POST ANALYZER
User's Manual

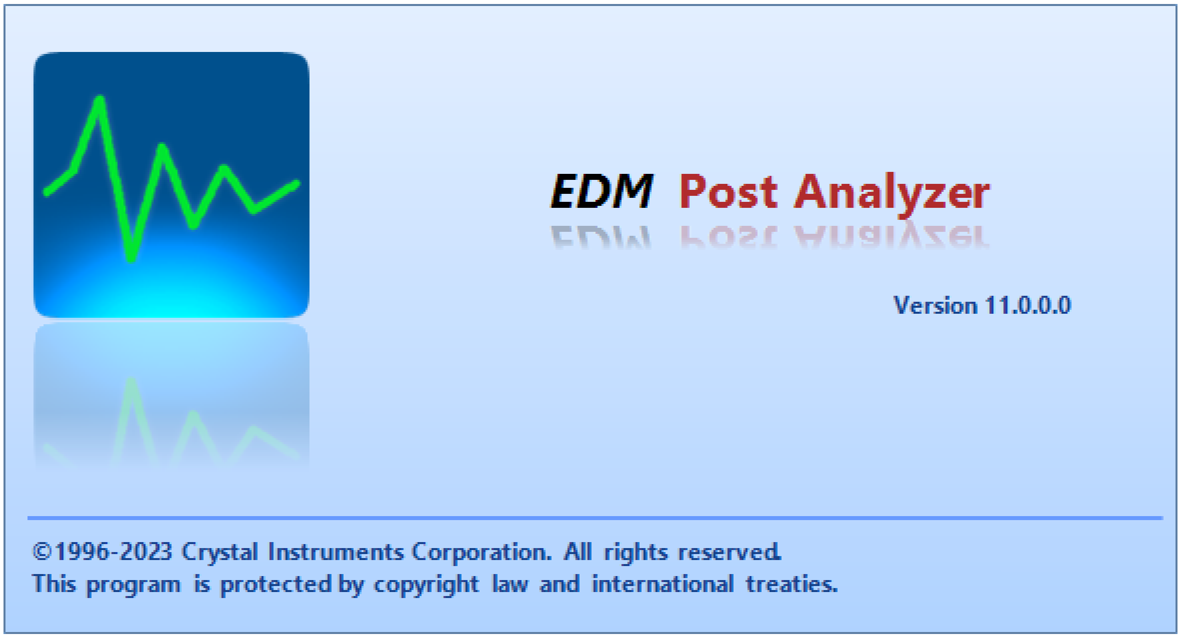
Based on Software Version 11.0

Table of Contents

Introduction

The **EDM Post Analyzer** application is used to post process previously recorded time stream data. Post processing includes data conditioning, Fourier transform operations, and specialized analyses such as order tracking, octave analysis, and more.

Crystal Instruments developed *EDM (Engineering Data Management) Post Analyzer, Waveform Editor*, and *File Converter* to provide a complete package for real-time analysis and post processing. **Post Analyzer (PA)** contains many powerful post processing tools with batch processing capabilities. This document describes **EDM PA** functionalities.



PA is an independent Windows application that analyzes data files on a computer using various algorithms. It has three versions: PA Viewer allows the user to view data and create reports; PA Basic has FFT spectral analysis and 3D signal display functions; PA Premium has more advanced functions such as octave filters and order tracking.

Batch processing is a powerful tool that allows PA software to automatically process compatible data files in batches. As long as the recorded time signal files are of the same settings, which is often the case, the recorded signal files can be fed into the same project and processed consecutively without manual operation.

The following table describes the functions in each of the three versions:

Functions	PA Viewer	PA Basic	PA Premium
Browse and display long waveform files	√	√	√
Signal display with different spectrum unit and X-Y scale	√	√	√
Signal annotation, cursor, play sound, calculate	√	√	√

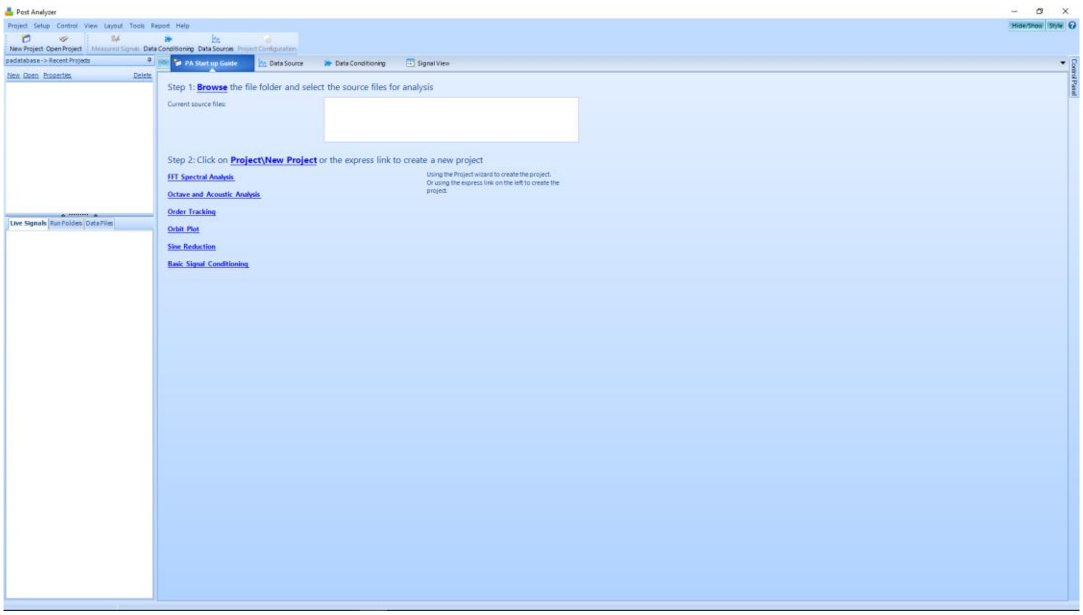
RMS, THD, ZOOM-in, ZOOM-out, auto scaling			
Create template-based report	√	√	√
Engineering unit conversion, dB reference	√	√	√
Export to standard formats including ASAM-ODS, UFF, BUFF, MATLAB, user-defined ASCII, and wave files	√	√	√
Import user-defined ASCII file, wave file, Pacific Instrument file		√	√
Acceleration, velocity, and displacement conversion		√	√
FFT Spectral analysis: FFT, auto power spectra, cross power spectra, frequency response function		√	√
3D display: waterfall, color map		√	√
Data file batch processing		√	√
Math Functions: abs, +, -, *, /, square, square-root, log, integration, differentiation, RMS, peak, offset and scale,			√
Digital Filters: IIR, FIR, Low-pass, High-pass, Band-pass			√
User defined data conditioning modules			√
Fractional octave filters: 1/1, 1/3, 1/6, 1/12			√
Order Tracking: RPM spectra, order spectra			√
Offline Sine Data Reduction			√

Getting Started

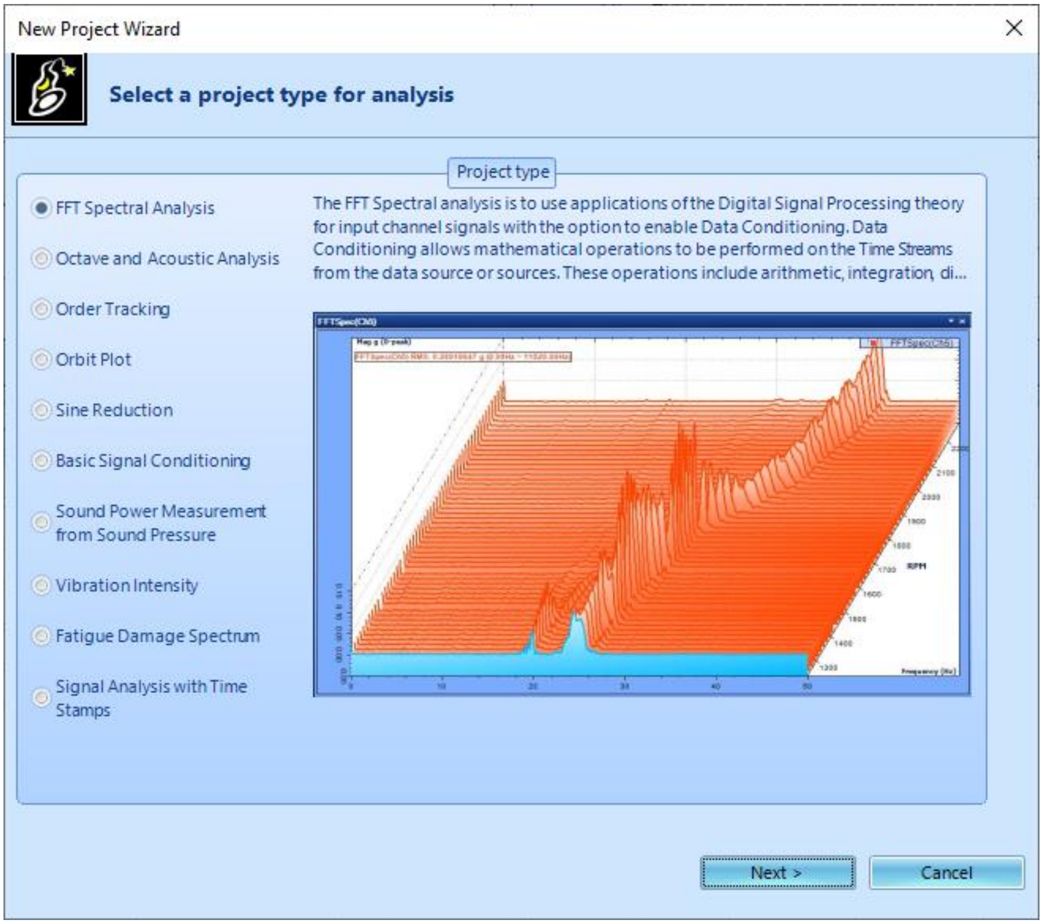
At a glance

Every analysis operation is performed in the context of a **Project**, which stores analysis settings and is associated with one or more source data files. An unlimited number of projects can be created, but only one can be active at a time.

The first time Post Analyzer is launched opens the following window containing a guide to create a new project.



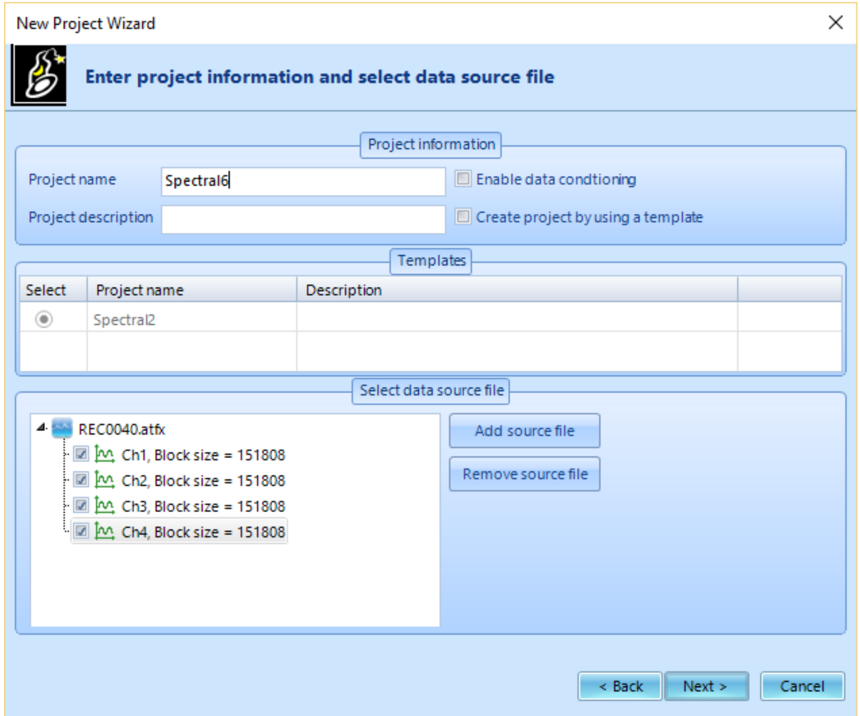
Create a new project by selecting **New Project** from the Project menu or click the **New Project icon**. This will open the **New Project Wizard** as shown in the following screenshot.



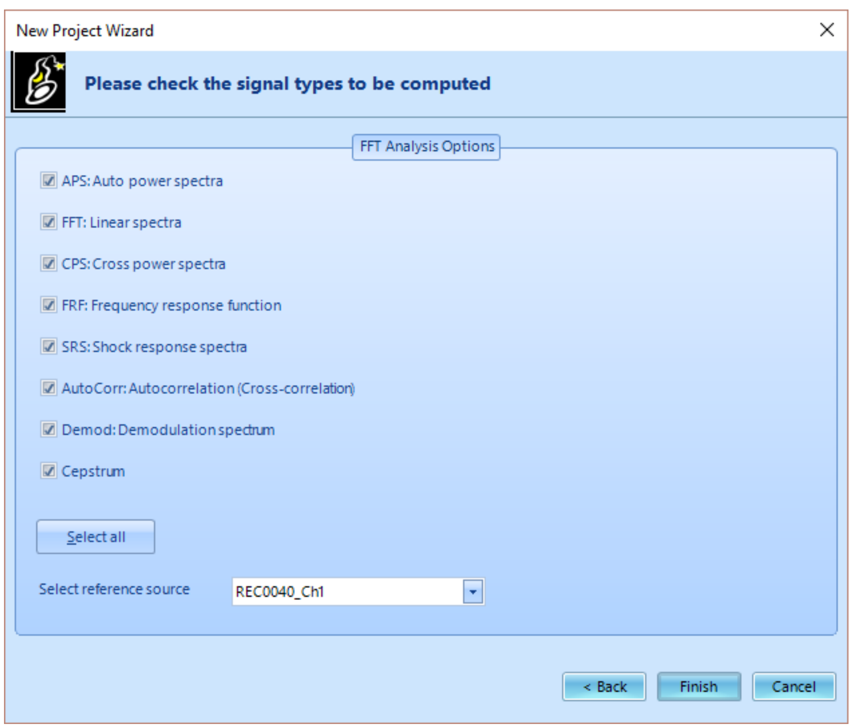
Select an analysis type to perform in the **New Project Wizard** window:

- FFT Spectral Analysis
- Octave and Acoustic Analysis
- Order Tracking
- Orbit Plot
- Sine Reduction
- Basic Signal Conditioning
- Sound Power from Sound Pressure
- Vibration Intensity
- Fatigue Damage Spectrum
- Signal Analysis with Time Stamps

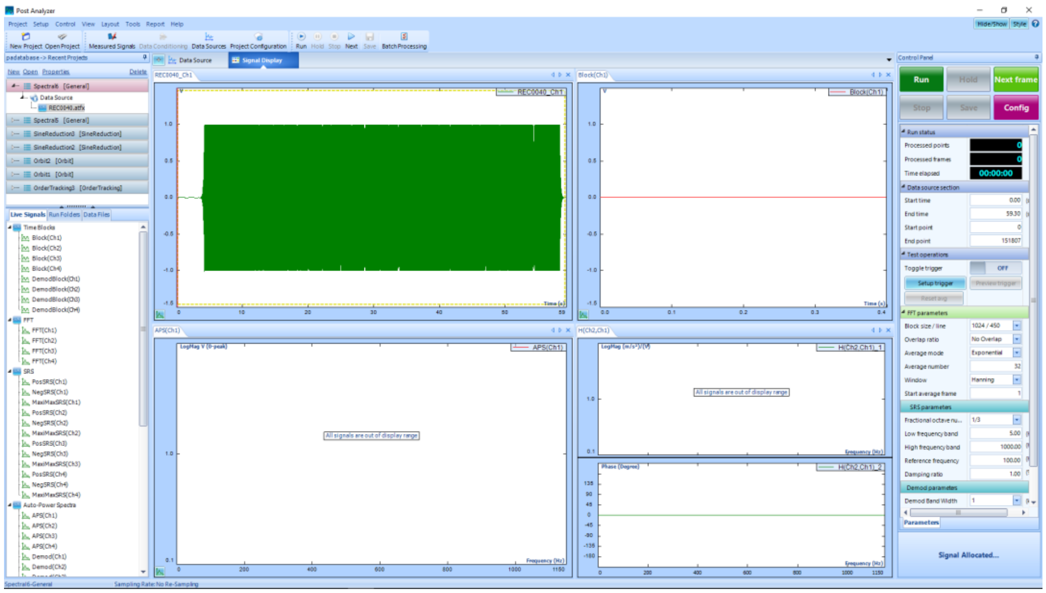
Click **Next** to open the following window.



Enter the project name and select a recorded file by clicking the **Add source file** button. Then click the **Next** button.

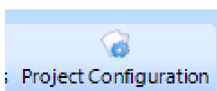


Select the signal types to be computed and click **Finish** to display the new project window as shown in the following screenshot.

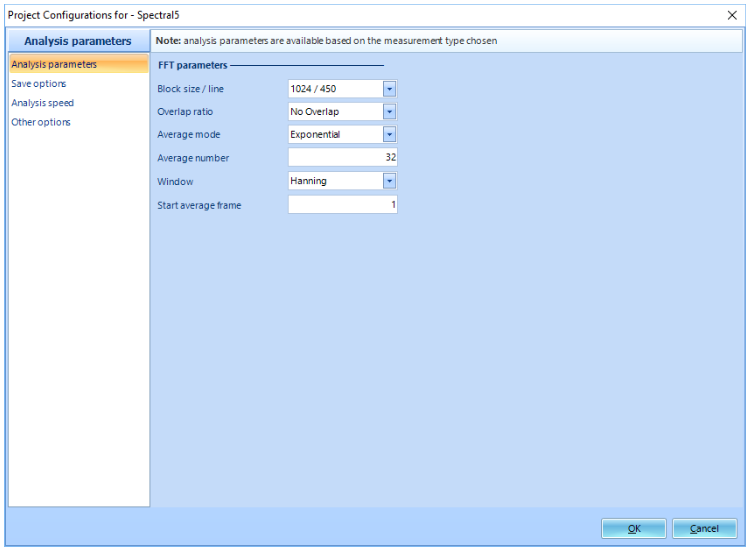


Project Configuration

Click on the **Config** icon or button to access project configuration parameters.

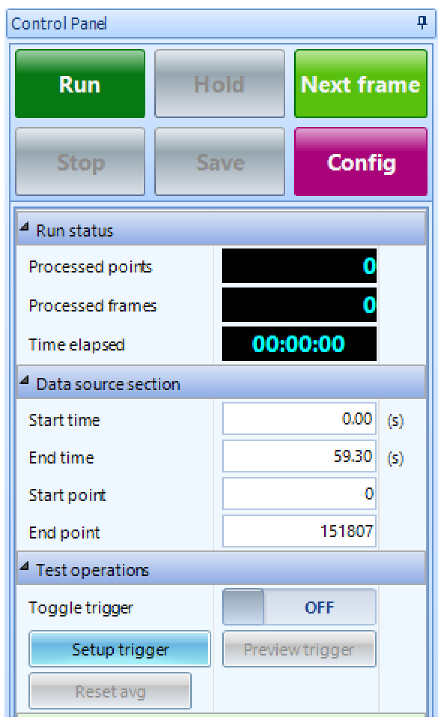


The **Project Configuration** section contains a multi-tab dialog box with options to set up the analysis parameters, save options, analysis speed, and more. Users can edit some of these parameters directly on the Control Panel during a test.



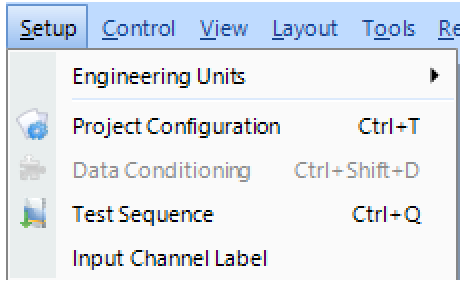
Control Panel

Users can control a project with buttons on the **Control Panel** and the **Control Toolbar**.

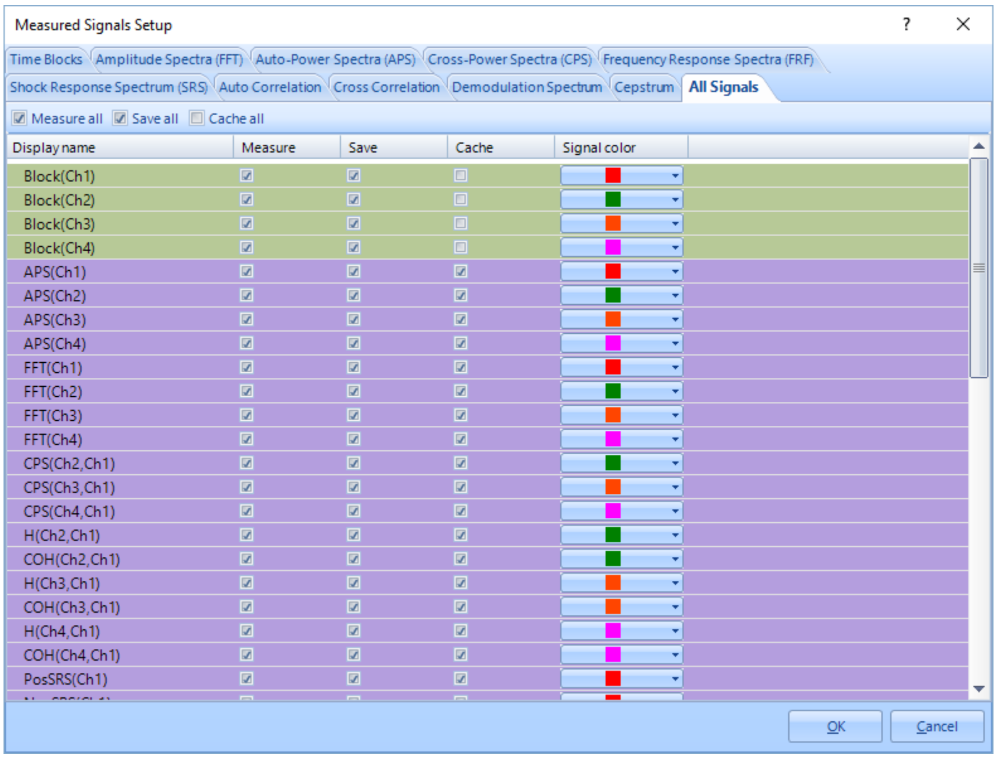


Measured Signals

Click on **Setup->Measured Signals** to select signals to measure, save, or cache.



Signals are organized according to type under this tab. Select any signal to measure, save, or cache.



Signals selected in the **Save** column are saved when the **Save button** is clicked in the Control Panel or when a save condition is met.

View Live Signals

Select a live signal to view from the list of available live signals on the left side of the window. Right-click on the signal and select **Display in New Window**.

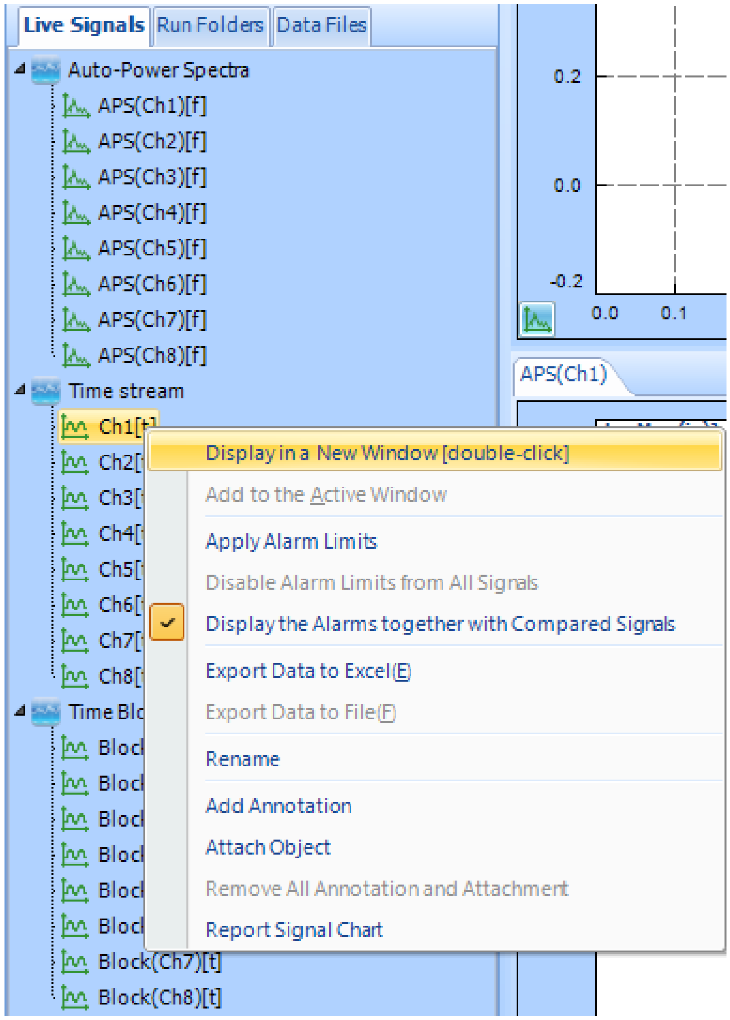
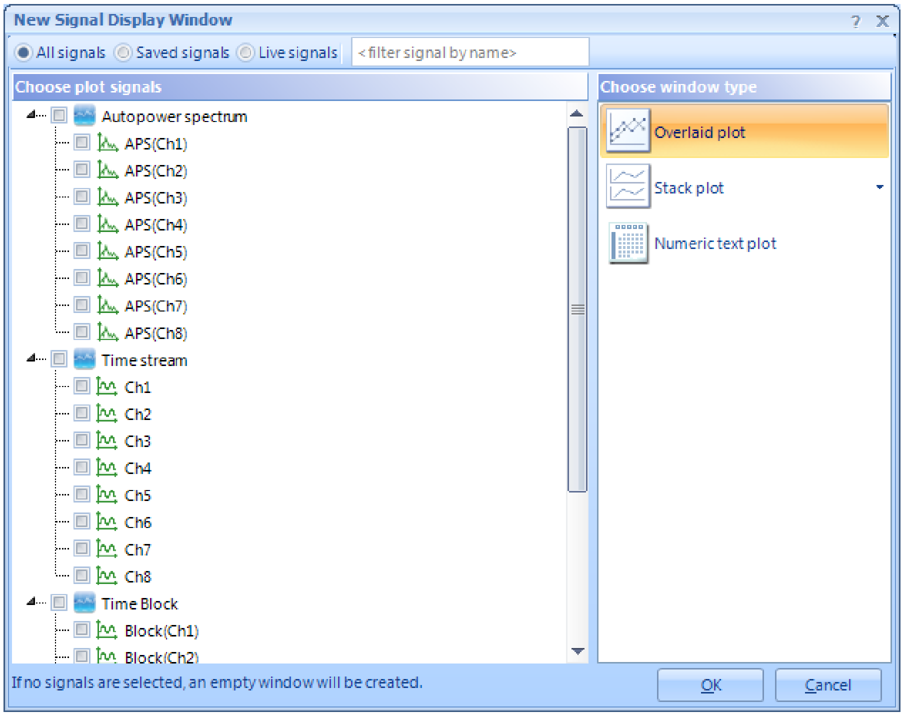


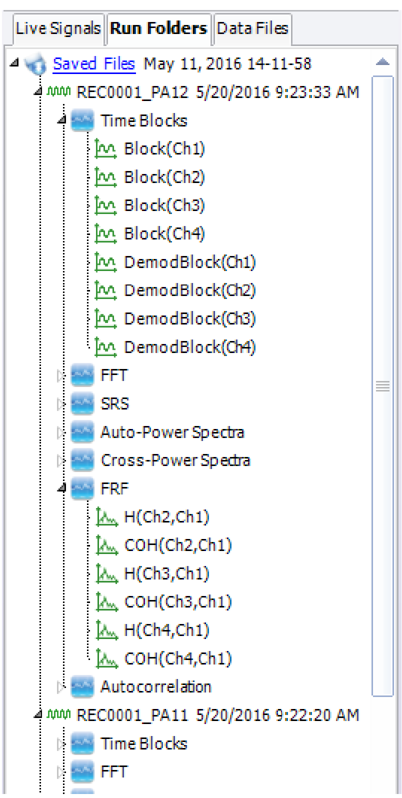
Figure 1: Signal display options

Users can also navigate to the **View** menu and select **New Window** to open the **Window Customizer** dialog box. Select signals to display and display window types.

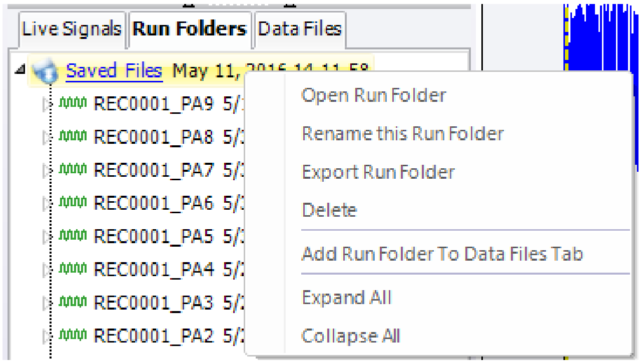


Run Folders

A **Run Folder** is created on the disk by default each time the Run button is clicked on the Control Panel. Data files and a runlog are saved in the Run Folder.

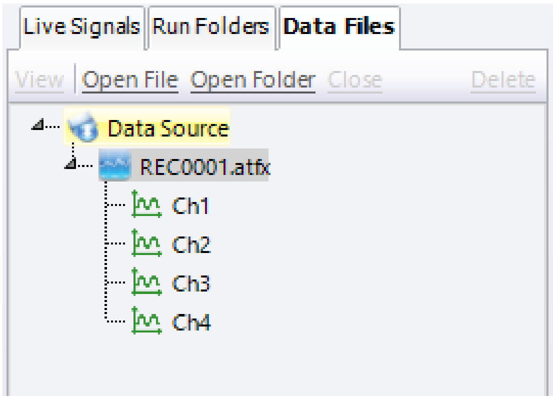


Right-clicking on a **Saved Files** entry opens a list of available entries for the saved file folder. Entries under **Run Folders** can be opened, renamed, exported, or deleted. When a Run Folder is exported, all of the data files in the Run Folder are copied to the target folder. The list of saved signals under Saved Files can be expanded or collapsed.

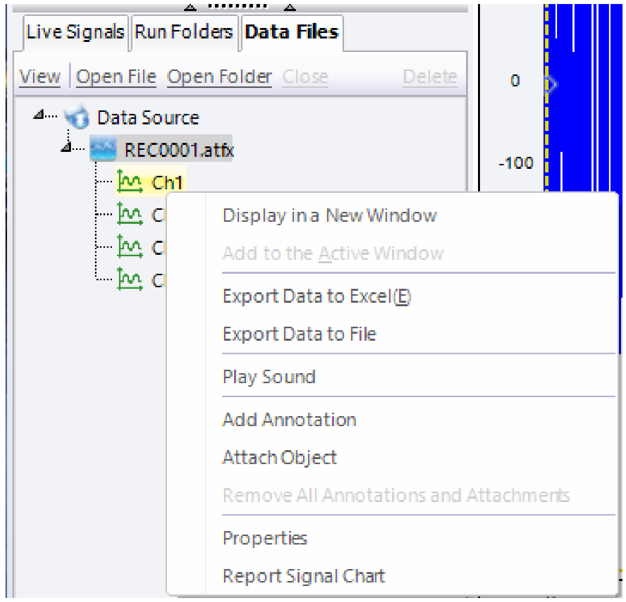


View Data File

Click on the **Data Files** tab to view the data source file opened. The available options are *View*, *Open File*, and *Open Folder*.

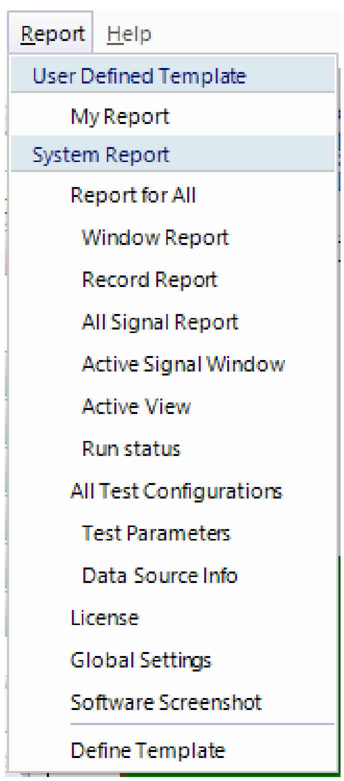


Right-click and highlight a channel of data to export into a new display window, Excel file, or data file. **Annotations and Objects** can be added to a selected channel of data. The properties window and report generation for the selected channel of data are also available.



Create a Report

Click on the **Report** menu. The information listed below is available to generate in a Word document report. Users can access previously defined templates using the **Define Template** feature or can create a new template.



FFT Spectral Analysis

Basics

FFT Analysis is included as a Dynamic Signal Analyzer operation. This operation focuses on using applications of Digital Signal Processing theory such as Auto Power Spectrum, Cross Power Spectrum, and Fourier Transform calculations for input channel signals.

This section provides an overview of the theory related to functions performed by EDM Post Analyzer. To read in further detail about this topic, refer to the "Dynamic Signal Analyzer Basics" document published by Crystal Instruments along with this manual.

Fourier Transform is the most fundamental and popular method of signal analysis. Frequency domain facilitates the signal analysis due to advantages like phase information and Transfer Function calculations. Fourier Transform involves an infinite sum over infinite time, which requires the signal to be broken into finite blocks of N samples. Each block is transformed with the Discrete Fourier Transform (DFT). The DFT sum is computationally intensive, so a more efficient algorithm referred to as Fast Fourier Transform (FFT) was developed.

Applications of FFT are listed below.

Power Spectrum

The magnitudes of the frequency components of signals are collectively called amplitude spectrums. In many applications, the quantity of interest is power or rate of energy transfer that is proportional to the squared magnitude of frequency components. The squared magnitudes of all the DFT frequency lines are collectively referred to as the **Power Spectrum**.

$$S_{xx}(k) = \frac{1}{2} S_x^*(k) \cdot S_x(k)$$

Cross Spectrum

Cross Spectrum characterizes the relationship between the spectra of two signals. For two signals x and y, with frequency components Sx and Sy, it is defined as:

$$S_{xy}(k) = S_x^*(k) \cdot S_y(k)$$

It is the correlation between the frequency components of two related signals. While the Power Spectrum is real-valued, the Cross Spectrum is complex. This means that it also describes the phase alignment of the two signals.

Frequency Response Function

An important application of Dynamic Signal Analysis is characterizing the input-output behavior of physical systems. This is the domain of network analysis. With linear systems, the output can be predicted from a known input if the Frequency Response Function of the system is known. The Frequency Response Function (f) relates the Fourier Transform of the output Y(f) to the Fourier Transform of the input Y(f) by the simple equation:

$$Y(f)= H(f)X(f)$$

The system is linear because H(f) is independent of the input. When measuring the input-output behavior of a system, there is always noise present that obscures the output. An important measure is how much of the output is actually caused by the input. Another correlation measure is called Coherence and it is defined as:

$$C_{xy} = \frac{|S_{xy}|^2}{S_{xx}S_{yy}}$$

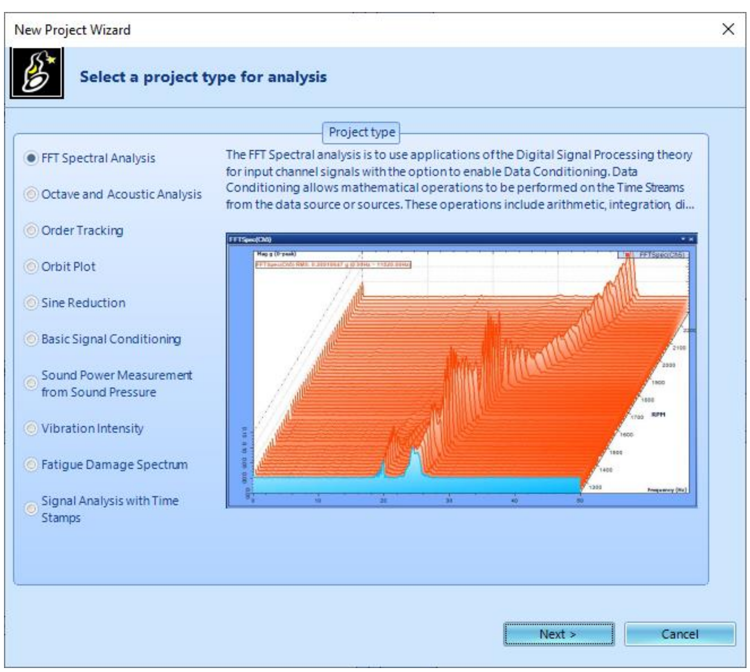
The coherence is between 0 and 1, with 1 meaning the output is perfectly explained by the input. A coherence of 0 means the output and input are uncorrelated.

Shock Response Spectrum

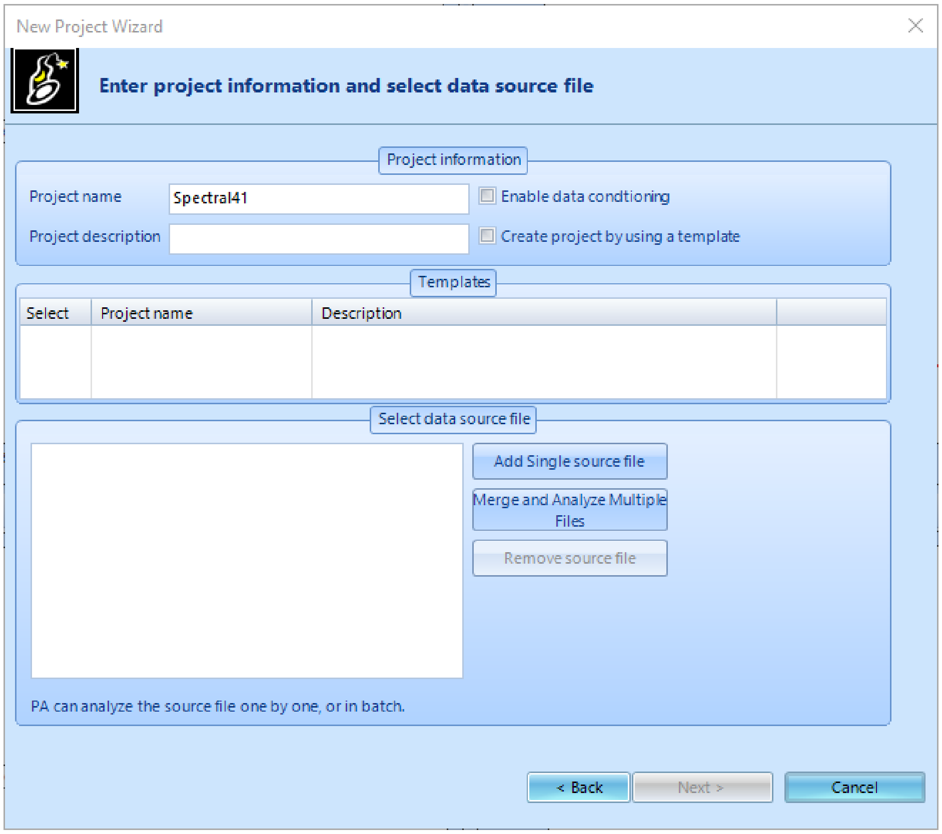
The **Shock Response Spectrum (SRS)** is used to characterize transient and shock waveforms in terms of their effect on single degree-of-freedom (DOF) mechanical systems. The SRS calculated from a time waveform can predict the effect of that waveform on more complex, multi-degree-of-freedom structures.

Create an FFT project

Open the **New Project Wizard** to create a new project.



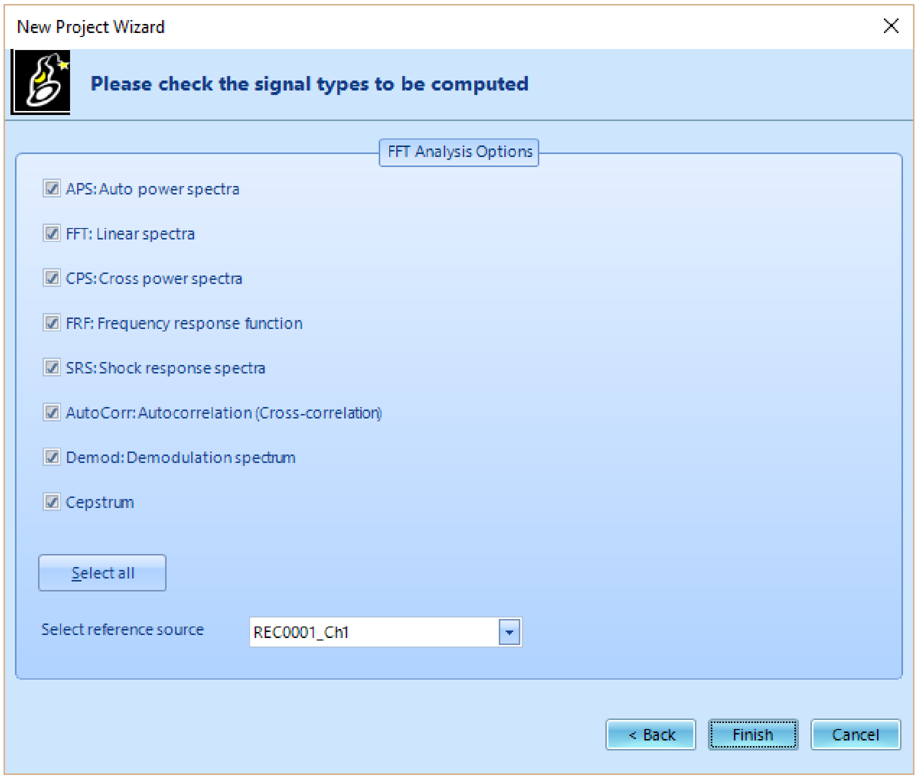
Select FFT Spectral Analysis and click **Next.**



Assign a project name and add a project description if desired in the Project Information section. Select the **Enable data conditioning** box to enable data conditioning (refer to the *Data Conditioning* section below).

In the same window, select the data source files to use with this project. A single source file is commonly used but multiple source files can be combined as well. The source file is usually a time stream recording in ATFX format. Select and highlight source files to remove from the data source file list, then click **Remove Source File**.

Click the **Next** button to accept all selections, and the following FFT analysis options window will open.



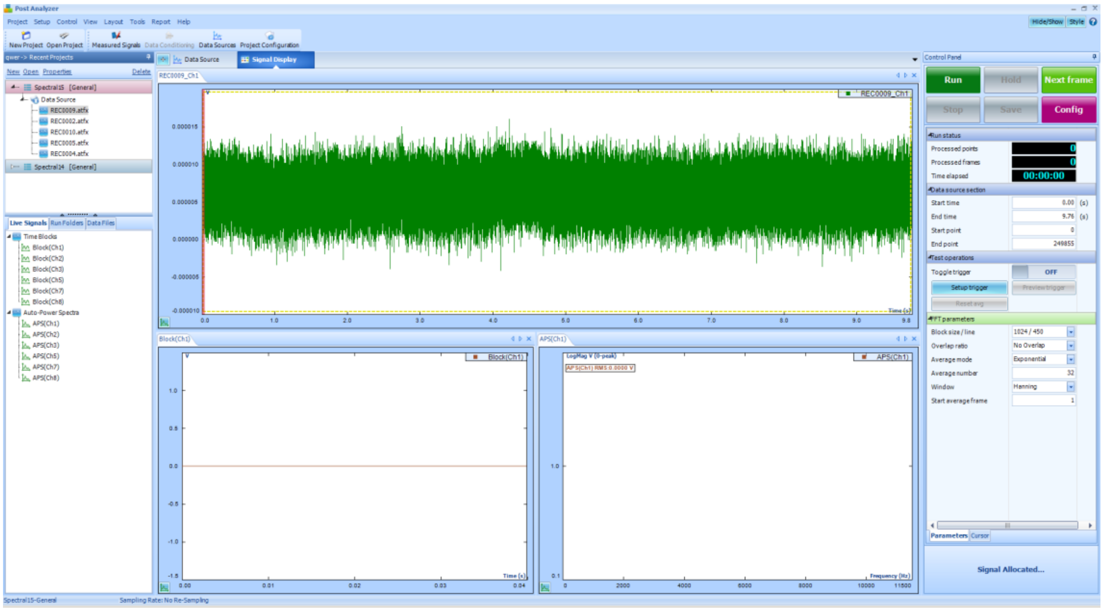
The final window in the Wizard provides options to specify the type of analysis spectrum being analyzed. For FFT projects, check the type of FFT or any spectrum data to be enabled. If only one channel is selected for post-processing, cross-channel spectrum signals will be grayed out.

The reference channel is required when a CPS or FRF option is selected. This allows the user to choose a channel as a reference for calculating the frequency response function, coherence, and cross power spectrum. Users can specify the selected channel as the response channel or the excitation channel. Signals such as CPS, Coherence and FRF are defined based on this selection.

Click the **Finish** button to close the New Project Wizard window and open the Post Analyzer window.

Overview

After creating or opening a project, the main Post Analyzer window is displayed. There is a toolbar across the top, as well as three main sections in the middle. The left side has the Recent Projects, Live Signals, Run Folder, and Source Files views; in the middle are one or more Signal Display views; and on the right is the Control Panel which also contains the analysis parameters.



Recent Projects

The **Recent Projects** list shows the currently active project and previously opened projects in the application. The current project will have a shortcut toolbar containing the Measured Signals setup, Data Conditioning setup (if data conditioning is checked with creating this project), Data Sources setup, and Project Configuration window. It also displays the data file associated with the project.

Live Signals, Run Folder, and Data Files

The **Live Signals, Run Folders, and Source Files** list is below the Recent Projects list. Live Signals includes the signals in the Data Source file or files, as well as new signals created as part of the analysis. The live signals are not saved to the host computer hard disk but are available for display in a Signal Display view tab. Anything saved to the hard disk is displayed in the Saved Files list, which is associated with a directory in the file system. Run Folders contains data for each run. Data is saved in a specified file format under the Source Files tab.

Signal Display

The middle of the main window contains the **Signal Display** view tabs, where live and saved data is displayed. More than one of these tabs can be created, but there always is at least one. Each of these views can contain one or more Display Windows, with a fully customizable layout. These display windows can move freely, resize, and display any valid combination of live or saved signals. New view windows are created by selecting an option in the View menu.

Control Panel and Analysis Parameters

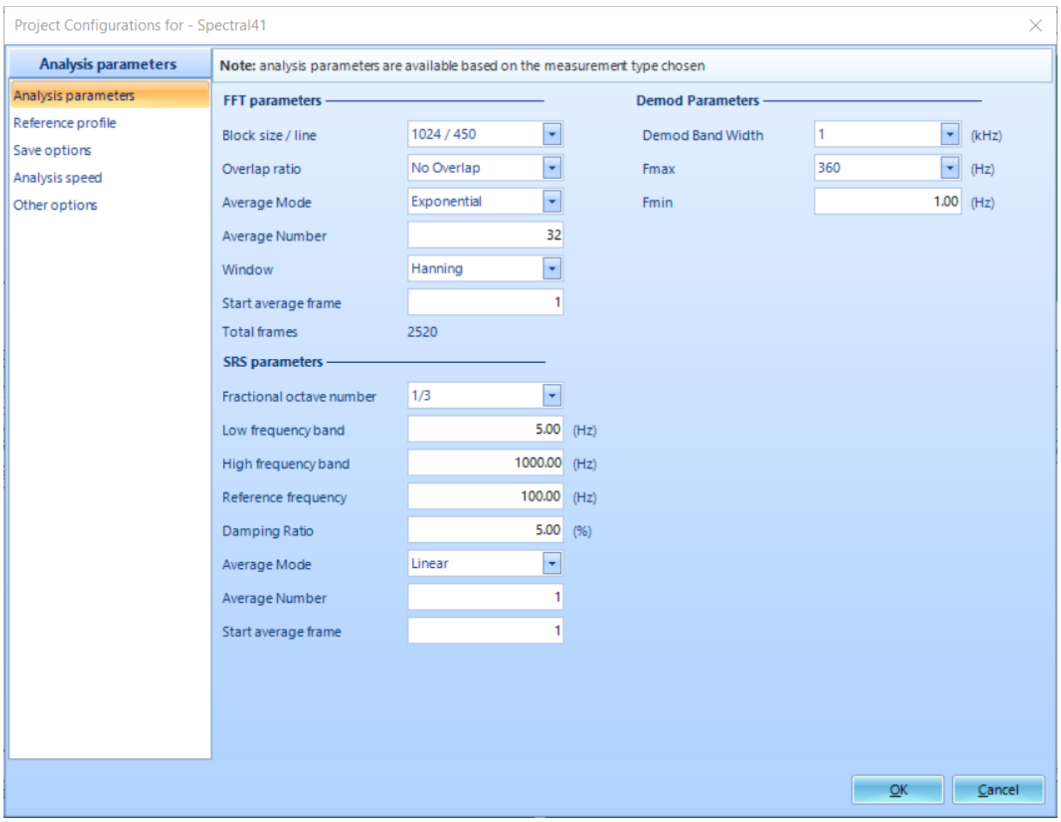
The right side of the window contains the **Control Panel** (which controls the post-analysis operation) and the **Analysis Parameters** (which contains setup parameters specific to the performed analysis type). The **Config** button on the control panel opens the Project Configuration window, which is described in the following section.

Project Configuration

The **Project Configuration** window has settings related to saving data, analysis speed, and other options that are mostly independent of the type of analysis being performed.

Analysis Parameters

Analysis Parameters consist of FFT, SRS and Demodulation parameters. The availability of SRS and Demodulation parameters depends on the selected FFT Analysis Options when the project is created.



FFT Parameters

Block Size/Lines defines the size of the time blocks in terms of the number of samples, and the number of spectral lines used in the frequency domain of the Fourier Transform of the signal. The duration of a time block can be calculated as follows:

$$T = \frac{Block\ Size}{Sampling\ Frequency}$$

Increasing the block size increases the resolution of the frequency transform and allows detection of lower frequencies. In turn, the increased block size also increases the calculation time and slows down the response.

The ratio between lines and block size is determined by the characteristics of the A/D converter and anti-aliasing filter. In general, the ratio is about 0.439 which means that a time block of 1,024 samples results in about 0.439 * 1,024 = 450 lines in the spectrum.

Overlap Ratio sets the proportion of the samples in the time blocks that are overlapped when calculating the FFT. Higher overlap ratios result in faster response time but increase processing requirements. The overlap ratio options are: no overlap, 25%, 50%, 75%, 80%, 87.5%, 90%, 95% and "as high as possible".

Average Mode provides options such as: exponential, linear, moving linear, and peak hold as the methods used to average the spectrum signals.

Average Number is the number of blocks that are averaged for the signal spectrum. Increasing the number of averages reduces the variance of the signal spectrum.

Window allows users to select a window to apply during FFT operation. Windowing functions can help reduce leakage and increase the precision of the frequency measurement.

Detailed descriptions about window types and average modes are provided in the DSA Basics document.

Start Average Frame defines which frame the average function starts from.

SRS Parameters

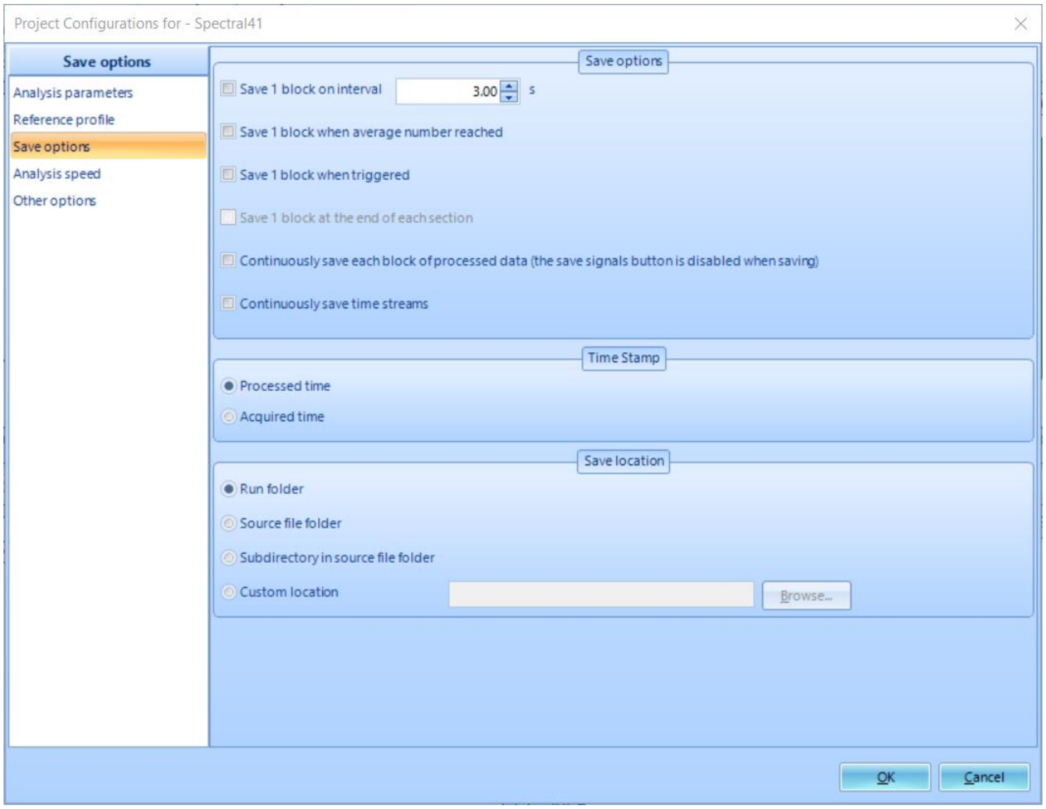
- **Fractional Octave Number** is chosen from 1/1, 1/3, 1/6, 1/12, 1/24, 1/48.
- **Low Frequency** defines the lowest frequency boundary of the SRS spectrum.
- **High Frequency** defines the highest frequency boundary of the SRS spectrum.
- **Reference Frequency** defines the where the reference level is.
- **Damping Ratio (%)** defines the damping ratio as a percentage.

Demod Parameters

- **Demod Band Width** defines the band width of the demod.
- **Fmax** defines the highest demod frequency.
- **Fmin** defines the lowest demod frequency.

Save Options

After running the analysis, computed signals are listed under the **Live Signals** tab. However, this data is only stored in a display buffer and must be explicitly saved to the disk to create new data files. There are multiple ways to do this. One option is to manually click the **Save** button on the control panel, which saves data according to the Save settings under the Measured Signals setup. Users can also enable an option (listed in the section below) to automatically save data while the analysis is running.



There are five options for saving block data and one option for saving stream data. Most data computed in Post Analyzer is block data, such as APS and SRS spectra. Some data, such as the output of digital filters or other data conditioning blocks, are continuous time streams. Only one type of data can be saved at a time – block or time stream.

Save 1 Block on interval saves one block of data for all computed signals with the save option selected in the *Measured Signals* setup per interval defined here.

Save 1 Block when average number reached saves one block of data every time the number of processed blocks reaches the average number set under the Analysis Parameters.

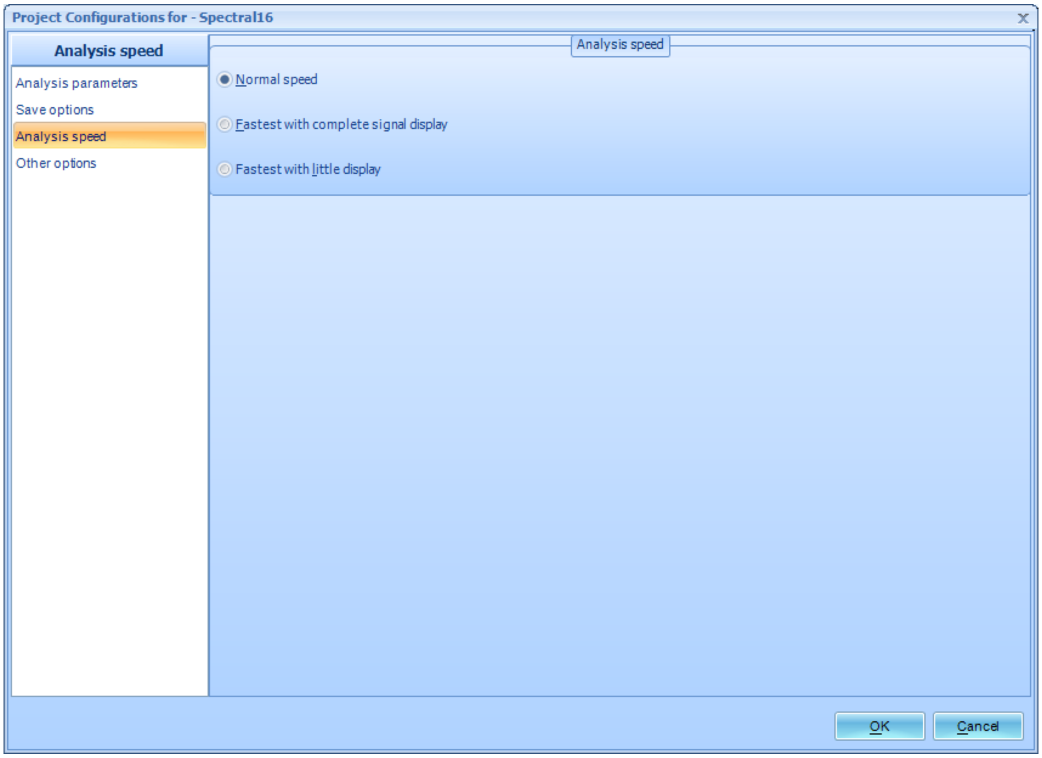
Save 1 Block when triggered saves one block of data when a trigger occurs. The trigger is set up by clicking the *Setup Trigger* button in Analysis Parameters.

Save 1 Block at the end of each section saves the last block of data analyzed.

Continuously save each block of processed data saves every block of data computed.

This window provides three options to choose a location in the file system used to create new data files: in the same directory as the data source file, in a sub-directory of the original directory, or in a custom location.

Analysis Speed



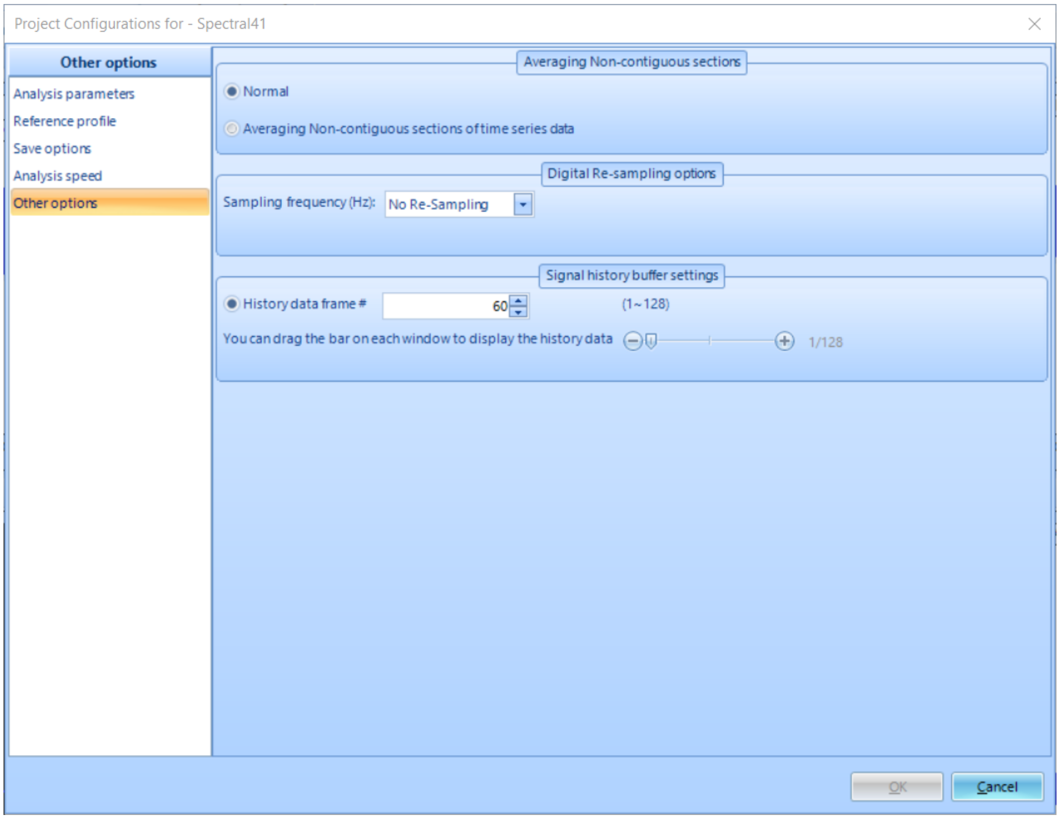
Analysis Speed adjusts the amount of processing time used for the signal display during post-processing. Increased display details result in less available processing resources for post processing data analysis. **Normal Speed** is the default setting that balances data processing and the signal display. **Fastest with Complete Signal Display** prioritizes the display, and **Fastest with Little Display** prioritizes data processing.

Other Options

Averaging Non-Contiguous Sections of Time Series Data allows the data source to be split into two or more non-contiguous blocks. When this option is selected, a Set Block Section button appears under the Data Source tab.

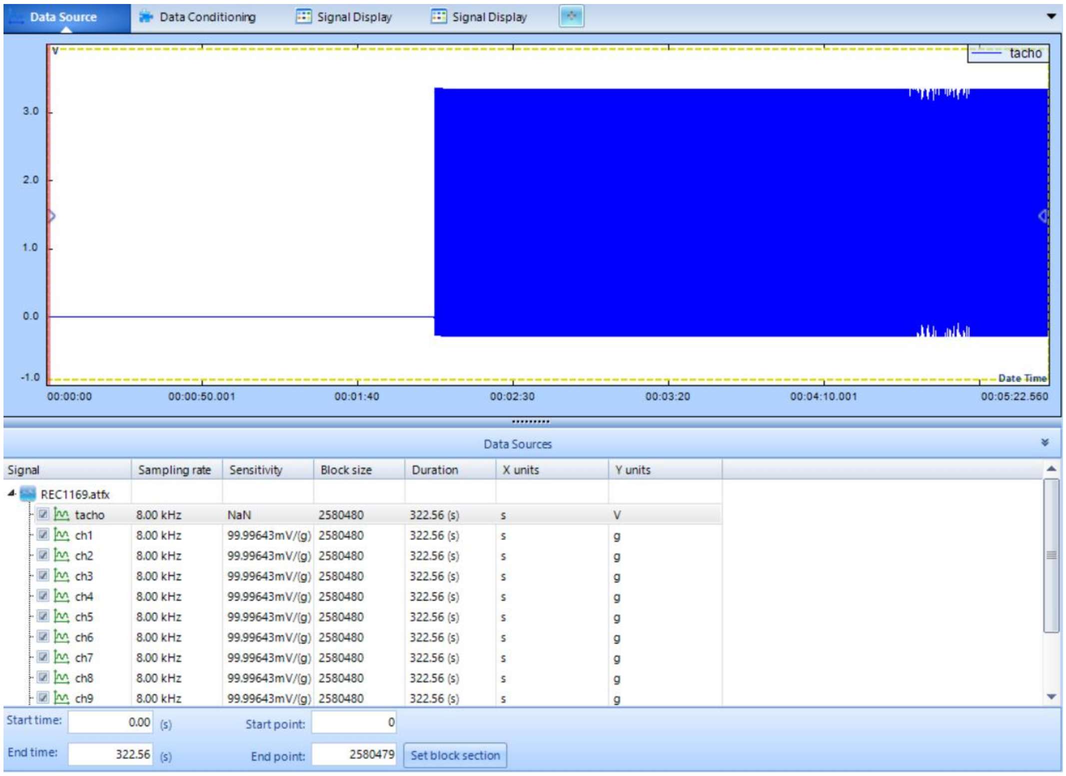
Digital Resampling allows resampling of all source signals to the selected sampling rate to meet the requirement of the other interface. Many sampling rate stages can be selected from the dropdown menu.

Signal History Buffer sets the size of the data buffer used to store and display the signals under the Live Signals tab. This buffer size is set according to a number of frames, a time duration, or data size in MB. If the buffer fills during analysis, old data is discarded. This does not affect data already saved to the disk.

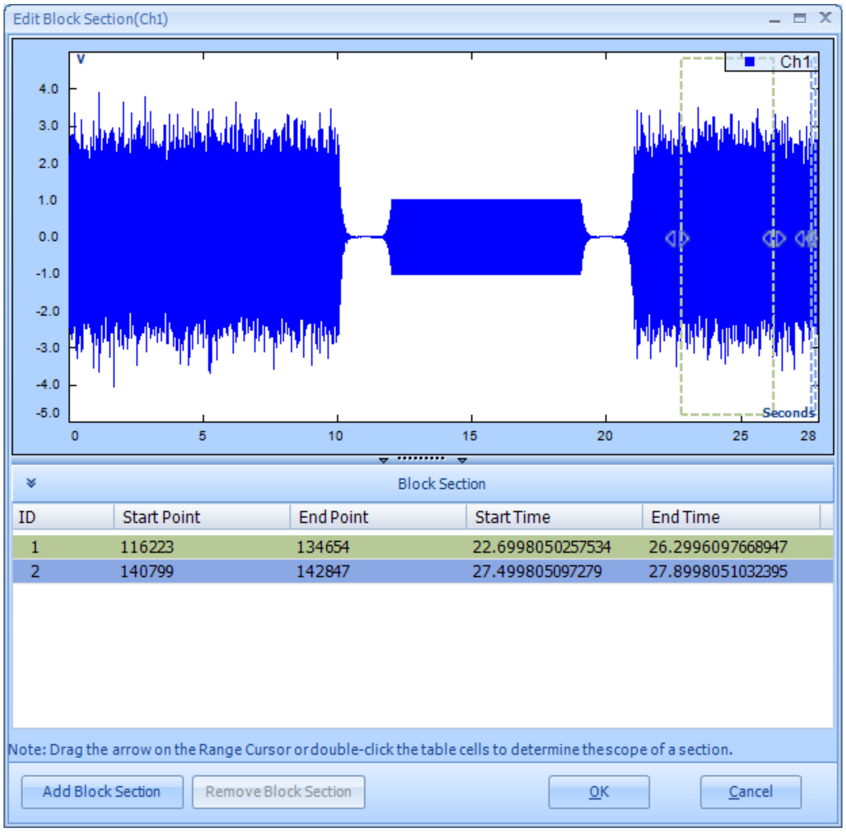


Using Non-Contiguous Sections

If the **Averaging Non-Contiguous Sections of Time Series Data** option is selected in the Project Configuration window under the Other Options tab, a Set Block Section button will appear in the lower section of the Data Source window.

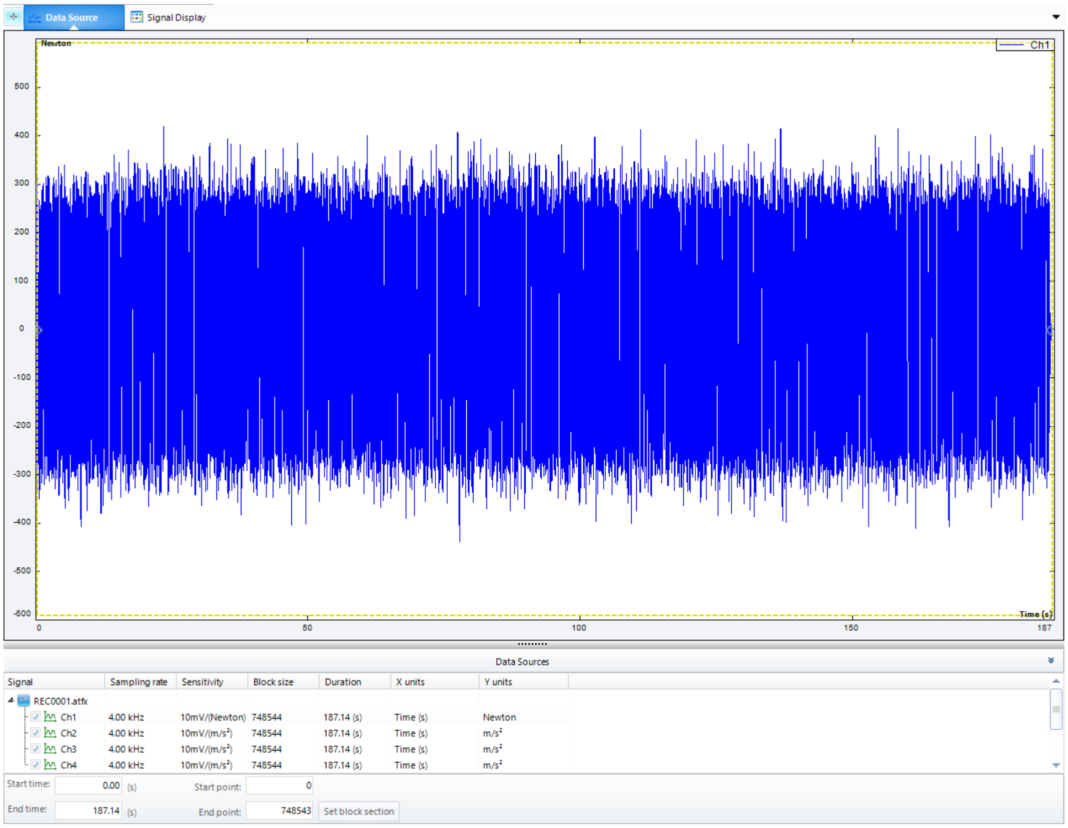


Use this to add block sections to the analysis domain. Each block section is specified with a start and stop point or time. The sections can also be adjusted graphically by moving the edges of the dashed boxes that appear around them.



Data Source

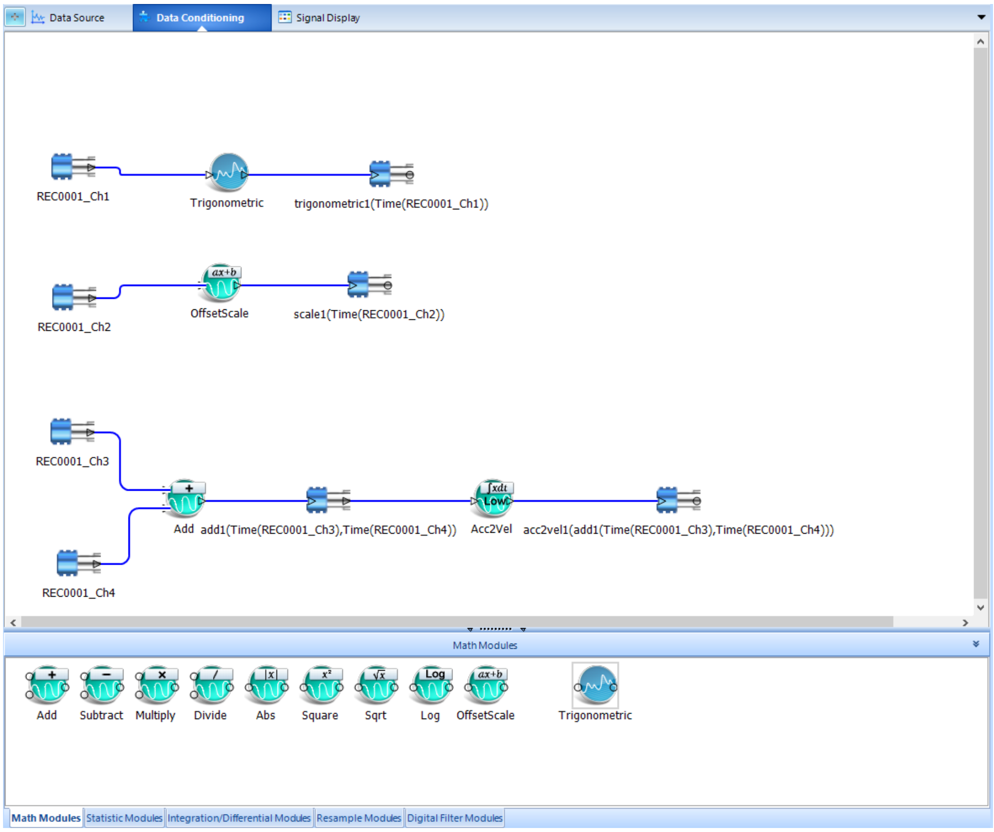
The **Data Source** layout tab displays source data from the recorded file or files associated with the current project. Enable or disable source signals in this tab and set the time domain for analysis by dragging the left and right edges of the yellow box. The start and stop point, or time, can be entered manually in the text field below.



Use the mouse to drag both boundaries shown in yellow to define the time segment to be analyzed.

Data Conditioning

Some analysis types provide the option to enable **Data Conditioning** modules. Data Conditioning allows mathematical operations to be performed on time streams from the data source or sources. Each plug icon is a signal source, and the conditioning modules can be dragged from the toolbox onto the conditioning workspace. To connect a source to a conditioning module, drag a line from the source output to the conditioning module input. Once they are connected, the output of the conditioning module becomes a new signal source, which will be used just like one original signal source. It can be connected to more conditioning modules.

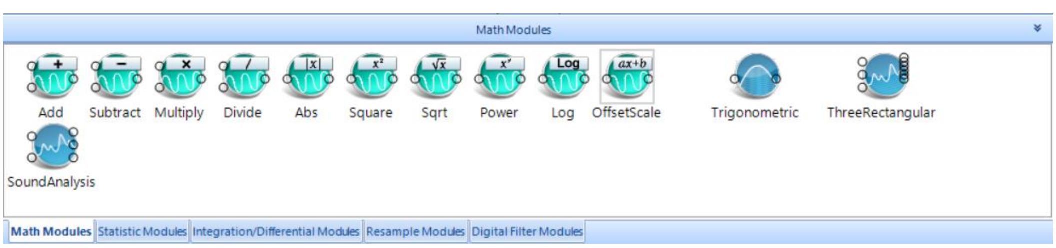


Data Conditioning modules are categorized into the following groups: Math Modules, Statistic Modules, Integration/Differentiation Modules, Resample Modules, and Digital Filter Modules. Users can configure these graphically by connecting conditioning modules and signal sources in the Data Conditioning tab of the Analysis Page. Each resulted output from the conditioning module is treated like a regular signal source. The corresponding signals will be added to the signal list and analyzed.

Math Modules

The following basic math operation modules are available under Math Modules: Add, Subtract, Multiply, Divide, Abs, Square, Sqrt, Log, OffsetScale, Trigonometric, ThreeRectangular, SoundAnalysis.

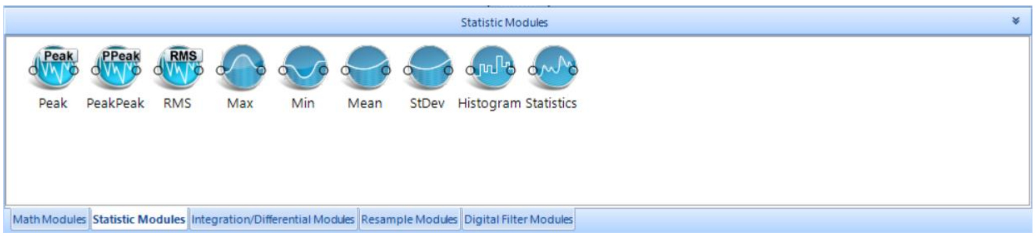
The setup interface is displayed in the Control Panel when the OffsetScale and Trigonometric modules are in use. Users can set the parameters for these modules there.



Statistic Modules

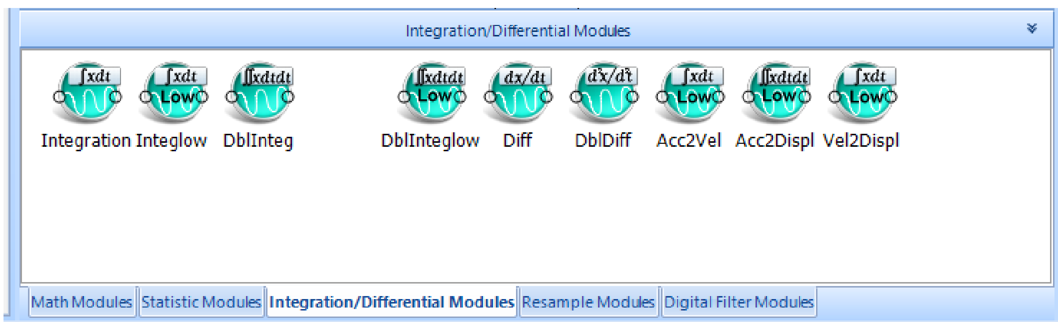
The following statistic modules are available: Peak, PeakPeak, RMS, Max, Min, Mean, StDev, Histogram, and Statistics.

Users can define the setup parameters for each module in the corresponding sections of the **Control Panel**.



Integration/Differentiation Modules

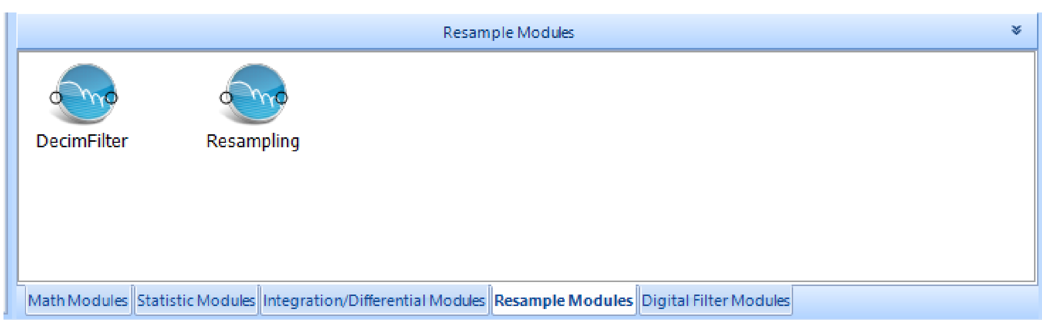
Integration and differentiation modules are displayed in the following screenshot.



Users can specify the cut-off frequencies for **Integlow, Acceleration to Velocity**, or **Displacement Integration** modules.

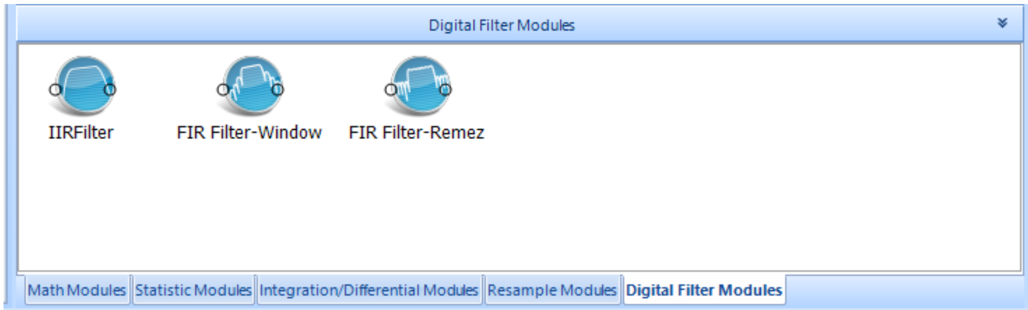
Resample Modules

DecimFilter and Resampling are provided in the **Resample Modules** tab.



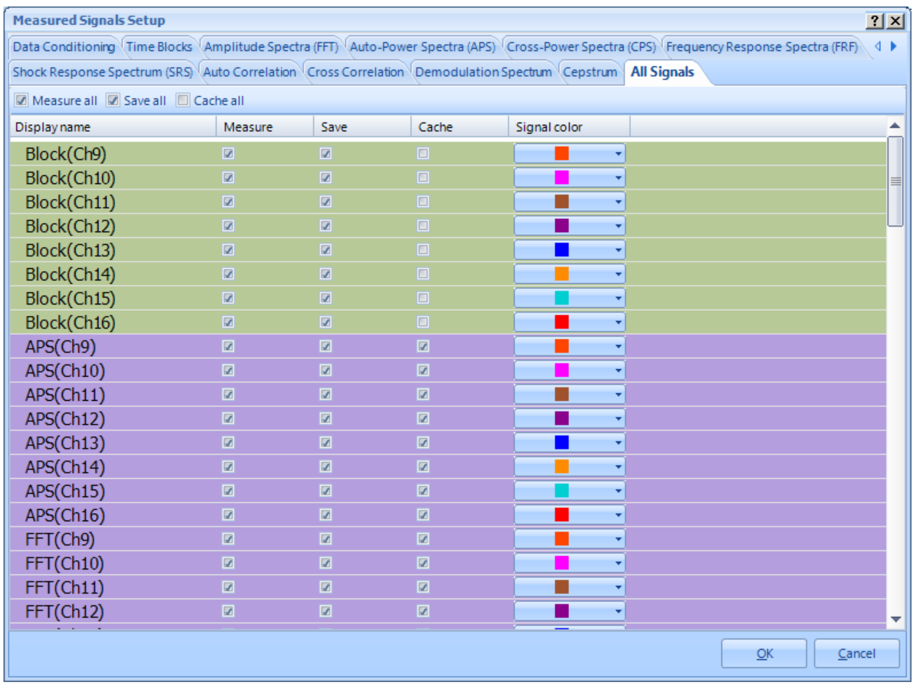
Digital Filter Modules

Three filters are provided, and the setup parameters can be specified in the corresponding section of **Control Panel**.



Measured Signals

All signals available for computing are listed under the **Setup->Measured Signals** tab.



This includes defined signals from each **Data Conditioning** module, as well as block data computed from the original data file, such as Auto-Power Spectra (APS).

Each signal has a **Measure, Save**, and **Cache** option. Signals with the **Measure** option checked are listed under the **Live Signals** tab and available for display. Signals with the **Save** option selected are saved to the disk when the **Save** button is clicked in the control panel, or when an automatic saving option is specified in **Project Configuration**. Signals with the **Cache** option are cached in the memory for faster access by the software.

Control Panel

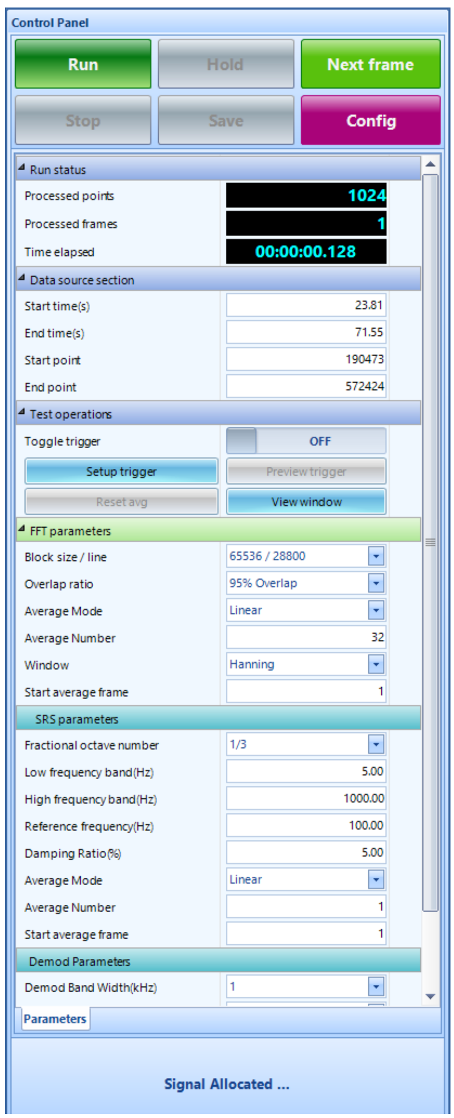
The **Control Panel** contains sections for post processing operation control, status display, and parameter setup.

The following control buttons are available for post analysis processing:

- **Run** starts the analysis
- **Hold/Continue** pauses and resumes the analysis
- **Stop** stops the analysis
- **Next Frame** processes one frame, and then stops
- **Save** saves block signals.
- **Config** opens the Project Configuration window

Run status contains the following information:

- **Processed Points** is the number of processed samples.
- **Processed Frame** is the number of processed frames.
- **Time Elapsed** is the total time duration of the input signals processed.



Octave and Acoustic Analysis

Basics

The **Octave and Acoustic Analysis** function applies a bank of real-time octave filters to the input time streams and generates two types of signals at the same time: fractional frequency band signals (e.g., octave spectra) and the RMS time history of each filter band. The output of each real-time filter bank is in fact a 3D waterfall signal that is arranged on the x-axis as logarithmic frequency and z-axis as time. Frequency weighting is applied in the frequency direction. Time-weighting is applied in the time axis.

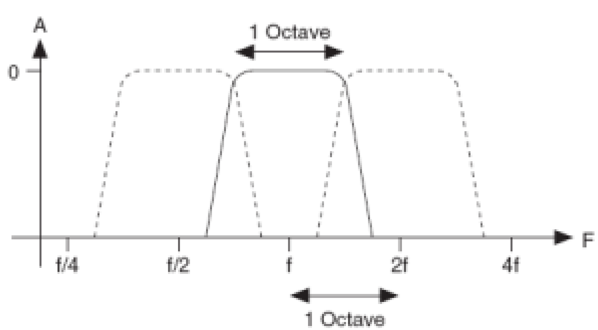
Octave Analysis provides 1/N octave analysis using true real-time digital filters that conform to ANSI std. S1.11:2004, Order 3 Type 1-D and IEC 61260-1995 specifications. A, B and C weighting filters can be applied to the input data. Output results are weighted or un-weighted RMS values. The output can be normalized with a calibration value. The results can be plotted on log or linear axes and exact or preferred frequency values are supported.

Acoustic Analysis provides 1/N octave analysis using true real-time digital filters that conform to ANSI S1.11 and IEC 61260 specifications. Each band filter is designed in accordance to ANSI S1.11 and IEC 61260 specifications by transforming the original analog transfer function to the digital domain by means of the bi-linear transform. The filter order can be specified, and the frequency ratio can be calculated using the binary or decimal system.

The RMS reading of each octave filter is usually represented by a "bar" in the spectrum plot. Keep in mind that the octave filters have "skirts" on both sides. They are not as straight as the bars depicts. The adjacent filters always overlap. Due to this reason, a sine tone at 1 kHz will not only excite the filter with a center frequency at 1 kHz, but also all other filters.

Full Octave Filters

An octave is a doubling of frequency. For example, frequencies of 250 Hz and 500 Hz are one octave apart, as are frequencies of 1 kHz and 2 kHz.



Full octave analysis (e.g., 1/1 octave) displays the frequency characteristics of a signal by passing the signal through a bank of band-pass filters where the center frequency of each filter is one octave apart. If the lower and upper cutoff frequencies of a band-pass filter are f_L andf_H, then the center frequency, f_c can be determined with:

$$f_c = \sqrt{(f_L * f_H)}$$

The nominal frequency ratio G is determined by:

$$G = f_H / f_L$$

Two systems are used in the industry, base-two or base-ten systems. For base-two systems, G = 2. For base-ten systems, G = $10^{3/10}$. The base-ten system is used for Spiders.

The proportional bandwidth property will divide the frequency information uniformly over a log scale. It is very useful in analyzing a variety of natural systems. For example, the human response to noise and vibration is very non-linear and many mechanical systems have a behavior that is best characterized by proportional bandwidth analysis.

Fractional Octave Filters

To gain finer frequency resolution, the frequency range can be divided into proportional bandwidths that are a fraction of an octave. For example, with 1/3 octave analysis, there are three band-pass filters per octave where each center frequency is $10^{1/10}$ the previous center frequency.

In general, for 1/N octave analysis, there are N band pass filters per octave such that:

$$\frac{f_H}{f_L} = \left(10^{3/10}\right)^{1/N}$$

$$f_{c\ j+1} = f_{c\ j} * \left(10^{3/10}\right)^{1/N}$$

where 1/N is called the fractional bandwidth resolution.

For Spiders, the equation and table below define the center frequency of each fractional filter.

$$f_c = 10^{3X/10N}$$

For example, for 1/1 Octave (N =1) the first center frequency (index X = 1) is computed as

$f_c = 10^{\frac{3*(-3)}{10*1}} = 0.125$ Hz

	1/1-Octave	1/3-Octave	1/6-Octave	1/12-Octave
Standard	IEC 225-1966 DIN 45651 ANSI S1.11-2004 Order 7 Type 1-D	IEC 225-1966 DIN 45651 ANSI S1.11-2004 Order 3 Type 1-D	N/A	N/A
X (index)	-3 ~14	-10 ~ 43	-20 ~ 86	-40 ~172
Total number of	18	54	107	213
f_c (Hz)	0.125 – 16k	0.1 – 20k	0.1 – 20k	0.1 – 20k

Nominal Center Frequencies (Mid-band Frequencies)

Nominal center frequencies are “round” numbers that were inherited from the old analog octave filters. These values are rounded mid-band frequencies for the designation of band pass filters. The nominal mid-band frequencies for 1/1-octave and 1/3-octave are listed in the ANSI S1.11-2004 Annex A. The standard also describes how to decide the nominal mid-band frequencies for other fractional octave bands.

The exact center frequency of the filter band is usually not the same as that of the nominal frequency. For example, in a 1/3 octave, the exact center frequencies 794.33 Hz, 1000 Hz and 1258.9 Hz are used to correspond to the filters with nominal frequencies 800 Hz, 1000 Hz and 1250 Hz.

Band Edge Frequencies of Fractional Filters

The low and high edge frequencies of a filter can be calculated based on the frequency ratio, G, and the fractional octave resolution N (=1, 3, 6, 12…)

$$Lower\ Edge\ Frequency \quad f_L = f_c * \left(10^{3/10}\right)^{-1/2N}$$

$$Upper\ Edge\ Frequency \quad f_H = f_c * \left(10^{3/10}\right)^{1/2N}$$

The bandwidth of the filter is: BW = $f_H - f_L$

When starting or resetting the filtering operation of the fractional-octave filters, time is required before the measurements are valid. This time is referred to as settling time and is related to the bandwidth of any particular filter. The lowest frequency band has the smallest bandwidth and defines the settling time required before the complete fractional-octave measurement valid can be considered valid. Settling time is approximately calculated as five divided by the bandwidth.

$$Settling\ time = \frac{5}{BW} = \frac{5}{f_H - f_L}$$

Note the settling time depends on the bandwidth which changes with the center frequency. A narrower filter and a lower frequency band requires a longer settling time.

Analysis Frequency Range

In Post Analyzer, the user can set the analysis range by changing the lowest and highest f_c as the Analysis Parameters.

Analysis Range	1/1 Octave	1/3 Octave	1/6 Octave	1/12 Octave	1/24 Octave
Lowest f_c (Hz)	0.1 1 10 100	0.1 1 10 100	0.1 1 10 100	0.1 1 10 100	0.1 1 10 100
Highest f_c (Hz)	125 250 500 1000 2000 5000 10000 20000	125 250 500 1000 2000 5000 10000 20000	125 250 500 1000 2000 5000 10000 20000	125 250 500 1000 2000 5000 10000 20000	125 250 500 1000 2000 5000 10000 20000

Frequency Weighting

The human hearing system is more sensitive to some frequencies than others, and its frequency response varies with the level. In general, low frequency and high frequency sounds appear to be less loud than mid-frequency sounds, and the effect is more pronounced at low pressure levels, with a flattening of response at high levels. Octave analysis and sound level meters therefore incorporate weighting filters, which reduce the contribution of low and high frequencies to produce a measurement that corresponds approximately to what humans hear.

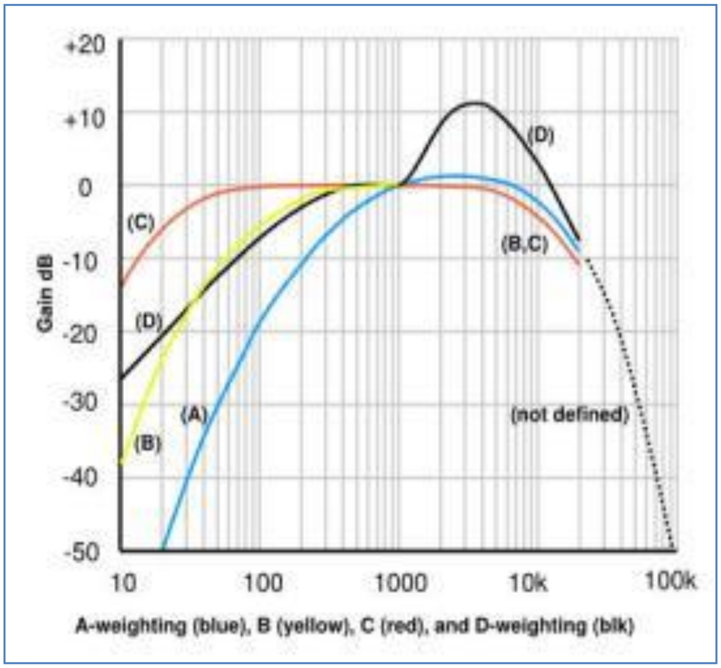


Figure 2. Frequency weighting filter shapes.

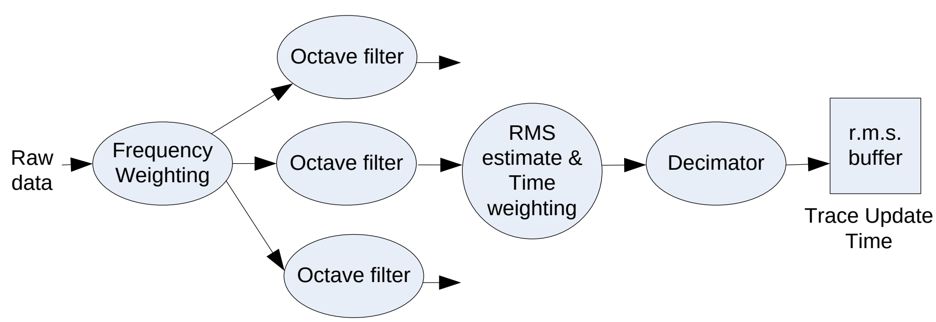
The Post Analyzer software provides A, C, Z weightings conforming to IEC 61672-1 2002 and B weighting conforming to IEC 60651 in both **Octave Analysis** and **Sound Level Meter**. The frequency weighting in the octave filters will affect the results of all filter bands.

Time or RPM based RMS Trace of the Octave Filters

The ANSI and IEC standard does not require storage of band pass filter output time history. However, users may want to view this information. In Post Analyzer, the RMS history of all band pass filters is stored in the RMS quantity.

The RMS history can be stored against one of two variables: Time or RPM.

Both the input and output of a digital filter are a series of data points. While it requires excessive memory to keep all the time data of all the filters, it is useful to keep the so-called RMS history of each filter output. The RMS time history is computed after the time weighting averaging operation.

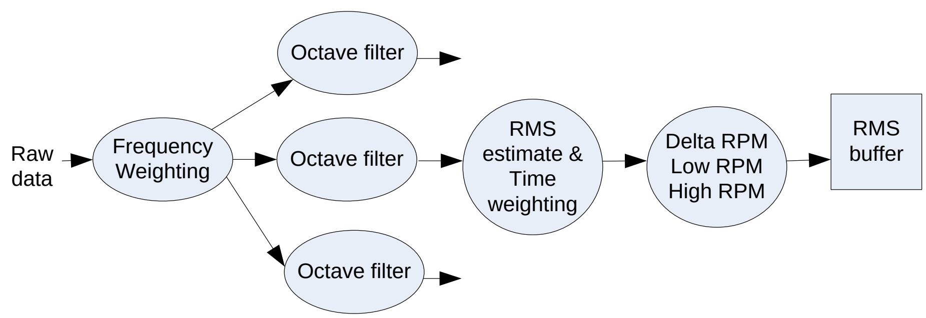


The Decimator is used to allow the user to choose the length of time to save RMS data. For example, given a buffer length of 1,024, a Trace Update Time of 5 ms will keep about 5 seconds of RMS history; if this update time is set to 5 seconds, it will record 1.4 hours of RMS history.

If a cut is made in the Z axis direction, the result is an Octave Spectrum. If a cut is made in the X-axis, the result is called a Time Trace.

The Time Trace stores the RMS history of each filter output. The spacing between two points of the Time Trace (in seconds) is referred to as Trace Update Time. Spider systems allocate one Time Trace for each channel to display. Keep in mind that this Time Trace buffer is the output of a specific filter. Users can change the center frequency of the Time Trace filter during a run, which will change the content of the time trace buffer display.

Alternatively, the RMS trace can be stored using RPM as a variable. This method is particularly useful in automotive NVH applications. The following diagram demonstrates how one of the filter outputs is stored in the RPM trace.



Exponential and Linear Averaging

Linear Average: The Linear Average method uses a fixed time period to sum up the historical power value of each filter and then takes the square-root to calculate the averaged RMS value. In Linear Average, the RMS trace update time is governed by the time period of averaging. For each time period of averaging, one RMS value per frequency bin is produced.

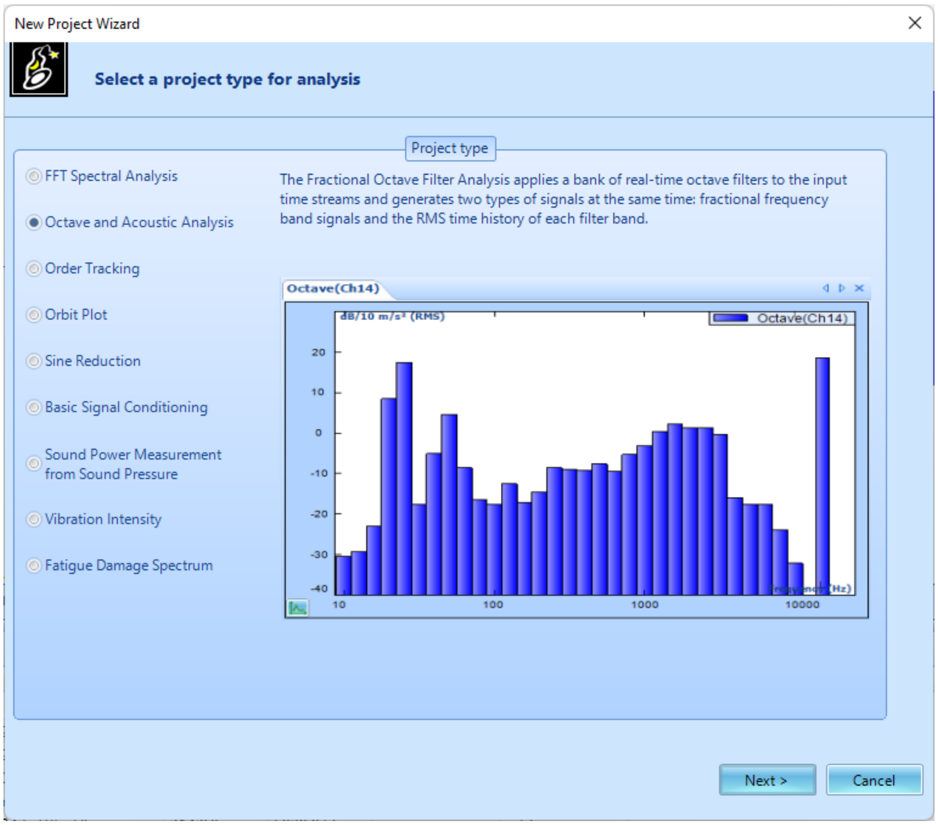
Exponential Average: Exponential Average applies an exponential time constant to the historical power values of each filter and takes the square-root of the averaged power value. A time constant of 0.125 seconds is equivalent to “Fast” averaging and 1.0 second is equivalent to “Slow” averaging in the acoustics. In Exponential Average, the RMS trace update time is independent of the time constant.

Peak Hold: Peak Hold retains the maximum value in each frequency bin over the period of time since the last “start” or “restart”.

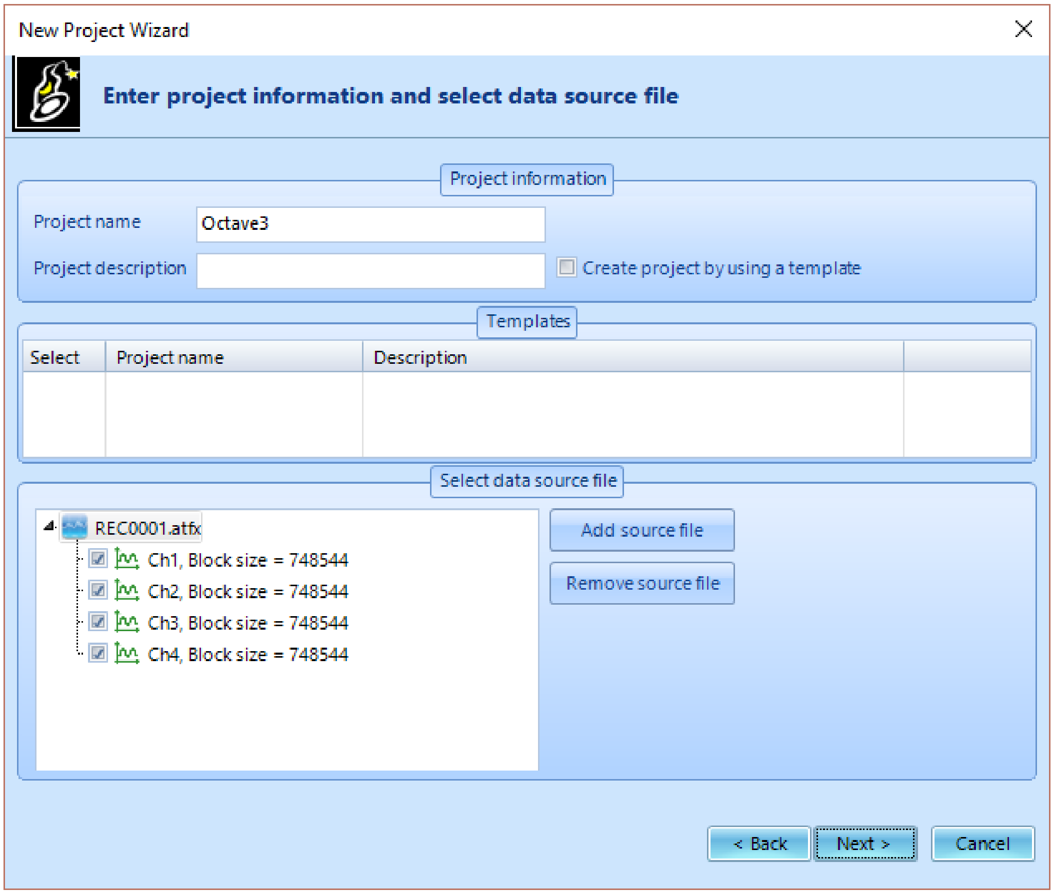
As discussed previously, each filter may have a different settling time.

Create a Project for Octave Analysis

Create a new project by clicking **New Project**. Select **Octave and Acoustic Analysis** as the project type. Click **Next** to continue.

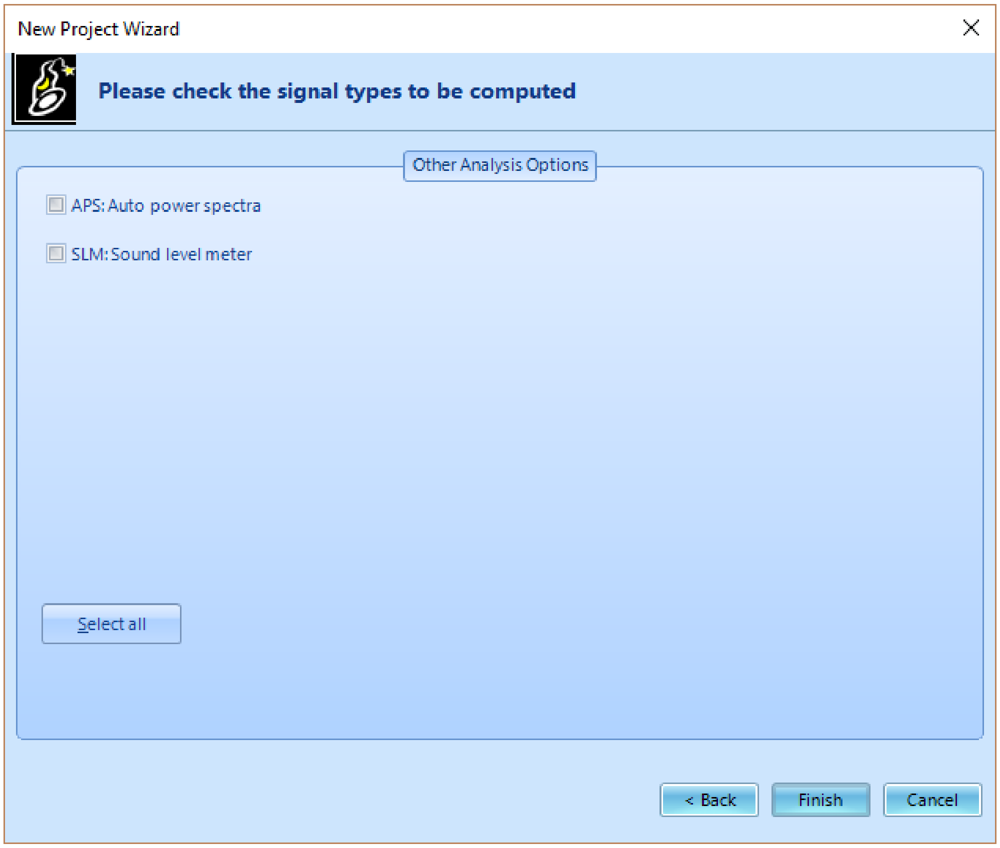


In the following window, enter a name for the project and optionally add a description. If **Create project by using a template** is selected, the previously defined template projects will be shown in the Templates section. Highlight a selection to use as a template and the new project will incorporate the same project setup and layout.



In the **Select data source file** section, select the source data files to recall with this project by clicking the **Add source file** button. The source file is usually a time stream recording in ATFX format. Once a source file is selected, the **Remove Source File** button becomes active. To remove a selected source file, highlight the entry and click the **Remove Source File** button.

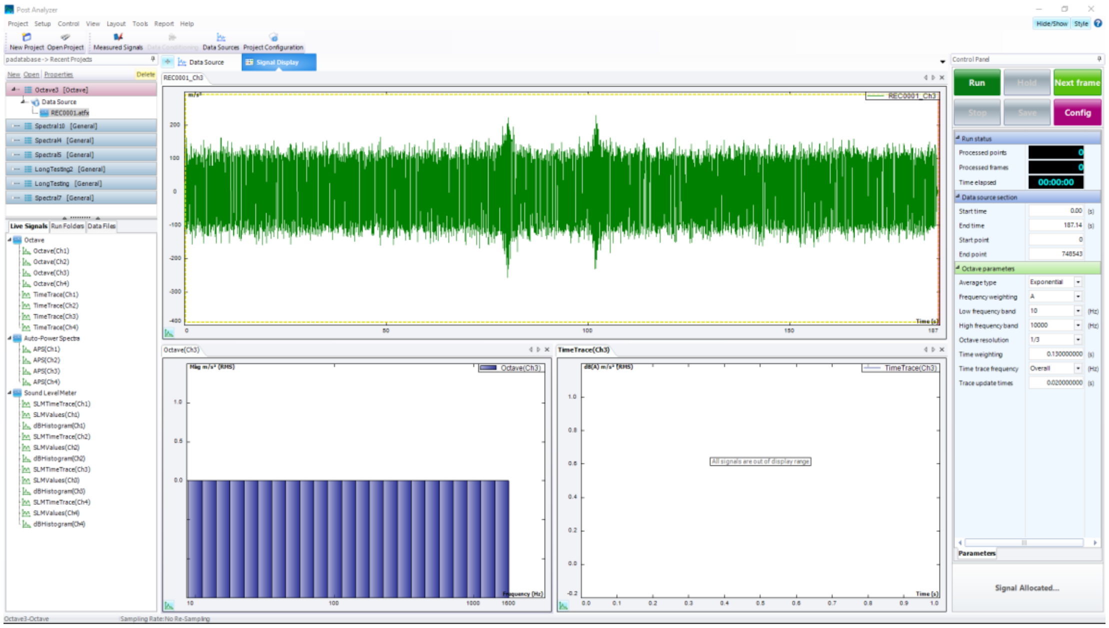
Check the channels requiring post analysis and click **Next** to open the next window.



Select APS and/or SLM to enable these processes or simply click the **Select all** button to select both. Click **Finish** and a new project is created.

Overview

After creating or opening a project, the main Post Analyzer window is displayed. There is a toolbar across the top, and 3 main sections in the middle. The left side contains **Recent Projects, Live Signals**, and **Saved Files** views; in the middle are one or more **Signal Display** views; and on the right is the **Control Panel** which also contains the analysis parameters.



Recent Projects

The **Recent Projects** list displays the currently active project and projects previously opened in the application. The current project provides a shortcut toolbar which contains the **Measured Signals** setup, **Data Conditioning** setup (if data conditioning is checked when creating the project), **Data Sources** setup, and **Project Configuration** window. It also displays the data file associated with the project.

Live Signals, Run Folder, and Source Files

Below the **Recent Projects list** is the **Live Signals, Run Folder, and Source Files list**. Live Signals includes the signals in the Data Source file or files, as well as new signals created as part of the analysis. Live signals are not saved to the host computer hard disk but are available for display in the **Signal Display** view tab. Anything saved to disk is displayed in the Saved Files list, which is associated with a directory in the file system. Run Folders contain data for each run, which is saved in a defined file format accessed from the Data Files tab.

Signal Display

The middle of the main window contains **Signal Display** view tabs, where live and saved data is displayed. More than one of these tabs can be created, but there always is at least one. Each of these views can contain one or more **Display Windows**, with a fully customizable layout. These display windows move freely, be resized, and display any valid combination of live or saved signals. Create new view windows by selecting an option from the View menu.

Control Panel and Analysis Parameters

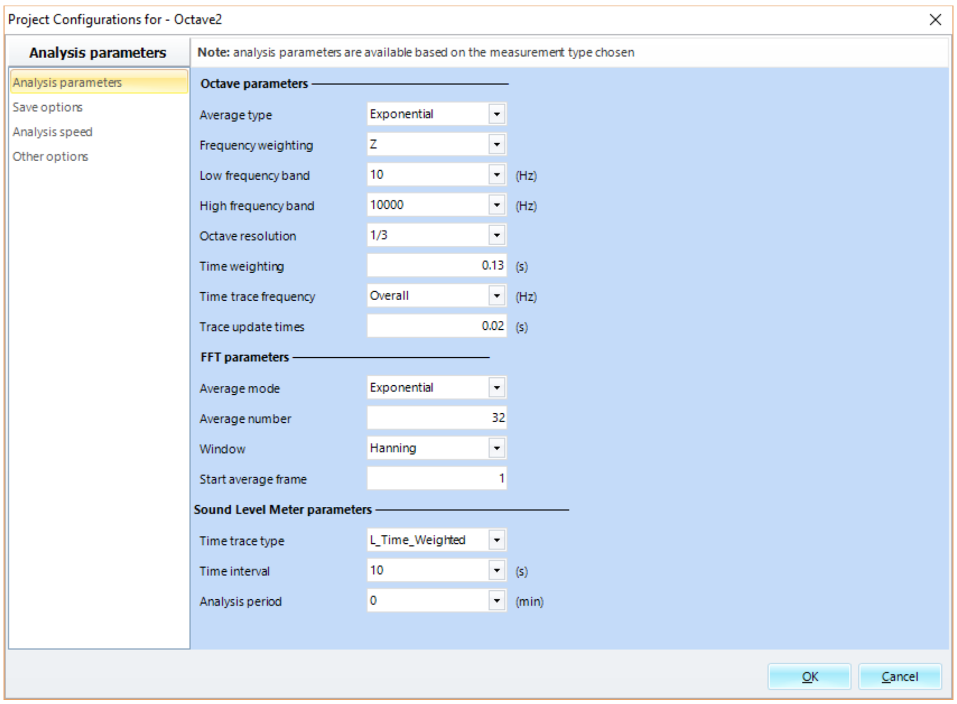
The right side of the window has the **Control Panel** (which controls the post-analysis operation) and the **Analysis Parameters** (which contains setup parameters specific to the analysis type). The **Config** button on the control panel opens the **Project Configuration** window, described in the following section.

Project Configuration

The **Project Configuration** window provides settings for saving data, analysis speed, and other options that are mostly independent of the analysis type to be performed.

Analysis Parameters

Analysis parameters consist of Octave, FFT, and Sound Level Meter parameters.

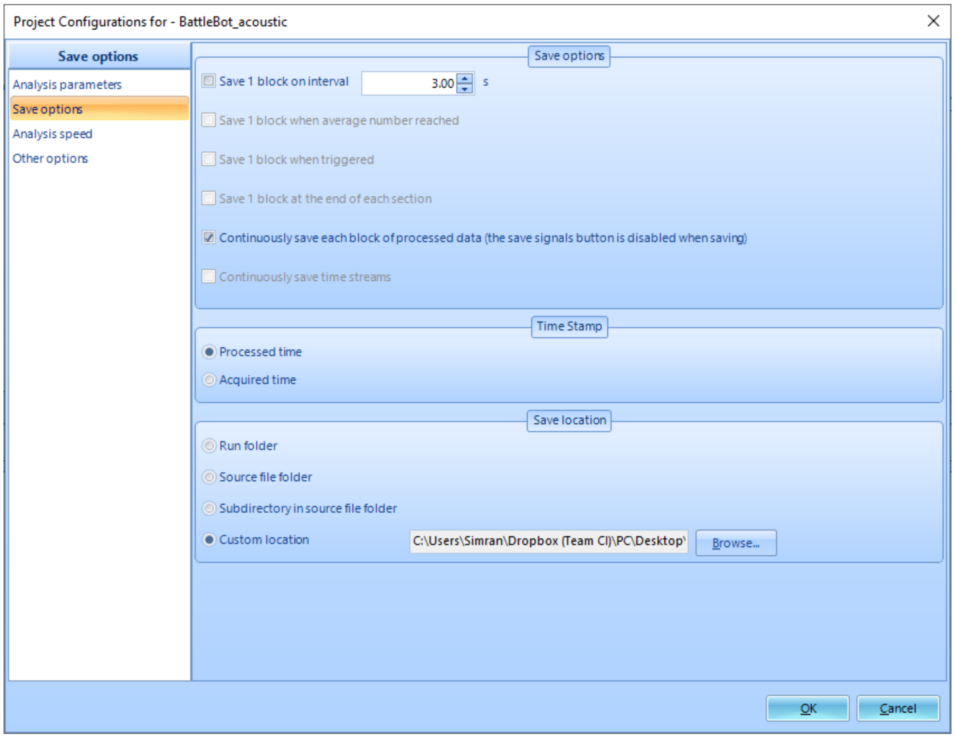


Octave parameters are listed below.

- **Average Type** defines the averaging type including exponential, linear and peak hold.
- **Frequency Weighting** defines the frequency weighting including A, B, C or Z.
- **Low Frequency Band (Hz)** defines the low frequency band of the measurement in Hz.
- **High Frequency Band (Hz)** defines the high frequency (span) of the measurement in Hz.
- **Octave Resolution** defines the octave resolution including: 1/1, 1/3, 1/6, 1/12, and 1/24.
- **Time Weighting** defines the time weighting for exponential averaging.
- **Time Trace Frequency (Hz)** defines which center band frequency or overall frequency weighted band is used to plot time traces.
- **Trace Update Time(s)** defines the time trace display duration. Select a larger update time to create a longer time trace display duration.

Save Options

After running the analysis, computed signals are listed under the **Live Signals** tab. However, this data is only stored in a display buffer and must be explicitly saved to a disk to create new data files. There are multiple ways to save. One method is to manually click the **Save** button in the control panel, which saves data according to the Save settings defined in the **Measured Signals** setup. The other method is to enable an option listed in the section below to automatically save data during the analysis.



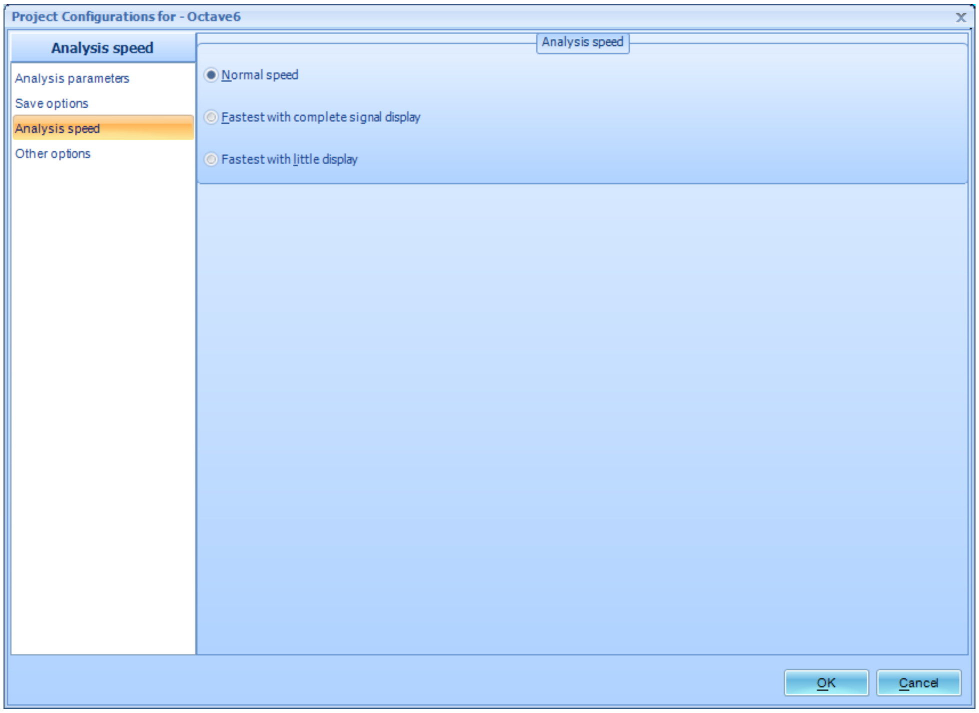
There are many options to save block data, and one option to save stream data. Most data computed in Post Analyzer is block data, such as APS and SRS spectra. Some data, such as the output of digital filters or other data conditioning blocks, are continuous time streams. Only one type of data can be saved at a time – block or time stream.

- **Save 1 Block on Interval** saves one block of data for every computed signal with the save option selected in the Measured Signals setup per the selected interval.
- **Save 1 Block when Average Number Reached** saves one block of data every time the number of processed blocks reaches the average number set under the Analysis Parameters.
- **Save 1 Block when Triggered** saves one block of data when a trigger occurs. The trigger is set up by clicking the Setup Trigger button in Analysis Parameters.
- **Save 1 Block at the end of Each Section** saves one block at the end of the analysis section.
- Continuously Save Each Block of Processed Data saves *every* block of data computed.
- **Continuously Save Time Streams** saves all time stream data. Selecting this option disables all options related to block data saving.

The **Time Stamp** section is used to set the time axis on time-based signals. The axis can either display the time relative to the start of the test or to the actual date and time the data was acquired.

The **Save Location** section allows users to select the location of the file system used to create new data files from the following options: **current run folder, same folder as data source file folder, subdirectory in source file folder,** or a **custom location**.

Analysis Speed

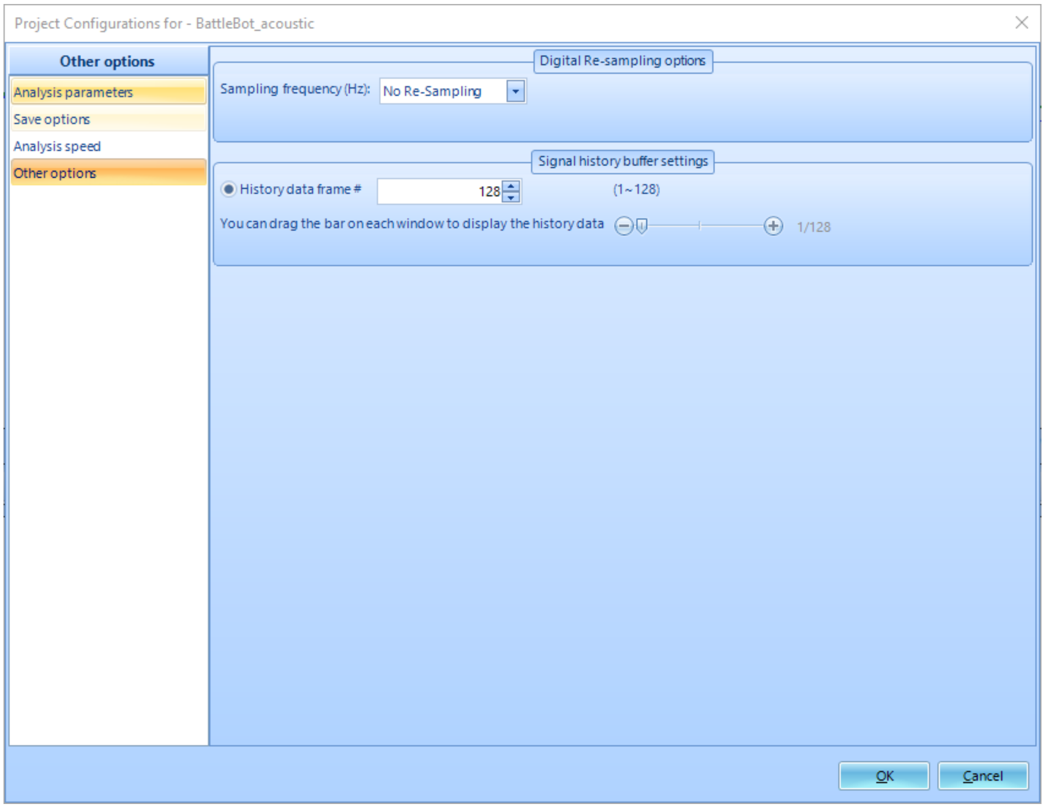


Analysis Speed adjusts the amount of processing time used for the signal display during post-processing. Increased display details result in less available processing resources for post processing data analysis. **Normal Speed** is the default setting that balances data processing and the signal display. **Fastest with Complete Signal Display** prioritizes the display, and **Fastest with Little Display** prioritizes data processing.

Other Options

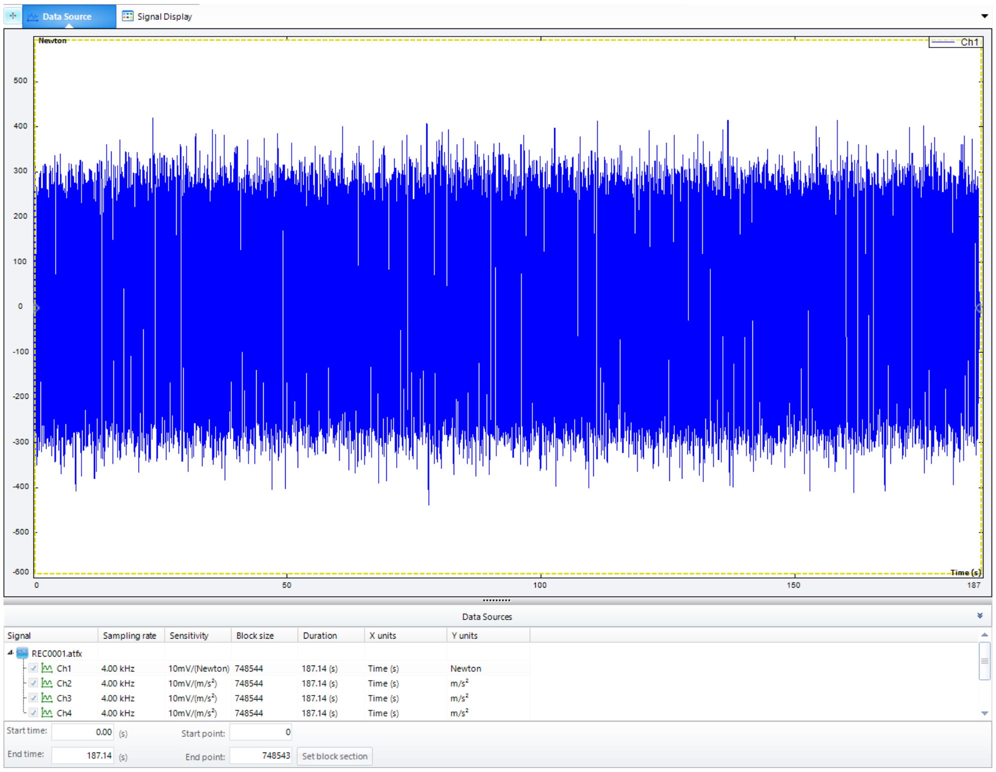
Digital Resampling allows resampling of all source signals to the selected sampling rate to meet the requirement of the other interface. Sampling rate stages can be selected from the dropdown menu.

Signal History Data Buffer sets the size of the buffer used to store and display the signals under the Live Signals tab. This buffer can be sized according to a number of frames. If the buffer fills during the analysis, the old data is discarded. This does not affect data already saved to disk.



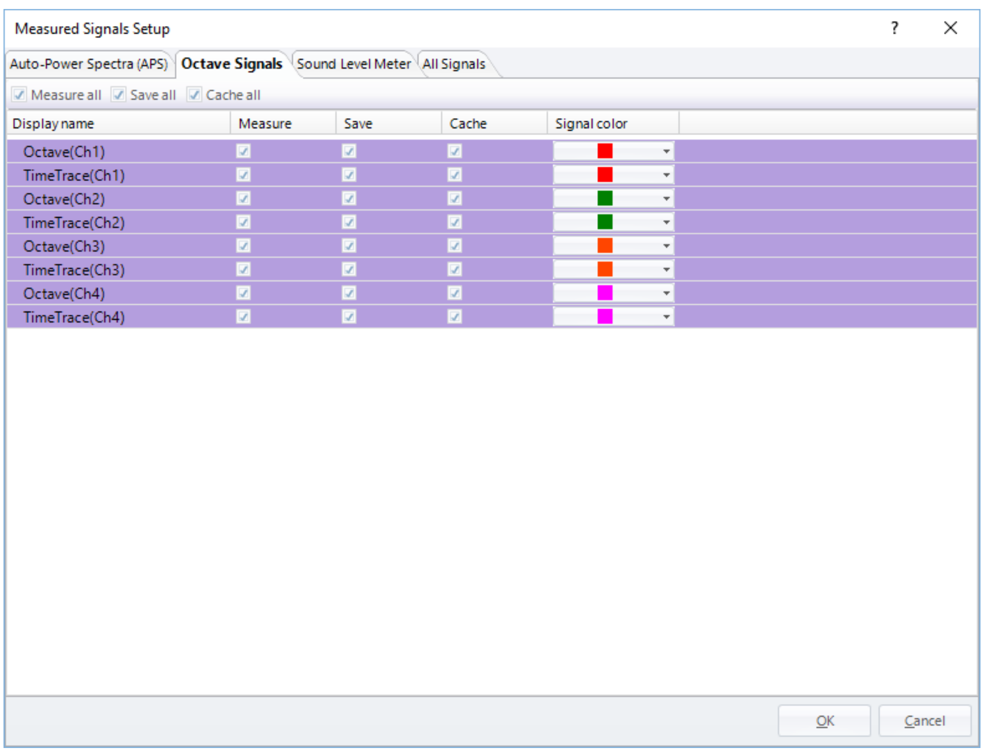
Data Source

The **Data Source** tab displays the source data from the file or files associated with the current project. Enable or disable source signals and set the analysis time range by dragging the edges of the yellow box. Users can enter the start and stop time, or point, manually in the provided text fields.



Measured Signals

All available signals are listed under the **Setup->Measured Signals** tab.



This includes the output from every Data Conditioning module used as well as block data computed from these, such as Auto-Power Spectra (APS).

Each signal has a **Measure, Save**, and **Cache** option. Signals with the **Measure** option checked are listed under the **Live Signals** tab and available for display. Signals with the **Save** option selected are saved to the disk when the **Save** button is clicked in the control panel, or when an automatic saving option is specified in **Project Configuration**. Signals with the **Cache** option are cached in the memory for faster access by the software.

Control Panel

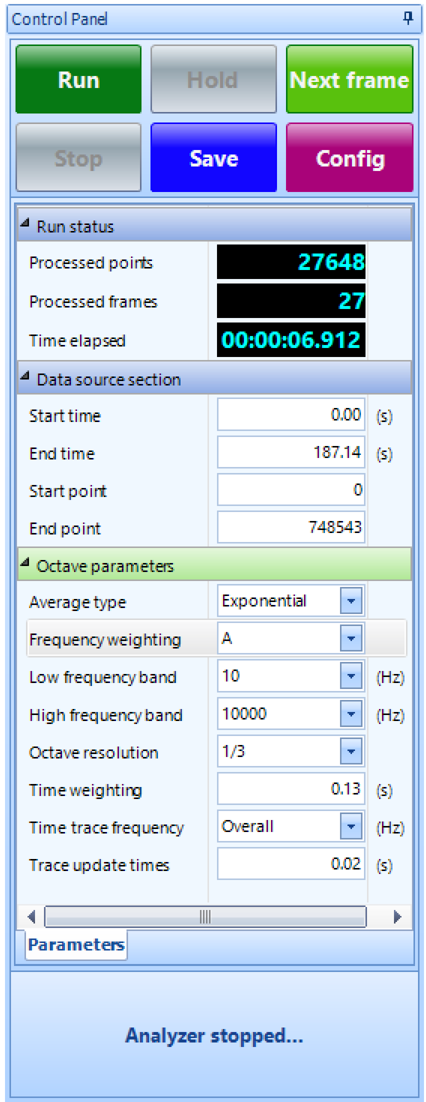
The Control Panel is used to control post processing operations. It also displays the analysis status. Analysis parameters can be quickly modified here as well.

The **Control Panel** provides the following control buttons:

- **Run** starts the analysis.
- **Hold/Continue** pauses and resumes the analysis.
- **Stop** stops the analysis.
- **Next Frame** processes one frame, and then stops.
- **Save** saves block signals.
- **Config** opens the Project Configuration window.

The **Run Status** panel displays the following status information:

- **Processed Points** # is the number of processed samples.
- **Processed Frame #** is the number of processed frames.
- **Time Elapsed** is the total time duration of the input signals processed.


Control Panel
Run
Hold
Next frame
Stop
Save
Config
Run status
Processed points
27648
Processed frames
27
Time elapsed
00:00:06.912
Data source section
Start time
0.00
(s)
End time
187.14
(s)
Start point
0
End point
748543
Octave parameters
Average type
Exponential
Frequency weighting
A
Low frequency band
10
(Hz)
High frequency band
10000
(Hz)
Octave resolution
1/3
Time weighting
0.13
(s)
Time trace frequency
Overall
(Hz)
Trace update times
0.02
(s)
Parameters
Analyzer stopped...

Order Tracking

Basics

Order Tracking is a general term describing a collection of software functions used for analyzing the mechanical dynamic behavior of rotating or reciprocating machinery for which the rotational speed can change over time. Unlike the power spectrum and other frequency-domain analysis standards where the changing variable is the frequency, Order Tracking functions presents data related to the variable rotating speed, i.e., RPM (revolution per minute).

The most useful measurements are order spectra and order tracks. An order spectrum gives the amplitude of the signal as a function of harmonic order of the rotation frequency. This means that a harmonic or sub-harmonic order component remains in the same analysis line independent of the speed of the machine.

The technique that observes the changes of any quantity vs. RPM is called **Tracking**, as the rotation frequency is being tracked and used for analysis. Most of the dynamic forces exciting a machine are related to the rotation frequency so the interpretation and diagnosis can thus be greatly simplified by use of order analysis.

Order tracks are simply the observations to the amplitude of the components with fundamental frequency or harmonics. It is a typical type of tracking. There are other types of tracking. For example, the user can track the FFT-based PSD spectra, a fixed band or an octave band, etc.; all these can be called tracking.

With Order Tracking, the PA software can:

- Process a tachometer signal and give a high fidelity RPM measurement
- Measure the order spectra
- Measure the order tracks
- Measure the RPM FFT spectrum
- Measure the energy in fixed bands vs. RPM
- Measure the amplitude and phase of an order relative to the tachometer.

There are several different applications for order tracking. A discussion of some is given below.

The first application, often referred to Run Up/Run Down, is used to evaluate the noise or vibration dynamic response when RPM is used as a changing variable. In this case, the RPM range can be very large, from a few RPM to 10,000 RPM. Typical applications of order tracking are for automotive or aircraft engines testing. The measurements can be any physical quantities such as sound, displacement, velocity, acceleration, torque, etc. The analysis measure can be the amplitude or the power of an order, the energy over a fixed frequency band, a bin of octave filter, etc. The phase information of the responses to tacho is less important in this type of application. In fact, the rotating element might be hidden inside of a mechanical system. The primary result for this type of measurement is the magnitude of the responses vs. RPM.

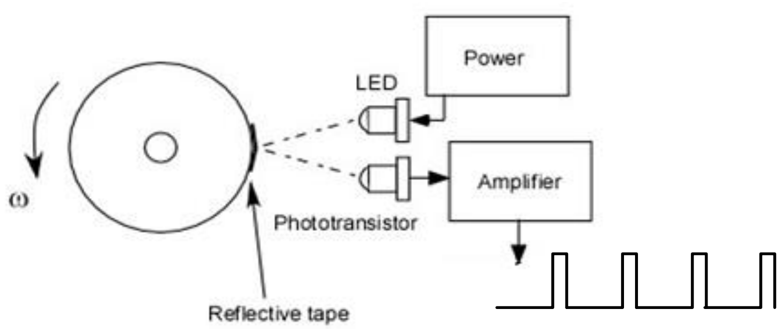
The second application is rotating machine analysis that focuses on the measurement of displacement or velocity of the rotors while it is rotating. The instrument measures the amplitudes of specific orders and their relative phase to a reference signal. The phase is calculated relative to the tachometer input or a separate reference input. This application is common for machine diagnosis and balancing. In this case, the RPM is stable or quasi-stable. Order tracking technology is useful to increase the accuracy of the estimation of orders.

Order signals with phase are useful in the testing of rotating machinery in the Run Up/Run Down process. This is often presented as a “Bode Plot”. The Bode Plot is a borrowed concept from control theory; it is a collection of Amplitude and Phase data over a changing speed range (i.e., Run Up or Coast Down). Some of the setup information depends on the rate of change of the RPM. The Run Up or Coast Down could take anywhere from a few minutes to a few hours (such as for a cold startup on a turbine). Other displays such as the orbit plot are useful as well.

The Post Analyzer by Crystal Instruments includes the ability to measure RPM based octave analysis and sound levels. This feature is similar to order tracks except that spectra are recorded in octave bands with A, B, C or Z frequency weighting. This feature is included in the Acoustic Analysis and Sound Level Meter CSA Templates instead of the Order Tracking Template. Refer to these sections for more details.

Tachometer Signal Processing and RPM Measurement

A **tachometer (tacho)** converts the angular velocity of a rotating shaft into an electrical signal, typically a voltage. It is common for calibrated instruments to provide a measurement of the shaft in units of revolutions per minute (RPM) or revolutions per second (RPS). Many modern rotating machines (electric motors, generators, pumps, turbines, IC engines, etc.) have integrated tachometers that can measure shaft angular velocity. An example of an optical tachometer is illustrated in the following diagram.



The goal of tacho signal processing is to obtain a clean and stable RPM reading. The tacho signal must be carefully processed to provide a base of tracking. Any order tracking results can only be thought of as being as accurate as the tachometer signals that were used to estimate the instantaneous frequency of the order in the analysis process. If the quality of the tachometer channel is poor, the results from all other channels will be poor or even unreliable.

Old analog methods conditioned tacho channels with a tracking ratio tuner using a phase lock loop. The disadvantage of this method is the limited slew rate and the use of complex hardware. Various digital tacho processing methods have been developed to overcome these limitations.

There are two ways to implement a tacho input channel from a hardware design perspective: use a dedicated tacho channel with a digital counter or use an analog input channel.

Dedicated Tacho Channel Using Counter

Using a dedicated tacho channel, usually without an A/D converter, has been popular. This hardware approach contains its own tacho clock which runs at a much higher speed, typically in MHz. This tacho hardware also contains special counters which maintain a continuous counter reading to avoid skipping any triggered cycles of the tacho signal. There is also an option to allow these counters to "average" several tacho periods for cases when the input tacho frequency is very high.

Using Analog Input Channel

Alternatively, some systems use an analog input data channel as a tacho channel. In this case, the tacho clock is actually the sample rate of the data channels. This sample rate usually limits the tacho frequency range since the tacho range is now set by the input data frequency range requirement. In addition, due to the "frame processing" nature of some not-so-well designed input sampling processes, some instruments may be limited to how they acquire the tacho signal. This restriction usually means they get several tacho cycles in every data frame. The result is often an "averaged" value which is acceptable unless the tacho signal is changing frequency during the data frame event, which is often the case.

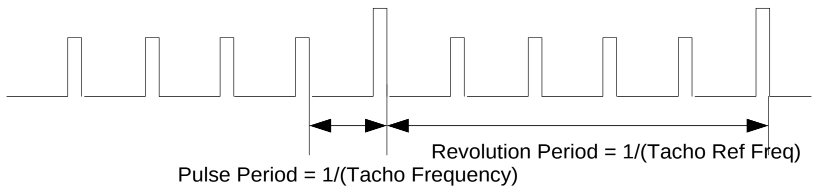
The advancements in electronics and the lowering cost of electronic components places expenses as less of a concern. The approach of a dedicated tacho channel with a digital counter, without A/D, may or may not be the best choice.

The PA software can use any data channel as a tacho channel. For simplicity, channel 1 is used as the tacho. While the data input channel is used as a tacho measurement, the special hardware circuitry allows this data channel to sample at the highest possible sampling rate. In other words, the accuracy of tacho speed measurement is dependent on the current range of the analysis frequency. This technique has several obvious advantages:

- The time domain signal of the tacho input is transformed by the A/D converter into a digital signal. The user can observe the pulse trains of the tacho signal and set the threshold arbitrarily.
- Accurate phase information can be obtained relative to each data channel because the tacho channel, which is fed by a high frequency sampling counter, is synchronized with data channels.
- The RPM estimation is not influenced by the current data sampling rate.

Pulse per revolution is defined as the number of pulses per revolution. It must be defined by the user to enable the software to calculate the tacho reference frequency using tacho frequency. This relationship is:

Tacho Reference freq = Tacho freq / Pulses per Rev



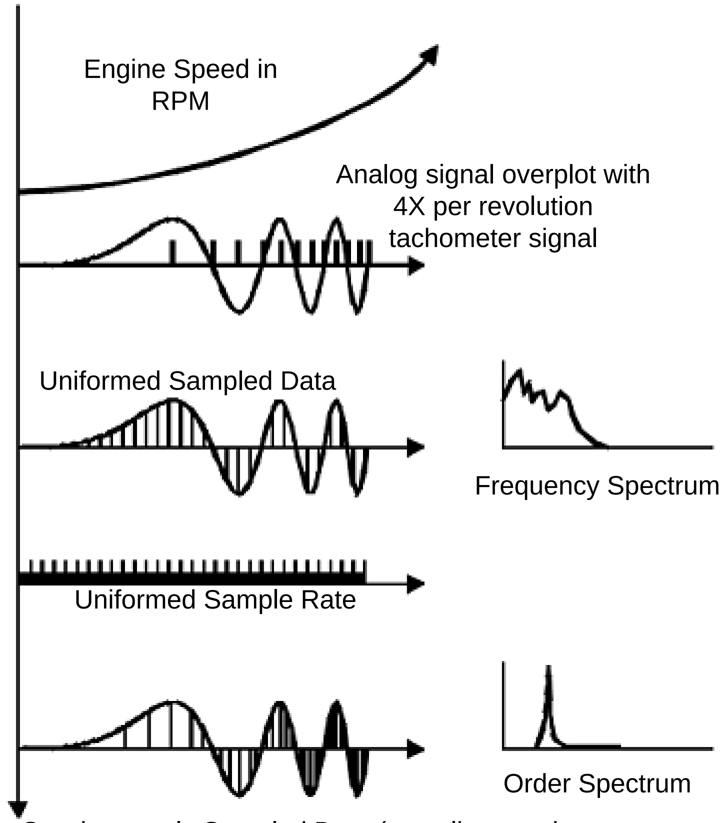
In most rotor tests, especially in balancing, the pulses per revolution is simply 1. However, in other cases, such as in flywheel or geared data measurement, the pulses per revolution can be as high as the hundreds. A dedicated tacho channel with a high-speed counter could work more effectively in this situation.

Order Tracks and Order Spectrum

Knowledge of the rotating speed allows presentation of measurement results in angle and order domains, corresponding to the time and frequency domains. An order is a frequency normalized with some reference frequency, e.g., the shaft frequency. This means that the order of a vibration component in the order spectrum indicates the number of vibration cycles per shaft revolution. The magnitude, which can be measured using EU_{pk}, EU_{rms} or EU_{rms}^2, of an order is the measurement extracted through a tracking filter with the center frequency located at this frequency. Multiple measurements of a range of orders will construct an **Order Spectrum**. An order power spectrum measurement gives a quantitative description of the amplitude, or power, of the orders in a signal. It provides a good view of all order components of a signal. This can help users find significant orders and compare the level of different order components.

There are two methods to perform rotationally coherent sampling: phase-locked frequency multipliers and digital resampling. Phase-locked frequency multipliers were mainly used in early work. They generate sampling pulses based on a rotational reference signal. These sampling pulses control the sampling process. Note that the sampling frequency will depend on the rotational speed, and thus an adjustable anti-aliasing filter is needed. This complicates the method considerably. In the digital resampling technique, the time signal is conventionally sampled together with some rotational reference signal. The time signal is then digitally re-sampled to the angle domain by interpolation techniques. The rotational reference signal can be acquired with a tachometer or an incremental pulse encoder.

The following diagram shows conceptually how angle data resampling can be used to analyze vibrations from an engine during start up. Once the signal has been transformed into its angle domain, the FFT can be applied to analyze the order spectrum of the vibrations.

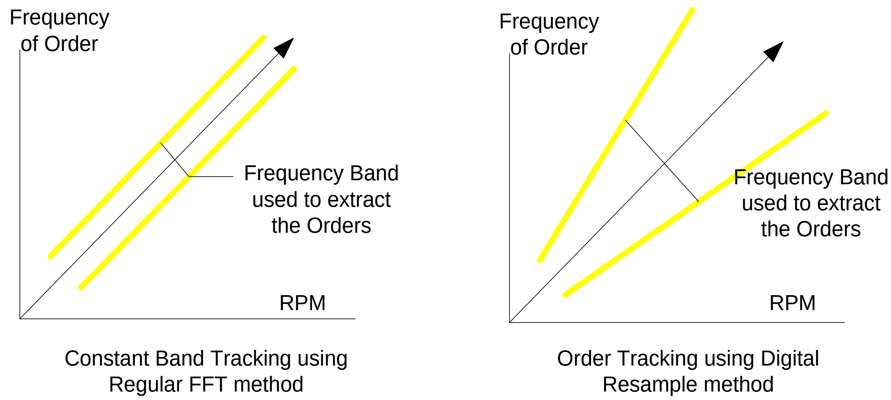


The last plot in the preceding diagram shows that the sampling rate will be determined both by instantaneous tacho speed and required analysis frequency range.

In PA software, the order tracks and order spectrum are computed with a proprietary technology that combines digital resampling, data decimation, interpolation, DFT and FFT calculations.

An important concept that must be introduced now is called delta order, ΔOrder. In the FFT based frequency spectrum analysis, the frequency span and frequency resolution are fixed. The capability of discriminating frequency components is equal in both low and high frequency. In rotating machine analysis, a better analysis resolution in the low frequency than that in high frequency is desirable. For example, if the rotating speed is at 60 RPM, the user would be concerned if the instrument can tell the difference between 1 Hz (order 1) and 2 Hz (order 2). On the contrary, if the rotating speed is at 6,000 RPM, the user probably will not be concerned if the instrument can discriminate the measurement between 100 Hz (order 1) and 101 Hz.

With the digital resampling technique, the order tracks and order spectrum are extracted based on a filter with equal ΔOrder instead of equal ΔFrequency. The concept is illustrated in the following diagram:



In the diagram above, the left side shows when the order tracks are extracted using the conventional FFT method with fixed resolution, the ΔFrequency of tracking filter will be fixed. The right side of the diagram shows that if the order tracks are extracted using digital resampling, the ΔFrequency tracking filter will be increased proportionally with the RPM. Obviously, the method of digital resampling is more desirable to extract order measurements.

RPM Frequency Spectrum

While the order tracks and order spectra are developed to analyze the characteristics of the system on the order space, the measures of fixed bands are also helpful for analysis. Similar to the RMS time trace for a given frequency band with time as a variable, the RMS trace can be extracted for a given frequency band with the RPM as an independent variable. This is simply called an RPM Spectrum.

Overall Level Measurement

In order tracking, it is important to monitor the overall RMS level or power level of the measurement versus RPM. The overall level is a good reference for comparison with other signals such as order tracks or fixed band RPM spectrum.

Overall level can be in unit of RMS (EU_{rms}) or power (EU_{rms}^2). The horizontal axis is RPM.

Order Tracks with Phase

Phase in Rotating Machine Analysis

Many mechanical faults are associated with certain orders. Analyzing order magnitude and phase can help users detect mechanical faults. For example, a strong first order magnitude indicates imbalance in most cases. Analyzing the first order magnitude can help users identify the imbalance. Moreover, the magnitude and phase of the first order can help users correct the imbalance by adding weights on the appropriate rotor positions. However, to fix such an imbalance problem requires phase information of order tracks. A list of vibration sources from the rotating machine is provided in the following table.

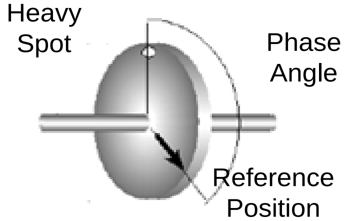
Order	Source of Problem
0.05X~0.35X	Diffuser Stall
0.43X~0.49X	Instability
0.5X	Rubbing
0.65X~0.95X	Impeller Stall
1X	Imbalance
1X+2X	Misalignment
(#Vane)X	Vane/Volute gap
(#Blades)X	Blade/Diffuser Gap

As previously discussed, an order track is the measurement taken for an order, i.e., normalized frequency, versus RPM. In most of applications of engine related tests, the phase information of order tracks is not important. In rotating machine analysis, the phase of the signal is vitally important.

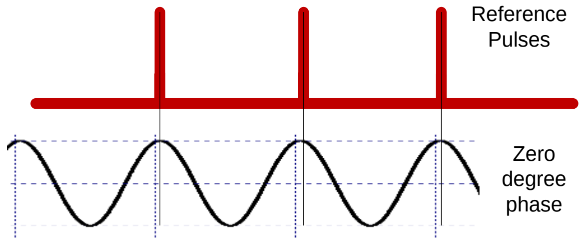
Phase is a relative measurement quantity and can only be measured with a pair of signals. It indicates the time delay at a certain frequency between two signals. The phase value can be translated into the difference of relative angle, relative position, or propagation time if additional information is given. When we refer to the phase information of one signal, we imply its phase is relative to a reference signal that was mentioned in context.

In rotating machine analysis, the phase of the first order of the rotor can be directly mapped to an angular difference between a signal and the reference signal. The reference signal can be another channel of measurement, or the tachometer signal. The phase difference between two waveforms is often called a phase shift or phase delay. A phase shift of 360 degrees is a time delay of one cycle, or one period of the wave, which actually amounts to no phase shift at all. A phase shift of 90 degrees is a shift of 1/4 of the period of the wave. The phase shift can be considered positive or negative, one waveform can be delayed relative to another one, or one waveform can be advanced relative to another one. These conditions are called phase lag and phase lead respectively.

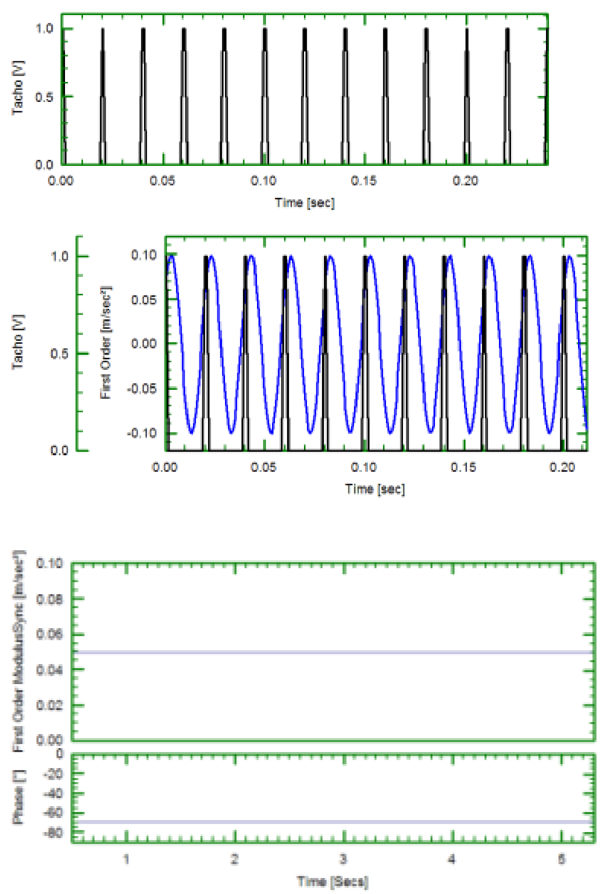
An example of this is the phase of an imbalance component in a rotor with reference to a fixed point on the rotor, such as a keyway. To measure this phase, a trigger-pulse must be generated from a certain reference point on the shaft. This trigger can be generated by a tachometer or some type of optical or magnetic probe that senses a discontinuity on the rotor and is sometimes called a "tach" pulse.



A zero-degree phase delay at a frequency can be depicted as a series of pulses overlaid with a sine wave where the pulse edge is exactly located in peak position of the sine wave.



In the figure above, a section of the tacho signal is shown on its own and then overlaid on the vibration signal. The tacho signal in this example crosses the vibration signal at exactly the same point on each cycle. If the phase of the vibration signal were to change, then its position relative to the tacho pulse would also change. Extracting the first order modulus and phase, as before, gives the curves shown in Figure Y. The phase is now constant near -60^o as it should be for such a signal. The rotating period of the signal is about 20 ms, which results in -60^o corresponding to a 20*60/360=3.3 ms delay.

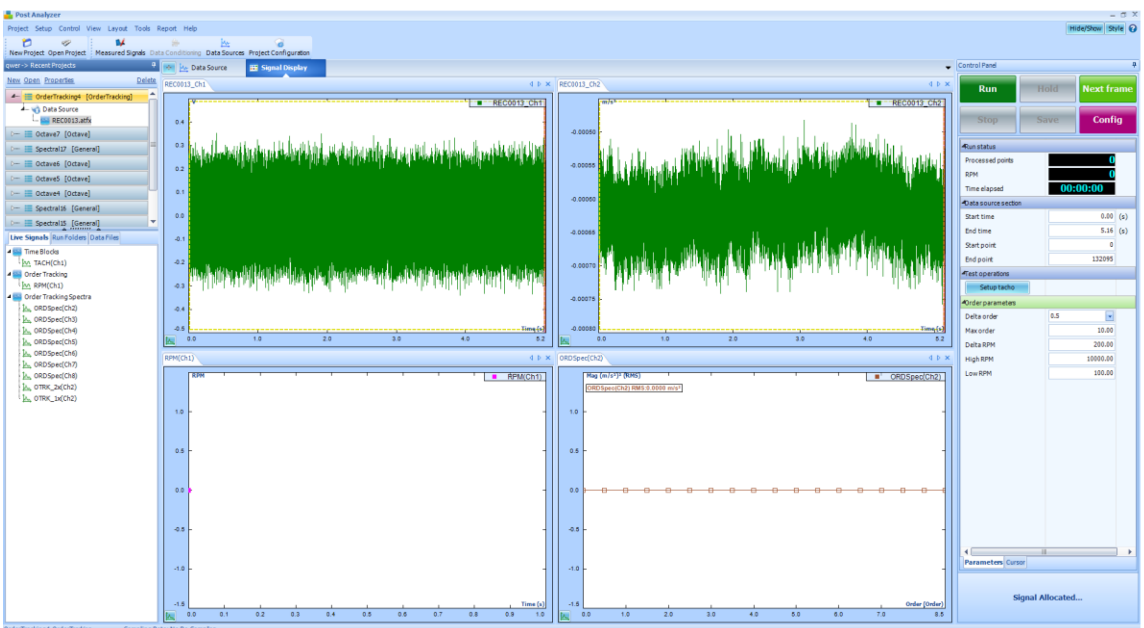


The phase measurement at higher orders will have a similar physical interpretation although they are difficult to comprehend intuitively.

It must be noted that the order tracks with phase, or Complex Order Tracks by name, are not regular complex signals as frequency response or cross spectrum. They are really auto spectra with assigned phase. These synthesized signals can certainly be viewed as complex signal using tools including bode plot, polar and orbit diagram. However, the user must keep in mind that the magnitude and phase of a complex order track are calculated separately.

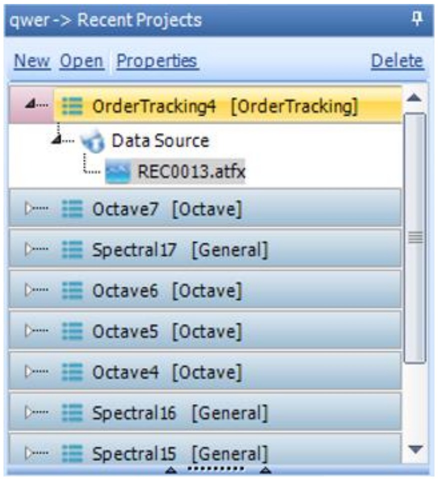
Overview

After creating or opening a project, the main Post Analyzer window is displayed. There is a toolbar across the top, and three main sections in the middle. The left side has **Recent Projects**, **Live Signals**, and **Saved Files** views; in the middle are one or more **Signal Display** views; and on the right is the **Control Panel** which also contains the analysis parameters.



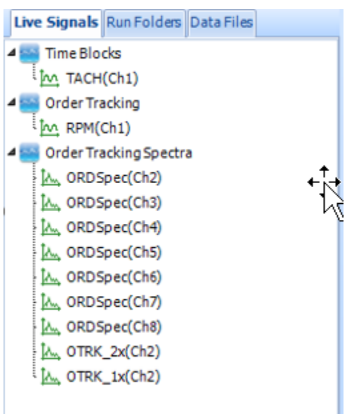
Recent Projects

The Recent Projects list shows the currently active project and previously opened projects. The current project displays a shortcut toolbar to access the **Measured Signals** setup, **Data Sources** setup, and **Project Configuration** window. It also displays the data file associated with the project.



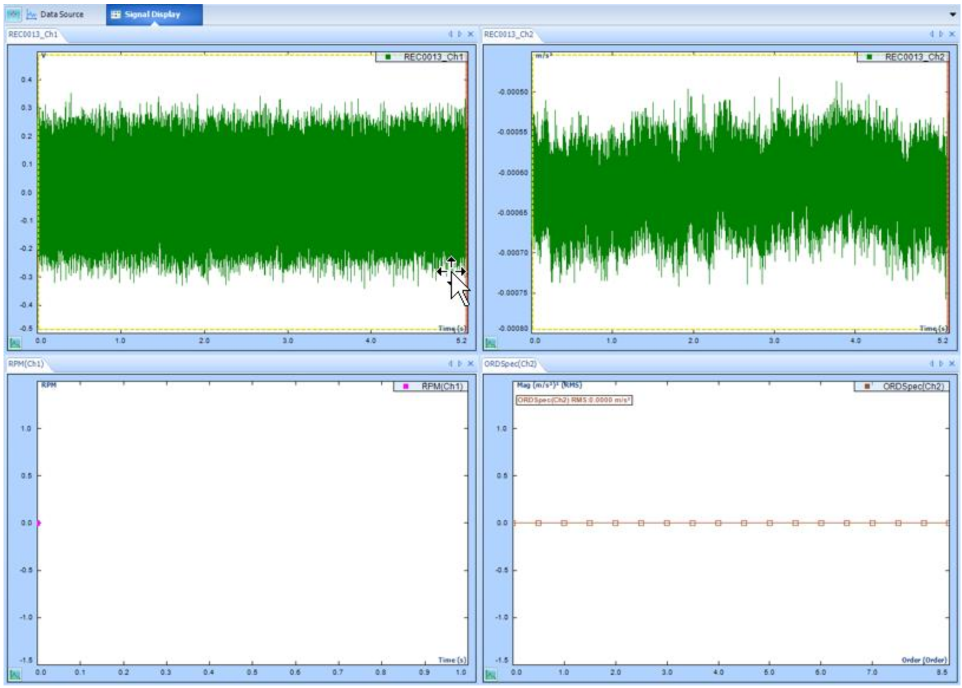
Live Signals, Run Folder, and Data Files

Below the Recent Projects list is the **Live Signals, Run Folder, and Source Files** list. Live Signals includes the signals in the Data Source file or files, as well as new signals created as part of the analysis. The live signals are not saved to the host computer hard disk but are available for display in the **Signal Display** view tab. Anything saved to disk will appear in the Saved Files list, which is associated with a directory in the file system. Run Folders contain data from each run. Data is saved in a file format under the Data Files tab.



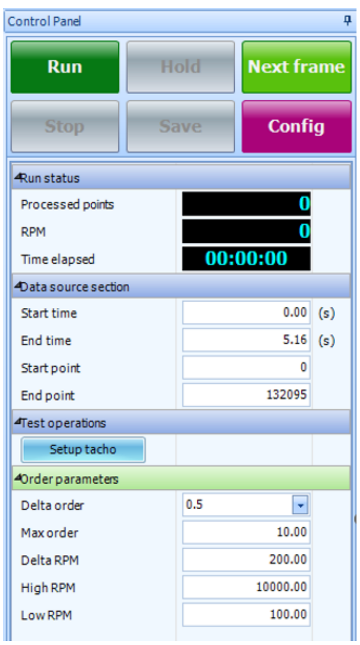
Signal Display

The middle of the main window contains the **Signal Display** view tabs, where live and saved data can be displayed. More than one of these tabs can be created, but there is always at least one. Each of these views can contain one or more **Display Windows** with a fully customizable layout. These display windows can move freely, be resized, and display any valid combination of live or saved signals. Create new view windows by selecting an option from the View menu.



Control Panel and Analysis Parameters

The right side of the window contains the **Control Panel** (which controls post-analysis operations) and the **Analysis Parameters** (which contains setup parameters specific to the analysis type being performed). The **Config** button on the control panel opens the **Project Configuration** window, which is described in the following section.

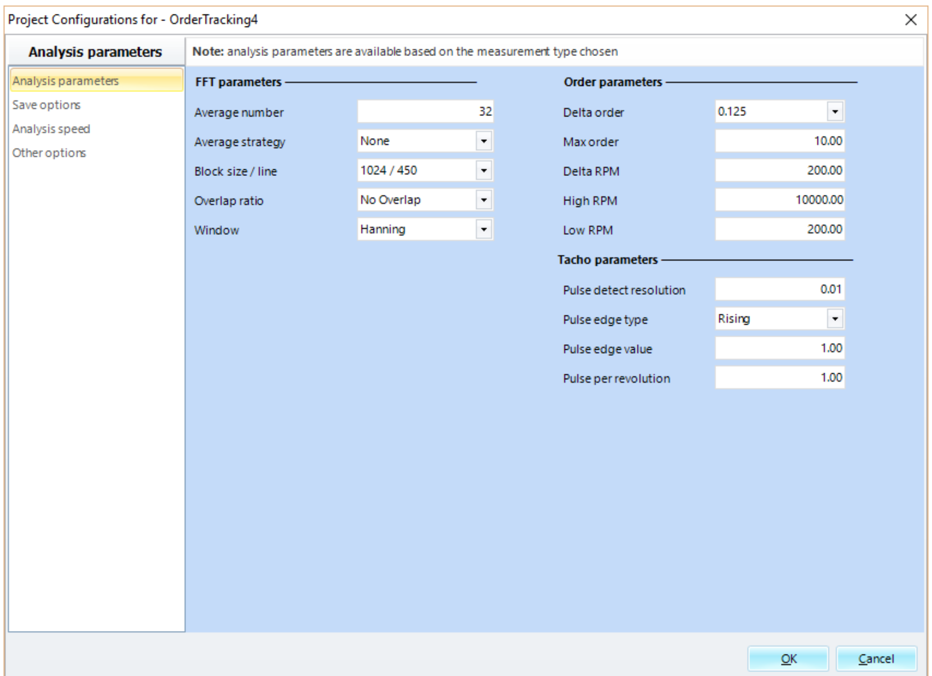


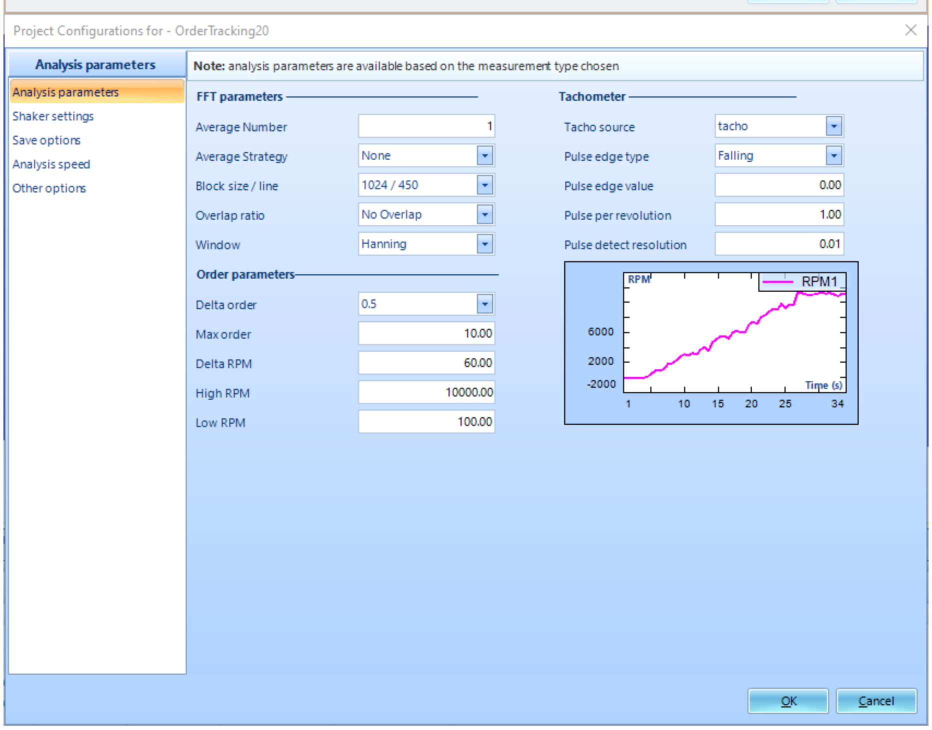
Project Configuration

The **Project Configuration** window has settings related to analysis parameters, shaker settings, saving data, analysis speed, and other options that are mostly independent of the type of analysis being performed.

Analysis Parameters

Analysis parameters contains FFT, Order, and Tacho parameters.





FFT Parameters

Average Number is the number of blocks that are averaged for the signal spectrum. Increasing the number of averages reduces the variance of the signal spectrum.

Average Strategy provides options such as None, Exponential, Linear and PeakHold as methods to average the spectrum signals.

Block Size/Line defines the size of the time blocks in terms of the number of samples, which the FFT algorithm transforms to a number of spectral lines in the frequency domain. Increasing the block size increases the resolution of the frequency transform and allows lower frequencies to be detected. However, this also increases the calculation time and slows down the response.

The ratio between lines and block size is determined by the characteristics of the A/D converter and anti-aliasing filter. In general, the ratio is about 0.439 which means that a time block of 1,024 samples results in about 0.439 * 1024 = 450 lines of spectrum.

Overlap Ratio sets the proportion of the samples in the time blocks that are overlapped when calculating the FFT. Higher overlap ratios result in faster response times but increases processing requirements. The Overlap ratio options are: no overlap, 10%, 25%, 40%, 50%, 75%, 80%, 87.5%, 90% and 95%.

Window allows users to select a window to apply during FFT operation. Windowing functions can help reduce leakage and increase the precision of the frequency measurement. Detailed descriptions about window types and average modes are provided in the "DSA Basics" document.

Order Parameters

Delta Order: The Delta Order defines the resolution of the order spectra. The Max Order and Delta Order together define the number of points in a normalized order spectrum. Users should define the minimum delta order required for an application to conserve computation resources.

Max Order: The Max Order defines the highest order number for an order spectrum. The software uses this value to determine the analysis frequency range. Users should define the minimum Max Order an application requires. If the Max Order is defined as a very high value, the system must sample at a very high frequency to cover the entire frequency range, which may cause poor accuracy of lower order estimations.

Delta RPM: The Delta RPM defines the resolution of the RPM trace or the resolution of the waterfall in the RPM axis. The higher the Delta RPM, the finer the signals will be when stored and displayed in the RPM axis. More storage will be required. The Delta RPM is typically chosen between 25 and 100.

High RPM: The High RPM defines the upper edge of RPM for any order signals, RPM waterfall, or RPM traces to be analyzed. If the software detects the current RPM is between the Low and High RPM, it will take the measurement and display it. Otherwise, the system will display the status as RPM High or RPM Low, and the required signals will not be computed or displayed.

Low RPM: The Low RPM defines the lower edge of RPM for any order signals, RPM waterfall or RPM traces to be analyzed. If the software detects the current RPM is between the Low and High RPM, it will take the measurement and display it. Otherwise, the system will display the status as RPM High or RPM Low, and the required signals will not be computed or displayed.

Tacho Parameters

Tacho Source defines the channel to be referenced as the tachometer. Additionally, the Setup tacho is used to define the pulse edge type, pulse edge value, and pulses per revolution.

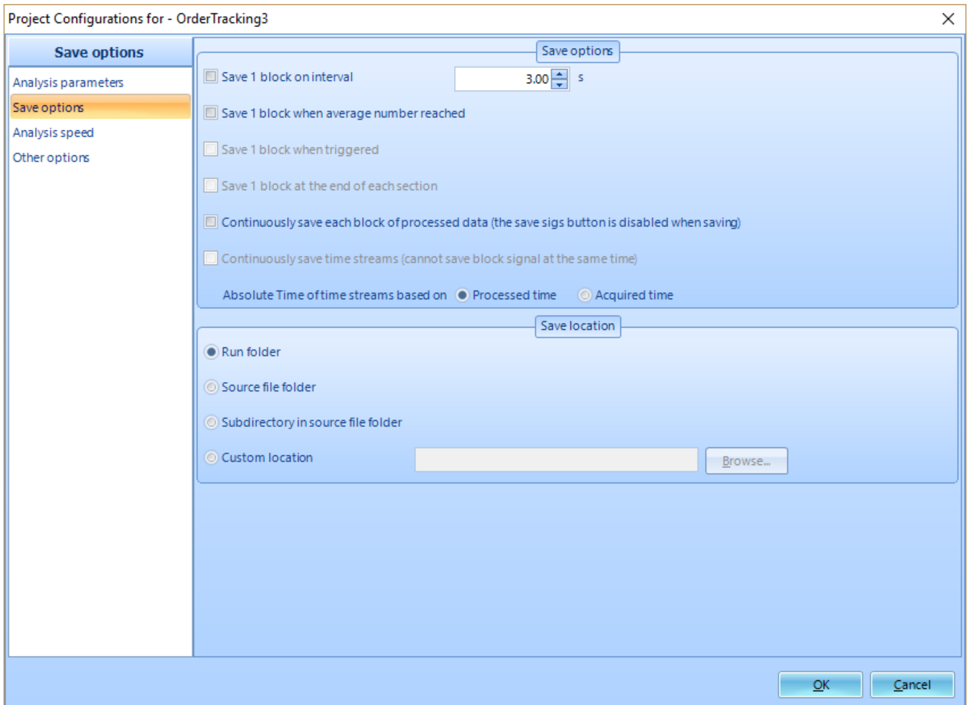
Pulse Detection Resolution is defined as the number of pulses per revolution. Pulse per revolution must be defined by the user so that the software can calculate the reference frequency of tacho using tacho frequency. In most rotor tests, especially in balancing, the pulse per revolution is simply 1. However, in other cases, such as in flywheel or geared data measurement, the pulses per revolution can be as high as hundreds. To address this situation, a dedicated tacho channel with a high-speed counter might work more effectively.

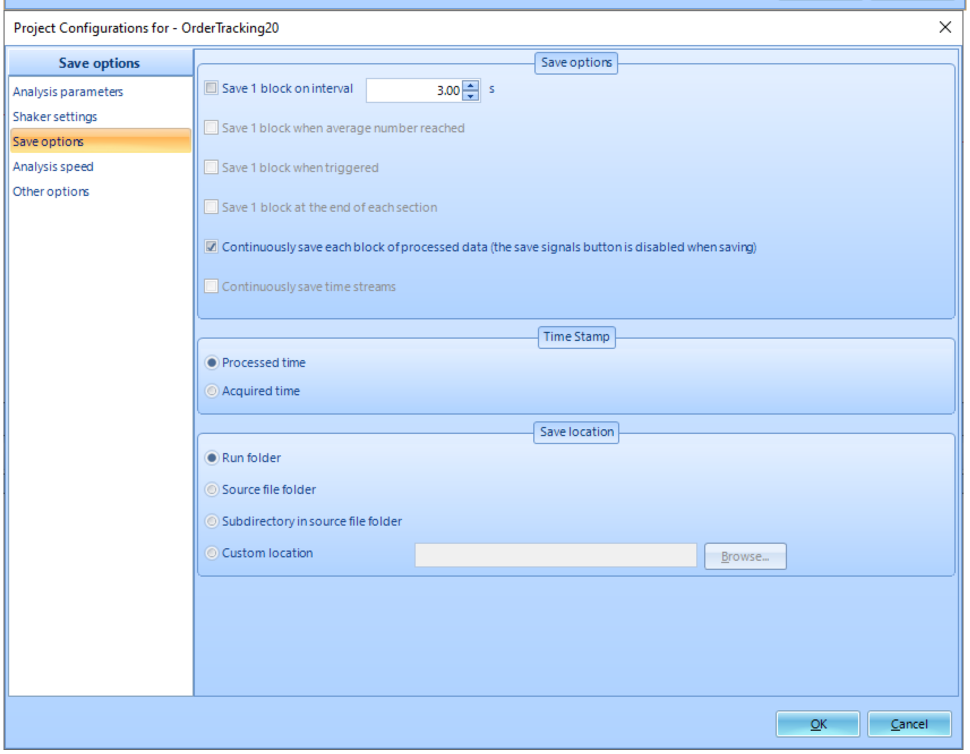
Pulse Edge Type controls the pulse trigger edge conditions. Rising sets the trigger condition to a rising slope. Falling sets the trigger condition to a falling slope.

- **Pulse Edge Value** defines the level of edge lines.
- **Pulse per Revolution** is defined as the number of pulses per revolution.

Save Options

After running the analysis, computed signals are listed under the **Live Signals** tab. However, this data is only stored in a display buffer and must be explicitly saved to disk to create new data files. There are multiple ways to do this. One method is to manually click the **Save** button on the control panel, which saves data according to the Save settings specified under the Measured Signals setup. The other method is to enable an option (listed in the following section) to automatically save data during the analysis.





There are many options to save block data, and one option to save stream data. Most data computed in Post Analyzer is block data, such as APS and SRS spectra. Some data, such as the output of digital filters or other data conditioning blocks, are continuous time streams. Only one type of data can be saved at a time – block or time stream.

Save 1 Block on Interval saves one block of data for every computed signal with the save option selected in the Measured Signals setup per the interval selected here.

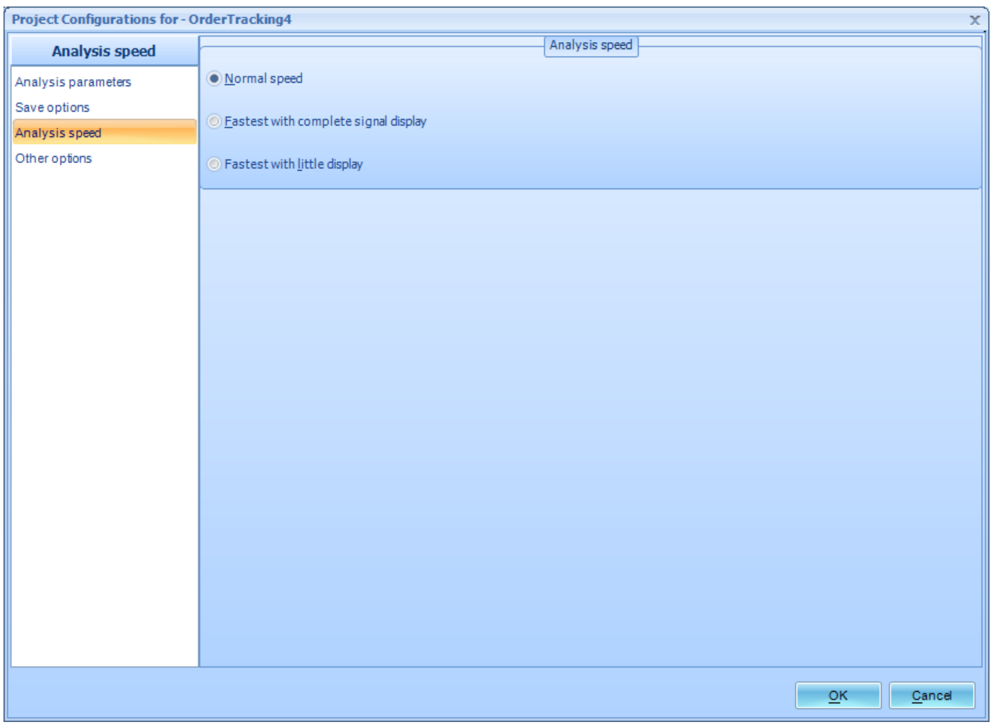
Save 1 Block when Average Number Reached saves one block of data every time the number of processed blocks reaches the average number set under the Analysis Parameters.

Continuously Save Each Block of Processed Data saves *every* block of data computed.

Absolute Time of time streams can be based on Processed time or data Acquired time.

The **Save Location** section allows users to select the location of the file system used to create new data files from the following options: **current run folder, same folder as data source file folder, subdirectory in source file folder,** or a **custom location**.

Analysis Speed

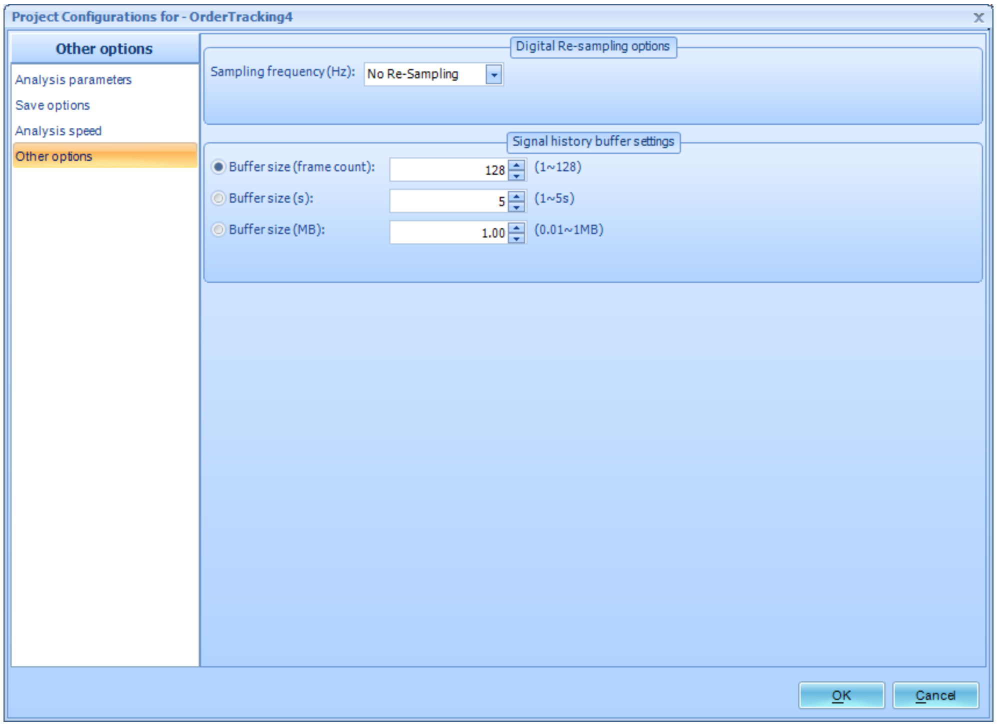


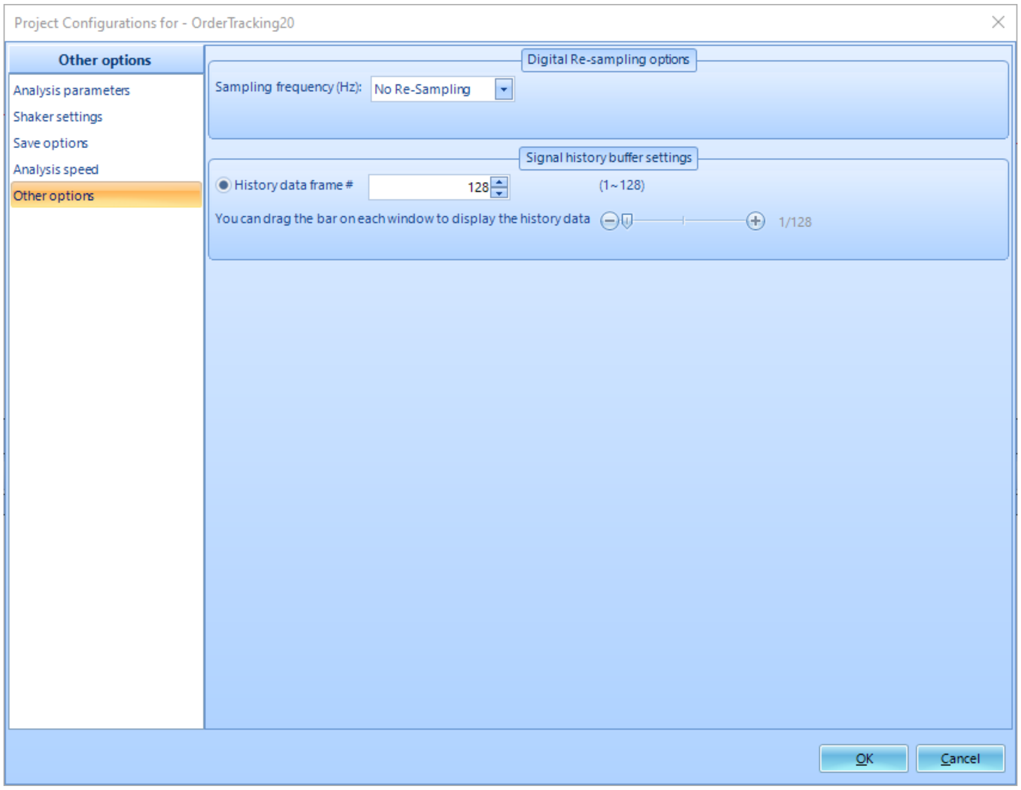
Analysis Speed adjusts the amount of processing time used for the signal display during post-processing. Increased display details result in less available processing resources for post processing data analysis. **Normal Speed** is the default setting that balances data processing and the signal display. **Fastest with Complete Signal Display** prioritizes the display, and **Fastest with Little Display** prioritizes data processing.

Other Options

Digital Resampling allows resampling of all source signals to the selected sampling rate to meet the requirement of the other interface. Many sampling rate stages can be selected from the dropdown menu.

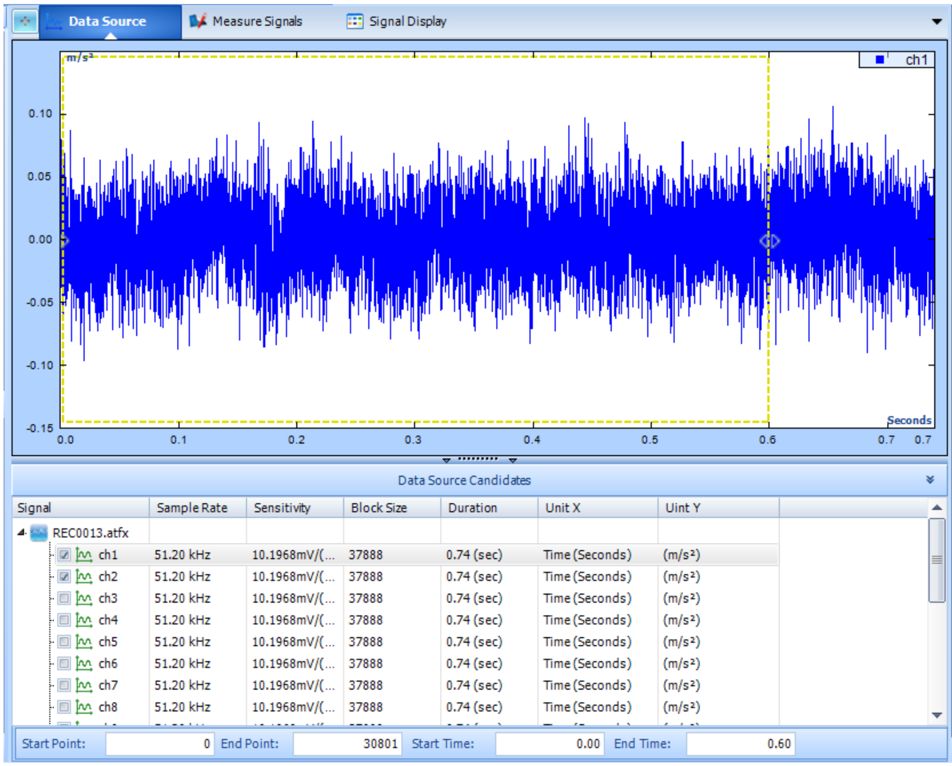
Signal History Data Buffer sets the size of the buffer used to store and display the signals under the Live Signals tab. This buffer can be sized according to a number of frames. If the buffer fills up during the analysis, old data will be discarded. This does not affect data already saved to disk.





Data Source

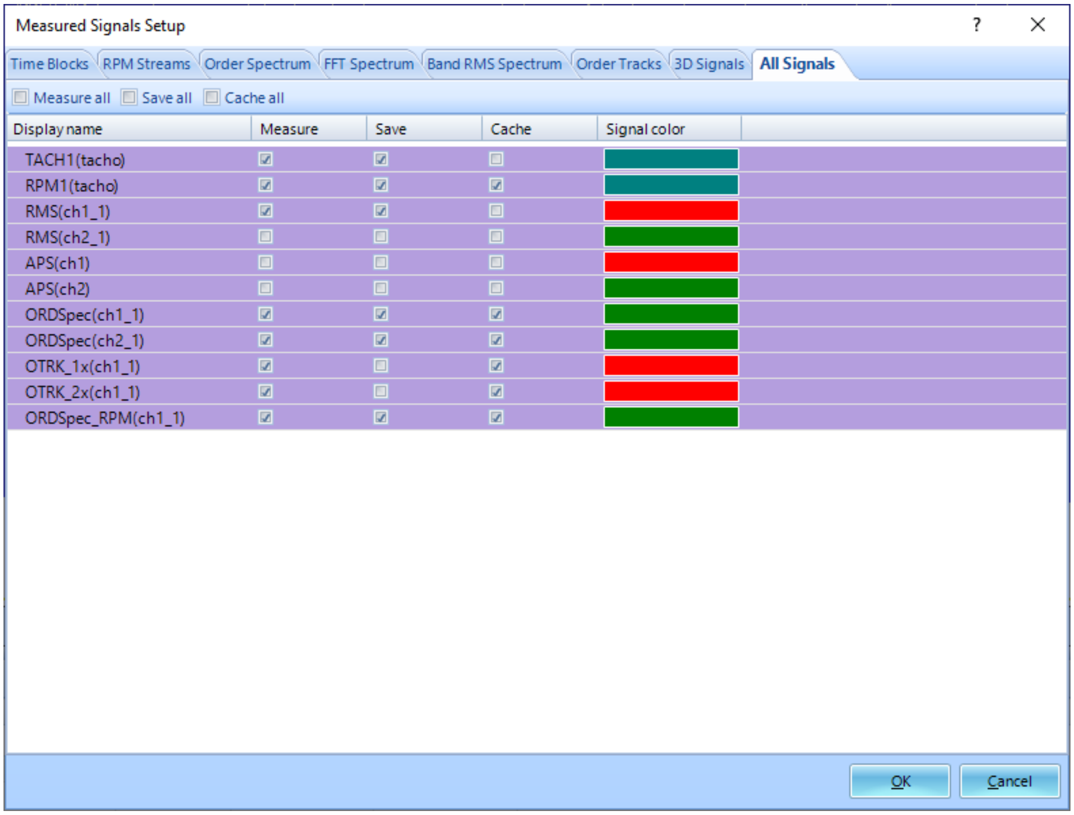
The Data Source tab displays the source data from the data file or files associated with the current project. Enable or disable source signals here and set the time range for analysis by dragging the edges of the yellow box. Enter the start and stop time (or point) manually in the text fields below.



Measured Signals

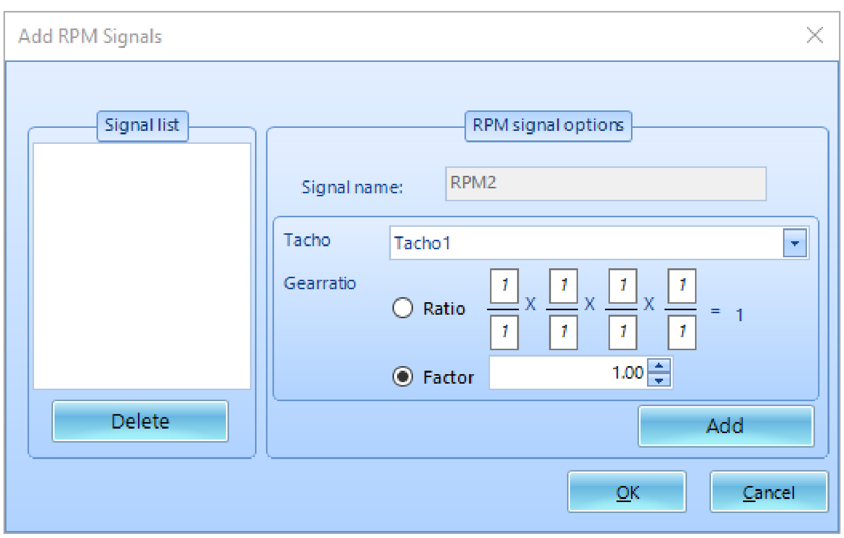
The different signal types are listed under the **Setup->Measured Signals** tab.

Each signal has a **Measure** and **Save** option. Signals with the **Measure** option selected are listed under the **Live Signals** tab and will be available for display. Signals with the **Save** option selected are saved to disk whenever the Save button is clicked in the Control Panel, or when an automatic save option is selected in the Project Configuration window.

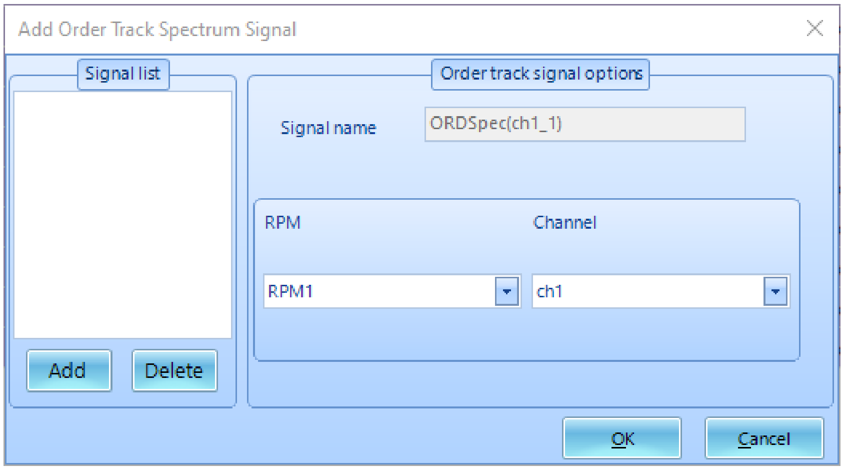


Time Blocks – The signal used as the tachometer is listed here.

RPM Streams – Signals based on the tachometer channel can be added here. The RPM signals take a gear ratio into account.

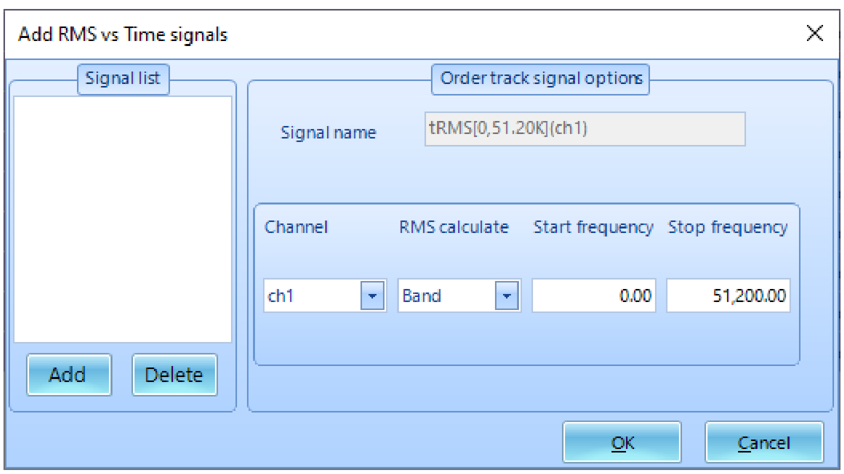


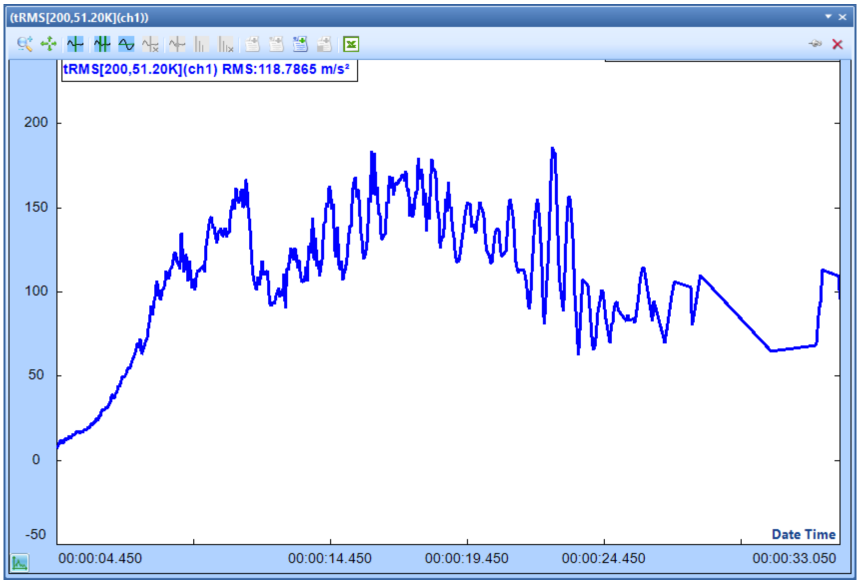
Order Spectrum – The signals here are tracked against one of the RPM signals.

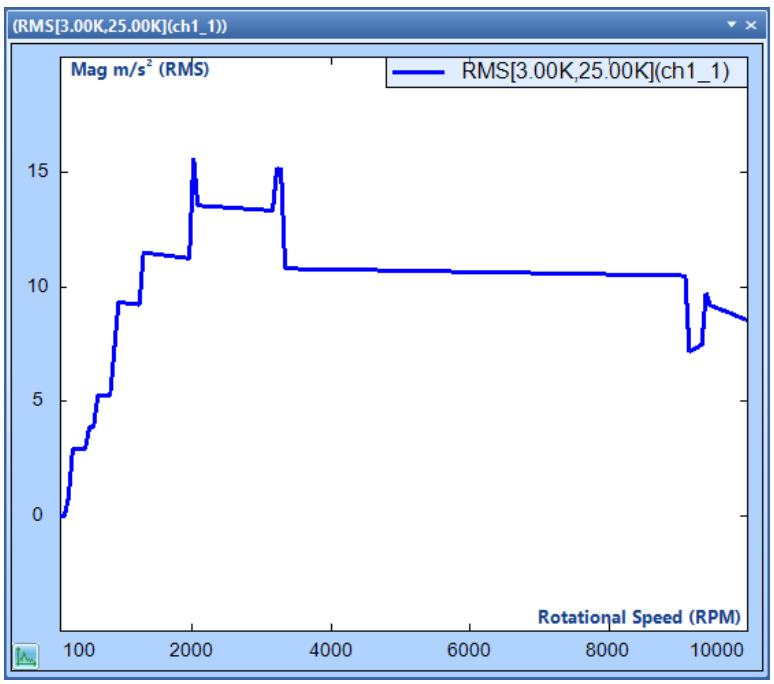


FFT Spectrum – Auto-Power Spectra signals are listed here.

Band RMS Spectrum – The root mean square is calculated in a block-by-block operation. The number of data points used is based on the block size/spectral lines. The RMS signals can reference either time or RPM. Signals based on RPM may be filtered before the RMS calculations are performed.

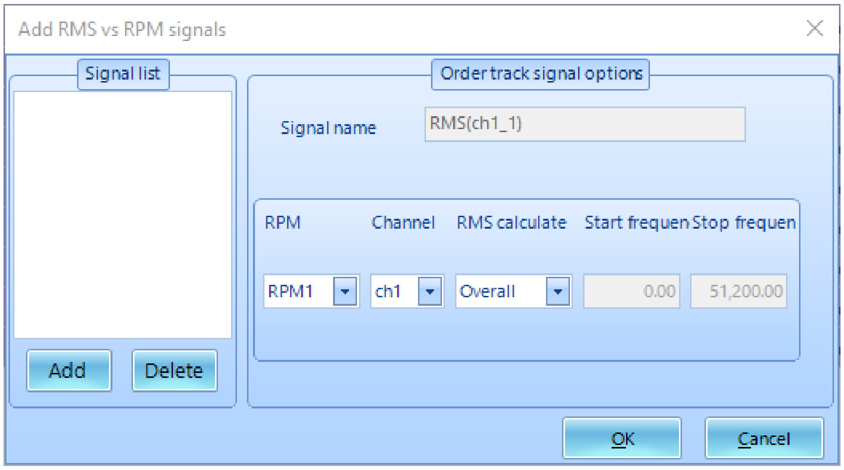






Order Tracks – These signals track the magnitude of an order against either RPM or time. Additionally, the order phase can also be measured.

3D Signals – APS and Order spectrum signals are referenced against frequency and order, respectively. 3D signals allow for the addition of a third axis of measurement, either time or RPM.



All Signals - This includes the output from every Data Conditioning module used as well as block data computed from these such as Auto-Power Spectra (APS).

Control Panel Functions

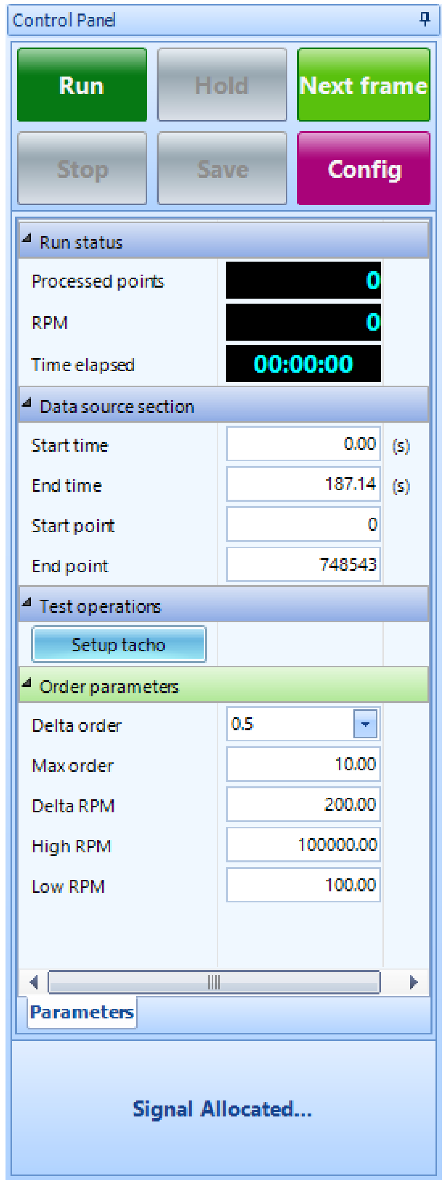
The **Control Panel** is used to control post processing operations. It also displays the analysis status information.

The following command buttons are provided:

- **Run** starts the analysis.
- **Hold/Continue** pauses and resumes the analysis.
- **Stop** stops the analysis.
- **Next Frame** processes one frame, and then pauses.
- **Save** saves block signals.
- **Config** opens the Project Configuration window

Run Status panel displays the following information:

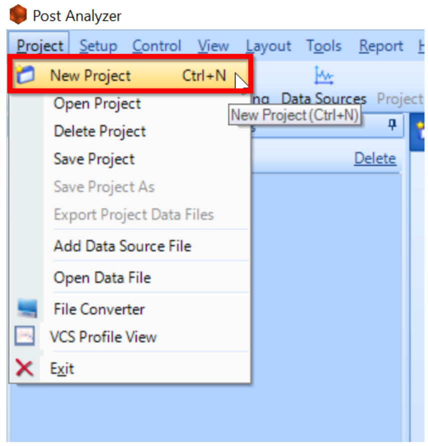
- **Processed Points** # is the number of processed samples.
- **RPM** is the speed of the current frame under processing.
- **Time Elapsed** is the total time duration of the input signals processed.



General Process for Conducting an Order Tracking Project

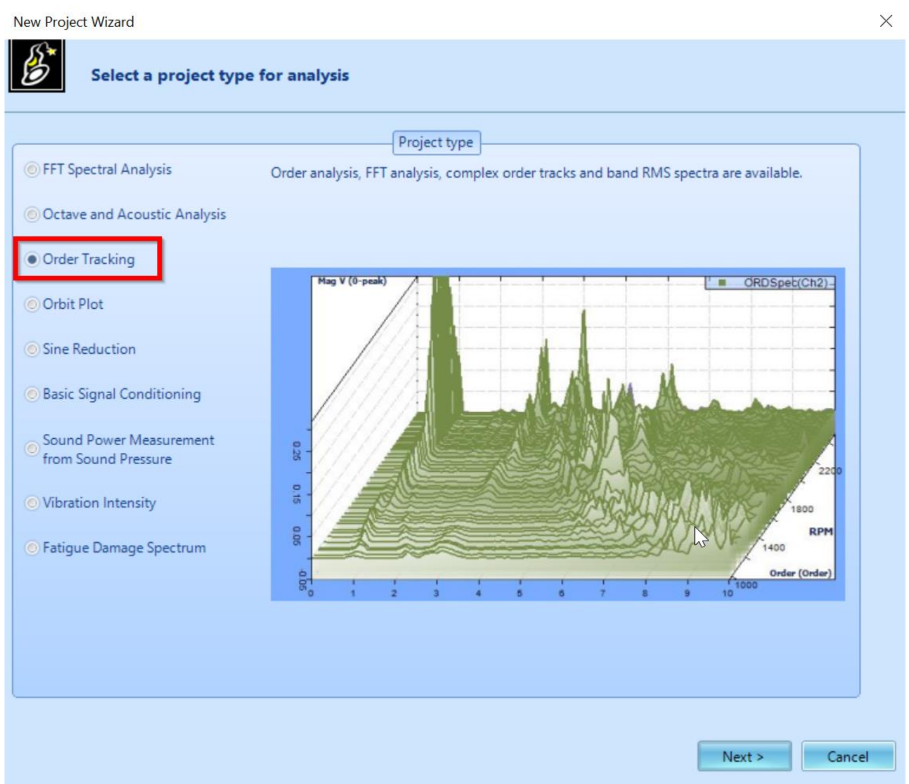
Create a new project

Locate and select **New Project** within the **Project** tab after opening the Post Analyzer software. This tab is located in the top left corner of the program.



Select the 'Order Tracking'

Select the **Order Tracking** project type in the **New Project Wizard** window and click **Next**.



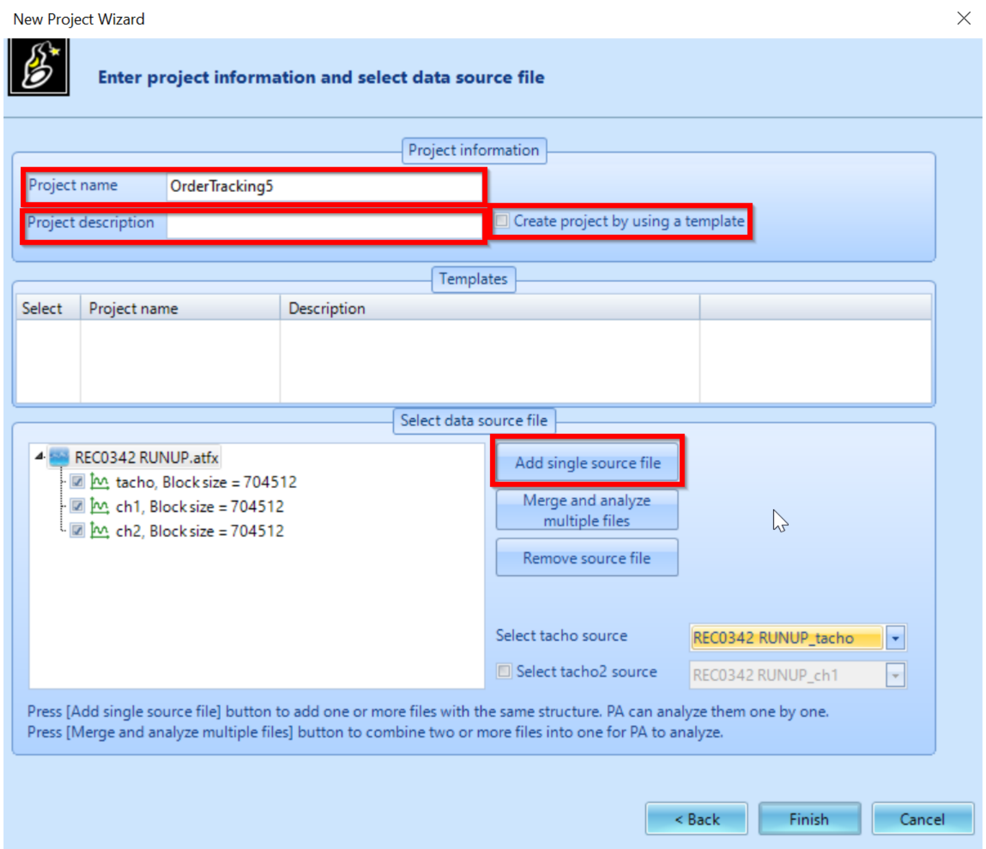
Enter Project Name, Description and Signal Source

This page allows users to add a project name and description. An option to create a project using a template is available if a template file exists.

Under **Select data source file** select **Add single source file**. Select the desired file and click open.

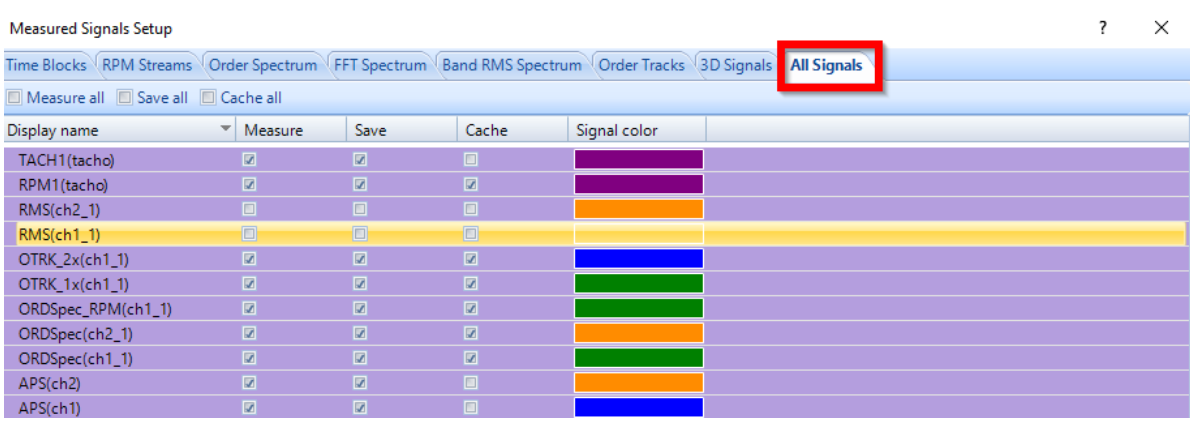
The file with associated signals will appear in the display box as seen in the following figure.

Click the **Finish** button.



Measured Signals Setup

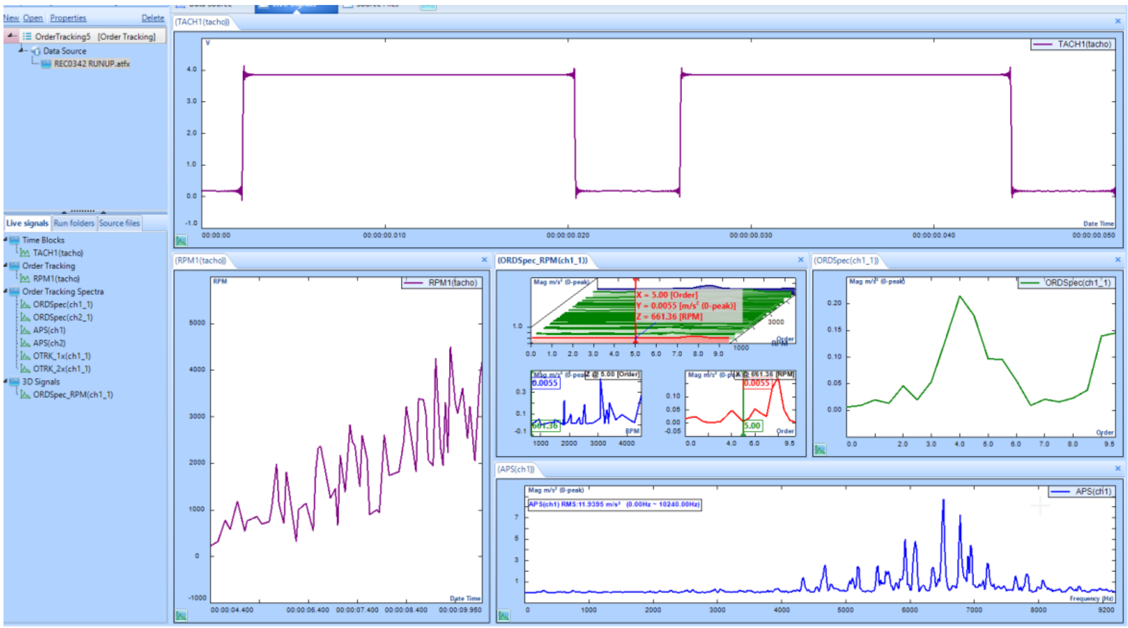
Locate the **Measured Signals** section in the top left ribbon portion and navigate to the **All Signals** tab to select signals to measure. A selection of signals to save and a cache is also available.



Setup Signal Display

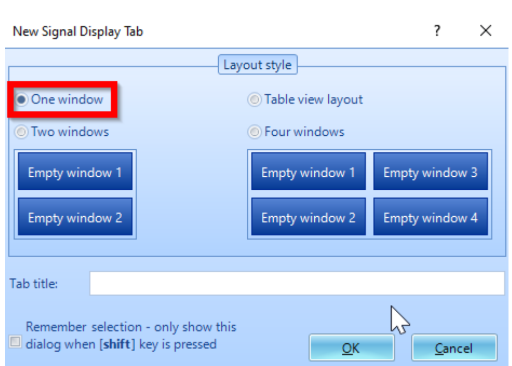
Double-click to select signals from the **Live Signals** pane, then drag and drop into the blank window. The signals can include but are not limited to the following:

- Tacho time block
- RPM order tracking
- Order tracking spectra – ORDSpec and APS
- 3D Signals – ORDSpec_RPM

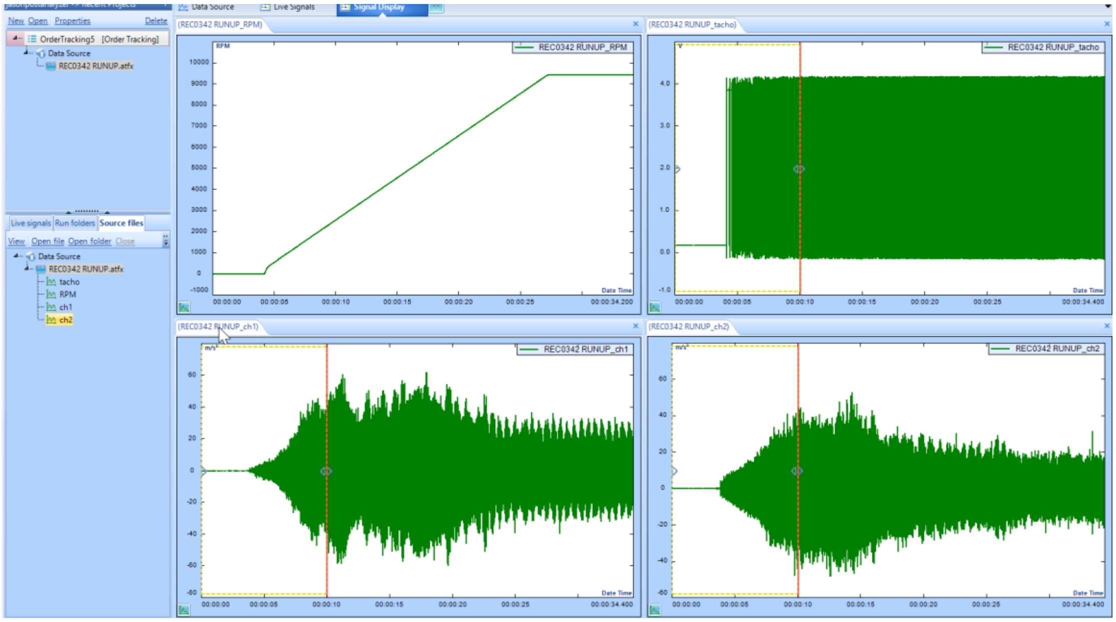


An additional signal display can be created for the imported **source files**. Select the **+** (plus) button in the signal display ribbon.

Select **One window** and click **OK**.

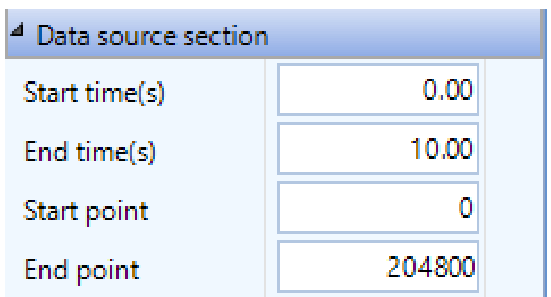


Double-click to select signals from the **source files** pane, then drag and drop into the blank window.



Data Source Section of Control Panel

Input the analysis range of interest. This range may be defined in terms of time or points.



Test Operations Section

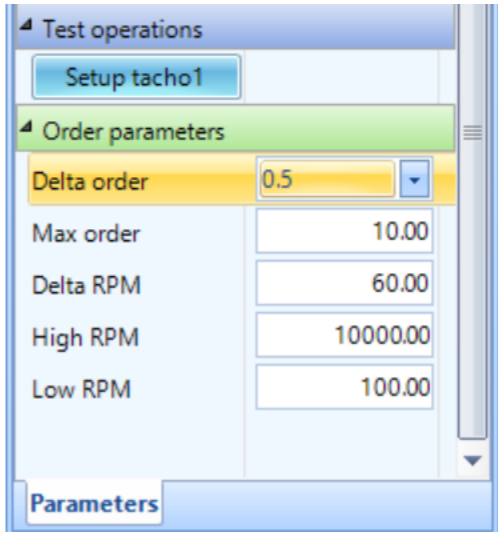
The **Delta Order** defines the resolution of the order spectra. Select and define the minimum Delta Order required for an application to conserve computation resources.

Next, select the **Max Order,** Users should define the minimum Max Order required from an application. If a Max Order value is defined very high, the system must sample at a very high frequency to cover the entire frequency range, which may result in poor accuracy of lower order estimations.

The Max Order and Delta Order combined define the number of points in a normalized order spectrum.

The **Delta RPM** defines the resolution of the RPM trace or the resolution of the waterfall in the RPM axis. The Delta RPM is typically chosen between 25 and 100.

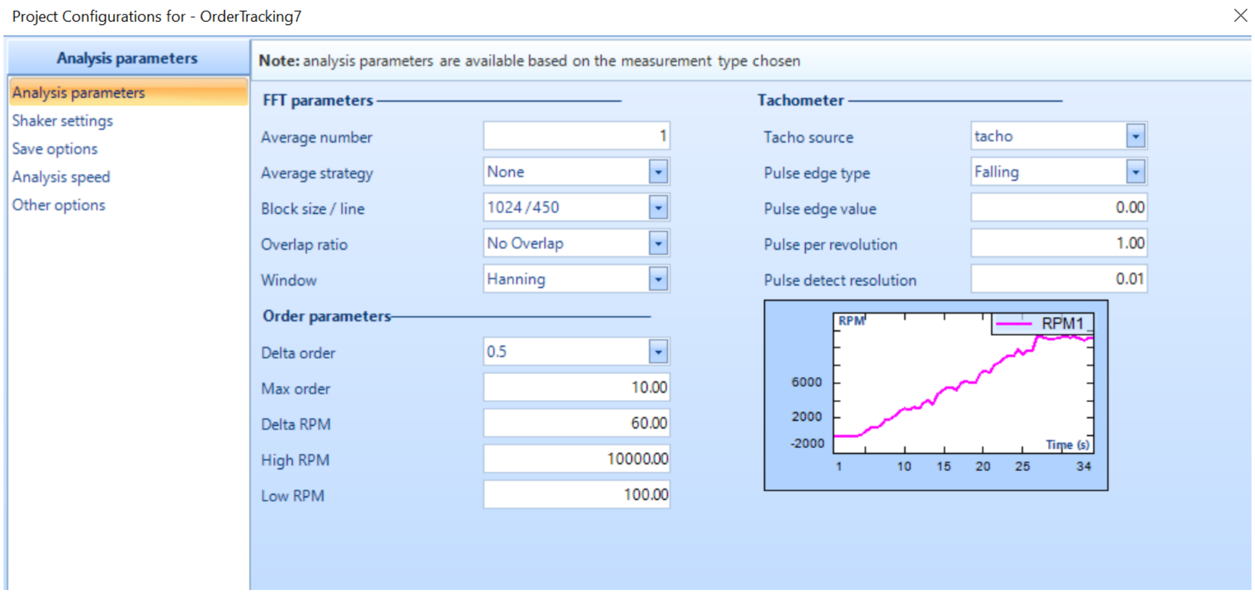
The **High RPM** defines the upper edge of RPM for any order signals, RPM waterfall or RPM traces to be analyzed. Likewise, the **Low RPM** defines the lower edge of RPM for any order signals, RPM waterfall or RPM traces to be analyzed.



Configuration Setup

Locate and click the purple **Config** button in the control panel.

Select the **Analysis Parameters** tab within the Configuration window. Refer to pages 15 – 17 for descriptions of these parameters.



Next, select the **Save Options** tab to define the location to save processed data.

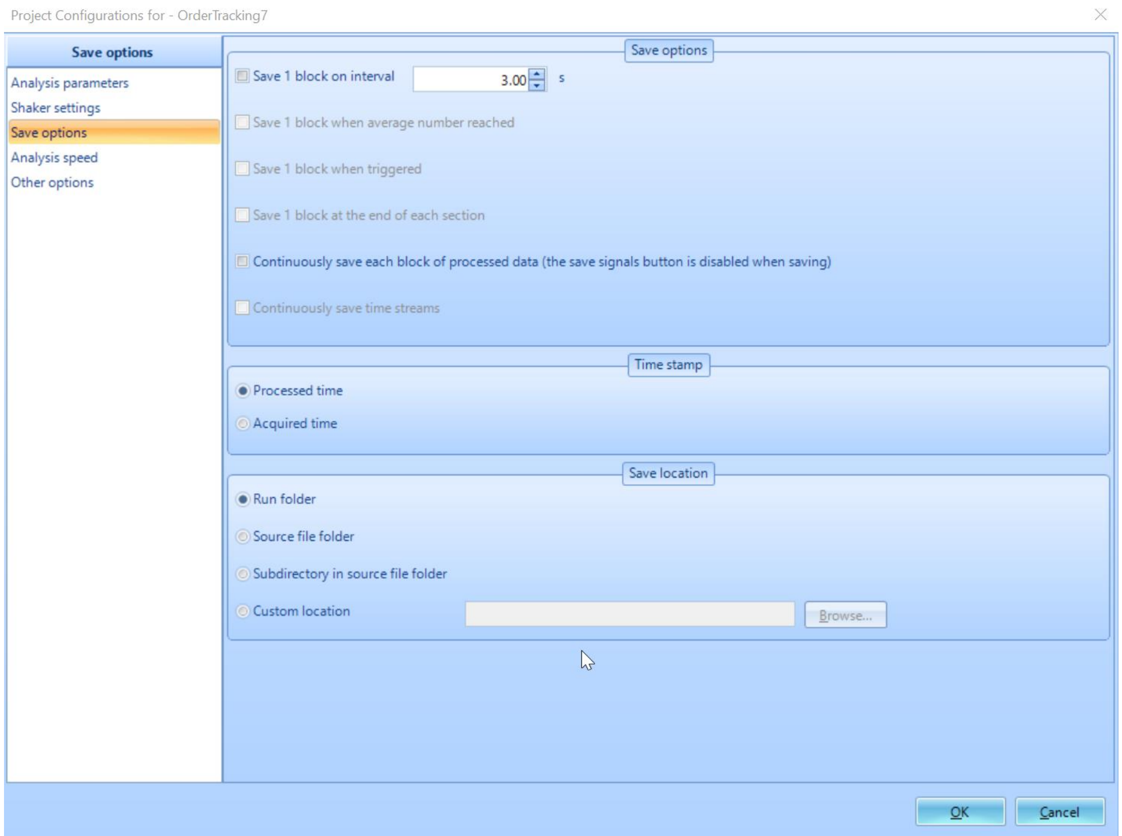
Save 1 Block on Interval saves one block of data for every computed signal with the save option selected in the Measured Signals setup per the selected interval.

Save 1 Block when Average Number Reached saves one block of data every time the number of processed blocks reaches the average number set under the Analysis Parameters.

Continuously Save Each Block of Processed Data saves every block of data computed.

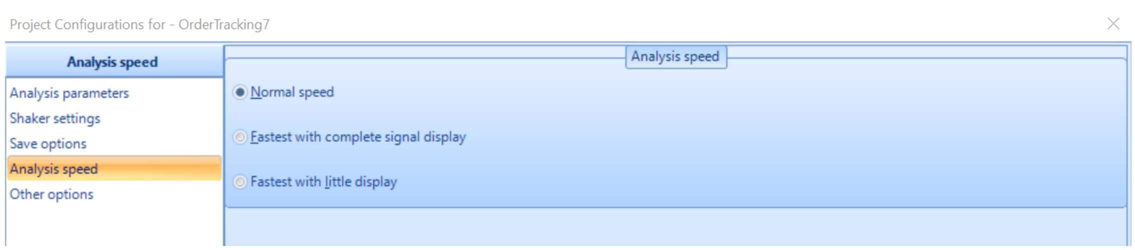
Absolute Time of time streams can be based on either Processed Time or data Acquired Time.

The **Save Location** section allows users to select the location of the file system used to create new data files from the following options: **current run folder, same folder as data source file folder, subdirectory in source file folder,** or a **custom location**.



Proceed to the **Analysis Speed** tab after save options are configured. **Analysis Speed** adjusts how much processing time is used for the signal display during post-processing.

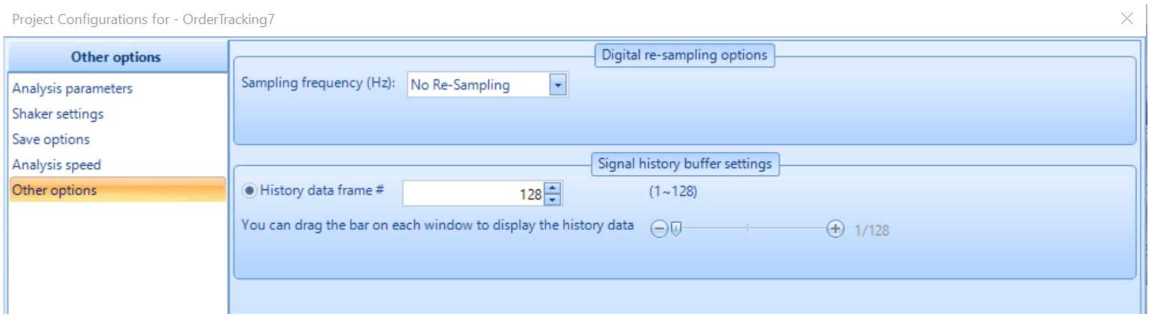
Increased display detail results in less available processing resources for post processing data analysis. **Normal Speed** is the default setting that balances the data processing and signal display. **Fastest with Complete Signal Display** prioritizes the display, and **Fastest with Little Display** prioritizes the data processing.



Select the **Other Options** tab for additional configurations.

Digital Resampling allows resampling of all source signals to the selected sampling rate to meet the requirement of the other interface. Many sampling rate stages are available for selection from the dropdown menu.

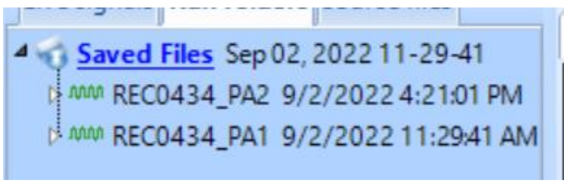
Signal History Data Buffer sets the size of the buffer used to store and display the signals under the Live Signals tab. This buffer can be sized according to a number of frames. Old data is discarded if the buffer fills up during the analysis. This does not affect data already saved to disk.



Run the Project

The last step is to run the project. Click the green **Run** button in the Control Panel.

After the run is completed, the recording will appear in the Run Folder.



Orbit Plot

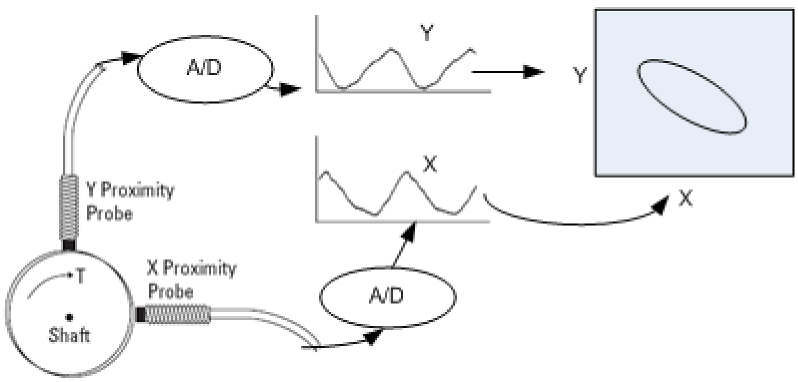
Basics

The **Orbit Display** uses two data channels in the time domain. The signals from two channels are drawn on an X and Y plane to display the shaft position change versus the angle of rotation. Orbit displays provide a two-dimensional visual of motion from a rotating shaft.

A well-balanced shaft without movement in any direction produces a dot in the middle of a plot. The shaft movement can give an indication of the vibration source (i.e., a lot of up/down movement may indicate that the machine feet are not bolted down securely).

Users can create an orbit plot by taking a simultaneous dual channel measurement to capture data at the horizontal and vertical axes at the same time. The displacement or acceleration sensors must be placed 90° away from each other.

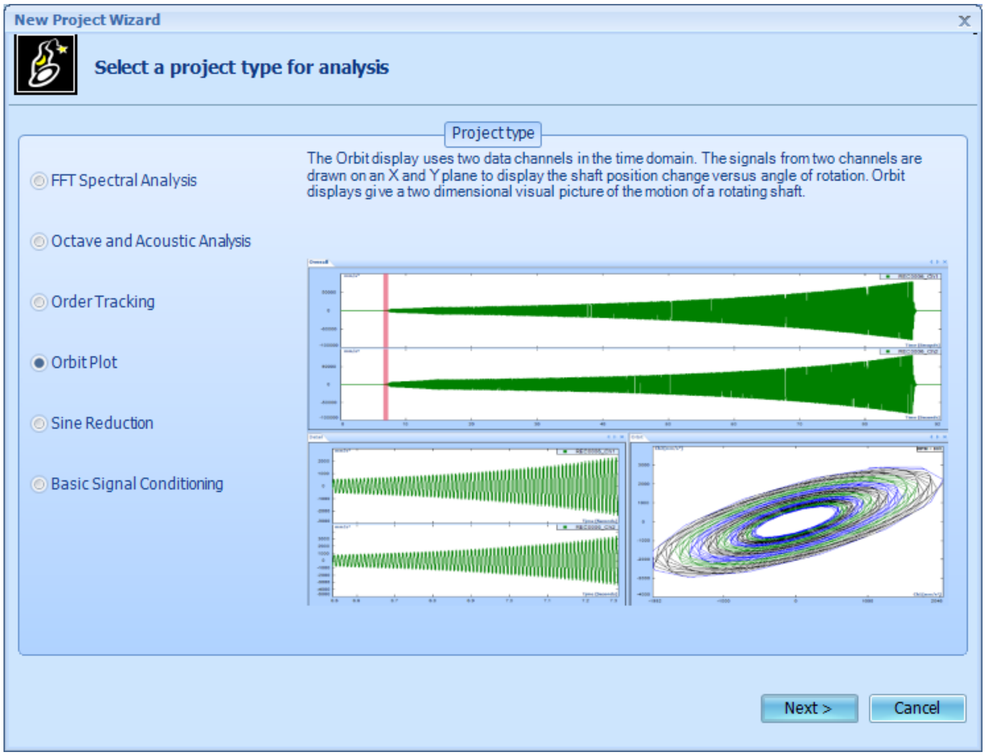
The Orbit Display uses measurement obtained from a pair of channels in the time domain. It does not require Order Tracking.



The Orbit Display is similar to the Polar Display in that it only displays the instantaneous status at the current RPM. In theory, the Orbit Display does not need a tachometer or another reference signal because X and Y reference to each other.

Create an Orbit Plot Project

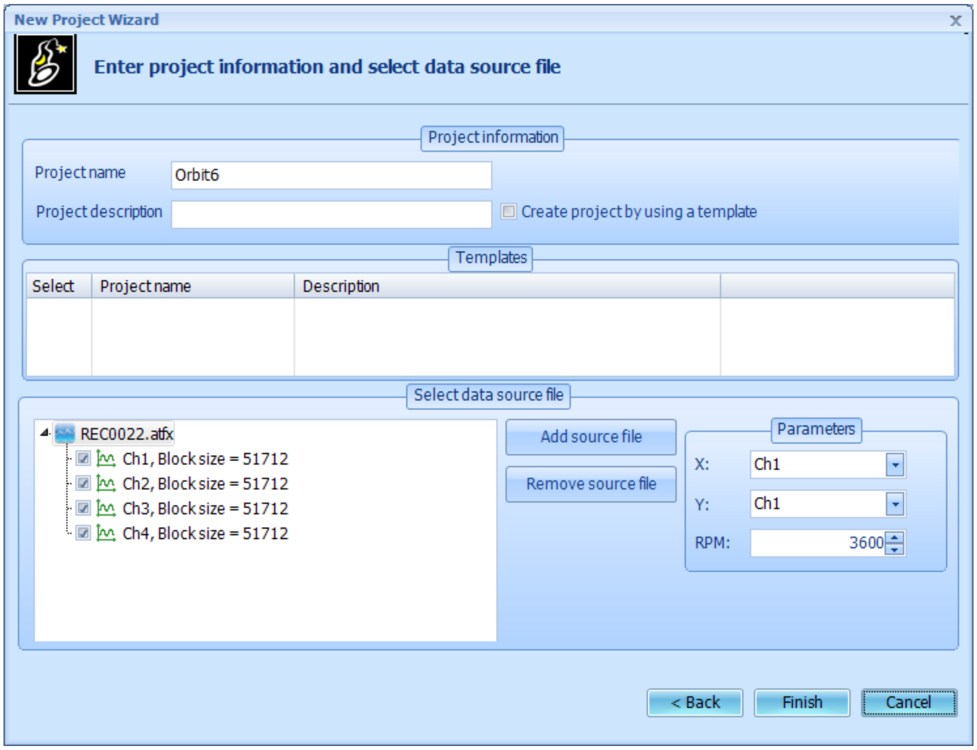
Create a new project by clicking **New Project** and select **Orbit Plot** as the project type. Click **Next** to continue.



In the following window, create a name for the project and optionally add a description. Both data conditioning and digital resampling features are not available in Orbit Plot projects.

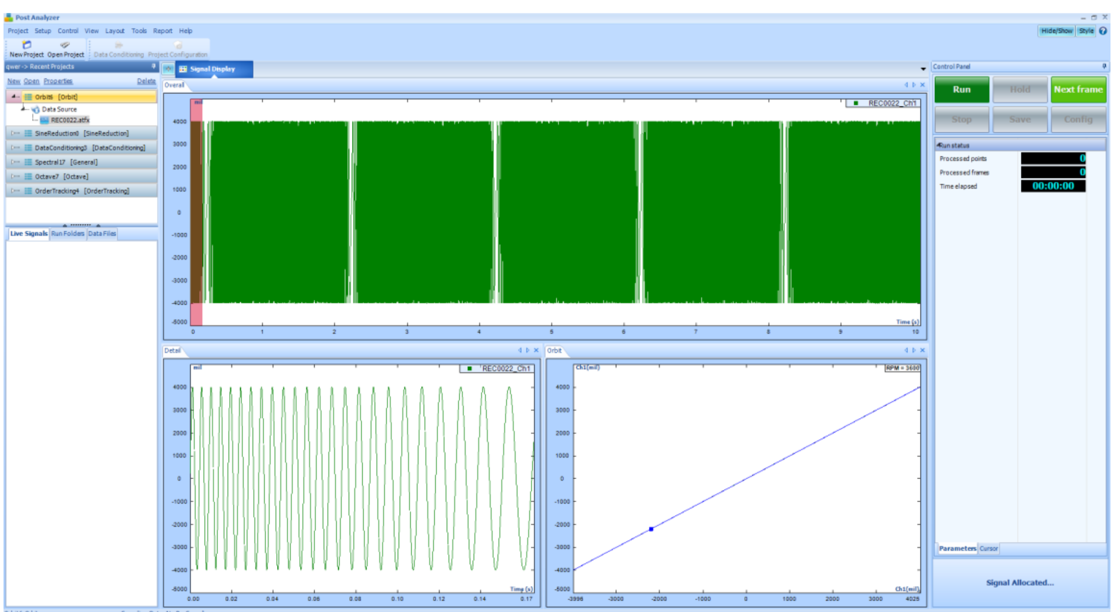
Select the data source files to associate with this project. The source file is usually a time stream recording in ATFX format. The time stream must be recorded in displacement to be validated. Existing source files can be selected for removal from the data source file list by clicking **Remove Source File**.

Set signal sources for the X-Y coordinate plotting. The RPM value is also specified at this step. Then click **Finish** to enter the main page. Only one signal source file can be added as the source file.



Overview

After creating or opening a project, the main Post Analyzer window is displayed. There is a toolbar across the top, and three main sections in the middle. On the left, there are the **Recent Projects**, **Live Signals**, and **Saved Files** views; in the middle are one or more **Signal Display** views; and on the right is the **Control Panel**. Compared with other project types, the Orbit Plot project does not require the analysis parameter due to its nature.



Recent Projects

The **Recent Projects** list shows the currently active project and previous projects recently opened in the application. The current project will have a shortcut to the Measured Signals setup, the Project Configuration window, and the Saved Files tab. It also displays data files associated with this project.

Live Signals, Run Folder, and Data Files

Below the Recent Projects list is the **Live Signals, Run Folder, and Data Files** list. Live Signals includes the signals in the Data Source file or files and new signals from the analysis. The live signals are not saved to disk but are available to display in a Signal Display view tab. Anything saved to disk will appear in the Saved Files list, which has an associated directory in the file system. Run Folders contains the data for each run. Data is saved in a file format specified under the Data Files tab.

Signal Display

The middle of the main window contains **Signal Display** view tabs, where live and saved data can be displayed. More than one of these tabs can be created, but there is always at least one. Each of these views contains one or more Display Windows, with a fully customizable layout. These display windows move freely, be resized, and can display any valid combination of live or saved signals. New view windows can be created by selecting an option in the Display menu.

Control Panel and Analysis Parameters

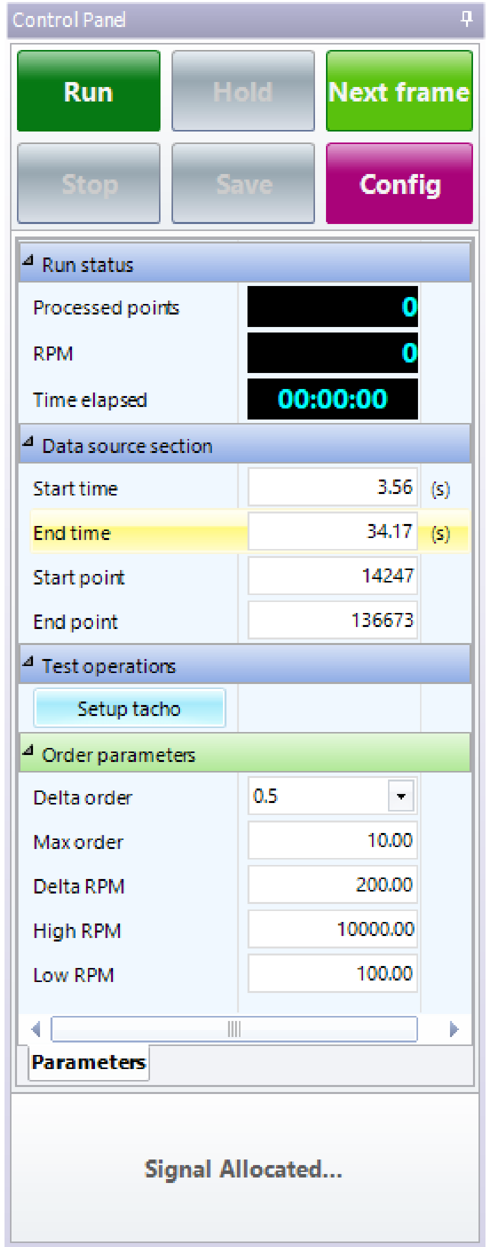
The right side of the window has the **Control Panel** (which controls the post-analysis operation), and the **Analysis Parameters** (which contains parameters specific to the analysis type being performed). The **Config button** on the Control Panel opens the **Project Configuration** window.

Control Panel

The **Control Panel** is used to control post processing operations. It also displays the analysis status information.

The Control Panel contains the following buttons:

- **Run**: starts the analysis
- **Hold/Continue:** pauses and resumes the analysis
- **Next Frame**: process one frame, and then pauses
- **Stop**: stops the analysis
- **Save**: N/A in Orbit Plot project
- **Config:** N/A in Orbit Plot project



The Run Status panel provides the following information:

- **Processed Points:** the number of processed sample points
- **Processed Frame:** the number of processed frames
- **Time Elapsed**: the total time duration of the input signals processed

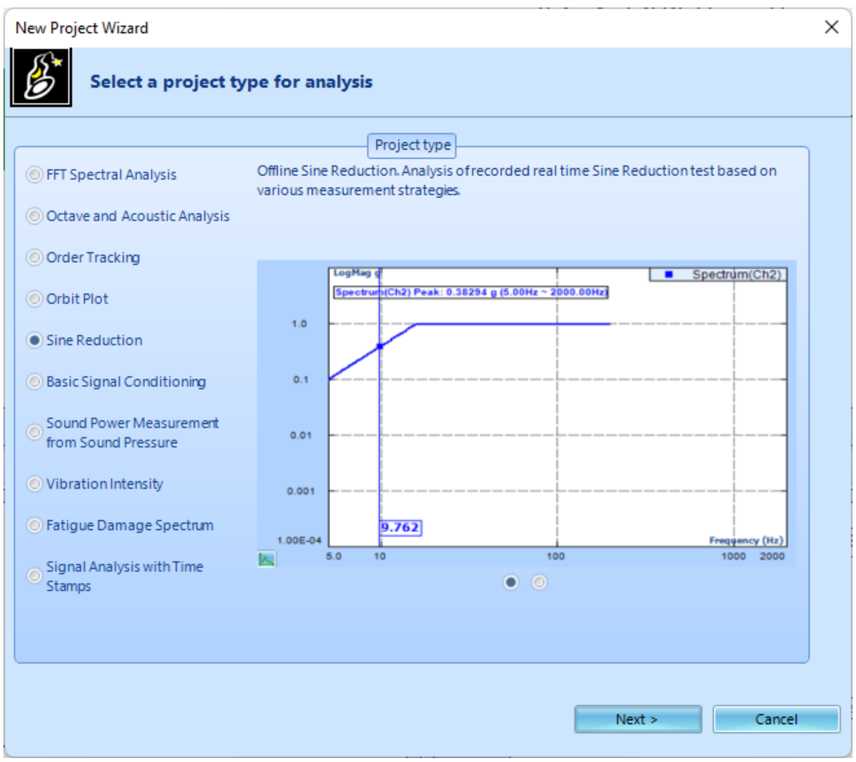
Sine Reduction

Sine Reduction allows users to "slave" a dynamic signal analyzer (DSA) to a vibration control system (VCS) to gain more processing channels for a Swept Sine test. The Sine Reduction software runs on the DSA system; no changes to the VCS are required. Two systems are synchronized by the controller's COLA (Constant Output Level Amplitude) signal. The COLA is a constant voltage sine wave that tracks the drive signal in frequency during a Sine test.

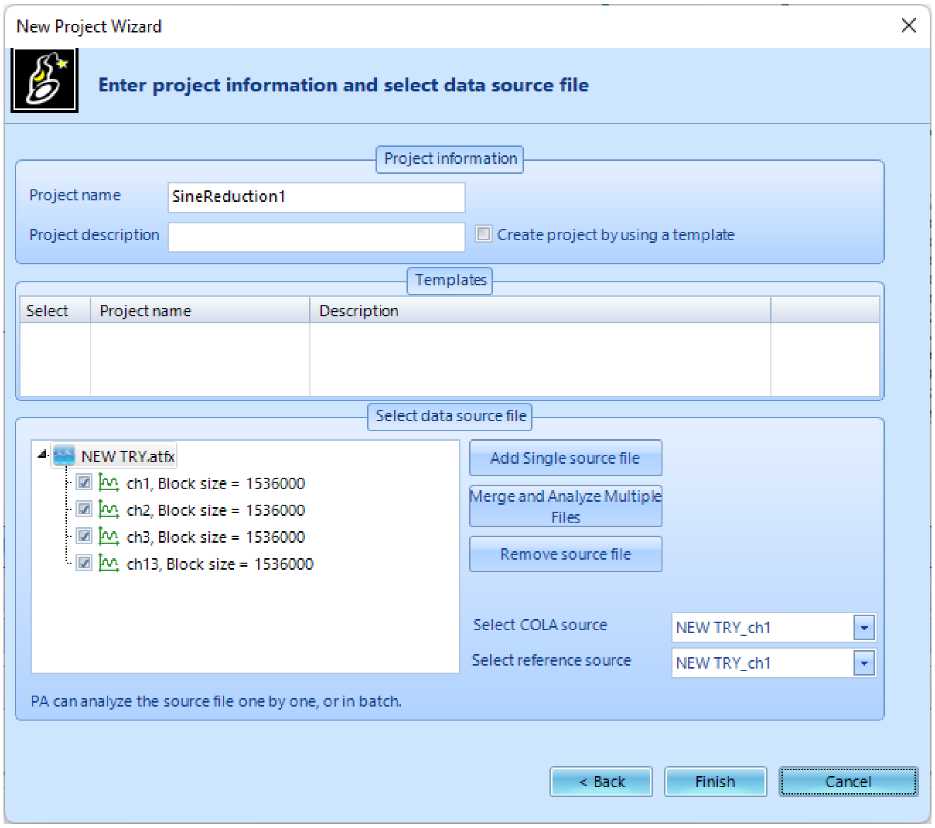
Unlike a real-time Sine Reduction test, PA Sine Reduction post-analyzes recorded signal data file offline for Sine Reduction analysis.

Create a Sine Reduction Project

Open the New Project Wizard and select **Sine Reduction** as the project type. Click **Next** to continue.



In the following window, enter a name for the project and optionally add a description.

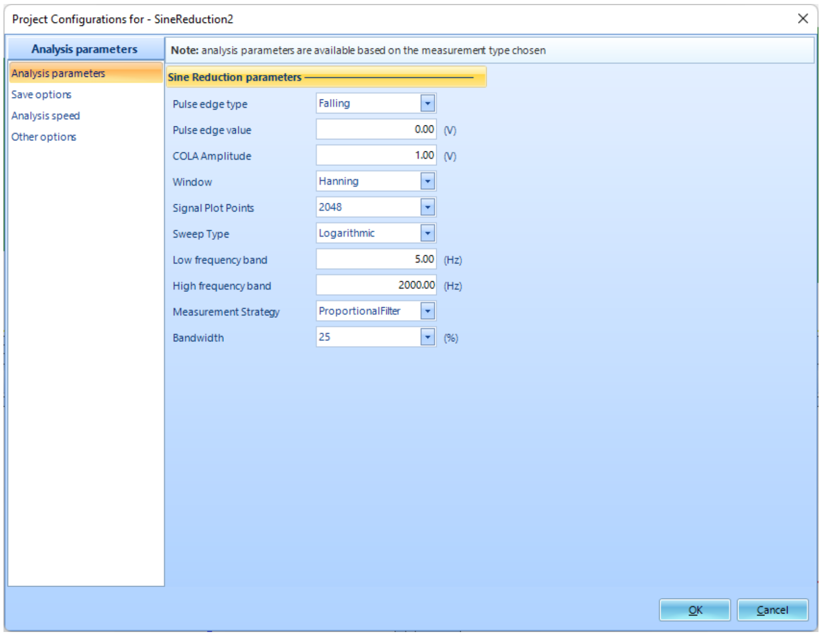


Select the source data files in this window to associate with this project. The source file is usually a time stream recording in ATFX format. Existing source files can be selected and removed from the data source file list by clicking **Remove Source File**.

Select the COLA source before finishing the new project creation. The COLA channel in the PA project must match the COLA channel in the real-time test when the data file is recorded.

Overview

After creating or opening a project, the main Post Analyzer window is displayed. There is a toolbar across the top, and three main sections in the middle. The left side features **Recent Projects**, **Live Signals**, and **Saved Files** views; in the middle are one or more **Signal Display** views; and on the right is the **Control Panel** which also contains the analysis parameters.



Recent Projects

The **Recent Projects** list shows the currently active project and previous projects recently opened in the application. The current project will have a shortcut to the Measured Signals setup, the Project Configuration window, and the Saved Files tab. It also displays data files associated with this project.

Live Signals, Run Folder, and Data Files

Below the Recent Projects list is the **Live Signals, Run Folder, and Data Files** list. Live Signals includes the signals in the Data Source file or files and new signals from the analysis. The live signals are not saved to disk but are available to display in a Signal Display view tab. Anything saved to disk will appear in the Saved Files list, which has an associated directory in the file system. Run Folders contains the data for each run. Data is saved in a file format specified under the Data Files tab.

Signal Display

The middle of the main window contains **Signal Display** view tabs, where live and saved data can be displayed. More than one of these tabs can be created, but there is always at least one. Each of these views contains one or more Display Windows, with a fully customizable layout. These display windows move freely, be resized, and can display any valid combination of live or saved signals. New view windows can be created by selecting an option in the Display menu.

Control Panel and Analysis Parameters

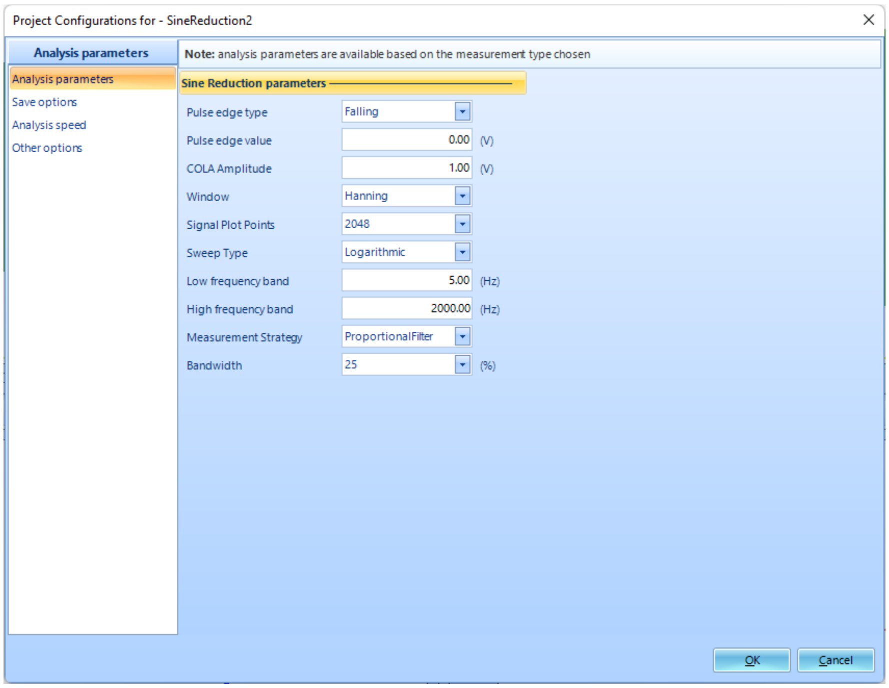
The right side of the window has the **Control Panel** (which controls the post-analysis operation), and the **Analysis Parameters** (which contains parameters specific to the analysis type being performed). The **Config button** on the Control Panel opens the **Project Configuration** window.

Project Configuration

The **Project Configuration** window provides settings for saving data, analysis speed, and other options that are mostly independent of the type of analysis being performed.

Analysis Parameters

Analysis Parameters consist of Sine Reduction analysis parameters and COLA settings.



Pulse Edge Type specifies triggering upon the rising or falling edge of the pulse.

Pulse Edge Value is the threshold voltage level that the input pulse needs to exceed to be detected as a pulse.

COLA Amplitude is used to set the COLA amplitude. Set the COLA amplitude as that was recorded from the Sine controller.

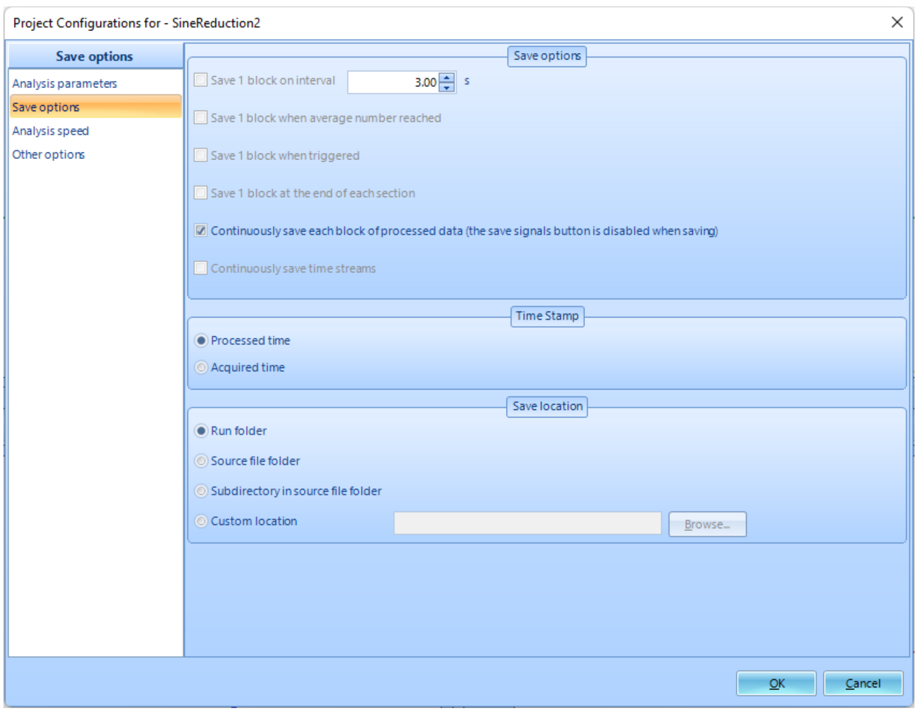
Window lets the user choose the window to be applied during FFT operation. Windowing functions can help reduce leakage and increase the precision of the frequency measurement. Detailed descriptions about window types and average modes are provided in the DSA Basics document.

- **Signal Plot Points** sets the number of signal points on the X-axis.
- **Sweep Type** has two options: Logarithmic and Linear.
- **Low Frequency Band** is the low limit of the sweeping range.
- **High Frequency Band** is the high limit of the sweeping range.

- **Measurement Strategy** provides the options of Proportional Filter, Fixed Filter, RMS, Mean, and Peak.
- **Bandwidth** is used to set the tracking filter width when Proportional Filter or Fixed Filter is selected as the Measurement Strategy.

Save Options

After running the analysis, computed signals are listed under the **Live Signals** tab. However, this data is only stored in a display buffer and must be explicitly saved to disk to create new data files. There are multiple ways to do this. One method is to manually click the **Save** button in the Control Panel, which saves data according to the Save Settings specified under the Measured Signals setup. The other method is to enable an option (listed in the following section) to automatically save data during the analysis.



Four options are provided to save block data and one option to save stream data. Most data computed in Post Analyzer is block data, such as APS and SRS spectra. Some data, such as the output of digital filters or other data conditioning blocks, are continuous time streams. Only one type of data can be saved at a time – block or time stream.

Save 1 block on interval saves one block of data for every computed signal with the save option selected in the Measured Signals setup per the selected interval.

Save 1 block when average number reached saves one block of data every time the number of processed blocks reaches the average number set under the Analysis Parameters.

Save 1 block when triggered saves one block of data when a trigger occurs. Click the Setup Trigger button in Analysis Parameters to set up a trigger.

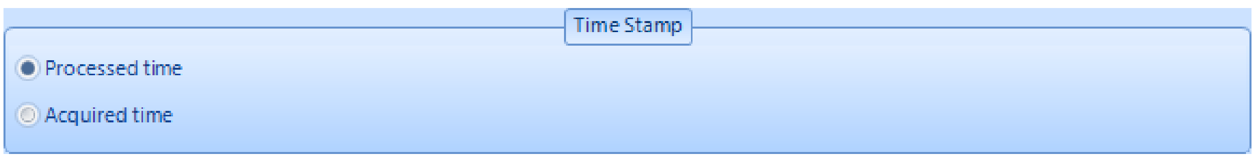
Save 1 block at the end of each section saves one block at the end of an analysis section.

Continuously save each block of processed data saves *every* block of computed data. Please note that the Save button in the Control Panel is disabled when this setting is used.

Continuously save time streams saves all time stream data. Selecting this option disables all options related to block data saving.

Time Stamp

Users can select Processed time or Acquired time in the **Time Stamp** section to time stamp data.

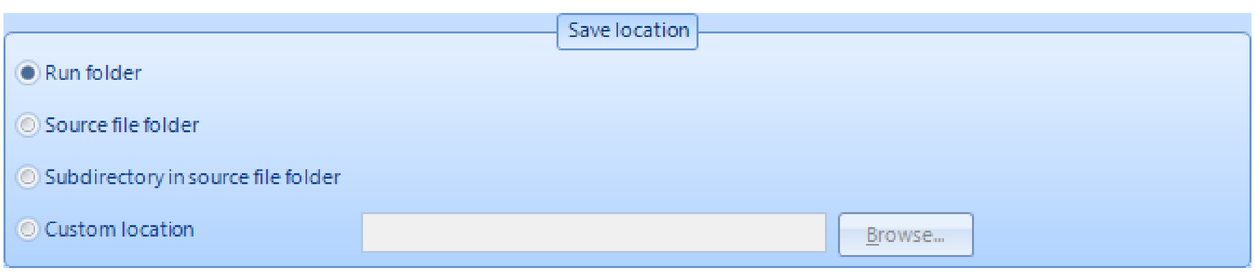


Processed Time: data is time stamped according to the time it was processed with Post Analyzer software on the computer.

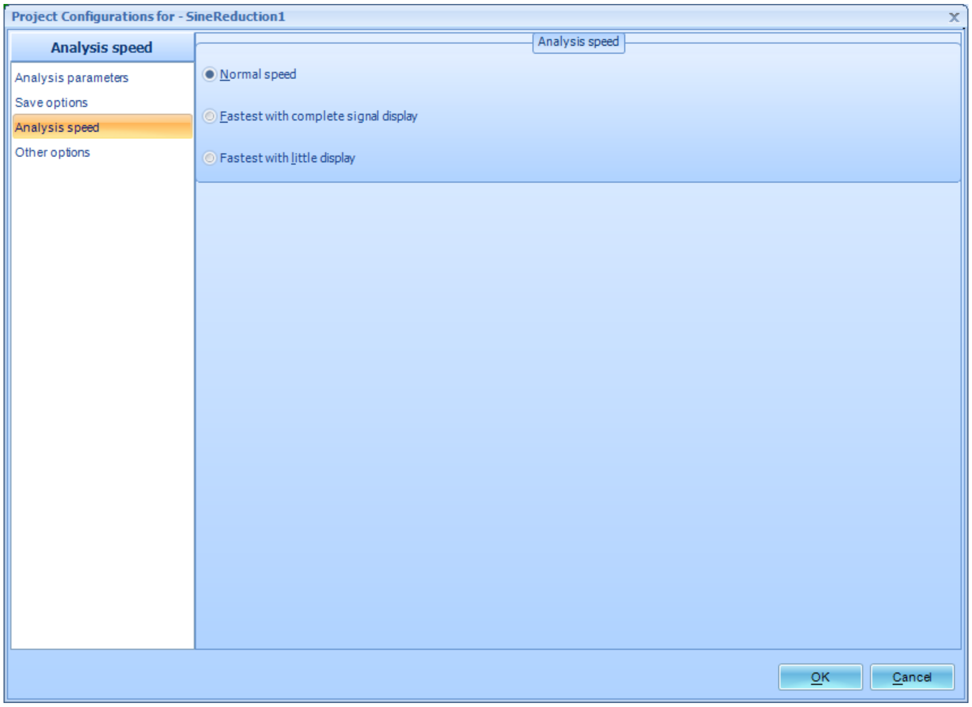
Acquired Time: data is time stamped according to when it was acquired with the data acquisition hardware.

Save location

The **Save Location** section allows users to select the location of the file system used to create new data files from the following options: **current run folder, same folder as data source file folder, subdirectory in source file folder,** or a **custom location**.



Analysis Speed

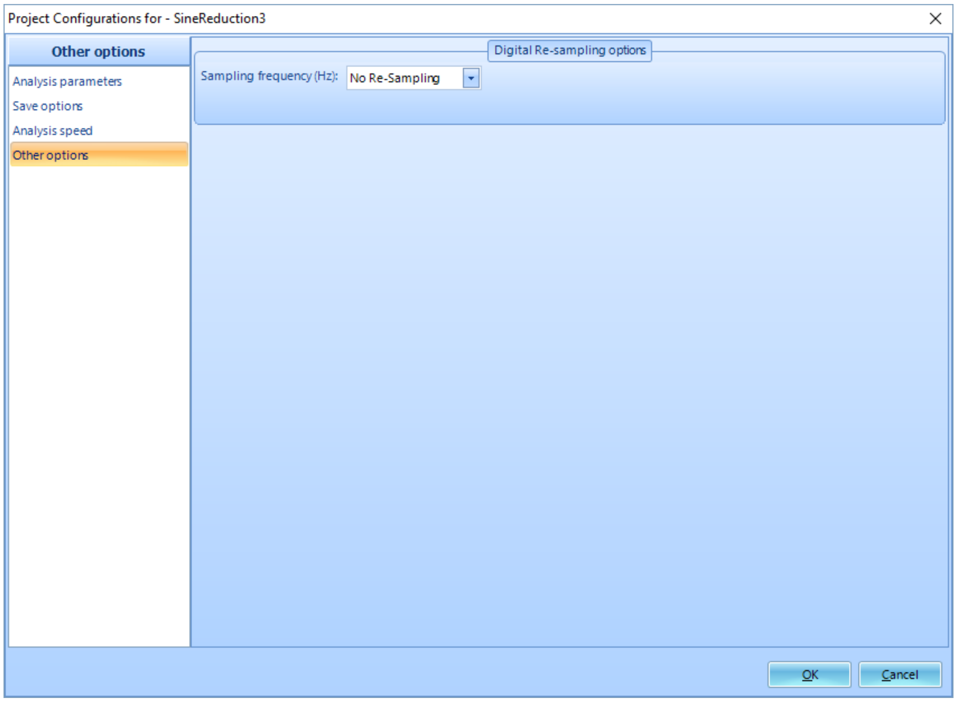


Analysis Speed adjusts the amount of processing time used for the signal display during post-processing. Increased display details result in less available processing resources for post processing data analysis. **Normal Speed** is the default setting that balances data processing and the signal display. **Fastest with Complete Signal Display** prioritizes the display, and **Fastest with Little Display** prioritizes data processing.

Other Options

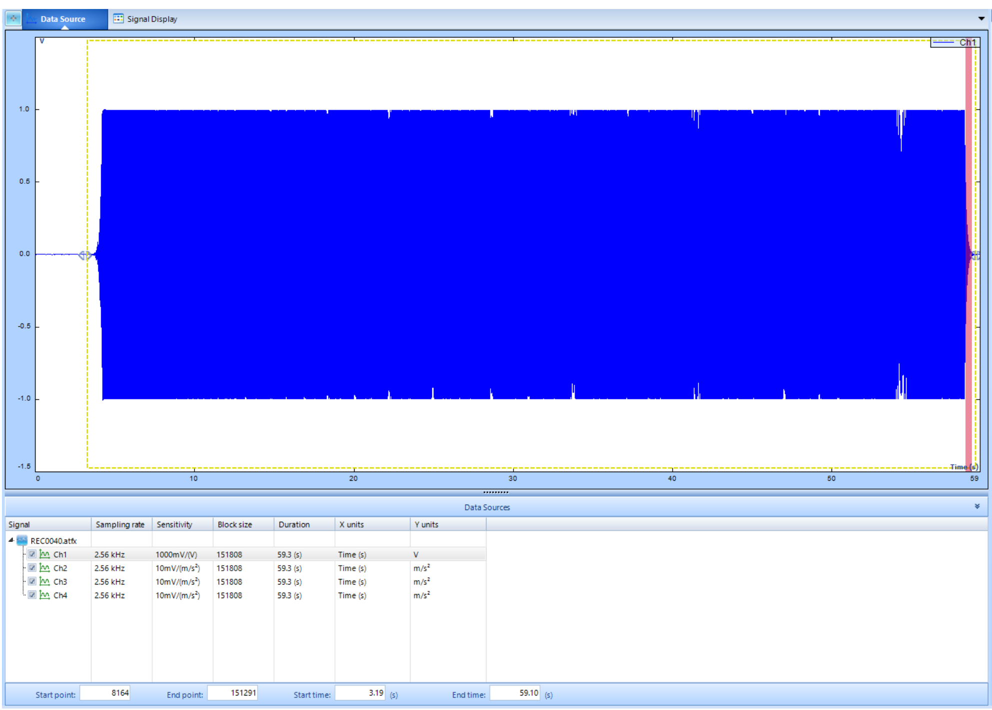
Digital Resampling allows all source signals to be resampled according to the selected sampling rate to meet another interface requirement. Sampling rate stages ranging from 20 Hz to 102.4 kHz can be selected from the dropdown menu.

Data can be automatically resampled with Post Analyzer software.



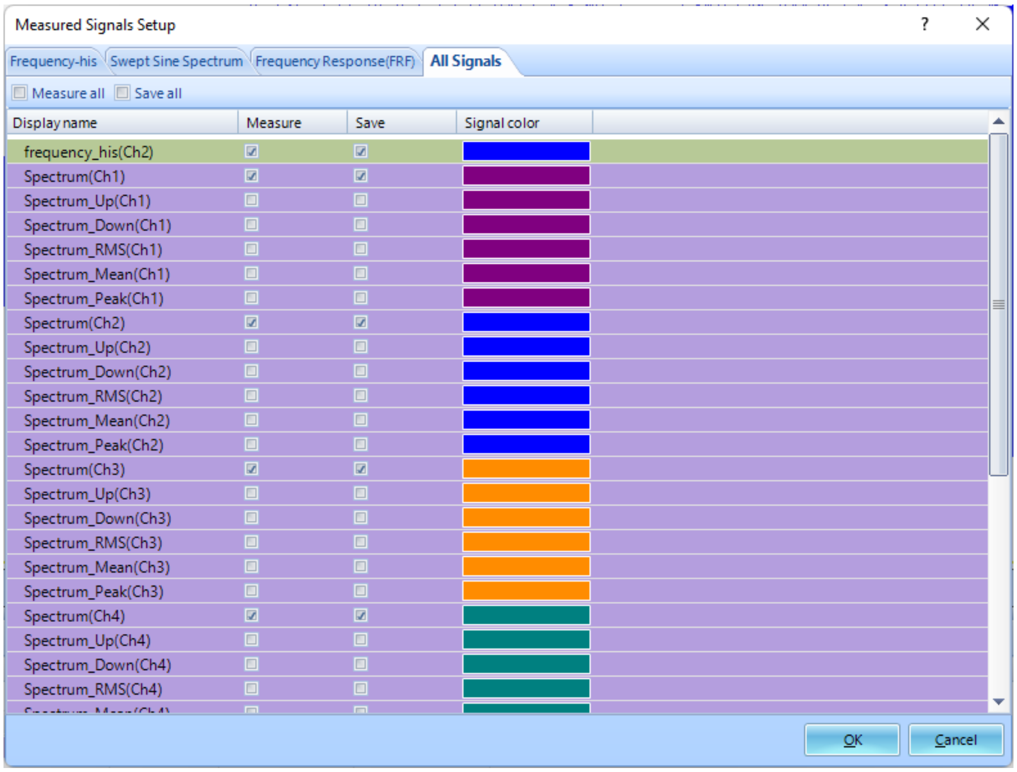
Data Source

The **Data Source** tab displays source data from the recorded file or files associated with the current project. Enable or disable source signals in this tab and set the time domain for analysis by dragging the left and right edges of the yellow box. The start and stop point, or time, can be entered manually in the text field below.



Measured Signals

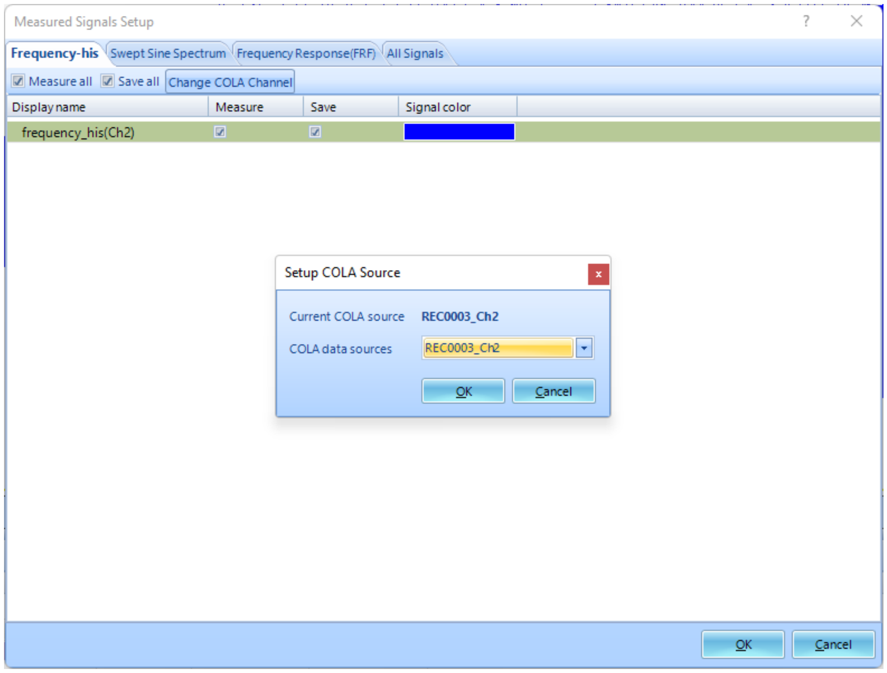
Every signal available for computing is listed under the **Setup->Measured Signals** tab.



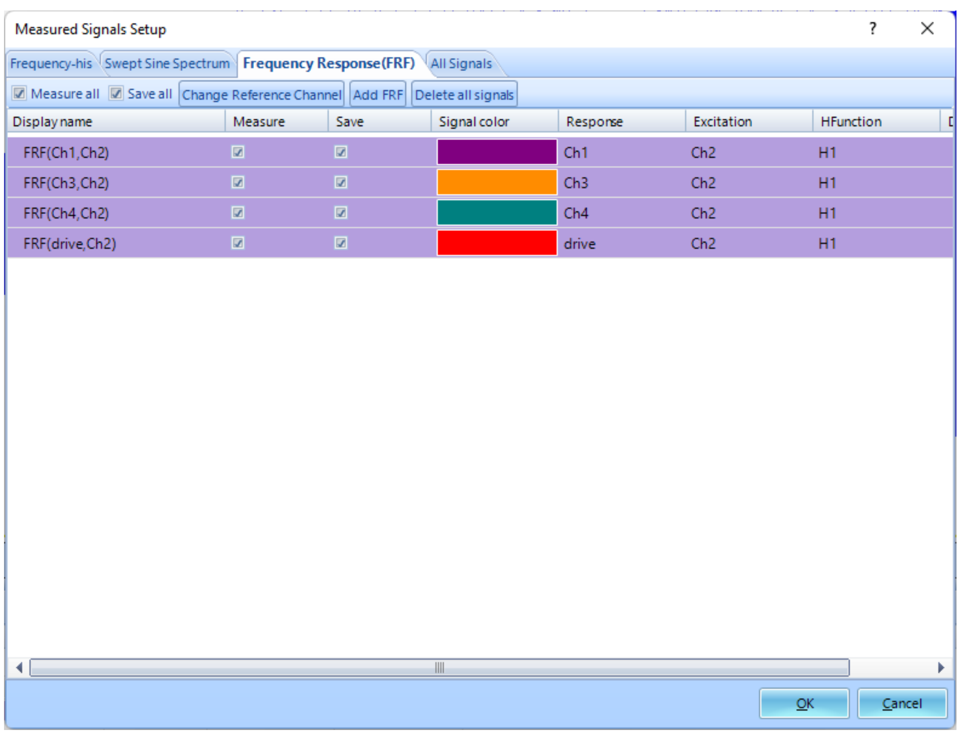
This includes Frequency-his, Swept Sine Spectrum, Frequency Response (FRF), and All Signals.

Each signal has a **Measure, Save**, and **Cache** option. Signals with the **Measure** option checked are listed under the **Live Signals** tab and available for display. Signals with the **Save** option selected are saved to the disk when the **Save** button is clicked in the control panel, or when an automatic saving option is specified in **Project Configuration**. Signals with the **Cache** option are cached in the memory for faster access by the software.

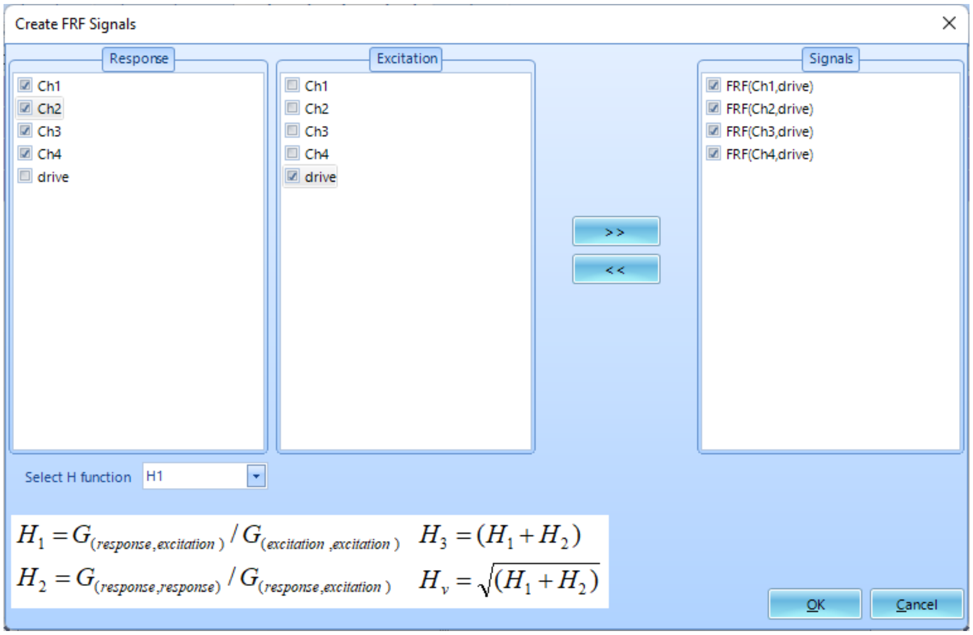
Select a COLA channel by clicking **Change COLA Channel** in the Frequency-his tab. Select a data channel as a COLA data source in the Setup COLA Source window.



Default FRF signals use a COLA channel as the reference channel after a new Sine Reduction project is created. Click the **Add FRF** button in the Frequency Response (FRF) tab to assign the response and excitation channels.



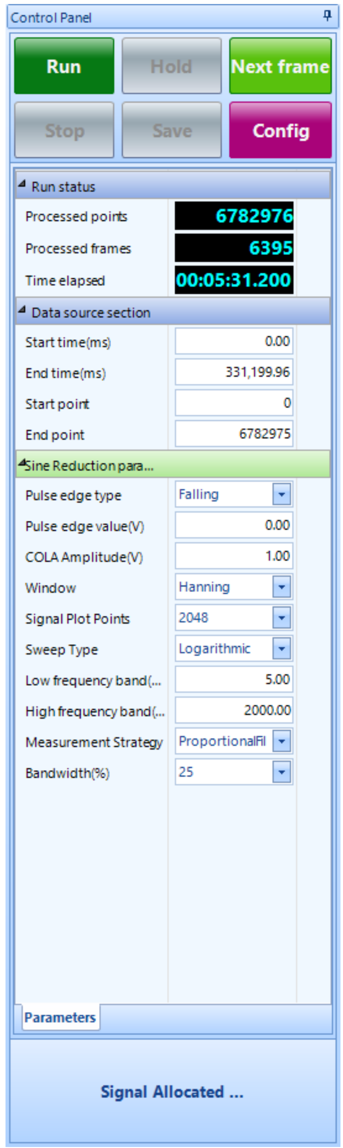
The **Create FRF Signals** window will appear as shown in the following screenshot.



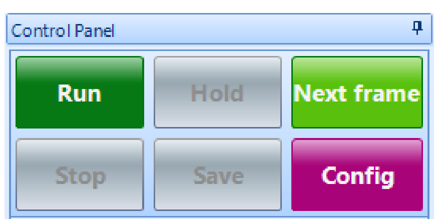
Select the **Response** and **Excitation** channel(s); click the >> button to add FRF signals. Remove selected FRF signals by clicking the << button. Selections of different frequency response functions are available. Users can select H_1, H_2 or H_3 a frequency response.

Control Panel

The **Control Panel** is used to control post processing operations. It also displays analysis status information. Quickly modify Analysis Parameters here as well.



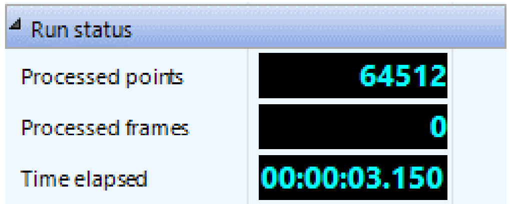
Control panel contains the following buttons:



- **Run** starts the analysis
- **Hold/Continue** pauses and resumes the analysis
- **Next Frame** processes one frame, and then the test is in a Hold state. The test can then process frame by frame via **Next Frame** or run as configured via **Run**.
- **Stop** stops the analysis

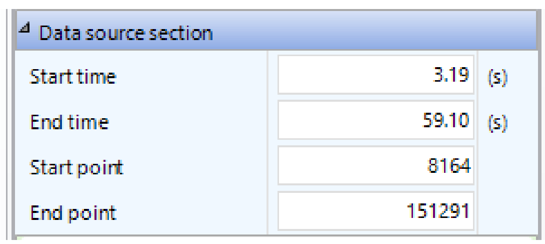
- **Save** saves block signals
- **Config** opens the Project Configuration window

Run Status contains the following information:

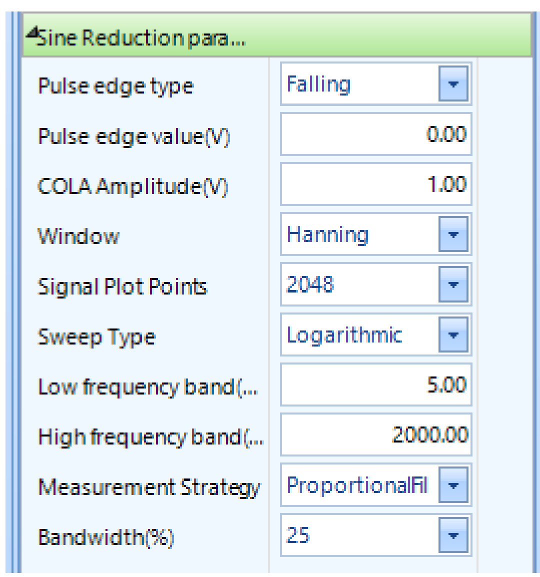


- **Processed Points** is the number of processed samples.
- **Processed Frame** is the number of processed frames.
- **Time Elapsed** is the total time duration of the input signals processed.

Data Source Section lists the following information: Start time, End time, Start point, and End point.



Sine Reduction Parameters contains setup parameters related to Sine Reduction analysis.

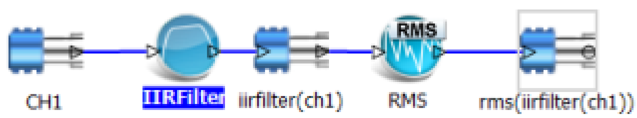


Digital Filters, Data Conditioning, and Resampling

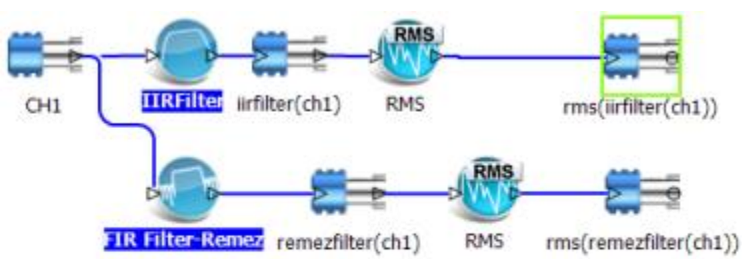
Digital filters are advanced signal conditioning modules that are applied in the data conditioning phase. Users can cascade digital filters or other data conditioning modules to construct powerful post analysis functions. The three main real-time digital filters are: Decimation filter, FIR filter, and IIR Filter.

Digital Filters is a powerful analysis tool that can filter a measured signal in real time and then apply the Spider system's built-in FFT and time-based analysis. Users can precisely define the filter characteristics to meet a specific application. Real-time digital filters are applied in the data conditioning phase. The user designs the filter model with a provided graphic design tool and uploads the filter design parameters to the front-end for real-time calculation. The graphic design tool draws the filter performance in vertical axis with dB unit and horizontal axis in relative frequency.

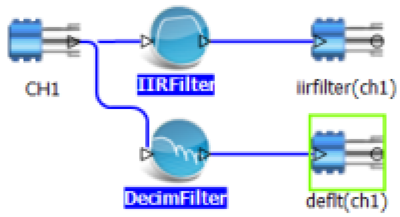
Some users may want to view the energy distribution over time for a specific band of frequencies instead of the entire frequency spectrum from zero to the maximum sampling rate. This is accomplished by creating a band-pass filter and then applying an RMS estimator to the output of the filter. The figure below shows the graphical representation of this process which is used to define the real-time filter in EDM Post Analyzer software. The CH1 icon on the left represents the native measured time stream and is connected to an IIR Filter that computes a signal connected to an RMS estimator labelled iirfilter (ch1). The output of the RMS estimator is a signal labelled rms (iirfilter(ch1)).



Users may also want to view the frequency energy over 100 Hz to 200 Hz and 1000 Hz to 2000 Hz separately. This is accomplished by deriving two output streams from the native channel 1, then applying the band-pass filter to each path as shown below.



In another example, users may want to view the very fast time characteristics of a channel at a high frequency, and the same channel at a very low sampling rate. This is accomplished by applying a decimation filter to the native time stream as shown below. The native channel time stream is split into two streams so that the signal from the same channel is recorded at both high and lower sampling rates.



The **Real Time Digital Filters** option includes three types of digital filters: FIR, IIR and decimation filters. Users can specify low-pass, high-pass, band-pass or band-stop types with several different methods for FIR and IIR filters. This chapter first explains the filter design theory and then introduces operations within the Post Analyzer software and Spider hardware.

The goal of filter design is to calculate a series of filter coefficients based on user specified criteria, which are often described by following variables:

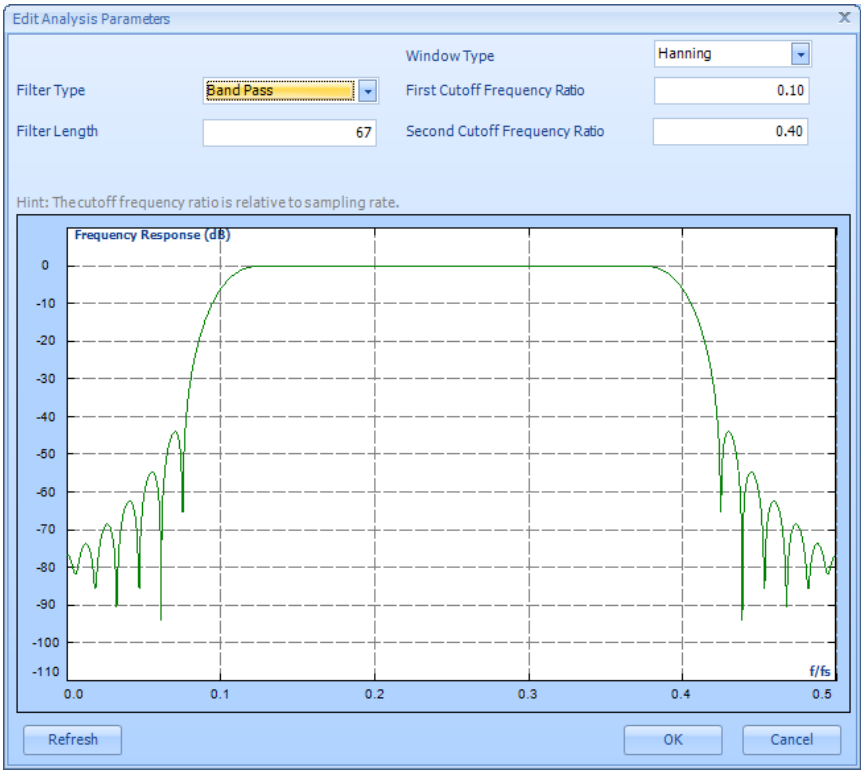
Number of Filter Coefficients: also known as the order of the filter. The filter order defines the number of delay elements used in the filter. A lower order filter consists of a fewer number of coefficients. A low order filter responds relatively faster than a higher order filter, i.e., the filter output has less lag time.

Cutoff Frequencies: low-pass or high-pass filters only need one cutoff frequency. Band pass or band-stop filters require two cutoff frequencies to fully define the filter shape. The following figure shows a typical band-pass filter design with the two cutoff frequencies set to approximately 0.1 and 0.2 Hz as indicated by the blue and yellow vertical lines.

Stop Band Attenuation: defines portion of input signal cut out of the output at rejected frequencies. In theory, the higher the attenuation, the better. The following figure shows the stop ban attenuation is >40 dB as seen from the highest side lobe just below 0.25 Hz.

Pass Band Ripple: Ripple is an unavoidable characteristic for a digital filter. It refers to the fluctuation in the filter shape outside the transition frequencies. If a very flat filter is required, then it can be specified by choosing a very low ripple. In the following figure, ripple is seen in the stop band and no ripple is evident in the pass band. Ideally the pass band should be very flat, and some ripple is tolerable in the stop band.

Width of Transition Bands: This refers to the filter shape between a band pass and a band stop region. Ideally this transition band should be very small. However, a very narrow transitional band requires a higher order filter which affects the filter response time and can also affect ripple. In the following figure, the transition bands are between 0.05 to 0.1 and 0.2 to 0.25.



In most cases, filter design includes making trade-offs between minimizing the filter order, ripple, transition band width, and response time. Not all variables can be satisfied at the same time. Filter design can be an iterative process and experience is helpful.

FIR Digital Filters

Finite Impulse Response (FIR) filters have the distinctive trait that their impulse response lasts for a finite duration of time as opposed to, an Infinite Impulse Response (IIR) filters whose impulse response is infinite in duration. This trait is due to the fact that there are no feedback paths in the FIR filter. FIR filters offer several advantages over IIR filters:

- Completely constant group delay throughout the frequency spectrum. Group delay refers to the time delay between when a signal goes into the filter and when it comes out. Constant group delay means that an input signal will come out of the filter with all parts delayed the same duration with no distortion.
- Complete stability at all frequencies regardless of the size of the filter.

FIR filters also have some disadvantages:

- The frequency response is not as easily defined as it is with IIR filters.
- The number of coefficients required to meet a frequency specification may be far larger than that required for IIR filters.

A digital filter can be understood by considering the difference equation which defines how the input signal is related to the output signal as:

$$y[n] = b_0 x[n] + b_1 x[n-1] + \cdots + b_N x[n-N]$$

where x[n] is the current input signal sample, x[n-1] is the previous signal sample and x[n-N] is the last sample in the series. The series multiplies the most recent N+1 samples associated with the N+1 filter coefficients. y[n] is the current output signal and b_i are the filter coefficients. The number N is known as the filter order; an N^{th}-order filter has (N + 1) terms on the right-hand side and N+1 filter coefficients also referred to as “taps”.

This equation illustrates why a higher order filter has a slower response time. It takes more samples and therefore more time for an event to work its way through the series until the output is no longer affected by the event as compared to a lower order filter with fewer coefficients.

The previous equation can also be expressed as a convolution of the filter coefficients and the input signal.

$$y[n] = \sum_{i=0}^{N} b_i x[n-i]$$

The impulse response of the filter shows how the historical data affects the current filtered value.

The longer the impulse response, the farther the old data will affect the current filtered value. To find the impulse response we set:

$$x[n] = \delta[n]$$

where $\delta[n]$ is the Kronecker delta impulse. The equation below shows that the impulse response for an FIR filter is simply the set of coefficients b_n, as follows:

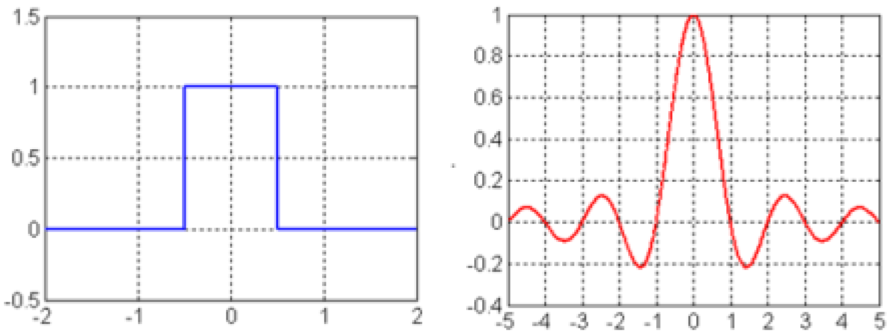
$$h[n] = \sum_{i=0}^{N} b_i \delta[n-i] = b_n \qquad for\ n = 0\ to\ N$$

FIR filters are clearly stable, since the output is a sum of a finite number of finite multiples of the input values, so can be no greater than $\sum_{n=0}^{N} |b_n|$ times the largest value appearing in the input.

Data Windows FIR Filters

Hundreds of methods are available in the academic world to design a FIR filter to meet various criteria. The Post Analyzer software includes the most popular filter design methods: Data Window and Remez. Both methods are discussed below.

The Data Window FIR Filter Design method is the easiest to understand. The name "Window" comes from the fact that these filters are created by scaling a sinc (SIN(X)/X) function with a window such as a Hanning, Flat Top, etc. to produce the desired frequency effect.

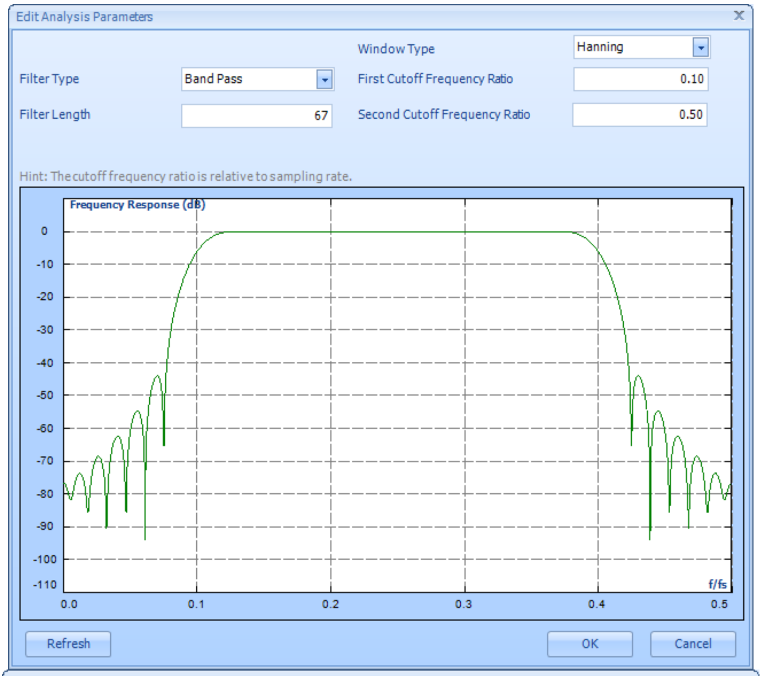


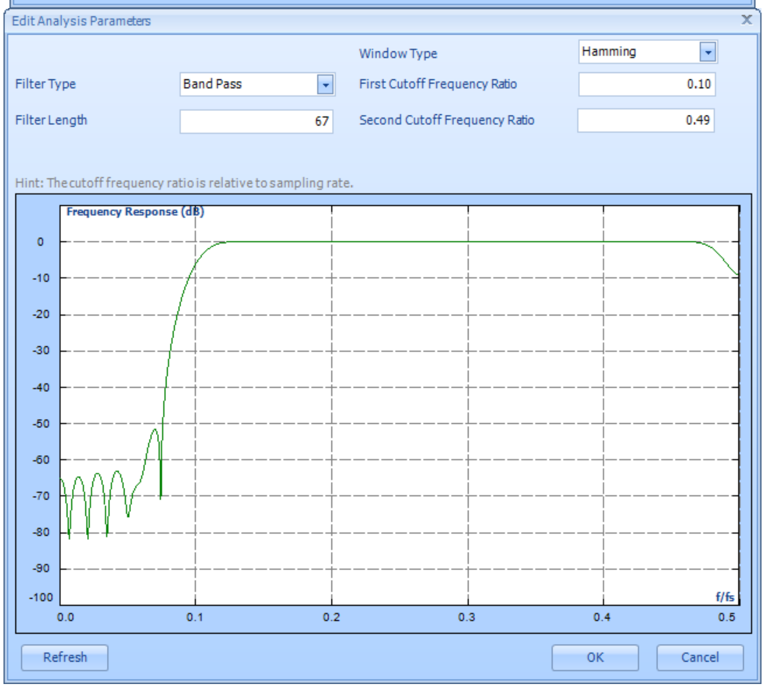
A data window FIR filter is generated by starting with an ideal “brick-wall” shaped filter, which is a filter with vertical edges, or zero transition band width as shown on the left. The brick-wall filter is specified by the cutoff frequencies and has a band-pass amplitude of 1 and stop band amplitude of 0. The problem with the ideal brick-wall filter is that the time response oscillates forever and it requires an infinite number of filter coefficients.

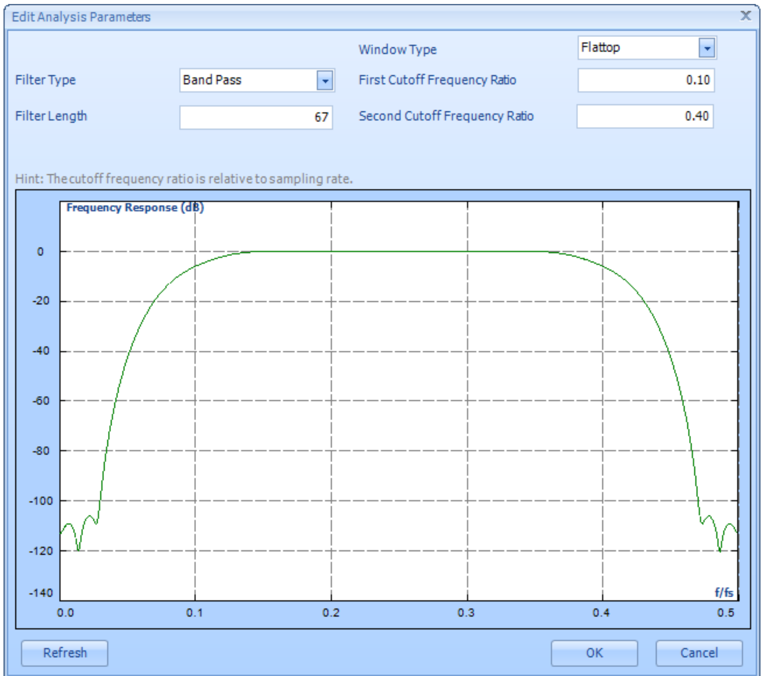
This ideal filter can be modified by applying a data window to force the time response to decay in a finite time. Of course, this degrades the shape of the ideal brick-wall filter performance. It introduces ripple, increases the transition band width, and decreases the stop band attenuation. However, it allows the filter to be defined by a finite number of filter coefficients.

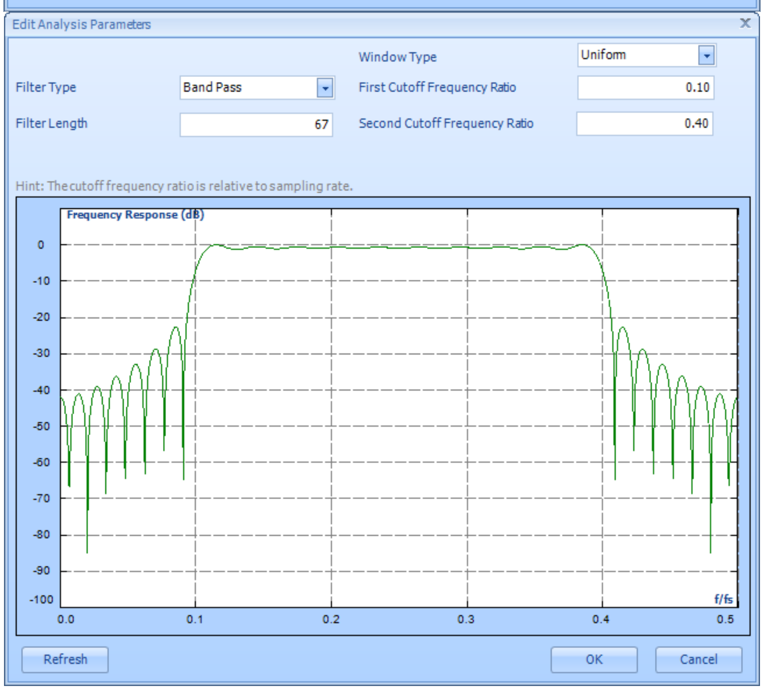
The filter performance can be modified by using different data windowing functions and making the tradeoff between filter order and response time. The user must choose these settings during the filter design.

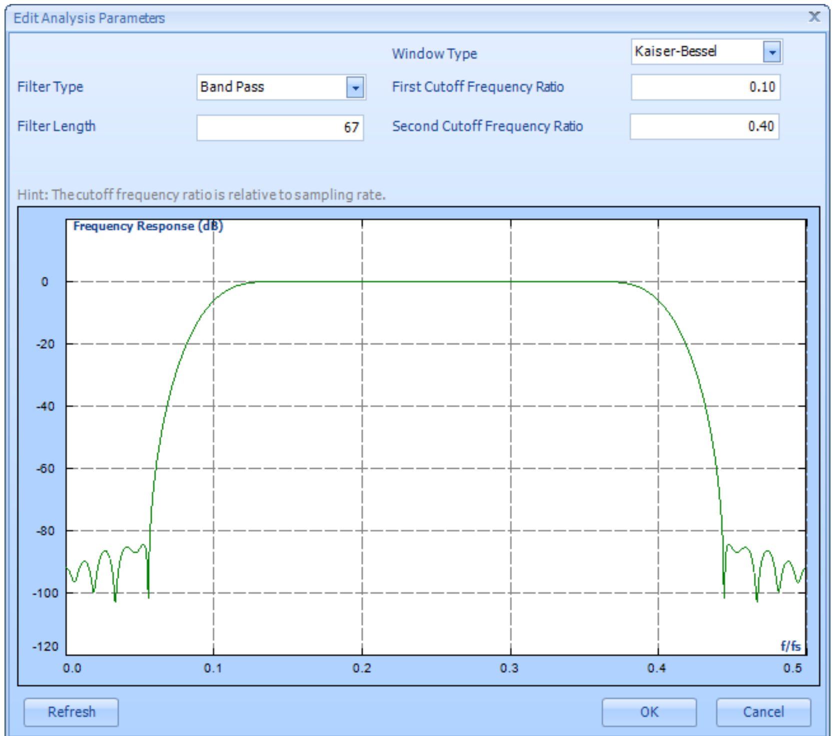
The following figures show a comparison of different data window choices for the same filter settings. In all cases the low and high cutoff frequencies are 0.1 and 0.2 relative to the sampling frequency. The number of filter taps is 51.

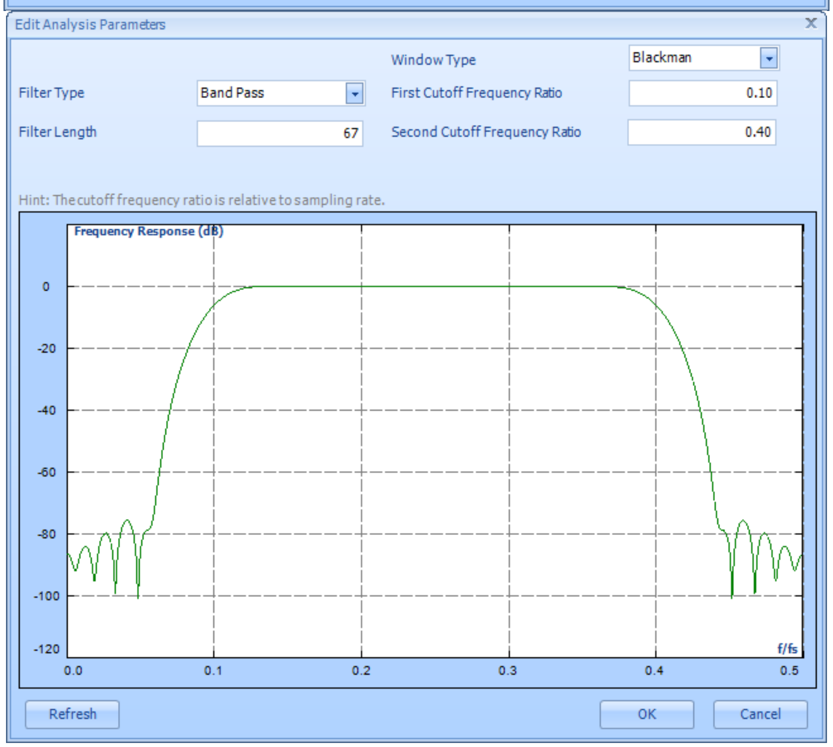

Edit Analysis Parameters
Window Type
Hanning
Filter Type
Band Pass
First Cutoff Frequency Ratio
0.10
Filter Length
67
Second Cutoff Frequency Ratio
0.50
Hint: The cutoff frequency ratio is relative to sampling rate.
Frequency Response (dB)
f/fs
Refresh
OK
Cancel


Edit Analysis Parameters
Window Type
Hamming
Filter Type
Band Pass
First Cutoff Frequency Ratio
0.10
Filter Length
67
Second Cutoff Frequency Ratio
0.49
Hint: The cutoff frequency ratio is relative to sampling rate.
Frequency Response (dB)
f/fs
Refresh
OK
Cancel


Edit Analysis Parameters
Window Type
Flattop
Filter Type
Band Pass
First Cutoff Frequency Ratio
0.10
Filter Length
67
Second Cutoff Frequency Ratio
0.40
Hint: The cutoff frequency ratio is relative to sampling rate.
Frequency Response (dB)
f/fs
Refresh
OK
Cancel


Edit Analysis Parameters
Window Type
Uniform
Filter Type
Band Pass
First Cutoff Frequency Ratio
0.10
Filter Length
67
Second Cutoff Frequency Ratio
0.40
Hint: The cutoff frequency ratio is relative to sampling rate.
Frequency Response (dB)
f/fs
Refresh
OK
Cancel



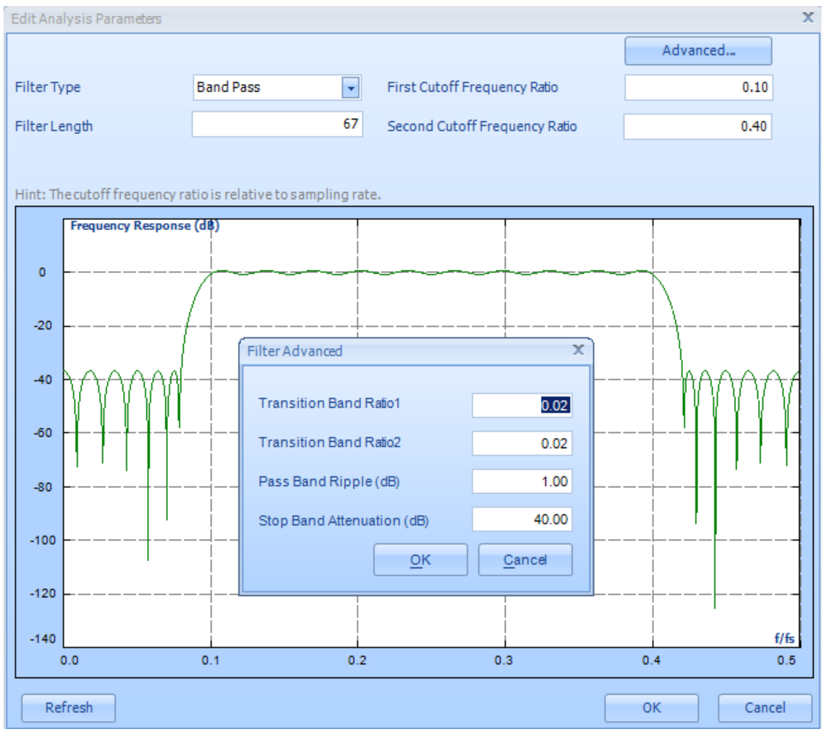


As shown in the preceding figures, different window methods produce different filter performances, i.e., different attenuation of the main lobe and side lobes. The best data window choice depends on the user's specific application.

Remez Filters

The Remez Filter is a different method for designing a FIR filter. It is more computationally intensive than the data window method. A Remez filter is generated with iterative error-reducing algorithms designed to reduce the pass band error. In addition to allowing stop band ratio and frequency definition, the Remez filter allows the "Ripple Ratio" to be defined as a user specified parameter.

The figures below show an example of a filter design using the Remez method in the Post Analyzer software. The low and high cutoff frequencies are 0.1 and 0.2 relative to the sampling frequency. The number of filter taps is 51.



The software is intelligent enough to automatically calculate the total FIR filter length based on these criteria. For example, if the user asks for very high attenuation, very small ripple, or a very sharp transition band, the filter length will be very high. The user must make trade-offs between these parameters so that the appropriate filter length can be generated and used.

IIR Real Time Digital Filters

Infinite impulse response (IIR) filters' impulse responses decay very slowly but theoretically lasts forever. This is due to the fact that the filter input includes the measured signal and also the filter output creating a feedback path which results in the infinite impulse duration. This is in contrast to Finite Impulse Response filters (FIR) which have fixed-duration impulse responses.

The design procedures for IIR filters are somewhat more complicated than FIR filter design because there is no direct design method like the data window method for FIR filters. Instead, IIR filters are typically designed by starting with an ideal analog filter in terms of the frequency response characteristics such as the Chebyshev, Butterworth, or Bessel filter. Then the analog filter is converted into a digital filter using a method known as the Bilinear transformation or impulse invariance method.

An IIR digital filter can be understood by considering the difference equation which defines how the input signal is related to the output signal as:

$$y[n] = b_0 x[n] + b_1 x[n-1] + \cdots + b_p x[n-P] - a_1 y[n-1] - \cdots - a_q y[n-Q]$$

where P is the feed-forward filter order, b_i are the feed-forward filter coefficients, Q is the feedback filter order, a_i are the feedback filter coefficients, x[n] is the input signal and y[n] is the output signal.

The previous equation can also be expressed as a convolution of the filter coefficients and the input signal:

$$y[n] = \sum_{i=0}^{P} b_i x[n-i] - \sum_{j=1}^{Q} a_j y[n-j]$$

which, when rearranged, becomes:

$$y[n] + \sum_{j=1}^{Q} a_j y[n-j] = \sum_{i=0}^{P} b_i x[n-i] \quad if\ we\ let\ a_0 = 1$$

To find the transfer function of the filter, we first take the Z-transform of each side of the above equation, where we use the time-shift property to obtain:

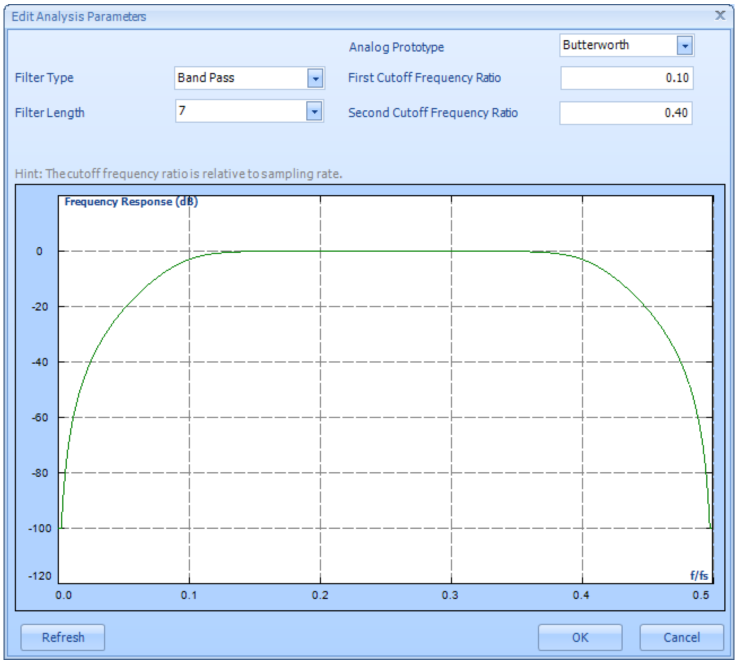
$$Y(z)(1 + \sum_{j=1}^{Q} a_j z^{-j}) = \sum_{i=0}^{P} b_i z^{-i} X(z)$$

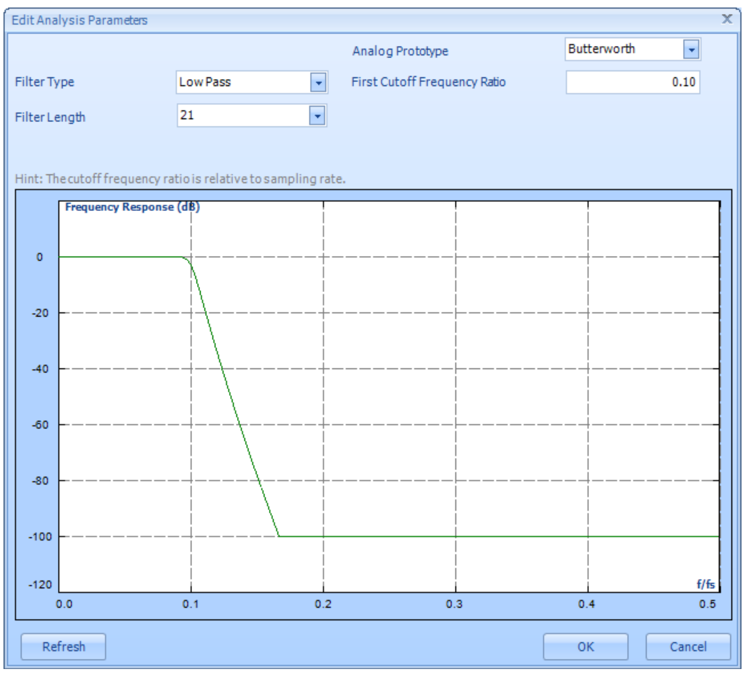
We define the transfer function to be:

$$H(z) = \frac{Y(z)}{X(z)} = \frac{\sum_{i=0}^{P} b_i z^{-i}}{1 + \sum_{j=1}^{Q} a_j z^{-j}}$$

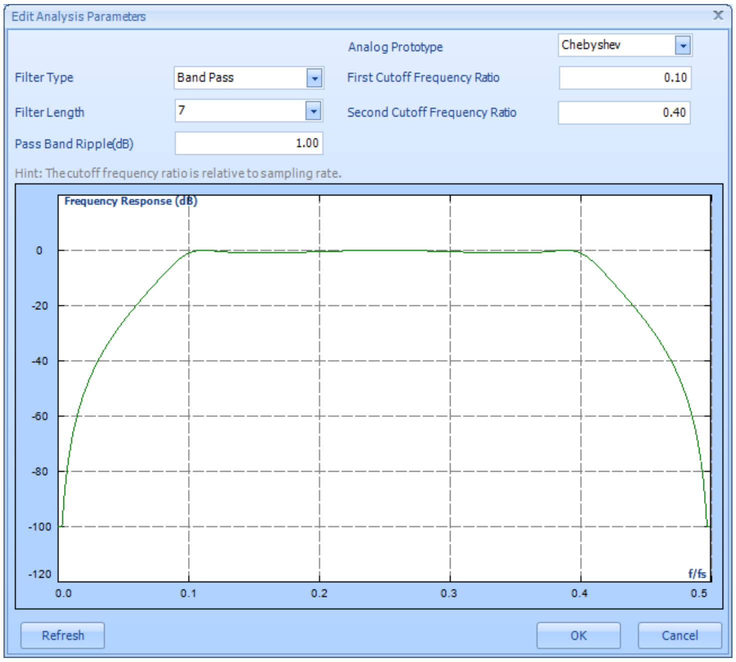
The transfer function gives the frequency response that relates the input to the output magnitude and phase relationship.

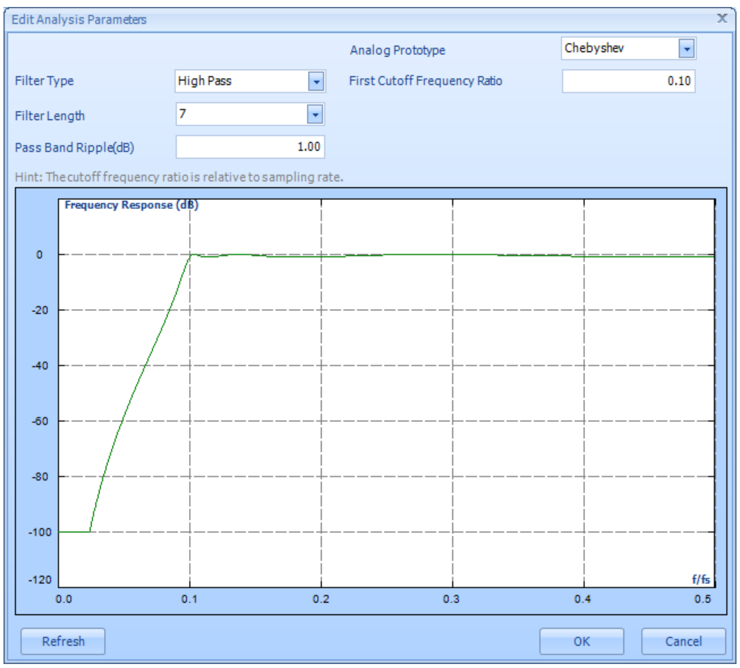
Various analog filter types can be used as the basis for the IIR filter. The Butterworth Filter is a filter type that results in the flattest passband and contains a moderate group delay. The following are examples of Butterworth low-pass and band-pass filters.



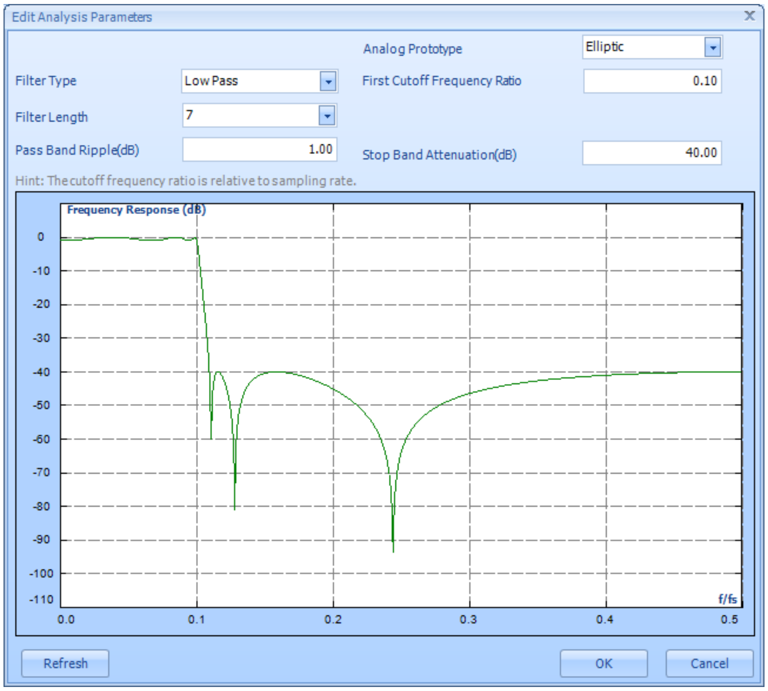


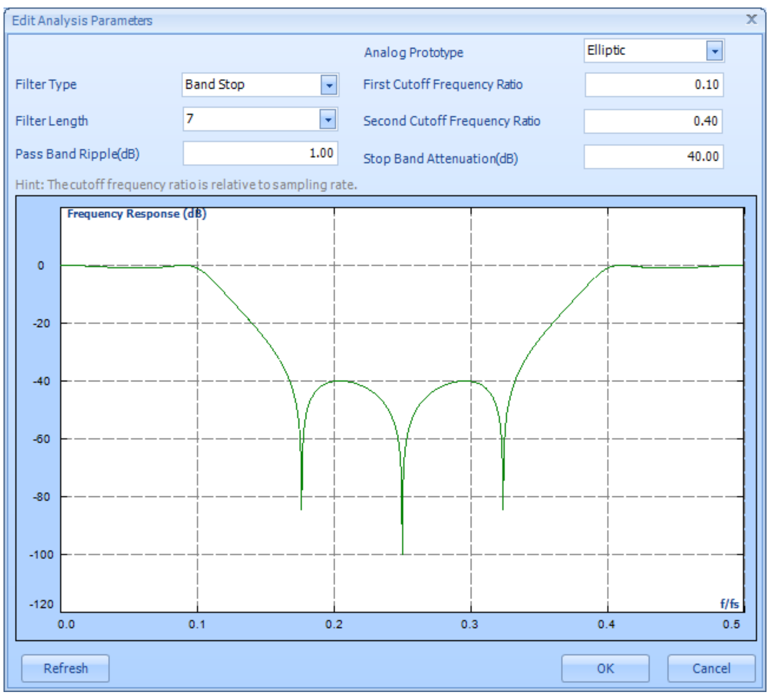
The Chebyshev Type I Filter results in the sharpest passband cut off and contains the largest group delay. The most notable feature of this filter is the significant ripple in the pass-band magnitude. A standard Chebyshev Type I Filter's pass-band attenuation is defined to be the same value as the pass-band ripple amplitude. The following are examples of Chebyshev Type I band-pass and high-pass filters.





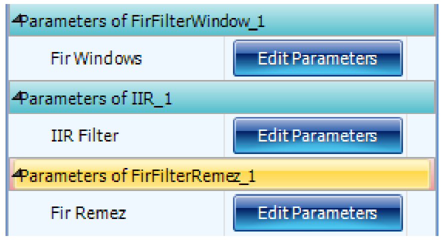
The Elliptic Filter contains a Chebyshev Type I style equi-ripple pass band, an equi-ripple stop band, a sharp cutoff, high group delay, and the greatest possible stop band attenuation. The following are examples of 7th order Elliptic low-pass, band-stop filters.



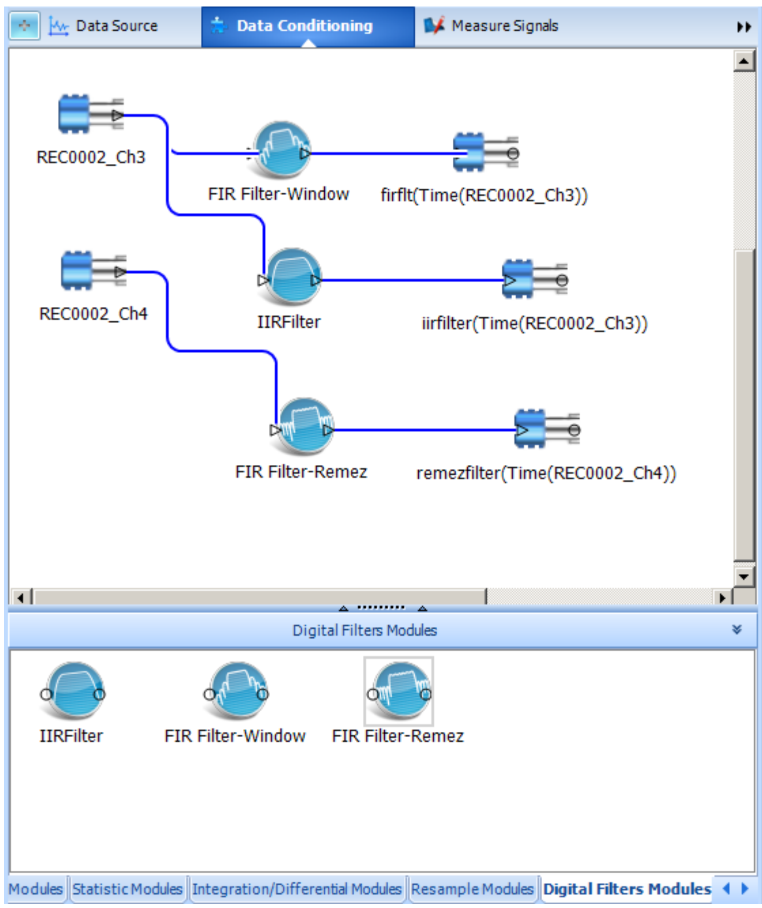


Applying Filters

PA software allows filters to be applied to any project supporting the Data Conditioning option. Once the data conditioning option is selected when creating a new project, filter modules can be found under Data Conditioning tab. **Edit Parameters** buttons are provided in the Control Panel. Click this button to open the edit window.

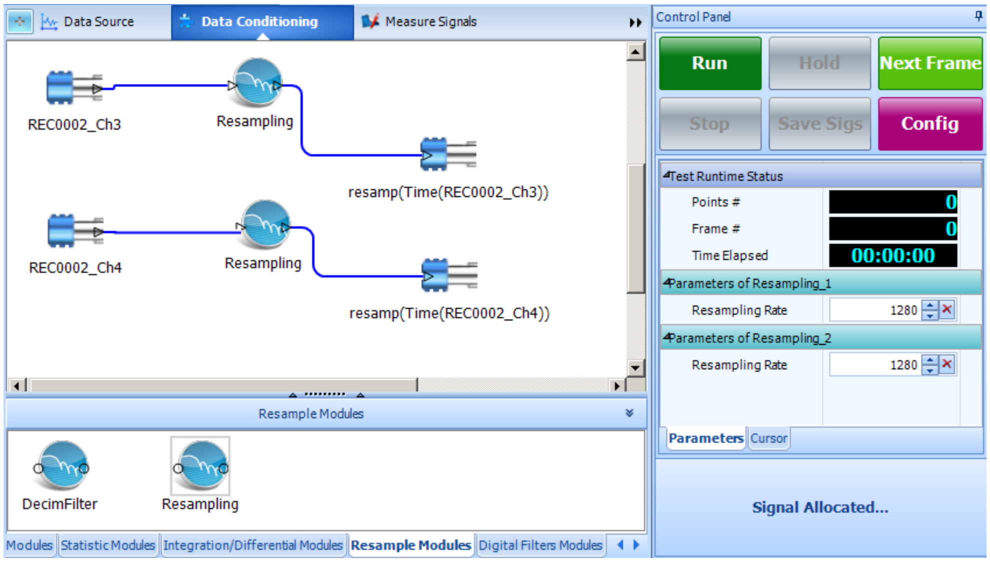


Users can also double-click the filter module in the data conditioning window to configure filter parameters as introduced in the theoretical section. Right-click the filter module to access a command to delete a module.



Applying Digital Resample

Resampling can be applied to any project supporting the Digital Resample option. Once the Data Conditioning option is selected when creating a new project, the Digital Resample modules are provided under the Data Conditioning tab. Editing resampling rates are located in the Control Panel.



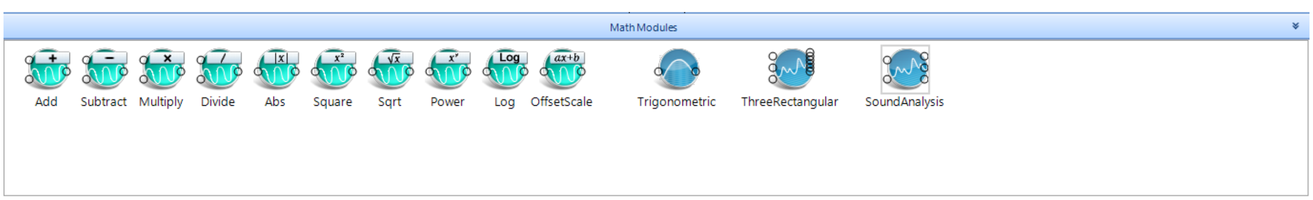
The digital resample can be set up during Project Configuration instead of adding the digital resample module at the Data Conditioning phase (see previous chapters for more details).

Data Conditioning Modules

Math

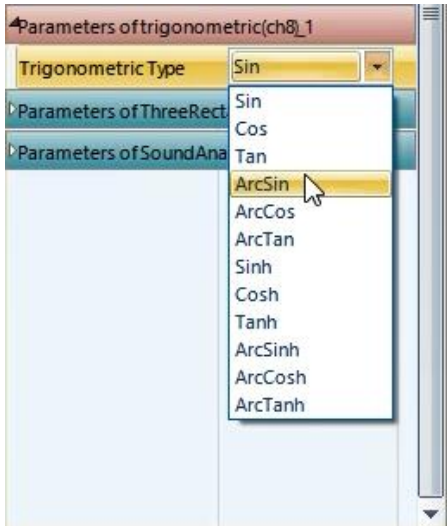
Math Modules, Statistic Modules, and Integration/Differential Modules are located in the Data Conditioning tab. Like Digital filters and Resampling, these modules can be applied to the source files selected in the New Test Wizard.

The Math Modules consist of arithmetic operations among several more complex operations. The arithmetic operations include add/subtract, multiple/divide, square/square-root, power/log, absolute value, and offset scale.

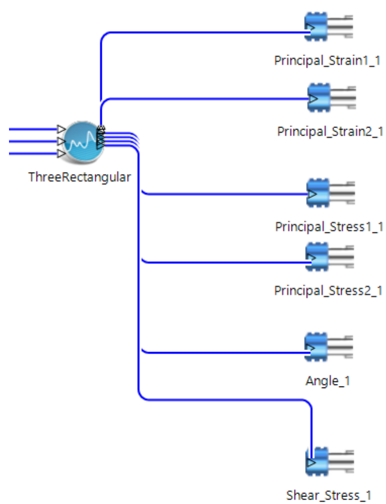


The more advanced functions include:

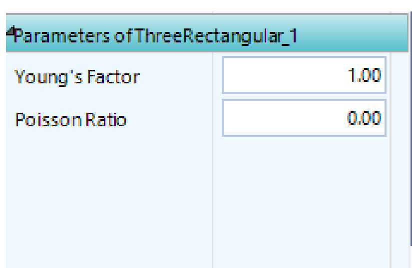
1. **Trignometric Operations:** Recording channels are used as arguments for trigonometric functions. The function to be applied is edited in the Control Panel.



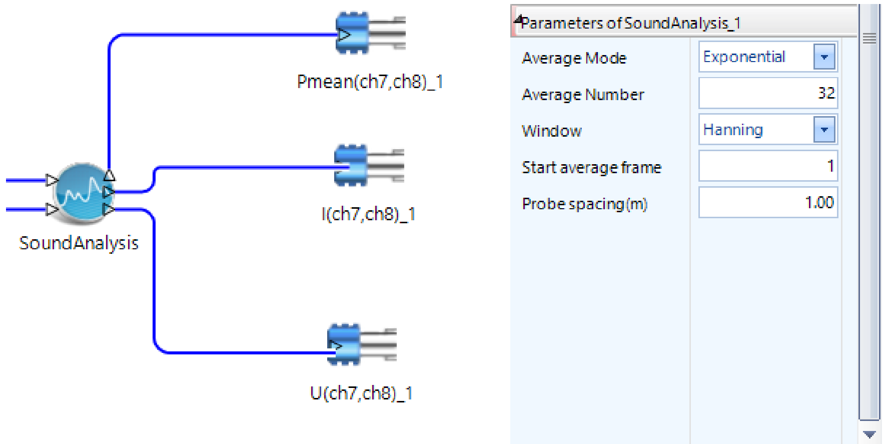
2. **ThreeRectangular:** This function takes 3 channels as inputs and outputs 6 channels.



The functions parameters are Young's Factor and a Poisson Ratio.

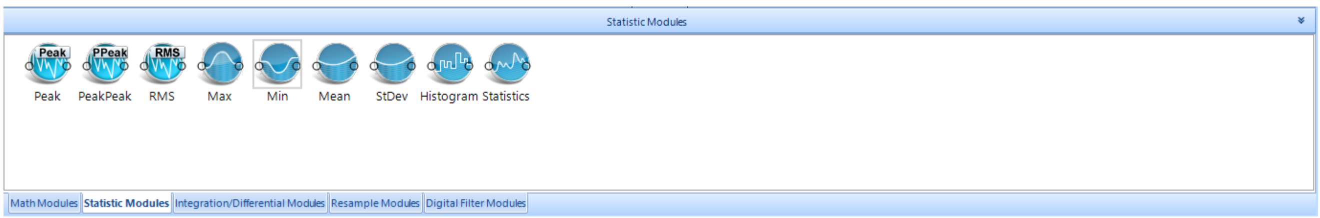


3. Sound Analysis:

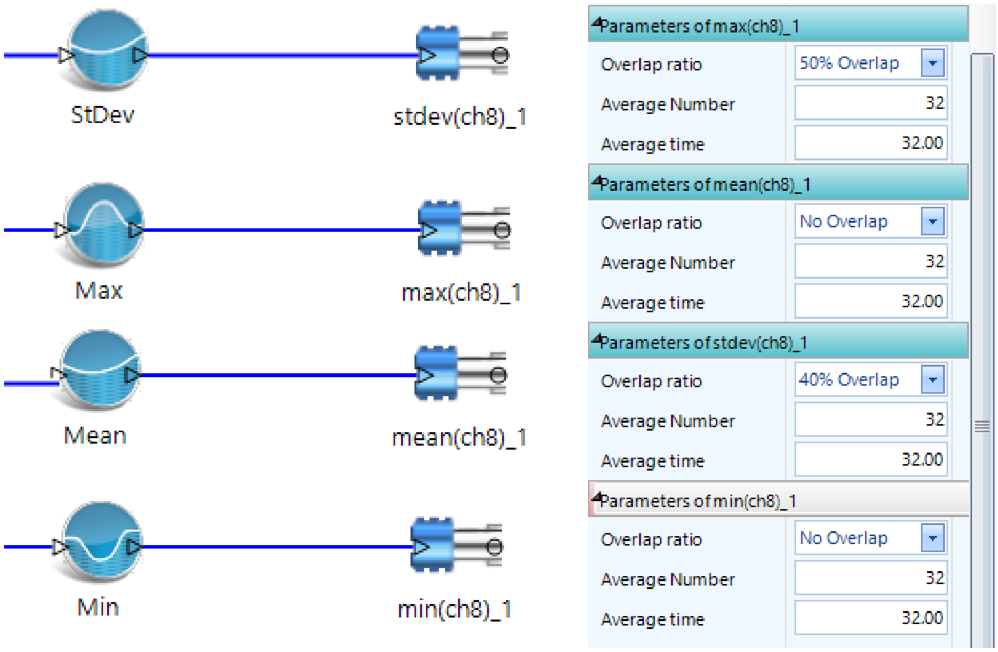


Statistic

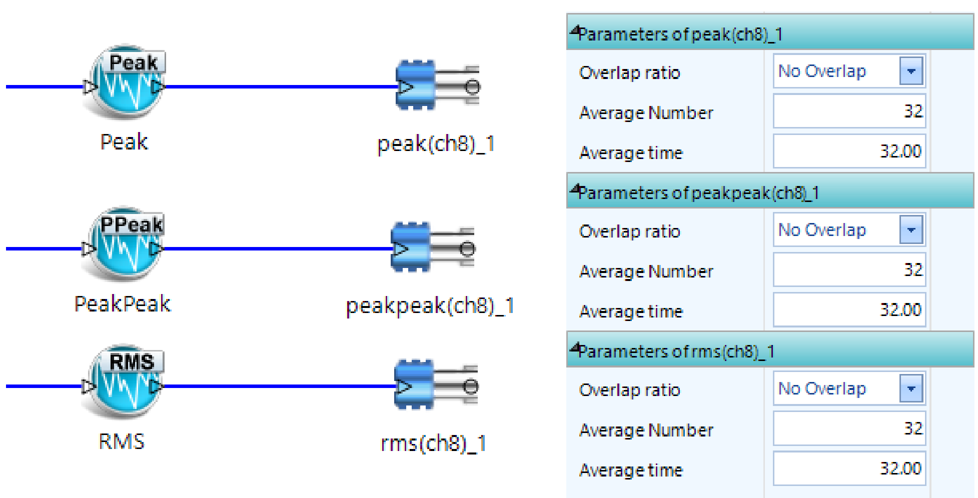
All statistic modules use one recording channel as an input and produce one corresponding output channel.



These modules can be applied to determine the mean, minimum value, maximum value, and standard deviation. Each operation uses the same parameters: Overlap Ratio, Average Number, and Average Time.



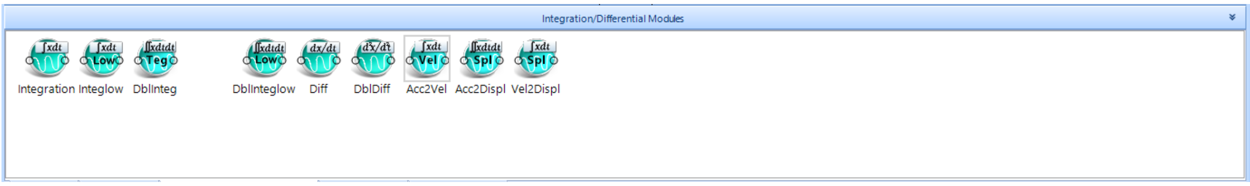
The PeakPeak, Peak, and RMS functions are block-based operations. Similar to the four statistic modules described earlier, these functions use Overlap ratio, Average number, and average time as parameters.



The histogram function, as its name suggests, produces a histogram based on all its input values. It uses three parameters to produce the plot: Minimum Value, Maximum Value, and Number of Bins.

Integration/Differential

There are several integration and differentiation modules with subtle differences among them. However, the operation of all these modules occurs in the time-domain. Frequency-based integration/ differentiation is available through the input channel settings and in the AVD function.



The Integration functions are:

1. **Integration:** This module simply integrates the incoming time stream.
2. **Integlow:** This module first filters out the low frequency components of the incoming time stream and then integrates that data. The cutoff frequency is defined in the function parameters.
3. **DblInteglow:** Similar to Integlow with the exception that the filtered data is integrated twice.
4. **Acc2Vel:** This module requires an acceleration time stream as an input. It performs a frequency-based integration on the input and outputs a timestream of velocity.
5. **Vel2Displ:** This module requires a velocity time stream as an input. It performs a frequency-based integration on the input and outputs a timestream of displacement.

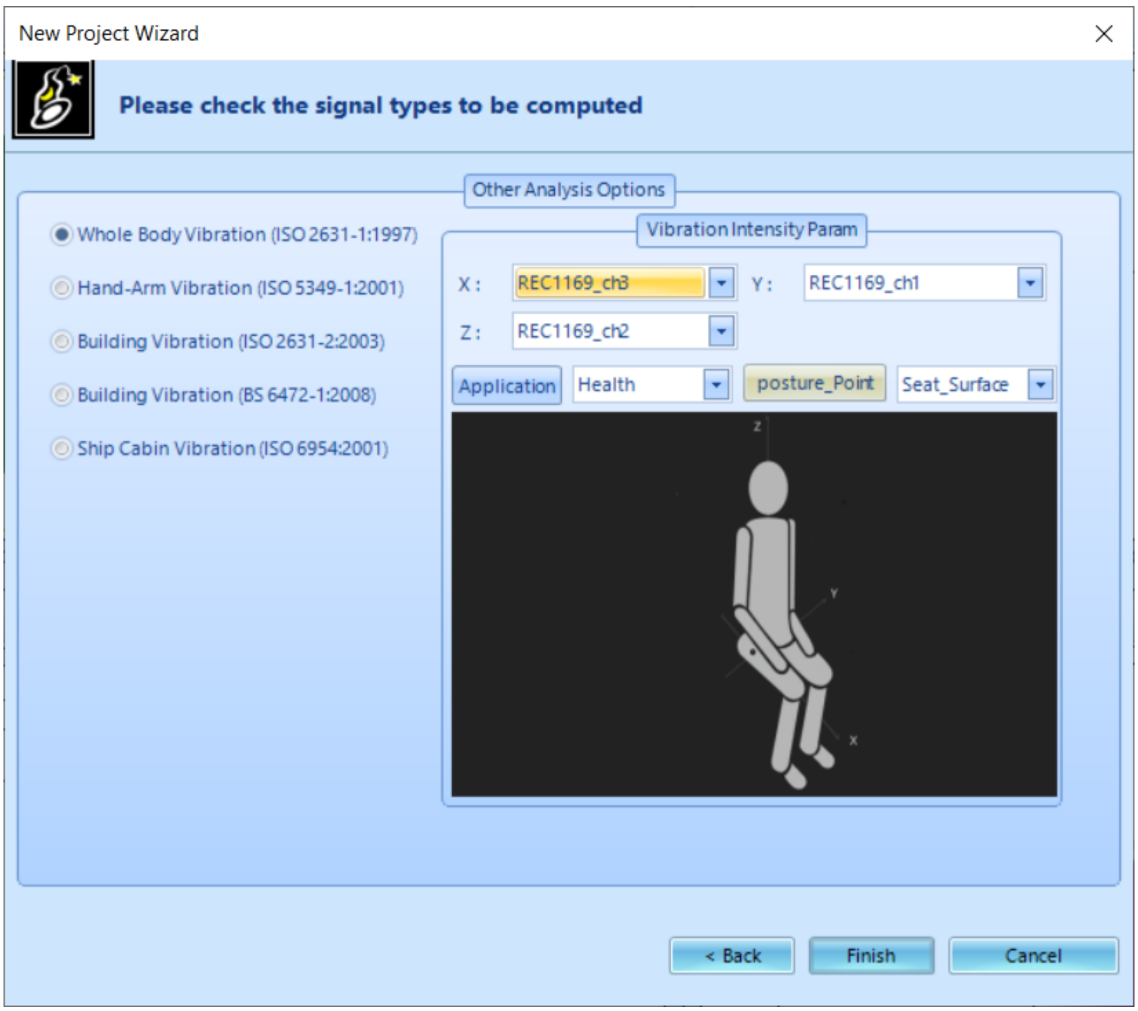
The Differentiation functions are:

1. **Diff:** This module simply differentiates the incoming time stream.
2. **DblDiff:** This module differentiates the incoming time stream twice.

Vibration Intensity

Post Analyzer is capable of processing signals related to **Vibration Intensity** testing. This feature is based on five different standards:

- Whole Body Vibration – ISO 2631 – 1:1997
- Hand-Arm Vibration – ISO 5349 - 1:2001
- Building Vibration – ISO 2631 – 2:2003
- Building Vibration – BS 6472 – 1:2008
- Ship Cabin Vibration – ISO 6954:2001



Each of these standards require at least three time-waveform recordings. In the **New Project Wizard,** a time-waveform recording is assigned to each axis of measurement specified in the Vibration Intensity Param. Moreover, each axis requires a unique recording, one recording cannot be assigned to multiple axes.

The relevant signals generated by Post Analyzer will be Weighted RMS Acceleration, Fourth Power Vibration Dose, Running RMS, Maximum Transient Vibration, and the Crest Factor. These signals are generated in all the standards in the preceding list. However, the weighting filter applied to the time-waveforms varies by standard.

Basics

The weighted RMS of these data sets is required for evaluation and is calculated by Post Analyzer according to the following equation:

$$a_w = \left[\frac{1}{T}\int_0^T a_w^2(t)dt\right]^{\frac{1}{2}}$$

Where $a_w(t)$ is the weighted acceleration as a function of time, and T is the duration of the measurement periods.

The **fourth power vibration dose** value is calculated similarly to the weighted RMS, but uses the fourth power instead of the second, making it more sensitive to peaks in the acceleration data sets. Post Analyzer uses the following equation for this calculation:

$$VDV = \left\{\int_0^T a_w^4(t)dt\right\}^{1/4}$$

The **Running RMS** value is important in cases where transient vibration and shocks are present. These cases can be accounted for by using a small integration time constant. Post Analyzer uses the following equation for this calculation:

$$a_w(t_o) = \left\{\frac{1}{\tau}\int_{t_0-\tau}^{t_0} a_w^2(t)dt\right\}^{1/2}$$

Where τ is the integration time constant for running averaging, and t_o is the observation time.

When the Running RMS values are known, the **Maximum Transient Vibration** value is simply computed with the following equation:

$$MTVV = max\{a_w(t_o)\}$$

The **Crest Factor** is defined as the absolute value of the ratio between the maximum instantaneous peak and the RMS for an acceleration signal. Post Analyzer performs this calculation for each frame of processed data.

Post Analyzer is preconfigured to calculate the RMS, Fourth Power Vibration Dose, Running RMS, Maximum Transient Vibration Value, and Crest Factor for all Vibration Intensity tests. These signals are computed for each axis of acquired acceleration data, and are located in the Measured Signals tab.

Whole Body Vibration

The ISO 2631-1 (1997) is a common standard referred to in Whole Body Vibration (WBV). It defines the procedure for measuring and interpreting WBV in terms of human health and comfort, the probability of vibration perception, and the incidence of motion sickness. Specifically, this standard applies to vibrations introduced to the human body through supporting surfaces.

A W_k filter is applied to x-axis data and y-axis data, and a W_d filter is applied to the z-axis data. The difference in weightings is due to the difference in human perception of vibration in the horizontal and vertical directions.

Hand-Arm Vibration

The ISO 5431-1 (2001) standard defines the procedure for measuring vibration intensity in human hands and arms. Vibration may be transmitted to hands and arms through the use of vibrating tools, vibrating machinery, etc. This standard is used to help establish safety guidelines for workers who may be subjected to hand-transmitted vibration. Additionally, this standard may aid in the advancement of hand-operated tools in an effort to reduce the hazards associated with vibration-related health effects.

The vibration frequencies that concern hand-arm vibration range from 8 Hz to 1000 Hz. Vibrations are measured in the x-axis, y-axis, and z-axis. A W_h filter is applied to the measurements acquired in each axis.

Building Vibration (ISO 2631-2 (2003))

Similar to the ISO 2631-1 (1997) standard, the ISO 2631-2 (2003) standard concerns human exposure to whole-body vibration. However, the posture of the subject is not defined in this standard. Like the two standards described earlier, vibration is measure in the x-axis, y-axis and z-axis, but unlike the ISO2631-1 (1997) standard, these axes are in reference to the structure instead of the human body.

The frequencies of concern range from 1 Hz to 80 Hz, and a W_m filter is used to weight the measurements in each axis of direction.

Building Vibration (BS 6472-1 (2008))

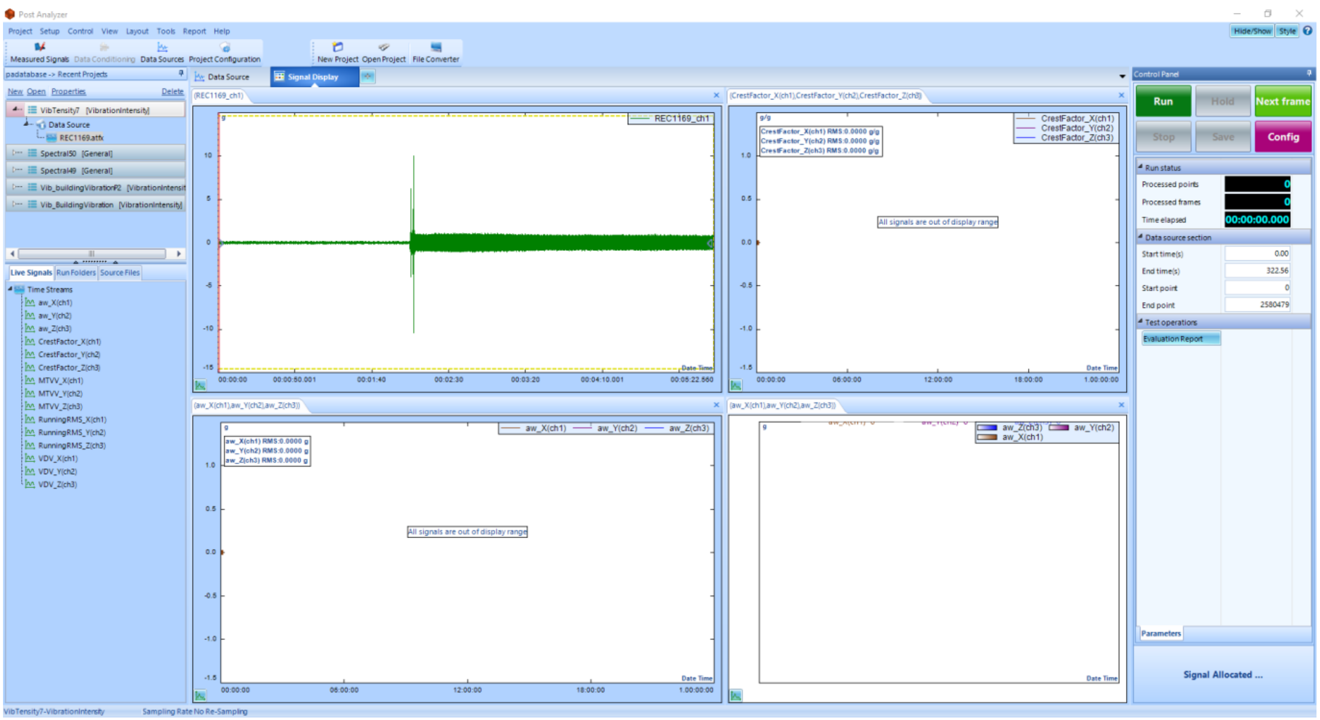
This standard provides guidance for predicting the human response to vibration in buildings within the frequency range 0.5 Hz to 80 Hz. The vibration is measured in three axes defined by the vertical direction, the lateral direction, and the fore and aft direction. The weighting curve applied to the horizontal direction is defined by the W_d filter, and the weighting curve applied to the vertical direction is defined by the W_b filter.

Ship Cabin Vibration (ISO 6954 (2001))

This standard provides a procedure for measuring and evaluating vibration with regards to habitability on passenger and merchant ships. The habitability is evaluated in the frequency range 1 Hz to 80 Hz, with a minimum spectral resolution of 0.2 Hz. The instrumentation calculates the weighted RMS value according to ISO 6954 in terms of vibration velocity. The velocity measurements should be given in mm/s. A velocity spectrum, created by an FFT of the measured time series, should be made available and storable.

Overview

After creating or opening a project, the main Post Analyzer window is displayed. There is a toolbar across the top, and three main sections in the middle. On the left, there are the Recent Projects, Live Signals, and Saved Files views; in the middle are one or more Signal Display views; and on the right is the Control Panel.



Recent Projects

The **Recent Projects** list displays the currently active project and projects previously opened in the application. The current project provides a shortcut toolbar which contains the **Measured Signals** setup, **Data Conditioning** setup (if data conditioning is checked when creating the project), **Data Sources** setup, and **Project Configuration** window. It also displays the data file associated with the project.

Live Signals, Run Folder, and Source Files

Below the **Recent Projects list** is the **Live Signals, Run Folder, and Source Files list**. Live Signals includes the signals in the Data Source file or files, as well as new signals created as part of the analysis. Live signals are not saved to the host computer hard disk but are available for display in the **Signal Display** view tab. Anything saved to disk is displayed in the Saved Files list, which is associated with a directory in the file system. Run Folders contain data for each run, which is saved in a defined file format accessed from the Data Files tab.

Signal Display

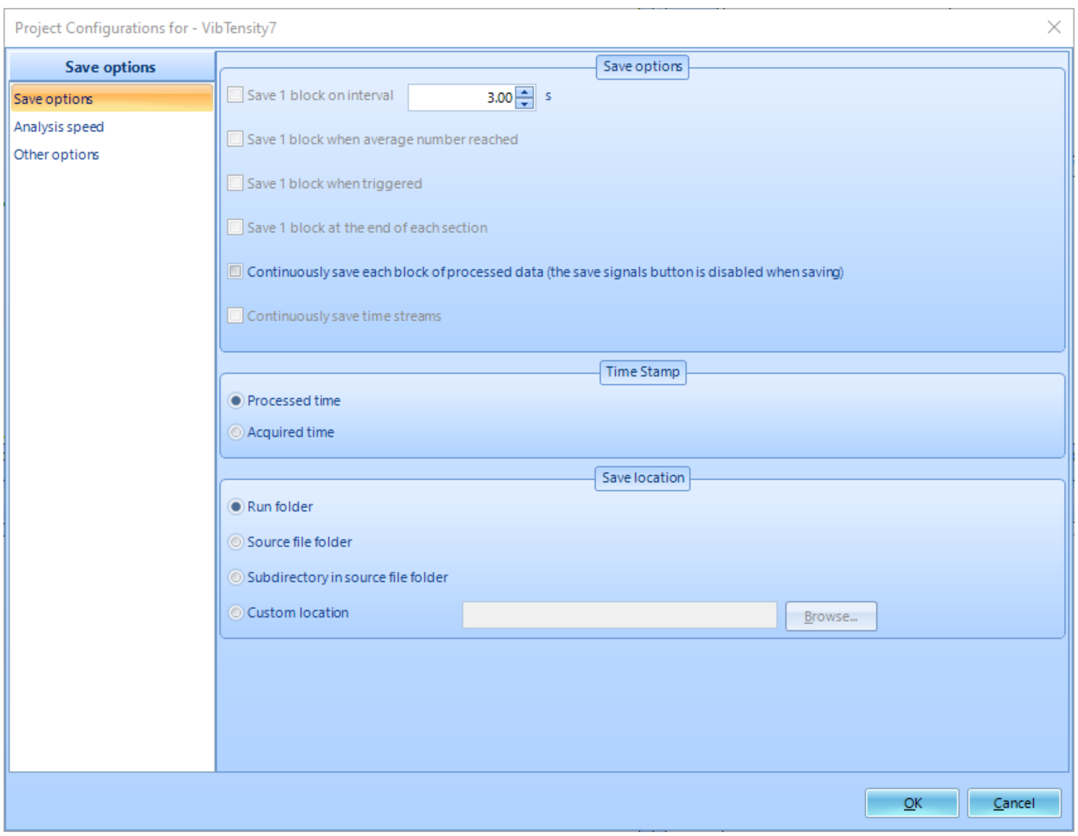
The middle of the main window contains the **Signal Display** view tabs, where live and saved data is displayed. More than one of these tabs can be created, but there always is at least one. Each of these views can contain one or more Display Windows, with a fully customizable layout. These display windows can move freely, resize, and display any valid combination of live or saved signals. New view windows are created by selecting an option in the View menu.

Project Configuration

The **Project Configuration** window has settings related to saving data, analysis speed, and other options related to resampling data.

Save Options

After running the analysis, computed signals are listed under the **Live Signals** tab. However, this data is only stored in a display buffer and must be explicitly saved to the disk to create new data files. There are multiple ways to do this. One option is to manually click the **Save** button on the control panel, which saves data according to the Save settings under the Measured Signals setup. Users can also enable an option (listed in the section below) to automatically save data while the analysis is running.



There are many options to save block data, and one option to save stream data. Most data computed in Post Analyzer is block data, such as APS and SRS spectra. Some data, such as the output of digital filters or other data conditioning blocks, are continuous time streams. Only one type of data can be saved at a time – block or time stream.

Save 1 Block on Interval saves one block of data for every computed signal with the save option selected in the Measured Signals setup per the selected interval.

Save 1 Block when Average Number Reached saves one block of data every time the number of processed blocks reaches the average number set under the Analysis Parameters.

Save 1 Block when Triggered saves one block of data when a trigger occurs. Set up the trigger by clicking the Setup Trigger button in Analysis Parameters.

Save 1 Block at the End of Each Section saves one block at the end of an analysis section.

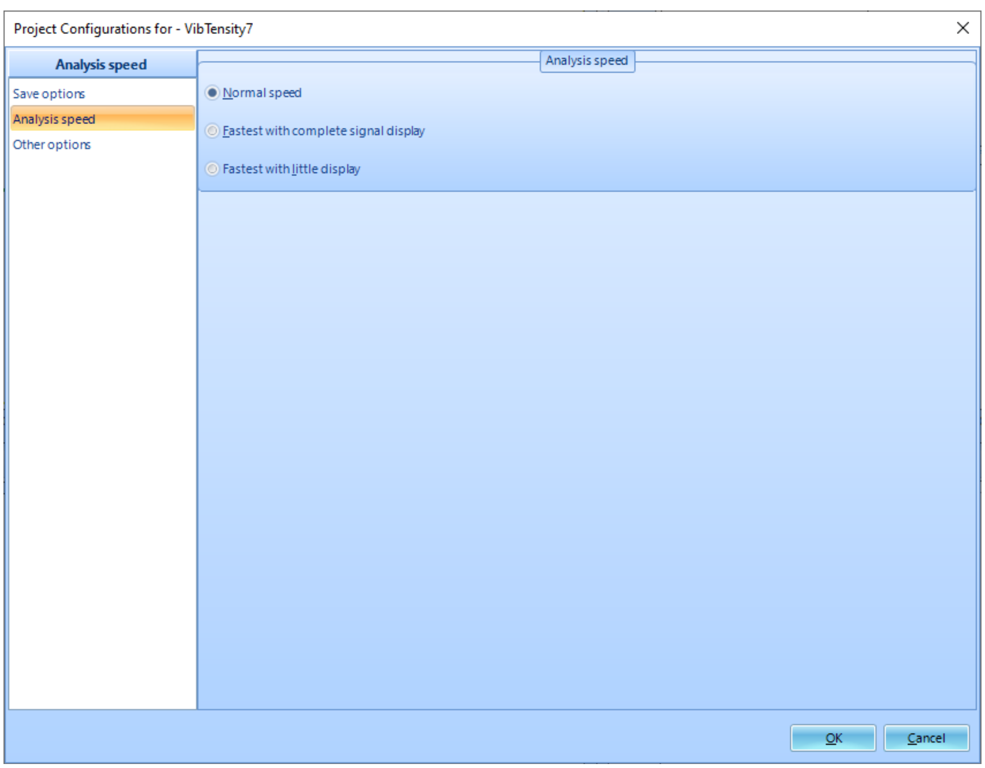
Continuously Save Each Block of Processed Data saves every block of computed data.

Continuously Save Time Streams saves all time stream data. Selecting this option disables all options related to block data saving.

The **Time Stamp** section is used to set the time axis on time-based signals. The axis can either display the time relative to the start of the test or to the actual date and time the data was acquired.

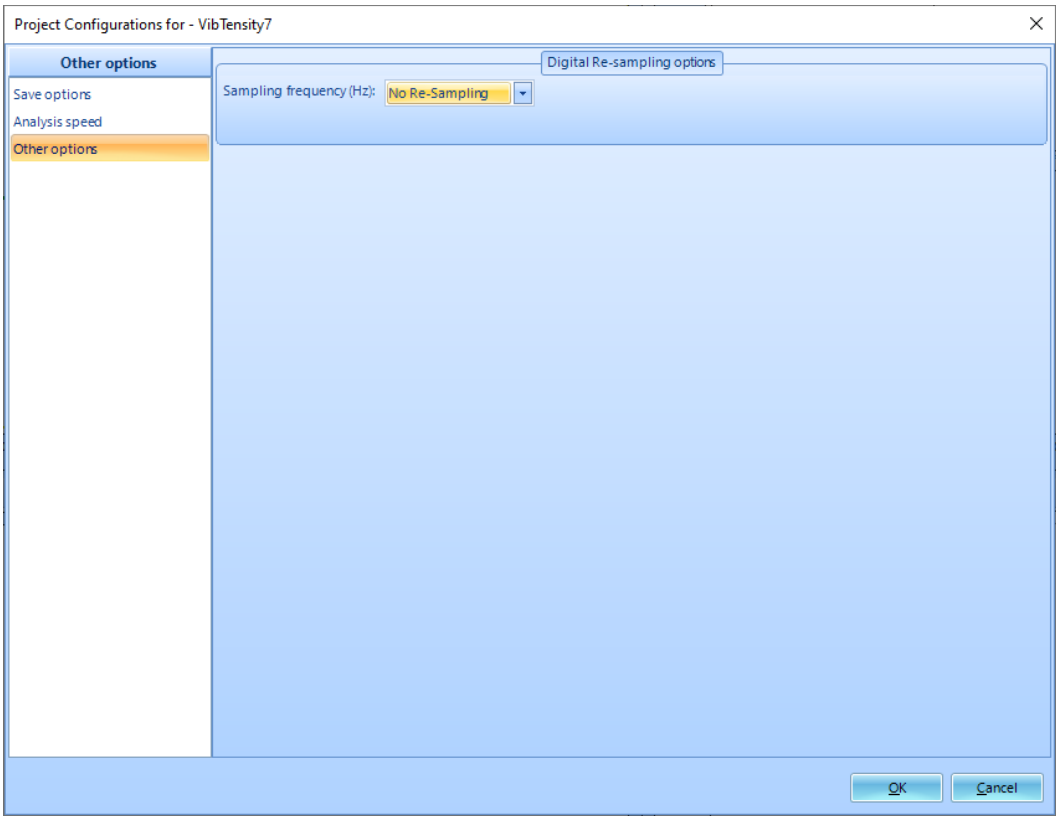
The Save Location section allows users to select the file system location used to create new data files from the following options: current run folder, same folder as data source file folder, subdirectory in source file folder, or a custom location.

Analysis Speed



Analysis Speed adjusts the length of processing time used for the signal display during post-processing. Increased display details result in less available processing resources for post processing data analysis. **Normal Speed** is the default setting that balances data processing and the signal display. **Fastest with Complete Signal Display** prioritizes the display, and **Fastest with Little Display** prioritizes data processing.

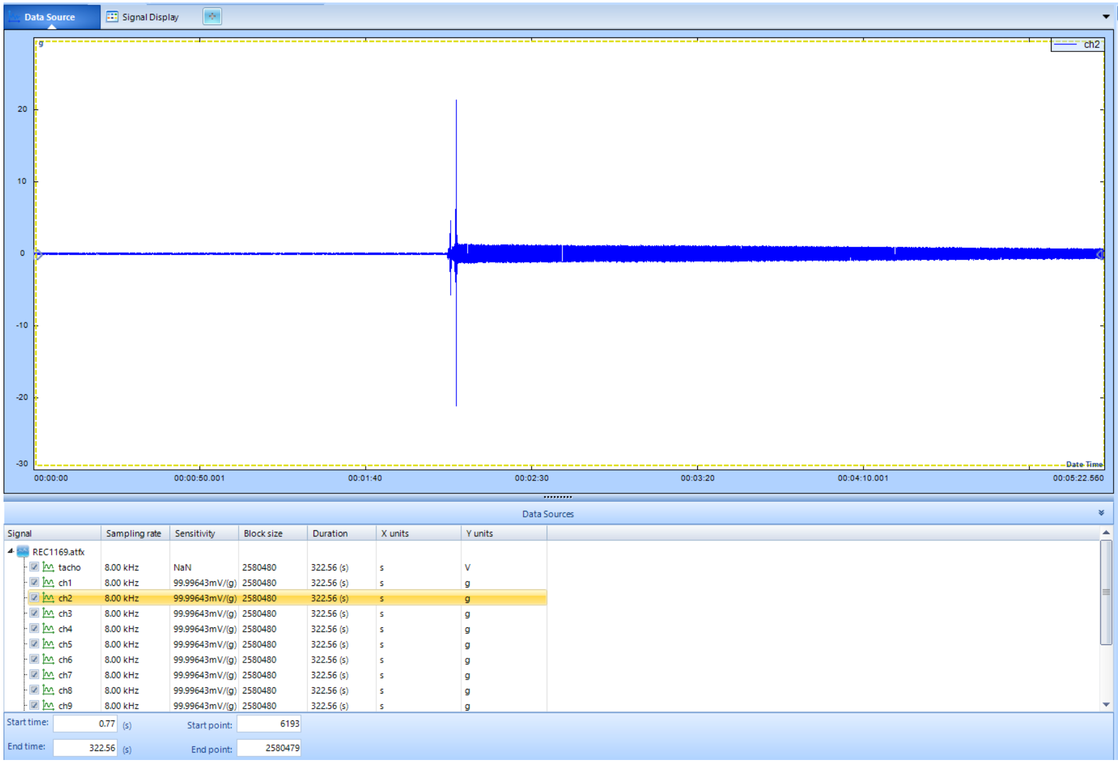
Other Options



Digital Resampling allows users to resample all source signals according to the selected sampling rate to meet the requirement of other interface. Sampling rate stages are available for selection from the dropdown menu.

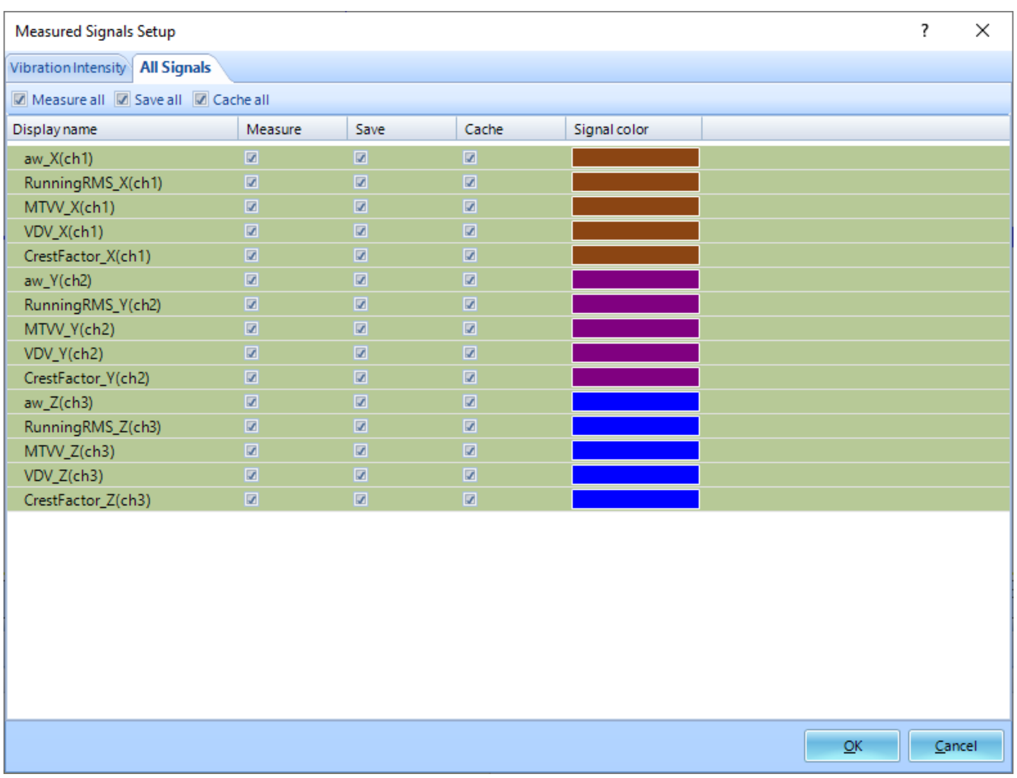
Data Source

The **Data Source** tab shows the source data from the file or files associated with the current project. Set the time range for analysis by dragging the edges of the yellow box. Enter the start and stop time or point manually into the text fields below.



Measured Signals

Every signal available for computing is listed under the **Setup->Measured Signals** tab.

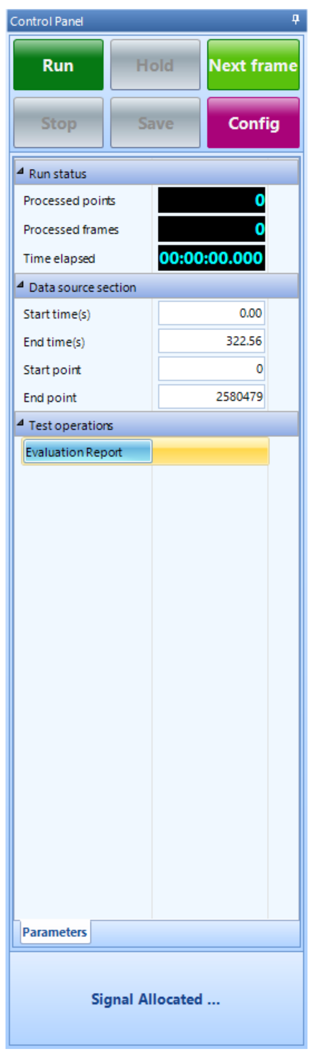


Control Panel

The **Control Panel** is used to control post processing operations. It also displays the analysis status information.

The following control buttons are provided from the Control Panel:

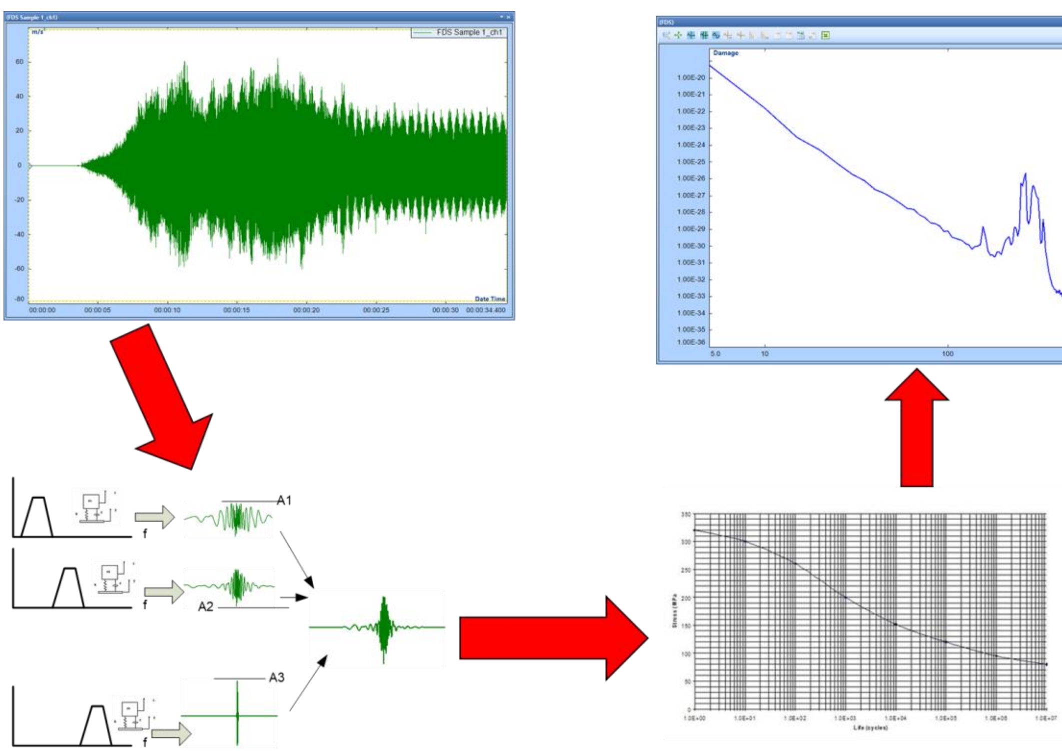
- **Run** starts the analysis
- **Hold/Continue** pauses and resumes the analysis
- **Stop** stops the analysis
- **Next Frame** processes one frame, and then stops
- **Save** saves block signals
- **Config** opens the Project Configuration window
- **Run Status** panel shows the following status information
- **Processed Points #** is the number of processed samples
- **Processed Frame #** is the number of processed frames
- **Time Elapsed** is the total time duration of the input signals processed



Fatigue Damage Spectrum

Fatigue Damage Spectrum (FDS) creates accelerated life cycle testing profiles for vibration tests. The FDS function reduces testing time by calculating the quickest path to destruction or damage. Testing time is accelerated by concentrating the random or swept sine energy, depending on the FDS calculation, to where it will induce the most fatigue damage. The FDS function can determine the amount of damage on a tested object. Combine information gleaned from the FDS function with other parameters to further reduce testing time.

The overall fatigue damage spectrum workflow is as follows. First, vibration data must be measured to obtain a time history. The obtained time history becomes an input to a system of SDOF (Single-Degree-Of-Freedom) resonators. Each resonator has a particular natural frequency. The different responses for each resonator are relative acceleration values. These values become the inputs into a cycle count process (called Rainflow Counting). The relative accelerations are used to calculate the stress that the system undergoes. Once this stress response is known, SN-Curve information is used to determine the stress-cycle relationship for the system. The Rainflow Counting algorithm then calculates the number of stress-loading cycles. This assortment of cycles then factors into the damage calculation, adjusted by the Fatigue Damage parameters. The result is the FDS.



$$\sum_{i=1}^{k} \frac{n_i}{N_i} = D$$

FDS Process

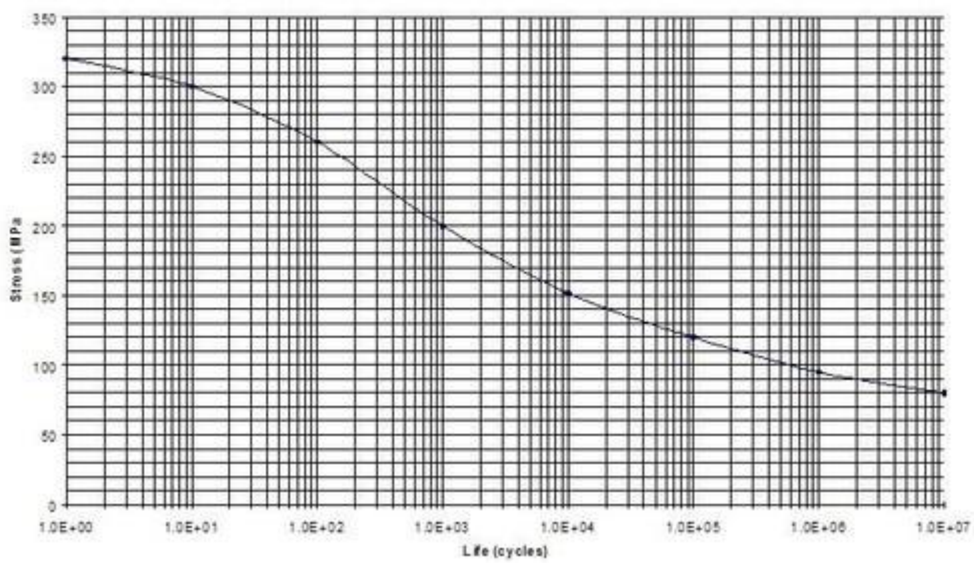
Additional information regarding different components of the FDS calculation process are provided in the following sections.

Damage

The damage caused by vibration can come from one of the following conditions:

- Damage caused by an extreme instantaneous stress on an object.
- Damage caused by fatigue when a high amount of stress is applied to the object.

FDS represents a spectrum of the fatigue damage on an object. The FDS function uses the S-N curve to construct the spectrum. The S-N curve represents the stress applied to a material (S) versus the number of cycles of applied stress. The S-N curve of aluminum is illustrated in the following figure:



S-N Curve of Aluminum

Rainflow Counting

Rainflow Counting is a method used to determine the number of fatigue cycles present in a load-time history. A fatigue cycle is the loading and unloading of a DUT. It will weaken and eventually fail with enough repeated cycles. Users can calculate the fatigue damage for a part subjected to a cyclical load of constant amplitude using the SN-Curve of the material and Miner's Rule. However, the number of cycles and their respective amplitudes are not easily determined in a real-world load-time history.

A **load-time history** typically consists of force versus time, or strain versus time. Rainflow counting is then used to extract the number of cycles, their respective range, and mean.

The Rainflow Counting method extracts fatigue cycles from any load-time history. The result is a Rainflow matrix and residue. The following information is preserved from the time history:

- Number of cycles

- Range of cycles
- Mean of cycles

Analytical Formulation of the Fatigue Damage Spectrum

Move on to the FDS analytical model after the number of fatigue cycles are extracted from the S-N curve. Start with an understanding of how the system under investigation responds to a shock to accomplish this task.

The Shock Response Spectrum (SRS) is a Frequency Response Function (FRF) that describes the frequency response of a system to a shock or a transient event. Its analog in the time domain is the Extreme Response Spectrum (ERS), but the ERS predicts a system response due to a lengthier loading duration. The equation for the ERS of an acceleration response for a system is:

$$ERS_{Accel}(f_n) = [G_z(f_n) \cdot \ln(T f_n) \cdot Q\pi f_n]^{\frac{1}{2}}$$

Where:

$$f_n \rightarrow \text{natural frequency of an SDOF system}$$
$$T \rightarrow \text{excitation duration}$$
$$Q \rightarrow \text{dynamic amplification factor}$$
$$G_z(f_n) \rightarrow \text{acceleration input PSD value at } f_n$$

The FDS builds on assumptions of the SRS and ERS. The FDS is an FRF that describes cyclic fatigue damage of a system. The equation for the FDS is:

$$FDS(f_n) = \Gamma\left(\frac{b}{2}+1\right) \cdot \frac{k^b T f_n}{C} \cdot \left[\frac{Q G_z(f_n)}{2(2\pi f_n)^3}\right]^{\frac{b}{2}}$$

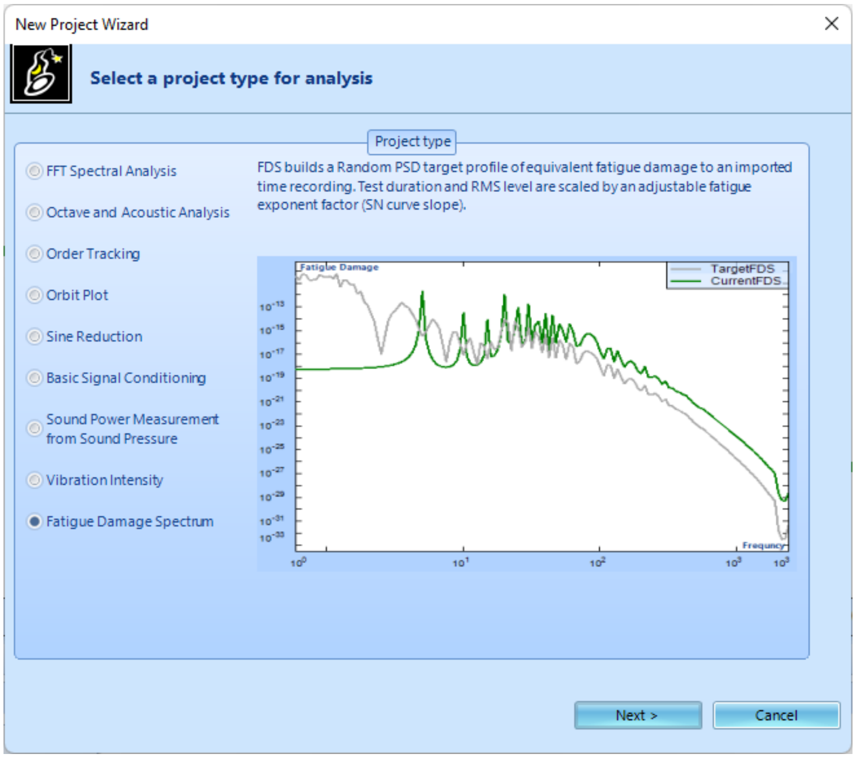
Where:

$$\Gamma \rightarrow \text{Gamma function,} \qquad \Gamma(g) = \int_0^\infty x^{g-1} \cdot e^{-x} dx$$
$$b, C \rightarrow \text{fatigue parameters related by } N = \frac{C}{S^b}$$
$$S \rightarrow \text{Cyclic Stress Amplitude}$$
$$N \rightarrow \text{\# of cycles to failure at } S$$
$$k \rightarrow \text{system spring stiffness using the SDOF model}$$

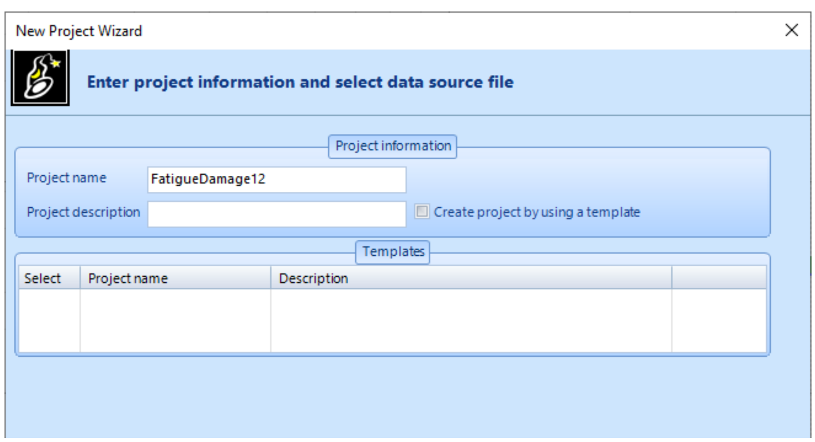
The FDS can also be calculated using a time-dependent input. Similar to the SRS, the FDS for a range of frequencies can add together to obtain the lifetime fatigue damage of a system. The obtained FDS can be inverted into a test PSD. This PSD can be used for accelerated vibration testing, with the potential to impart a whole lifecycle of damage to a system with a single test.

Create a Fatigue Damage Spectrum Project

Open **the New Project Wizard** to start a new project and select **Fatigue Damage Spectrum** as the project type. Click **Next** to continue.



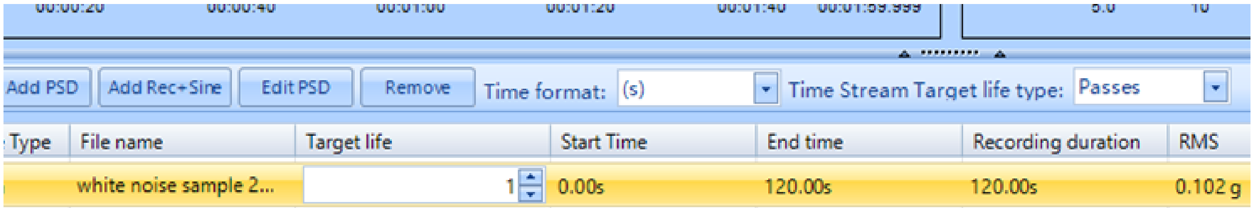
In the following window, enter a name for the project and optionally add a description or set up the test based on a template.



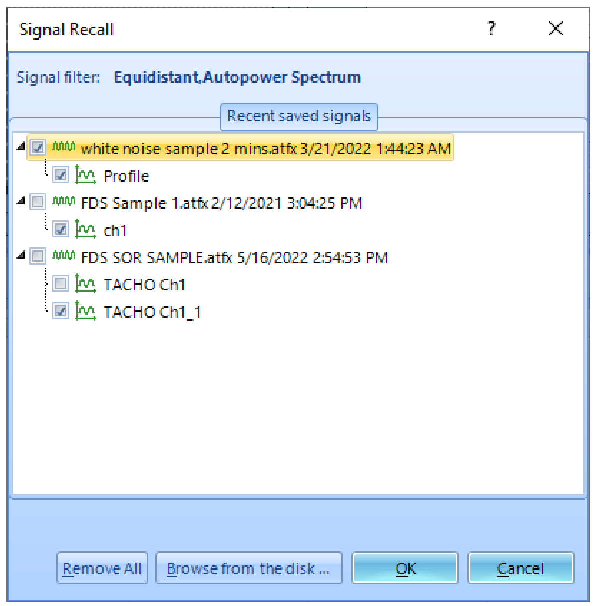
Overview

Importing Signals

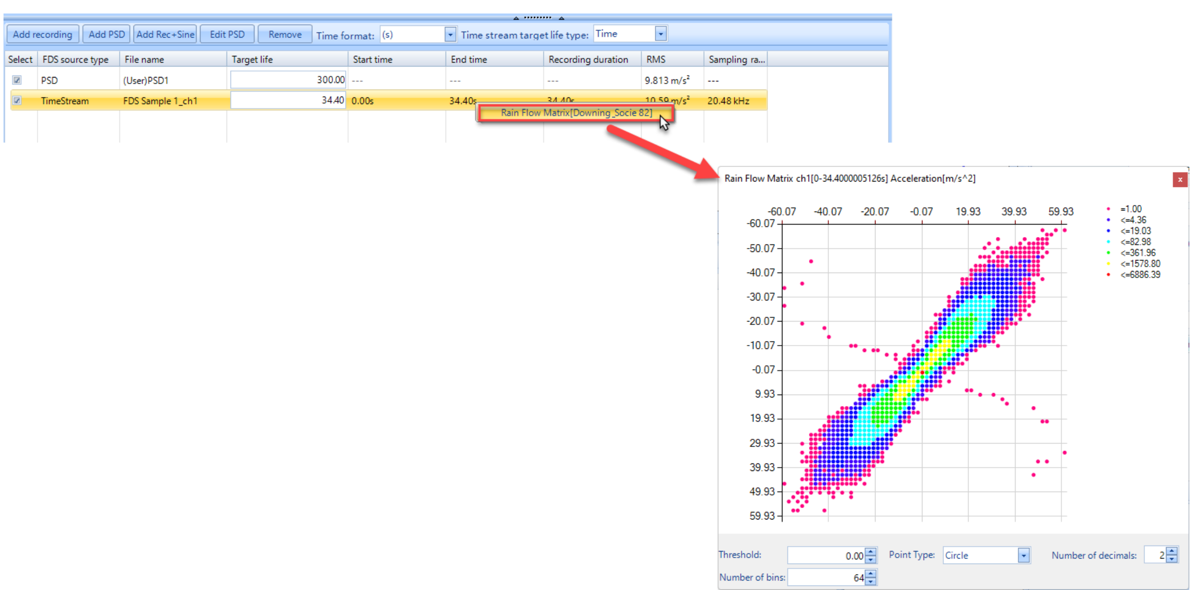
Users can select and add one or more signals to the FDS analysis on the start page of a new test.



Add Recording – add any time-based recording signal from .atfx, .csv, .txt or other universal file formats.



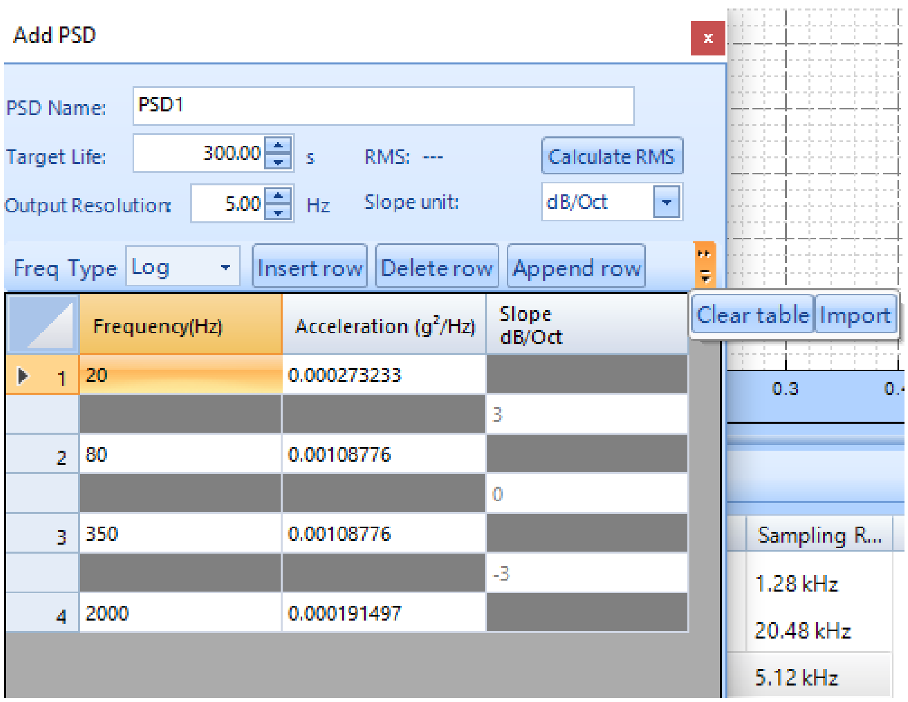
Rainflow Counting – after a time-based signal is added, right-click the signal to generate a Rainflow Matrix according to the Downing-Socie 82 algorithm.



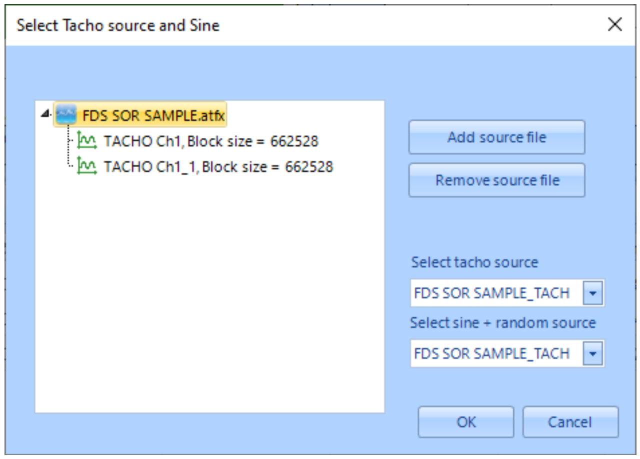
The Rainflow Matrix settings are:

- **Threshold:** determines the difference between a peak or a valley for the Rainflow counting algorithm. This is an absolute value number for the desired Engineering Unit.
- **Point Type:** setting can be used to change the shape of the points to either circles or squares.
- **Number of decimals:** adjusts the amount of decimal places for all numbers.
- **Number of bins:** increase or decrease the amount of fixed amplitude ranges used to map the data.

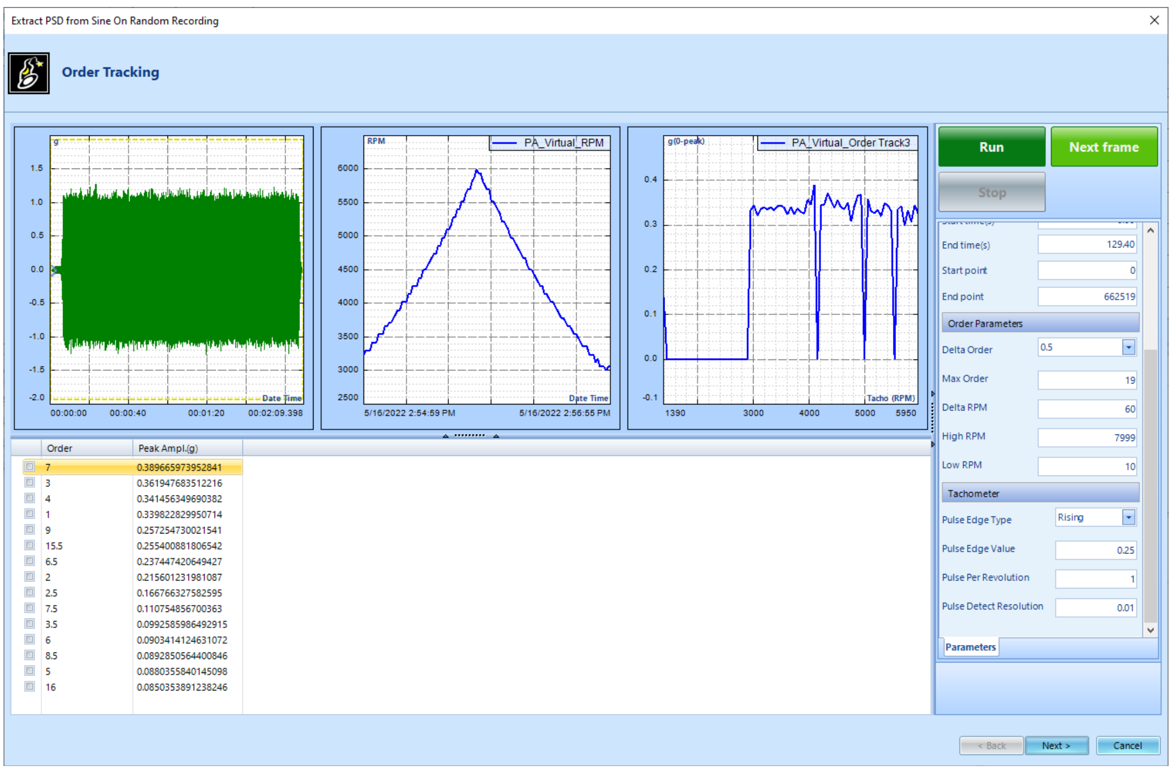
Add PSD – add a PSD signal through the breakpoint table. Import a CSV or a PSD with any universal file format.



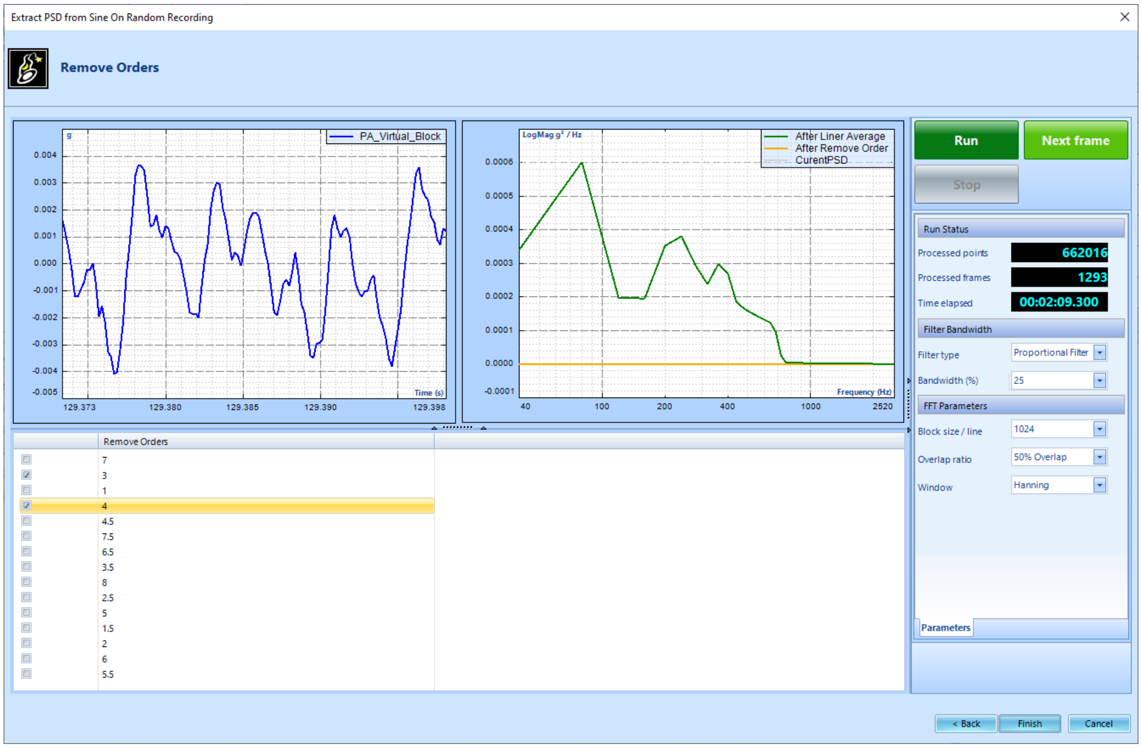
Add Rec + Sine – add a signal with a tachometer signal to extract sine tones from the recording and individually analyze random and sine components for Fatigue Damage analysis.



Specify the tachometer parameters and PA will extract the top 15 orders.



Users can then select orders to filter out of the broadband as independent Sine tones.



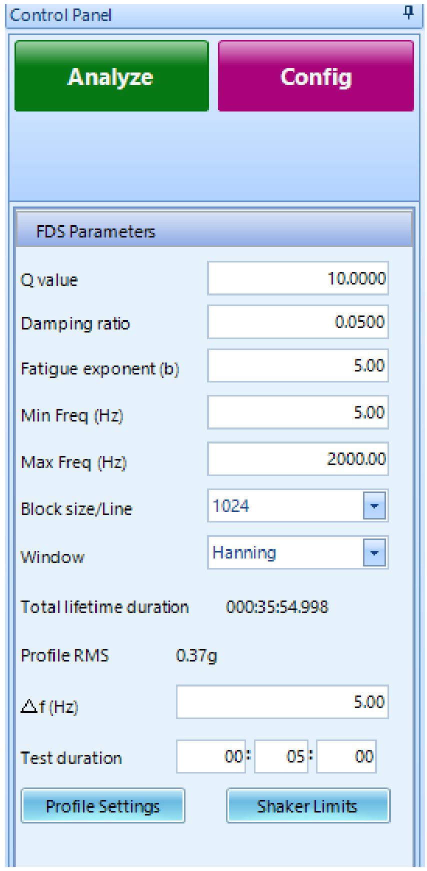
PA will then create 2 signals one with the filtered broadband signal and the other with the damage from the sine tones.

Add Recording | Add PSD | Add Rec+Sine | Edit PSD | Remove | Time format: (s) | Time Stream Target life type: Passes

Select	FDS Source Type	File name	Target life	Start Time	End time	Recording duration	RMS	Sampling R...
☑	TimeStream	white noise sample 2...	1	0.00s	120.00s	120.00s	0.102 g	1.28 kHz
☑	TimeStream	FDS Sample 1_ch1	1	0.00s	34.40s	34.40s	1.079 g	20.48 kHz
☑	TimeStream	FDS SOR SAMPLE_TAC...	1	0.00s	129.40s	129.40s	0.4366 g	5.12 kHz
☑	PSD	FDS SOR SAMPLE_TAC...	129.40	---	---	---	0.4072 g	5.12 kHz
☑	SineDamage	FDS SOR SAMPLE_TAC...	129.40	0.00s	129.40s	129.40s		---

Control Panel

The Control Panel is located to the right of the main work display which contains parameters for FDS analysis and controls the analysis process flow.



Q Value – Q factor determines the sharpness of the filter used and is inversely correlated to the damping factor of the system. These measurements are typically obtained at the primary resonance of the DUT.

Damping Ratio – usually refers to the damping ratio of the primary resonance of a DUT. Users will only need to the set Q or damping ratio, since these are interlinked quantities.

Q = 1/ (2*damping ratio)

Fatigue Exponent - calculated from the slope of the high cycle SN curve in log-log format. The mathematical definition for the fatigue exponent is $\frac{1}{b}$, where b is:

$$b = \frac{Log\left(\frac{S_L}{S_E}\right)}{Log\left(\frac{N_L}{N_E}\right)}$$

Where S_L is the stress amplitude, low cycle fatigue becomes high cycle fatigue, S_E is the stress amplitude, high cycle fatigue becomes the endurance limit, N_L is the number of cycles that it takes to reach high cycle fatigue, and N_E is the number of cycles required to reach the endurance limit.

The convention for the fatigue exponent varies from source to source. Careful attention must be given to ensure that the value used for the fatigue exponent is derived using the provided conventions (in preceding paragraph). If not, the proper recalculations must be made.

Min – Max Frequency – determines the max limits of the PSD generated after FDS analysis is performed.

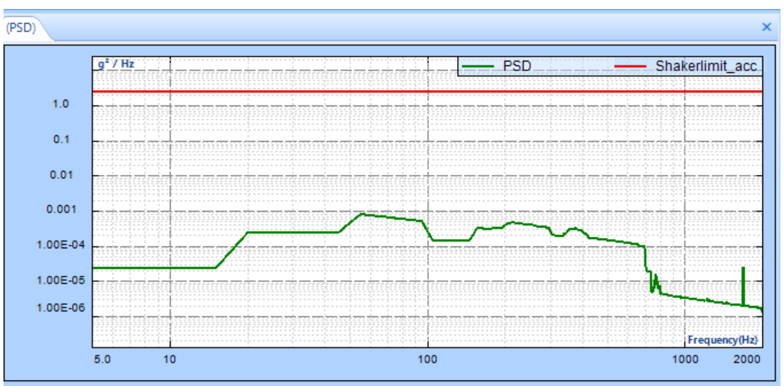
Block Size/Lines – defines the size of the time blocks as a number of samples and the number of spectral lines used in the Fourier Transform frequency domain of a signal.

Window – allows users to select a window to apply during FFT operation. Windowing functions can help reduce leakage and increase the precision of a frequency measurement.

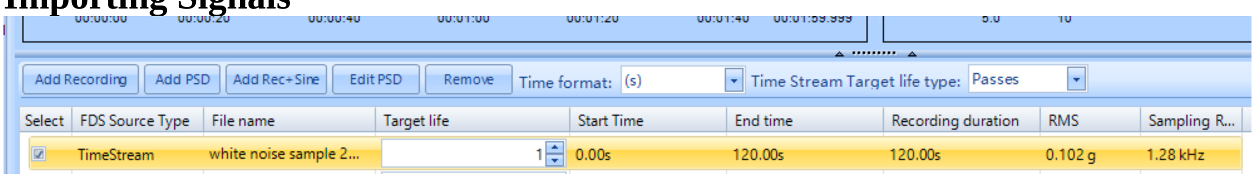
Delta f – allows users to define the frequency resolution of a time accelerated profile that can be exported.

Test Duration – allows users to select the test duration used to accelerate the PSD to run within a specified time while matching the overall damage imparted to the DUT over an expected lifetime operation.

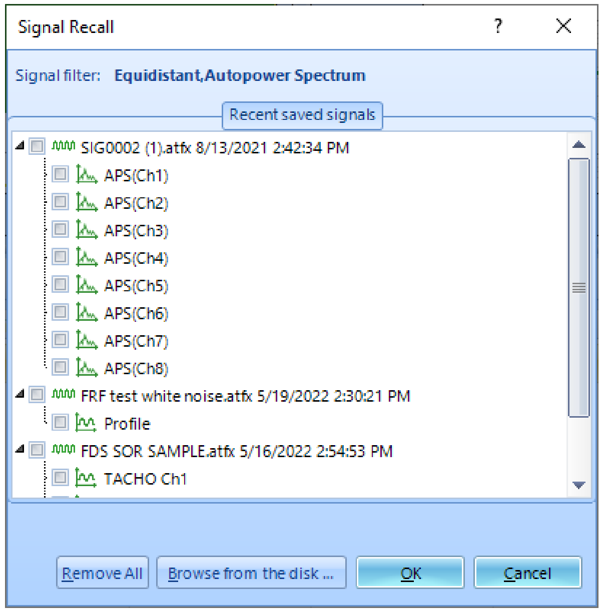
Shaker Limits – displays shaker limits overlaid with the time accelerated PSD profile.



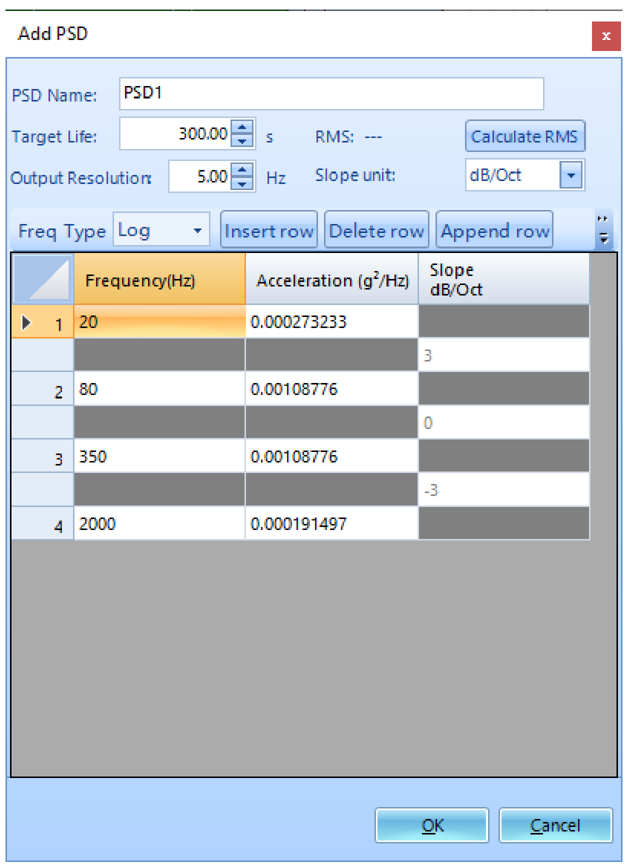
Importing Signals



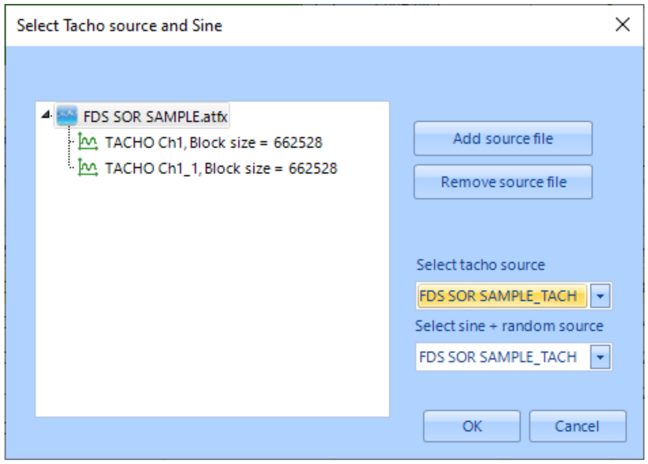
Add Recording – users can select already loaded signals (Time or PSD) to recall or browse to load signals from the computer.



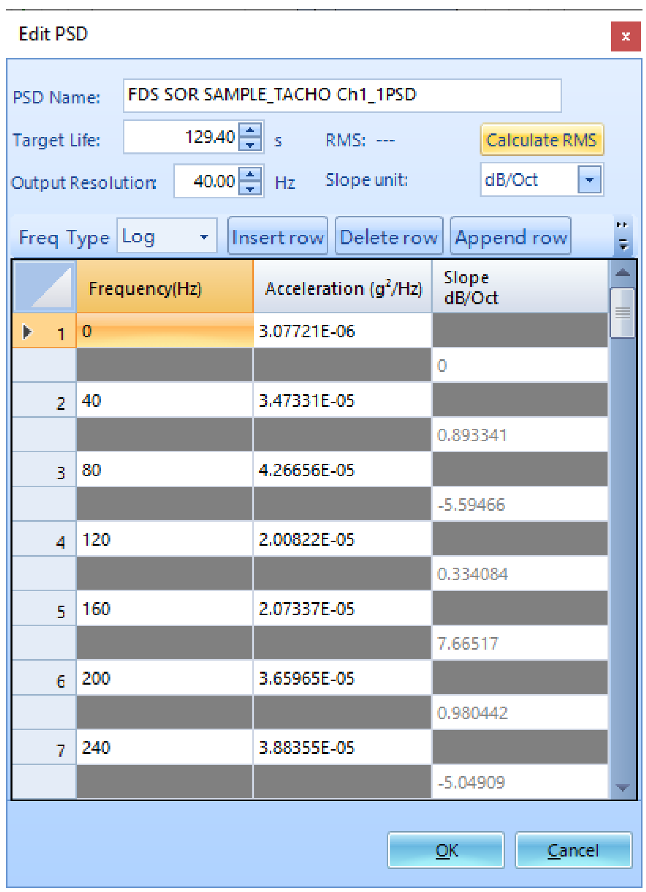
Add PSD – users can import a PSD by manually entering a breakpoint table or importing it from a .csv file.



Add Rec +Sine – allows the user to add a time waveform with dominant sine tones and a tachometer signal. Provides option to filter out and extract sine tones.



Edit PSD – select any existing PSD from the list and edit the profile of the signal.

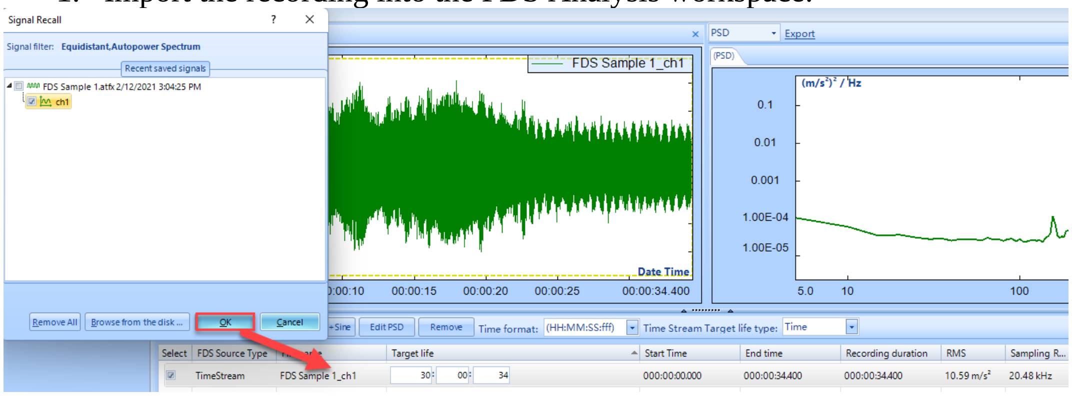


FDS Workflow

FDS of Random Broadband Profile

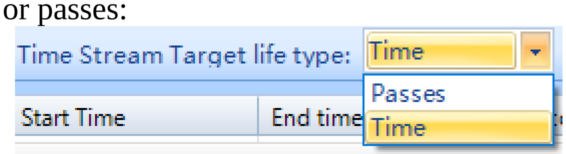
The process to perform FDS analysis of a raw time recording is as follows:

1. Import the recording into the FDS Analysis workspace:



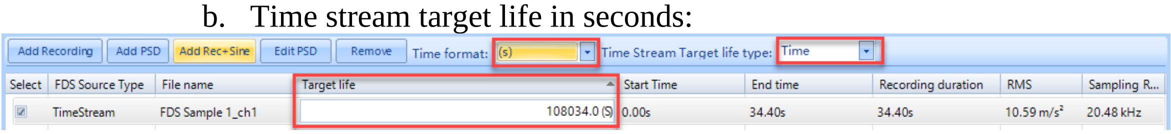
   a. Multiple recordings can be imported. The RMS levels and sampling rates of the imported recordings do not have to be identical.

2. Configure the Time Stream Target life of all imported time streams in terms of either time or passes:

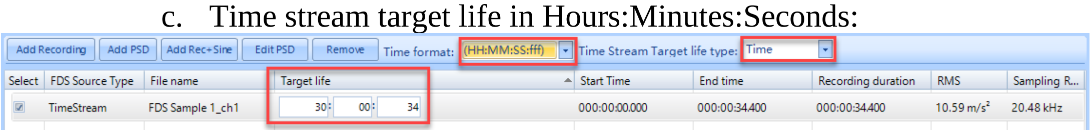


   a. The target life type can be configured in terms of time if the duration of the time waveform vibration needs to be adjusted.
   b. The target life type can be configured in terms of passes if the time waveform vibration is more cyclic or path dependent.

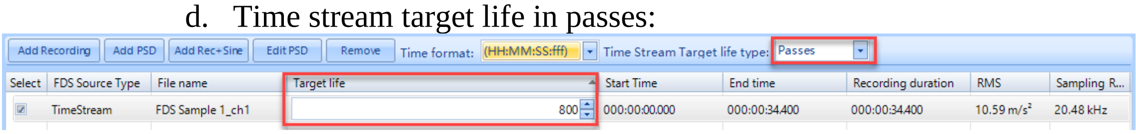
3. Set the target life of the time waveform recording in the FDS workspace:
   a. This is based on what is selected in Step 2.
   b. Time stream target life in seconds:



   c. Time stream target life in Hours:Minutes:Seconds:



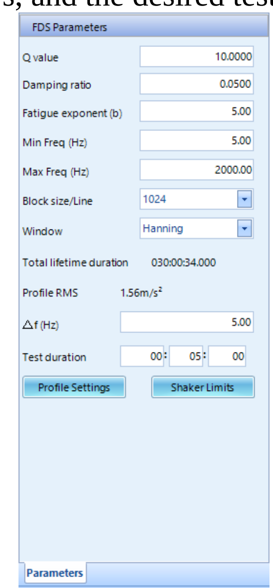
   d. Time stream target life in passes:



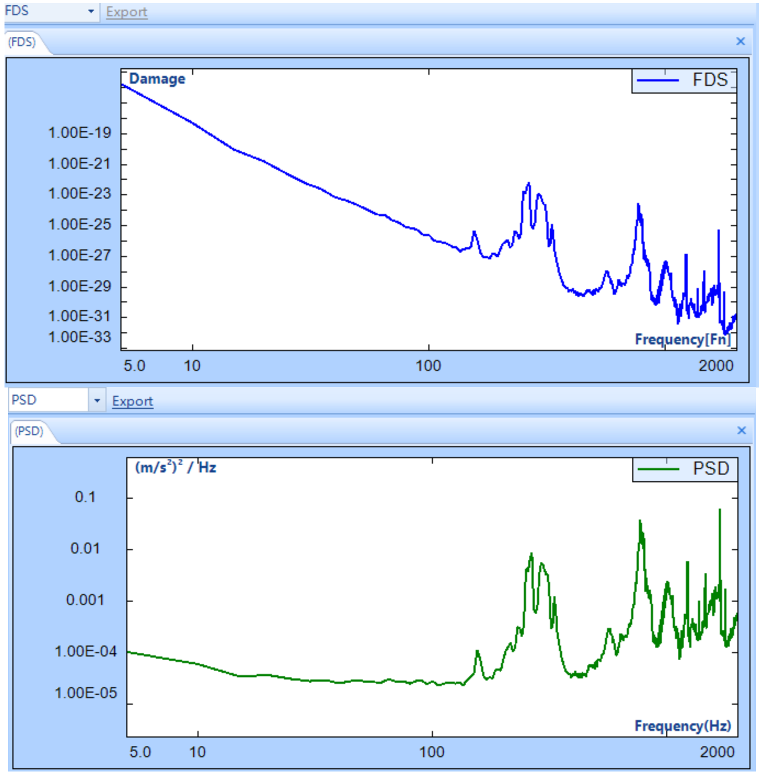
   e. Next to the imported file space is the Signal Properties pane. This pane provides a comprehensive understanding of the mechanical, statistic, and signal processing properties of any signal that is selected in the imported file space:

Signal Properties	
Signal name	ch1
Duration	34.400 s
Total points	704513
Sampling rate	2.048E+04 Hz
Acceleration (Positive)	62.18 m/s²
Acceleration (Negative)	-60.07 m/s²
Acceleration (RMS)	10.59 m/s²
Velocity (Positive)	0.005008 m/s
Velocity (Negative)	-0.005348 m/s
Velocity (RMS)	0.000789 m/s
Displacement (Positive)	0.0002558 m
Displacement (Negative)	-0.000353 m
Displacement (RMS)	7.042E-05 m
Kurtosis	4.404
Skewness	0.0004023

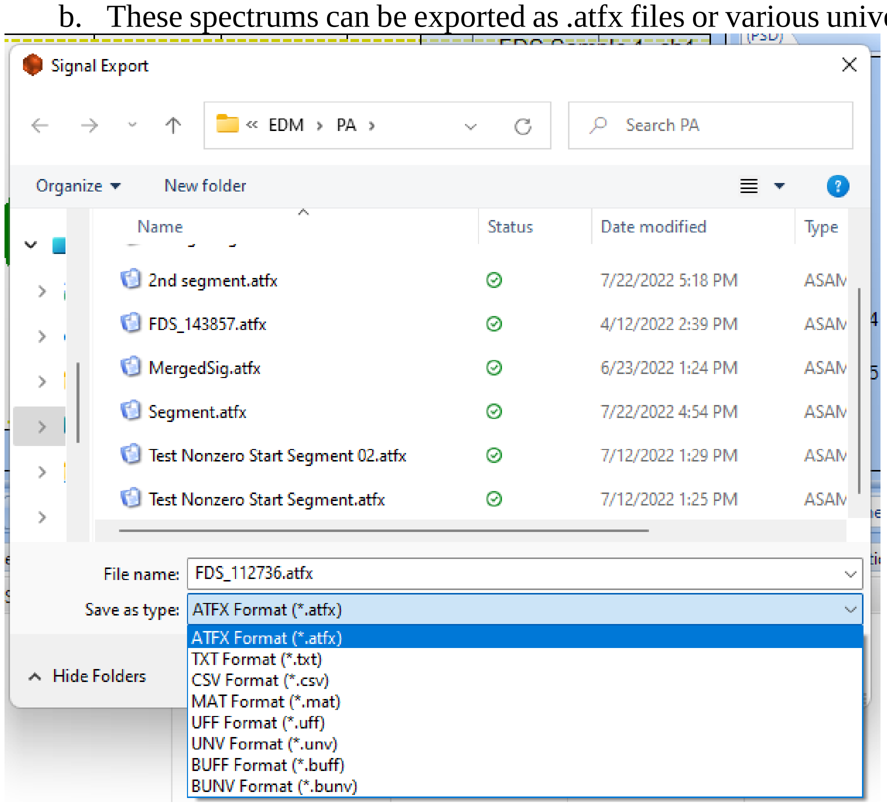
4. Configure the FDS parameters.
    a. In this step, the settings will be configured based on the S-N curve, desired FDS parameters, and the desired testing duration:



    b. Saving and various other settings can be set in **Config.**
5. Select Analyze.
    a. The calculated FDS and PSD can be viewed and then exported via **Export:**

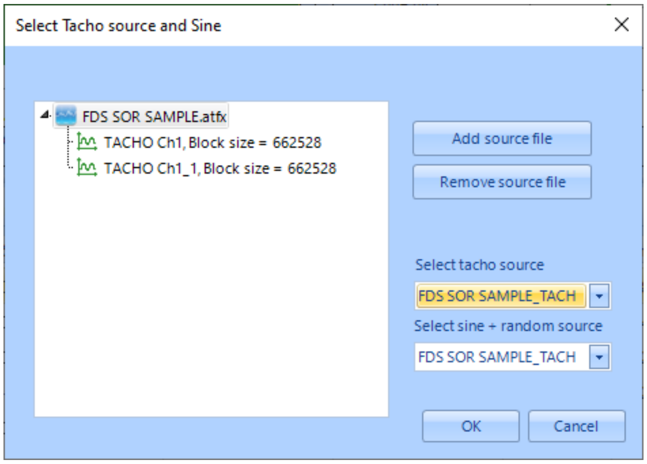


b. These spectrums can be exported as .atfx files or various universal file types:

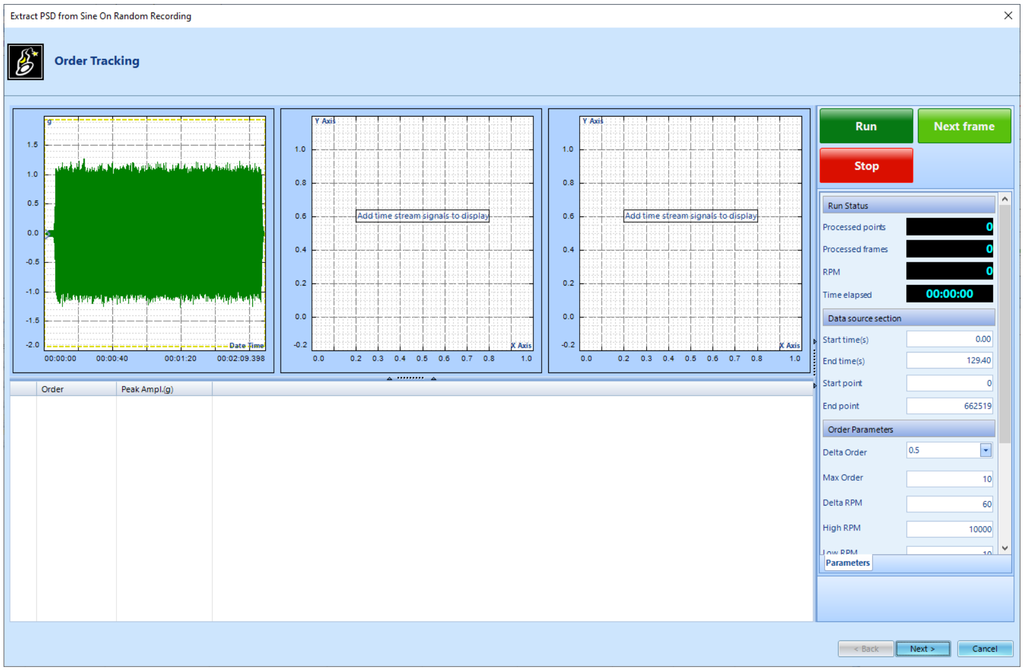


Extracting Sine tones from Sine-dominated broadband signals

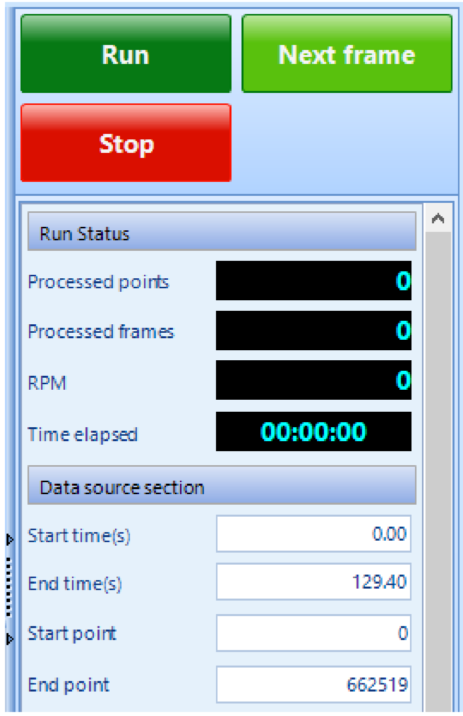
Add Rec +Sine – allows the user to add a time waveform with dominant sine tones and a tachometer signal. Provides an option to filter out and extract the sine tones.



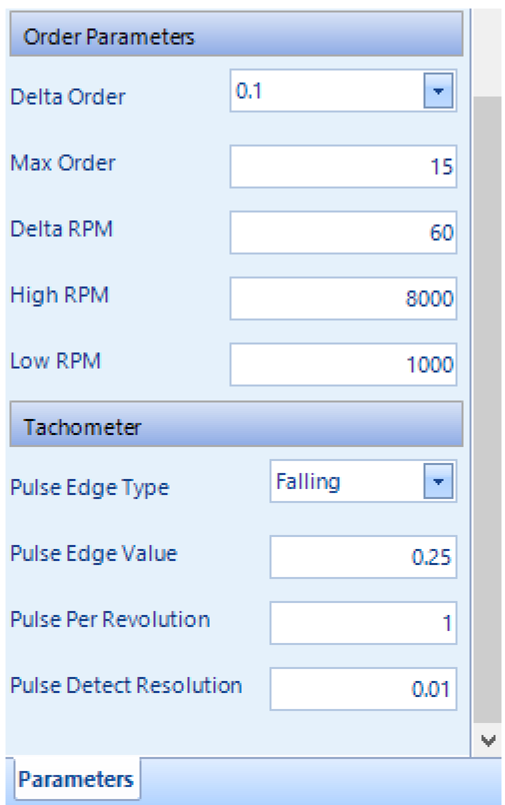
Select the Tach channel and then the Sine+Random signal.

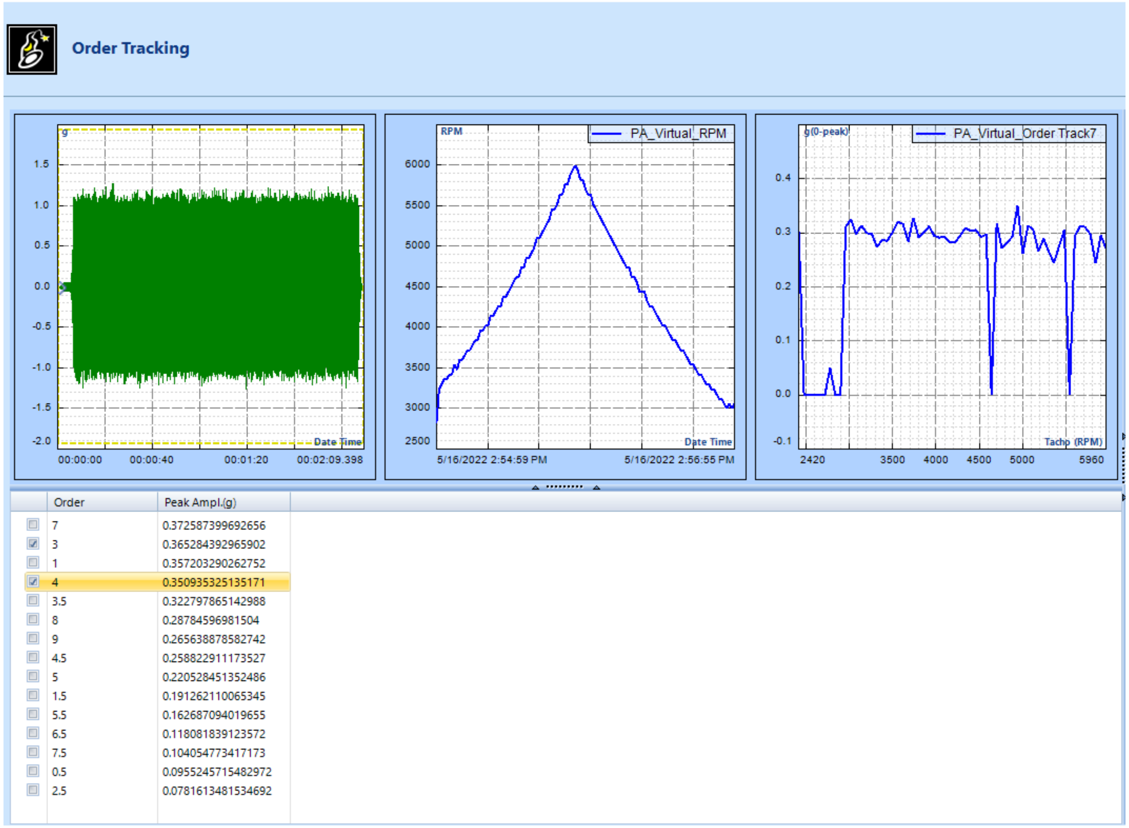


Users can set the start and the end time for recording and the analysis controls.



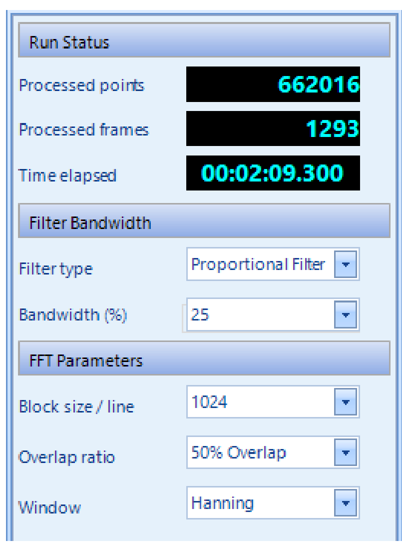
Set the order and tachometer parameters for sine extraction.



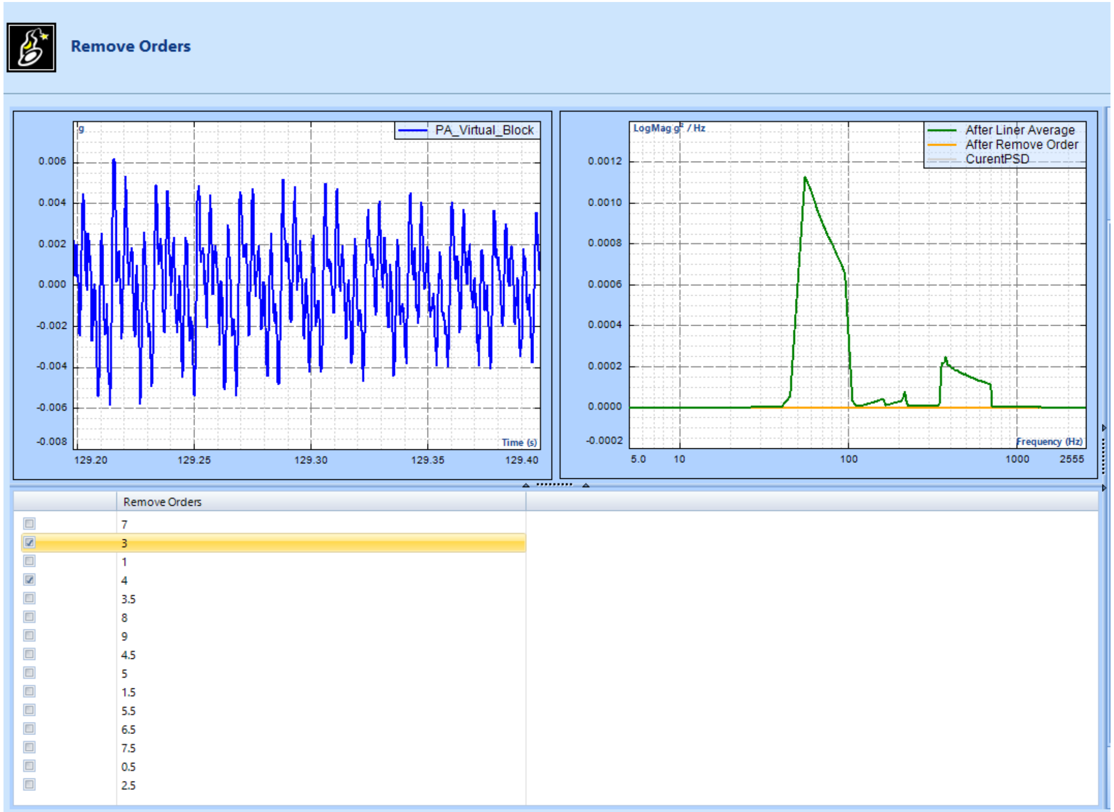


The top 15 orders are displayed for the user to select and filter.

Then set the parameters for the Sine filter and the broadband parameters.



PA will display the PSD of the filtered and unfiltered PSD for the user to review. The user can then add or remove tones and rerun the analysis.



Click **Finish** to import the PSD and Sine damage.

FDS Accelerated Profile

Add Recording | Add PSD | Add Rec+Sine | Edit PSD | Remove | Time format: (s) | Time Stream Target life type: Time

Select	FDS Source Type	File name	Target life	Start Time	End time	Recording duration	RMS	Sampling R...
☑	TimeStream	white noise sample 2...	120.0 (S)	0.00s	120.00s	120.00s	0.102 g	1.28 kHz
☐	TimeStream	FDS Sample 1_ch1	34.4 (S)	0.00s	34.40s	34.40s	1.079 g	20.48 kHz
☑	TimeStream	FDS SOR SAMPLE_TAC...	10000.0 (S)	0.00s	129.40s	129.40s	0.4366 g	5.12 kHz
☑	PSD	FDS SOR SAMPLE_TAC...	129.4 (S)	---	---	---	0.4072 g	5.12 kHz
☑	SineDamage	FDS SOR SAMPLE_TAC...	129.4 (S)	0.00s	129.40s	129.40s		---
☑	PSD	FDS SOR SAMPLE_TAC...	129.4 (S)	---	---	---	0.3284 g	5.12 kHz
☑	SineDamage	FDS SOR SAMPLE_TAC...	129.4 (S)	0.00s	129.40s	129.40s		---

The user can then add the **Target Lifetime** in either seconds or passes and based on the individual target life. The cumulative damage is computed from each of the profiles and then accelerated to reduce the testing time while maintaining the cumulative damage.

PA will display the accelerated PSD. Users can export the profile to run on a shaker using VCS software.

Time Stamped Processing

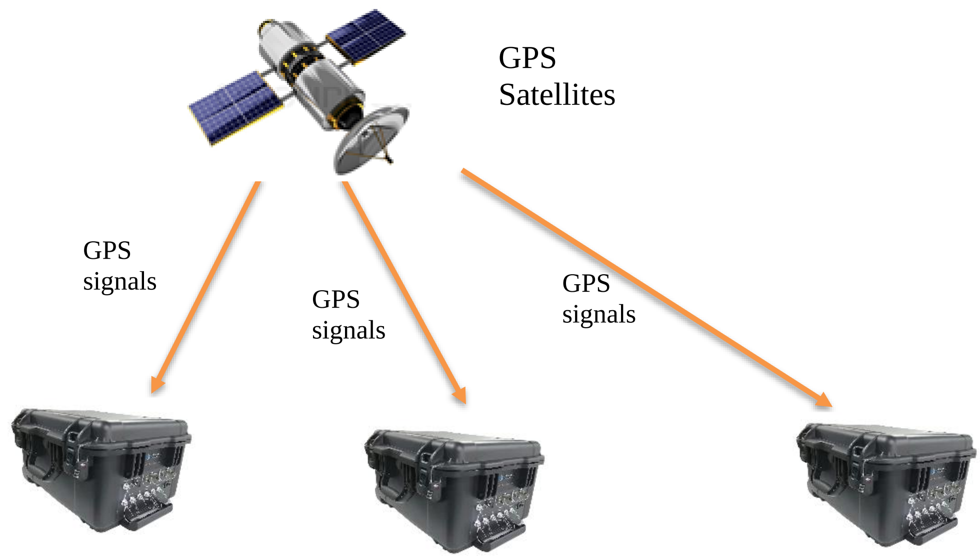
Technology Background of Time Stamping

The Global Navigation Satellite System (GNSS) installed in the CI-GRS delivers high integrity, precision timing in demanding applications worldwide. Support for BeiDou, GLONASS and Galileo constellations in addition to GPS (Global Position System) enables compliance with national requirements. Enhanced sensitivity and concurrent constellation reception extend coverage and integrity to challenging signal environments. Survey-in and fixed-position navigation reduce timing jitter, even at low signal levels, and enable synchronization to be maintained with as few as one single satellite in view.

Due to popular use of GPS (Global Position System) in history, this document simply adopts the usage of GPS instead of GNSS.

If the time accuracy of the sampling clock for a data acquisition system is not very demanding, say no better than seconds, then the time base can be derived from the “computer time”, which can be set manually through a time server on a network or through the internet. If the time accuracy of the sampling clock demands millisecond resolution, the digital input paths of a data acquisition system, especially its ADC, has to be designed with control from a more accurate time base, such as GPS or IEEE 1588 PTP (precision time protocol). This is necessary to calculate the cross-channel spectrum, or any signal property related to the time delays between all measurement signals.

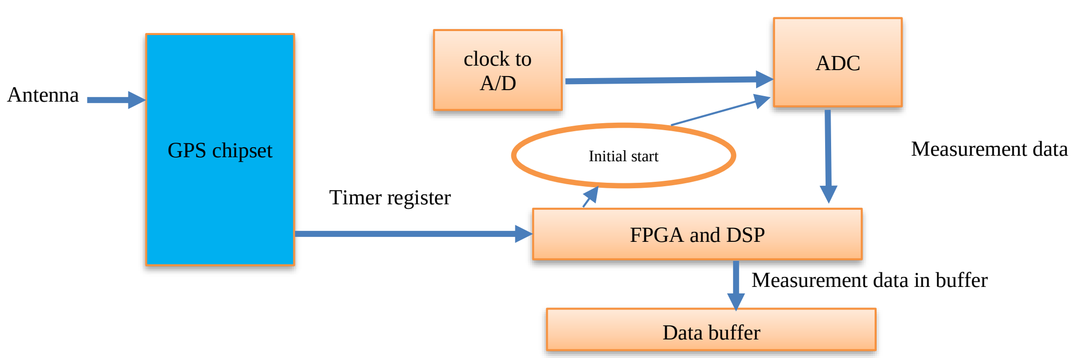
If the ADC sampling clocks cannot accurately synchronize between different hardware units, it is possible to achieve the similar signal processing results if all recorded data is accurately time stamped. Applying time stamps and synchronizing the ADC clocks are two different strategies. CI believes accurate time stamping allows users to achieve the same measurement goals, including the calculation of cross-spectra between all measurement channels.
Multiple CI-GRS data acquisition systems can acquire data simultaneously while they are physically spread out over hundreds of miles. These units do not share any direct hardware connections except each will receive GPS signals. Accurate time stamping technology is implemented with the GPS time base to line up acquired signals on the same time base for post processing in PA software (from CI).



Global Positioning System (GPS) consists of 24 satellites revolving around the earth every 12 hours. Each satellite has an extremely accurate atomic clock onboard. The GPS satellites continuously transmit their coordinates in space along with a time message on a 1.575 GHz carrier frequency. The accuracy of the time signals sent for civilian use is guaranteed to be within ±170 ns of UTC. The GPS receiver used in the CI-GRS claims to provide ±60 ns time accuracy at 99% of time.

Clock signals, timing, and location signals are delivered from the GPS chip inside each GRS unit. A real-time, zero latency hardware logic is used to time stamp the A/D sampling clock with the measured GPS time base.

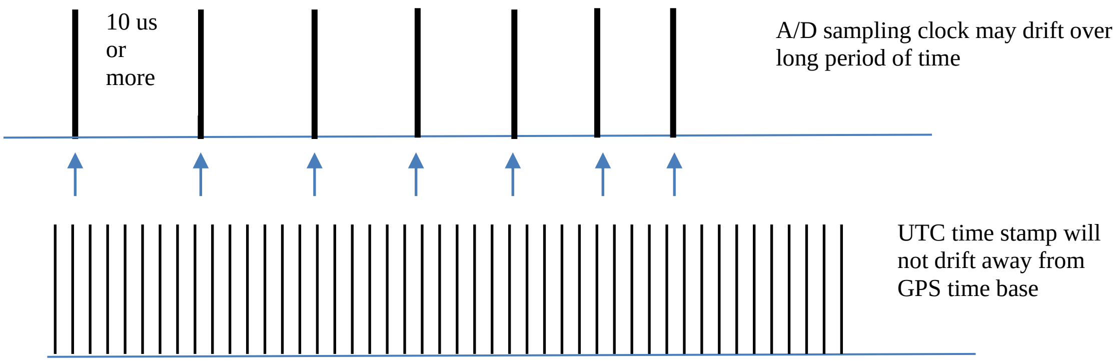
The following is a simplified diagram of how time stamping works in GRS.



The CI-GRS has a time register derived from the GPS receiver chip that is directly accessible by the FPGA and the processor. The GPS timing accuracy is better than ±60 ns with 99% probability according to the GPS chipset specifications. The time stamping method described in the preceding results features time stamping accuracy within the sub-microsecond range, typically less than 100 ns.

Given such an accurate technique of applying time stamps, we are able to analyze the error of actual sampling clocks vs. the GPS time base, which is described as follows.

An ADC requires a clock with a specific oscillator frequency to drive its sampling. The diagram below shows the ADC clock drift in concept. Here the ADC clock has a period of more than 10 µs (102.4 kHz ⇔ 10 µs sampling). The drift might be caused by the environment, especially changes in temperature.



Besides the short term **drift**, the ADC sampling clock, which is driven by an internal oscillator in the GRS, may have an inherit **offset** compared to the oscillators on the other GRS units, or compared with the atomic clock of GPS. The combination of offset and drift constructs a **bias** of the clock. For a recording time of a few minutes this bias may not be a major problem. However, if the recording time is as long as hours or days, this sampling clock bias may cause sampled data from different units to have mismatches when comparing time stamps.

Bias error = short term drift + offset

Suppose the oscillator to drive the A/D sampling has a stability of 50 ppm (part per million). After one hour the clock bias will be the following:

(0.000050 bias) * (3600 seconds per hour) = 180 ms bias error per hour

In the worst case scenario, the time clock of sampled data will vary by a factor of 180 ms per hour. This is a situation when all oscillators on different GRS units are running without clock constraints, such as periodic synchronization through a GPS signal.

In order to discuss the timing issues with clarity, a few terms used in this manual are introduced:

<u>Nominal Sampling Rate</u> is the sampling rate labeled by the product specifications, for example, 64 kHz, 102.4 kHz etc. These are not accurate because of the bias and drift from ADC clocks.

<u>Nominal Time</u> is the time of acquired signal samples calculated based on a starting time, number of samples, and **nominal sampling rate**.

<u>Measured Time</u> is a series of time stamps taken from the GPS clocks corresponding to the samples of acquired signals. If GPS works properly, the Measured Time is a much better time base than Nominal Time.

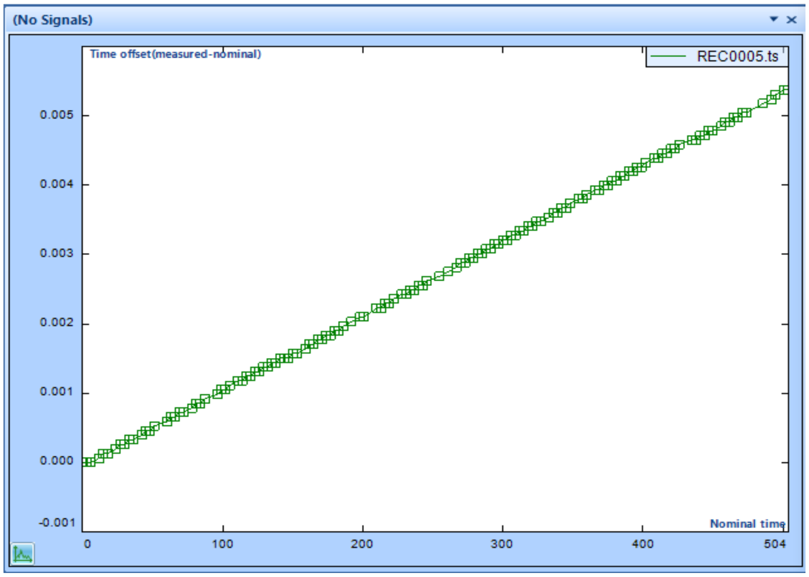
<u>Corrected Sampling Rate</u> is a single value for the sampling rate used in the entire duration of a recording. It is calculated by removing the offset between local clocks against that of GPS based on Nominal Sampling Rate, the duration, and time offset.

In other words, the **Corrected Sampling Rate** is calculated by removing the first order error between the local clocks and GPS time base.

CI-GRS is able to record time stamps and show them in the following format:

	Nominal Time(REC0078.ts)	Actual Time(REC0078.ts)
0	00:00:00.000.000.000	00:00:0
1	00:00:03.000.000.000	00:00:02.999.606.7...
2	00:00:06.000.000.000	00:00:05.999.213.5...
3	00:00:09.000.000.000	00:00:08.998.885.8...
4	00:00:12.000.000.000	00:00:11.998.492.6...
5	00:00:18.000.000.000	00:00:17.997.706.2...
6	00:00:24.000.000.000	00:00:23.996.985.3...
7	00:00:30.000.000.000	00:00:29.996.198.9...
8	00:00:48.000.000.000	00:00:47.993.905.1...
9	00:00:57.000.000.000	00:00:56.992.791.0...
10	00:01:03.000.000.000	00:01:02.992.004.6

When drawn in a plot using (Measured-Nominal) vs. Nominal, it looks like the following screenshot:



In this example of roughly 500 seconds of measurement, the nominal time is off by 5 milliseconds against the GPS time clock. In this particular case, the local clock is slightly slower.

The 5 milliseconds error of 500 seconds of measurement is within the range of expected error of the oscillator, but too large to conduct signal processing when cross-channel computation is needed. The time base difference between data samples of any two channels must be within a few microseconds of each other for vibration and acoustic applications.

Since we know the error between nominal time and actual measured is mainly an offset instead of random drift, this offset error can be corrected after the measurement. The technology developed in CI-GRS first applies a correction to the sampling rate, then uses the signals with the corrected sampling rate in post processing. The following describes the correction process in more detail:

1) Calculate the Time Offset as shown in the previous plot for the entire duration of the measured signals, 504 seconds in this example.

2) Based on the duration of measurement and the time difference between that of nominal and GPS time stamps, the actual measured sampling rate is calculated as:

   Corrected Sampling Rate = Nominal sampling rate / (1.0 + Time Offset/Duration)

   For example, if the Nominal Sampling Rate is 64 kHz, the Time Offset is 0.005 second and Duration is 504 seconds,

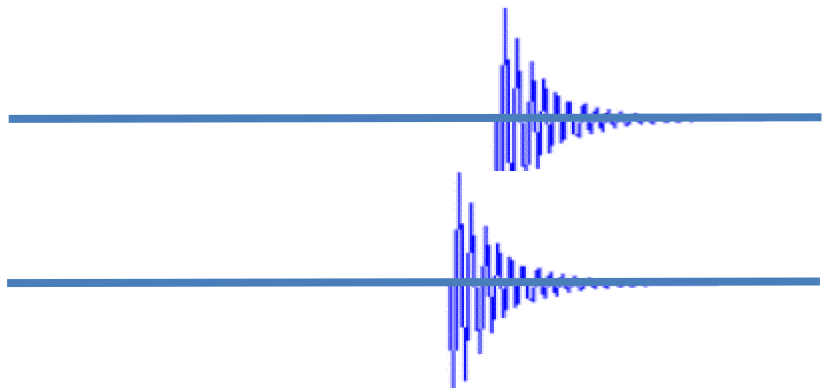
   Corrected Sampling Rate = 64kHz / (1.0 + 0.005/504.0) = 63.999365 kHz.

3) This **Corrected Sampling Rate** is computed by removing the first order error to the nominal sampling rate. It did not take into consideration the sampling rate fluctuation during this time period.

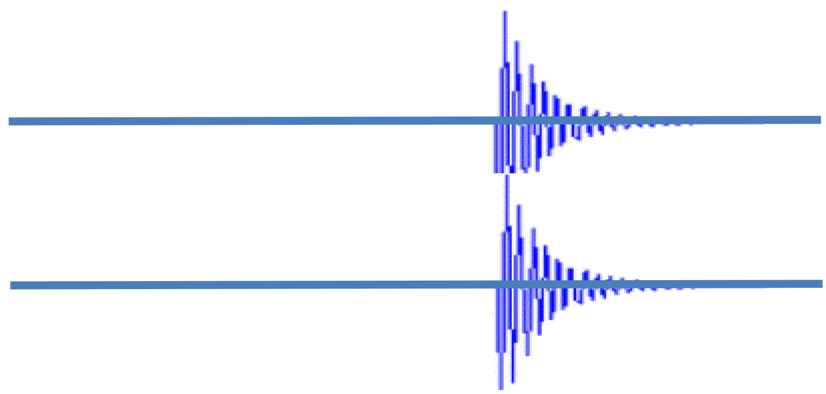
4) Since the signals acquired on this unit are from a slower clock with 63.999365 kHz, we are able to adjust it in the display and post processing algorithm.

The following is an example of how the corrected sampling rate can help in data processing.

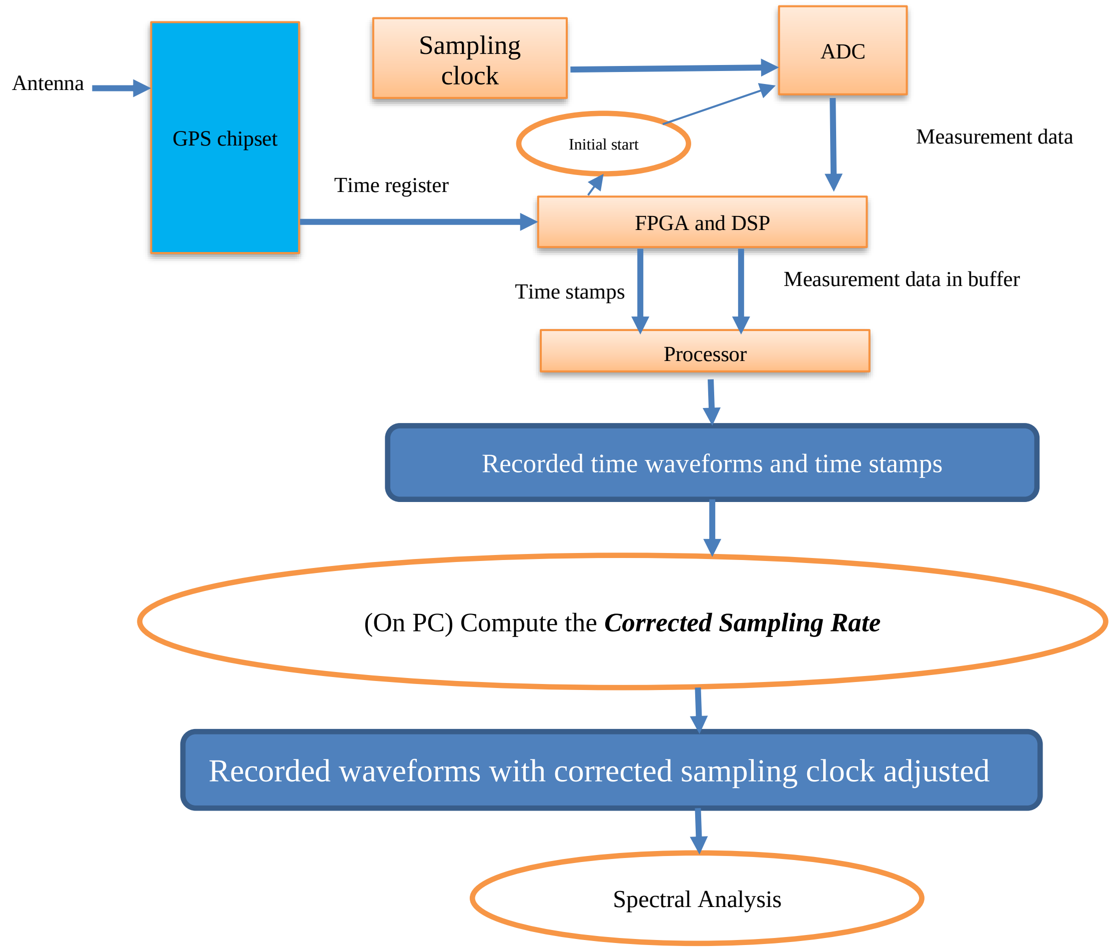
The plot below shows that when an impulse event is recorded by two data acquisition systems with slightly deviated sampling clocks, the signals will be off by a certain duration even with the same starting time-base.



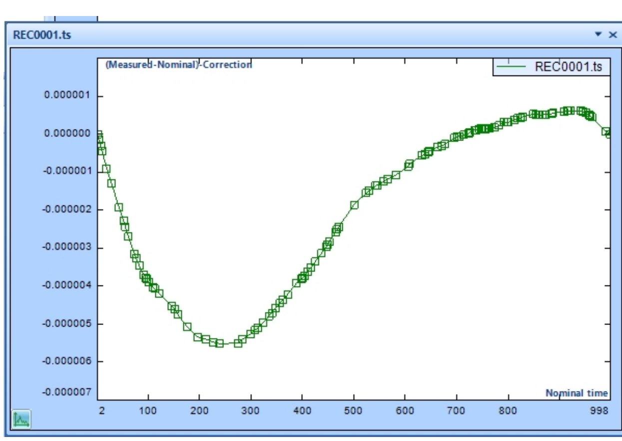
But after a correction against the GPS time base, the sampled points can be lined up on the plot.

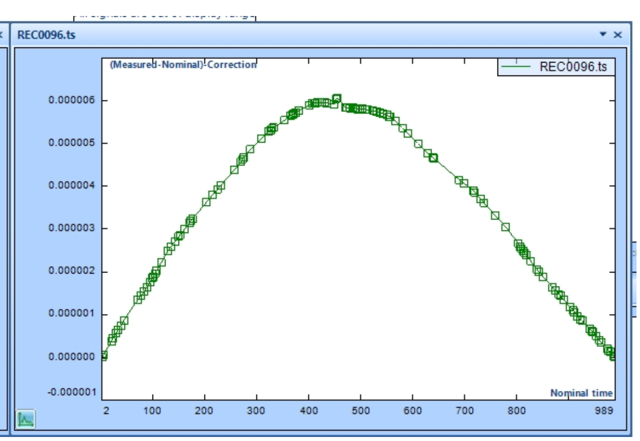


After the adjustment, all classical signal processing methods such as FFT, FRF (Frequency Response Function), Coherence Function etc., can be applied in the post processing software. The whole process is illustrated in the following diagram.



The next question addresses the accuracy of the Corrected Sampling Rate. The post processing software is able to display the clock fluctuation after the first order correction. The following two graphs display the "drift" of the time stamps after the correction is applied.





Both vertical and horizontal are displayed in units of seconds. In these graphs, the quantity displayed vertically is:

(Measured – Nominal) – Correction

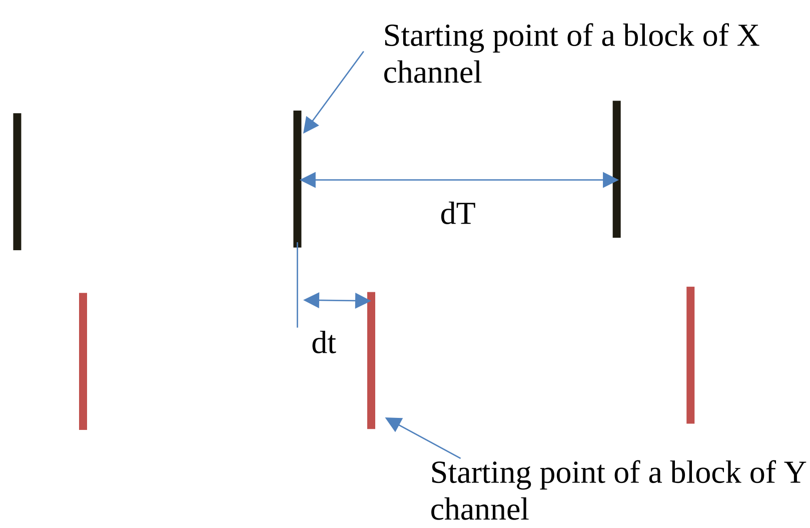
The **Correction** is a straight line drawn between the two time stamps, the very first and last time stamps of the entire recording. Both signals show that the fluctuation range is roughly 6 μs while the recording time is roughly 1000 seconds. This fluctuation can be further corrected when a more sophisticated clock correction algorithm is used, which is already implemented in the PA software.

Technology Background on Phase Correction

In most typical FFT analyzers, the following spectra are always computed: auto power spectra of all measurement channels; cross-spectra of any pair of channels; transfer function (or FRF, the Frequency Response Function) of any pair of channels; coherence functions of any pair of channels; phase spectra of any pair of channels. Once the auto power spectra and cross-spectrum of a pair of channels are computed, the other signals can be derived.

Spectral Analysis is performed on a block-by-block basis, meaning a fixed number of sample points is used to calculate an FFT, APS, FRF, etc. Now, with data acquired using two different GRS systems, a phase correction is required.

Up until this point, we have seen that the GPS timestamps are used to apply a first-order correction to a nominal sampling rate, allowing us to 'line-up' data streams acquired by different GRS systems with different ADC clock rates. However, since these ADC clock rates are ultimately different, samples collected on two different systems will not be acquired at the exact same instance in time, and consequently blocks from two different systems will not share the same starting point in time.



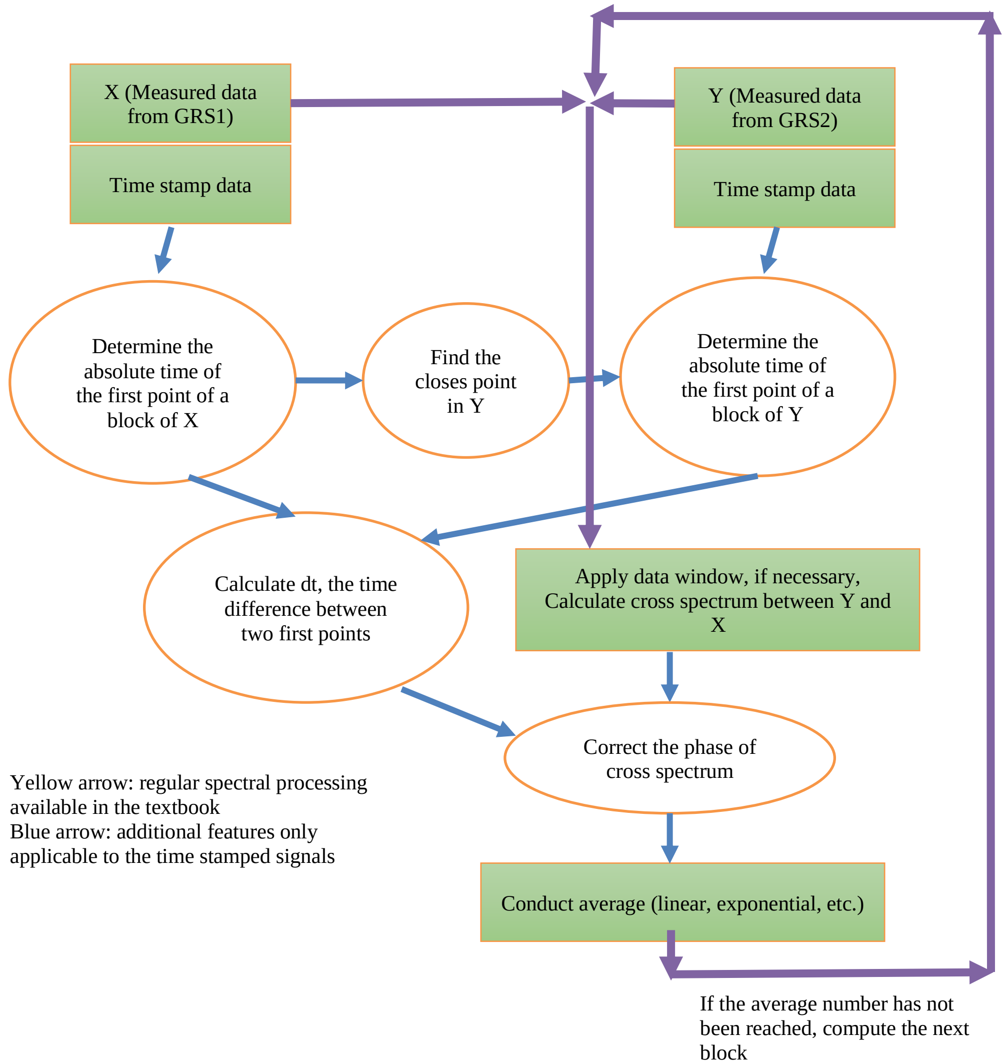
The figure above depicts blocks of data acquired by two different DAQs, with starting points separated by some time, denoted by dt. The following describes the process to perform an analysis using these two blocks of data:

1. First, the starting point of each block is determined as follows:
    a. The starting point of the block from the first channel (X channel) is arbitrarily selected.
    b. The starting point of the block from the second channel (Y channel) is selected using GPS time stamps. It will be the data point sampled closest in time to the starting point chosen for the block in the first channel.
2. A discrete Fourier Transform for each block is calculated, and an instantaneous Cross Spectrum is derived from that calculation.
3. A phase change is applied to each element of the cross spectrum based on the following formula:

$$Phase\ Change[n] = 2\pi \cdot \frac{dt}{dT} * \frac{n}{Block\ Size}$$

4. After the phase correction is made, the Averaged Cross Spectrum may be computed, and the other Spectrums (FRF, Coherence, etc.) may be derived.

The spectral analysis portion of the flow diagram above can be described by the following flow diagram.



With signals taken from separate CI-GRS units, a new algorithm was developed where the phase mismatch between any pair of FFT spectra can be corrected based on certain calculations of time stamps. Test results have indicated that the CI-GRS can achieve the following specification of phase estimation to the cross-spectral calculation:

Phase match between two CI-GRS units:	±5°(degree) at 40 kHz ±2.5°(degree) at 20 kHz, ±0.5°(degree) at 2 kHz.

This is an extraordinary result. While many companies claim to use the GPS time base for their data acquisition, not many companies claim to achieve such an outstanding specification.

Traditionally, in order to measure the transfer function between any two sensors, all ADC (Analog to Digital Converters) have to be accurately synchronized through hardware connection. The GRS technology presented by Crystal Instruments provides an alternative approach.

Quick Guide: View time stamps and signals in Post Analyzer

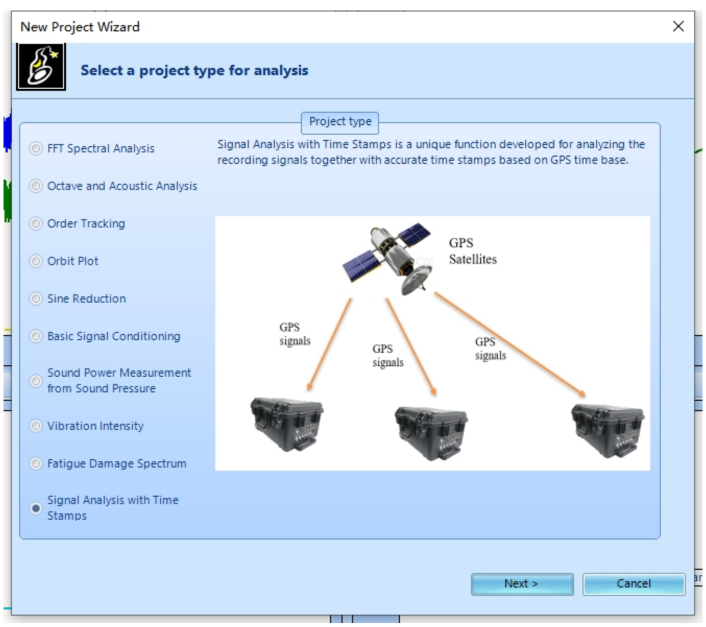
Merge recording files and apply correction to the Sampling Rate

To analyze time stamped signals, first use the GRS-Host software to download the recorded files with the time stamped files to a computer. Then use the PA (Post Analyzer) software to analyze the data. This section provides a quick guide to view time stamps together with their measurement recording files.

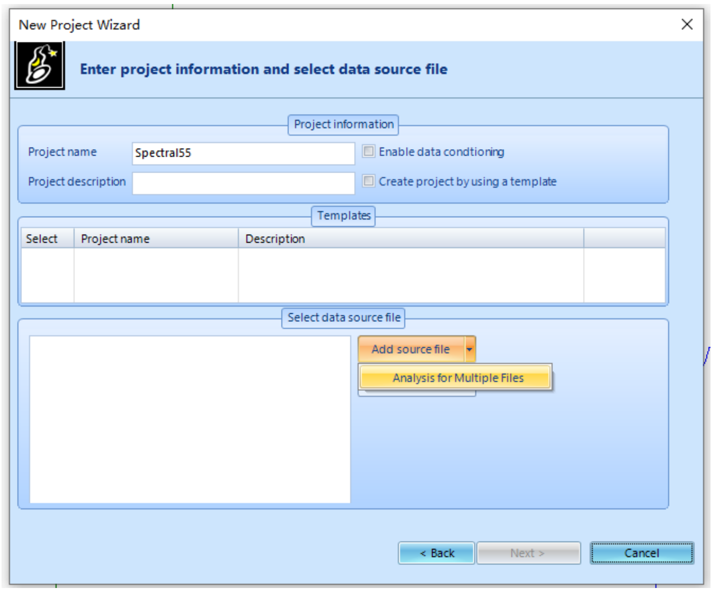
The analysis function in PA is applied to a recording file that contains the time domain signals from multiple channels. Therefore, to post process data recorded from different CI-GRS units, first merge them into one data file. To view the recording data only without conducting spectral analysis, do not merge the data into a single file.

The following describes how to merge the recording into one file and view it.

Open the PA software and select **Signal Analysis with Time Stamps** testing type, click **Next** and create a New Project:

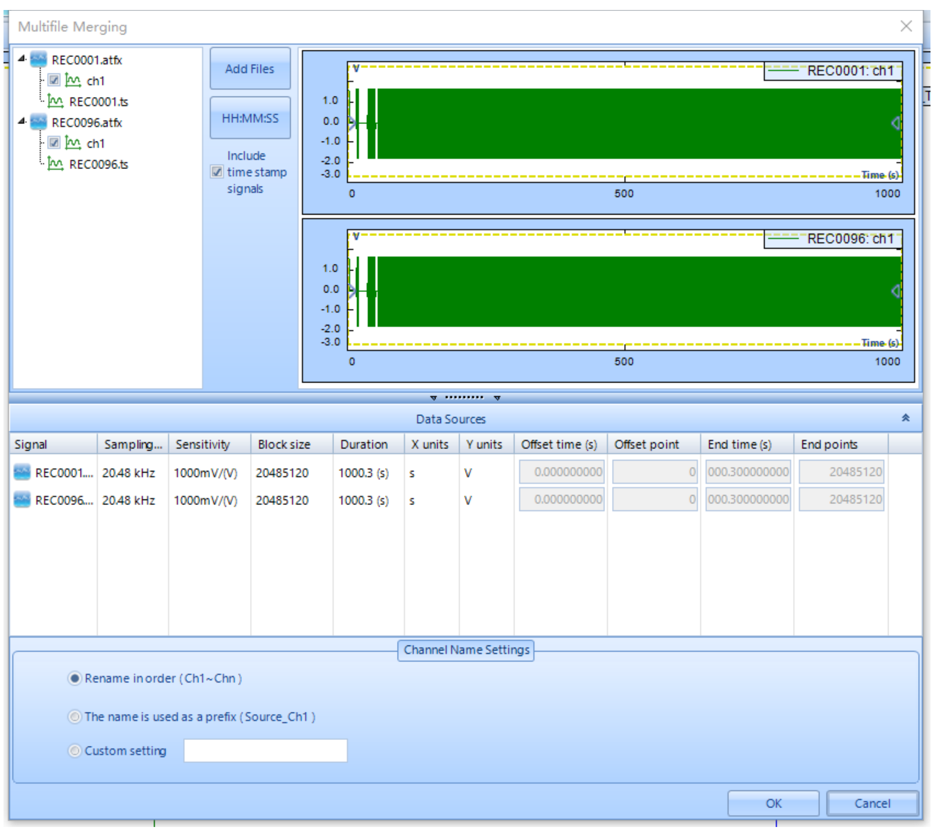


To merge the signal file, select **Analysis for Multiple Files** as shown in the following screenshot.

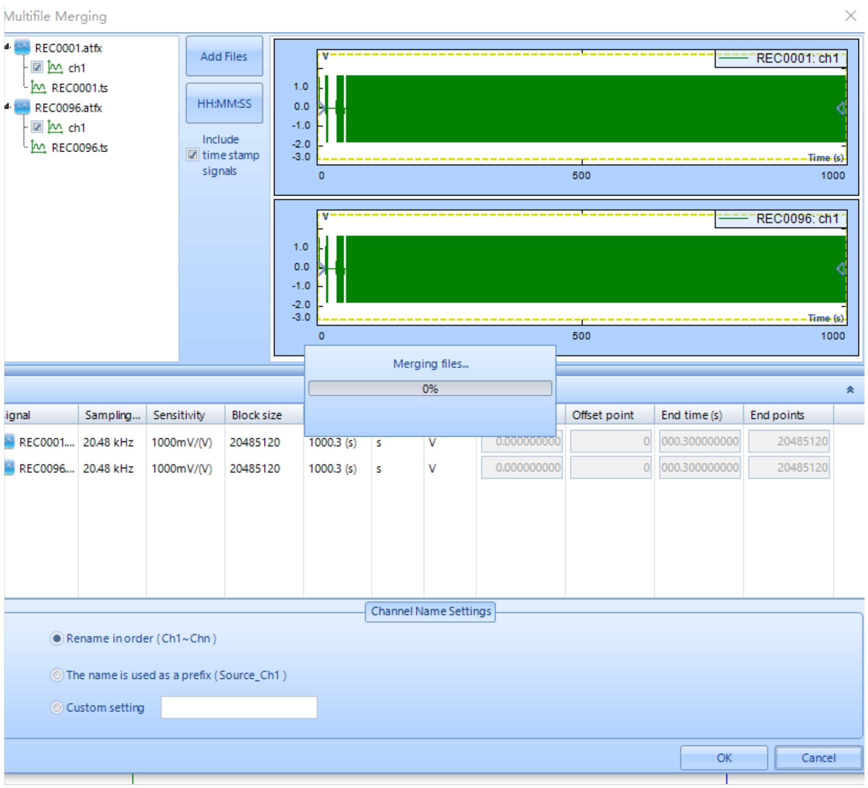


To add files with time stamps, use the GRS-Host software to download the recording files together with their time stamp files and save them into a folder.

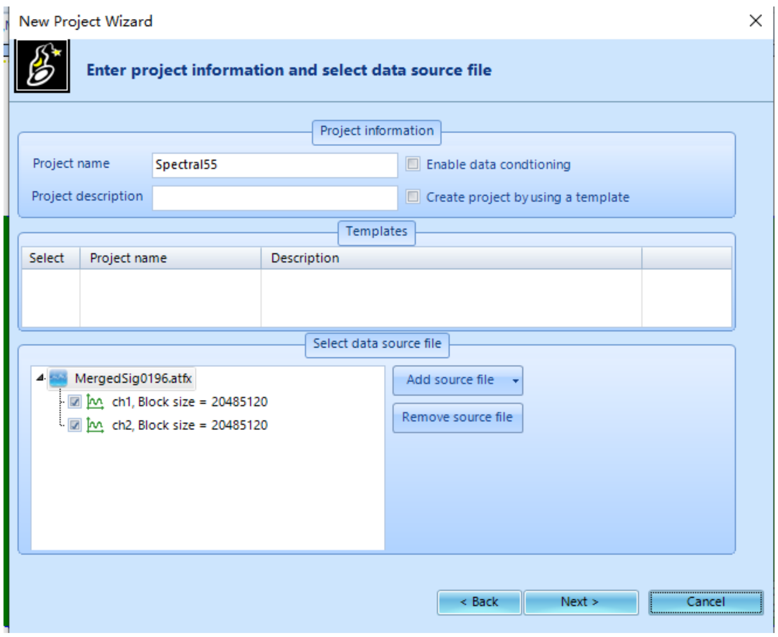
Check the **[x] include time stamp signals** box below, then add the files one by one into the list. Click OK and wait for merging process to finish.



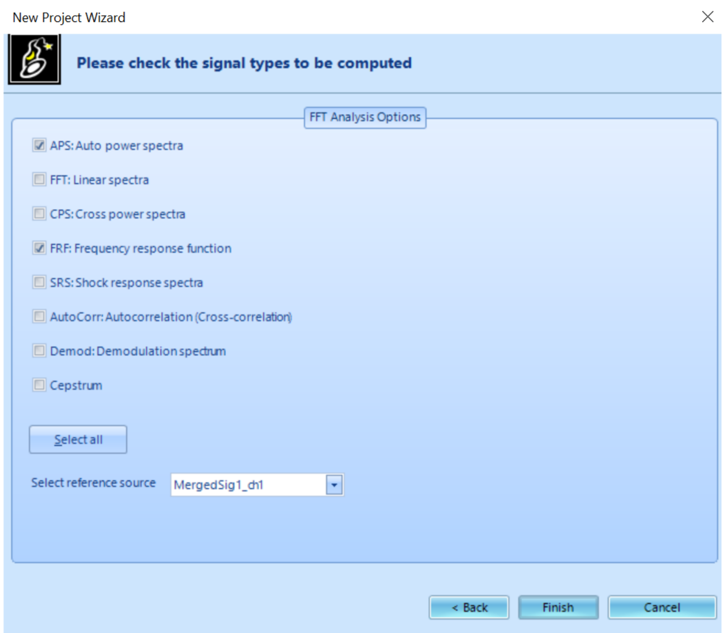
When the progress bar completes, the software will ask the user to choose a path to save the file. Please set the path and wait for completion. For long duration recordings, it will take some time to create the Merged Signal, so a progress bar is displayed.



After merging is finished, the merged file is added to New Project Wizard as shown in the following screenshot:

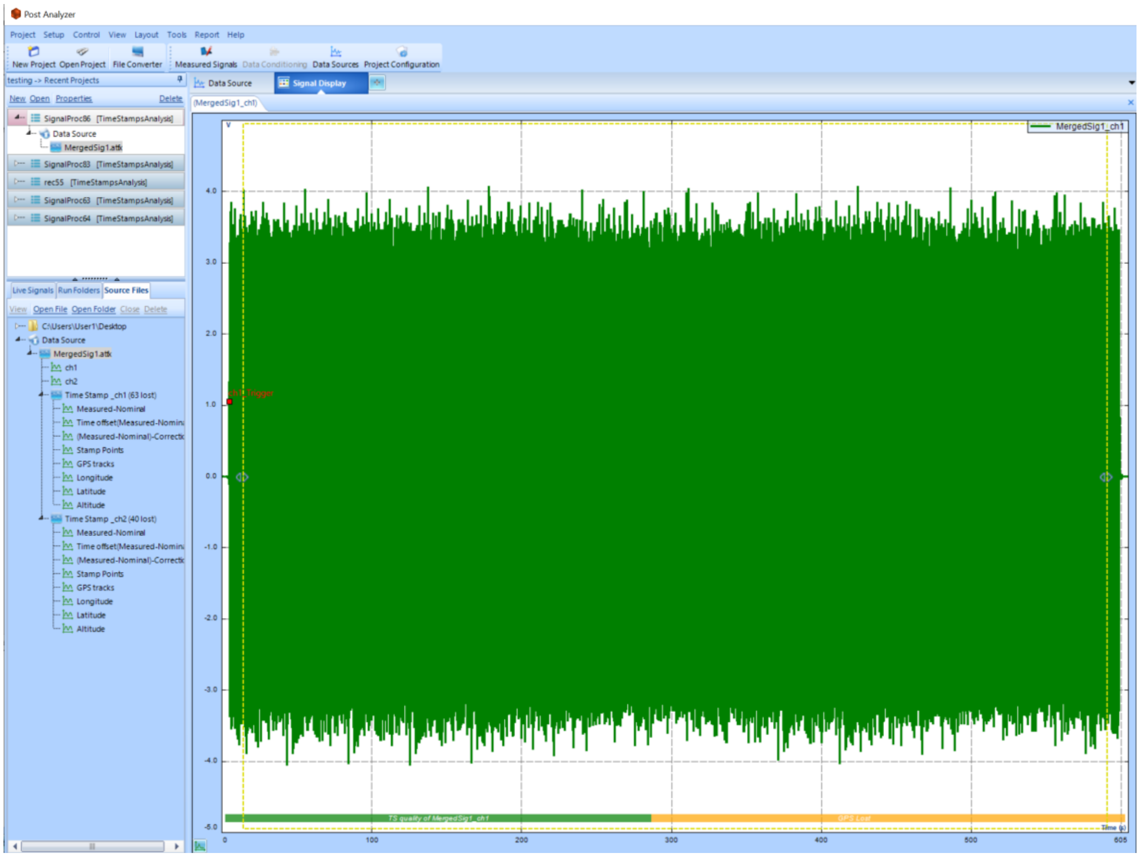


Click **Next** to open the final window in the New Project Wizard, which is the selection of signal types. To view the phase shift between the recordings, select **FRF: Frequency Response Function**.

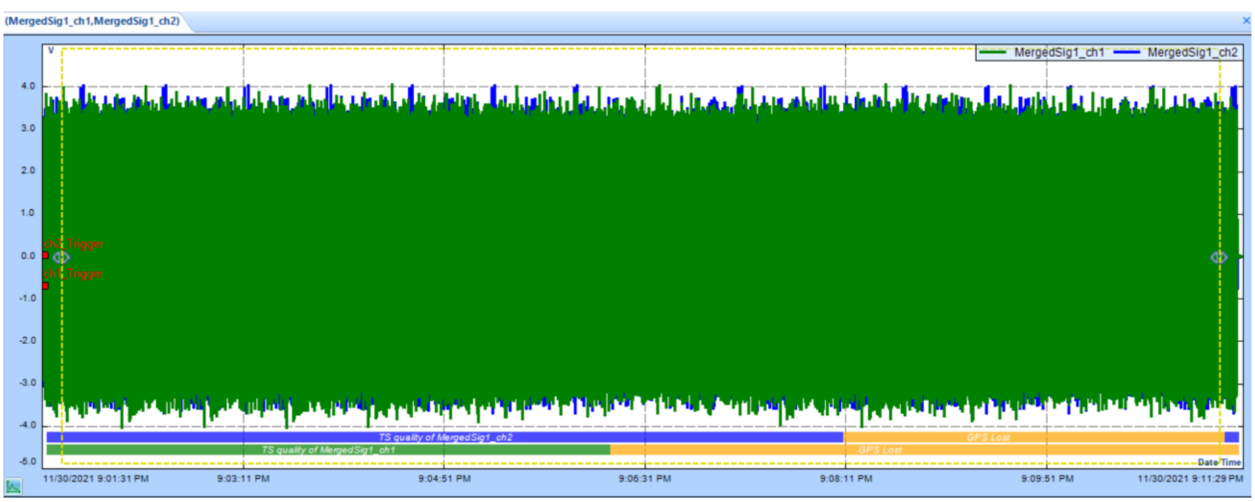


The PA software can now operate on files that are merged with recording signals from multiple GRS units.

The Signal Display will show at least one time waveform by default.

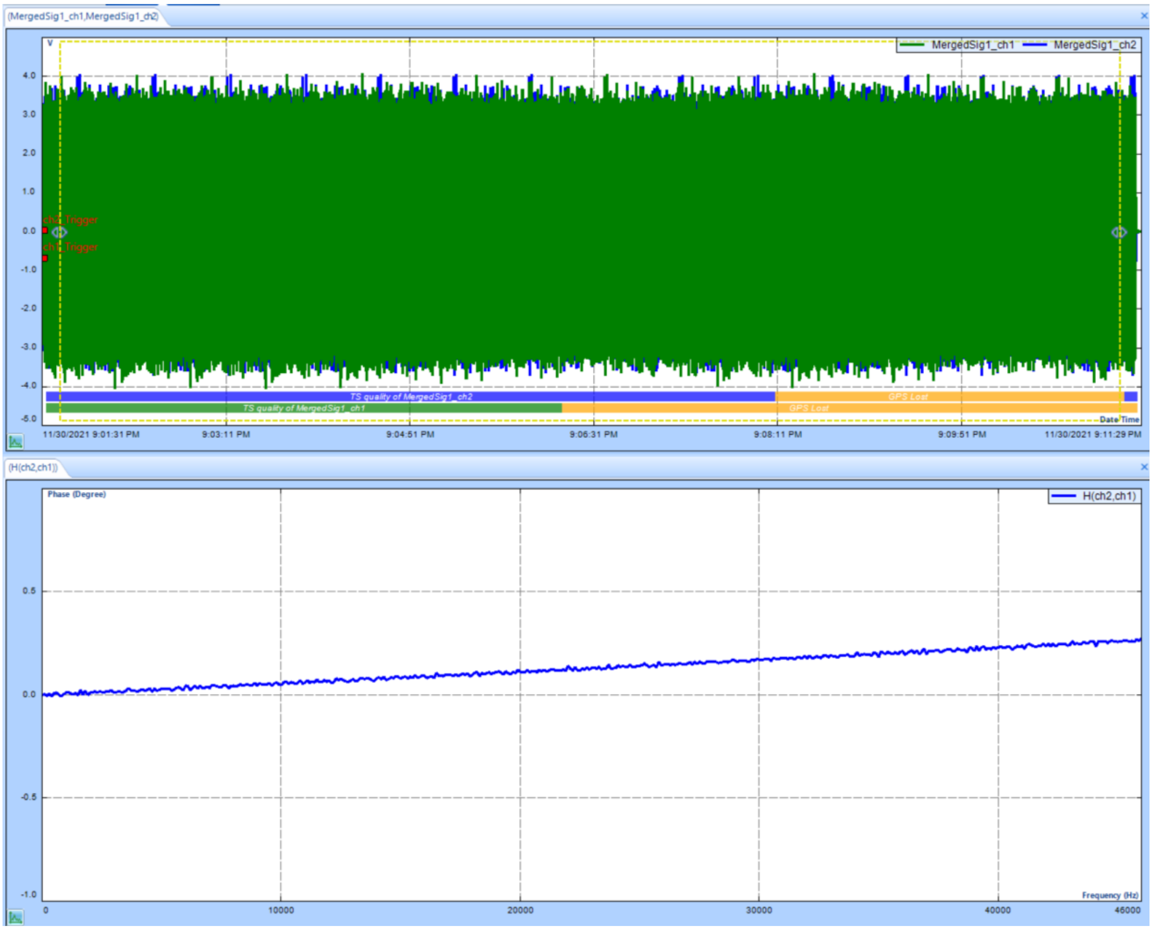


To view both waveforms, go to the Source Files tab on the left side, below Recent Projects. Click and drag the other file into the Signal Display.



The lower portion of the plot shows intervals of time when no valid time stamps are available. Additional information regarding time stamps and GPS data such as latitude and longitude is found under the Source Files section.

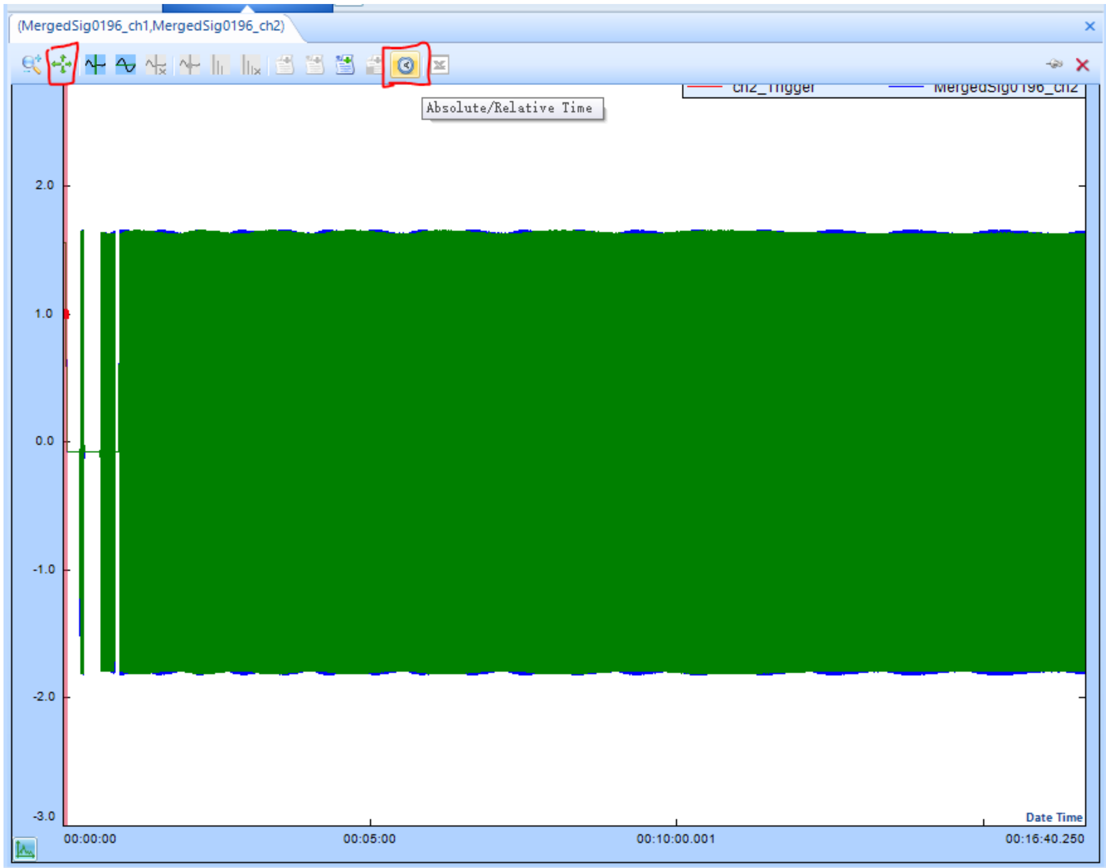
In the Live Signals tab, select the FRF to view the phase difference between the two recordings. Set the Vertical Axis to Phase, configure the analysis parameters, then click Run to see the phase difference.



In this example, the phase shift at 46 kHz is just above 0.5 degrees. The value may vary throughout the duration of the recording, but it is typically less than 1 degree at 46 kHz between two systems that have their time bases synchronized to the GPS time base. If a section is chosen for analysis that is marked with *"GPS Lost"*, the phase shift between the two signals will typically be larger.

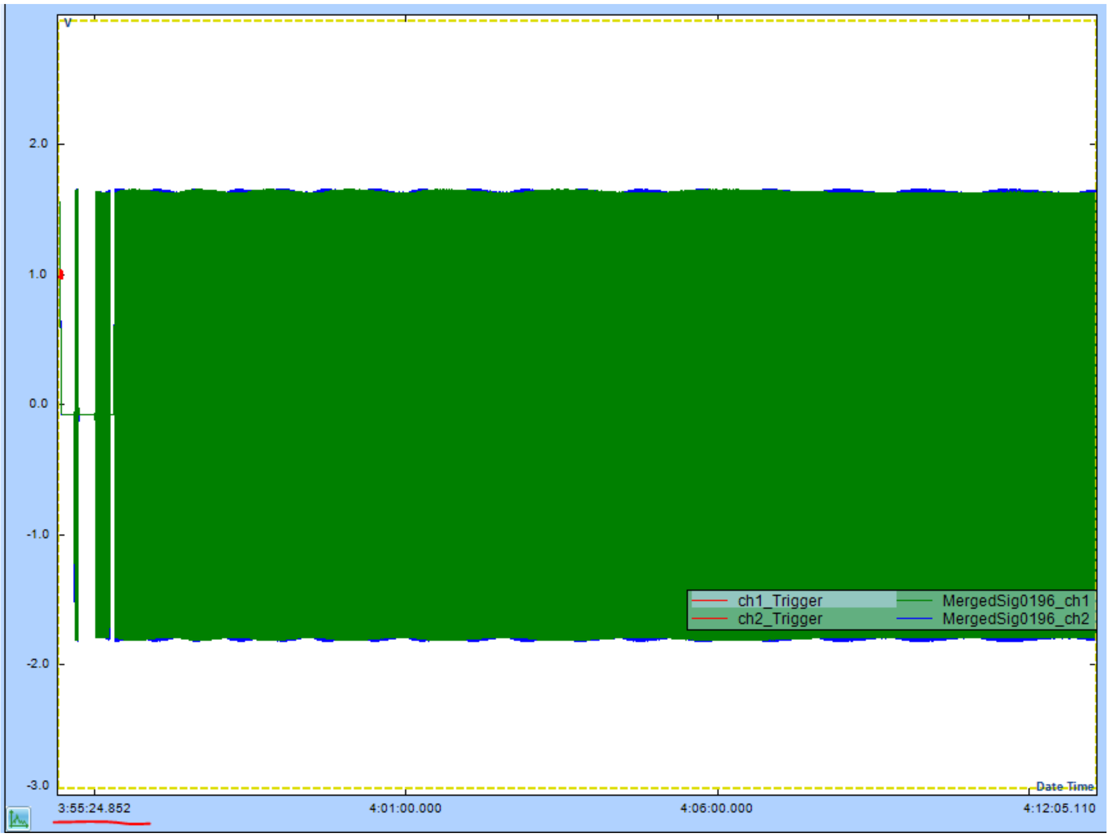
Viewing Absolute or Relative Time

An option is available to set the time axis to either Absolute Time or Relative Time. The default display includes the time from zero to the length of the recording. Selecting this option allows users to toggle the time shown between UTC time and the relative recording time. This option is revealed on the hidden top panel of a plot window. Place the mouse near the top and move down to reveal the panel. Or click on the time format selection to display the horizontal axis as Absolute Time or Relative Time.

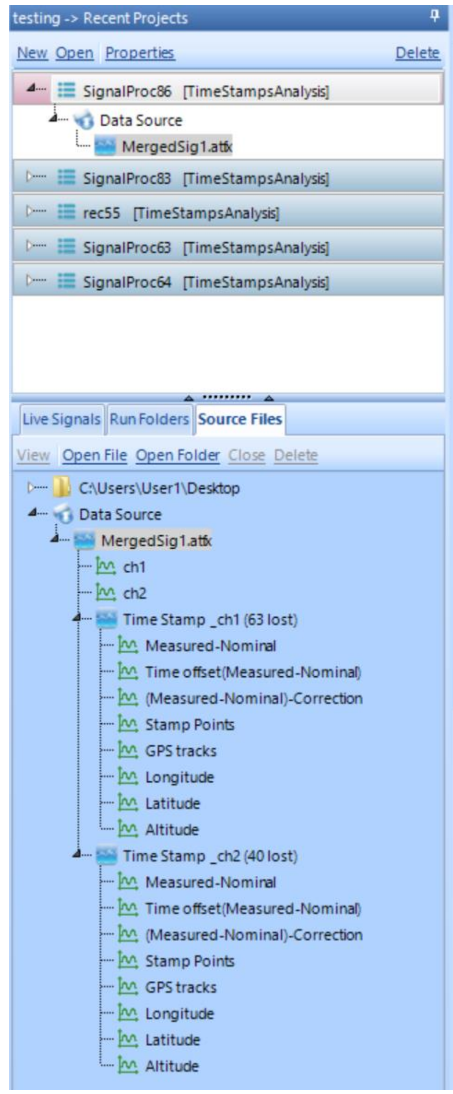


The format of Absolute Time shows the calendar time signals are acquired while the Relative Time sets the initial time to zero.

If the signals are starting at different times, the left boundary shows the earliest signal's start time.


V
2.0
1.0
0.0
-1.0
-2.0
-3.0
ch1_Trigger
ch2_Trigger
MergedSig0196_ch1
MergedSig0196_ch2
Date Time
3:55:24.852
4:01:00.000
4:06:00.000
4:12:05.110

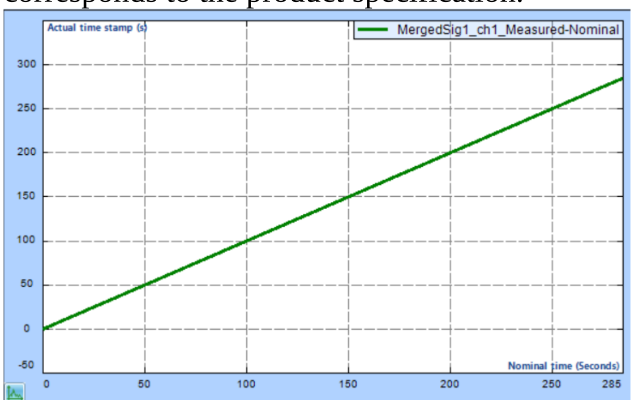
Additional Viewable Signals



The Source Files tab on the left side provides additional signal files and details of time stamps and acquired GPS signals. The following list describes what PA displays when opening these signals as plots:

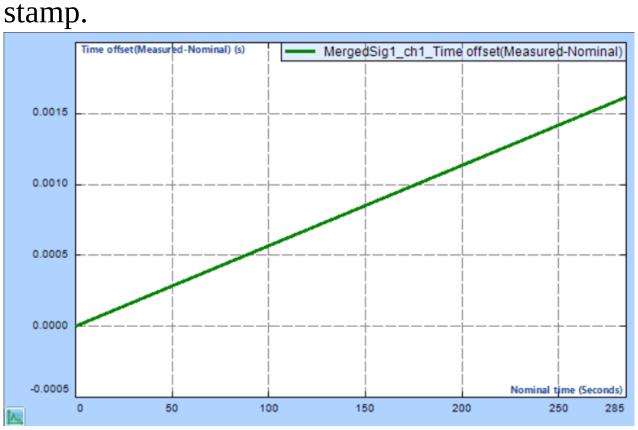
1) Measured-Nominal

   This plot shows the difference between the measured and nominal sampling rates. The measured sampling rate corresponds to the GPS clock, the nominal sampling rate corresponds to the product specification.



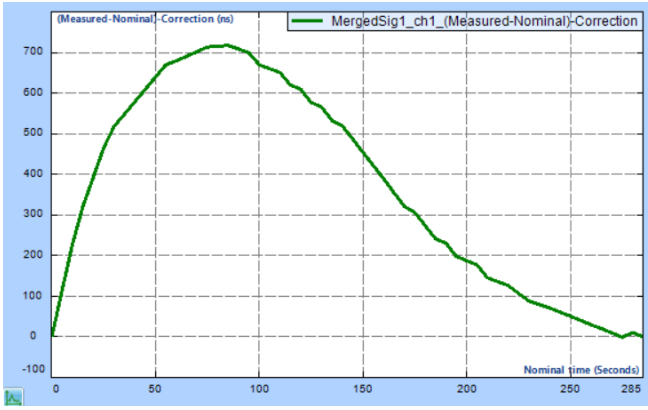
2) Time Offset

   This plot is identical to Measured-Nominal except that the units on the vertical axis are displayed in seconds, excluding the time stamp interval of 5 seconds per time stamp.



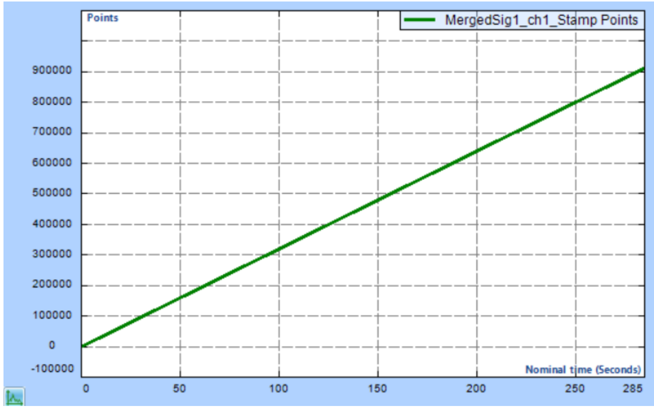
3) (Measured-Nominal)-Correction

   The corrected sample rate is a single value assigned as the sampling rate for the duration of the recording. It is typical for this curve to be nonlinear because of the variation in the Measured-Nominal curve relative to the single corrected value.



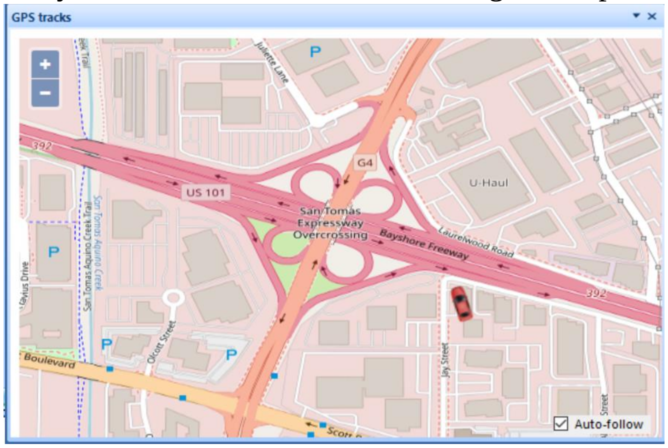
4) <u>Stamp Points</u>

Displays the number of GPS timestamps received as a function of nominal time.

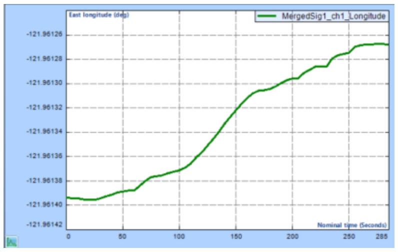


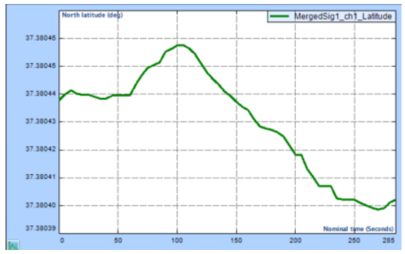
5) <u>GPS Tracks</u>

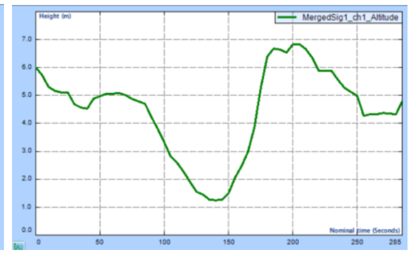
Displays the GPS coordinates on a map with a car icon to represent the location. When PA analysis is run, the car will move along the map with the GPS coordinates.



GPS coordinates can also be displayed as plots with Longitude, Latitude, and Altitude.





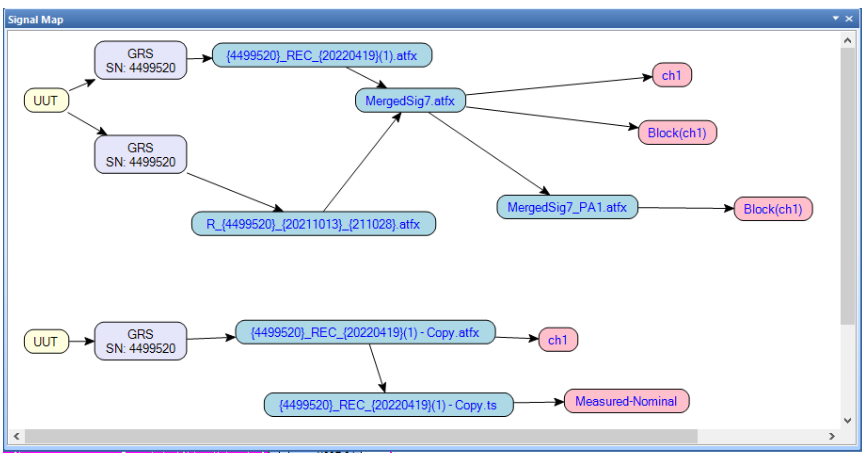


Signal Map

The Signal Map allows the origin of the signal to be displayed. The path from the Unit Under Test (UUT) to the Spider data acquisition unit, to the source file, to the signal is mapped out. This is a powerful tool that greatly simplifies the troubleshooting process if data discrepancies are suspected to be hardware issues.

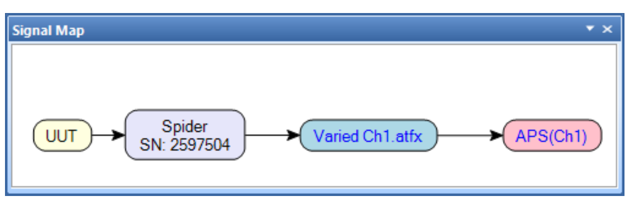
Adding Multiple Signals to a Signal Map:

A single signal map is used to display multiple test files and signals. To add multiple signals, right click on a signal and click **Signal Map** or drag and drop to add a signal. Depending on the test file, another hierarchy of nodes may be added to the map.



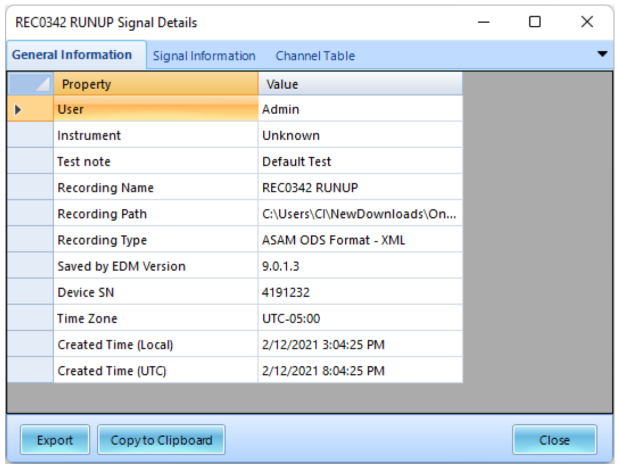
Right clicking on any node reveals a menu with an option to **Remove** that removes the entire node tree from the signal map.

Signal Map of a Live Signal:

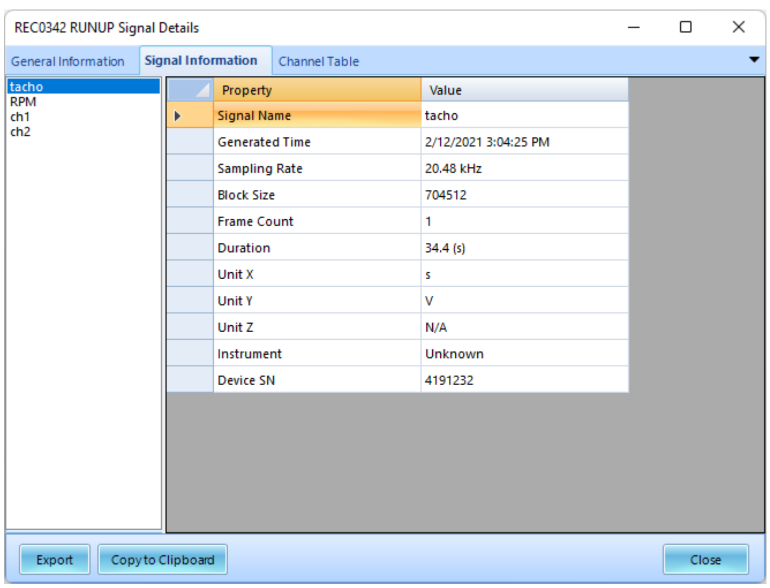


Selecting the test file (.atfx) will open the **Signal Details** window. This window displays the General Information, the Signal Information, and the Channel Table for a particular test or recording. All information in this window can be exported as a text (.txt) file or copied to the clipboard.

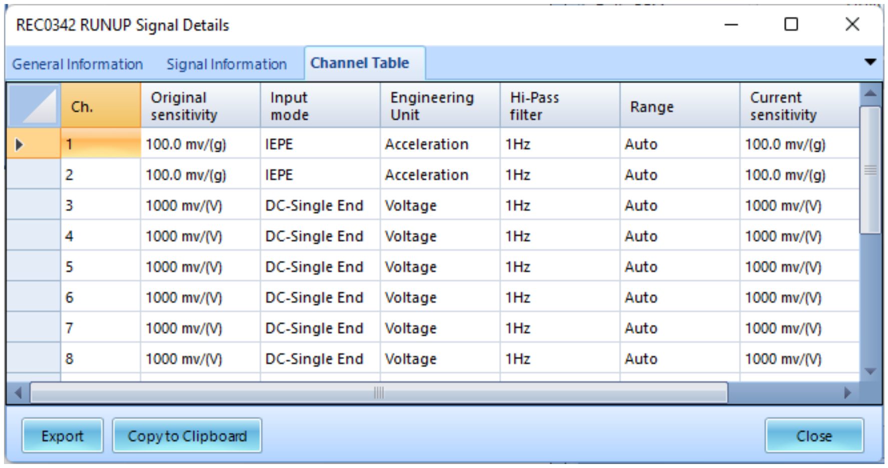
The **General Information** tab displays data that pertains to the recording properties, such as the acquisition system, test type, and creation time.



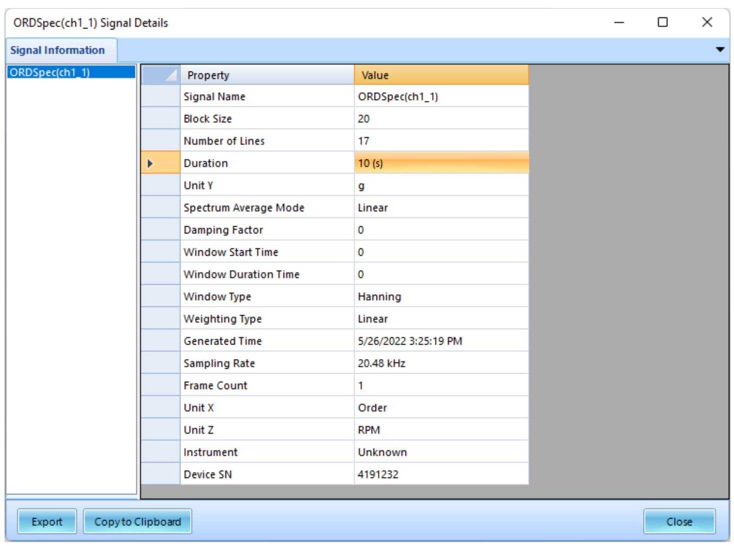
The **Signal Information** tab displays data that describes the signal properties for all signals:



The **Channel Table** displays the input channels settings that were configured on the data acquisition system for the particular test and signal being viewed:



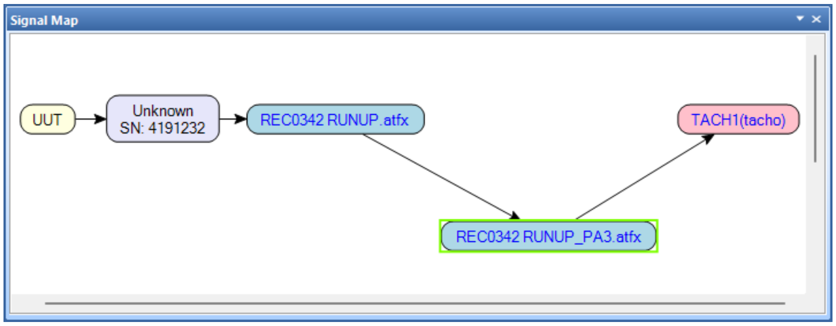
Selecting a particular live signal will open the **Signal Details** window for that signal. Information pertaining to the FFT analysis parameters, acquisition settings, and the front-end details will be displayed:



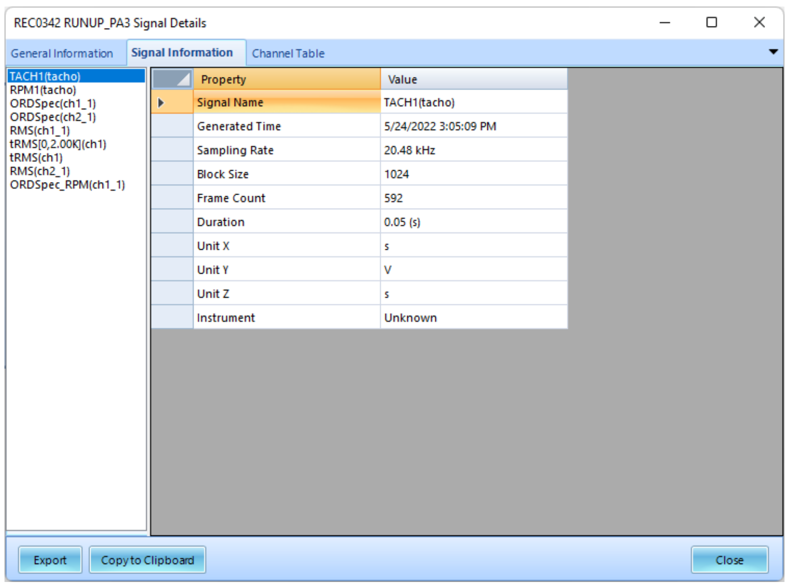
All information in this window can be exported as a text (.txt) file or copied to the clipboard.

Signal Map of a Recorded Signal:

The **Signal Map of a Recorded Signal** displays the path from the UUT to the front-end hardware, the acquired data test file, the post-processed data test file, and the post-processed signal file.

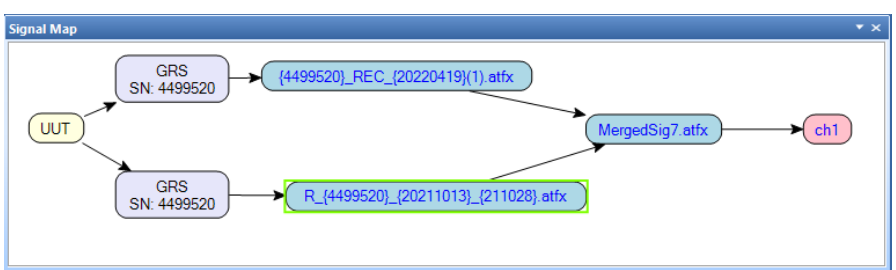


The **Signal Details** window for the post-processed test file and the post-processed signal file is practically identical to the Signal Map for live signals. Information for post-processed signals is displayed in the following screenshot.



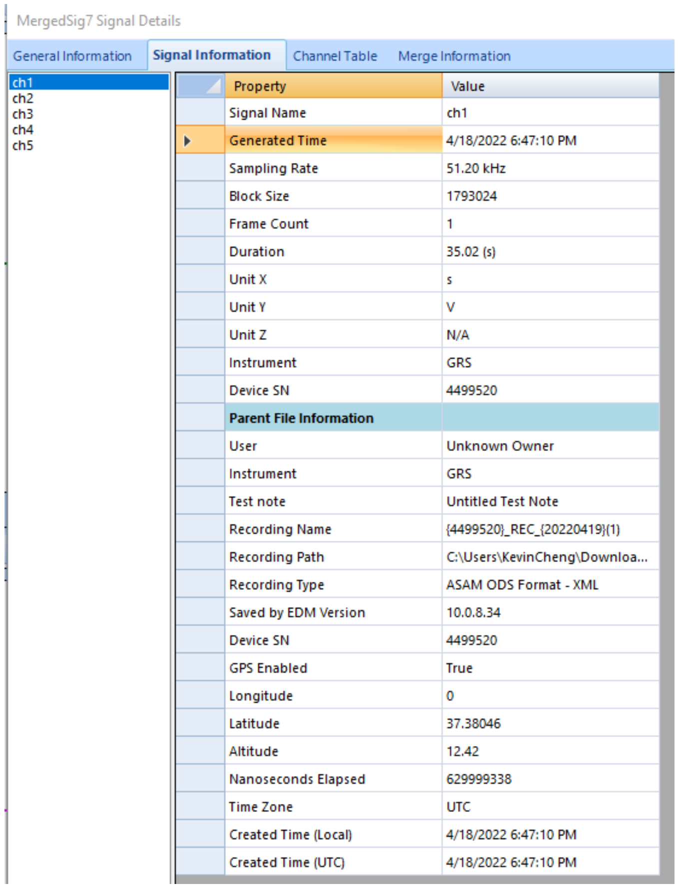
Signal Map of a Merged Signal

The **Signal Map of a Merged Signal** displays the path from the UUT to the front-end hardware, the acquired data test file, the merged test file, and a signal from one of the test files.

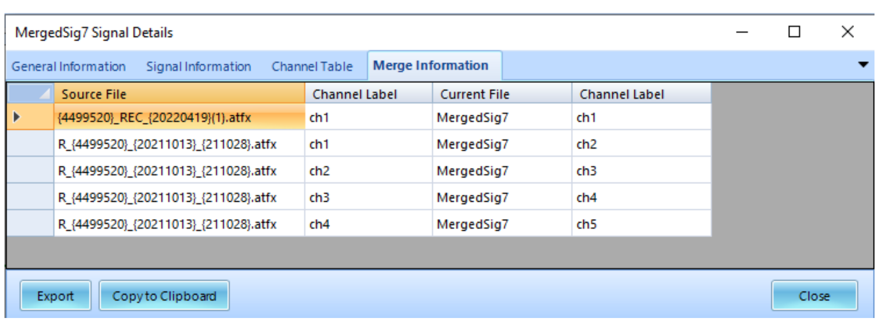


Clicking on any of the following test files or signals will open windows similar to Signal Map for live signals with some differences.

The **Signal Information** tab clearly displays the signal properties and parent test file properties.

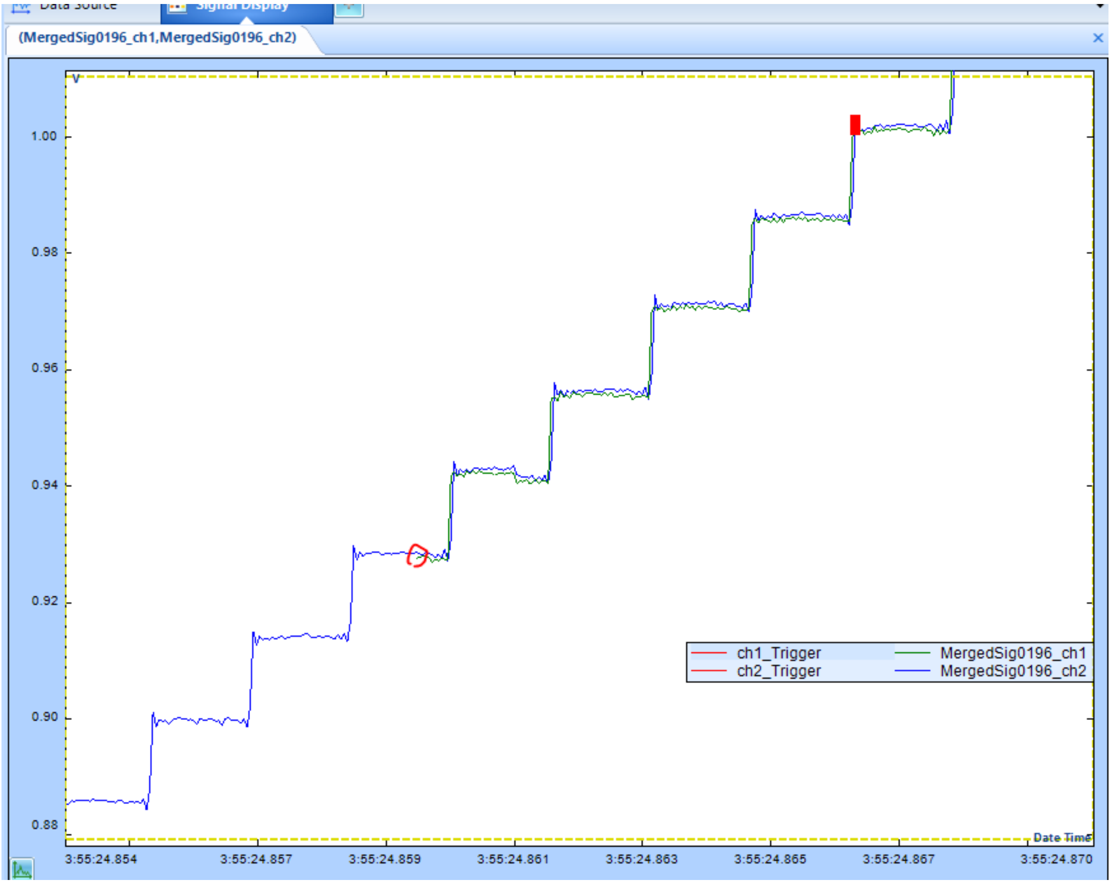


The new **Merge Information** tab displays the source test file, test file signals, and their new channel label in the merged test file.

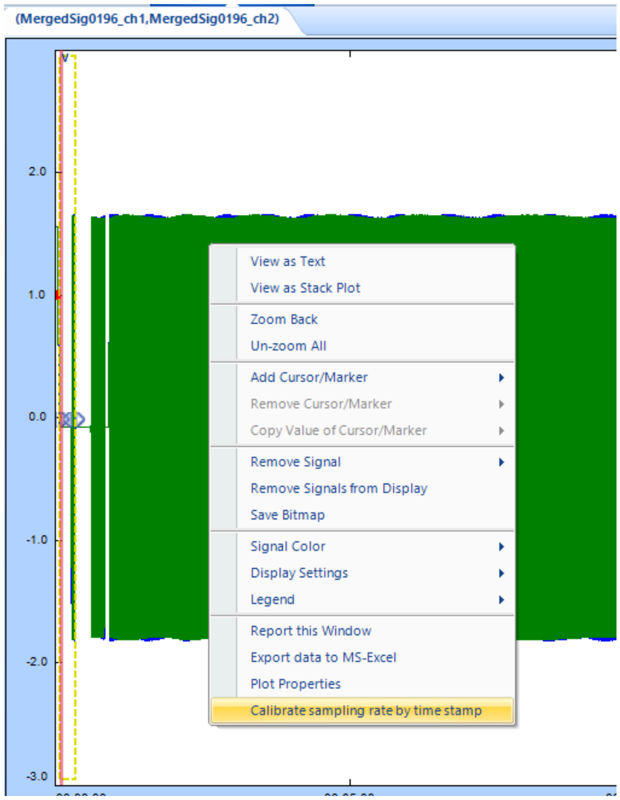


Signal Display

Zoom in and out of the display using the mouse to drag a small rectangular area. A small red dot indicates the exact trigger point.



The software can display signal plots in the calibrated sampling rate, i.e., a sampling rate after the correction is applied as described earlier. To view, right-click on the display window and click on **Calibrate sampling rate by time stamp**.



After the correction is complete, choose **Absolute Time** and click on **Auto Scale** to obtain the right display range.

Plotting Signals Example

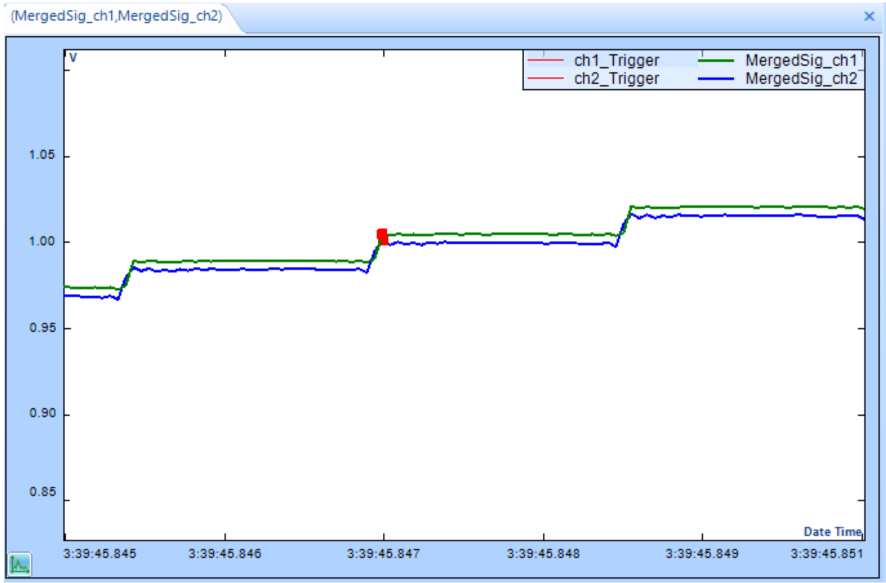
The following example demonstrates how signals are plotted after a sampling rate correction is applied using attached time stamp signals.

Original file name: REC0058, REC0131. These are recorded from two GRS units. When recorded, a source signal is fed into the input ends of both units using a T-splitter port. Both GRS units have installed GPS receivers to communicate with GPS satellites. The recording duration is about 40 minutes.

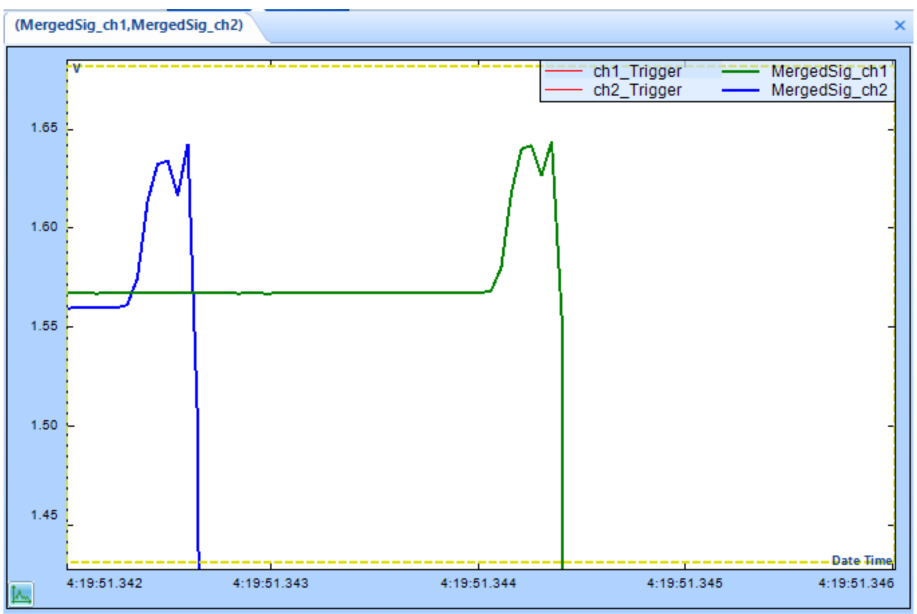
Nominal sampling rate:
REC0058: 20,480Hz
REC0131: 20,480Hz

Sampling rate after the correction:
REC0058: 20,480.026214158832 Hz
REC0131: 20,480.011221929413 Hz

The following plot shows these two signals in the time zone of the triggering point:

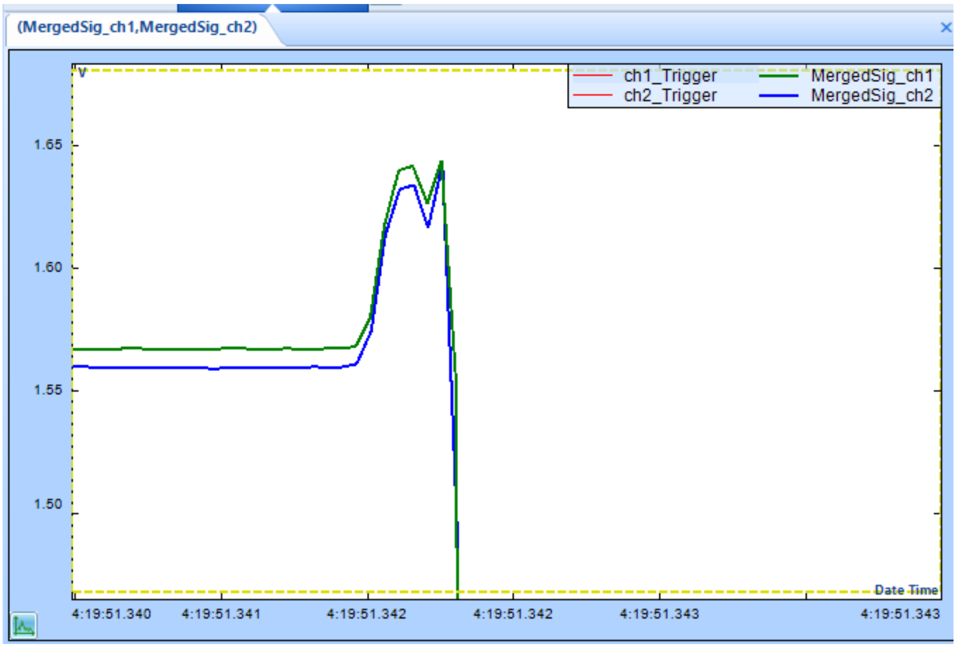


The following plot shows the transient event at the end of a recording without a sampling rate correction. It shows the time difference between two signals is about as large as 36 sampling points:



The time difference shown above is caused by using the nominal sampling rate which is inaccurate.

The following plot shows the same transient event at the end of recording while the first order correction is applied to the sampling rate of both signals:

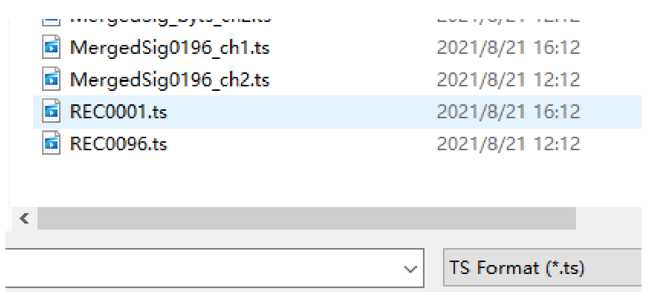


The two signals are lined up beautifully in the horizontal axis.

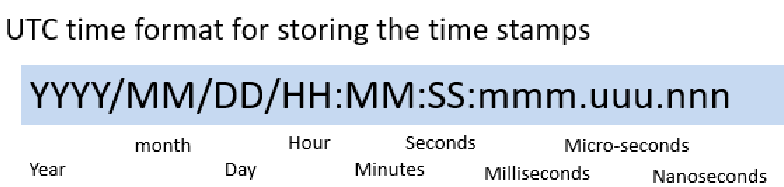
Show Time Stamp Files

This section demonstrates how to display time stamp signals.

Click **Data Files** ->**Open File**, select **TS Format**, which is a special file format created to store a sequence of time stamps.



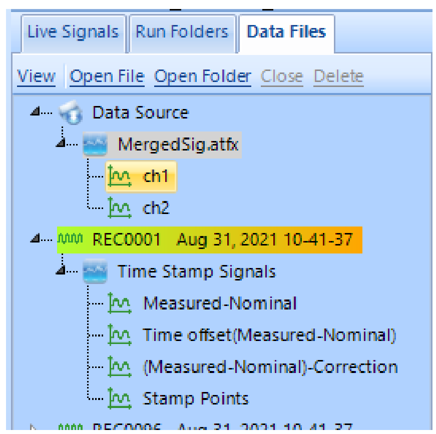
The software uses the following format to store time stamps:

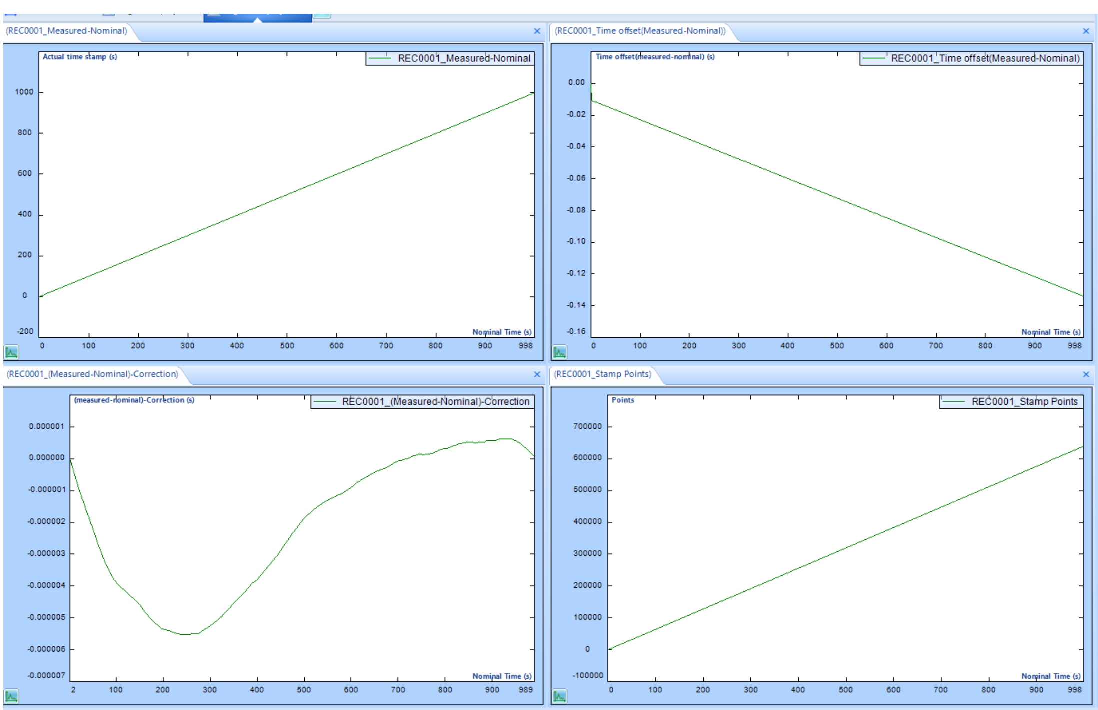


In the software, each time stamp piece is an instance of a structure:

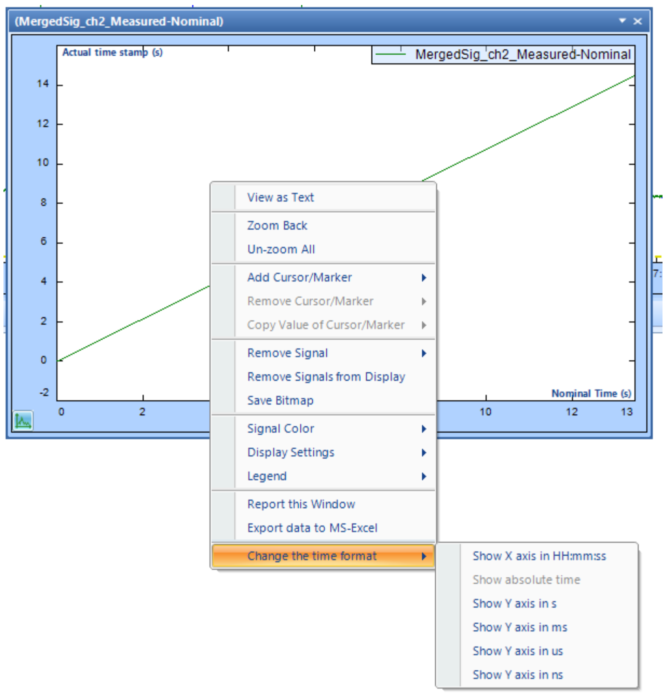
```
typedef struct tagSystemTime
{
        unsigned short wYear;
        unsigned short wMonth;
        unsigned short wDay;
        unsigned short wHour;
        unsigned short wMinute;
        unsigned short wSecond;
        unsigned int   wNanoSec;
} SYSTEMTIME, *LPSYSTEMTIME;
```

There will be four derived time stamp signals under the Time Stamp Signal node in tree view:

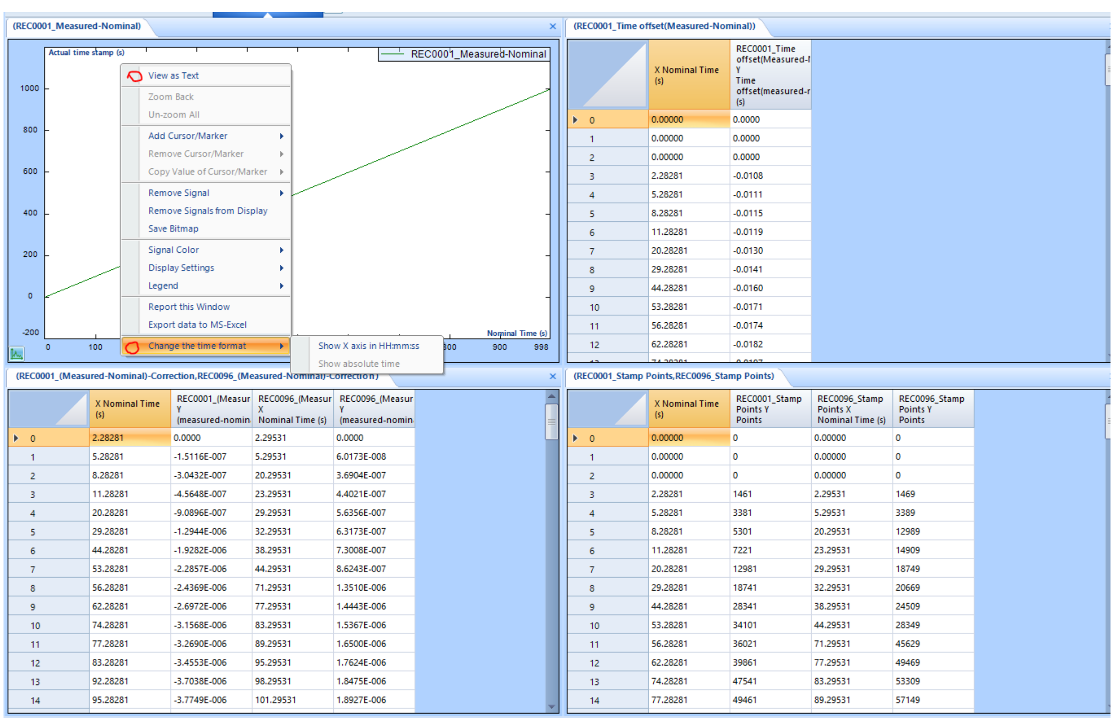




Right-click on the display window to select a different time unit:



Time stamp signals can also be viewed in text mode. The format of time stamps can be shown as seconds or in the format as HH:mm:ss. Set these values in the right-click menu.



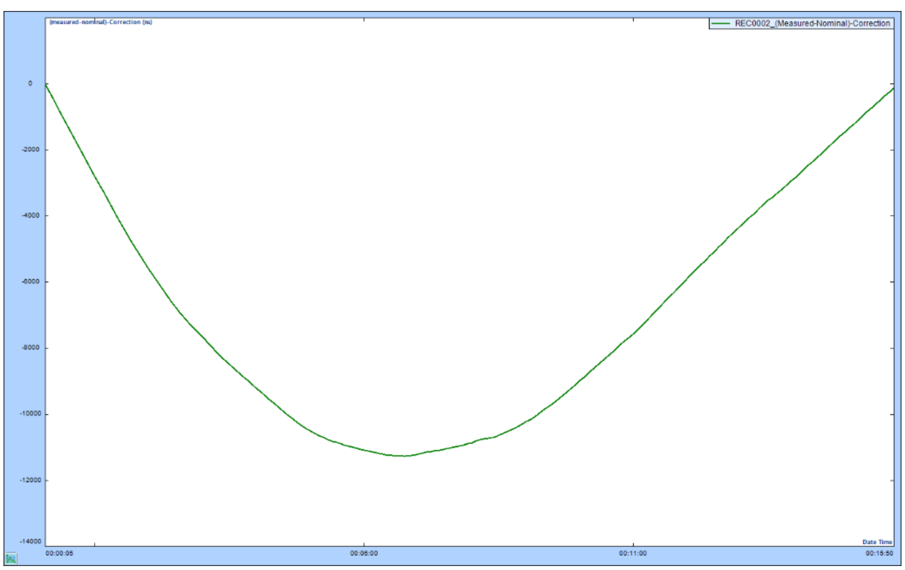
More than one time stamp signal can be viewed together.

(REC0001_Measured-Nominal,REC0001_Time offset(Measured-Nominal),REC0001_Stamp Points)

	X Nominal Time (s)	REC0001_Measured-Nominal Y Actual time stamp (s)	REC0001_Time offset(Measured-Nominal) Y Time offset(measured-nominal) (s)	REC0001_Stamp Points Y Points
0	0.0000	0.000000	0.000000	0
1	0.0000	0.000000	0.000000	0
2	0.0000	0.000000	0.000000	0
3	2.2828	2.272049	-0.010764	1461
4	5.2828	5.271678	-0.011135	3381
5	8.2828	8.271306	-0.011506	5301
6	11.2828	11.270935	-0.011877	7221
7	20.2828	20.269822	-0.012991	12981
8	29.2828	29.268709	-0.014104	18741
9	44.2828	44.266853	-0.015960	28341
10	53.2828	53.265740	-0.017073	34101
11	56.2828	56.265369	-0.017444	36021
12	62.2828	62.264626	-0.018186	39861
13	74.2828	74.263142	-0.019671	47541
14	77.2828	77.262771	-0.020042	49461
15	83.2828	83.262029	-0.020784	53301
16	92.2828	92.260915	-0.021897	59061
17	95.2828	95.260544	-0.022268	60981
18	98.2828	98.260173	-0.022639	62901
19	101.2828	101.259802	-0.023010	64821
20	110.2828	110.258689	-0.024123	70581
21	113.2828	113.258318	-0.024494	72501

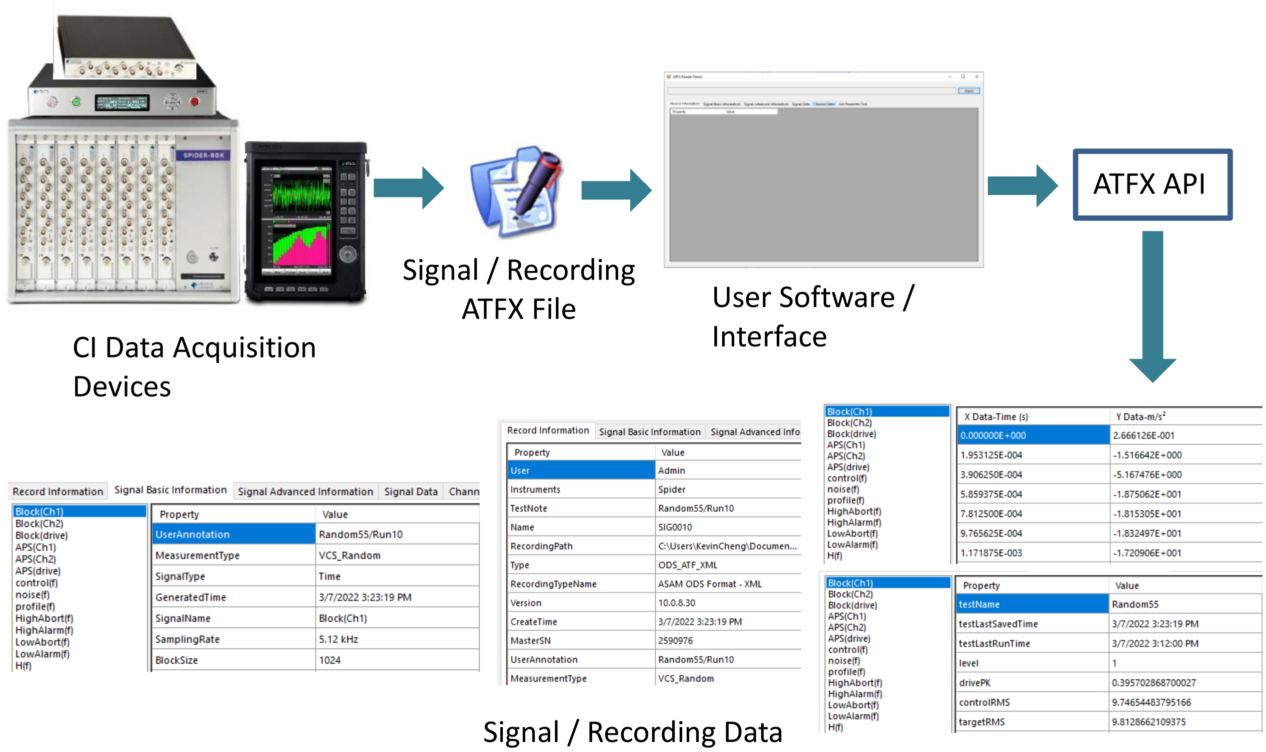
What if you see the corrected time stamps have a parabola shape?

The single direction parabola curve of time stamps after the first order correction indicates that the bias between the local clock of ADC sampling and GPS time base is still changing into one direction, as seen in the following screenshot.



If the temperature of the CI-GRS is not settled, the frequency of the local clock may still change. When the CI-GRS is in a stable temperature environment, this problem usually does not occur. This problem will not affect the phase match calculation of cross-spectrum based on the algorithm developed in post processing.

Post Analyzer



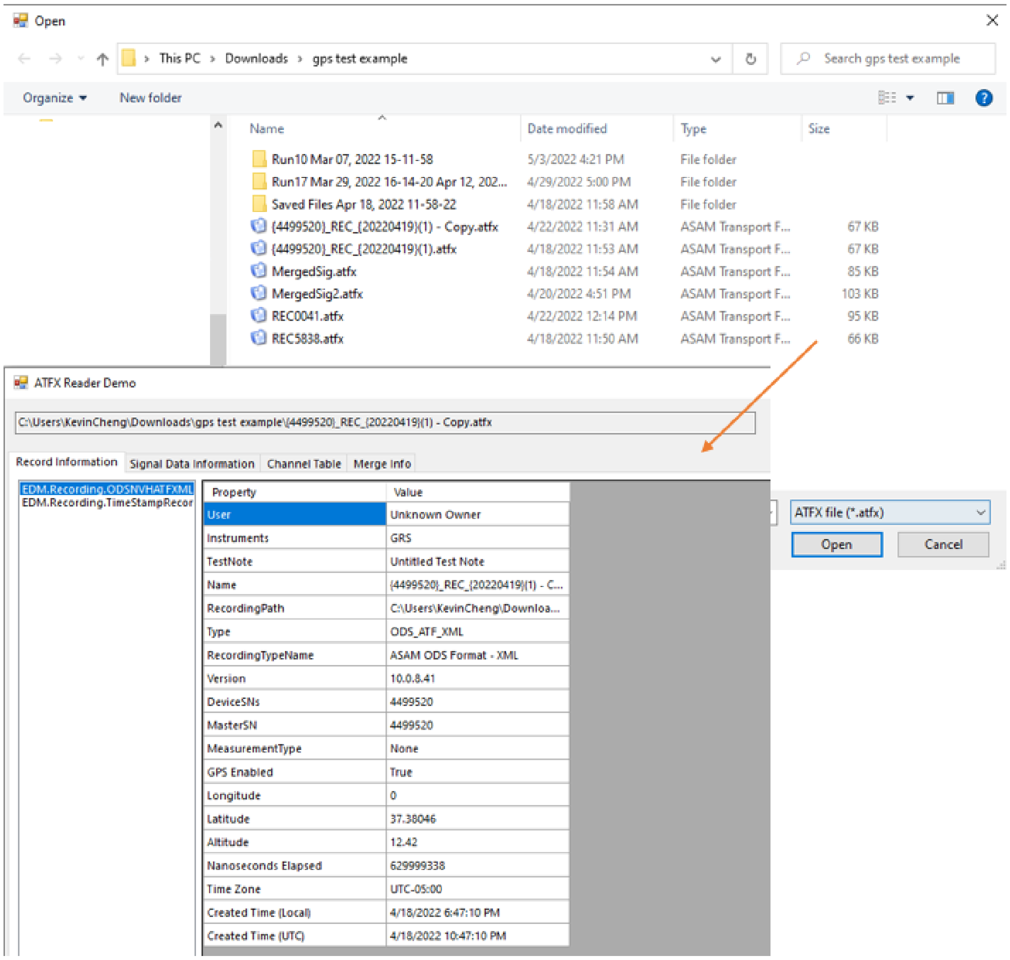
The Crystal Instruments (CI) ATFX ODS Signal Reader Application Programming Interface (API) consists of 2 Windows Dynamic-Linked Libraries (DLL) providing third-party applications an interface to access the signal data stored in the ASAM Transport Format XML (ATFX) files.

ATFX files are formatted according to the Association for Standardization of Automation and Measuring Systems (ASAM) Open Data Services (ODS) standardization. This is a standard dedicated for storing vibration data and its different forms. CI software natively stores its data using the ATFX format, for both signals and recordings.

For details about the ATFX ODS format please refer to the official website:
https://www.asam.net/standards/detail/ods/wiki/

The .atfx files are xml-based files which store the signal data along with all the attributes of the signal data including recording properties and time created, length of recording, number of channels, channel parameters (e.g., input channel sensor and sensitivities), geographic coordinates, sampling rate, high pass filter, etc. The .atfx files contain a reference to a .dat file that are well-defined for storing both raw time data as well as processed spectral data, calculated from methods including Fourier Transform, Frequency Response Functions, Cross-Power Spectrum, Octave Spectrum, etc.

Two additional file types referenced by the .aftx file contains raw data: .ts and .gps. The .ts file is a TimeStamp recording containing an accurate measure of when a recording was saved with accuracy down to nanoseconds. The .gps file is a GPS recording containing locational data of where a recording was saved (e.g., latitude, longitude, altitude).



The CI Data File Reader API provides end-users with a streamlined file reading and browsing library to decode ATFX, TS and GPS files. Users can integrate the API with their own custom developed application. CI currently supports Windows-based programs, ideally written in C#. The same API also supports Python, MatLab and LabView.

The API offer direct calls to the ASAM ODS model classes and objects used to store data saved in the ATFX file, such as calling the recording NVHMeasurement and NVHEnvironment to read the DateTime with nano seconds elapsed. The API also provides a Utility class that has functions to return data from the ATFX file without the user needing to understand the complexity of the ASAM ODS model classes. Examples include the Utility GetListOfAllSignals which returns a list of signals contained in an ATFX file or the Utility GetChannelTable function which returns a 2D list of strings, where each list is an input channel row.

It is also possible to read any signal, time, or frequency, in other engineering units (EU) such as Acceleration m/s^2 to g. Frequency domain signals can also be read in other spectrum types such as

EU_{rms} to EU_{Peak}. The signal function GetFrame accomplishes these tasks and users can pass in parameters to return a converted signal frame data saved in the ATFX file.

When the ATFX API reads the ATFX file, there may be some differences in the signal frame data due to some display related parameters such as the spectrum type not being saved into the ATFX file. The spectrum type is $EUrms^2$ by default. Engineering units are saved into the ATFX file and should be the default EU when reading the signal frame.

For detailed information regarding the ATFX API, please refer the CI Data File Reader API Documentation available for download on the CI site:

https://www.crystalinstruments.com/basics-of-test-and-measurement
https://www.crystalinstruments.com/programming-corner

Post Analyzer Standalone and License Management Website

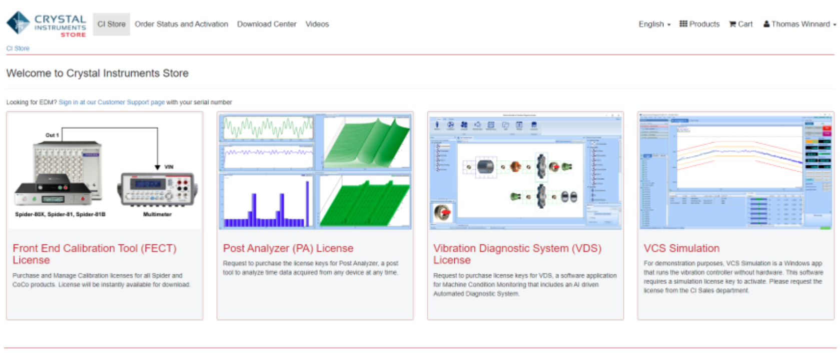
Overview

Users are offered a new option to request, purchase, install and utilize post analysis functions. Post Analyzer software starting from the 10.0 release will not be associated with a hardware serial number. Its installation will be separate from EDM to operate as a standalone feature, offering users the benefits of Crystal Instruments' unique post data processing ability.

The new Online License Management website will further expedite and simplify the license key process. Convenient features such as online payment options are planned for future releases. Each OLM issued license will be proprietary to the computer on which the subscription was ordered, and the CI server will keep track of the total activated licenses. This format provides one license for each computer and is not sharable across multiple computers. New customers will need to purchase one license for each machine intended to use the software. The same license can be used on a different computer as long as it is deactivated from the previous one. Current customers and licenses with existing Post Analyzer privileges will not be affected.

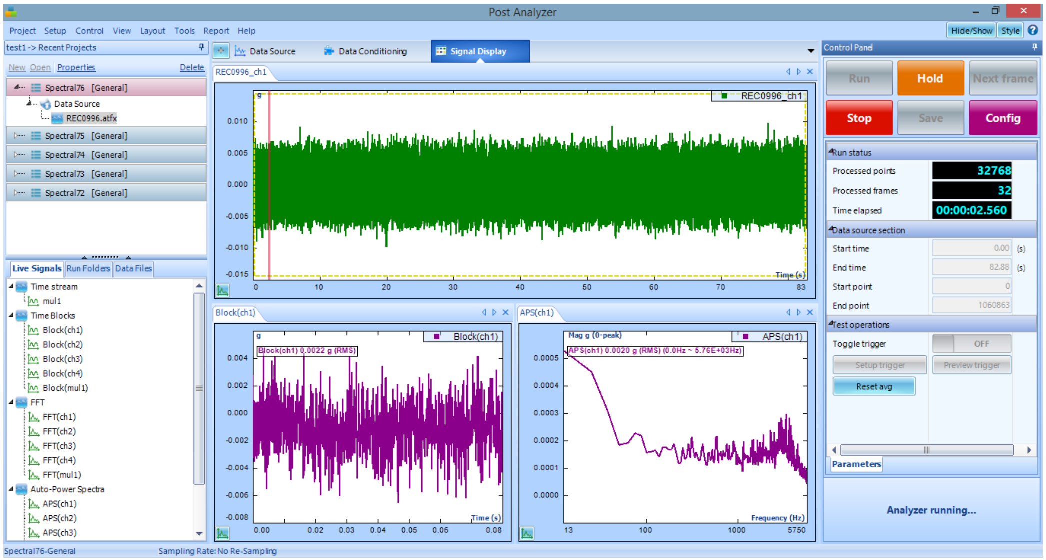
Online License Management – OLM

Starting with PA version 10.0, the license file is managed on Crystal Instruments' Online License Management website at the following address: http://license.go-ci.com/



Customer can begin by creating an account in OLM and submitting a request for Post Analysis functions. A quotation response will be sent through email from CI Sales. Upon receiving a payment, the sales team will create a license and send an email. The customer may then download the Post Analysis software and use the newly issued license key.

Post Analyzer 10.0 Configuration Update

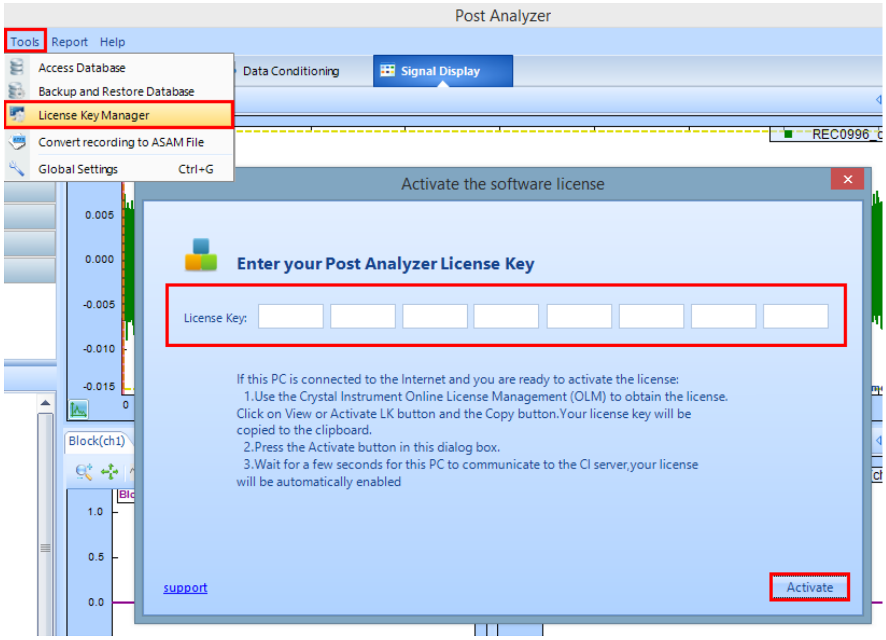


Three bundles of PA are offered for ordering convenience: **PA Viewer** allows the user to view data and create reports; **PA Basic** has FFT spectral analysis, curve fitting, demodulation spectrum and 3D signal display functions; **PA Premium** has more advanced functions including File Converter, offline sine reduction, real-time filters, octave filters and order tracking.

Advanced functions can be ordered separately after the user has purchased PA Basic. This major change in this release allows the user to spend less money and obtain the advanced processing functions as necessary.

Activating a License

Similar to EDM, the customer begins by installing the PA software. Once installed, PA will not function until the user inputs their individual license key code. Unlike EDM, this license will no longer be in the format of a LIC file and instead will follow the typical software licensing format of a long string of characters. The OLM website will provide users access to their individual code once payment is received. Users can then copy and paste the license code within the Post Analyzer LK Manager Tool.



Next Steps

<u>CI end user customers:</u> If you believe your system is eligible for an upgrade, please go online to license.go-ci.com, register an account, and submit a request to upgrade. When submitting the request, please mention the hardware serial number associated with the PA software you wish to upgrade. CI will follow up with you shortly.

<u>Distributors and CI business partners:</u> please inform your end user about this change and coordinate with Crystal Instruments to upgrade their software.