Software Manual for TVIB

TSAP 201 TVM 203 TSLM 204 TIST 205 TWGM 206 TOA 207 TB 208 TOA 209

This user guide is designed to provide documentation for TVIB:TSAP 201, TVM 203, TSLM 204, TIST 205, TWGM 206, TOA 207, TB 208, TOA 209 . Read these instructions carefully

Description: Software Manual for TVIB: TSAP 201 TVM 203 TSLM 204 TIST 205 TWGM 206 TOA 207 TB 208 TOA 209

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Contents

1

General Instructions 6
- Installing the Software 6
  - System Requirements 6
    - Installing the TVIB Software 6
- Launcher 8
- Channel Settings 8
- Calibrator function 10
- Software Modules 12

Time & FFT Spectrum Analyzer with Post Processor (TSAP 201) 13
- Module Interface 15
- Menubar 16
- Function Bar 19
- Settings Recorder Mode 22
  - Run Control 24
  - Triggering 25
  - Filter 27
- Settings Post Processor Mode 32
- Notification Bar 33
- Time Signal 35
- FFT Spectrum 38
- Phase Analysis 44
  - Relative Phase Measurements 45
  - Absolute Phase Measurements 46
- Envelope Detection tile 46

Advanced Vibration Meter (TVM 203) 48
- Module Interface 48
- Function Bar 48
- Settings 49
  - General Settings tab 50
  - Vibration Level Meter Settings tab 53
- Vibration Parameter 56
- Saving the Measurement 58
- Vibration Level Meter 59
- Time Signal 60

Sound Level Meter (TSLM 204) 65
- Module Interface 66
- Function bar 67
- Settings 68

General Settings tab 69
Octave band Setting 71
Acoustic Parameter 73
Octave Band Analyzer 75
Saving the measurement 76
FRF Test (TIST 205) 77
Module Interface 77
Menu bar 78
Function Bar 79
Settings 79
Run Control 81
Sampling 82
FRF 86
Triggering 88
Filter 91
Notification Bar 99
Signal Indicator 100
Real and Imaginary 102
Bode Plot 109
Waveform Generator (TWGM 206) 114
Module Interface 115
6.1 Random Noise 116
Parameters 116
Burst Mode 116
Usage Notes 116
Order Analysis (TOA 207) 122
Field Balancer (TB 208) 123
Orbit Analysis (TOA 209) 124
Troubleshooting 125
Support 127

General Instructions

Installing the Software

System Requirements

Processor	Pentium 4M/Celeron 866 MHz (or equivalent) or later (32-bit) Pentium 4 G1 (or equivalent) or later (64-bit)
RAM	4 GB
Screen Resolution	1024 x 768 pixels
Operating System	Windows 10 or higher
Disk Space (Min)	2 GB

For installing the software in Windows 10/ higher versions is recommended, follow the steps below.

Installing the TVIB Software

1. Download T-ViB Installer.& Driver.
2. Run the Install.exe file and complete the installation.

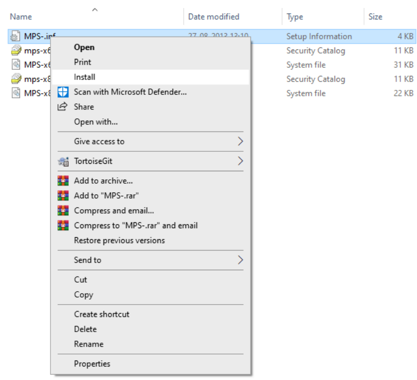
Installing the Driver file

Note:

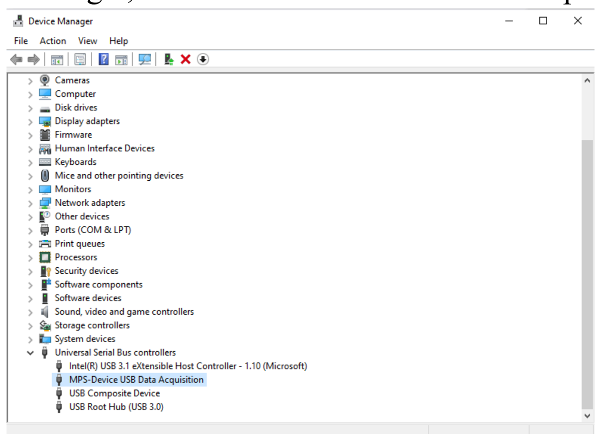
Only for Phonovibe Q/O/HD
Phonovibe D *does not require any additional driver file installation.*

Step 1: Download & Extract the driver, and install the **.inf** file.

Note: Make sure the device is not connected to the PC while the driver is installed.

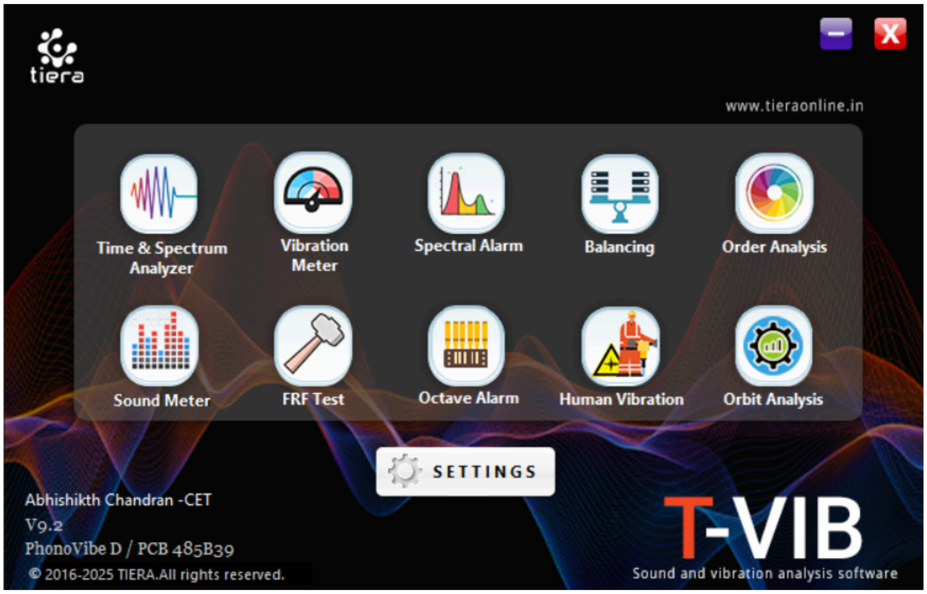


Step 2: After the installation is complete, connect the DAQ.
Step 3: Check in the Device Manager, make sure the device is detected properly.

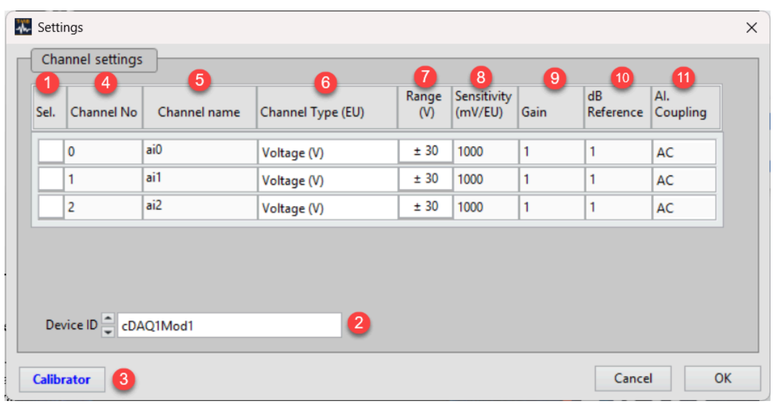


3. Open T-VIB from the desktop or Start menu to begin.

Launcher



Channel Settings



1 - Active Channel: Click the check box to enable a channel.

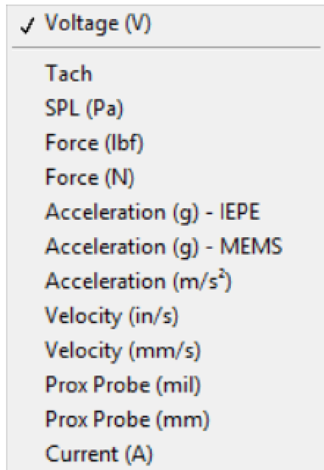
2 - Device ID: This displays the ID of the Data Acquisition Device connected to the software. 0 is the Default ID for Phonovibe Q/O/HD.

3 - Calibrator button: Click to calibrate the sensor in real-time with an independent calibrator (Vibration Calibrator/ Microphone Calibrator) connected to the Data Acquisition device. This option is only available for Microphone & Accelerometer

4 - Channel No. : Number specified for channel

5 - Channel Name: Give a suitable channel name to denote the data from a particular sensor in the graphs or measurement.

6 - Channel Type: Specifies units of measurement & the Sensor connected to the channel. Following are the general units of measurement & Sensor type used.



7 - Range: Enter the maximum range of voltage, the sensor would output to the Data Acquisition device.

8 - Sensitivity: Sensitivity is the voltage output (in millivolts) per unit of the Engineering Unit.

9 - Gain: The signals which are amplified using external hardware can be accounted for by the Gain option .

10 - dB Reference: Is the reference with which the logarithmic (in decibels) is measured. The decibel is a logarithmic ratio between two numbers – a measured value and a reference value. It is shown in two forms below: Equation 1 for POWER quantities, and Equation 2 for field AMPLITUDE quantities.

$$dB = 10\log(\frac{W}{W_{ref}})$$

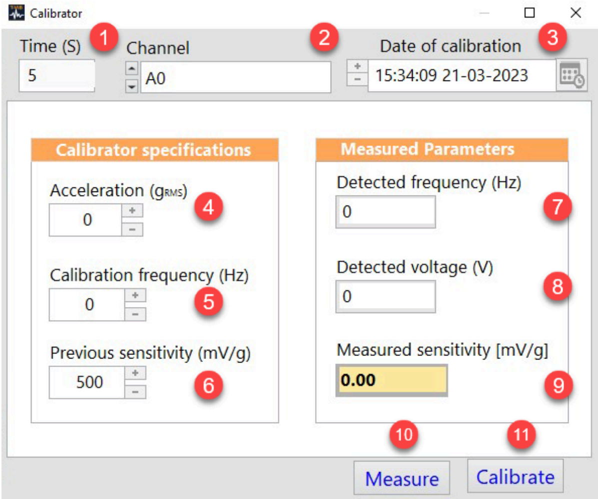
$$dB = 10\log(\frac{X}{X_{ref}})^2 \quad or \quad dB = 20\log\left(\frac{X}{X_{ref}}\right)$$

11 - AI Coupling: Analog Input coupling refers to the coupling options available in the Data Acquisition device.

Calibrator function

Software option to be used with the sensor mounted on an independent calibrator with the input signal parameters to be entered in the calibration window.

Note: Only for Microphones & Accelerometers



1 - Time (s): Time frame of the signal collected from the sensor.

2 - Channel: The name of the channel which is to be calibrated.

3 - Date of Calibration: Set the date on which calibration is done.

4 - Acceleration (g rms): The acceleration level of vibration in the calibrator.

5 - Calibration frequency: The frequency of vibration set in the calibrator.

6 - Previous Sensitivity: The previous sensitivity value entered by the user or obtained in the last calibration.

7 - Detected frequency: The frequency of vibration measured from the calibrator by the sensor in real time.

8 - Detected voltage: The Voltage Output from the sensor corresponds to the input measured from the calibrator by the sensor in real time..

9 - Measured Sensitivity: The sensitivity value of the sensor measured from the Calibration procedure.

10 - Measure: Press the Measure button to calculate the sensitivity value from the calibration procedure.

11 - Calibrate: Press the calibrate button to automatically set the sensor sensitivity value measured from the calibration procedure.

Software Modules

T VIB Software is used for Sound & Vibration Testing applications. It is modular in functionality. The user can purchase the modules which suit their testing requirements.

No.	Module	Ordering Code	Brief Description
1.	Time & Spectrum Analyzer with Post processor	**TSAP 201**	● Real-time vibration measurement and analysis ● Post-processing ● Frequency Filtering ● Software Triggering ● Time Analysis ● Frequency Analysis
2.	Phase Analysis	**PH 212**	● Relative Phase : Max, Min, Range, Coherence ● Absolute Phase : Max, Min, Range, Coherence
3.	Envelope Detection	**EV 213**	● Detect the centre frequency automatically ● Envelope signal ● Envelope Spectrum
4.	Basic /Advanced Vibration meter	**TVM 202/203**	● Measuring Vibration Levels. ● Trending the Vibration Levels. ● Classifying the Health of the Machine using **ISO 10816-3** based filters or based on custom settings. ● Apply custom filter settings to calculate vibration levels. ● Apply custom time weighting settings to calculate vibration levels.
5.	Sound Level meter	**TSLM 204**	● Measuring the Sound Levels. ● Measuring the Octave Spectrum .
6.	FRF Test	**TIST 205**	● Measuring the transmissibility ● Measuring of resonant frequencies.
7.	Order Analysis Basic/Advance	**TOA 207**	● Measuring Rotating machinery, non stationary signals ● Measure the Order Spectrum ● Order Tracking for Ramp up / Coast down ● Envelope Analysis for Bearing/Gear Faults/Electric Motor Fault Detection
8.	Balancer	**TB 208**	● Balancing rotating machines ● Single Plane ● Dual Plane ● Influence Coefficient Method
9.	Orbit Analysis	**TOA 209**	● Analyzes Shaft Motion. ● Orbit Plot ● Time base plot
10.	Human Vibration meter	**THVM 210**	● Measure Vibration levels ● Use Hand Arm/ Human Body weighting filters

11.	Waveform Generator	**TWGM 206**	• Waveform generator is used for generating signals internally for the T-Xcite Series Electrodynamic Shakers . • Signals such as Pink/White/random noise/NoiseBurst FrequencySineSweep,FrequencySineSweepburst , Log/Linearsweep Band pass filter for noise

Time & FFT Spectrum Analyzer with Post Processor (TSAP 201)

The Time & FFT Spectrum Analyzer with Post Processor Module has functions such as Waveform Recording & Post-processing, Spectrum, Signal Generator, Triggering, Filtering, Windowing, Integrating, etc. In the coming sub-sections, we will explore these features in details

Module Interface

The interface can be divided into the following features

1. Menu bar
2. Function bar
3. Signal Generator *(reserved for T-Xcite Series)*
4. FFT Spectrum tile
5. Time Signal tile
6. Notification bar
7. Envelope Detection tile

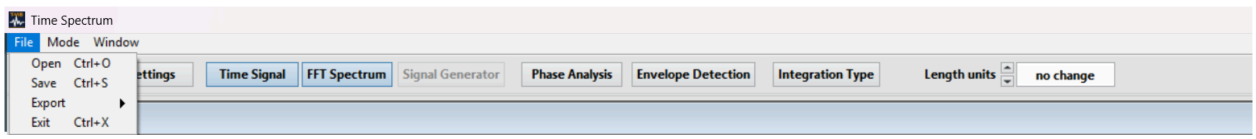
Menubar

The menu bar deals with basic features such as saving the measurement, opening the saved data, switching modes of operation (recorder or post processor), automatically arranging the tiles, etc.

The options are briefly described below

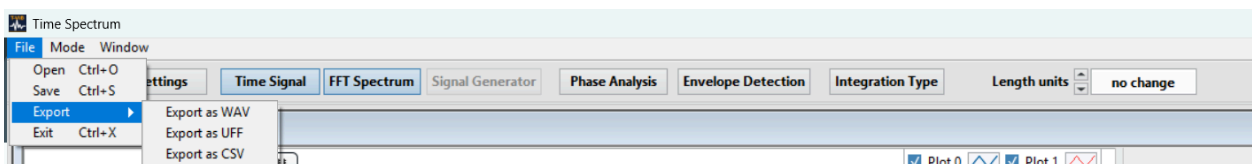
1. File

The file menu consists of the following options



a. Open: This menu option allows the user to select a saved file in the **.tdms** format.

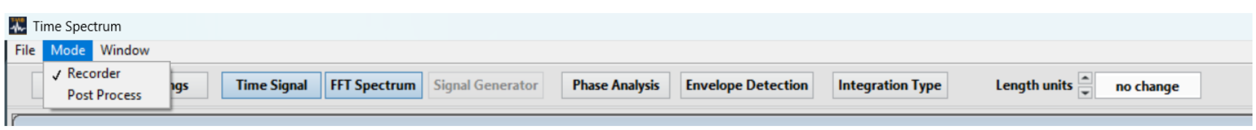
b. Save: This menu option allows the user to save data acquired in the recent acquisition in the **.tdms** format.



c. Export as WAV/ UFF: This option allows the user to export the data acquired in **.wav** & **.uff** File format.

d.Exit: This menu option allows users to exit the module and go to the application launcher.

2. Mode: The mode menu consists of the following options. Each mode has a unique setting.

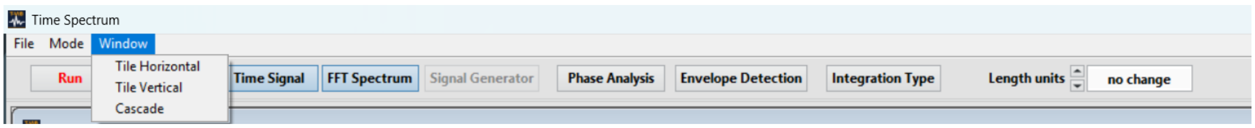


a. Recorder: In this mode, the user can measure data in real-time and save the data. This allows the user to take measurements onsite/ perform a test in the laboratory.

b. Post Process: This mode allows the user to post-process the data recorded offsite using signal processing techniques such as windowing, filtering, etc.

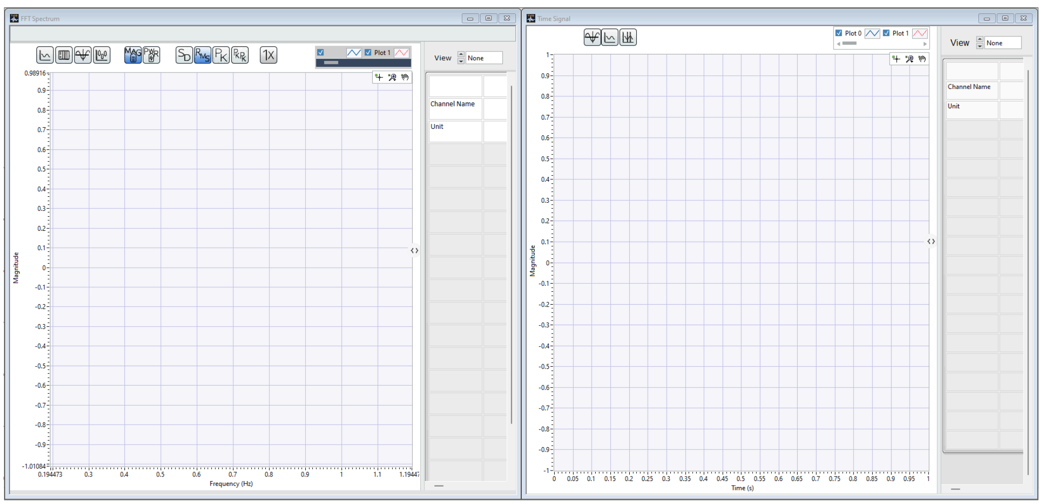
3. Window

The window menu basically arranges the tiles symmetrically and consists of the options as below

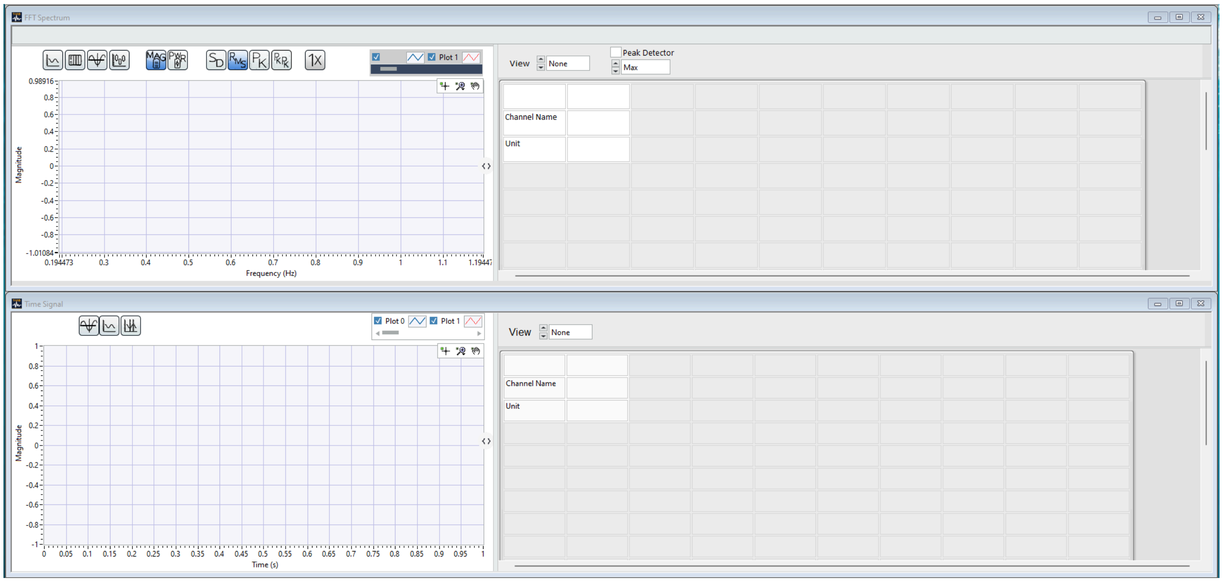


Note: The software allows the user to use single/multiple tiles simultaneously (each tile has a functionality, in this module Time signal tile gives the waveform, FFT Spectrum tile gives the frequency spectrum).

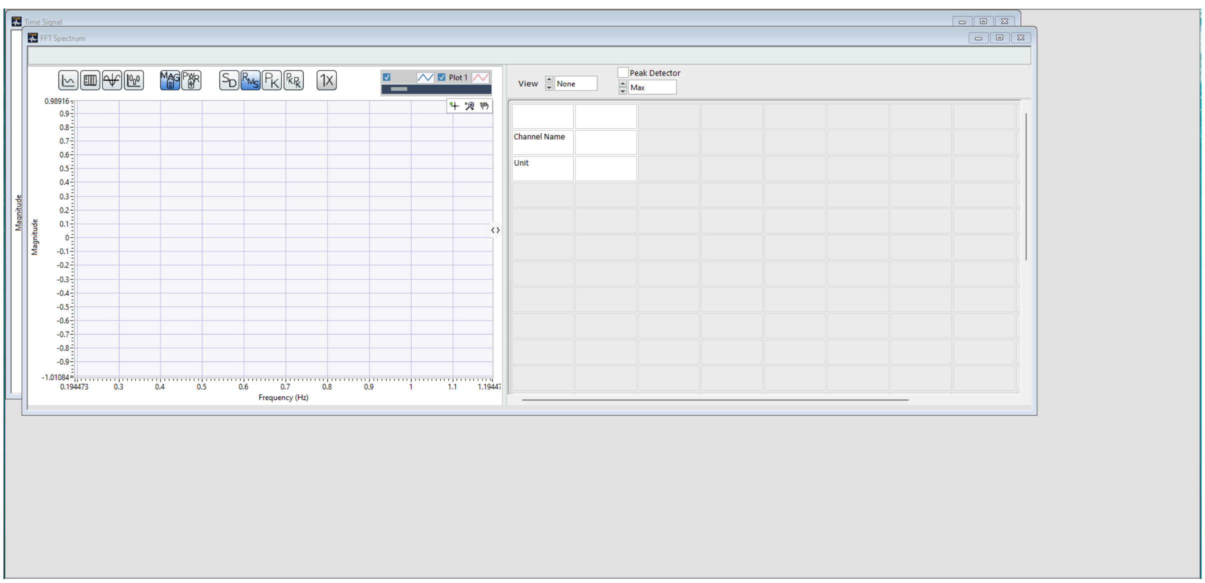
a. Tile Horizontal: This option arranges the tiles (whichever tile is being used for measurement) horizontally.



b. Tile Vertical: This option arranges the tiles (whichever tile is being used for measurement) vertically.

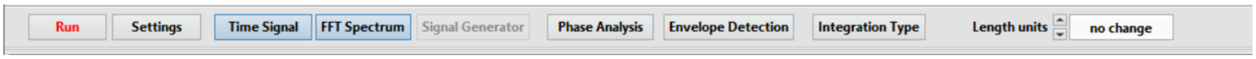


c. Cascade: This option arranges the tiles (whichever tile is being used for measurement) one over the other.



Function Bar

The function bar is composed of buttons; it performs a variety of functions as detailed below.



The blue highlight on Time Signal & FFT Spectrum buttons indicates that these tiles have been chosen by the user for the waveform and the spectrum Measurement.

1. Run button

In the recorder mode, this button can be used to start & stop the acquisition.

In the post-process mode, this button can be used to open the recorded .tdms format measurements.

2. Settings

In the recorder mode, settings can set up an acquisition & various functionality of the software. We will discuss the various options in the settings in the next section.

3. Time Signal

On pressing the time signal button the Time signal tile pops up on the screen. The tile comprises the graph with options like cursors, the tabular section allows the users to take note of the precise values of the peaks, cursor values & units.

4. FFT Spectrum

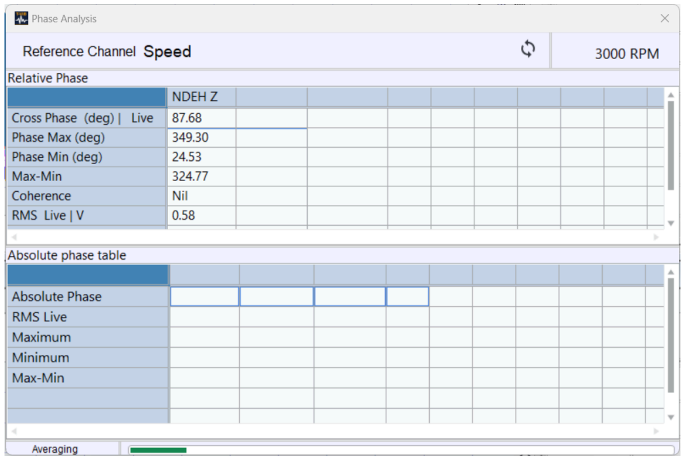
On pressing the time signal button the FFT spectrum tile pops up on the screen. The tile comprises the graph with several options for selecting the amplitude value type (such as RMS, peak, peak to peak etc.) and cursors, the tabular section allows the users to take note of the precise values of the peaks, cursor values & units.

5. Signal Generator

This option is reserved for the T-Excite Series (i.e. electrodynamic shaker series).

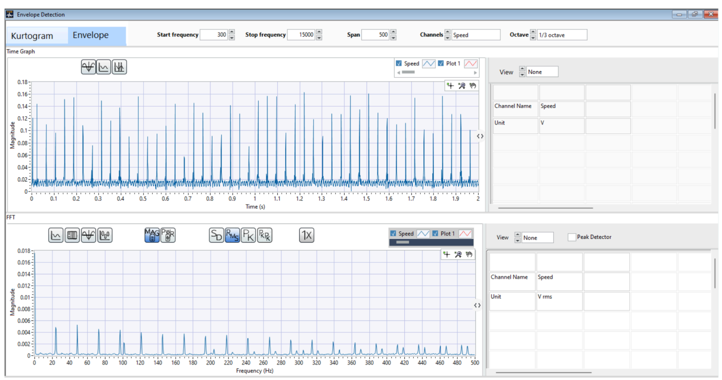
6. Phase Analysis

On pressing the phase analysis button the phase analysis tile pops on the screen. It consists of a tabular column with the values of the parameters of the relative and absolute phase measurements of each channel listed in the column.



7. Envelope Detection

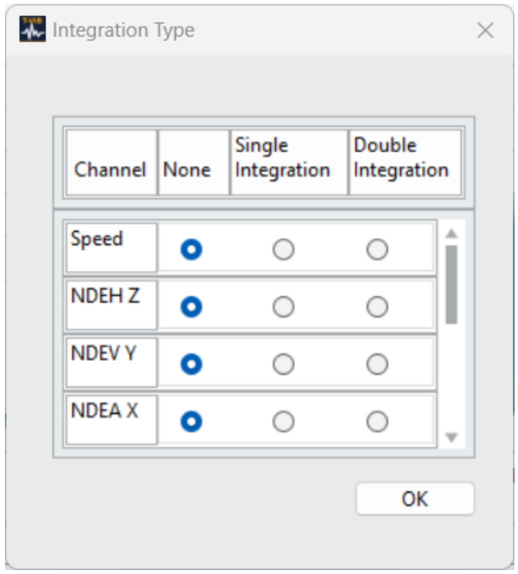
On pressing the envelope detection button the envelope detection tile pops on the screen. This consists of two sections: the kurtogram and the envelope.



The kurtogram displays the kurtosis over a frequency range and is used for calculating the centre frequency automatically for performing the envelope operation. The user has the option to manually pick the centre frequency.

8. Integration type

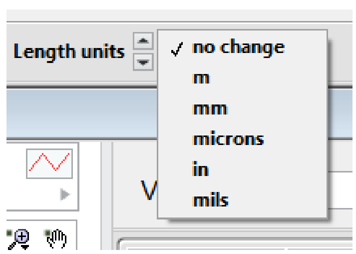
This allows the user to perform the software integration on each channel as required.



None is the default option, this indicates that no integration is performed on the data. The user has the option to perform single or double integration as required in a separate channel.

9. Length Units

We can choose the desired length units needed for the integration type selected.



Settings Recorder Mode

Below we will discuss the settings in detail of the recorder mode.

The settings tab consists of four control panes,

1. Run control settings: This setting allows the user to adjust the running mode of the acquisition **i.e.** either run the acquisition continuously and stop when the user presses the stop button or run the acquisition for a time/number of data blocks acquired as specified by the user.

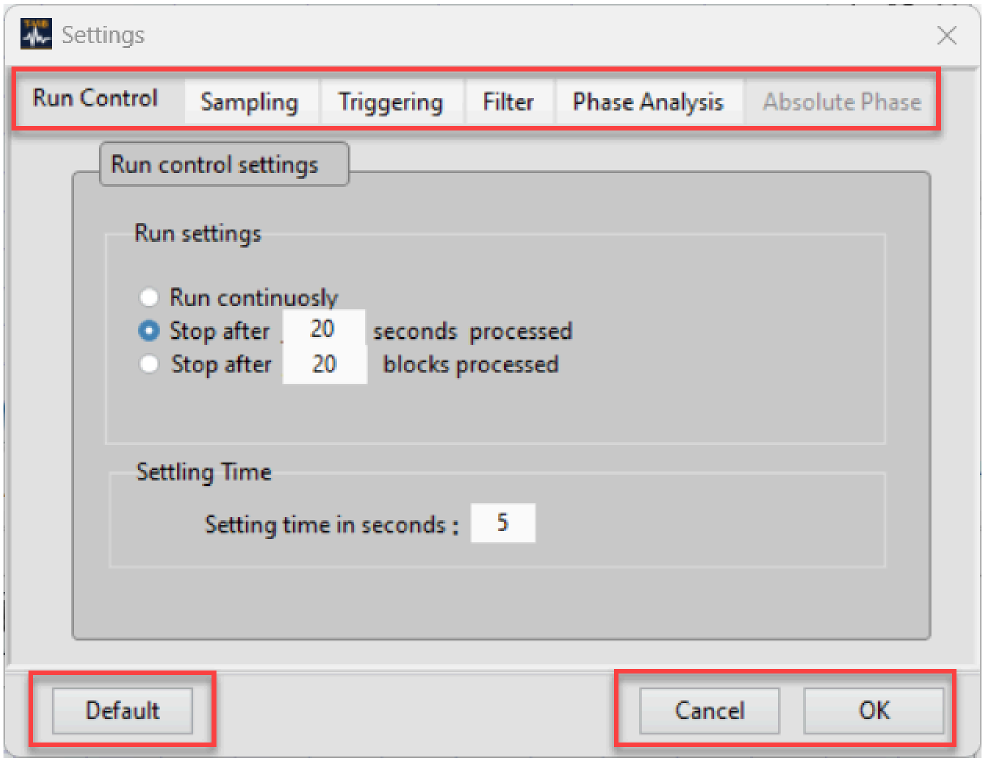
2. Sampling settings: This enables the user to fine-tune the block size (sample size), the sampling rate, and the decimation ratio. Additionally, FFT parameters such as the window type, the averaging method, the weighting mode, and the number of averages are available.

3. Triggering: This tab is for setting the Trigger Settings in case if the triggered acquisition is required. The user may configure the trigger channel, type, slope, levels, sampling rate, and hysteresis.

4. Filter: This tab is for setting the filtering option. The user may choose the type of filter, order, and design technique of the filter.

5. Phase Analysis: This tab is for setting the relative phases measurement. The user has the option to select the reference channel and the measured channel, the option to select the speed, tolerance and the averaging.

6. Absolute Phase : This tab is for setting the absolute phase measurement. The option is enabled only when one of the channels configured in the channel settings is a Tach. The user has the option to select the measured channel and the phase notation.

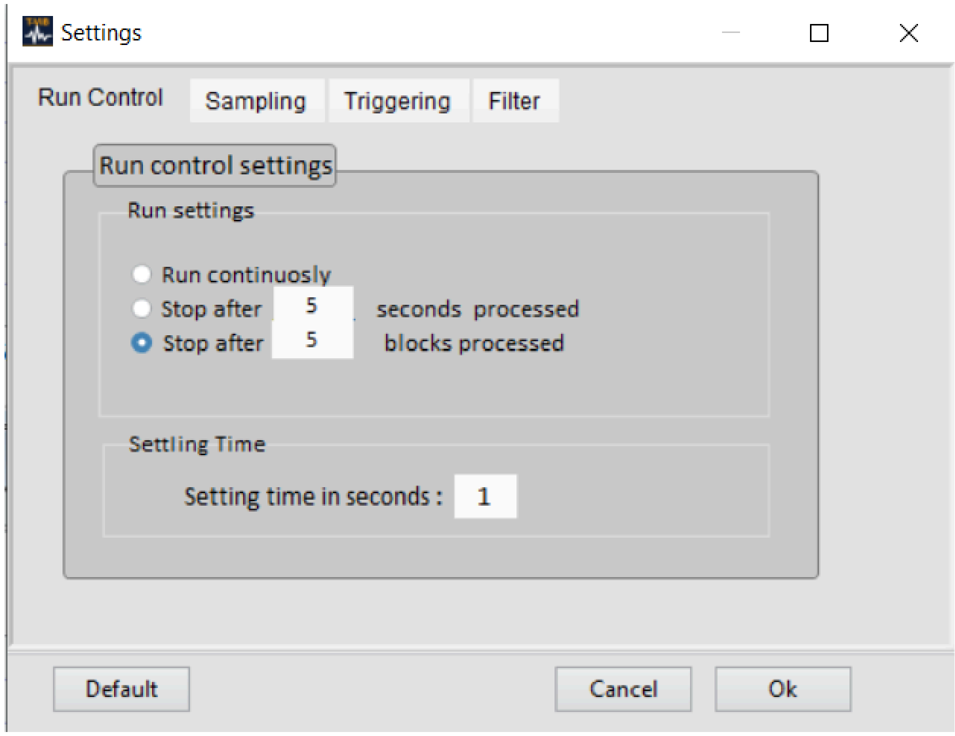


Default: This will load the settings' default values in every tab.

Cancel: Closes the settings tab. The changes made by the user on the settings will be lost and the previous settings will be loaded. Pressing the Close button on the settings window would have the same effect.

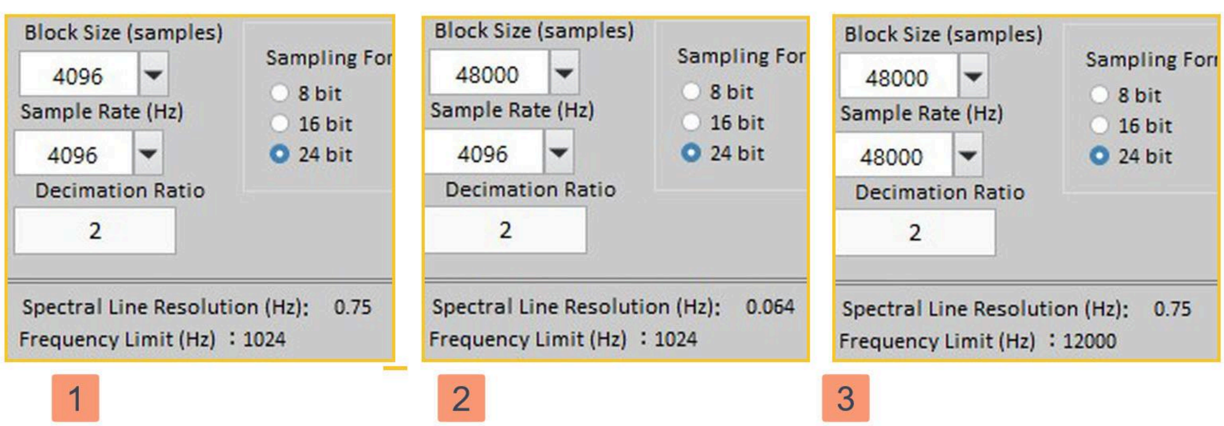
Ok: Saves the selected configuration as the present settings.

Run Control

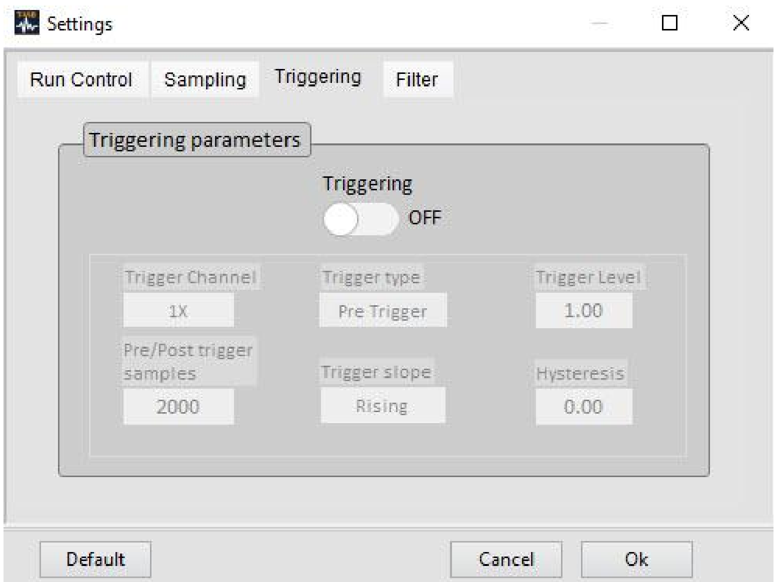


Run Settings: Enabling **Run continuously** reads samples indefinitely, whereas **Stop after** reads samples for a specified time period or a specified number of blocks.

Settling Time: Settling Time is defined as the time (in secs) required for the accelerometer's internal electronics to stabilize its bias voltage and achieve its operational condition. In this example, if we specify a time period, the software will acquire the data only after the specified settling time period has crossed.

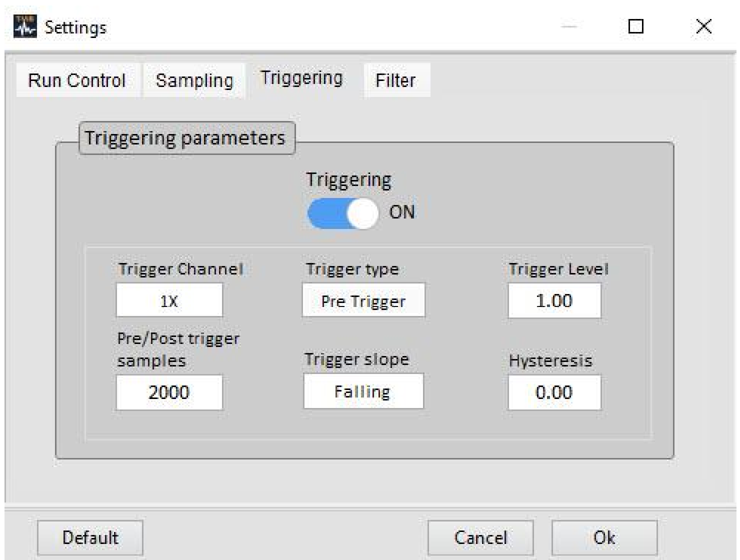


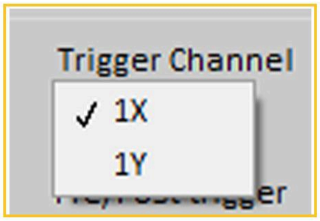
Triggering

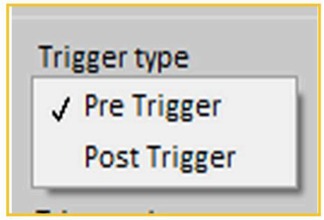


Triggering referred to here is a software triggering, this option allows to acquire a frame only when a specified level in the input signal is crossed. Signals are monitored at random periods in the absence of triggering. In the default settings triggering is disabled.

When it is enabled we can choose different channels to setup trigger, type of triggers like **Pre-Trigger** or **Post-Trigger** (pre-trigger samples refer to the samples to be acquired before the trigger has occurred & post-trigger samples would refer to the samples after which the data is to be acquired after the trigger), **Triggering level** (amplitude value at which the trigger occurs), and **Number of Samples** to read in each trigger type, **Trigger slope** (Rising or Falling) and **Hysteresis**.

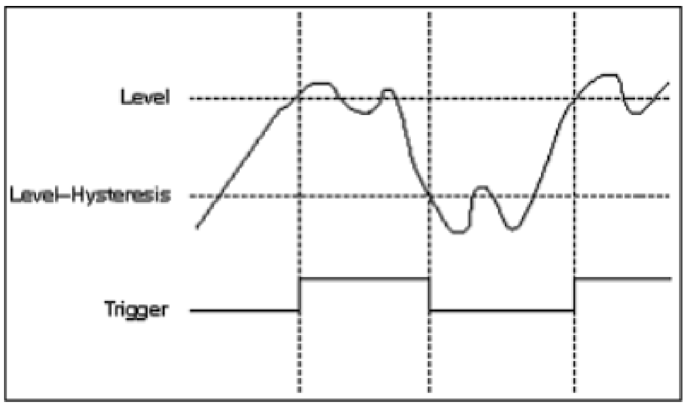






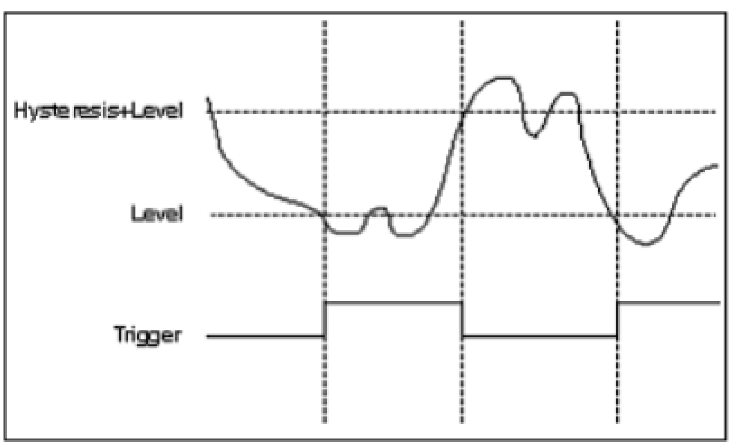
The **Hysteresis** defines the distance between the firing level and the arming level and determines the sensitivity of the trigger system.

It is often used to reduce false triggering in signals caused by noise or jitter. When hysteresis is used at a rising slope, the trigger occurs when starting under Level-Hysteresis and crossing over Level. The trigger is released when the signal passes below the Level-Hysteresis.



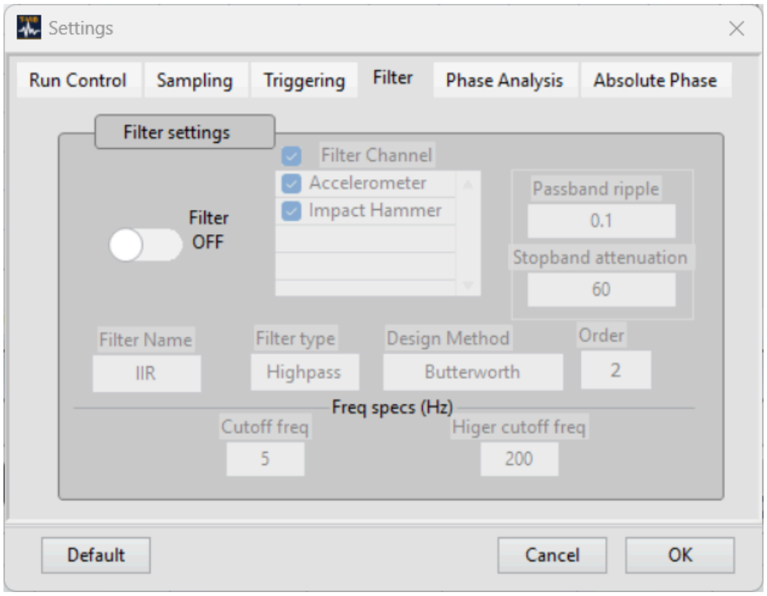
Hysteresis with rising slope

When hysteresis is used at a falling slope, the trigger occurs when starting above Level+Hysteresis and crossing under Level. The trigger is released when the signal passes above the Level+Hysteresis.

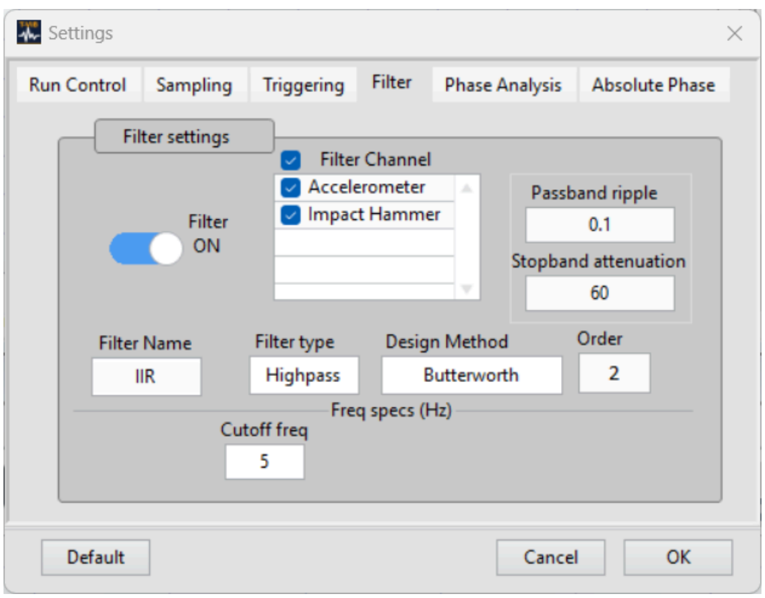


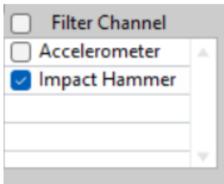
Hysteresis with falling slope

Filter



In the default case the filtering is disabled. Here, the available filter mode is IIR (Infinite Impulse Response). Infinite impulse response (IIR) filters, also known as recursive filters, operate on current and past input values and current and past output values.





This option allows the user to choose the channels to be filtered

The various modes of filters available are as follows:

Lowpass (LP): It is used to allow only low frequencies to pass. It is frequently used to isolate the high frequencies of the signal. It is widely used to suppress high-pitched sounds in speakers, some electric guitars, and radio transmitters.

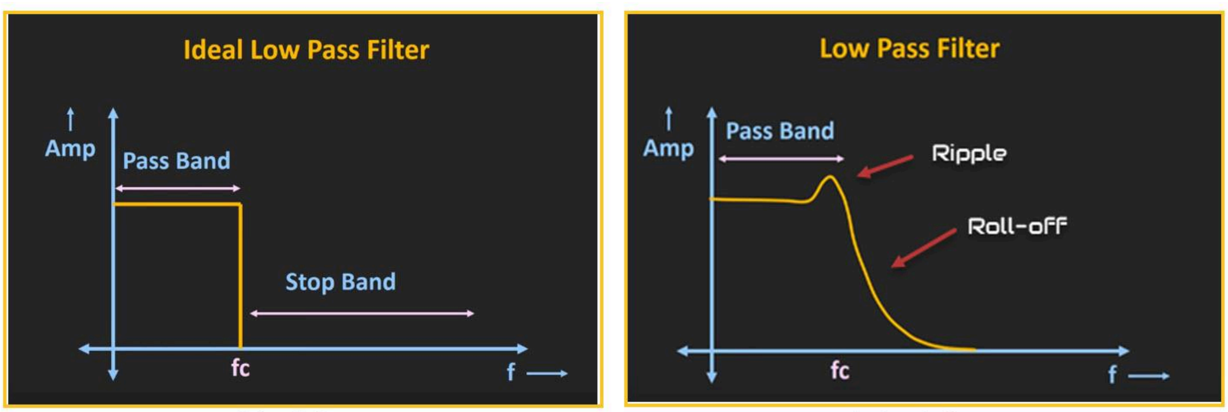
Highpass (HP): A high pass filter is used to ensure that only high frequencies of the signals are allowed for operation. It cancels out the majority of low frequencies and DC components.

Bandpass (BP): A band pass filter is a filter that combines the characteristics of a high pass and a low pass filter. It is constructed in such a manner that the LPF's cutoff frequency is greater than the HPF's cutoff frequency, enabling only a limited range of frequencies to pass through.

Bandstop (BS): It is also known as a notch filter, and is a type of filter that is used to block some frequencies of the signals while allowing others to pass through. It is the inverse of a bandpass filter and may well be made by utilizing the same input with a high pass filter and a low pass filter.

Design Method

In the case of a filter, it is nearly difficult to obtain an Ideal filter's frequency response and to overcome this limitation, several filter approximations, referred to here as Design methods, are adopted.



Ideal Case　　Actual Case

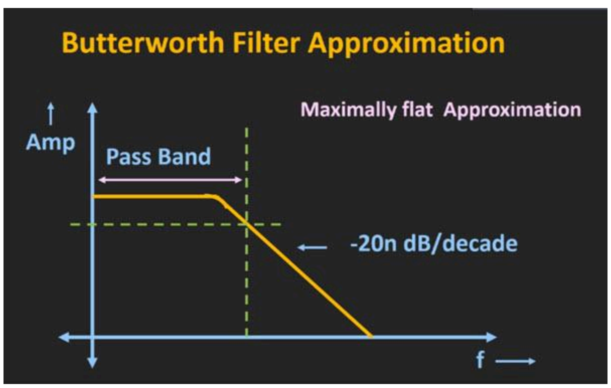
We can understand the different filter approximations by taking the example of an Ideal low-pass filter. The ideal low-pass filter passes only the low-frequency components in the input signal and it rejects all the frequency components beyond some cutoff frequency.

The frequency band which is being supported by this filter is known as the **Pass Band** and the frequencies which are being rejected by this filter are known as the **Stop Band**.

And in this Pass Band, the frequency response is very flat. While above this cutoff frequency, it rejects all the frequencies and it provides infinite attenuation for those frequencies. So there is a vertical transition between this Pass Band and the Stop Band. And such a kind of response is known as the brick wall response.

But whenever we are designing the practical filters, it is almost impossible to achieve this vertical transition from the Pass Band to the Stop Band. Hence the transition from the Pass Band to the Stop Band will be always gradual and we use different design methods.

Butterworth: It exhibits a nearly flat Pass Band with no ripple. The roll-off is smooth and monotonic, with a low-pass or high-pass roll-off rate of 20 dB/decade (6 dB/octave) for every pole. This approximation is also known as the maximal flat approximation.

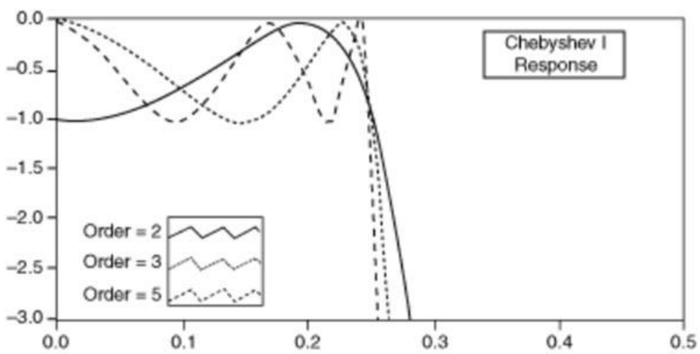


Suppose you are designing the Butterworth filter of a second **order** then it will have a roll-off of 40dB per decade. Similarly, designing a fifth **order** will have a roll-off of at the rate of 100dB per decade. Also in this design, you will not see any kind of ripple in the stop band as well.

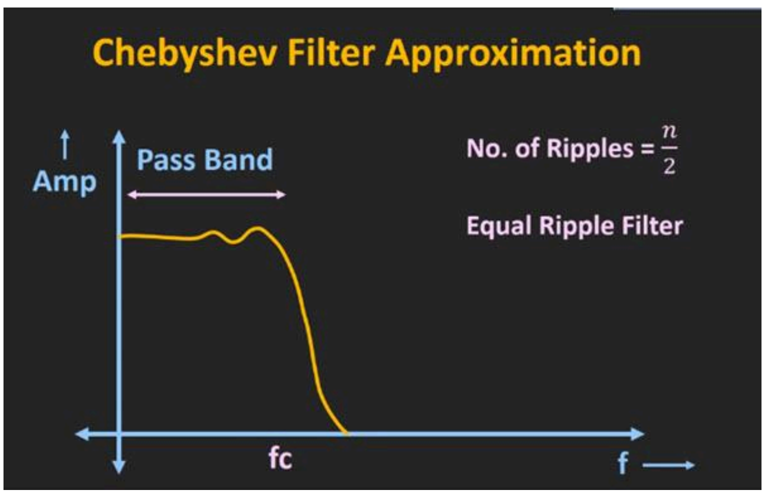
This filter design is quite common, particularly in audio processing.

Chebyshev: This design is a mathematical strategy for achieving a faster roll-off by allowing ripple in the Pass Band. As the ripple increases, the roll-off becomes sharper. In Stop Band you will not see any ripples. The Chebyshev response is an optimal trade-off between these two parameters.

If maintaining a flat response throughout your Pass Band is not a concern, and if you want a rapid transition, this design filter can be used. (Compared to a Butterworth filter, a Chebyshev filter can achieve a sharper transition between the Pass Band and the Stop Band with a lower-order filter. The sharp transition between the Pass Band and the Stop Band of a Chebyshev filter produces smaller absolute errors and faster execution speeds than a Butterworth filter).



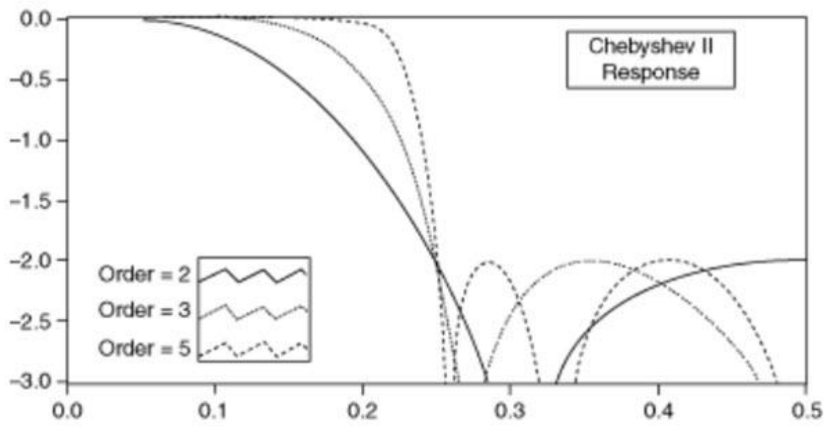
In the case of this Chebyshev filter design, the number of ripples in the Pass Band will be equal to **n** divided by 2, where **n** is the **order** of the filter. For example, the fourth order will have 2 ripples.



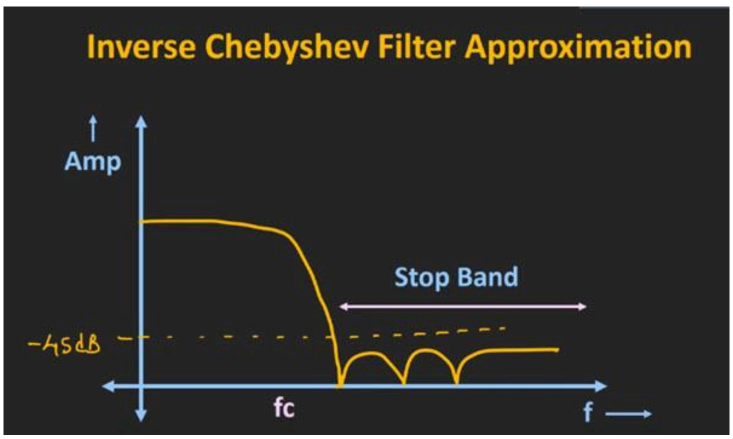
Also, we need to provide the maximum amount of **ripple(Passband ripple)** which is acceptable by this filter design. In most cases, this **ripple(Passband ripple)** is in the range of 0.1dB up to 3dB. The amplitude of these ripples is the same for all the ripples. Hence it is also known as the equal ripple filter.

Inverse Chebyshev: This design filter is having a flat Pass Band and it is having ripples in the Stop Band. And the transition or roll-off in this inverse Chebyshev filter is the same as the Chebyshev filter.

So whenever in your application you require the flat frequency response in the Pass Band, as well as you required the faster transition and you are not bothered about any kind of ripple at the Stop Band then these case is used.

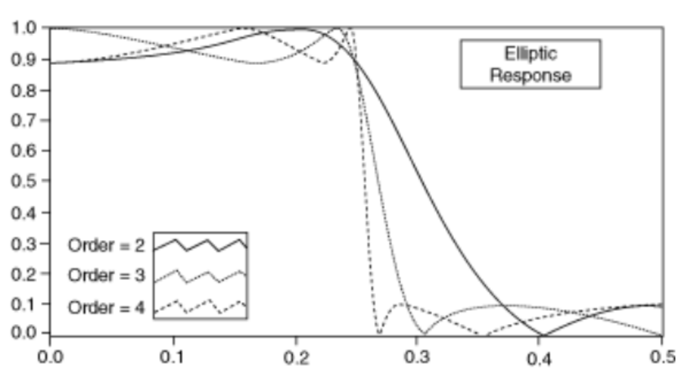


But whenever you have a **ripple** in the Stop Band then you also need to provide the amount of attenuation**(Stopband attenuation)** because it is quite possible that the amplitude of the ripple might touch that value of a Stop Band rejection.

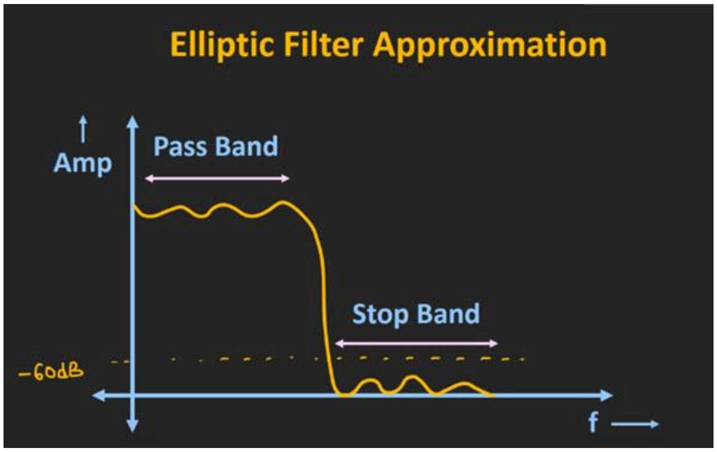


In this case, the filter is providing the **attenuation** in the stop band at minus 45 dB. So we need to make sure that the amplitude of the ripple is always less than this value.

Elliptic: The cut-off slope of an elliptic filter is steeper than that of a Butterworth, Chebyshev, or Bessel, but the amplitude response has a ripple in both the Pass Band and the Stop Band, and the phase response is very non-linear.

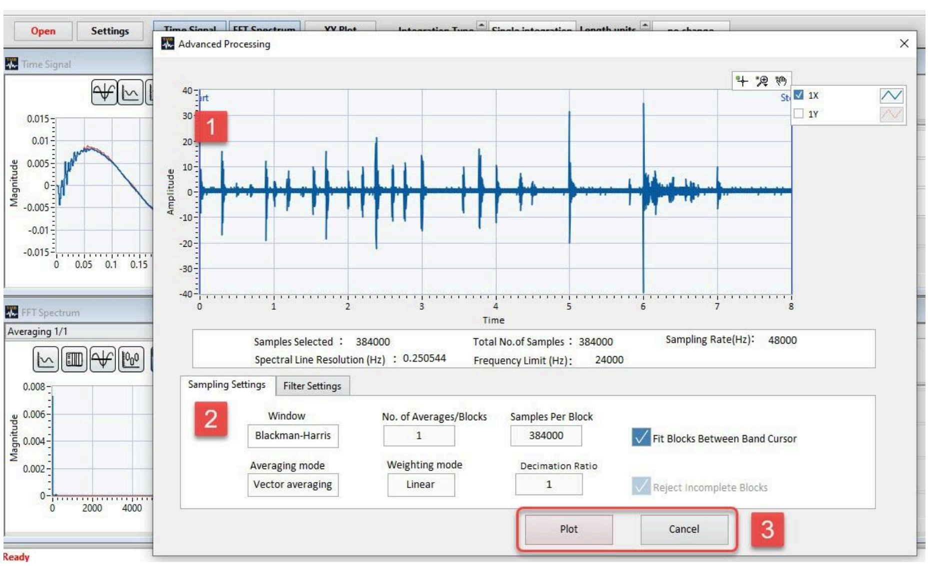


However, if the primary concern is to pass frequencies falling within a certain frequency band and reject frequencies outside that band, regardless of phase shifts or ringing, the elliptic response will perform that function with the lowest-order filter.



Whenever in application the ripple in the Pass Band and Stop Band is not a concern and you require the fastest possible transition then in such cases you can use this design.

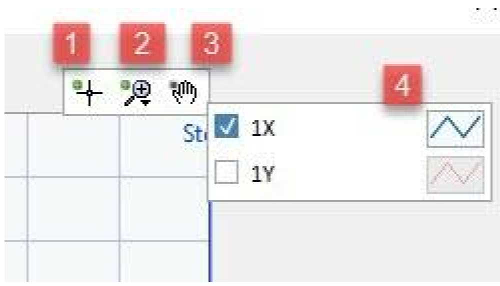
Settings Post Processor Mode



In Post processor Mode the Run/Stop Button changes to Open Button to choose the save .tdms files for further processing.

1

The data is plotted from the loaded .tdms file. The Start & Stop cursors can be used to select the data required for processing and plotting. Likewise, you can enable or disable current channels on the right side of the graph.



The little green light in the top left corner indicates whether or not the item has been activated.

1. This locks the graph in place, and the user would find the cursor locked.

2. Different zoom options like selected region, vertical region, horizontal region, fit to graph, pan zoom in, and pan zoom out are available. (NB: This functionality is accessible on all graphs).

3. Holding and move option.

4. Channel selection for display.

2

This part consists of Sampling and Filter Settings. Users can choose to enable or disable both functions.

Details of the above can be found in Sampling and Filtering settings.

3

The **Plot** option applies the changes made with the settings, while **Cancel** discards them.

Notification Bar

The notification bar lets you know the current status of the software.

1. **Ready** indicates the software is ready for acquisition..

2. **Initializing** indicates the measurement has started and is processing for acquiring the measurement.

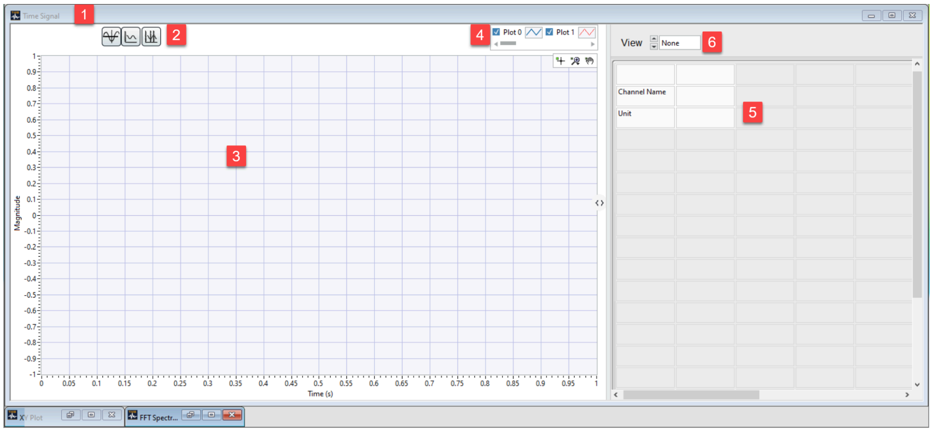
3. **Settling** indicates that the acquisition system is waiting for the sensor's electronics to settle. All sensors require a settling time before the measurement starts. The settling time can be input in the settings.

4. **Output Ready** indicates that the acquisition is ongoing and the real-time measurement is being displayed in the tiles.

5. **Stopped** indicates that the acquisition has been stopped.

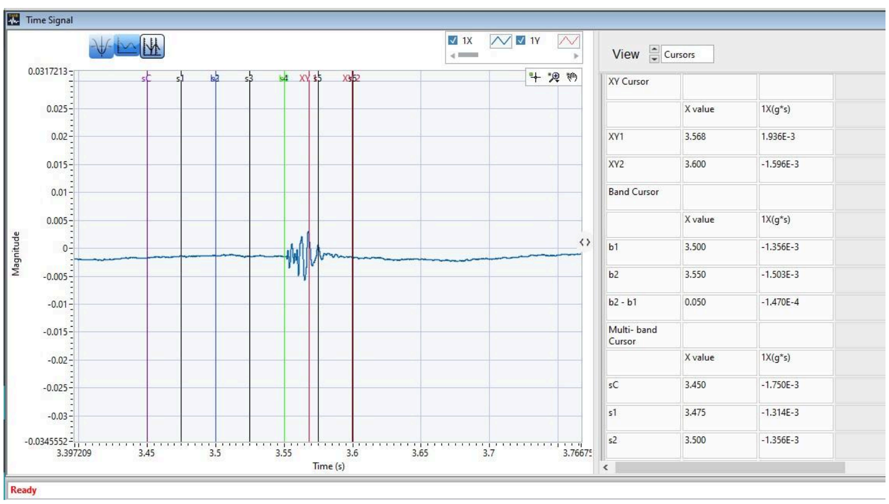
Time Signal

The Time signal tile displays the measurements in real-time.

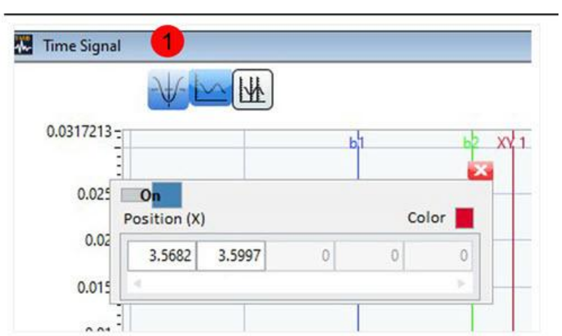


1. Tile Name: Time Signal

2. Cursor Options: The cursor options are the XY cursor, the band cursor, and the multi-band cursor.



The settings of the above example are as follows:



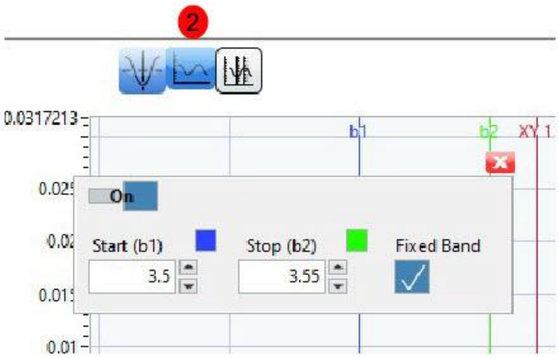
(1) XY cursor

The **slide button** allows the user to switch ON/OFF the cursors**.**

The **Position (X)** allows the user to input the position of the cursor which places the cursor to the input position. For example: In the above figure 3.5682 is the position of the first cursor & 3.5997 is the position of the second cursor on the time graph. For more cursors, the user can input the corresponding position values.

To change the color of the cursor click the **Color** button and choose the appropriate color.

Once configured the user can press the **close** button.



(2) Band Cursor

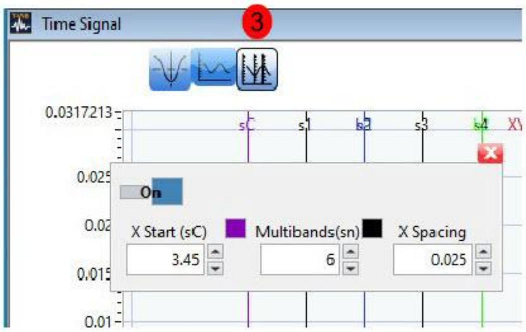
The **slide button** allows the user to switch ON/OFF the cursors**.** The functionality of the band cursor is to determine the period/ width of a band, for example, this feature allows the user to check the period of an oscillating signal (the period between the cursor is displayed in the **Info tab**).

The **Start(b1)** and **Stop(b2)** are the two cursors available in this feature. The user can input the position of the cursors.

The **Fixed band** allows the user to fix the difference between the band, on moving any of the cursors the period or the width between the cursors would be preserved.

To change the color of the cursor click the **Color** button and choose the appropriate color.

Once configured the user can press the **close** button.



(3) Multi-band cursor

The **slide button** allows the user to switch ON/OFF the cursors**.** The functionality of the multi-band cursor is to determine a recurring event, for example, this feature allows the user to check the periodicity of bearing impacts.

Input the following details to use this cursor. **X Start (sC)** the first cursor position, **Multibands(sn)** the number of cursors & **X Spacing** the width between consecutive cursors.

To change the color of the cursor click the **Color** button and choose the appropriate color.

Once configured the user can press the **close** button.

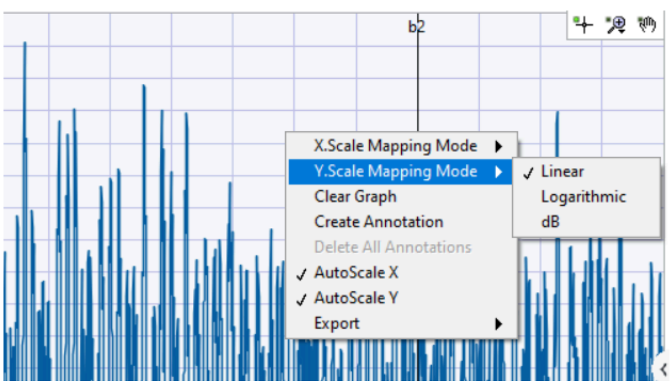
3. Graph: The data acquired is displayed on the **Graph,** with the magnitude of the measurement on the Y-axis and time on the X-axis. We can use the standard tools on the top right toolbar such as zoom out/in, pan, etc.

Additional features are available on the graphs. To access the tools, right-click on the graph and select the required option from the list.

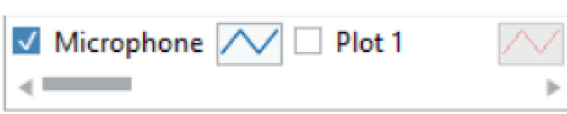
The user can change the scales of the X axes & Y axes between logarithmic/ linear/dB. To do so right click on the graph and select the **X Scale/ Y Scale Mapping Modes**. Choose **logarithmic/ linear/dB** as per user requirements.

The graph also allows the user to create annotations to the plots, export the graph as a picture or data points to Excel, enable/ disable auto scale X/Y, etc. To do so right click on the graph, and select the required option.

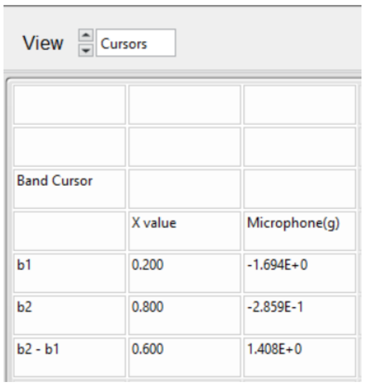
Enabling the auto scale will allow the user to view the graph only at the maximum X/Y axes range whereas disabling would allow the user to select the ranges in the graph to be viewed.



4. Plots details: This option allows the user to choose the channels on which the plots are to be viewed. For example in the below picture, only the Microphone channel has been selected to display the data on the graph.

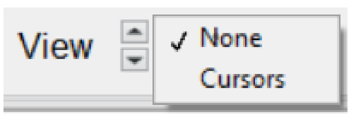


5.Info tab: Displays the details of the plots such as channel name, units & cursors values etc.



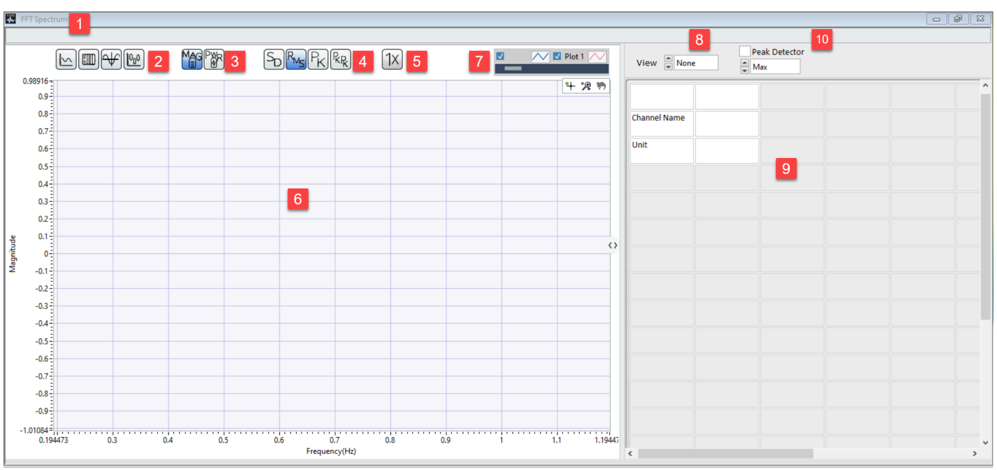
Band Cursor		
	X value	Microphone(g)
b1	0.200	-1.694E+0
b2	0.800	-2.859E-1
b2 - b1	0.600	1.408E+0

6. View Selection: The view selection allows the user to select what details are to be displayed on the Info Tab. Selecting the None option would allow the user to view the channel name & unit of the value display. To view the cursor details on the info tab the user would require to select the cursor in the view selection.



FFT Spectrum

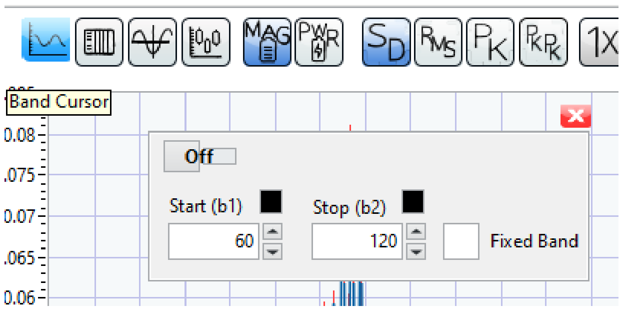
The Fast Fourier Transform Spectrum tile computes the frequency content of the time domain Measurements.



1. Tile Name: FFT Spectrum

2. Cursor Options: The cursor options available are Band Cursor, Harmonic Cursor, XY Cursor, and Side Band Cursor.

Band Cursor:



The **slide button** allows the user to switch ON/OFF the cursors**.** The function of the band cursor is to determine the period/ width of a band, for example, this feature allows the user to check the period of an oscillating signal (the period between the cursor is displayed in the **Info tab**).

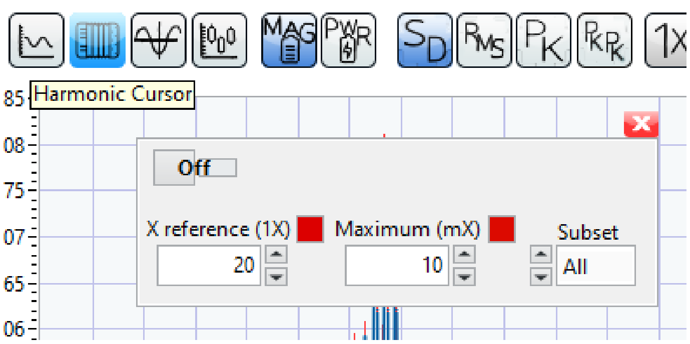
The **Start(b1)** and **Stop(b2)** are the two cursors available in this feature. The user can input the position of the cursors.

The **Fixed band** allows the user to fix the difference between the band, on moving any of the cursors the period or the width between the cursors would be preserved.

To change the color of the cursor click the **Color** button and choose the appropriate color.

Once configured the user can press the **close** button.

Harmonic Cursor:



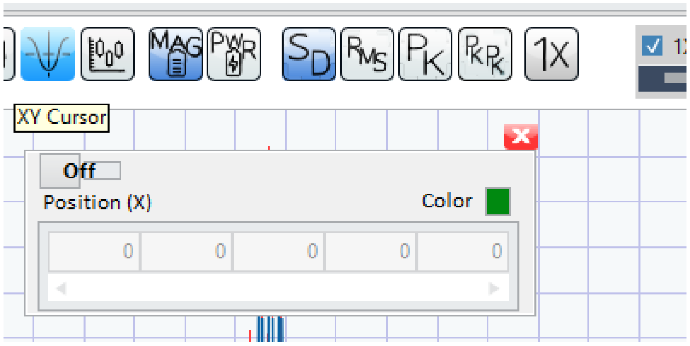
The **slide button** allows the user to switch ON/OFF the cursors**.** The function of the Harmonic cursor is to find the harmonic peaks in the spectrum, for example, this feature allows the fault frequencies like bearing ball pass frequencies outer(the magnitudes of the harmonics are displayed in the **Info tab**).

The **X Reference (1X)** is the position of the fundamental frequency where the cursor is to be placed.**Maximum (mX)** is the number of cursors required to view the harmonics of the fundamental frequency. **Subset** is a feature of the harmonic cursor that allows the user to specifically view even/odd/all harmonics.

To change the color of the cursor click the **Color** button and choose the appropriate color.

Once configured the user can press the **close** button.

XY Cursor:



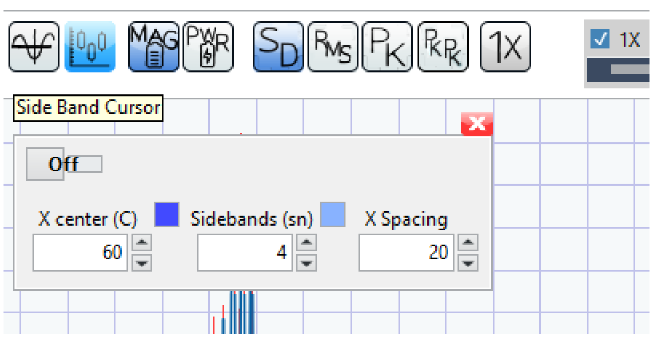
The **slide button** allows the user to switch ON/OFF the cursors**.**

The **Position (X)** allows the user to input the position of the cursor which places the cursor to the input position. For enabling cursors the user has to input the corresponding position values to each column.

To change the color of the cursor click the **Color** button and choose the appropriate color.

Once configured the user can press the **close** button

Side Band Cursor:



The **slide button** allows the user to switch ON/OFF the cursors**.** The functionality of the sideband cursor is to determine sidebands around a center frequency, for example, this feature allows the user to check the modulations at the Gear Mesh Frequency.

Input the following details to use this cursor. **X center (sC)** is the center cursor position, **Sidebands (sn)** is the number of cursors & **X Spacing** is the width between consecutive cursors.

To change the color of the cursor click the **Color** button and choose the appropriate color.

Once configured the user can press the **close** button.

3. Magnitude/ Power: To view the power spectrum choose power else Magnitude, this allows the user to view the magnitude of the peaks.

4. Amplitude Selection: This option allows the user to view the amplitude in the following values

A. SD/Spectral Density: This button allows the user to view the Power Spectral density (along with the **Power** selected). This is useful for random signals. **Power spectral density** is a scaled version of the **Power spectrum**, where the power present within each spectral bin is normalized by the frequency bin width. You typically use this measurement to examine the noise floor of a signal or the power in a specific frequency range. Normalizing the power spectrum by the bin width makes this measurement independent of the signal duration, or number of samples.

B. **RMS/Root Mean Square**: The root mean square is defined as the square root of the mean square (the arithmetic mean of the squares of a set of numbers). The RMS (root mean square) value is generally the most useful because it is directly related to the energy content of the vibration profile.

C. **Peak:** The peak is the maximum amplitude in time duration. It is used to measure shocks or waves**.**

D. **Peak to Peak:** Peak to Peak is the difference between the maximum positive and the maximum negative amplitudes of a waveform.

5. 1X Peak: This option works with the harmonic cursor and allows the user to set or place the cursor over the fundamental frequency without manually placing it(that is the running speed).

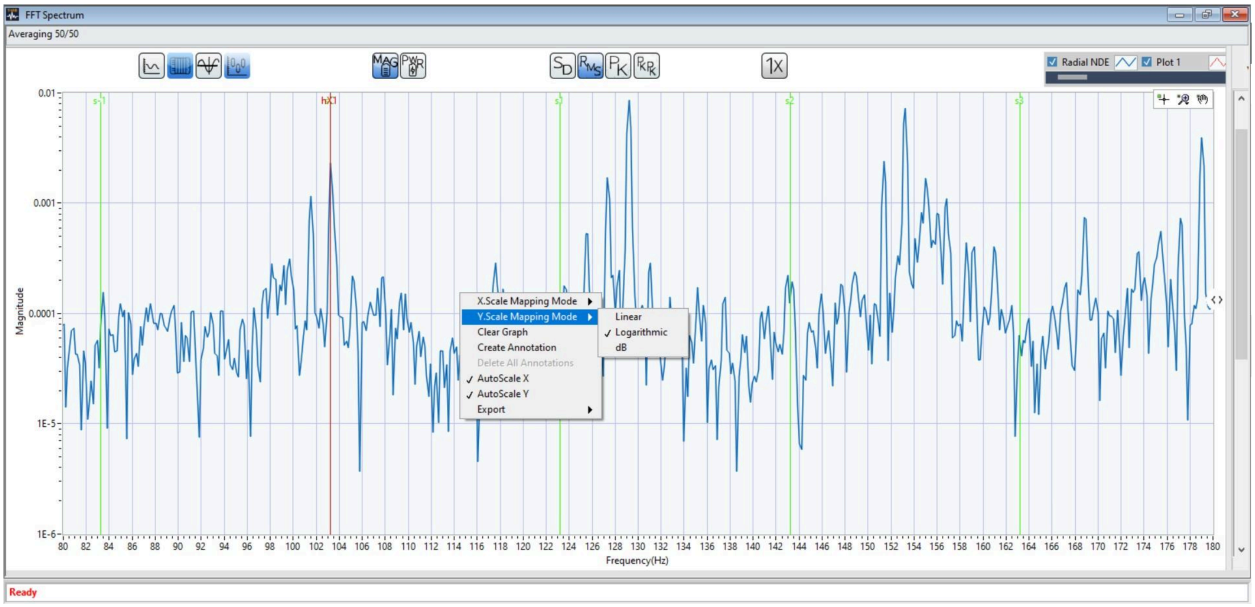
For example: To measure and denote the harmonics of the running speed the harmonic cursor is activated and placed in the position near the running speed, then the 1X button is pressed. This aligns the 1X cursor to place over the fundamental frequency and allows the following cursors to be aligned to it's harmonics**.**

6. Graph: The plots are shown here. Here using configurations **3** & 4 we can change the Y-axis value to either magnitude/power, RMS/Peak/Peak to Peak. The X-axis is the frequency in Hertz.

Additionally, in the graphical area, extra features are available by right-clicking the graph. The user has the option to set the scale to linear/logarithmic/dB Scale by setting the **X Scale & Y Scale Mapping Modes**.

Set the scale range automatically using the **Auto-scale** option.

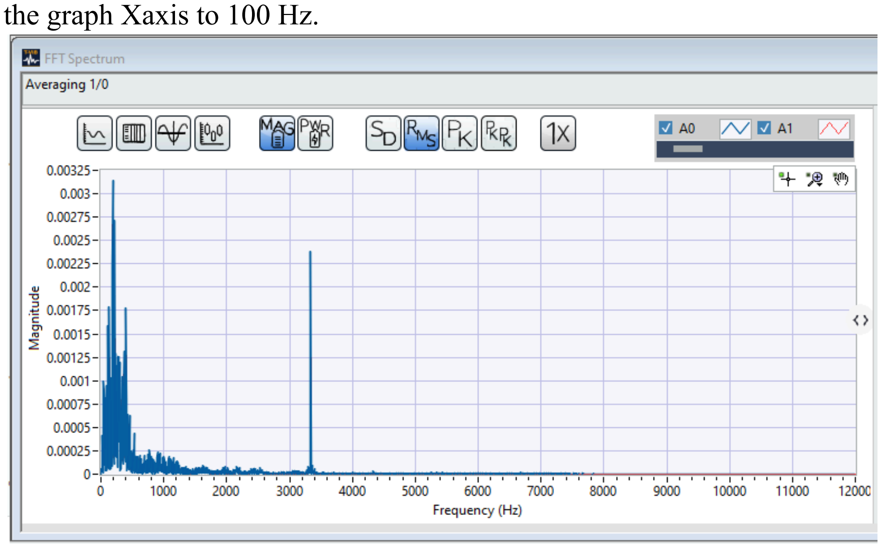
Also, the user can **Export** the data in Excel or picture formats, create annotations, etc.



Setting the Graph Scale - Autoscale

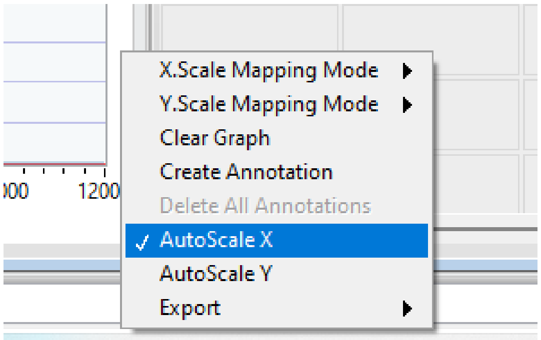
Autoscale is the option for automatically displaying the graph scale based on the data available, enabling this function disables the user's control over the graph.
To manually set the range of the graph switch off the Auto scale function and double click the max value and enter the required value.

For example, my spectrum is showing data up to 12,000 Hz in the Xaxis and I want to set the scale of the graph Xaxis to 100 Hz.



Follow the below steps

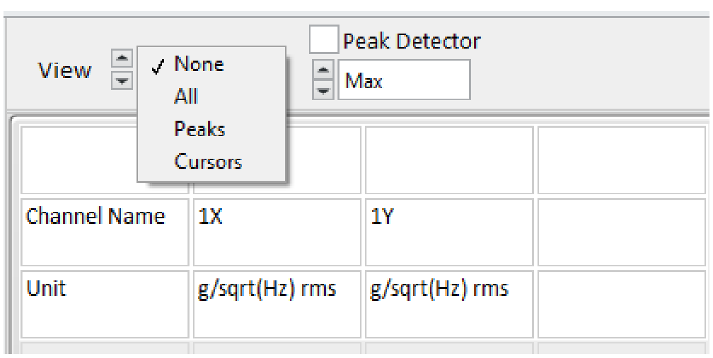
1. Switch off the Autoscale X.



2. Enter the maximum value on the graph as 100 Hz.

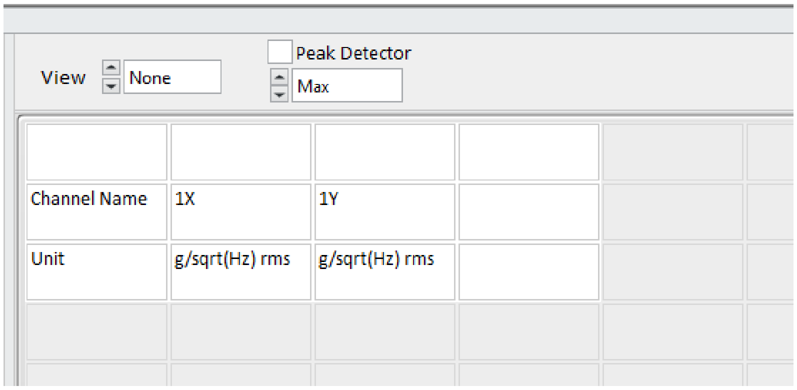
7. Plot Selection: Enables/disables channels based on user selection.

8. View Selection: Using the selection we can change the values displayed on the table.

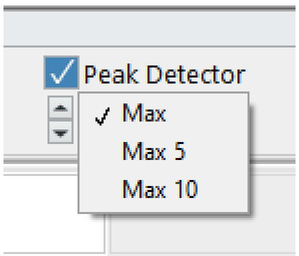


9. Info Tab: The table shows values according to view selection.

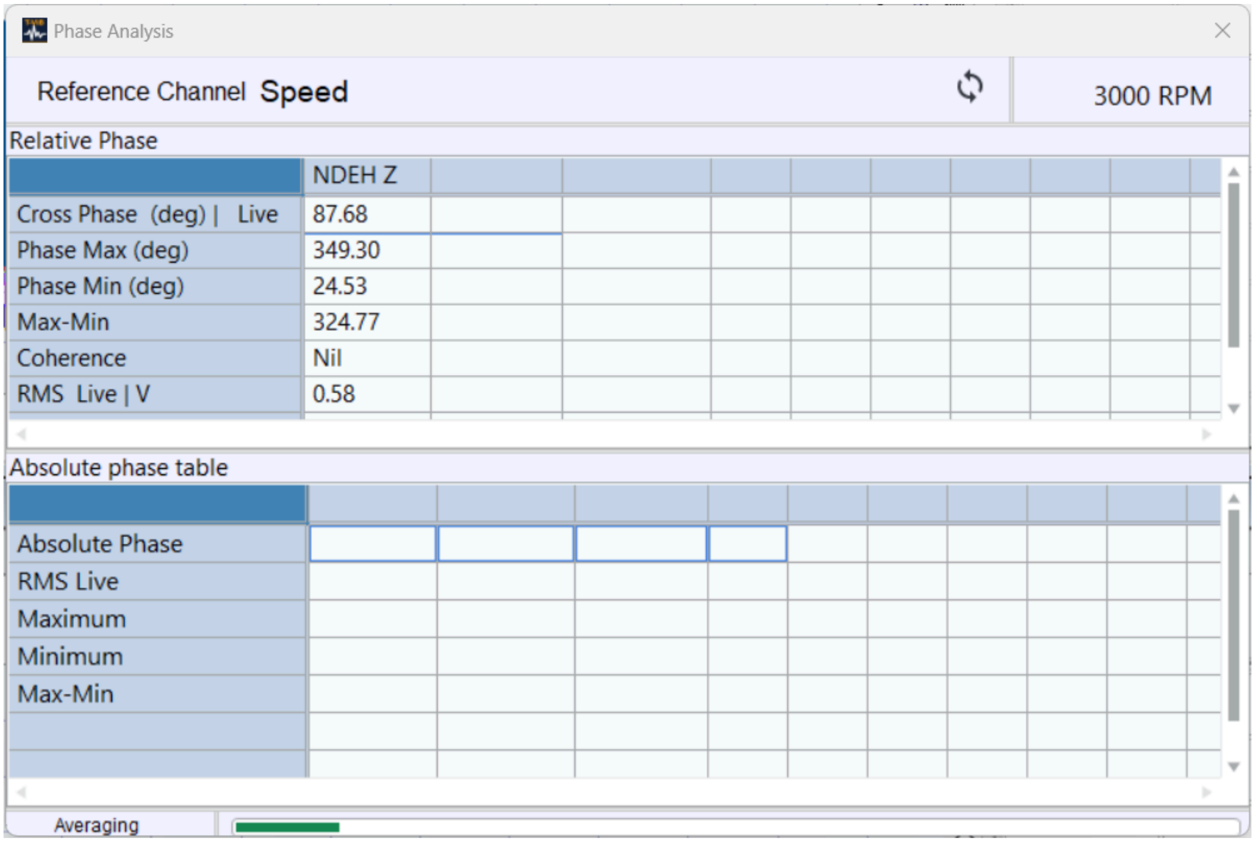
When none is selected it shows the channel name and measurement unit.The **peak detector** shows the peak values if it is enabled.



10. Peak Detector: When enabled, the peak value is displayed. Additionally, a maximum of 5 or 10 peak values are displayed.



Phase Analysis



The phase analysis window allows the user to view the phase measurement, they can measure the Relative phase and the absolute phase (only when a tachometer is connected).

1. Relative Phase: Relative Phase refers to the phase difference between two signals, typically measured between the vibration of two points on a machine. This measurement helps identify misalignment, looseness, or other mechanical issues by showing how different parts of a system move in relation to each other.Relative phase, which is the phase difference between two signals, is usually measured between the vibration at two points on a machine or between a vibration signal and a reference signal.
2. Absolute Phase :Absolute Phase represents the phase angle of a vibration signal relative to a fixed reference, like time or a machine's rotational position. While absolute phase provides insights into a machine's overall dynamics, relative phase is particularly valuable for diagnosing specific relationships between components.Absolute phase is the phase angle of a vibration signal measured against a fixed reference, such as time or a machine's rotational position. It offers insights into the overall dynamics of a machine. Relative phase, on the other hand, indicates the phase angle between two vibration signals and is particularly useful for diagnosing the relationships between specific machine components.

Relative Phase Measurements

Below are the brief description of the parameters displayed in the relative phase measurements

1. **Cross Phase:** Cross channel phase is a process of using a two-channel instrument to determine the phase relationship between two measurement locations without using a fixed reference tachometer. The cross channel phase uses two-time waveform signals: Using channel “A” as a reference it takes one cycle in the time waveform as one revolution of the shaft and plots the “B” channel time waveform against channel “A” to determine the relative phase.Cross channel phase is a method that determines the phase relationship between two measurement locations using a two-channel instrument. This technique does not require a fixed reference tachometer. The cross channel phase utilizes two-time waveform signals. By taking one cycle in the time waveform of channel "A" as a reference, which represents one revolution of the shaft, it plots the "B" channel time waveform against channel "A". This process allows for the determination of the relative phase.
2. **Phase Max:** This parameter displays the maximum phase value measured.
3. **Phase Min:** This parameter displays the minimum phase value measured.
4. **Max-Min:** This parameter displays the difference between the maximum and minimum phase.
5. **Coherence:** This parameter is a measure of how well two signals maintain a stable phase relationship across multiple measurements.

   Coherence = 1 → Signals are perfectly correlated (fully phase-coherent).

   Coherence = 0 → No correlation (random or unrelated signals).

   Intermediate Values → Partial correlation, affected by noise, nonlinearities, or external influences.

6. **RMS Live:** This parameter shows the live RMS Value.

Absolute Phase Measurements

Below are the brief description of the parameters displayed in the absolute phase measurements

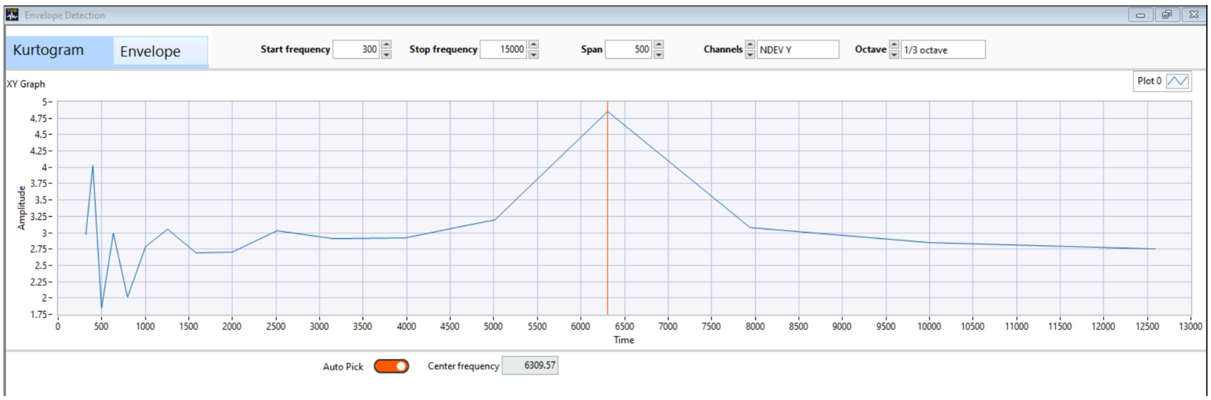
1. **Absolute Phase:** Absolute Phase represents the phase angle of a vibration signal relative to a fixed reference, like time or a machine’s rotational position. While absolute phase provides insights into a machine’s overall dynamics, relative phase is particularly valuable for diagnosing specific relationships between components.
2. **RMS Live:** This parameter shows the live RMS Value.
3. **Maximum:** This parameter displays the Maximum value of the Absolute Phase
4. **Minimum:** This parameter displays the minimum value of the Absolute Phase
5. **Max-Min:** This parameter displays the difference between the Maximum and Minimum.

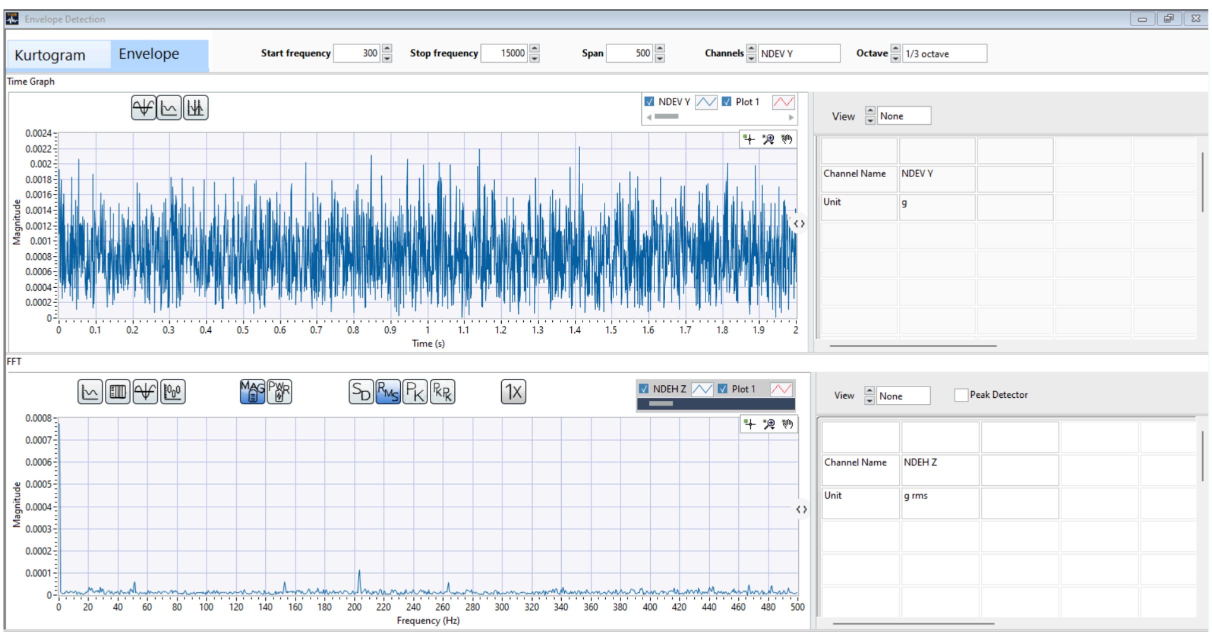
Envelope Detection tile

The envelope Detection is the technique to extract the modulating or envelope signal from an amplitude modulated signal. This software feature has an algorithm for setting the centre frequency for extracting the envelope signal. The kurtogram tab allows the user to view the kurtosis over frequency range. The user has the option to set the centre frequency by manual picking by switching off the Auto pick option. The envelope tab allows the user to view the time and spectrum of the envelope signal.

There are options that the user can configure

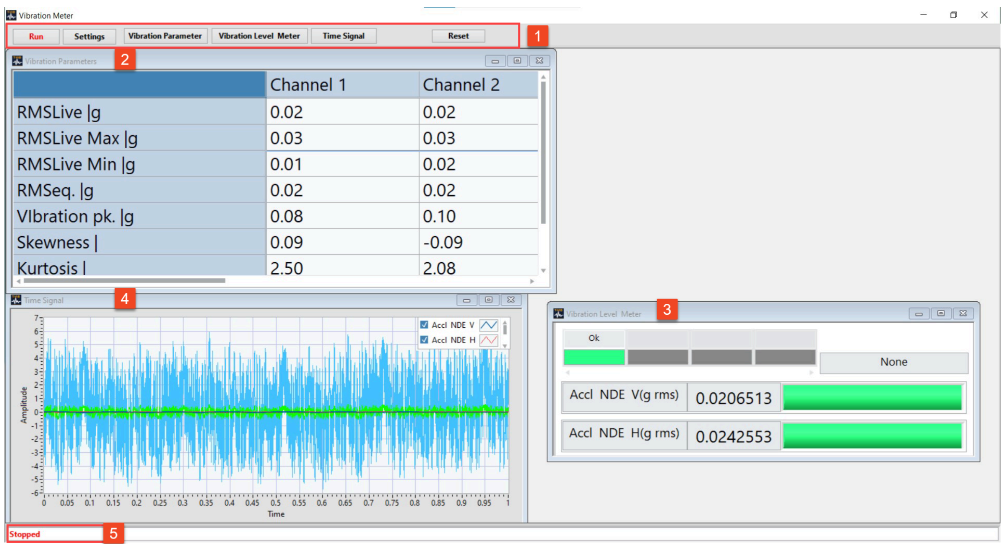
- Start frequency: The starting frequency for computing the kurtogram
- Stop frequency: The stop frequency for the computing the kurtogram
- Span: The span of the frequency spectrum
- Channel : The user can select the channel to perform the enveloping.
- Octave: The user can select the band of the octave 1/, 1/6, 1/12, 1/24





Advanced Vibration Meter (TVM 203)

Module Interface



1. Function bar

2. Vibration Parameter *tile*

3. Vibration Level Meter *tile*

4. Time Signal *tile*

5. Notification bar

In the following subsections we will discuss in detail the various features & functionality of the above components.

Function Bar

The function bar is composed of buttons; it performs a variety of functions as detailed below.

1. Run button

This button can be used to start & stop the acquisition.

2. Settings

The *Settings* can set up an acquisition & various functionality of the test. We will discuss the various options in the settings in the next section.

3. Vibration Parameter

On pressing the *Vibration Parameter* button, a tile pops up on the screen. The tile comprises of parameters such as RMS Live, Max, Min, RMS eq. , Vibration peak, Skewness, Kurtosis etc.

4. Vibration Level Meter

On pressing the *Vibration Level Meter* button, a tile pops up on the screen. The tile shows the vibration level and indicates the status based on ISO 10816-3 or custom entry, as per the user selection. This helps user to identify the present health status of the machine.

5. Time Signal

On pressing the *Time signal* button the Time signal tile pops up on the screen. The tile comprises of graph with several options for setting the amplitude value *(such as RMS, peak etc.)* and cursors, the tabular section allows the users to take note of the precise values of the peaks, cursor values & units.

6. Reset Button

This button resets the measurements.

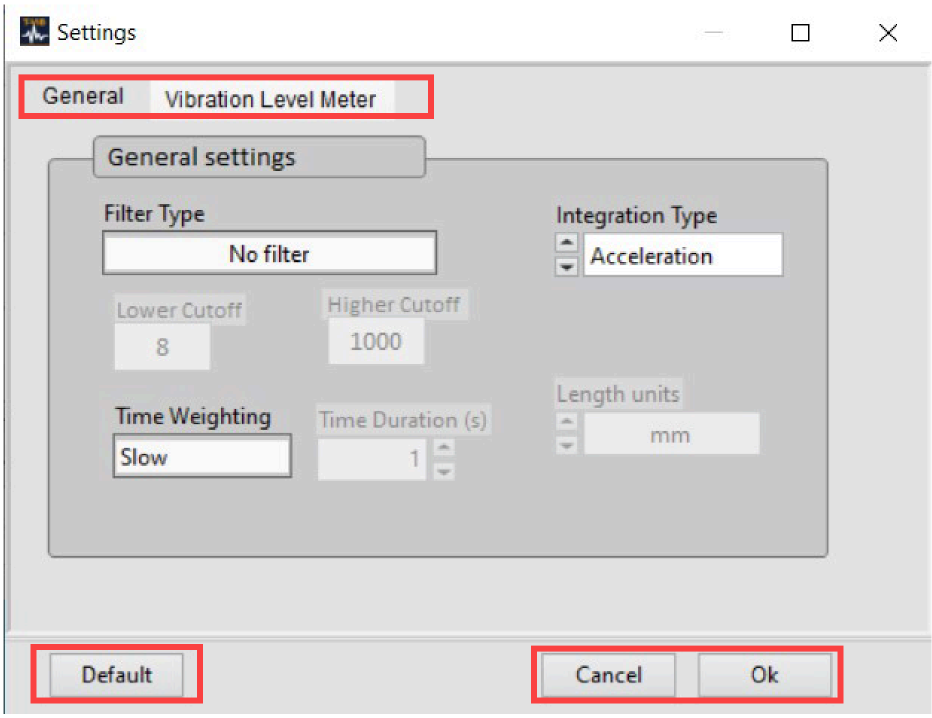
Settings

Below we will discuss the settings in detail.

The Settings tab consists of two panes.

1. **General Settings:** This allows the user to select the general acquisition settings such as the Filter type, time weighting , parameter to be displayed,etc.

2. Vibration Level Meter Settings : This enables the user to adjust the settings of the vibration level indicator such as setting the ISO based vibration indications of Machine Health Status, Machine Class to custom vibration alarm setting.

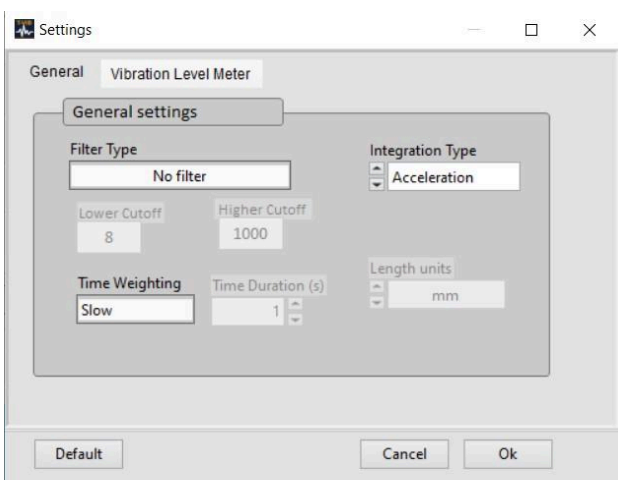


Default : This will load the settings' default values in every tab.

Cancel : Closes the settings tab, also the changes made by the user on the settings will be lost and the previous settings would be loaded. Pressing the Close button on the settings window would have the same effect.

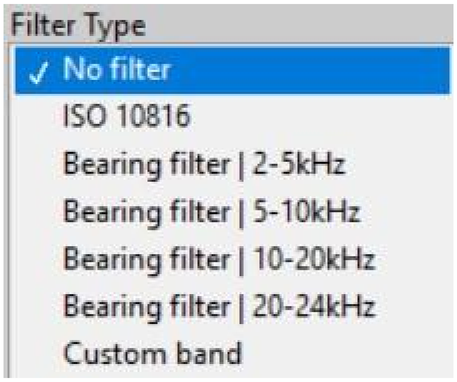
Ok : Save the selected configuration as the present settings .

General Settings tab



The options in the General Settings are discussed below

1. Filter Type: There are 7 filter settings



No filter: This option does not have any filter, the inbuilt filters like anti-aliasing filters would decide the frequency range of the signal.

ISO 10816: The ISO 10816 standard specifies that the measured signal should be within the frequency range of 10-1000 Hz.

Bearing filter : Bearing vibration signal has a high frequency range. There are four preset bearing filters i.e

a) 2-5 kHz b) 5-10 kHz c) 10-20 kHz d) 20-24 kHz

Custom Band : This option allows the user to select the frequency range. Once this option is selected the user can input the lower and higher cut off frequencies. The signal within the range would be acquired and reflected in the measurements.

2. Integration Type: This option allows the user to select between acceleration, velocity and displacement. The integration type depends on the filter type chosen. The following table shows the Integration type options corresponding to the Filter types.

Filter Type	**Integration Type**
ISO 10816	Velocity
Bearing Filters	Acceleration
Custom Band/ No Filter	Acceleration/ Velocity/ Displacement

3. Length Units : The user can select the following units based on the integration type selected. The table shows the integration type and the unit options available**.**

Integration Type	**Unit**
Acceleration	No change
Velocity	m,mm,microns,inch, mils
Displacement	m,mm,microns,inch, mils

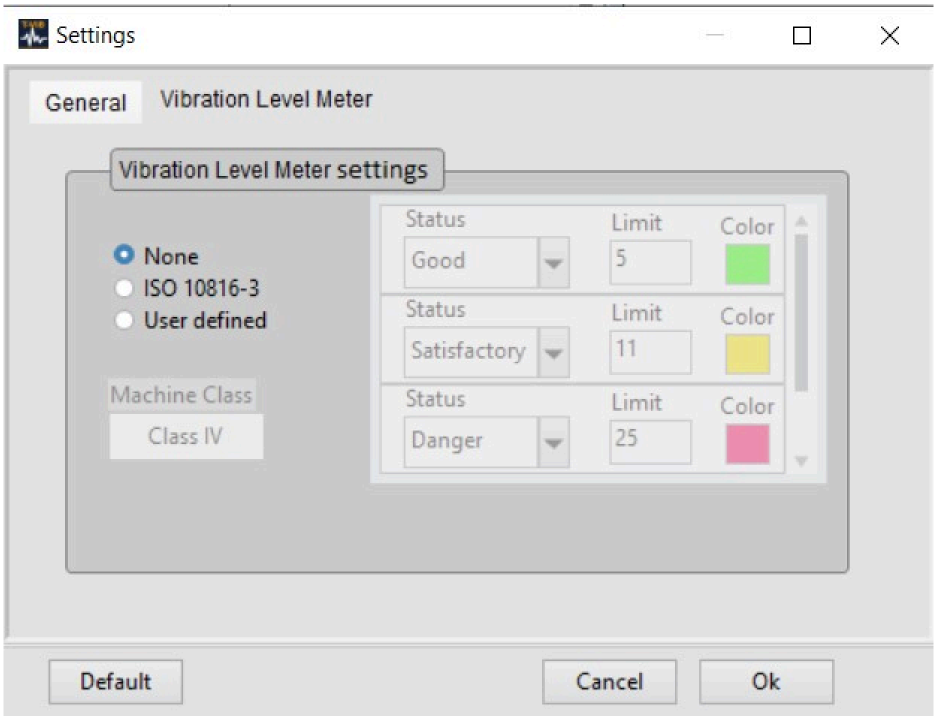
No change option specifies that the unit in the channel settings would be used for the measurements.

4. Time Weightings : There are two time weightings available as per the options

Slow weighting: 1 second time weighting

Custom: As per the user choice *(limit would be the buffer of the DAQ)*

Vibration Level Meter Settings tab



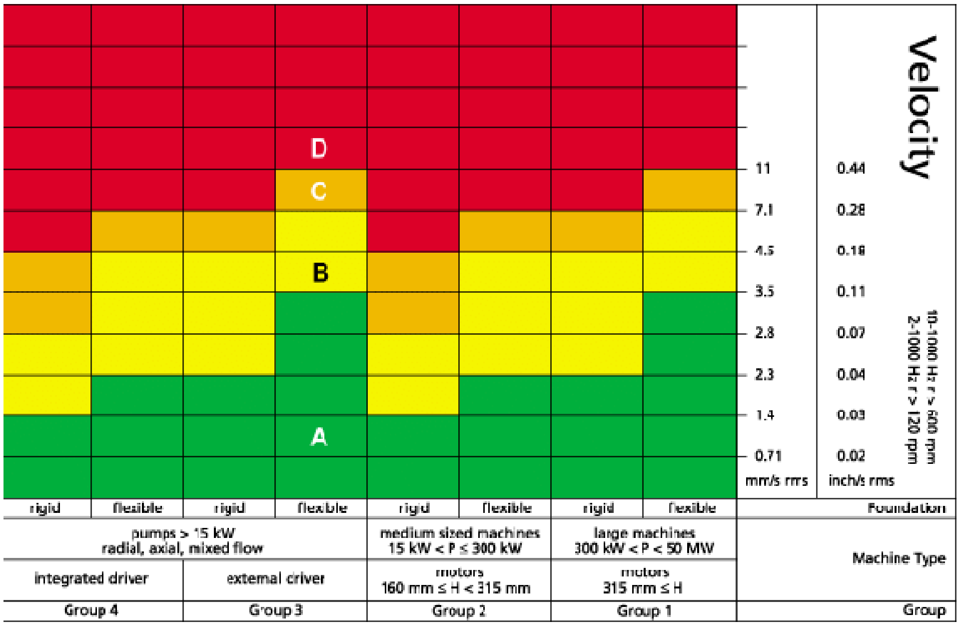
Vibration Limits: The user can select the following options

None: if the user does not want to see the Vibration Level Meter.

ISO 10816-3: if the status, limits & Color are as per that of the ISO standards.

User defined: if the user wants to set the vibration meter indicator as per the vibration levels observed in the trends of their machine.

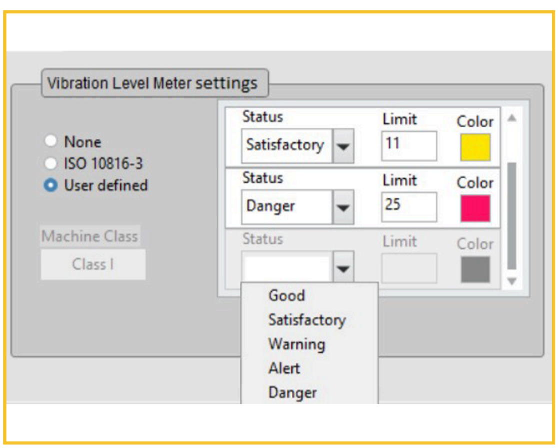
Machine Class: The user can select the Machine Class when the Vibration Limits are configured for ISO 10816-3 chart below.

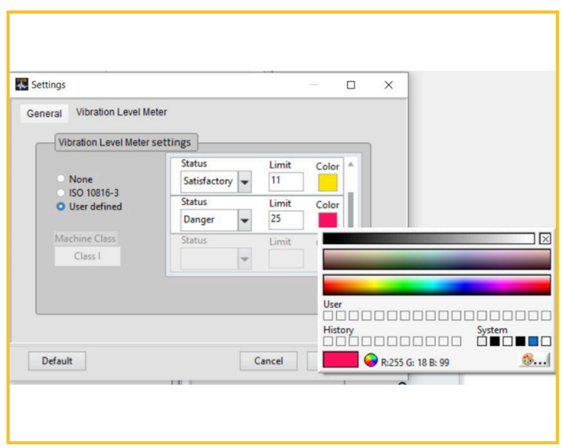


Vibration Level Meter Indicator:

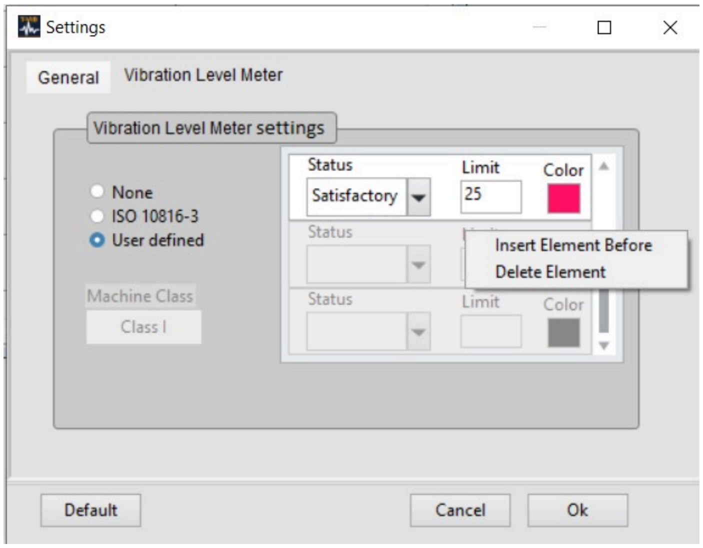
The User defined option allows the user to set the Status, Limit & the Color as per the user experience and historical data from the machine.

To enter a level of the indicator , choose the status from the list, then enter the value of the Limit below which the vibration level meter is to indicate the selected status.





If you want more levels, then right click and choose Insert Element Before. Then enter the details of the level as to be displayed in the Vibration Level Meter.


Settings
General
Vibration Level Meter
Vibration Level Meter settings
None
ISO 10816-3
User defined
Machine Class
Class I
Status
Limit
Color
Satisfactory
25
Status
Insert Element Before
Delete Element
Status
Limit
Color
Default
Cancel
Ok

Vibration Parameter

	Channel 1	
RMSLive |mm/s	3.34	
RMSLive Max |mm/s	10.60	
RMSLive Min |mm/s	0.83	
RMSeq. |mm/s	4.76	
VIbration pk. |mm/s	45.86	
Skewness |	-0.07	
Kurtosis |	4.48	
Crest factor |	9.64	

The Vibration Parameters *tile* displays the parameters on tabular form.

Root Mean Square: The RMS (Root Mean Square) value is a measure of the energy content in a vibration signal. It is one of the most accurate statistical parameters for assessing the severity of machine failures. While it's excellent for monitoring overall vibration levels, it doesn't pinpoint the specific faulty component. However, the RMS value is highly effective in detecting significant imbalances in rotating equipment systems and tends to increase with the occurrence of shock pulses.

The RMS value is calculated by considering the time history of the vibration wave. It provides valuable insights into the vibration's energy content, but it won't identify which part of the machine is causing the issue. The following equations are used to calculate RMS values for both discrete and continuous time signals.

$$\mathrm{RMS} = \sqrt{\frac{1}{N}\sum_{i=1}^{N} x_i^2}$$

$$\mathrm{RMS} = \sqrt{\frac{1}{T}\int_0^T x^2(t)\,dt}$$

RMS Live: This is the real time Root Mean Square Value of the present block of vibration signal acquired.

RMS Live Max: This is the maximum RMS value calculated in the Vibration Signal.

RMS Live Min: This is the minimum RMS value calculated in the Vibration Signal.

RMS eq : This is the average of the RMS value calculated for each block of Vibration signal acquired.

Vibration pk: This is the peak value of the vibration signal measured.

Skewness: Skewness is a parameter that assesses the asymmetrical behavior of a vibration signal by analyzing its probability density function. It measures how unsymmetrical the signal is around its mean value. If a signal is perfectly symmetrical, its skewness value is zero. In the context of most vibration signals, the probability distribution is typically symmetrical, resembling a normal distribution. Therefore, a non-zero skewness value often signals that something might be amiss.

Skewness serves as a dimensionless measure to indicate the absence of symmetry in the distribution of the signal. It's a valuable tool for identifying deviations from the expected patterns in vibration data.

$$\text{Skewness} = \frac{\frac{1}{N}\sum_{i=1}^{N}(x_i-\bar{x})^3}{[\frac{1}{N}\sum_{i=1}^{N}(x_i-\bar{x})^2]^{3/2}}$$

Kurtosis: The kurtosis factor is a statistical measure that helps us understand the pulse-like characteristics of a signal. A high kurtosis factor suggests the presence of repetitive impulses within the signal. It aids in determining whether the spectrum displays small peaks spread across a wide frequency range or a few distinct peaks at specific frequencies. Kurtosis is especially useful when monitoring low-speed rotating shaft bearings, where traditional frequency-based techniques may have limitations.

A normal bearing typically has a kurtosis factor of 3. Signals with higher kurtosis values exhibit more pronounced peaks, often exceeding three times the RMS value of the signal. This can indicate the presence of significant impulses or shocks within the data.

$$\textbf{Kurtosis Factor} = \frac{\frac{1}{N}\sum_{i=1}^{N}(x_i-\bar{x})^4}{[\frac{1}{N}\sum_{i=1}^{N}(x_i-\bar{x})^2]^2}$$

Crest factor:

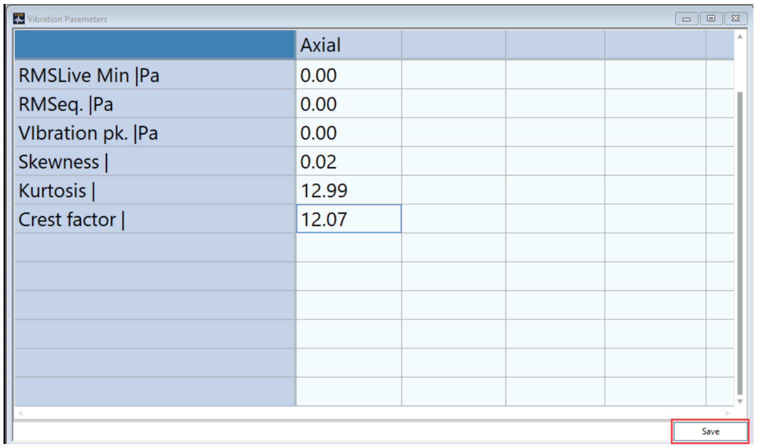
Crest Factor is a crucial parameter defined as the ratio of a waveform's peak value to its RMS value. It's a dimensionless measure. In Figure 3, the expression provides the specific definition of the crest factor. For a sine wave, the crest factor is 1.414. When receiving a typical vibration signal from a machine with a significant imbalance but no other issues, the crest factor reaches 1.5. However, as bearings wear and lead to impacts, the crest factor can increase significantly.

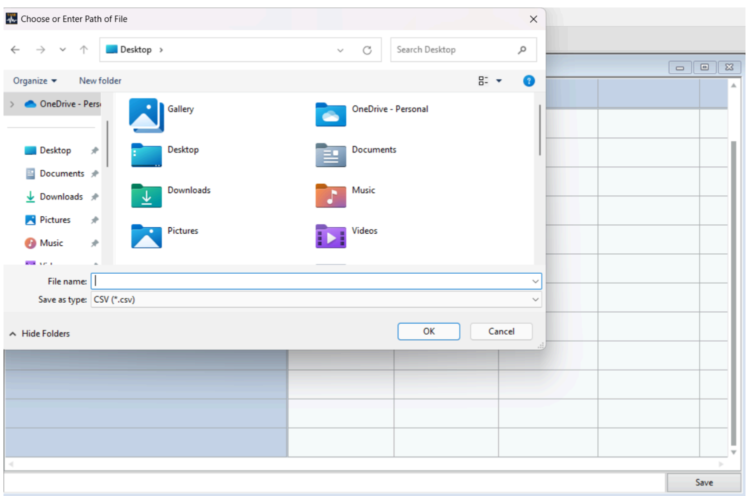
The underlying principle is that when bearings deteriorate, the peak levels of acceleration rise faster than the RMS levels due to increased impulsiveness. Calculating the crest factor is straightforward, and it isn't greatly affected by bearing speed and load. During the early stages of bearing damage, the inner race, bearing housing, rolling elements, and cage can generate periodic impact signals, causing the crest factor to rise. As the damage worsens, the RMS value increases, leading to a decrease in the crest factor value.

$$\textbf{Crest Factor} = \frac{x_{max}}{x_{RMS}}$$

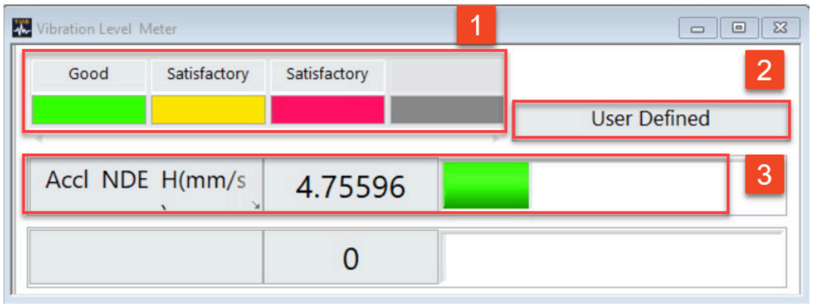
Saving the Measurement

The values in the software can be saved to .csv format. Click the save button and name the file and select the location.





Vibration Level Meter

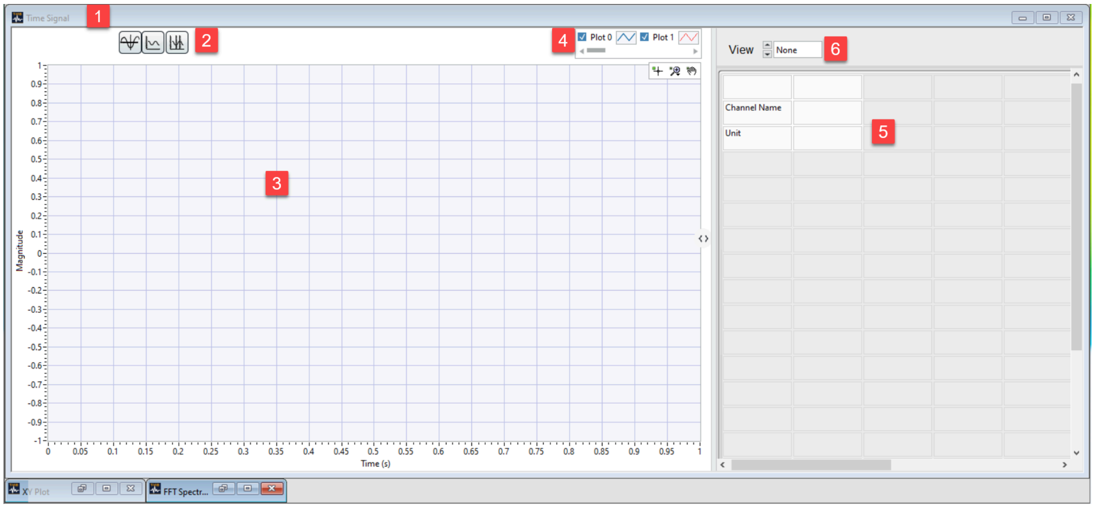


The Vibration Level Meter displays

1. Status and the color set for each status
2. The filter used for the measurement
3. The vibration level of each channel with it's unit

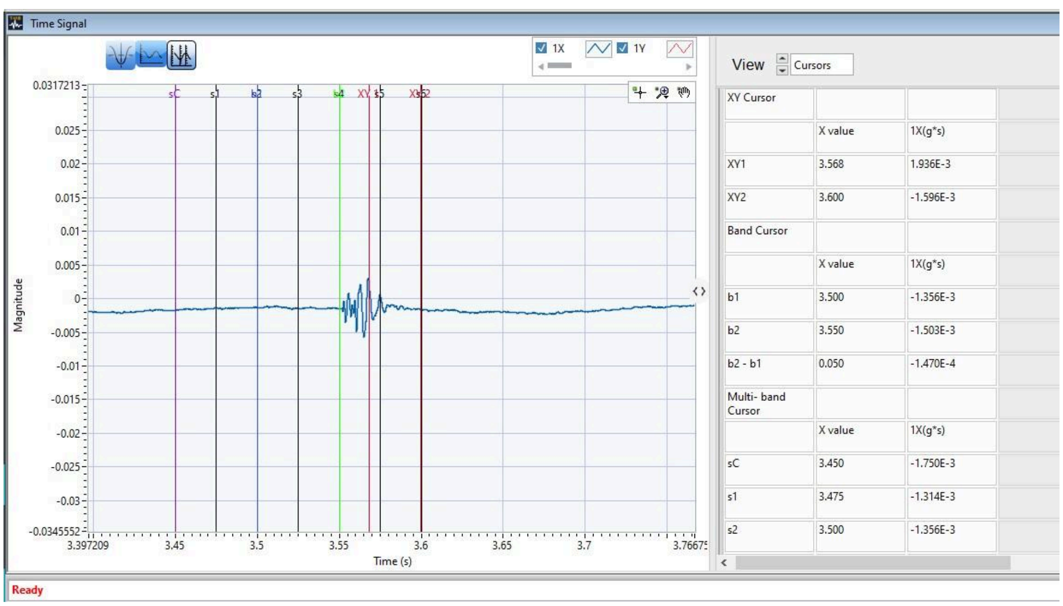
Time Signal

The Time signal tile shows the measurements of the sensor with respect to time.

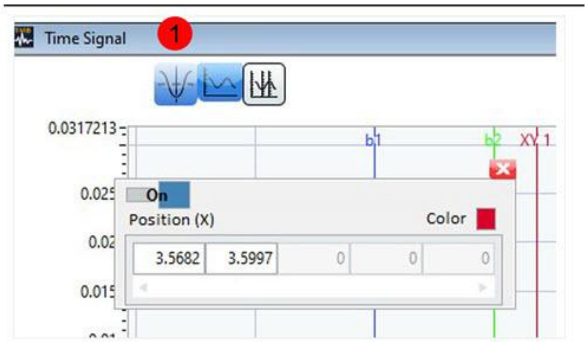


1. Tile Name: Time Signal

2. Cursor Options: These include the XY cursor, the band cursor, and the multi-band cursor.



The settings of the above example is as follows:



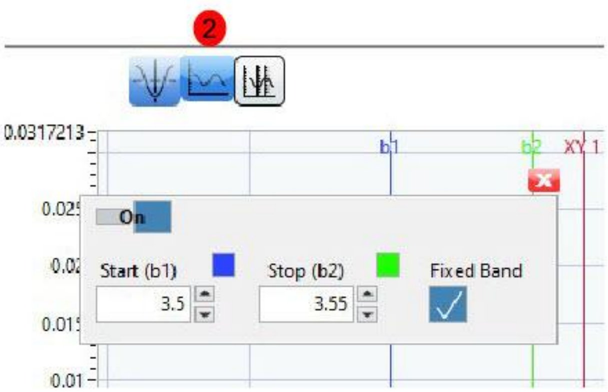
(1) XY cursor

The **slide button** allows the user to switch ON/OFF the cursors**.**

The **Position (X)** allows the user to input the position of the cursor which places the cursor to the input position. For example: In the above picture the 0 value has been replaced with 3.5682 which is the position of the first cursor & 3.5997 which is the position of the second cursor. For more cursors the user can input the corresponding position values.

To change the color of the cursor click the **Color** button and choose the appropriate color.

Once configured the user can press the **close** button .



(2) Band Cursor

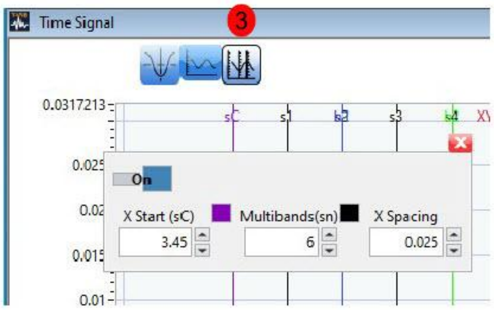
The **slide button** allows the user to switch ON/OFF the cursors**.** The functionality of band cursor is to determine the period/ width of a band, for example this feature allows the user to check the period of an oscillating signal (the period between the cursor is displayed in the **Info tab**).

The **Start(b1)** and **Stop(b2)** are the two cursors available in this feature. The user can input the position of the cursors.

The **Fixed band** allows the user to fix the difference between the band, on moving any of the cursor the period or the width between the cursors would be preserved.

To change the color of the cursor click the **Color** button and choose the appropriate color.

Once configured the user can press the **close** button .



(3) Multi band cursor

The **slide button** allows the user to switch ON/OFF the cursors**.** The functionality of multi band cursor is to determine a recurring event, for example this feature allows the user to check the periodicity of bearing impacts.

Input the following details to use this cursor. **X Start (sC)** the first cursor position , **Multibands(sn)** the number of cursors & **X Spacing** the width between consecutive cursors.

To change the color of the cursor click the **Color** button and choose the appropriate color.

Once configured the user can press the **close** button .

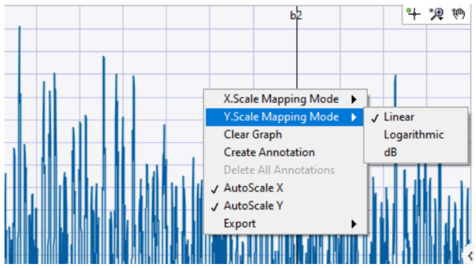
3. Graph: The data acquired is displayed on the **Graph ,** with configured function as Y-axis and time at X-axis.We can use the standard tools on the top right toolbar such as zoom out/in, pan etc.

Additional features are available on the graphs. To do so right click on the graph select the required option from the list.

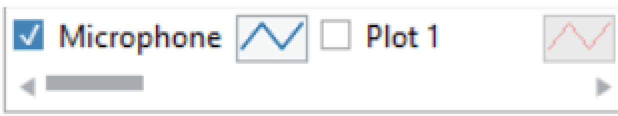
The user can change the scales of X axes & Y axes between logarithmic/ linear/dB. To do so right click on graph select the **X Scale/ Y Scale Mapping Modes**. Choose **logarithmic/ linear/dB** as per user requirements.

The graph also allows the user to create annotations to the plots , export the graph as a picture or data points to excel, enable/ disable auto scale X/Y etc. To do so right click on graph, select the required option.

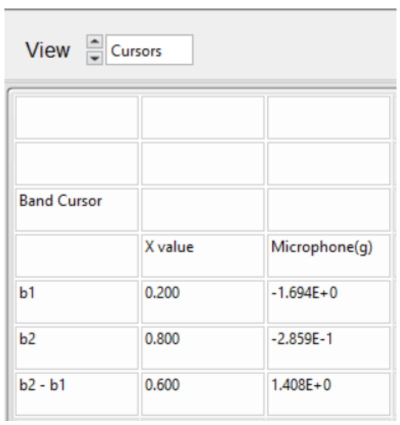
Note: Enabling the auto scale will allow the user to view the graph only at the maximum X/Y axes range whereas disabling would allow the user to select the ranges in the graph to be viewed.



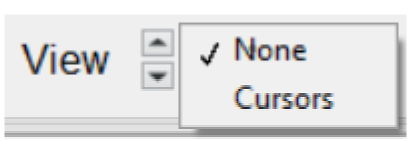
4. Plots details: This option allows the user to choose the channels of which the plots are to be viewed.For example in the below picture only the Microphone channel has been selected to display the data on the graph.



5.Info tab: Displays the details of the plots such as channel name, units & cursors values etc.



6. View Selection:The view selection allows the user to select what details are to be displayed on the Info Tab. Selecting the None option would allow the user to view the channel name & unit of the value display. To view the cursor details on the info tab the user would require to select cursor in the view selection .



Sound Level Meter (TSLM 204)

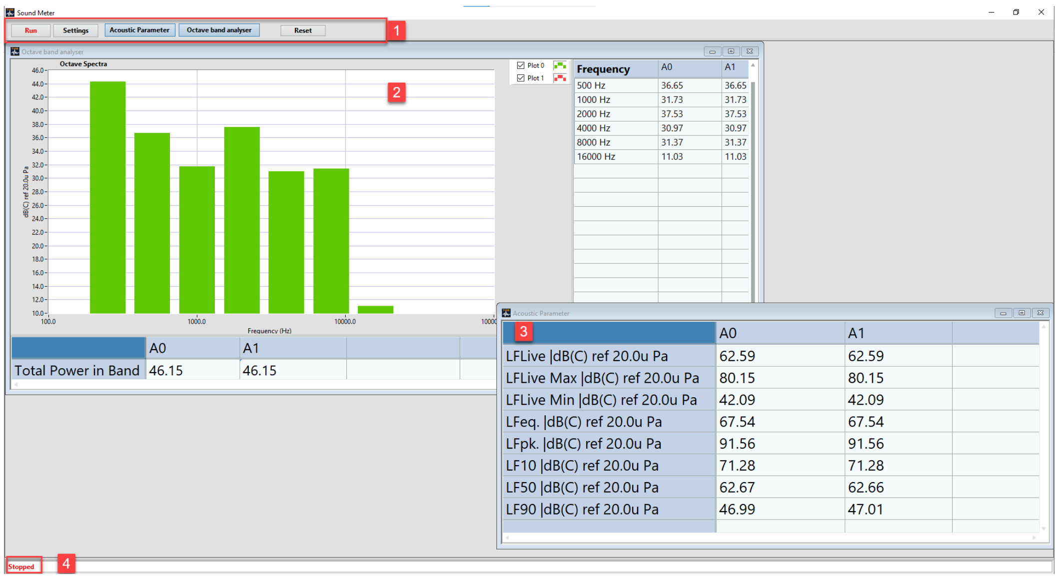
The sound level meter module in TVIB is used for acoustic measurements. The software is capable of below measurements

Broadband Level: The single value measured by a sound level meter is referred to as the "broadband value" as it uses all values across the audio frequency bands (20 Hz to 20 kHz) to calculate the level. It is typically measured in decibels (dB), which is a logarithmic unit.

Octave Band Analyser: The octaves provided suits for tasks such as optimization of sound systems and rooms. It measures with a resolution of 1/1 or 1/3 octave.

Module Interface

The module interface consists of below options



1. Function Bar

2. Octave Band Analyzer *tile*

3. Acoustic Parameter

4. Notification Bar

Function bar

1. Run button

This button can be used to start & stop the acquisition.

2. Settings

The settings can set up an acquisition & various functionality of the test. We will discuss the various options in the settings in the next section.

3. Acoustic Parameter

On pressing the Acoustic Parameter button, a tile pops up. Parameters such as Equivalent, Minimum & Maximum sound pressure Levels measured will be displayed in this window.

4. Octave band Analyzer

On pressing the Octave band Analyzer button, the tile with the octave spectrum graph pops up.

5. Reset

The reset button resets and starts the measurement again.

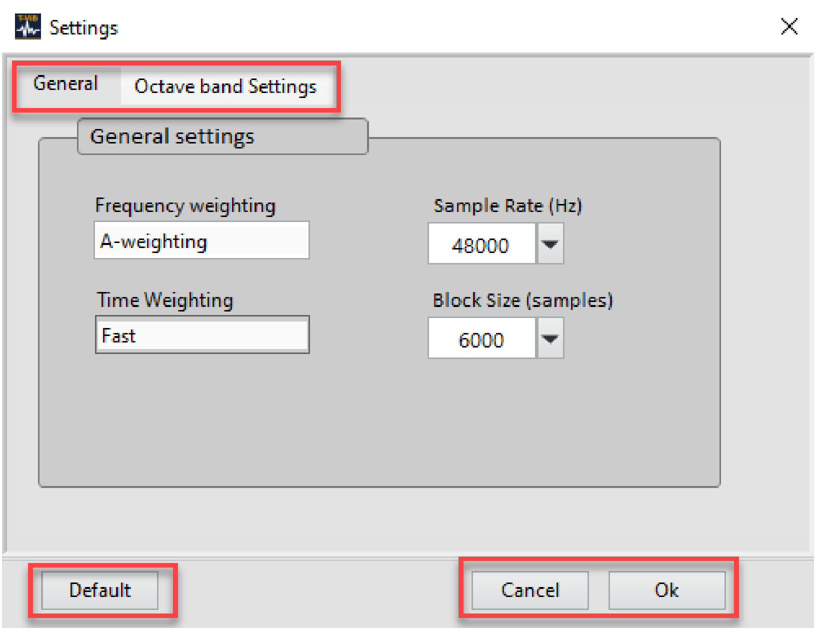
Settings

Below we will discuss the settings in detail.

The Settings tab consists of two panes.

1. **General Settings:** This allows the user to select the general acquisition settings such as the Frequency Filter type, time weighting , Sample Rate and Block Size etc.

2. Octave band Settings : This allows the user to adjust the settings of the octave spectrum such as octave bandwidth, scale setting etc.

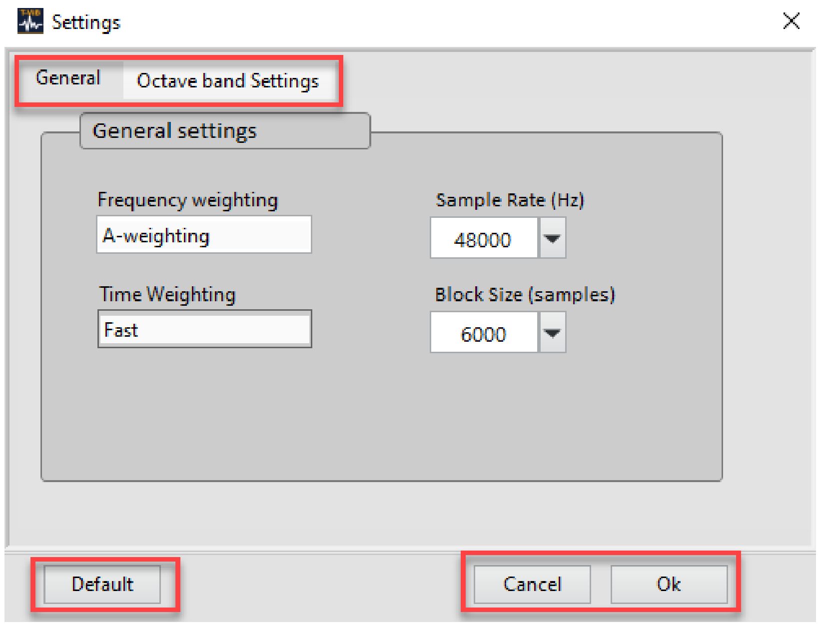


Default : This will load the settings' default values in every tab.

Cancel : Closes the settings tab, also the changes made by the user on the settings will be lost and the previous settings would be loaded. Pressing the Close button on the settings window would have the same effect.

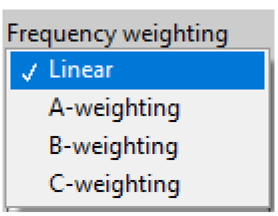
Ok : Save the selected configuration as the present settings .

General Settings tab



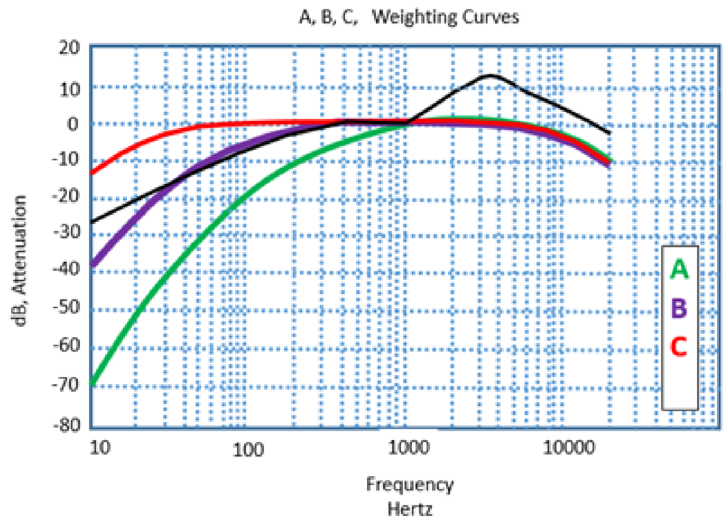
The options in the General Settings are discussed below

1. Frequency weighting : There are four Frequency weightings options available

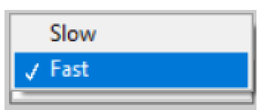


Linear : This option does not have any filter, the inbuilt filters like anti-aliasing filters would decide the frequency range of the signal.

A-weighting, B-weighting & **C-weighting** : :The ANSI S1.4 standard defines the frequency responses of the A, B and C Weighting filters. A-weighting is a frequency dependent curve (or filter) which is applied to sound pressure microphone measurements to mimic the effects of human hearing.Used in various applications and industries, the B and C weighting curves are similar to A-weighting, but do not have as much attenuation below 1000 Hz. The C weighting curve is the flattest of the A, B and C curves.B-weighting is the best weighting to use for music listening purposes.C-weighting is used for high-level noise measurements.



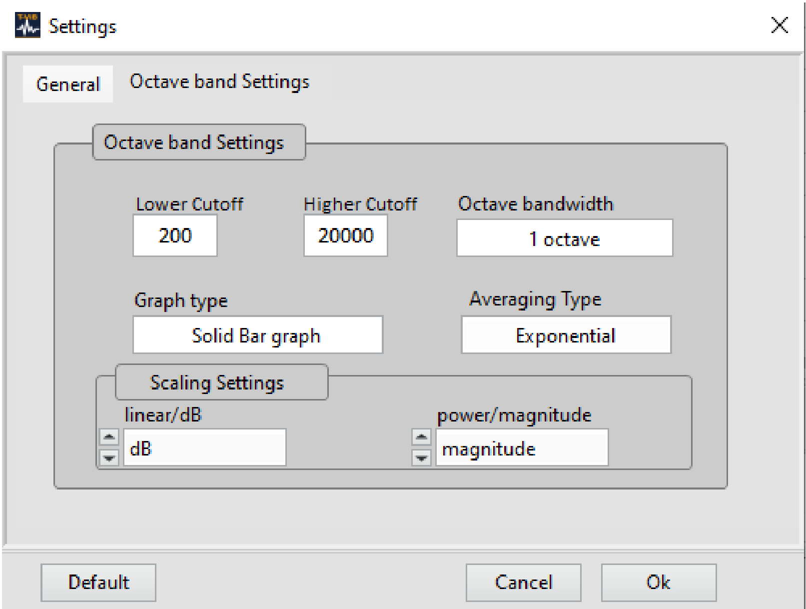
4. Time Weightings : There are two time weightings available



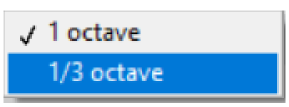
The IEC 61672-1 standard describes two different time weightings, Fast (F) and Slow (S). They both dampen the reaction of the displayed level to a sudden change in the sound level. Fast reacts quicker than Slow. If for instance in a quiet environment a loud, constant sound is suddenly switched ON, the 'F' weighted level display would take approximately 0.6 seconds to reach the new level, while the 'S' weighted level display would reach the new level only after approx. 5 seconds. These values are given by the time constants for the 'F' (t = 125 ms) and the 'S' weighting (t = 1 s), which are defined in the standard.

5. Sample Rate and Block Size: The Maximum sampling rate of the device would be the best option, the block size would be automatically updated based on the time weighting applied.

Octave band Setting



1. **Lower Cutoff :** The minimum frequency to be displayed in the octave spectrum.
2. **Higher Cutoff :** The maximum frequency to be displayed in the octave spectrum.
3. **Octave bandwidth**: There are two options 1 octave bandwidth, ⅓ octave bandwidth.



The options allow the user to view the spectrum in 1 octave or ⅓ octave bands

1/1 Octave Band Noise Measurements

1/1 Octave Band measurements are used when the frequency composition of a sound field is needed to be determined. Octave analysis is often used in noise control, hearing protection and sometimes in environmental noise issues.

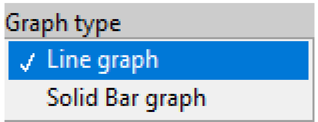
The common octave frequency bands are: — 31Hz, 63Hz, 125Hz, 250Hz, 500Hz, 1kHz, 2kHz, 4kHz, 8kHz and 16kHz – and their composition is made up of the Lower Band Limit, Centre Frequency and Upper Band Limit.

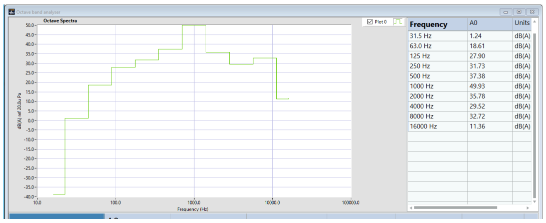
1/3 Octave Band Noise Measurements

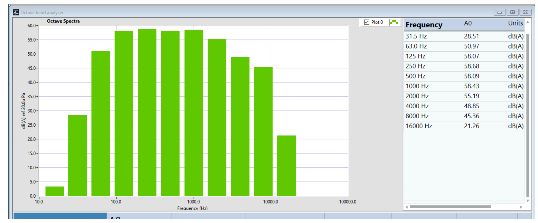
Mainly used in environmental and noise control applications, 1/3 Octave Bands provide a further in-depth outlook on noise levels across the frequency composition.

Each 1/1 (single) Octave is further split into three, providing a more detailed view of noise content.

4. **Graph type:** There are two types of graph options available, Line and Solid Bar graphs

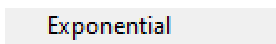






Line graph   Solid Bar graph

5. **Averaging:** This option has been configured as exponential averaging to the octave measurements



Exponential averaging: It is a continuous averaging process that weights current and past data differently. The amount of weight given to past data as compared to current data depends on the exponential time constant. In exponential averaging, the averaging process continues indefinitely.

The user can reset the measurement using the **Reset** button.

6. **Scale Settings:** The scale settings provide the user to view the measurements in dB and Linear Scales. Also in Linear scale the user can view the magnitude or Power values of the octaves.

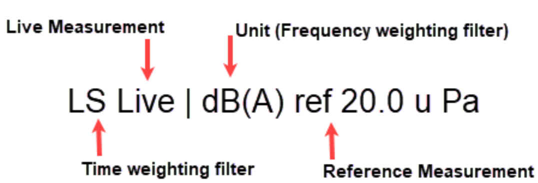
Acoustic Parameter

	A0		
LSLive |dB(A) ref 20.0u Pa	54.27		
LSLive Max |dB(A) ref 20.0u Pa	58.98		
LSLive Min |dB(A) ref 20.0u Pa	53.00		
LSeq. |dB(A) ref 20.0u Pa	56.17		
LSpk. |dB(A) ref 20.0u Pa	77.92		
LS10 |dB(A) ref 20.0u Pa	58.87		
LS50 |dB(A) ref 20.0u Pa	55.87		
LS90 |dB(A) ref 20.0u Pa	0.00		

The Acoustic Parameters *tile* displays the parameters in tabular form.

1. **Channel Name**: The measurement will be displayed as columns, with each channel value corresponding to the parameter in each column of the table. The first row displays the channel names.
2. **Parameter:** The parameters calculated will be visible in the first column.

   The format is as follows



   Note: The dB reference is configured as default 20 uPa for Sound Measurements

3. **Value:** This is the value computed from the measurement.

The parameters change based on the time weighting filter, frequency weighting filter and the scaling selected.

L Live is the sound level value measured real time. The max, min values are the maximum & minimum values captured during the measurement.

Leq (or LAeq) is the Equivalent Continuous Sound Pressure Level

Equivalent Continuous Sound Pressure Level, or Leq/LAeq, is the constant noise level that would result in the same total sound energy being produced over a given period.

LAeq is a fundamental measurement parameter designed to represent a varying sound source over a given time as a single number. This number is a measure of the energy contained within the sound at the point of the receiver. This is useful in terms of the potential for sound to damage or disturb and is extensively used in environmental noise standards as well as many other regulations and documents.

L_{10}, L_{50} and L_{90}

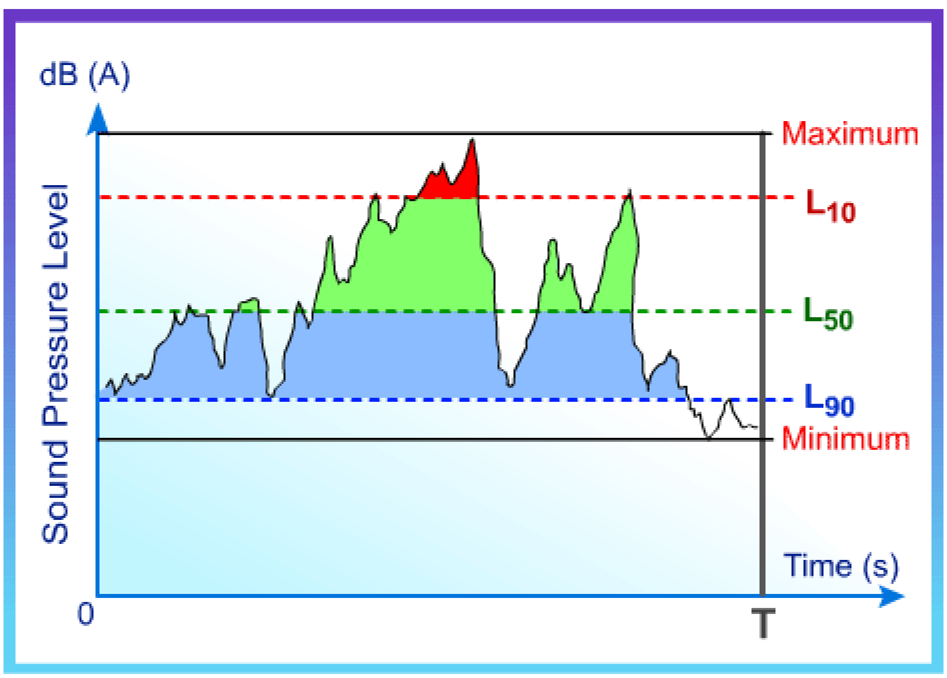
The commonly used value of n for the n-percent exceeded level, L_n, are 10, 50, and 90.

L_{10} is the level exceeded for 10% of the time. For 10% of the time, the sound or noise has a sound pressure level above L_{10}. For the rest of the time, the sound or noise has a sound pressure level at or below L_{10}. These higher sound pressure levels are probably due to sporadic or intermittent events.

L_{50} is the level exceeded for 50% of the time. It is statistically the mid-point of the noise readings. It represents the median of the fluctuating noise levels.

L_{90} is the level exceeded for 90% of the time. For 90% of the time, the noise level is above this level. It is generally considered to be representing the background or ambient level of a noise environment.

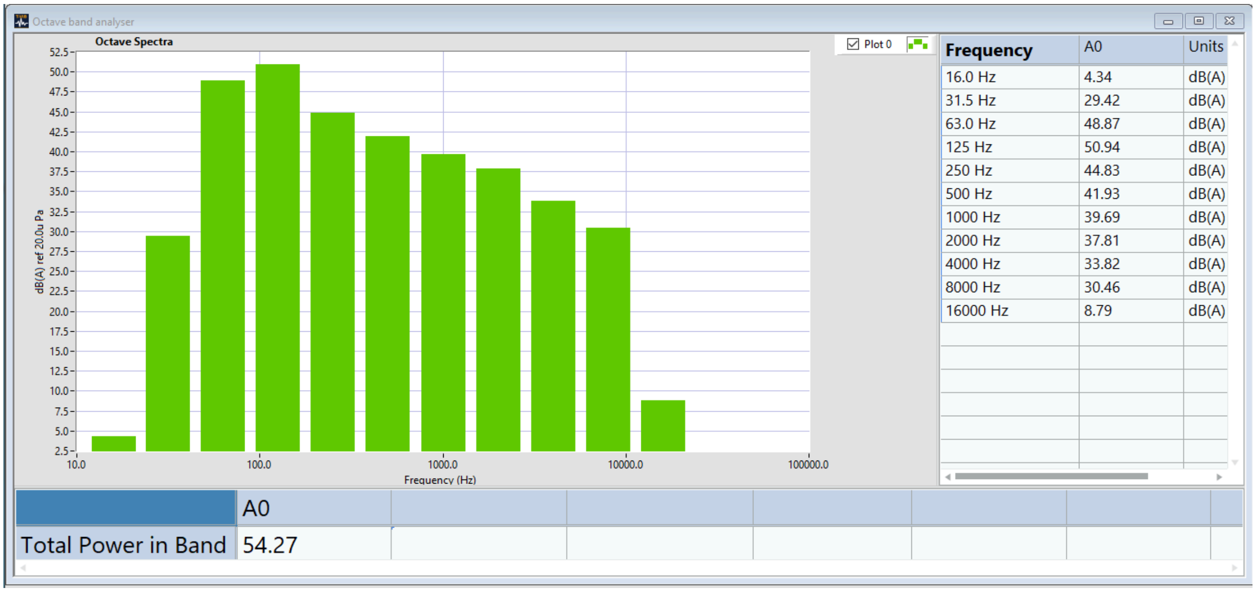
For a varying sound, L_{10} is greater than L_{50} which in turn is greater than L_{90}. The following graph illustrates L_{10}, L_{50} and L_{90}.



Please note that $L_{10} > L_{50} > L_{90}$ for the same sound or noise.

Octave Band Analyzer

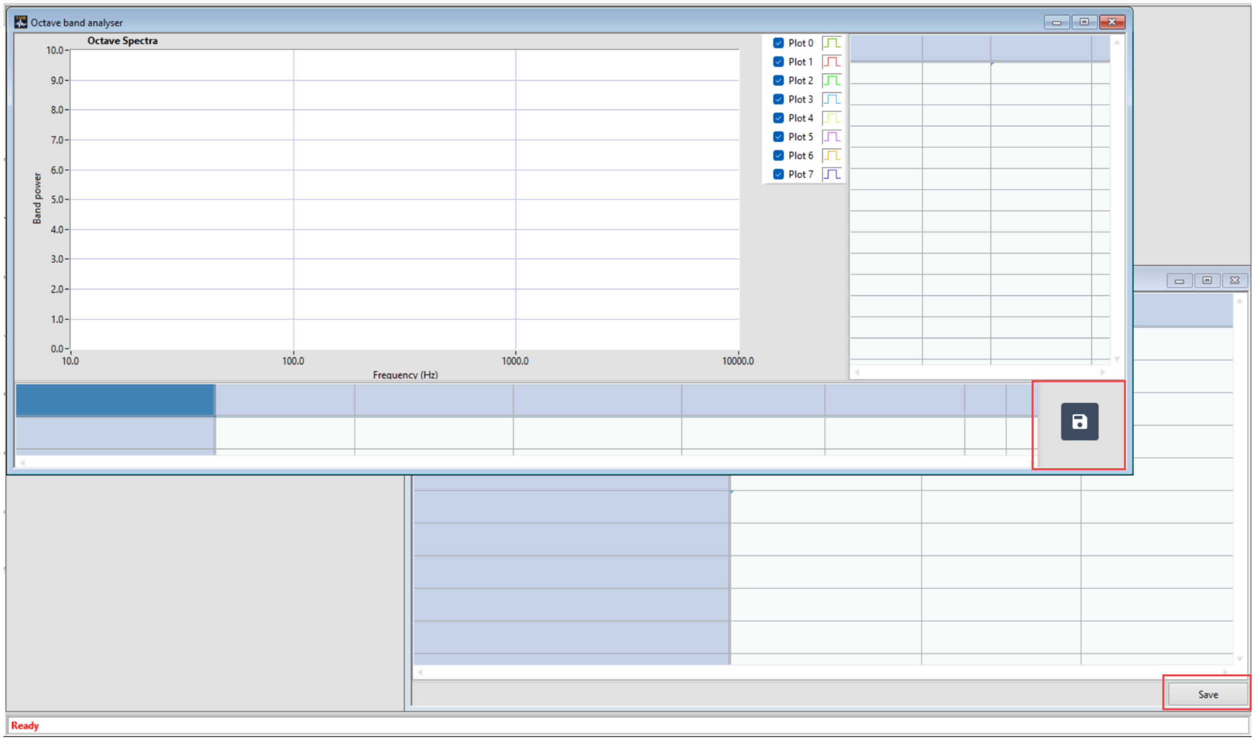
The Octave Band Analyzer tile computes the octave spectrum and the Total Power of the octave Measurements.



The band width 1 octave / ⅓ octave can be selected using the octave band settings and the values of each band can be viewed in the table along with the unit.

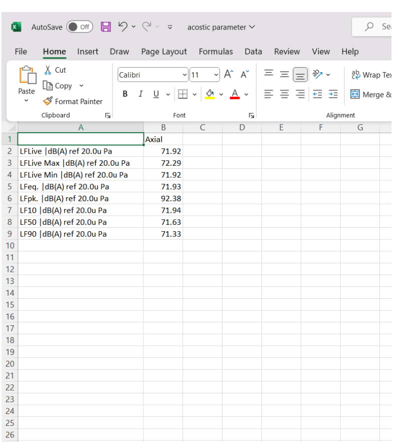
Saving the measurement

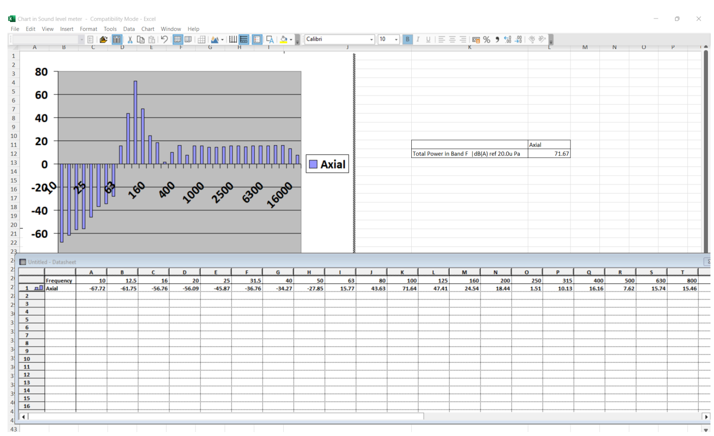
The data captured can be saved in the .csv format, click the save button and enter the name and location of the file on the browser.



.

Saved Measurement can be opened in the Excel etc.





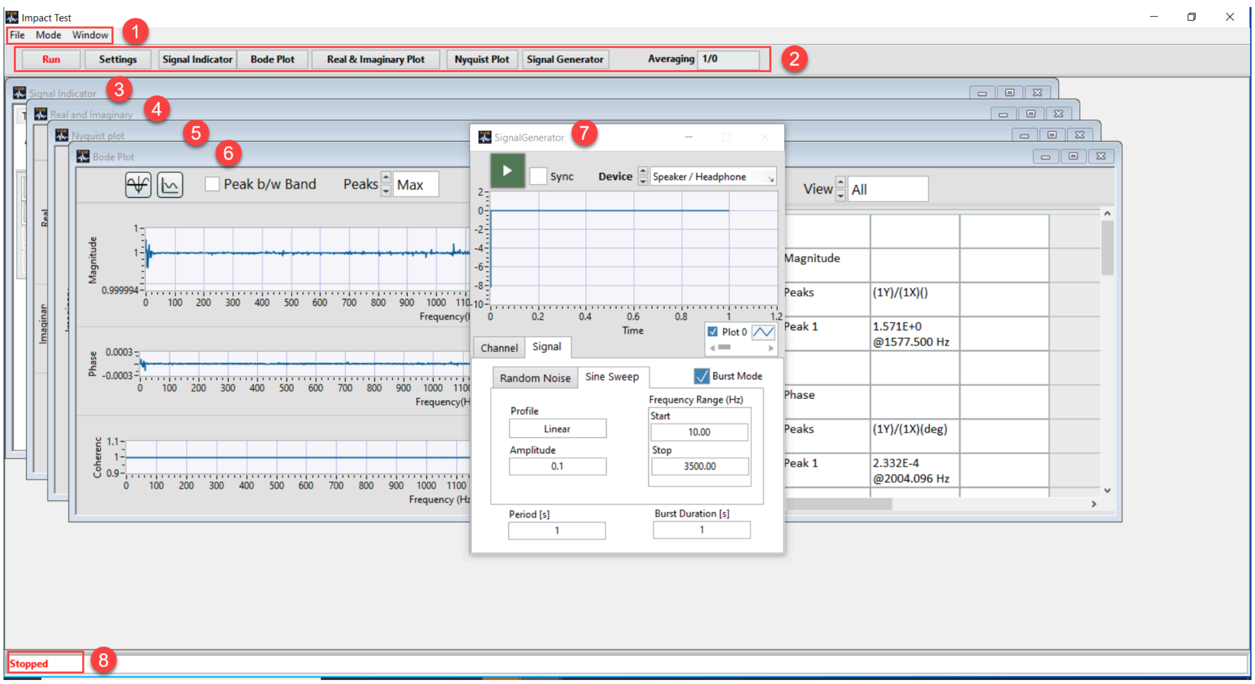
Acoustic Parameter Octave Band Analyzer

FRF Test (TIST 205)

The FRF Test Module allows the user to do Frequency Response Function Measurements especially used in Modal frequency determination by Impact / Shaker Test, Transmissibility etc.

Module Interface

The interface can be divided in to following components



1. Menu bar

2. Function bar

3. Signal Indicator tile

4. Real and Imaginary tile

5. Nyquist Plot tile

6. Bode Plot tile

7. Signal Generator tile

8. Notification bar

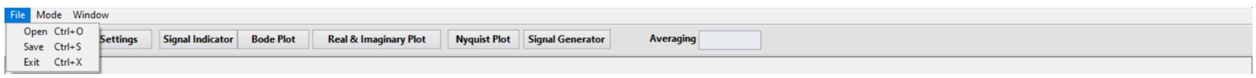
In the following subsections we will discuss in detail the various features & functionality of the above components.

Menu bar

The menu bar deals with the basic features such as saving , opening saved data, switching modes, automatically arrange the tiles etc. , the options are briefly described below

1. File

The file menu consists of the following options



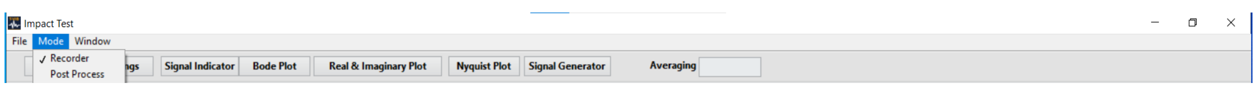
a. Open: This menu option is disabled as of now, will be updated in future revisions.

b. Save: This menu option allows the user to save a measurement acquired in the recent acquisition in the uff format.

c. Exit : This menu option allows the user to exit the module and go to the application launcher.

2. Mode

The mode menu consists of the following options.Each mode has a unique settings tile.

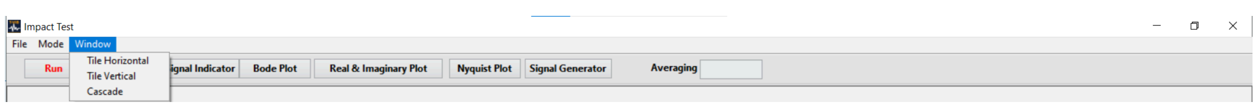


a. Recorder: This is the measurement recording mode. This allows the user to take measurements onsite/ perform a test in laboratory.

b. Post Process: This mode is disable. Will update in future revisions.

3. Window

The window menu basically is to arrange the tiles and consists of the options as below



Note: The software allows the user to use single/multiple tiles simultaneously (each tile has a functionality, in this module Bode plot tile gives the Bode plot, Real & Imaginary Plot tile gives the Real & Imaginary Plots etc.).

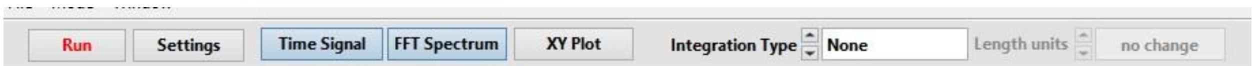
a. Tile Horizontal: This option arranges the tiles (whichever tile is being used for measurement) horizontally.

b. Tile Vertical : This option arranges the tiles (whichever tile is being used for measurement) vertically.

c. Cascade : This option arranges the tiles (whichever tile is being used for measurement) one over the other.

Function Bar

The function bar is composed of buttons; it performs a variety of functions as detailed below.



The blue highlight on Time Signal & FFT Spectrum buttons indicates that these tiles have been chosen by the user for viewing the waveform and the spectrum of the Measurement from the sensor in the software.

1. Run button

In the recorder mode, this button can be used to start & stop the acquisition.

In the post process mode, this button can be used to open the recorded .tdms format measurements.

2. Settings

In the recorder mode , settings can set up an acquisition & various functionality of the software. We will discuss the various options in the settings in the next section.

Settings

Below we will discuss the settings in detail.

The settings tab consist of four control panes,

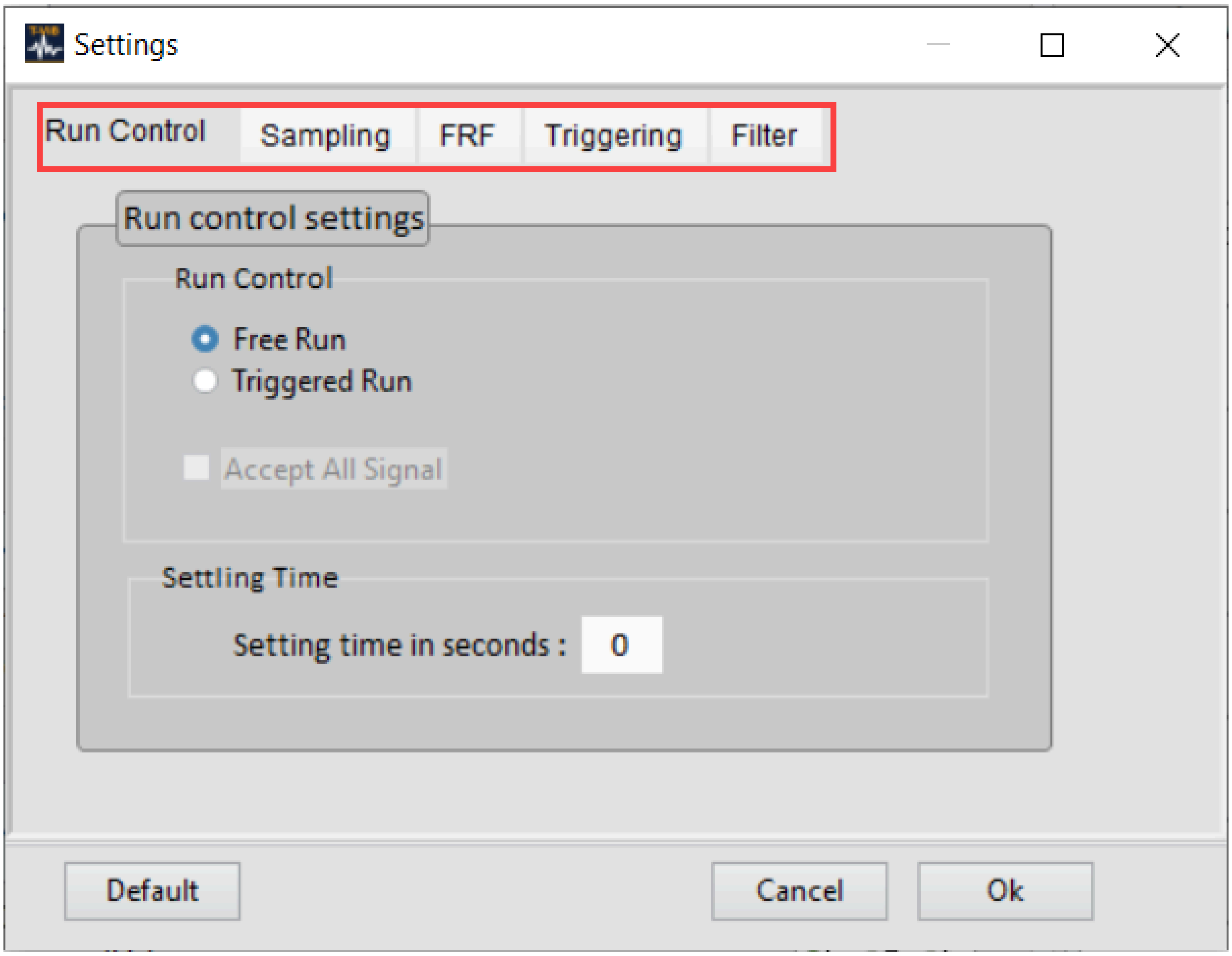
1. Run control settings : This setting allows the user to adjust the running mode of the acquisition **i.e.** either run the acquisition continuously and stop when the user press the stop button or run the acquisition for a time/number of data blocks acquired as specified by the user.

2. Sampling settings : This enables the user to fine-tune the block size (sample size), the sampling rate, and the decimation ratio.Additionally, FFT parameters such as the window type, the averaging method, the weighting mode, and the number of averages are available.

3. FRF settings: This settings allows the user to choose the stimulus & Response Channel, FRF estimates,Window function & the option for viewing the amplitude in dB Scale.

4. Triggering : This tab has the Trigger Settings.The user may configure the trigger channel, type, slope, levels, sampling rate, ahysteresis.

5. Filter : This tab enables and disables filtering.The user may choose the type of filter,order, and design technique of the filter.

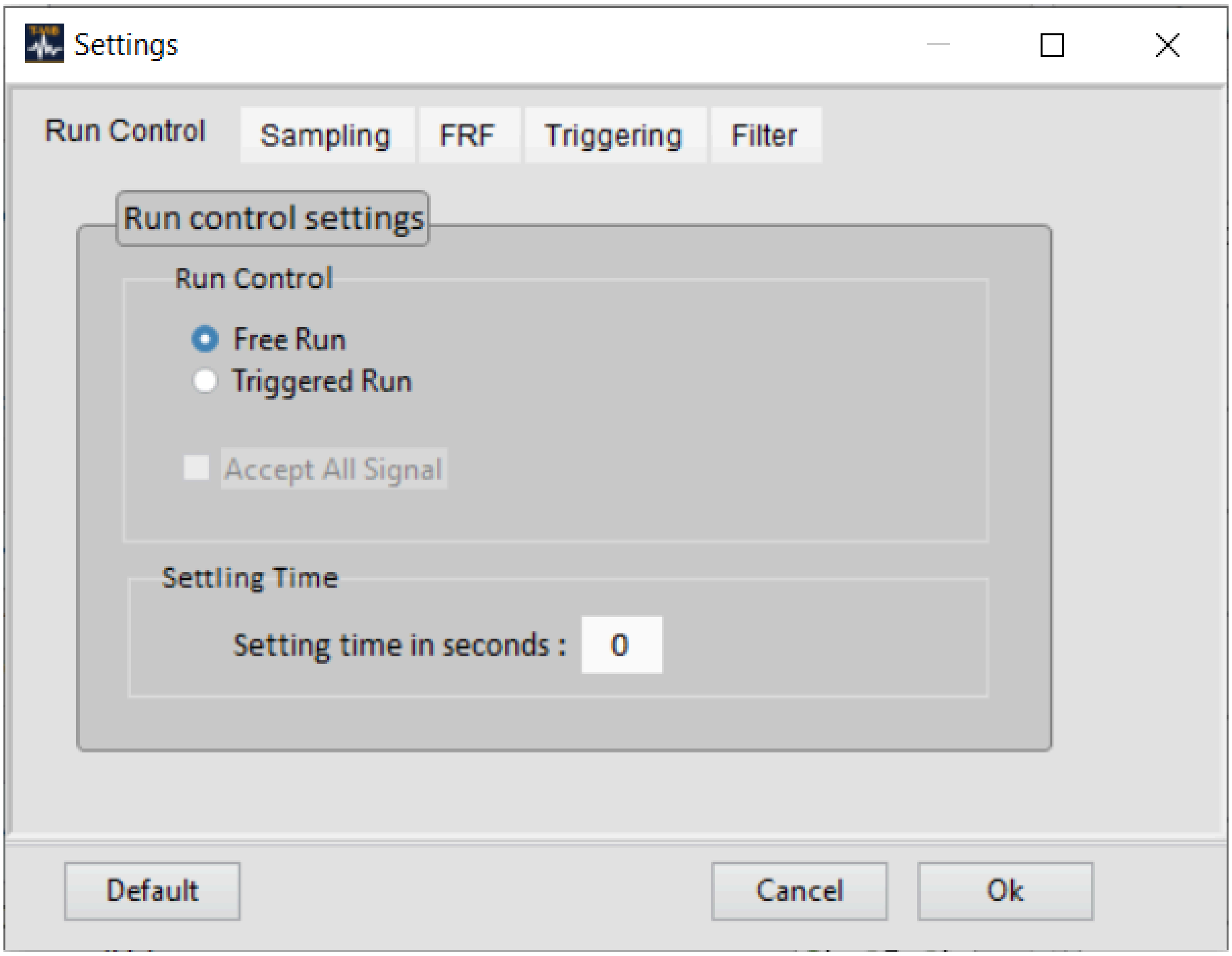


Default : This will load the settings' default values in every tab.

Cancel : Closes the settings tab, also the changes made by the user on the settings will be lost and the previous settings would be loaded. Pressing the Close button on the settings window would have the same effect.

Ok : Saves the selected configuration as the present settings .

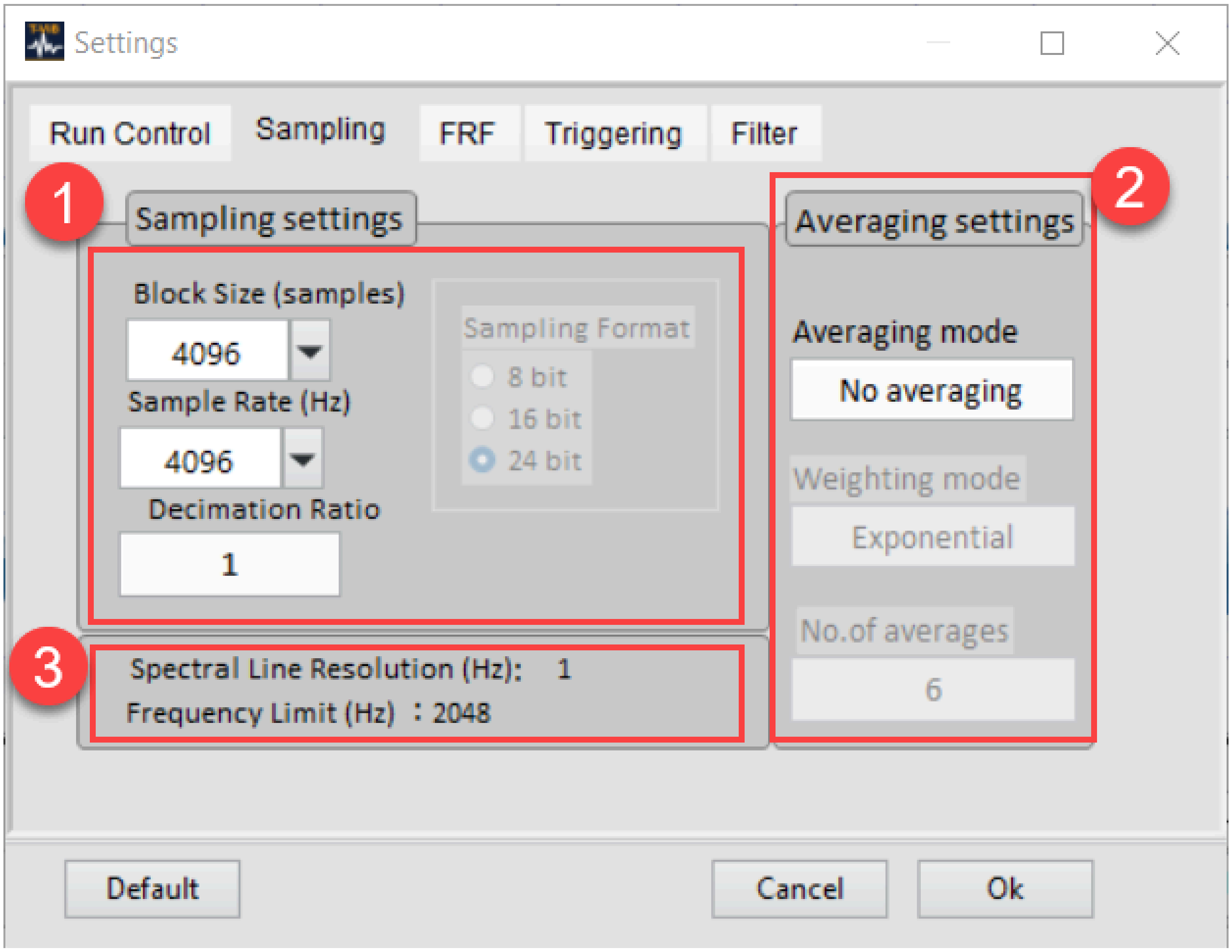
Run Control



Run Control: Enabling **Free Run** reads samples indefinitely, whereas **Triggered Run** reads samples only when triggered for a specified time period and a specified number of blocks as per the sampling settings.In Triggered Run, the user has the option to accept all the signals/ accept on user intimation only (This allows the user to accept/Reject the acquired data).

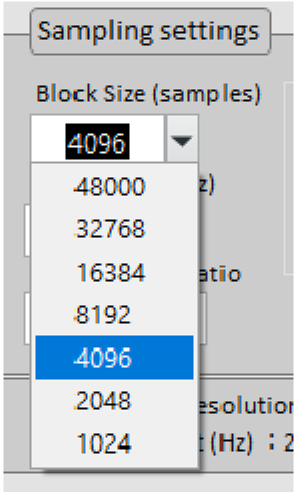
Settling Time: Settling Time is defined as the time (in secs) required for the accelerometer's internal electronics to stabilize its bias voltage and achieve its operational condition. In this example, if we specify a time limit, the software will generate readings only after that period.

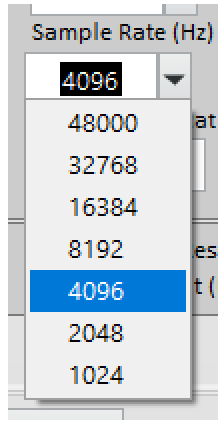
Sampling



1

We can select the required **Block Size**(no of samples) and **Sample Rate** from the drop down list.





They default values to 48000, 32768, 16384,8192,4096,2048, and 1024, which correspond to the most popular sound card. We can enter in any values we require.

The decimation ratio is used to achieve a lower sampling rate than that supported by the digitizer. This feature controls the ratio with which the digitized signal is down-sampled. We can use natural numbers till 50. This value is used to divide the selected sample rate.

A value of 1 will not decimate the data at all.

For example : if we select sample rate 48000Hz and decimation ratio of 2, the sampling rate of the decimated data would be 24000Hz (48000/2). The resulting measurement range(frequency limit) will then be 0 to 12000 Hz and the frequency resolution improved by a factor of 2. [In short, Decimation improves the resolution of the Spectrum but the max frequency limit would be reduced.]

The **Sampling Format** is the number of bits used to describe each sample.

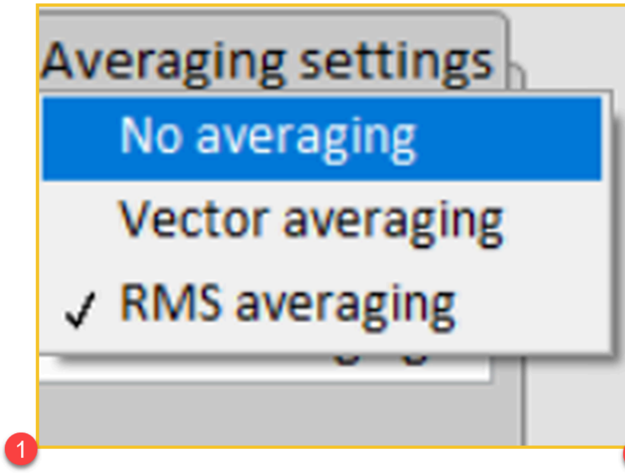
The greater the number of bits, the more data will be stored in each sample. Common sample formats are 8 bit, 16 bit and 24 bit. 8 bit samples are low-quality, and not used often.

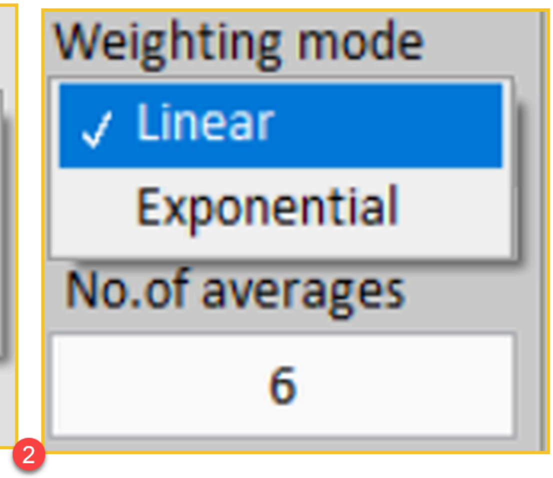
2

Averaging Settings : This section allows us to enable or disable averaging.

Additionally, linear and exponential weighting modes for averaging with the specified number of averages are offered.

Using None option will disable windowing.





(1). Averaging Mode

No averaging will not average the FFT Spectrum.

Vector averaging, also called coherent averaging, or time or angle synchronous averaging, can reduce the noise floor in an even-angle signal.

RMS averaging averages the power of a signal. The averaged RMS spectrum does not contain phase information. Thus, performing. RMS averaging on spectra can reduce signal fluctuation but not the noise floor.

(2). Weighting Mode

- Linear weighting—Weights each individual spectrum by the same amount in the averaged spectrum. Use linear weighting for analysis purposes.
- Exponential weighting—Weights the most recent spectrum more than the previous spectra, which makes the averaged spectrum more responsive to changes in the input signal. This responsiveness makes exponential weighting ideal for the configuration phase of a measurement. Exponential weighting also is useful for monitoring applications because the averaged spectrum responds to a singular event. A linearly averaged spectrum might not respond noticeably to a singular event, especially with a large number of averages.

3

Spectral Line Resolution: It refers to a sensor's capacity to distinguish between very small wavelength intervals. The higher the spectral resolution, the more condensed the wavelength range of a given channel.

$$\textbf{Spectral Line Resolution} = \frac{Sample\ rate\ \times fft\ wf}{block\ size\ \times decimation\ ratio}$$

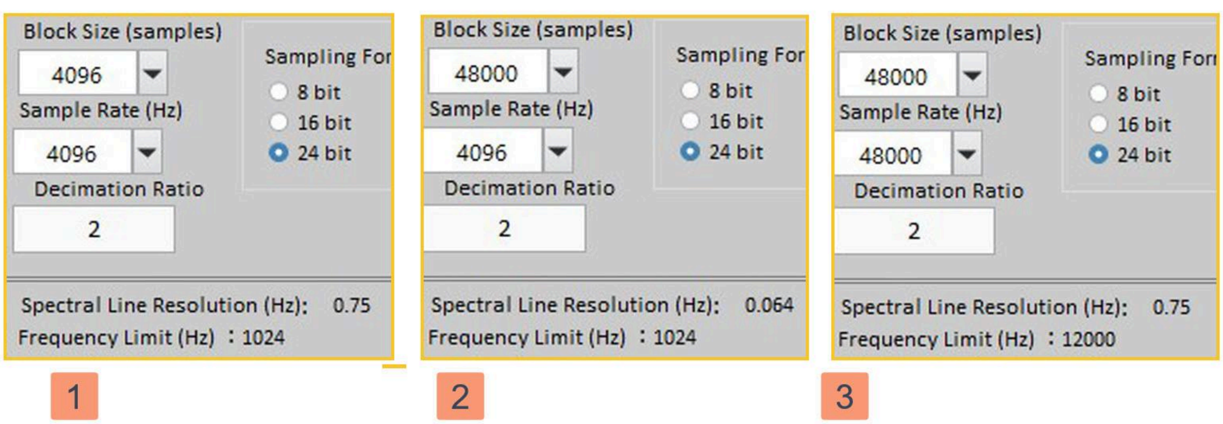
*** fft wf** implies the normalized equivalent noise bandwidth of FFT window type chosen. The values for the different FFT windows are mentioned on table below,

FFT Window Type	**Normalized equivalent noise bandwidth (fft wf)**
None	1
Hanning	1.5
Blackman-Harris	2.00435
Flat Top	3.81936
Force	1
Exponential	1
Gaussian	2.02119

Frequency Limit: provide the range across which calculations will take place.

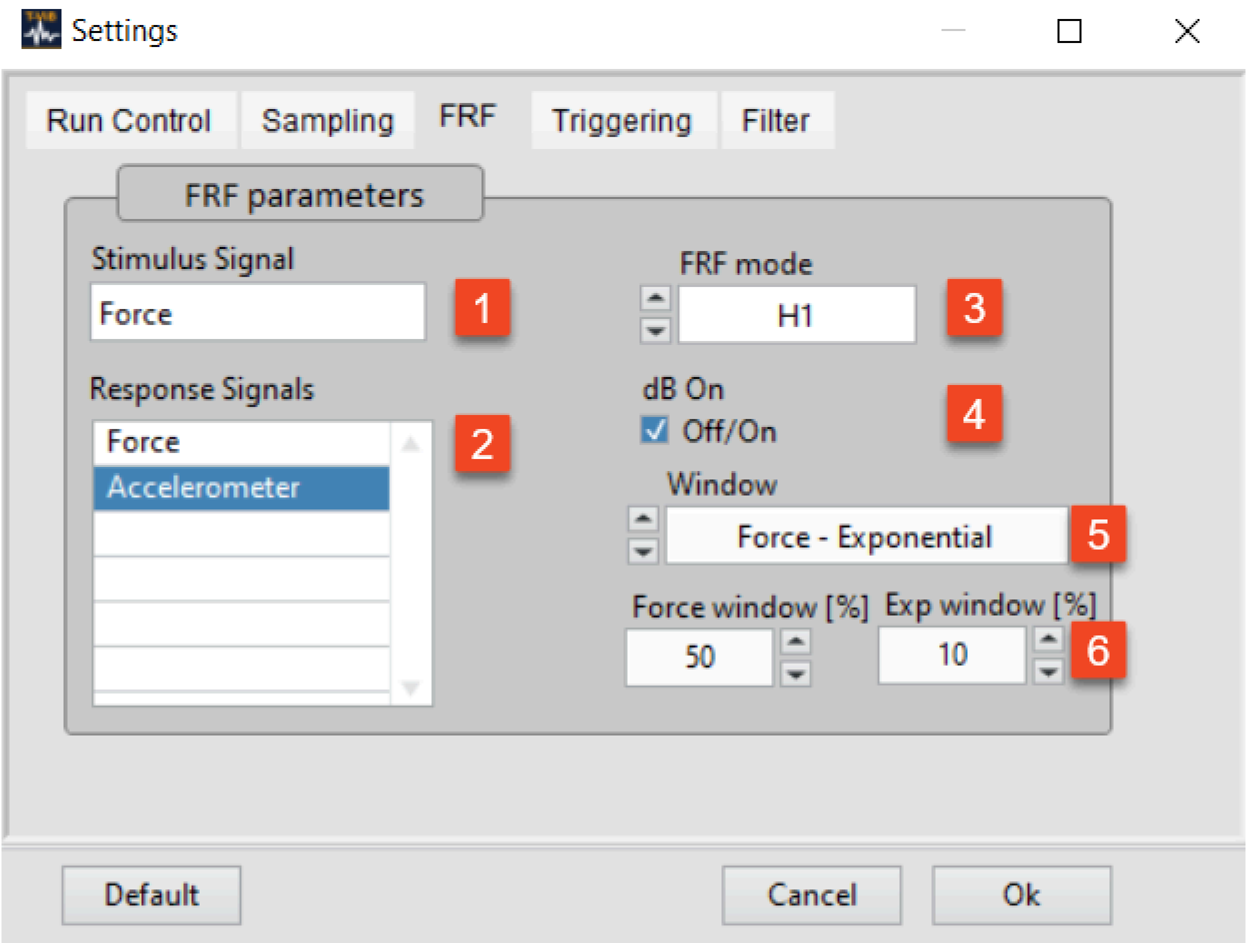
$$\textbf{Frequency Limit} = \frac{Sample\ rate}{2 \times decimation\ ratio}$$

The **Spectral Line Resolution** and **Frequency Limit** values will update to automatically reflect the current Sampling Rate, FFT window and Decimation ratio settings.



FRF

This settings allow the user to set the parameters of the Frequency Response Function Plot.



1. Stimulus Signal : Select the channel to which the excitation (like force sensor, impact hammer etc.) is measured.

2. Response Signal: Select the channel to which the response (like accelerometer, microphone etc.) is measured.

3. FRF Mode: Select the FRF estimation mode based on the below conditions

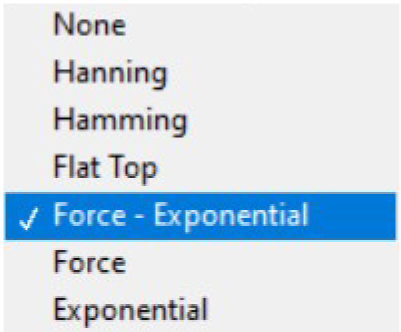
a) H1 : Minimizes errors in the result when noise contaminates the output signal.

b) H2 :Minimizes errors in the result when noise contaminates the input signal.

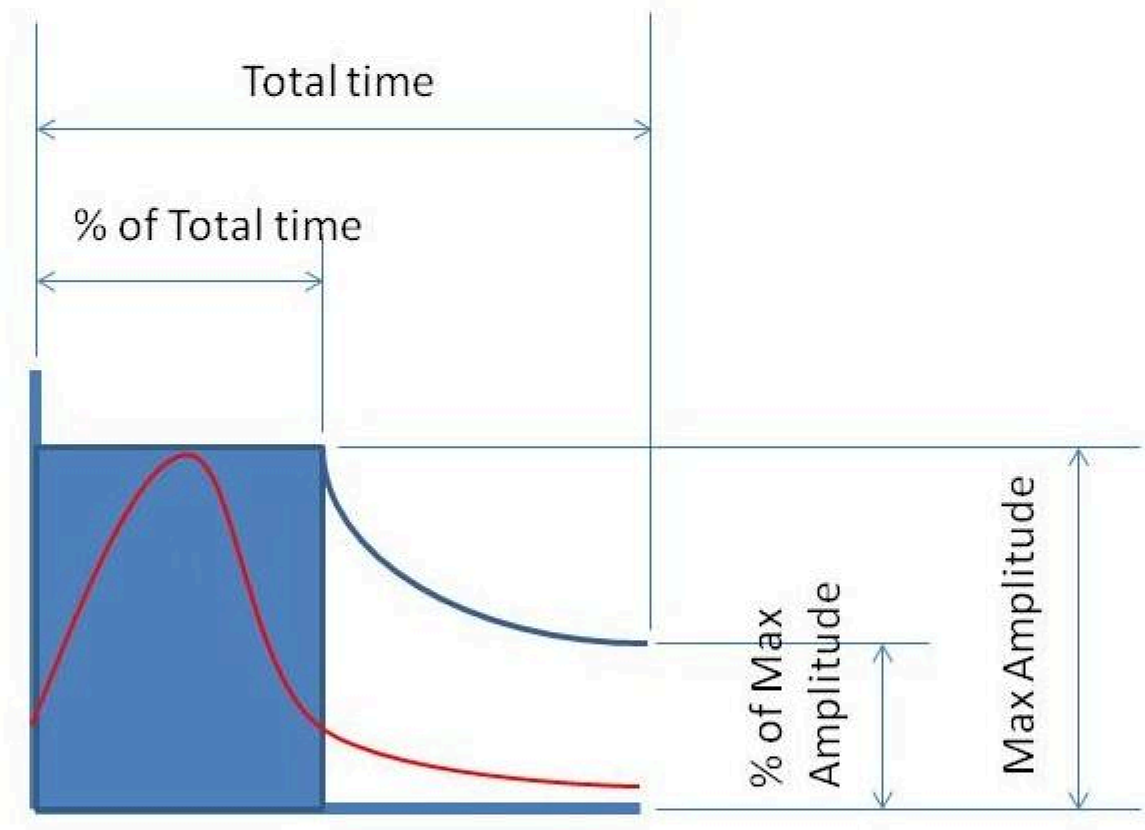
c) H3 : Minimizes errors in the result when noise contaminates both the input signals and the output signals.

4. dB On : Select this option to view the plots (Y-Axes) in decibel scale.

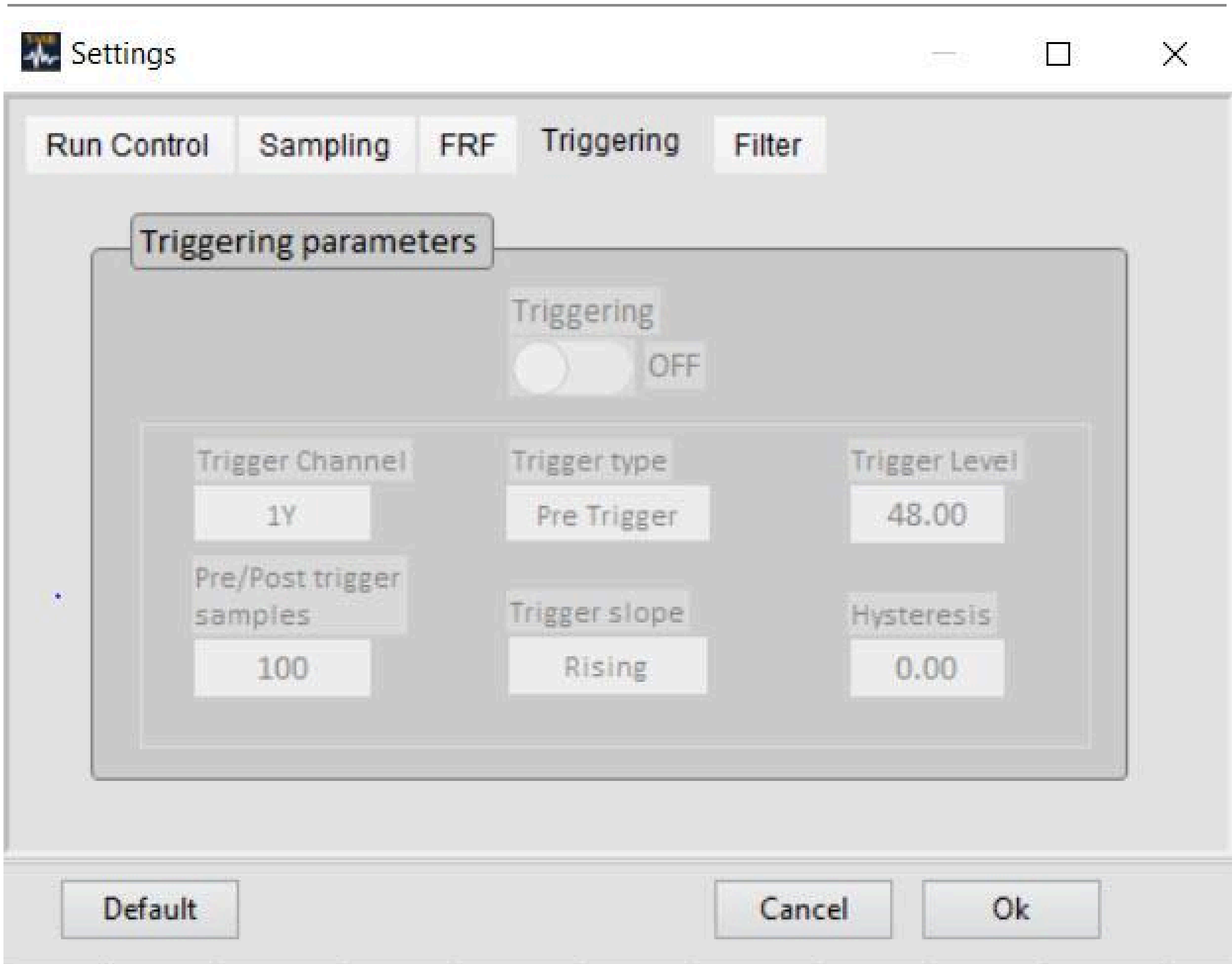
5.Window: Select the required windowing function.For Example: Force exponential is the preferred windowing function for Impact test.



6. Force Window (%) & Exponential Window (%) : This allows the user to set the force- exponential windowing function as a percentage of time.



Triggering

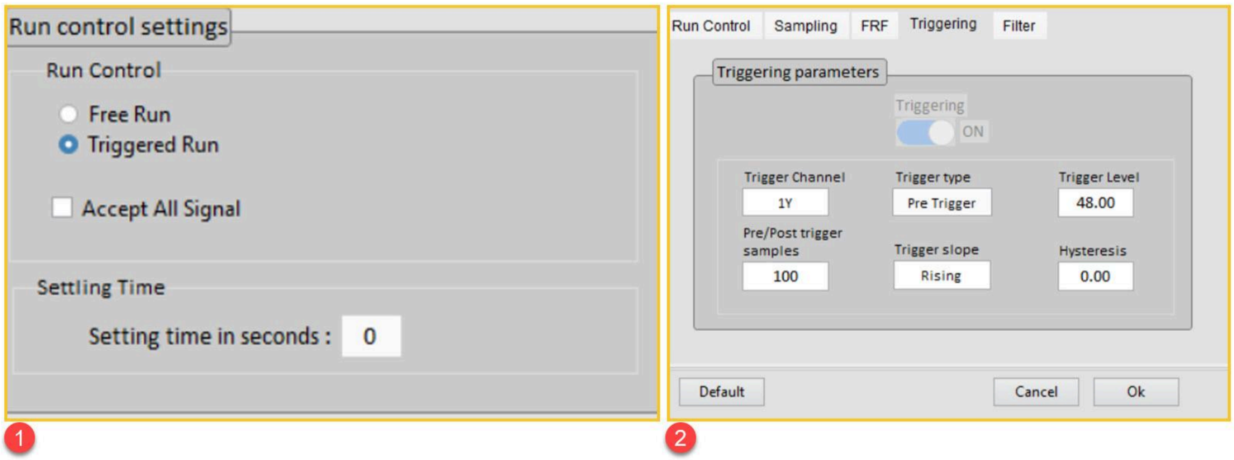


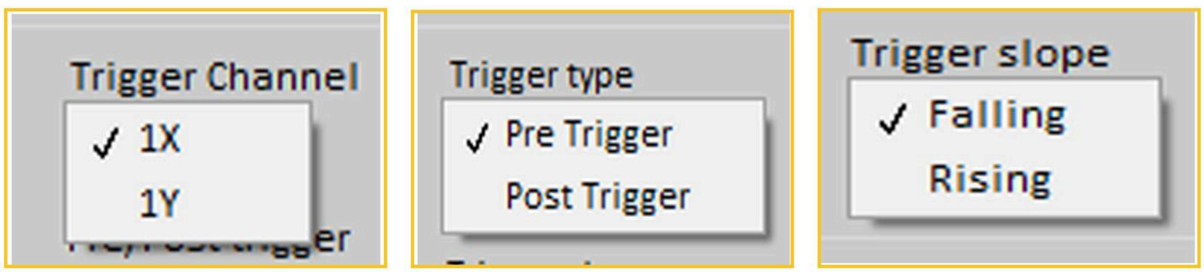
Triggering is used to initiate measurement only when a specified condition in the input signal occurs.

Signals are monitored and presented at random periods in the absence of triggering.

In default case triggering is disabled.

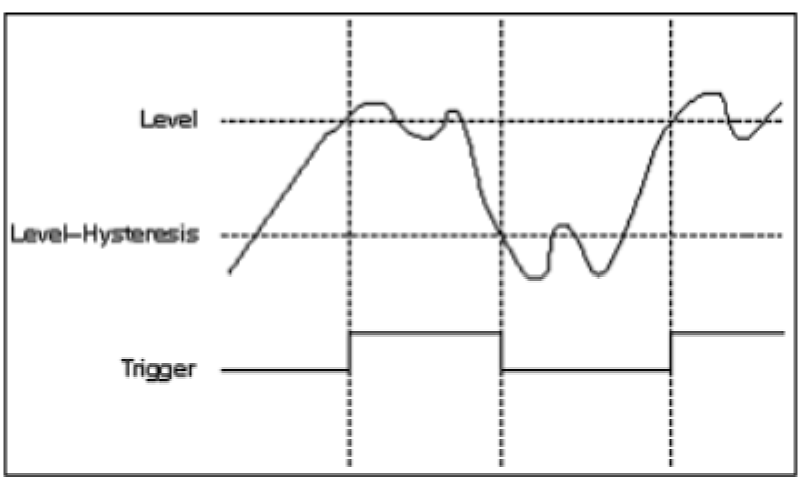
When it is enabled we can choose different channels to setup trigger, type of triggers like **Pre-Trigger** or **Post-Trigger**, **Triggering level** (amplitude value at which trigger occur), **Number of Samples** to read in each trigger type (pre-trigger samples refers the samples to be acquired before the trigger has occurred & post trigger samples would refer to the samples after which the data is to be acquired after the trigger), **Trigger slope** (Rising or Falling) and **Hysteresis**.





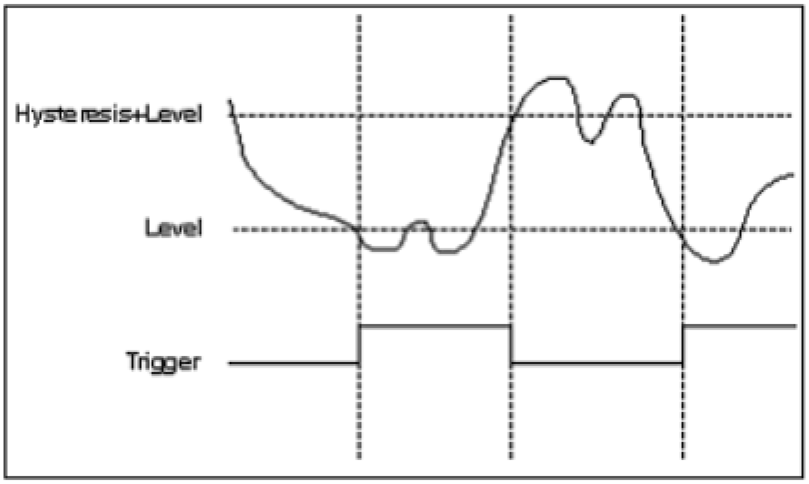
The **Hysteresis** defines the distance between the firing level and the arming level and determines the sensitivity of the trigger system.

It is often used to reduce false triggering in signals caused by noise or jitter. When hysteresis is used at a rising slope, the trigger occurs when starting under Level-Hysteresis and crossing over Level. The trigger is released when the signal passes below the Level-Hysteresis.



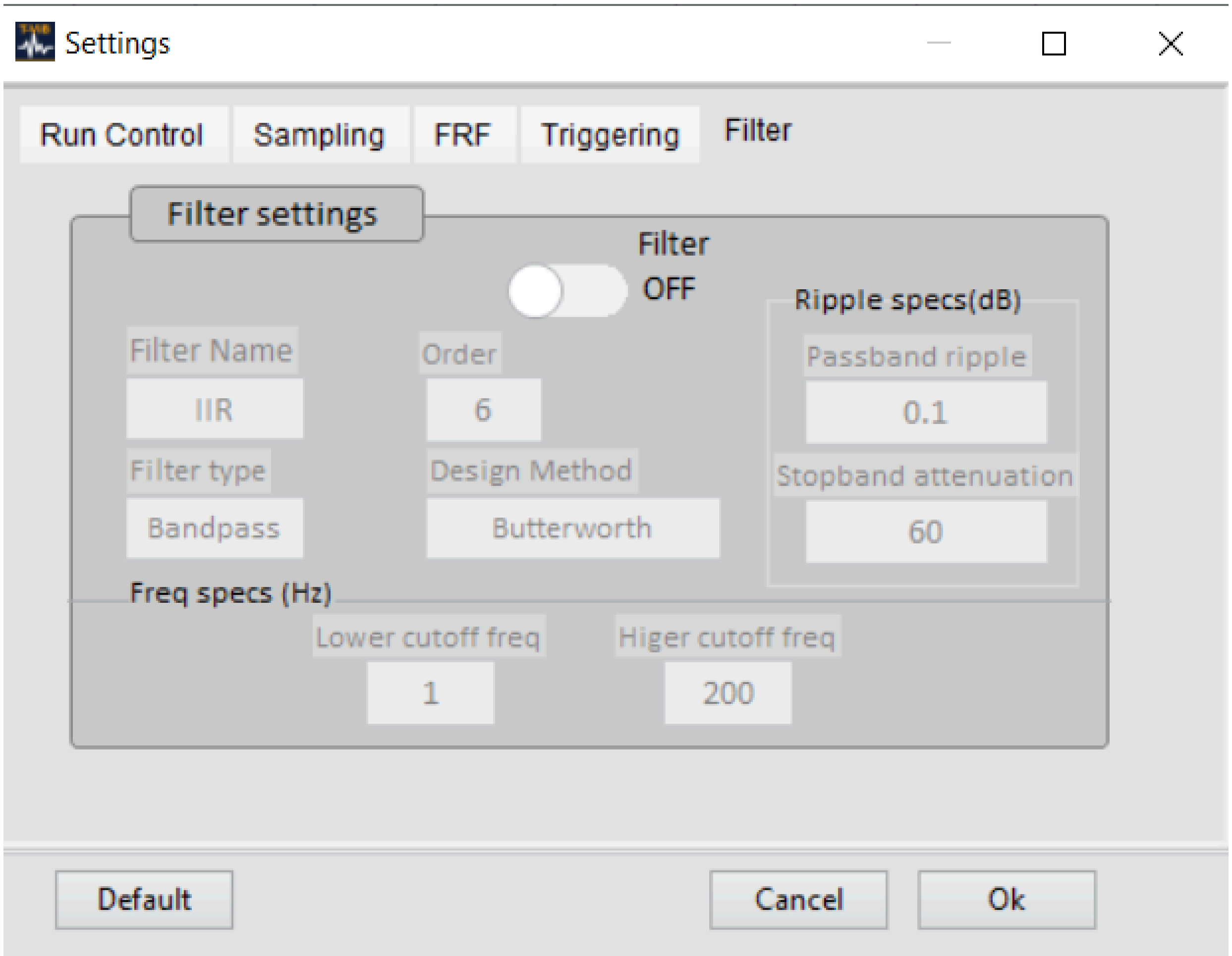
Hysteresis with rising slope

When hysteresis is used at a falling slope, the trigger occurs when starting above Level+Hysteresis and crossing under Level. The trigger is released when the signal passes above the Level+Hysteresis.

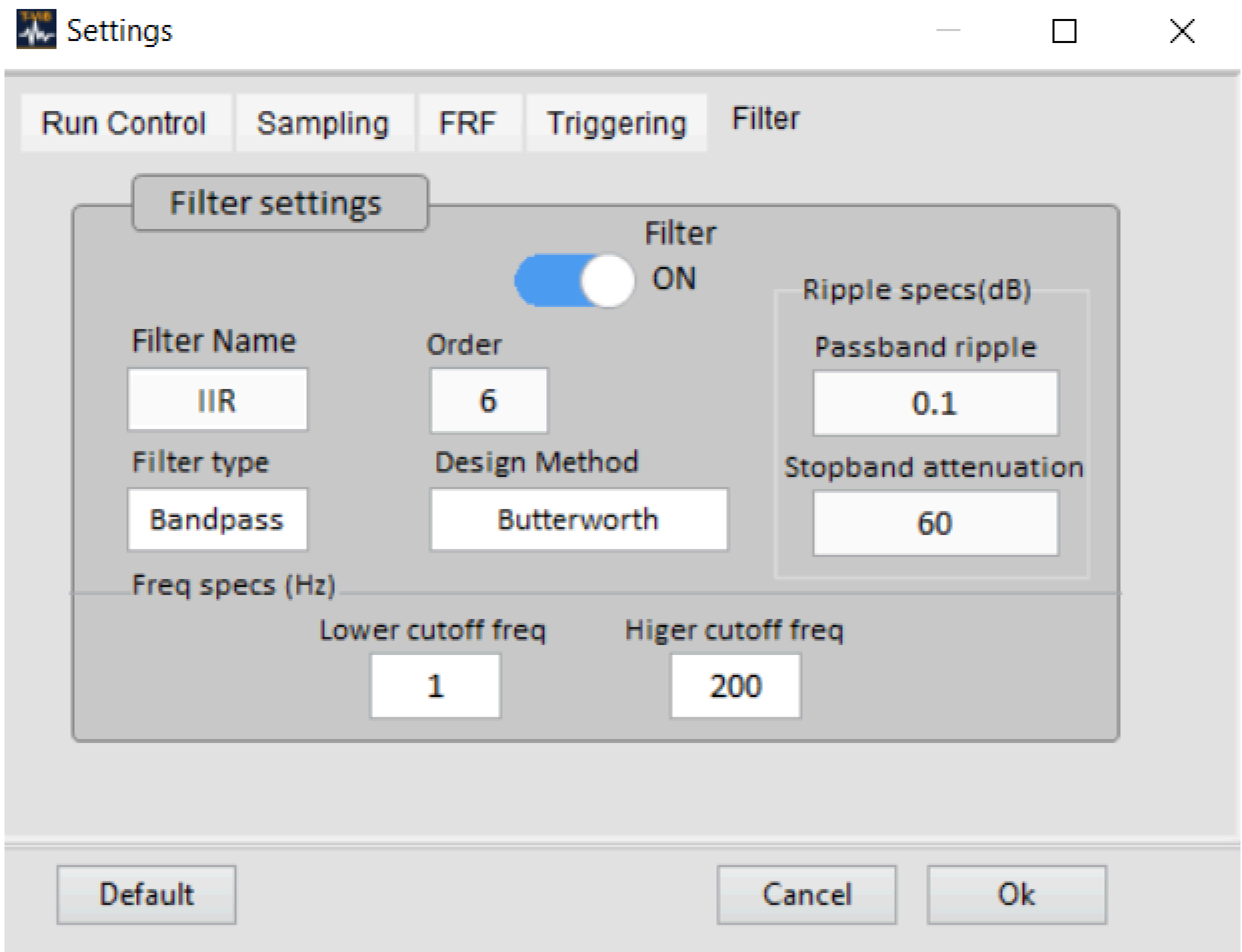


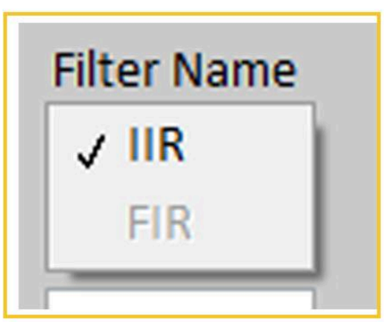
Hysteresis with falling slope

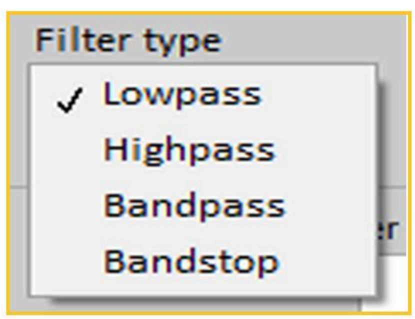
Filter

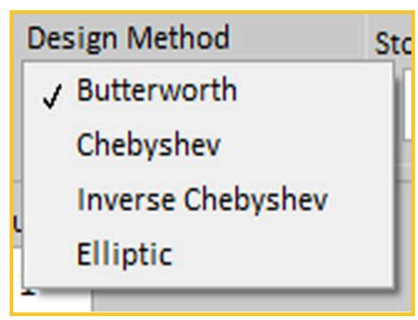


In default case the filtering are disabled. Here the available filter mode is IIR (Infinite Impulse Response). Infinite impulse response (IIR) filters, also known as recursive filters, operate on current and past input values and current and past output values.









The various modes of filters available are as follows:

Lowpass (LP): It is used to allow only low frequencies to pass. It is frequently used to isolate the high frequencies of the . It is widely used to suppress high-pitched sounds in speakers, some electric guitars, and radio transmitters.

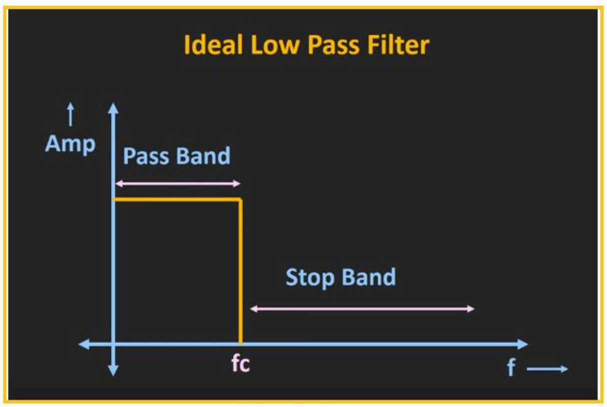
Highpass (HP): A high pass filter is used to ensure that only high frequencies of the signals are allowed for operation. It cancels out the majority of low frequencies and DC components.

Bandpass (BP): A band pass filter is a filter that combines the characteristics of a high pass and a low pass filter. It is constructed in such a manner that the LPF's cutoff frequency is greater than the HPF's cutoff frequency, enabling only a limited range of frequencies to pass through.

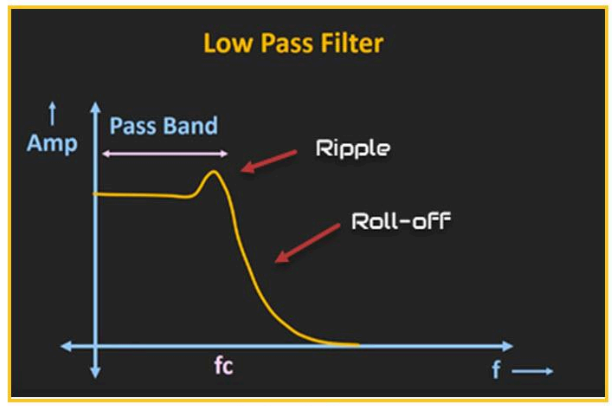
Bandstop (BS): It is also known as a notch filter,and is a type of filter that is used to block some frequencies of the signals while allowing others to pass through. It is the inverse of a band pass filter and may well be made by utilizing the same input with a high pass filter and a low pass filter.

Design Method

In the case of a filter, it is nearly difficult to obtain an Ideal filter's frequency response, and to overcome this limitation, several filter approximations, referred to here as Design methods, are adopted.



Ideal Case



Actual Case

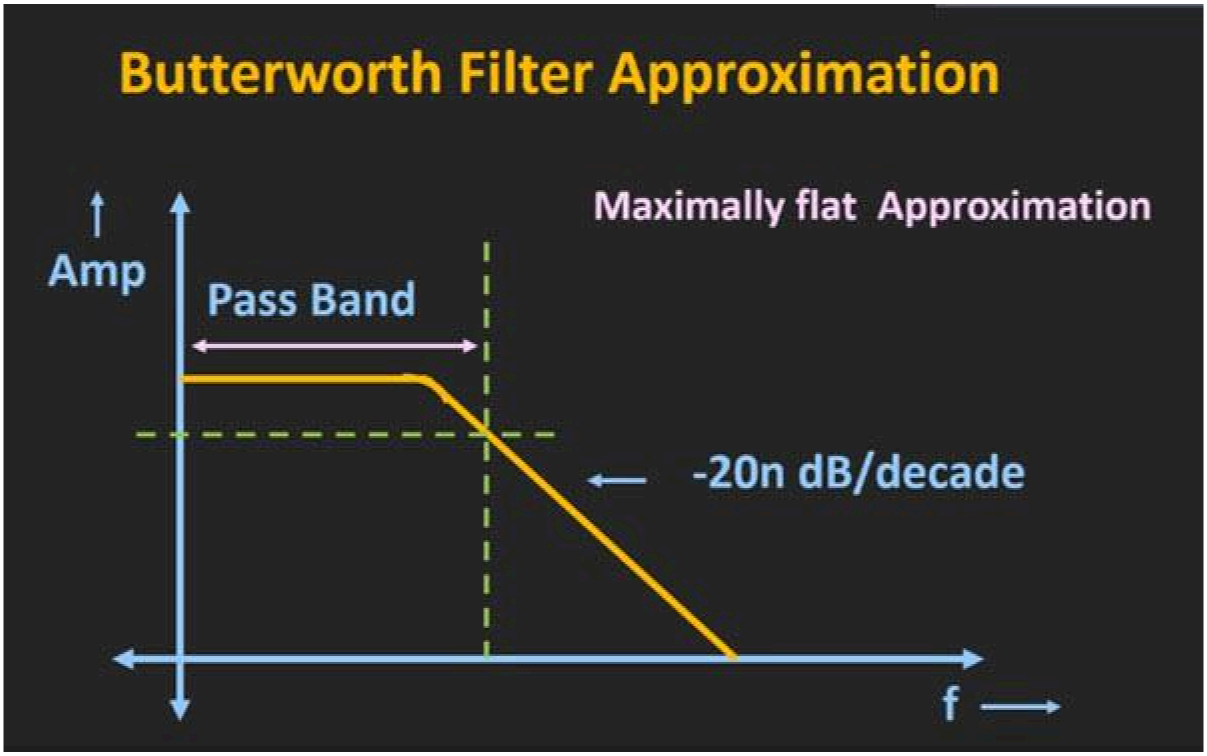
We can understand the different filter approximations by taking the example of Ideal low pass filter. The ideal low pass filter passes only the low frequency components in the input signal and it rejects all the frequency components beyond some cutoff frequency.

The frequency band which is being supported by this filter is known as the **Pass Band** and the frequencies which is being rejected by this filter is known as the **Stop Band**.

And in this Pass Band the frequency response is very flat. While above this cutoff frequency, it rejects all the frequencies and it provides the infinite attenuation for those frequencies. So there is a vertical transition between this Pass Band and the Stop Band. And such kind of response is known as the brick wall response.

But whenever we are designing the practical filters, it is almost impossible to achieve this vertical transition from the Pass Band to Stop Band. Hence the transition from the Pass Band to Stop Band will be always gradual and we use different design methods.

Butterworth: It exhibits a nearly flat Pass Band with no ripple. The roll-off is smooth and monotonic, with a low-pass or high-pass roll-off rate of 20 dB/decade (6 dB/octave) for every pole. This approximation is also known as the maximal flat approximation.

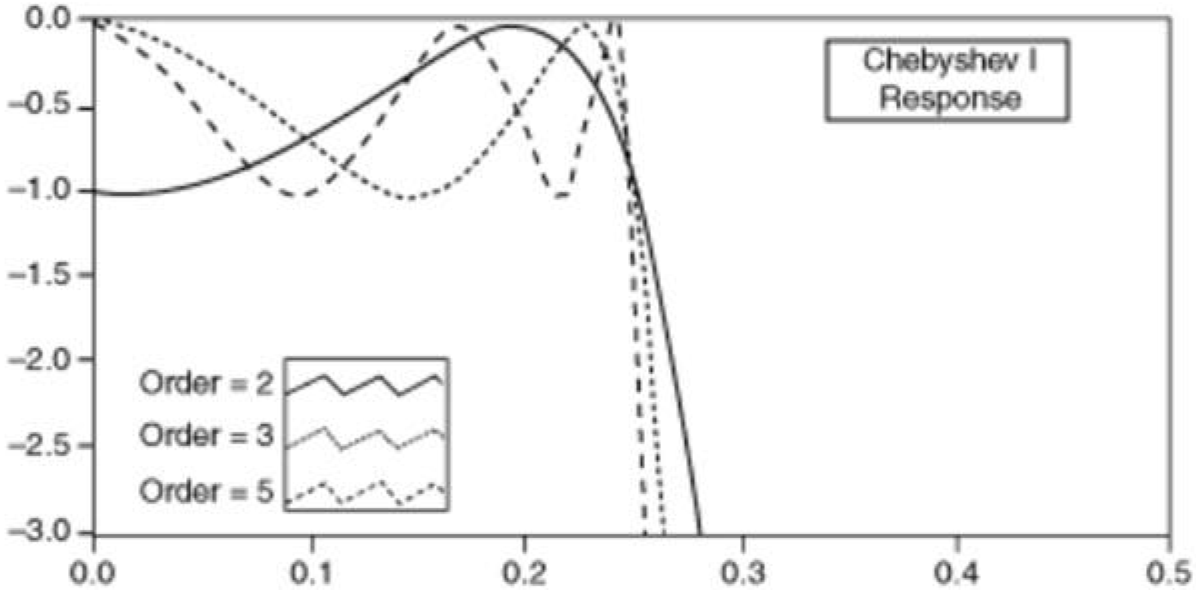


Suppose if you are designing the Butterworth filter of a second **order** then it will have a roll-off of 40dB per decade. Similarly, designing a fifth **order** will have a roll-off of at the rate of 100dB per decade.Also in this design you will not see any kind of ripple in the stop band as well.

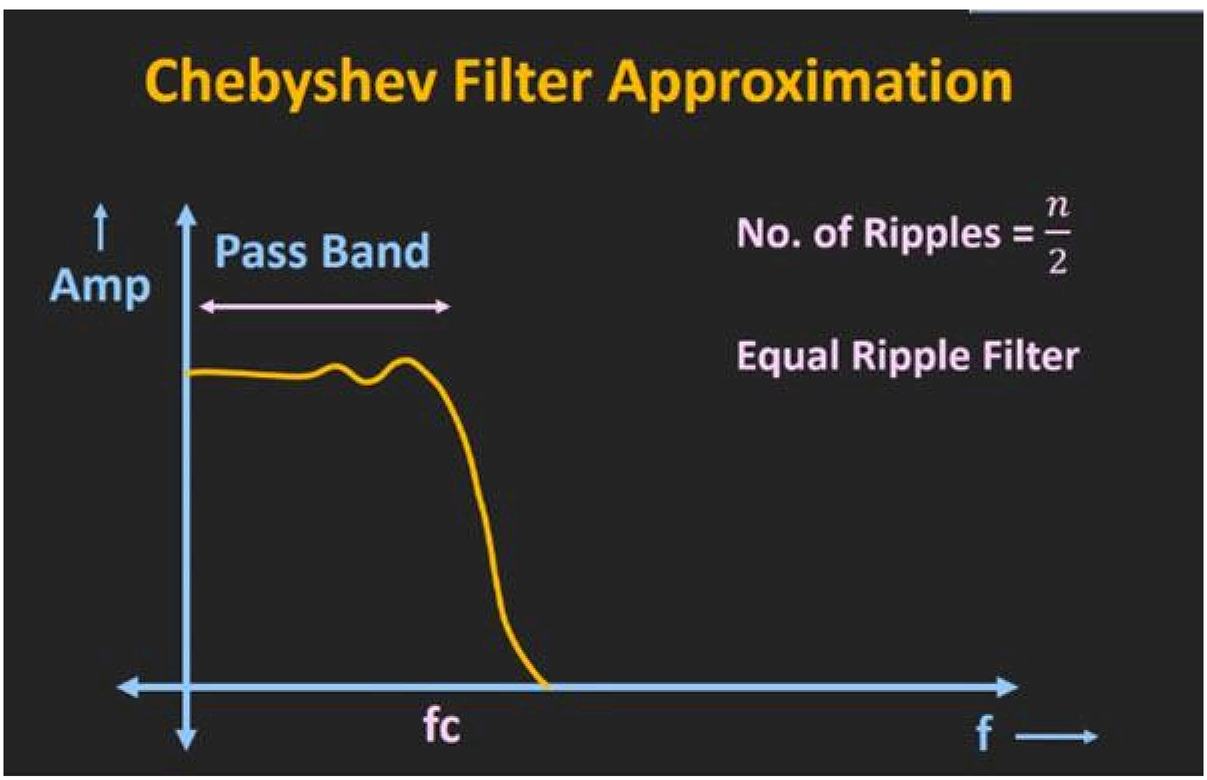
This filter design is quite common, particularly in audio processing.

Chebyshev: This design is a mathematical strategy for achieving a faster roll-off by allowing ripple in the Pass Band. As the ripple increases, the roll-off becomes sharper. In Stop Band you will not see any ripples. The Chebyshev response is an optimal trade-off between these two parameters.

If maintaining a flat response throughout your Pass Band is not a concern, and if you want a rapid transition, this design filter can be used. (Compared to a Butterworth filter, a Chebyshev filter can achieve a sharper transition between the Pass Band and the Stop Band with a lower order filter. The sharp transition between the Pass Band and the Stop Band of a Chebyshev filter produces smaller absolute errors and faster execution speeds than a Butterworth filter).



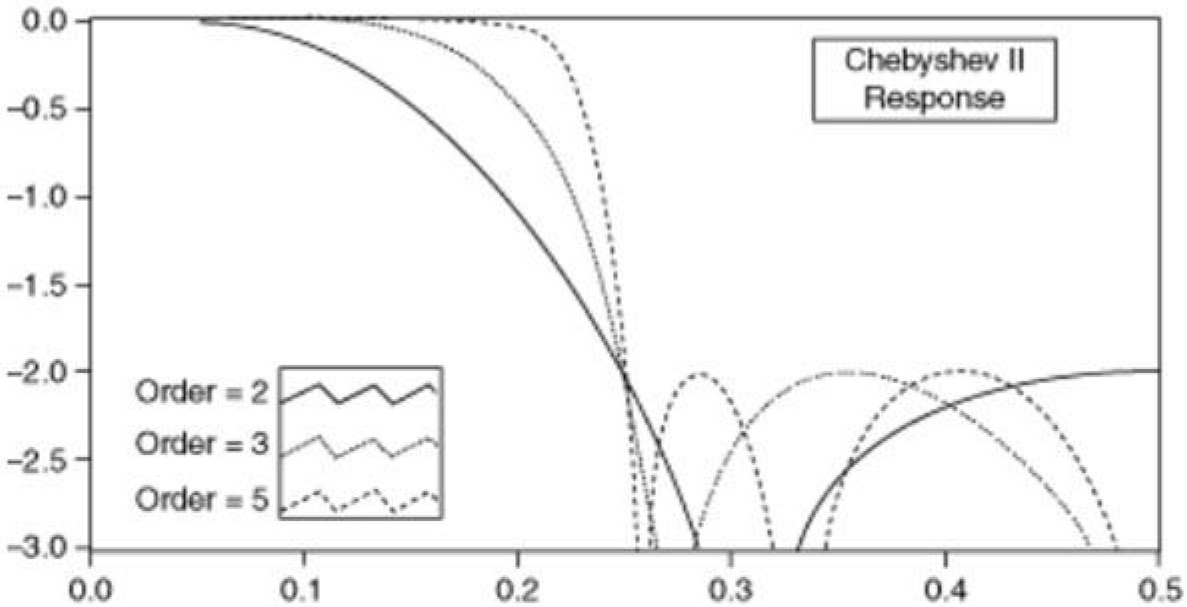
In case of this Chebyshev filter design, the number of ripples in the Pass Band will be equal to the **n** divided by 2, where **n** is the **order** of the filter. For example the fourth order will have 2 ripples.



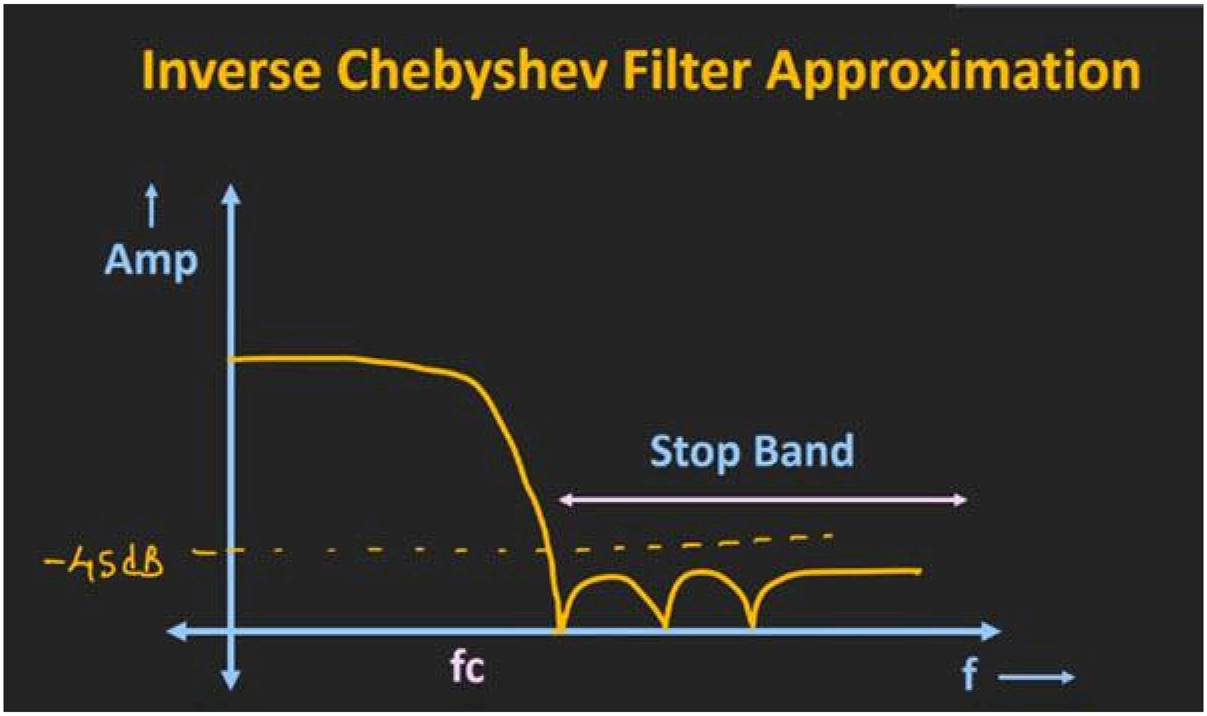
Also we need to provide the maximum amount of **ripple(Passband ripple)** which is acceptable by this filter design. In most cases, this **ripple(Passband ripple)** is in the range of 0.1dB up to 3dB. The amplitude of these ripples is same for all the ripples. Hence it is also known as the equal ripple filter.

Inverse Chebyshev: This design filter is having a flat Pass Band and it is having ripples in the Stop Band. And the transition or roll-off in this inverse Chebyshev filter is same as the Chebyshev filter.

So whenever in your application you require the flat frequency response in the Pass Band, as well as you required the faster transition and you are not bothered about the any kind of ripple at Stop Band then these case is used.

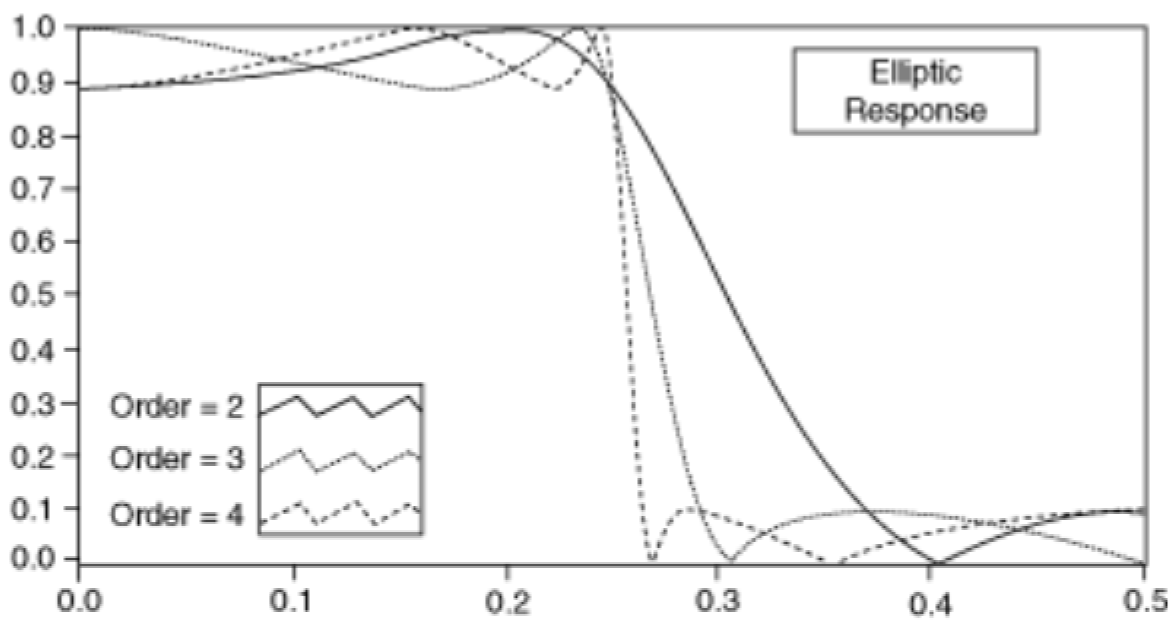


But whenever you have **ripple** in the Stop Band then you also need to provide the amount of attenuation**(Stopband attenuation)** because it is quite possible that the amplitude of the ripple might touch that value of a Stop Band rejection.

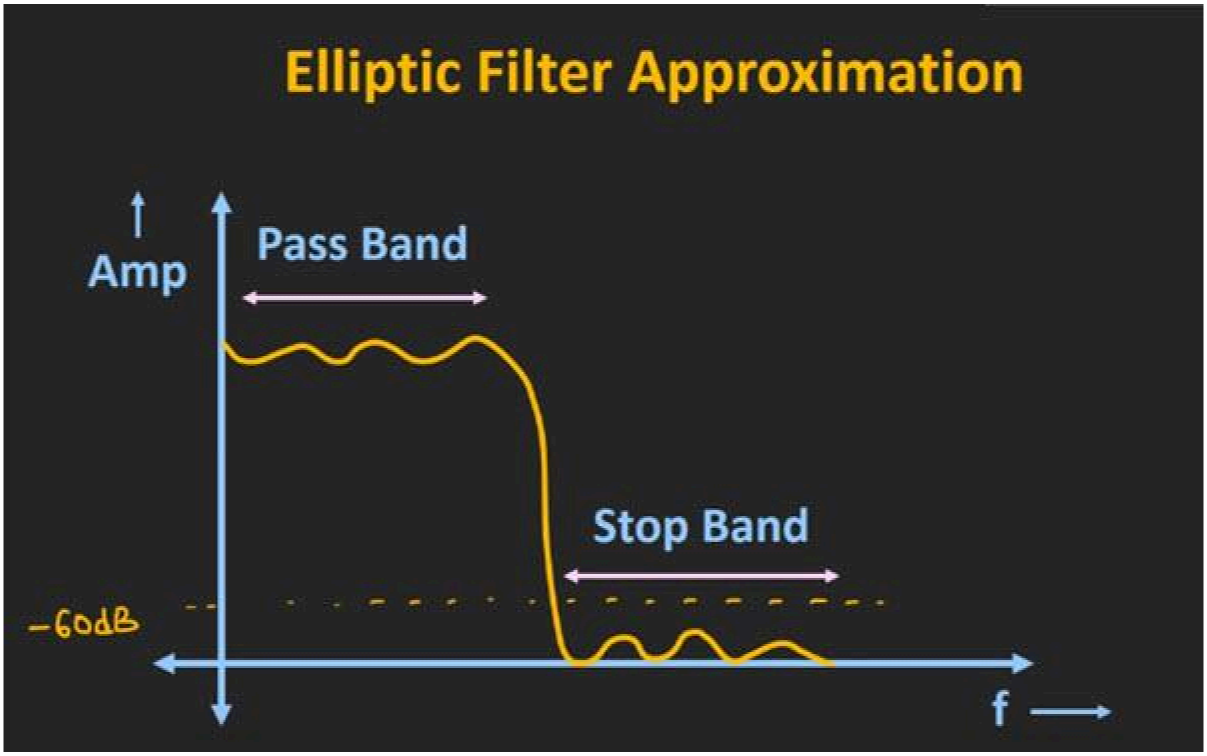


In this case the filter is providing the **attenuation** in the stop band at minus 45 dB. So we need to make sure that the amplitude of the ripple is always less than this value.

Elliptic: The cut-off slope of an elliptic filter is steeper than that of a Butterworth, Chebyshev, or Bessel, but the amplitude response has ripple in both the Pass Band and the Stop Band, and the phase response is very non-linear.



However, if the primary concern is to pass frequencies falling within a certain frequency band and reject frequencies outside that band, regardless of phase shifts or ringing, the elliptic response will perform that function with the lowest-order filter.



Whenever in application the ripple in the Pass Band and Stop Band is not a concern and you require the fastest possible transition then in such cases you can use this design.

Notification Bar

The notification bar lets you know the current status of the software.

1. **Ready** indicates the software is ready for an acquisition. To begin the measurement press **RUN** button.

2. **Initializing** indicates the measurement has started on pressing the **RUN**.

3.**Settling** indicates that the acquisition system is waiting for the sensors to settle. All sensors require a settling time before the measurement starts. The settling time can be input in the settings.

4. **Output Ready** indicates that the acquisition is on going and the real time measurement is being displayed in the graphs in the tiles.

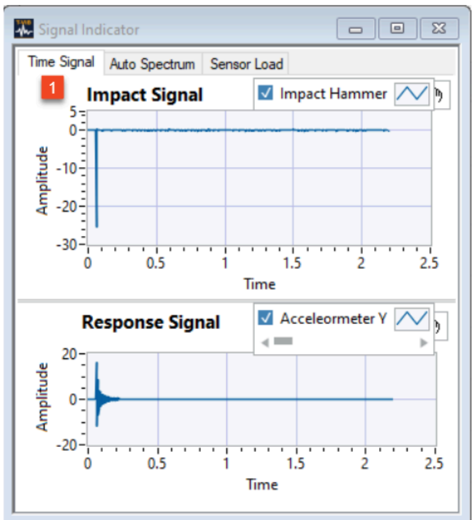
5.**Stopped** indicates that the acquisition has been stopped on pressing the stop button.

Signal Indicator

To view the signal characteristics during triggered acquisition, click the signal Indicator button on the function bar. The **Signal Indicator** tile pops up which allows the user to view the time signal, auto spectrum & Sensor Load.

1. Time Signal

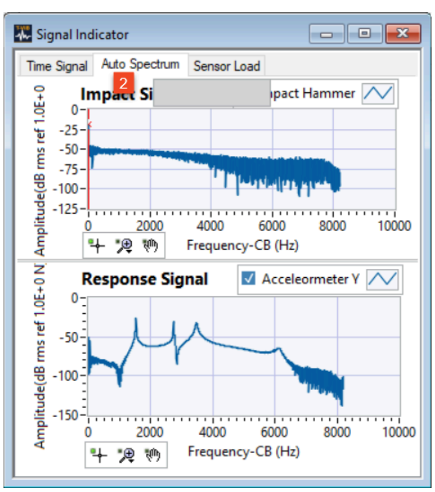
To view the data of the stimulus and response data w.r.t time, choose the **Time Signal** Tab.



2. Auto Spectrum

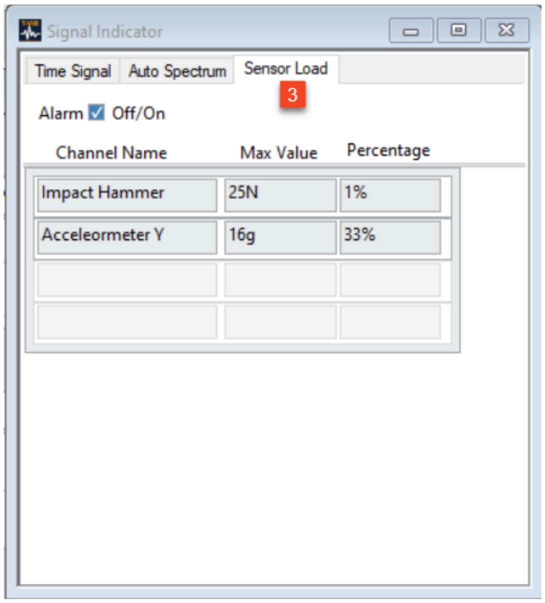
To view the data w.r.t frequency choose the **Auto Spectrum** Tab,

An Auto Spectrum (Power Spectrum) is a spectral display of the power (acceleration squared) at each frequency. Phase is ignored.



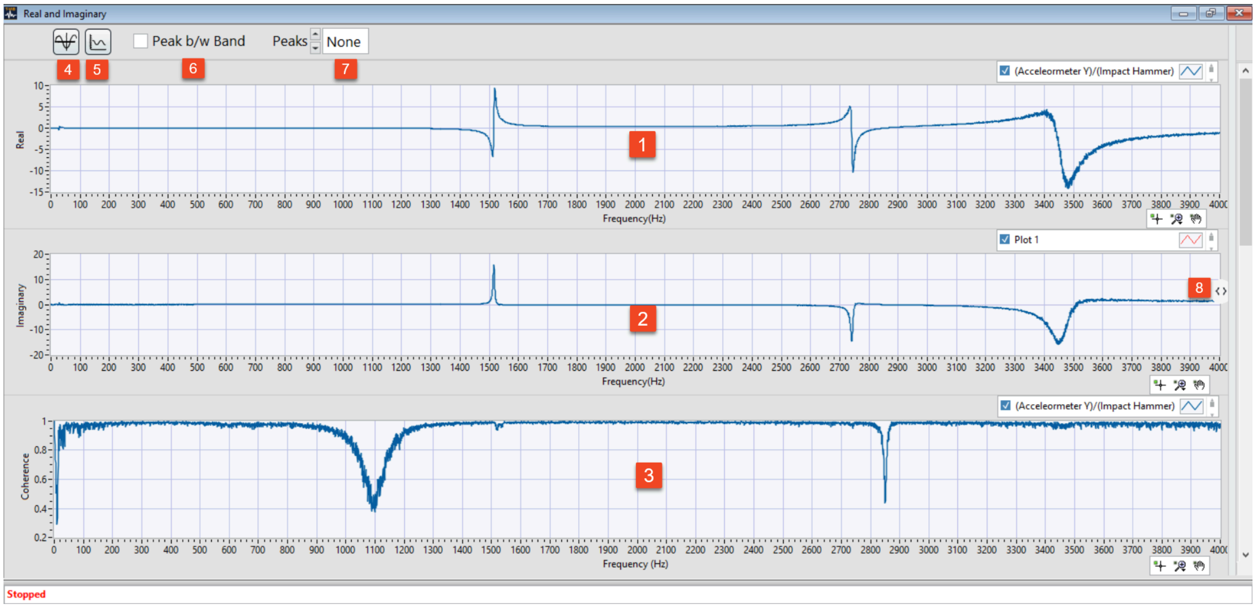
3. Sensor Load

The percentage load on the sensor can be viewed on the **Sensor Load** tab. The user can configure the Alarm to switch on/off. If the alarm is switched **On**, when the sensor load crosses 100% the alarm will be triggered.



Real and Imaginary

The Real and Imaginary tile gives the real,imaginary & coherence plots

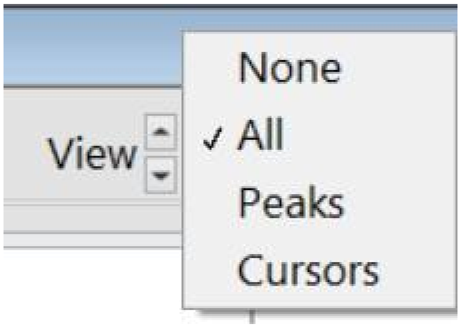


1. Real Plot
2. Imaginary Plot
3. Coherence
4. Info Tab
5. XY Cursor
6. Band Cursor
7. Peak between Bands
8. Peak Selection

Plots: [1] Real , [2] Imaginary & [3] Coherence

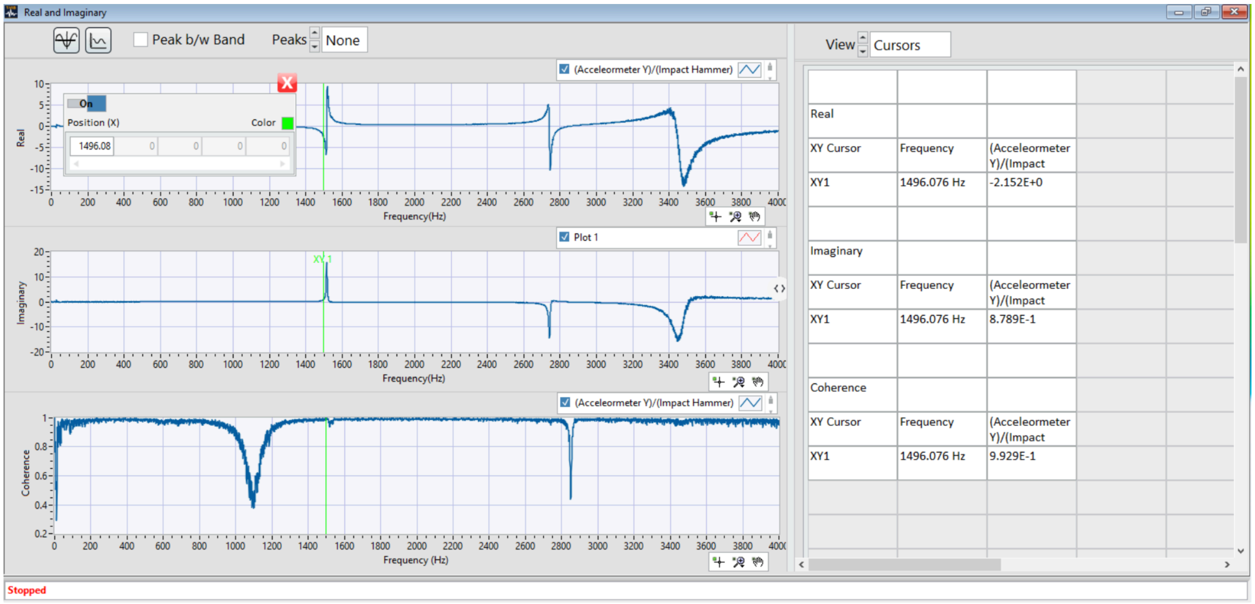
The Plots give the Real ,Imaginary & Coherence values w.r.t frequency.

4. Info Tab



Info tab displays the information regarding the peaks, Cursor, Units of the Amplitude of the Plots. Choose the **View** Selection to configure which data to be displayed.

5. XY Cursor



The **slide button** allows the user to switch ON/OFF the cursors**.**

The **Position (X)** allows the user to input the position of the cursor which places the cursor to the input position. For example: In the above picture the 0 value has been replaced with 1496.08 which is the position of the first cursor. For more cursors the user can input the corresponding position values.

To change the color of the cursor click the **Color** button and choose the appropriate color.

Once configured the user can press the **close** button.

6. Band Cursor

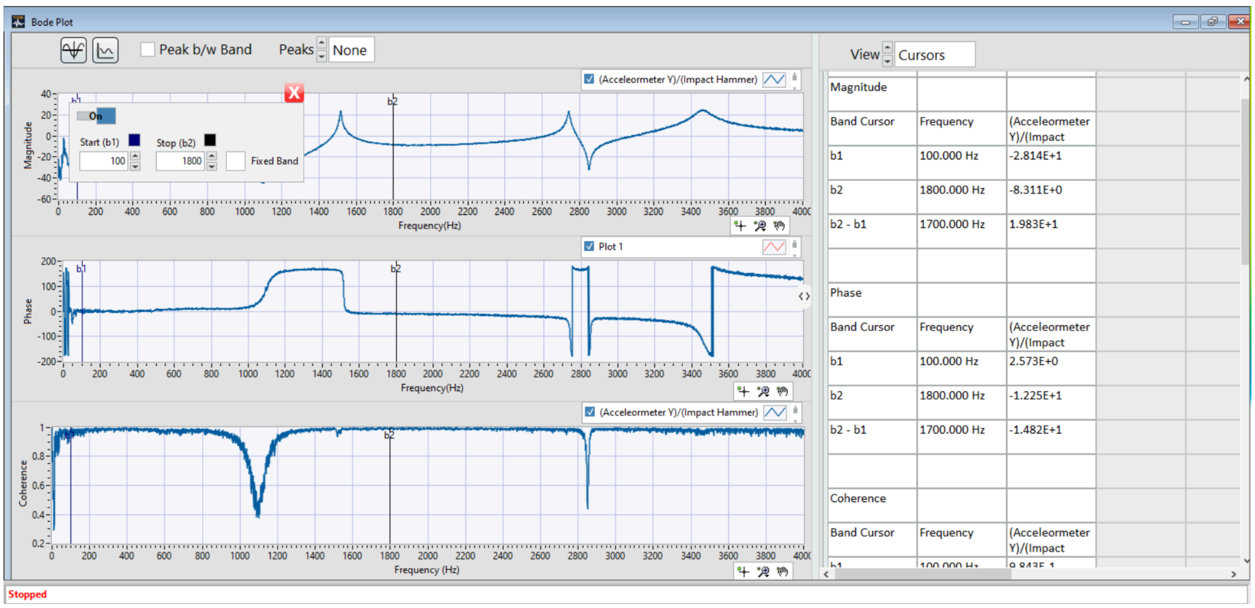
The **slide button** allows the user to switch ON/OFF the cursors**.** The functionality of band cursor is to determine the period/ width of a band, for example this feature allows the user to check the period of an oscillating signal (the period between the cursor is displayed in the **Info tab**).

The **Start(b1)** and **Stop(b2)** are the two cursors available in this feature. The user can input the position of the cursors.

The **Fixed band** allows the user to fix the difference between the band, on moving any of the cursor the period or the width between the cursors would be preserved.

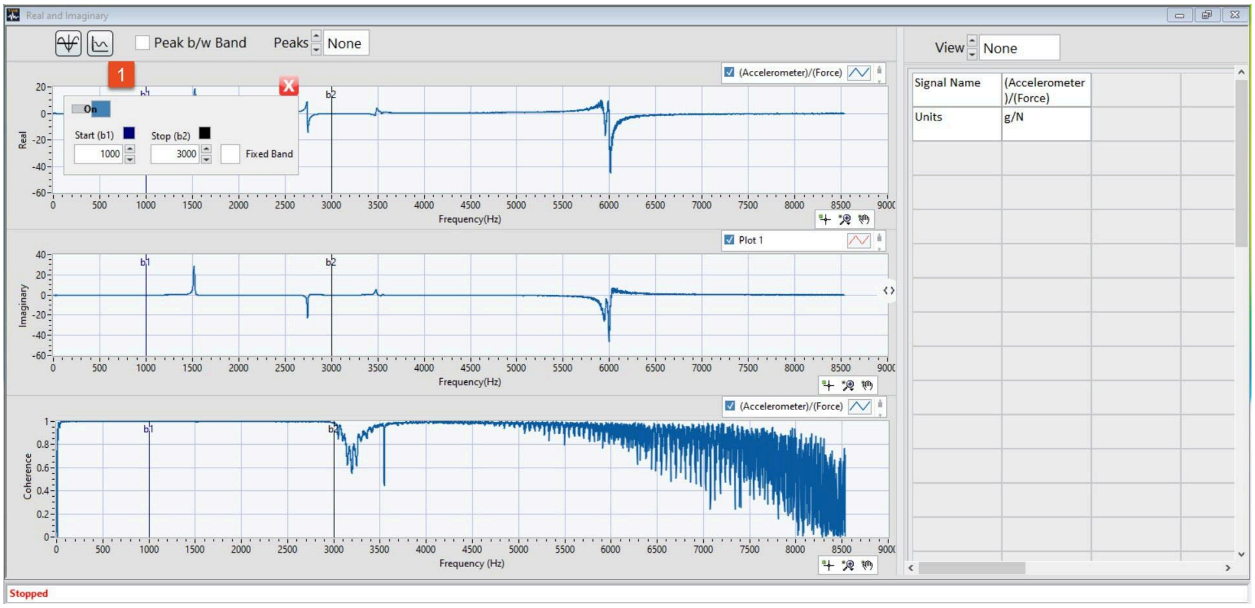
To change the color of the cursor click the **Color** button and choose the appropriate color.

Once configured the user can press the **close** button .



7. Peak between Bands

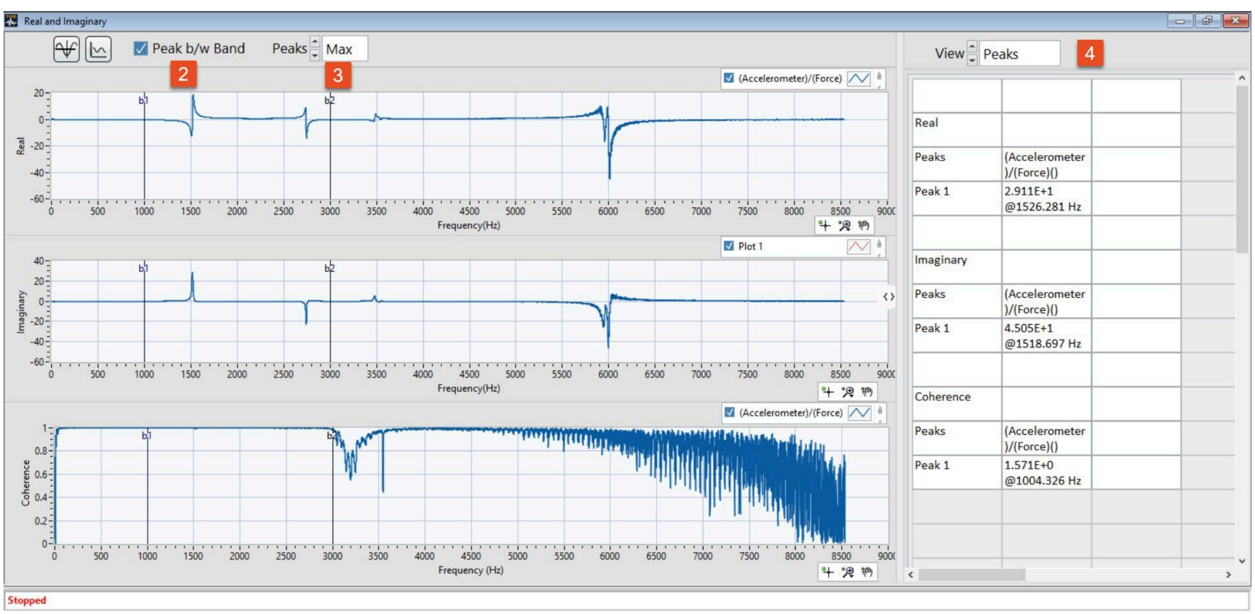
1. Configure **Band** Cursor.



2. Select the **Peak b/w Bands** to be switched on.

3. Configure the **Peaks** Selection to the number of required peaks.

4. Choose the **View** selection to All (if the cursor values to be displayed in the Info Tab) /Peaks (if only the peak values to be displayed in the Info Tab).



8. Peak Selection

The Peak Selection allows the user to view the Max Amplitude values & its frequency.

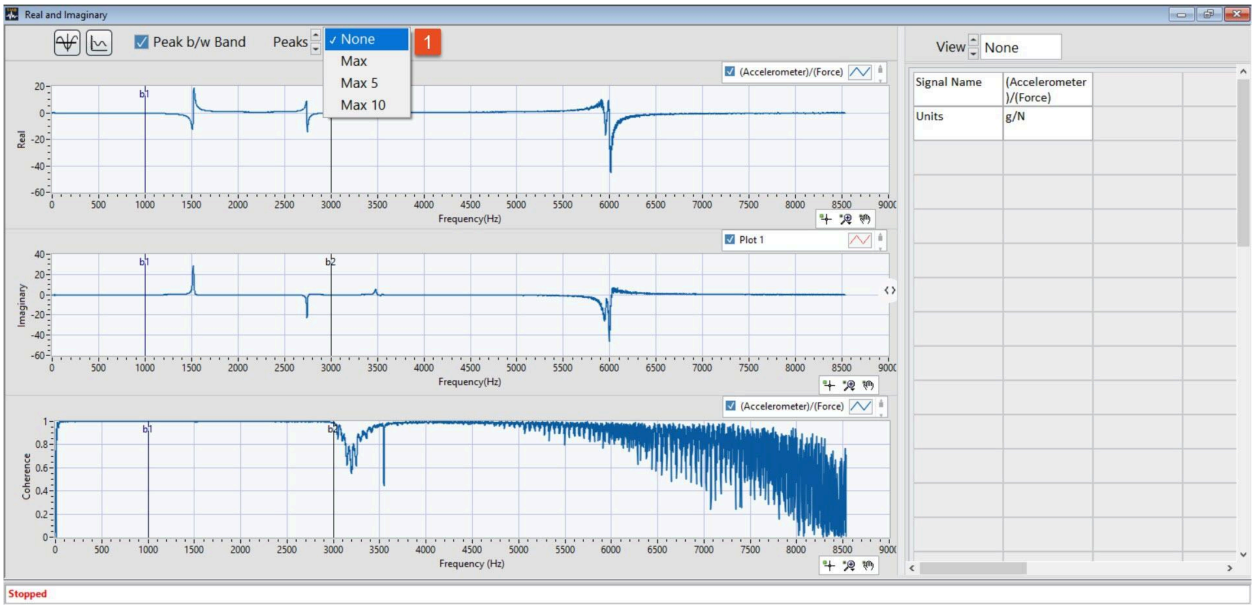
1. The user can choose the **Peak** Selection from below

None: No peaks will be displayed. Default setting.

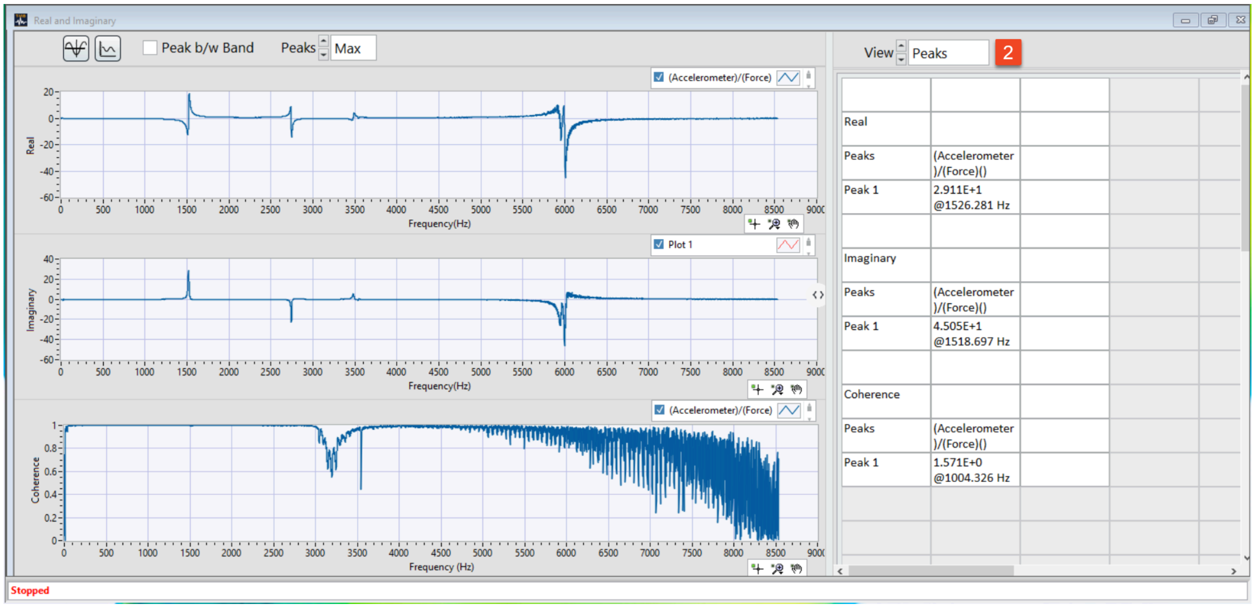
Max: To display the Maximum Amplitude value & its frequency.

Max 5: To display the Maximum Amplitude value of 5 peaks & its frequency.

Max 10 :To display the Maximum Amplitude value of 10 peaks & its frequency.

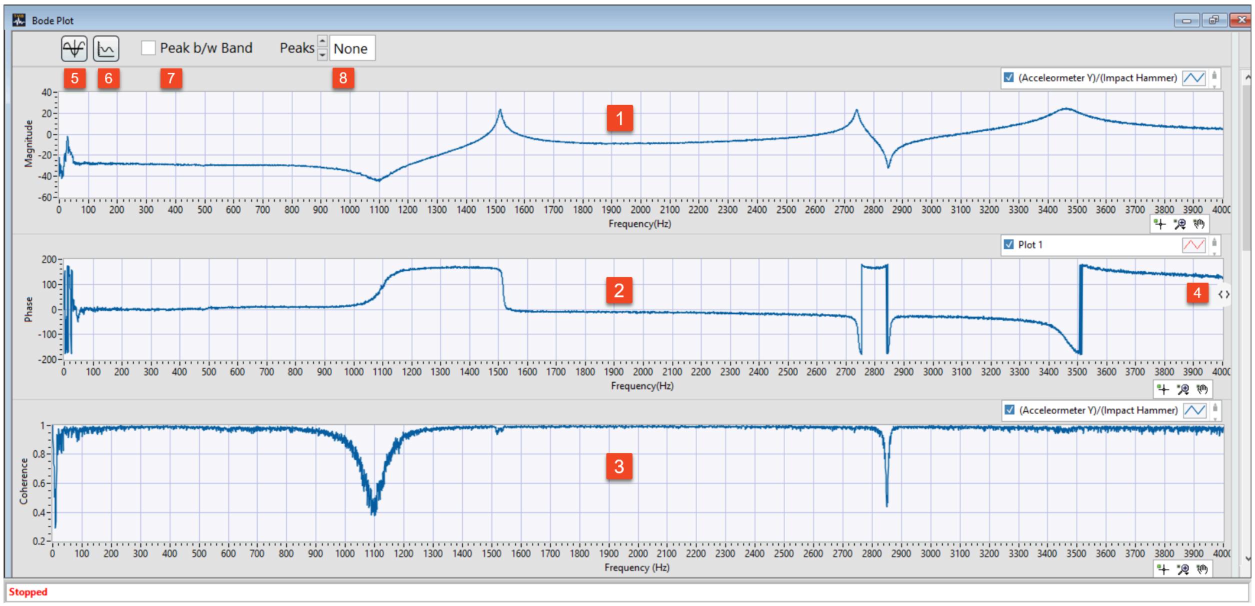


2. Choose the **View** Selection to All/ Peaks for the Info Tab to display the peak information.


Real and Imaginary
Peak b/w Band
Peaks
Max
View
Peaks
2
(Accelerometer)/(Force)
Real
Frequency(Hz)
Plot 1
Imaginary
Frequency(Hz)
(Accelerometer)/(Force)
Coherence
Frequency (Hz)
Real
Peaks
(Accelerometer)/(Force)()
Peak 1
2.911E+1 @1526.281 Hz
Imaginary
Peaks
(Accelerometer)/(Force)()
Peak 1
4.505E+1 @1518.697 Hz
Coherence
Peaks
(Accelerometer)/(Force)()
Peak 1
1.571E+0 @1004.326 Hz
Stopped

Bode Plot

The Bode plot tile gives the magnitude, phase & coherence plots



1. Magnitude

2. Phase

3. Coherence

4. Info Tab

5. XY Cursor

6. Band Cursor

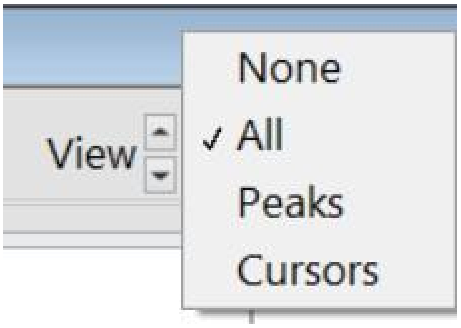
7. Peak between Bands

8. Peak Selection

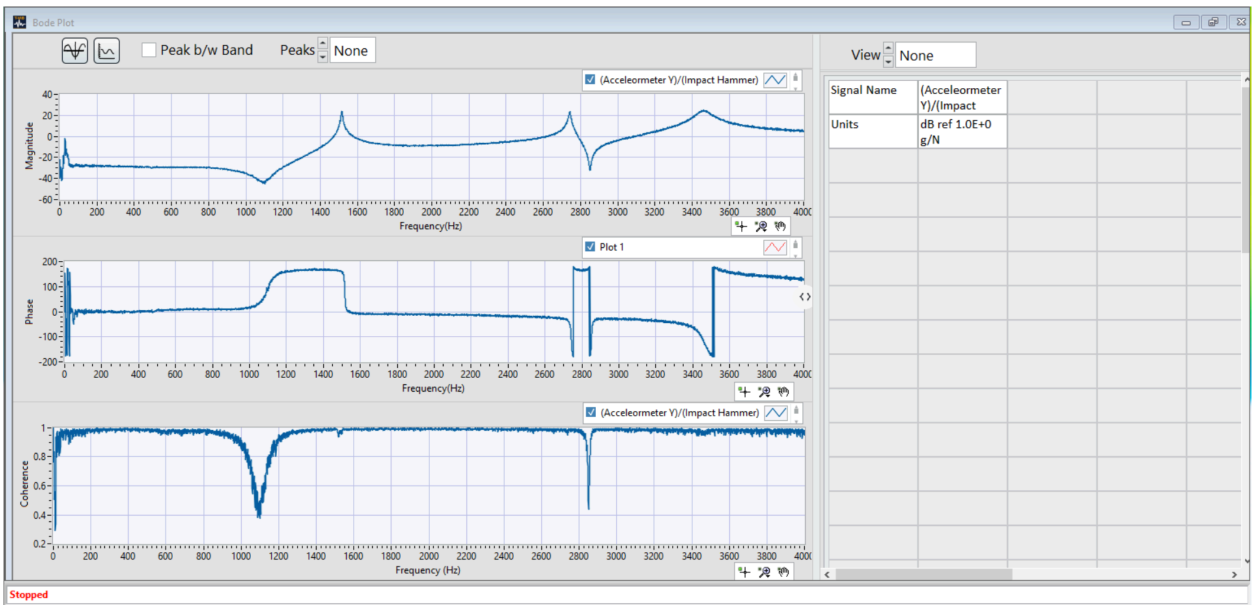
Plots: [1] Magnitude [2] Phase [3] Coherence

The Plots give the Real ,Imaginary & Coherence values w.r.t frequency.

4. Info Tab



Info tab displays the information regarding the peaks, Cursor, Units of the Amplitude of the Plots. Choose the **View** Selection to configure which data to be displayed.



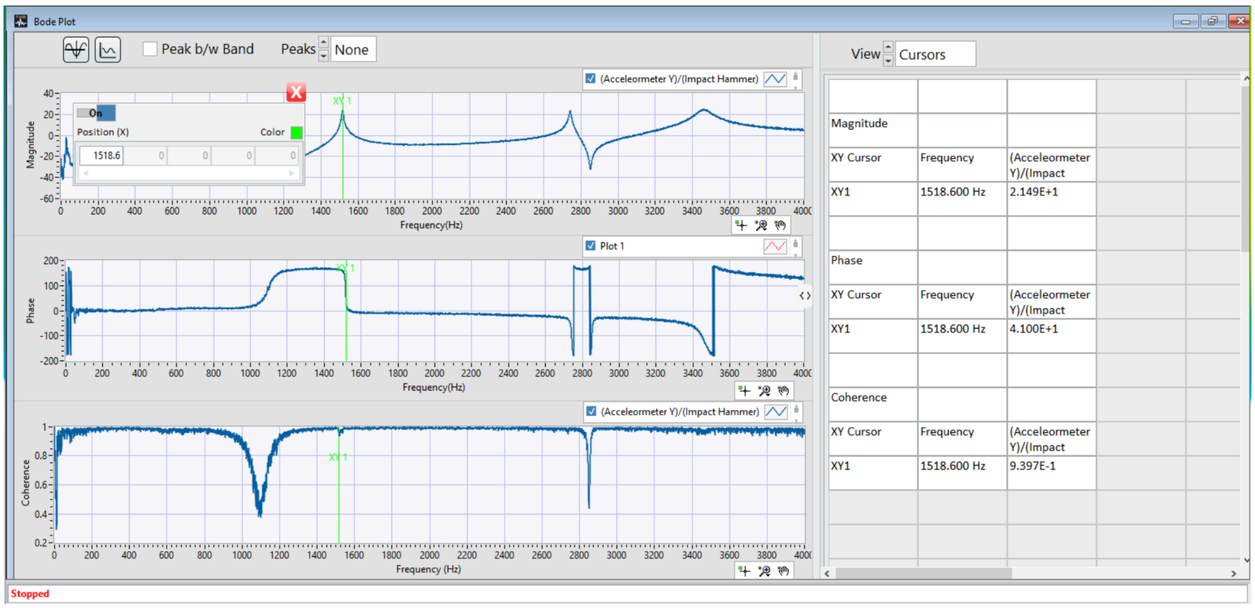
5. XY Cursor

The **slide button** allows the user to switch ON/OFF the cursors**.**

The **Position (X)** allows the user to input the position of the cursor which places the cursor to the input position. For example: In the above picture the 0 value has been replaced with 1496.08 which is the position of the first cursor. For more cursors the user can input the corresponding position values.

To change the color of the cursor click the **Color** button and choose the appropriate color.

Once configured the user can press the **close** button.



6. Band Cursor

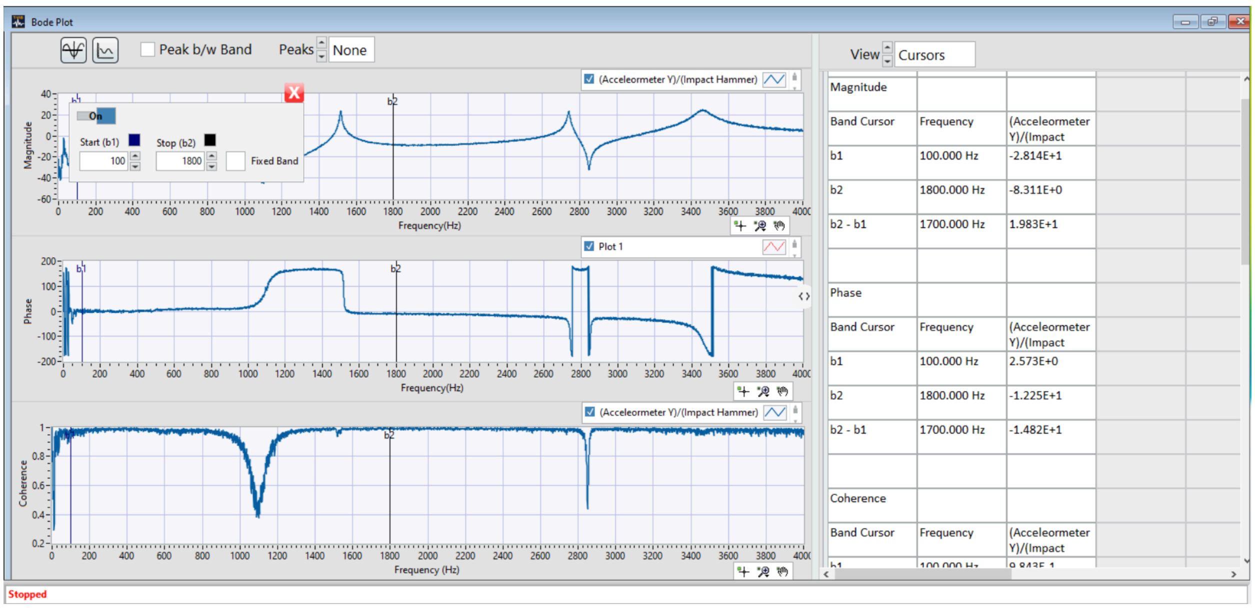
The **slide button** allows the user to switch ON/OFF the cursors**.** The functionality of band cursor is to determine the period/ width of a band, for example this feature allows the user to check the period of an oscillating signal (the period between the cursor is displayed in the **Info tab**).

The **Start(b1)** and **Stop(b2)** are the two cursors available in this feature. The user can input the position of the cursors.

The **Fixed band** allows the user to fix the difference between the band, on moving any of the cursor the period or the width between the cursors would be preserved.

To change the color of the cursor click the **Color** button and choose the appropriate color.

Once configured the user can press the **close** button .



7. Peak between bands

1. Configure **Band** Cursor.

2. Select the **Peak b/w Bands** to be switched on.

3. Configure the **Peaks** Selection to the number of required peaks.

4. Choose the **View** selection to All (if the cursor values to be displayed in the Info Tab) /Peaks (if only the peak values to be displayed in the Info Tab).

8. Peak Selection

The Peak Selection allows the user to view the Max Amplitude values & its frequency.

1. The user can choose the **Peak** Selection from below

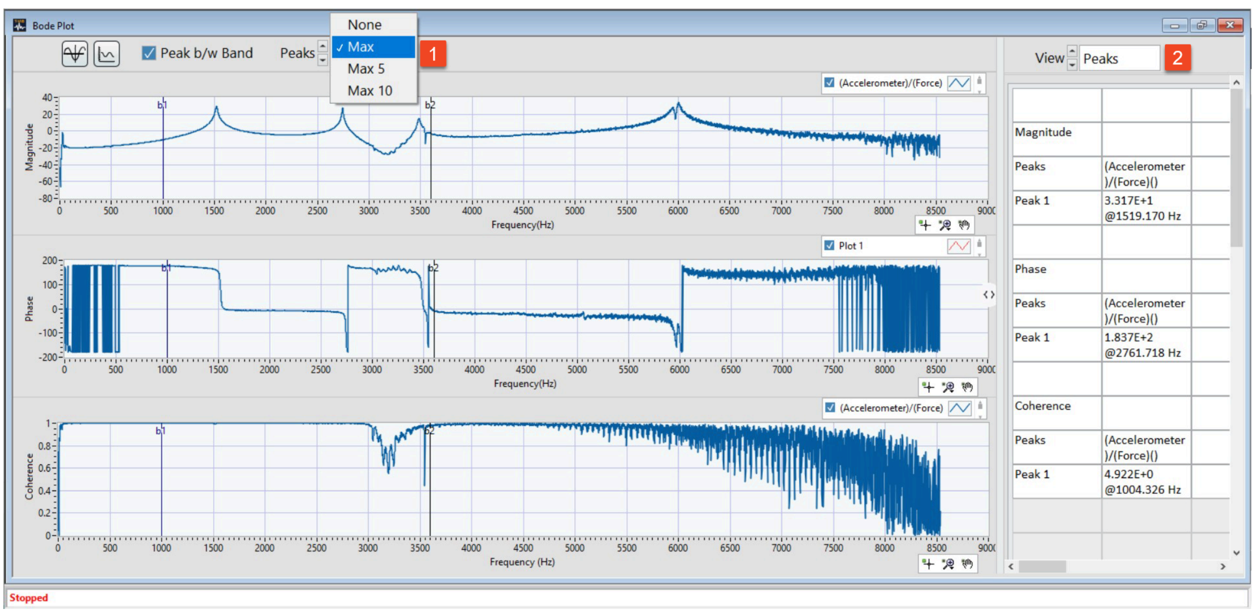
None: No peaks will be displayed. Default setting.

Max: To display the Maximum Amplitude value & its frequency.

Max 5: To display the Maximum Amplitude value of 5 peaks & its frequency.

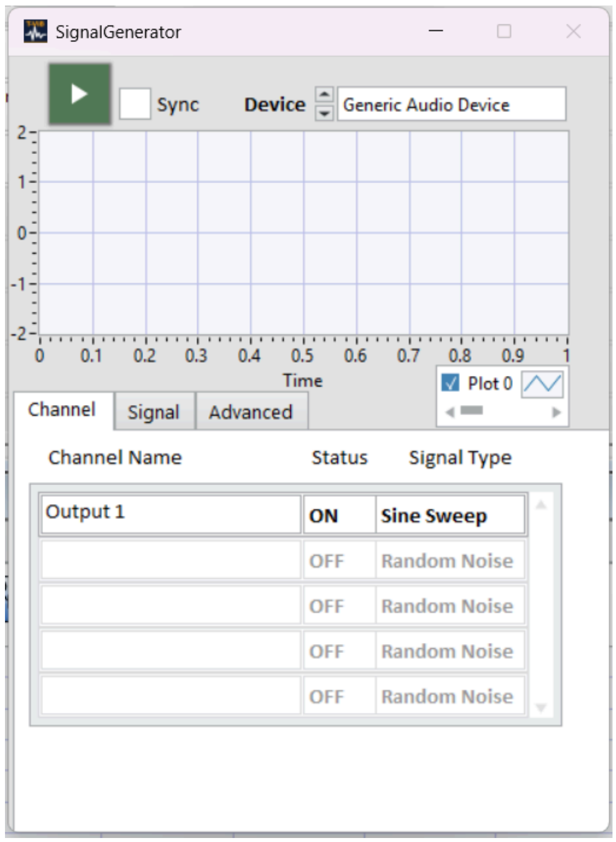
Max 10 :To display the Maximum Amplitude value of 10 peaks & its frequency.

2. Choose the **View** Selection to All/ Peaks for the Info Tab to display the peak information.

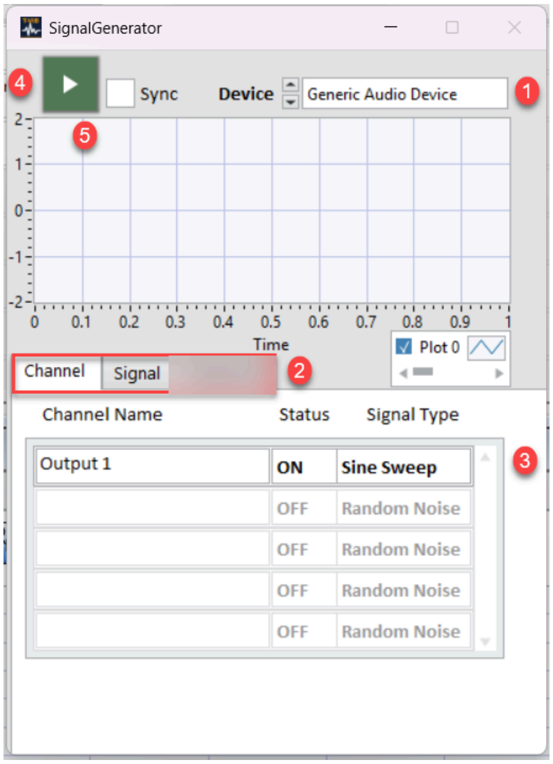


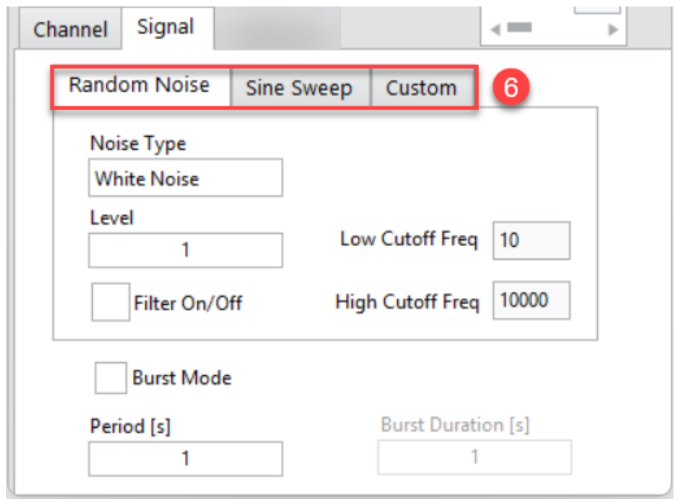
Waveform Generator (TWGM 206)

The waveform generator (TWGM 206) of TVIB is a software module used for accessing the internal signal generator of the TXcite Series amplifiers. The module has the option to generate random, sine, sine sweep and custom signals (optional burst mode) as stimulus for driving the electrodynamic shaker.



Module Interface





1. Device Name
2. Settings Ribbon
3. Channel Settings
4. Start button
5. View Waveform
6. Signal Settings Ribbon (in Signals option)

1. **Device Name:** Select the hardware device from the list
2. **Settings Ribbon:** Configure the settings of the Channel and Signal by clicking the channel or signal tab.
3. **Channel Settings:** The channel settings consist of the channel name, status and the signal type. Switch the status to ON and the signal type to the corresponding signal that is to be generated.
4. **Start button:** Press the Start button to start the signal generation.
5. **Signal Profile Graph** : The signals generated can be viewed in this graph

Based on the customer need, they can choose the tab

- Random Noise
- Sine Sweep
- Custom

Random Noise

The **Random Noise** tab allows generation of broadband noise signals for testing and analysis purposes.

Parameters

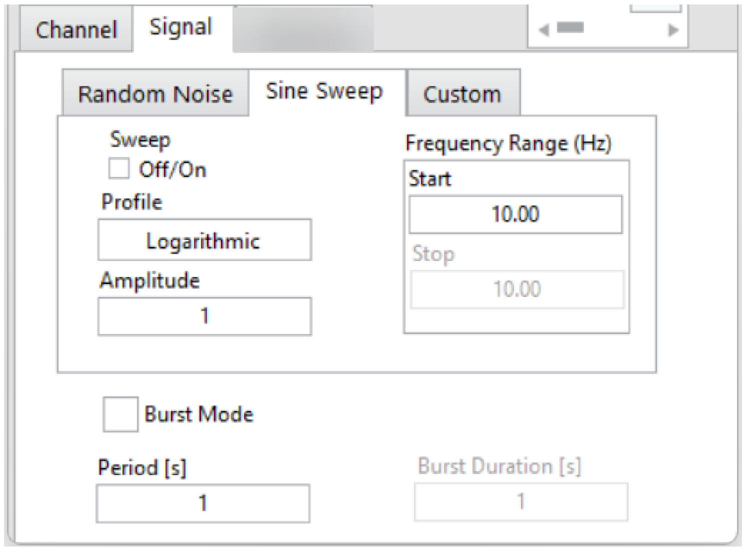
- **Noise Type**
  Select the type of noise to be generated.
  Example: *White Noise* (equal energy across frequencies)
- **Level**
  Defines the amplitude level of the generated noise signal.
- **Low Cutoff Frequency (Hz)**
  Sets the lower frequency limit of the noise band.
- **High Cutoff Frequency (Hz)**
  Sets the upper frequency limit of the noise band.
- **Filter On/Off**
  Enables or disables band-limiting of the noise signal based on the cutoff frequencies.

Burst Mode

- **Burst Mode (Enable/Disable)**
  When enabled, the noise signal is generated in bursts instead of continuously.
- **Period (s)**
  Defines the total cycle time for one burst sequence.
- **Burst Duration (s)**
  Specifies the duration within the period during which the signal is active.

Usage Notes

- Ensure **Low Cutoff Frequency < High Cutoff Frequency** for valid operation.
- When **Filter is OFF**, the signal may include the full bandwidth of the system.
- Burst Mode is useful for transient testing and system response analysis.



Sine Sweep Mode

The **Sine Sweep** tab allows the generation of a sine wave whose frequency changes smoothly over time. This is commonly used for system identification and testing resonant frequencies.

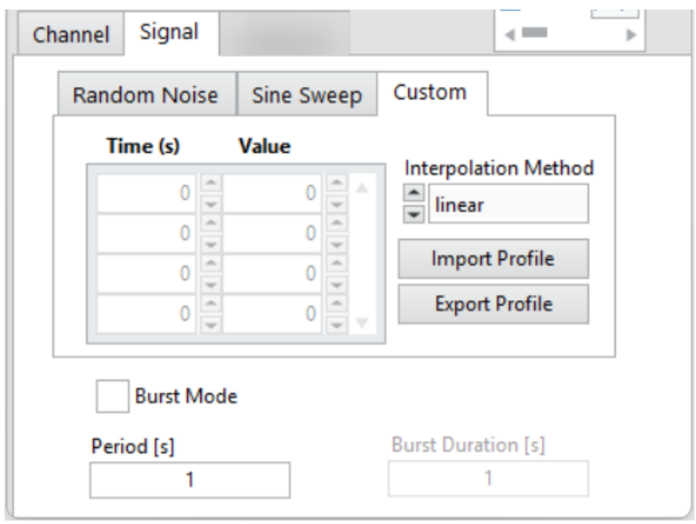
Parameters

- Start Frequency (Hz)
  - Defines the initial frequency of the sine wave sweep.
- Stop Frequency (Hz)
  - Defines the final frequency of the sine wave sweep.
- Period (s)
  - Specifies the total duration for the frequency to sweep from the start frequency to the stop frequency.
- Sweep Type
  - Selects the method of frequency change over time:
    - Linear Sweep: Frequency changes at a constant rate (Hz/s).
    - Logarithmic Sweep: Frequency changes at a constant rate in relation to octaves/decades. Typically used to excite all resonances equally.
- Level
  - Defines the constant amplitude level of the generated sine signal during the sweep.
- Number of Cycles

    - Specifies the number of sweep cycles to run. Setting this to 0 typically means continuous operation.
- Filter On/Off
    - Enables or disables filtering (band-limiting) of the generated signal.

Usage Notes

- A logarithmic sweep provides better excitation across a wide frequency range, especially at lower frequencies.
- Ensure Start Frequency is less than Stop Frequency for a forward sweep.
- Setting the Number of Cycles is useful for controlled, repeatable tests.



Custom Mode

The **Custom** tab allows the user to import a user-defined time-domain waveform to be generated by the device. This is useful for playing back recorded signals or custom transient waveforms.

Parameters

- **Import Waveform (.csv)**
    - Button to browse and load a user-defined time-domain signal file in .csv format.
    - **Format Requirement:** The .csv file must contain a single column of amplitude values representing the waveform samples. The waveform's sample rate will be dictated by the device settings (e.g., Block Size and Sample Rate).

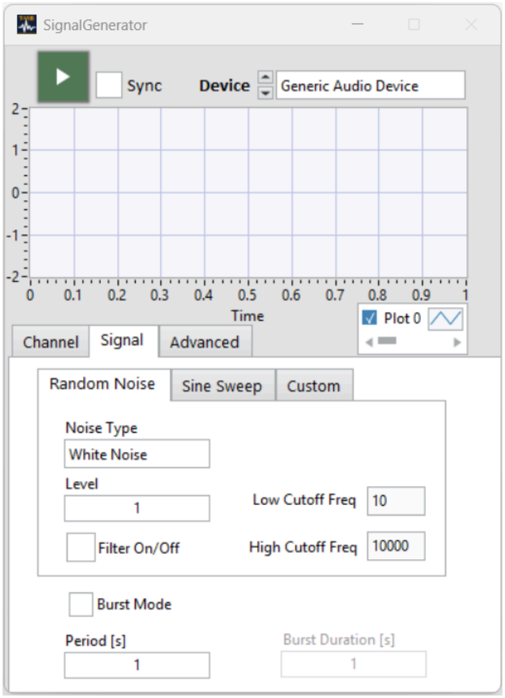
- **Signal Name**
  - Displays the name of the imported waveform file.
- **Level**
  - Defines the maximum amplitude scaling factor applied to the imported signal before output. This allows the user to adjust the output voltage level.

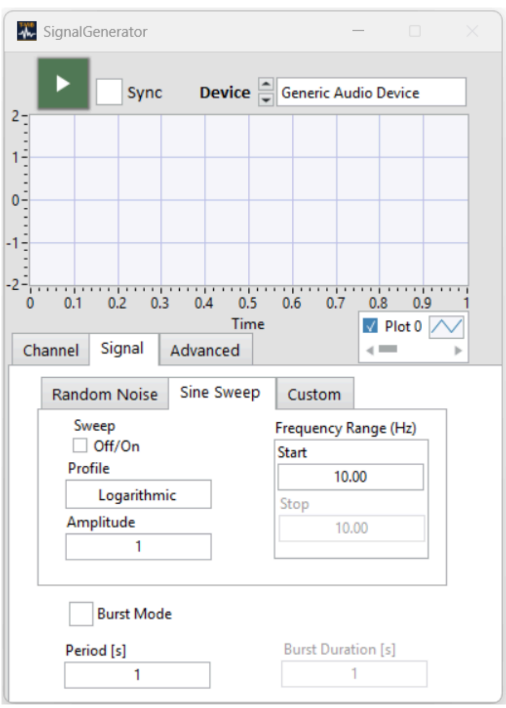
Burst Mode

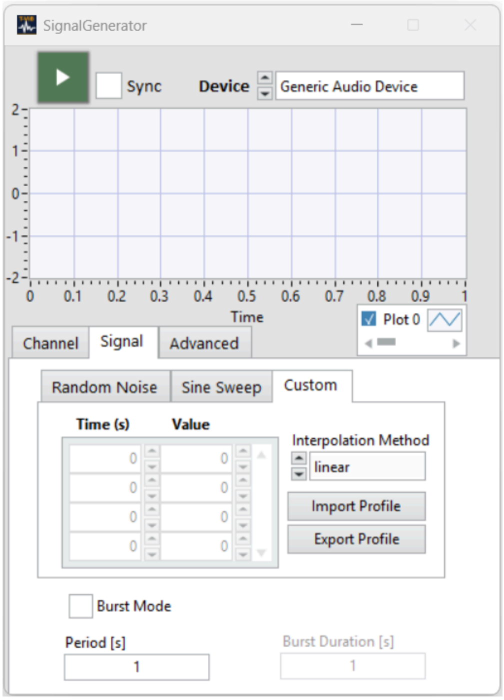
- **Burst Mode (Enable/Disable)**
  - When enabled, the imported signal is generated as periodic bursts.
- **Period (s)**
  - Defines the total cycle time for one burst sequence.
- **Burst Duration (s)**
  - Specifies the duration within the period during which the signal is active. The duration must be less than or equal to the total length of the imported waveform.

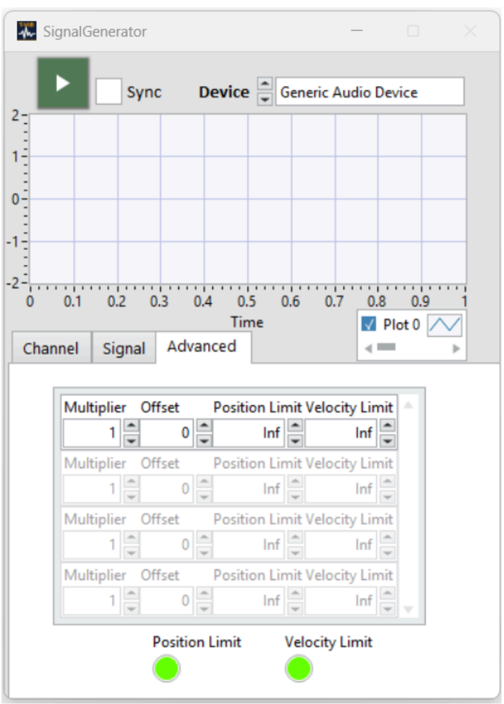
Usage Notes

- The imported waveform will be scaled to match the defined **Level**.
- Ensure the imported .csv file contains only amplitude data.
- This mode is ideal for reproducing specific transient events or complex non-standard signals.

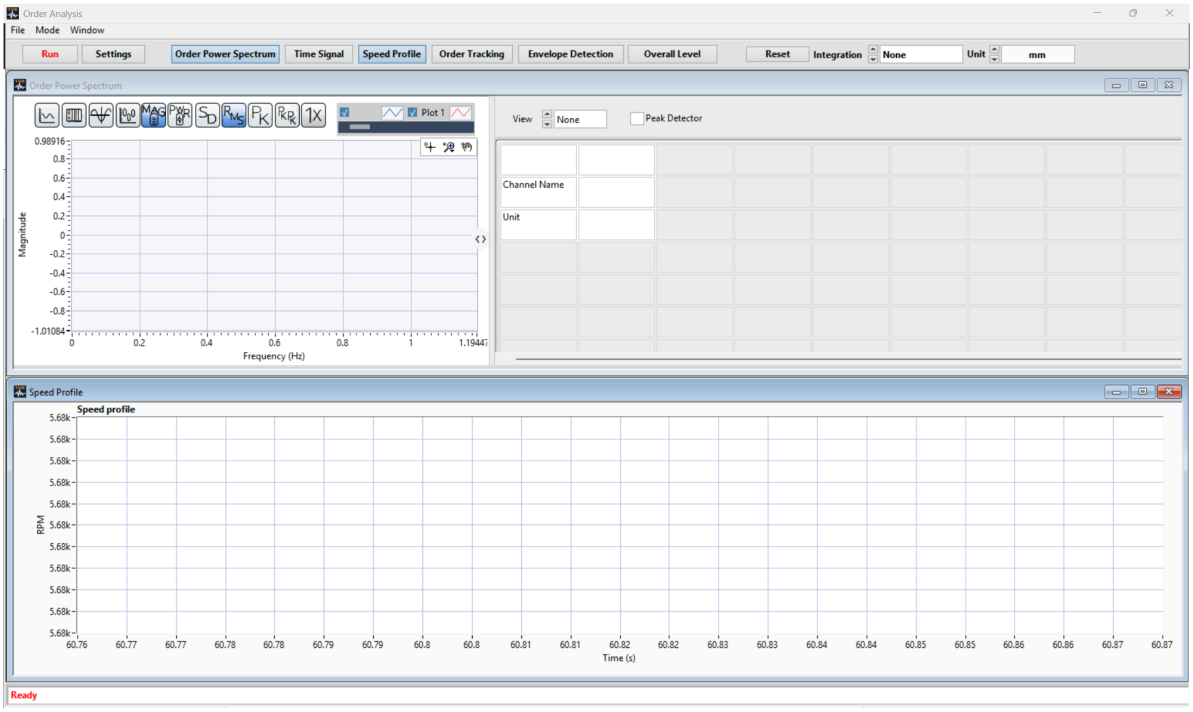

SignalGenerator
Sync
Device
Generic Audio Device
Time
Plot 0
Channel
Signal
Advanced
Random Noise
Sine Sweep
Custom
Noise Type
White Noise
Level
1
Low Cutoff Freq
10
Filter On/Off
High Cutoff Freq
10000
Burst Mode
Period [s]
1
Burst Duration [s]
1


SignalGenerator
Sync
Device
Generic Audio Device
Time
Plot 0
Channel
Signal
Advanced
Random Noise
Sine Sweep
Custom
Sweep
Off/On
Profile
Logarithmic
Amplitude
1
Frequency Range (Hz)
Start
10.00
Stop
10.00
Burst Mode
Period [s]
1
Burst Duration [s]
1

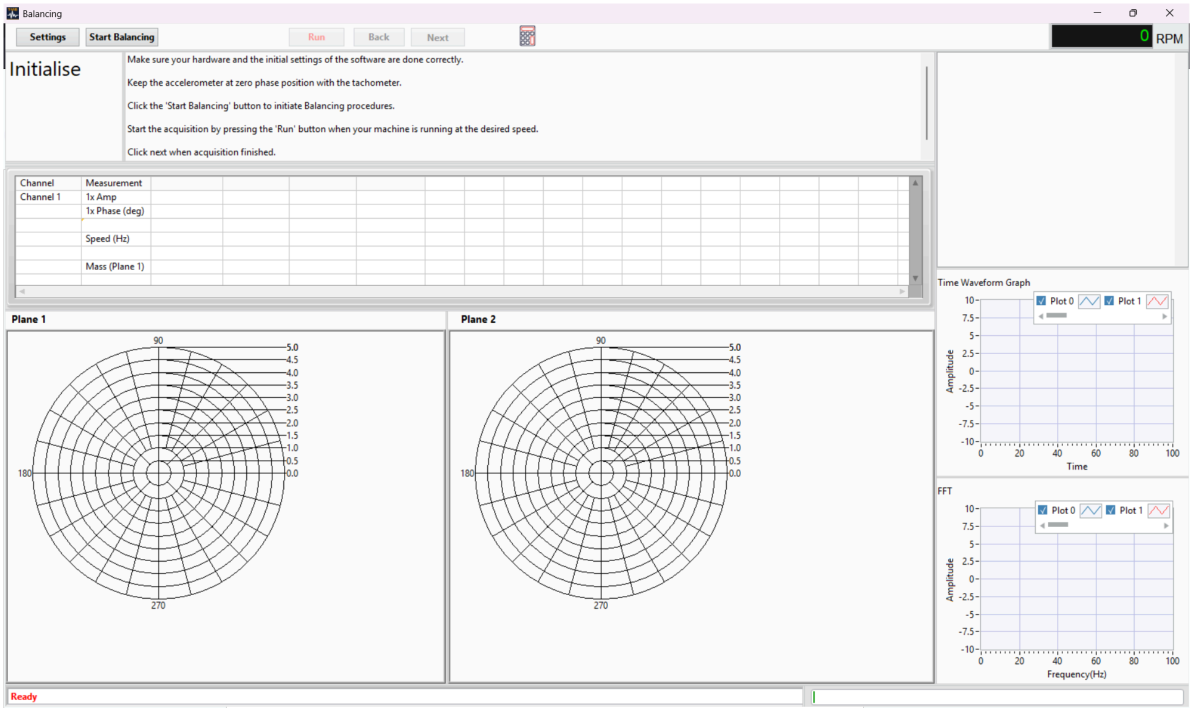

SignalGenerator
Sync
Device
Generic Audio Device
Time
Plot 0
Channel
Signal
Advanced
Random Noise
Sine Sweep
Custom
Time (s)
Value
Interpolation Method
linear
Import Profile
Export Profile
Burst Mode
Period [s]
Burst Duration [s]


SignalGenerator
Sync
Device
Generic Audio Device
Time
Plot 0
Channel
Signal
Advanced
Multiplier
Offset
Position Limit
Velocity Limit
Position Limit
Velocity Limit

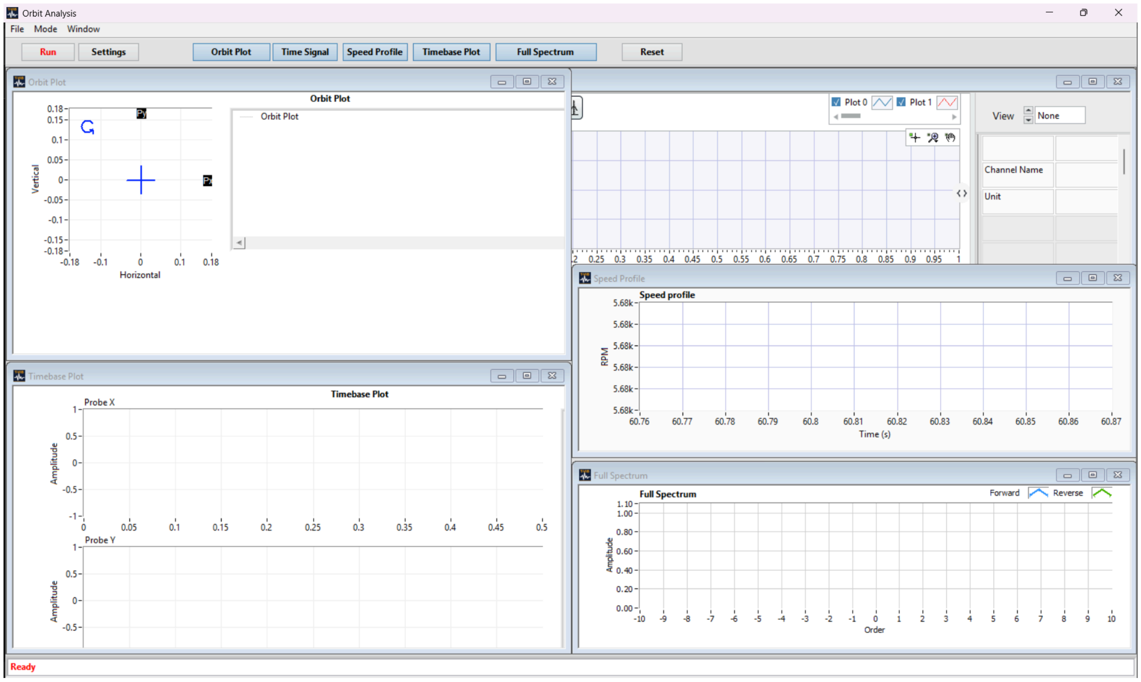
Order Analysis (TOA 207)



Field Balancer (TB 208)

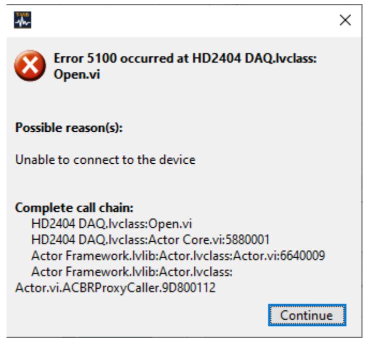


Orbit Analysis (TOA 209)



Troubleshooting

- **What should I do if the software shows the error "Unable to connect to the device"?**

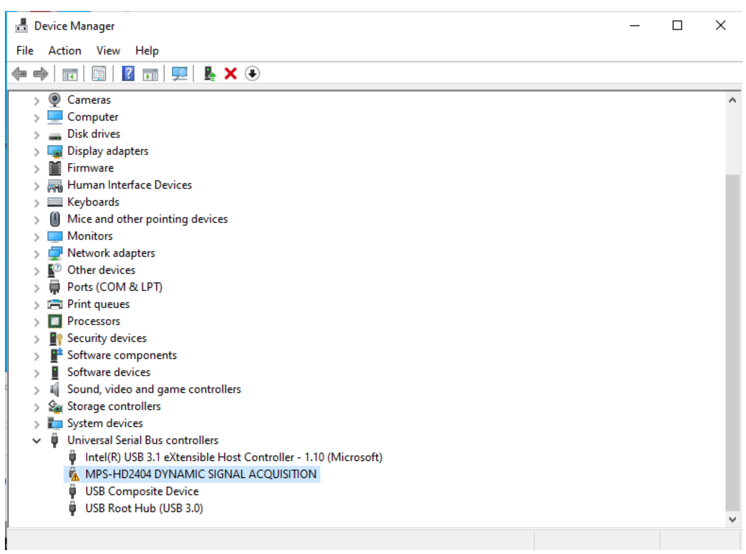


This error could occur

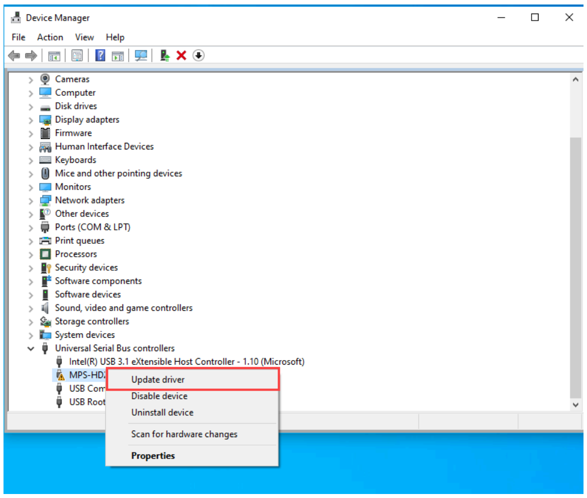
- If there is insufficient power in the port. Check plugging to a different Port and confirm whether the port is working properly.
- If the driver is not detected. Follow the below steps for checking and rectifying this issue.

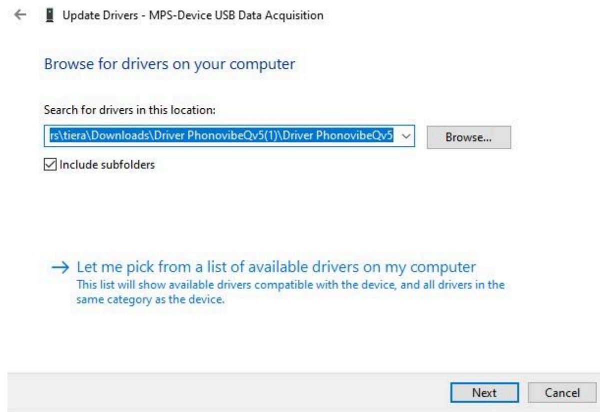
<u>Steps to check and rectify the driver issue</u>

Step 1 : Open the Device Manager

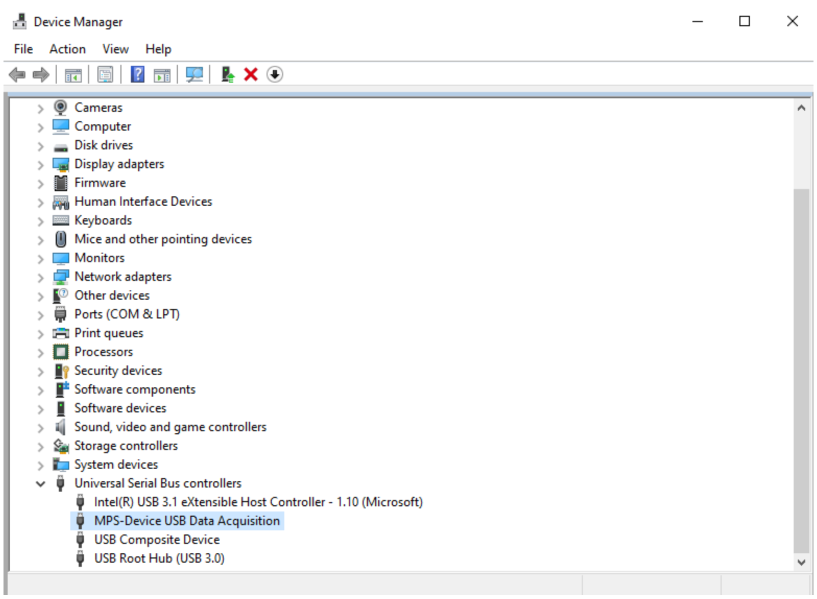


Step 2 : If there is a warning signal, right click and choose the update driver and select the driver folder.





Once the driver is updated the device manager window shows the device configured properly.



Step3: If the error is still occurring, then check for the sample rate in the settings. This should be the sampling rate compatible with the device.

Support

For any queries or assistance, contact support@tieraonline.in

For self-learn courses/ training, visit our website https://learn.tieraonline.in/

CONTACT US

Tyrannus Innovative Engineering and Research Academy Pvt. Ltd.
TrEST Research Park, TC- 4/2322, GEM building, (Opp) CET Thiruvananthapuram, Kerala
695016
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