Sound & Vibration Training

National Instruments Germany GmbH
Konrad-Celtis-Str. 79
81369 München
Tel.: 089 7413130
Fax: 089 7146035

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Agenda

- Uses of Sound and Vibration
- System Architecture – Sensors, Data Acquisition, Analysis
- Introduction – LabVIEW + Sound and Vibration Measurement Suite
- Analyze and Present Data with NI DIAdem
- Exercises

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What You Need To Get Started

Sound & Vibration Course Manual

CD with Exercises and Solutions

- NI LabVIEW 8.5
- NI Sound and Vibration Measurement Suite 6.0
- NI DIAdem 10.2

- NI USB-9233
- Sound and Vibration Signal Simulator
- Microphone
- BNC cables

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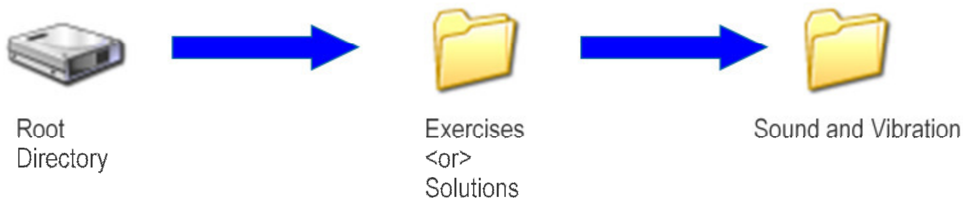
File Locations



Why Measure Sound & Vibration?

- **R&D** – during research and development, the noise & vibration of a device is studied and attempts are made to reduce its vibration, or improve acoustics thus permitting longer service life and greater appeal to end users. e.g. Appliances, Vehicles, Tools.

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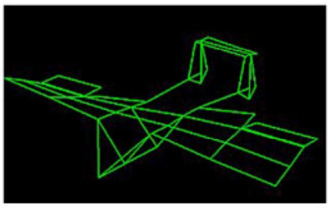
Why Measure Sound & Vibration?

- **Structural Testing** – measuring the vibration response of a structure permits a determination of the integrity of the structure, material properties, and mode shapes. e.g., Fatigue, Stiffness, Cracking.



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Why Measure Sound & Vibration?

- **Control** – detect the presence or change of noise or vibration and initiate appropriate action. e.g., Active Suspension Systems, Cabin Noise Suppression, Vibration Shaker Control

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Why Measure Sound & Vibration?

- **Machinery Protection** - monitor vibration and initiate alarm or shutdown when levels exceed a pre-determined threshold.

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Why Measure Sound & Vibration?

- **Predictive Maintenance** – trend and analyze machinery performance to determine when maintenance will be necessary to avoid a catastrophic failure.

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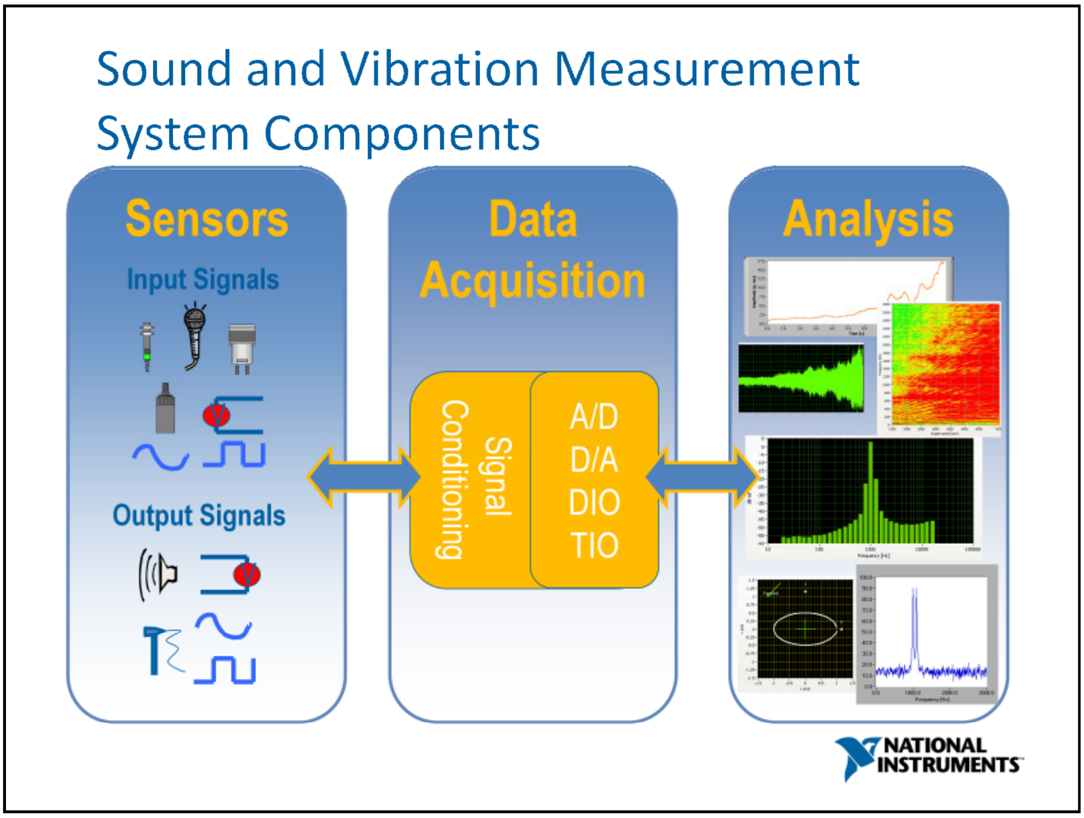
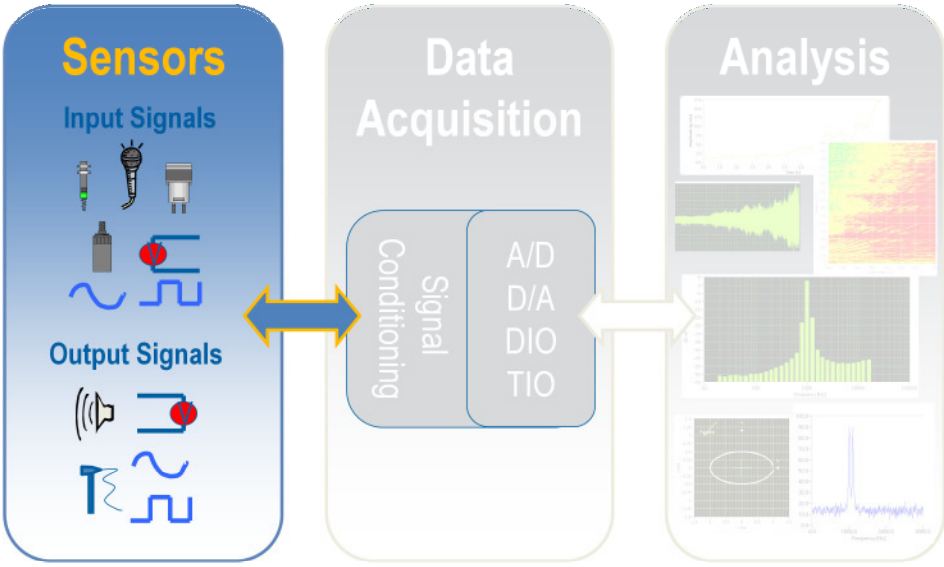


You can divide a Sound and Vibration Measurement System into three parts:

- Sensors
- Data Acquisition
- Analysis

Sound and Vibration Measurement System Components



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Transducers used in Sound and Vibration Applications

Sensor	Physical Phenomena
Microphone	Sound Pressure
Accelerometer	Acceleration
Proximity Probe	Displacement
Tachometer	Rotating Speed

Microphones

- Measure
  - Sound pressure (a variation around the atmospheric pressure)
  - Converted to Sound Pressure Level (SPL in dB)
    (via a dB reference to 20 uPa)
- Result is expressed in Pascals (Pa)
- Calibration with a reference noise source

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Microphones are basically pressure sensors, but they are dedicated to the measurement of very small variations of pressure around the atmospheric pressure.

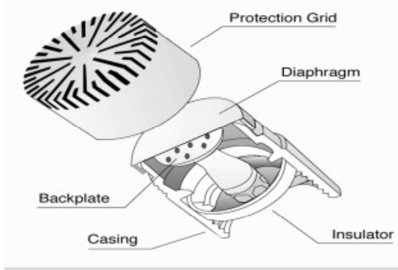
The sound pressure level (SPL) measured by a microphone is expressed in decibels (dB ref 20 μPa), as given by the following formula:

$$SPL = 20 \log_{10}(p/p_0) \qquad \text{with } p_0 = 20\ \mu\text{Pa}$$

This reference value (giving 0 dB) corresponds roughly to the minimal sensitivity of the human ear. By opposition, a value around 120 dB corresponds to the threshold of pain.

The calibration of a microphone is performed with a reference noise source generating a constant and known sound pressure level (typically 94 dB, which corresponds almost to 1 Pa) at a particular frequency.

Types of Microphones



- Prepolarized (Electret)
  - Powered by constant current (2-20 mA)
  - Two wire connection (BNC, microdot, etc,)
- Externally Polarized
  - Powered by up to 200 V supply
  - 7 pin LEMO connectors

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Microphone Systems

PCB microphones can be broken down into two categories, Externally Polarized microphones and Prepolarized microphones. The cartridge from the condenser microphone operates on basic transduction principles and will transform the sound pressure to capacitance variations, which are then converted to an electrical voltage. This conversion process requires a constant electrical charge, which is either applied by a preamp or built into the microphone. Externally Polarized microphones will differ, when compared to the Prepolarized microphones, in the relationship of how the constant charge of the capacitance between the diaphram and backplate is applied. Externally Polarized and Prepolarized microphones will each require different components for optimum operation.

Externally polarized microphones are based on a capacitive transduction principle. These high precision condenser microphones require a constant electrical charge for polarization from an external source. This voltage source comes from an external power supply, which ranges from 0V (and can be used with Prepolarized microphones) to 200V. PCB’s Externally Polarized microphone set-up requires the use of 7-conductor cabling with LEMO connectors. Externally polarized microphones are the traditional design, and are still utilized for compatibility reasons.

Insert Externally Polarized set-up picture

Prepolarized microphones are also high precision condenser type microphones. The polarization process is accomplished by adding a polymer that is applied to the backplate. This permanently charged polymer contains frozen electrical charges and is commonly referred to as an electret. The prepolarized microphones are ICP® powered by an inexpensive and easy-to-operate, constant current signal conditioner (or directly by a readout that has constant current power built-in). This enables the owner to use low impedance coaxial cables with BNC or 10-32 microdot connectors (rather than 7 Pin conductor cabling with LEMO connectors), for both current supply and signal to the readout device. This newer design has become very popular in recent years due to its cost savings and ease of use characteristics.

Insert Prepolarized set-up drawing here

Array microphones, are Free Field Type microphones, which are designed to offer a cost effective solution for multiple channel sound measurement. This makes Nearfield Acoustic Halography (NAH) measurements practical. Grids can be constructed to take 2D mapping measurements. The 130D20 and 130D21 have an integrated Microphone and Preamp. The 130 series utilizes the Prepolarized microphone design, and incorporate ICP®, powered by a constant current signal conditioner. The 130 series provide an inexpensive alternative to the 377 series. The 130 series are accurate for frequency responses and great for trending, but are more sensitive to changes in temperature, and less accurate than the 377 series of high precision condenser microphones, when measuring dB.

Insert multiple channel array set-up drawing

Sound Pressure Level (SPL)

- Sound level is most commonly expressed in dB
- This is a reference to 20 uPa which is the threshold of human hearing
  - Audible frequency range: 20 Hz to 20 kHz
- Convert to dB using this equation:

$$dB = 20 \log \left(\frac{P}{Po} \right)$$

Where:
P = Pressure in Pascals
Po = Reference Pascals (.00002 Pa)

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Sound Pressure Level (SPL)

Sound Pressure	Sound Level	Comparison
0.00002 Pa	0 dB	Threshold of Hearing
0.02 Pa	60 dB	Business Office
0.2 Pa	80 dB	Shop Noise
1 Pa	94 dB	Large Truck
2 Pa	100 dB	Jackhammer
20 Pa	120 dB	Airplane Take-Off
200 Pa	140 dB	Threshold of Pain

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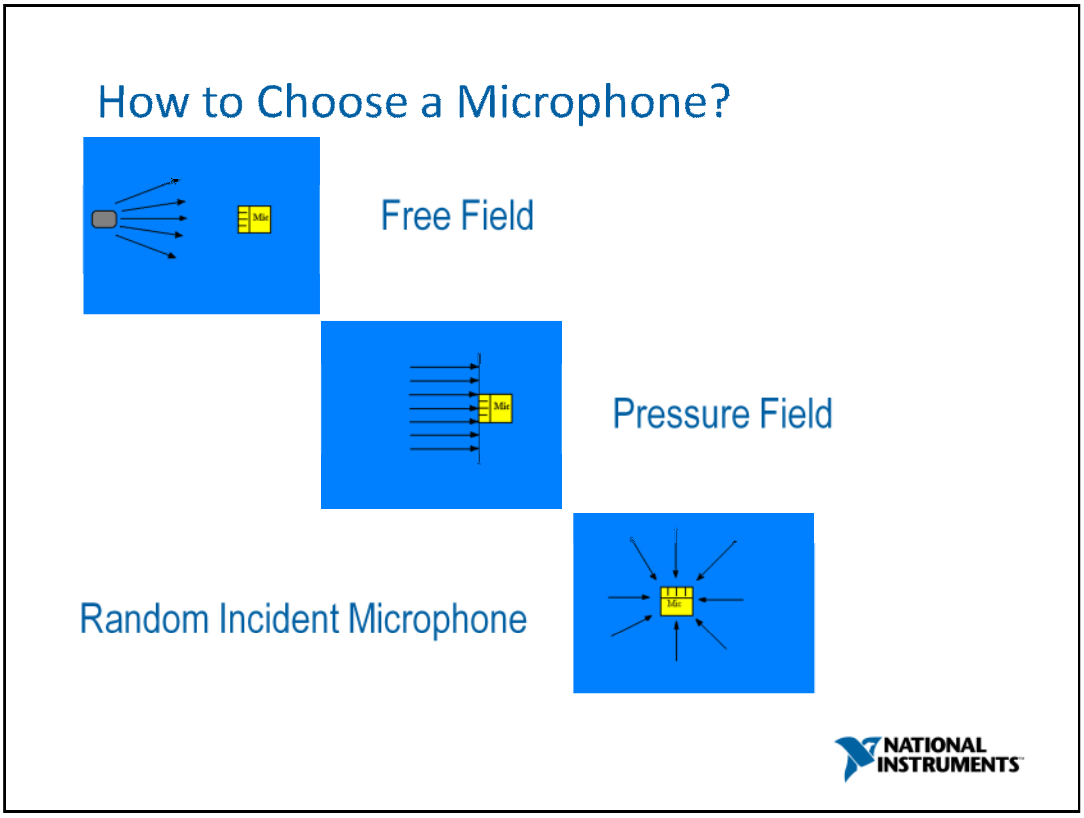
High Precision microphones are used in acoustical test and measurement applications in order to determine the amount of sound pressure, in decibels (dB) that are exerted on an object at different frequencies and wavelengths. Acoustical testing is performed for a variety of reasons, ranging from the design of new products, to monitoring products, to predictive maintenance functions, to personal protection. Pressure from sound not only can damage material items, but also can damage the most precious and delicate mechanism designed to perceive it, the human ear.

The main criteria to describe sound, is based upon the amplitude of the pressure fluctuations. The lowest amplitude that a healthy human ear can detect is 20 millionths of a Pascal (20µPa). Since the pressure numbers represented by Pascals are generally very low and not easily managed, another scale was developed and is more commonly used, called the Decibal (dB). The decibal scale is logarithmic and more closely matches the response reactions of the human ear to the pressure fluctuations

PCB specifies its microphone on the maximum decibal level based on the design and physical characteristics of the microphone. The specified maximum dB level will refer to the point where the diaphram will approach the backplate. The maximum decibals that a mic will output in a certain application is dependant upon the voltage supplied, and the particular microphones sensitivity. In order to calculate the maximum output for a microphone, using a specific preamp and its corresponding peak voltage, use the following formulas:



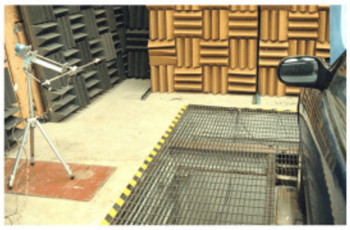
PCB for example offers microphones with the three most common application fields. The first is the Free-Field Type. The Free-Field microphone is designed to measure sound coming from one direction and source, which is pointed directly at the microphone. The sound waves propagate freely without any disturbing objects. The Free-Field microphone measures the sound as it was exited, without the influence of the microphone itself. Thus, it has been compensated for its own presence. These microphones work best in open areas, where there is no hard or reflective surfaces. Anechoic rooms are ideal for these Free Field microphones

The second type is called a Pressure Field. A Pressure Field microphone is designed to measure the sound pressure that exists in front of the diaphram. It is described to have the same magnitude and phase at any position in the field. It is usually found in an enclosure, or cavity, which is small when compared to wavelength. The microphone will include the measurement changes in the sound field caused by the presence of the microphone. The sound being measured is coming from one source at a direction pointing directly at the microphone. Testing of pressure exerted on walls, structures or pressure exerted on airplane wings, are examples of Pressure Type microphone applications.

The third type is called a Random Incident Microphone. This is also referred to as a “Diffuse Field Type.” The Random Incident type of microphone was designed to be omni-directional and measure sound pressure coming from multiple directions, with similar level. The Random Incident type microphone will measure the sound as if it existed before the introduction of the microphone itself into the diffuse field. When taking sound measurements in a church or in a shop with hard, reflective walls, you would utilize this type of microphone.

Microphone Applications

- Acoustic array
- Sound quality
- Sound power
- Pass-by-noise (Fahrgeräusch)



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Example Application with Acoustic Array: Flyover Test

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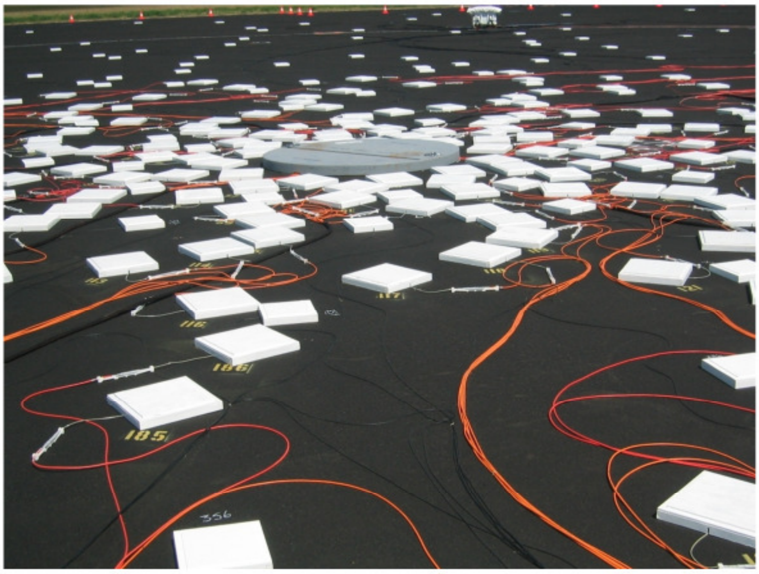
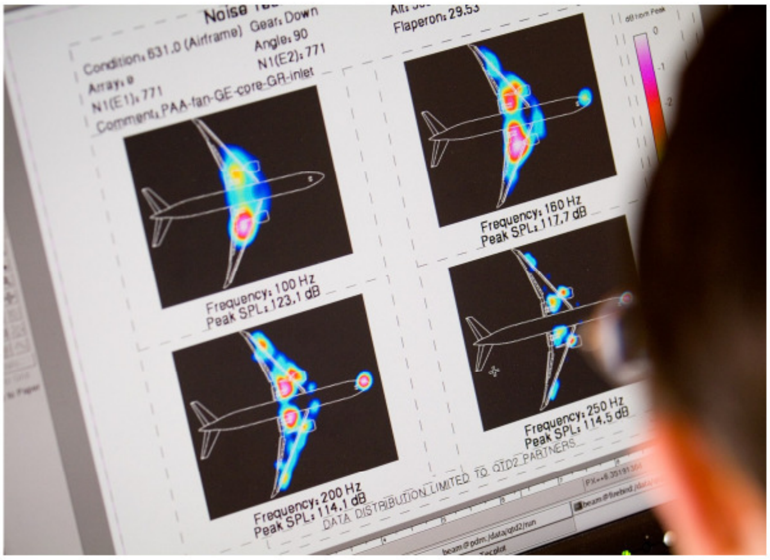
In an airplain application, during a typical test cycle, the aircraft would fly past the target approximately every 6 minutes. The system had to be capable of uploading the data acquired to the master data storage engine and be ready to acquire more data. During the test sequence they conducted over 300 tests comprising 78 minutes of data and over 1 terabyte of data total.

Microphone Arrays



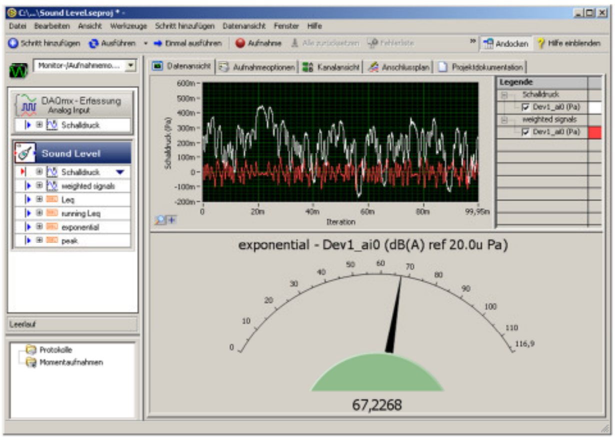
In this application customer used 448 microphones for the Microphone Arrays.

Noise Maps

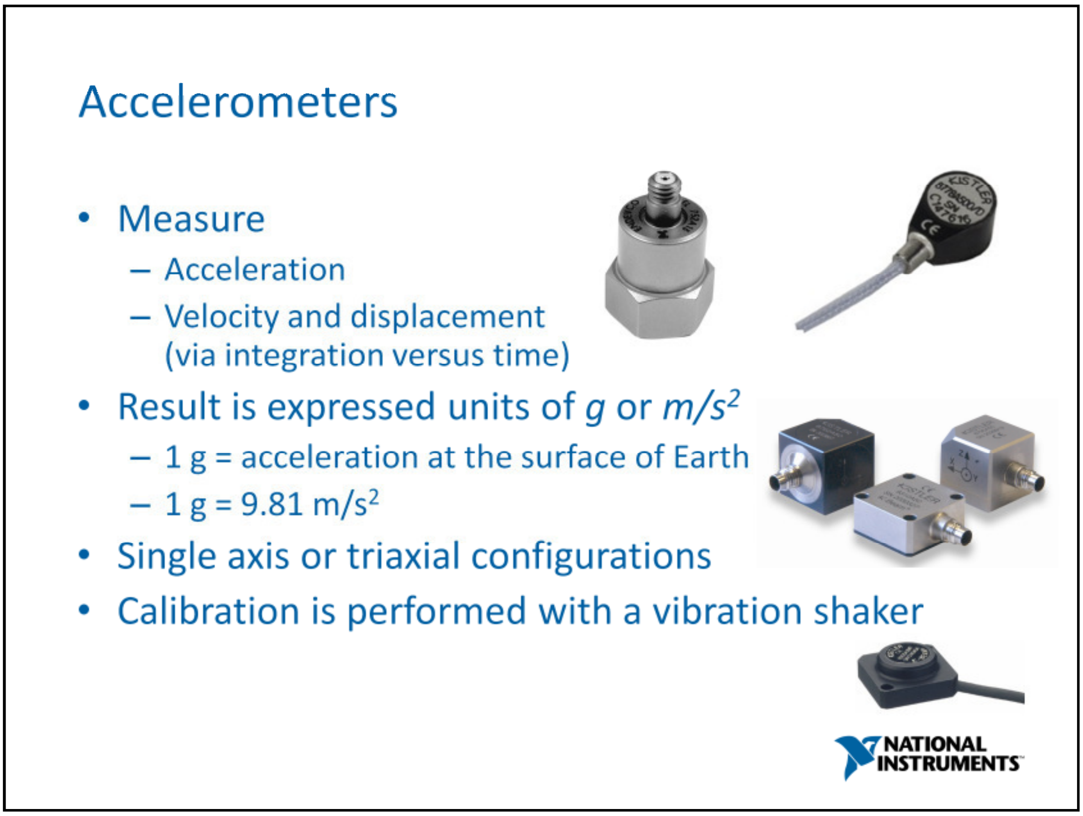


With the data of the microphone array you can create Noise Maps to get information which element produce the noise.

Exercise 1 – Microphone Measurement



In this exercise you will use the Sound and Vibration Assistant to measure the sound level from a microphone.



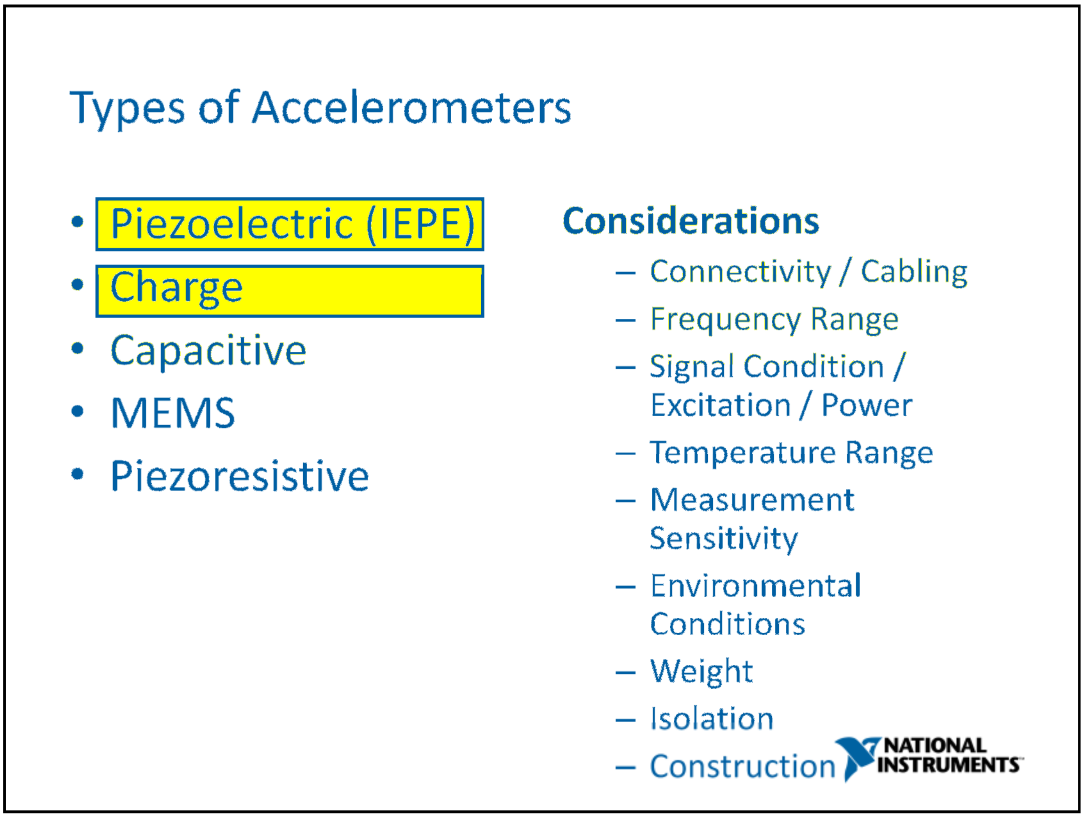
The original purpose of any accelerometer is of course to measure the acceleration level at a particular point of the system under test. But this signal, if properly integrated, also allows the simultaneous measurement of the velocity and the displacement.

Vibration parameters are almost universally measured in metric units (m/s^2). The gravitational constant "g" is often used for acceleration, but is not a standard metric unit of measure. Fortunately, a conversion factor exists with which you can perform quick mental calculations with only 2% of error.

$$1.0g = 9.81\ m/s^2$$

Simple (1D) accelerometers measure the component of the acceleration in one particular direction. If three of these accelerometers are combined in a single unit (3D accelerometer), the acceleration can me measured simultaneously in three orthogonal directions.

The calibration of an accelerometer is typically performed with a shaker generating a constant and known level of acceleration at a particular frequency.



IEPE signal conditioning provides a constant-current source to power internally amplified sensors. All major accelerometer suppliers have their own equivalent solution, such as ICP from PCB Piezotronics, Isotron from Endevco, DeltaTron from Brüel & Kjaer, and Piezotron from Kistler.

The major advantage of the IEPE accelerometer is that it is simple and easy to use due to the built-in microelectronics and the simple conditioner required.

Limitations are a limited temperature range, especially for high temperatures, and a fixed sensitivity.

Charge mode accelerometers are used essentially for measurements in high-temperature environments. They also offer variable sensitivity.

However, as they do not have built-in signal conditioning, they require an external charge amplification and conditioning to convert charge to voltage.

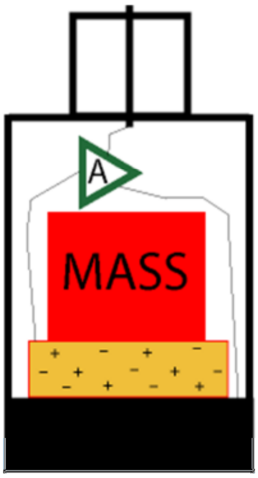
$$V_{out} = q/C_f$$

They also require low noise cabling and are sensitive to environmental influences.

MEMS	Micro Electro Mechanical Systems
Piezoresistive	Change in resistance

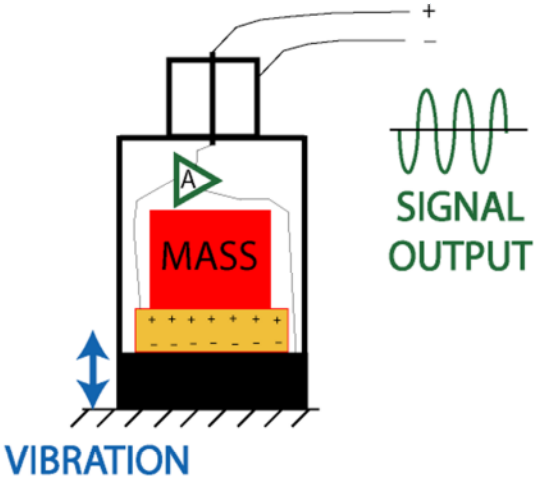
Operating Principles for Piezoelectric Accelerometers

- The acceleration sensing element is installed into a protective housing
- An amplifier, which serves to provide signal conditioning, is typically included
- For high temperature requirements, this amplifier is not included, and signal conditioning is performed remotely



Operating Principles for Piezoelectric Accelerometers

- When exposed to vibration, the accelerometer generates an analog output signal, which is proportional to the acceleration of the applied vibration.



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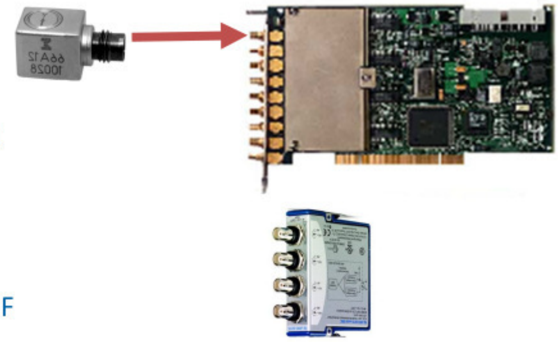
IEPE Accelerometers

- Advantages
  - Simple and easy to use
  - Built-in microelectronics
  - Simple constant current signal conditioner (18-30 VDC ; 2 mA)
- Limitations
  - Temperature range: max 250 °F (some 325 °F)
  - Fixed sensitivity

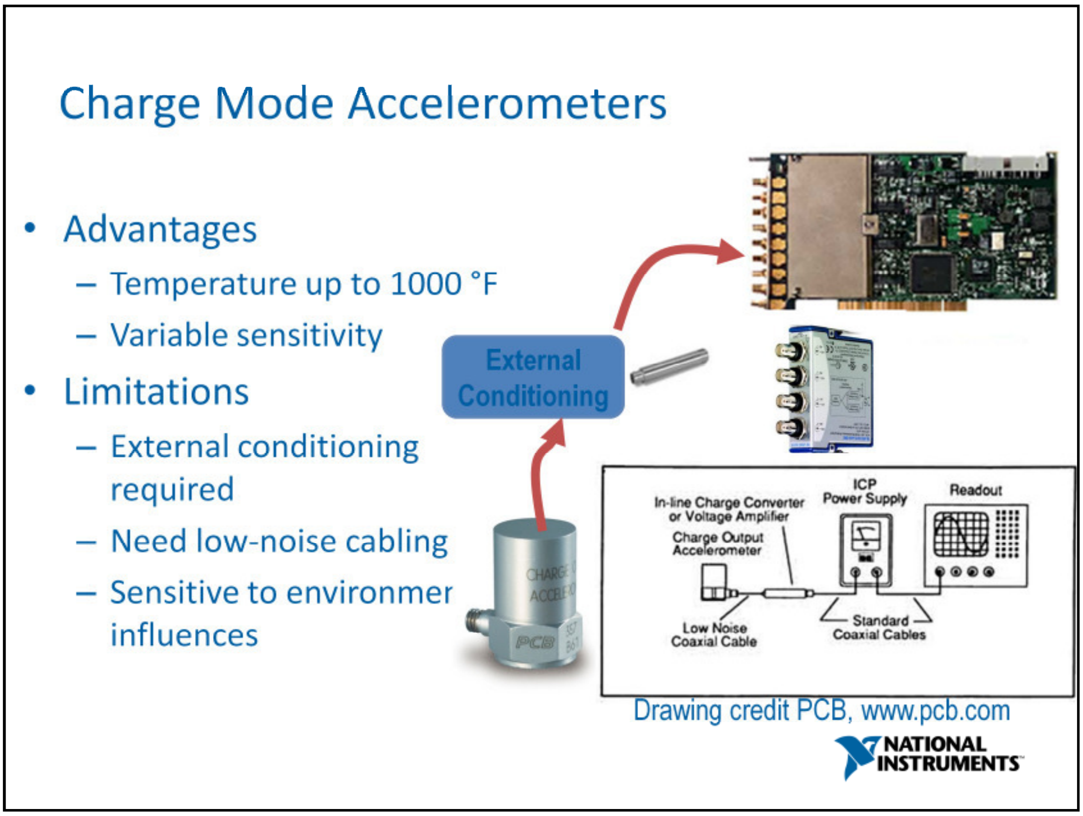
Direct Connection



IEPE signal conditioning provides a constant-current source to power internally amplified sensors. All major accelerometer suppliers have their own equivalent solution, such as ICP from PCB Piezotronics, Isotron from Endevco, DeltaTron from Brüel & Kjaer, and Piezotron from Kistler.

The major advantage of the IEPE accelerometer is that it is simple and easy to use due to the built-in microelectronics and the simple conditioner required.

Limitations are a limited temperature range, especially for high temperatures, and a fixed sensitivity. However, 21st century technologies are raising the temperature range of accelerometers.



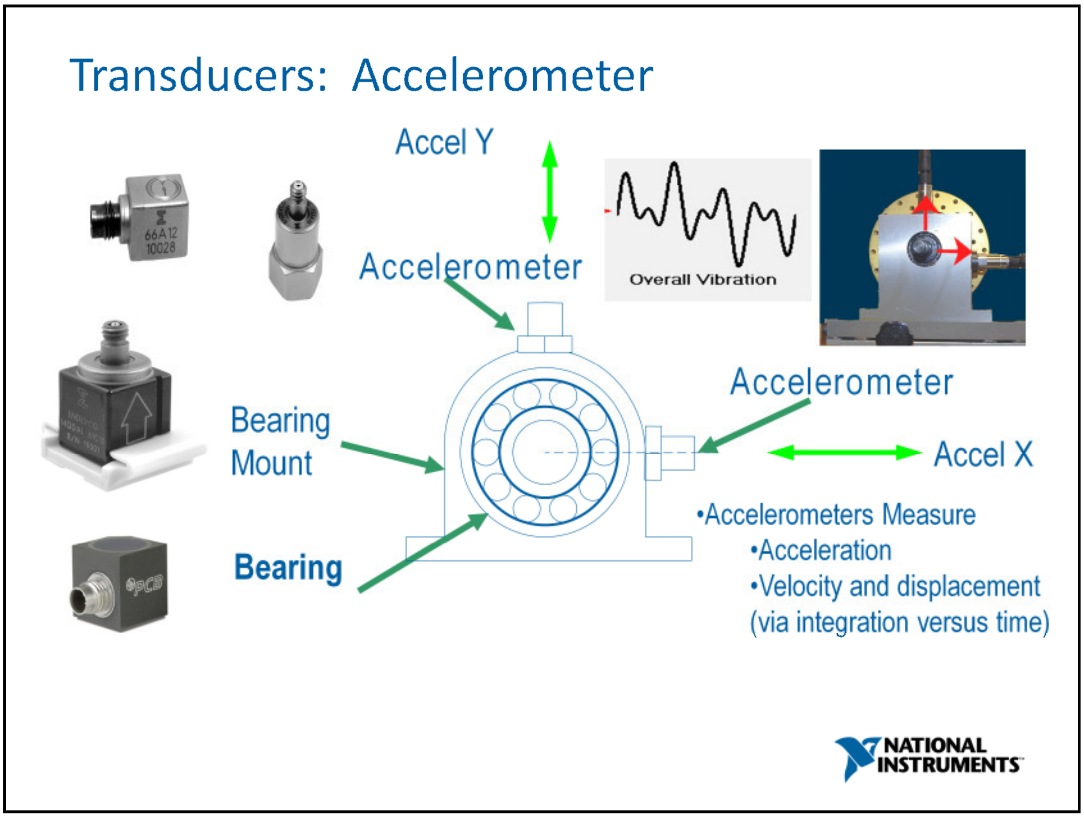
Charge mode accelerometers are used essentially for measurements in high-temperature environments. They also offer variable sensitivity.

However, as they do not have built-in signal conditioning, they require an external charge amplification and conditioning to convert charge to voltage.

$$V_{out} = q/C_f$$

They also require low noise cabling and are sensitive to environmental influences.

Charge mode accelerometers are common in electrical power generation applications where high temperature turbines are used.

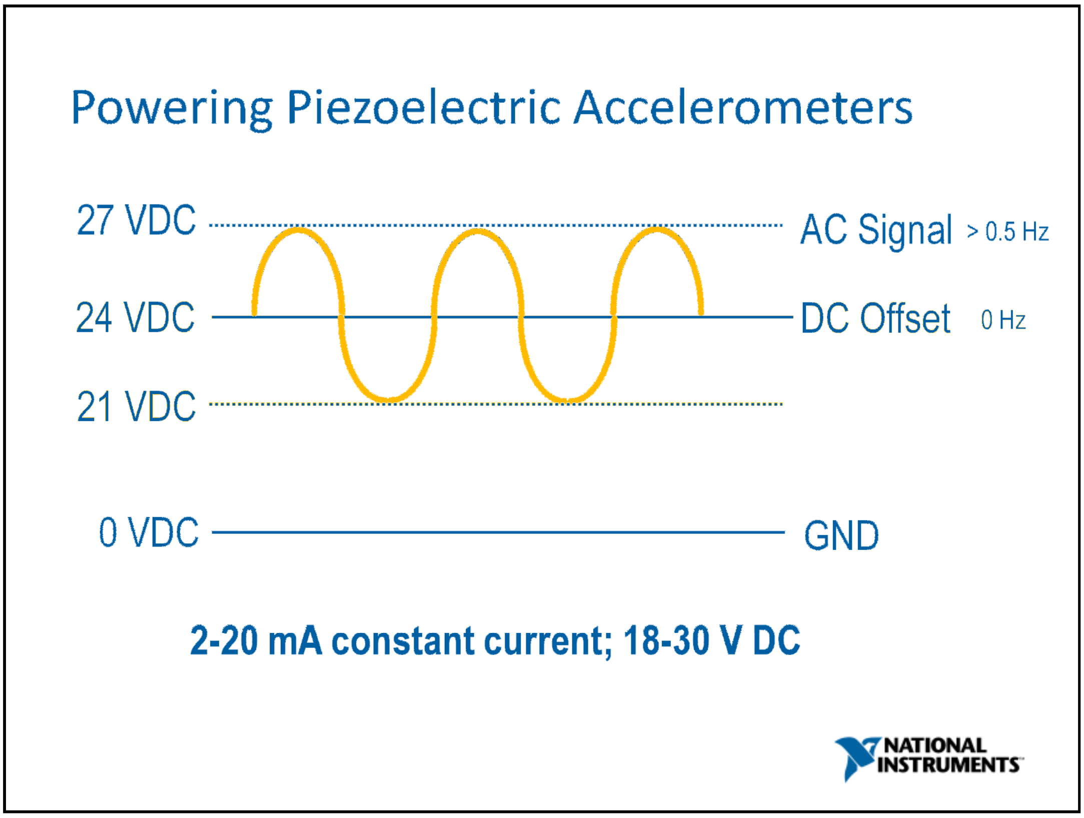


The common stimulus for accelerometers is the normal movement of the machine or unit under test. In the case of a rotating machine, rotation is the stimulus that creates vibration.

Accelerometers measure vibration that is transmitted thru the mechanical structure of the machine. Typical accelerometer applications include bearings, gearboxes, electrical motor, frame or footing vibration.

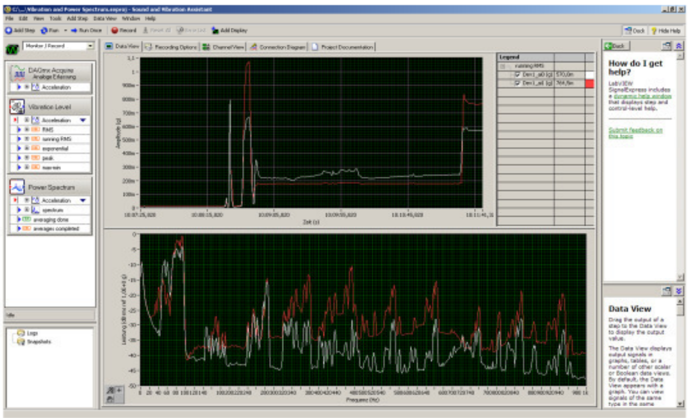
Because of their proximity to bearings, bearing mounts are a common location for accelerometer sensors to measure vibration signals. The diagram shows a bearing mount with accelerometers that measure acceleration in the X and Y directions shown in green.

Bearing wear is a common problem with rotating machinery because the bearings are commonly the transmission path for rotor forces caused by other vibration forces. As such, worn bearings are often only symptoms of other problems. For instance, stress caused by a bent shaft often contribute to bearing wear.



Piezoelectric accelerometers are powered by a constant current at DC. This is why these types of accelerometers have no response at DC, because there is no signal. The typical signal has frequency content only above DC.

Exercise 2 - Accelerometer Measurement



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In Exercise 2 you will use the Sound and Vibration Assistant to measure the vibration level of accelerometers and perform a power spectrum during live acquisition.

Displacement and Velocity Probes

- Non-contact displacement probes
  - High-frequency oscillator generating eddy currents whose energy is a function of displacement
  - Used when low level of acceleration is generated (for example, turbomachines)
- Velocity transducers
  - Moving coil in permanent magnetic field
  - Replaced by accelerometers, offering the same functionality through a wider frequency range

In turbo machinery, shaft vibrations are very important to monitor. This is because turbo machinery rotates at much higher speeds than electrical motor or engine drive machinery. To measure vibrations on the shaft a non-contact vibration sensor is used.

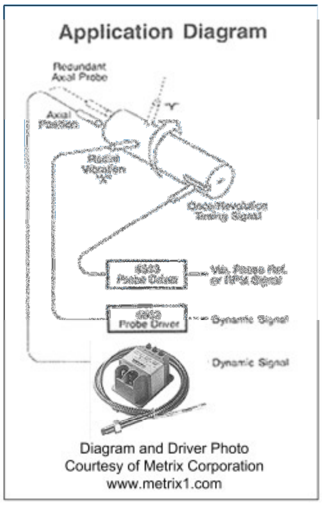
Non-contact displacement probes measure relative motion between the shaft and the mount. They use a high-frequency oscillator to generate eddy currents in the shaft. The energy of these eddy currents is function of the distance between the shaft and the probe, and so modulates the oscillator voltage. After demodulation and removing the DC offset, the output signal is proportional to the displacement.

Displacement probes are especially useful when a low level of acceleration is generated, as is the case for rotating machines with flexible bearings (for example, turbomachines).

Similar to accelerometers, Velocity transducers use a moving coil in a permanent magnetic field to generate a signal that is proportional to the velocity of the vibration. This magnetic field is produced by a permanent magnet mounted on springs so that this spring/mass/damper system maintains the magnet essentially fixed in space (very low natural frequency). This type of sensor is now often replaced by accelerometers, which offer the same functionality (via integration as seen before) through a wider frequency range. Velocity sensors are less common today, but are still in use. The IEEE vibration limits table is listed in velocity, or inches per second.

Transducers: Proximity Probes

- **Measure:** Distance or displacement
- **Output:** An AC signal proportional to the distance from the tip of the probe to a metallic surface
- **Excitation:** -15 to -24 V power source
  - The AC signal is transmitted on this DC offset



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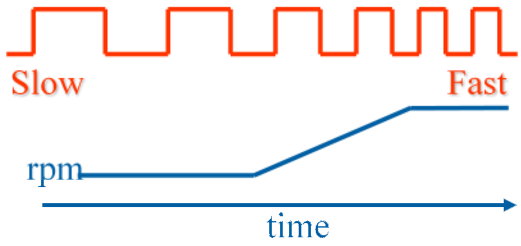
Here we have a diagram of five proximity probes on a machine shaft. Shaft vibrations are typically indicative of unbalance, misalignment, and mechanical looseness. Proximity probes are probably one of the more common sensors used for larger machinery applications and for rotor dynamics research. Radial position of the shaft is typically measured with two proximity probes, one in the horizontal or X position and one in the vertical or Y position. Axial position measurements the sliding of the shaft along the axis of the shaft. Proximity probes are also used to measure machine speed, by indicating the passing of a notch or key (a bump) in the shaft.

Proximity probes are a bit challenging electrically to NI data acquisition systems. First, the probe driver that provides the signal conditioning is powered with -24V. So a separate power supply is used in order to avoid ground loops. Second, the DC value of the measured displacement signal rides on a -12V DC offset. This requires a wide analog input range from the DAQ device, at least +/-24V.

Transducers: Tachometers

- Uses a wheel with N teeth (N pulses/rev)
- Or a notch in shaft (1 pulse/rev)
- Measure
  - Angular position
  - Angular speed (rpm)
- Used for
  - Order analysis
  - Order tracking
  - Angular averaging
- Sensor Types
  - Proximity Probe, Magnetic Sensor, Optical / Laser Diode



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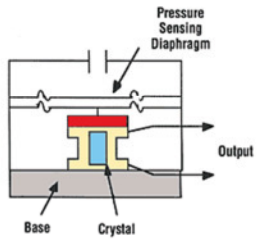
When dealing with rotating machines, it is important to be able to relate any measurement from an accelerometer or a microphone to the actual rotational speed of the machine under test. A tachometer tracks the angular position of a shaft versus time and so lead to the measurement of the angular speed (commonly expressed in rpm, revolutions per minute).

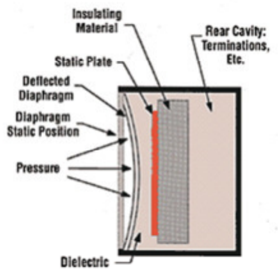
More than providing this angular speed information, the signal from the tachometer is needed to perform advanced analysis such as order tracking, order analysis, and angular averaging.

Dynamic Pressure

- Uses
  - Compressor
  - Gas Turbine
- Characteristics
  - High Speed
  - Pressure Pulsations at Machine Speed
- Piezoelectric types are best for higher frequency pulsations
- www.pcb.com
- www.kistler.com



Piezoelectric Pressure Transducer



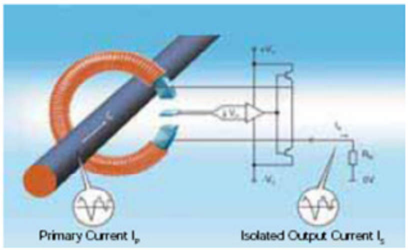
Capacitance Pressure Transduce

In compressor and gas turbine applications, dynamic pressure measurements are used. Dynamic pressure indicates gas turbine combustor health in the case of a gas turbine. In the case of a compressor, it indicates the operational range and performance of the compressor.

Similar to accelerometers, dynamic pressure transducers are available in IEPE style format.

Current Transducer

- Used in electric motor machinery applications
- Motor condition
  - Rotor bar and winding faults



The magnetic flux created by the primary current IP is balanced by a complementary flux produced by driving a current through the secondary windings. A hall device and associated electronic circuit are used to generate the secondary (compensating) current that is an exact representation of the primary current.

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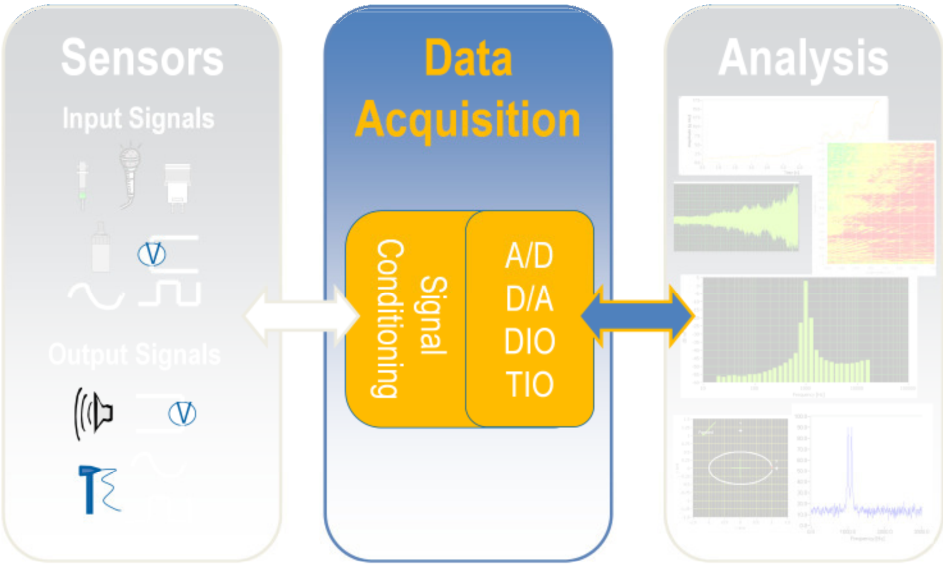
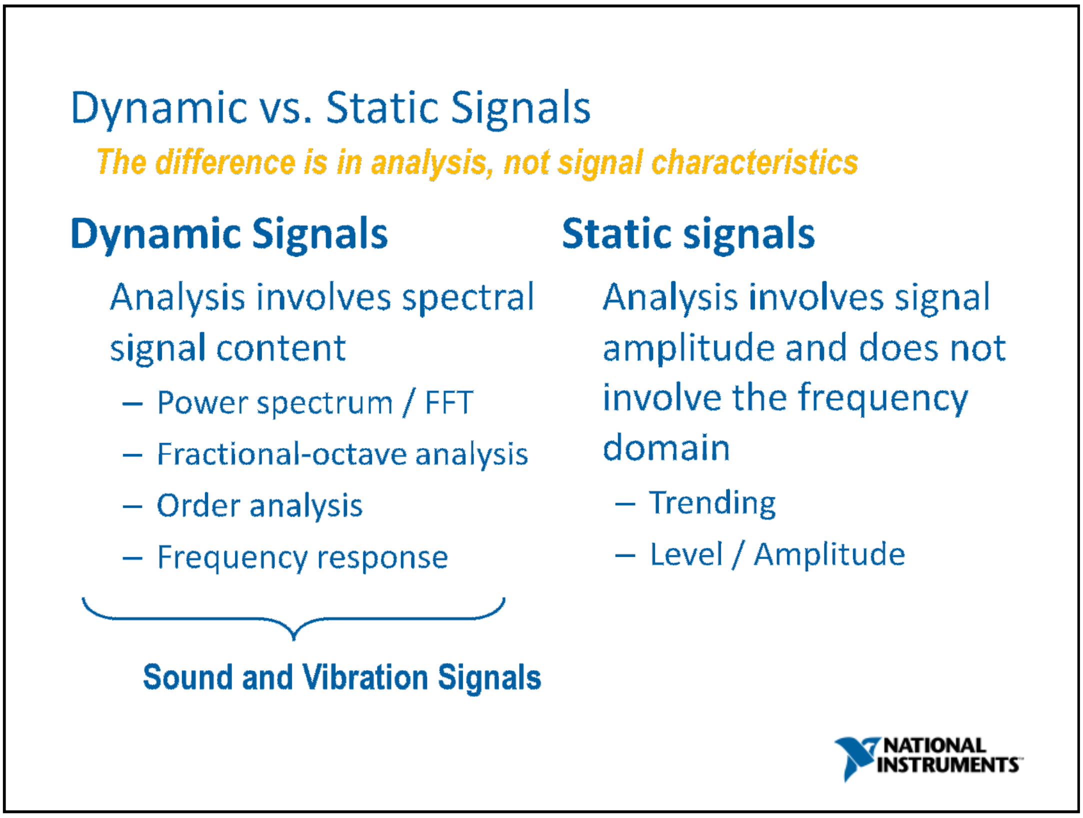
When machines are driven my electrical motors, the electrical motor becomes another mechanical part of the system. Rotor bars, stators, windings, bearings, and other components of the electrical motor can be measured with a current sensor. As mechanical vibrations impact the magnetic flux fields of the electrical motor, the motor current signature carries these variations. A current transducer can measure these electrical fluctuations. The application of motor current signature analysis, leverages this mechanical electrical relationship of the industrial machine.

The current sensor leverages flux fields in current flow and a hall effect device to generate a current or voltage output that is an exact scaled representation of the primary current. Typical outputs of current transducers are +/- 5V.

Sound and Vibration Measurement System Components





The acquisition considerations that we'll discuss apply to dynamic signals, as opposed to static signals. Somewhat surprisingly, the difference between the two doesn't involve a particular signal characteristic—i.e. how rapidly the signal varies. Instead, the difference between static and dynamic signals involves the type of analysis you intend to apply.

Dynamic Signals – those in which you wish to examine the spectral (frequency-domain) content using analysis such as power / phase spectrum, octave analysis, order analysis, frequency response. Examples include sound, vibration, power consumption, and signal level. Engineers often associate dynamic signals with those that vary rapidly with time. While such signals often do so, this isn't always the case. For instance, you can examine the frequency content of a slowly varying signal such as the vibration generated by a slowly rotating drum.

Static Signals – those in which your interest in the signal is in amplitude component and doesn't involve it's spectral content. Analysis for static signals often involves trending or process control.

Understanding these signal viewpoints is important because each has different acquisition requirements. Acquiring a static signal using dynamic hardware is possible, but is often overkill. Acquiring a dynamic signal using a measurement system intended for static signals can lead to misleading results such as the aliasing shown earlier.

Considerations for Acquiring Dynamic Signals

D *High **D**ynamic Range*

S *Multi-channel **S**imultaneous Sampling*

A ***A**nti-aliasing*

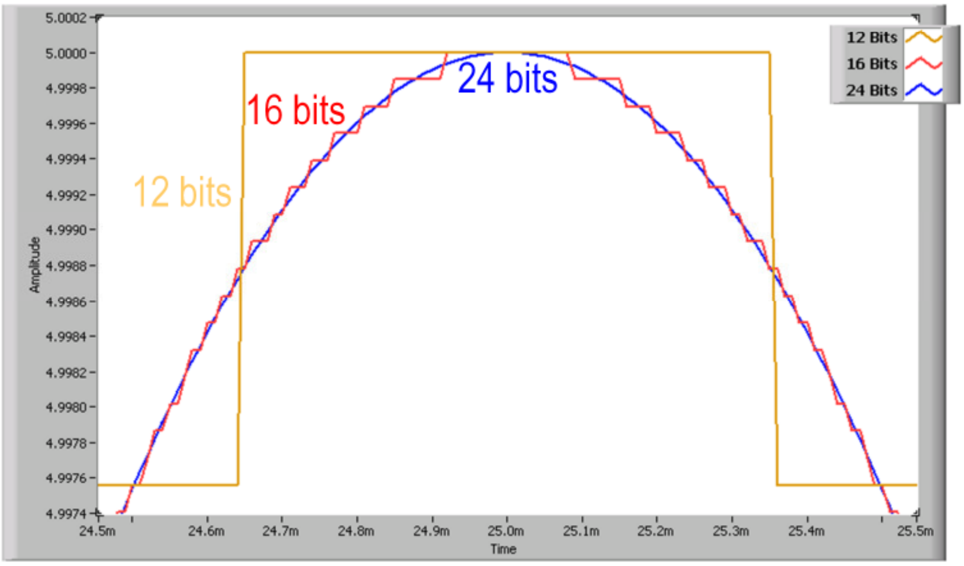
NATIONAL INSTRUMENTS

•With a high dynamic range, you will be able to measure small signal components in the presence of larger signal components. This capability ensures that you won't miss low amplitude components that are sometimes the key to a successful analysis (e.g. predictive maintenance). A common requirement is usually around 90 dB (based on 16-bit converters) with a tendency in this industry to quickly take advantage of the higher dynamic range (around 110 dB) offered by the new 24-bit converters.

•Simultaneous sampling between several channels and boards preserves phase information and reduce measurements uncertanties. This information is useful for applications such as frequency response measurements, order analysis, balancing, and modal analysis.

•Anti-aliasing is required to avoid aliasing when sampling dynamic signals such as sound and vibration. Aliasing is an unavoidable consequence of sampling that can add false signal components to your measurement. Anti-aliasing protection can be offered at fixed sampling frequencies by analog filters or, when simpler analog filters are combined with delta-sigma converter with built-in digital filtering that allows the anti-aliasing protection to follow arbitrary sampling frequencies.

A/D Resolution & Dynamic Range



NATIONAL INSTRUMENTS

With a high dynamic range, you will be able to measure small signal components in the presence of larger signal components. This capability ensures that you won't miss low amplitude components that are sometimes the key to a successful analysis (e.g. predictive maintenance). A common requirement is usually around 90 dB (based on 16-bit converters) with a tendency in this industry to quickly take advantage of the higher dynamic range (around 120 dB) offered by the new 24-bit converters.

Understanding ADC Performance

- Resolution = # of bits the ADC uses to represent a signal
  - Determines how many different voltage changes can be measured:

$$\#\text{ of levels} = 2^{\text{resolution}} = 2^{16} = 65{,}536 \text{ levels}$$

- Code width is the smallest signal change your system can detect
  - Smaller Code Width = more precise representation of your signal

$$\text{code width} = \frac{\text{range}}{\text{amplification} * 2^{\text{resolution}}} = \frac{20\text{ V}}{1 * 2^{16}} = 305\ \mu\text{V} \qquad \textit{16-bit ADC}$$

$$= \frac{20\text{ V}}{1 * 2^{24}} = 1.19\ \mu\text{V} \qquad \textit{24-bit ADC}$$

NATIONAL INSTRUMENTS

As we learned earlier an Analog-to-Digital Converter (ADC) takes an analog signal and turns it into a binary number. Therefore, each binary number from the ADC represents a certain voltage level. The ADC returns the highest possible level without going over the actual voltage level of the analog signal. Resolution refers to the number of binary levels the ADC can use to represent a signal. To figure out the number of binary levels available based on the resolution you simply take $2^{\text{Resolution}}$. Therefore, the higher the resolution, the more levels you will have to represent your signal.

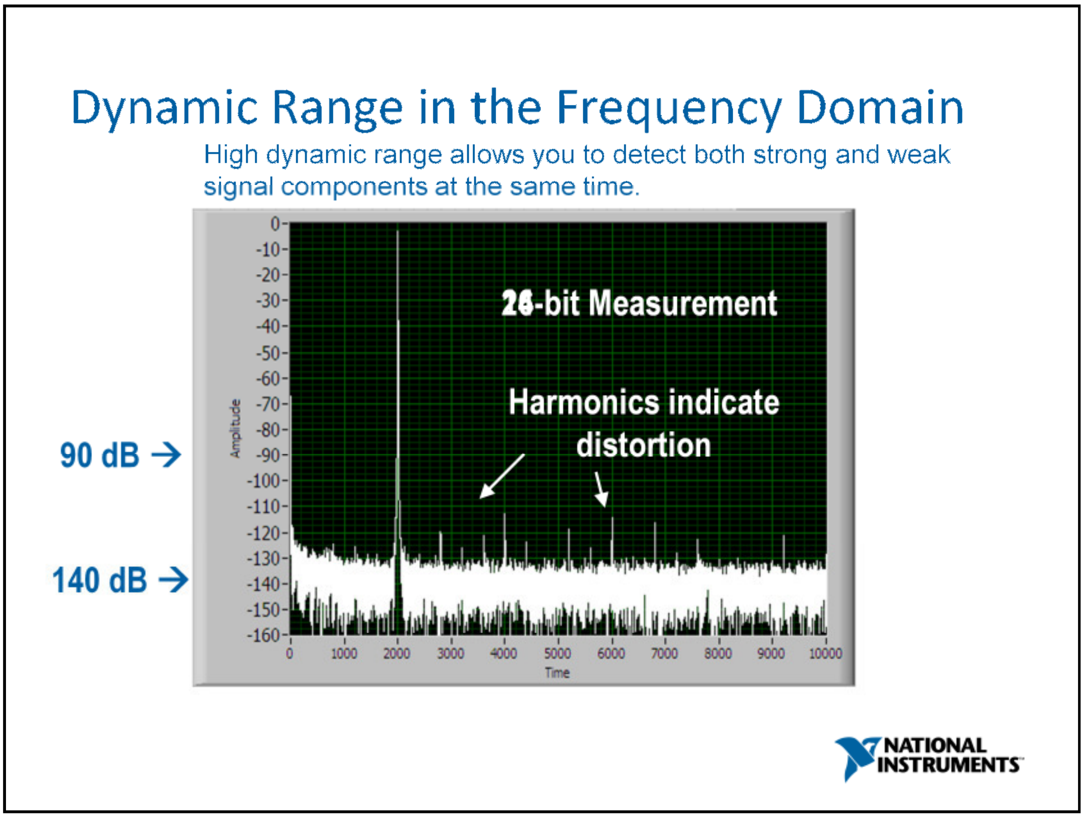
Code width is the smallest change in your signal that your system can detect. The formula for the code width is shown above. As you can see the code width is a property of the resolution, range, and amplification. The smaller our code width is the better we can represent our signal. The formula confirms what we have already learned in our discussion of resolution, range, and gain:

• Larger resolution = smaller code width = better representation of the signal

• Larger gain = smaller code width = better representation of the signal

• Larger range = larger code width = worse representation of the signal

An example is shown above.



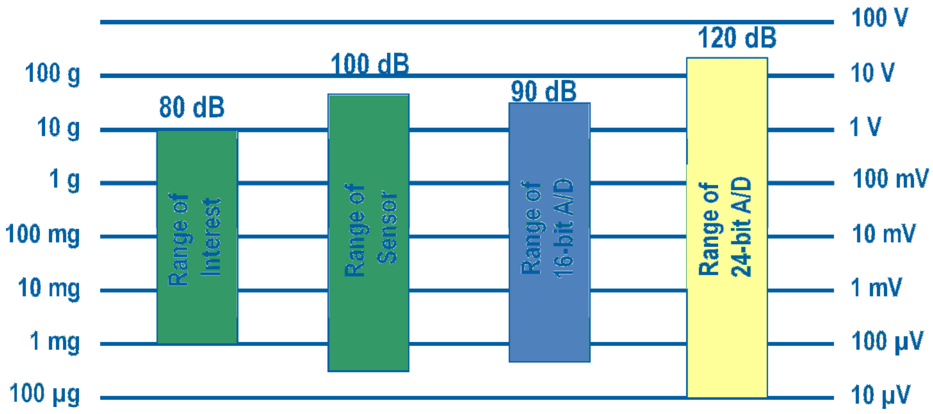
Dynamic range is the ratio (usually expressed in decibels) of the highest and lowest signal amplitudes that can be resolved simultaneously. The 24-bit ADCs on the DSA devices provide excellent dynamic range. This is important to approximating the performance of the human ear for acoustic applications and for resolving very low-level vibrations that may signal a future mechanical failure in machine monitoring applications.

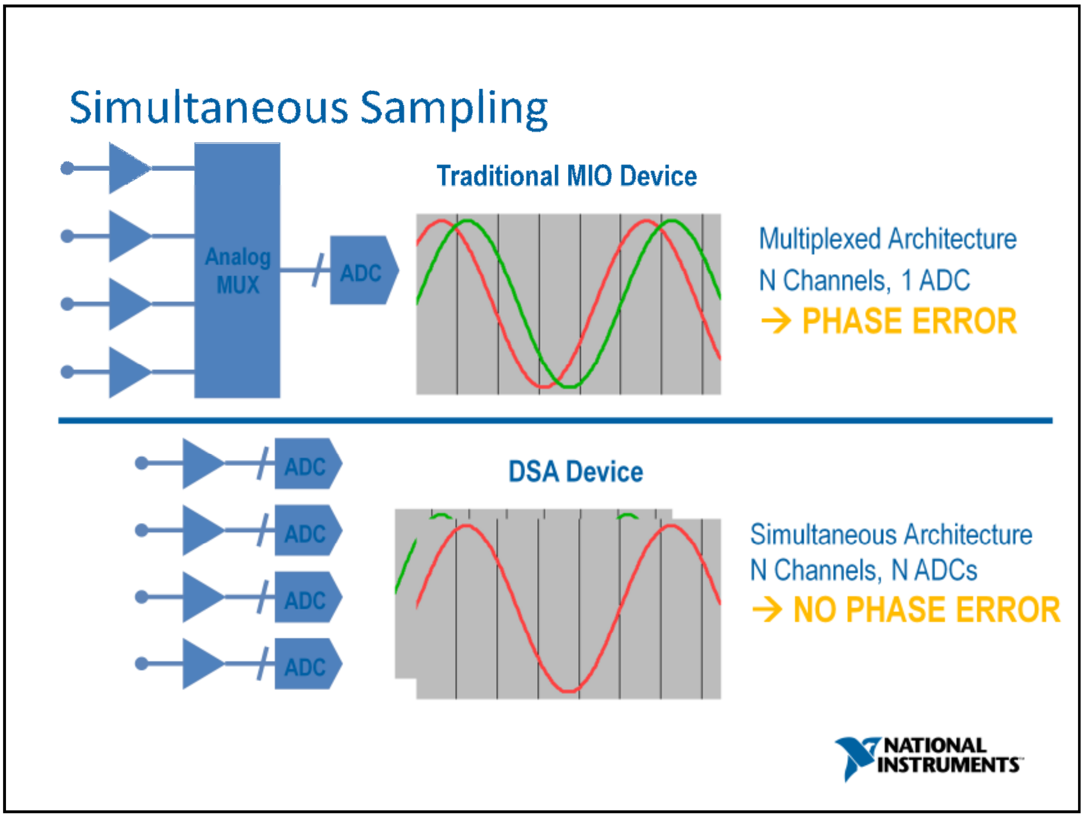
With regards to the human ear, dynamic range is the ability to hear sounds that are both loud and soft in volume at the same time

Usable Dynamic Range

Range of interest : 1 mg to 10 g

Sensor: 100 mV/g



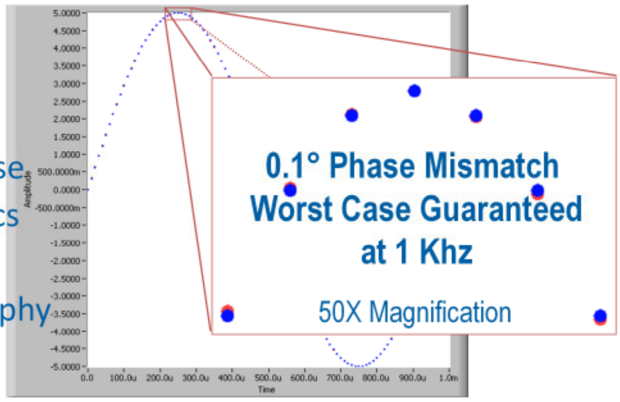


Tight synchronization between all channels is usually a requirement in DSA applications. The reason for this is that phase between channels can be measured accurately only when all channels are simultaneously sampled. Phase relationships are essential to the data processing in most acoustic and structural test applications.

Most MIO devices use a multiplexed architecture for multi-channel acquisitions. In these setup, an analog MUX “moves” the ADC rapidly from one channel to the next. Because the individual channels are not sampled at exactly the same time, this introduces an artificial phase delay when signals are compared between channels. DSA devices (like S-Series devices) use a separate ADC for each input channel to guarantee accurate phase measurements.

Multi-channel Simultaneous Sampling

- A/D conversion is performed at the same instant from 2 to 5,000 channels
- No skew between channels to guarantee phase matching
- Required for
  - Sound intensity
  - Frequency response
  - Structural dynamics
  - Order analysis
  - Acoustical holography



NATIONAL INSTRUMENTS

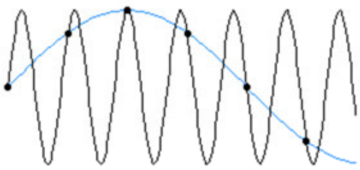
In some applications, such as modal analysis and sound intensity measurements, the inter-channel phase information is crucial. These types of measurements thus require simultaneous sampling, which means that the A/D conversion must be performed at the same instant for every channel.

In the case of Δ/Σ converters, the concept of simultaneous sampling loses its literal meaning, but the phase relationship between the channels is guaranteed by the fact that the converters are run from the same clock.

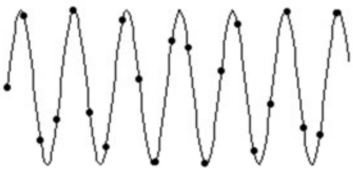
Aliasing

- Shannon sampling theorem
  - The maximum frequency (Nyquist frequency: f_N) that can be analyzed is given by

$f_N = f_s/2$ $\qquad$ f_s: sampling frequency



Improperly sampled



Properly sampled

NATIONAL INSTRUMENTS

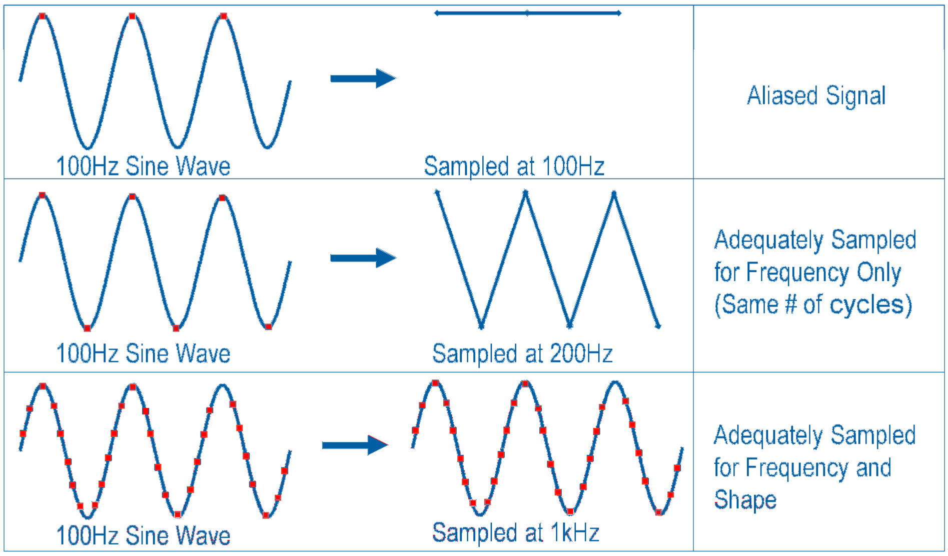
According to Shannon sampling theorem, the highest frequency (Nyquist frequency: f_N) that can be analyzed is $f_N = f_s/2$, where f_s is the sampling frequency. Any analog frequency greater than f_N will, after sampling, appear as a frequency between 0 and f_N. Such a frequency is known as an “alias” frequency. In the digital (sampled) domain, there is no way to distinguish these alias frequencies from the frequencies that actually lie between 0 and f_N. Therefore, these alias frequencies need to be removed from the analog signal before sampling by the A/D converter.

Nyquist Theorem

- You must sample at greater than 2 times the maximum frequency component of your signal to accurately represent the FREQUENCY of your signal.

- NOTE: You must sample between 5 - 10 times greater than the maximum frequency component of your signal to accurately represent the SHAPE of your signal.

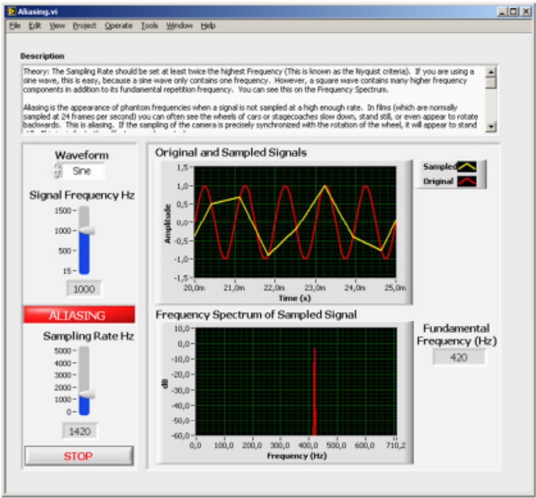
Nyquist Example



NATIONAL INSTRUMENTS

Assume you are measuring a 100Hz sine wave. First we will try sampling our signal at exactly 100Hz. Keep in mind that the signals shown above are theoretical approximations. It is often very difficult for both the signal and this sampling rate to be at exactly the same frequency. According to the Nyquist Theorem 100Hz is not fast enough to correctly represent the frequency of our signal. If our signal frequency is exactly 100Hz and we are measuring at exactly 100Hz we will get a straight line. So we are obviously not correctly representing either the shape or the frequency of our signal. Therefore the Nyquist Theorem and our guideline for shape both hold true. Keep in mind that the signals shown above are theoretical approximations. It is often very difficult for both the signal and the sampling rate to be at exactly the same frequency. Now let us sample our signal at 200Hz. Note that this is exactly twice the frequency of our signal, so according to the Nyquist Theorem this is just fast enough to correctly represent the frequency of our signal. However, it is not fast enough to correctly represent the shape of our signal. If our signal is exactly 100Hz and if we sample at exactly 200Hz we will get the triangle wave shown above. Notice that the 100Hz sine wave and the triangle wave have different shapes, but the same frequency. So the Nyquist Theorem and our guideline for shape still hold true. Finally, we will sample our signal at 1kHz. Since we are sampling at 10 times our frequency, we should be able to accurately represent both the frequency and shape of our signal. As you would expect, our sampled signal does look like a sine wave, and it has the same frequency as our measured signal. So the Nyquist Theorem and our guideline for representing the shape of our signal both hold true.

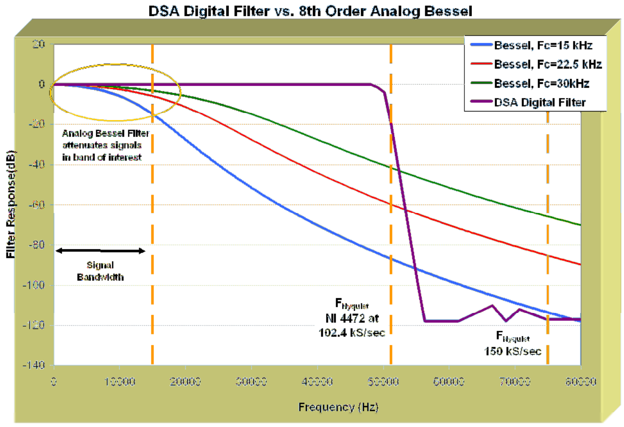
Exercise 3 – Alaising



Open and run Exercise 3.

Anti-Aliasing Filter

- Removes frequency components higher than Nyquist frequency
- Needs to be performed before signal is digitized
- Mix of Analog and Digital filters
- Required characteristics:
  - Flat in-band frequency response
  - Good high frequency alias rejection
  - Fast roll-off in the transition band

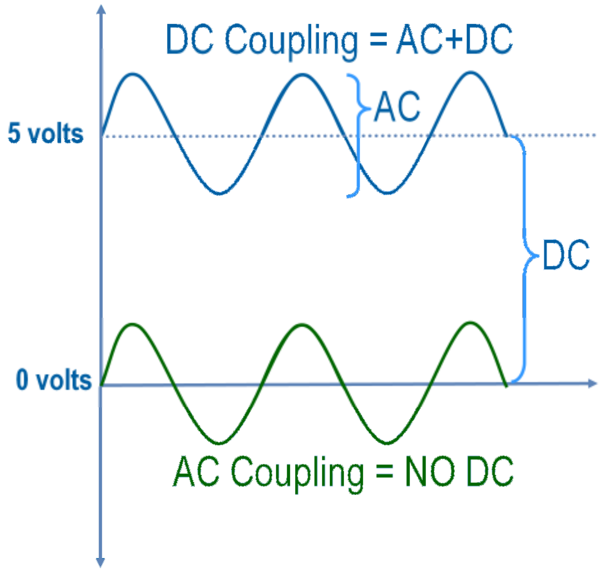


NATIONAL INSTRUMENTS

To remove these components present at higher frequencies than the Nyquist frequency, an analog lowpass filter must be used. This anti-aliasing filter should exhibit a flat in-band frequency response with a good high frequency alias rejection and a fast roll-off in the transition band.

Additional Signal Conditioning

- DC and AC Coupling
  - AC Coupling = AC
  - DC Coupling = AC+DC
- IEPE = Integrated Electronic Piezoelectric
  - Constant current power supply



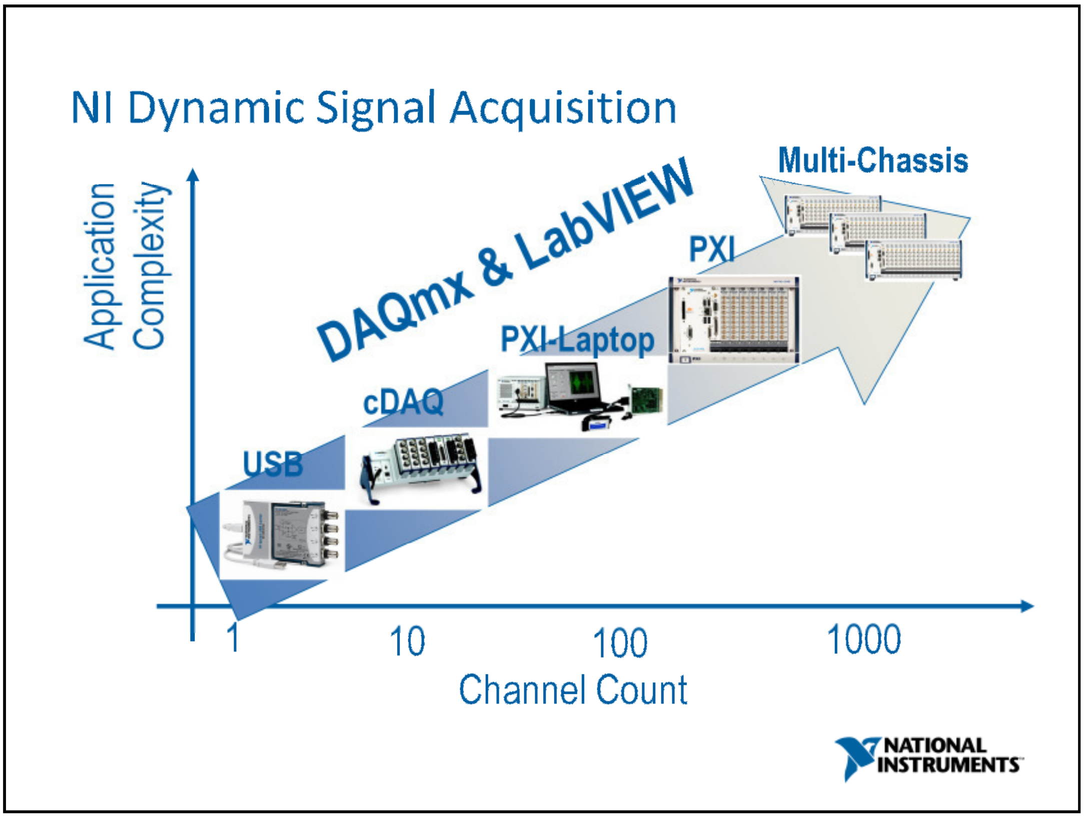
NATIONAL INSTRUMENTS

AC coupling consists of using a capacitor to filter out the DC signal component from a signal with both AC and DC components. The capacitor must be in series with the signal. AC coupling is useful because the DC component of a signal acts as a voltage offset, and removing it from the signal can increase the resolution of signal measurements. AC coupling is also known as capacitive coupling.

DC coupling describes any voltage signal acquisition in which both AC and DC components are measured.

IEPE refers to a type of transducers that are packaged with a built-in charge amplifier or voltage amplifier. IEPE is an acronym for Integral Electronic Piezoelectric. Because the charge produced by an accelerometer is very small, the electrical signal produced by the transducer is susceptible to noise, and sensitive electronics must be used to amplify and condition the signal. IEPE makes the logical step of integrating the sensitive electronics as close as possible to the transducer to ensure better noise immunity and more convenient packaging.

IEPE technology is marketed under several more recognizable brand names. Although other companies make IEPE based accelerometers, this does not imply that they are compatible with equipment that claims Integrated Circuit Piezoelectric compatibility. One of the key variations from company to company is the amount of current necessary to power the sensor.


NI Dynamic Signal Acquisition
Application Complexity
DAQmx & LabVIEW
Multi-Chassis
PXI
PXI-Laptop
cDAQ
USB
1
10
100
1000
Channel Count
NATIONAL INSTRUMENTS

USB-9233/9234

- **Maximum portability**
  - No external power (USB bus powered)
- **Capabilities**
  - 4 simultaneous inputs
  - 24-bit A/D
  - 102 dB dynamic range
  - Up to 51.2 kS/s/ch sample rate
  - +/- 5V input range

- **Signal conditioning**
  - IEPE and AC-coupled*
  - Continuously variable anti-aliasing filters

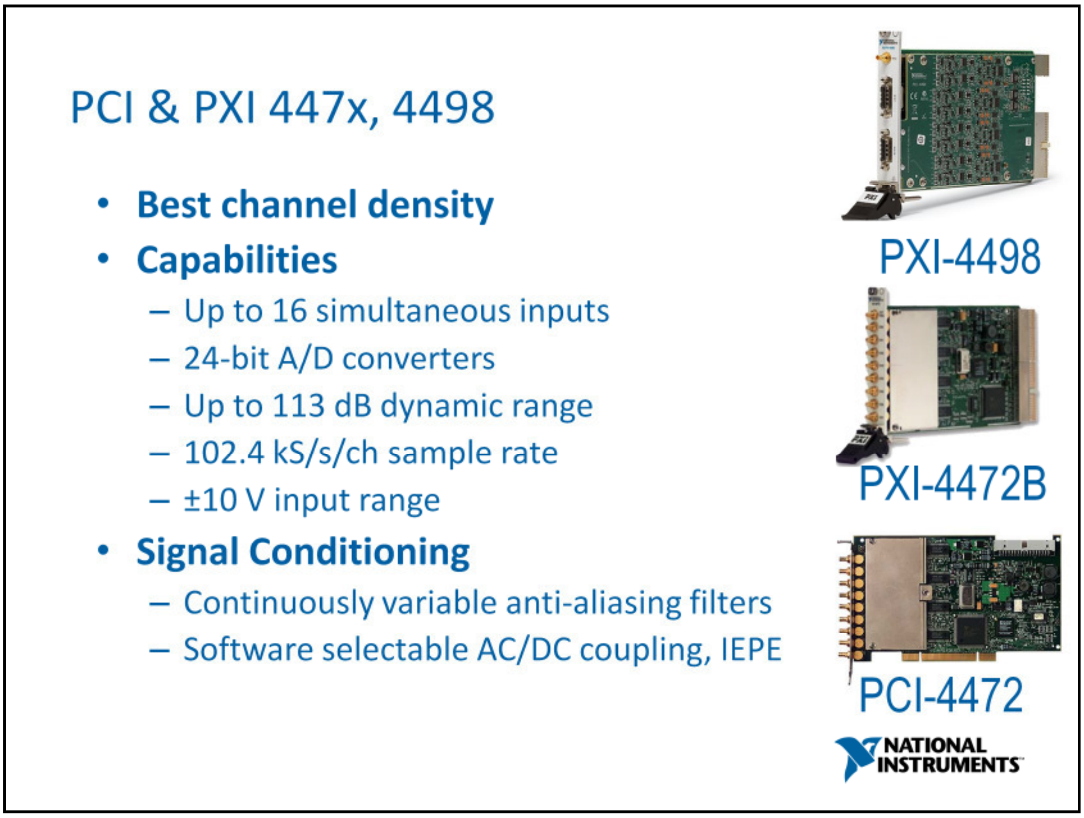
*software selectable

NATIONAL INSTRUMENTS

CompactDAQ for Sound & Vibration

- **Silent, portable, DC powered**
- **Capabilities**
  - Up to 32 simultaneous inputs
  - 24-bit A/D converters
  - 102 dB dynamic range
  - 50 kS/s/ch sample rate
  - +/- 5V input range
- **Signal conditioning**
  - Always-on IEPE and AC-coupled
  - Continuously variable anti-aliasing filters

NATIONAL INSTRUMENTS



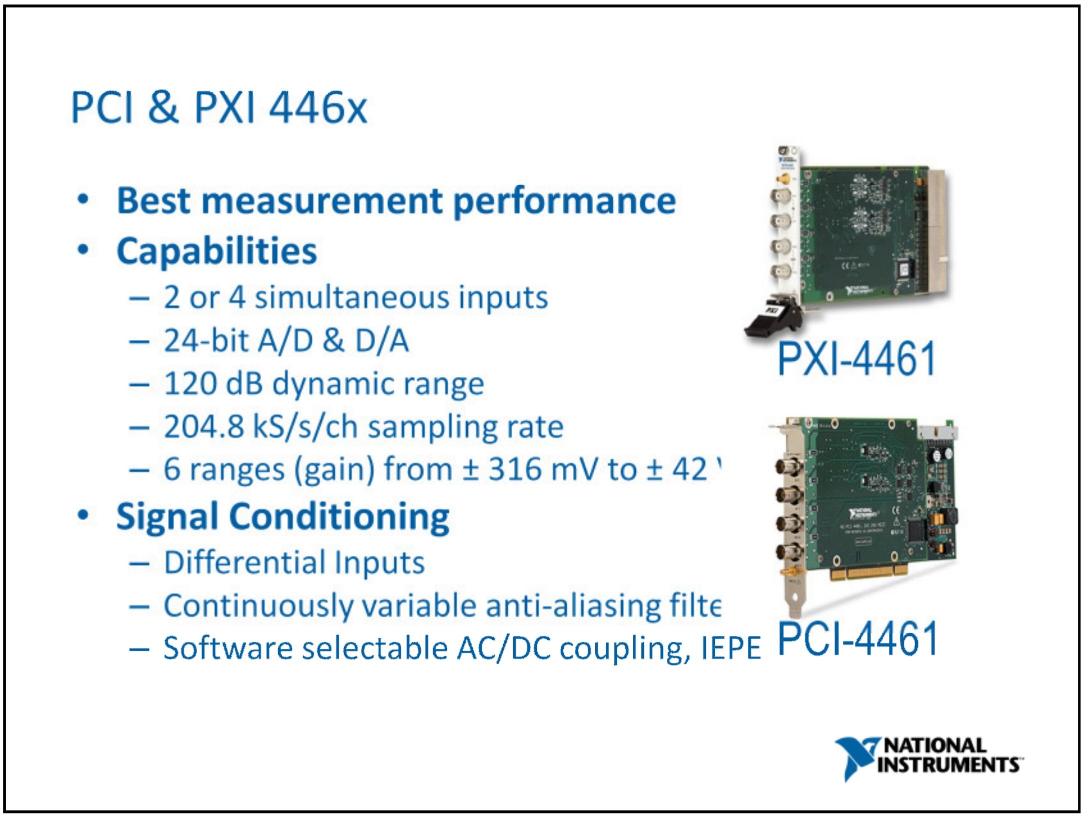
The National Instruments NI 4472 is an 8-channel and the NI 4498 a 16-channel dynamic signal acquisition module for making high-accuracy frequency domain measurements. The input channels of the NI PXI-4472 simultaneously digitize input signals over a bandwidth from DC to 45 kHz. You can synchronize NI PXI-4472/4498 modules with each other for high-channel-count applications, or with other modules using the PXI star trigger bus. When used with the sound and vibration toolset or other software analysis tools, the NI PXI-4472/4498 can obtain a variety of accurate time and frequency measurements for your application.

NI 4472

- 8-channel dynamic signal acquisition module
- 102.4 kS/s maximum sampling rate, simultaneously sampled
- 110 dB dynamic range
- 45 kHz alias-free bandwidth
- 24-bit resolution
- Vibration optimized version with a high pass filter of 0.5 Hz to be introduced

NI 4498

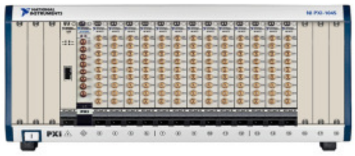
- 16 simultaneously sampled analog inputs at up to 204.8 kS/s
- 24-bit resolution ADCs with 113 dB dynamic range
- 4 gain settings up to +30 dB for input ranges from ±316 mV to 10 V
- Software-configurable 4 mA IEPE and TEDS for microphones and accelerometers
- Variable antialiasing filters
- AC-coupled analog inputs at 0.5 Hz



With the PXI-4461, NI delivers a powerful dynamical signal acquisition device for high precision audio measurements. The 4461 has 2 simultaneously sampled input channels that digitize signals from DC to 92kHz with anti-aliasing protection and balanced differential inputs. Input gains enable measurement ranges from +/-316mV to 42V. The 4461 provides IEPE conditioning for microphones and accelerometers and uses sigma-delta technology to deliver better than 120dB dynamic range.

The PXI-4461 also provides 2 output channels that generate frequencies up to 92 kHz at 110dB dynamic range.

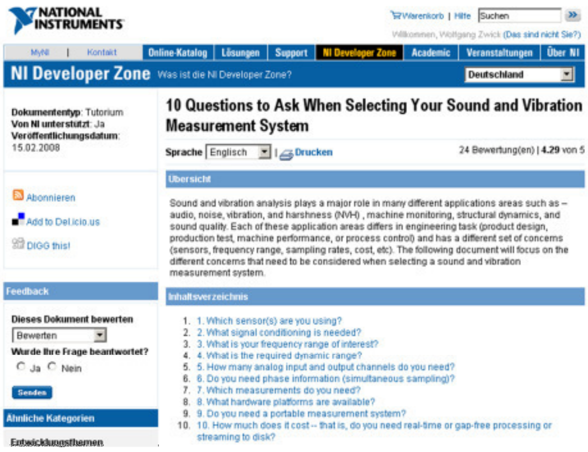
PXI Platform for Sound & Vibration



- **Up to 112 synchronized channels in a single chassis**
  - < 0.1 degree phase mismatch at 1 kHz
- **Mixed signal inputs**
  - Accelerometers, microphones
  - Tachometers, proximity probes
  - Thermocouple, strain gauges, etc
- **Embedded or laptop controlled**

NATIONAL INSTRUMENTS

Selecting Your Sound and Vibration Measurement System



NATIONAL INSTRUMENTS

Warenkorb | Hilfe | Suchen

Willkommen, Wolfgang Zwick (Das sind nicht Sie?)

MyNI | Kontakt | Online-Katalog | Lösungen | Support | NI Developer Zone | Academic | Veranstaltungen | Über NI

NI Developer Zone — Was ist die NI Developer Zone? — Deutschland

Dokumententyp: Tutorium
Von NI unterstützt: Ja
Veröffentlichungsdatum: 15.02.2008

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Entwicklungsthemen

10 Questions to Ask When Selecting Your Sound and Vibration Measurement System

Sprache Englisch | Drucken — 24 Bewertung(en) | **4.29** von 5

Übersicht

Sound and vibration analysis plays a major role in many different applications areas such as – audio, noise, vibration, and harshness (NVH) , machine monitoring, structural dynamics, and sound quality. Each of these application areas differs in engineering task (product design, production test, machine performance, or process control) and has a different set of concerns (sensors, frequency range, sampling rates, cost, etc). The following document will focus on the different concerns that need to be considered when selecting a sound and vibration measurement system.

Inhaltsverzeichnis

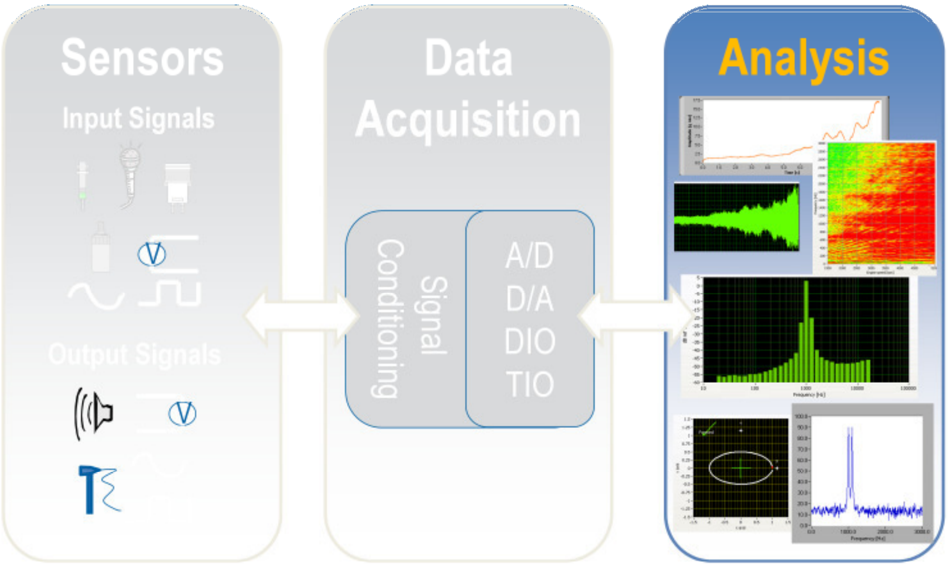
1. 1. Which sensor(s) are you using?
2. 2. What signal conditioning is needed?
3. 3. What is your frequency range of interest?
4. 4. What is the required dynamic range?
5. 5. How many analog input and output channels do you need?
6. 6. Do you need phase information (simultaneous sampling)?
7. 7. Which measurements do you need?
8. 8. What hardware platforms are available?
9. 9. Do you need a portable measurement system?
10. 10. How much does it cost -- that is, do you need real-time or gap-free processing or streaming to disk?

http://zone.ni.com/devzone/cda/tut/p/id/2955

NATIONAL INSTRUMENTS

If you need help to selecting your Sound and Vibration Measurement System check Tutorium on http://zone.ni.com/devzone/cda/tut/p/id/295.

Sound and Vibration Measurement System Components



NATIONAL INSTRUMENTS

Analysis

Acquisition:

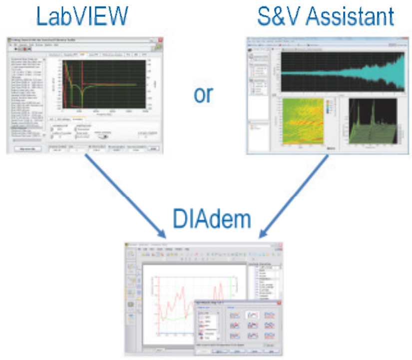
- LabVIEW or
- Sound and Vibration Assistant

Analysis online:

- LabVIEW or
- Sound and Vibration Assistant

Analysis offline, Data Navigation and Reporting:

- DIAdem



NATIONAL INSTRUMENTS

Use LabVIEW or Sound & Vibration Assistant for Acquisition, Data Logging and Online Analysis. For Offline Analysis, Data Navigation and Reporting use DIAdem.

Analysis Funktions in LabVIEW and Sound and Vibration Assistant

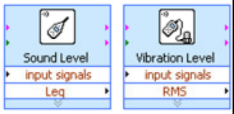
	NI Sound and Vibration Toolkit		NI Sound and Vibration Measurement Suite	
	NI Sound and Vibration Assistant	LabVIEW Analysis VIs	NI Sound and Vibration Assistant	LabVIEW Analysis VIs
Interactive analysis	✓	–	✓	–
Code generation	✓	–	✓	–
UFF58 file I/O	✓	✓	✓	✓
FFT analysis	✓	✓	✓	✓
Zoom FFT	✓	✓	✓	✓
Subset FFT	–	✓	–	✓
Frequency response function (FRF)	✓	✓	✓	✓
Peak search	✓	✓	✓	✓
Sound level	✓	✓	✓	✓
Vibration level	✓	✓	✓	✓
1/n octave	✓	✓	✓	✓
Weighting filters	✓	✓	✓	✓
Integration	✓	✓	✓	✓
Harmonics	✓	✓	✓	✓
Distortion (THD, THD + N, SINAD)	✓	✓	✓	✓
Intermodulation distortion (IMD)	✓	✓	✓	✓
Tone detection	✓	✓	✓	✓
Spurious-free dynamic range (SFDR)	✓	✓	✓	✓
Crosstalk	✓	✓	✓	✓
SNR	✓	✓	✓	✓
Swept sine	–	✓	–	✓
Limit testing	✓	✓	✓	✓
Pink noise	–	✓	–	✓
Shock response spectrum (SRS)	–	✓	–	✓
Short-time Fourier transform (STFT)	–	✓	–	✓
Human vibration filters	–	–	–	✓
Torsional vibration	–	–	–	✓
Tachometer processing	–	–	✓	✓
Order power spectrum	–	–	✓	✓
Order tracking	–	–	✓	✓
Order extraction	–	–	✓	✓
Angular resampling	–	–	–	✓
Envelope detection	–	–	–	✓
Waterfall plot	✓	✓	✓	✓
Spectral map	–	–	✓	✓
Bode plots	✓	✓	✓	✓
Orbit and shaft centerline plot	–	–	–	✓

NATIONAL INSTRUMENTS

Sound & Vibration Analysis

- Time Domain Measurements
  - RMS Level
  - Sound Level
- Frequency Domain Measurements
  - Power Spectrum
  - Harmonics & Octave Analysis
  - Frequency Response / Swept Sine
- Order Analysis

Level Measurements

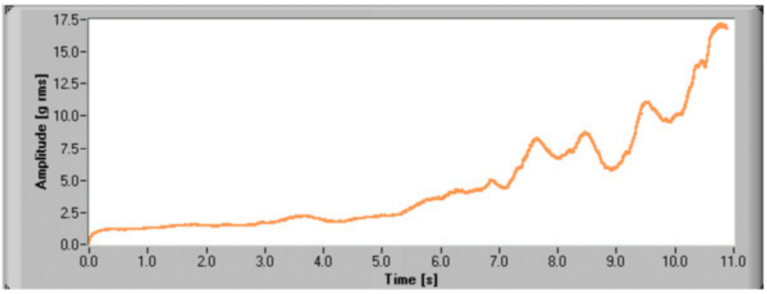


Sound Pressure Level

- Logarithmic scale (dB)
- 20 uPa reference
- Frequency weighting: A, B, C

Vibration Level

- g rms, g peak
- 1 ug reference
- Integration: velocity & displacement



Perhaps the most basic measurement analysis related to sound and vibration is level. Root Mean Square (RMS) gauges the energy content of a dynamically varying signal.

A common sound level measurement is sound pressure level. This value is always expressed relative to a reference pressure of 20 micropascals, the threshold of human hearing, using logarithmic scaling based on the formula mentioned on this slide.

With sound level measurements, there is often some effort made to account for the for the frequency response of human hearing. Simply stated, the human ear does not hear all frequencies equally. With this in mind, some standard filters account for the response of the ear.

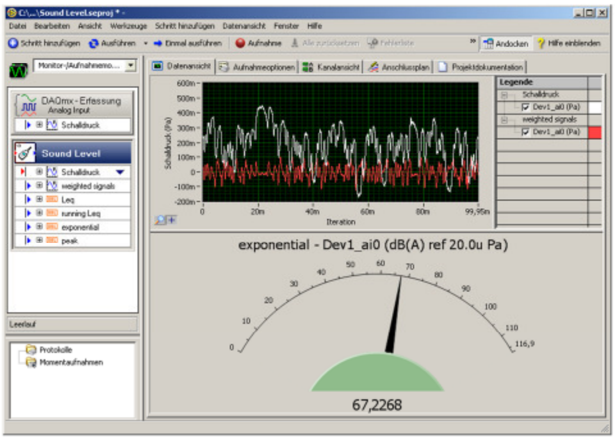
Vibration level measurements report on linear or logarithmic scales. If a vibration level reports in dB, the reference is 1 micro-g.

In the context of vibration measurement, you might measure a DC/RMS level on a proximity probe. Such a measurement could assist applications including:

•If you are installing a proximity probe by adjusting the distance of the tip of the probe to a shaft, the DC/RMS value will vary with the tip distance. Proximity probe manufacturers will specify a valid range of voltages in which the output of the proximity probe will be linear. You should work to ensure that the bias falls within that range by adjusting the tip distance.

•On a calibrated proximity probe, the DC/RMS value multiplied by the probe sensitivity will dictate the average distance of the probe tip to the shaft.

Demo – Sound Level



NATIONAL INSTRUMENTS

Remeber Exercise 1.

Open Solution >> Exercise 1 >> Sound Level.seproj

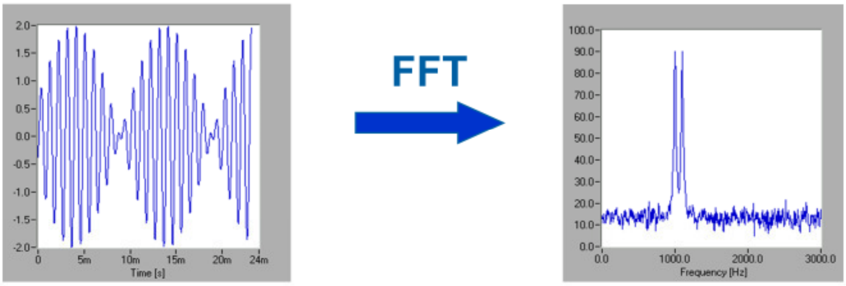
or

Open Example Finder in LabVIEW

Help >> Example Finder >> Toolkits and Modules >> Sound and Vibration >> Analyzing and Processing Signals >> Sound Level Measuremens >> SVXMPL_Sound Level Meter.vi

Frequency Analysis

- The Fast Fourier Transform (FFT) converts the time-domain signal into the frequency domain



Time-Domain Signal

Spectrum

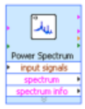
Unlike octave analysis, frequency analysis examines the spectrum based on a set of linearly-spaced frequency “lines” (also called “bins”). Such analysis is commonly implemented on a computer using an FFT (Fast Fourier Transform) algorithm. FFT algorithms are an efficient means of applying a DFT (Discrete Fourier Transform) and convert a set of time-domain samples into frequency-domain samples.

The use of frequency analysis implies two important relationships:

The first relates the highest frequency that can be analyzed to the sampling frequency (Nyquist theorem).

The second links the frequency resolution to the total acquisition time, which is related to the sampling frequency and the block size of the FFT. This means that better frequency resolution can only be achieved by acquiring data for a longer period of time. This also tells us that if the frequency content of a signal is changing with time, there will always be a compromise between frequency resolution and time resolution (improving one will deteriorate the other).

Baseband Frequency Analysis



Analysis	Results	Phase Data
Power spectrum	Computes the power present within each spectral bin	All phase information is lost
Power spectral density	Computes the power present within each bin normalized by the bin width	All phase information is lost
FFT spectrum	Computes either the magnitude and phase or the real and imaginary parts of the spectrum	Phase information is retained depending on the averaging mode

NATIONAL INSTRUMENTS

You can perform the following single-channel measurements with the Frequency Analysis VIs:

• **Power spectrum** computes the power present within each spectral bin. All phase information is lost in the computation. This is a useful tool for examining the various frequency components in a signal.

• **Power spectral density** computes the power present within each bin normalized by the bin width. All phase information is lost in the computation. This is a useful tool for examining the noise floor in a signal or the power in a specific frequency range. Normalizing the power spectrum by the bin width decouples the result of this measurement from the block size *N*.

• **FFT spectrum** computes either themagnitude and phase or the real and imaginary parts of the spectrum of the input signal. Phase information is retained depending on the selected averaging mode. This measurement is most often used by more advanced measurements that require magnitude and phase information.

Frequency Analysis Considerations

1. The highest frequency that can be analyzed is dictated by the Nyquist theorem

Highest analysis frequency

$$F_{\max} = \frac{f_s}{2}$$

2. Frequency resolution is related to acquisition time, which is related to sampling frequency and block size

Frequency Resolution

$$\Delta f = \frac{1}{T} = \frac{f_s}{N}$$

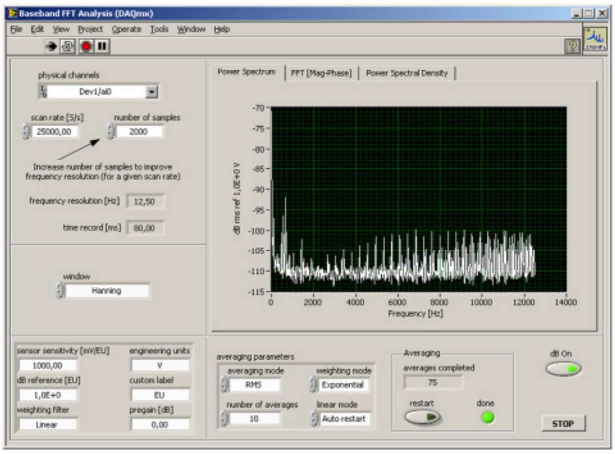
NATIONAL INSTRUMENTS

The use of frequency analysis implies two important relationships:

The first relates the highest frequency that can be analyzed to the sampling frequency (Nyquist theorem).

The second links the frequency resolution to the total acquisition time, which is related to the sampling frequency and the block size of the FFT. This means that better frequency resolution can only be achieved by acquiring data for a longer period of time. This also tells us that if the frequency content of a signal is changing with time, there will always be a compromise between frequency resolution and time resolution (improving one will deteriorate the other).

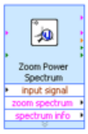
Demo – Baseband FFT



Open Example Finder in LabVIEW

Help >> Example Finder >> Toolkits and Modules >> Sound and Vibration >> Analyzing and Processing Signals >> Frequency (FFT) Measuremens >> SVXMPL_Baseband FFT.vi

Zoom Frequency Analysis



- Achieves arbitrarily fine frequency resolution
- Decouples number of lines in FFT from block size
  - data is accumulated until enough samples have been acquired to compute zoom spectrum

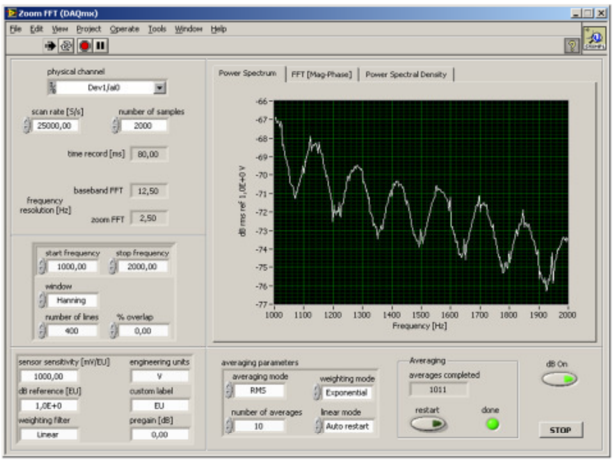
Frequency Resolution

$$\Delta f = \frac{1}{T} = \frac{f_s}{N}$$

In some applications, you need to obtain the spectral information over a limited portion of the baseband span and with greater resolution. In other words, you need to zoom in on a spectral region to observe the details of that spectral region. You can use the zoom FFT to obtain spectral information over a limited portion of the baseband span and with greater resolution. Just as in baseband analysis, the acquisition time determines the frequency resolution of the computed spectrum. The number of samples used in the transform determines the number of lines computed in the spectrum.

Zoom FFT analysis achieves a finer frequency resolution than the baseband FFT. The Zoom FFT VI acquires multiple blocks of data and downsamples to simulate a lower sampling frequency. The block size is decoupled from the achievable frequency resolution because the Zoom FFT VI accumulates the decimated data until the required number of points are acquired. Because the transform operates on a decimated set of data, you only need to compute a relatively small spectrum. Because the data is accumulated, do not think of the acquisition time as the time required to acquire one block of samples. Instead, the acquisition time is the time required to accumulate the required set of decimated samples.

Demo – Zoom FFT

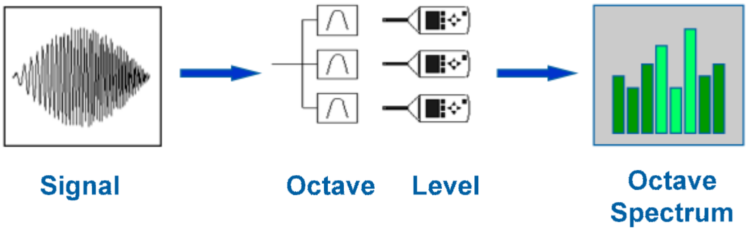


Open Example Finder in LabVIEW

Help >> Example Finder >> Toolkits and Modules >> Sound and Vibration >> Analyzing and Processing Signals >> Frequency (FFT) Measuremens >> SVXMPL_Zoom FFT.vi

Fractional Octave Analysis

- Used when final customer is the human ear
  - amplitude scale in dB
  - logarithmic frequency scale
- Implementation: Bank of bandpass filters followed by level measurements



- Compliance with ANSI and IEC standards

NATIONAL INSTRUMENTS

In many cases, when dealing with sound measurement and analysis, the final “customer” is the human ear. And like most human senses, the ear exhibits a response based on a logarithmic scale for both the level and the frequency. So, to produce results that are somewhere related to this human perception, sound levels are expressed in decibels and frequency content measured with a logarithmic scale. A lot of research efforts are currently focused on this field of psycho-acoustics, but octave band analysis remains a technique of prime choice.

One octave band corresponds to the frequency range between two frequencies with a ratio of 2 to 1. A typical example of this is a piano keyboard, where two consecutive C tones are exactly separated by an octave. The reference center frequency for the audio range is at 1 kHz. Other midband frequencies below 1 kHz are thus at 500 Hz, 250 Hz, …, and above 1 kHz at 2 kHz, 4 kHz, …

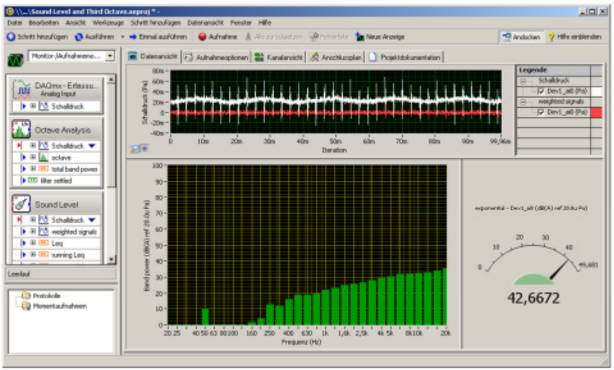
Octave band analysis is performed through a parallel bank of bandpass filters. The time-domain output of each filter is then averaged to compute the power in each band and displayed as a bar graph.

With octave analysis, the audio frequency range (roughly 20 Hz to 20 kHz) will be covered with 10 filters. This offers only a limited frequency resolution. In order to improve frequency resolution, more filters can be used per octave, leading to fractional-octave analysis. If 3 filters are used to cover a single octave (instead of one), this analysis is then called 1/3 octave. Other typical numbers of filters per octave are 6, 12, and 24 for 1/6, 1/12, and 1/24 octave analysis.

Fractional-octave analysis is defined by 2 major standards (ANSI S1.11 and IEC 1260) and is used by numerous standards (sound power measurements, reverberation time, noise attenuation properties, …).

Octave analysis is mainly used for acoustical measurements when the noise source has a broadband spectrum. Spectral analysis (FFT) is generally preferred when the signal exhibits some components at specific frequencies (prominent “tones”).

Exercise 4, Demo – Third Octave Analysis



Accomplish Exercise 4.

Additional you can open examples in LabVIEW Example Finder.
Help >> Example Finder >> Toolkits and Modules >> Sound and Vibration >> Analyzing and Processing Signals >>
Octave Measurement >>

•SVXMPL_Present Octave Results.vi

•SVXMPL_Third-Octave Analysis.vi

Typical Octave and Third Octave Band Filter Applications.

In particular these filters are clearly used to identify the frequency content of the noise and to help you to control and reduce the noise. Here are some examples:

•Hearing protection - the Octave Band spectrum is used for calculating the level at the ear when wearing hearing protectors.

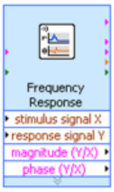
•Noise reduction and control - identifying the problem areas and then focusing the control on those levels.

•Building Acoustics

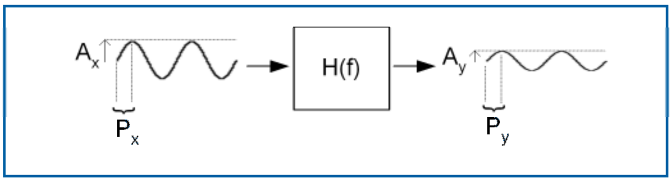
•Machine and product testing

Frequency Response

Stimulus-Response



- The Frequency Response Function (FRF) characterizes the UUT
- The FRF compares the amplitude and phase of the stimulus and response at various frequencies



Stimulus Types

Broadband (FFT-based)

- Excitation includes all frequencies at once
- Shorter test time
- Examples:
  - White noise
  - Chirp
  - Pink noise
  - Impact hammer

Discrete

- Excitation consists of only one frequency at a time
- Generally longer test time (allows for settling & averaging)
- Can be used to measure frequency-dependent distortion
- Examples:
  - Single tone
  - Multi-tone sine
  - Swept sine or stepped-frequency sine

NATIONAL INSTRUMENTS

Broadband measurements include all frequencies of interest in a single block of data which is analyzed using an FFT (Fast Fourier Transform). Separate FFTs are performed on the stimulus and response signals. All the excitations from the previous slide **except** the stepped frequency sine are used with broadband frequency response measurements. The major advantage of this type of FRF measurement is that you can analyze all frequencies simultaneously, helping to reduce test time.

Stepped-Frequency measurements take longer than FFTs because a separate measurement must be performed for each excitation frequency. However, they can provide distortion measurements in addition to a simple FRF measurement. Harmonic distortion is very important when characterizing the performance of amplifiers. Most of FFT rules apply for this measurement.

Two of the most common stimulus-response measurements are Broadband Frequency Response (FRF) and Swept Sine.

Frequency response is defined as the gain and phase response of a circuit or other unit under test (UUT) at all frequencies of interest. Although the formal definition of frequency response includes both the gain and phase, in common usage, the frequency response often only implies the magnitude (gain).

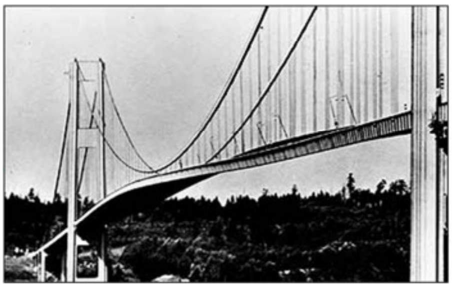
Broadband Frequency Response measurements require the excitation of the UUT with energy at all relevant frequencies. The fastest way to perform the measurement is to use a broadband excitation signal that excites all frequencies simultaneous, and use FFT techniques to measure at all of these frequencies at the same time. Noise and non-linearity is best minimized by using random noise excitation, but short impulses or rapid sweeps (chirps) may also be used.

Swept Sine Analysis provides an advanced method to measure the frequency response of a speaker, amplifier, filter, or other DUT. Swept sine tests are sometimes employed in mechanical vibration tests using shaker tables.

A swept sine test works by stepping through individual discrete excitation tones and measuring both the DUT stimulus and response signals. Unlike a broadband measurement, it allows for settling time at each individual frequency and for extensive averaging, if desired. It can also measure THD as a function of frequency. The only disadvantage of Swept Sine as compared to a traditional broadband measurement is that it is generally slower because each frequency is generated and acquired individually.

Testing the Physical Model

- Looking for resonant frequencies
- Information gained:
  – Where/how to mount a component
  – What frequencies to avoid exciting
    - Increase stiffness
    - Additional mass



- Inject stimulus energy and look at the response

- Hammer/Impact	- Accelerometer
- Wind (Wind Tunnel)	- Microphone

NATIONAL INSTRUMENTS

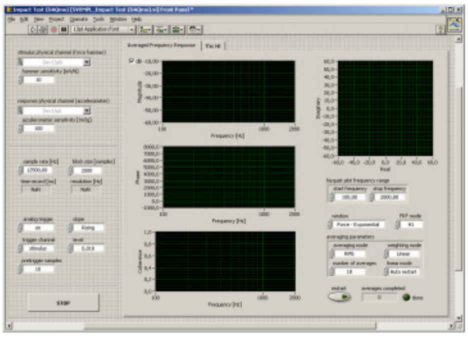
Analogy to Electrical Systems

Mass = Inductor

Spring = Capacitor

Damping = Resistance

Demo - Impact Test with Impact Hammer



The first channel should use a sensor mounted on an impact hammer. The second channel should use an accelerometer mounted on the structure under test.

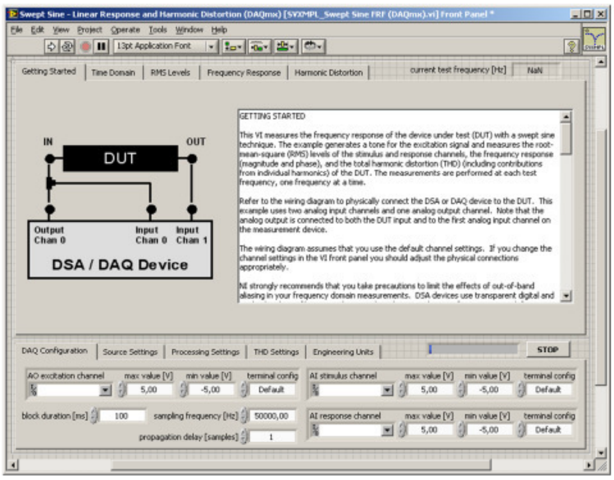
The data from each channel is windowed and the frequency response is computed. The results of the frequency response measurement are displayed as magnitude and phase. The coherence is also graphed to evaluate the validity of the measured frequency response.

NATIONAL INSTRUMENTS

Help >> Example Finder >> Toolkits and Modules >> Sound and Vibration >> Analyzing and Processing Signals >> Frequency Response Measurements

• SVXMPL_Impact Test (DAQmx).vi

Demo – Swept Sine FRF

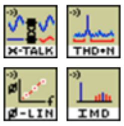


Help >> Example Finder >> Toolkits and Modules >> Sound and Vibration >> Analyzing and Processing Signals >> Swept Sine Measurements

- SVXMPL_ Swept Sine FRF.vi
- SVXMPL_Getting Started with Swept Sine.vi
- SVXMPL_Getting Started with Swept Sine Express VI.vi

These VIs measures the frequency response of the device under test (DUT) with a swept sine technique. The example generates a tone for the excitation signal and measures the magnitude and phase response of the DUT. The frequency response is measured at each test frequency, one frequency at a time.

Audio Analysis



- Distortion (non-linearity)
  - THD, THD+Noise, SINAD: Measured with a pure tone (fundamental)
    - Total Harmonic Distortion = Distortion / Fundamental
    - SINAD = Signal / (Noise + Distortion)
      Signal = Fundamental + Noise + Distortion
  - IMD (Intermodulation Distortion): dual tone excitation
- Crosstalk
- Gain/phase

NATIONAL INSTRUMENTS

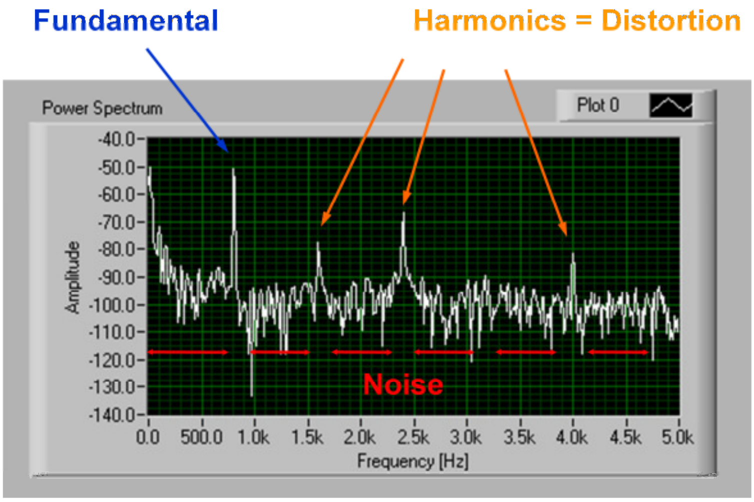
•Phase Linearity: In a perfectly linear system, the phase response will be a exactly linear with frequency (the Bose plot for phase response will be straight line). This VI quantifies how the phase response differs from the linear ideal. The measurement is important in characterizing analog filters.

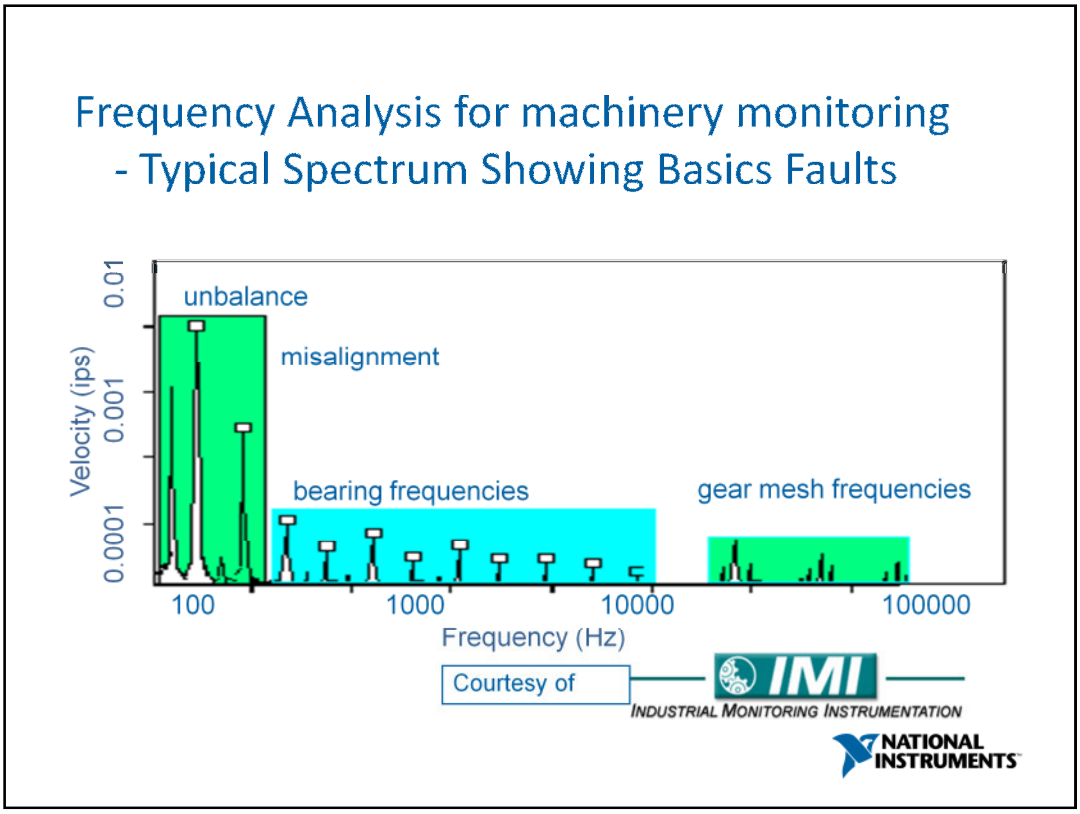
•THD: THD (total harmonic distortion) analyzes the non-linear harmonics produced in response to a single excitation tone. It is important in characterizing speakers and amplifiers.

•IMD: Intermodulation Distortion, or IMD, analyzes the harmonics produced by non-linear interaction of two excitation tones. It is useful in characterizing amplifiers, especially in analyzing the distortion characteristics at high frequencies.

•Crosstalk measures the inductive coupling between two channels on an amplifier or other circuit.

Distortion Measurements





The core analysis for machinery monitoring vibration signatures is Frequency Analysis.

Here is the same summary plot, yet with colored groupings of key frequencies and magnitudes. An important note with this slide is that there are many features in a single vibration signature. From low frequency high amplitude vibrations to high frequency low amplitude vibrations.

A wide dynamic range data acquisition board does the best job in digitizing vibration signals such that the high frequency and low amplitude signatures can be captured in the presence of low frequency high amplitude signatures. A single device makes this cost effective.

NI Dynamic Signal Acquisition devices, DSA, provides the needed High Dynamic Range.

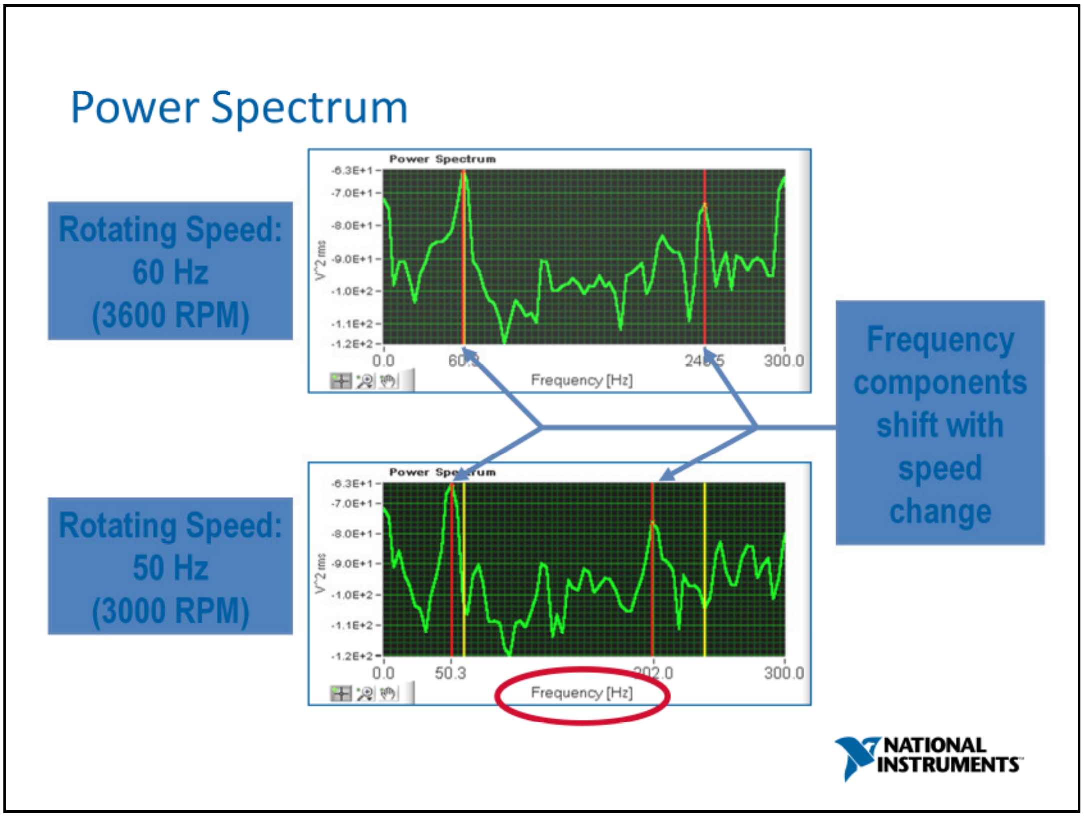
Order Analysis

- For analysis of mechanical systems with **rotating** components:
  - Engines, gearboxes, transmissions, motors, rotors, turbines, pumps, compressors
- Many noise and vibration signal components are directly related to running (**rotating**) speed:
  - Imbalance, misalignment, gear mesh, bearing defects, loose coupling
- Order analysis normalizes the measurements to the **rotating** speed to better dissect these signal components

NATIONAL INSTRUMENTS

Order analysis is similar to frequency analysis, in that it examines the frequency content of signals. But in the case of order analysis the frequency content is viewed as harmonics of a fundamental frequency, the rotational speed of the system under investigation. Thus an order is nothing else than a harmonic of the rotational speed of a system (fundamental).

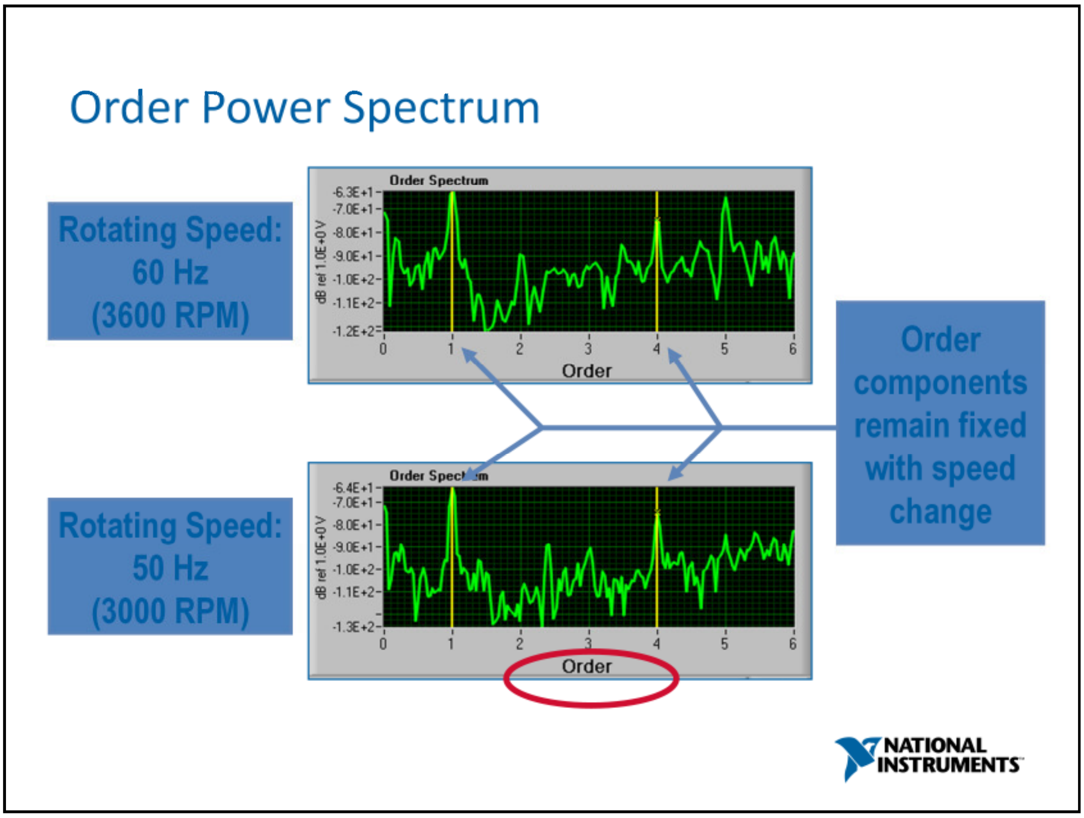
In other words, instead of looking at signals in terms of frequency, order analysis examines them in terms of orders. Orders are signal components that are generated by the rotating or reciprocating components of mechanical systems. This makes order analysis the first choice for the analysis of rotating or reciprocating machines. If a particular problem (such as a high level of vibration or noise) can be related to a constant order, then you know that this problem has something to do with a rotating part of the machine. Usually, knowing which particular order is involved will also help you to track down the problem (for example it could be that the shaft is unbalanced, first order, or misaligned, second order). If the problem is always appearing at a constant frequency, assuming that you change the rotational speed, then you will know that the source is to be found among non-rotating parts (a good example in this case would be a resonance frequency of the case of the machine).



The rotational speeds of machine components correspond to the periodic forcing frequencies that make up sound and vibration signals. As such, **changing the rotational speed will shift the component frequencies.** Nonetheless, we're still interested in following the level of a particular peak, even during speed changes. Changes in peak level that occur at different speeds might indicate vibration associated with a resonance (of rotating components in this case).

The graphs show the vibration spectra from a motor running at 60 [rps] and 50 [rps]. On the top graph, the two red cursors show the peaks associated with the primary rotational frequency and with an event occurring four times per revolution (approx 240 [rps]). On the 50 [rps] graph, the red cursors show the location of the primary and four-event peaks for this speed. A frequency shift is clear when you compare these peaks to those of the 60 [rps] case, which are designated with the yellow cursors.

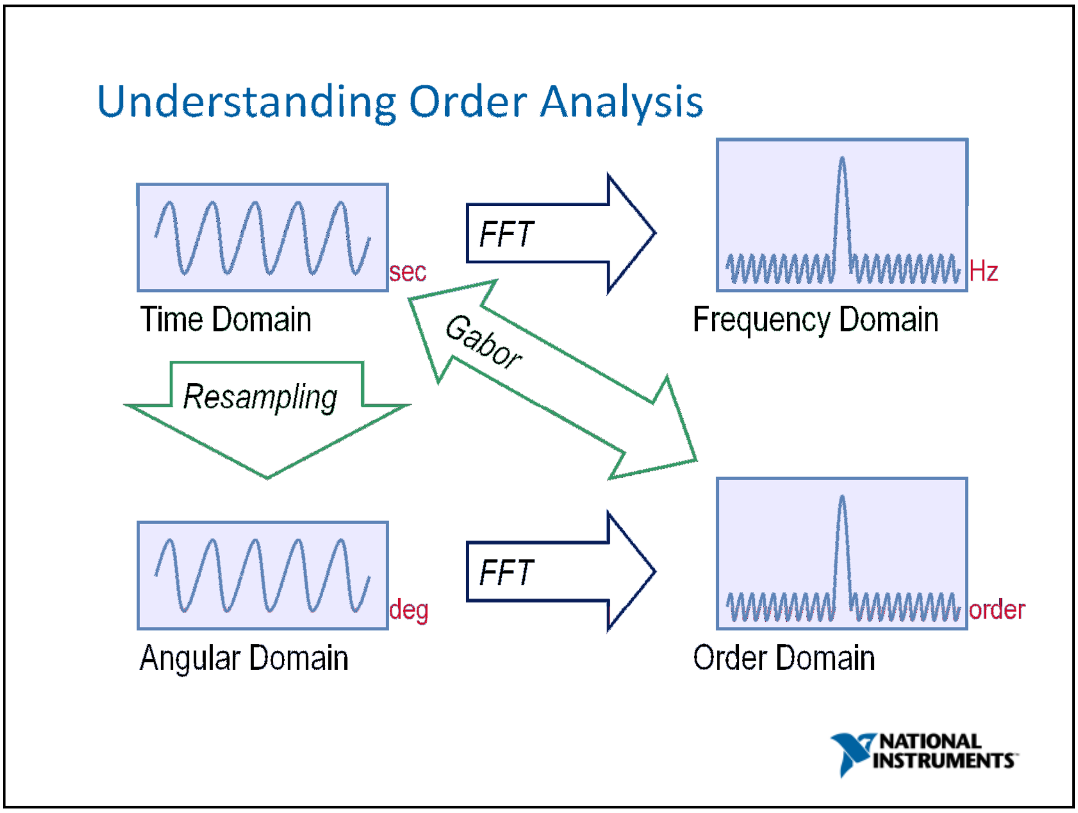
Another example would be a 4-cylinder engine during a run-up test. The combustion process is a major source of noise and vibration for this engine. Because this combustion process occurs twice per revolution, we see a strong contribution in the second order. At 3,000 rpm the second order corresponds to 100 Hz, but at 6,000 rpm it will show up at 200 Hz. Being able to track the level associated with the second order will tell you immediately how the combustion process is contributing to the overall level.



Because of the relationship between the frequency components of interest and rotational frequencies, it's useful to normalize the frequency axis to the rotational speed. To do so, we divide the scale of that axis by the periodic forcing frequency.

The two graphs above show the renormalized versions of the frequency domain graphs from the previous slide. Notice that now the peaks remain in the same x-location on the graph, despite the change in rotational speed. In the top graph, the x-location labeled 1 (first order) corresponds to about 60 [Hz] and the 4 (fourth order) corresponds to about 240 [Hz] (4)(60). With it's 50 [rps] rotational speed, the 1 and 4 on the bottom graph correspond to 50- and 200-Hz, respectively.

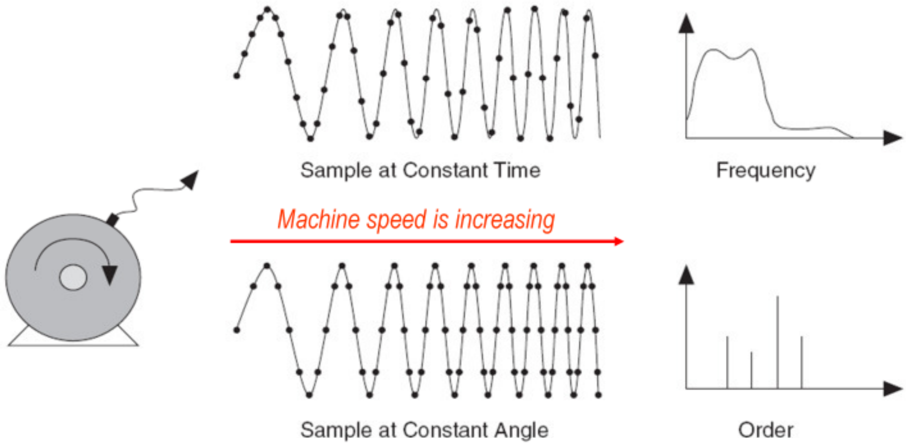
Order Analysis involves examination of a signal in terms of angular frequency, rather than in Hertz (Hz), which is the typical scale for spectral analysis. Instead of looking at how many times per second an event happens, you look at how many times per revolution this event happens. Just as the definition of a Hertz is "per second," angular frequency is "per revolution."



In both time domain and frequency domain analysis, the FFT is the workhorse of analysis. With order domain analysis, the FFT is done on angular based data. This provides the benefit that data is provided in integer number of cycles and can be done without windowing. In order to convert data to the angular domain, a phase and frequency reference is required to indicate the beginning and end of a cycle.

Resampling for Changing Machine Speed

- Data is converted to angular domain from time domain with the use of a tachometer (speed) signal



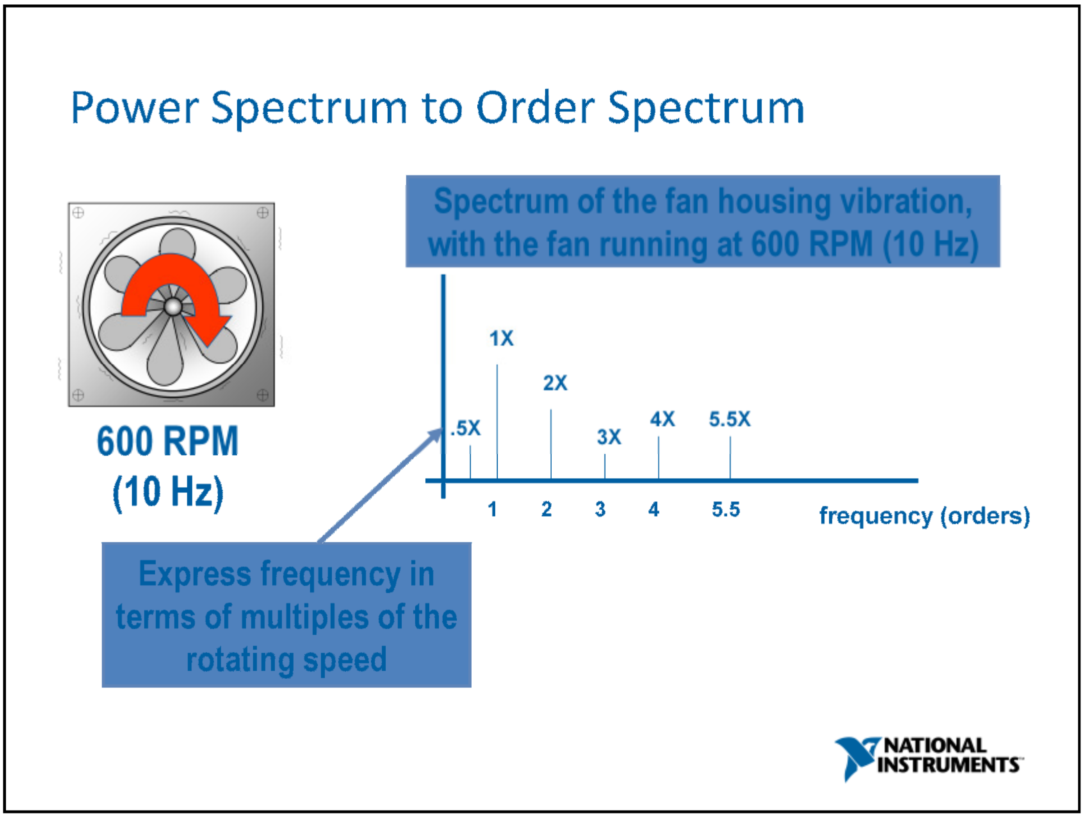
NATIONAL INSTRUMENTS

Resampling converts from time domain into angular domain.

Example: if you want to analyze up to 5 orders, you require 2*HighestOrder ≈ at least 10 samples per cycle. (Note: the 2 is the Nyquist multiplier.)

The point of resampling will force there to be a fixed amount of data per cycle. Hence, the points will always be separated by a fixed angle.

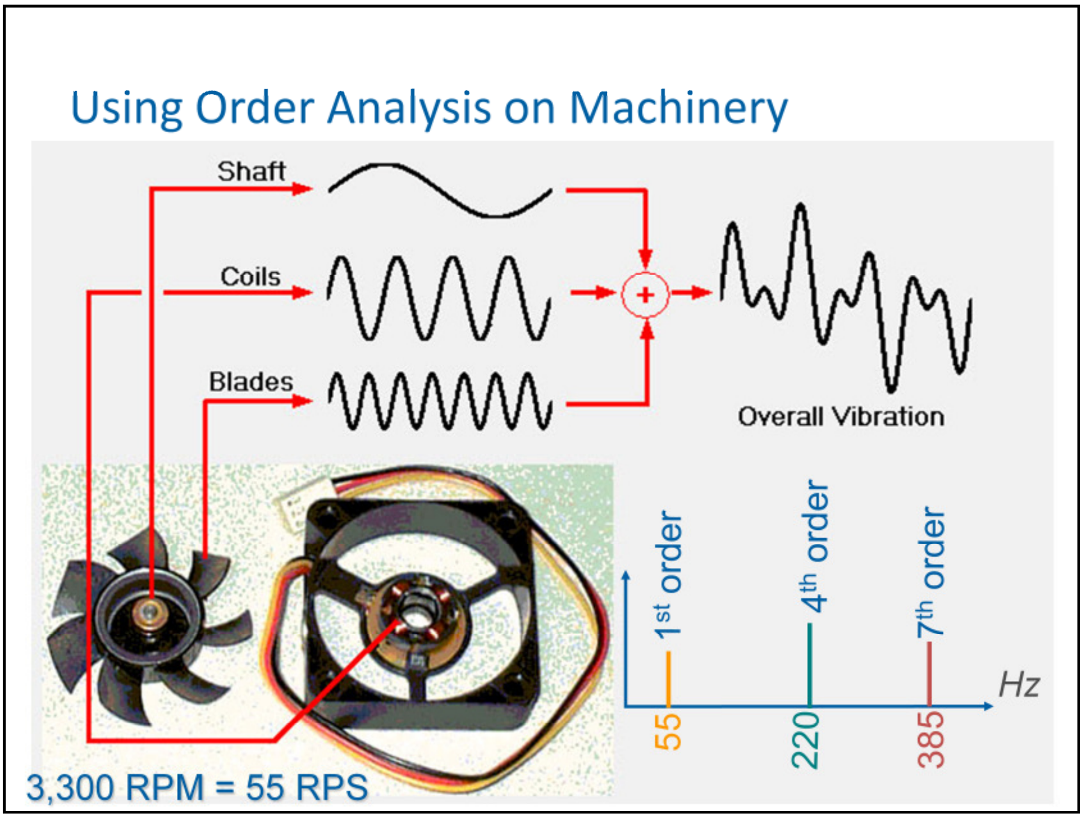
You must sample at least as fast as the highest RPM * the highest order * 2.56. At this high speed, you will not need to adjust anything. At the lower speeds, the resampling algorithms will remove unnecessary points, only maintaining the points at the required angular locations.



Another extension to the FFT is Order Analysis. With order analysis, the rotational speed of the machine is used to normalize the x-axis of the resulting FFT to show harmonics or rotation or angular position rather than harmonics of time, or frequency.

In this example, the tachometer signal us used to identify the rotational speed of the fan, and to normalize the frequency axis of the resulting FFT.

A more complete order analysis technique is to transform the data from the time domain to angular domain with a tool called Resampling. With Resampling, any FFT skew in the graph is removed by analyzing equal angular sampled data. This signal processing technique is very common for machinery which changes speed such as variable speed drives, turbo machinery, and so on.



Lets look at a quick example:

When we acquire the vibration of a PC cooling fan. The major sources include:

1. Imbalance of the rotor. It generates the vibration at the frequency of the rotating speed. This is often called it the 1st order harmonic of rotating speed.
2. Electric-magnetic force from the coil. This fan has 4 coils, which drives the fan 4 times per revolution. So the vibration contributed by coil is at the frequency 4 times the rotating speed, so we call it the 4th order.
3. 7 blades of the fan. They generates the vibration at 7th order.
4. The harmonics of E-M force, 8th, 12th, and so on orders.

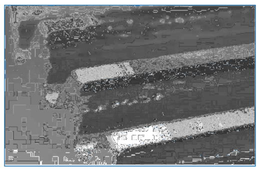
So if the machine is running at 3300 RPM (55Hz), we would set our sampling rate at 11 times 55 times 12 or 7260 samples per second.

Relationship of Orders and Faults

- We can diagnose machine faults by knowing the order:
  - Imbalance
  - Misalignment
  - Loose Coupling
  - Valve Noise
  - Bearing Defects / Wear
  - Blade Pass Frequency
  - Gear Mesh

Low Order

High Order

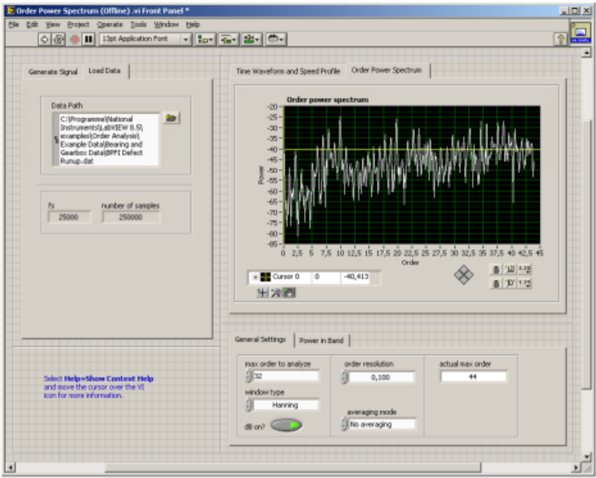


NATIONAL INSTRUMENTS

Useful analysis starts with an understanding of the relationship between the signal components and the corresponding physical system.

Machine operation often involves repetitive movement and rotation of interacting components. Motion of shafts, gears, couplings, fan blades, magnets, coils, and other elements generate **forced vibration**, which occurs periodically at **forcing frequencies**. For instance, a shaft turning an unbalanced mass will generate a forcing frequency equal to the rate of rotation. Machine components that have coupled motion will have related forcing frequencies.

Demo – Order Power Spectrum



You can find examples for Order

Help >> Example Finder >> Toolkits and Modules >> Sound and Vibration >> Analyzing and Processing Signals >> Order Analysis

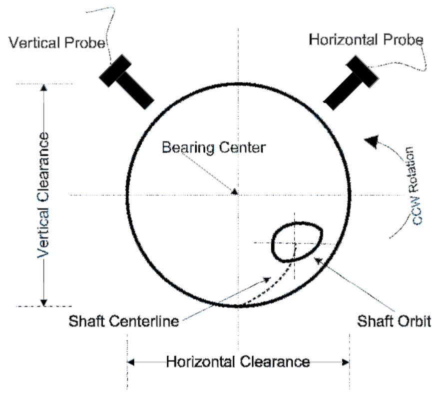
•Order Power Sepctrum (Analog Tacho, DAQmx).vi (use this example with Sound and Vibration Signal Simulator)

•Order Power Spectrum (Offline).vi (use this example if you do not have the Sound and Vibration Signal Simulator).

In Exercise 6 you will programm an Order Analysis with LabVIEW Express VIs .

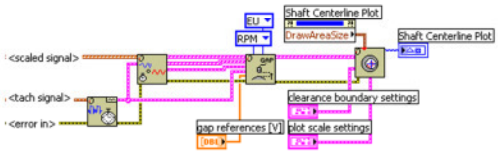
Shaft Measurement Plots

- Plots include:
  - Orbit Plot
  - Timebase Plot
  - Shaft Centerline Plot
- Measured by two orthogonally placed, DC coupled proximity probes.
- Need to use 10dB attenuation cable with 447x

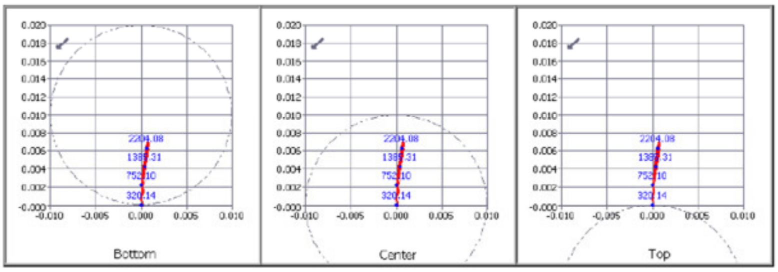


Some common machine vibration plots include vibration associated with shaft lateral movements. These vibration show the rotation of the center of mass and the location of the shaft within its bearing housing.

Shaft Centerline Plot

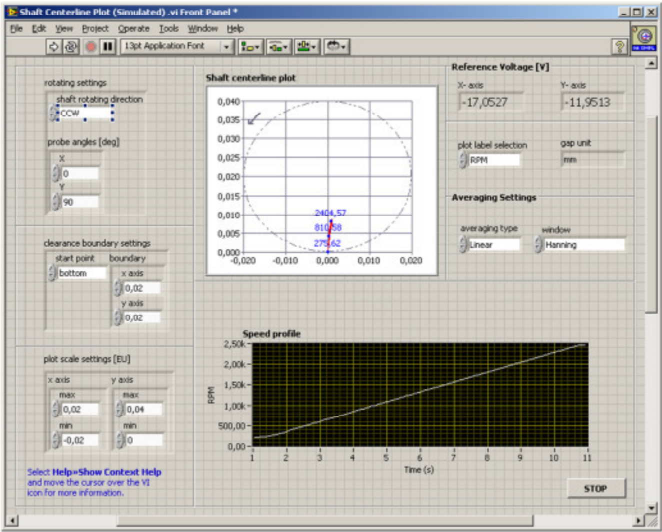


- Compute gap reference first
- Bearing clearance start point
  - Bottom
  - Center
  - Top



The shaft centerline plot indicates the movement of the shaft from start-up to running state in the machine.

Demo - Shaft Centerline and Polar Plot



Help >> Example Finder >> Toolkits and Modules >> Sound and Vibration >> Graphs and Charts

• Shaft Centerline Plot (Simulated) .vi

• Polar Plot (Analog Tacho, DAQmx) .vi

What is Machine Monitoring?

- Predict maintenance requirements and protect machines
  - Collecting machine electrical and mechanical signatures to analyze maintenance requirements in a predictive nature
- Machine and Process Performance
  - Collecting process information to provide insight to process optimization opportunities
- Other names
  - Machine Condition Monitoring (MCM), Vibration Monitoring
  - Machine Test

NATIONAL INSTRUMENTS

From a high level, machine monitoring is performed for two major purposes.

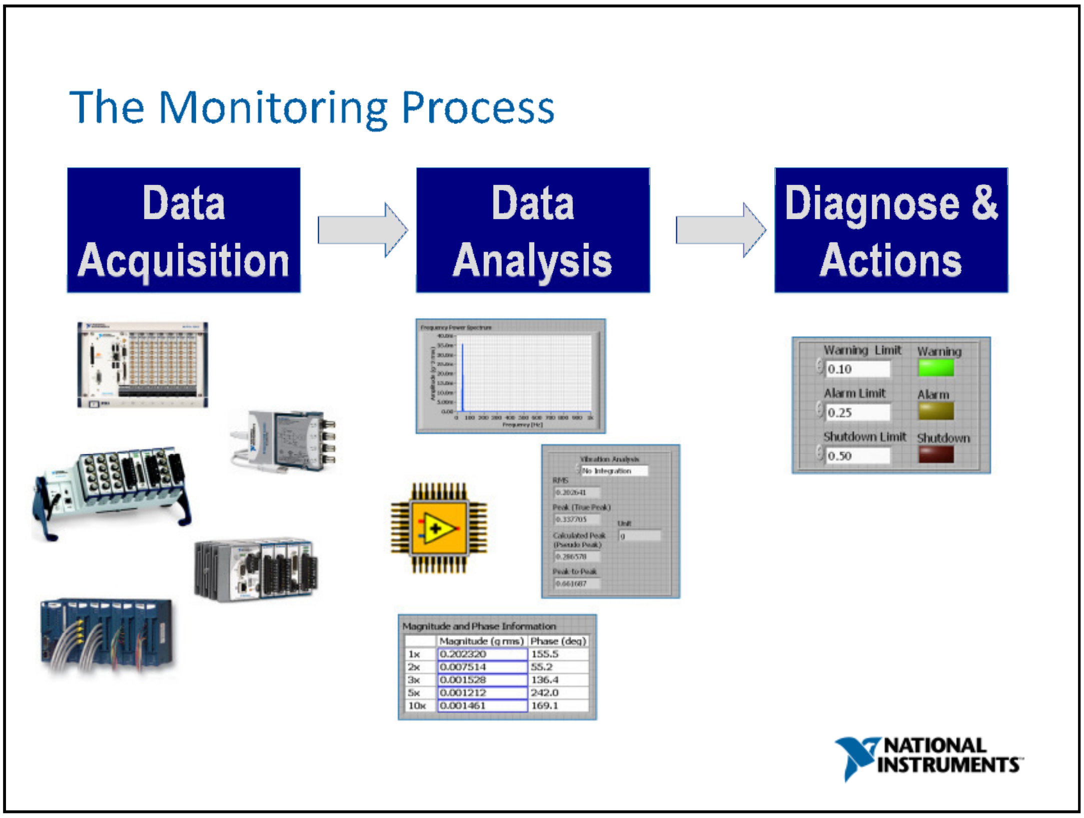
First, and most importantly, machines are monitored to collect electrical and mechanical signatures from the machines to determine maintenance requirements. Machine monitoring is an important maintenance cost control resulting in cost savings in a) repair costs, b) parts storage, and/or c) interrupted production caused by breakdowns. Overall, these activities keep a machine running during production. Ideally, machinery is powered off only when you schedule it to shutdown.

To keep machinery running and available for production, you must predict when faults will occur. After you learn when the machine will need to be powered off, repairs can be made during scheduled shut downs.

Capturing vibration signatures and related machine parameters provides insight into machine performance and optimal operational speeds and settings. This insight can lead to lower operational costs and increased consistency in product quality.

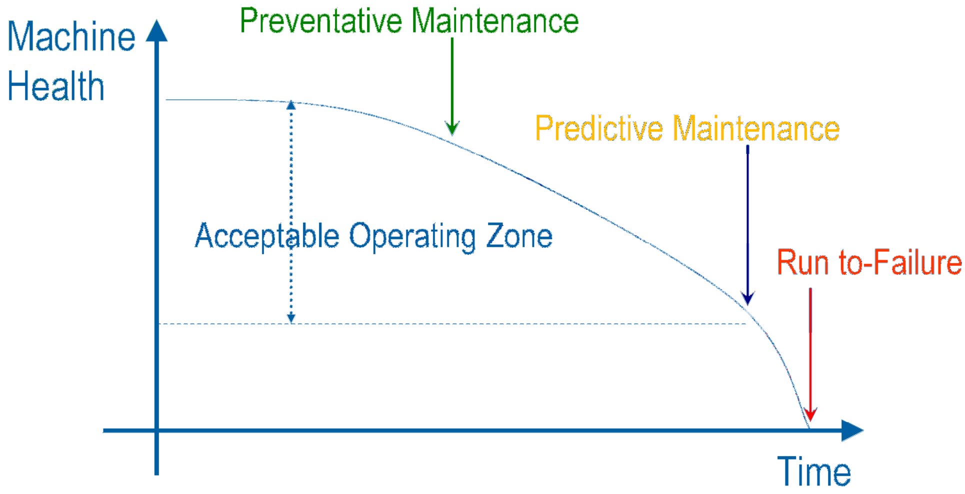
Second: Machine monitoring activities collect data that can be used to help optimize the production process. Machine performance and health conditions can play a role in the loading of a machine and the selection of raw materials. There are a number of physical characteristics to monitor on any given machine. Common signatures to monitor are temperature, process variables, and digital status events. Trending these parameters over time provides indications of physical or mechanical feature failures if trends move outside acceptable limits.

Often in machine monitoring for predictive maintenance or machine health, dynamic or rapidly changing signatures are captured and analyzed. These signatures provide a closer look, as compared to static signatures, at the actual frequency make-up of the signature. By looking at vibration or electrical signatures in their frequency content, individual forcing functions become apparent. In other words, the dynamic vibration or electrical signatures reveal information regarding such mechanical components as bearings, gears, rotor bars, and so forth.



If we look again at the overall monitoring process, National Instrument provides a variety of hardware for data acquisition, a variety of software for data analysis, and output and control options for diagnostics. The machine monitoring process is very similar to the traditional acquire, analyze and present capabilities of NI tools.

MCM is Increasing Availability



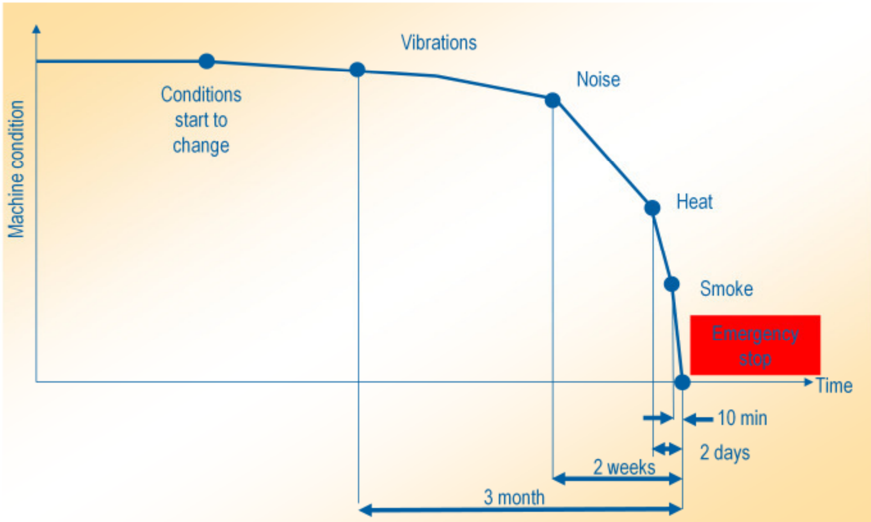
NATIONAL INSTRUMENTS

-Preventative maintenance is completely dependent on time, not machine performance

-Run to failure is completely dependent on machine performance (failure) but extends the amount of time

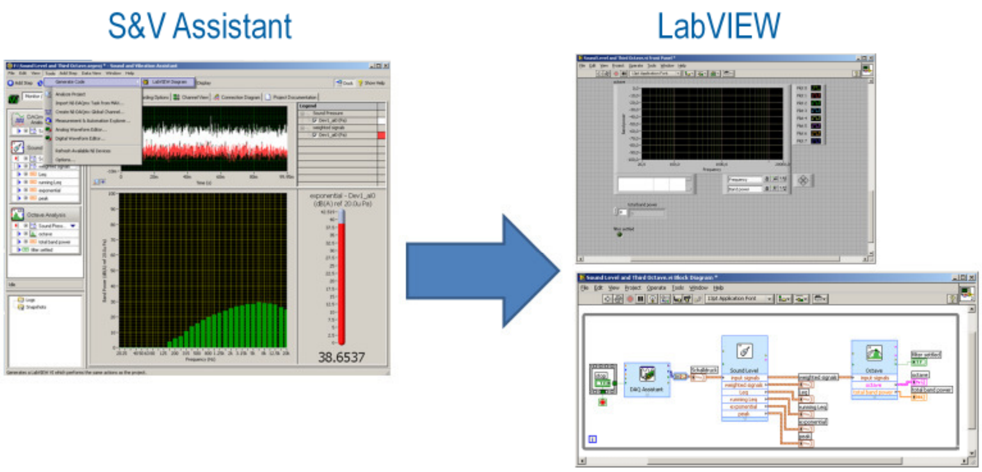
-Predictive maintenance attempts to extend machine uptime through acceptable performance

Lead Time of Options



The trick in machine condition monitoring is to predict failures in plenty of time to plan and schedule resources such as parts and people for downtime. With good planning, downtime is minimized and the machine is back up and running sooner and thus making money again sooner.

Exercise 5 - Generate LabVIEW Code with S&V Assistant

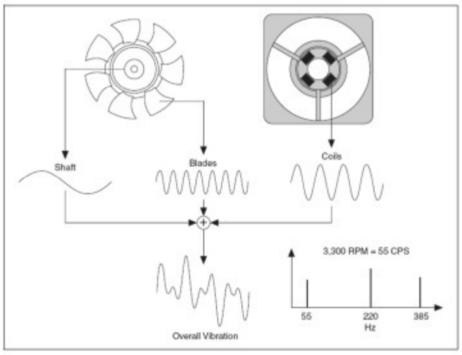


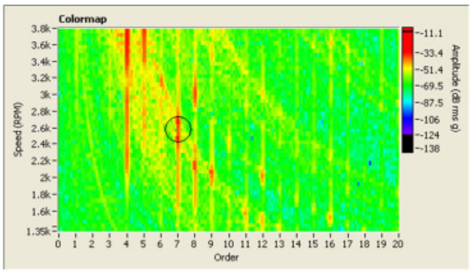
NATIONAL INSTRUMENTS

You can generate LabVIEW Code from S&V Projekt. In S&V Assistant click on Tools >> Generate Code > LabVIEW Diagram to generate LabVIEW Code.

Accomplish Exercise 5.

Exercise 6: Order Analysis



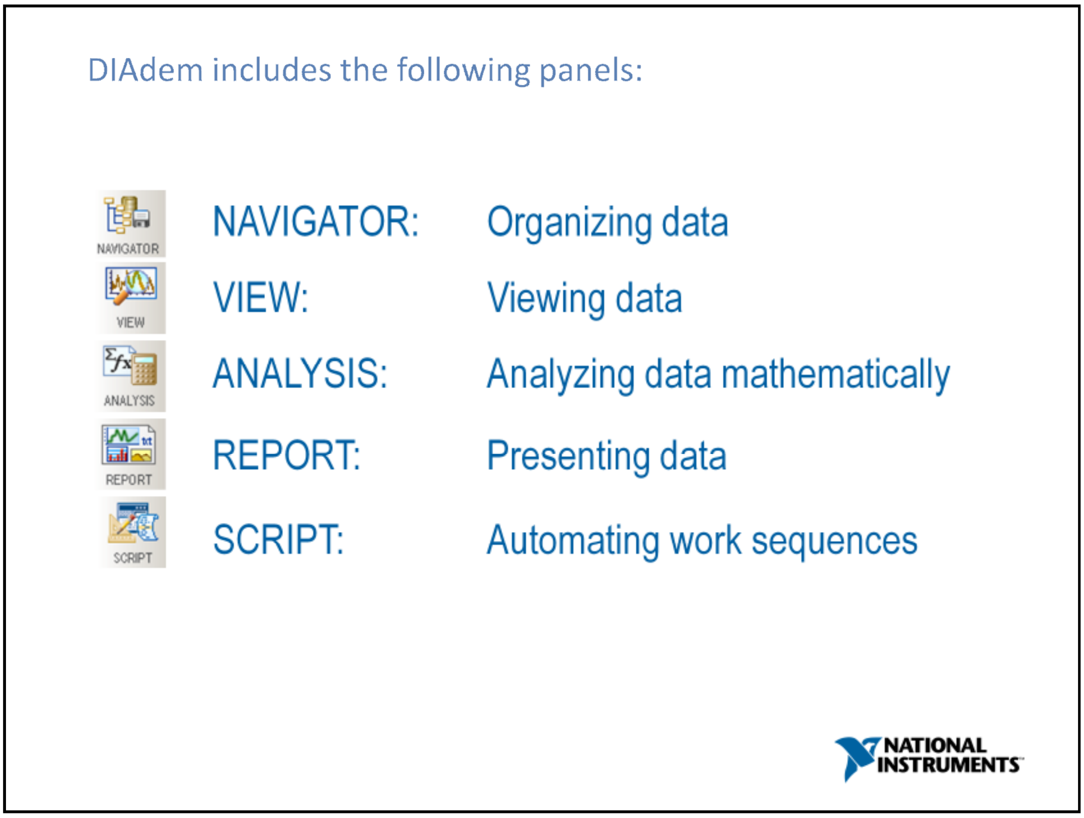


Use the NI-9233 and the Order Analysis Express VIs to make some basic rotating machinery measurements such as a speed profile and order power spectrum as well as use a color map to display results.

Accomplish Exercise 6.

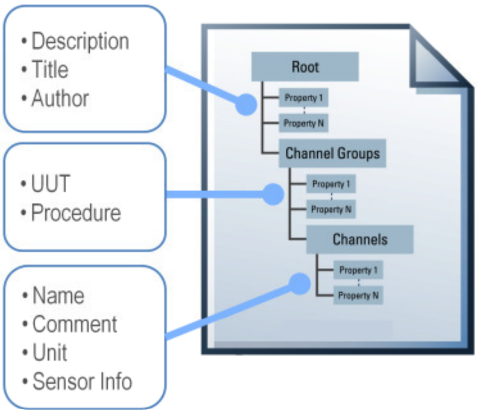
NI DIAdem

For Offline Analysis, Data Navigation and Reporting use DIAdem.



The DIAdem panels constitute the main level, where functions are grouped into related areas. The panels are arranged on the left side of the screen and are always visible, enabling you to switch from one panel to another. All the DIAdem panels read data into and out of the Data Portal.

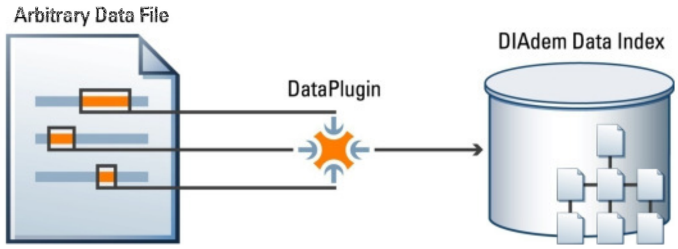
The TDM Data Model



- 3 levels of hierarchy
  - File, Groups and Channels
- Each level has associated properties
- The data model is saved with file

NATIONAL INSTRUMENTS

DIAdem DataFinder: How does it work?



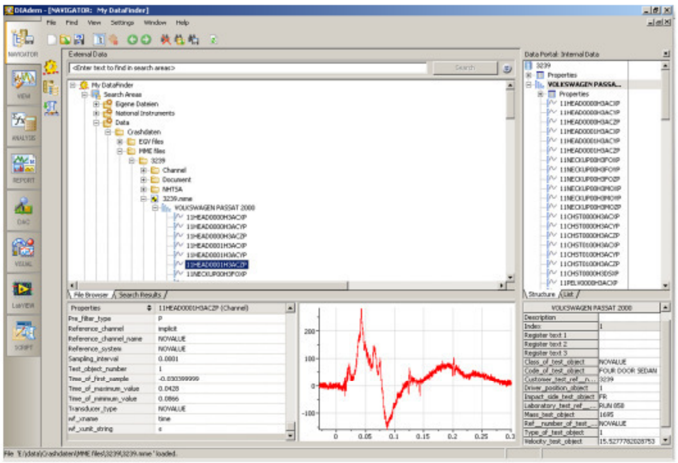
- The DIAdem Data Index stores descriptive information of your data files
- Works with **any** data file you have a DataPlugin for *
- It builds automatically and updates periodically
- No IT support required to install, configure or maintain

*visit www.ni.com\dataplugins

NATIONAL INSTRUMENTS

Many companies have developed test file formats that capture essential test information in header files. NI DataPlugin technology makes your custom file formats and legacy data available for searching by reading in the attributes from any arbitrary, binary, or ASCII file into the DIAdem data index. You also can use the DataPlugin wizard in DIAdem 10.0 to automatically generate DataPlugins for ASCII files or use the programmer's API to create your own DataPlugin using the DIAdem VBScript interface. The DIAdem DataFinder works natively with TDM files.

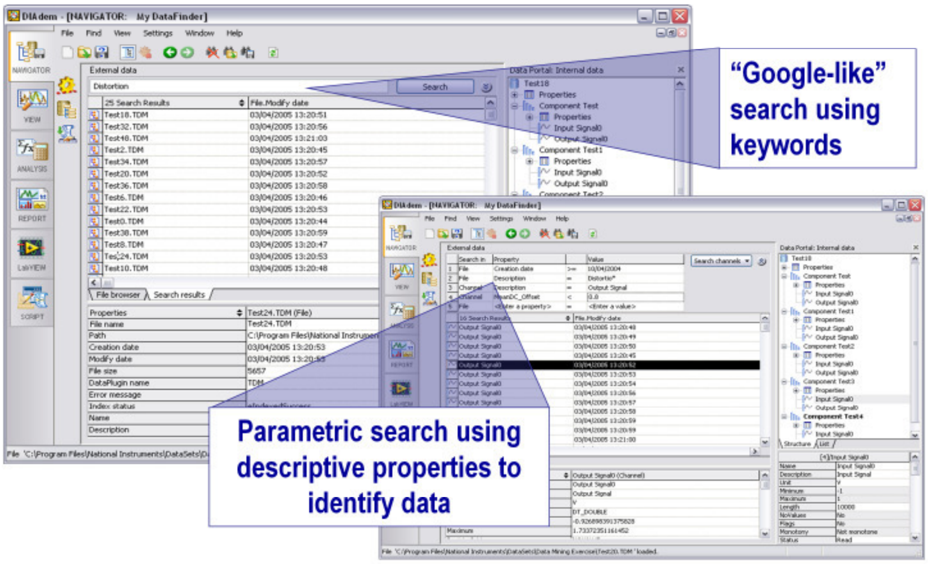
Data Navigation



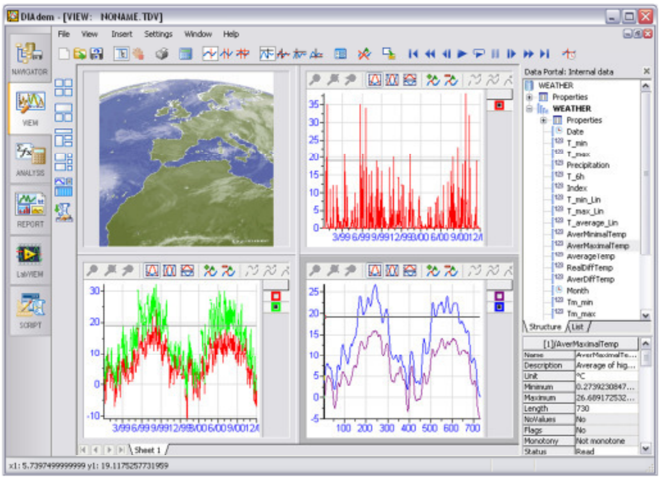
- Organize and search data using key search or an interactive query builder
- Find and load data using an drag & drop environment
- Connect to over 100 data formats
- Work with up to 2 billion values per channel

NATIONAL INSTRUMENTS

DIAdem DataFinder



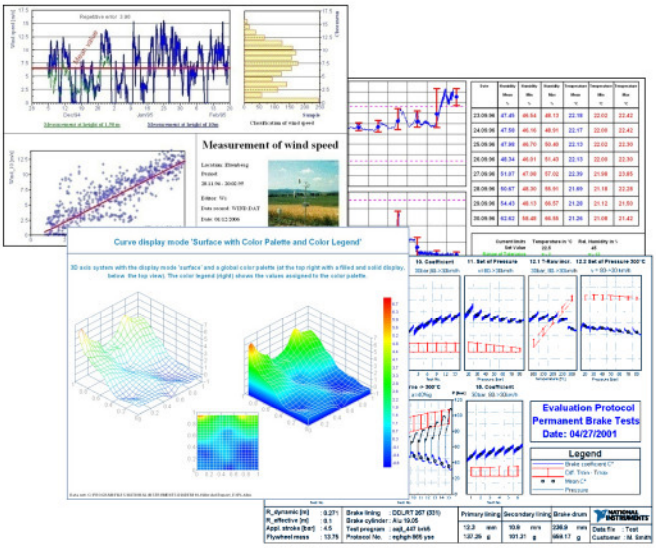
Interactive Data Inspection



- Inspect data in various different ways - graphs, tables, text, images and videos
- Synchronize data and videos
- Zoom into details
- Customize via scriptable object oriented API

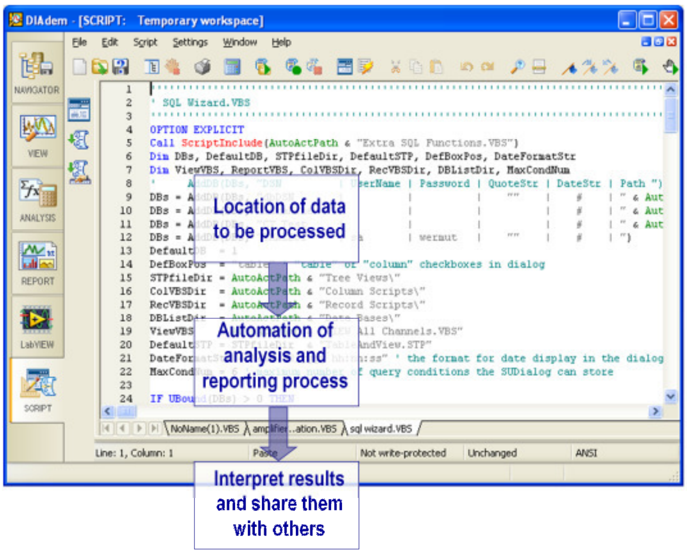
NATIONAL INSTRUMENTS

Report Generation



- Design interactively reusable, multi page report templates
- Integrate 2D, 3D Graphs and Tables, Graphics, Text, Variables and Commands
- Export as Graphic, HTML page or PDF document

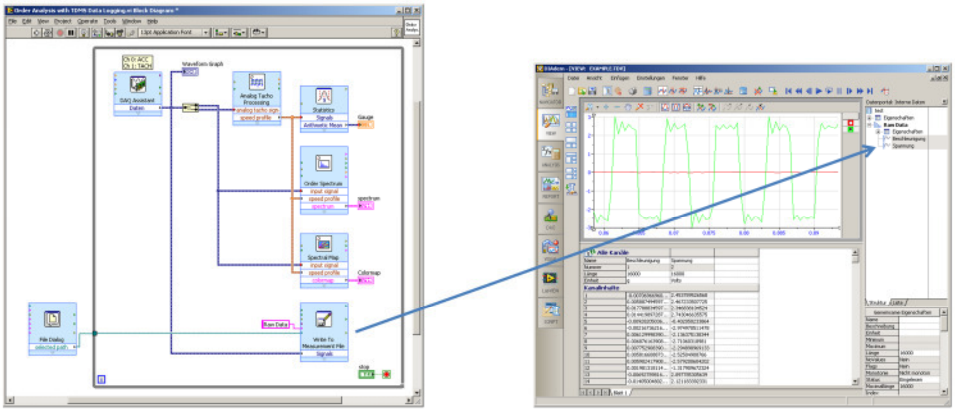
Automation and Customaization



- Automate analysis tasks and standard process using Visual Basic Script
- Create graphically application specific dialogs
- Interface with 3rd party software via Active-X
- Customize DIAdem standard menus

NATIONAL INSTRUMENTS

Exercise 7: Save date in TDMS format, load and analyze data with DIAdem



In this Exercise you will expand the **Order Analysis.vi** from Exercises 6, Part 1 with the **Write To Measurement File Express VI**.

Save data as TDMS file and analyze data in DIAdem.

Accomplish Exercise 7.

How To Learn More Out Of The Classroom

- ni.com/support
  - Access product manuals, KnowledgeBases, example code, tutorials, application notes and discussion forums
  - Request technical support
- Collection of tutorials, webcasts, case studies, and example code: http://zone.ni.com/devzone/cda/tut/p/id/6602
- Alliance Program: ni.com/alliance
- Publications: ni.com/reference/books/
- Practice!

NATIONAL INSTRUMENTS

Thank you for attending the course

Please fill out the course survey

NATIONAL INSTRUMENTS

Exercises

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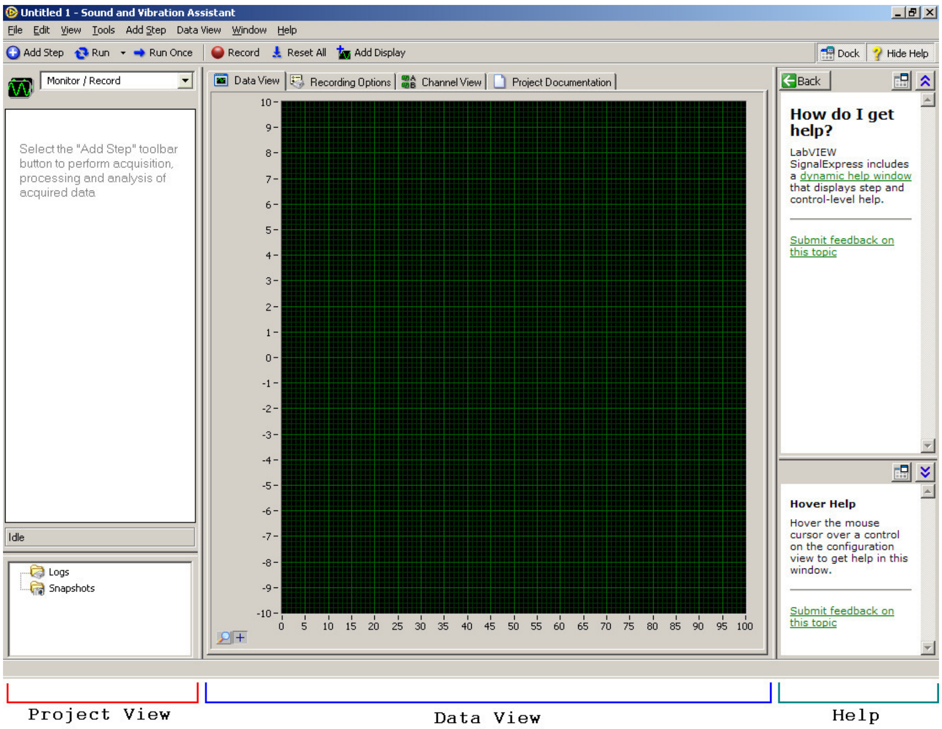
Exercise 1: Sound Level Measurement with Sound and Vibration Assistant

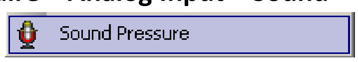
In this exercise you will use the Sound and Vibration Assistant to measure the sound level from a microphone.

1. Connect the microphone to channel 0 of the NI-9233.
2. Select **Start >> Programs >> National Instruments >> Sound and Vibration >> Sound and Vibration Assistant** to Launch the Sound and Vibration Assistant.

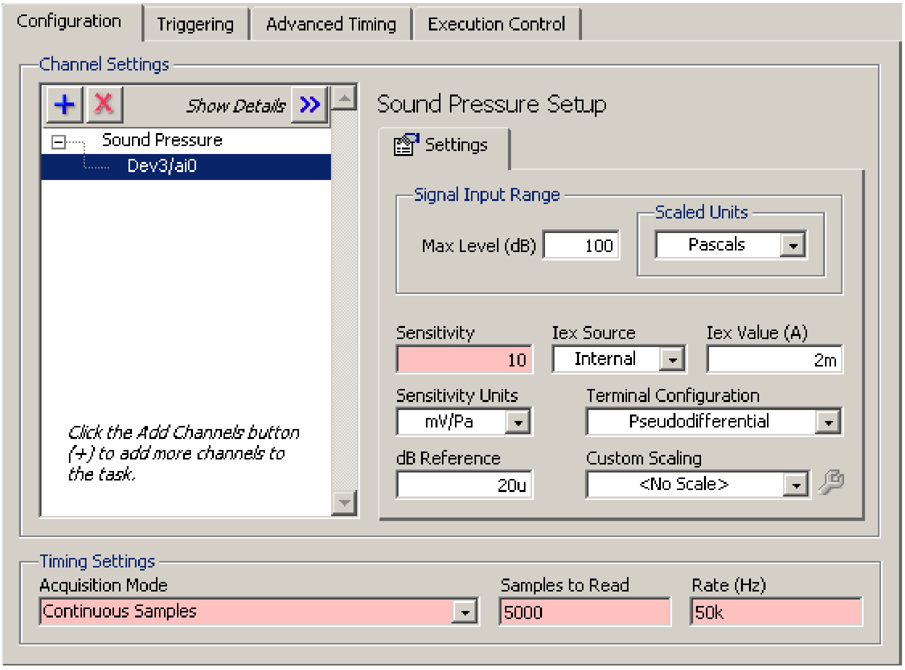
   Notice that the Sound and Vibration Assistant is split into three main views: the **Project View** on the left, the **Data View** in the middle, and the **Help** on the right.



3. The Sound and Vibration Assistant uses **Steps** to acquire, generate, analyze, load, and save signals. Place a step that acquires sound pressure data from the NI 9233 by right clicking on the Project View and selecting **Acquire Signals > DAQmx Acquire > Analog Input > Sound Pressure**

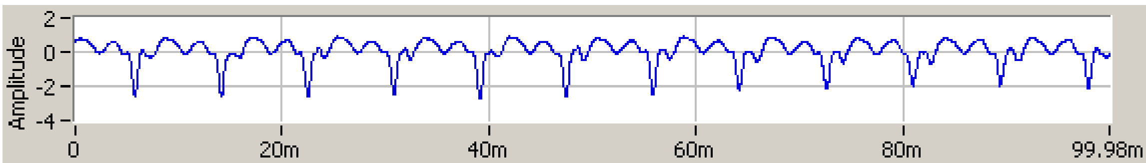


4. When you place the DAQmx Acquire Step, the Data View tab will automatically change to the **Step Setup** tab. This view allows you to configure all aspects of your acquisition step including number of channels, excitation levels, and timing information.
5. Select Channel **ai0** on USB-9233.
6. Configure the DAQmx Acquire step with the following settings:

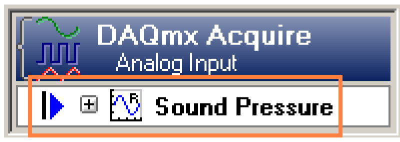


7. Enter the microphone sensitivity which is listed on the calibration certificate for your microphone. If you don't have a certificate then just leave it at "10".

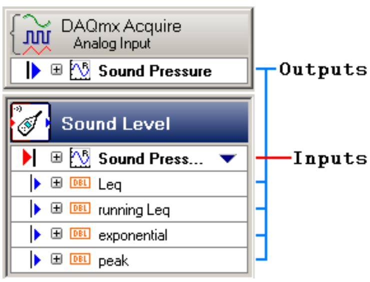
8. Before going any further, you can preview the time domain data from the **Step Setup** tab by clicking the **Run** button near the top of the screen.



9. Click on the **Data View** tab and notice that it is blank. You can configure the Data View to show only the signals that you wish to see. Use your mouse to click on the Step output labeled "Sound Pressure" and drag it onto the blank data view. Your time domain data should immediately appear in the **Data View**.



10. Use an Analysis Step to take a Sound Level Measurement from this time domain data. Right click on the **Project View** and select **Analysis >> Time Domain Measurements >> Sound Level**. Again you will notice that the view tab automatically changes to **Step Setup**.

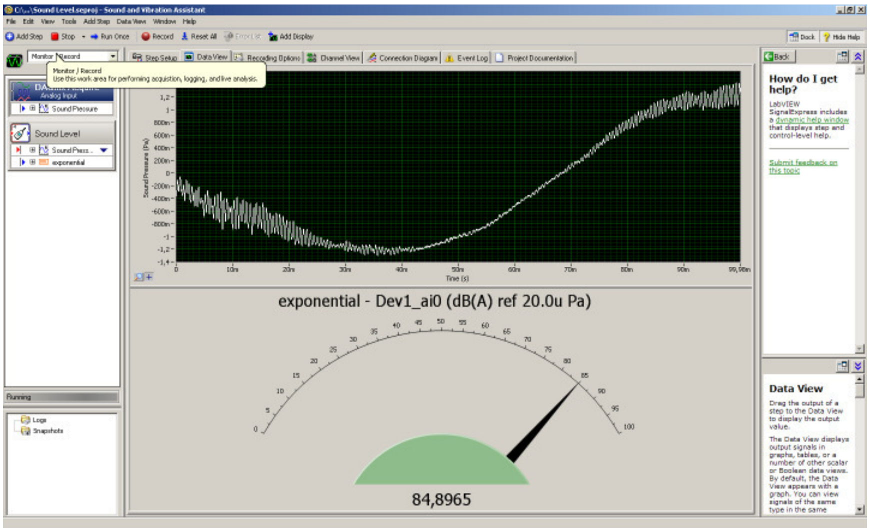


Note: This is a good place to mention that every step in the Sound and Vibration Assistant has inputs and/or outputs. The DAQmx Acquire step has only one output, **Sound Pressure** data. The Sound Level step however has several sound level outputs and requires one input. **Sound Pressure** has been chosen by default but you can change the source (if applicable) by clicking on the input's drop down menu.

11. This particular step has three configuration tabs near the bottom of the screen: Input, Weighting, and Averaging. Configure this step to include "A Weighting" and to only compute "Exponential" averaging.

12. Switch back to the Data View. Click on the Sound Level output "exponential" and drag it onto the data view. Notice that a new chart and table was created to display this data. Since the "exponential" output is a different data type than your currently displayed "Sound Pressure", the Sound and Vibration Assistant automatically creates a new display in the Data View.

    Right click the sound level chart and select **View As >> Meter**. Right click the meter and turn off Autoscale. You can then double click the maximum value on the meter and change it to 100.



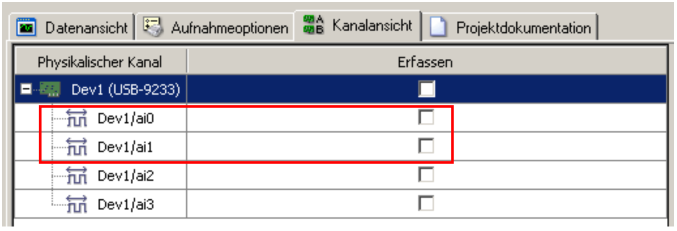
13. Stop and Project. 

14. Save this Project, you will use it later.
    Select File >> Save As to save the file as **Sound Level.seproj** in Exercise 1 folder.

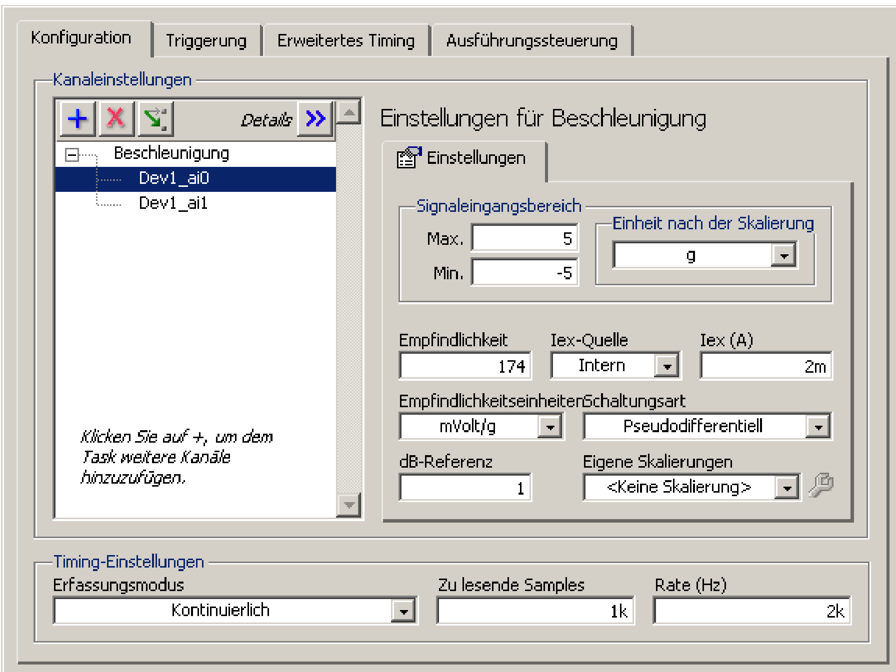
Exercise 2: Vibration Measurement with Sound and Vibration Assistant

The DSA Demo Box has two accelerometers which measure the vibrations caused by an onboard unbalanced fan (see Appendix). In this exercise you will use the Sound and Vibration Assistant to measure the vibration level of these accelerometers and perform a power spectrum during live acquisition.

1. Connect channel 0 to the X acceleration and channel 1 to the Y acceleration of the DSA Demo Box.
2. Launch the Sound and Vibration Assistant.
3. Place a DAQmx Acquire step on the Project View, but this time let's use the **Channel View**. The Channel View is a convenient way to organize and configure all National Instruments hardware for your project. Select channel 0 and channel 1 from the Channel View.



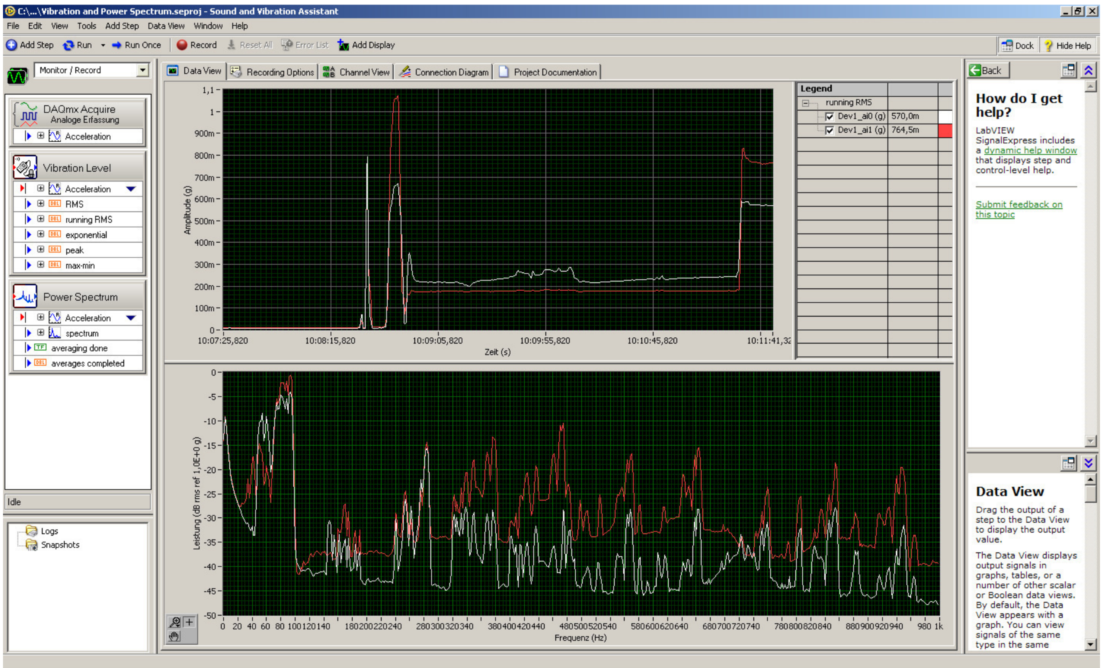
4. Notice how a DAQmx Acquire step was created and it automatically included channels 0 and 1 from your NI 9233. From the channel view, change the measurement type of each channel from "Voltage" to "Acceleration."
5. Double click the newly created DAQmx Acquire step to quickly switch to the **Step Setup** tab. Configure the acquisition with the following settings:



***Note**: Make sure to set the sensitivity for both accelerometers (174mV/g).*

6. Click the **Run** button and preview the time domain data. Switch to the **Data View** and plot both channels of Acceleration output onto the blank display.

7. Place a **Vibration Level** analysis step from the **Time Domain Measurements** palette. Configure the step to calculate only the Running RMS level over a 1 second period.
8. Drag the “running RMS” output onto the **Data View** to display the vibration level for both channels.
9. Place a **Power Spectrum** analysis step from the **Frequency Domain Measurements** palette. Choose how you would like the units to be displayed (RMS, Peak, etc.) and if you would like any averaging. If you have no preferences, just use the default settings.
10. Drag the “spectrum” output onto the **Data View** to display the Power Spectrum data for both channels.

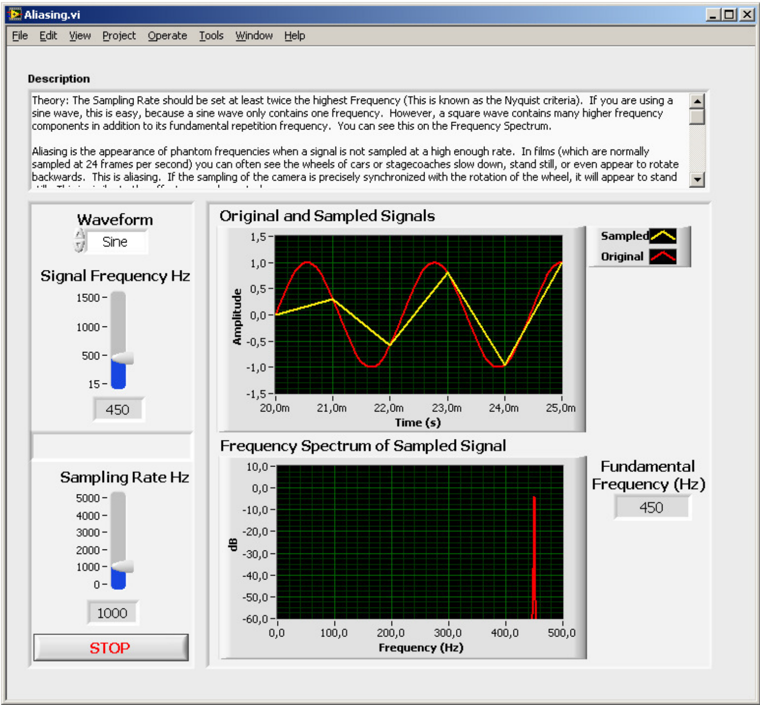


11. Vary the speed of the fan and watch how the power spectrum changes.
12. Stop the Project. 

13. Save the Project.
    Select File >> Save As to save the file as **Vibration and Power Spectrum.seproj** in Exercises 2 folder.

Exercise 3: Aliasing

Open “Aliasing.vi” in Exercise 3 folder and change Sampling Rate and Signal Frequency.



Theory: The Sampling Rate should be set at least twice the highest Frequency (This is known as the Nyquist criteria). If you are using a sine wave, this is easy, because a sine wave only contains one frequency. However, a square wave contains many higher frequency components in addition to its fundamental repetition frequency. You can see this on the Frequency Spectrum.

Aliasing is the appearance of phantom frequencies when a signal is not sampled at a high enough rate. In films (which are normally sampled at 24 frames per second) you can often see the wheels of cars or stagecoaches slow down, stand still, or even appear to rotate backwards. This is aliasing. If the sampling of the camera is precisely synchronized with the rotation of the wheel, it will appear to stand still. This is similar to the effect causes by a stroboscope.

Experiment 1: Select the Sine Waveform and set the Signal Frequency to 450 Hz and the Sampling Rate at 1000. Note that the Sampled (Yellow) waveform is very jagged due to the relatively few samples per cycle. However, even with these few samples we can still make a correct measurement of the signal as can be seen in the Frequency Spectrum containing only one tone at 1 kHz. Click the mouse to the right of the last digit in the box below the Signal Frequency Slider. You can now change the last digit of the Signal Frequency, using the Up/Down keys on your keyboard. Hold down the Down Arrow key, and note how the points of the waveform that are sampled change quite dramatically.

Experiment 2: Using the Arrow key as in Experiment 1, increase the frequency by holding down the Up arrow key on your keyboard. Notice that when you pass 500 Hz, three things happen. The Sampled waveform has only two samples per period, and looks almost flat, the Frequency Spectrum now shows the signal moving backwards (down in frequency) as you increase the frequency. This is Aliasing, which is why the Aliasing indicator comes on. Keep increasing the Frequency until you get close to 1000 Hz. As you slowly approach 1000 Hz, note that the Tone on the Frequency Spectrum finally disappears at 0 (DC) and then re-emerges as you increase the frequency, this time running the right direction, but with measured value that is 1000 Hz too low on the Frequency Spectrum.

Experiment 3: Set the Signal Frequency to just below 1500 Hz. Notice that the sampled waveform repeats only 1/3 times as often as the original! With aliasing, you cannot tell the aliased components from the real ones! Therefore to make correct measurements you must always make sure you do not violate the Nyquist criteria. You can do this by using analog filtering of the signal to be measured. This is done using a low pass filter which removes undesired higher frequency components. It is normally called an anti-aliasing filter. Or, if you have a knowledge that the signal does not contain unwanted higher frequencies you can also get by without a filter. Alternately, you can set the sampling rate very high compared to the frequency you want to measure.

Experiment 4: Repeat the above experiments using a Square Waveform. Since the square wave contains many other frequencies (harmonics) this will alias dramatically, and you may see various portions of the spectrum both moving up and down at the same time as you change the Frequency of the Square Wave. This is a dramatic illustration of the mess that aliasing can get you into!

Experiment 5: (Advanced) With a Square wave, set the Signal Frequency to 250 Hz and the Sampling Rate to 5000 Hz. The Spectrum looks nice and free from Aliasing. But this is an illusion! Increase the Signal Frequency to 260 Hz in steps of 1 Hz and notice all the aliasing components that show up. Notice also how the harmonic levels of the spectrum drops! This is because the signal at 250 Hz also was aliased, but the frequencies of the aliasing components fell on top of the existing frequency components in the square wave, thus making the aliasing invisible, but still ruining the accuracy of the result.

Close the VI without saving.

Exercise 4: Third Octave Analysis

1. Open **Sound Level.seproj** you have programmed in Exercise 1.
2. Now let's add a 1/3 Octave Analysis. Place an **Octave Analysis** step from the Frequency Domain Measurements.
3. Configure the Octave Analysis step to perform third-octave analysis over the frequency range of 20Hz to 20kHz with A-weighting[1]. Notice how the step's preview adjusts to reflect the changes that you make.
4. Switch back to the **Data View** and drag the "octave" output onto the Data View.

5. Stop the Project.

6. Save the Project.
Select File >> Save As to save the file as **Sound Level and Third Octave.seproj** in Exercises 4 folder.

[1] In the measurement of loudness, for example, an A-weighting filter is commonly used to emphasize frequencies around 3–6 kHz where the human ear is most sensitive, while attenuating very high and very low frequencies to which the ear is insensitive. A-weighting is the most commonly used of a family of curves defined in the International standard IEC61672:2003 and various national standards relating to the measurement of sound level, as opposed to actual sound intensity.

Exercise 5: Generate LabVIEW Code with S&V Assistant

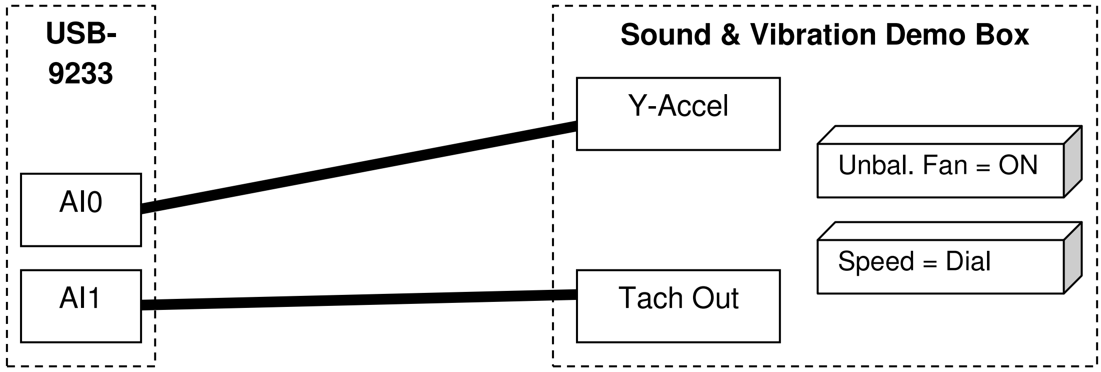
1. Open **Sound Level and Third Octave.seproj** you have programmed in Exercise 4.
2. In S&V Assistant click on **Tools >> Generate Code > LabVIEW Diagram** to generate LabVIEW Code.
3. Save VI as **Sound Level and Third Octave.vi** in Exercise 5 folder.
4. Run and test the VI.

Exercise 6: Order Analysis

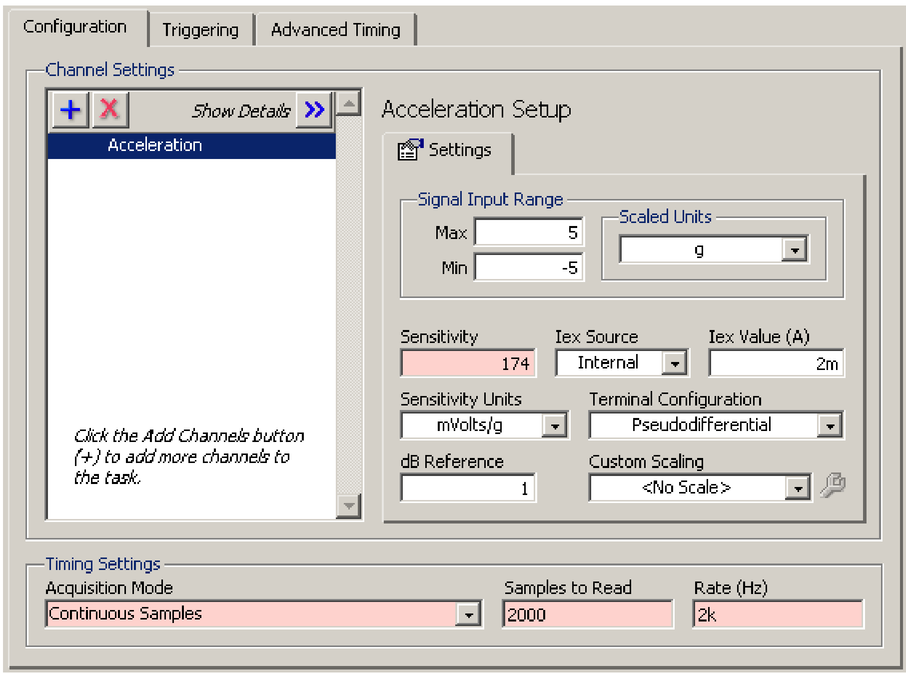
Use the NI-9233 and the Order Analysis Express VIs to make some basic rotating machinery measurements such as a speed profile and order power spectrum as well as use a color map to display results.

Part 1- Build Order Analysis VI

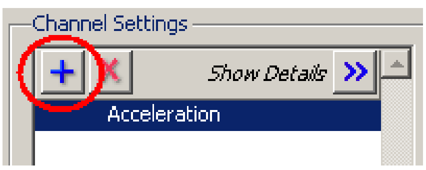
1. Connect the Y acceleration output of the DSA Demo Box to channel 0 of the NI-9233 and connect the tach out to channel 1.



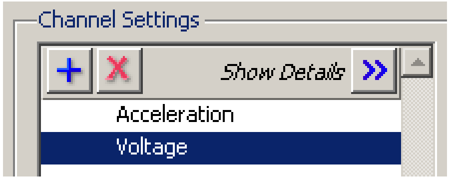
2. Open a new VI in LabVIEW.
3. Go to the block diagram and place down the **DAQ Assistant**
   You can find this function at **Measurement I/O >> NI-DAQmx >> DAQ Assistant**.
4. Configure the DAQ Assistant to acquire an "acceleration" measurement on Ch 0. Click **Finish**.
5. Configure the measurement with the following settings:



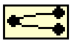
6. Now we'll add the tachometer channel to the same task. Click on the button for creating a new virtual channel.



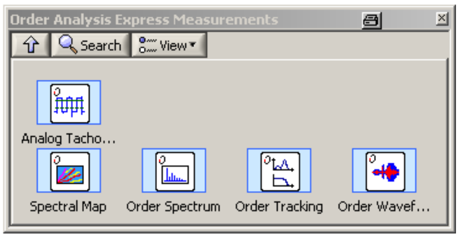
7. Select the measurement type to be “voltage” and from channel 1 of the NI-9233. You should now see two different measurement types in your task.



8. Click OK to apply your settings to the DAQ Assistant and press **No** when prompted about Automatic Loop Creation.
9. Right-click on the data output terminal and select **Create >> Graph Indicator**.
10. Place the Split Signal function next to the DAQ Assistant. You can find this function at **Express palette >> Signal Manipulation >>Split Signal**. Expand the split signal so that it has two outputs.



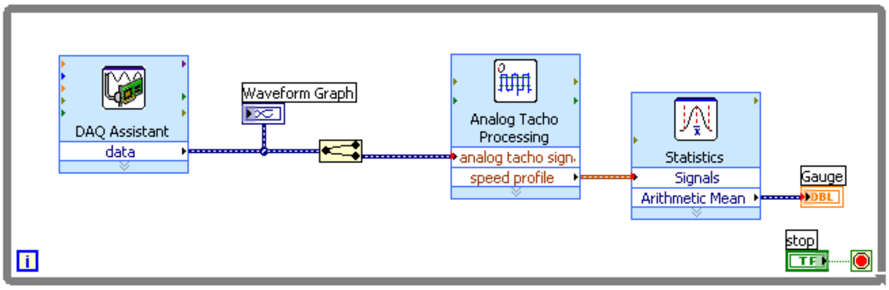
11. All Order Analysis Express VIs can be found in the palette located at **Sound and Vibration >> S&V Express Measurements >> Order Analysis Express Measurements**.



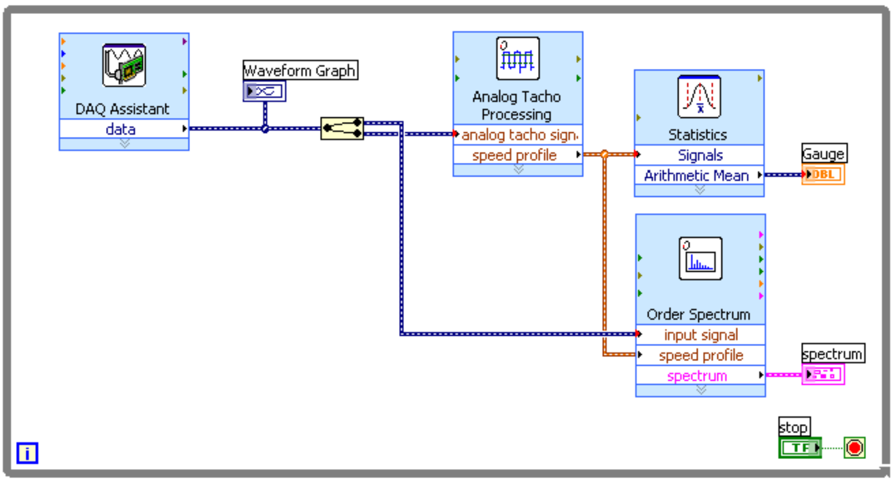
12. Place an **Analog Tacho Processing** Express VI on your block diagram. The tachometer from the fans in the DSA Demo Box has 2 pulses per revolutions. Change the **Pulse/Revolution** to 2 and select OK.
13. This Express VI calculates the speed profile of the machinery from the tachometer signal. Wire the tachometer signal (bottom half of the Split Signal VI) into the **analog tacho signal** input.
14. Next place the **Statistics Express VI**, located on the **Express palette >> Signal Analysis >> Statistics.** Configure the VI to calculate only arithmetic mean.
15. Wire the **speed profile** to the **Signals** input of the new express VI.
16. Go to the front panel and place down a gauge numeric indicator. You can find this on the **Express palette >> Numeric Indicators**. Increase the maximum value of the gauge to be 6000. You can do this by double clicking on the value 10 and then typing 6000. We will use the gauge indicator to display the current speed of the fan.

17. Return to the block diagram. Wire the output of the Statistics Express VI to the gauge indicator.

18. To make this program run continuously, add a While Loop around all of your code. Now your VI should look similar to this:

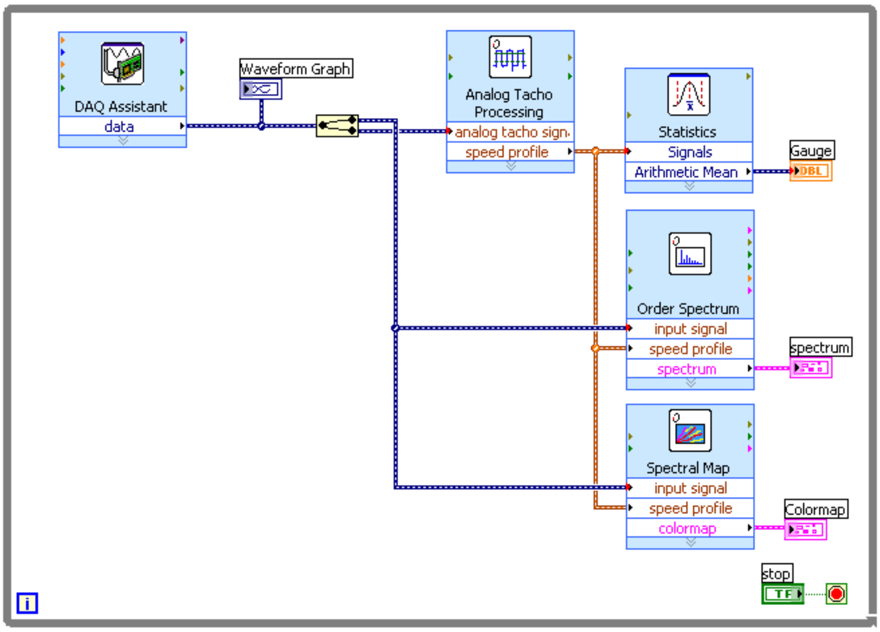


19. Go to the front panel and run the VI. Use the knob on the Demo Box to change the fan speed. When you are finished testing your VI save it as **Order Analysis.vi**.
20. Next we will add an Order Power Spectrum to our VI. Retrieve the **Order Spectrum Express VI** and place it on your block diagram. Press OK and keep all of the default settings.
21. Wire the **input signal** terminal to the top half of the Split Signal VI and also connect the **speed profile** to the output of the Analog Tacho Processing VI.
22. Right-click on the **spectrum** output and select **create >> indicator**.
23. Your VI should now look something like this:



24. Go to the Front Panel and run the VI. Vary the speed of the fan and switch between the balanced and unbalanced fans.
    - Hint: Turn off AutoScaling on the Order Power Spectrum graph so you can easily see how things are changing. Set the scale on the X-axis to be 0 to 10 and the Y-axis to be from -100 to 0.
25. The last thing we will add is a color map. Retrieve the **Spectral Map Express VI** and place it on your Block Diagram.
26. Configure the Plot type to be **Frequency – Time** and select OK.
27. Wire the **input signal** and **speed profile** inputs in the same way as you did with the Order Spectrum VI.
28. Go to the Front Panel and place a color map from the **Sound and Vibration >>Colormap.xctl** palette.

29. To finish the VI go back to the Block Diagram and wire the color map to the output of the Spectral Map VI. Here is what your finished VI should look like:

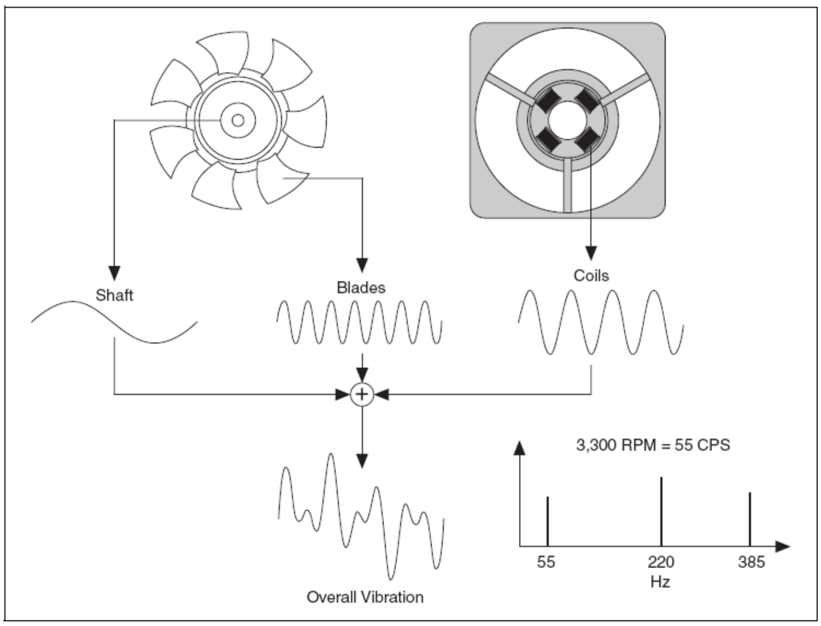


30. Run the VI and vary the speed of both the balanced and unbalanced fans.

31. Save the VI.
    Select File >> Save As to save the file as **Order Analysis.vi** in Exercises 6 folder.

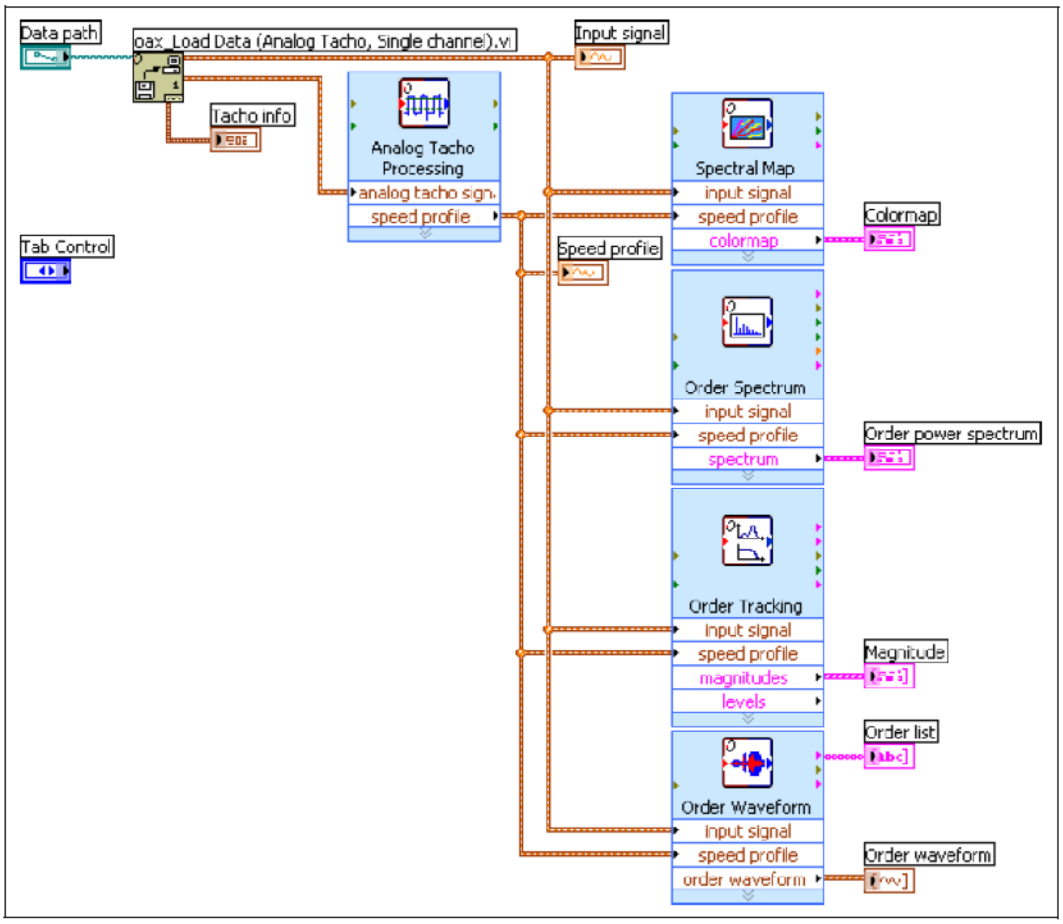
Part 2 – Analyze Fan Run Up

In this section, you use the order analysis tools in the NI Sound and Vibration Measurement Suite to use it for noise, vibration, and harshness (NVH) testing. You use these tools to perform offline order analysis on example data. This example data represents the speed and vibration amplitude of a computer fan during a run-up test. The exercises in this section illustrate the common analysis methods you can use to acquire information about the orders in the data.



1. Select **Help » Find Examples** in LabVIEW to launch the NI Example Finder. Select **Toolkits and Modules » Sound and Vibration » Getting Stated » Getting Started with Order Analysis Express VIs.vi.**
2. Select **File » Save As » Getting Started with Order Analysis Express VIs.vi** in folder **Exercise 6**.

The **Getting Started with Order Analysis Express VIs.vi** uses the colormap, order power spectrum, magnitude, and order waveform plots to analyze the sound and vibration data from the computer fan. Each plot displays the same information from a different perspective. You can use these plots to analyze the example data and gain a fuller understanding of the DUT (computer fan).

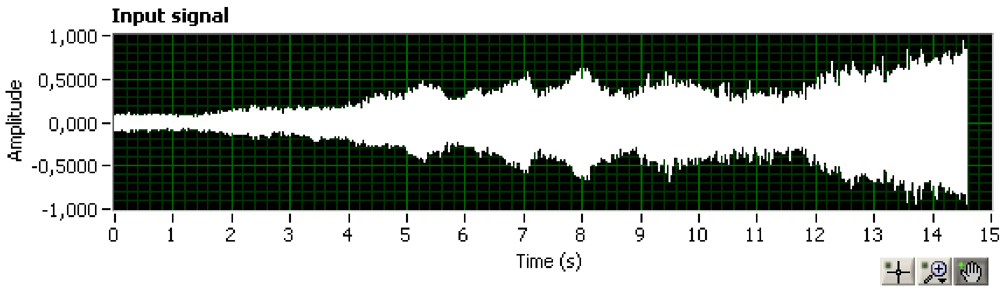


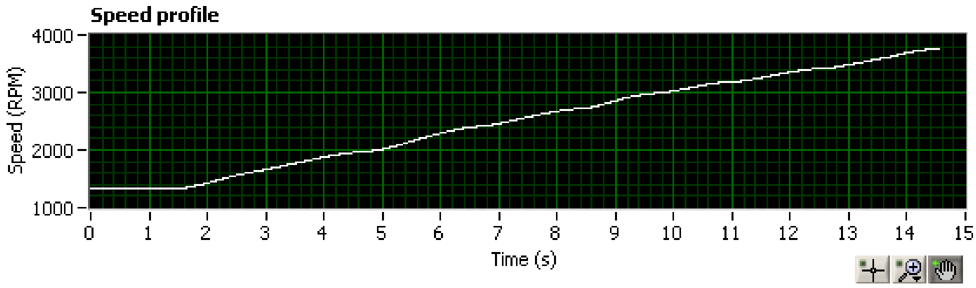
Complete the following steps to load the computer fan run-up data to use in the **Getting Started with Order Analysis Express VIs.vi**:

3. In Front Panel click the Browse button next to the Data path text box and navigate to the PC Fan runup.dat file in the labview\examples\Order Analysis\Example Data directory. (For your information: This data was logged with the example **Log Data (Analog Tacho, DAQmx).vi** you can find in Example Finder **Toolkits and Modules » Sound and Vibration » File Input and Output**.)
4. Click the OK button. The filename appears in the Data path text box.
5. Click the Run button to run the Getting Started with Order Analysis Express VIs.vi.

6. The **Getting Started with Order Analysis Express VIs.vi** displays the waveform of the computer fan run-up data in the Input signal plot on the front panel. This VI also returns the tachometer information for the input signal in the Tacho info indicator. Notice that the number of pulses per revolution of the tachometer is two (like the Part 1 in this exercise).

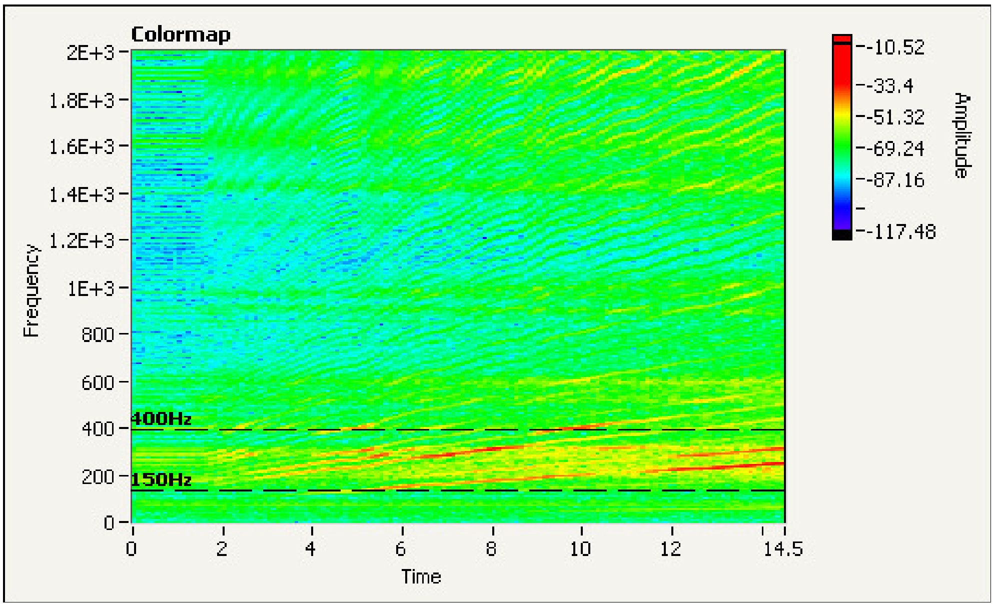
7. **Analyzing the Speed Profile**





The Input signal plot is a vibration signal plot. Vibration signal plots illustrate how sound or vibration signals change over time. You can use a speed profile plot to illustrate how rotational speed changes over time. You then can use these two plots in parallel to observe how raw sound or vibration changes with speed. The Analog Tacho Processing Express VI uses an analog tachometer signal to determine the rotational speed of a DUT.

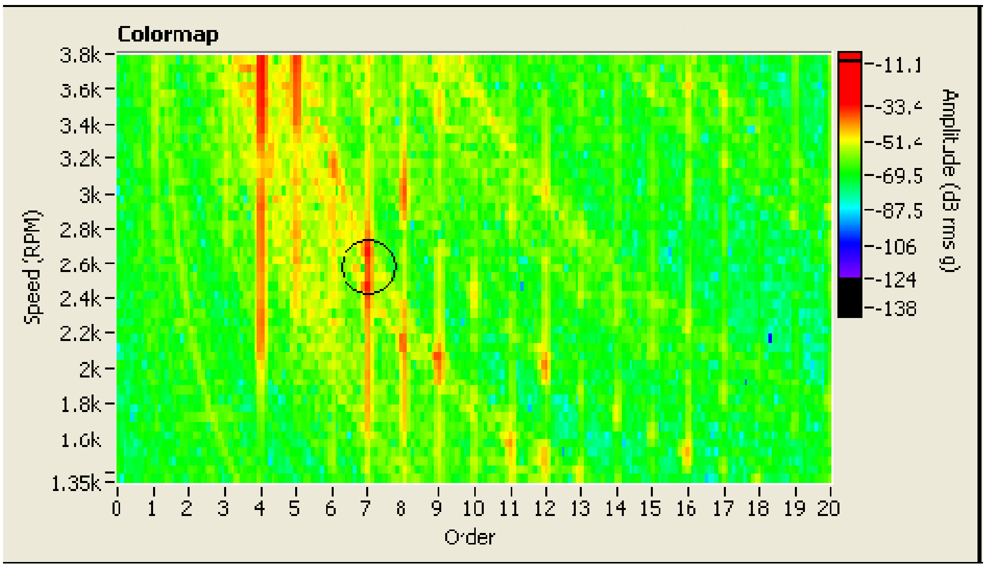
8. **Analyzing the Colormap**
On the front panel, click the **Colormap** tab to view a colormap of the computer fan run-up data. A colormap is a three-dimensional display of a sound or vibration spectrum as a function of time or speed. The spectrum can be a frequency or order spectrum. Use the **Spectral Map Express VI** to compute the spectral map of a sound or vibration signal. This Express VI can return the spectral map in a **colormap** or a **waterfall graph**. You can use this Express VI to create a colormap of the computer fan run-up data.



9. By default, this plot displays frequency against time. Red portions of the colormap indicate areas of strong amplitudes. Notice that several red lines appear on this colormap. These red lines correspond to strong vibrations at different frequencies in the data. Notice that the strongest vibrations occur at frequencies between 150 Hz and 400 Hz. This range most likely includes several resonant frequencies from the DUT.

   You also can customize a colormap to display RPM against order. Complete the following steps to compute a colormap for the example data that displays RPM against order.

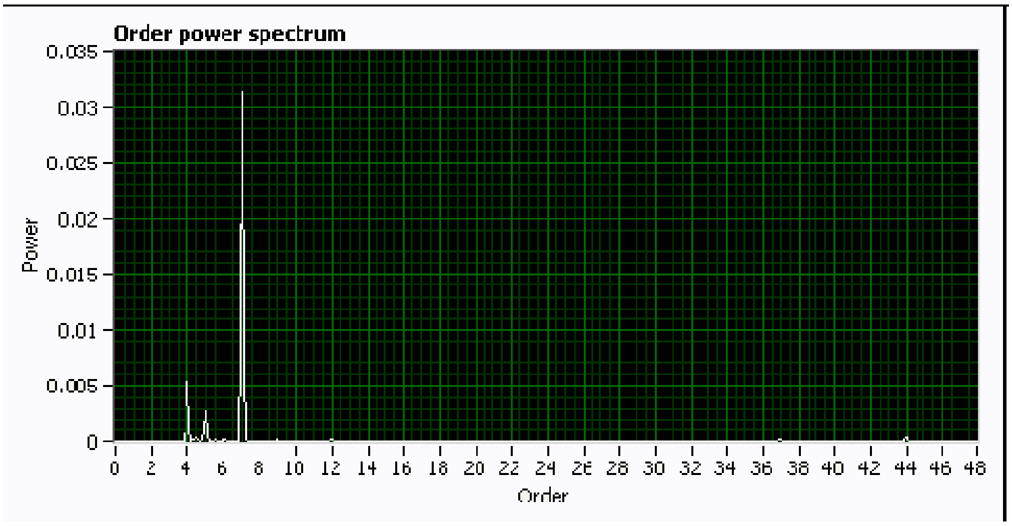
10. On the block diagram, double-click the **Spectral Map Express VI** to display the configuration dialog box.
11. On the Configuration page of the configuration dialog box, change the Plot type to **RPM-Order**. An RPM-Order plot displays how the vibration amplitude at different orders changes with the rotational speed. From the Colormap plot, you can see that the amplitude of the seventh order is the strongest around 2,600 RPM.



With this Colormap plot, you also can identify orders that correspond to loud noises or strong vibrations that you observe. For example, suppose the computer fan generates loud noises between 2,800 RPM and 3,000 RPM. From the RPM-Order plot, you can see that the fourth and the eighth order contribute the most to the loud noises in this speed range.

12. Click the OK button to save the current configuration and close the configuration dialog box. A colormap plot therefore provides an overview of how the intensity of a signal relates to time, speed, frequency, and order.

13. **Analyzing the Order Power Spectrum**
    An order power spectrum provides a quantitative description of the rotation-related components of a signal. You can use an order power spectrum to find and compare significant orders.
    You can use the **Order Spectrum Express VI** to compute the order power spectrum of a sound or vibration signal. As you observed with the **Colormap** plot, the vibration amplitude of the seventh order is the strongest around 2,600 RPM. You can use the **Order Spectrum Express VI** to compute the order power spectrum of the example data at 2,600 RPM and to find the significant orders.
14. On the front panel, click the **Order Power Spectrum** tab to display the **Order power spectrum** plot for the computer fan run-up data.

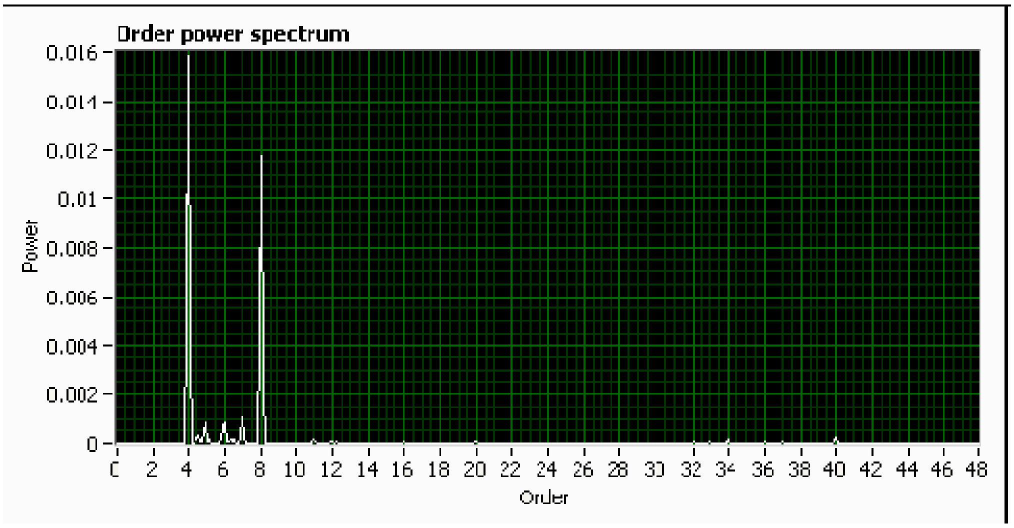


The Order power spectrum plot displays the order amplitudes at 2,600 RPM. Notice that the seventh order has the strongest amplitude on the plot. The fourth order also is strong at 2,600 RPM. The seventh and fourth orders correspond to the **seven blades** of the fan and **four coils** of the electric motor, respectively.
Similarly, as you observed with the RPM-Order plot in the Spectral Map Express VI, the seventh order loses amplitude at around 2,900 RPM, and the eighth order gains amplitude around this rotational speed.

Complete the following steps to compute and display the order power spectrum of the computer fan run-up data around 2,900 RPM.

15. On the block diagram, double-click the **Order Spectrum Express VI** to display the configuration dialog box.
16. On the **Processing Settings** page of the configuration dialog box, set the **Start** speed (RPM) to **2900**, and set the **End** speed (RPM) to **2900**.
17. Click the OK button to save the current configuration and close the configuration dialog box.
18. Run the VI.

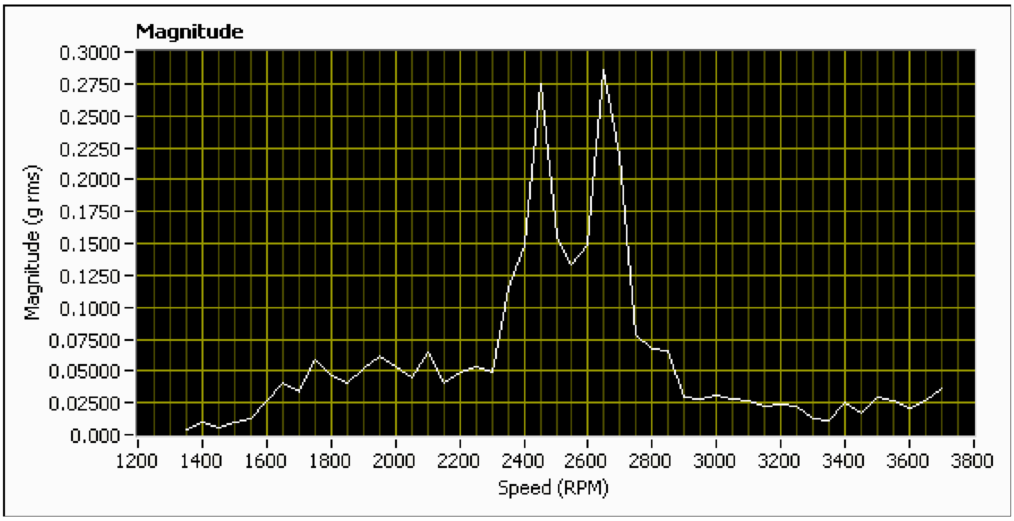
    On the front panel, you can see that the eighth order now has a much greater amplitude than the seventh order does.



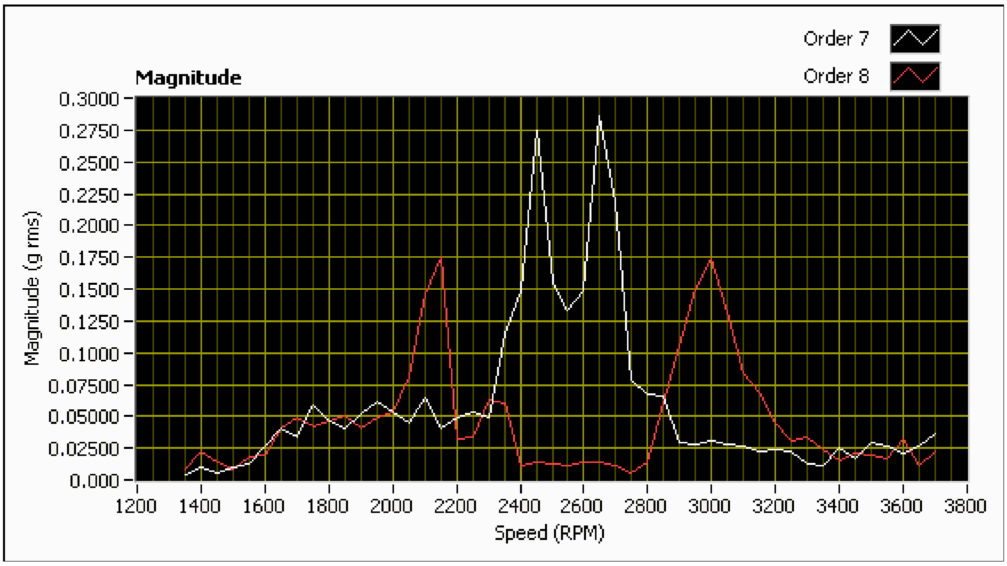
    Again, the fourth order also is significantly strong at around 2,900 RPM. The eighth order and other order multiples of four correspond to the four coils of the electric motor of the computer fan. An order power spectrum plot therefore provides detailed information about the strength of each order at a specific speed. You also can use the Order Spectrum Express VI to compute the strength of each order at a specific time and to perform spectrum averaging.

19. **Analyzing the Magnitude Plot**

    A magnitude plot can help you analyze a sound or vibration signal by focusing on particular orders. Use the **Order Tracking Express VI** to compute the magnitude of designated orders. Recall that the **Colormap** plot showed the seventh order of the computer fan run-up data has strong amplitude at 2,600 RPM. You can use a magnitude plot to observe this order in more detail.
20. On the front panel, click the **Order Tracking** tab to display the **Magnitude** plot for the seventh order of the computer fan run-up data. You can see that this order has greater amplitude around 2,600 RPM, with a dip between 2,450 RPM and 2,650 RPM. This dip is difficult to observe with the colormap alone.



You also can use the **Order Tracking Express VI** to compare the magnitudes of different orders. For example, if you add a second element to the Orders to track input and set the element to 8, the Magnitude plot displays the seventh and eighth orders together.

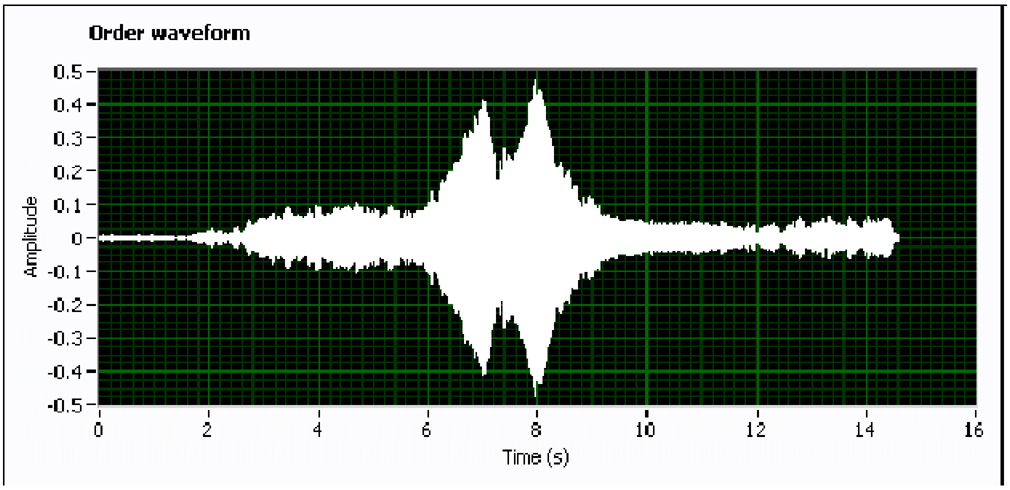


As you observed with both the **Colormap** plot and the **Order power spectrum** plot, the amplitude of the eighth order increases as the amplitude of the seventh order decreases between 2,800 RPM and 3,000 RPM. The **Magnitude** plot provides a detailed view of exactly how the two orders change in magnitude with changing speed. In the configuration dialog box of the Order Tracking Express VI, you can set the **X-axis selection** of the **Magnitude** plot to display the order magnitude against time, speed, or number of revolutions. You also can use the **Magnitude View** options to specify the type of quantitative measurement you want to use to calculate and display the magnitude. Whereas an order power spectrum shows the magnitude or power values of all orders at a specific period in time or at a specific speed, the **Magnitude** plot provides detailed information about particular orders that you specify.

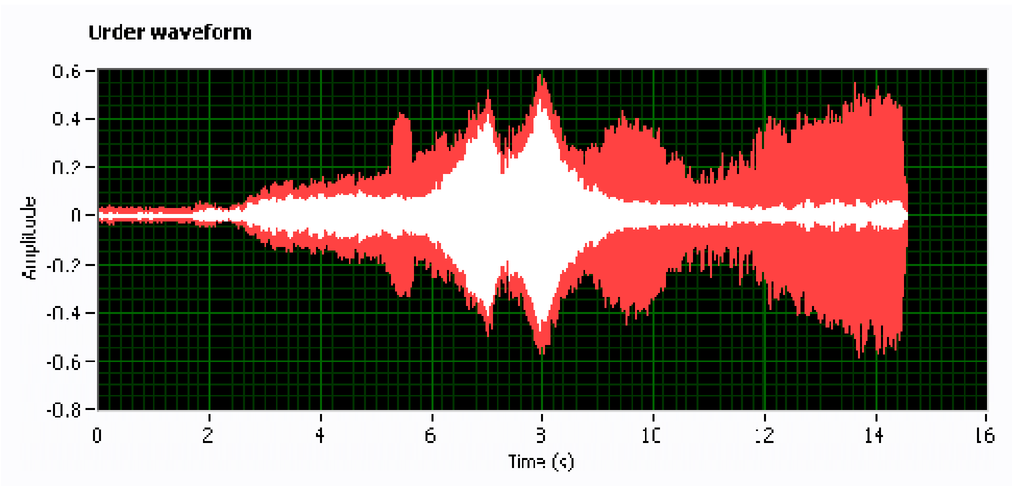
21. **Extracting Order Waveforms**

    You can extract order waveforms to isolate specific orders from a sound or vibration signal. The order waveform provides detailed information about the specific order you extracted from the input signal. You then can perform further analysis, such as sound playback and sound synthesis, on these significant orders. Use the Order Waveform Express VI to extract order waveforms. As you observed with the **Colormap, Order power spectrum**, and **Magnitude plots**, the seventh order is one of the orders that has strong amplitude in the computer fan run-up data. You can use the Order Waveform Express VI to extract and display the order waveform of the seventh order of the computer fan run-up data.

22. On the front panel, click the Order Waveform tab to display the Order waveform plot of the seventh order.



You also can use the Order Waveform Express VI to extract multiple orders of a signal that generate the greatest amplitude, or produce the strongest noise. As you observed with the **Colormap, Order power spectrum**, and **Magnitude** plots, the fourth, seventh, and eighth orders have relatively strong amplitudes in the computer fan run-up data.
On the front panel, the **Order waveform** plot now displays both the waveform of the seventh order and the waveform of the combination of the fourth, seventh, and eighth orders.

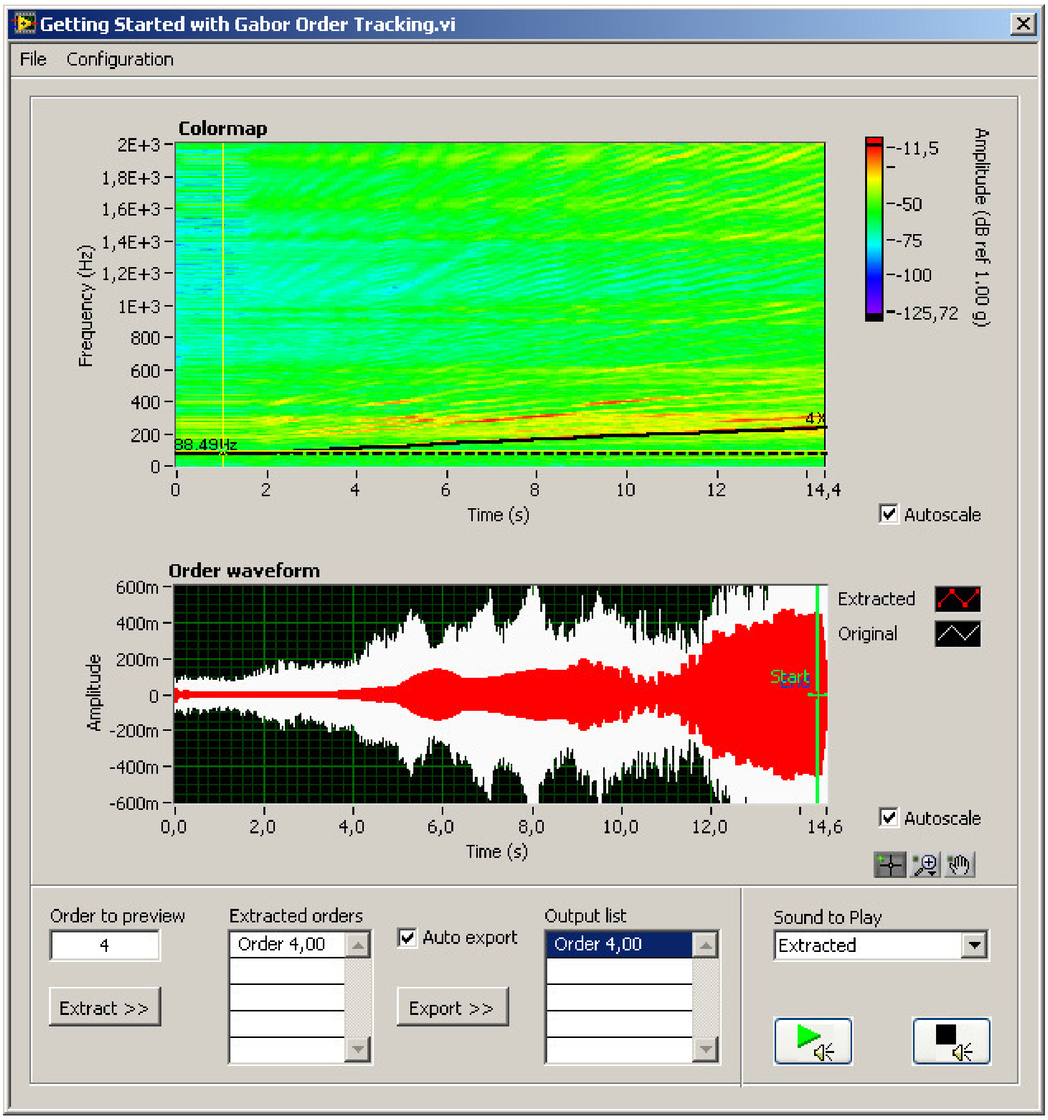


You can add different order waveforms together in different combinations. You then can play back the resulting waveforms as sounds and determine which combination is the most pleasing to the ear. This way, you can determine how to reduce the harshness of the signal and how to remove undesirable sounds and vibrations.

A good example for Extracting Order Waveforms is the **Getting Started with Gabor Order Tracking.vi** in Example Finder.

Open Example Finder in LabVIEW.
Help >> Example Finder >> Toolkits and Modules >> Sound and Vibration >> Getting Started >> Getting Started with Gabor Order Tracking.vi.
Select an order with the mouse or write an order in the **Order to preview** control.

Click on **Extract >>** button to extract this order.
Turn the PC speakers on.
Press **Play** button to listen the extracted order.

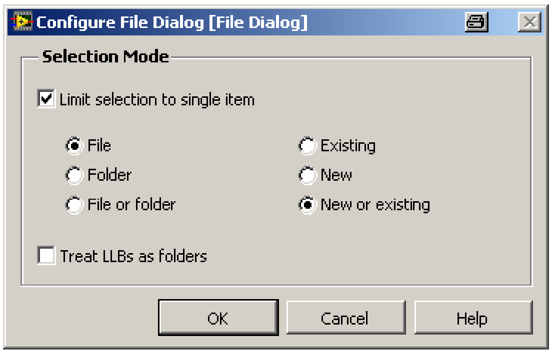


Exercise 7: Save date in TDMS format, load and analyze data with DIAdem

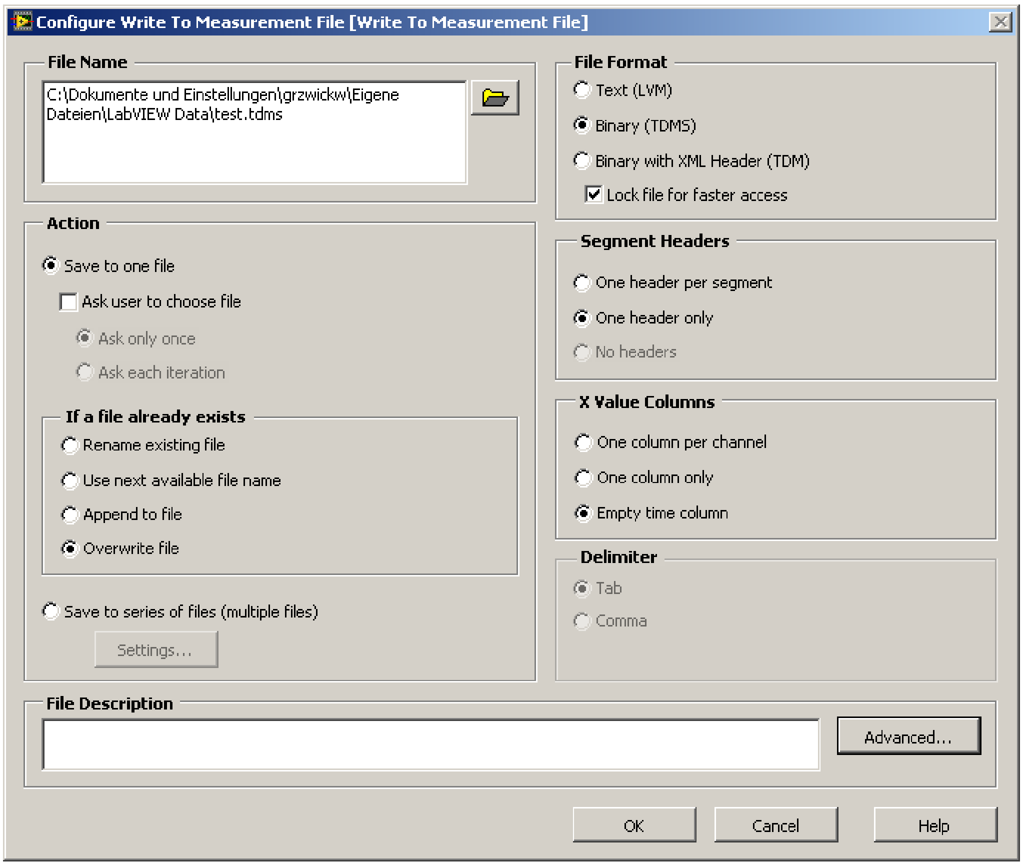
In this Exercise you will expand the **Order Analysis.vi** from Exercises 6, Part 1 with the **Write To Measurement File Express VI**.

1. **Open** the **Order Analysis.vi** from Exercise 6.
2. On the **Programming** palette, select the **File I/O**. There select **Advanced File Functions >> File Dialog** and place it on the left side of the while loop on block diagram.

3. Configure File Dialog.

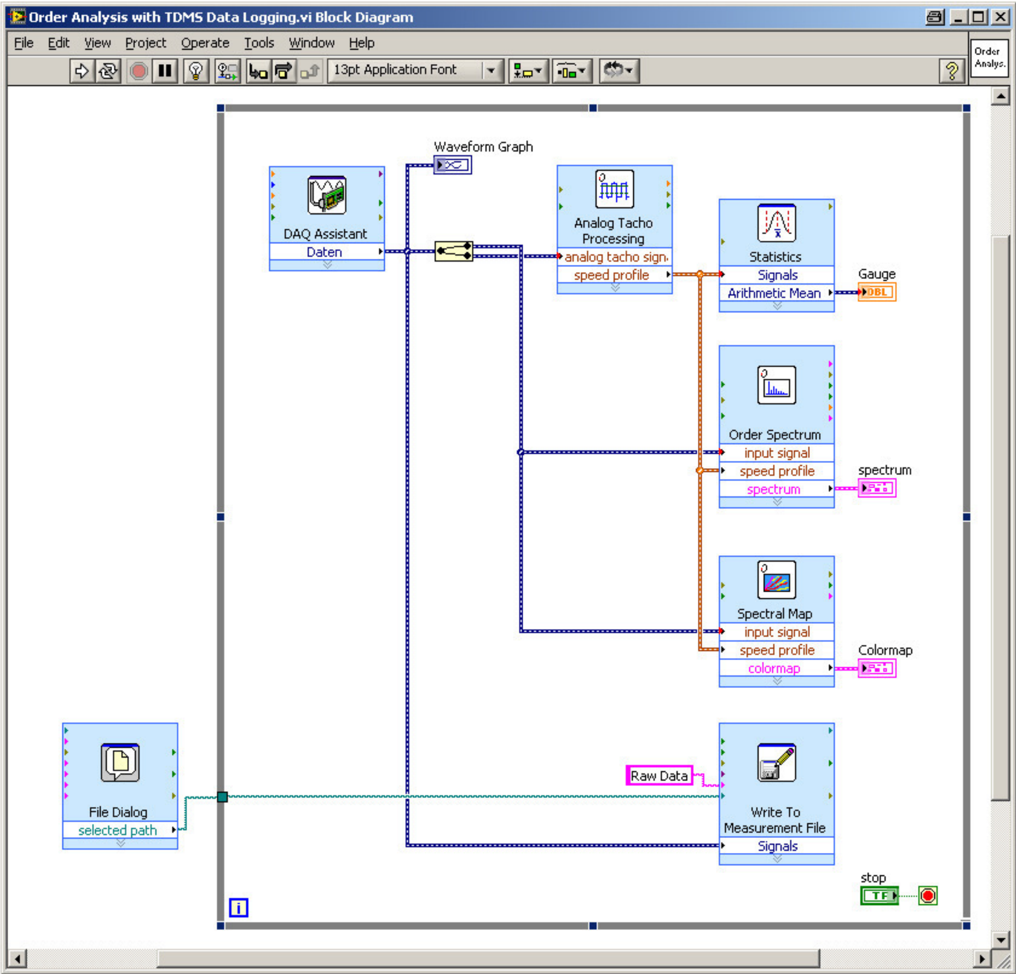


4. **Ok**.
5. On the **Programming** palette, select the **File I/O >> Write To Measurement File Express VI**.
6. Configure Write to Measurement File Dialog.



7. To finish the VI wire the File Dialog Express VI and the Write to Measurement File Express VI. Right click on **Comment** input connector of **Write to Measurement Express VI** and **Create >> Constant.** Write **Raw Date** in constant.

   Here is what your finished VI should look like:



8. Save VI as **Order Analysis with TDMS Data Logging.vi** in **Exercise 7** folder.
9. Run VI circa 5 to 10 seconds.
   Save logged date in **Exercise 7** folder. Choose any name.
10. Open DIAdem.
11. Select module **Navigator**.
12. Drag & Drop the logged data in **Datenportal**.
13. Select module **View** to analyze the data. Zoom in the data to discover the signal from analog tacho.

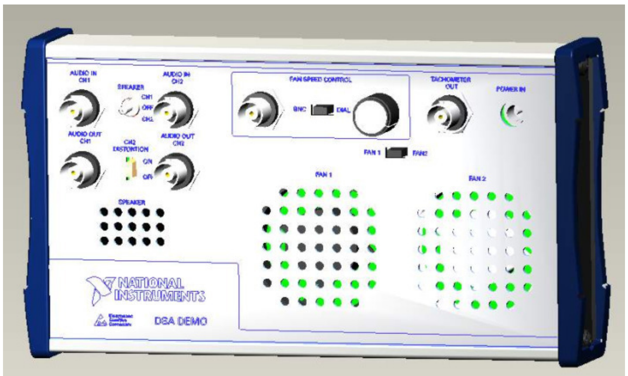
Appendix

Topics

A. DSA Demo Box
B. NI-USB 9233

A. DSA Demo Box

SPECIFICATIONS



Analog Inputs:

Audio Input	
Expected:	± 2.83V
Absolute Rating:	± 10V
Fan Speed Input	
Expected:	0 -> 5 VDC
Absolute Rating:	± 10V

Analog Outputs:

Audio Output	2.83 V peak, 6V DC offset
Accelerometer Outputs	2.5 V peak, 2.5V DC offset
Tachometer Output:	0 -> 5 V square wave, 2 pulses per revolution

All inputs and outputs can be shorted to ground indefinitely.

Power Supply:	15 VDC, 0.8A
Dimensions:	20.3 by 11.2 by 5.5 cm (8.0 by 4.4 by 2.2 in.)
Required Software:	LabVIEW with NI-DAQmx
	Sound and Vibration Toolkit
	Order Analysis Toolkit
I/O Connectors:	BNC
IEPE Compatibility:	Accelerometer and Tachometer Outputs Only

BLOCK DIAGRAMS

The high-level functionality of the DSA demo box can be broken up into two sections: audio and vibration. Block diagrams depicting each can be seen in figures 1 and 2 below. More extensive explanation of the schematic and internal circuitry can be found in the design analysis document.

AUDIO SECTION

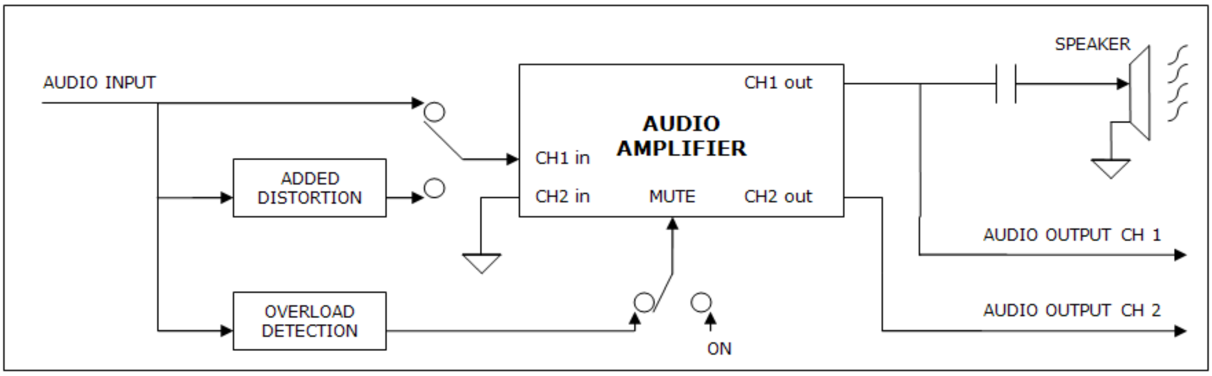


Figure 1. Audio Section Block Diagram

An audio signal is generated from an analog output channel on a DSA board. This signal is connected to the audio input of the DSA demo box. The audio signal then passes through a unity gain audio amplifier IC which drives a speaker with the input waveform. The audio output can also be connected to the analog input channel of a DSA board.

The user has the option (in the form of a slide switch) of adding additional distortion into the signal before it enters the audio amplifier. This distortion circuitry clips the input waveform if it rises above a diode drop, ~ 0.6Vpeak. The resulting signal is then amplified and then appears on the speaker and analog input of the DSA board.

If the input audio signal is too high, the overload detection circuitry will automatically mute the audio amplifier. The user may also choose to mute the system at any time, so no signal will appear on the output of the audio amplifier.

VIBRATION SECTION

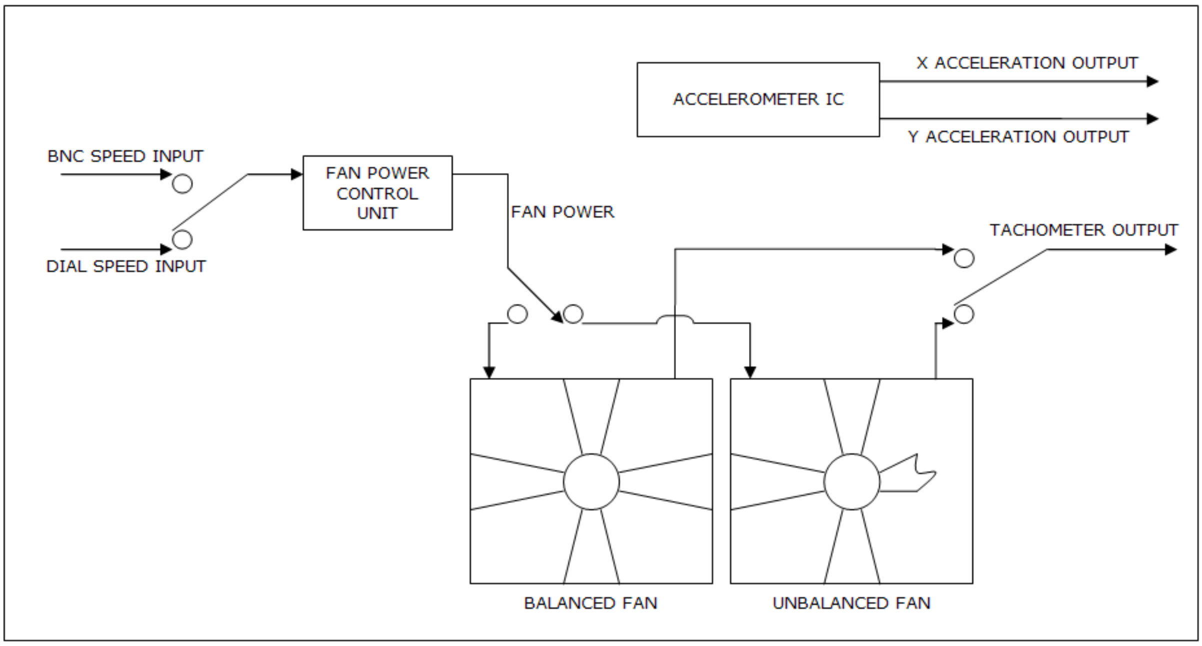


Figure 2. Vibration Section Block Diagram

Using either the BNC input or dial, a user can control the voltage applied to either a balanced or unbalanced fan. The fans are identical except that the unbalanced fan has one of its fins cut off. A tachometer pulse is generated from the fans and sent to an analog input of a DSA board. The tachometer pulse occurs twice per revolution of the fan. Slide switches control whether the BNC or dial input is selected, and which fan will be active.

An accelerometer IC returns a voltage proportional to the acceleration it feels in both the X and Y directions of the box. These voltages are connected to the analog inputs of a DSA board along with the tachometer pulse.

POWER SUPPLY CONSIDERATIONS

Originally, the demo box was intended to be compatible with any voltage supply from 15VDC to 30VDC. However, due to power dissipation on the board, the demo box will only take an input of 15VDC. The minimum requirement is also 15VDC. Therefore, only a 15VDC supply will be input into the demo box.

The unregulated power connected to the DC power jack on the demo box is stepped down to 12VDC using a voltage regulator. This particular device can dissipate about 2.4W of power without a heat sink. Due to space limitations on the board, a heat sink was not added. If a 10˚C/W heat sink were added, the regulator could dissipate about 6.5W of power, allowing for an unregulated DC supply of 15VDC up to 20VDC.

SPEAKER OVERLOAD PROTECTION

The speaker is rated at a maximum power dissipation of 2W. It is an 8Ω speaker which means that no more than a 4V peak signal should be applied across its leads.

The audio circuitry is set to have unity gain, so if a 4V peak signal is applied to the audio input of the demo box, it will appear on the speaker. However, analog outputs of DSA cards reach peak amplitudes of 10V. To ensure that the speaker is not damaged, special circuitry was added to detect the peak amplitude of the input signal.

If the voltage applied at the input rises above 3.5V, then the audio amplifier's mute feature will become active, and will not disengage until the input signal falls back into the rated range for the speaker. The 'overload' LED will light up on the demo box if this protection feature is active.

AUDIO AMPLIFIER

The audio amplifier IC used in the demo box is the TPA1517 from Texas Instruments. It is a two-channel IC capable of driving audio at 6W continuously. It has three modes of operation: normal, mute, and standby. Normal mode is when the audio signal on the input appears on the output with a fixed gain of 20dB. Mute mode disables the output of the amplifier but keeps the quiescent current at the same level. Standby mode minimizes power consumption and effectively turns off the audio amplifier IC. Normal mode and mute mode are used in the DSA demo box.

ACCELEROMETER

The ADXL320JCP from Analog Devices serves as the accelerometer for the demo box. Since both the fans and IC will be mounted to the PCB, the vibrations of the fan should be picked up by the ADXL320. The IC detects acceleration in two directions, X and Y. It has been placed directly in between the balanced and unbalanced fan, so it pick up vibrations from both fairly well. The X and Y axes of the demo box are shown in figure 3.

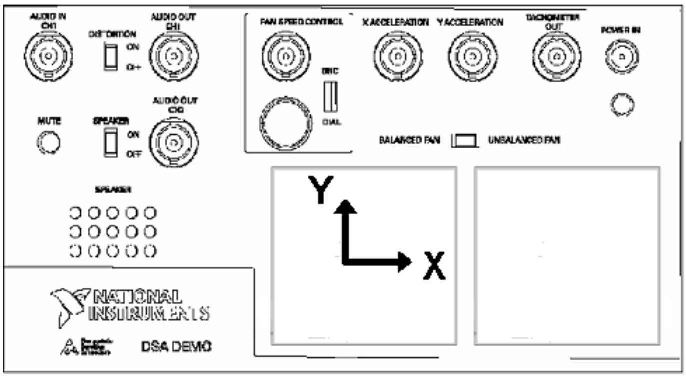


Figure 3: X and Y axes of the demo box

DC FAN - VIBRATION EXPECTATIONS

The demo box fans are 50mm square DC fans from Delta Products, rated at 1.2W @ 12VDC. They are four pole fans with seven blades each. Therefore, 4th and 7th order vibrations are expected. The unbalanced fan has one of its blades cut off so it should have significant 1st order vibrations as well. The fans will be screwed between the top plate and the PCB so their vibrations should shake the PCB.

B. NI USB-9233

Specifications

The following specifications are typical for the range 0 to 60 °C unless otherwise noted.

Input Characteristics

Number of channels........................... 4 analog input channels
ADC resolution .... 24 bits
Type of ADC .... Delta-sigma
(with analog prefiltering)

Data rate (fs)
Minimum... 2 kS/s
Maximum.. 50 kS/s

Master timebase (internal)
Frequency.. 12.8 MHz
Accuracy .... ±100 ppm max

Input coupling .... AC

AC cutoff frequency
-3 dB.. 0.5 Hz typ
-0.1 dB.. 4.2 Hz max

AC voltage full-scale range
Typical .... 5.4 V_{peak}
Minimum.. 5 V_{peak}
Maximum... 5.8 V_{peak}

Common-mode voltage
(AI- to earth ground)...................... ±2 V

IEPE excitation current
Minimum.. 2.0 mA
Typical... 2.2 mA

IEPE compliance voltage.................... 19 V max

Overvoltage protection (with respect to chassis ground)
For an IEPE sensor connected
to AI+ and AI- .... ±30 V
For a low-impedance source connected
to AI+ and AI- .... -6 to 30 V

Input delay (in seconds)
≤ 25 kS/s .... 12.8 / fs
> 25 kS/s... 9.8 / fs

Accuracy (0 to 60 °C)

Error	Accuracy
Calibrated max	±0.3 dB
Calibrated typ	±0.1 dB
Uncalibrated max	±0.6 dB

Accuracy drift
Typical .... 0.001 dB/°C
Maximum.. 0.0045 dB/°C

Channel-to-channel matching
Gain
Maximum.................................... 0.27 dB
Typical... 0.07 dB
Phase (*fin* in kHz)............................ *fin* · 0.077° + 0.067°

Dynamic characteristics

	Passband			
fs	Freq	Flatness (pk-to-pk max)	Freq	Phase Nonlinearity
≤25 kS/s	0.45 · *fs*	0.05 dB	0.45 · *fs*	±3.4°
>25 kS/s	0.42 · *fs*	0.05 dB	0.41 · *fs*	±1.3°

	Stopband		Oversample	Alias-Free
fs	Freq	Attenuation	Rate	Bandwidth
≤25 kS/s	0.58 · *fs*	95 dB	128 · *fs*	0.42 · *fs*
>25 kS/s	0.68 · *fs*	92 dB	64 · *fs*	0.32 · *fs*

Crosstalk
Paired channels
(0 and 1, 2 and 3)........................... -100 dB at 1 kHz
Nonpaired channels....................... -110 dB at 1 kHz

Common-mode rejection ratio (CMRR)
Minimum... 54 dB, *fin* ≤ 1 kHz
Typical .... 80 dB, *fin* ≤ 1 kHz

Spurious-free
dynamic range (SFDR)..................... 120 dB (*fin* = 1 kHz, -60 dB FS)

Idle channel noise and noise density

Idle Channel	50 kS/s	25 kS/s	2 kS/s
Noise	95 dB FS	98 dB FS	102 dB FS
Noise density	400 nV/√Hz	400 nV/√Hz	900 nV/√Hz

Input impedance
Differential (AC)............................. >300 kΩ
AI– (shield) to chassis ground........ 50 Ω

Distortion
Harmonic (THD)

	1 kHz, -40 to 70 °C	10 kHz, 25 to 70 °C	10 kHz, -40 to 25 °C
-1 dB FS	-90 dB	-80 dB	
-20 dB FS	-95 dB	-90 dB	-80 dB

Intermodulation (full-scale input)
DIN 250 Hz/8 kHz
4:1 amplitude ratio..................... -80 dB
CCIF 11 kHz/12 kHz
1:1 amplitude ratio..................... -93 dB