
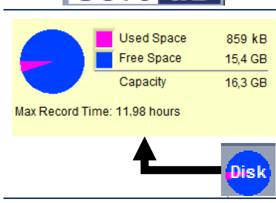
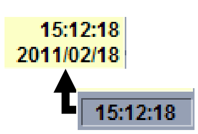
File Menu

	**Open Data**	Allows opening a **.wav data file**. This launches the Record Viewer Interface (section 8, p.23).
	Export Graph	Allows exporting graphs in a spreadsheet file. Available only in the **Acquisition Interface** (section 7.5, p.22) and the **Rec Viewer Interface** (see section 8.5, p. 27)
	File Manager	Launches the **File Manager** (see section 10, p. 50)
	Open Config	Allows restoring a saved software configuration.
	Save Config	Allows saving the software configuration.
	Quit:	Allows to quit the **Data Logger module** and to return to the **Opus Suite Interface**.

Setup / Acquisition / Rec Viewer Buttons

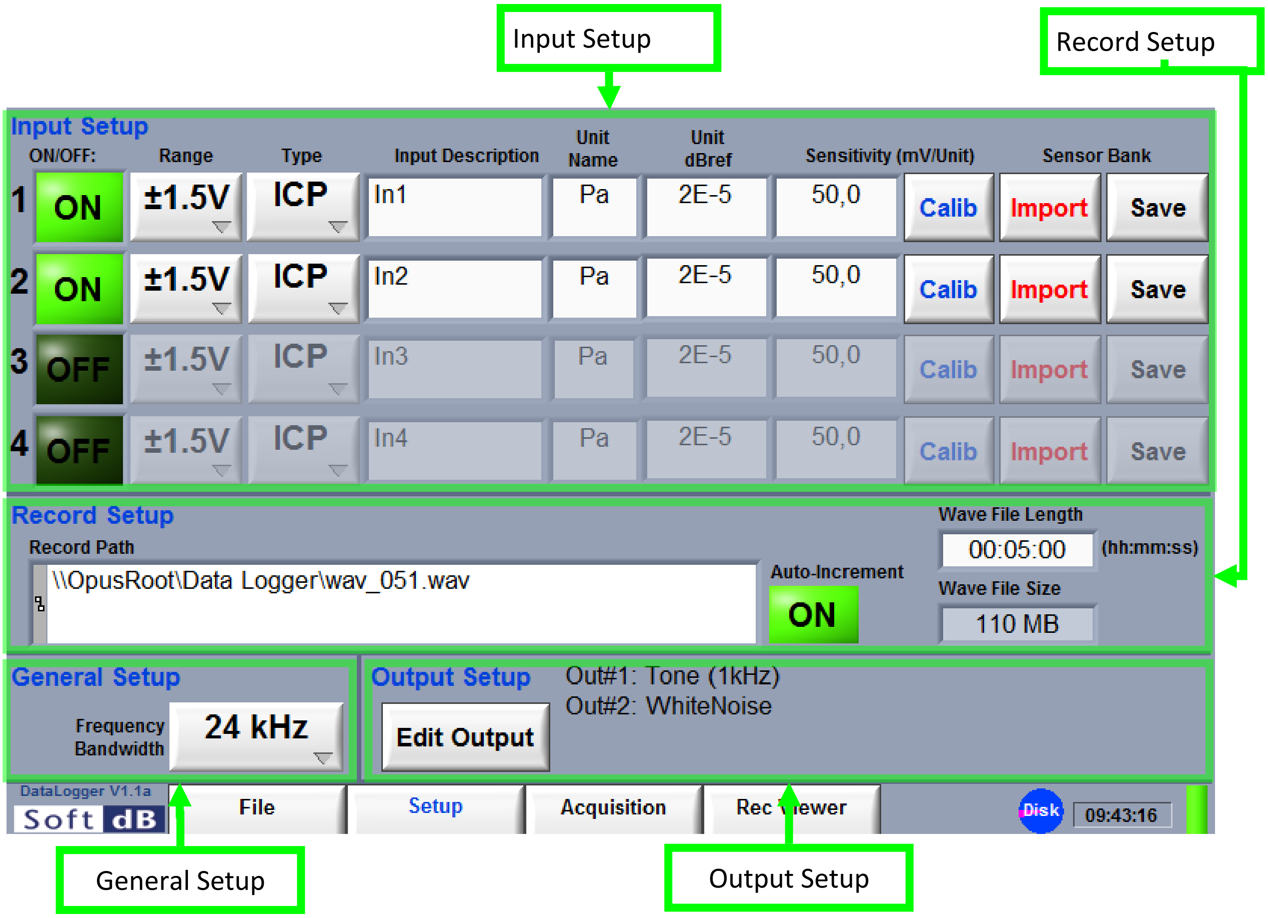
	Launches the **Setup Interface** (see section 0, p.11)
	Launches the **Acquisition Interface** (see section 7, p.17)
	Launches the Rec Viewer Interface (see section 8, p.23)

Other Menu Mar Indicators

	Software name and version.
	**Disk memory usage**. Point the indicator to pop up the disk info window that gives more details about the disk memory usage. The estimated maximum record time is also displayed. This estimation obviously depends on the acquisition setup.
	**Clock** indicator. Point the indicator to display the full date. In order to adjust the date/time of the clock on a Concerto, quit this module and proceed through the **Opus Environment** (see section 1, p. 3).

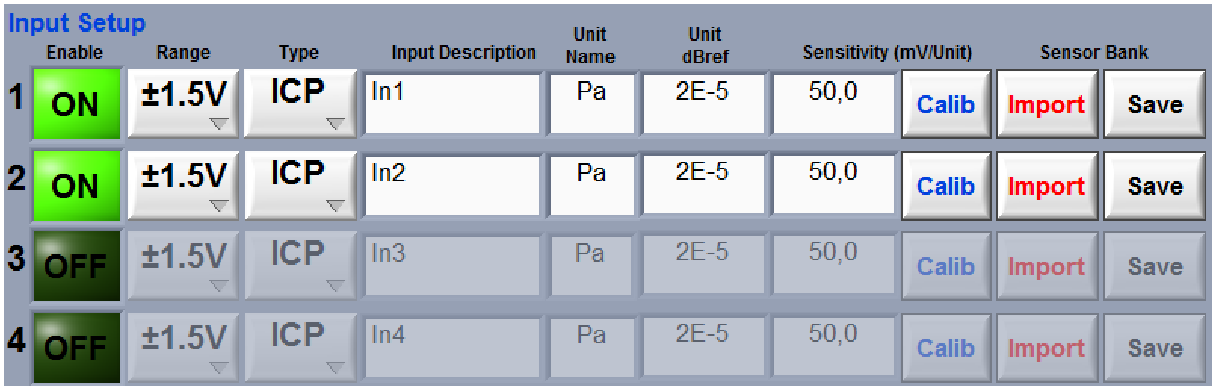
6 Setup Interface

Click the Setup button on the menu bar to access the **Setup Interface**.

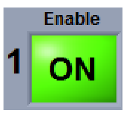


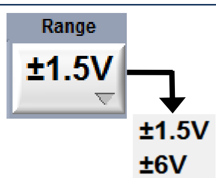
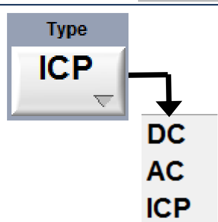
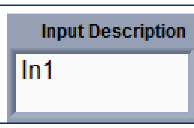
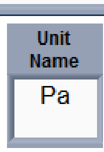
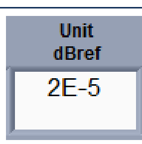
The setup can be saved through the menu **File→Save Config** or it can be loaded through the menu **File→Open Config**. The configuration files have a .cfg extension.

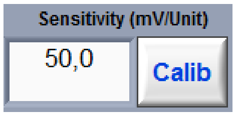
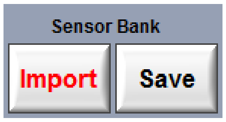
6.1 Input Setup



Input Setup

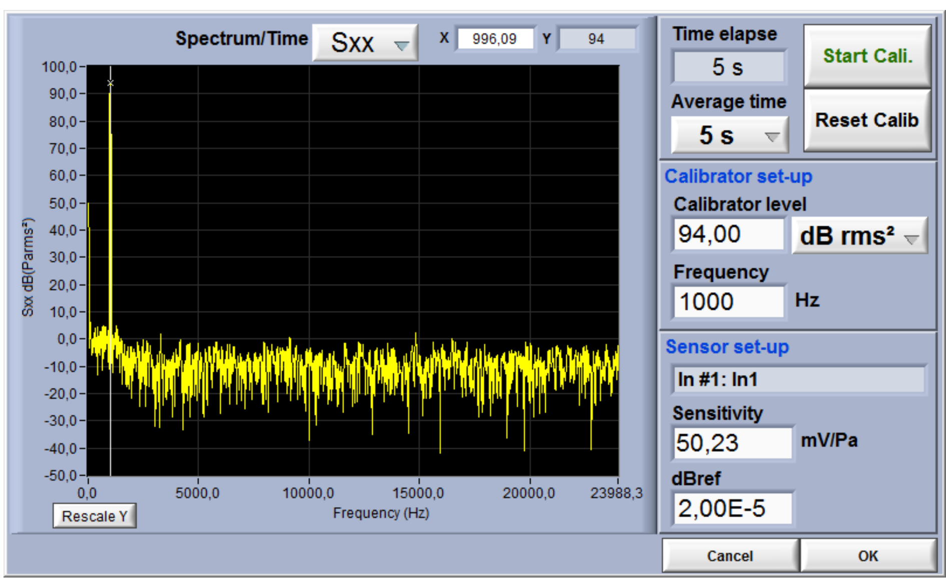
<table>
<tr><td>
</td><td>Input Enable

Set the input switch to ON or OFF to enable or disable the corresponding input. 1 to 4 inputs can be used simultaneously.</td></tr>
<tr><td>
</td><td>Input Range:
• +/- 1.5 volt: low range
• +/- 6 volts: high range (only available on inputs 1 and 2)</td></tr>
<tr><td>
</td><td>Input Type
• DC input: both AC and DC components are present in the signal.
• AC input: the DC component is filtered from the input signal.
• ICP input: polarized for an ICP sensor and DC component is filtered.
• The DC filter is a high-pass filter at Fc=0.5Hz. Common ICP sensors are ICP microphones and accelerometers.</td></tr>
<tr><td>
</td><td>Input Description
This text control allows the user to identify the input. This information has no impact on the behaviour of the acquisition system.</td></tr>
<tr><td>
</td><td>Unit name
This text control is used by the acquisition system for the label of the global level, time and spectrum graphs. For instance, if a Pa unit is specified by the user for input #1, the Y label of the Spectrum will be dB(Parms^2).</td></tr>
<tr><td>
</td><td>Unit dBref
The dB reference value is used by the interface for dB computation. For acoustic measurement with a microphone, the standard dB reference value is 2E-5.</td></tr>
</table>

	Sensor **Sensitivity** in mV/Unit. This sensitivity is used to present a calibrated signal in the **Acquisition** and **Rec Viewer** interfaces. The sensitivity can be manually entered (if known) or can be determined with the calibration function (see section 6.1.1, p.13).
	Import/Save **Sensor Bank** The save function allows saving the complete configuration of a specific input (range, type, input description, unit name, unit dBref and sensitivity). The Import function allows recalling the input configuration saved in a .sen file. Before starting to use the data logging system, we suggest generating a .sen file for all your sensors. Note also that the entire set-up of the data logging instrument can be saved in a .cfg file and recalled from one measurement session to another.

6.1.1 Calibration Interface

The input sensitivity can be manually changed in the text field or it can be calibrated using the calibration function and a sensor calibrator. Click the  button within the **Setup Interface** in order to launch the **Calibration Interface**.



Step 1 Adjust the calibration parameters

The defaults values are:

- Averaging time: 5 s
- Calibrator Level: 94 dB rms^2
- Frequency: 1 kHz

Step 2 Install the calibrator device on the microphone

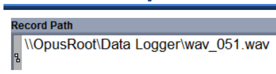
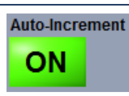
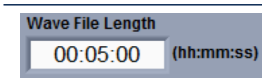
Step 3 Click START

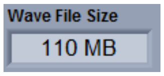
After the average time is elapsed, the sensitivity value will update.

Step 4 Click OK to accept the sensitivity value

6.2 Record Setup

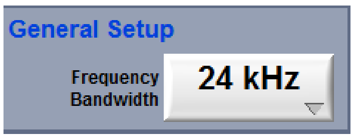
Record Setup

	Record Path Click on the control to adjust the destination path and name of the .wav to be recorded. This uses the **Explorer Window** (see section 10, p. 50).
	Auto-Increment Set the switch to  or  to enable or disable the auto-increment of the record file. When the auto-increment is on, the name must be defined only for the first measurement. All others measurements will be logged in a wave file with a four-digit extension number. For instance: First measurement file name: test.wav Second measurement file name: test_0001.wav Third measurement file name: test_0002.wav
	Wave File Length in the format hh:mm:ss (hour:min:sec). In order to avoid large files that would be heavy to post-process, the data logging will switch wave file as it reaches this length. The length is limited to a minimum five minutes and is further limited to a Wave File Size of 1 GB. The partition index will be had to the specified measurement name as new partitions are created. For instance: Measurement name (length = 5 min): test_0002.wav 1st partition (from 0 to 5 min) : test_0002.wav 2nd partition (from 5 to 10 min): test_0002_2.wav Nth partition (from 5·(N-1) to 5·N min): test_0002_N.wav

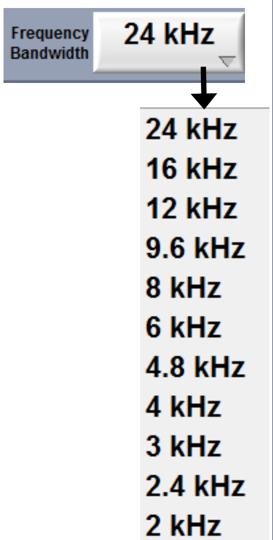


Wave File Size
This indicator informs the user about the maximum size of the wave files (partitions) that will be created. The size depends on the **Wave File Length**, the number of enabled inputs and the **Frequency Bandwidth**.

6.3 General Setup

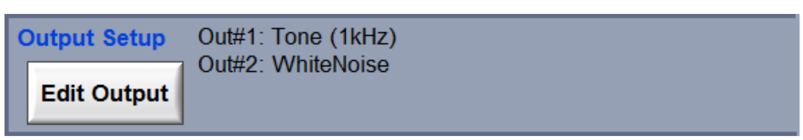


General Setup

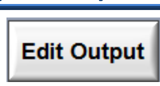


Frequency Bandwidth
This control sets the sampling rate of the acquisition board (both inputs and outputs). The actual sampling rate is simply twice the bandwidth selection (24 kHz bandwidth means a sampling rate of 48 kHz).

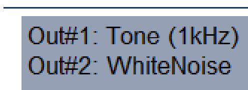
6.4 Output Setup



Output Setup



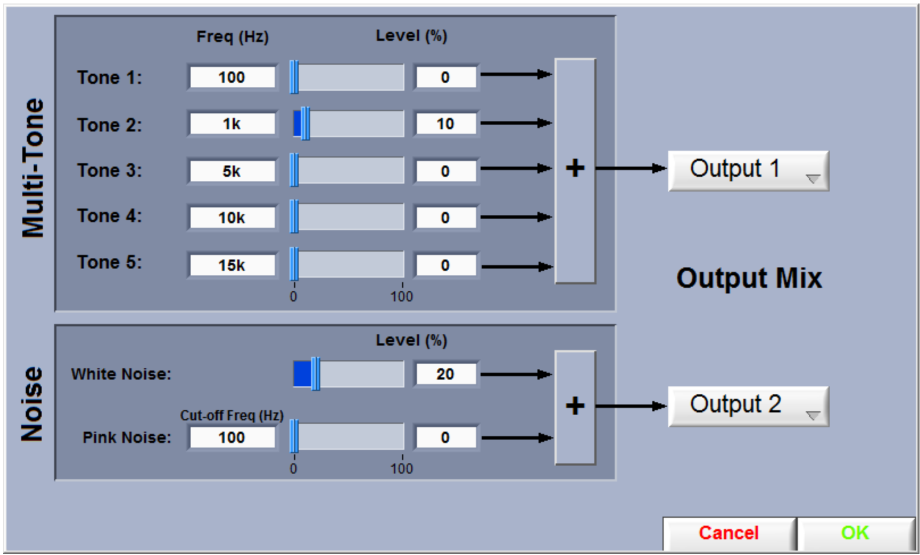
Edit Output button
This button launches the **Output Generator Interface** (see section 6.4.1, p.16)



Output Information
This indicator informs the user about the current output generator setup.

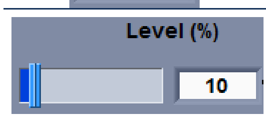
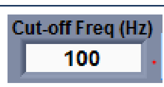
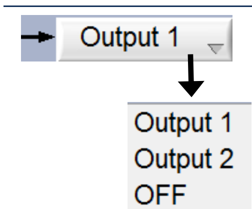
	Click the **Edit Output** button for more details or to modify the generator.

6.4.1 Output Generator interface



The output generator is made of a multi-tone generator (up to five tones) and of a noise generator (white noise and/or pink noise). Each part of the generator can be mixed to outputs 1 and 2.

Output Generator

	**Tone Frequency** in Hertz. This control sets the frequency of to tone to be generated.
	**Signal Level** in percent of full amplitude. The value of this level can be adjusted through the slide control or fine tuned with the numeric control.
	**Pink Noise Cut-off Frequency** in Hertz. This controls set the frequency at which the power density of the pink noise starts to fall (10 dB/decade). The default value is 100 Hz.
	**Output Mix** This control maps its part of the generator to the specified output. Both parts of the generator can be summed to the same output, mapped to different outputs or simply turned off.

Care should be taken to avoid output saturation. Each **Level(%)** can be adjusted to the full output range (100%). When the signals are added at the output, there is a risk that the resulting signal would be higher than the maximum range. This is possible when the sum of the **Level(%)** of all the signals that are mapped to an output is higher than 100%.

6.5 Default Setup

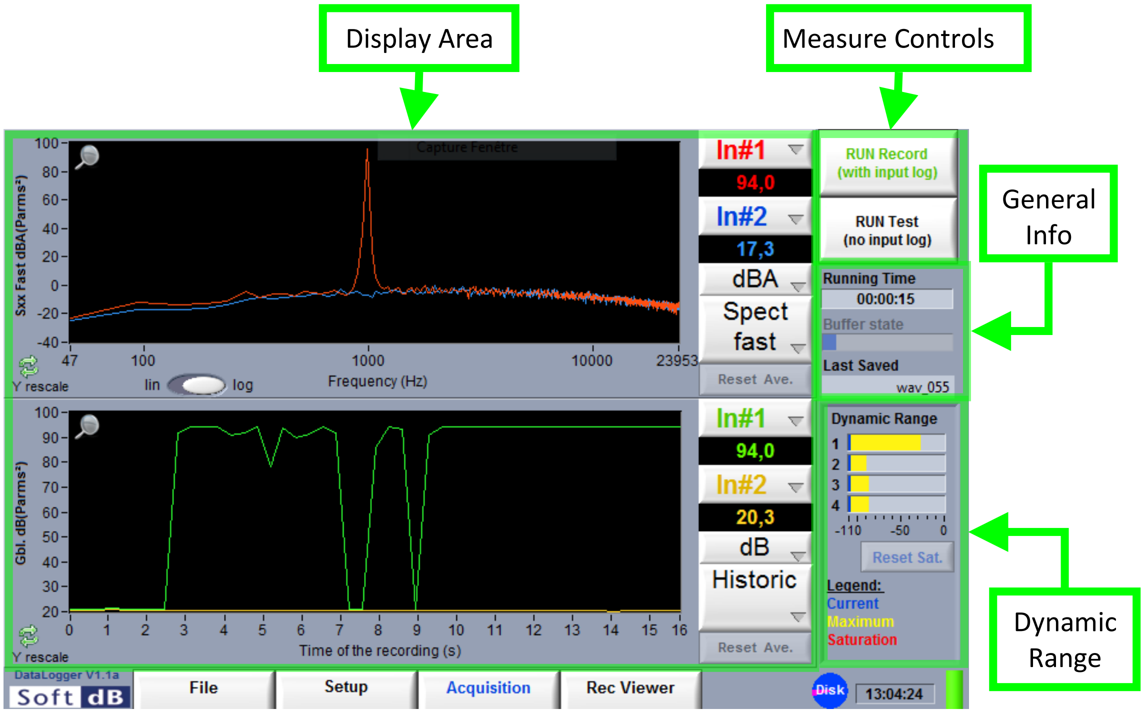
At the delivery, the default setup is:

Input Setup	Channel Selection:	Channel 1
	Dynamic Range:	+/-1.5 V
	Input Type:	ICP
	Description:	In1, In2, In3, In4
	Unit name	Pa
	Unit dBref	2E-5
	Sensitivity:	50 mV/Unit
	Record Path:	\\OpusRoots\Data Logger\wave.wav
	File Auto-Increment	On
	Length of wave file	5 minutes
General Setup	Frequency Bandwidth:	24 kHz
Output Setup	Output generator:	Sinus 1 kHz, 10% of full amplitude
	Output Mix:	Out#1 = Out#2 = OFF

The setup can be modified using the Setup menu. The resulting configuration setup can be saved and recalled using **File → Save Config** and **File → Open Config**.

7 Acquisition Interface

Click the Acquistion button on the menu bar to access the **Acquisition Interface**.



7.1 Measure Controls

The **RUN Record** button starts the acquisition and record the signal into a .wav file. Beside the wave file, a .cfg file is also created in order to provide the measurement information (mainly sensitivity) for the post-analysis. This configuration file is named the same as its wave file. Because both files are closely linked to each other, they should be copied and moved together.

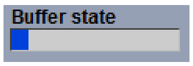
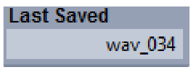
Run/Stop Record (with input log)

RUN Record (with input log)	Starts the measurement and becomes the **Stop Record** button when clicked. The signal of the enabled inputs is saved in the selected wave file.
STOP Record (with input log)	Stops the measurement and becomes the **Run Record** button when clicked.

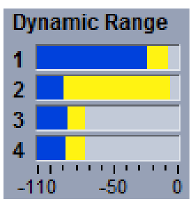
Run/Stop Test (no input log)

RUN Test (no input log)	Starts the measurement and becomes the **Stop Test** button when clicked. No data are saved in a wave file. This is useful to test the acquisition before starting the data logging for real.
STOP Test (no input log)	Stops the measurement and becomes the **Run Test** button when clicked.

7.2 General Info

	**Running Time** This indicator displays the running time in the format hh:mm:ss (hour:min:sec).
	**Acquisition Buffer State** This buffer should stay low and never turn red (buffer full). If it were to occur this would mean that the acquisition failed in keeping a real-time communication.
	**Last Saved Measurement** This indicator displays the name of last saved file (excluding the current record in progress if any).

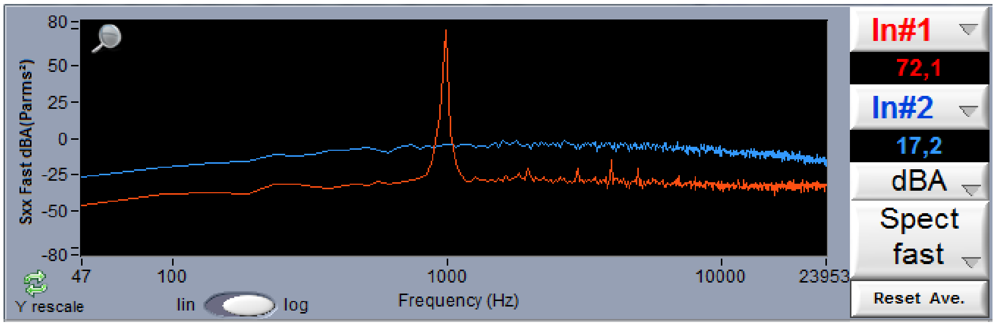
7.3 Dynamic Range

	**Dynamic Range** These vu-meters display the level of the inputs in dBFS (decibels relative to full scale). Color legend: - blue is the current level - yellow is the maximum level measured - red indicates that a saturation occurred
	**Reset Saturation** This button resets the saturation and the maximum level from the **Dynamic Range** vu-meters.

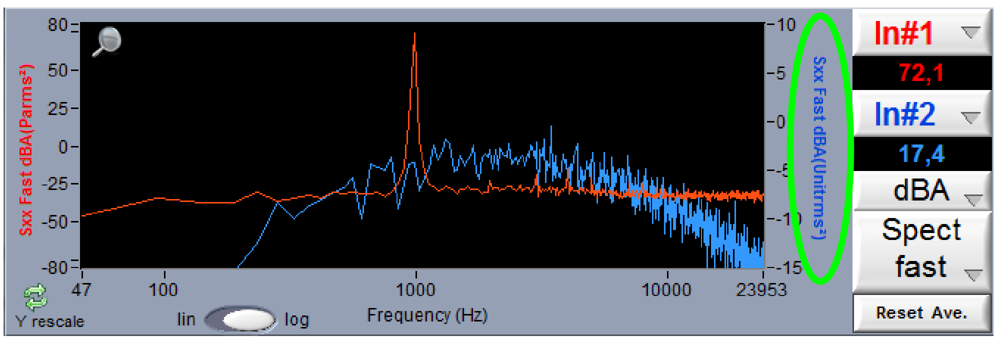
When the input type selection is AC or ICP, the interface removes the DC component of the input signal before the dBFS computation. In this way, the dynamic indicator presents the range occupied by the AC part of the signal. For the Direct DC input selection, the DC is included in the dBFS computation.

In some rare circumstances when the DC signal is relatively high for the Direct AC or ICP input selections, saturation can occur before the DC subtraction. In these circumstances, no AC saturations are detected but a saturation problem is present. To avoid this saturation detection problem, the interface tests the saturation before removing the DC component. If saturation is detected at this stage, the number of the corresponding input on the top of the dBFS indicator turns red.

7.4 Display Area

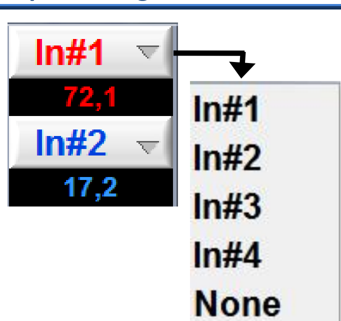


One or two graph areas can be displayed in the Display Area. Each graph contains two plots. The graph settings can be changed during the measurement. Data is displayed in real-time with a refresh of 1/3 s.



If two plots with different units are displayed, a second Y axis appears on the right side of the graph. Then, each plot has its own Y axis which is the same color as its corresponding plot.

Graph settings

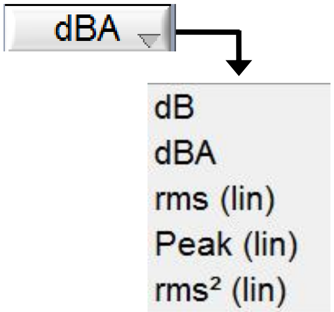
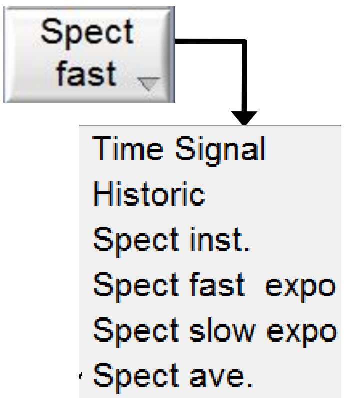
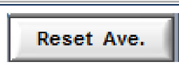
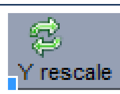


Display Input

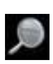
These controls select the inputs to be display.

Below the input selection is its global level.

The color of the control is the same as the corresponding plot on the graph.

	**Display Units** This control sets the units for the graph and for the global levels. • dB: decibels. • dBA or dB(A): decibels after A frequency weighting is applied. • rms : root mean square in Unit rms. • Peak: maximum level. • rms^2
	**Display Graph Type** This control applies to both plots of the graph. The type can be time signal, global historic or one of the FFT spectrum[3]. • Time Signal: frames of 1/3 s of time signal. • Historic: History of global levels vs time. • Instant Spectrum: FFT Spectrum on time signal frames. • Fast exponential Spectrum: average filter with τ = 1/8 s. • Slow exponential Spectrum: average filter with τ = 1 s. • Average Spectrum: linear average.
	**Reset Average** This button resets the averaging of signal (exponential spectrums and average spectrum) and also resets the historic graph.
	**Frequency lin/log** This control sets the X axis of the spectrum graph in linear or logarithmic scale.
	**Y rescale** This button adjusts the Y scale to the graph data. If two Y scales are present on the same graph, both axes are rescaled independently.
 and 	**Magnifier buttons** See section 7.4.1 below for details.

7.4.1 Magnifier

Each graph has a  icon at the top left corner. Clicking on this icon pops-up the graph on a **magnified** display.

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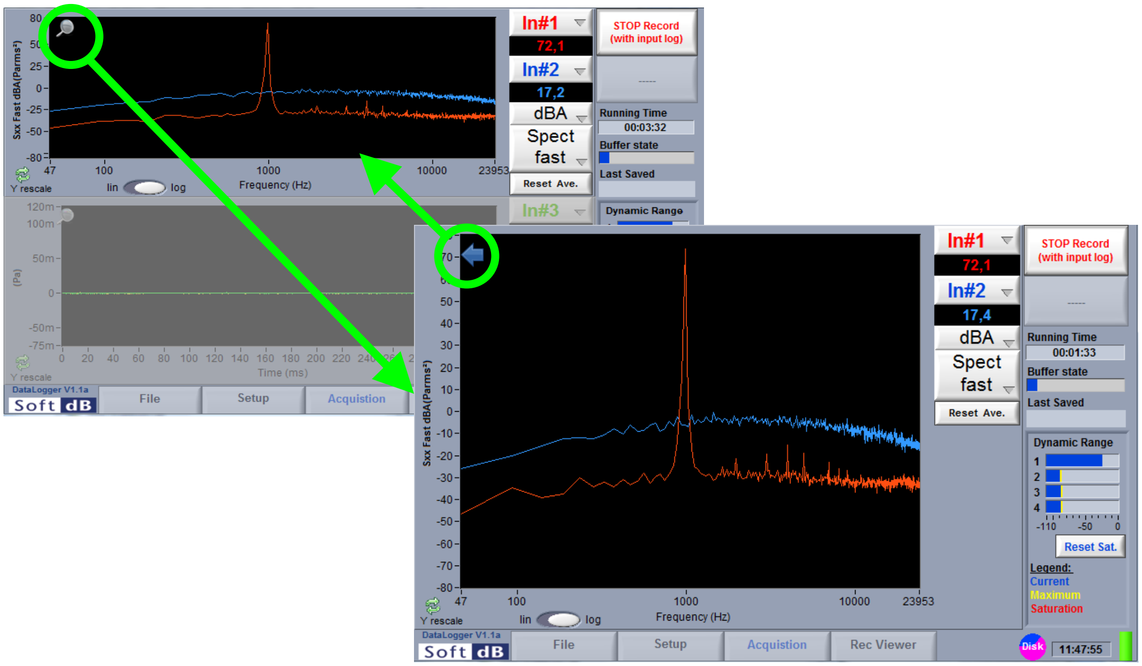
[3] In the particular case where the Frequency Bandwidth is 24 kHz and 4 inputs are enabled, the bottom graph of the Acquisition Interface will not allow spectrum graph types or dBA units.

The **magnified** display has a icon at the top left corner. Clicking on this icon pops back the graph to its initial size and position.



7.5 Export Acquisition Graphs

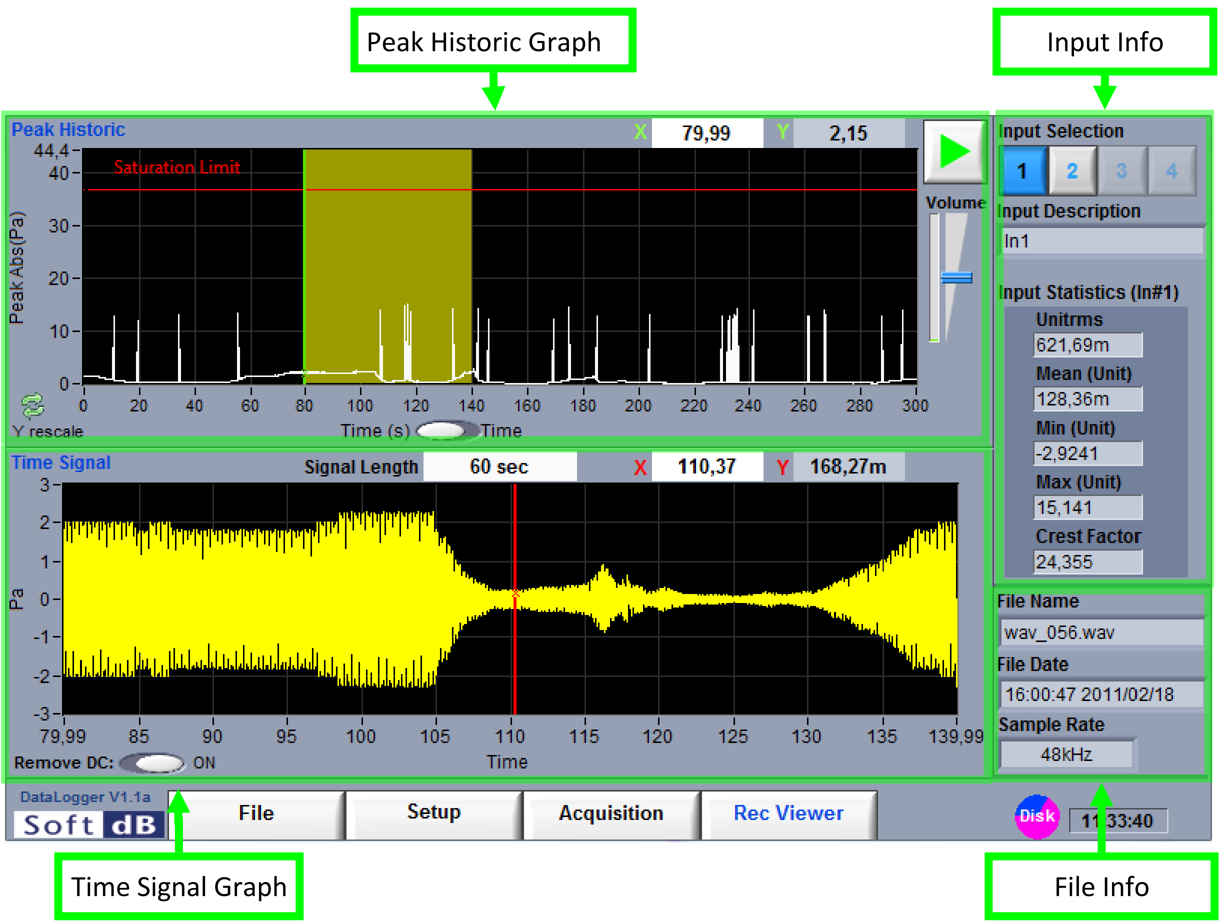
The graph plots can be exported into text file of extension .xls (compatible with Excel).

The command **File→Export Graph** exports both graphs into a single file. Data are exported with the same resolution as seen on display.

8 Record Viewer Interface

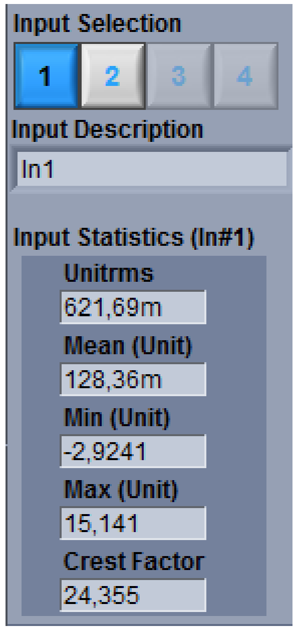
The **Record Viewer Interface** allows a basic analysis of a previously recorded wave signal on a Concerto Unit. Further analysis of the acquired data should take place on a PC. When run on a PC the “Record Viewer tab” becomes the “Post-Process tab” and the advanced post-processing functions are unlocked (refer to the section 9 of this document for details).

To access the **Record Viewer Interface,** go to **File→Open Data** or click the Rec Viewer button on the menu bar.

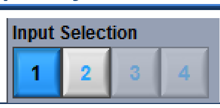
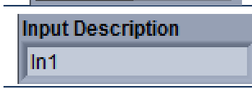
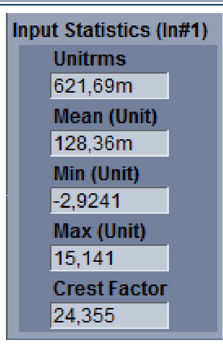


This interface uses a streaming approach that never completely loads the signal into the memory. In order to keep the overview of the entire signal contained in the wave file, the interface presents the peak values of the entire wave. The entire signal is divided in 4096 blocks and the peak values of each block are used to form the **Peak Historic Graph** (section 8.3, p.25). For a more detailed view of some portions of the signal, the interface presents a **Time Signal Graph** (section 8.4, p.26) with a time span specified by the yellow area on the **Peak Historic Graph**.

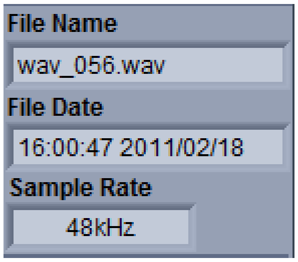
8.1 Input Info



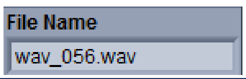
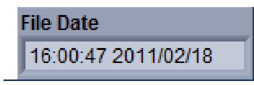
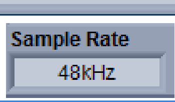
Input Info

	**Input Selection** This control selects the input to be displayed among the available ones.
	**Input Description** Description of the input as described in the **Input Setup.**
	**Input Statistics** This indicator displays the standard statistics of the input measurement. The whole file length is considered no matter the length displayed on the graphs. The *Unit* refers to the **Unit Name** set in the Input Setup. • rms: The AC component of the signal in Unit rms. • Mean: The DC component of the signal in Unit. • Min: The minimum level of the signal in Unit. • Max: .The maximum level of the signal in Unit. • Crest Factor: The peak-to-average ratio.

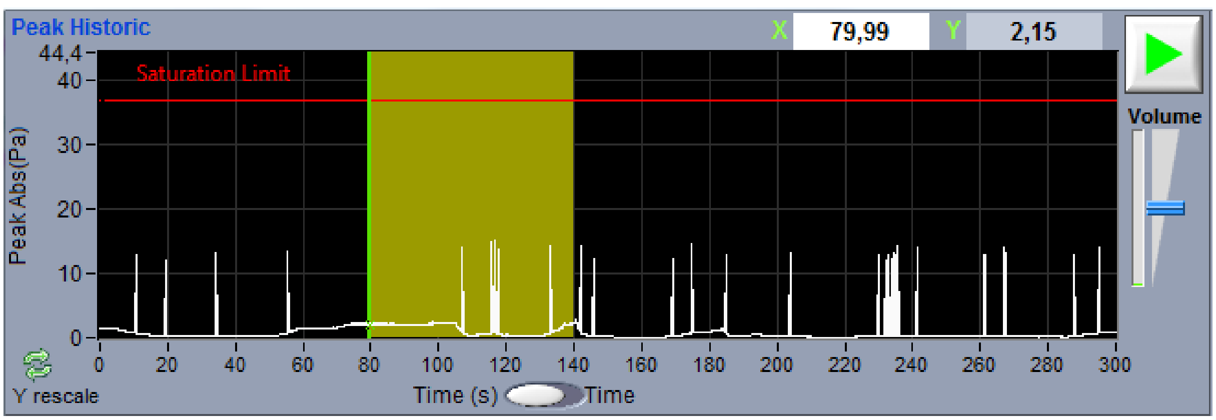
8.2 File Info



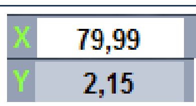
Input Info

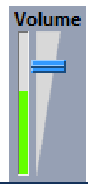
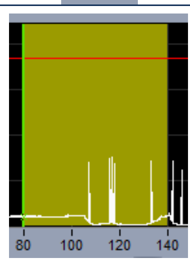
	**File Name** This indicator displays the name of the file being viewed.
	**File Date** This indicator displays the starting time of the file being viewed.
	**Sample Rate** This indicator displays the sample rate of the file being viewed.

8.3 Peak Historic Graph

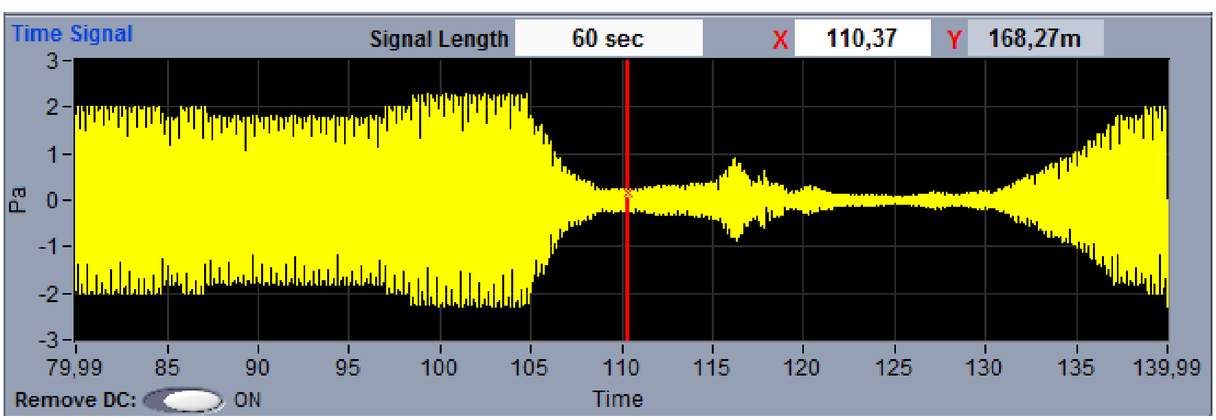


Peak Historic Graph

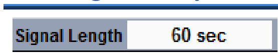
	**Saturation Limit** The red line at the top of the graph indicates the level of saturation of the input.
	**Green cursor X and Y values** The green X (time) and Y (amplitude) values are linked to the green cursor on the graph. The X value can be entered manually or defined by pointing to the position on the graph. The cursor also defines the beginning to the yellow area for the **Time Signal Graph.**

	**Y rescale** This button adjusts the Y scale to the graph data.
	**Time format** This control sets the time axis format in seconds or in format *hh:mm:ss*.
 and 	**Playback Start/Stop** This button starts the playback of the record from the green cursor until the end of the wave or the user stop. If the file is a multi-channel wave, the playback first proceeds to a pre-reading stage in order to guaranty the quality of the listening.
	**Playback Volume** This control allows the user to adjust the volume of the playback. The green bar at the left shows the current level of the playback. The color of the bar changes to red as saturation is seen to indicate that the volume is too loud.
	**Time Signal span** The yellow area is the selected time span to be displayed into the **Time Signal Graph** (section 8.4, p.26). The span can be selected intuitively by pointing (press) on the graph at the starting time and then sliding to the end time (release). Alternatively, the start time of the time span can be fine tuned with the green X cursor  and with the **Signal Length** on the **Time Signal Graph**.

8.4 Time Signal Graph



Time Signal Graph

	**Signal Length** The control sets the length of the signal on the **Time Signal Graph**.
	**Red cursor X and Y values** The red X (time) and Y (amplitude) values are linked to the red cursor on

	the graph. The X value can be entered manually or defined by pointing to the position on the graph.
	**Remove DC** This button can be used to subtract the mean (DC component) from the signal to be displayed on the **Time Signal Graph**.

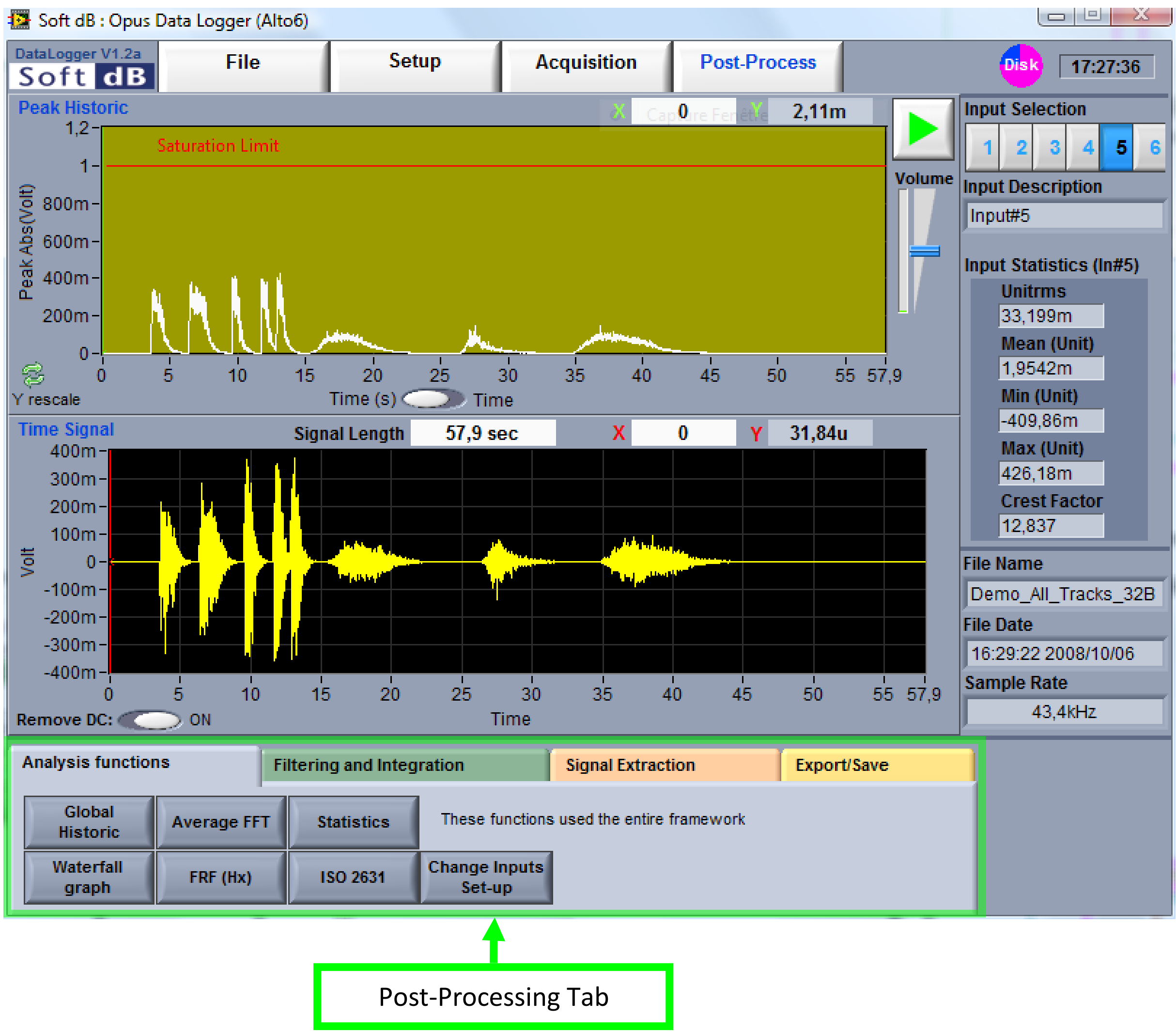
8.5 Export Record Viewer Graphs

The graph plots can be exported into text file of extension .xls (compatible with Excel).

The command **File→Export Graph** exports both graphs into a single file. Data are exported with the same resolution as seen on display. For this reason, the exported data of the time signal should not be used for further analysis.

9 Post-Processing Interface

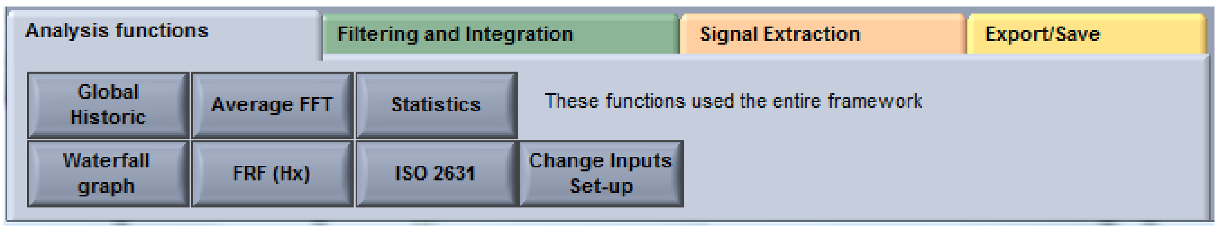
As describe in the table of the section 0, the advanced post-processing functions are not available on the Concerto hardware because the unit is intended for the signal acquisition. The advanced post-analysis of the acquired data must take place on a stand-alone PC. Below is the presented the Post-Process Interface. It is the same as the Rec Viewer Interface (on a Concerto), but it also unlocked the Post-Processing tab at the bottom of the panel.



The post-processing tab includes a complete software tool allowing the analysis of a multi-channel wave file. The post-processing can be used even if the acquisition unit is not present. The post-processing interface accepts wave files recorded with the *Opus Data Logger* software and standard wave files. The recorded wave files always have an associated configuration file that contains the complete configuration of the instrument. This configuration file is automatically saved in the same folder as the wave file at the beginning of the recording. This configuration file allows recovering a calibrated time signal. So, any wave files not recorded with the *Opus Data Logger* system will not be calibrated (the *Change Input Set-up* function can be used to calibrate the signal, see section 9.1.7 for more details).

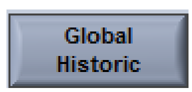
9.1 Analysis Functions Tab

The post-processing module includes six analysis functions, which are applied on the entire selected framework (see section 9.3 for more information about the framework selection). These functions are grouped under the *Analysis Tab* at the bottom of the main post-processing interface.



The next sections describe these functions in detail.

9.1.1 Global Historic

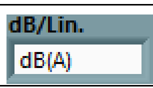
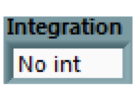
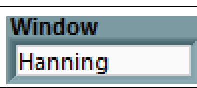


This function allows building the global level historic with an adjustable time resolution for the selected channel. The global level calculation is done with power spectra using blocks with a number of samples adjusted to match the requested time step. The power spectrum approach is used to facilitate the A weighting and integration options. The interface of the *Global Historic* function is presented in the next figure.

At any time, the computation parameters can be changed and the *Recompute* button can be used to update the results.

The *Do Graph Autoscale* button allows adjusting the Y and X scales for the new data of the global historic graph.

The next table explains the computation parameters:

Computation parameters	Description
	The time step of the global level historic can be adjusted between 50ms and 60000ms.
	The dB/Lin. parameter allows choosing the amplitude scale format. The choices are dB, dB(A), Lin. rms, Lin. peak or Lin. rms2.
	The Integration parameter allows application of simple or double integration in the frequency domain for an acceleration measurement. For the simple integration option, the power spectrum is divided by jω. For the double integration option, the spectrum is divided by –ω2.
	This is the time window used for the power spectrum computation.

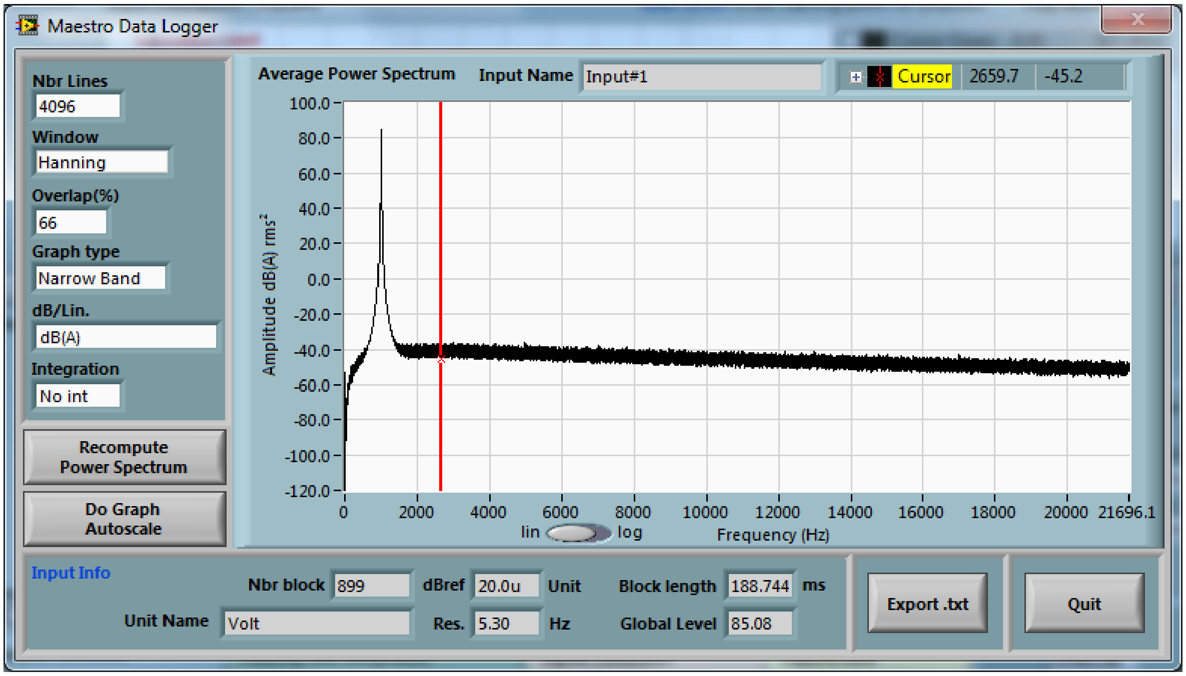
	This parameter allows adjusting the overlap between the time blocks used for the global levels historic calculation. This parameter can be adjusted from 0% to 90%. This parameter is used to increase the number of blocks and the time resolution of the historic graph. We suggest keeping this parameter to 0% if standard acoustic signals are used. A larger overlap can be used to avoid aliasing problem when the signal is impulsive.

The Export .txt function allows exporting the global historic data in text format.

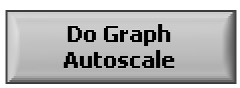
9.1.2 Average FFT

This function computes the average power spectrum for the entire framework of the selected channel. First, the time signal of the entire framework is divided into blocks of an adjustable number of samples. The power spectrum is computed for each block and a linear average is obtained. Also, the *Average FFT* module allows the overlap of the blocks used for the average. For instance, the use of an overlap of 66% and the Hanning time window is a classic set-up that compensates for data lost in the time window.

The interface of the *Average FFT* function is presented on the next figure.

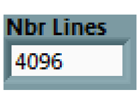
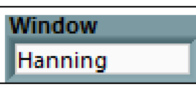
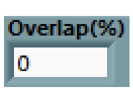


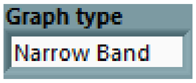
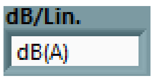
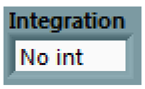
At any time, the computation parameters can be changed and the recompute button can be used to update the results.



The *Do Graph Autoscale* button allows adjusting the Y and X scales for the new data of the average power spectrum graph.

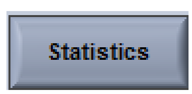
The next table explains the computation parameters:

Computation parameters	Description
	This is the number of lines of the average power spectrum. The selections are: 1024, 2048, 4096 or 8192 lines. The FFT calculation is done with a block size of 2**Nbr Lines*.
	This is the time window used for the power spectrum computation.
	This parameter allows adjusting the overlap between the time blocks used for the average power spectrum calculation. This parameter can be adjusted from 0% to 90%. This parameter is used to increase the number of blocks for the average and to compensate for the loss of data in the time window. The classic set-up is an overlap of 66% and the Hanning window.

	This is the type spectrum. Narrow Band, octave, 1/3-octave and 1/12-octave set-ups are possible. The octave, 1/3 octave and 1/12-octave types are computed with a frequency bank of filters applied on the narrow band spectrum. To avoid low frequency imprecision, we suggest using the maximum number of lines (8192) set-up when an octave type is selected.
	The *dB/Lin.* parameter allows choosing the amplitude scale format. The choices are dB, dB(A), lin. rms, lin. peak, lin. rms^2, ISO2631-XY (dB), ISO2631-Z (dB), ISO2631-XY (lin. rms et lin. peak) and ISO2631-Z (lin. rms et lin. peak). The ISO2631 selections allow applying standard frequency weighting to the average power spectrum.
	The *Integration* parameter allows applying a simple or double integration in the frequency domain for an acceleration measurement. For the simple integration option, the power spectrum is divided by $j\omega$. For the double integration option, the spectrum is divided by $-\omega^2$.

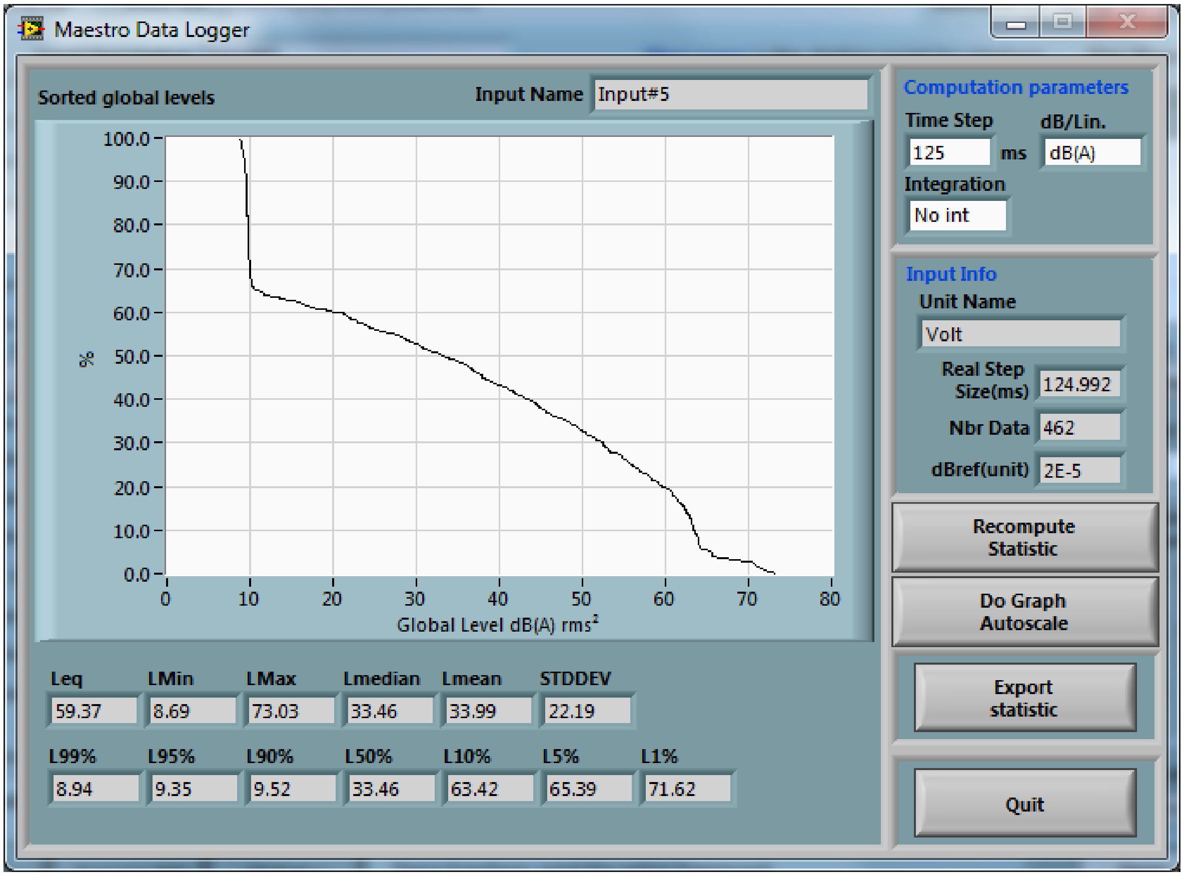
The function Export .txt allows exporting the average power spectrum in text format.

9.1.3 Statistics



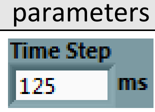
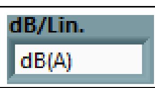
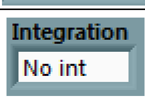
This module allows computing the standard Lx% statistics on the global level historic for the entire framework of the selected channel. The *Statistic* function starts by computing the global levels of the framework with an adjustable time resolution. The global level calculation is done with power spectra using blocks with a number of samples adjusted to match the requested time step. A Hanning time window is used for the FFT computation. The power spectrum approach is used to facilitate the A-weighting and integration options. When the global level historic is computed, the *Statistics* module computes the statistics of the global levels and presents a graph of the sorted global levels.

The interface of the *Statistics* function is presented in the next figure.



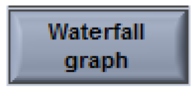
At any time, the computation parameters can be changed and the *Recompute* button can be used to update the results.

The next table explains the computation parameters:

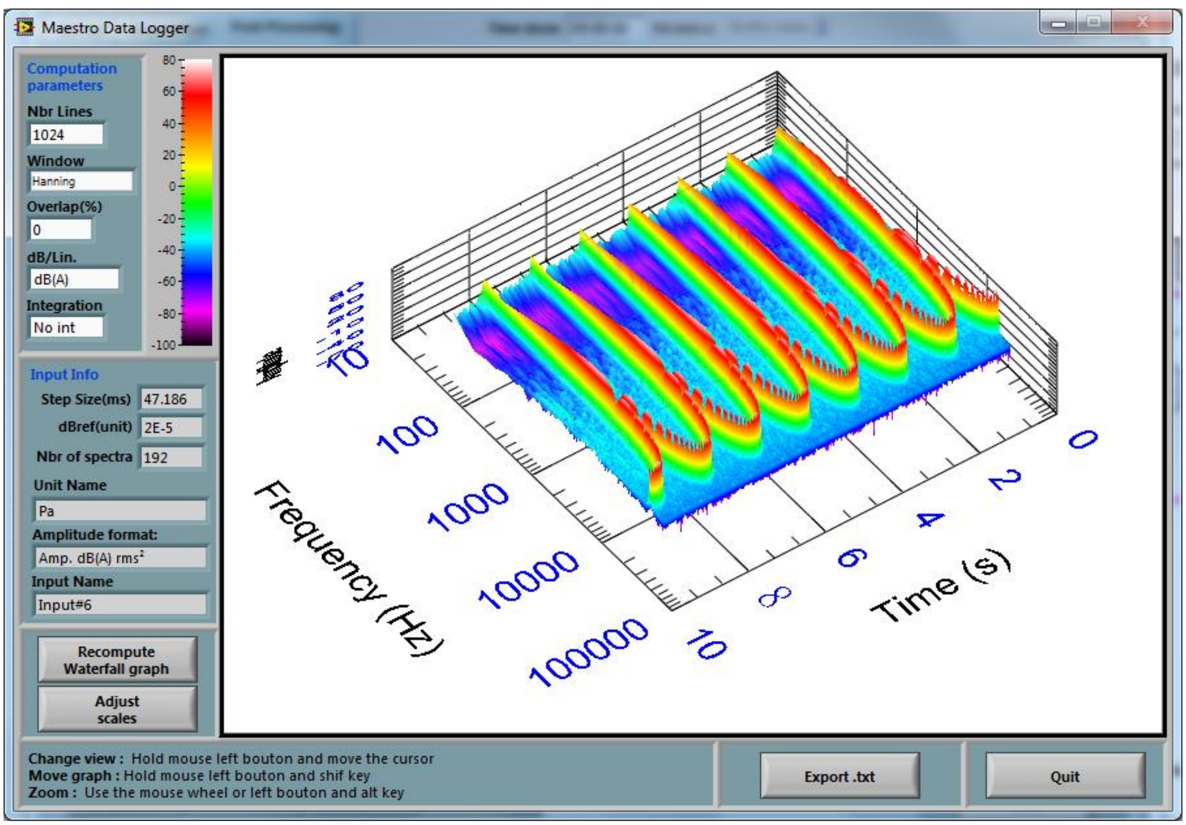
Computation parameters	Description
	The statistics are computed for all global levels of historic. The time step of the global level historic can be adjusted between 50ms and 1000ms.
	The dB/Lin. parameter allows choosing the amplitude scale format. The choices are dB, dB(A), lin. rms, lin. peak or lin. rms2.
	The Integration parameter allows applying simple or double integration in the frequency domain for an acceleration measurement. For the simple integration option, the power spectrum is divided by jω. For the double integration option, the spectrum is divided by –ω2.

The *Export statistic* function allows exporting the statistics in text format.

9.1.4 Waterfall graph

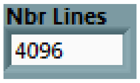
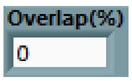
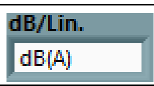
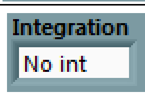


The *Waterfall graph* module presents the power spectra for the entire time-length of the framework. The *Waterfall graph* module computes the power spectra with an adjustable frequency resolution. Each power spectrum is presented in a three dimensional graph along a time axis. As well as the numerical amplitude scale, a color scale is employed to help in visualizing the high and low levels of the power spectrum amplitudes. The three dimensional graph can be rotated, zoomed and moved. The next figure presents the waterfall interface.

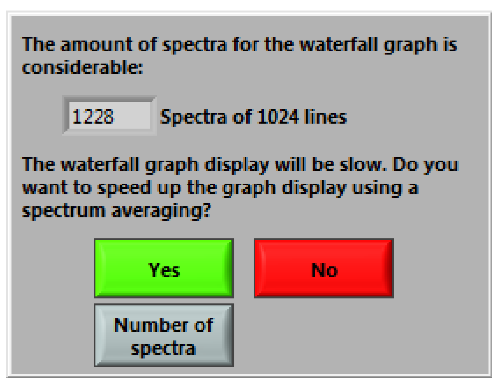


At any time, the computation parameters can be changed and the recompute button can be used to update the results.

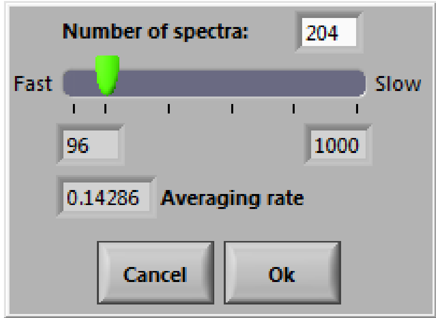
The next table explains the computation parameters:

Computation parameters	Description
	This is the number of lines of the average power spectra. Possible selections are 1024, 2048, 4096 or 8192 lines. The FFT calculation is done with a block size of 2**Nbr Lines*. For a waterfall graph, a large number of lines mean a poor time resolution. Conversely, greater time resolution results in lower frequency resolution. Note that the overlap parameter can be used to limit this problematic.
	This is the time window used for the power spectrum computation.
	This parameter allows adjusting the overlap between the time blocks used for the power spectrum calculation. This parameter can be adjusted from 0% to 90%. This parameter can be used to increase the time resolution of the waterfall graph.
	The *dB/Lin.* parameter allows choosing the amplitude scale format. The choices are dB, dB(A), lin. rms, lin. peak or lin. rms^2.
	The *Integration* parameter allows applying a simple or double integration in the frequency domain for an acceleration measurement. For the simple integration option, the power spectra are divided by $j\omega$. For the double integration option, the spectra are divided by $-\omega^2$.

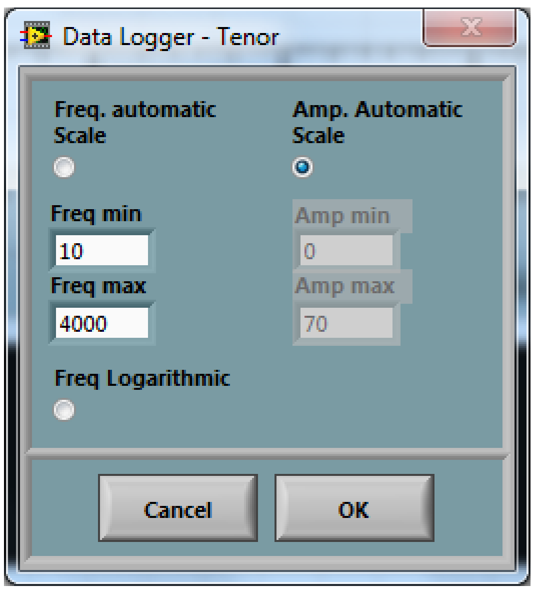
At the time of the computation and if the framework has a considerable amount of data, the interface asks the user for using a spectrum averaging:



If the user clicks *Yes*, the software will perform a spectrum averaging to reduce the amount of data. In this case, the time resolution of the waterfall graph will be reduced. However, the user can adjust the averaging rate with the button *Number of spectra.* Then, the number of spectra and the averaging rate can be adjusted with the following sliding control:



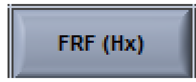
The *Adjust scales* button allows setting the frequency and amplitude scales of the waterfall graph. Unlike on the other two-dimensional graphs of the interface, minimum and maximum scale values cannot be changed directly on the waterfall graph. The minimum and maximum values for each scale can be changed using the following dialog box:



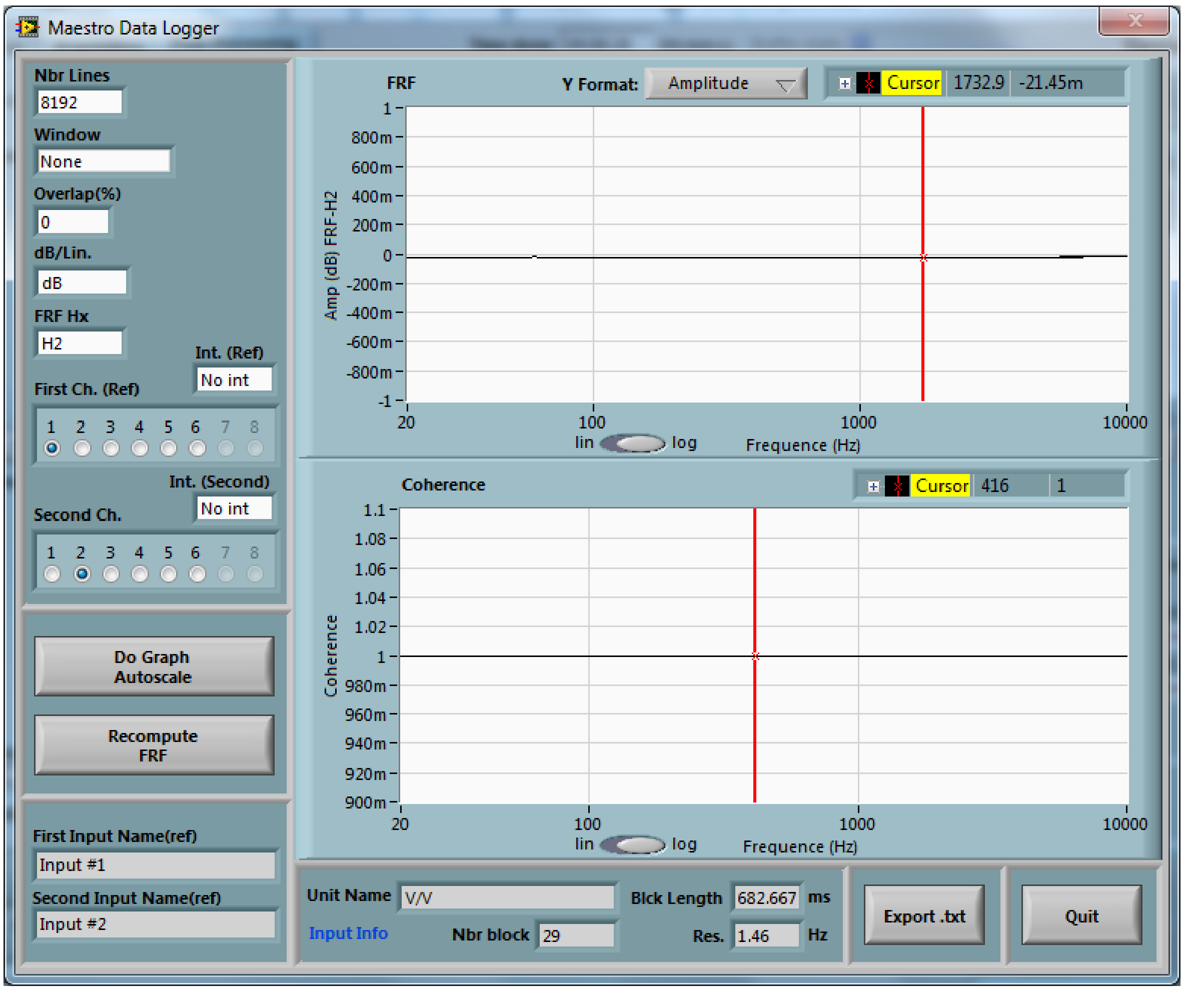
The frequency and amplitude scales can be adjusted through the dialog box. The frequency scale can also be put in logarithmic mapping mode. Note that no options are available for the time scale. The waterfall graph is always generated for the entire framework. The starting point and the length of the framework can be adjusted in the main interface (see section 9.3).

The *Export .txt* function allows exporting the waterfall data in a text format.

9.1.5 FRF (Hx)



The *FRF (Hx)* module allows computing the average frequency response between channels. The following figure presents the FRF module interface:



At any time, the computation parameters can be changed and the *Recompute FRF* button can be used to update the results.

The FRF interface has two graphs: 1) The FRF graph to present the amplitude (or the phase) versus the frequency, and 2) the coherence graph. Three types of FRF can be computed:

H1 using this equation: H1=y/x=Sxy/Sxx

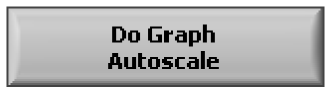
H2 using this equation: H2=y/x=Syy/Sxy

H3 using this equation: H3=y/x=(H1+H2)/2

Where x is the reference channel and y the second channel. In theory, all Hx must give the same result. However, when the FRF contains resonance and/or anti-resonance, the result can vary from one equation to another. In effect, noise on the X or Y channel on the anti-resonance or resonance can distort the results. The H1 equation is better when the noise is on the Y channel while the H2 equation gives better results when the noise is on the X channel.

The coherence graph allows determining if the signal measured on both channels is linked. The equation used for the coherence is:

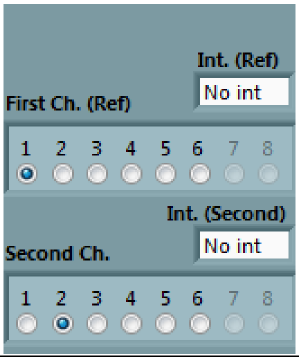
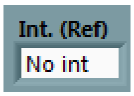
Coherence=H1/H2=Sxy^2/(Sxx*Syy)

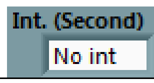


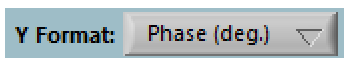
The *Do Graph Autoscale* button allows adjusting the Y and X scales of both the FRF and coherence graphs.

The next table explains the computation parameters:

Computation parameters	Description
Nbr Lines 4096	This is the number of lines of the average FRF. Possible selections are 1024, 2048, 4096 or 8192 lines. The FRF calculation is done with a block size of 2**Nbr Lines*.
Window Hanning	This is the time window used for the Sxx, Syy and Sxy computation.
Overlap(%) 0	This parameter allows adjusting the overlap between the time blocks used for the FRF calculation. This parameter can be adjusted from 0% to 90%. This parameter is used to increase the number of block used for the average calculation and to compensate for the loss of data that causes the time window. The classic set-up is an overlap of 66% and the Hanning window.
dB/Lin. dB	The *dB/Lin.* parameter allows choosing the amplitude scale format. The choices are dB or Lin.
FRF Hx H2	This parameter allows selecting the Hx (H1, H2 or H3). Refer to the beginning of this section for computation technique detail. H1 must be used if noise is on the second channel while H2 must be used if noise is on the reference channel. H3 is an average of both H1 and H2 and can be used if the noise is on both channels.

<table>
<tr>
<td>
</td>
<td>These controls allow selecting the reference channel (x) and the second channel (y).</td>
</tr>
<tr>
<td>

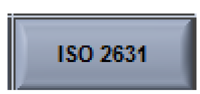
or

</td>
<td>These controls allow the integration of the inputs in the FRF calculation process.</td>
</tr>
</table>



This control can be used to select the format of the Y-axis for the FRF graph. The amplitude, phase (deg./rad), Im and Re formats can be used.

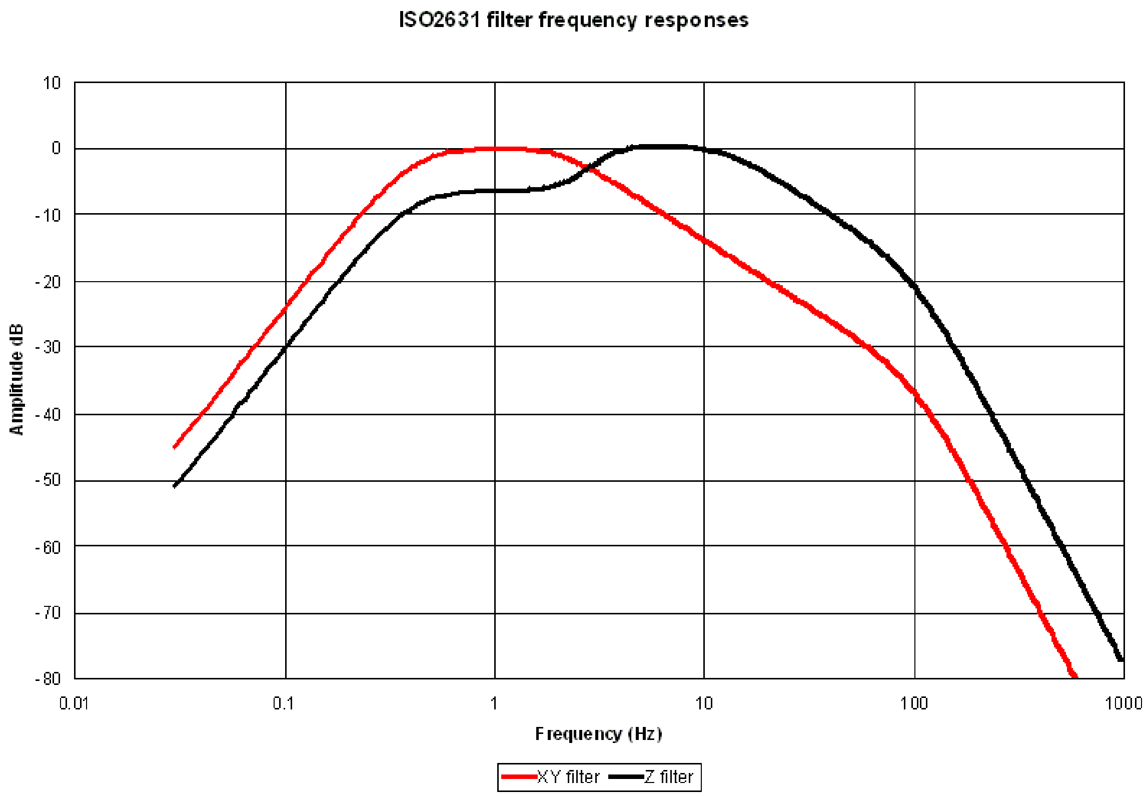
The function *Export .txt* allows exporting the FRF (the Phase (Deg./Rad), Amplitude, Im and Re) and the coherence data in a text format.

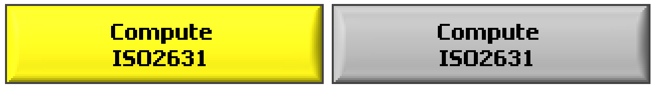
9.1.6 ISO2631



This function applies a filter and computes the statistics on three selectable channels following the ISO2631 standard. The next figure presents the interface of the ISO2631 module:

The ISO2631 standard has one filter for the XY-axes and another filter for the Z-axis. The following curves present the frequency response of these filters.



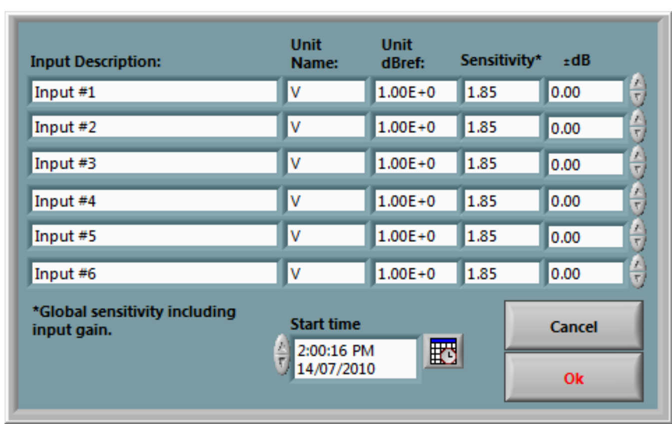


At start-up, the *Compute ISO2631* button is yellow. This means that the ISO2631 filters have not been applied and the statistics are not up-to-date. After computation, button *Compute ISO2631* the turns grey to indicate that the filtering operation has been done and the statistics are up-to-date. Use the X,Y and Z tab to see the statistics for all axes.

The *Export .txt* function allows exporting the X,Y and Z statistics in text format.

9.1.7 Change Inputs Set-up

This function allows changing the input sensitivity, the input description, the unit name and the dB reference set-up. This function has been added to allow the calibration of an existing wave file, which is not recorded with the *Opus Data Logger* system. When opening an external wave file, the software generates a standard associated configuration file with an input sensitivity of 1. Then, the function *Change Inputs Set-up* can be used to change the default sensitivity to obtain a calibrated signal. We suggest to estimate the sensibility correction with a spectrum analysis through the function *Average FFT.* The dialog box to change the inputs set-up is shown on the next figure:

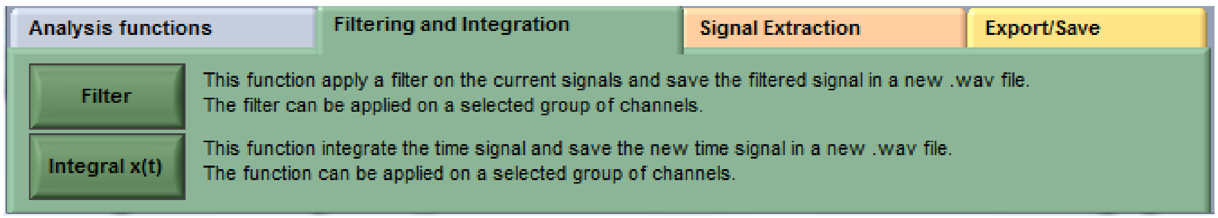


The input sensitivities can be changed with the *Sensitivity* control or using the *±dB* control if the dB correction value is known in dB.

Note: the Start Time control allows changing the starting time of the wave file. This modification can be useful if absolute time scale for the global historic graph is desired.

9.2 Filter and Integration Tab

The *Filtering and Integration* tab at the bottom of the main post-processing interface includes two filter functions that can be applied across the entire selected framework (refer to section 9.3 for more information about the framework selection).



The Filter function is a generic filtering module while the Integral x(t) function allows making simple or double-time integration on an acceleration signal. Both functions filter the signal since the Integral x(t) makes the time integration with a filter.

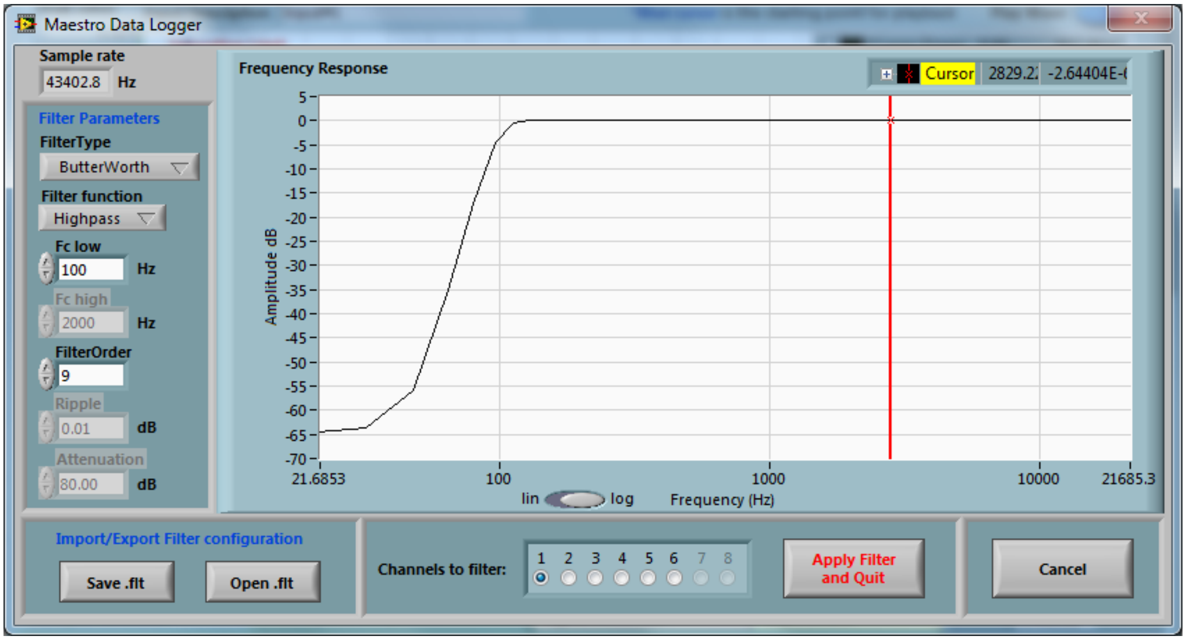
A streaming technique is used to apply a filter on the entire framework. Both filtering functions use the current wave file as a source for the filter and save the filtered signal in a new wave file. The filtering operation starts at the beginning of the framework and is done on the entire framework. If the current framework does not cover the entire signal of the original wave file, the filtering operation will generate a smaller filtered wave file that contains only the filtered signal of the selected framework. In this case and after a filtering operation, the starting point of the framework is automatically reset to zero and the entire filtered signal is presented on the peak value curve. Note that the original wave file is always conserved on the PC’s hard disk.

Note: After the filtering operation, the beginning of the signal can be distorted because of the impulse response of the filter applied. We suggest removing the beginning of the time signal by using the time extraction function (see section 9.3 for details).

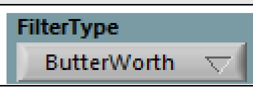
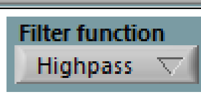
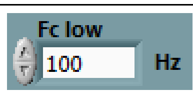
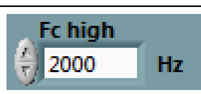
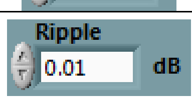
The next two sections describe the filtering functions in detail.

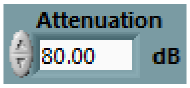
9.2.1 Filter

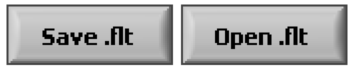
The *Filter* function is very useful for removing some frequency components on a group of selected channels. Some typical tasks that can be done with this function are a DC filter, a notch filter to remove the 60 Hz and its harmonics and a low-pass filter to remove high-frequency noise for clearer playback. This function has a filter design interface, which is presented in the next figure.



The frequency response graph represents the filter response for the current filter parameters. You don't have to use any button to update the frequency response graph; the graph is automatically updated when the filter parameters change. The filter parameters are described in the following table:

Filter Parameters	Description
	There are five available filter types: 1) ButterWorth 2) Chebyshev 3) Inverse Chebyshev 4) Elliptic and 5) Bessel.
	The filter function control allows selecting a Lowpass, Highpass, Bandpass or Bandstop filter. When a Lowpass or Highpass filter is selected, only the Fc low parameter is available. The Fc high parameter is available for the Bandpass and Bandstop selections only.
	This parameter allows specifying the cut-off frequency of Lowpass and HighPass filters and the low cut-off frequency of Bandpass and Bandstop filters.
	This parameter allows specifying the high cut-off frequency of the Bandpass and Bandstop filters. This parameter is not used for the Lowpass and Highpass selections.
	This is the filter order. This parameter is available for all filter type selections.
	This parameter is used only for the Elliptic and Chebyshev filter types. This parameter is the tolerable passband ripple of the filter. It must be greater than zero and expressed in decibels.

	This parameter is used only for the Elliptic and Inverse Chebyshev filter types. This parameter is the desired attenuation in the stop band of the filter. It must be greater than zero and you must express it in decibels.



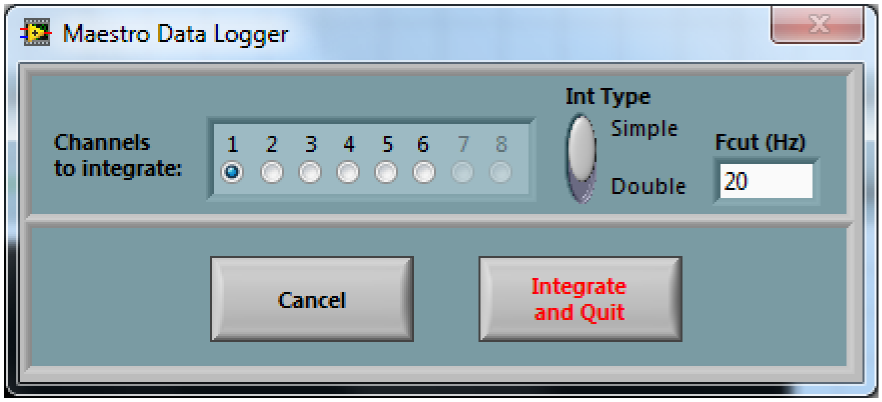
All filter parameters can be saved and recalled using these buttons. Note that the filter interface automatically creates a .flt file when the filter operation is launched with the *Apply Filter and Quit* button. This automatically saved .flt file will have the name of the filtered wave file and it will be in the same folder. This feature is very useful for following the filtering historic of a wave file.

This control allows selecting the group of channels to which the filter will be applied. Note that the non-filtered channels will be included in the wave file created by the filter function.

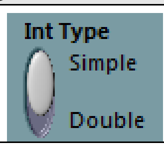
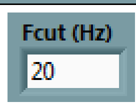
This button launches the filtering operating. The original wave file will be used as a source for the filter and the filtered signal will be saved in a new wave file. Also, a .flt file containing the filter parameters is automatically created in the same folder that the filtered wave.

9.2.2 Integral x(t)

This function performs a time-integration on the signal of a group of selected channels. Some modules in the analysis functions tab allow integration in the frequency domain. This function is useful for analyzing the signal coming from an accelerometer, especially if the speed and position must be analyzed in the time domain. There are many available techniques for performing time integration. We select the filtering approach, which has the advantage of being numerically stable. So, a simple first order or second order low-pass filter is used to perform a simple or double integration. The user must specify a cut-off frequency for the filter. We suggest using an *Fcut(Hz)* parameter that fits with the sensor used. A typical accelerometer sensor has a flat frequency response that starts at around 20 Hz. The interface of the *Integral x(t)* function is shown in the next figure:



The parameters are described in the following table:

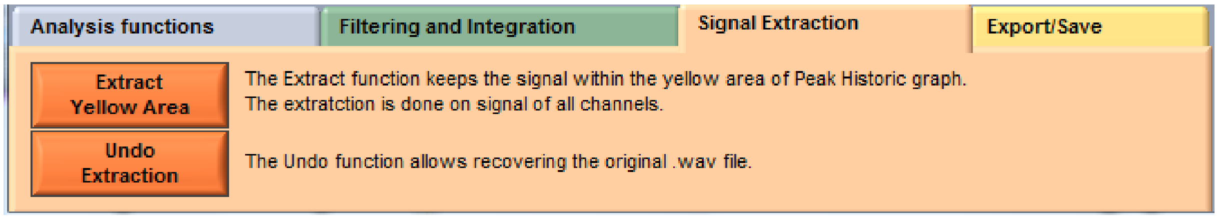
Integration Parameters	Description
	A simple or double operation can be done. A first (simple integration) or second order (double integration) filter is used following this parameter.
	This is the cut-off frequency for the filter used in the integration process. We suggest using an Fcut(Hz) parameter that fits with the sensor used. A typical accelerometer sensor has a flat frequency response that starts at around 20 Hz.

This control allows selecting the group of channels for which the integration will be done.

This button launches the integration operation.

9.3 Signal Extraction Tab

The signal extraction tab includes a function for extracting a portion of the signal. This feature is very useful for analyzing a portion of the wave file and for removing the beginning of a wave file after a filtering operation. Also, a saturated portion of the signal can be excluded for better analysis.



The extraction can be cancelled at any time, and the entire framework of the original wave file will be recovered. The extraction technique is based on the use of file pointers. This means that the start and end-points of the selected framework are sample numbers in the wave file. This way, the original wave file is never changed and an *Undo* operation is always available.

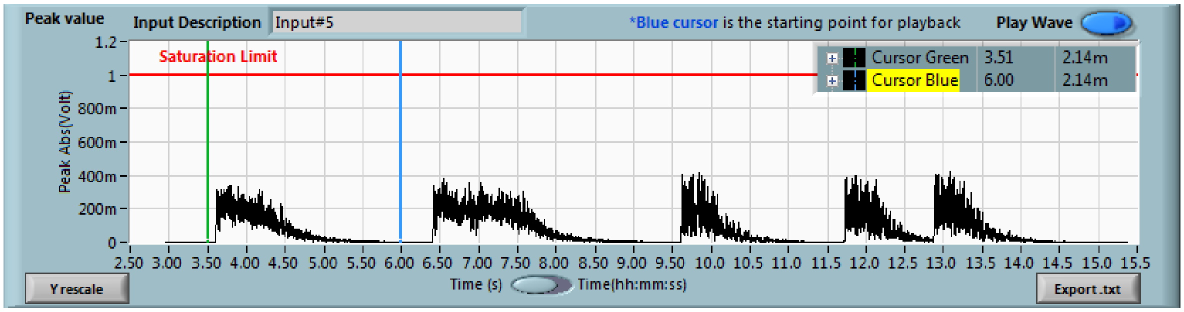
When an extraction has been done using the *Extract Yellow Area* function, all functions and modules of the post-processing interface will work with the new start and end-points of the framework. The save .wav function will also take into account the starting point and the ending points. After saving for instance, the new wave file will contain only the selected portion of the signal of the original wave file.

Here is the detail of the extraction functions:

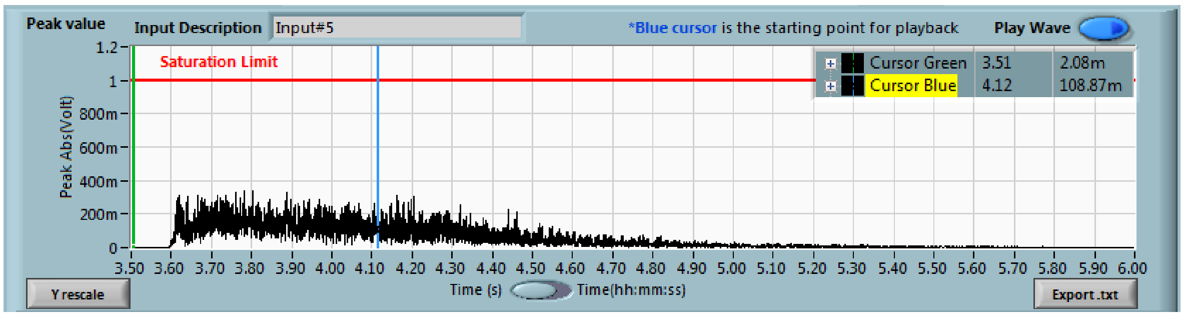
Functions	Description
Extract Yellow Area	This function changes the start and end-points of the framework following the yellow area on the peak value. The start and end points are changed for all channels. Use the Undo Extraction function to cancel the extraction.
Undo Extraction	This function cancels the extraction(s) and resets the start point to zero and the end point to the end of the wave file. For multiple extractions, the undo operation always retrieves the entire wave file signal and not the preceding selection.

9.3.1 Extraction example

The objective of this example is to keep the signal between time 00:03:50 and 00:06:00. First, the blue and green cursors are moved to the desired start and end-points:

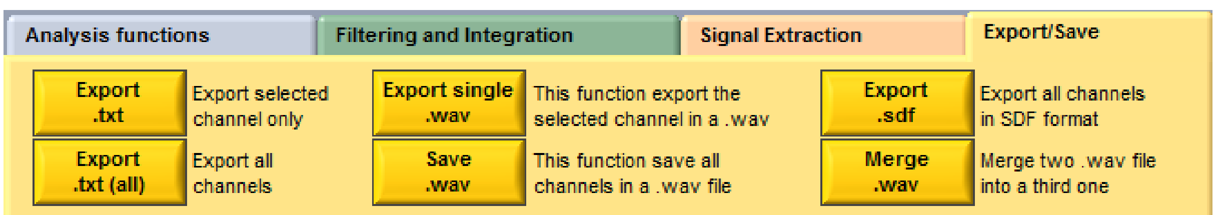


Then, the *Extract between cursors* button is used to launch the extraction. After extraction, the peak value graph looks like this:



9.4 Export/Save Tab

The *Export/Save* tab includes f functions to export the time signal data to a text file, a wave file or a SDF file (Standard Data File from HP).

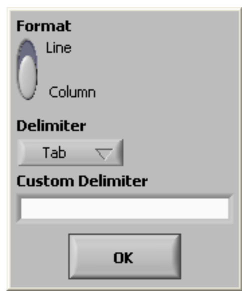


Here is a description of the *Export/Save* tab functions:

Functions	Description
	This function exports the selected channel to a text file. The exported signal is calibrated and the format of the text file can be adjusted through a dedicated dialog box*. This function saves only the selected framework.
	This function exports all channels in a text file. The exported signals are calibrated and the format of the text file can be adjusted through a dedicated dialog box*. This function saves only the selected framework.

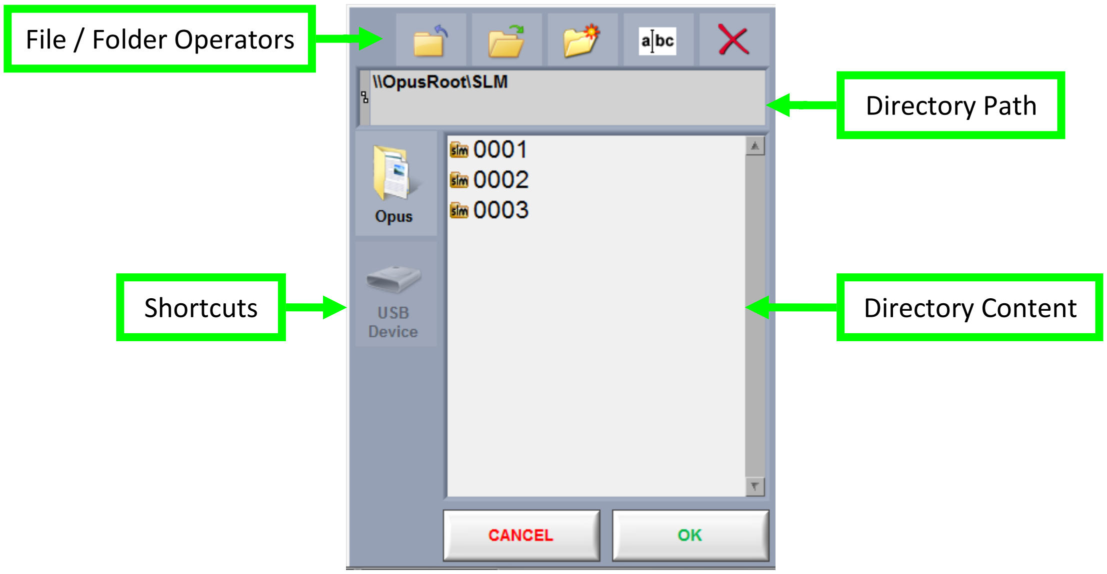
Export single .wav	This function allows saving the selected channel in a wave file. This function also saves the associated configuration file for further analysis with the post-processing interface.
Save .wav	This function allows saving all channel signals of the selected framework in a wave file. This function also saves the associated configuration file for further analysis with the post-processing interface.
Export .sdf	This function allows saving all channel signals of the current selected .wav file in a SDF format. The SDF format is a standard for the major spectrum analyzer manufacturers.
Merge .wav	This function allows merging two wave files into a third one. The resulting wave file is the sum of the two input file vs time.

*When a text format is chose for the export, the following dialog box is used to specify the format of the data in the file:



The data can be saved in line or column and the delimiter can be 1) a standard tab 2) a semi-colon (;) or 3) a custom delimiter. If the custom delimiter option is selected, the user has to specify the custom delimiter character.

10 Explorer Dialog



Explorer Window Controls and Indicators

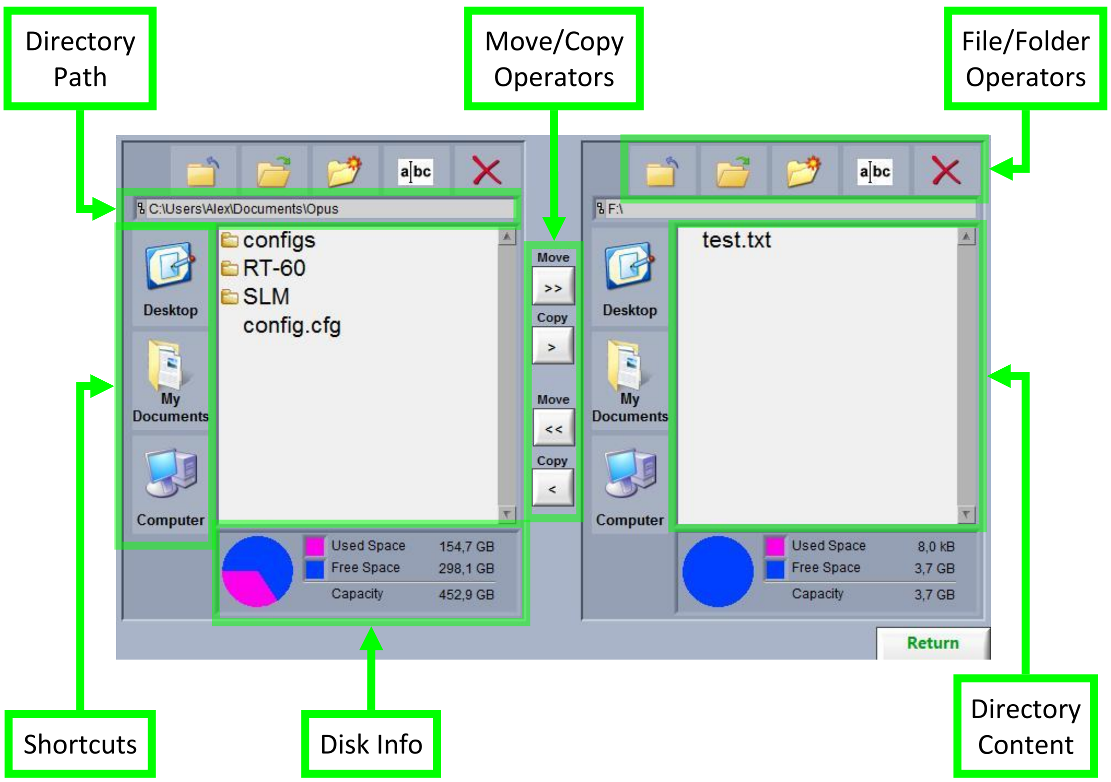
File/Folder Operators	• Go to parent directory • Open directory • Create new folder • Rename folder or file • Delete folder or file
Directory Path	Displays the path of the active directory.
Shortcuts	Accesses to **common directories**. When used on a stand-alone computer, those shortcuts are linked to: • Desktop • My Documents • Computer When used on a ***Concerto***, the shortcuts are linked to: • Opus Root • USB Device.
Directory Content	Displays the content of a directory and responds to common actions: • Single clicking on an element selects it. • Double clicking on a directory opens it.

11 File Manager

The File manager is used to perform most file operations:

- Navigate the directory structure
- Create folders
- Rename files and folders
- Move or copy files and folders from one place to another
- Delete a file or a folder

Although not very useful on a stand-alone computer, this manager is necessary on the *Concerto,* on which Windows explorer is unavailable. Its primary function is to allow the user to manage the *Concerto* directory structure and to export files and folders to a USB memory stick.



Directory Path	Displays the path of the active directory.
Shortcuts	Allows easy access to common directories. When the **File Manager** is used on a stand-alone computer, these shortcuts are linked to: • Desktop • My Documents • Computer When the **File Manager** is used on a ***Concerto***, the shortcuts are linked to: • Opus Root • USB Device.
Move/Copy Operators	Copies or moves a file or folder from a source to its destination.
File/Folder Operators	Allows user to: • Go to parent directory • Open directory • Create new folder • Rename folder or file • Delete folder or file
Directory Content	Displays the content of a directory and responds to common actions from the user: • Single clicking on an element will select it • Double clicking on a directory will open it • Dragging an element from one side to the other will copy it.
Disk info	Displays the disk information of the associated hardware.

Appendix 1: Concerto Hardware

Connections

Power on/off

Turn On	Press the trigger button located at the back of the unit This key has two (2) functions: 1. To turn the unit ON. 2. Start a measurement once the SLM Module is loaded After a few seconds, the **Opus Environment Interface** will appear.
Stand-by	The **stand-by** mode allows fast load time. • To put the unit on **stand-by**, click the **Turn Off** button. Note: The unit can be in stand-by for more than three days without recharging, provided batteries are fully charged prior to storage.
Shutdown	To **Shut down** the unit, click and **hold** the **Turn Off** button for five seconds.

Power Reset

If the Concerto happens to crash and it is not possible to take back the control, a power reset might be necessary. To complete the power reset, the three buttons on the front of the Concerto must be used.

Here is the procedure:

Step 1 ***Press and hold the Function, Enter and Down Arrow button for 5 seconds until the Concerto shuts down***

Step 2 ***Wait 5 seconds and press the power button***

Step 3 ***Wait 5 seconds and press the power button a second time to restart the Concerto from a power reset.***

Step 1

Press and hold to trigger the power reset

Step 2 and 3

Power-on

Press 2 times to restart from power reset

Inputs and Signal Processing Specifications (Embedded Signal Ranger MK3 DSP Board)

DSP Processor	Texas Instruments TMS320C6424
Inputs	4
Outputs	2
Linear Range	2 x (25-120 dBA or 30-130 dBA) + 2 x (25-120 dBA)
Conditioning	AC, DC, ICP (4 mA)

Physical (DAP Tech 9000 Tablet PC)

Operating system	Intel Atom E660T 1.3 GHz
Storage	16 GB SSD
Data Transfer	USB
Display	180 mm (7 inches) WVGA (800 x 480)
Dimensions	230 x 185 x 60mm (9.0 x 7.3 x 2.4 inches)
Weight	1350 g (2.96 lb)
Battery	2 x Li-ion battery, 7.4 V, 3100 mAh, (1 internal + 1 hot-swappable)
Power	10-20 VDC, 2A
Protection rating	IEC 68-2-32 method 1 (Multiple 1m drops on concrete) IP67 (Rain, Humidity, 1 meter immersion) MIL-STD-810F method 506.4 procedure I (windblown rain) Humidity: 95% non-condensing Temperature: MIL-STD-810-F (-20 °C ... +50 °C); Vibration: MIL-STD 810E 514.5