Soft dB

ALTO Instrument

Spectrum Analyzer, SLM and Datalogger

User Manual, version 2.7c

ROHS COMPLIANT

July 2015

Contents

1.0 Introduction

1.1 About this manual

This user manual explains in detail the use of the Alto instrument. The Alto can work with a standard PC sound card or with the Soft dB professional acquisition unit. This manual describes the operation of the Soft dB acquisition unit but the information is also useful for the PC sound card configuration. Note that the SLM (Sound Level Meter) function and the generator are not available with the PC sound card.

Note: Be sure to install the USB driver before launching the Alto software. See section 2.2 for more details.

1.2 Features

The Alto instrument from Soft dB is a **2-Channel Spectrum Analyser**, a **Class 1 1-Channel SLM** (Sound Level Meter) and a **1-Channel Datalogger.** The Alto instrument is suitable for acoustic and vibratory measurements. Since it includes three different instruments in one, it can be used for a very large range of applications.

The Alto instrument has been designed as a high quality instrument, available at a reasonable price. This has been achieved by careful selection of electronic components and by the design of an efficient communication architecture. Combined with an easy-to-use PC interface[1], the Alto is a great alternative to the expensive vibro-acoustic systems of most manufacturers.

1.3 Hardware Features

The following are the main hardware features of the Alto instrument. More hardware specifications can be found in Appendix B.

- 2 analog inputs with BNC connectors and status LEDs
- Input dynamic range (+-25 mV to +-3 V)
- 3 input types: Direct DC, Direct AC and ICP
- 2 analog outputs with a white noise generator (±1V)
- Anti-aliasing filter on outputs and inputs

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[1] Note: The PC used with the Alto must have a minimum memory (RAM) of 512 MB.

- Sampling rate up to 43 kHz per channel
- SNR: 85 dB (inputs) and 88 dB (outputs)
- SNR + Input Gain: 110 dB
- USB 2.0 full speed interface
- Power supply: 24Volt-0.5A or 12Volt-2A external AC/DC power supply included
- LCD screen for level display (more recent units only)
- Embedded battery backup for up to 4 hours of autonomy (more recent units only)
- Dimension (LxHxD): 77 x, 217 x 274 mm (3 x 8½ x 10¾ in.)
- Weight: 1.8 kg (4 lb)
- Dust-tight and splash-proof

1.4 Software features

The following table presents a summary of the software features for each software module available with the Alto instrument:

Feature	Spectrum analyzer	SLM	Data Logger
Precision integrating sound level meter with SPL, Leq, Lmax, Lmin, SEL, Lpeak and Lpeak(min/max) measurements (compliance with the standard IEC 61762 – class 1)		X	
Total dynamic range between 28 to 125 dB (with a 50mV/Pa microphone) through 3 selectable ranges	X	X	X
Slow and Fast detectors		X	
Frequency Weighting: A, C & Z		X	
Peak Frequency Weighting: A, C & Z		X	
Historic graph for all global value indicators (SPL, LEQ and Peak)		X	
1/3-octave frequency analysis through a real-time filter bank		X	
Octave, 1/3-octave and 1/12-octave analysis through a frequency filter bank applied on a high resolution narrow	X		X

band spectrum			
Auto-store for a long period or a continuous measurement with a complete post-processing module		**X**	**X**
Sensor calibration module	**X**	**X**	**X**
Time signal monitoring in real time	**X**	**X**	**X**
Real-time narrow band spectrum analysis (FFT)	**X**	**X**	
Real-time narrow band inter-spectrum analysis (S12 and S21) and cross correlation (coherence, frequency response H_1, H_2 (magnitude and phase))	**X**		
LN statistics and histogram tables		**X**	**X**
Waterfall analysis			**X**
Real-time acquisition/data-logging using the standard wave file format		**X**	**X**
Real-time dynamic and saturation monitoring of all input channels	**X**	**X**	**X**
Trigger functions with an accept/reject mode for impact measurements	**X**		
White noise generator	**X**		**X**
Automatic white noise mute with delay for RT60 measurements			**X**
Automatic RT60 computation module in octave			**X**
Export function in text file format compatible with spreadsheet software such as Excel	**X**	**X**	**X**

2.0 First use

2.1 Unpacking and inspection

Your Alto is shipped in a protective package. Please, verify that the package contains the items listed below:

- The Alto unit
- A USB cable
- A 24 Vdc power-supply
- An installation CD

2.2 Software and USB driver installation

> **!** **Do not connect the Alto unit to your PC before installing the software and the Alto USB driver *(Windows does not automatically recognize the Alto unit, and the wrong driver could be installed).***
>
> **Follow each step below precisely to ensure proper installation of the software and drivers on your PC.**
>
> **See Appendix A for more details.**

STEP 1: Installation of the Alto Software

1) If a CD is provided with the instrument, insert it in the CD drive of the computer.

   You can also download the latest release of the Alto software from Alto page of the Soft dB website :

   http://www.softdb.com/acoustic-products-alto.php

2) Launch the installer and follow the installation procedure. By default, the Alto software is installed in the **C:\Program Files\Alto** directory. Do not start the Alto software after this step. The USB driver has to be installed beforehand first (Step 3).

STEP 2: Power the Alto unit

Connect the provided power supply to the Alto unit using the corresponding connector at the rear of the unit. The Board Status LED on the rear of the Alto unit should turn yellow.

STEP 3: Connection of the Alto unit to the PC

Connect the Alto unit to a computer using the USB cable provided.

Windows will then ask for the Signal Ranger Mk2 Drivers. The Signal Ranger Mk2 is the DSP (Digital Signal Processor) board on which the Alto system is based.

The Signal Ranger Mk2 drivers are located in the **C:\Program Files\Alto\Driver** directory.

Follow the steps in Windows to select the appropriate directory and continue the installation.

! Windows may display a warning message indicating that the driver is not certified. Ignore it and continue with the installation.

At this point, the Board Status LED on the rear of the Alto should turn green and the software can be launched (see section 4.0).

Note: If you have Windows Vista, the following step must be performed to ensure compatibility:

Go to the **C:\Program Files\Alto** directory and right click on the file **Alto.exe** and then select the option "*Run this program as an administrator"* in the Compatibility tab**:**

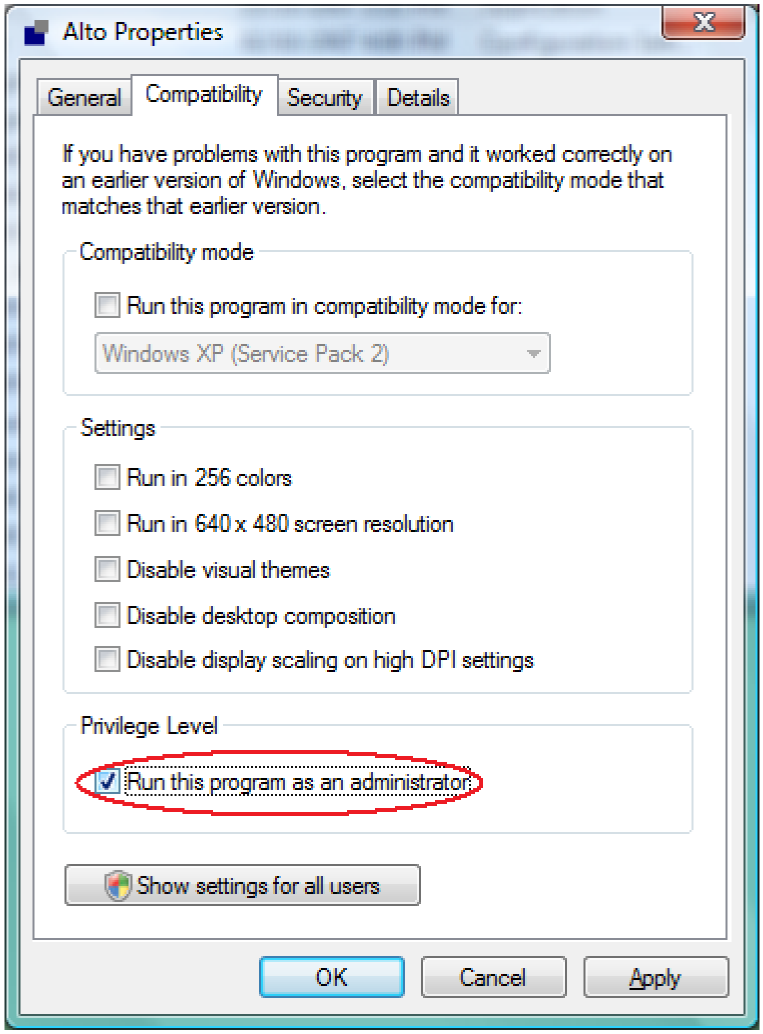


Figure 1: Compatibility option for Windows Vista

3.0 Hardware Description

The Alto system is based on an external USB acquisition board. This acquisition board includes an ICP conditioning board. The metal box that contains the USB acquisition board is equipped with 4 BNC connectors (2 inputs/2 outputs). The internal switches on the conditioning board allow the selection of Direct DC, Direct AC or ICP input acquisition paths. Each input and output has a LED to indicate saturation or a low-level problem.

Figure 2: Alto Hardware Unit (front)

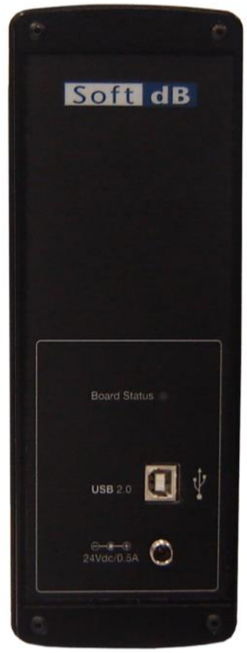


Figure 3: Alto Hardware Unit (rear)

3.1 Technical details

Here are the hardware technical details:

- External board with a USB link that allows real-time acquisition
- 24Vdc/0.5A or 15Vdc/2A power supply connector depending of the model[2]
- LCD screen for level display (more recent units only)
- Embedded battery backup for up to 4 hours of autonomy (more recent units only)
- 2 analog inputs/outputs with BNC connectors and status LEDs
- Sampling rate of up to 43 kHz per channel
- SNR: 85 dB (input) and 88 dB (output)
- SNR + Input Gain: 110 dB
- 3 input types: Direct DC, Direct AC and ICP
- Input dynamic range (+-25 mV to +-3 V)
- Output dynamic range (+-1V)
- Anti-aliasing filter on outputs and inputs

Note: See Appendix B for more input/output specifications.

[2] The first Alto model (without screen and battery) uses a 24V power supply. The more recent model (with a LCD screen and battery) uses a 15V power supply. This model can also be power with 12V but the charger may fail to charge the battery.

3.2 LED meaning

The LED colour and description are presented in the following table:

	Colour meaning			
LED	**Black**	**Red**	**Yellow**	**Green**
Input	Not used	Saturated	Low Level	Used
Output	Not used	---	---	Used
Measuring/Overload	Acquisition OFF	Input(s) saturated	---	Acquisition ON
System Status	Board OFF	---	Software OFF	Software ON
Board Status (rear)	Board OFF	Board Error	Board ON	PC connected

Additional note for the Board Status LED:

- A **Green LED** indicates that the computer is connected to the Alto unit and the USB driver is operational.
- A **Yellow LED** indicates that either the computer is not connected to the Alto unit, or the USB driver is not operational. See **Appendix A** for more information.
- A **Red LED** indicates that the Alto unit has detected an error. Try switching off the Alto power supply for 5 seconds, and then back on. If the Board Status LED remains red, contact Soft dB.

4.0 2-Channel Spectrum Analyzer

The first module of the Alto instrument is a 2-channel spectrum analyzer. This module has been designed for advanced sound and vibration spectrum analysis. In addition to the auto spectrum, the cross-spectrum and the 1/n octave functions, this instrument includes a trigger and output source generators.

The following figure shows the main interface of the 2-channel analyzer module. This module is automatically selected at start-up.

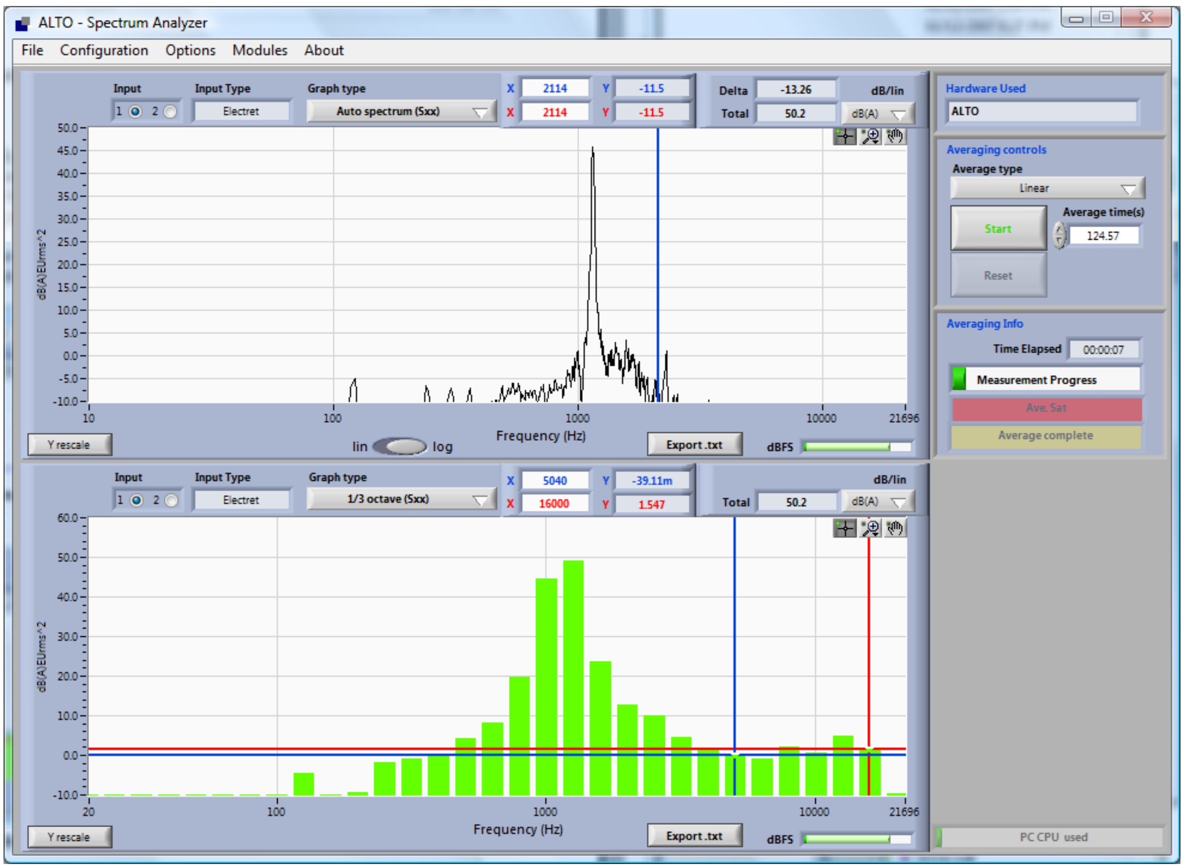


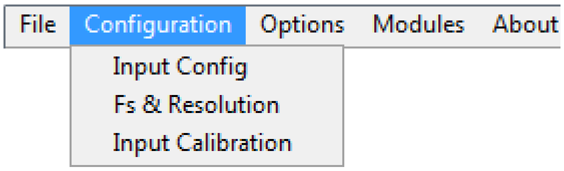
Figure 4: Main interface of the 2-channel spectrum analyzer

The configuration of the analyzer can be performed through the *Configuration* menu while the *Options* menu allows some advanced features of this portion of the instrument to be accessed.

The following sections will present all the features of the spectrum analyzer, in detail.

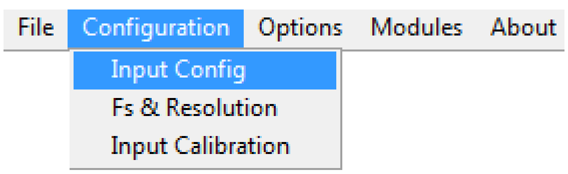
4.1 Instrument Set-up

The entire configuration of the instrument is done through the *Configuration* menu:



Each function of this menu is explained in following sections.

4.1.1 Input Configuration



The *Input Config* function of the *Configuration* menu allows the inputs to be set through the following dialog box:

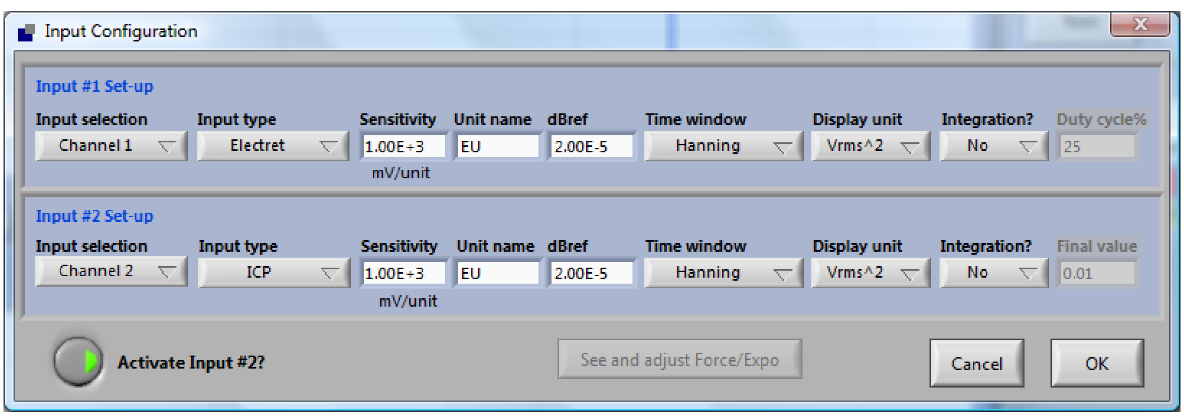
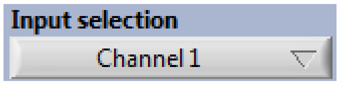
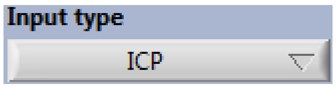
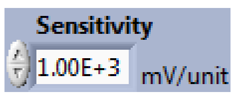
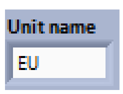
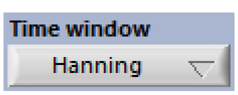
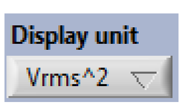
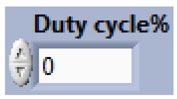
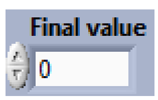
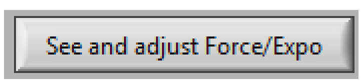
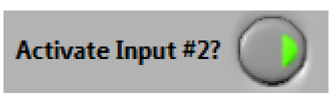


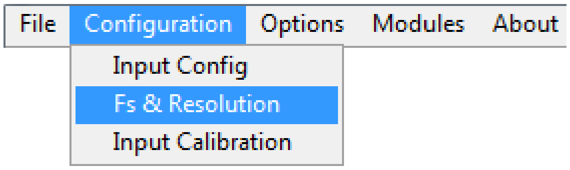
Figure 5: Input configuration dialog box

Parameter/control	Description
	The channel number for each input of the spectrum analyzer can be selected through this control. The Alto hardware has only two input channels. However, the Alto software can work with the Tenor hardware that has 8 input connectors (see http://www.softdb.com/a-acous_20_3_4.html for more information about the Tenor). Note: in any case, the first input and second input can not be the same physical input channel.
	There are three input types: *Direct DC*: This input type can measure the DC signal (no high pass filter). *Direct AC*: This input type uses a high pass filter at 0.5 Hz and is suitable for general AC signal measurements. *ICP:* This input type is for an integrated circuit preamplifier (ICP). It has a high pass filter at 0.5 Hz. When this type of input is selected, a compatible ICP/DeltaTron microphone or accelerometer can be connected directly to the BNC connector.
	This is the sensitivity of the input in mV/Unit. The sensitivity is used by the interface to present a calibrated signal. The sensitivity can be entered manually if the value is known, or determined automatically with the *Input Calibration* function (see section 4.1.3 for more information on this function).
	This text control can be used to specify the name of the unit. The unit name is used by the acquisition system to label 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).
	This is the reference value for the dB calculation. For an acoustic measurement, the reference is generally 2E-5 Pa. Each input has its own reference. Note: this value can also be adjusted in the input calibration dialog box.
	This is the time window used for the FFT computation. If the time window “Force” is selected for input #1, the second input time window will be automatically set to “Expo”. This particular set-up is used exclusively for impact measurements (see section 4.4.2 for more details about impact measurement).
	This control allows the selection of the unit format. Time signal, Frequency Response and Coherence graphs are not affected by this control. The following list shows each format type and presents the result for a 2-Volt sine wave and a 2 Hz frequency resolution narrow band spectrum: Voltage: 1) Vrms: 1.414 Vrms 2) Vpk: 2 V Energy:

	3) Vrms^2: 2 Vrms^2 4) Vpk^2: 4 V^2 Spectral Density: 5) Vrms/rtHz: 1 Vrms/Hz^(1/2) 6) Vpk/rtHz: 1.414 V/Hz^(1/2) 7) Vrms^2/Hz: 1 Vrms^2/Hz 8) Vpk^2/Hz: 2 V^2/Hz
	The *Integration* parameter allows a simple or double integration in the frequency domain to be applied for an acceleration measurement. For the simple integration option, the power spectra are divided by iω. For the double integration option, the spectra are divided by $-\omega^2$.
 or 	These two controls are available only if the first input time window is set to ”Force”. The duty cycle (%) of the “Force” window and the final value of the “Expo” window can be set through these controls. These parameters are used for impact measurement. Note that the button “*See and adjust Force/Expo”* can be used to perform a visual adjustment of these parameters (see section 4.4.2 for more details about impact measurement).
	This button “*See and adjust Force/Expo”* can be used to perform a visual adjustment of the time window parameters for the impact measurement case (see section 4.4.2 for more details about impact measurement).
	This control allows a second input to be added to the spectrum analyzer. Then, the instrument becomes a 2-channel analyzer. If the “*Activated Input #2?* “ option is not set, none of the Input #2 set-up controls are accessible.

Note: When using ICP microphone, the sensitivity given by the manufacturer doesn’t usually take into account the ICP preamplifier attenuation, which is usually less than 0.5 dB and typically around 0.3 dB. In order to enter the sensitivity manually you will need to calculate the sensitivity viewed at the unit input. For instance, for a given manufacturer sensitivity of 50mV/Pa and a known ICP preamplifier attenuation of 0.5 dB, the sensitivity entered in the software will be 50mV/Pa*10^(0.5/20) = 53 mV/Pa. A better solution would be to evaluate the sensitivity within the calibration Module. The resulting input sensitivity will consider both the microphone and its preamplifier so that you don’t have to worry about the preamplifier amplification.

4.1.2 Frequency Span & Resolution configuration



The "*Fs & Resolution*" function of the *Configuration* menu allows the setting of the frequency span and the number of lines of the narrow band spectrum graphs through the following dialog box:

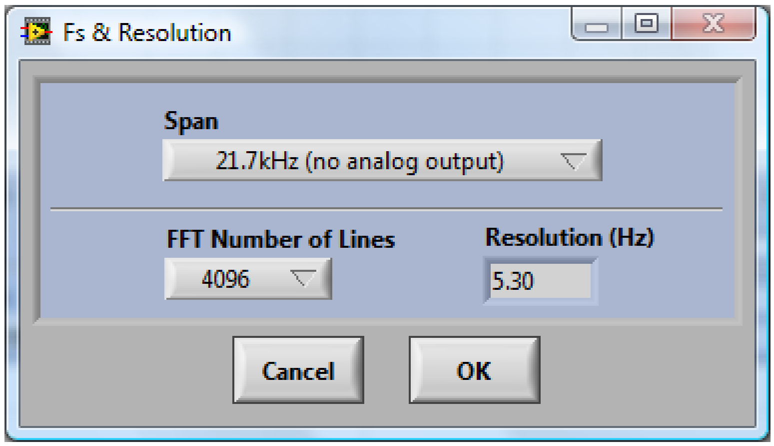
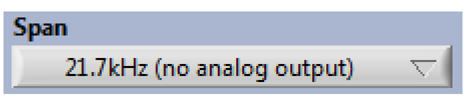
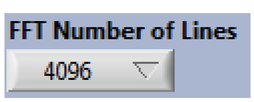
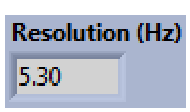
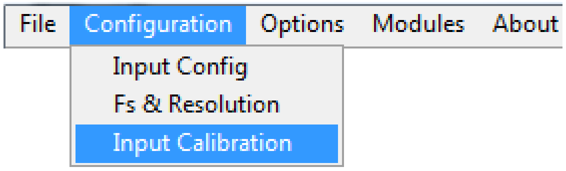


Figure 6: Sampling Frequency and Resolution dialog box

Parameter/control	Description
	The *Span* control allows the selection of the bandwidth of the spectrum graphs. The first selection (21.7 kHz) does not allow a signal to be generated with the two outputs. Special frequency response compensation is used for the two first span selections to extend the span to the audio bandwidth (see Appendix B for input frequency response graphs).
 	The "*FFT Number of Lines*" control allows the selection of the number of lines and the resolution of the spectrum graphs. To achieve a good resolution, the number of lines of the FFT must be increased. The number of lines for the FFT can be adjusted from 512 to 4096. Following the "*FFT Number of Lines*" selection, the *Resolution (Hz)* indicator shows the width of each band of the narrow band spectrum graphs.

4.1.3 Input Calibration Module



This function can be used to automatically determine the input sensitivity. Note that a microphone or accelerometer calibrator is required. The interface of the calibration function is presented in Figure 7.

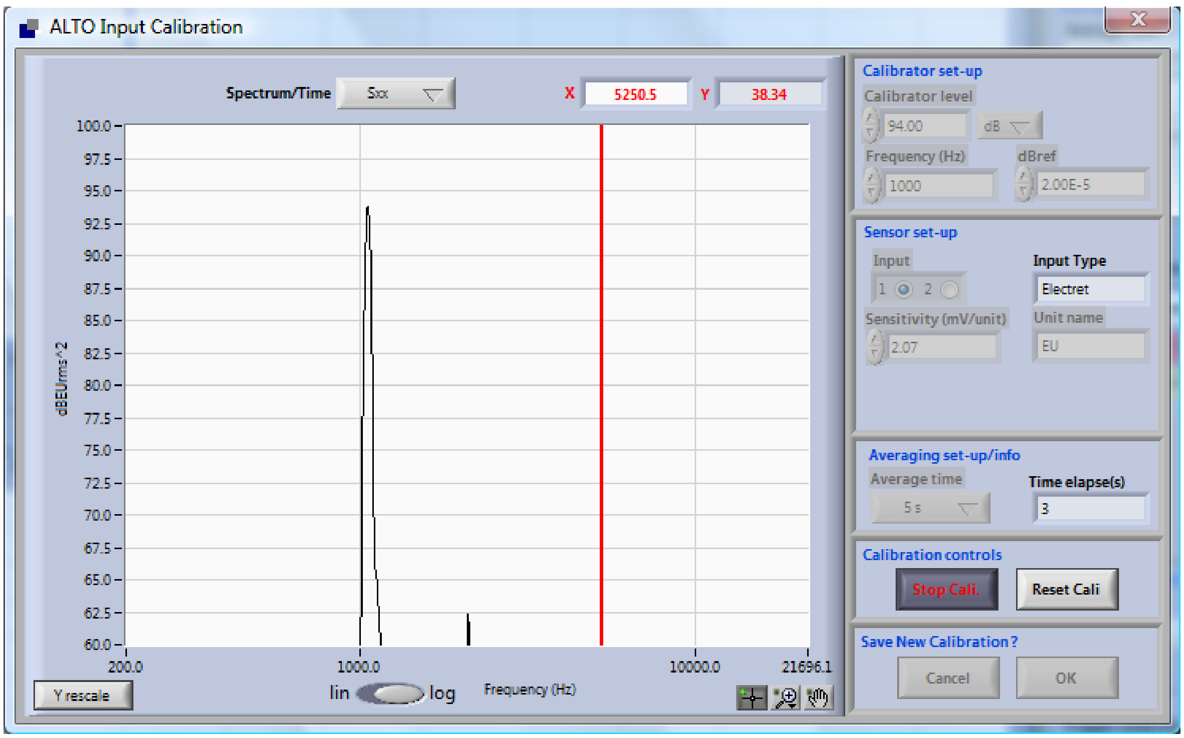
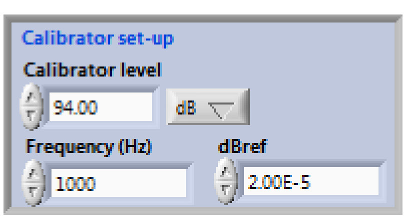


Figure 7: Input calibration interface

Before starting the calibration measurement, the following calibrator parameters must be set:



Calibration level: The calibrator level can be defined in dB or in a linear value. Use the little control on the right to specify if the calibrator level is defined in dB or linear.

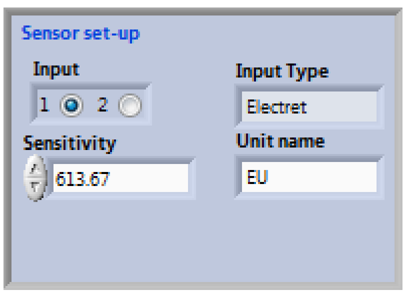
Frequency (Hz): This is the calibrator emission frequency (typically 1000 Hz in the case of a microphone).

dBref: This is the reference value used for the dB computation (typically 2E-5 for a microphone). Note: this value can also be adjusted in the input configuration dialog box.

When the calibrator set-up has been carried out, the average time parameter must be set. A short average of 5 seconds is usually appropriate.

Then, the averaging process can be launched with the button "*Start Cali.*"

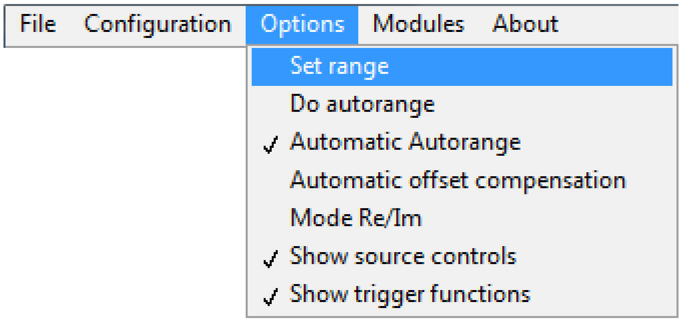
At the end of the averaging process, the interface computes the required sensitivity to reach the calibrator level at the specified frequency.



Click the *Ok* button to accept the new sensitivity value and to return to the main interface.

4.1.4 Input Range Set-up

The spectrum analyzer has an input gain that must be adjusted to optimize the input range for the measured signal. By default, the spectrum analyzer uses the option *Automatic Autorange* (this option can be changed in the *Options* menu). With this option, the analyzer performs an auto range at the beginning of each averaging process. However, the user can disable the *Automatic Autorange* option and force a specific range set-up with the following function:



The “*Set range”* function of the *Options* menu launches the following interface:

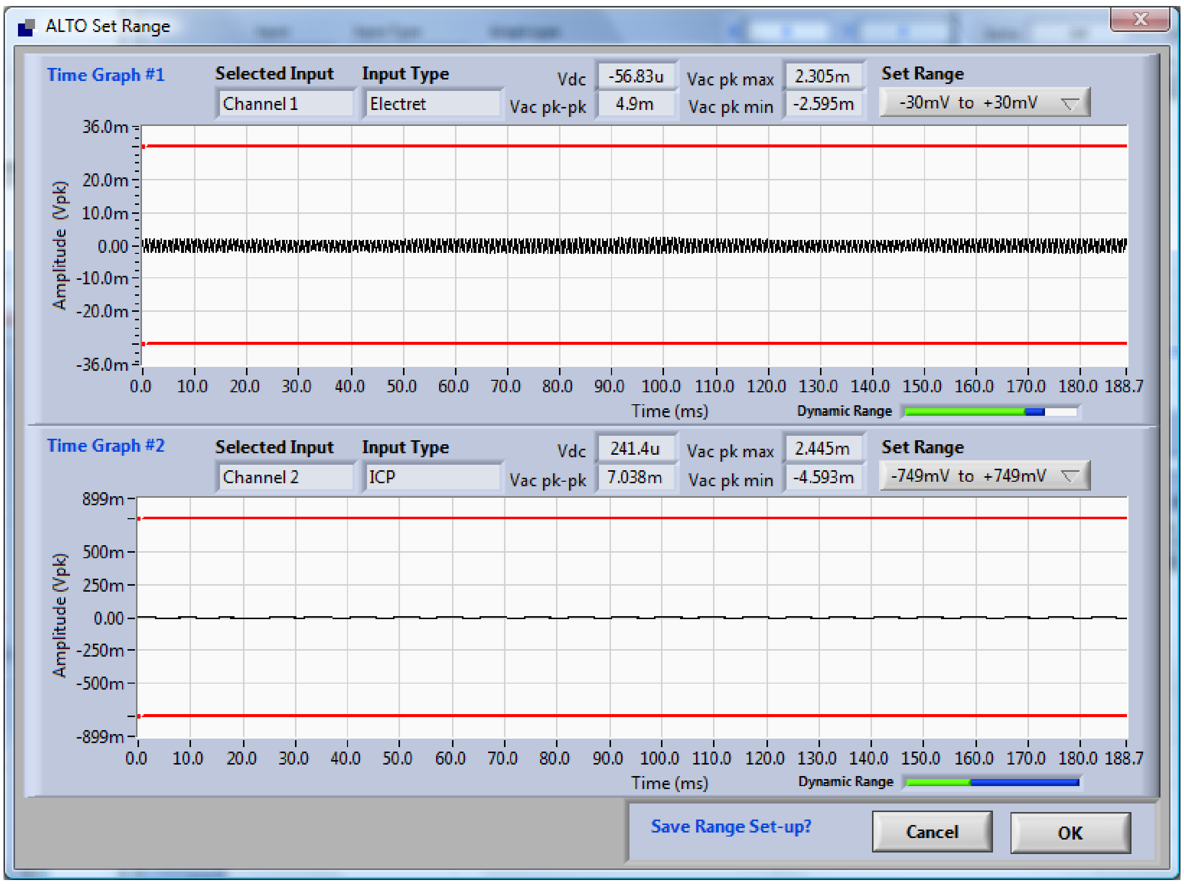
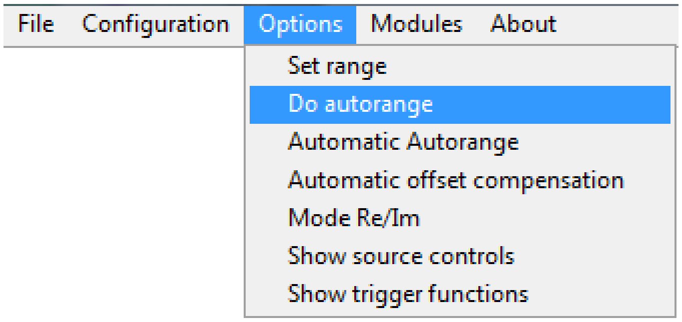


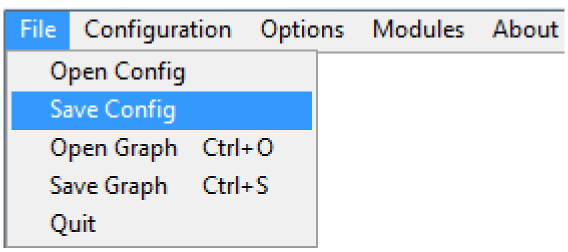
Figure 8: Set input range interface

The interface presents the time graph for both inputs and the negative and positive limits of the current selected input range shown by the red lines. The controls “*Set Range”* allows the selection of the adequate input range. The selection “*Automatic Autorange”* can be used to return to the automatic auto range mode.



If the "*Automatic Autorange"* option is not selected, the "*Do autorange"* function in the *Options* menu can be used to automatically adjust the input range. It means that the optimal range for the current input signal will be found and fixed until the user manually changes the input range.

4.1.5 Save & Recall Instrument Configuration



The entire set-up of the spectrum analyzer and the datalogger can be saved in a .cfg file and recalled from one measurement session to another. All graph selections of the main interface and all instrument options are saved in the configuration file.

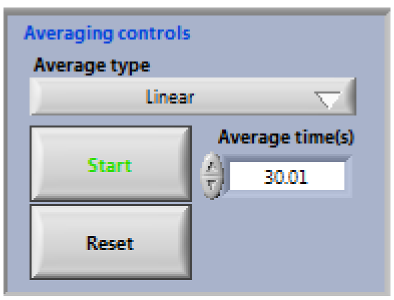
Note: the configuration files used with the spectrum analyzer are not compatible with the configuration files of others modules. The configuration files of the spectrum analyzer have the .cfg extension.

4.2 Measurement Procedure

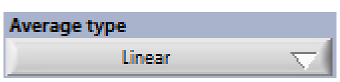
The following section explains in detail the standard procedure to complete a measurement/averaging process. Be sure that the configuration of the instrument has been done properly before launching the averaging process (see section 4.1 for more details).

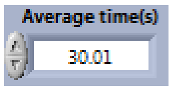
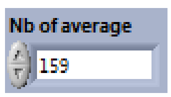
4.2.1 Launching the Averaging Process

These controls allow the selection of the average type and the launching of the averaging process. During the averaging, the Alto performs the acquisition of the input #1 and input #2 (if both channels are used). Then, the spectrum averages are computed and presented by the interface.

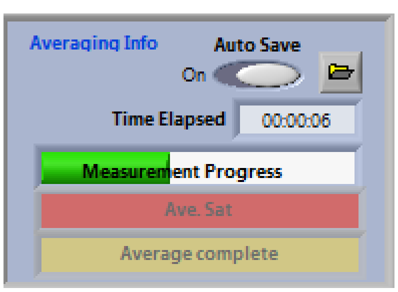


The following table explains the averaging controls:

Control	Description
	This control allows the selection of the average type. There are three types of average: 1) Linear: With this type of average, the value of the "Average time(s)" control is used to determine the time length of the average. At the end of the averaging process, the acquisition stops by itself. 2) Expo.: With this type of average, the value of the "Constant time(s)" control is used as a time constant for exponential averaging. The acquisition never stops by itself. The user must use the "Stop" control to stop the acquisition process. 3) Accept/Reject: This is like the Linear average type but the user must confirm each new signal block during the averaging process. With this type of average, the "*Nbr of averages"* control is used to specify the number of signal blocks for the average. During the averaging process, the acceptance or rejection is done through these controls:

	 This average mode is useful for impact measurements since it allows the removal of a bad impact from the average (see section 4.4.2 for more details about impact measurements).
 or  or 	This is the "***Average time(s)"*** control used to set the average time length for the linear averaging mode. This is the "***Constant time(s)"*** control used to define the time constant of the exponential averaging mode. This is the "***Nbr of averages"*** control that allows the number of time blocks for the average to be specified when the "*accept/reject*" average type is selected or/and when the trigger is activated (see section 4.4 for more details about trigger operation).
 	The "*Start"* button allows launching the averaging process. When pressed, this button switches to a *Stop* button that allows stopping the average. Note: if the "*accept/reject"* mode is selected, the user must use the "*Abort average"* control instead of the "*Stop"* button (see section 4.4.2).
	This button allows the averaging process to be reset when the linear or exponential mode is selected.

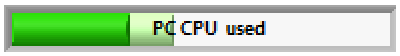
During the averaging process, the following indicators allow information to be obtained about the averaging:



The *Ave. Sat* indicator lights up if saturations occur on input #1 or input #2.

The *Auto Save* option can be set to On to allow an automatic save of all graphs at the end of the average. If no file has been specified yet, the interface asks for the name at the end of the average. If the name has been defined, the interface adds a number at the end of the current file name and automatically saves the graphs.

During the acquisition/averaging process, the “*PC CPU used”* indicator (lower right corner of the main interface) provides visual information if the acquisition and the processing are in real time:



This meter shows a light green bar and a green bar. The green bar is the current CPU usage while the light green bar is the maximum CPU usage from the beginning of the averaging process. If the CPU of the PC is not fast enough to allow real-time acquisition and processing, the bar turns red.

The 1/3-octave and 1/12-octave spectra require a lot of CPU power from the PC. We suggest avoiding the monitoring of this type of graph during the averaging process if real-time acquisition is required. Note that the 1/3-octave and 1/12-octave spectra can be selected and analyzed at the end of the averaging process to avoid this problem.

4.2.2 Monitoring the Results

The main interface of the spectrum analyzer presents two sets of graph, controls and indicators that allow the monitoring of the measurement results:

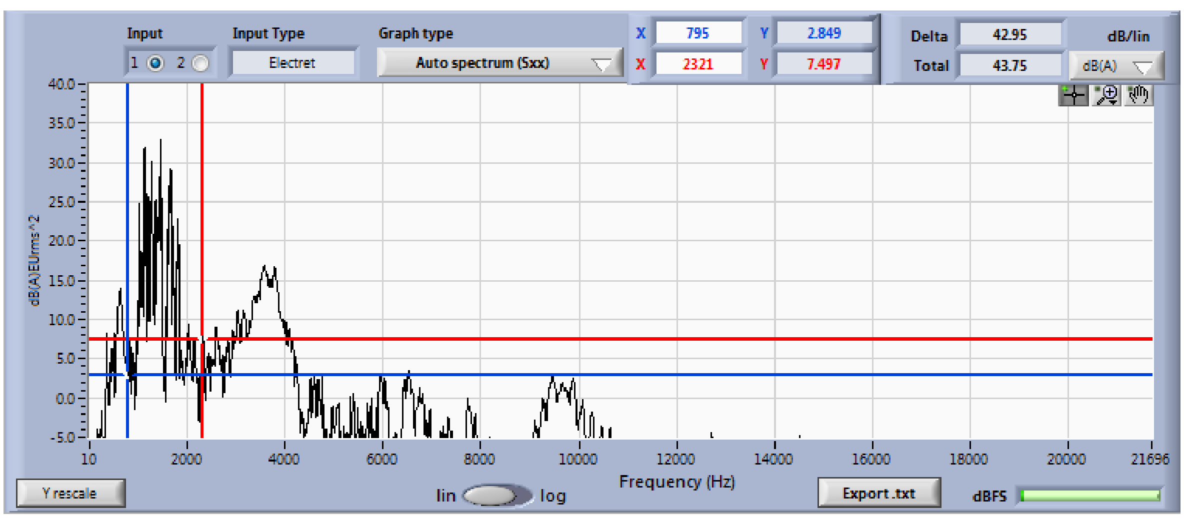
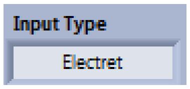
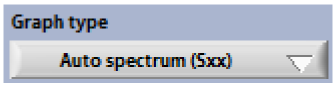
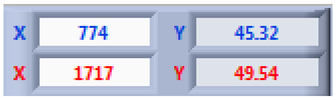
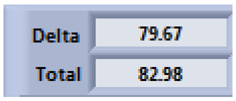
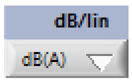
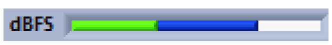


Figure 9: Graph and controls to monitor measurement results

The following table explains the controls and indicators:

Indicator/control	Description
	This control allows the selection of the input signal shown in the graph. The input #2 is available only if the second input has been turned on in the input configuration dialog box.
	This indicator shows the input type of the selected input.
	This control allows the selection of the type of graph: X is the input selected by the *Input* control Y is the other channel (if input=1 → y=2) 1) Time Signal: Time signal of the last acquisition block presented in “Unit” 2) Windowed Time Signal: Time signal weighted by the time window 3) Instant spectrum (Sxx): Instantaneous auto spectrum of the signal x 4) Auto spectrum (Sxx): Average auto spectrum of the signal x 5) 1/12octave (Sxx)*: Average 1/12-octave spectrum of the signal x 6) 1/3octave (Sxx)*: Average 1/3-octave spectrum of the signal x 7) Octave (Sxx)*: Average octave spectrum of the signal x 8) Cross spectrum (Sxy) Mag**: Magnitude of the cross-spectrum between the reference x and the signal y 9) Cross spectrum (Sxy) Phase**: Phase of the cross-spectrum between the reference x and the signal y 10) Fr. Response H1 Mag**: Magnitude of the transfer function y/x calculated with Sxy/Sxx 11) Fr. Response H1 Phase**: Phase of the transfer function y/x calculated with Sxy/Sxx 12) Fr. Response H2 Mag**: Magnitude of the transfer function y/x calculated with Syy/Sxy 13) Fr. Response H2 Phase**: Phase of the transfer function y/x calculated with Syy/Sxy 14) Fr. Response H3 Mag**: Magnitude of the transfer function y/x calculated with (H1+H2)/2 15) Fr. Response H3 Phase**: Phase of the transfer function y/x calculated with (H1+H2)/2 16) Coherence: Coherence between the signals x and y calculated with H1/H2 * The octave, 1/3-octave and 1/12-octave spectra are computed with a bank of frequency filters applied on the Sxx narrow band spectrum. We

	suggest the use of the maximum number of lines (see Fs & Resolution dialog box) to obtain best results at low frequencies. ** It can also be: Re and Im instead of Mag and Phase following the option (*Mode Re/Im*) in the menu *Options*.
	These indicators present the X and Y values of the blue and red cursors.
	These indicators present the global level (Total) of the entire spectrum and the global level in between the cursors (Delta) in dB, dB(A) or lin following the *dB/lin* control. If the time signal graph type is selected, only the rms unit of the current time signal is presented on the total indicator.
	This control allows the selection of the amplitude format for the spectrum graph.
	This control applies an auto scaling on the Y-axis of the current selected graph.
	This control allows the use of a linear or a logarithmic scale for the X-axis. This control is available for the narrow band spectrum graph type only.
	This indicator presents the dynamic used by the selected input. This value is the peak value in dB (the reference is the full scale of the input). The green bar is the instantaneous peak value while the blue one is the maximum peak value since the beginning of the average. Note that the reset operation during the average resets this indicator.
	This button allow the data of the selected graph to be exported in a text file format. This file can be used in a standard spreadsheet program such as Excel. The text file contains a header that describes the content of the file. For the auto spectrum Sxx, the text file includes a correction factor for total and delta total computations. This value has been added to allow compensation for the time windows correction applied on the amplitude values of the spectrum. The linear value of the global level must be divided by this correction factor.

Both graphs of the main interface can be customized with a right click over the curve. Then, a menu appears as illustrated in the following figure:

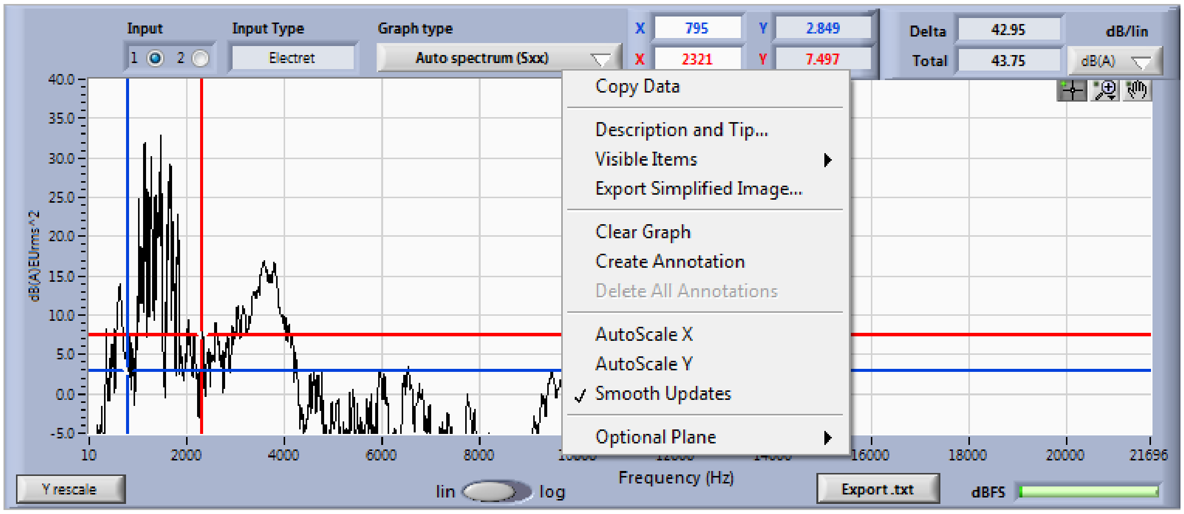


Figure 10: Option menu of the spectrum analyzer graph

When the *AutoScale* option for X or Y is not selected, the user can change the minimum and the maximum values of the scales as shown in the following figure:

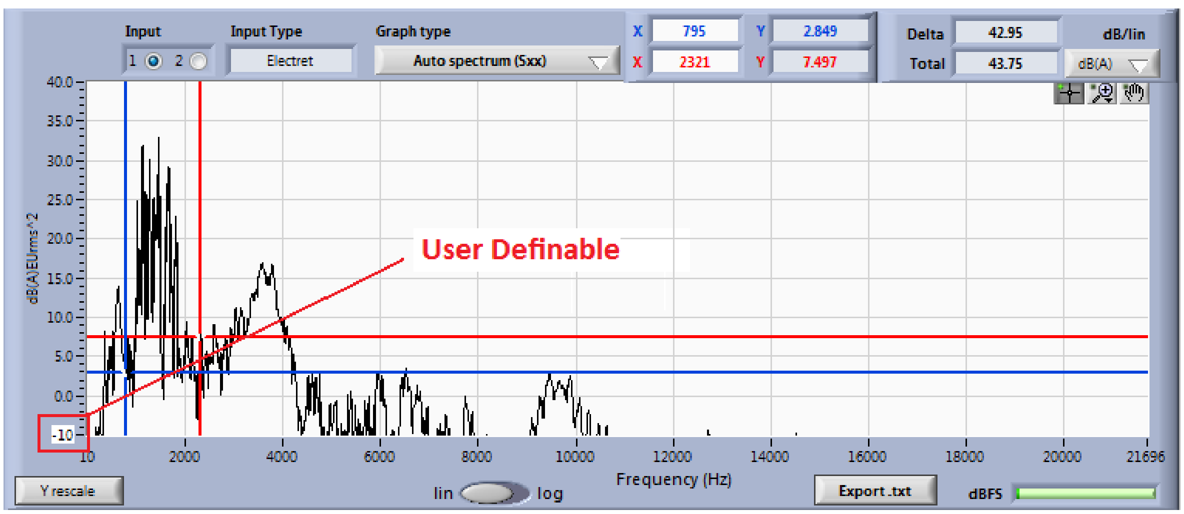
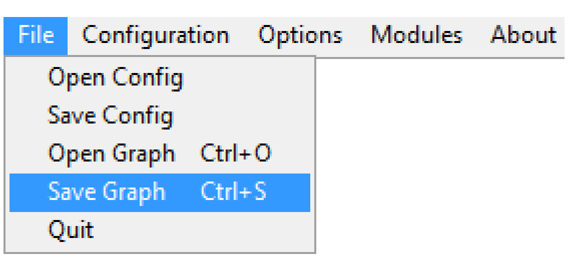


Figure 11: Editing the Y-scale minimum value

4.2.3 Export and Recall Results

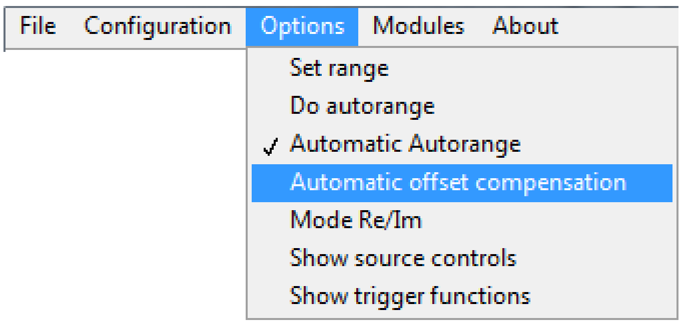


All results can be saved in a .gph file using the *Save Graph* function of the *File* menu. All results for both inputs are saved and can be recalled for post analysis.

4.3 Advanced Options

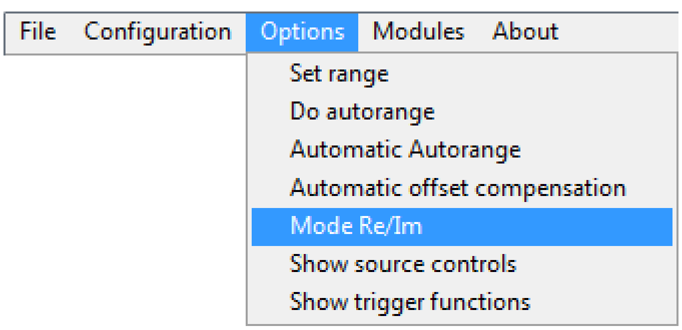
This section describes two advanced options that can be set through the *Options* menu.

4.3.1 Automatic Offset Compensation



With this option activated, the offset of each acquisition block is evaluated and subtracted before performing a computation of the spectrum. This way, the DC component is not present in the spectra. This option is recommended for most cases, given that acoustic and vibratory measurements never have a DC component. This option can be set to OFF for DC measurements. In this case, the direct DC input type must be selected.

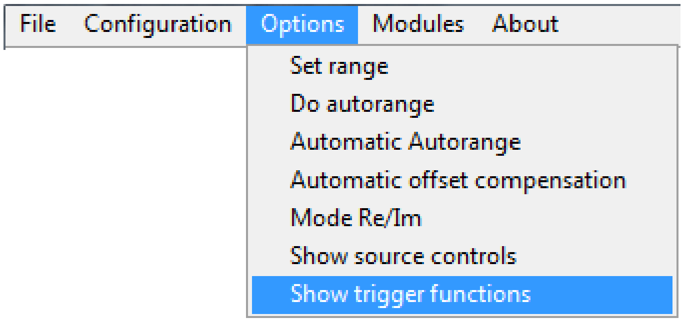
4.3.2 Mode Re/Im



This option manages the displays of FRF and Sxy spectrum graphs. These graph types are in complex values and can be presented in magnitude/phase or in Real/Imaginary format.

4.4 Trigger Functions

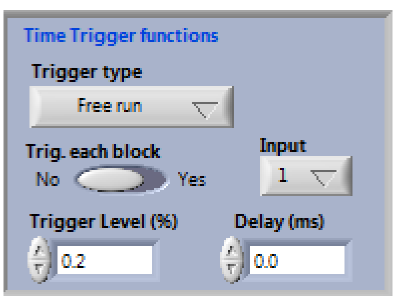
The spectrum analyzer includes a trigger that can be used to measure a transient signal or to perform impact measurements. The trigger controls are accessible when the "*Show trigger functions"* option is set in the *Options* menu.



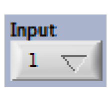
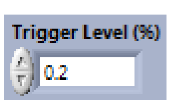
The following sections present the normal and the impact trigger operations.

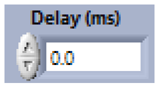
4.4.1 Normal Trigger Operation

When the "*Show trigger functions"* option is set, the following controls are presented in the lower right corner of the main interface:

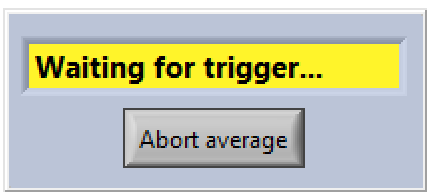


Here are the descriptions of the trigger parameters:

Parameter/control	Description
	This control allows the selection of whether the trigger is applied for each acquisition block or only for the first block of the averaging process.
	This control allows the selection of the input number for the trigger.
	This is the trigger threshold of the input dynamic, defined as a %. This control can vary from 0.25% to 100 % of the input dynamic. The lower limit is due to the hysteresis. The trigger level together with the hysteresis allow two thresholds to be defined:

	Higher threshold = trigger level + h/2.0 Lower threshold = trigger level - h/2.0 Where h is the hysteresis value (0.5 %) and the trigger level is defined as a % of the input dynamic in the “Trigger” control.
	This control allows the trigger delay to be determined. This is the delay (in ms) between the trigger location and the beginning of the time block. A negative delay implies that the acquisition block will begin before the trigger location and a positive delay implies that the acquisition will begin after the trigger location. For a negative delay, a maximum of 1000 input samples is used for the delay. Thus, due to the sampling frequency used, the delay is limited. For instance, for the maximal sampling frequency of the Alto system (43 kHz), the maximal negative delay is -23 ms. The positive delay is not limited.

When the trigger is activated (trigger set-up other than “Free run”), the average is computed for an adjustable number of blocks. Also, the following dialog box appears while the analyzer waits for a trigger event. The user can abort the averaging process.



4.4.2 Trigger Operation for Impact Measurements

The spectrum analyzer includes a set of features to facilitate the impact measurement. The spectrum analyzer comes with a configuration file (impact.cfg) that can be used as a configuration starting point for impact measurement. The following paragraphs review all the parameters involved.

Time windows and input configuration

Impact measurement implies the use of a hammer equipped with an accelerometer on input #1 and a second accelerometer on input #2. The time windows used for both inputs can be adjusted through the following dialog box (use the button “*See and adjust Force/Expo”* of the “*Input Config”* function):

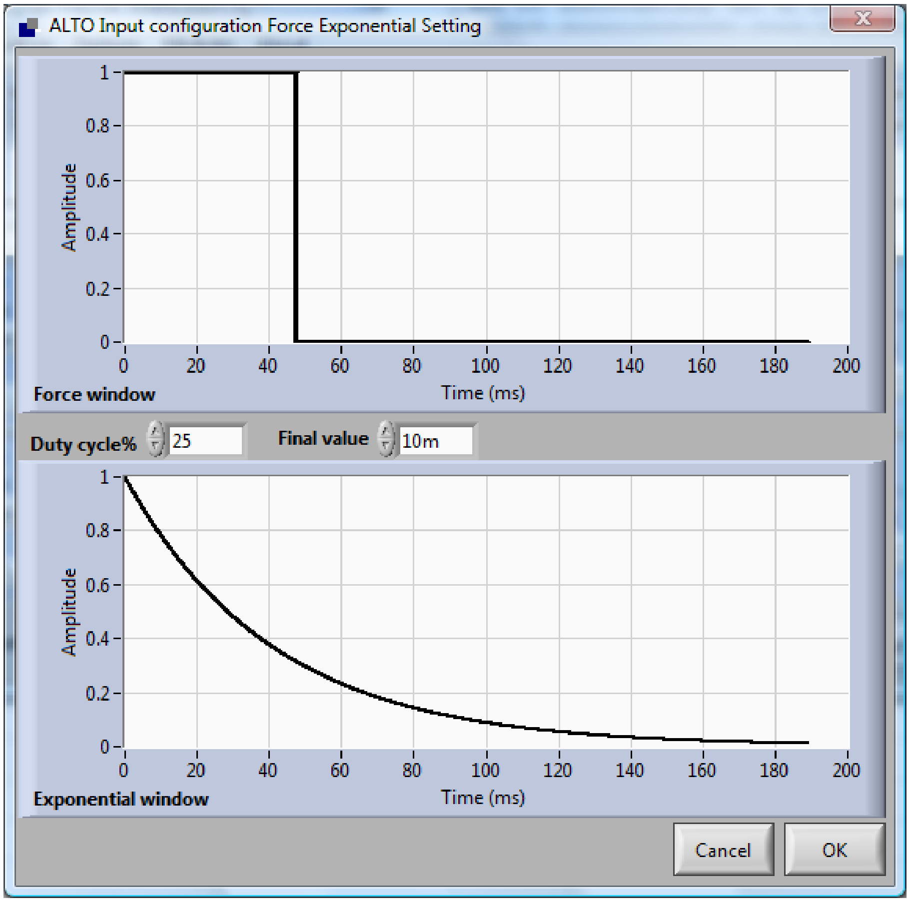
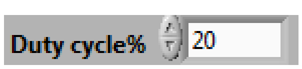
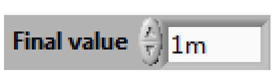


Figure 12: Adjustment interface of the Force and Exponential windows

The Force window allows a double impact to be avoided by limiting the width of the signal. The Exponential window allows the final value to be forced. To be able to perform an FFT averaging process without spectrum leak, the final value parameter must be adjusted to contain the accelerometer impulse signal in the time block. Don't use an excessively small final value to avoid adding artificial damping to the mechanical system measured. We suggest the use of the maximum number of lines for the FFT to work with a longer time block. This way, the measured impulse signal of input #2 will have better changes to fit the time block length.

Parameter/control	Description
	This is the Duty cycle of the Force window defined as a %.
	This is the final value of the Exponential window.

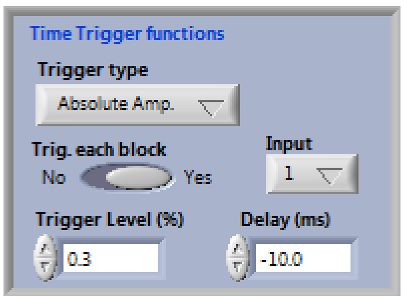
Note: These two parameters can be adjusted during the measurement in the main interface when the time signals are presented for both inputs. This way, the user can adjust the parameter of both time windows on the real signal.

Input range

Since transient signals are acquired during an impact measurement, the automatic auto range option can't be used. The input range must be set manually before the averaging process using the "*Set range"* function of the *Options* menu. For some special cases, input #1 range must be adjusted to a higher level since the accelerometer is on the hammer where the acceleration is more significant.

Trigger set-up

The following trigger set-up is the one found in the impact.cfg configuration file:



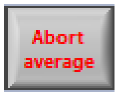
These parameters can be adjusted following the requirements of your set-up.

Averaging set-up

The average type must be set to "*Accept/Reject"* and the average length is defined in "number of blocks". During the averaging process, each block can be accepted or rejected with the following controls:

If saturations occur, or a double impact is observed, the user can reject the block. At any time, the user can abort the averaging process.

Control	Description
	This control allows the current acquisition block to be accepted.

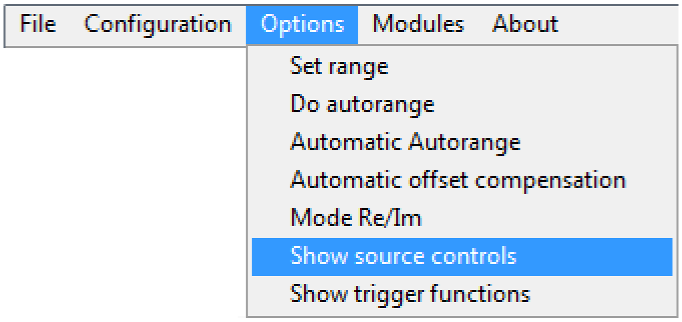
	This control allows the current acquisition block to be rejected.
	This control allows the averaging process to be stopped.

Monitoring during the averaging process

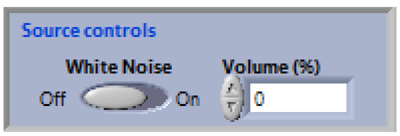
To facilitate the rejection or the acceptance of a time block, the impact.cfg configuration file presents the windowed time signal (input #1) on the first graph and the average FRF on the second graph.

4.5 Source Function

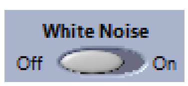
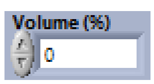
The spectrum analyzer includes a white noise source that can be activated with the "*Show source controls"* option of the *Options* menu.



Then, the following controls appear at the lower right corner of the main interface:



The following table describes these controls:

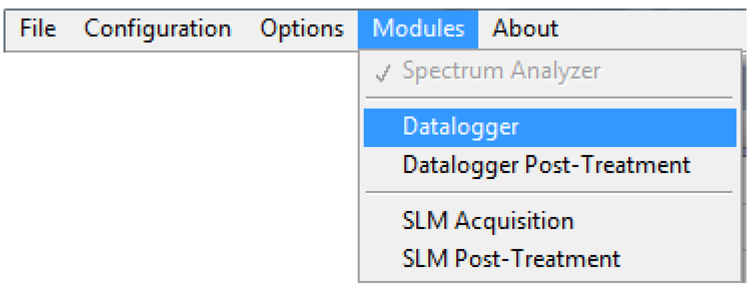
Parameter/control	Description
	This control allows the source to be turned on or off.
	This control allows the source output volume to be adjusted. The control can vary from 0% (source off) to 100%. At 100%, the white noise output signal dynamic is ±1V.

5.0 1-Channel Datalogger

The 1-channel datalogger module is a tool that allows the real-time recording of the time signal. This datalogger also includes a special feature to allow RT60 measurement. Coupled with a complete set of post-analysis functions, the datalogger of the Alto system is a powerful analysis tool.

Note: During the measurement, the entire recorded signal is loaded into the memory of the PC. This approach requires a great amount of memory from the PC and we suggest that the recording time be limited to 10 minutes. The Tenor datalogger software from Soft dB (http://www.softdb.com/a-acous_20_3_4.html) can be used for a longer recording time. This software is compatible with the Alto hardware and it uses a streaming approach that allows a longer recording.

The datalogger module can be launched with the *Datalogger* function in the *Modules* menu:



The following figure presents the main acquisition interface of the datalogger module:

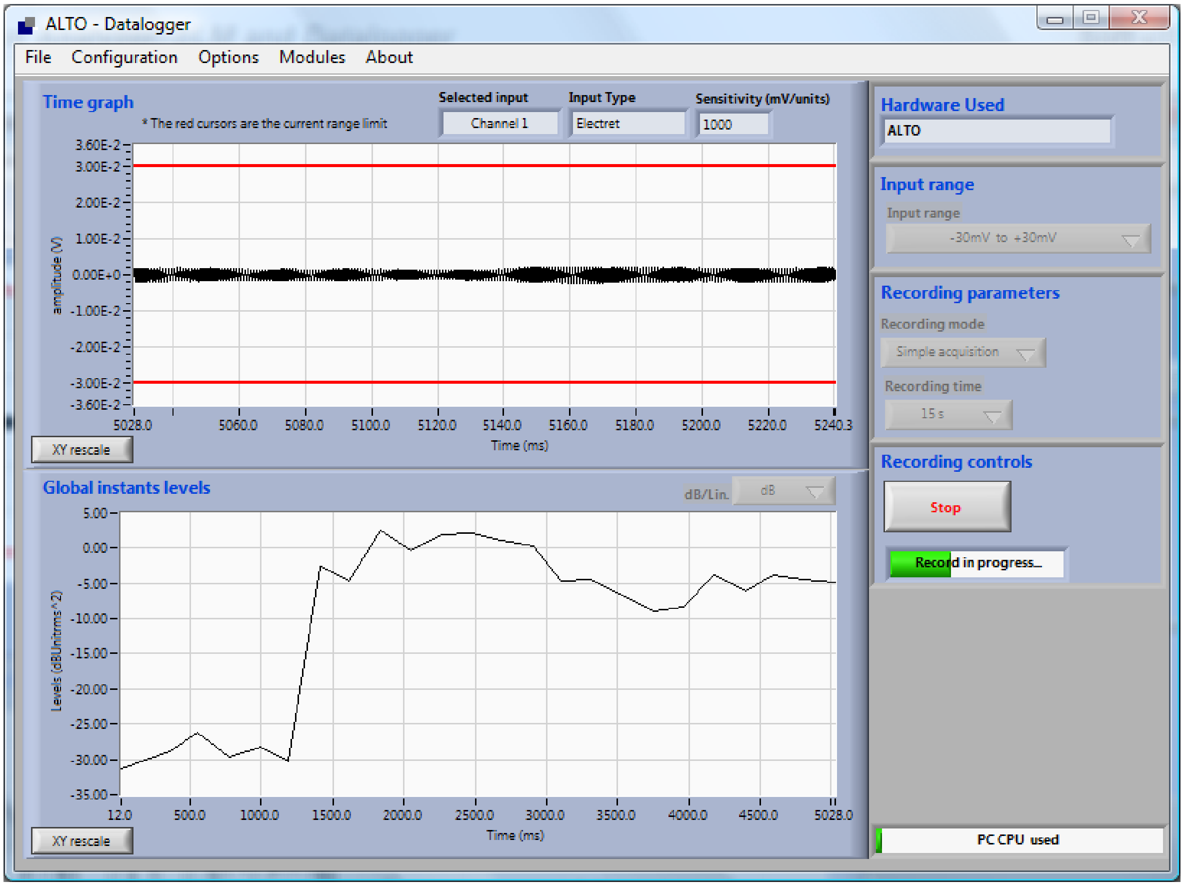


Figure 13: 1-Channel datalogger main acquisition interface

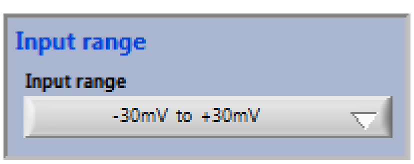
5.1 Configuring the Acquisition

The datalogger always uses the first input. The configuration of this input can be done through the dialog boxes of the "*Input Config*", "*Frequency Span*" and "*Input calibration*" functions of the Configuration menu (see section 4.1 for input configuration details).

The following sections present the set-up parameters of the datalogger module.

5.1.1 Input Range Set-up

The input range can be set manually using the "*Input range*" control.

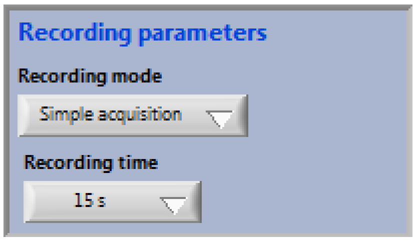


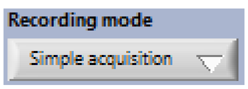
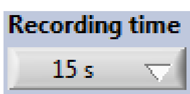
During the configuration of the datalogger or between recordings, the time signal is always acquired and presented in the upper graph. This graph includes two red lines that represent the lower and higher limits of the input range, which are useful to select the optimal input range.

The input range can also be automatically selected through the function "*Do autorange"* of the Options menu.

5.1.2 Recording Set-up

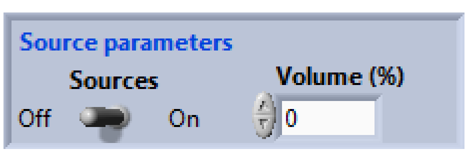
The recording parameters allow the selection of the type and the duration of the recording:

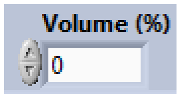


Parameter/control	Description
	This control allows the selection of the recording mode: 1) Mode RT60: The output source will be automatically stopped after the first 10000 samples of the recording (see section 5.1.3 for more details). 2) Simple acquisition: to record the signal of input #1. In this mode, the output source stays either on or off.
	This is the recording time duration. Note that the "*Start"* control changes to "*Stop"* when the recording is launched and it allows the recording to be stopped whatever the set-up for the recording time.

5.1.3 Source Set-up

The datalogger source can be used for RT60 measurement or other purposes. If the RT60 recording type is used, the white noise generator is automatically muted 10000 samples after the beginning of the recording (250 ms at 39 kHz). The following controls can be used to set the white noise generator:

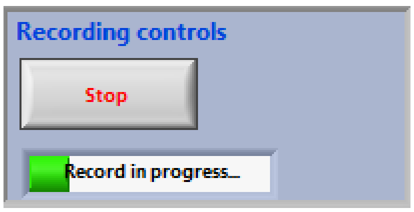


Parameter/control	Description
	This control allows the source to be turned on or off. If the recording mode is RT60, the source will be automatically muted after 10000 recording samples.
	This control allows the generator output volume to be adjusted. The control can vary from 0% (source off) to 100%. At 100% the white noise output signal dynamic is ±1V.

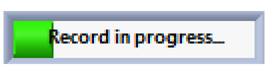
5.2 Recording Process

Before launching the recording process, the format of the global levels presented in the historic graph during the recording must be set. This parameter is not available once the recording is launched.

The recording is launched with the "*Start*" button:



The following table describes the recording controls and indicators:

Indicator/control	Description
 	This control allows the starting of the recording process. Once the acquisition is started, the button becomes a "*Stop*" control. The acquisition stops by itself when the recording time is elapsed or if the user clicks on "*Stop*".
	This indicator presents the recording progression.

During the acquisition, the global level historic is presented in the lower graph. The format of the global level must be set before launching the recording.

5.3 Datalogger Post-Treatment Module

At the end of the acquisition process, the interface automatically launches to the post treatment module. Note that the datalogger post-treatment interface can be launched from the *Modules* menu. The next figure presents the datalogger post-treatment interface:

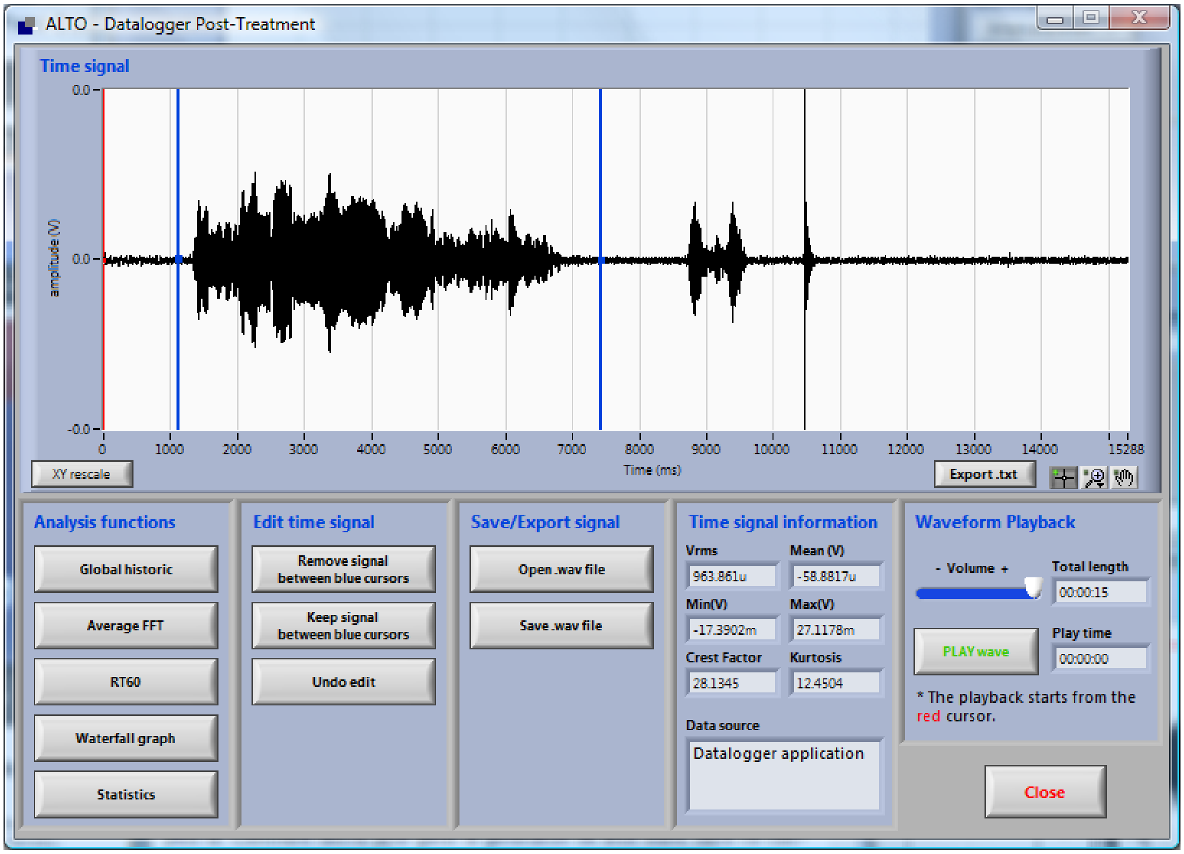
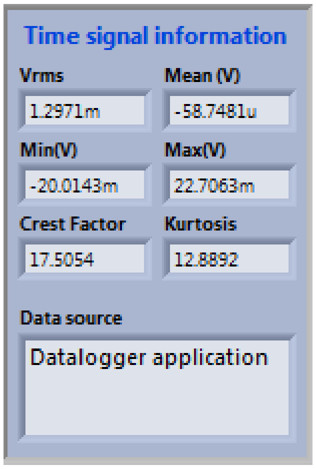
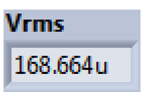
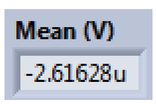
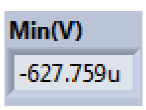
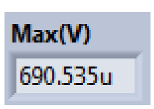
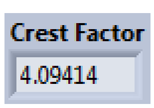
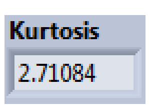
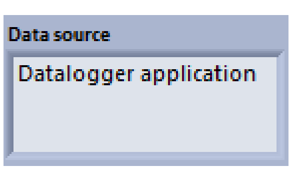


Figure 14: 1-Channel datalogger main analysis interface

The entire recording time signal is presented in this interface along with the low and high limits of the input range (red lines). In addition, the time signal information is also presented:



Indicator	Description
	RMS (Root mean square) indicator
	Time signal mean value indicator (DC)
	Minimum value indicator.
	Maximum value indicator.
	Crest factor indicator: VMax/Vrms.
	Kurtosis indicator. Kurtosis is a measure of the peakedness and corresponds to the fourth-order moment about the mean.
	This indicator shows the source of the signal. It can be either the data of the last acquisition or a wave file.

The analysis module includes a complete set of functions described in the following sections.

5.3.1 Global Historic

This historic function allows the analysis of the global levels over time. After launching the function, the interface asks for computation parameters through this dialog box:

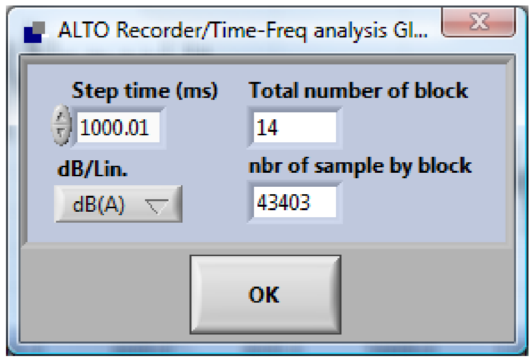
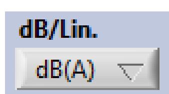
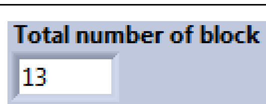
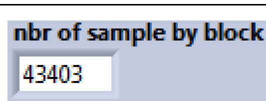


Figure 15: Global historic function parameters

Parameter/control	Description
	This is the step time between two analysis blocks. The step time can be adjusted to fit the standard “Slow” (1000ms) and “Fast” (125 ms).
	This control allows the selection of the amplitude format. The following formats are available: 1) Lin: Linear 2) dB: The amplitude is presented in decibels: 20*log(amp/ref) 3) dB(A): Same as dB but with a frequency A weighting Ref: By default this is 2E-5 but it can be modified with the Input Config function.
	Indicator of the total number of blocks used for the global analysis.
	Indicator of the number of samples per block used for the global analysis.

Then, the global level historic is presented in this interface:

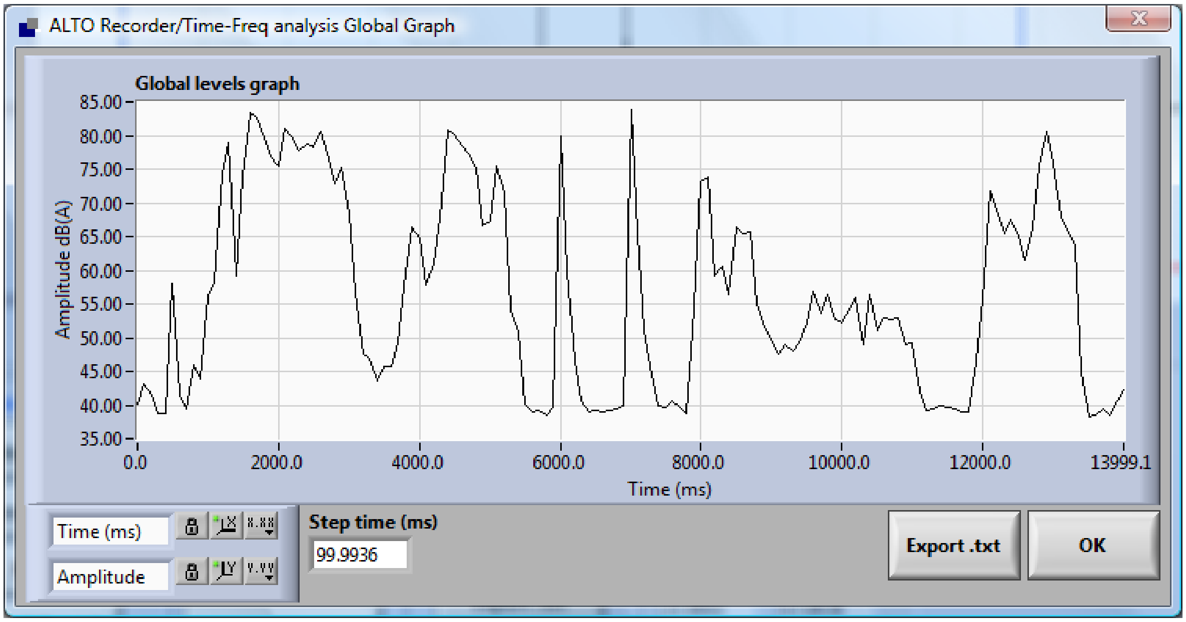
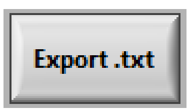
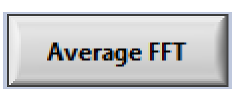


Figure 16: Global level historic interface



This control allows the current global historic graph to be saved in a text file format.

5.3.2 Average FFT



This function computes the average spectrum of the entire time signal. The interface asks for the computation parameters through this dialog box:

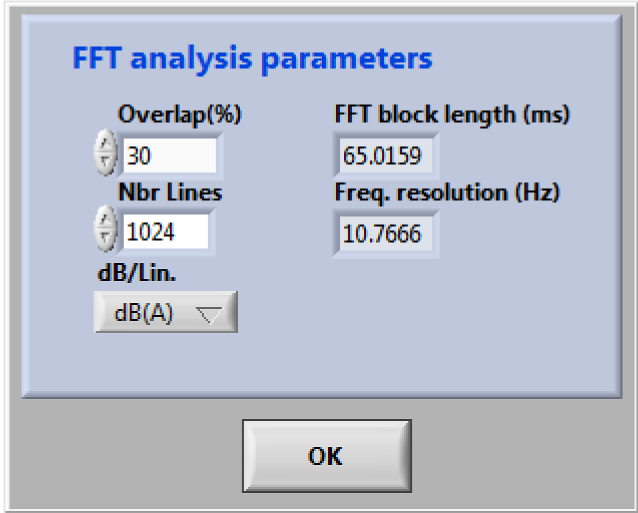
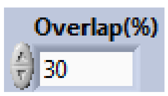
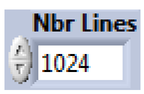
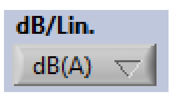
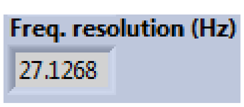


Figure 17: Average FFT function parameters

Indicator/control	Description
	This is the overlap between each analysis block defined as a %. The average FFT computation is performed block by block and the length of each block depends on the “Nbr Lines” control. The overlap allows the reduction of the effect of the time window used on time blocks during the FFT average computation (hanning). The hanning window combined with a 66 % overlap is a common set-up.
	This is the number of lines of the FFT. This control can be adjusted from 512 to 4096 lines. To achieve a good resolution, the number of lines of the FFT must be increased.
	This control allows the selection of the amplitude format. The following formats are available: 1) Lin: Linear 2) dB: The amplitude is presented in decibels: 20*log(amp/ref) 3) dB(A): Same as dB but with a frequency A weighting Ref: By default this is 2E-5 but it can be modified with the Input Config function.
	This is the step time duration between two FFT blocks. This value depends on the set-up of the “Overlap (%)” and the “Nbr Lines” controls.
	This is the frequency resolution (Hz). This value depends on the set-up of the “Nbr Lines” control.

The interface of the “*Average FFT”* function is presented in the following figure:

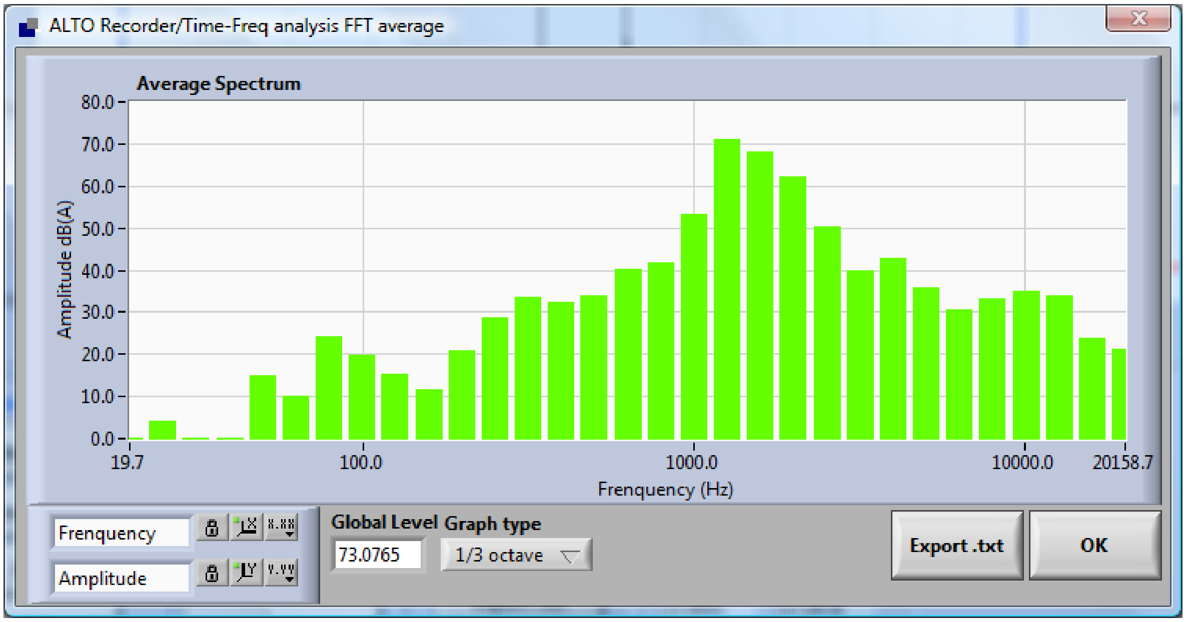
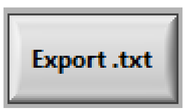
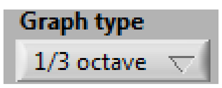


Figure 18: Average FFT interface



This control allows the current average spectrum to be saved in a text file format.



This control allows the selection of the type of graph. The octave, 1/3-octave and 1/12-octave spectra are computed with a frequency domain filter bank applied on the narrow band spectrum. Always use the maximum number of lines for the FFT to obtain precise results at low frequencies on the 1/n-octave spectra.

1) Auto spectrum (Sxx): Average auto spectrum of the time signal

2) Octave (Sxx): Average octave spectrum of the time signal

3) 1/3octave (Sxx): Average 1/3-octave spectrum of the time signal

4) 1/12octave (Sxx): Average 1/12-octave spectrum of the time signal

5.3.3 RT60

This function allows the reverberation time to be computed for all the octave bands and the global level, through a dedicated interface. The following dialog box allows the computation parameters to be specified:

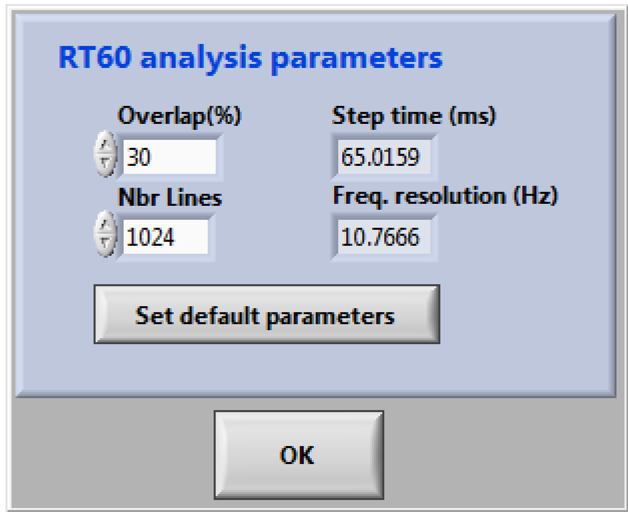
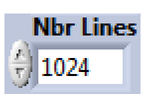
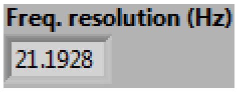


Figure 19: RT60 function parameters

Indicator/control	Description
	This is the overlap between each analysis block defined as a %. The RT60 computation is performed block by block and the length of each block depends on the Nbr Lines control. To obtain a better time resolution, an overlap between each block of the analysis can be used. Also, the RT60 analysis applies a hanning window on each block. To obtain a precise estimation of RT60, it's recommended to use an overlap of 30 % or less.
	This is the number of lines of the narrow band spectra used for the RT60 analysis. The RT60 analysis is performed for each octave band computed with the narrow band spectrum filtered with an octave filter bank. Note that the RT60 estimation is not good at low frequencies if the number of lines is too small. If this is the case, the RT60 analysis interface indicates the problematic octave bands; the band indicator turns red.
	This is the step time duration between two analysis blocks. This value depends on the set-up of the "Overlap (%)" and the "Nbr Lines" controls. Note that the RT60 estimation is not good at low frequencies if the step time duration is too short. The RT60 analysis interface indicates the problematic octave bands; the band indicator will turn red.

	This is the frequency resolution (Hz). This value depends on the set-up of the “Nbr Lines” control.

The following figure shows the RT60 interface:

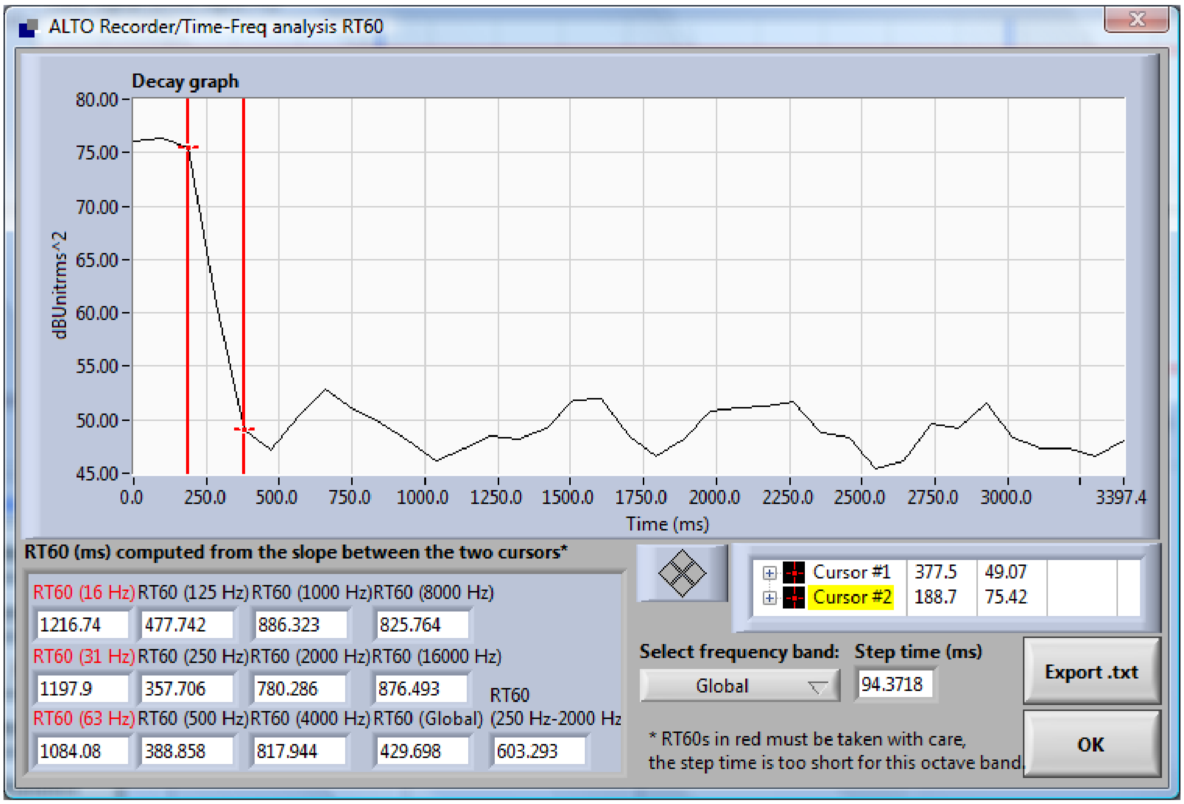
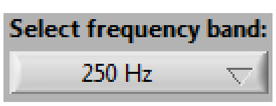


Figure 20: RT60 interface

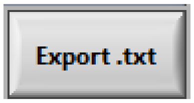
The RT60 for all octave bands and the global level are computed following the two red cursors of the decay curve. The user must manually set the red cursors for each selection of the “*Select frequency band”* control:



This control allows the selection of the octave band presented in the decay graph.

After the selection of the octave band, the red cursors must be set at the beginning and the end of the decay curve as illustrated in the Figure 20. Then, the interface will compute the reverberation time for a reduction of 60 dB, by extrapolation.

This is the step time of the decay graph.



This control allows the RT60 estimations to be saved in a text file format.

5.3.4 Waterfall Graph

The *Waterfall graph* function presents the power spectra for the entire recorded time signal. This function requires a large amount of PC memory since a lot of data is used to generate the waterfall graph. To avoid memory problems when using this module, we suggest limiting the time signal duration.

The *Waterfall graph* function computes the power spectra with an adjustable frequency resolution. All power spectra are presented in a three dimensional graph along a time axis. The three dimensional graph can be rotated, zoomed and moved.

The following dialog box allows the waterfall computation parameters to be set:

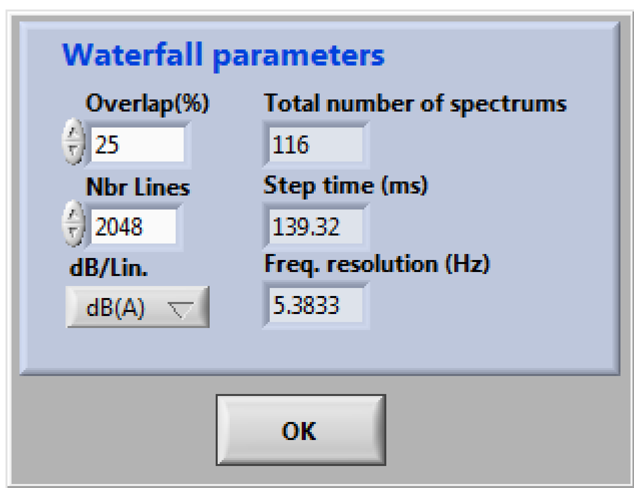
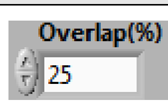


Figure 21: Waterfall function parameters

Indicator/control	Description
	This is the overlap between each analysis block defined as a %. The waterfall analysis is performed block by block and the length of each block depends on the "Nbr Lines" control. To obtain a better time resolution, an overlap between each block of the analysis can be used. Also, the waterfall analysis applies a hanning window on each block. The hanning window combined with a 66 % overlap is a common set-up.

Nbr Lines 1024	This is the number of lines of the FFT. This control can be adjusted from 512 to 4096 lines. To achieve a good frequency resolution, the number of lines of the FFT must be increased.
dB/Lin. dB(A)	This control allows the selection of the amplitude format. The following formats are available: 1) Lin: Linear 2) dB: The amplitude is presented in decibels: 20*log(amp/ref) 3) dB(A): Same as dB but with a frequency A weighting Ref: By default this is 2E-5 but it can be modified in the spectrum analyzer calibration module.
Total number of spectrum: 189	Indicator of the total number of spectra used to form the waterfall graph.
Step time (ms) 70.7789	This is the step time duration between two analysis blocks. This value depends on the set-up of the “Overlap (%)” and the “Nbr Lines” controls.
Freq. resolution (Hz) 10.5964	This is the frequency resolution (Hz). This value depends on the set-up of the “Nbr Lines” control.

The following figure presents the waterfall interface:

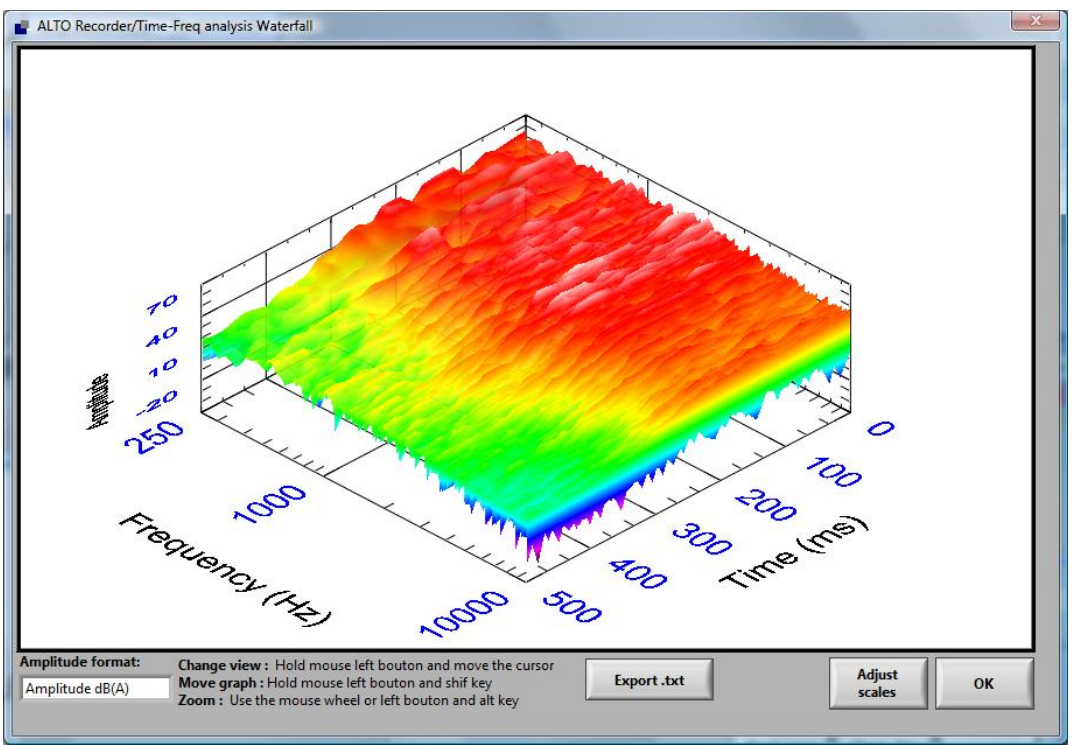
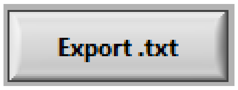
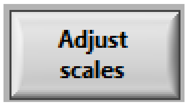


Figure 22: Waterfall interface



This control allows the current waterfall graph to be saved in a text file format.



This button allows the scale of the waterfall graph to be adjusted through the following interface:

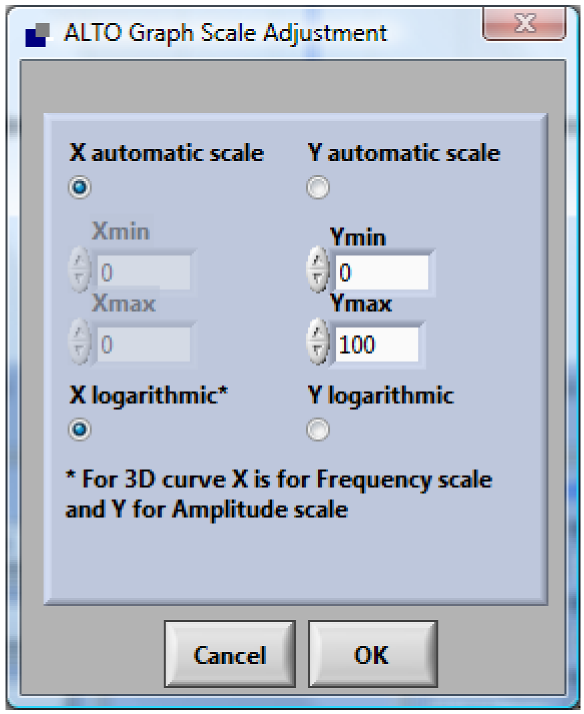
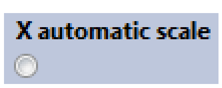
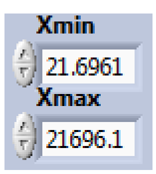
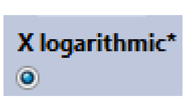
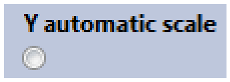
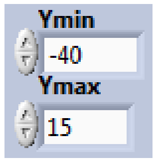
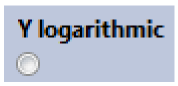


Figure 23: Waterfall scale dialog box

Parameter/control	Description
	This radio button allows the selection of the auto scale for the X-axis. Note: the X scale is the frequency scale of the waterfall graph.
	This is the minimum value for the X-axis of graph #1. This control is used only when the X automatic scale mode is not selected. This is the maximum value for the X-axis of graph #1. This control is used only when the X automatic scale mode is not selected. Note: the X scale is the frequency scale of the waterfall graph.
	This radio control allows the selection of a logarithmic scale for the X-axis. Note: the X scale is the frequency scale of the waterfall graph.

	This radio button allows the selection of the auto scale for the Y-axis. Note: the Y scale is the amplitude scale of the waterfall graph.
	This is the minimum value for the Y-axis. This control is used only when the Y automatic scale mode is not selected. This is the maximum value for the Y-axis. This control is used only when the Y automatic scale mode is not selected. Note: the Y scale is the amplitude scale of the waterfall graph.
	This radio control allows the selection of a logarithmic scale for the Y-axis. Note: the Y scale is the amplitude scale of the waterfall graph.

Note: the time scale can be adjusted by initially selecting a portion of the recorded time signal. See section 5.3.6 for more details about the editing of the time signal.

5.3.5 Statistics Lx%

This function allows the computing of the standard Lx% statistics on the global levels for the entire time signal. The *Statistics* function starts by computing the global levels on the entire time signal with an adjustable time resolution. The global level calculation is performed with power spectra using time 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 option. When the global level historic is computed, the *Statistics* module computes the statistics of the global levels and presents a histogram graph of the sorted global levels.

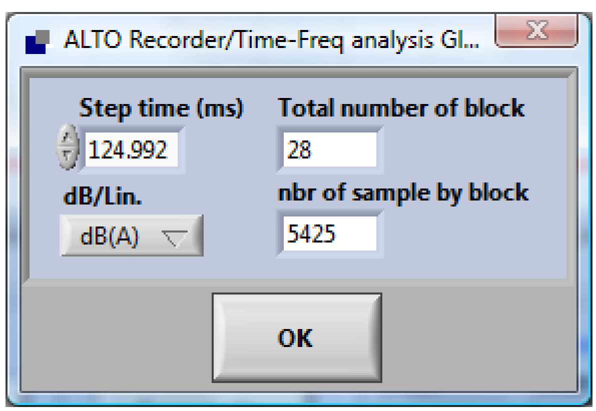
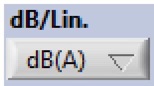
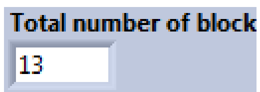
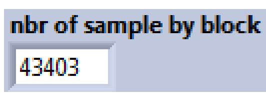


Figure 24: Statistics function parameters

Parameter/control	Description
	This is the step time duration between two analysis blocks. The step time can be adjusted to fit the standard “Slow” (1000ms) and “Fast” (125 ms).
	This control allows the selection of the amplitude format. The following formats are available: 1) Lin: Linear 2) dB: The amplitude is presented in decibels: 20*log(amp/ref) 3) dB(A): Same as dB but with a frequency A weighting Ref: By default this is 2E-5 but it can be modified in the spectrum analyzer calibration module.
	Indicator of the total number of blocks used for the statistical analysis.
	Indicator of the number of samples per block used for the statistical analysis.

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

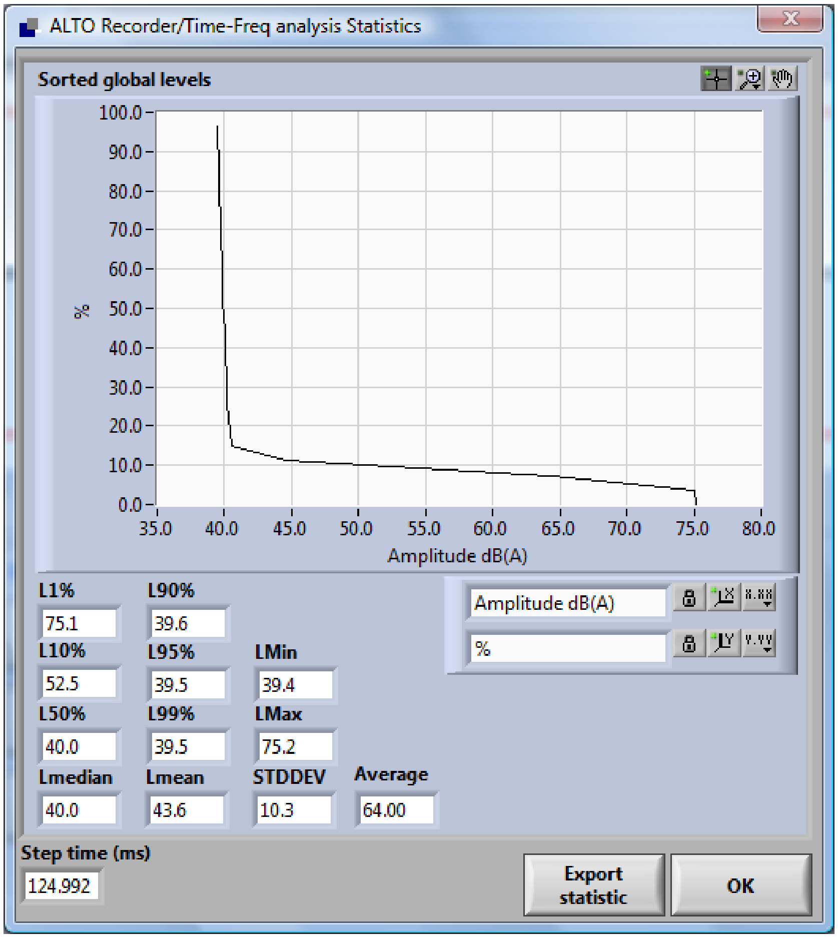


Figure 25: Statistics interface

This control allows all the Lx% to be saved in a text file format.

5.3.6 Edit Time Signal

The time signal can be edited to select or to remove a desired portion of the recording. The two blue cursors of the time curve are used to select the desired portion of the signal. Then, the following functions can be used to edit the signal:

Control	Description
Remove signal between cursors	This control allows the portion of the signal between the blue cursors to be removed. Use the “Undo edit” control to reload the original time signal. Note that this operation is delicate because of the discontinuity caused by this type of editing. Be sure to minimize the discontinuity while using this function.
Keep signal between cursors	This control allows the portion of the signal between the blue cursors to be kept and the rest of the signal to be removed. Use the “Undo edit” control to reload the original time signal.
Undo edit	This control allows the reloading of the original time signal.

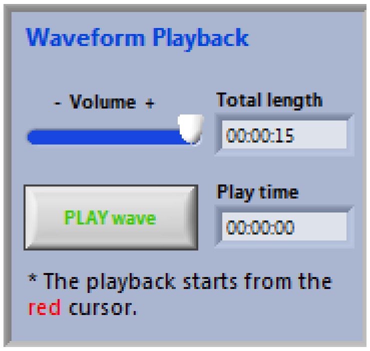
5.4 Save and Open Recorded Signal

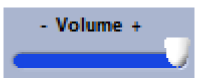
The time signal can be saved and recalled using the standard wave format (.wav). The following table presents the "save" and "open" functions descriptions:

Control	Description
Save .wav file	This control allows the current time signal to be saved in a wave file format. The file can be used for a simple playback using a standard wave reader. A configuration file is also automatically recreated to allow a post-processing on a calibrated signal. The Alto datalogger post-treatment interface along with the Soft dB Tenor software (see http://www.softdb.com/a-acous_20_3_4.html for more information about this software) can be used to analyze the wave files recorded with this function.
Open .wav file	This function allows a .wav file to be opened. This function seeks a configuration file in the same folder as the .wav file in order to retrieve a calibrated signal. This configuration file is automatically created with the "*Save .wav file*" function. If the configuration file is not present and since input sensitivity is not included in a .wav format, the time signal will be normalized between -1 Volt and +1 Volt.

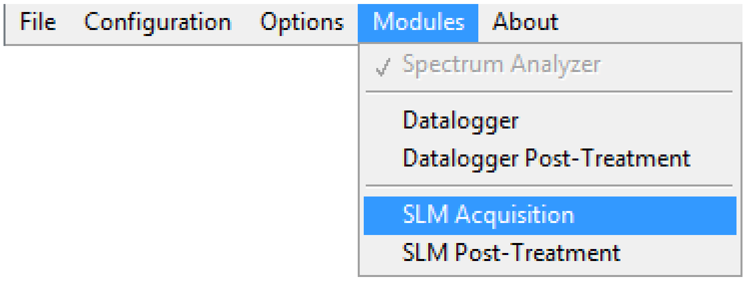
5.5 Playback Function

The following group of controls and indicators allow the playback of the current recorded signal:



Control/Indicator	Description
	This is volume of the sound card of the PC during the playback.
 	This control allows the starting of the playback. The playback always starts at the red cursor position. When the red cursor reaches the end of the recorded signal, it returns to the beginning of the recording. The user can stops the playback with the "*STOP wave*" button.
	These indicators show the total duration of the recording and the current playback time. The current playback time is also illustrated by the red cursor on the time signal graph.

6.0 1-Channel SLM Instrument



Section 6.0 explains the SLM (Sound Level Meter) module of the Alto instrument. This module has been developed and tested to comply with the IEC 61672 class 1 standard. The user can refer to appendix C for more details about the SLM specifications in accordance with the IEC 61672 standard (*Electroacoustics-Sound Level Meters*).

The following figure presents the main interface of the SLM module (launched with the "*SLM Acquisition"* function in the *Modules* menu):

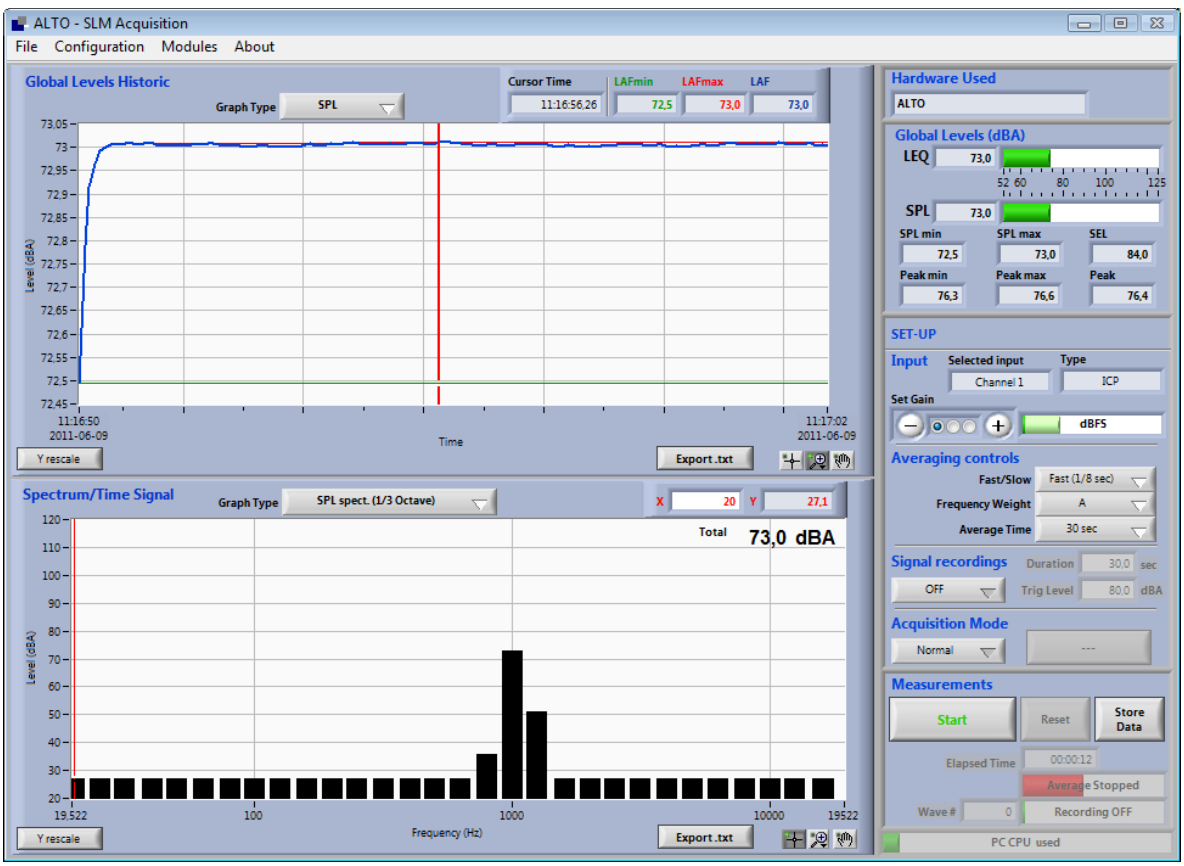


Figure 26: SLM module main interface

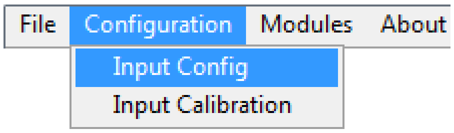
The SLM module of the Alto is a 1-channel Sound Level Meter (SLM) that allows linear and exponential averages to be made. Also, the time signal and the average spectrum (in 1/3-octave and in narrow band) are computed and can be saved during the measurement. The historic of the global levels (Peak, LEQ and SPL) can be monitored and saved during the measurement.

The SLM module has auto store capability (global levels, historic data and spectra). This unique feature allows long period or continuous measurements. Coupled with a powerful post-processing interface, the auto store capability makes the Alto SLM module more powerful and polyvalent than any other SLM on the market.

The following paragraphs explain the set-up and the functionalities of the Alto SLM module.

6.1 Input Set-up

Before starting a measurement, two input parameters must be set: the input sensitivity and the input range (or gain). These parameters are accessible through the "*Input CFG"* function in the *Configuration* menu:



The entire set-up of the input can be done through this dialog box:

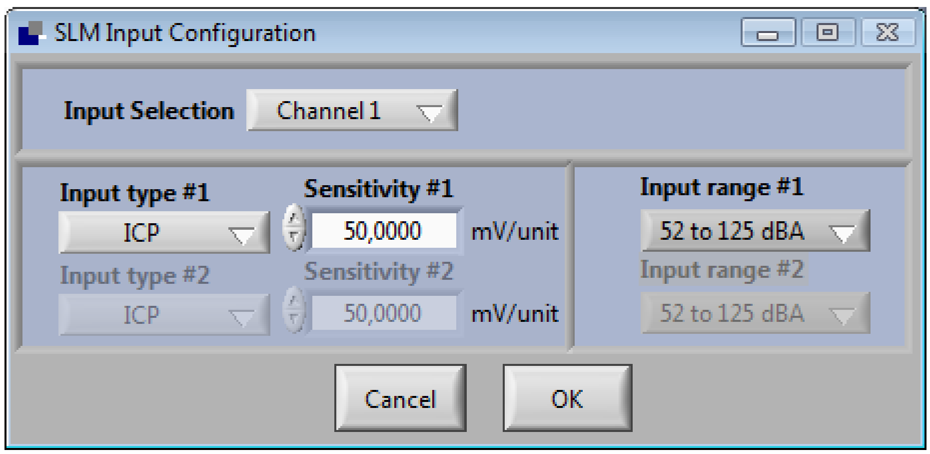
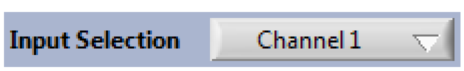
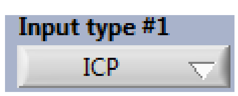
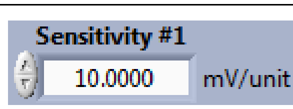
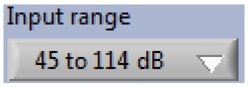


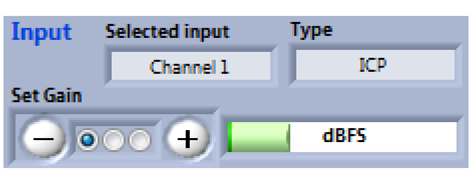
Figure 27: SLM Input Configuration dialog box

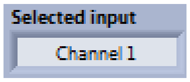
Parameter/control	Description
	The SLM module can use the first or second channel of the Alto unit. The SLM module can also work with the Soft dB Tenor unit (see http://www.softdb.com/a-acous_20_3_4.html for more information about the Tenor). In this case, only the first or the second channel can be selected. It means that channels 3 to 8 of the Tenor unit can't be used with the SLM module.
	There are three input types: *Direct AC*: This input type uses a high pass filter at 0.5 Hz and is suitable when an external preamplifier is used. *ICP:* This input type is for an integrated circuit preamplifier (ICP). It has a high pass filter at 0.5 Hz. When this type of input is selected, a compatible ICP/DeltaTron microphone or accelerometer can be connected directly to the BNC connector.
	This is the sensitivity of the input in mV/Unit. The sensitivity can be entered manually if the value is known, or determined automatically through the use of the calibration function via the "*Input Calibration"* function (see section 4.1.3 for more information

	on this function). The suggested BK microphone/preamplifier (see Appendix C) has been used during the tests for the IEC 61672-3 standard and its sensitivity is 50 mv/unit.
	There are three input range selections. The input range must be selected following the expected range of global levels to be measured. The input range written on the "*Input range*" control is estimated with the current input sensitivity and voltage range of the input. The technique used to estimate the input range is based on a sine signal of 8 kHz (A-weighting). Also, this is the linear operating range (see Appendix C for more details). Following the frequency contents and the type of frequency weighting used, the actual range can vary. During the measurement and if a global level is over or under the estimated input range limits, a low-level or saturation indicator appears on the lower graph of the main SLM interface. In these cases, the user must readjust the input range.   Note that the input range can also be adjusted through the following control in the main SLM interface (see following paragraph for more information): 

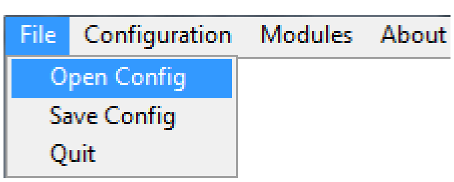
Note: When using ICP microphone, the sensitivity given by the manufacturer doesn't usually take into account the ICP preamplifier attenuation, which is usually less than 0.5 dB and typically around 0.3 dB. In order to enter the sensitivity manually you will need to calculate the sensitivity viewed at the unit input. For instance, for a given manufacturer sensitivity of 50mV/Pa and a known ICP preamplifier attenuation of 0.5 dB, the sensitivity entered in the software will be 50mV/Pa*10^(0.5/20) = 53 mV/Pa. A better solution would be to evaluate the sensitivity within the calibration Module. The resulting input sensitivity will consider both the microphone and its preamplifier so that you don't have to worry about the preamplifier amplification.

The main SLM interface presents the current input set-up for this group of indicators and controls:



Indicator/control	Description
 & 	These indicators show the input channel selection and the type of input used. Use the "*Input CFG"* function in the *Configuration* menu to change these parameters.
	This control can be used to change the current input gain selection. Use "+" to increase the input gain for the measurement of lower global levels. Use "–" to decrease the input gain for the measurement of higher global levels. The *dBFS* indicator can be used as a guideline to select the adequate input gain. If the *dBFS* is too high, it turns red and a lower input gain must be selected. If the dBFS is too low, a higher input gain can be selected. During the measurement and if a global level is over or under the input range limits, a low-level or saturation indicator appears in the lower graph of the main SLM interface. In these cases, the input gain must be readjusted. The "*Input Range"* parameter in the input configuration dialog box (the "*Input CFG"* function in the *Configuration* menu) can be used to select the input gain or range.
	This indicator presents the dynamic used. This value is the peak value in dB (the reference is the full scale of the input). The green bar is the instantaneous peak value, while the light green bar is the maximum peak value from the beginning of the averaging process. Note that the reset operation during the averaging process resets this indicator. This indicator turns red when an input saturation is detected.

Note that the input configuration along with the average, recording and auto store configurations can be saved and recalled through the "*Open Config"* and "*Save Config"* options in the *File* menu:



The extension of the SLM configuration file is .cfl. This file format is not compatible with the configuration file of the spectrum analyzer and the datalogger.

6.2 Average Set-up

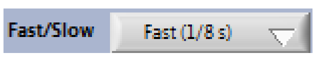
The Alto SLM module computes two types of average:

1) Standard fast or slow exponential average

2) Linear average (Leq)

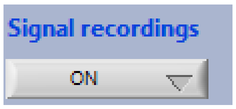
Both average types are computed at each measurement. The following group of controls allow the averaging process to be configured:

Figure 28: Averaging Controls

Parameter/control	Description
	This control allows the selection of the exponential average type. The fast set-up uses an exponential time constant of 1/8 second while the slow set-up uses an exponential time constant of 1 second. Since an exponential average takes time to stabilize, the SLM module waits for this stabilization at the beginning of the averaging process. The waiting time is 2 seconds in slow mode and 0.25 second in fast mode.
	The frequency weighting selection is performed through this control. The A type is by far the most commonly applied and it is used for all levels of sound. The C type is mainly used for high sound pressure levels. The Z type is a flat frequency weighting.
	This is the average time duration. It can vary from 10 seconds to 24 hours. The linear average will be obtained for this time period while the exponential average data will be collected for this time period.

6.3 Signal Recording Set-up

The SLM module allows the time signal to be recorded during the averaging process. This feature allows the playback of the sound measured during the average. It can be useful for detecting special events or to perform a more advanced post-analysis (using the datalogger post-treatment module of the Alto or the Tenor software from Soft dB).



The recording can be done for the entire average period when the "*Signal Recordings* "control is set to ON. Then, the SLM interface will create a standard wave file with a satellite configuration file. The configuration file contains the necessary information to format the data in the wave file and to retrieve a calibrated time signal.

Both files are saved in the current "*Data Storage Directory"* path. This path can be set with the "*Set AutoStore"* button. For each new measurement, the SLM interface creates a new subfolder, a .wav file and a .cfg file, with the following name:

YYYYMMDD_HHMMSS

Where:

YYYY is the year

MM is the month

DD is the day of the month

HH is the hour

MM is the minute

SS is the second

Also, the interface manages the maximal wave size to avoid files larger than 1 Gigabyte. If this size is achieved, the interface closes the current wave and configuration files and open new files with a suffix (_partx, where x is the part number).

Note that the "*On with Trig"* option is not allowed for the "*Signal recordings"* control if the auto store mode is not activated. The following section explains the auto store mode in detail.

6.4 AutoStore

The auto store functions of the SLM module allow a continuous saving of all LEQ average results. The information saved includes the SPL global level historic and the linear average 1/3-octave and narrow band spectra. The time signal can also be saved continuously or on trig events. The SLM module has a dedicated post-processing interface that allows a complete analysis of recorded data (see section 6.8). This auto store feature along with the post-processing interface is a powerful tool for the continuous or long period analysis of sound.

The Acquisition Mode can also be set to Standalone. Refer to section 6.7 for details.

The auto store set-up is done through this group of controls:

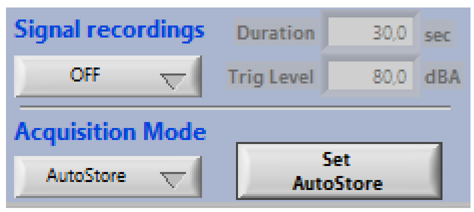
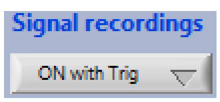


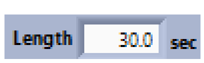
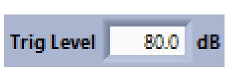
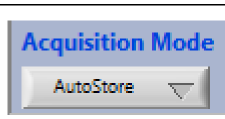
Figure 29: Signal Recording and AutoStore controls

<table>
<tr><th>Parameter/control</th><th>Description</th></tr>
<tr><td>
</td><td>This control allows the time signal recording behaviour to be specified.

It can be Off if no time signal recording is desired.

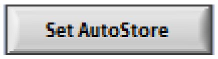
The set-up ON can be used for a continuous recording of the time signal.

The recording can also be performed for triggered events (set-up “ON with Trig”). In this case, the Length and the Trig control are used to specify the trigger set-up (see the following item in this table for more information).

When the signal recording option is activated (“ON” or “ON with Trig”), the SLM interface automatically saves a wave file containing the time signal along with a satellite configuration file. After the measurement process, the wave file(s) can be played or analyzed with the Alto or Tenor software from Soft dB.</td></tr>
<tr><td>
</td><td>This is the duration of the individual recording if the “ON with Trig” option is used. The maximum duration has been fixed at 6734 seconds to avoid wave files larger than 1 Gigabyte.</td></tr>
<tr><td>
</td><td>This is the threshold trigger level for starting a new time signal recording. The trigger uses the peak global level to start the recording.</td></tr>
<tr><td>
</td><td>When the auto store mode is activated, the “Start average” control is renamed “Start AutoStore”. Then, a series of LEQ averages are measured and automatically saved. The auto store process stops when the user clicks on “Stop AutoStore” or if the total storage time is reached. The total storage</td></tr>
</table>

	time can be set with the "*Set AutoStore*" function (see the following paragraph).

The following button allows the setting of the file location and the duration of the auto store measurement:



The following figure shows the dialog box for the "*Set AutoStore*" function:

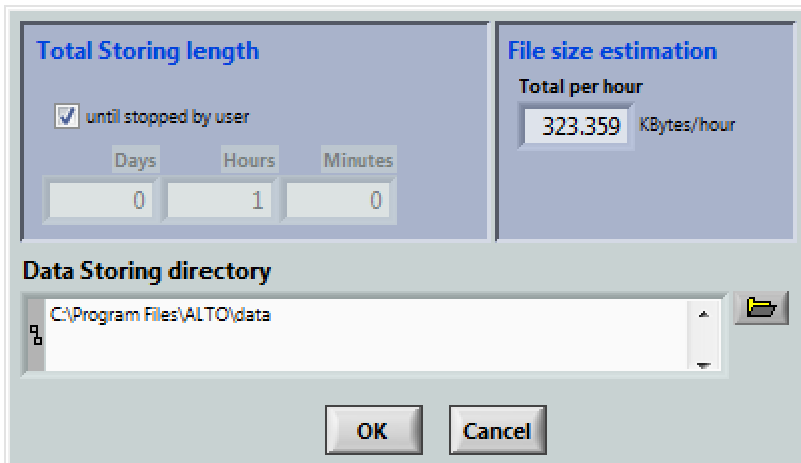
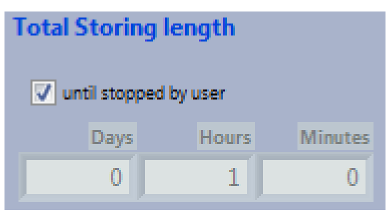
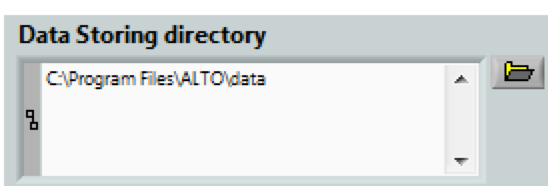


Figure 30: Auto store dialog box

Parameter/control	Description
	This is the total storage time. The option "*Until stopped by the user*" can be used for a continuous auto store process. If the option "*Until stopped by the user*" is used, be sure to have enough hard disk space. As a guideline, the following set-ups require 66 Mbytes: - A total storage time of 24h - LEQ Average Time: 10s - No time signal recording (Signal Recording set to "Off")
	This is the root folder for all recorded data. At the beginning of the auto store process, the SLM interface creates a subfolder with the following name: YYYYMMDD_HHMMSS Where: *YYYY* is the year

	MM is the month *DD* is the day of the month *HH* is the hour *MM* is the minute *SS* is the second This subfolder will contain all the data including the wave files if the signal recording option is activated.
	This indicator presents the size of the data file (.slm) in Kbytes per hour.

6.5 Measurements

Once the SLM is configured, the measurement can be launched. The measurement can be a single LEQ average or a series of LEQ averages if the auto store mode is used. The following buttons and indicators allow the controlling and monitoring of the measurement process:

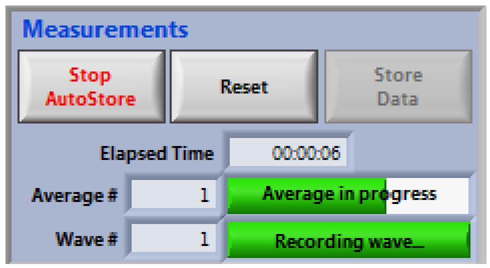
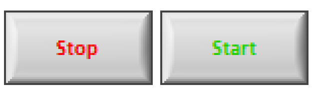
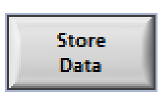
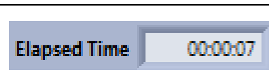
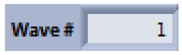


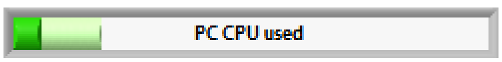
Figure 31: SLM measurement controls and indicators

Note that some controls or indicators can be hidden if the auto store mode is not used. The following table explains all the measurement controls and indicators.

Indicator/control	Description
 or (if auto store mode): 	These controls allow the starting and stopping of the measurement process. The name of the button changes if the auto store mode is activated.
	This control is available at the end of the averaging process and it allows the manual saving of all the data from the last LEQ average (SPL historic, 1/3-octave and narrow band spectra). The file created can be opened in the post-processing module of the SLM (see sections 6.6.3 and 6.8 for more details).
	This button allows the current LEQ average to be reset. If the auto store mode is activated, the reset operation will stop the current measurement, create a new folder for storing data and restart the measurement. For the normal single LEQ measurement type, the interface resets the average data and restarts the measurement.
	This is the time elapsed since the beginning of the measurement process.

	If the auto store mode is used, this indicator shows the number of the current LEQ average.
	This bar indicates the current LEQ average progression. If the auto store mode is activated, this bar presents the progression for the current LEQ average and not the entire recording progress.
	If the signal recording is activated with the trigger option, this indicator presents the number of wave files recorded since the beginning of the measurement.
	This indicator lights up during the recording of the time signal. This indicator is activated only if the signal recording option is used. If the "*ON with Trig"* option is used, this indicator presents the progression of the current recording.

During the measurement, the "*PC CPU used"* indicator (lower right corner of the main SLM interface) allows evaluation if the acquisition and the processing are in real time:



This meter shows a light green bar and a green bar. The green bar is the current CPU usage while the light green bar is the maximum CPU usage from the beginning of the measurement. When the PC's CPU is not fast enough to allow real-time acquisition and processing, the bar turns red. We suggest avoiding the use of software other than the Alto during the measurement. Also, the use of PC power saving functions must be avoided during a measurement. If a long period measurement is performed with the auto store option activated, we suggest that the main window of the SLM interface be minimized in order to avoid overburdens during the measurement.

The two following sections present the graphs and the global level indicators, which allow the results of the averaging process to be monitored during a measurement.

6.5.1 Global Indicators

During the measurement, the following group of indicators shows the global level results:

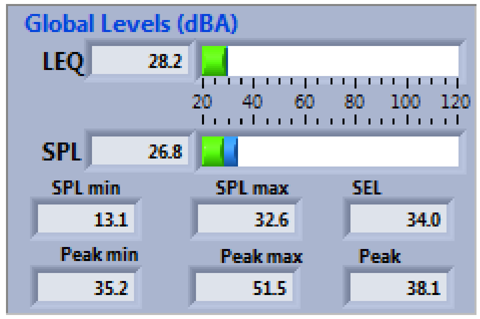
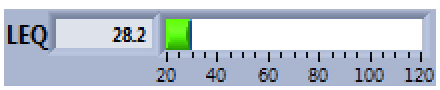
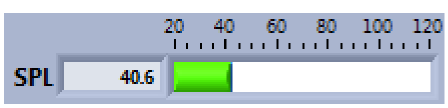
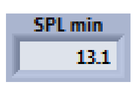
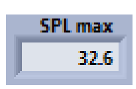
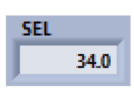
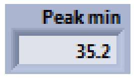
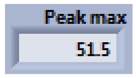
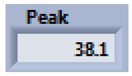


Figure 32: Global Level indicators

The following table explains all the global level indicators.

Parameter/control	Description
	The LEQ is the equivalent sound pressure level: the steady sound level that, over a specified period of time (the average time), would produce the same energy equivalence as the fluctuating sound level that is actually occurring. The LEQ is computed with a linear average and is updated every 1/10 second during the measurement. The time base for the LEQ measurement is the "*Average Time"* specified in the "*Averaging Set-up"* group of controls. If the auto store mode is used, the LEQ average is reset after each average period.
	This is the instantaneous sound pressure level measured with an exponential time average. The SPL indicator (with a slow or fast exponential time average) is updated every 1/10 second during the measurement.
	This is the minimum SPL measured since the beginning of the averaging process.
	This is the maximum SPL measured since the beginning of the averaging process.
	The SEL (Sound Exposure Level) is the total energy normalized over a period of 1 second. The total average energy of the LEQ global level is multiplied by the elapsed time of the current LEQ average to estimate the equivalent sound level for 1 second (SEL). Since the SEL is based on the LEQ value and if the auto

	store mode is used, the SEL is reset after each average period.
	This is the minimum peak value measured since the beginning of the measurement.
	This is the maximum peak value measured since the beginning of the measurement.
	The peak value is the maximum instantaneous global level of the last 1/10-second.

6.5.2 Global Level Historic and Spectrum/Time Graphs

During the measurement, the two graphs of the SLM interface allow the average results to be monitored. The upper graph presents the Leq, SPL and Peak global level historics while the lower graph presents the time signal, the linear narrow band and 1/3-octave average spectra and the exponential 1/3-octave average spectrum.

The following figure presents the upper graph of the SLM interface with the global level historic. The historic graph presents the Leq, the SPL or the Peak global levels over time since the beginning of the measurement. The maximum (red curve) and minimum (green curve) are also presented on the historic graph. The maximum duration for the historic graph is 1 hour. After this period, the oldest global levels are erased from the historic graph to make room for new global level results.

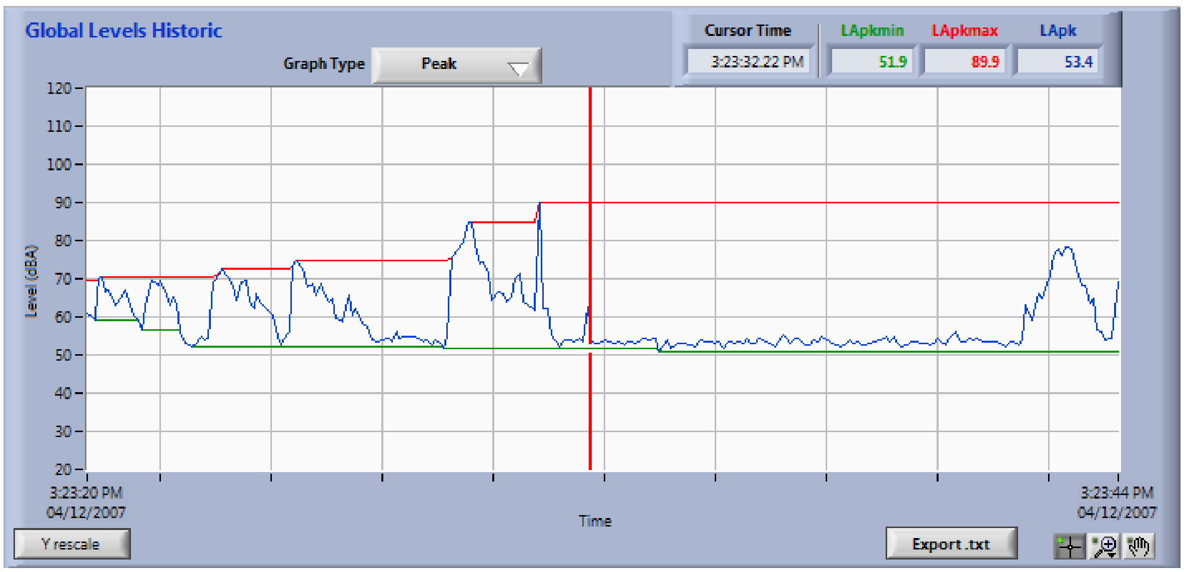
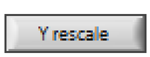
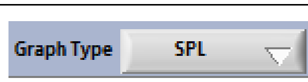


Figure 33: Global Level Historic graph

Parameter/control	Description
	This control applies auto scaling on the Y-axis of the current selected graph.
	This button allows all the historic data of the selected graph to be exported into a text file format. The minimum and maximum historic curves for the selected graph are also saved. This file can be used in a standard spreadsheet program such as Excel.
	This control allows the selection of the global levels to be presented in the historic graph: Leq: This selects the historic of the equivalent sound pressure levels. In auto store mode, the LEQ value is reset for each new average. The historic graph reflects

	this but the minimum and maximum curves are not reset. SPL: This selects the historic of the exponential time average sound pressure levels. Peak: This selects the historic of the instantaneous peak levels.
	These are the cursor indicators. The red cursor in the historic graph can be moved to obtain a precise reading for the selected global level.

The following figure presents the lower graph of the SLM interface with the time and spectrum graphs.

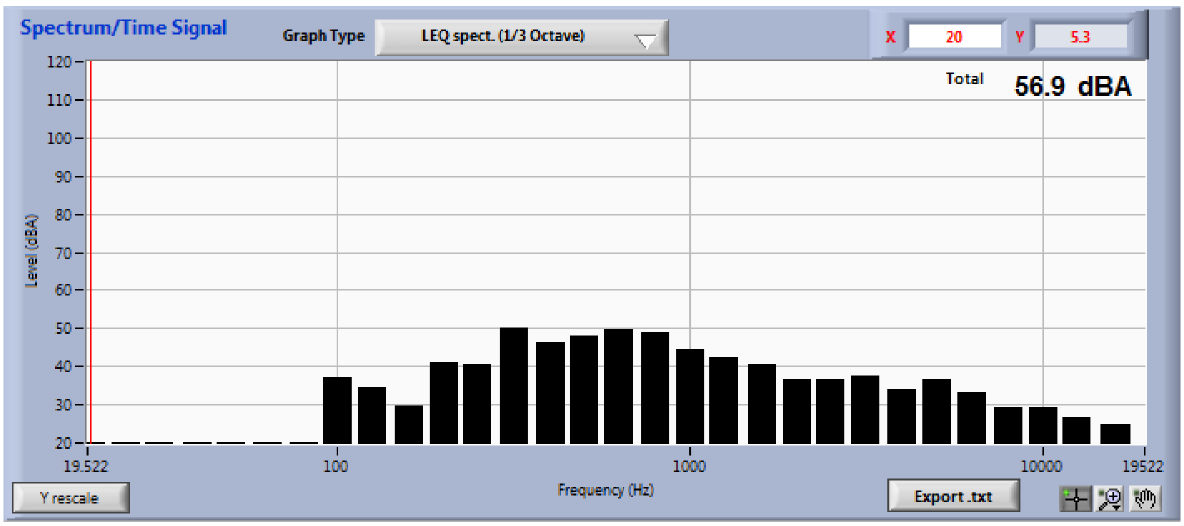
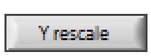
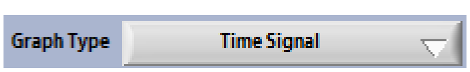
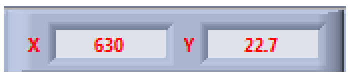


Figure 34: Time/Spectrum graph

Parameter/control	Description
	This control applies auto scaling on the Y-axis of the currently selected graph.
	This button allows the selected spectrum (or the current time signal block) a text file format. This file can be used in a standard spreadsheet program Excel.
	This control allows the selection of the time or spectrum graph type: Time Signal: This is the instant time signal block. The length of the block is 90 ms. LEQ (1/3 octave): This is the linear average 1/3-octave spectrum. The 1/3-octave spectrum is obtained from a bank of filters running in real time on the DSP (Digital Signal Processor) of the Alto. In auto store mode, the LEQ average 1/3-octave spectrum

	is reset at each new average. SPL (1/3 octave): This exponential time average 1/3-octave spectrum. The 1/3-octave spectrum is obtained from a bank of filters running in real time on the DSP (Digital Signal Processor) of the Alto. LEQ (narrow band): This is the linear average narrow band spectrum. The PC with the instant time signal computes the narrow band spectrum. In auto store mode, the LEQ narrow band spectrum is reset for each new average.
	These are the cursor indicators. The red cursor of the time/spectrum graph can be moved to obtain a precise reading of the spectrum or time signal values.
	This is the global level of the selected spectrum.

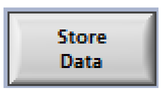
6.6 Export Functions

The SLM module allows the measurement results to be exported with the automatic store mode. However, the user can manually export the results with two functions described in the following sections.

6.6.1 Export Graph Function

The lower and upper graphs of the SLM interface have this button which allows the selected spectrum or historic data to be exported to a text file format. The file can be used in a standard spreadsheet program such as Excel.

6.6.2 Store Data Function

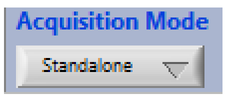


This button, which is close to the “Start/Stop” controls, can be used to store all the data of the last measurement in a .slm file. The file saved can be used with the post-processing interface of the SLM for analysis (see section 6.8 for more details).

6.7 Standalone Acquisition

In the SLM Acquisition interface the the alto unit can be set to run in standalone. The standalone acquisition do not require to be connected to a PC in order to work. The SLM process is embedded in the unit and the data are recorded in its flash memory. The unit can be powerdown without the risk of loosing any recorded data. At any time during the measurement it is possible connect a PC to the Alto unit in order to view and/or download the recorded data.

6.7.1 Standalone Setup

	When the standalone mode is activated, the "*Start average*" control is renamed "*Start Standalone"*. As the Acquisition Mode is switch to standalone, the standalone setup panel is automatically called. Note that the Standalone acquisition will not be activated until the Start Standalone is pressed.
	The Set Standalone access the standalone setup (figure below). The Start Mode is set here. The panel also let you review the acquisition parameters that will be loaded in the unit. An indication of the available measurement duration is also display.

The following figure shows the dialog box for the "*Set Standalone"* function:

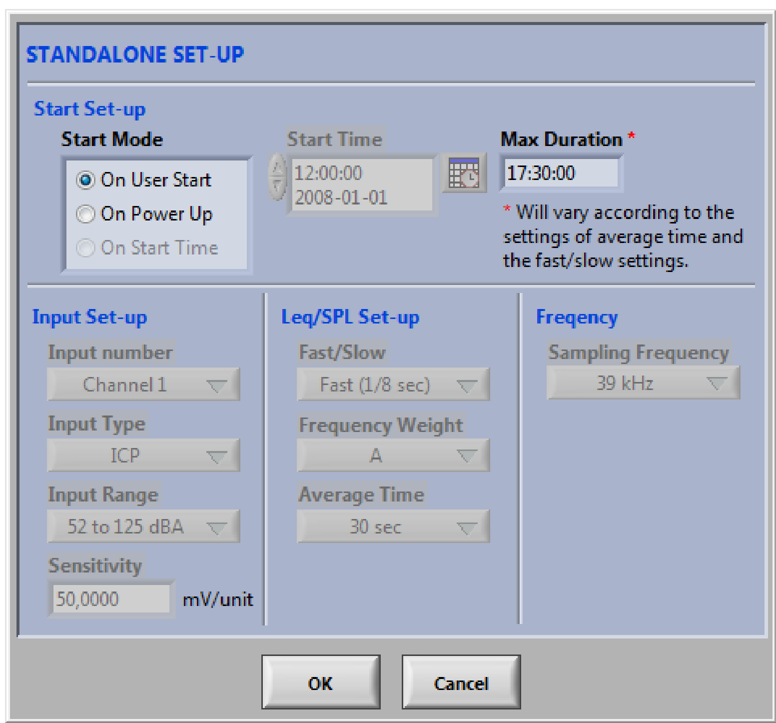


Figure 35: Auto store dialog box

When the start mode is set to "On User Start", the standalone process will be started with the PC. When th start mode is set to "On Power Up", the standalone process will started at the next power up of the Alto Unit.

6.7.2 Standalone Acquisition Interface

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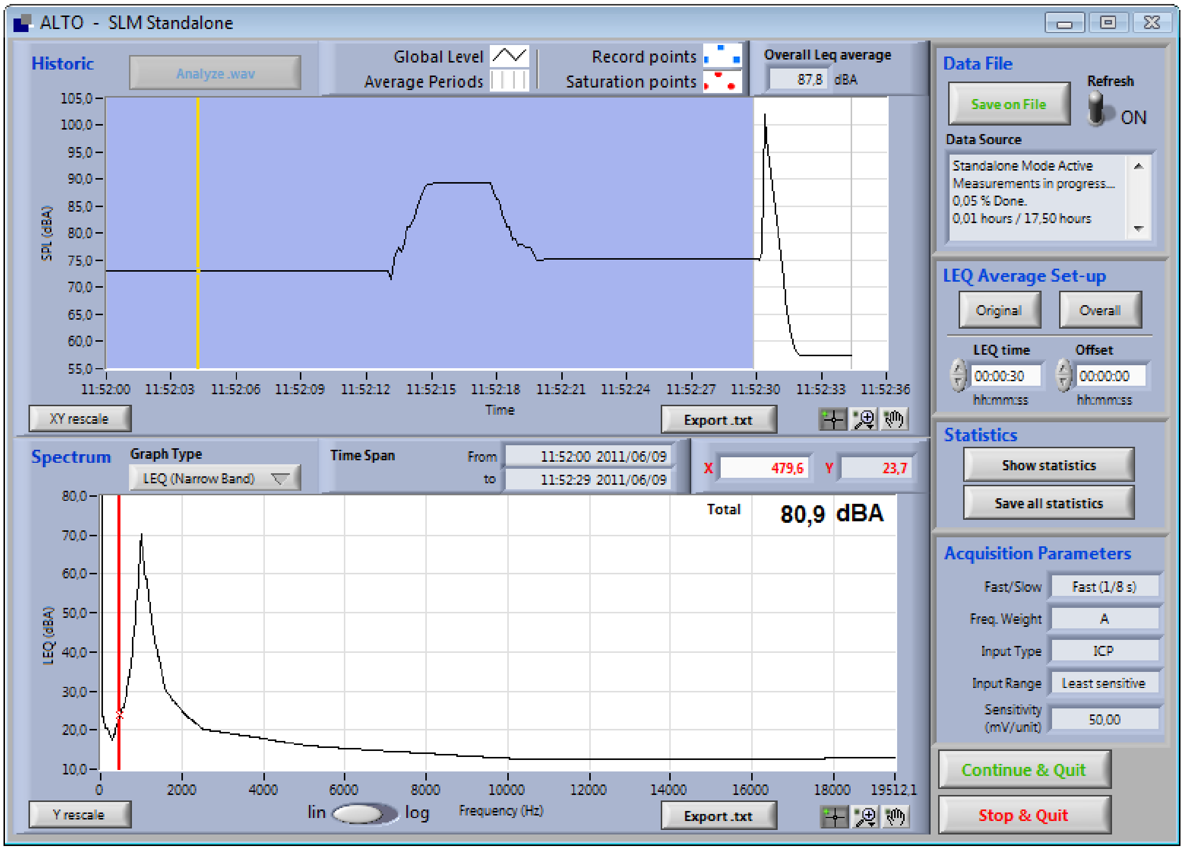


While the Alto unit is in standalone acquisition, it is possible to connect a PC to the unit and run the Alto software. It will automatically open the SLM Standalone interface, which is basically the same as the Post-Treatment Interface (section 6.8).

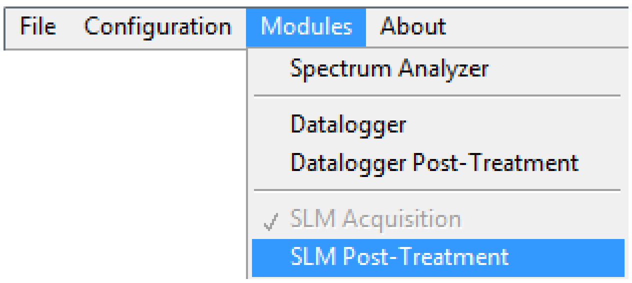
The progress of the measurement is display in the Data Source indicator. The graphs periodically refresh themselves as new data are available. It is possible to lock the display by disabling the Refresh button.

There is two ways to close the application. The “Continue & Quit” will let the acquisition continue while the “Stop & Quit” will stop the acquisition. Be sure to save the data before stopping the acquisition because the data will be cleared from the unit.

6.8 Post-Treatment Module

The post-treatment function along with the auto-store feature of the Alto SLM module is a very useful tool for the analysis of sound measurement over long periods of time. Section 6.4 explains how to configure and take a long-period measurement with the SLM module. The current section explains how to analyze the data files created during the measurement.

The SLM post-treatment interface is launched with the "*SLM Post-Treatment"* function in the *Modules* menu of the SLM main interface.



The following figure presents the main interface of the post-treatment module of the Alto SLM:

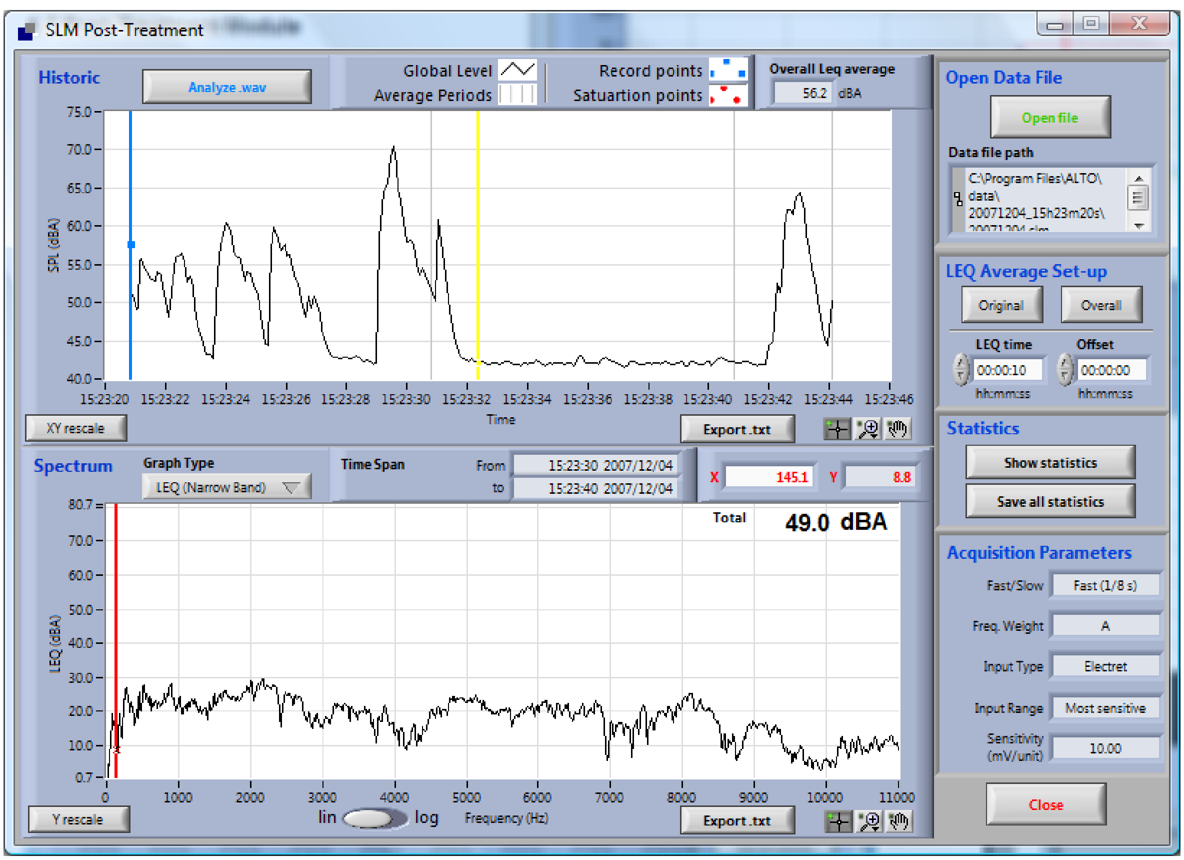
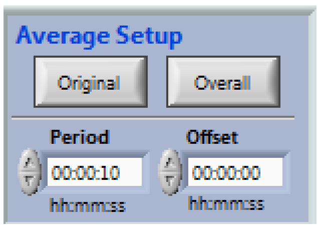


Figure 36: Main Post-Treatment SLM interface

This interface starts by asking for a .slm file. The user can also open a new .slm file in the post-processing using the "*Open File"* button. The Alto SLM automatically saves this type of file when the auto-store mode is used (see section 6.4 for more details). Note that the user can manually save a .slm file with the "*Store Data"* function of the main SLM interface (see section 6.6.2).

The post-treatment interface presents two graphs: 1) the historic of the SPL global levels and 2) the LEQ spectrum for a selected average. The LEQ average is selected with the yellow cursor in the historic graph. Sections 6.8.1 and 6.8.2 explain both graphs and the related options, in detail.

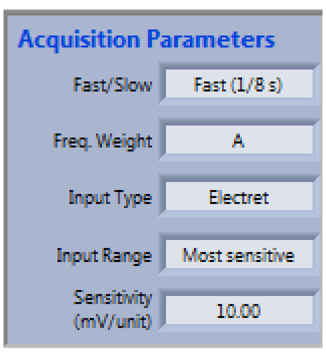
The LEQ averages can be regrouped to increase the LEQ average duration with these controls (see section 6.8.1 for more details):



The historic graph also presents the recording events with blue dots. The blue cursor can be used to select a specific recording (.wav file) in the historic graph and the "*Analyze .wav"* button allows a dedicated function to be launched for the playback and analysis of the selected wave file. Section 6.8.3 presents this function.

The statistics of the selected LEQ average can be analyzed with the "*Show Statistics"* function. Section 6.8.4 presents this feature.

In the main interface, the user can obtain the information about the set-up of the SLM instrument used during the recording with the following group of indicators:



6.8.1 SPL Historic Graph

The graph in the following figure presents the SPL global level historic for the entire recording. This graph also presents the LEQ average time limits with the gray vertical lines. The input saturations are illustrated with red dots. The blue dots and the blue cursor allow the identification and selection of a specific recording event in the historic graph. Then, the "*Analyze .wav"* button can be used to analyze and to listen to the selected wave file.

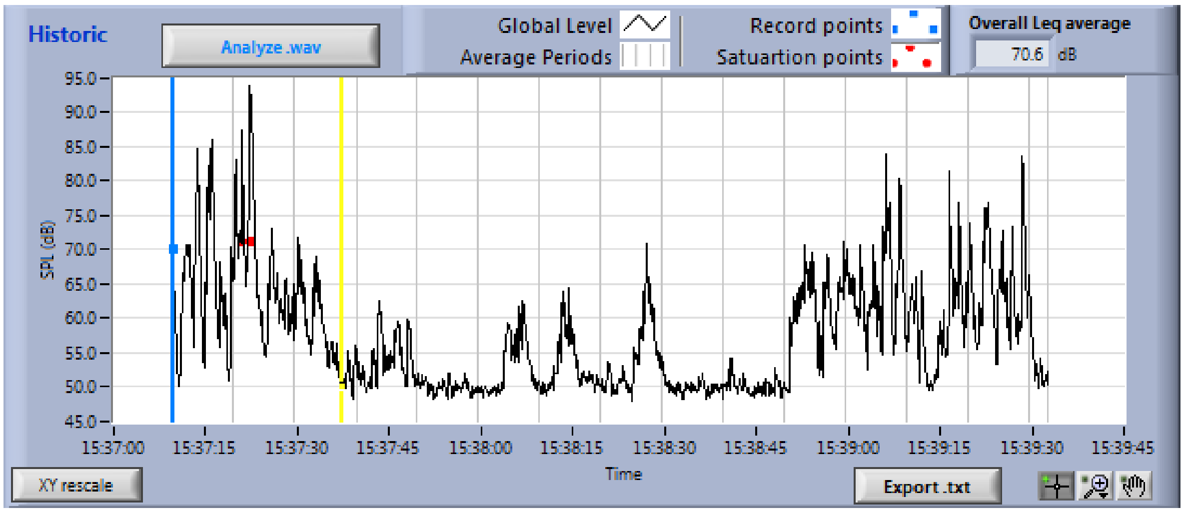
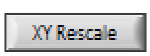
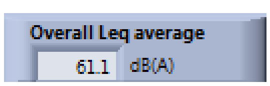
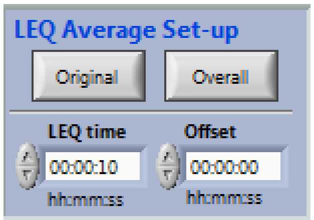


Figure 37: Historic graph of the post treatment SLM interface

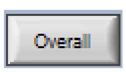
The following table explains the controls and indicators of the historic graph:

Indicator/control	Description
	This button allows the rescaling of the Y and X scales of the historic graph. The interface computes an auto scale for the X and Y scales.
	This indicator presents the overall LEQ average value for the entire recording. This value represents the overall equivalent global level of the whole recording.
	This function allows the SPL historic to be exported to a text file format. The file contains a header with the starting time, step time and LEQ average duration.
	This button allows the starting of an analysis and playback interface for the selected recording event (.wav file). The recording event is selected with the blue cursor in the historic graph. Section 6.8.3 explains the wave file analysis function.
	This group of controls can be used to move cursors, zoom, and pan the display. By default, the cursor setting is used to allow the selection of the recording events and the LEQ average with the blue and yellow cursors.

The following controls allow the LEQ average time to be changed:



Just after the opening of the .dat file, the LEQ duration is set to the original LEQ average duration used during the recording and the offset is set to zero. The post-processing interface of the SLM module allows the historic data to be reformatted by grouping the averages to increase the duration of the LEQ averages. For instance, if the original LEQ duration used during the recording is 10 seconds, the historic graph can be reformatted to extend the LEQ average length by multiples of 10 seconds. It is recommended to use a short LEQ duration during the measurement process. This allows a more precise reformatting. The following table explains each control that can be used to reformat the historic graph:

Control	Description
	This button can be used to retrieve the original LEQ duration.
	This button allows the LEQ duration to be increased to the entire recording period. Then, the "*LEQ time length"* control is set to the total recording duration and the offset is reset to zero. After this operation, the historic graph contains only one LEQ average.
	The *LEQ time* can be changed by multiples of the original LEQ duration. Then, the historic graph is updated and the LEQ periods illustrated with the white vertical lines reflect the new LEQ duration.
	This control can be used to change the starting time of the historic graph. The minimum step size for changing the starting time of the historic graph is the original LEQ duration. When the function Overall is used, the offset is reset. The offset is also reset if a new .dat file is opened.

6.8.2 LEQ Spectrum Graph

The yellow cursor of the historic graph is used to select a specific LEQ average of the recording. Then, the lower graph of the interface presents the corresponding LEQ spectrum (narrow band or 1/3 octave) recorded during the measurement. The following figure presents the spectrum graph:

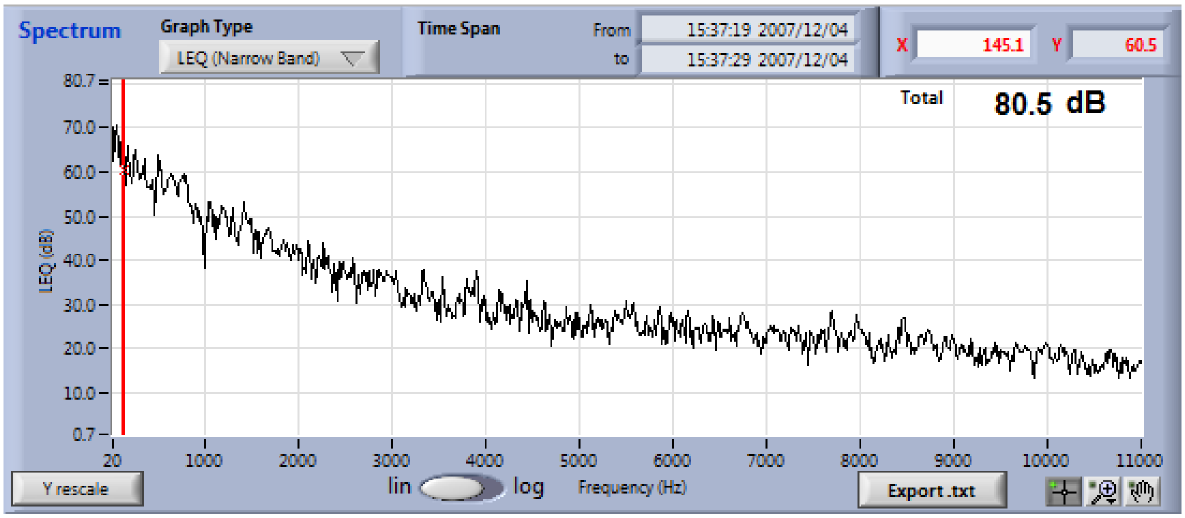
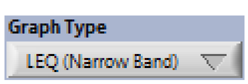


Figure 38: Spectrum graph of the post treatment SLM interface

The following table explains the controls and indicators related to the LEQ spectrum graph:

Indicator/control	Description
	This button allows auto scaling to be applied for the Y scale of the spectrum graph.
	This control allows a linear or a logarithmic scale to be used for the X-axis. This control is available for the narrow band spectrum graph type only.
	This function allows the selected spectrum to be exported to a text file format. The file contains a header with the start and stop times of the selected LEQ average and the global level of the spectrum.
	Two types of spectrum can be presented: LEQ (Narrow band): This is the average narrow band spectrum of the selected LEQ average. The PC has computed this average spectrum during the recording from the raw time signal. LEQ (1/3-octave): This is the average 1/3-octave spectrum of the selected LEQ average. This average 1/3-octave spectrum has been computed by the DSP (Digital Signal

	Processor) of the Alto unit with a bank of filters.
	These indicators present the start and stop times of the selected LEQ average. The LEQ average is selected with the yellow cursor of the historic graph.
	These indicators show the X and Y values of the spectrum graph cursor.
	This is the global level of the spectrum. It corresponds to the equivalent global level (LEQ) of the selected LEQ average. This information is also available in the statistic analysis function.
	This group of controls can be used to move cursor, zoom, and pan the display. By default, the cursor setting is used to allow a precise reading of the spectrum levels.

6.8.3 Playback/Analysis Wave Function

The post-processing module of the Alto SLM includes a dedicated function that allows the playback and the analysis of the recorded wave files. The SLM instrument allows the automatic storing of the time signal in a wave file format. The time signal recording can be performed continuously during the entire measurement or for triggered events (see sections 6.3 and 6.4 for more information about the time signal recording).

The following figure presents the playback and analysis interface. This interface is launched with the "*Analyze .wav"* button. The function starts with the selected .wav file. The selection is done with the blue cursor in the historic graph. The user can also change the wave selection with the "*#Wave"* control of the wave analysis interface.

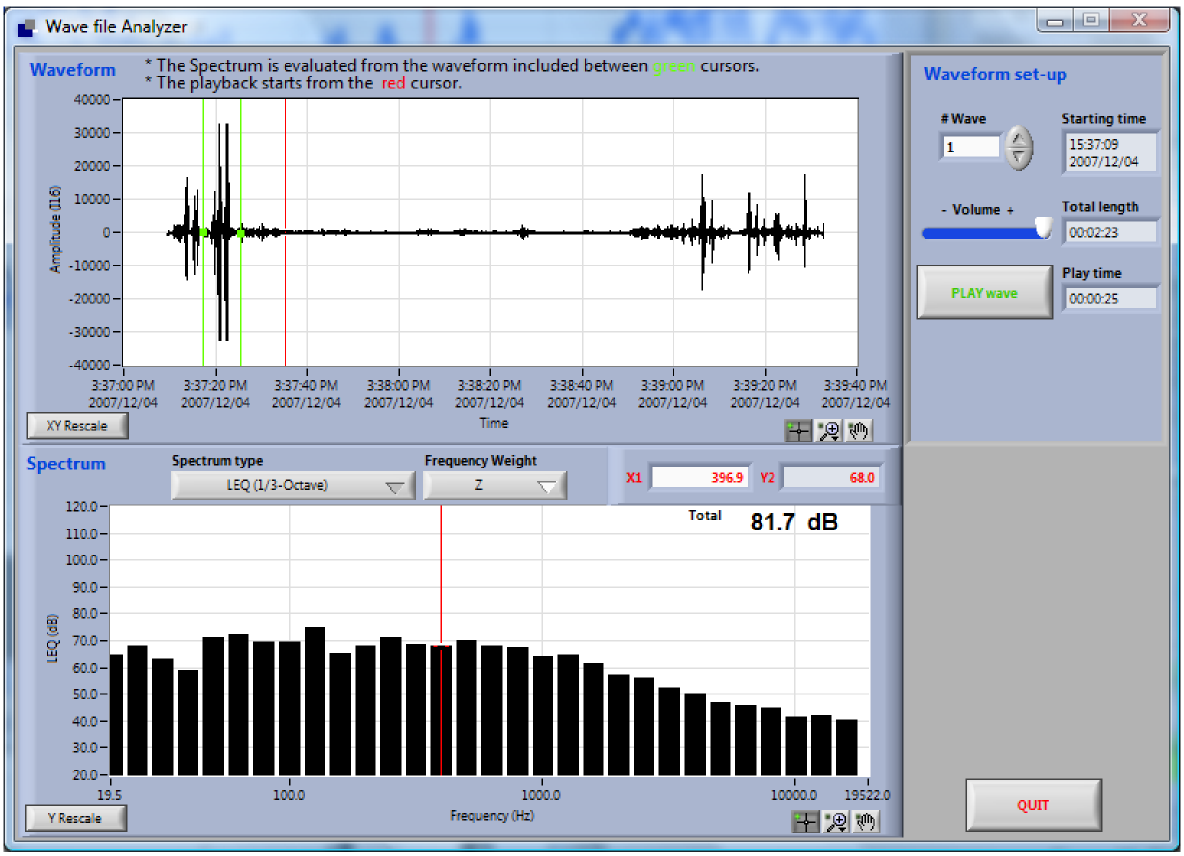


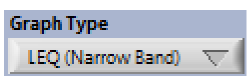
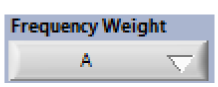
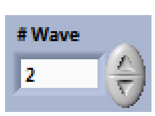
Figure 39: Playback and Analysis interface

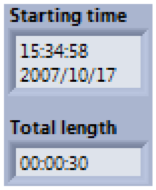
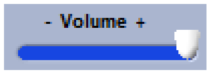
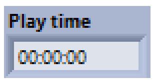
This interface has two graphs:

1) The waveform graph that contains the time signal and two green cursors, which allow the selection of a specific portion of time.

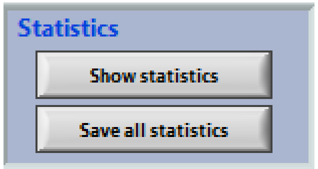
2) The spectrum graph that presents the average spectrum for the selected time portion of the waveform. The time portion selection is done with the two green cursors in the waveform graph.

The following table explains all the controls and indicators of the wave file analysis function:

Indicator/control	Description
 and 	This button allows auto scaling to be applied for the Y and X scales of the waveform graph. This button allows auto scaling to be applied for the Y scale of the spectrum graph.
	This group of controls can be used to move cursors, zoom, and pan the display. By default, the cursor setting is used to allow the selection of the time portion in the waveform graph. If the selection of the time portion changes, the spectrum graph is automatically updated for the new selection.
	This control allows the use of a linear or a logarithmic scale for the X-axis. This control is available for the narrow band spectrum graph type only.
	Three types of spectra can be presented: LEQ (Narrow band): This is the average narrow band spectrum of the selected time portion of the waveform graph (selected with the two green cursors). The average is computed block by block and with a hanning time window. The size of the block is fixed to allow an FFT graph with 4096 lines. LEQ (Octave): This is the average octave spectrum of the selected time portion in the waveform graph (selected with the two green cursors). This average octave spectrum is computed with frequency filters applied on the narrow band spectrum. LEQ (1/3-Octave): This is the average 1/3-octave spectrum of the selected time portion in the waveform graph (selected with the two green cursors). This average 1/3-octave spectrum is computed with frequency filters applied on the narrow band spectrum.
	This control allows the selection of the frequency weighting for the average spectrum graph. The A, C or Z weightings are available.
	This control allows the selection of the recording event if the recording has more than one wave file. The playback/analysis interface starts with the recording event selected in the historic graph of the main SLM post-processing interface.

	This is the starting time and the duration of the selected recording event.
	This control allows the selection of the output volume of the PC sound card during playback.
 	This button allows the playback of the recording event to start. The playback starts at the position of the red cursor in the waveform graph. Once the playback is started, this button changes to a “Stop” control.
	This indicator presents the current playtime. It corresponds to the red cursor position in the waveform graph.

6.8.4 Statistics Analysis Function



These functions allow the standard Lx% statistics to be computed on the SPL global levels of the selected LEQ average or on all LEQ averages of the recording.

The "*Save all statistics"* button can be used to export the statistics of all the LEQ averages of the entire recording. Then, a text file format is created with the statistics results for all the LEQ averages as well as a header describing the content of the file.

The "*Show statistics"* button of the main post-processing interface launches an interface, which allows the analysis of the statistics for a specific LEQ average selected with the yellow cursor in the historic graph. The following is the statistics analysis interface:

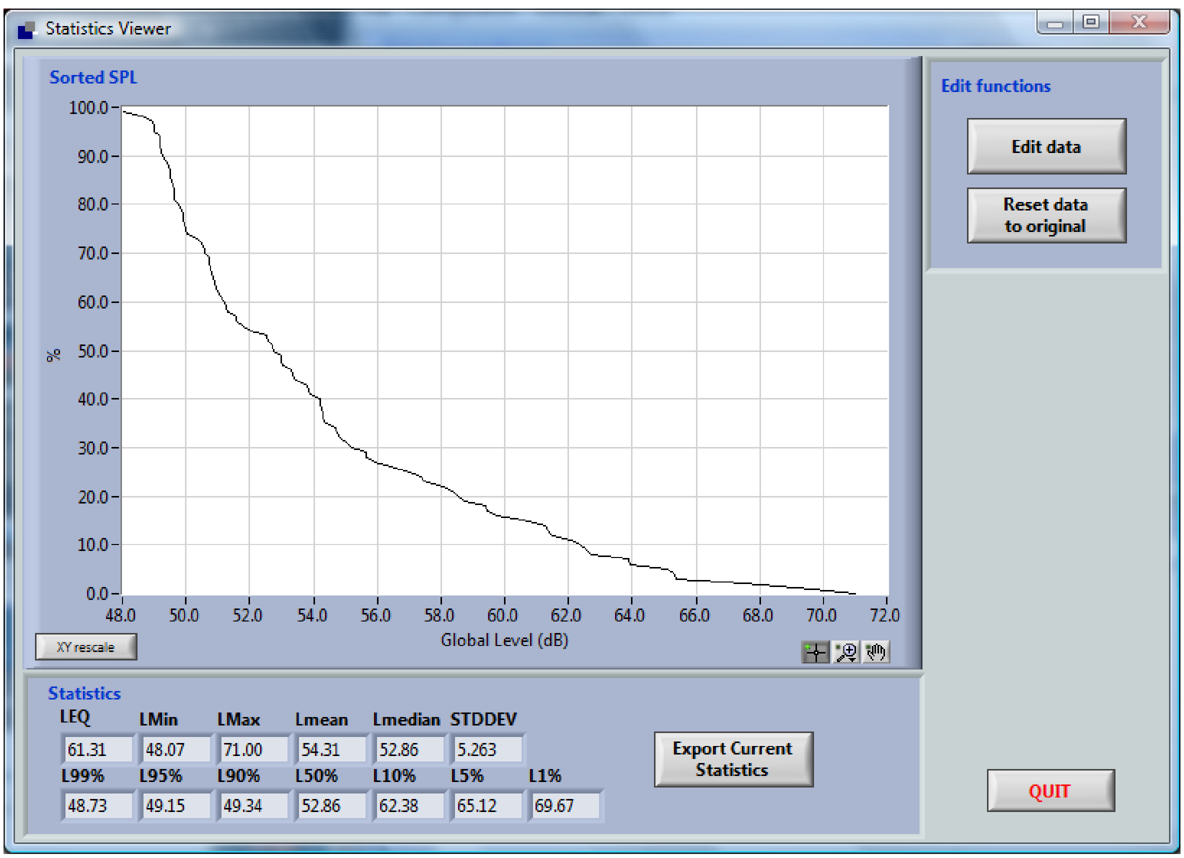


Figure 40: Statistics Analysis Interface

The following table explains the controls and indicators of the statistics function:

Indicator/control	Description
Leq 55.29	This value is the equivalent sound pressure level for the LEQ average.
LMin 16.52 LMax 67.83	These are the maximum and the minimum SPL values for the selected LEQ average.
Lmedian 50.34	This value is the median SPL computed with the following equation: (LMin+LMax)/2
Lmean 49.60	This is the average SPL value computed for all SPL values of the selected LEQ average.
STDDEV 8.00	This is the standard deviation of SPL values of the selected LEQ average.
L1% 64.87 to L99% 28.45	These are the standard Lx% statistics of the SPL values of the selected LEQ average.
Export Current Statistics	This button can be used to export the statistics of the selected LEQ average. Then, a text file format is created with the statistic results and a header describing the content of the file.
XY Rescale	This button allows the rescaling of the Y and X scales of the sorted SPL graph. The interface computes an auto scale for the X and Y scales.
Edit data	This button allows the SPL historic to be edited. A portion of the SPL historic can be removed or kept with this function. This function is useful when saturations or special events are present in the SPL historic of the selected LEQ average. See the following paragraph for more information about this function.
Reset data to original	This button allows the original SPL historic to be retrieved if the *Edit data* function has been used.

The following figure presents the interface of the "*Edit Data"* function. This function allows some portions of the SPL historic to be removed or kept for problematic LEQ average. For instance, if the selected LEQ average contains saturations or PC overruns (illustrated with red and orange dots on the historic curve), the user can use the *Keep* or *Remove* functions to avoid these events. Then, the statistic results will be recomputed for the new SPL historic.

Note: Once an SPL historic has been edited, the LEQ value is not valid anymore since the LEQ value is not computed with the SPL historic data but with the linear average obtained during the measurement.

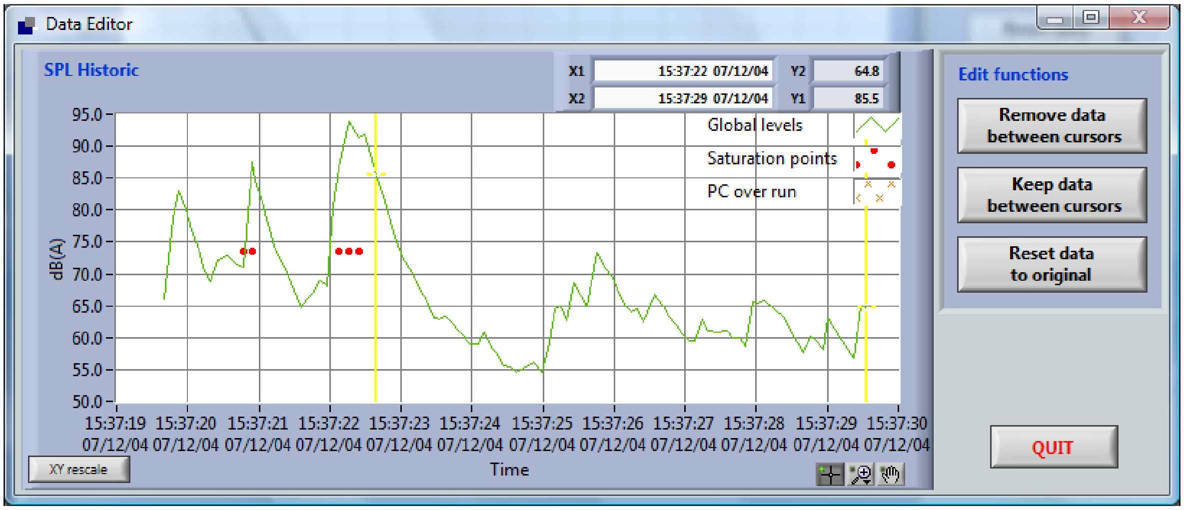
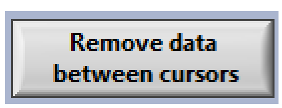
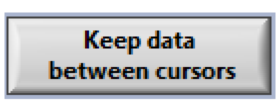
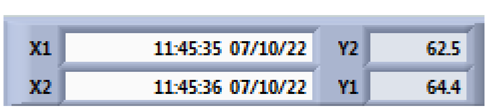


Figure 41: SPL Historic Edit Interface

The following table explains the controls and indicators of the "*Edit data"* function:

Indicator/control	Description
	This button allows the rescaling of the Y and X scales in the historic graph. The interface computes an auto scale for the X and Y scales.
	This function allows the removal of the SPLs contained in between the two yellow cursors.
	This function allows the SPLs contained in between the two yellow cursors to be kept.
	This button allows the original SPL historic to be retrieved.
	These indicators present the X and Y position of both yellow cursors.
	This group of controls can be used to move cursors, zoom, and pan the display. By default, the cursor setting is used to allow a precise reading of the position of the yellow cursors.

Appendix A: Alto Driver and Software Installation Procedure

! The Alto unit is not automatically recognized by Windows.

Follow each step below precisely to ensure proper installation of the system.

1) Find the installation CD or the installation file on the Soft dB website.

2) Lauch the installation and follow the procedure. By default, the Alto software is installed in the C:\Program Files\Alto directory. **Do not launch the software at this point; the USB driver must be installed first.**

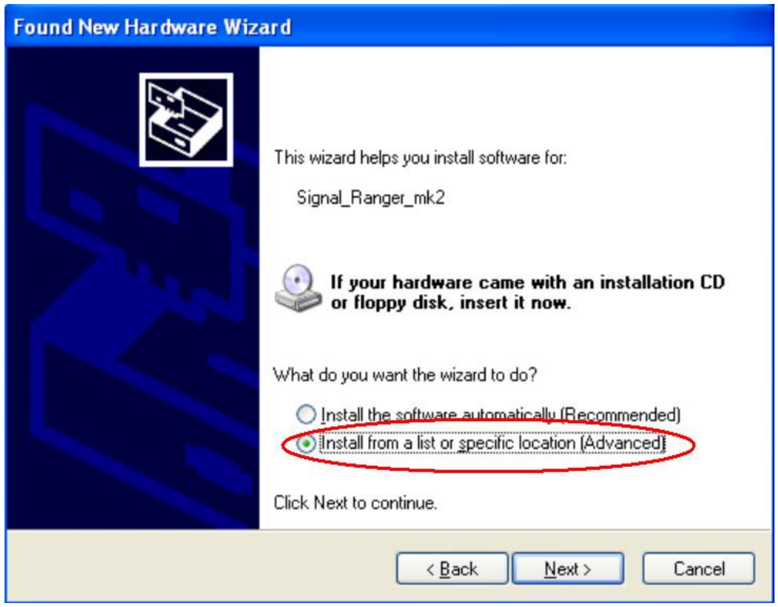
3) Apply power to the Alto unit (power connector on the rear of the unit). Use the power pack provided by Soft dB. The Status Board LED of the Alto unit should turn yellow.

4) Connect the USB cable to the unit (USB input on the rear of the Alto unit).

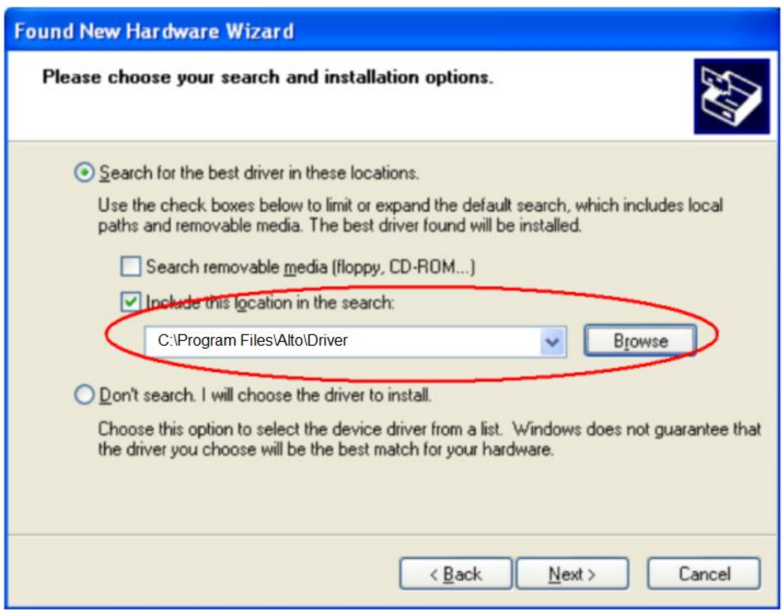
5) Connect the other end of the USB cable to the USB port of your PC.

Windows indicates that a new peripheral has been detected. Follow the procedure below:

Check "*No, not this time"* and then click on "*Next"*

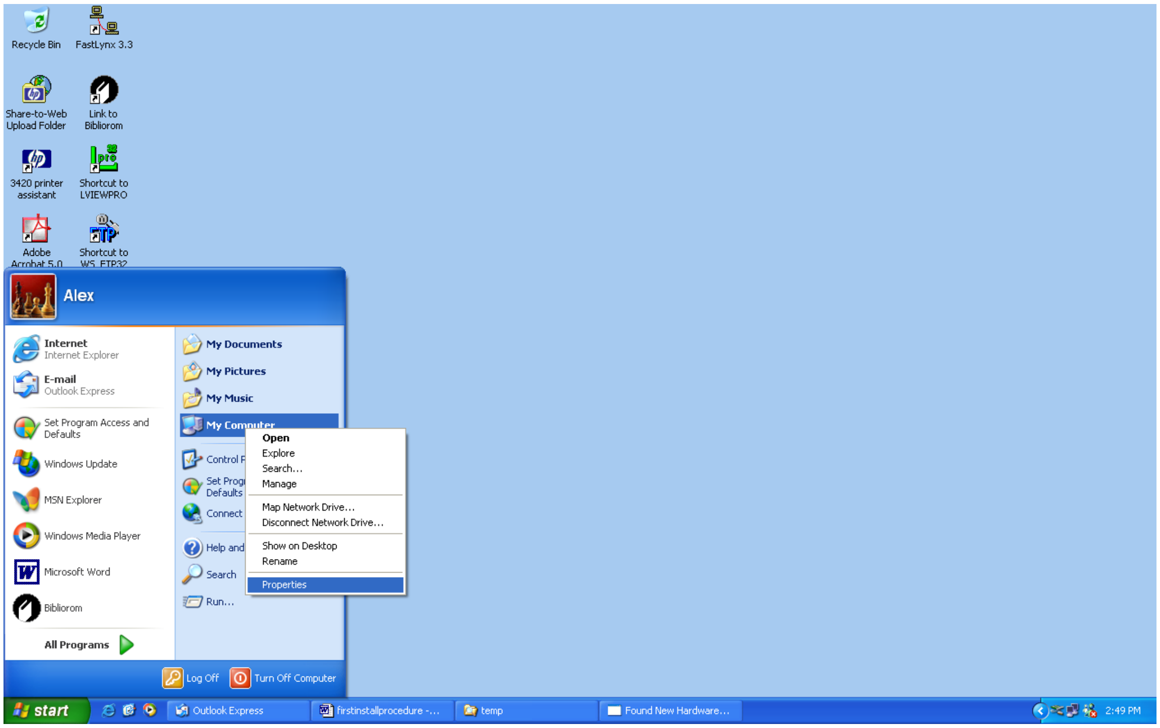


Check "***Install from a list or specific location (Advanced)"*** and then click on "*Next*"

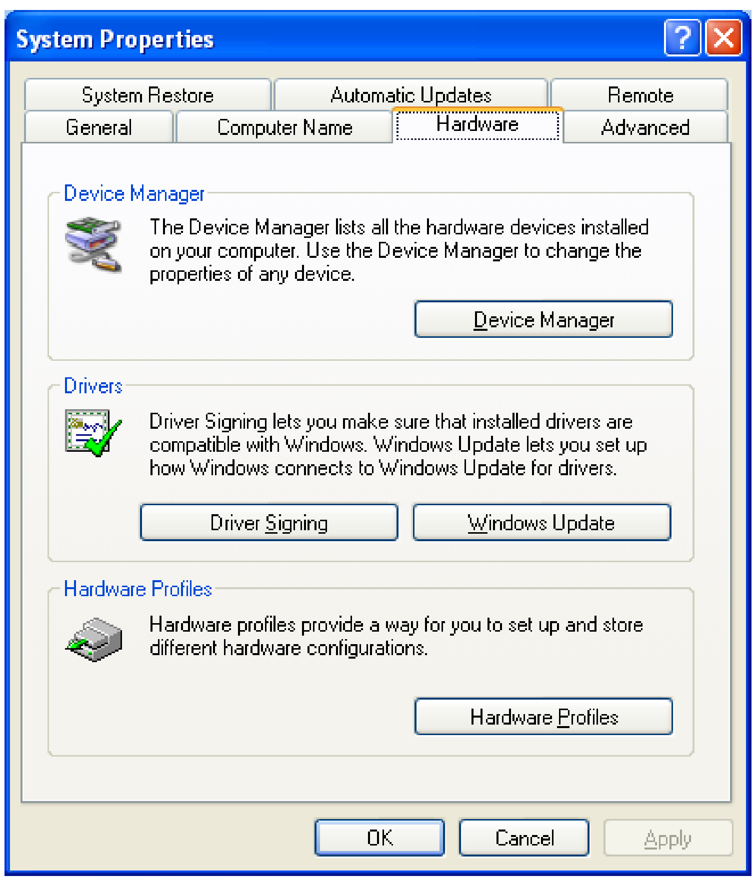


Use the "Browse" button to specify the location of the USB driver for the Alto unit.

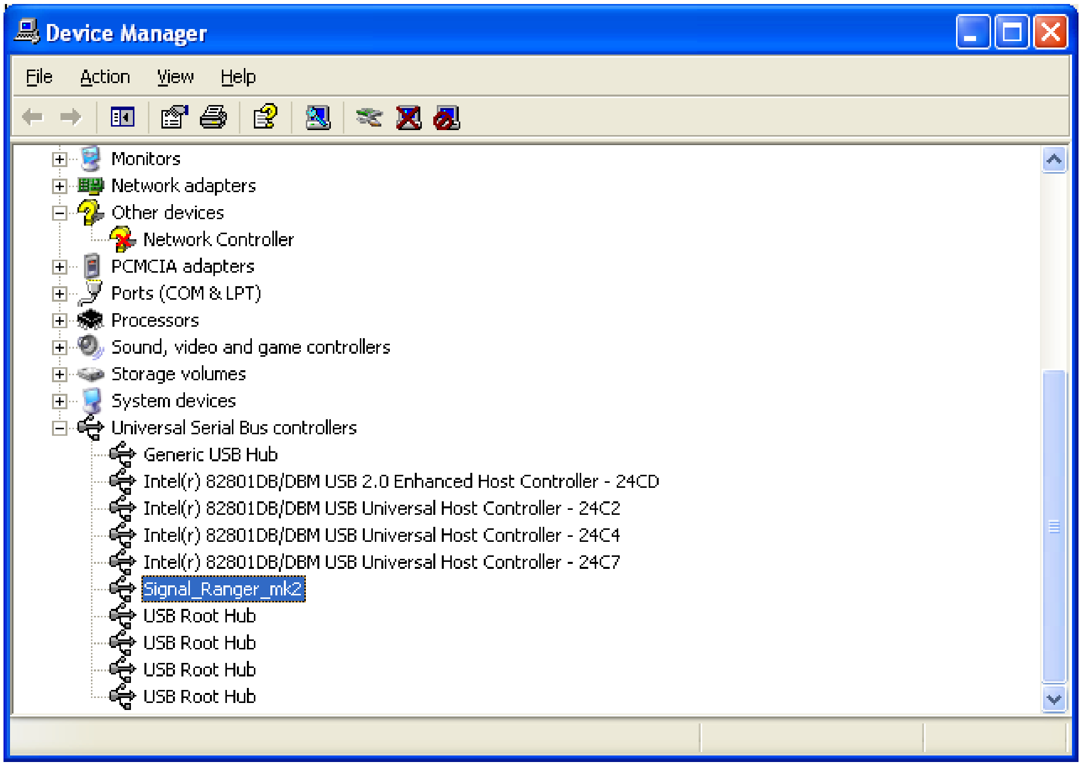
Select the C:\Program Files\Alto\Driver folder (or the one where you installed the software in step 2) and then click on "*Next".* After that, Windows may inform the user that the USB device does not have the logo. Click on "*Continue Anyway".* The driver is installed and the Alto unit is ready to be used. Click on "*Finish".* To test whether the Alto Unit is properly installed, go to "My Computer/Properties".



Right click on "*My Computer*" and select "*Properties*"



Click on "*Device Manager*" in the *Hardware* tab



Expand the USB tree and check for the following entry: Signal_Ranger_mk2. The Signal Ranger DSP board is the processor of the Alto unit.

Appendix B: Alto input/output specifications

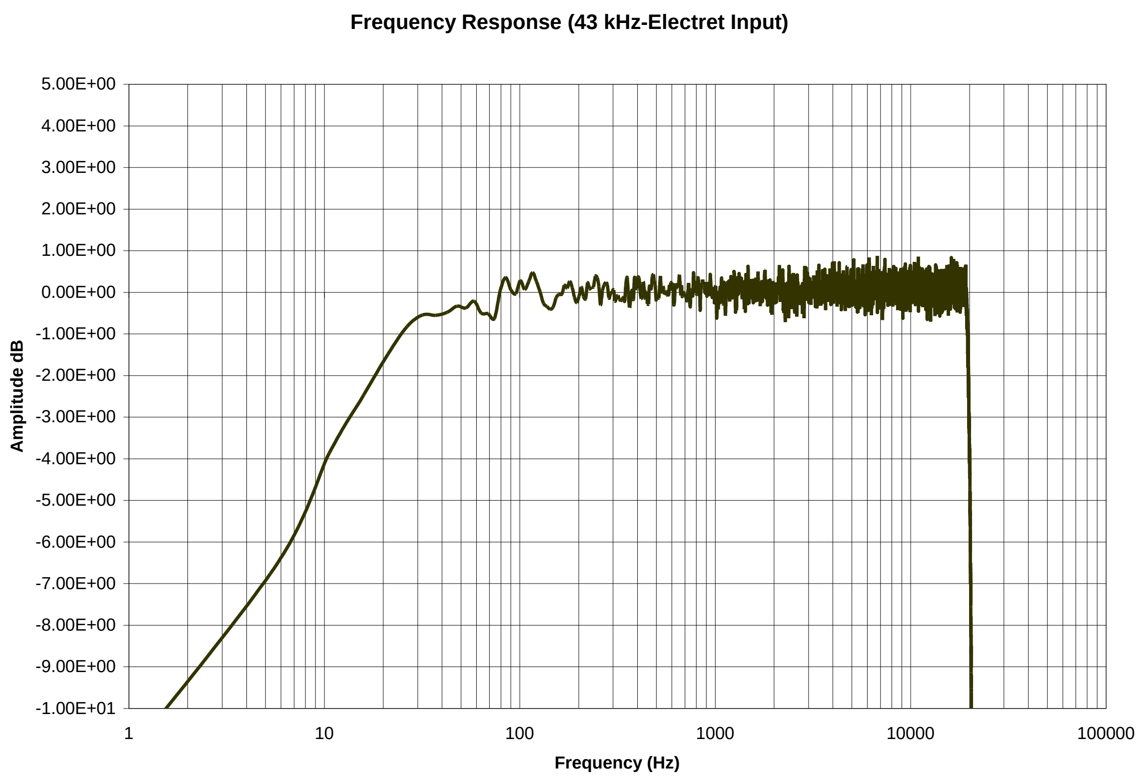
	Specification	Commentary
Direct AC, Direct DC and ICP inputs SNR	85.0 dB	This is the signal–to-noise ratio. This result is for an input dynamic of ±3V and for a sampling frequency of 43 kHz (span of 21.7 kHz). The total dynamic range including input gain is 85+25: 110 dB.
Output SNR	88.0 dB	This is the signal–to-noise ratio for the analog output. This result is for an output dynamic of ±1V and for a sampling frequency of 39 kHz (span of 19.5 kHz).
Input impedance	85 kOhms	For the direct AC and direct DC inputs.
Input Polarization Voltage	2.35 V	For the direct AC and direct DC inputs. Maximum 1 mA.
Output impedance	50 Ohms	Maximum 1 mA.
Input gain accuracy	0.05 dB	This is the precision of the input gain for different input range settings. This result is for a calibrated Alto. The worse case is close to 0.05 dB.
Input offset accuracy	0.01%	This is the precision of the offset. Once calibrated, the input offset should be close to zero with a precision of 0.01%. This result is for a calibrated Alto.
Input distortion	-93.5 dB	This is the maximum amplitude of the harmonics when a full scale 1000 Hz sine wave is present on an analog input.
Output distortion	-90.0 dB	This is the maximum amplitude of the harmonics when a full scale 1000 Hz sine wave is generated by the analog output.
Input crosstalk	-95.0 dB	This is the crosstalk between both inputs of the Alto.
Output crosstalk	In noise	On all output channels, the bandwidth noise is greater than the cross talk.

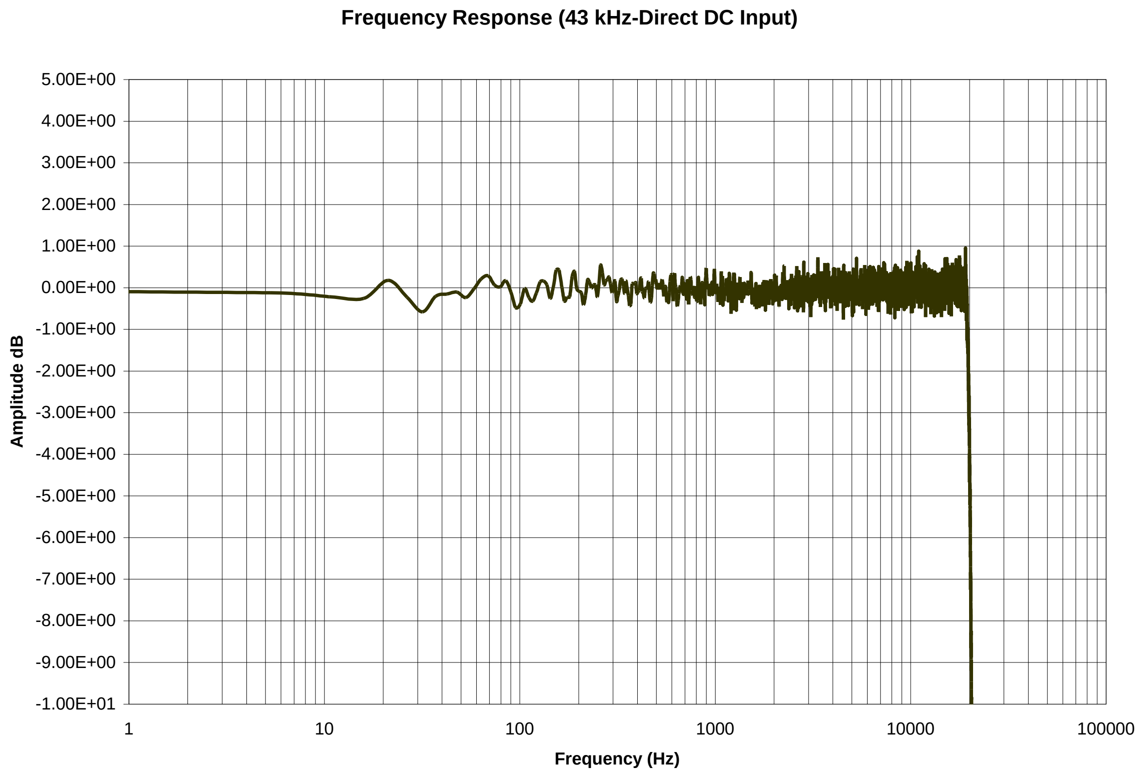
Input Frequency Response

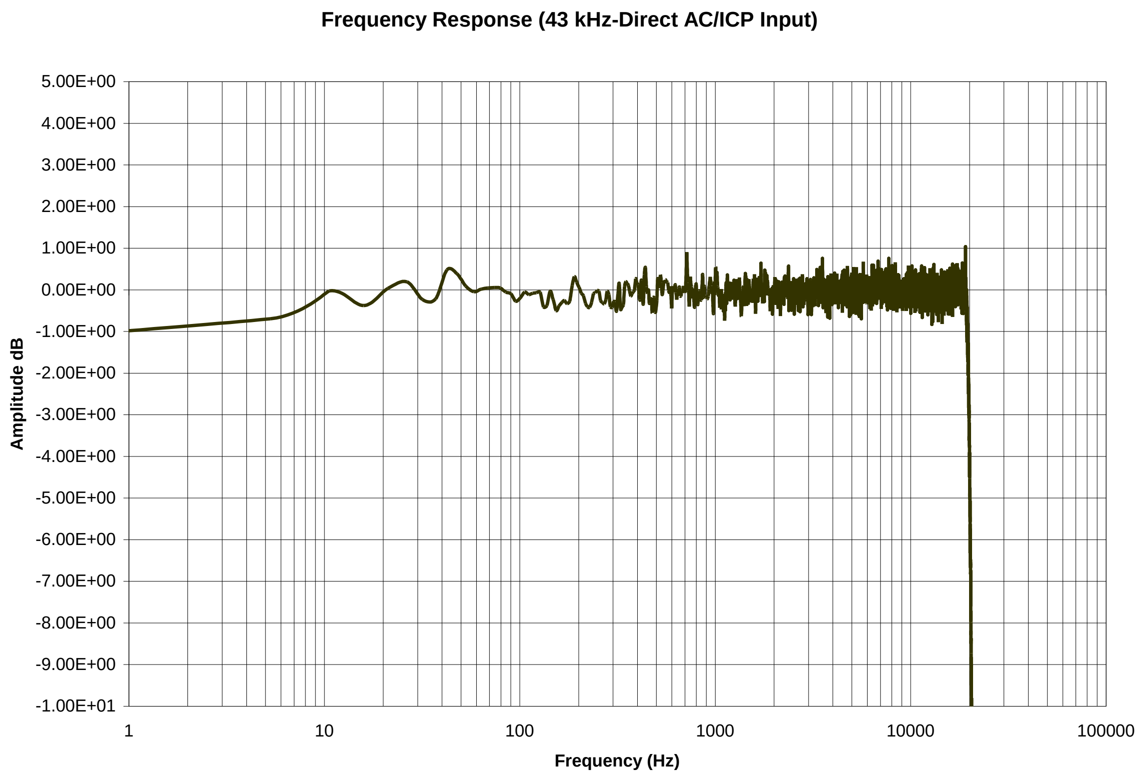
The Alto system has three distinct sampling modes: 1) 43kHz (span of 21.7 kHz with output disabled) 2) 39 kHz (span of 19.5 kHz) and 3) all other frequency span set-ups. The following graphs present the input frequency response for these three sampling modes:

Mode 1: 43 kHz (span of 21.7 kHz with output disabled)

In this sampling mode, the sampling frequency is fixed at 43 kHz. The following figures present the frequency response in this sampling mode for all input types.

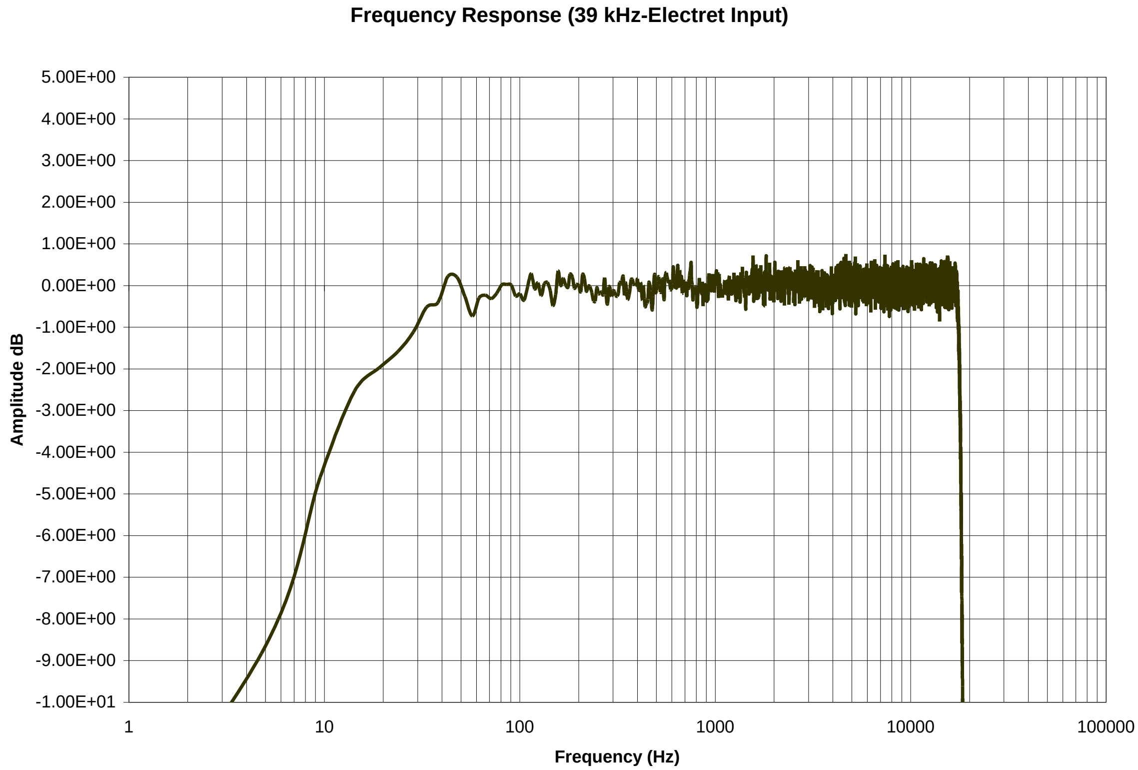


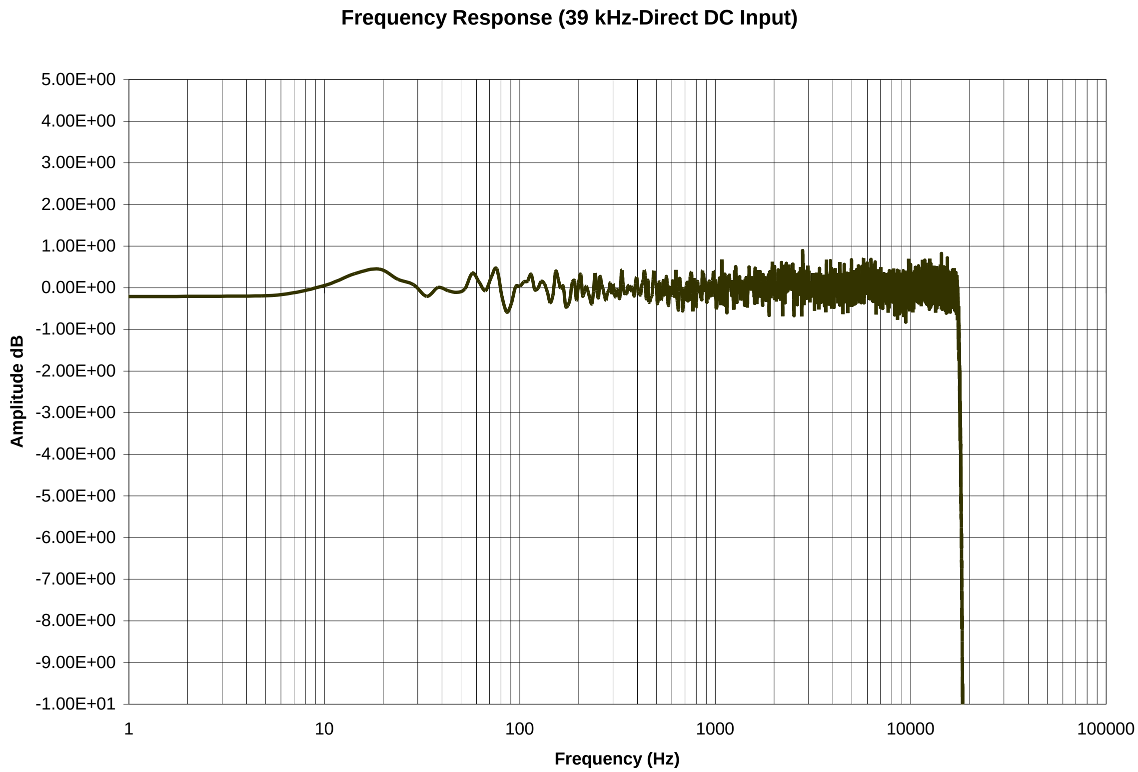

Frequency Response (43 kHz-Direct DC Input)
Amplitude dB
5.00E+00
4.00E+00
3.00E+00
2.00E+00
1.00E+00
0.00E+00
-1.00E+00
-2.00E+00
-3.00E+00
-4.00E+00
-5.00E+00
-6.00E+00
-7.00E+00
-8.00E+00
-9.00E+00
-1.00E+01
1
10
100
1000
10000
100000
Frequency (Hz)

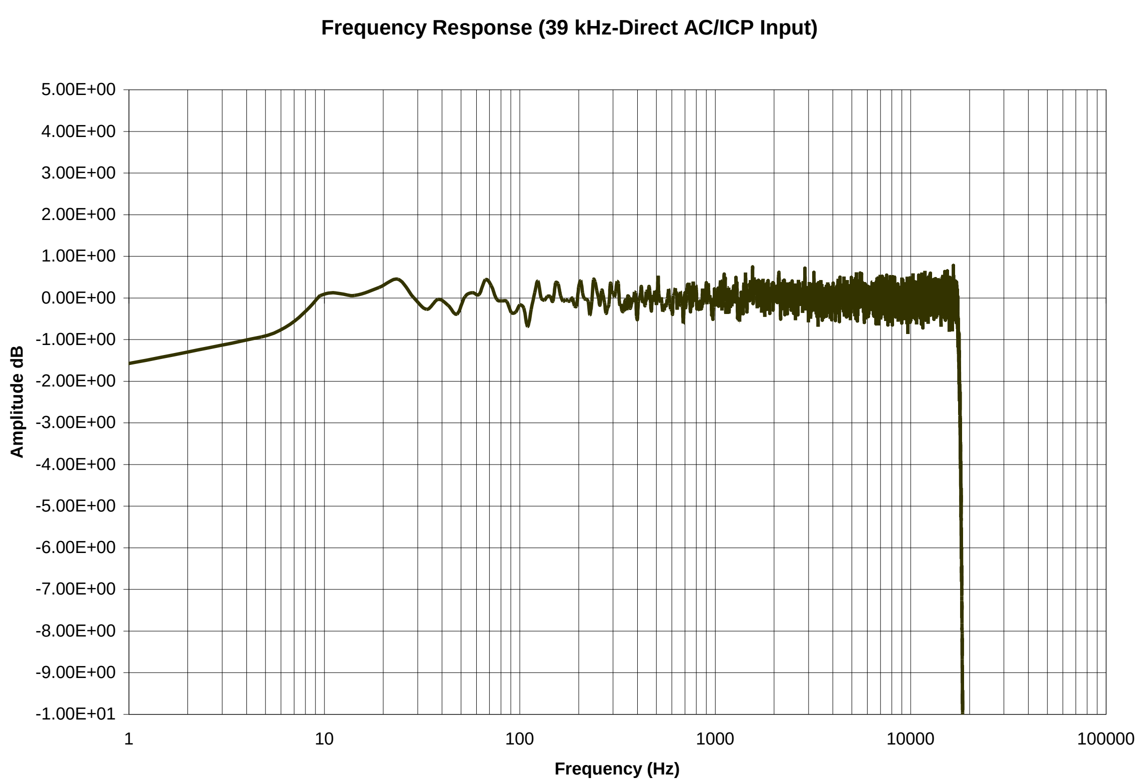

Frequency Response (43 kHz-Direct AC/ICP Input)
Amplitude dB
5.00E+00
4.00E+00
3.00E+00
2.00E+00
1.00E+00
0.00E+00
-1.00E+00
-2.00E+00
-3.00E+00
-4.00E+00
-5.00E+00
-6.00E+00
-7.00E+00
-8.00E+00
-9.00E+00
-1.00E+01
1
10
100
1000
10000
100000
Frequency (Hz)

Mode 2: 39 kHz (span of 19.5 kHz)

In this sampling mode, the sampling frequency is fixed at 39 kHz. The following figures present the frequency response in this sampling mode for all input types.

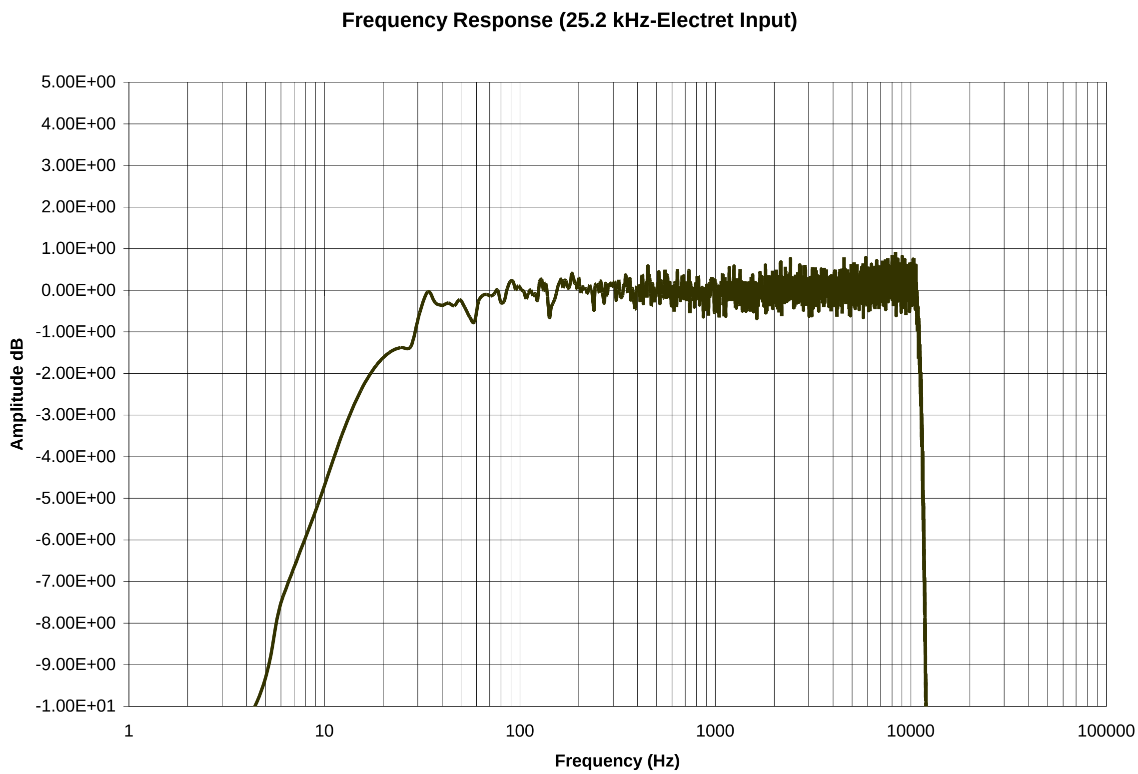


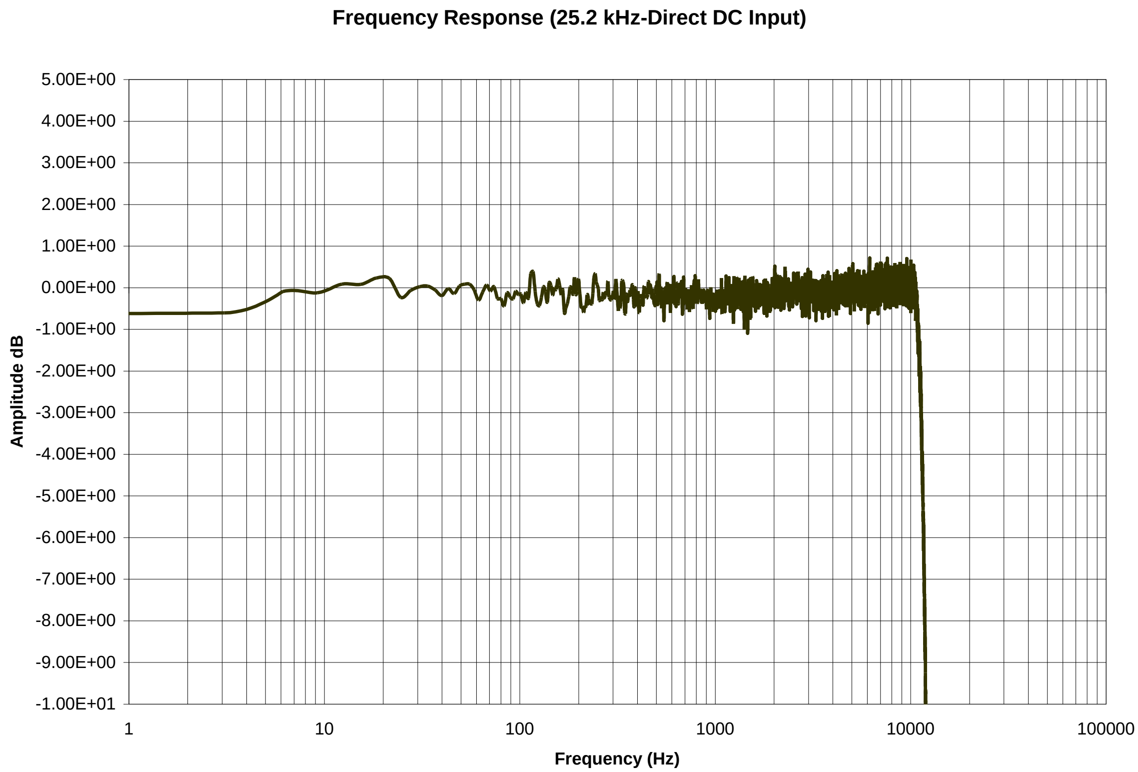

Frequency Response (39 kHz-Direct DC Input)
Amplitude dB
5.00E+00
4.00E+00
3.00E+00
2.00E+00
1.00E+00
0.00E+00
-1.00E+00
-2.00E+00
-3.00E+00
-4.00E+00
-5.00E+00
-6.00E+00
-7.00E+00
-8.00E+00
-9.00E+00
-1.00E+01
1
10
100
1000
10000
100000
Frequency (Hz)

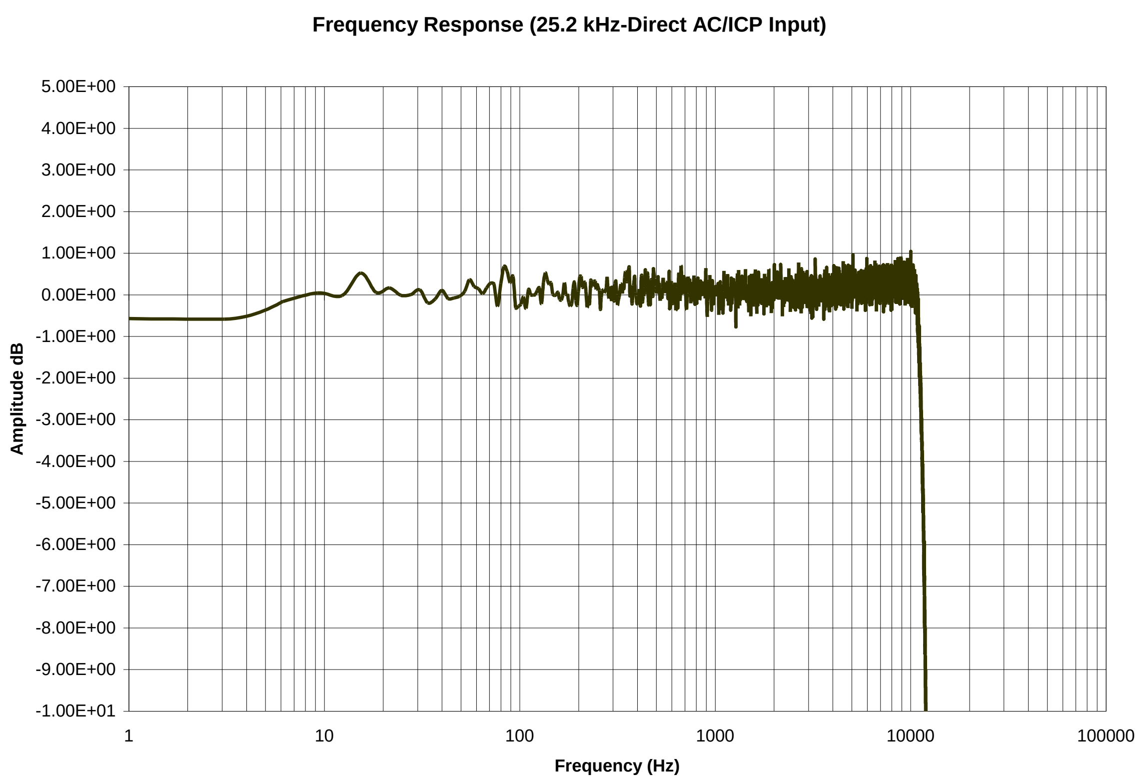

Frequency Response (39 kHz-Direct AC/ICP Input)
Amplitude dB
5.00E+00
4.00E+00
3.00E+00
2.00E+00
1.00E+00
0.00E+00
-1.00E+00
-2.00E+00
-3.00E+00
-4.00E+00
-5.00E+00
-6.00E+00
-7.00E+00
-8.00E+00
-9.00E+00
-1.00E+01
1
10
100
1000
10000
100000
Frequency (Hz)

Mode 3: All frequency span selections except 21.7 kHz and 19.5 kHz

In this sampling mode, the frequency span can be adjusted from 112 Hz (Fs at 286 Hz) to 9536 Hz (Fs at 25.2 kHz). The following figures present the frequency responses for the higher frequency sampling (Fs) set-up, but the shape of the curve remains the same for lower settings.




Frequency Response (25.2 kHz-Direct DC Input)
Amplitude dB
5.00E+00
4.00E+00
3.00E+00
2.00E+00
1.00E+00
0.00E+00
-1.00E+00
-2.00E+00
-3.00E+00
-4.00E+00
-5.00E+00
-6.00E+00
-7.00E+00
-8.00E+00
-9.00E+00
-1.00E+01
1
10
100
1000
10000
100000
Frequency (Hz)

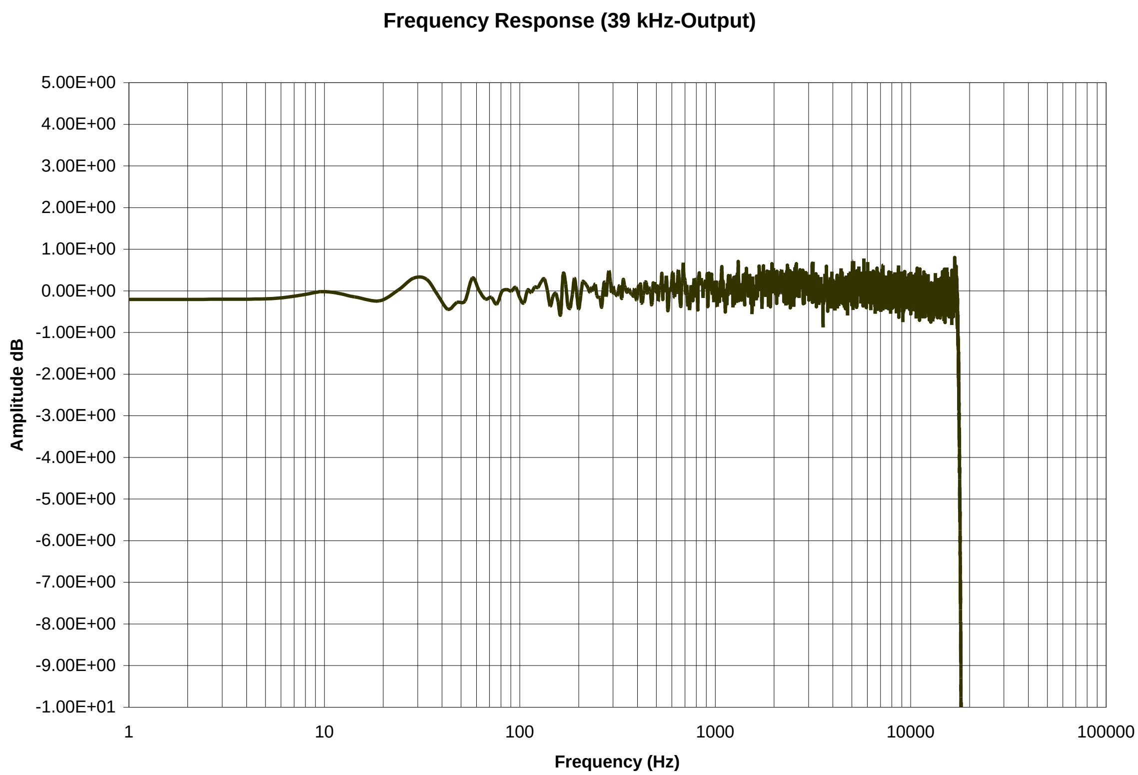

Frequency Response (25.2 kHz-Direct AC/ICP Input)
Amplitude dB
5.00E+00
4.00E+00
3.00E+00
2.00E+00
1.00E+00
0.00E+00
-1.00E+00
-2.00E+00
-3.00E+00
-4.00E+00
-5.00E+00
-6.00E+00
-7.00E+00
-8.00E+00
-9.00E+00
-1.00E+01
1
10
100
1000
10000
100000
Frequency (Hz)

Output Frequency Response

Only two sampling modes can be used with the output enabled: 1) 39 kHz (span of 19.5 kHz) and 2) All frequency span selections except for 21.7 kHz and 19.5 kHz. The following graphs present the output frequency response for these two sampling modes:

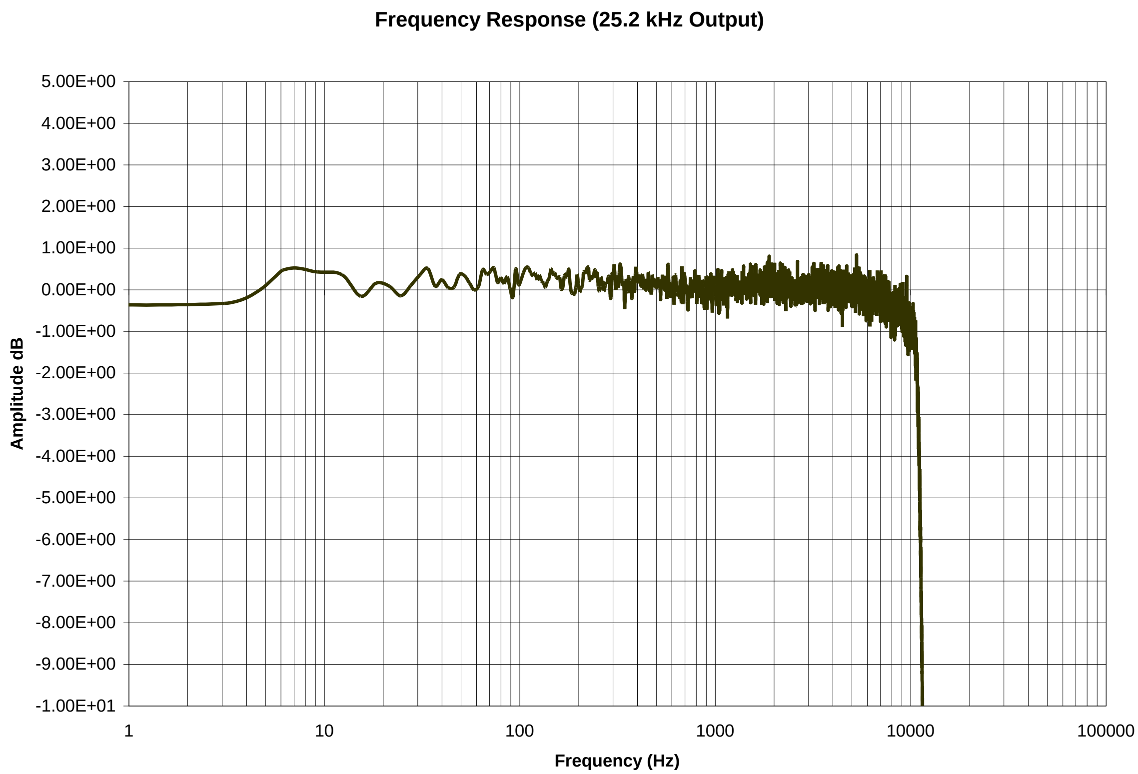
Mode 1: 39 kHz

In this sampling mode, the sampling frequency is fixed at 39 kHz. The following figure presents the output frequency response.



Mode 2: All frequency span selections except for 21.7 kHz and 19.5 kHz

In this sampling mode, the frequency span can be adjusted from 112 Hz (Fs at 286 Hz) to 9536 Hz (Fs at 25.2 kHz). The following figure presents the frequency response for the higher frequency set-up, but the shape of the curve remains the same for lower settings.



Appendix C: IEC 61672-Class 1 standard specifications

1) Temperature and Humidity Range

All specifications mentioned in this appendix (except where specified) are for the following conditions:

- *Temperature:* 23 C
- *Humidity: 58%*
- *Atmospheric pressure: 101.325 kPa*

Note: For all specifications, we consider that the Alto SLM hardware has been powered-up for at least 1 minute.

If the Alto SLM is subject to important ambient condition changes, we suggest waiting for at least 2 minutes (hardware powered) before starting a new measurement.

The Alto SLM complies with class 1 tolerances of the IEC-61672 standard for the following temperature and humidity range:

- *Between +5C and +35C*
- *Between 25% and 90% of humidity (under the dew limit)*

We suggest reading the specifications of the PC used with the Alto to determine the acceptable ambient condition range for the PC.

2) General description of the SLM instrument

The Alto SLM instrument includes:

- An external metal box with a digital signal processor (DSP), BNC connectors and a ICP/DeltaTron conditioning board
- A USB 2.0 cable
- A switching-mode 24 Volts DC power supply pack (500 mA)
- A PC (not included) with a dedicated software

Note: The maximum voltage that can be applied on the BNC connectors without any permanent hardware damage is ±20 Volts. However, the maximum voltage for linear measurements is ±3 Volts.

The signal processing performed by the Alto SLM is:

By the DSP (Digital Signal Processor) of the Alto unit:

- The Alto hardware accepts an ICP/DeltaTron compatible microphone (see following paragraph for the recommended ICP/DeltaTron microphone and preamplifier). This input has a DC filter with a frequency cut-off of 0.5 Hz
- The analog signal at the output of the ICP/DeltaTron microphone is digitalized (with a sample rate of 43 kHz) by a 16-bits A/D converter that includes an anti-aliasing filter
- The digital signal processor applies a frequency weighting filter A, C or Z on the digitalized signal
- The square value of the filtered input signal is computed
- An exponential time average is computed for square values with a time constant of 125ms (mode fast) or 1s (mode slow)
- An linear average is computed on square values
- The average global levels are transferred to the PC

By the PC:

- The average global levels are read in the memory of the Alto unit
- The PC formats the average data and presents the results to the user with the dedicated Alto software

Note: The PC used with the Alto must have a minimum memory (RAM) of 512 MB.

3) Microphone suggested in accordance with the IEC 61672 Class 1 Standard

The SLM module of the Alto is a class 1 instrument only if it is used with a class 1 microphone (according to the IEC 651 standard). The table below shows a few suggestions. The microphone sets must be used with the ICP input of the Alto SLM. The resulting microphone set can be used with a 10-meter (BNC-BNC) cable.

Microphone set	**Microphone**	**Preamplifier**
BSWA 50mV/Pa	MP201	MPA231 or MA221
BSWA 40mV/Pa	MP231	MPA231 or MA221
BK 50mV/Pa	#4189	#2671

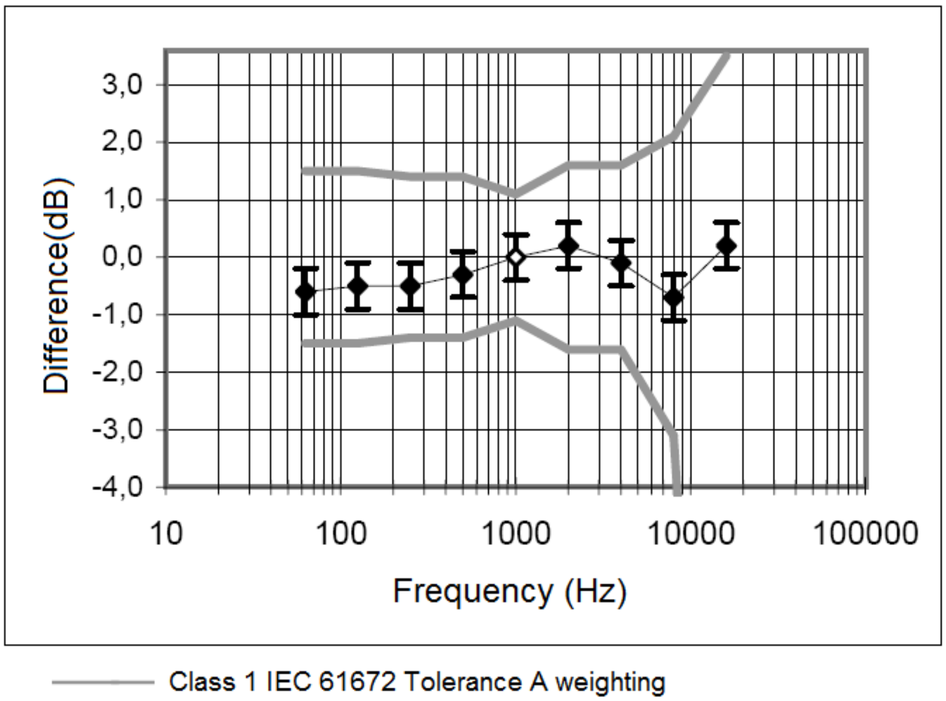
4) Input calibration

The Alto SLM has a dedicated module for the calibration of the microphone (see section 4.1.3). For an acoustic calibration, a sound level calibrator with a calibration frequency of 1 kHz and a calibration level of 94 dB are needed. The calibrator has to conform to the class 1 specifications of the International Standard IEC 60942, Electro acoustics – Sound Calibrators.

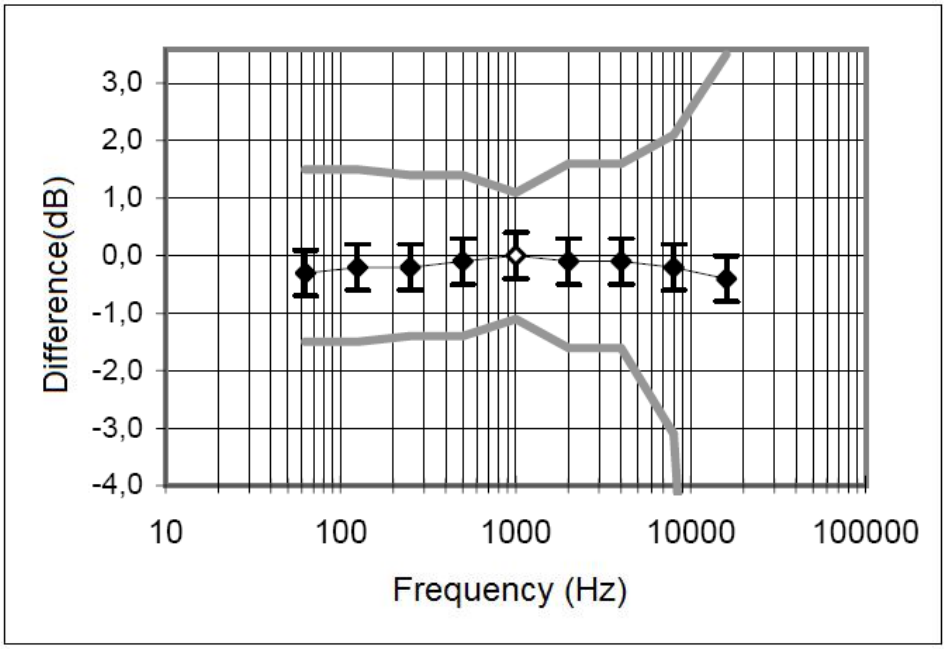
5) A, C and Z Frequency Weightings

The following graphs show the A, C and Z frequency weighting test results done for the IEC 61672-3 Alto SLM certification:

A-Weighting (Reference Input Range)

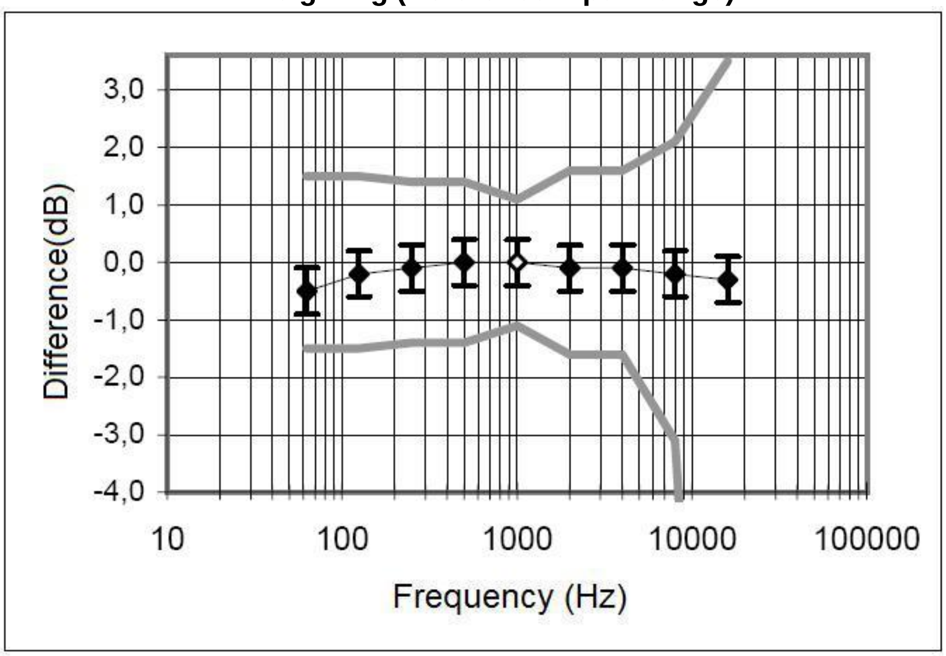


C-Weighting (Reference Input Range)



Class 1 IEC 61672 Tolerance C weighting

Z-Weighting (Reference Input Range)



Class 1 IEC 61672 Tolerance Z weighting

6) Linear Operating Input Ranges

The following tables present the linear operating input ranges for all input range selections, for all frequency weightings and for 31.5 Hz, 1 kHz, 4 kHz, 8 kHz and 12.5 kHz sine signals[3].

	Gain 1 (least sensitive range)				
Frequency-Weighting	31.5 Hz	1 kHz	4 kHz	8 kHz	12.5 kHz
A-weighting	52.5.0-88.0	52.5-126.2	52.5-127.0	52.5-124.7	52.5-121.9
C-weighting	53.0-121.1	53.0-126.2	53.0-125.4	53.0-123.2	53.0-120.0
Z-weighting	56.5-125.3	56.5-126.1	56.5-126.3	56.5-126.3	56.5-126.3
	Gain 2 (reference range)				
Frequency-Weighting	31.5 Hz	1 kHz	4 kHz	8 kHz	12.5 kHz
A-weighting	45.0-76.0	45.0-114.2	45.0-114.8	45.0-112.5	45.0-109.8
C-weighting	44.0-109.1	44.0-114.2	44.0-113.3	44.0-111.1	44.0-108.0
Z-weighting	47.5-113.2	47.5-114.1	47.5-114.2	47.5-114.2	47.5-114.1
	Gain 3 (most sensitive range)				
Frequency-Weighting	31.5 Hz	1 kHz	4 kHz	8 kHz	12.5 kHz
A-weighting	35.5-63.5	35.5-101.7	35.5-102.6	35.5-100.1	35.5-97.3
C-weighting	35.0-96.6	35.0-101.7	35.0-100.9	35.0-98.6	35.0-95.5
Z-weighting	37.0-100.8	37.0-101.7	37.0-101.7	37.0-101.8	37.0-101.7

Table 1: Linear operating range (50mV/Pa sensitivity)

Note that the minimum values for all linear operating input ranges are not the inherent noise levels of the input (see following paragraph for inherent noise values). Actually, the lower limit of the linear operating input range is measured for a sine signal, which emphasizes the distortion problem for a low level digitalized signal. Also, according to the IEC 61672 standard, the lower limit is determined when the global level measured is 1 dB over or under the real global level at the BNC connector.

Note: the low level indicator of the SLM module appears when the global level is equal to, or below, the lower limit of the selected range for the 1 kHz case.

[3] The levels stand for a microphone with a typical sensitivity of 50 mV/Pa. If a different sensitivity is used, the levels are offset accordingly. For example, a sensitivity of 40 mV/Pa means an offset of +2 dB on the table (20*log[50mV/Pa / 40mV/Pa]=+1.94dB)

7) Inherent Noise

The following table presents the microphone and electrical noises for all input ranges and frequency weightings[4].

	Gain 1 (least sensitive range)	
	Maximum Noise	
Frequency-Weighting	Microphone[5]	Electronic[6]
A-weighting	45.1	45.1
C-weighting	45.0	45.0
Z-weighting	49.0	49.0
	Gain 2 (reference range)	
	Maximum Noise	
Frequency-Weighting	Microphone	Electronic
A-weighting	30.5	30.5
C-weighting	30.0	30.0
Z-weighting	33.5	33.5
	Gain 3 (most sensitive range)	
	Maximum Noise	
Frequency-Weighting	Microphone	Electronic
A-weighting	28.0	28.0
C-weighting	28.0	28.0
Z-weighting	32.0	32.0

Table 2: Inherent Noise (50mV/Pa sensitivity)

8) Global Level Indicators Update Delay

For the fast or slow SPL global levels, the exponential average requires a delay before obtaining a representative result. For the slow mode, the exponential averaging process is stable after 2 seconds. For the fast mode, the exponential averaging process is stable after 0.25 second. Following the instrument set-up, the Alto SLM automatically waits for these delays at the beginning of a measurement.

After this first delay, the global level indicators are updated every 1/10-second on the PC interface.

9) Frequency corrections

Since the microphone is remote, there are no additional corrections besides the ones pertaining to the microphone, which are published by the manufacturers.

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[4] The levels stand for a microphone a typical sensitivity of 50 mV/Pa. If a different sensitivity is used, the levels are offset accordingly. For example, a sensitivity of 40 mV/Pa means an offset of +2 dB on the table (20*log[50mV/Pa / 40mV/Pa]=+1.94dB)

[5] The microphone is placed in a calibrator (turned off) and in an anechoic chamber. The BK 4189 microphone and the BK 2671 preamplifier are used for this test.

[6] Electronic noise of the instrument, measured with a 50 ohms resistance instead of the microphone.

10) Glossary

A Frequency Weighting Filter:

Frequency weighting representing the human ear response at low to medium sound levels; It is by far the most commonly applied frequency weighting and is used for all levels of sound.

Anti-Aliasing Filter:

This is a filter used before the digitalization of an analog signal to restrict the bandwidth of a signal to approximately satisfy the sampling theorem. Since the theorem states that unambiguous interpretation of the signal from its samples is possible only when the power of frequencies outside the Nyquist bandwidth is zero, the anti-aliasing filter would have to have an adequate stop-band rejection to satisfy the theorem. For the SLM case, the anti-aliasing filter is adjusted to avoid aliasing under 20 kHz.

C Frequency Weighting Filter:

Frequency weighting representing the human ear response at fairly high sound levels; It is mainly used when assessing peak values of high sound pressure levels.

Decibel (dB):

This is the measurement unit for expressing the relative intensity of sound. A direct application of linear scales (in Pa) to the measurement of sound pressure leads to large and unwieldy numbers. Since the ear responds logarithmically, rather than linearly, to stimuli, it is more practical to express acoustic parameters as a logarithmic ratio of the measured value to a reference value. This logarithmic ratio is called a decibel or dB. The linear scale with its large numbers is converted into a manageable scale from 0 dB at the hearing threshold (20 µPa) to 130 dB at the pain threshold (100 Pa).

DC Filter:

A DC filter is a high-pass filter with a very low frequency cut-off. The SLM module uses a frequency cut-off of 0.5 Hz for ICP and Direct AC input. This filter removes the DC component of the signal not associated with a real acoustic or vibratory signal. The removed DC signal comes from the preamplifier voltage supply or an offset in the acquisition chain.

Fast and Slow Time Weighting:

Time weighting defines how the exponential averaging is performed. It defines how the heavily fluctuating sound pressure variations are smoothed or averaged to allow useful readings. The standards define two time weightings: F (Fast) and S (Slow). Most measurements are carried out using the Fast time weighting, which uses a 125ms time constant.

Z-weighting:

The Z (Zero) frequency weighting is without any weighting, that is, equivalent to a flat weighting.