

LabVIEW™

Sound and Vibration Toolset User Manual

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About This Manual

This manual contains information about the LabVIEW Sound and Vibration Toolset (SVT), including brief discussions of the different measurements you can perform, brief discussions of the theory related to those measurements, and suggestions for getting started with the toolset.

Conventions

The following conventions appear in this manual:

[]

Square brackets enclose the units of measure specified or returned by a parameter—for example, [dB].

»

The » symbol leads you through nested menu items and dialog box options to a final action. The sequence **File»Page Setup»Options** directs you to pull down the **File** menu, select the **Page Setup** item, and select **Options** from the last dialog box.



This icon denotes a note, which alerts you to important information.

bold

Bold text denotes items that you must select or click on in the software, such as menu items and dialog box options. Bold text also denotes parameter names, controls and buttons on the front panel, dialog boxes, menu names, and palette names.

italic

Italic text denotes variables and cross references.

monospace

Text in this font denotes text or characters that you should enter from the keyboard, paths, directories, variables, and filenames and extensions.

Related Documentation

The following documents contain information that you might find helpful as you read this manual:

- *LabVIEW Sound and Vibration Toolset Help*, available by selecting **Help»Sound & Vibration**
- *Getting Started with LabVIEW*
- *LabVIEW User Manual*
- *LabVIEW Help*, available by selecting **Help»VI, Function, & How-To Help**
- *LabVIEW Measurements Manual*

Introduction

This chapter introduces the Sound and Vibration Toolset, its **Functions** and **Controls** palettes, and where to find examples to help you get started using the toolset.

Sound and Vibration Toolset

The Sound and Vibration Toolset is a collection of virtual instruments (VIs) for LabVIEW that you can use to perform typical measurements required by sound and vibration applications. Use the Sound and Vibration Toolset to perform the following sound and vibration measurements:

- Scaling a signal to engineering units (EU)
- Calibrating a measurement channel
- Applying weighting filters
- Integrating time-domain signals
- Performing level measurements
- Performing fractional-octave analysis
- Performing frequency analysis
- Performing transient analysis
- Displaying results

You can use the Sound and Vibration Toolset to perform measurements on digitized or simulated data.

Figure 1-1 illustrates the sound and vibration measurement process.



Note In Figure 1-1, the measurement operations shown on the Analysis line are not necessarily performed simultaneously. The dashed boxes in Figure 1-1 indicate optional measurement operations.

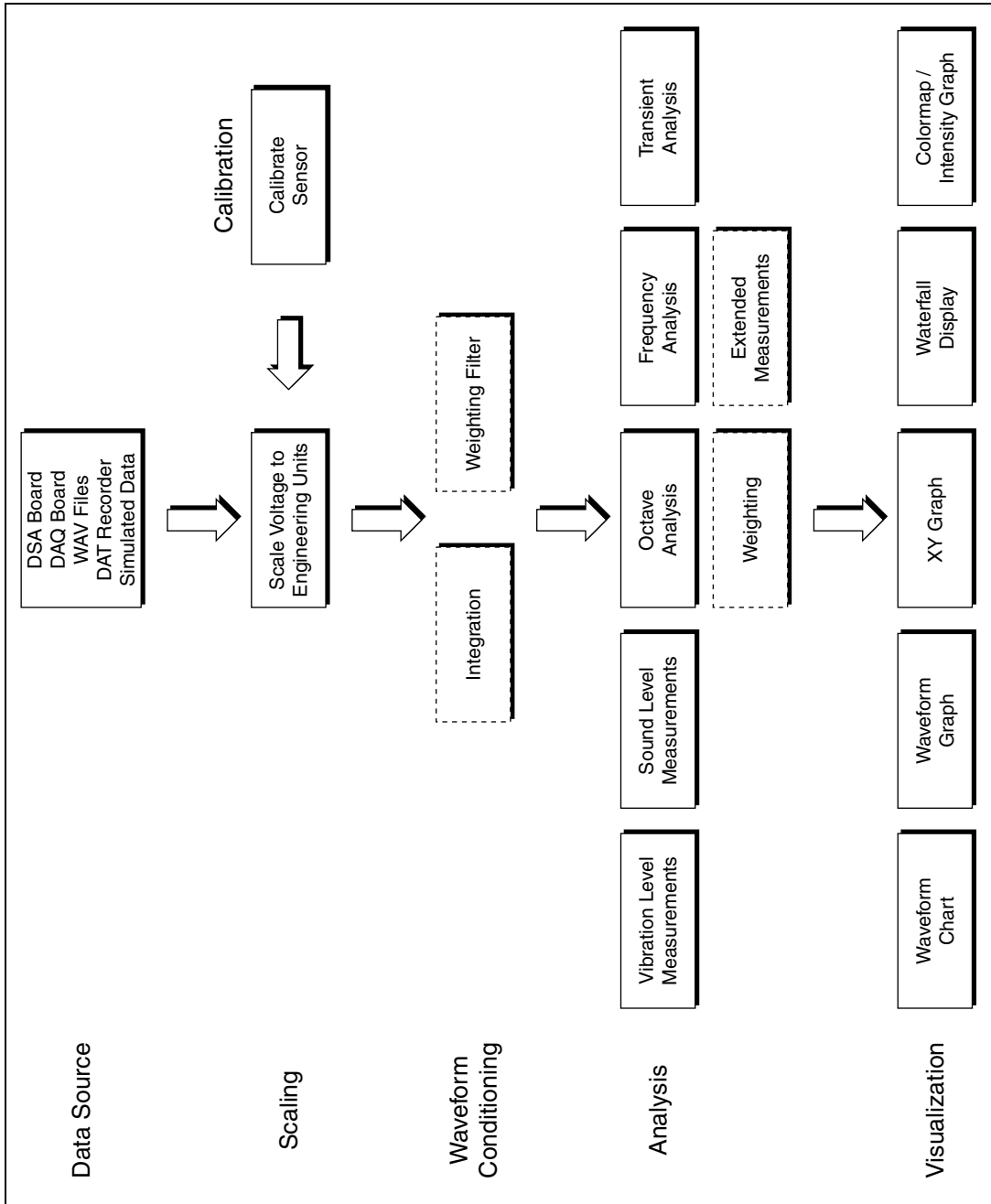


Figure 1-1. Sound and Vibration Toolset Overview

Toolset Palettes

Installing the Sound and Vibration Toolset adds **Sound & Vibration** palettes to both the LabVIEW **Functions** and **Controls** palettes. This section briefly introduces the different palettes that comprise the Sound and Vibration Toolset.

All the high-level VIs in the Sound and Vibration Toolset are designed to offer measurement capabilities. The high-level VIs perform the selected analysis and allow you to view the results with the appropriate engineering units in standard displays, such as magnitude/phase, real/imaginary part, or decibels on/off.

Scaling

The SVS Scale Voltage to EU VI located on the **Functions»Sound & Vibration»Scaling** palette allows you to scale the original signal to engineering units. The SVS Scale Voltage to EU VI is part of the Sound and Vibration Suite (SVS). The SVS is a collection of VIs shared by the Sound and Vibration Toolset and other National Instruments toolsets. Refer to Chapter 2, *Scaling and Calibration*, and to the *LabVIEW Sound and Vibration Toolset Help* for more information about the SVS Scale Voltage to EU VI.

Calibration

The Calibration VIs located on the **Functions»Sound & Vibration»Calibration** palette allow you to perform an end-to-end calibration on a selected channel. Refer to Chapter 2, *Scaling and Calibration*, for information about the calibration process. Refer to the *LabVIEW Sound and Vibration Toolset Help* for information about the individual Calibration VIs.

Weighting

The Weighting VIs located on the **Functions»Sound & Vibration»Weighting** palette allow you to apply A, B, or C-weighting filters on the time-domain signal. Refer to Chapter 3, *Weighting Filters*, for more information about applying weighting to a signal. Refer to the *LabVIEW Sound and Vibration Toolset Help* for information about the individual Weighting VIs.

Integration

The SVT Integration VI on the **Functions»Sound & Vibration»Integration** palette allows you to perform single or double integration on the time-domain signal. Refer to Chapter 4, [Integration](#), for information about the integration process. Refer to the *LabVIEW Sound and Vibration Toolset Help* for information about the SVT Integration VI.

Generation

The SVT Pink Noise Waveform VI located on the **Functions»Sound & Vibration»Generation** palette allows you to generate a continuous pink noise waveform. The **Generation** palette also contains a subpalette linked to the standard **Functions»Analyze»Waveform Generation** palette in LabVIEW. Refer to the *LabVIEW Sound & Vibration Toolset Help* for information about the Generation VIs.

Vibration Level

The Vibration Level VIs located on the **Functions»Sound & Vibration»Vibration Level** palette offer level measurements typically used for vibration measurements, including measuring the crest factor. Averaging modes include RMS averaging, exponential averaging, and peak hold. Refer to Chapter 5, [Vibration Level Measurements](#), for more information about performing vibration level measurements. Refer to the *LabVIEW Sound and Vibration Toolset Help* for information about the individual Vibration Level VIs.

Sound Level

The Sound Level VIs located on the **Functions»Sound & Vibration»Sound Level** palette offer typical sound level measurements, including equivalent continuous averaging (L_{eq}), exponential averaging, and peak hold. Refer to Chapter 6, [Sound Level Measurements](#), for information about performing sound level measurements. Refer to the *LabVIEW Sound and Vibration Toolset Help* for information about the individual Sound Level VIs.

Octave Analysis

The Octave Analysis VIs located on the **Functions»Sound & Vibration»Octave Analysis** palette offer a set of tools to perform fractional-octave analysis, including 1/1, 1/3, 1/6, 1/12, and 1/24 octave-band analysis. The Octave Analysis VIs can accommodate any sampling frequency and any number of fractional-octave bands. Refer to Chapter 7, [Fractional-Octave](#)

Analysis, for information about performing octave analyses. Refer to the *LabVIEW Sound and Vibration Toolset Help* for information about the individual Octave Analysis VIs.

Frequency Analysis

The Frequency Analysis VIs located on the **Functions»Sound & Vibration»Frequency Analysis** palette are a collection of frequency analysis tools based on the discrete Fourier transform (DFT) and the fast Fourier transform (FFT). The Frequency Analysis VIs also provide zoom-FFT frequency measurements and extended measurements. Refer to Chapter 8, *Frequency Analysis*, for information about performing frequency analyses. Refer to the *LabVIEW Sound and Vibration Toolset Help* for information about the individual Frequency Analysis VIs.

Transient Analysis

The Transient Analysis VIs located on the **Functions»Sound & Vibration»Transient Analysis** palette use the short-time Fourier transform (STFT). You can directly call the Transient Analysis VIs to obtain the frequency content of a signal versus time. Alternatively, you can pass the information simultaneously acquired by a tachometer to the Transient Analysis VIs to obtain the frequency content as a function of the rotational speed. Refer to Chapter 9, *Transient Analysis*, for information about performing transient analyses. Refer to the *LabVIEW Sound and Vibration Toolset Help* for information about the individual Transient Analysis VIs.

Waterfall Display

The Waterfall Display VIs located on the **Functions»Sound & Vibration»Waterfall Display** palette allow you to display the results of frequency analyses and octave analyses as waterfall graphs. The Waterfall Display VIs generate and manage the external window of the waterfall graph. Refer to Chapter 10, *Waterfall Display*, for information about displaying analysis results in a waterfall graph. Refer to the *LabVIEW Sound and Vibration Help* for information about the individual Waterfall Display VIs.

Front Panel Displays

The customized graphs located on the **Controls»Sound & Vibration** palette give you a choice for displaying results from octave and transient analyses. Use the Octave Graph and the Multiplot Octave Graph to display results from the Octave Analysis VIs. Use the Colormap to display results from the Transient Analysis VIs. Refer to the *Sound and Vibration Toolset Help* for information about the individual displays.

Examples

The Sound and Vibration Toolset includes examples to help you get started using the toolset. The examples are located in the `examples\Sound and Vibration` directory. The examples demonstrate the following Sound and Vibration Toolset functionality:

- Display
- Frequency analysis
- Integration
- Level measurements
- Octave analysis
- Scaling
- Transient analysis
- Weighting filters

Scaling and Calibration

This chapter discusses using the SVS Scale Voltage to EU VI located on the **Functions»Sound & Vibration»Scaling** palette to scale a signal to engineering units (EU) and using the Calibration VIs located on the **Functions»Sound & Vibration»Calibration** palette to perform an end-to-end calibration on a specific channel.

Figure 2-1 illustrates how scaling and calibration fit into the sound and vibration measurement process

Refer to the *LabVIEW Sound and Vibration Toolset Help* for more information about the SVS Scale Voltage to EU VI and the Calibration VIs.

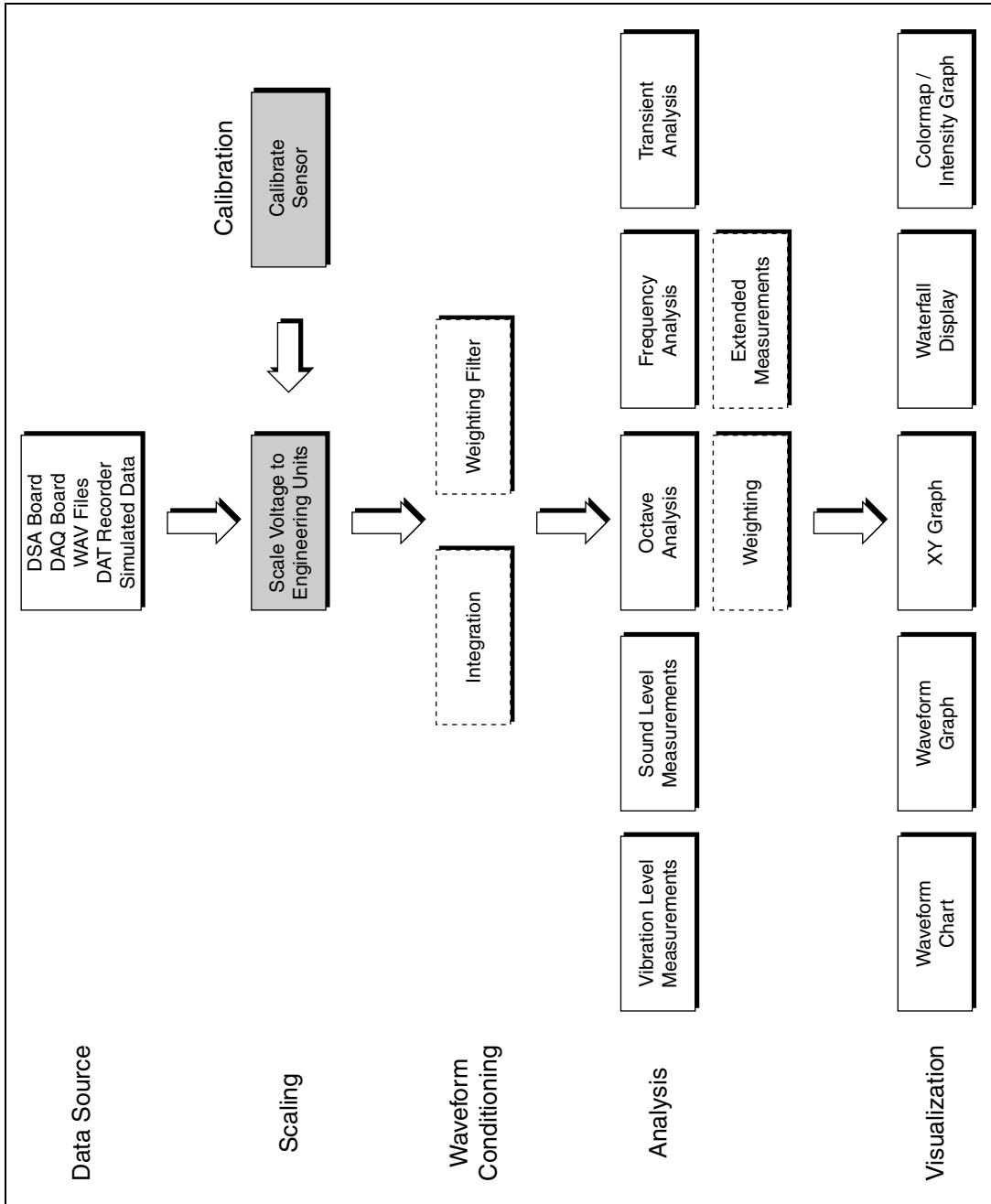


Figure 2-1. Relationship of Scaling to the Sound and Vibration Measurement Process

Scaling to Engineering Units

Typically, scaling a signal to the appropriate engineering units occurs before any analysis is performed. Use the SVS Scale Voltage to EU VI to scale the signal to the appropriate engineering units.

All measurement VIs in the Sound and Vibration Toolset return results with the appropriate units, such as time-domain signals in the correct engineering units, frequency spectra in decibels, phase information in degrees or radians, and so on. To handle units properly, the high-level VIs need the signal to be scaled to the appropriate engineering units.



Note Using a virtual channel in Measurement & Automation Explorer (MAX) is an alternative method of applying scaling information to a waveform. If MAX is used to scale the time-domain signal to the appropriate engineering units, National Instruments does not recommend using the SVS Scale Voltage to EU VI. However, if MAX is used only to name a channel and the units associated with that channel are left in volts, use the SVS Scale Voltage to EU VI to convert the signal from volts to the selected engineering units.

Performing System Calibration

You typically perform system calibration with a dedicated calibrator, such as a pistonphone for microphones or a hand-held shaker for accelerometers. If you are calibrating a microphone, consider using the SVS Calibrate Microphone VI. If you are calibrating an accelerometer, consider using the SVS Calibrate Accelerometer VI. These VIs are very similar to the general-purpose SVS Calibrate Sensor VI but offer the advantage of having default values commonly found for pistonphones or hand-held shakers. All the Calibration VIs use the characteristics of the calibrator, such as reference calibration value and frequency, to perform the calibration.

Weighting Filters

This chapter discusses the use of weighting filters in sound and vibration analysis, including the purpose of weighting filters, the types of weighting filters, and applying weighting to time-domain data, FFT-based spectra, and octave spectra.

The following Sound and Vibration Toolset palettes contain weighting filter VIs:

- **Functions»Sound & Vibration»Weighting**
- **Functions»Sound & Vibration»Octave Analysis»Extended Measurements**
- **Functions»Sound & Vibration»Frequency Analysis»Extended Measurements**

Refer to the *LabVIEW Sound and Vibration Toolset Help* for information about individual weighting filter VIs.

Purpose of Weighting Filters

In many applications involving acoustic measurements, the final sensor is the human ear. In other words, acoustic measurements typically attempt to describe the subjective perception of a sound by the human ear. Because instrumentation devices are usually built to provide a linear response and the ear is a nonlinear sensor, special filters, known as weighting filters, are used to take into account these nonlinearities. A typical microphone frequency response is shown in Figure 3-1 and serves as a good example of the linear response of a sensor.

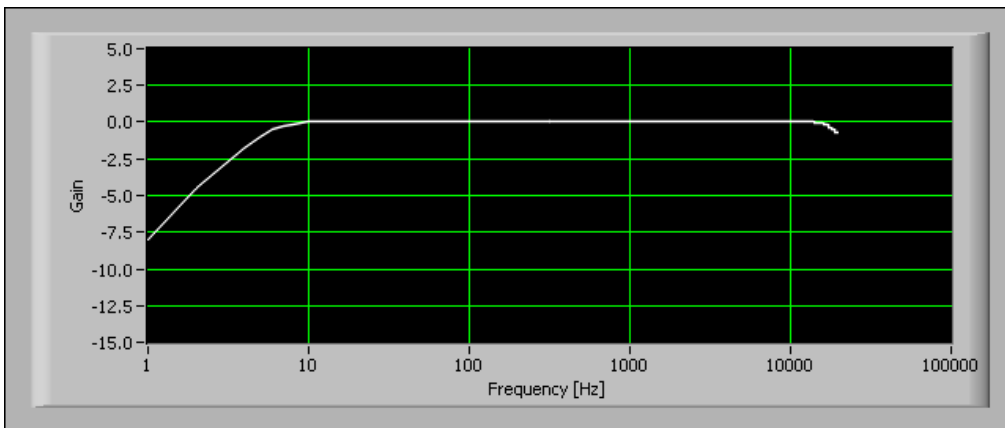


Figure 3-1. Frequency Response of a Typical Instrumentation-Grade Microphone

The frequency response of this microphone is designed to be as flat as possible in the frequency range of 10 Hz to 10 kHz. Compare this frequency response with the equal loudness curves shown in Figure 3-2.

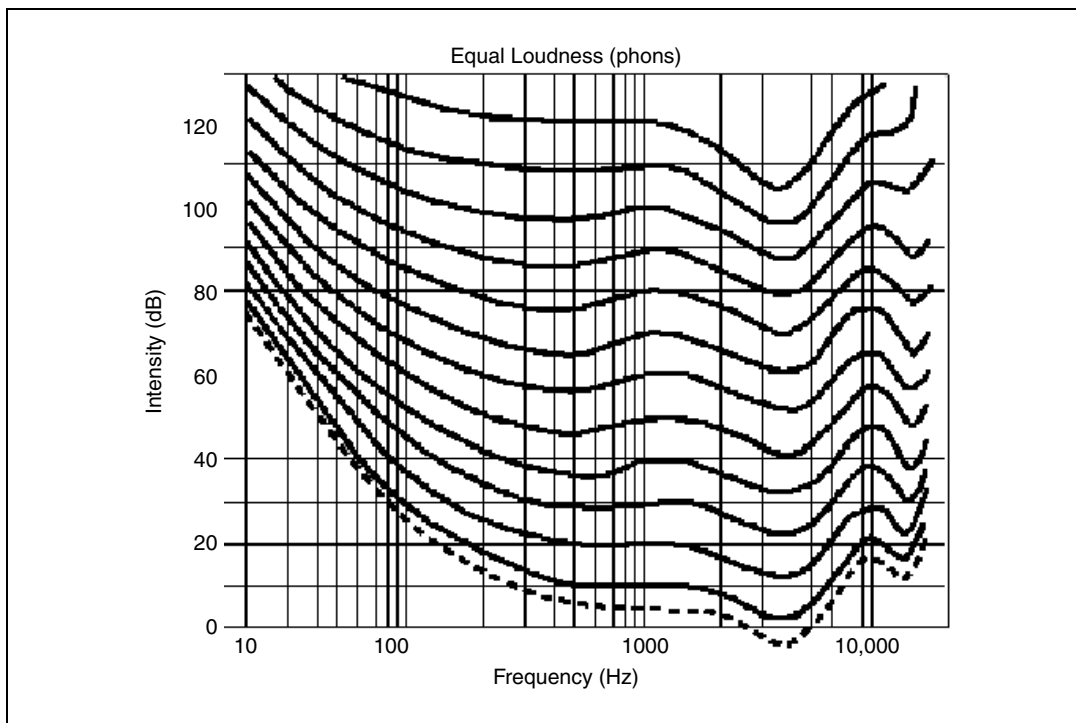


Figure 3-2. Equal Loudness Curves

Loudness is a subjective indicator of the perceived noise level expressed in phons. The loudness level in phons is the sound pressure level in dB of a 1 kHz tone having the same perceived loudness as the tone being evaluated. Thus, a 1 kHz tone with a loudness level of 30 phons is equally as loud as a 1 kHz tone with a sound pressure level of 30 dB ref 20 μ Pa. However, a signal frequency of 100 Hz requires a sound pressure level of 44 dB ref 20 μ Pa to provide the same loudness level.

A-, B-, and C-Weighting Filters

A-, B-, and C-weighting filters were originally designed for the following uses:

- A-weighting is a highpass filter designed to simulate perception of the loudness of low-level tones. A-weighting progressively de-emphasizes frequencies below 500 Hz.
- B-weighting simulates perception of the loudness of medium-level tones. B-weighting is used infrequently.
- C-weighting removes sounds outside the audio range of 20 Hz to 20 kHz and simulates perception of the loudness of high-level tones.

The frequency responses of these filters have been standardized according to the ANSI S1.4 standard. Figure 3-3 shows the relative attenuation defined for A-, B-, and C-weighting filters.

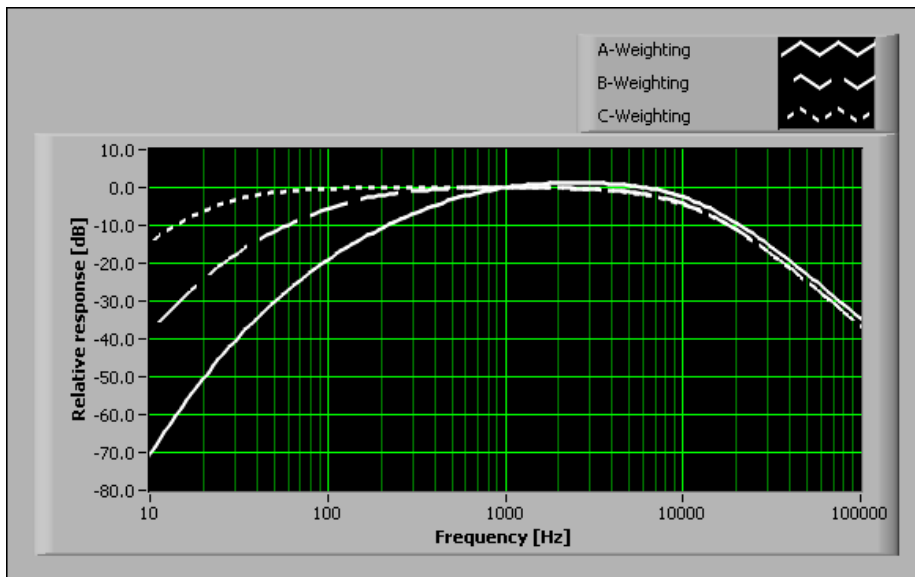


Figure 3-3. Relative Attenuation of A-, B-, and C-weighting Filters



Note Each of these filters has a relative attenuation of 0 dB at 1,000 Hz.

Applying Weighting Filters

Use the Sound and Vibration Toolset to apply weighting on time-domain signals, on FFT-based spectra, or on fractional-octave spectra. Figure 3-4 illustrates how applying weighting fits into the sound and vibration measurement process.

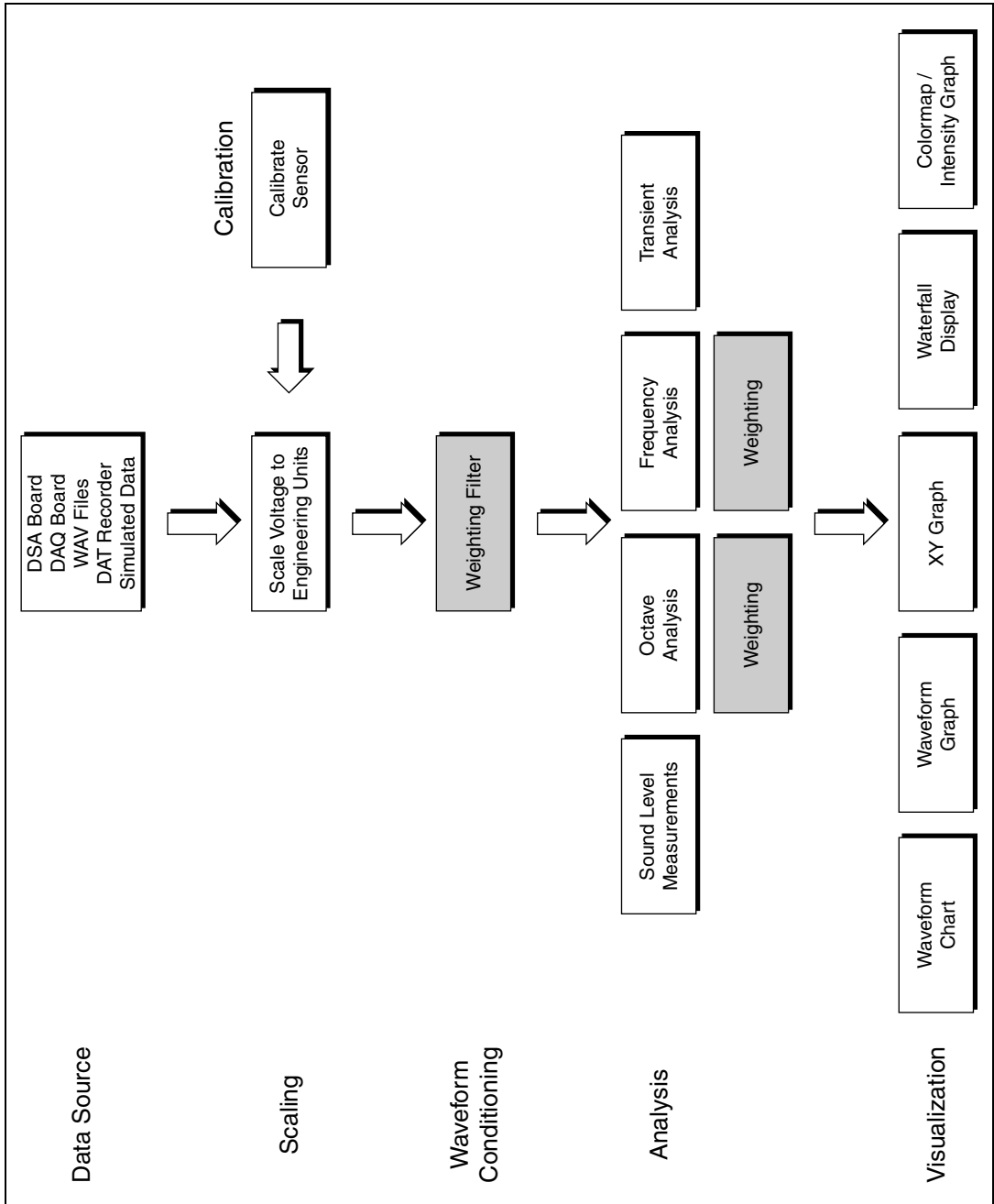


Figure 3-4. Relationship of Weighting to the Sound and Vibration Measurement Process

Traditionally, weighting filters are built using analog components. If you use an external weighting filter, use the **weighting filter** parameter in the **channel info** control of the SVS Scale Voltage to EU VI to ensure proper display of the selected units. Also, the **channel info** parameter reports an error if an attempt is made to apply additional weighting to an already weighted signal.



Note The **weighting filter** parameter assigns the correct units to the value the SVS Scale Voltage to EU VI returns but does not cause the VI to perform any filtering.

Refer to Chapter 2, *Scaling and Calibration*, for information about scaling a signal.

Applying Weighting to Time-Domain Data

This section discusses the ANSI standards pertaining to weighting filters, how to perform A-weighted sound level measurements, and the importance of selecting an appropriate sampling frequency.

ANSI Standards

Use the SVT Weighting Filter VI located on the **Functions»Sound & Vibration»Weighting** palette to apply an A, B, or C-weighting filter on time-domain signals. When combined with any National Instruments dynamic signal acquisition (DSA) board, the weighting filter designed by the SVT Weighting Filter VI complies with the following standards:

- ANSI S1.4-1983
- ANSI S1.42-1986

The SVT Weighting Filter VI accommodates any sampling frequency and designs the filter coefficients to target the attenuation curves defined by the ANSI standards. Given the selected sampling frequency, compliance with a particular filter type, either Type 1 or Type 0, is ensured up to a specific frequency. The specific frequency up to which compliance is ensured is the maximum frequency within tolerances. Use the SVT Max Frequency Within Tolerances VI located on the **Functions»Sound & Vibration»Weighting** palette to determine the maximum frequency within tolerances.



Note Selecting Type 1 or Type 0 has no influence on the design process but specifies how the actual attenuation must be tested against the ideal attenuation. Type 0 leads to a lower maximum frequency within tolerances, or a smaller valid frequency range.

Performing A-Weighted Sound Level Measurements

The block diagram in Figure 3-5 shows a VI that uses weighting filters to perform an A-weighted sound level measurement (L_{AS}) on a simulated signal.

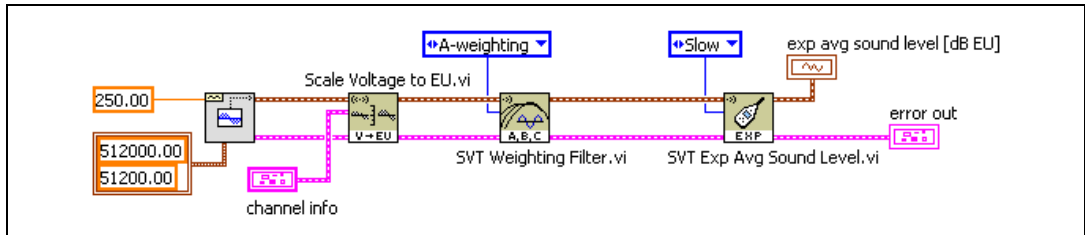


Figure 3-5. A-Weighting of a Simulated Time-Domain Signal Block Diagram

Time-domain data is simulated and scaled before being sent to the SVT Weighting Filter VI. The weighted signal is sent to the Sound Level Measurement VI.

Figure 3-6 shows the time-domain input and output waveforms when a 250 Hz sine wave is input to the SVT Weighting Filter VI and the A-weighting filter is applied.

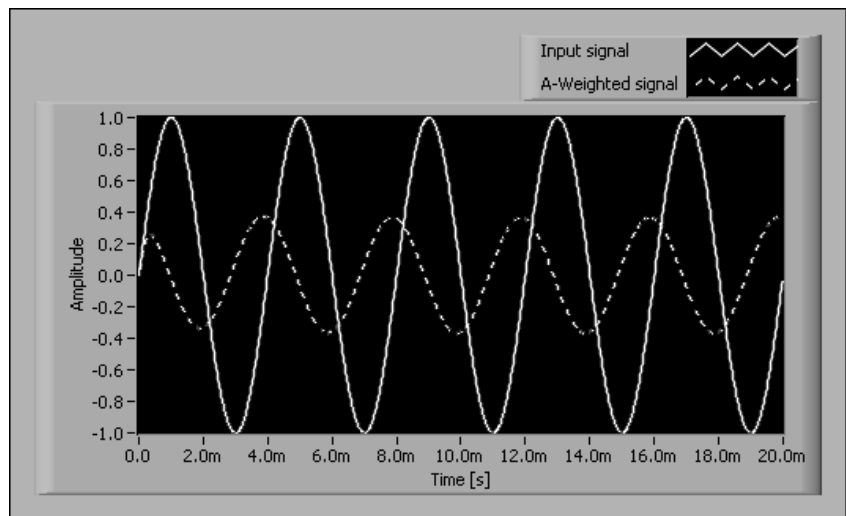


Figure 3-6. A-Weighting Response in the Time Domain

Because weighting in the time domain is applied by a filter, there is a phase difference between the input and output signals. Also, the transient behavior at the beginning of the filtered waveform corresponds to the settling time of the filter.

Selecting the Appropriate Sampling Frequency

The ANSI S1.4 standard defines several filter types based on the accuracy of the weighting filters when compared to the ideal frequency response function. Because the implementation of the weighting filters is based on digital filters, select a sampling frequency that provides the required accuracy. Use the SVT Max Frequency Within Tolerances VI to determine the maximum frequency that maintains results within Type 0 or Type 1 compliance. Figure 3-7 shows the block diagram of an example using the SVT Max Frequency Within Tolerances VI.

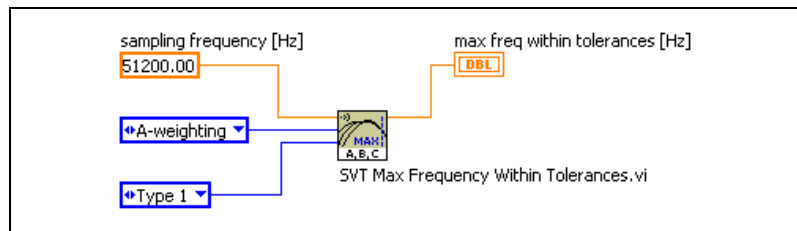


Figure 3-7. Block Diagram Using the SVT Max Frequency Within Tolerances VI

In this example, the sampling frequency is 51,200 Hz. When Type 1 is selected, the SVT Max Frequency Within Tolerances VI returns the maximum frequency within tolerances as 19,987.5 Hz. When Type 0 is selected, the maximum frequency within tolerances is 16,525.0 Hz. If you are trying to make measurements in the 16 Hz–16 kHz range and want to achieve Type 0 compliance for the weighting filters, select a sampling frequency high enough to ensure compliance. For this particular example, the sampling frequency achieves Type 0 compliance.

Applying Weighting to an Octave Spectrum

When performance, such as CPU usage, is an issue, applying weighting in the frequency domain can increase the efficiency of the measurement process. The block diagram in Figure 3-8 uses the SVT Weighting Filter (octave) VI located on the **Functions»Sound & Vibration»Octave Analysis»Extended Measurements** palette to apply weighting to a third-octave spectrum. The weighting method illustrated in Figure 3-8 requires less processing time than applying weighting in the time domain and subsequently computing the third-octave spectrum.

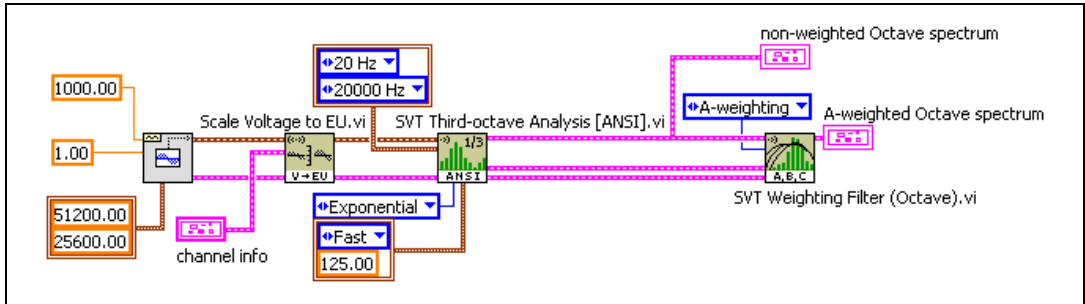


Figure 3-8. Applying Weighting to a Third-Octave Spectrum

Errors Due to Uniform Corrections

When applying weighting to a fractional-octave spectrum, remember that attenuation of the weighting filter is defined by a continuous frequency response function. When mathematically weighting a spectrum consisting of data from a fractional-octave spectrum, the correction values applied for the weighting are equal to the theoretical values at the center frequency of the fractional-octave band. The application of the correction values creates an essentially rectangular filter that is not continuous in its response. The filter applies the same correction to all energy within each fractional-octave band. Applying the correction values to a signal containing a pure tone near one of the fractional-octave filter edges might introduce a measurement error. The error is usually negligible unless an A-weighting filter is used at frequencies below 500 Hz. At frequencies below 500 Hz, the slope of the A-weighting curve is steep, as shown in Figure 3-9.

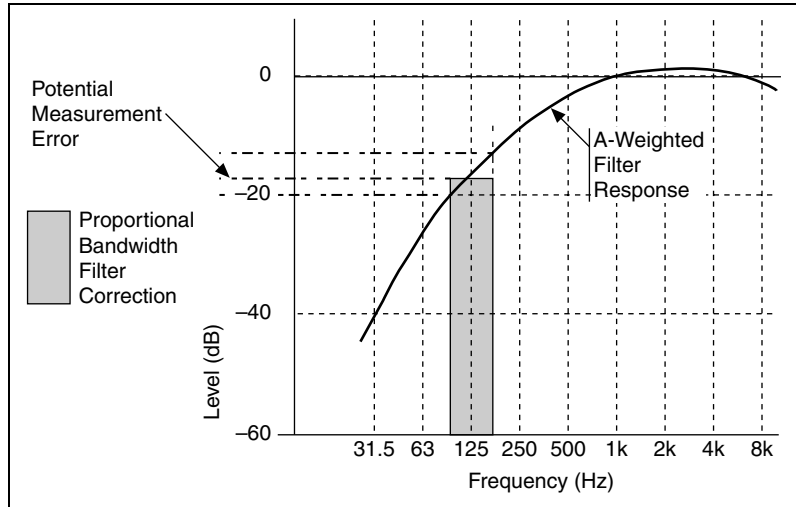


Figure 3-9. Potential Measurement Error for A-Weighting at Low Frequency



Note The same type of measurement error seen in Figure 3-9 can occur when applying weighting to FFT-based spectra. However, the error is almost always negligible as long as the frequency resolution of the spectrum is reasonable. For example, the error is negligible with a frequency resolution of 10 Hz or finer.

Applying Weighting to an FFT-Based Spectrum

The most efficient way to compute a frequency-weighted spectrum is to apply the weighting in the frequency domain. For example, compare the power spectrum of a signal with the power spectrum of the same signal after applying a weighting filter. You can use the SVT Weighting Filter (frequency) VI located on the **Functions»Sound & Vibration»Frequency Analysis»Extended Measurements** palette to apply weighting in the frequency domain, as shown in Figure 3-10.

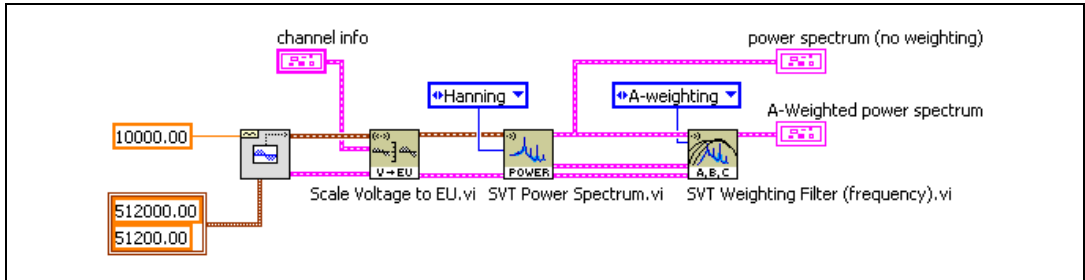


Figure 3-10. Applying Frequency Weighting to a Power Spectrum

Figure 3-11 shows a different implementation based on applying the weighting filter on the time-domain signal and then computing the power spectrum.

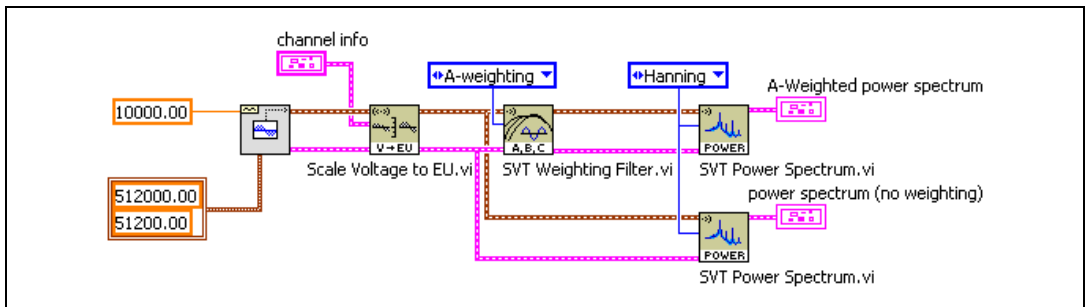


Figure 3-11. Applying Weighting Before Computing the Power Spectrum

In Figure 3-10, a single power spectrum is computed. In Figure 3-11, the power spectrum is computed twice, leading to more CPU usage and increased processing time. By applying the weighting filter in the frequency domain, as shown in Figure 3-10, you can decrease CPU usage and processing time.

Integration

This chapter discusses the integration process, including basic theory and implementation in the time and frequency domains.

The Sound and Vibration Toolset contains the following integration VIs:

- SVT Integration located on the **Functions»Sound & Vibration»Integration** palette for time-domain integration
- SVT Integration (frequency) located on the **Functions»Sound & Vibration»Frequency Analysis»Extended Measurements** palette for frequency-domain integration

Refer to the *LabVIEW Sound and Vibration Toolset Help* for information about the integration VIs.

Introduction to Integration

The conversion between acceleration, velocity, and displacement is based on one of the fundamental laws in Newtonian physics, represented by the following equations.

$$\dot{x} = \frac{d}{dt}(x)$$

$$\ddot{x} = \frac{d}{dt}(\dot{x}) = \frac{d^2}{dt^2}(x)$$

Velocity is the first derivative of displacement with respect to time. Acceleration is the first derivative of velocity and the second derivative of displacement with respect to time. Therefore, given acceleration, perform a single integration with respect to time to compute the velocity and a double integration with respect to time to compute the displacement. For a single integration, you integrate once. For a double integration, you integrate twice.

When representing the acceleration of a point by a simple sinusoid, the velocity and the displacement of the point are well known and represented by the following equations.

$$a = A \sin(\omega t)$$

$$v = -\frac{A}{\omega} \cos(\omega t) = \frac{A}{\omega} \sin\left(\omega t - \frac{\pi}{2}\right) \quad (4-1)$$

$$d = -\frac{A}{\omega^2} \sin(\omega t) = \frac{A}{\omega^2} \sin(\omega t - \pi) \quad (4-2)$$



Note Initial condition is arbitrarily set to zero in Equations 4-1 and 4-2.

The amplitude of the velocity is inversely proportional to the frequency of vibration. The amplitude of the displacement is inversely proportional to the square of the frequency of vibration. Furthermore, the phase of the velocity lags the acceleration by 90° . The phase of the displacement lags the acceleration by 180° . Figure 4-1 illustrates the relationship between acceleration, velocity, and displacement.

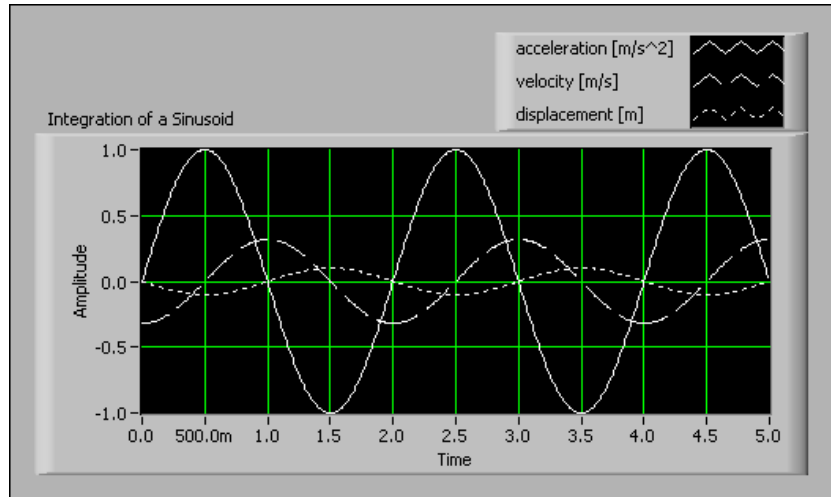


Figure 4-1. Integration of a 0.5 Hz Sine Wave

The integration of a sinusoid is known in closed form. Integration of an arbitrary waveform typically requires a numerical approach. You can use several numerical integration schemes to evaluate an integral in the time domain.

In the frequency domain, you can define any arbitrary band-limited waveform as a sum of sinusoids. Because the amplitude and phase relationships are known for sinusoids, you can carry out the integration in the frequency domain, as seen in Figures 4-1 and 4-3.

Implementing Integration

If you need to perform measurements on velocity or displacement data when only acceleration or velocity data have been acquired, integrate the measured signal to yield the desired quantity. You can perform integration either in the time domain as a form of signal conditioning or in the frequency domain as a final stage of analysis. When performed in the frequency domain, integration is one of the Extended Measurements for Frequency Analysis. Figure 4-2 illustrates how integration fits into the sound and vibration measurement process.

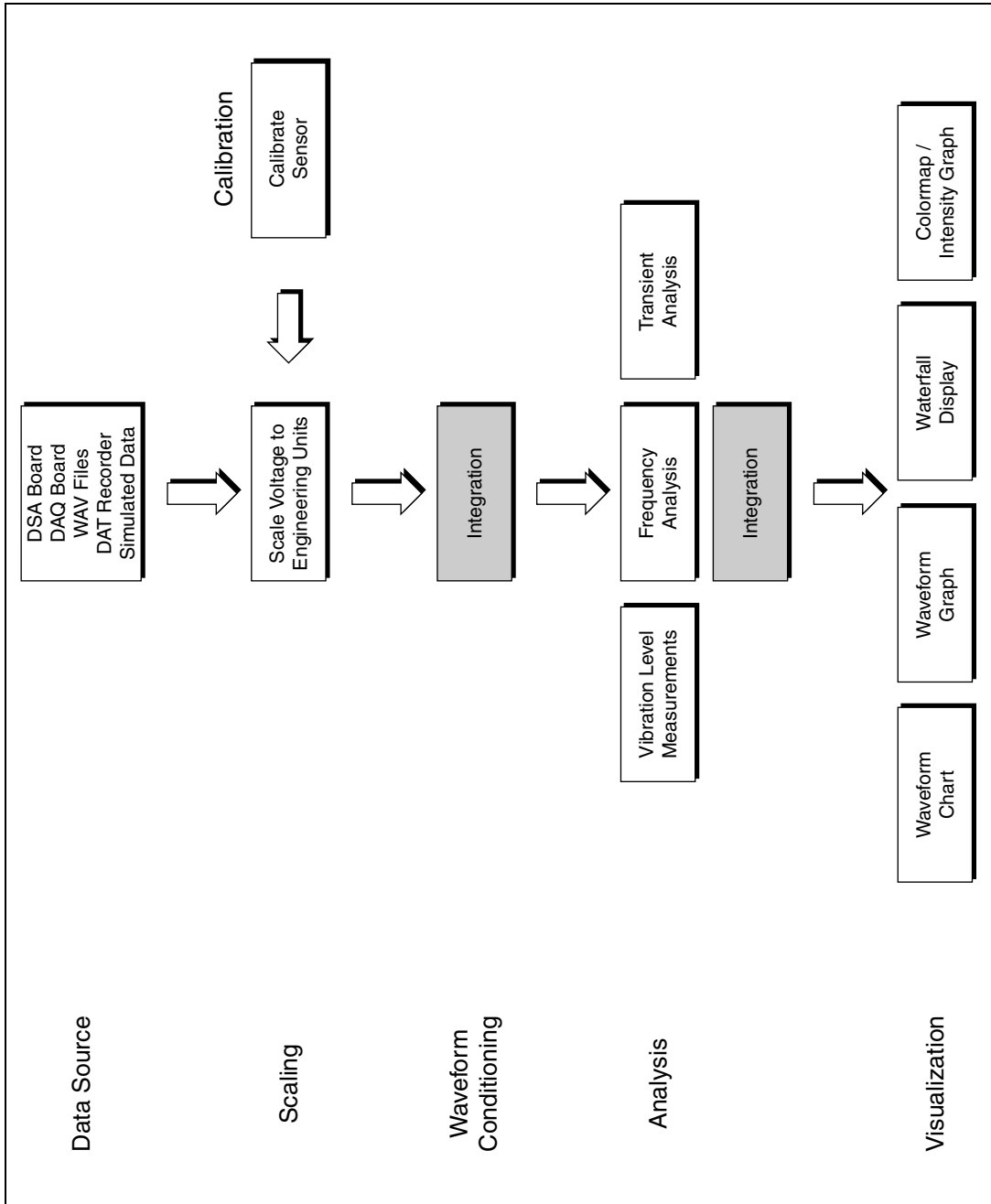


Figure 4-2. Relationship of Integration to the Sound and Vibration Measurement Process

Challenges

Conversion of acceleration data to velocity or displacement data presents the following unique challenges:

- Measured signals typically contain some unwanted DC component.
- Many transducers, especially vibration transducers, have lower-frequency limits. A transducer cannot accurately measure frequency components below the lower-frequency limit of the transducer.

DC Component

Even though a DC component in the measured signal might be valid, the presence of a DC component indicates that the unit under test (UUT) has a net acceleration along the axis of the transducer. For a typical vibration measurement, the UUT is mounted or suspended in the test setup. The net acceleration of the UUT is zero. Therefore, any DC component in the measured acceleration is an artifact and should be ignored.

Transducers

Most acceleration and velocity transducers are not designed to accurately measure frequency components close to DC. Close to DC is relative and depends on the specific transducer. A typical accelerometer can accurately measure components down to about 10 Hz. A typical velocity probe can accurately measure components down to 2–3 Hz. Inaccurately measured low-frequency vibrations can dominate the response when the signal is integrated because integration attenuates low frequency components less than high frequency components.

Implementing Integration using the Sound and Vibration Toolset

Both the SVT Integration VI and the SVT Integration (frequency) VI address the challenges of converting acceleration data to velocity or displacement data.

Time-Domain Integration

This section presents examples of and discussion about time-domain integration.

Single-Shot Acquisition and Integration

The following example shows how you can use integration to convert acceleration data to displacement data in a single-shot acquisition and integration. In this example, the acquired waveform is sampled at 51.2 kHz, and double integrated. Figure 4-3 shows the block diagram for the VI.

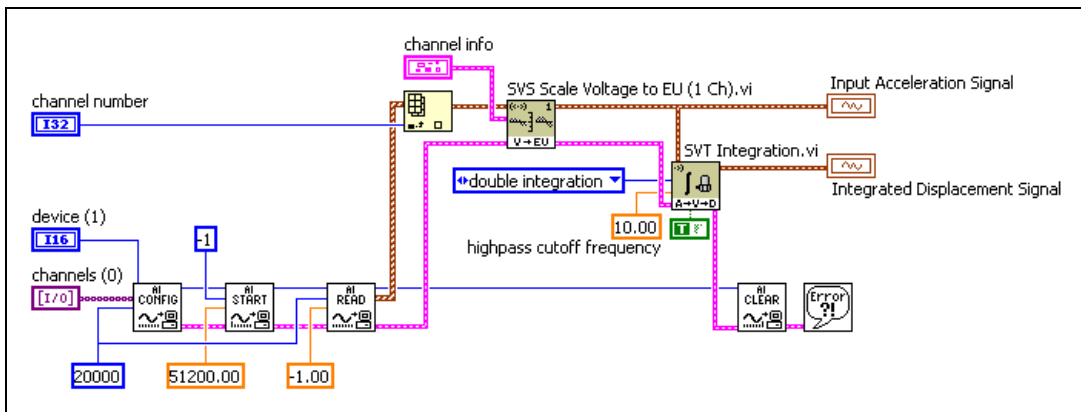


Figure 4-3. Block Diagram for Single-Shot Acquisition and Integration

Because the integration is implemented with filters, there is a transient response associated with integration while the filters settle. You should take care to avoid the transient region when making further measurements. Figure 4-4 shows the results of a single-shot acquisition and integration of a 38 Hz sine wave. You can see the transient response in the first 200 ms of the integrated signal.

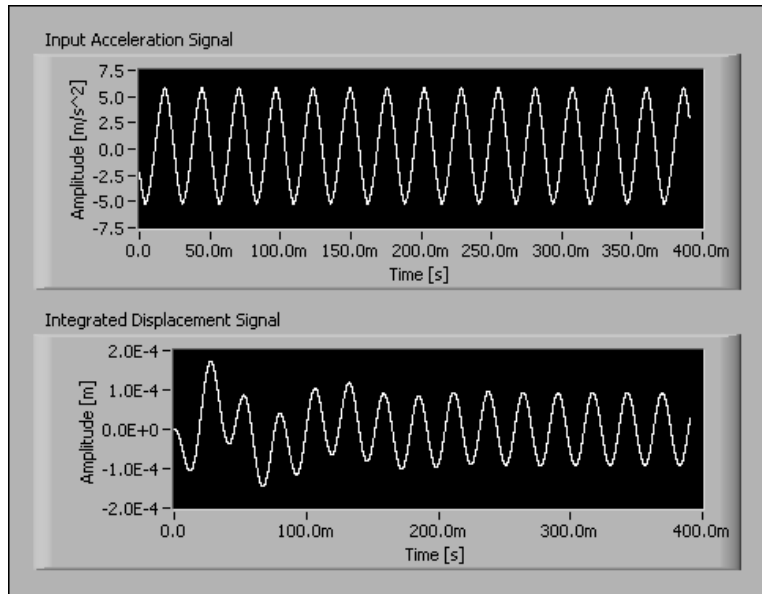


Figure 4-4. Transient Response in Single-Shot Acquisition and Integration

Continuous Acquisition and Integration

The more common case for time-domain integration occurs with continuous acquisition. Figure 4-5 shows the block diagram for a VI designed for continuous acquisition and integration.

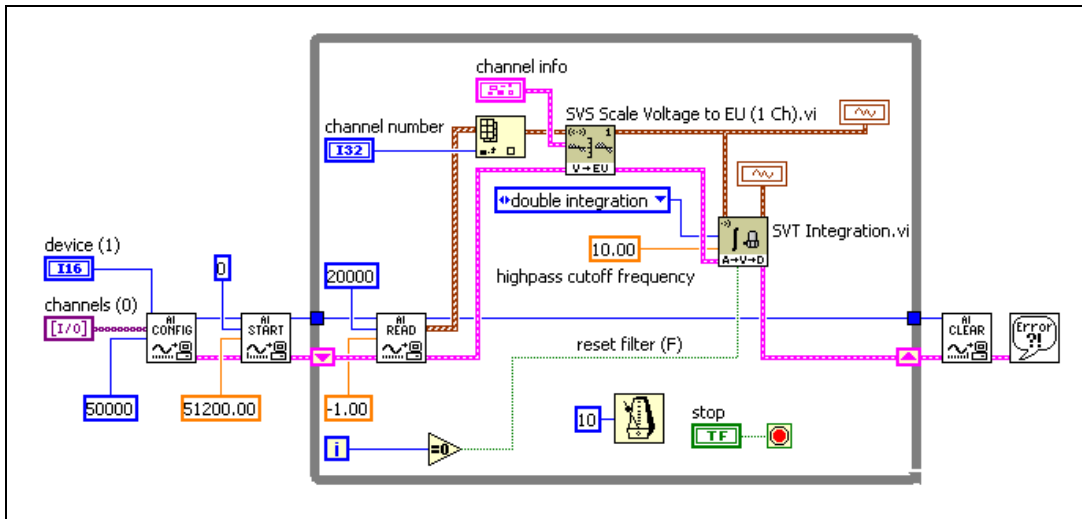


Figure 4-5. Continuous Acquisition and Integration

In this example, the highpass cutoff frequency used for the integration is 10 Hz. Additionally, the integration is explicitly reset in the first iteration of the VI and performed continuously thereafter. In this example, this additional wiring is optional because the SVT Integration VI automatically resets the first time it is called and runs continuously thereafter.

If you use the block diagram in Figure 4-5 in a larger application that requires starting and stopping the data acquisition process more than once, National Instruments suggests setting the **reset filter** control to TRUE for the first iteration of the While Loop. Setting the **reset filter** control to TRUE causes the filter to reset every time the data acquisition process starts. Set the **reset filter** control to FALSE for subsequent iterations of the While Loop.

Figure 4-6 shows the results of the continuous acquisition and integration of the same 38 Hz sinusoid used in the single-shot acquisition and integration example.

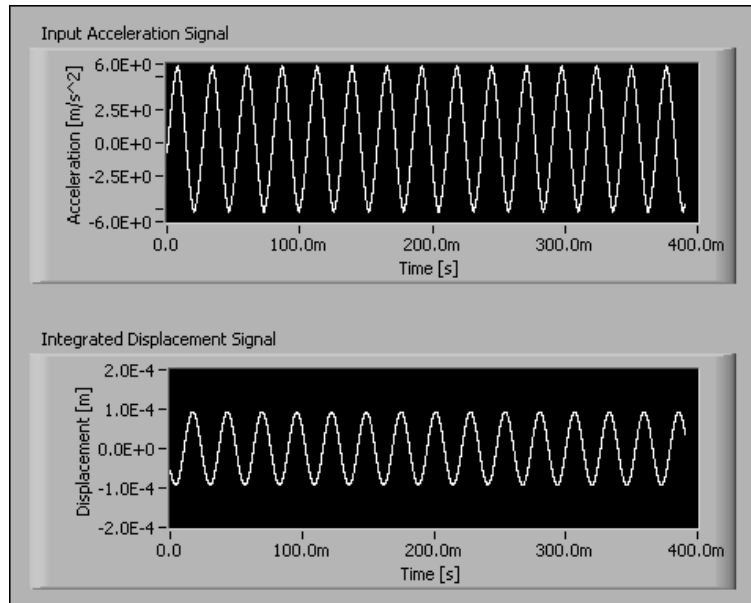


Figure 4-6. Settled Response of Continuous Acquisition and Integration

As in single-shot acquisition and integration, continuous acquisition and integration has an initial transient response. Take care to avoid making additional measurements until the response of the filters settles. Once the filters are settled, you can use the integrated signals for additional analysis with confidence.

Figure 4-7 shows the frequency response for single integration. Figure 4-8 shows the frequency response for double integration.

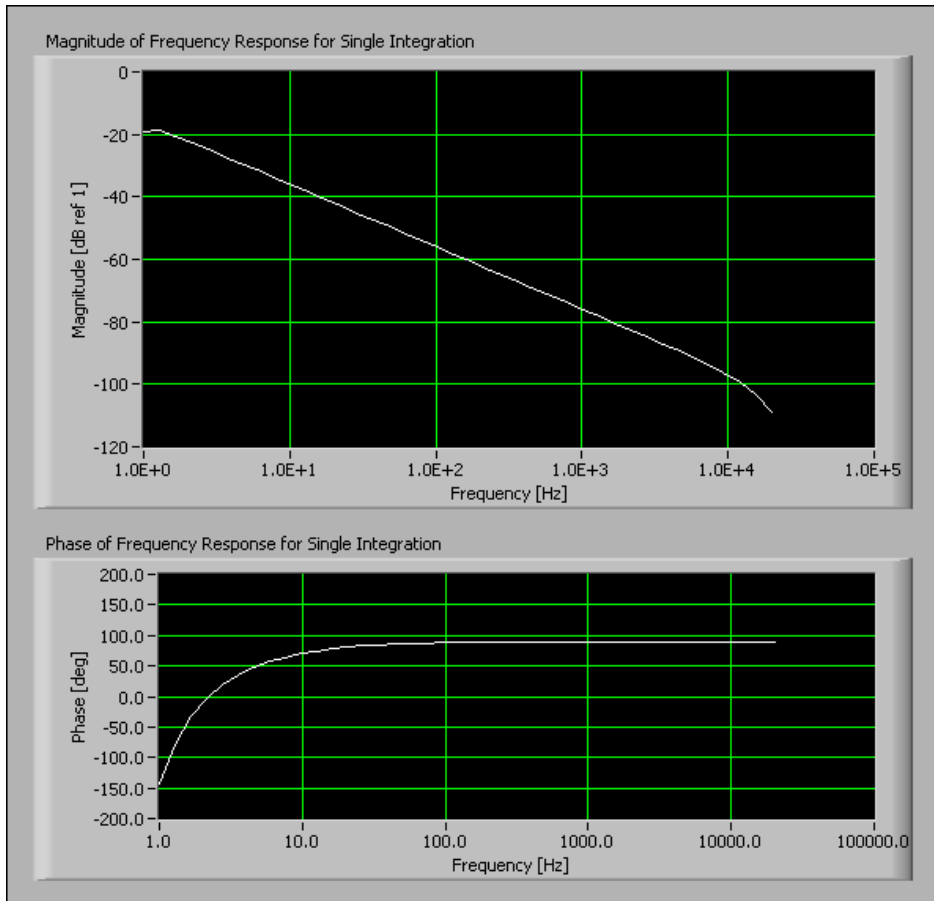


Figure 4-7. Frequency Response for Single Integration

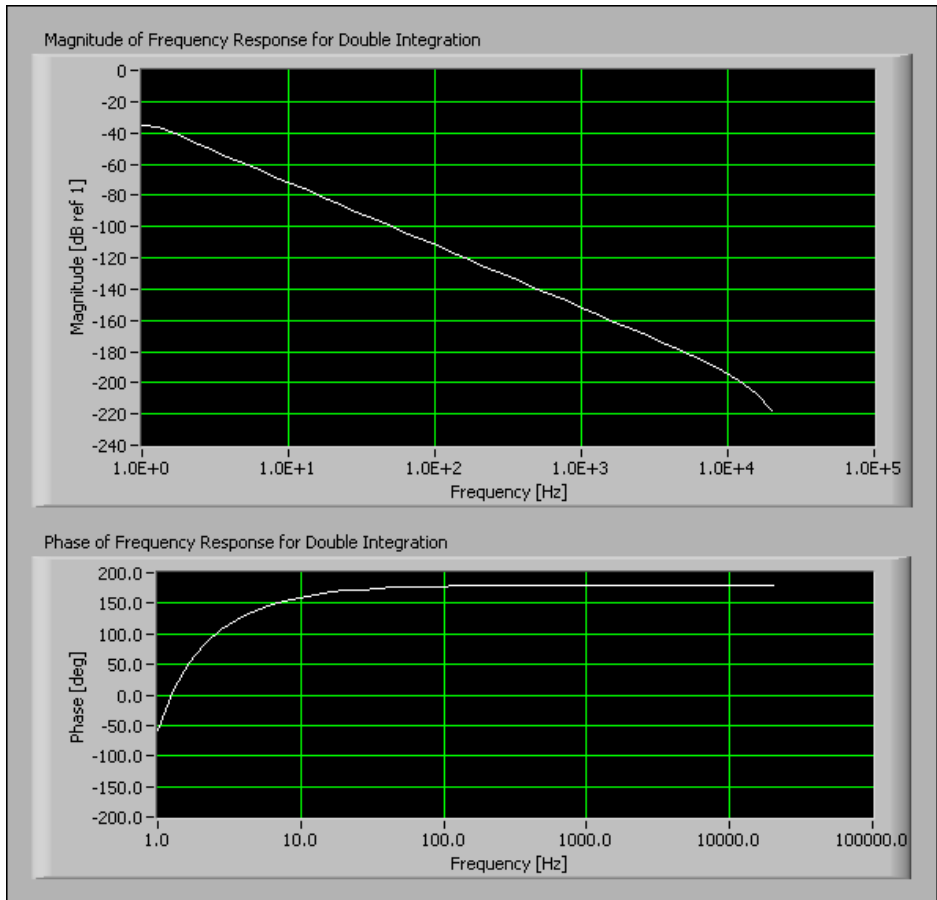


Figure 4-8. Frequency Response for Double Integration

In Figure 4-7, you can see the characteristic 20 dB per decade roll-off of the magnitude response of the single integration. In Figure 4-8, you can see the characteristic 40 dB per decade roll-off of the magnitude response of the double integration.

Upper and lower frequency limits exist for which you can obtain a specified degree of accuracy in the magnitude response. For example, sampling at a rate of 51.2 kHz, the magnitude response of the integrator is accurate to within 1 dB from 1.17 Hz to 9.2 kHz for single integration and from 1.14 to 6.6 kHz for double integration. The accuracy ranges change with the sampling frequency and the highpass cutoff frequency. The attenuation of the single integration filter at 9.2 kHz is -95 dB. The attenuation of the

double integration filter at 6.6 kHz is -185 dB. Accordingly, accuracy at high frequencies usually is not an issue.

Frequency-Domain Integration

You can use the following strategies to obtain the spectrum of an integrated signal:

- Perform the integration in the time domain before computing the spectrum.
- Compute the spectrum before performing the integration in the frequency domain.

The following example demonstrates the implementation of the strategies used to obtain the spectrum of an integrated signal. Figure 4-9 shows the block diagram for the example VI.

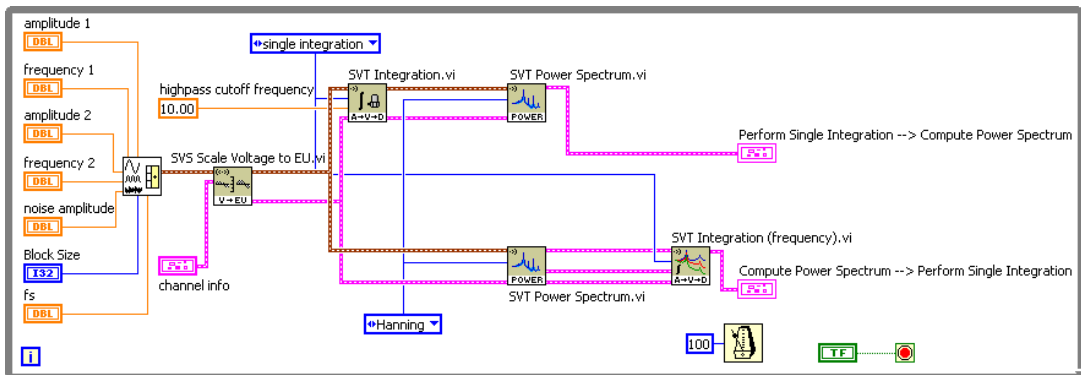


Figure 4-9. Integration in the Time Domain and in the Frequency Domain

The **highpass cutoff frequency** parameter of the SVT Integration VI is wired with a constant of 10 Hz. The SVT Integration (frequency) VI does not have a **highpass cutoff frequency** parameter. Instead, the SVT Integration (frequency) VI sets the DC component of the integrated signal to zero if the spectrum scale is linear or to negative infinity ($-\text{Inf}$) if the spectrum scale is in dB. Figure 4-10 shows the results of integrating in the time and frequency domains.

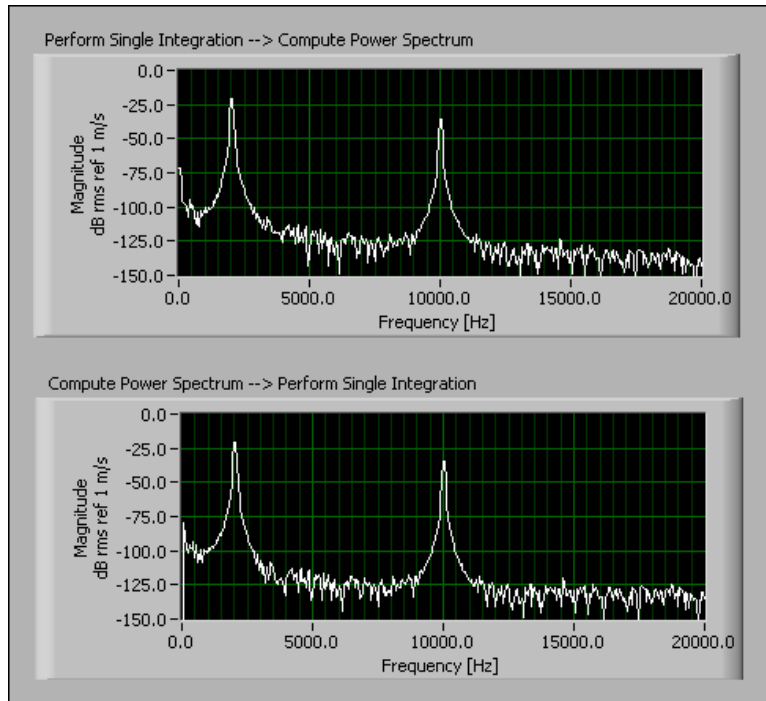


Figure 4-10. Power Spectra of the Integrated Signal

The power spectrum is computed after the time-domain integration filters settle. The frequency-domain integration scales the spectrum at each frequency line. Because filters are not involved in the frequency-domain integration, no settling time is necessary for the frequency-domain integration.

Perform frequency-domain integration in the following situations to maximize performance:

- When the integrated signal is not needed in the time domain
- When spectral measurements are made

Vibration Level Measurements

This chapter briefly discusses the analysis concepts associated with performing vibration level measurements and how you can use the Vibration Level VIs located on the **Functions»Sound & Vibration»Vibration Level** palette to perform vibration level measurements.

Refer to the *LabVIEW Sound and Vibration Toolset Help* for information about individual vibration level VIs.

Figure 5-1 illustrates how vibration level measurement fits into the sound and vibration measurement process.

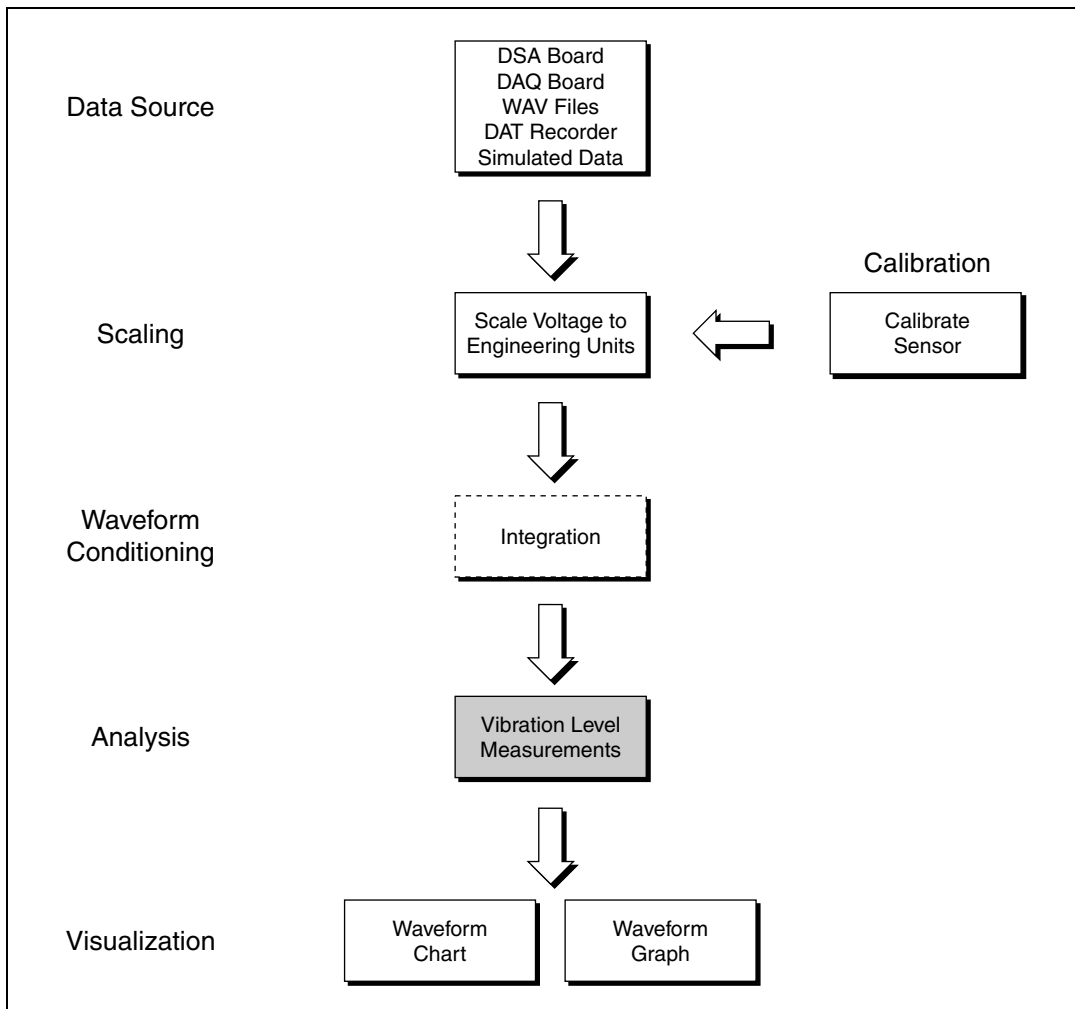


Figure 5-1. Relationship of Vibration Level Measurements to the Sound and Vibration Process

Measuring the Root Mean Square (RMS) Level

A basic requirement of vibration measurements is measuring the level of the signal returned by an accelerometer. The level of the accelerometer signal generally is expressed in root-mean-square acceleration (g_{rms}).

Single-Shot Buffered Acquisition

The block diagram shown in Figure 5-2 illustrates a VI designed to perform a single-shot acquisition and compute the RMS levels.

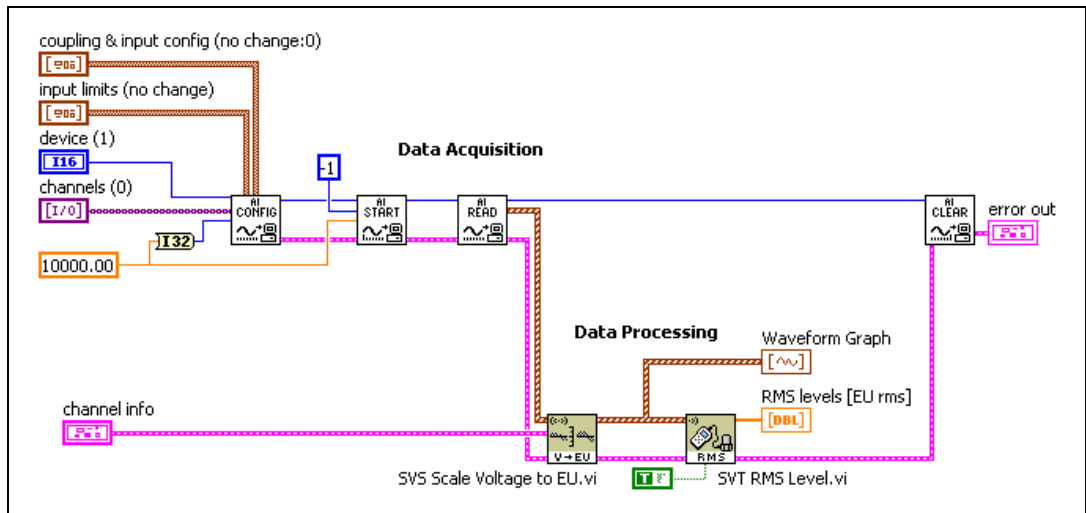


Figure 5-2. Single-Shot Buffered Acquisition and RMS Level VI

The sampling frequency is 10 kilo samples per second (kS/s). A buffer containing 1 second of data is returned by the AI Read VI.



Note Depending on the type of accelerometer you use, you might have to adjust some additional parameters for the AI Config VI. For example, if you use a voltage-mode accelerometer, you must set the **coupling** control to **AC**. Refer to the *LabVIEW Help* for more information about the AI Config VI.

Continuous Signal Acquisition

You can use the block diagram in Figure 5-2 with a While Loop to continuously acquire signals from an accelerometer and display the vibration level in a chart. The block diagram in Figure 5-3 illustrates how to measure the RMS value once every 100 ms and display the results in a

strip chart. In this example, the RMS value is computed based on the last 100 ms of acquired data.

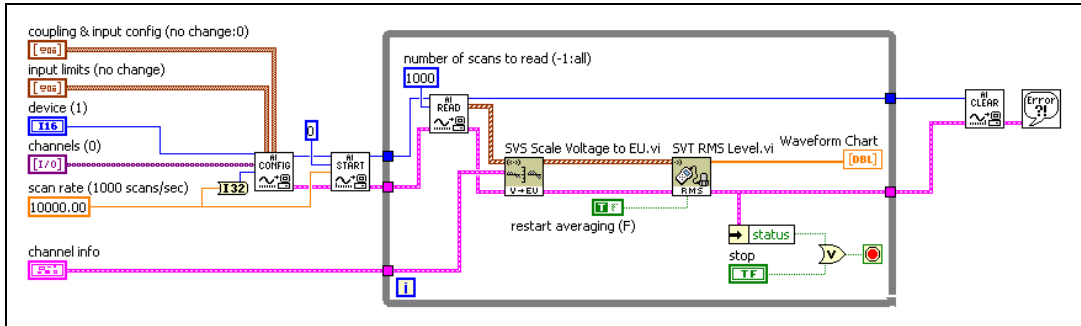


Figure 5-3. Continuous Data Acquisition and RMS Level VI



Note You should set the **restart averaging** control on the SVT RMS Level VI to TRUE. Otherwise, the SVT RMS Level VI accumulates intermediate results in order to compute the RMS vibration level over the entire data acquisition instead of just over the last block of data.

Performing a Running RMS Level Measurement

The SVT Running RMS Level VI returns the RMS value computed over the last *N* seconds, or integration time. The block diagram in Figure 5-4 illustrates an application using the SVT Running RMS Level VI. The sampling frequency is 10 kS/s. The AI Read VI reads 1,000 points at a time.

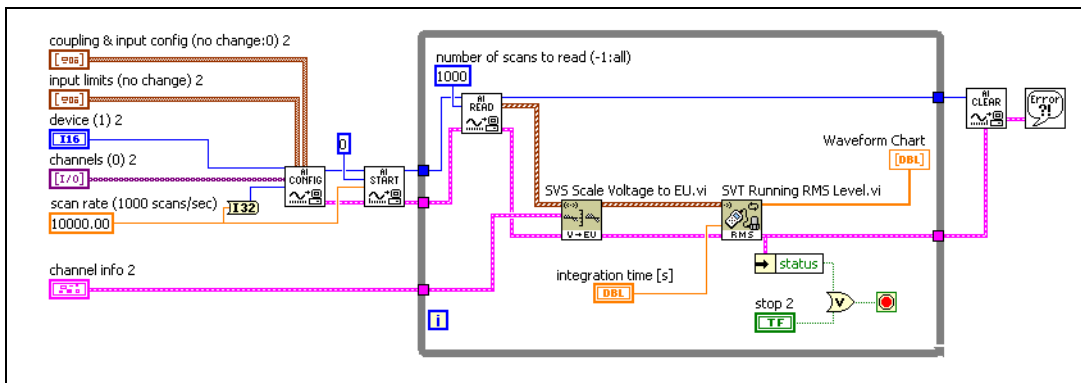


Figure 5-4. Running RMS VI

Computing the Peak Level

Use the SVT Peak Level VI to compute the peak level of a signal. In peak-hold averaging, the largest measured level value of all previous values is computed and returned until a new value exceeds the current maximum. The new value becomes the new maximum value and is the value returned until a new value exceeds it. Refer to Chapter 6, *Sound Level Measurements*, for more information about peak-hold averaging.

Computing the Crest Factor

The crest factor is the ratio of the peak value over the RMS value of a given signal and indicates the shape of the waveform. The crest factor is defined by the following equation.

$$F_c = \frac{V_{pk}}{V_{rms}}$$

where F_c is the crest factor,
 V_{pk} is the peak value of the signal,
 V_{rms} is the RMS value of the signal.

The block diagram in Figure 5-5 illustrates an application using the SVT Crest Factor VI. Along with the crest factors, the SVT Crest Factor VI also returns the peak and RMS levels.

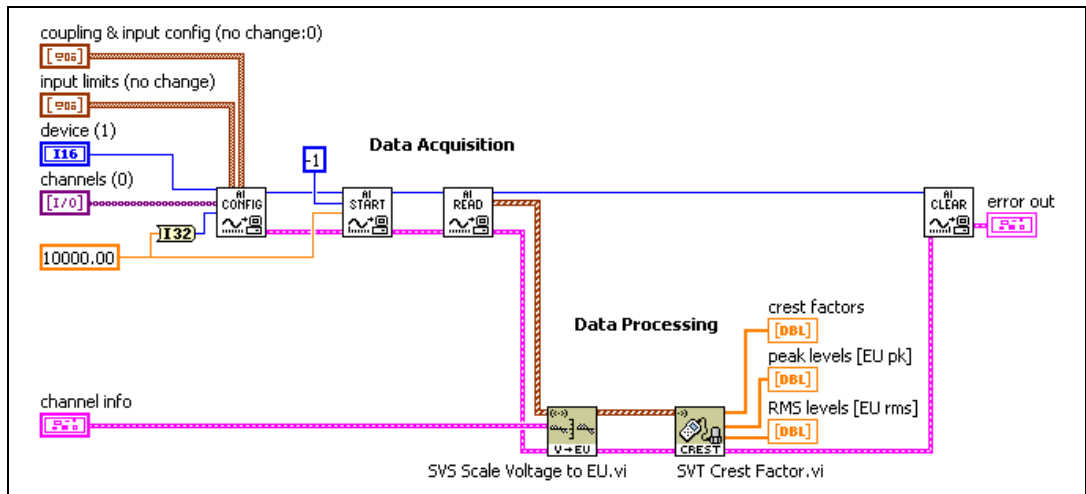


Figure 5-5. Crest Factor VI

Sound Level Measurements

This chapter discusses some of the analysis concepts associated with performing sound level measurements and how you can use the Sound Level VIs located on the **Functions»Sound & Vibration»Sound Level** palette to perform sound level measurements. You can combine different sound level measurements and use them simultaneously to provide flexibility when dealing with acoustic measurements.

Refer to the *LabVIEW Sound and Vibration Help* for information about individual Sound Level VIs.

Figure 6-1 illustrates how sound level measurement fits into the sound and vibration measurement process.

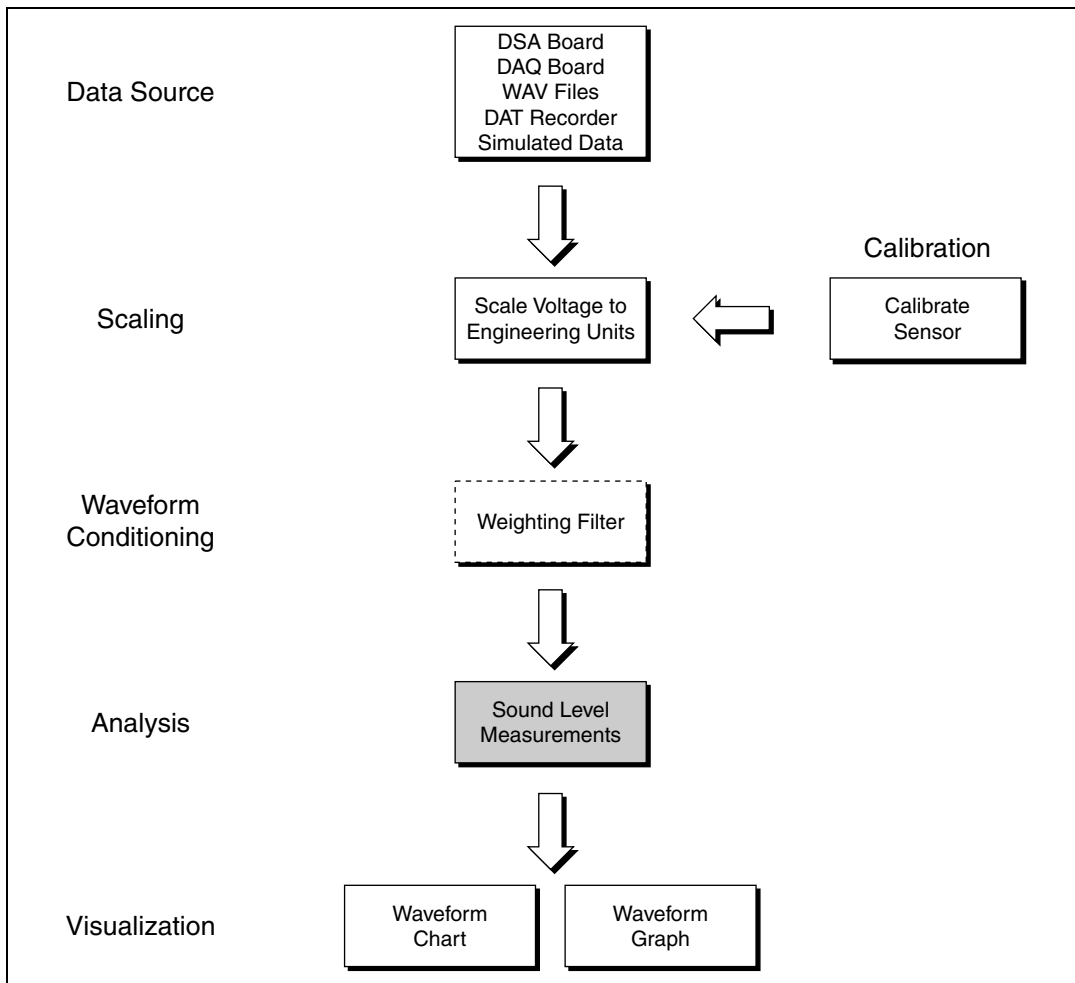


Figure 6-1. Relationship of Sound Level Measurements to the Sound and Vibration Process

Time Averaging Modes

The Sound Level VIs offer the following time averaging modes:

- Linear, which is equivalent continuous sound level (L_{eq})
- Exponential
- Peak hold

Linear Averaging

You compute the L_{eq} by integrating the square of the signal over a fixed time interval and dividing by the time interval. When you select linear averaging, the Sound Level VIs return a single value. The value returned represents the continuous dB level that would have produced the same sound energy in the same time T as the actual noise history. To obtain intermediate results, you must split a long time record into several smaller records. Linear averaging is represented by the following equation.

$$L_{eq} = 10 \log_{10} \frac{1}{T} \int_0^T \left(\frac{P^2_{rms}}{P_0^2} \right) dt,$$

where P_0 is the reference pressure, which is 20 μPa for acoustics.

Simple Linear Averaging

Figure 6-2 illustrates a simple example of linear averaging.

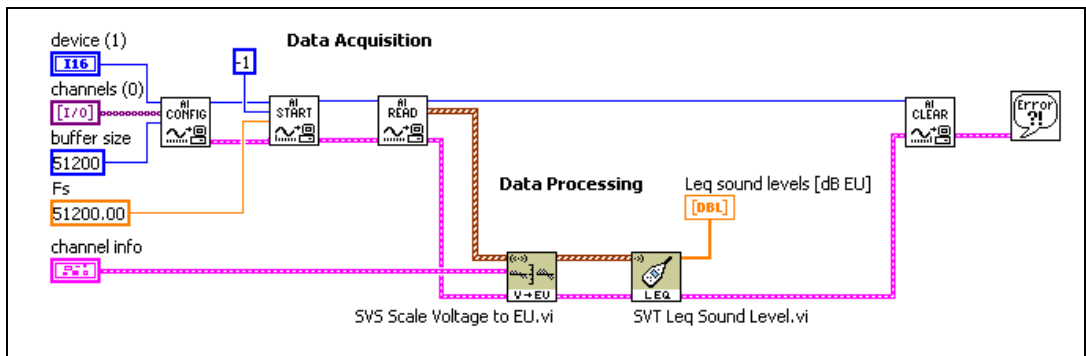


Figure 6-2. Simple L_{eq} VI

The upper part of the VI in Figure 6-2 uses intermediate-level VIs to perform a single-shot buffered acquisition. The SVT Leq Sound Level VI is configured with the following parameters:

- Sampling frequency $F_s = 51,200$ samples per second (S/s).
- Buffer size = 51,200 samples, which is one second of data.

The VI in Figure 6-2 completes the following steps to perform the single-shot buffered acquisition.

1. The AI Read VI returns a waveform containing one second of data.
2. The SVS Scale Voltage to EU VI scales the signal to the appropriate engineering units, such as Pa if a microphone is used to acquire the signal.
3. The SVT Leq Sound Level VI computes the L_{eq} .

Figure 6-3 displays the resulting L_{eq} measurement and the instantaneous sound pressure level.

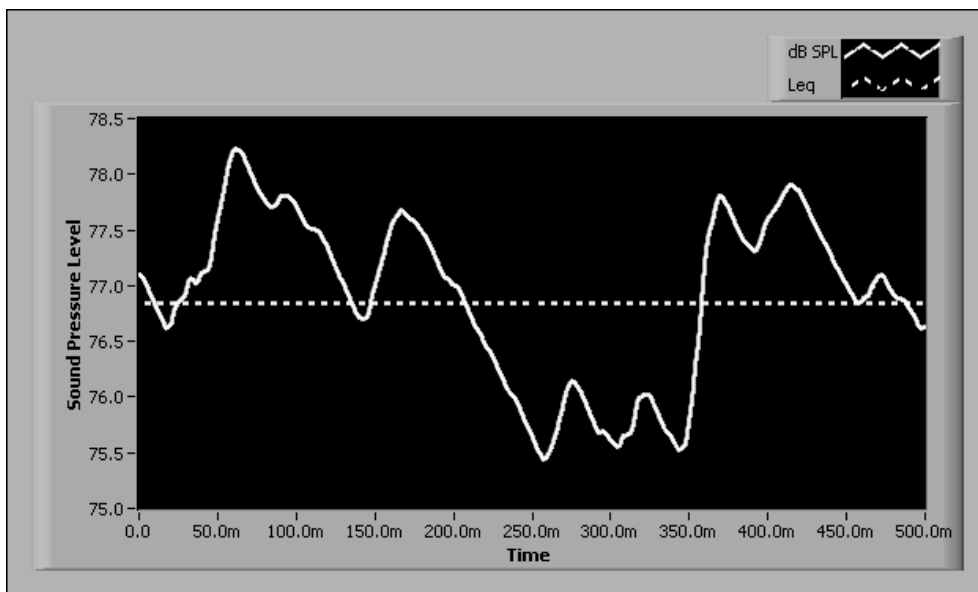


Figure 6-3. L_{eq} versus Instantaneous Sound Pressure Level

Measuring L_{eq} Over a Longer Time Period

You might need to compute the L_{eq} over a longer period of time. For example, you might need to measure the L_{eq} over one hour in the entire audio range of 20 Hz–20 kHz. To measure the L_{eq} over an hour in the

audio range, select a sampling frequency that allows you to perform measurements up to 20 kHz. According to the Shannon Sampling Theorem, the minimum sampling frequency is 40 kS/s, which is twice the 20 kHz maximum frequency of interest. Depending on the hardware you use, you might have additional considerations, such as an anti-aliasing filter, when selecting the sampling frequency. Traditionally, a sampling frequency of 51.2 kS/s is used to perform measurements up to 20 kHz.

Over one hour with a sampling frequency of 51.2 kS/s, you need to acquire 184,320,000 samples, as shown in the following equation.

$$\left(\frac{60 \text{ minutes}}{\text{hour}}\right)\left(\frac{60 \text{ seconds}}{\text{minute}}\right)\left(\frac{51,200 \text{ samples}}{\text{second}}\right) = \frac{184,320,000 \text{ samples}}{\text{hour}}$$

The memory required to accumulate such a large number of samples might prohibit you from accumulating the number of samples required for your analysis. An alternative method is to process small chunks of data, keep track of the intermediate results, and integrate the measurement over time. You can use the Sound and Vibration Toolset to perform this alternative method of L_{eq} measurement. Figure 6-4 shows the block diagram for a VI designed to compute the L_{eq} for a one-hour period.

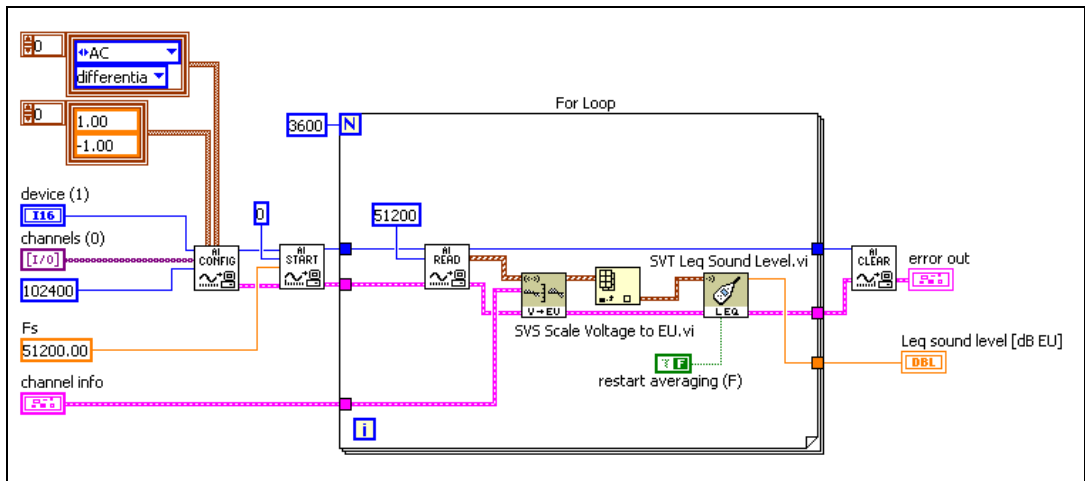


Figure 6-4. One Hour L_{eq} VI

The VI shown in Figure 6-4 performs an L_{eq} over one second and repeats the operation 3,600 times using a For Loop. The last result returned by the SVT Leq Sound Level VI is the L_{eq} over the one-hour period.

In order for the SVT Leq Sound Level VI to accumulate the intermediate results, set the **restart averaging** control to FALSE or leave the control unwired. You can make the intermediate results available by using the auto-indexing capability of the For Loop. Refer to the *LabVIEW Help* for information about auto-indexing.

Instead of performing an L_{eq} over one second and repeating the operation 3,600 times, you can perform the measurement over two seconds and repeat it 1,800 times or perform it over four seconds and repeat it 900 times.

Restart Averaging and Advanced Concepts

Some applications require you to perform L_{eq} measurements continuously, using a specific integration time. For example, you need to measure a reverberation time that requires the instrument to return an L_{eq} measurement every 50 ms for 10 seconds. You can use a For Loop and its auto-indexing capabilities to perform a continuous L_{eq} measurement, as illustrated in Figure 6-5.

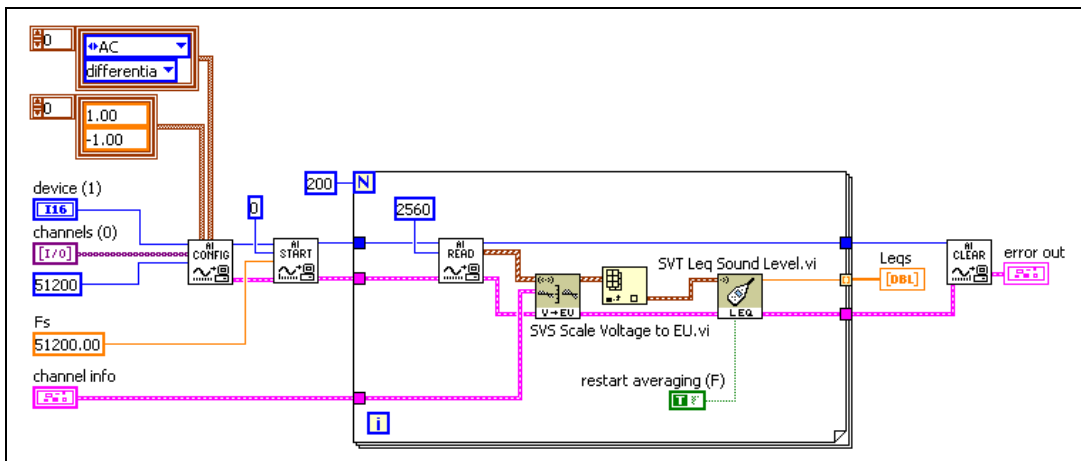


Figure 6-5. Block Diagram for Reverberation VI

Because you divide the 10 second measurement period into 50 ms blocks, you need 200 iterations (10 s/50 ms) of the For Loop. To continuously measure the L_{eq} , set the **restart averaging** control to TRUE. When the **restart averaging** control is set to TRUE, the SVT Leq Sound Level VI does not accumulate intermediate results but restarts the averaging process with each iteration of the For Loop. Figure 6-6 shows the results of performing a reverberation time measurement for 10 seconds.

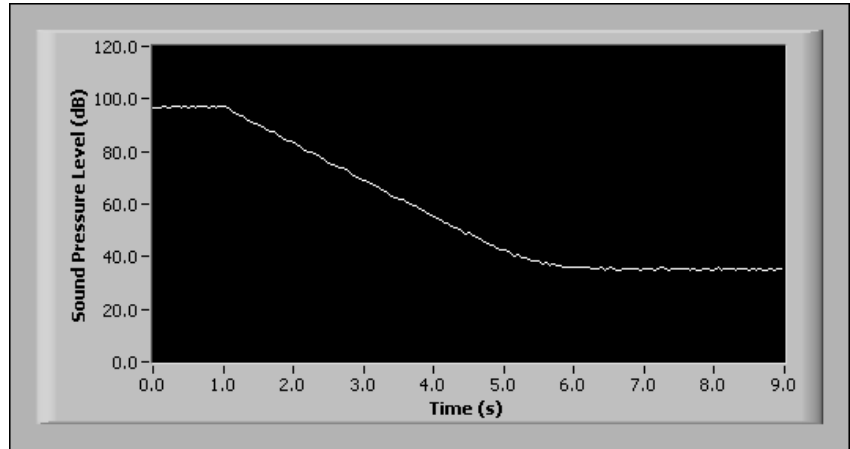


Figure 6-6. Ten-Second Reverberation Time Measurement

Performing a Running L_{eq}

Use the SVT Running Leq VI to perform a running L_{eq} over a specified period of time, or integration time. The result returned by the SVT Running Leq VI is the L_{eq} computed over the last N seconds. A new running L_{eq} value is returned each time you call the SVT Running Leq Sound Level VI.

Exponential Averaging

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

The exponential averaging mode supports the following time constants:

- **Slow** uses a time constant of 1,000 ms. Slow averaging is useful for tracking the sound pressure levels of signals whose sound pressure levels vary slowly.
- **Fast** uses a time constant of 125 ms. Fast averaging is useful for tracking the sound pressure of signals whose sound pressure levels vary quickly.

- **Impulse** uses a time constant of 35 ms if the signal is rising and 1,500 ms if the signal is falling. Impulse averaging is useful for tracking sudden changes in the sound pressure level and recording the changes so they do not disappear too fast.
- **Custom** allows you to specify a time constant suitable for your particular application.

Use the following sound level measurement VIs for exponential time averaging:

- SVT Exp Avg Sound Level VI
- SVT Decimated Exp Avg Sound Level VI

Refer to the *LabVIEW Sound and Vibration Toolset Help* for information about the SVT Exp Avg Sound Level VI and the SVT Decimated Exp Avg Sound Level VI.

Figure 6-7 illustrates a block diagram for a VI that performs an exponential averaged measurement using the SVT Exp Avg Sound Level VI with a **Slow** time constant. Use the **restart averaging** control to reset the averaging.

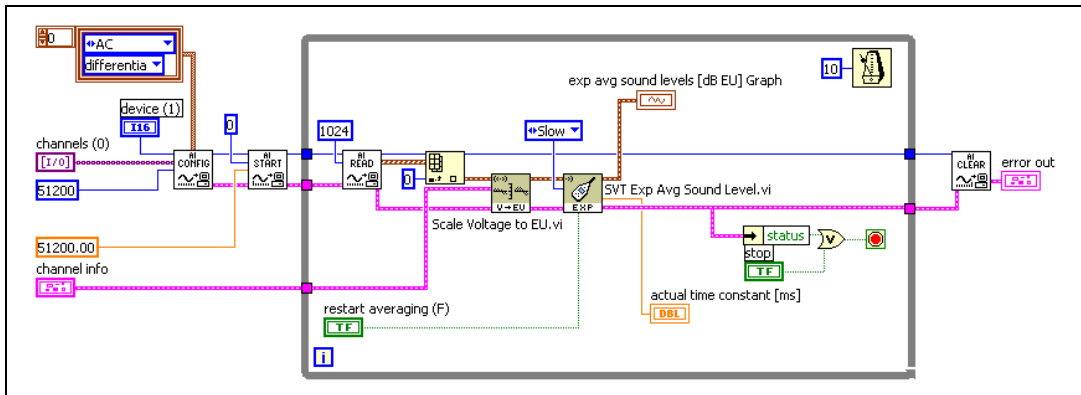


Figure 6-7. Exponential Averaging with Slow Time Constant VI

Peak Hold

In peak-hold averaging, the largest measured sound pressure level value of all previous values is computed and returned until a new value exceeds the current maximum. The new value becomes the new maximum value and is the value returned until a new value exceeds it. Peak hold actually is not a true form of averaging because successive measurements are not mathematically averaged. However, as with other averaging processes,

peak-hold averaging combines the results of several measurements into one final measurement and can be considered a kind of averaging. As with exponential averaging, the averaging process continues indefinitely. The formula for peak averaging is defined by the following equation.

$$y[k] = \max(y[k - 1], x[k]),$$

where $x[k]$ is the new measurement,
 $y[k]$ is the new average,
 $y[k - 1]$ is the previous average.

Considerations for Making Sound Level Measurements

The selection of a sampling frequency depends on the highest frequency of interest in your signal and whether you plan to use weighting filters. You can use A-, B-, and C-weighting filters in conjunction with sound level measurements. Refer to Chapter 3, [Weighting Filters](#), for information about weighting filters.

You can perform several types of sound level measurements on the same channel in order to compare different results.

Fractional-Octave Analysis

This chapter discusses fractional-octave analysis, including fractional-octave analysis theory, averaging modes supported by the Octave Analysis VIs, settling time, and ANSI and IEC standards.

Use the Octave Analysis VIs located on the **Functions»Sound & Vibration»Octave Analysis** palette to perform the following analyses:

- Octave analysis
- Third-octave analysis
- Fractional-octave analysis, including 1/1, 1/3, 1/6, 1/12, and 1/24 octave

The Octave Analysis VIs can accommodate any sampling frequency and any number of fractional-octave bands.

Refer to the *LabVIEW Sound and Vibration Toolset Help* for information about the individual Octave Analysis VIs.

Figure 7-1 illustrates the relationship of fractional-octave analysis to the sound and vibration measurement process.

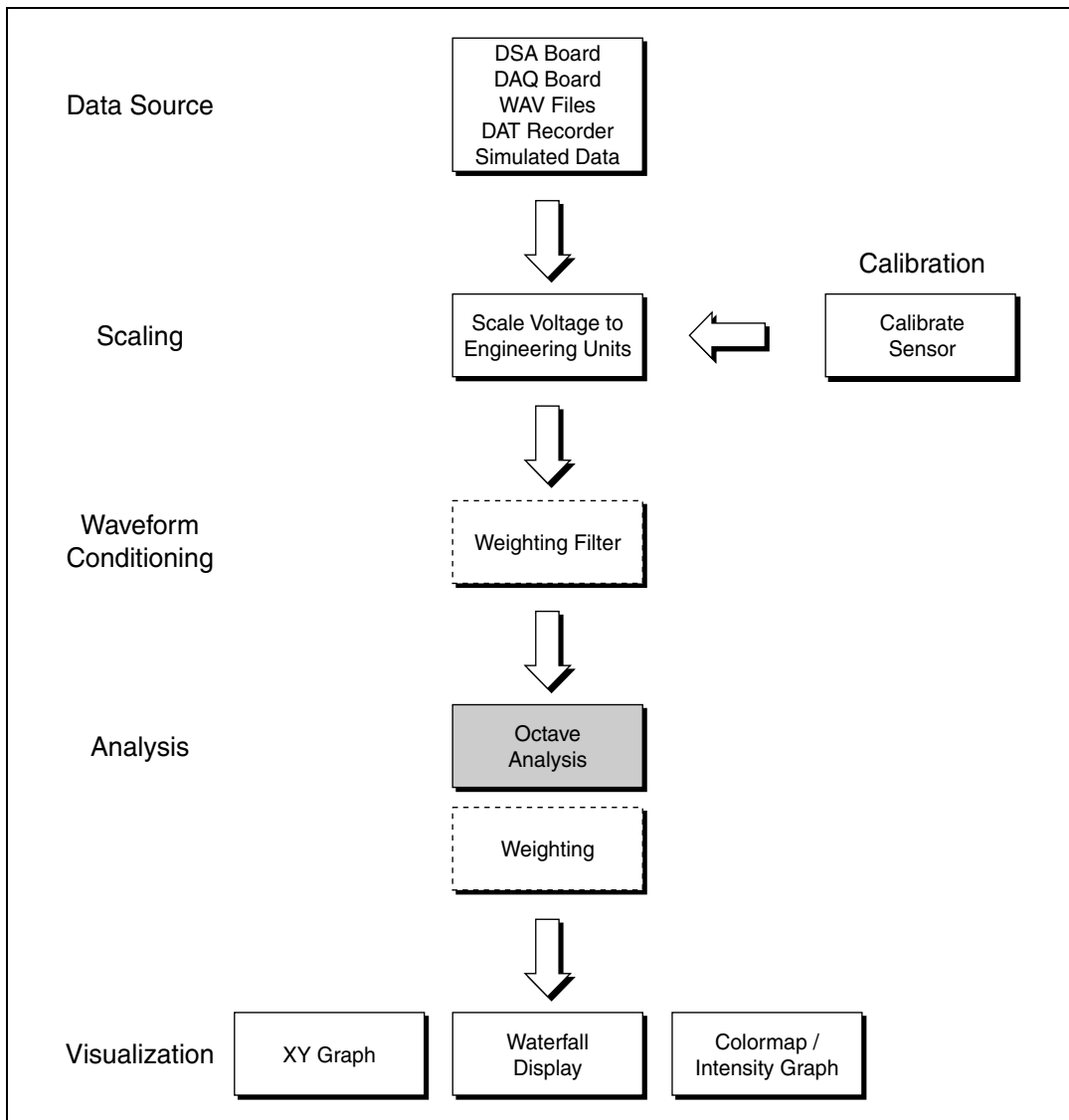


Figure 7-1. Relationship of Fractional-Octave Analysis to the Sound and Vibration Process

Fractional-Octave Analysis Overview

Fractional-octave analysis is a widely used technique for analyzing audio and acoustical signals. Fractional-octave analyses, especially 1/3 and 1/12 octave, exhibit characteristics analogous to the response of the human ear. Fractional-octave analysis is also required by many industry standards, such as ANSI and IEC.

The following steps summarize fractional-octave measurement.

1. Send a time domain signal through a bank of N band-pass filters.
2. Compute the instantaneous square value for the outputs of the N filters.
3. Average the instantaneous square values according to a selected averaging mode.
4. Display the averaged values in a bar graph.

The number of filters to apply as well as their passband is determined by the type of fractional-octave measurement you are making, 1/1 through 1/24 octave. The filters used are known as constant Q filters. Refer to the [Bandedge Frequencies](#) section of this chapter for more information about constant Q filters.

Octave Analysis in the 31.5 Hz–16 kHz Band

The following example is a simple illustration of octave analysis. In this example, an input signal acquired by a microphone is sent through 10 band-pass filters centered at the frequencies shown in Figure 7-2.

center frequency
31.5 Hz
63.0 Hz
125 Hz
250 Hz
500 Hz
1000 Hz
2000 Hz
4000 Hz
8000 Hz
16000 Hz

Figure 7-2. Full Octave Center Frequencies

The power in each band is computed and displayed in a bar graph with a log scale for the x-axis, as shown in Figure 7-3.

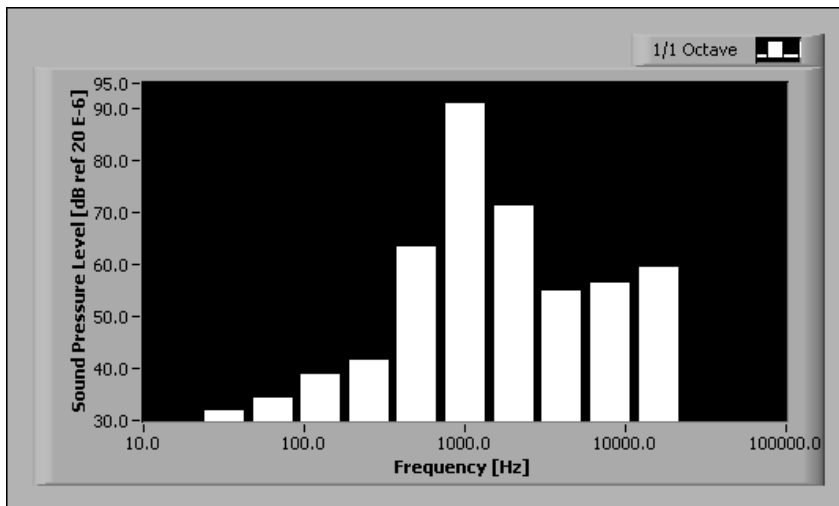


Figure 7-3. 1/1 Octave Spectrum

Bandwidth and Filter Banks

Fractional-octave analysis depends primarily on filters. This section briefly discusses filter design in relationship to fractional-octave analysis.

The Octave Filter

An octave is defined as the interval between two frequencies, one of which is twice the other. For example, frequencies of 250 Hz and 500 Hz are one octave apart, as are frequencies of 1 kHz and 2 kHz. An octave filter is a filter whose passband covers one octave, as shown in Figure 7-4.

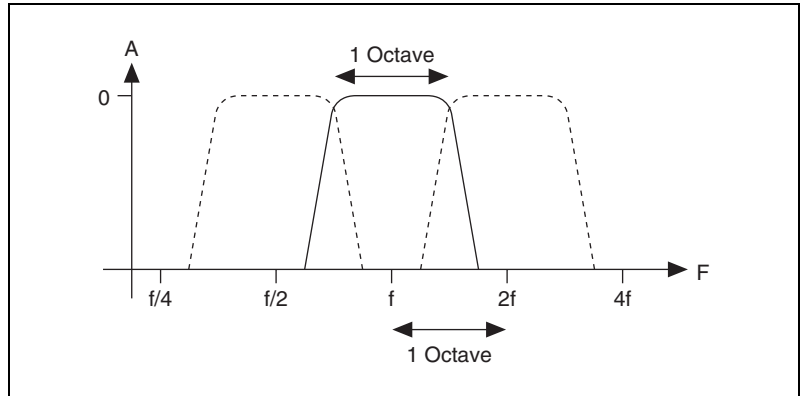


Figure 7-4. Octave Filter

Just as a tuning fork provides a 440 Hz reference frequency for musicians, a reference frequency is needed to fully define octave filters. Instrumentation equipment typically uses a 1 kHz reference frequency.

Bandedge Frequencies

Filters used in fractional-octave analysis are known as constant Q filters. The quality constant Q is defined as the ratio of the bandwidth over the center frequency of the filter. Q remains constant for octave filters. For example, an octave filter with a center frequency of 1,000 Hz leads to the following bandedge frequencies.

$$f_1 = \frac{1,000}{\sqrt{2}} = 707 \text{ Hz}$$

$$f_2 = (1,000)(\sqrt{2}) = 1,414 \text{ Hz}$$

$$\text{BW} = f_2 - f_1 = 707 \text{ Hz}$$

$$Q = \frac{707}{1,000} = 0.707$$

where f_1 and f_2 are bandedge frequencies,
 BW is the bandwidth,
 Q is the quality constant Q.

An octave filter with a center frequency of 8,000 Hz leads to the following bandedge frequencies.

$$f_1 = \frac{8,000}{\sqrt{2}} = 5,657 \text{ Hz}$$

$$f_2 = (8,000)(\sqrt{2}) = 11,314 \text{ Hz}$$

$$\text{BW} = f_2 - f_1 = 5,657 \text{ Hz}$$

$$Q = \frac{5,657}{8,000} = 0.707$$

where f_1 and f_2 are bandedge frequencies,
 BW is the bandwidth,
 Q is the quality constant Q.

The results obtained from calculating the bandedge frequencies indicate the following bandwidth characteristics:

- The bandwidth of the octave filters is narrow if the center frequency is low.
- The bandwidth of the octave filters is wider when the center frequency is higher.

Because of the bandwidth characteristics, fractional-octave analysis uses a logarithmic frequency scale to compute and display octave spectra.

Fractional-Octave Filters

The resolution offered by octave filters can be limited, being only 11 octaves in the 16 Hz–16 kHz range. To overcome the limited resolution of octave filters, you can use other filters known as fractional-octave filters. Rather than covering one octave with a single filter, N filters are applied per octave in order to improve resolution. Of the fractional-octave filters, the third-octave (1/3) filter is widely used for fractional-octave analysis. Figure 7-5 shows the 1/3 octave response at frequencies of 500 Hz, 630 Hz, and 800 Hz.

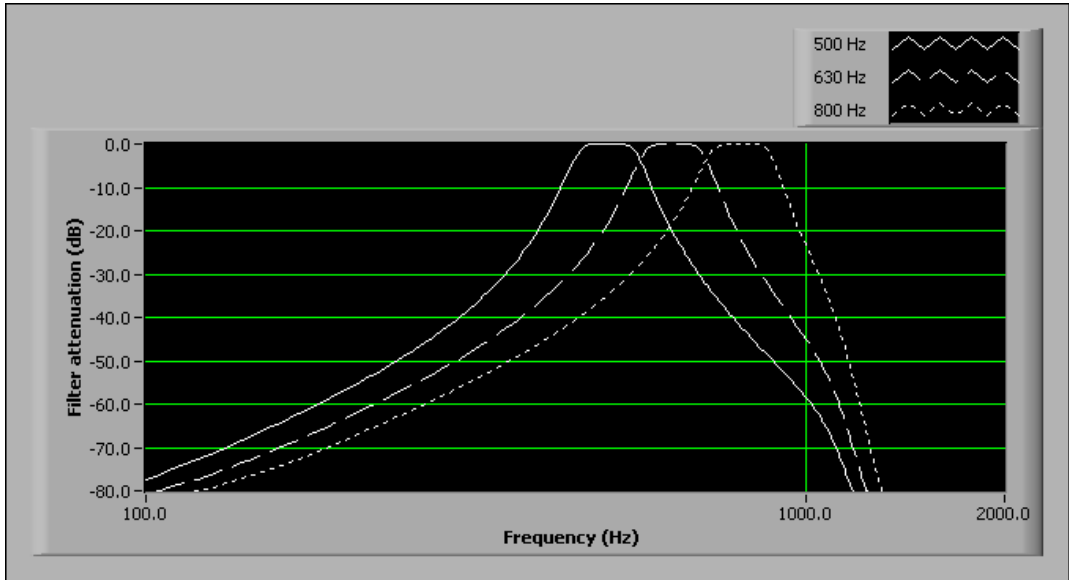


Figure 7-5. Response of 1/3 Octave Filters

Fractional-octave analysis is a CPU-intensive operation. Increasing the number of filters applied to a signal increases the demands placed on the CPU and can result in an increased computation time.

Filter Settling Time

When starting or resetting the filtering operation of the fractional-octave filters, a certain time is required before the measurements are valid. This time is called the settling time and is related to the bandwidth of any particular filter. The lowest frequency band has the smallest bandwidth and defines the settling time required before the complete fractional-octave measurement can be considered valid. In the Sound and Vibration Toolset, settling time is defined as five divided by the bandwidth of the filter used for the lowest frequency band. The settled? indicator returns TRUE as soon as all the filters are settled.

Averaging

The Octave Analysis VIs support the following averaging types:

- Linear averaging
- Exponential averaging
- Equal confidence averaging
- Peak hold

Linear Averaging

Linear averaging is computed by integrating the square of the filtered signal over a fixed time interval and dividing by the time interval. Refer to Chapter 6, *Sound Level Measurements*, for more information about linear averaging.

Exponential Averaging

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

Figure 7-6 illustrates the block diagram for a VI performing 1/3 octave analysis in the 20Hz–20kHz range using fast exponential averaging.

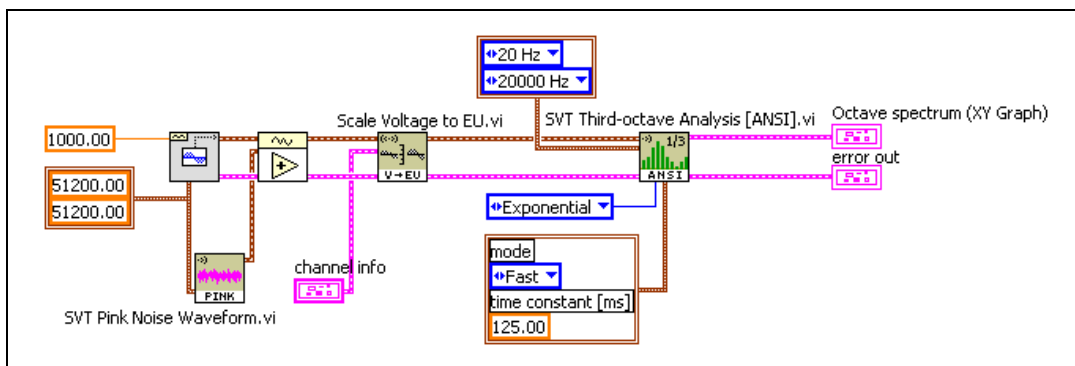


Figure 7-6. 1/3 Octave Analysis with Fast Exponential Averaging VI



Note In order to use exponential averaging, you must connect the **exp avg settings** control. Refer to the *LabVIEW Sound and Vibration Toolset Help* for information about the **exp avg settings** control.

Refer to Chapter 6, *Sound Level Measurements*, for more information about exponential averaging.

Equal Confidence Averaging

Equal confidence averaging is exponential averaging with a specific time constant set for each individual band. Each time constant is set so that the standard deviation of the band power measurement equals the confidence value in decibels specified by the **equal confidence level [dB]** control.

Peak-Hold Averaging

In peak-hold averaging, the largest measured level value of all previous values is computed and returned for each band until a new value exceeds the current maximum. The new value becomes the new maximum value and is the value returned until a new value exceeds it. Refer to Chapter 6, *Sound Level Measurements*, for more information about peak-hold averaging.

Resetting the Filter and Restarting the Averaging Process

The Octave Analysis VIs reset the filter the first time the VI executes. You can use the **reset filter** control to programmatically reset the filter. After the filter is reset, a transient response occurs while the filter settles.

You can use the **restart averaging** control to restart the averaging process without resetting the filter. Restarting the averaging without resetting the filter avoids the transient phase associated with the settling of the filter.

Refer to the *Filter Settling Time* section of this chapter for information about filter settling time.

Performing Third-Octave Analysis Outside the Audio Range

You might need to perform third-octave analysis on signals outside of the 20 Hz–20 kHz range. For example, you might need to evaluate human exposure to whole body vibration with potential signals of interest in the 0.5–80 Hz range. To perform third-octave analysis outside the 20 Hz–20 kHz range, use either the SVT Fractional-octave Analysis [ANSI] VI or the SVT Fractional-octave Analysis [IEC] VI and select a bandwidth equal to 1/3 octave, as illustrated in Figure 7-7.

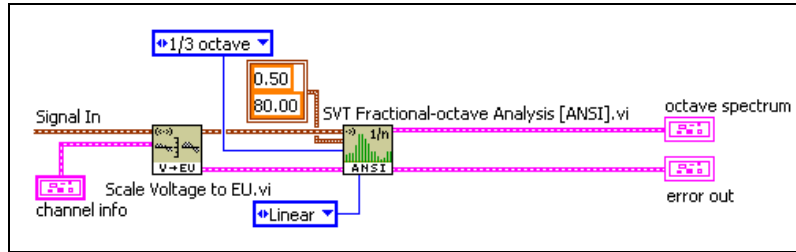


Figure 7-7. 1/3 Octave Analysis Outside Standard Bands VI

The fractional-octave analysis VIs accept numeric values for the low frequency and high frequency limits and enable the measurement of signals outside the 20 Hz–20 kHz range.

ANSI and IEC Standards

This section discusses the ANSI and IEC standards regarding midband frequencies, the nominal frequencies returned by the Octave Analysis VIs, and compliance with ANSI and IEC standards by the Octave Analysis VIs.



Note The ANSI and IEC Octave Analysis VIs use the base 2 system for midband frequencies instead of the base 10 system. Using the base 2 system means that the ratio of two midband frequencies is a fractional power of 2 instead of a fractional power of 10.

ANSI Standard

According to the ANSI S1.11-1986 standard, the midband, or center, frequency of the band-pass filter is defined by the following equation.

$$f_i = 1000 \cdot 2^{ib},$$

where f_i is the center frequency of the i^{th} band-pass filter expressed in hertz,
 i is an integer when $i = 0$, $f_0 = 1$ kHz, which is the reference frequency for the audio range,
 b is the bandwidth designator and equals 1 for octave, 1/3 for 1/3 octave, 1/6 for 1/6 octave, 1/12 for 1/12 octave, and 1/24 for 1/24 octave.

IEC Standard

According to the IEC 1260:1995 standard, the midband frequency, or center, frequency of the band-pass filter is defined by the following equations.

$$f_i = 1000 \cdot 2^{ib} \text{ for } 1/N \text{ octave filters when } N \text{ is odd}$$

$$f_i = 1000 \cdot 2^{\frac{(i+1)b}{2}} \text{ for } 1/N \text{ octave filters when } N \text{ is even}$$

where f_i is the center frequency of the i^{th} band-pass filter expressed in hertz,
 i is an integer when $i = 0$, $f_0 = 1$ kHz, which is the reference frequency for the audio range,
 b is the bandwidth designator and equals 1 for octave, 1/3 for 1/3 octave, 1/6 for 1/6 octave, 1/12 for 1/12 octave, and 1/24 for 1/24 octave.

Nominal Frequencies

The exact midband frequencies are used to design the filters for fractional-octave analysis. However, all the Octave Analysis VIs return the nominal midband frequencies, also called the preferred frequencies. In the case of octave and 1/3 octave analyses, the nominal frequencies are calculated in accordance with the ANSI and IEC standards. In the case of 1/6, 1/12, and 1/24 octave analyses, the nominal frequencies are calculated in accordance with the *Annex A (informative)* of the IEC 1260:1995 standard.

Compliance with ANSI and IEC Standards

When combined with any National Instruments DSA board, the appropriate microphone, and proper signal conditioning, the filters produced by the Octave Analysis VIs comply with the following standards:

- ANSI S1.11-1986: Specification for octave-band and fractional-octave-band analog and digital filters, Order 3, Type 1-D, optional range
- IEC 1260 (1995 - 07): Electroacoustics - Octave-band and fractional-octave-band filters, Class 1

In the case of the ANSI standard, the default order of the band-pass filter is 3, which leads to a Type 1-D filter.

For octave filters, choose a sampling frequency at least 3 times the exact center frequency of the highest frequency band. For fractional-octave filters, choose a sampling frequency at least 2.5 times the exact center frequency of the highest frequency band.

Displaying Results

The Octave Analysis VIs produce a cluster containing center frequencies and band power. The **Controls»Sound & Vibration** palette contains two graphs designed to display octave results, the Octave Graph and the Multiplot Octave Graph. Refer to the *Sound and Vibration Toolset Help* for information about the Octave Graph and the Multiplot Octave Graph.

You can connect the cluster containing the center frequencies and band power directly to an XY graph. However, you must modify the XY graph in order to display octave results.

Complete the following steps to modify an XY graph for displaying octave results.

1. Right-click the XY graph.
2. Select **X Scale»Mapping»Logarithmic** from the shortcut menu.
3. Right-click the Plot Legend.
4. Select **Bar Plots** and the style for the bar graph from the shortcut menu.
5. Right-click the Plot Legend.
6. Select **Fill Base Line»-Infinity** from the shortcut menu.

Weighting Filters

Use the SVT Weighting Filter (octave) VI to apply A-, B-, or C-weighting filters to a previously computed octave spectrum or spectra. Refer to the *Sound and Vibration Toolset Help* for more information about the SVT Weighting Filter (octave). Refer to Chapter 3, [Weighting Filters](#), for more information about weighting filters.

Frequency Analysis

This chapter discusses methods used by the Frequency Analysis VIs located on the **Functions»Sound & Vibration»Frequency Analysis** palette for windowing, averaging, and performing frequency-domain measurements.

Refer to the *Sound and Vibration Toolset Help* for information about individual Frequency Analysis VIs.

The Frequency Analysis VIs offer various frequency measurements based on the discrete Fourier transform (DFT). Refer to the *LabVIEW Measurements Manual* for information about the DFT and fast FFT sizes.



Note For simplicity, the remainder of this document uses the term FFT to denote both the fast Fourier transform (FFT) and the DFT.

Figure 8-1 illustrates the relationship of frequency analysis to the sound and vibration measurement process.

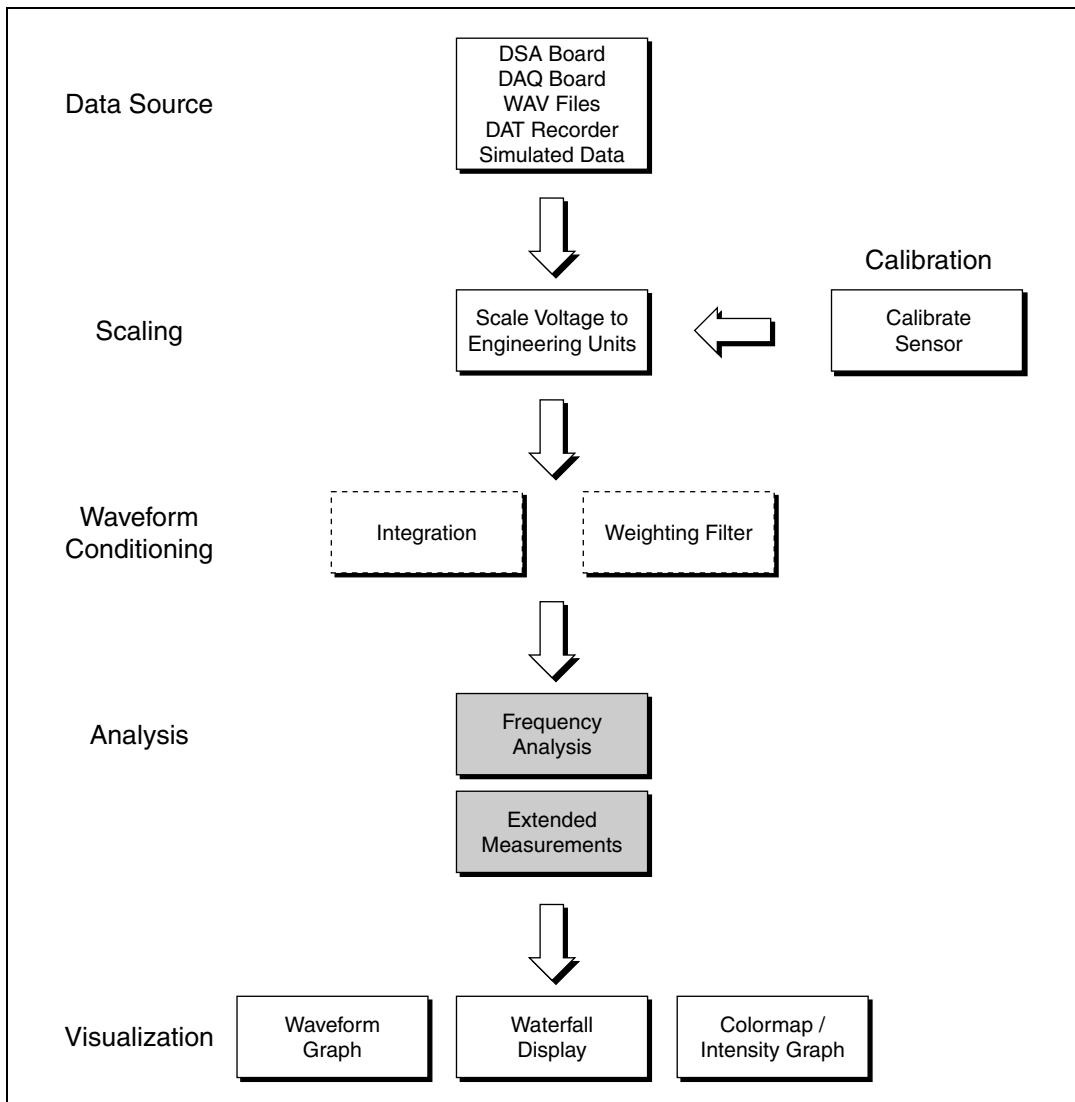


Figure 8-1. Relationship of Frequency Analysis to the Sound and Vibration Process

FFT Fundamentals

The FFT resolves the time waveform into its sinusoidal components. The FFT takes a block of time-domain data and returns the frequency spectrum of that data. The FFT is a digital implementation of the Fourier transform. Thus, the FFT does not yield a continuous spectrum. Instead, the FFT returns a discrete spectrum where the frequency content of the waveform is resolved into a finite number of frequency lines, or bins.

Number of Samples

The computed spectrum is completely determined by the sampled time waveform input to the FFT. If an arbitrary signal is sampled at a rate equal to f_s over an acquisition time T , N samples are acquired. T is given by the following equation.

$$T = \frac{N}{f_s}$$

where T is the acquisition time,
 N is the number of samples acquired,
 f_s is the sampling frequency.

N is given by the following equation.

$$N = T \cdot f_s$$

where N is the number of samples acquired,
 T is the acquisition time,
 f_s is the sampling frequency.

Frequency Resolution

Because of the properties of the FFT, the spectrum computed from the sampled signal has a frequency resolution df . The frequency resolution is given by the following equation.

$$df = \frac{1}{T} = \frac{f_s}{N}$$

where df is the frequency resolution,
 T is the acquisition time,
 f_s is the sampling frequency,
 N is the number of samples.



Note The frequency resolution is determined solely by the acquisition time. The frequency resolution improves as the acquisition increases.

Maximum Resolved Frequency

The sampling rate of the time waveform determines the maximum frequency that can be resolved. According to the Shannon Sampling Theorem, the maximum frequency that can be resolved must be half the sampling frequency and is given by the following equation.

$$f_N = \frac{f_s}{2}$$

where f_N is the maximum frequency that can be resolved and is the Nyquist frequency,
 f_s is the sampling frequency.

Minimum Resolved Frequency

The minimum frequency that can be resolved is 0 (DC). The term baseband analysis is often used to describe analysis from 0 to f_N .

Number of Spectral Lines

The number of lines in the spectrum is half the number of samples, N , in the waveform.

For the Zoom FFT VIs in the Sound and Vibration Toolset, you directly specify the number of lines in the spectrum. For the Baseband FFT and the Baseband Subset VIs, you control the number of spectral lines by specifying the number of data samples.

Relationship between Time-Domain and Frequency-Domain Specifications and Parameters

Table 8-1 summarizes the relationship of time-domain specifications to frequency-domain parameters.

Table 8-1. Time-Domain Specifications to Frequency-Domain Parameters

		Frequency Domain			
		f_N	f_{\max}	# lines	df
Time Domain	f_s N T	$\frac{f_s}{2}$	$f_s \cdot E_b$	$E_b \cdot N$	$\frac{1}{T} = \frac{f_s}{N}$

where f_s is the sampling frequency,
 E_b is the effective bandwidth,
 N is the number of samples acquired,
 T is the acquisition time.

Use the information contained in Table 8-2 if you prefer to specify the spectrum parameters and from those specifications determine the required data-acquisition parameters.

Table 8-2. Frequency-Domain Specifications to Time-Domain Parameters

		Time Domain		
		fs	N	T
Frequency Domain	f_{\max} # lines df	$\frac{f_{\max}}{E_b}$	$\frac{\# \text{ lines}}{E_b}$	$\frac{1}{df} = \frac{\# \text{ lines}}{f_{\max}}$

where f_{\max} is the maximum frequency that can be resolved,
 E_b is the effective bandwidth,
lines is the number of lines in the spectrum,
df is the frequency resolution.

In Tables 8-1 and 8-2, E_b is the ratio of the alias-free bandwidth to the sampling frequency. In traditional instruments, E_b is typically $\frac{1}{2.56}$. However, the alias-free bandwidth depends on the hardware used to acquire the dynamic signal. E_b can have a maximum value of 0.5. This maximum value corresponds to a perfect anti-aliasing filter. For example, if 1,024 samples are input to the FFT algorithm, the computed spectrum has 512

non-DC spectral lines. The computed spectrum has a total of 513 lines including the DC component. If these same 1,024 samples are acquired on an instrument with a standard $\frac{1}{2.56}$ effective bandwidth, use the equation in Table 8-1 to find the expected number of alias-free lines in the computed spectrum, as shown in the following equations.

$$\# \text{ lines} = E_b \cdot N$$

$$\# \text{ lines} = \frac{1}{2.56} \cdot 1,024 = 400 \text{ lines}$$

E_b is entirely a property of the hardware. However, mathematically, you can use the FFT to compute the frequency spectrum up to the Nyquist frequency. You must account for the presence or absence of an anti-aliasing filter when performing frequency analysis. The Frequency Analysis VIs compute every spectral line, alias-free or not. With FFT subset and zoom FFT measurements, you can use the frequency range to limit the analysis to the alias-free region of the spectrum.



Note The sampling frequency and the block size acquired during each cycle of a continuous acquisition completely determine the frequency-domain parameters in baseband FFT analysis, as in Table 8-1. However, many stand-alone instruments are operated by specifying the frequency range of interest and the number of lines in the FFT. The frequency range of interest and the number of lines in the FFT are used internally by the stand-alone instrument to determine an appropriate sampling frequency and block size, as in Table 8-2.

Increasing Frequency Resolution

Increasing the frequency resolution helps you distinguish two individual tones that are close together. For example, a signal being analyzed contains two tones at 1,000 Hz and 1,100 Hz. A sampling frequency of 10,000 Hz is used to perform the analysis. If data is acquired for 10 ms, the frequency resolution is 100 Hz. Figure 8-2 shows the results of this analysis. In this particular case, you cannot distinguish one tone from the other.

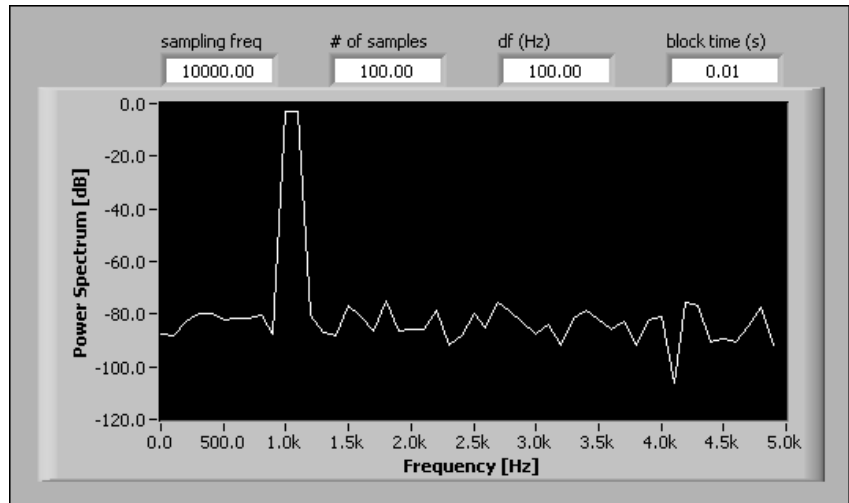


Figure 8-2. Power Spectrum Obtained with an Acquisition Time of 10 ms

By increasing the acquisition time to 1 second, you can achieve a frequency resolution of 1 Hz. Figure 8-3 shows the results obtained with a 1 second acquisition time. With the increased acquisition time, you can distinguish the individual tones.

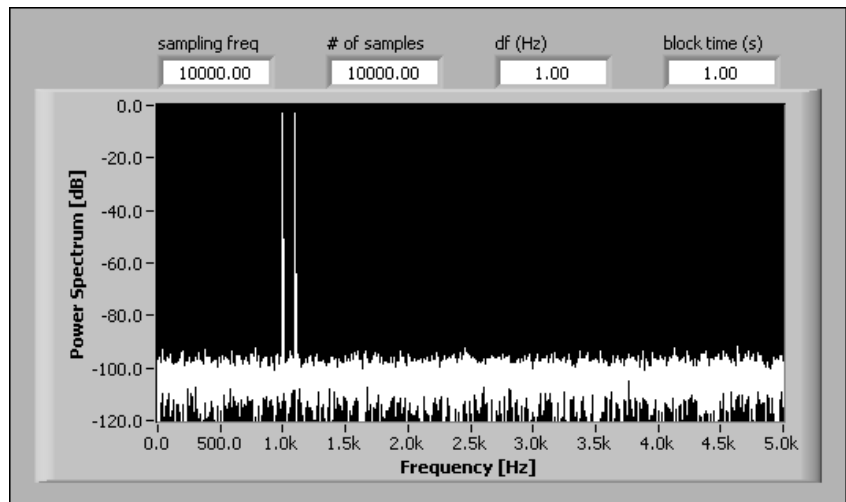


Figure 8-3. Power Spectrum Obtained with an Acquisition Time of 1 s

The following strategies achieve a finer frequency resolution:

- Decreasing the sampling frequency f_s
- Increasing the number of samples N

Decreasing f_s usually is not practical because decreasing f_s reduces the frequency range. Increasing N yields an increased number of lines over the original frequency range.

You can implement the decreased f_s strategy with zoom FFT analysis. You can use baseband FFT and FFT subset analyses to implement the increased N strategy. Baseband FFT analysis and FFT subset analysis both achieve the same frequency resolution. However, FFT subset analysis only computes a narrow subset of the spectrum.

The *Zoom FFT Analysis* and the *Subset Analysis* sections that follow contain examples that demonstrate the importance of frequency resolution in frequency analysis. The examples illustrate how to achieve a finer frequency resolution with the frequency analysis tools in the Sound and Vibration Toolset.

Zoom FFT Analysis

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.

The Zoom FFT VIs complete the following steps to process the sampled data.

1. Modulate the acquired data to center the analysis band at 0 frequency.
2. Filter the modulated data in the time domain to isolate the analysis band and prevent aliasing when the data is resampled at a lower sampling frequency.
3. Decimate the filtered data to reduce the effective sampling frequency.
4. Accumulate the decimated data until sufficient samples are available to compute the spectrum.
5. Use the Discrete Zak Transform (DZT) to efficiently compute the desired spectral lines.
6. Demodulate, or shift, the computed spectrum.

Frequency Resolution of the Zoom FFT VIs

You can use the Zoom FFT VIs to compute the spectrum of a signal over a narrow frequency range with an arbitrarily fine frequency resolution. Use the following simple formula to get an approximation of the frequency resolution seen with the Zoom FFT VIs.

$$\text{frequency resolution} \approx \frac{\text{stop frequency} - \text{start frequency}}{\text{number of lines}}$$



Note The exact frequency resolution is returned as **df** in the spectrum computed by the Zoom FFT VIs.

Zoom Measurement

The following example demonstrates a zoom measurement of the power spectrum.

A sine wave at 1,390 Hz is acquired with a DSA card and the VI shown in Figure 8-4.

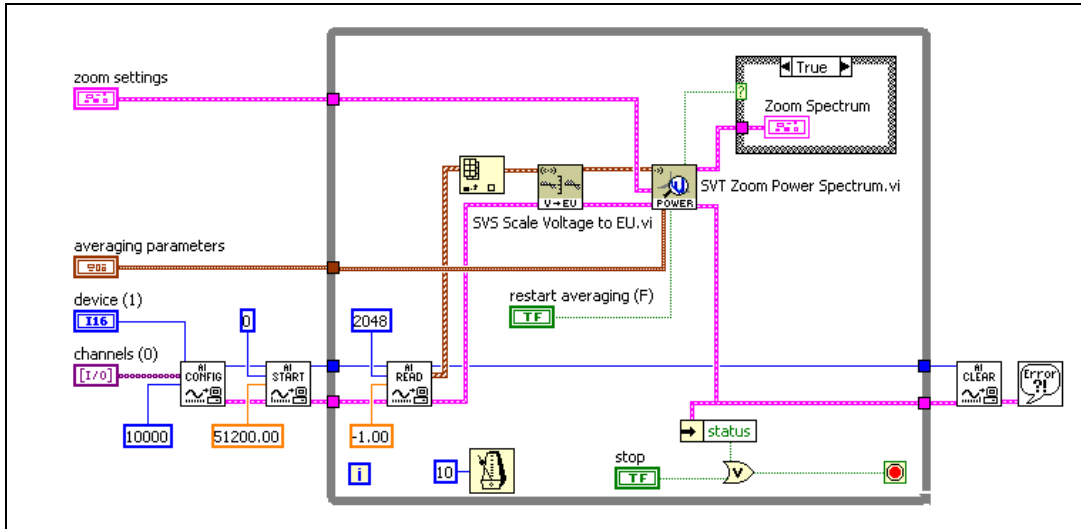


Figure 8-4. Zoom Power Spectrum VI Block Diagram

The signal is acquired at 51.2 kHz. The data is read in blocks of 2,048 samples. The frequency resolution of this measurement using baseband analysis is computed by the following equation.

$$\frac{51,200 \text{ Hz}}{2,048} = 25 \text{ Hz}$$

You can use the SVT Zoom Power Spectrum VI located on the **Functions» Sound & Vibration» Frequency Analysis» Zoom FFT** palette to analyze a narrow band with a much finer frequency resolution. Figure 8-5 shows the result of limiting the measurement to the frequency band between 1 kHz and 2 kHz and computing 400 lines. The frequency resolution of the computed spectrum is derived with the following equation.

$$\frac{2,000 - 1,000 \text{ Hz}}{400 \text{ lines}} = 2.5 \text{ Hz}$$

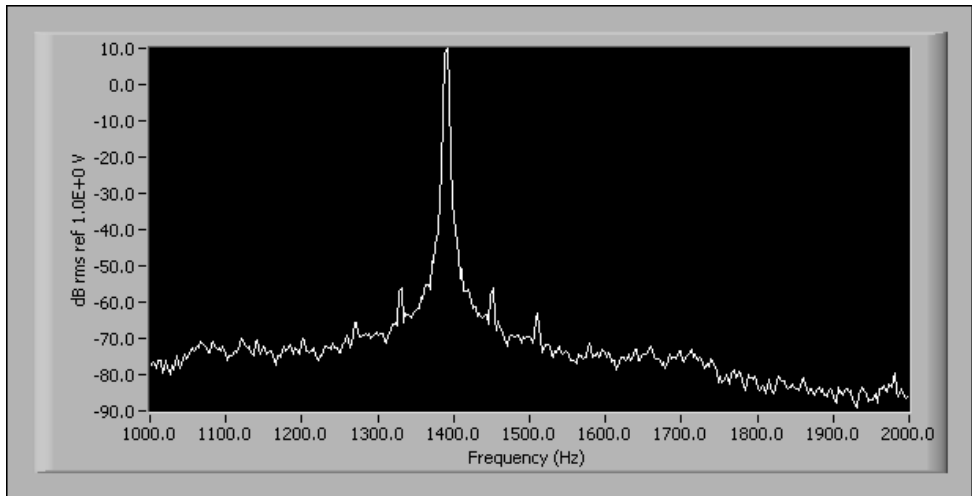


Figure 8-5. Zoom Power Spectrum VI Results

Zoom Settings

Use the **zoom settings** control, shown in Figure 8-6, to specify the frequency range, window, number of lines, and percent overlap used in the zoom analysis. Refer to the *LabVIEW Sound and Vibration Toolset Help* for information about the **zoom settings** control.

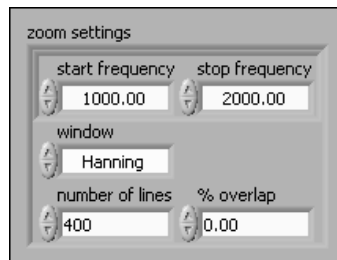


Figure 8-6. Zoom Settings Control

Subset Analysis

The Baseband Subset VIs located on the **Functions»Sound & Vibration»Frequency Analysis»Baseband FFT»Baseband Subset** palette allow you to compute subset of the baseband FFT measurement. Subset analysis uses the Discrete Zak Transform (DZT) to compute a subset of the baseband FFT. The frequency resolution for spectral measurements computed with the Baseband Subset VIs equals the

frequency resolution for measurements made with the Baseband FFT VIs. The acquisition time determines the frequency resolution. The only way to achieve a finer frequency resolution is to increase the length of the time record. In the case of baseband or subset analysis, a longer time record implies a larger block size.

The algorithm used by the Baseband Subset VIs computes only the desired spectral lines. The only programming difference between the Baseband Subset VIs and the Baseband FFT VIs is the additional parameter **frequency range**. **frequency range** specifies which spectral lines the Baseband Subset VI computes. The computed spectral lines are always inclusive of the start frequency and the stop frequency.



Note Setting the start frequency to 0 Hz and the stop frequency to the Nyquist frequency yields the same spectrum as the corresponding Baseband FFT VI.

The following consideration can help you decide when to use the Baseband Subset VIs instead of the Baseband FFT VIs:

- The required block size yields an acceptable frequency resolution.
- Analyzing a narrow subset of the baseband span requires better performance than the Baseband FFT VI can provide.

Using the Frequency Analysis VIs

This section provides specific information about using the Frequency Analysis VIs in your measurements.

Available Measurements

The following Frequency Analysis palettes offer the same basic measurements but designed for specific measurement needs:

- **Functions»Sound & Vibration»Frequency Analysis»Baseband FFT**
- **Functions»Sound & Vibration»Frequency Analysis»Baseband FFT»Baseband Subset**
- **Functions»Sound & Vibration»Frequency Analysis»Zoom FFT**

For example, each of the three palettes contains a VI for measuring the power spectrum. The SVT Power Spectrum VI located on the **Baseband FFT** palette computes the power spectrum of the input signal. The SVT Power Spectrum Subset VI located on the **Baseband Subset** palette computes a subset of the power spectrum of the input signal. The SVT

Zoom Power Spectrum VI located on the **Zoom FFT** palette computes a zoom power spectrum of the input signal. Refer to the *LabVIEW Sound and Vibration Toolset Help* for a complete listing of the Frequency Analysis VIs and information about each VI.

The Baseband FFT, Baseband Subset, and Zoom FFT VIs all share the same basic relationships between the input signal and the computed spectrum. For the baseband and subset analyses, you can obtain a tighter frequency resolution only by increasing the block size. Increasing the block size results in an FFT computed with more lines. Zoom analysis internally reduces the sampling frequency by decimating the data. In baseband FFT, baseband subset, and zoom FFT analyses, the frequency resolution is the inverse of the measurement duration.

Single-Channel Measurements

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 the magnitude 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.

Power Spectrum Measurement

You can use the Frequency Analysis VIs to perform a single-shot acquisition, as illustrated by the block diagram in Figure 8-7, or in a continuous acquisition, similar to the block diagram shown in Figure 8-8.

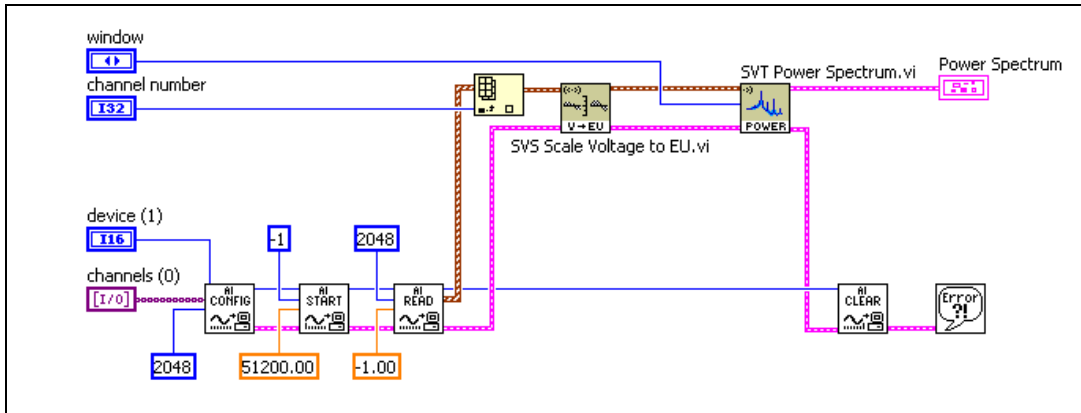


Figure 8-7. Single-Shot Acquisition

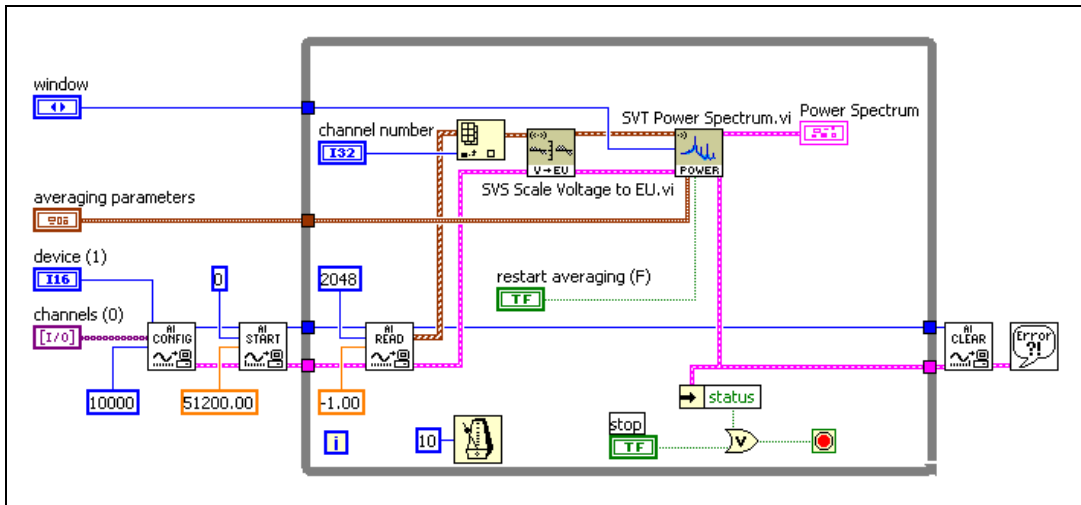


Figure 8-8. Continuous Acquisition



Note The output of the AI Read VI is an array of waveforms. You can wire the output array from the AI Read VI directly to the SVS Scale Voltage to EU VI. If the AI Read VI output array contains only a single waveform, index the array out of AI Read VI before

wiring the waveform to the SVS Scale Voltage to EU VI, as shown in Figures 8-7 and 8-8. Refer to the *LabVIEW Help* for information about indexing arrays and the AI Read VI.

Dual-Channel Measurements

Dual-channel measurements differ from single-channel measurements because the output spectrum of the dual-channel measurements is dependent on the relationship between two input channels. Typically, the input signals are a stimulus and a response. Some form of broadband excitation usually is required to obtain accurate results. Broadband signals include noise, chirps, multi-tone signals, impulses, and so on.

You can make the following dual-channel measurements with the Frequency Analysis VIs:

- Frequency response computes the transfer function of the response to the stimulus. You can use the frequency response as a general tool to characterize the dynamic response of a system. The coherence often is used to validate the frequency response results. The coherence quantifies the portion of the response that is linearly dependent on, or coherent with, the stimulus. The coherence ranges from 0 to 1.
- Cross spectrum computes the cross spectrum of the two inputs. You use the cross spectrum in some advanced analyses. Dynamic data often is stored in terms of cross spectra. You can use the cross spectra to compute other useful measurements, such as frequency response functions.
- Coherent output power computes the portion of the response power that is coherent with the stimulus.

You can view most dual-channel measurement results as magnitude and phase or as real and imaginary parts. You can use the **view** parameter of the Frequency Analysis VIs returning magnitude and phase to specify the following viewing options:

- Whether the magnitude is expressed in decibels
- Whether the phase is unwrapped
- Whether the phase is returned in degrees or radians

Refer to the *LabVIEW Sound and Vibration Toolset Help* for more information about the **view** parameter.

Frequency Response Function Measurement

Frequency response measurements compare a stimulus signal and a response signal in order to compute the frequency response function (FRF) of the device under test (DUT). The FRF represents the ratio of output-to-input in the frequency domain and fully characterizes linear, time-invariant systems.

Performing FRF measurements requires a signal source. You can use several stimulus signals. The stimulus signals usually are broadband signals such as white noise, pink noise, impulses, or chirp signals. Broadband signals ensure that all the frequencies are excited during acquisition.

The following example illustrates FRF measurement. In this example, an NI 4451 DSA board is used to generate the stimulus signal. The stimulus signal is white noise. The breakout box is a National Instruments BNC-2140. The DUT is a notch filter centered at 1 kHz. Figure 8-9 illustrates the connection scheme used in this example.

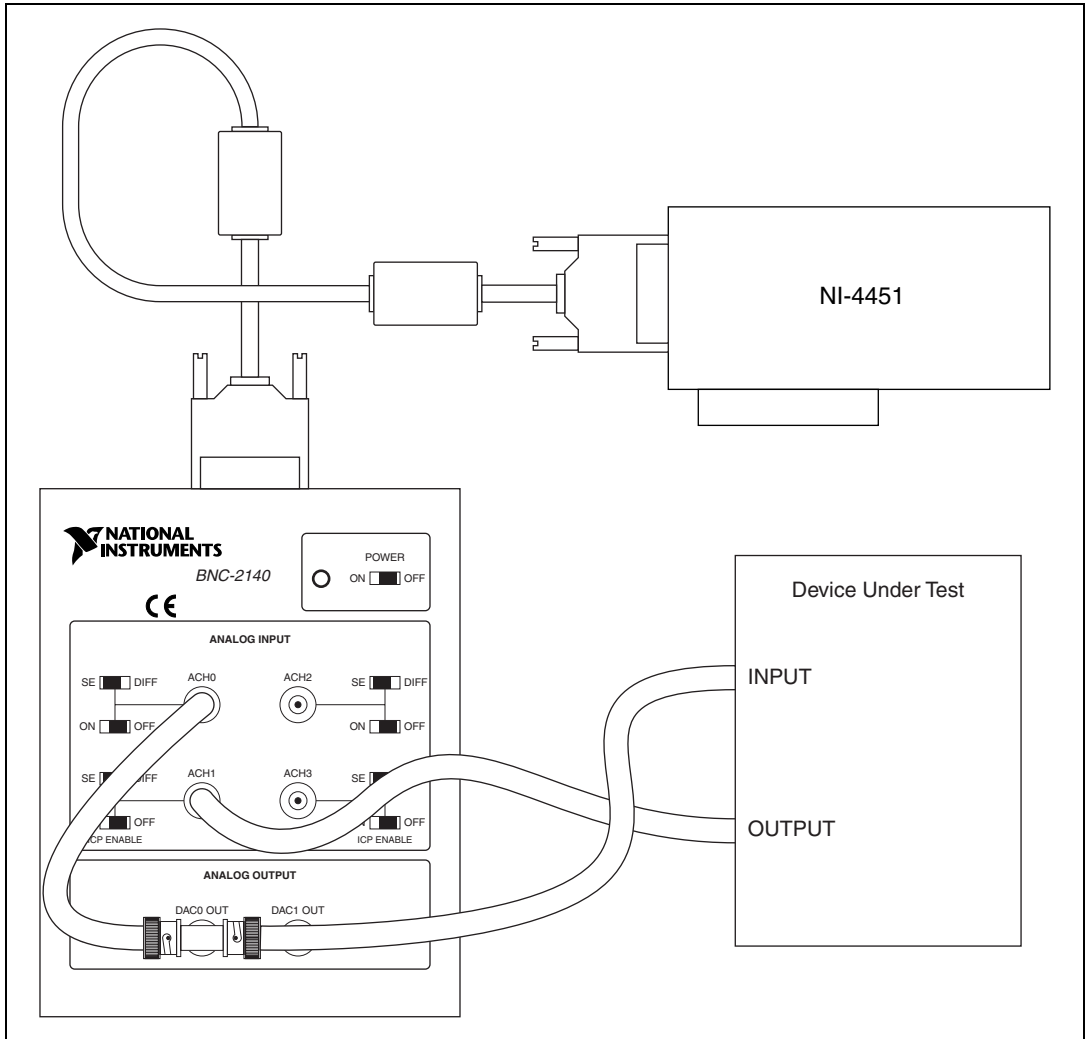


Figure 8-9. DSA Connection Scheme for FRF Measurement

The DSA board converts the stimulus signal from digital to analog. Analog output DAC0 OUT on the BNC-2140 sends the stimulus signal to the DUT. Analog input channel ACH0 receives the stimulus signal. Analog input channel ACH1 receives the response of the DUT.

Figure 8-10 shows the block diagram of the VI used to perform the FRF measurement.

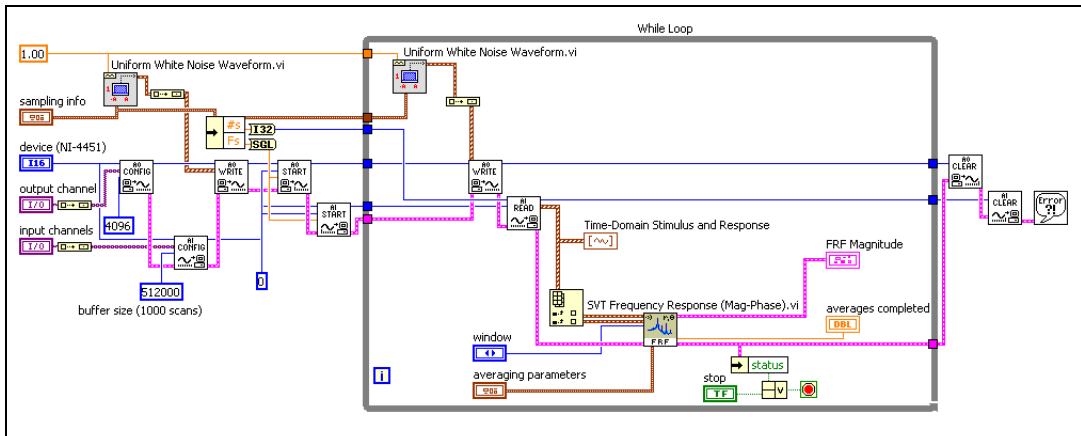


Figure 8-10. FRF Block Diagram

The **output channel** control is set to 0. The **input channels** control is set to 0, 1. The sampling frequency (f_s) is 51,200 Hz and is the same for the output and input channels. This sampling frequency means the measurement is performed in the audio domain. The buffers are configured to hold up to 10 seconds of data. The buffer size is equal to $(10)(f_s)$.

The While Loop in Figure 8-10 controls both the generation and the acquisition of the signal. For each iteration of the While Loop, the Uniform White Noise Waveform VI generates a white noise signal. The white noise signal is sent to the DUT on analog output channel 0, DAC0 OUT in Figure 8-9. Analog input channel 0, ACH0 in Figure 8-9, acquires the same number of samples as the buffer that is generated. Simultaneously, Analog input channel 1, ACH1 in Figure 8-9, acquires the response signal from the DUT. Figure 8-11 shows the measured time-domain stimulus and response signals.

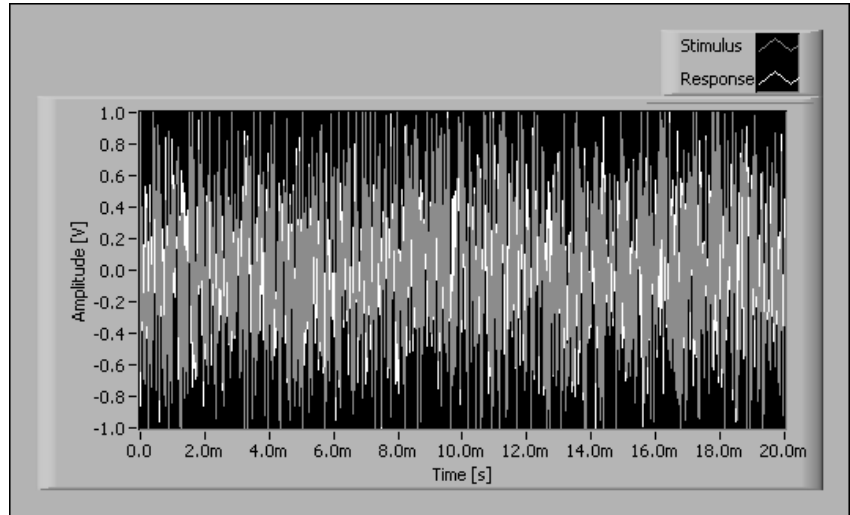


Figure 8-11. Time-Domain Stimulus and Response Signals

It is difficult to make any interpretation of the time-domain signals in Figure 8-11.

Effect of Averaging and Frequency Resolution on FRF Results

Understanding the effects averaging and frequency resolution have on FRF measurement results is important when performing FRF measurements. The following examples illustrate the effects averaging and frequency resolution have on FRF measurement results.

Both examples use the VI illustrated in Figure 8-10 with a sampling frequency of 51,200 Hz and a block size of 1,024 samples. Audio or acoustical applications commonly use this sampling frequency. The 51,200 Hz sampling frequency and the 1,024 sample buffer size permit alias-free measurements up to 20 kHz and result in a 400 line FFT.



Note The SVT Frequency Response [Mag-Phase] VI uses the stimulus and response signals from the DUT to compute the FRF. In the following examples, only the magnitude of the frequency response function is displayed.

FRF without Averaging

Figure 8-12 shows the frequency response of the notch filter measured without averaging.

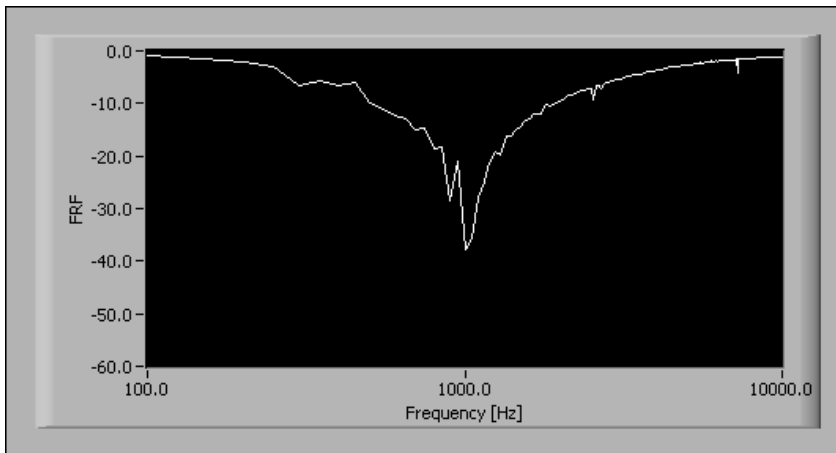


Figure 8-12. Frequency Response Without Averaging

FRF with Averaging

You can use averaging to improve the FRF measurement. Figure 8-13 shows the frequency response obtained by applying RMS averaging over 100 measurements.

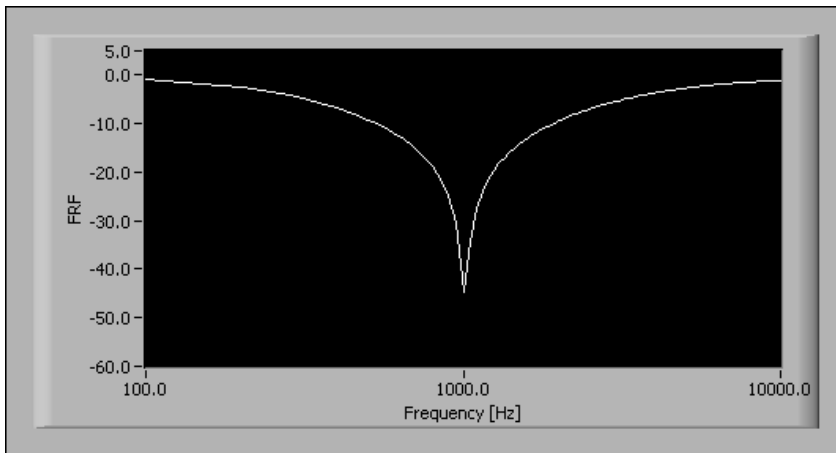


Figure 8-13. Frequency Response with Averaging

Using averaging improves the results as compared to the results obtained with no averaging.

Refer to the *Averaging Parameters* section of this chapter for more information about averaging and frequency analysis.

Improving Frequency Resolution by Increasing the Number of Samples

At a constant sampling rate, the number of samples used to process the FFT determines the frequency resolution of any FFT-based measurement. For example, assume that the filter used in the averaged FRF example is supposed to provide attenuation better than 50 dB at the notch frequency. The notch frequency is 1 kHz in this example. The averaged measurement shown in Figure 8-13 only indicates an attenuation slightly better than 40 dB at 1 kHz. The 1,024 samples and the 51,200 Hz sampling frequency lead to a 20 ms block of data and a frequency resolution of 50 Hz. However, the filter has a very narrow stopband. A better frequency resolution is needed to perform a more accurate measurement in the region of the notch.

If the processing buffer is increased to 4,096 samples and the sampling frequency is kept at 51,200 Hz, the frequency resolution becomes 12.5 Hz. The averaged measurement obtained with the new block size is shown in Figure 8-14.

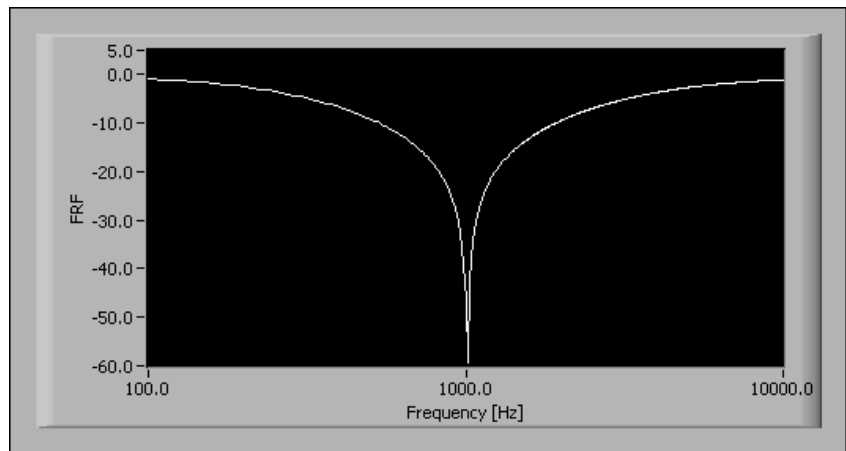


Figure 8-14. Frequency Response with 4,096 Samples

Increasing the frequency resolution results in a more accurate measurement of the attenuation of the filter around its notch frequency.

Windowing

Periodicity is one of the basic assumptions made in FFT-based frequency analysis. The FFT algorithm implicitly assumes that every block of acquired data indefinitely repeats in both positive and negative time. Windowing is one method of ensuring periodicity.

Windowing multiplies the time-domain data by a window function before the FFT is performed. Window functions typically have a value of zero at the start and end of the measurement period. Figure 8-15 shows how a signal that is not the same at the start and end of the measurement period appears not to be periodic and how this same signal becomes periodic when multiplied by a window function.

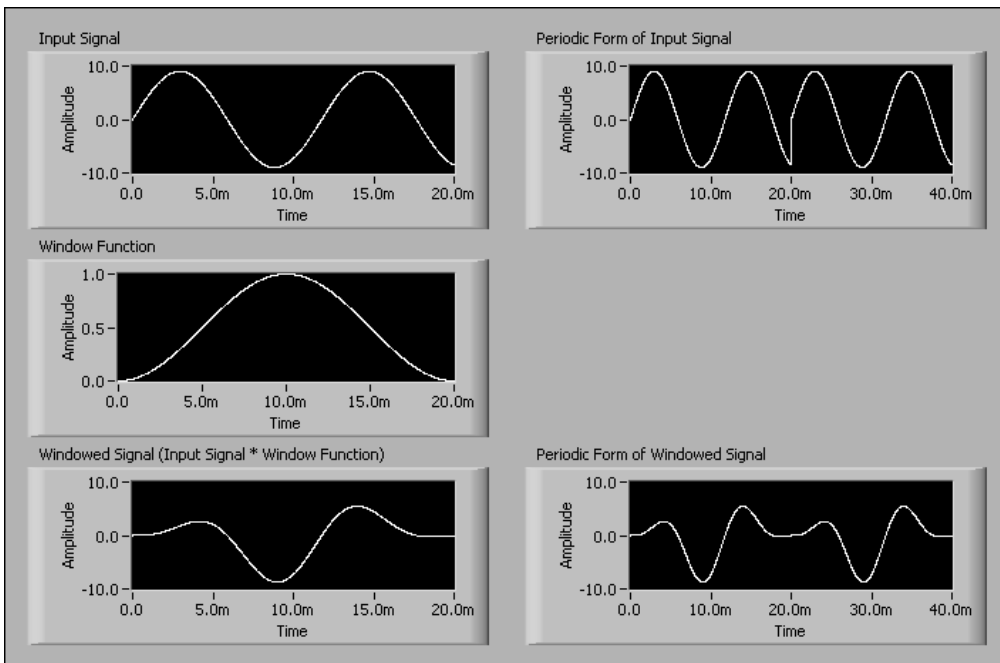


Figure 8-15. Effect of Windowing on Periodicity

The windows supported by the Sound and Vibration Toolset and their equivalent noise bandwidths (ENBW) are listed in Table 8-3. ENBW is a property of the window applied to the signal.

Table 8-3. Supported Time-Domain Windows

Window	ENBW
none	1
Hanning	1.50
Hamming	1.36
Blackman-Harris	1.71
Exact Blackman	1.69
Blackman	1.73
Flat Top	3.77
Four Term Blackman-Harris	2.00
Seven Term Blackman-Harris	2.63
Low Sidelobe	2.22
Force-Exponential	N/A

Typically, a window with a lower ENBW is used to resolve peak frequencies. A window with a higher ENBW is used to resolve peak amplitudes. The Force-Exponential window is used for shock and impulse testing where the response of the signal being tested does not decay to zero in the measurement period.



Note The Force-Exponential window is only supported for dual-channel measurements made with the Baseband FFT and Baseband Subset VIs.

Averaging Parameters

Each of the Frequency Analysis VIs supports averaging. The **averaging parameters** control, shown in Figure 8-16, defines how the averaged spectrum is computed. **averaging mode**, **weighting mode**, **number of averages**, and **linear mode** each control a particular feature of the averaging process.

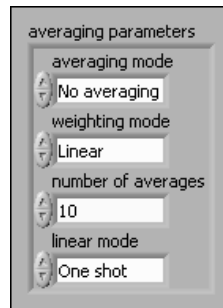


Figure 8-16. Averaging Parameters Control

Refer to the *LabVIEW Sound and Vibration Toolset Help* for more information about the **averaging parameters** control and for information about **number of averages** and **linear mode**.

Special Considerations for Averaged Measurements

When performing averaged frequency measurements, place the Frequency Analysis VI in a For Loop or a While Loop and wire a new block of time data to the VI. The Frequency Analysis VIs return the averaged results based on all the data sent since the first call to the VI or since the averaging process was restarted.

Keep in mind the following conditions when performing averaged measurements with the Baseband FFT and Baseband Subset VIs:

- You can use either contiguous or overlapping blocks of time data.
- The number of samples in each block of time data determines the FFT block size.
- The averaging automatically restarts if the size of the block of time data changes.

Averaging Mode

You can choose from the following averaging modes when performing frequency analysis with the Frequency Analysis VIs:

- No averaging
- RMS averaging
- Vector averaging
- Peak hold



Note Not all of the Frequency Analysis VIs support all of the averaging modes listed above. Refer to the *LabVIEW Sound and Vibration Help* for information about the averaging modes supported by a specific VI.

No Averaging

No averaging is the default setting and does not apply any averaging to the measurement. You can use the **No averaging** setting for quick computations or when the signal-to-noise ratio is high.

RMS Averaging

RMS averaging reduces signal fluctuations but not the noise floor. The noise floor is not reduced because RMS averaging averages the power of the signal. Because RMS averaging averages the power of the signal, averaged RMS quantities of single-channel measurements have zero phase. RMS averaging for dual-channel measurements preserves important phase information.

RMS averaged measurements are computed according to the following equations:

$$\begin{aligned}
 \text{FFT spectrum} & \quad \sqrt{\langle X^* \cdot X \rangle} \\
 \text{power spectrum} & \quad \langle X^* \cdot X \rangle \\
 \text{cross spectrum} & \quad \langle X^* \cdot Y \rangle \\
 \text{frequency response} & \quad H1 = \frac{\langle X^* \cdot Y \rangle}{\langle X^* \cdot X \rangle} \\
 & \quad H2 = \frac{\langle Y^* \cdot Y \rangle}{\langle Y^* \cdot X \rangle} \\
 & \quad H3 = \frac{(H1 + H2)}{2},
 \end{aligned}$$

where X is the complex FFT of the stimulus signal x ,
 Y is the complex FFT of the response signal y ,
 X^* is the complex conjugate of X ,
 Y^* is the complex conjugate of Y ,
 $\langle X \rangle$ is the average of X , real and imaginary parts being averaged separately,
 df is the frequency resolution/bin width of the spectrum.

Vector Averaging

Vector averaging, also called coherent averaging or time synchronous averaging, reduces the amount of noise in synchronous signals. Vector averaging computes the average of complex quantities directly. The real and imaginary parts are averaged separately, which preserves phase information. However, for single-channel measurements, the use of vector averaging without a triggered acquisition can cause strong spectral components to be eliminated in the averaged spectrum.

Vector averaged measurements are computed according to the following equations.

$$\begin{array}{ll}
 \text{FFT spectrum} & \langle X \rangle \\
 \text{power spectrum} & \langle X^* \rangle \cdot \langle X \rangle \\
 \text{cross spectrum} & \langle X^* \rangle \cdot \langle Y \rangle \\
 \text{frequency response} & \frac{\langle X \rangle}{\langle Y \rangle} \quad (H1 = H2 = H3)
 \end{array}$$

where X is the complex FFT of the stimulus signal x ,
 Y is the complex FFT of the response signal y ,
 $\langle X \rangle$ is the average of X , real and imaginary parts being averaged separately,
 $\langle Y \rangle$ is the average of Y , real and imaginary parts being averaged separately.

RMS versus Vector Averaging

Figure 8-17 compares the effect of RMS averaging and vector averaging on a typical signal. The input signal is a two-tone signal. The dominant tone is a 10 kHz sine wave with an amplitude of $1V_p$. The smaller component is a 15 kHz sine wave with an amplitude of $0.01 V_p$. In addition to the tones, noise is present in the signal. The signal is sampled at 51.2 kHz in blocks of 1,000 samples. A Hanning window is applied to reduce leakage. The VI completes 100 averages.

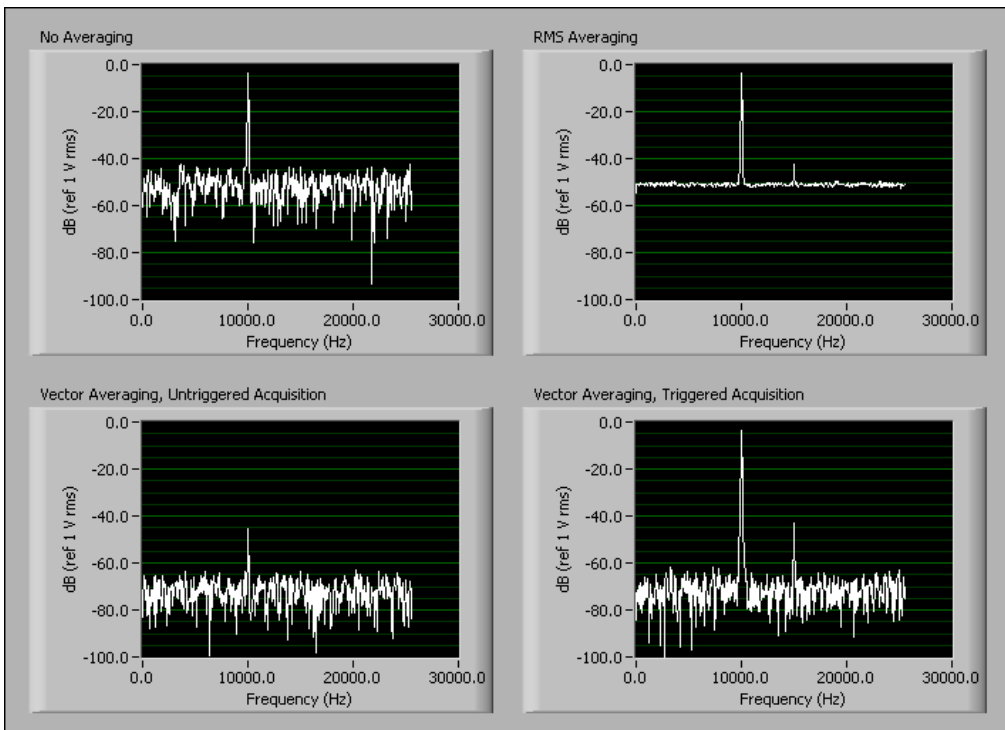


Figure 8-17. Effect of Averaging Mode on Averaged Spectrum

No Averaging only identifies the dominant tone.

RMS averaging does not reduce the noise floor. However, RMS averaging does smooth the noise out enough to unmask the tone at 15 kHz.

Vector averaging without a trigger underestimates the energy present at 10 kHz. Also, the tone at 15 kHz is indistinguishable from the noise.

Vector averaging with triggered acquisition accurately computes the energy of the tones, reduces the noise floor by 20 dB, and reveals the tone at 15 kHz. The 20 dB reduction in the noise floor corresponds to a factor of 10, or $\sqrt{100}$, where 100 is the number of averages completed.

Peak Hold

Peak hold is performed at each individual frequency line and retains the RMS peak levels of the averaged quantities from one FFT record to the next. Peak-hold averaging is most useful when configuring a measurement system or when applying limit, or upper limit, testing to a frequency spectrum.

Peak hold averaged measurements are computed according to the following equations.

$$\text{FFT spectrum} \quad \text{MAX} \sqrt{X^* \cdot X}$$

$$\text{power spectrum} \quad \text{MAX}(X^* \cdot X)$$

where X is the complex FFT of the stimulus signal x ,
 X^* is the complex conjugate of X .

Weighting Mode

Linear weighting weights each individual spectrum by the same amount in the averaged spectrum. Linear weighting is most often used for analysis.

Exponential weighting weights the most recent spectrum more than previous spectra. Weighting the most recent spectrum more than previous spectra makes the averaged spectrum more responsive to changes in the input signal. This responsiveness makes exponential weighting ideal for the configuration phase of a measurement. Exponential weighting also is useful for monitoring applications because the averaged spectrum responds to a singular event. A linearly averaged spectrum might not respond noticeably to a singular event, especially with a large number of averages.

Weighting is applied according to the following equation.

$$Y_i = \frac{N-1}{N} Y_{i-1} + \frac{1}{N} X_i,$$

where X_i is the result of the analysis performed on the i^{th} block,
 Y_i is the result of the averaging process from X_1 to X_i ,
 $N = i$ for linear weighting,
 N is a constant for exponential weighting with $N = 1$ for $i = 1$.

Coherence and Coherent Output Power

Coherence and coherent output power only support RMS averaging and are computed according to the following equations.

$$\text{Coherence} \quad \gamma^2 = \frac{|\langle X^* \cdot Y \rangle|^2}{\langle X^* \cdot X \rangle \langle Y^* \cdot Y \rangle}$$

$$\text{Coherent output power} \quad \text{COP} = \gamma^2 \langle Y^* \cdot Y \rangle,$$

where X is the complex FFT of the stimulus signal x ,
 Y is the complex FFT of the response signal y ,
 X^* is the complex conjugate of X ,
 Y^* is the complex conjugate of Y ,
 $\langle X \rangle$ is the average of X , real and imaginary parts being averaged separately.

Extended Measurements

You can use the VIs located on the **Functions»Sound & Vibration»Frequency Analysis»Extended Measurements** palette to perform the following measurements:

- Frequency-domain weighting
- Spectrum peak search
- Power in band
- Units conversion
- Frequency-domain integration

Refer to Chapter 3, [Weighting Filters](#), for information about weighting.

Refer to Chapter 4, [Integration](#), for information about frequency-domain integration.

Refer to the *Sound and Vibration Toolset Help* for information about the individual Extended Measurements VIs.

Power in Band

The SVT Power in Band VI measures the total power within the specified frequency range. Table 8-4 shows the equations for computing power in band based on the type of input spectrum.

Table 8-4. Power in Band Equations Based on Input Type

Type of Input Spectrum	Equation for Power in Band
Power Spectrum (PS)	$\frac{\sum_{\text{start frequency}}^{\text{stop frequency}} \text{PS}}{\text{ENBW}}$
Power Spectral Density (PSD)	$\left(\sum_{\text{start frequency}}^{\text{stop frequency}} \text{PSD} \right) \cdot df$
Magnitude Spectrum (MS)	$\frac{\sum_{\text{start frequency}}^{\text{stop frequency}} (\text{MS})^2}{\text{ENBW}}$
where ENBW is the equivalent noise bandwidth, df is the frequency resolution of the measurement.	

Transient Analysis

This chapter discusses performing transient analysis with the Transient Analysis VIs located on the **Functions»Sound & Vibration»Transient Analysis** palette.

Refer to the *LabVIEW Sound and Vibration Toolset Help* for information about individual Transient Analysis VIs.

Figure 9-1 illustrates the relationship of transient analysis to the sound and vibration measurement process.

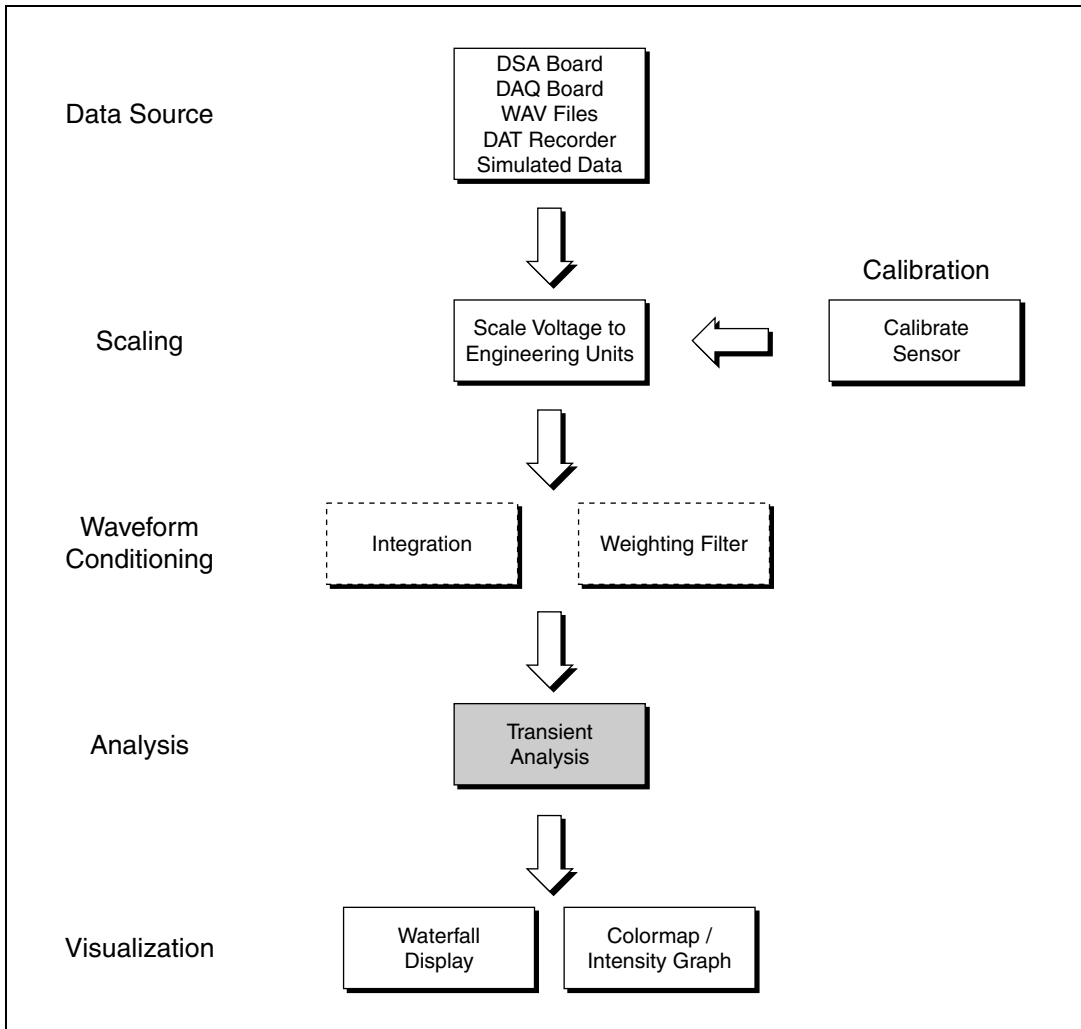


Figure 9-1. Relationship of Transient Analysis to the Sound and Vibration Process

Transient Analysis with the Sound and Vibration Toolset

Transient analysis is the analysis of non-stationary signals. The Transient Analysis VIs use the short-time Fourier transform (STFT). You can use the Transient Analysis VIs directly to extract frequency information as a function of time. In the case of rotating machines where a tachometer signal is simultaneously acquired, the Transient Analysis VIs can extract frequency information as a function of the rotational speed.

The results generated by the STFT are typically displayed on a waterfall display or on a colormap. The Transient Analysis VIs return the information needed to properly scale the axes of the displays. You can pass the information directly to a Waterfall Display VI. Property nodes are used for the colormap display. Refer to Chapter 10, [Waterfall Display](#), for more information about the waterfall display. Refer to the *Sound and Vibration Toolset Help* for information about the colormap display. Refer to the *LabVIEW Help* for information about property nodes.



Note Other LabVIEW toolsets are available that provide additional transient analysis capabilities. The Order Analysis Toolset is designed for rotating machinery analysis and monitoring. The Signal Processing Toolset has tools, such as wavelets and joint time-frequency analysis (JTFA), for the analysis of fast transients.

Performing an STFT versus Time

The STFT available in the Sound and Vibration Toolset can compute N Fourier transforms on the time-domain signal with or without overlapping.

For example, analyze a waveform containing 10 seconds of data acquired at 51.2 kS/s. The signal is a chirp signal with the following attributes:

- Start frequency = 10 Hz
- End frequency = 10,000 Hz

Figure 9-2 shows the signal corresponding to the first 200 ms of the waveform.

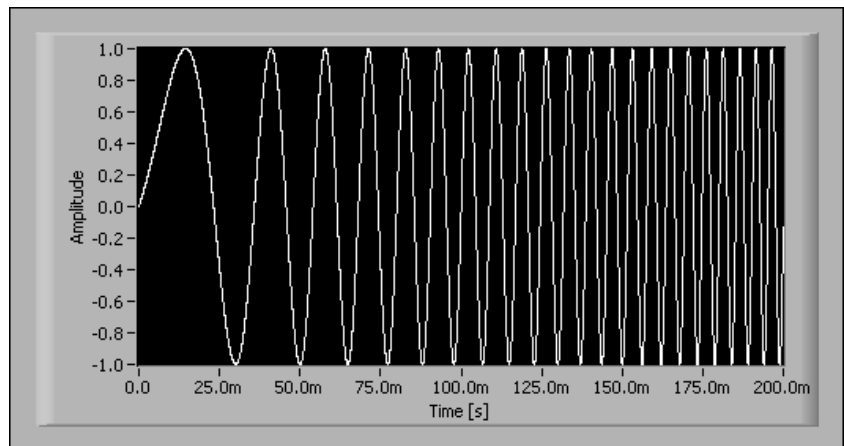


Figure 9-2. From FFT to STFT

Figure 9-3 shows the result of applying a baseband FFT on the entire waveform.

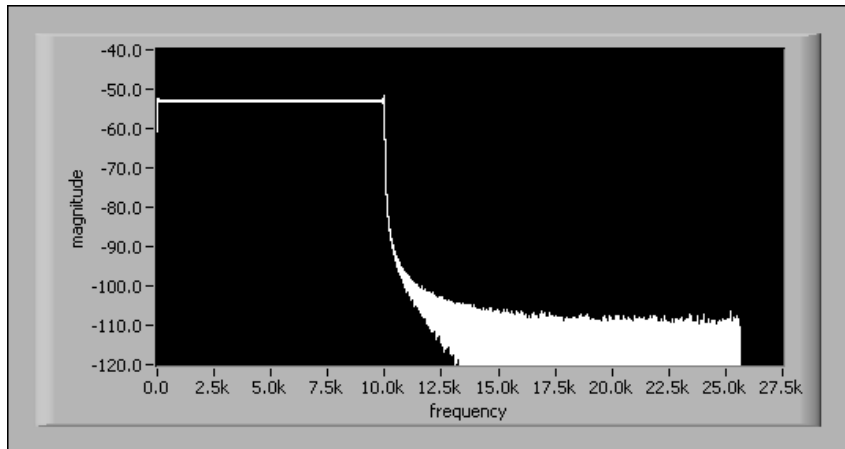


Figure 9-3. Baseband FFT



Note No window is applied on the signal.

The spectrum is flat from 10 Hz to 10 kHz. Only noise is measured at frequencies above 10 kHz. Unfortunately, this measurement does not provide any information about how the frequency content of the signal changes with time. However, the STFT can reveal useful information about the time dependence of the frequency content.

Instead of computing a single FFT on the whole data set, you can divide the data set into smaller blocks and compute N FFTs on these smaller data blocks. For example, divide the signal into 100 ms blocks and perform an FFT on each of the blocks with the SVT STFT vs Time VI.

Subdivide the time-domain signal appropriately by configuring the **time segment** control, as shown in Figure 9-4.

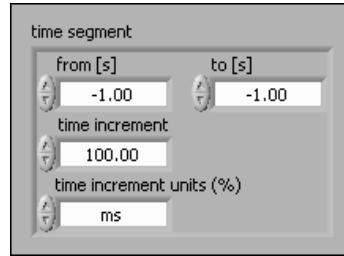


Figure 9-4. Time Segment Control

Leave **from [s]** and **to [s]** each equal to -1.00 to ensure that all of the signal is used in the STFT computation. In this particular example, the -1.00 setting in both **from [s]** and **to [s]** is equivalent to setting **from [s]** to 0 and **to [s]** to 10.

Set a 100 ms time increment by setting **time increment** to 100.00 and **time increment units (%)** to **ms**. The 100 ms time increment causes the SVT STFT vs Time VI to compute one FFT every 100 ms. Setting **time increment** is independent from selecting the FFT block size.

Selecting the FFT Block Size

In addition to the time segment, you can adjust the FFT block size. For example, analyze a chirp signal having the following attributes:

- Start frequency = 10 Hz
- End frequency = 10,000 Hz

The measurement is performed using the following settings:

- Acquisition time = 10 s
- Sampling frequency = 51.2 kS/s
- FFT block size = 1,024 samples, or 400 lines
- Time increment = 100 ms

Based on the sampling frequency of 51,200 Hz, a 1,024 sample FFT requires a 20 ms block of data, leading to a frequency resolution of 50 Hz.

Because the time increment is 100 ms and a 1,024 sample FFT only requires a 20 ms block, only one block out of five is used for computation. Figure 9-5 shows the result obtained with a 1,024 sample FFT.

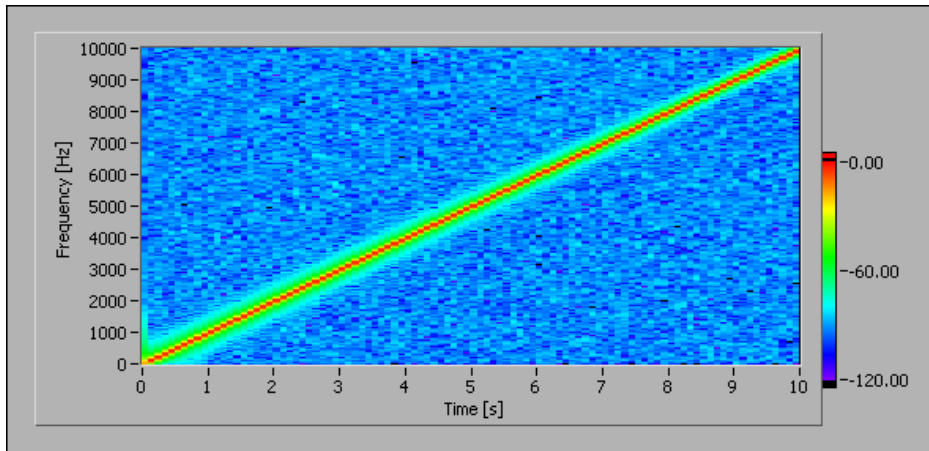


Figure 9-5. STFT Using a 1,024 Sample Block Size

If you select an FFT Block size of 4,096 samples, or 1,600 lines, the resolution improves, as shown in Figure 9-6. However, the increased resolution comes at the expense of processing.

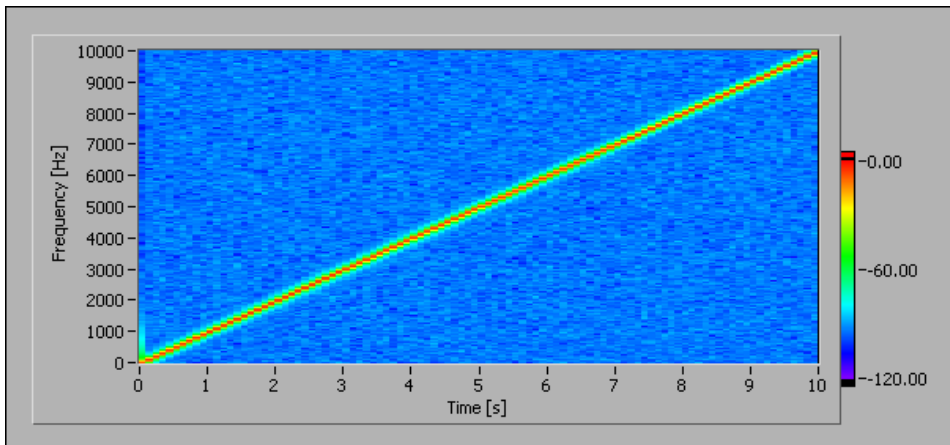


Figure 9-6. 4,096 Sample FFT

Overlapping

Overlapping is a method that uses a percentage of the previous data block to compute the FFT of the current data block. When combined with windowing, overlapping maximizes the use of the entire data set. If no overlapping is used, the part of the signal close to the window edges becomes greatly attenuated. The attenuation of the signal near the window

edges could result in the loss of information in the region near the window edges.



Note Specify % in the **time increment units (%)** section of the **time increment** control, shown in Figure 9-4, to set the desired overlapping rate. No overlapping, or 0%, corresponds to a time increment of 100%. An overlapping of 75% corresponds to a time increment of 25%. An overlapping of 50% corresponds to a time increment of 50%, and so forth. The advantage of using the time increment control is that you can specify values greater than 100%. For example, a time increment of 200% corresponds to computing an FFT on every other block of data.

Figure 9-7 and Figure 9-8 illustrate the overlapping process. Figure 9-7 shows a 50% overlap.

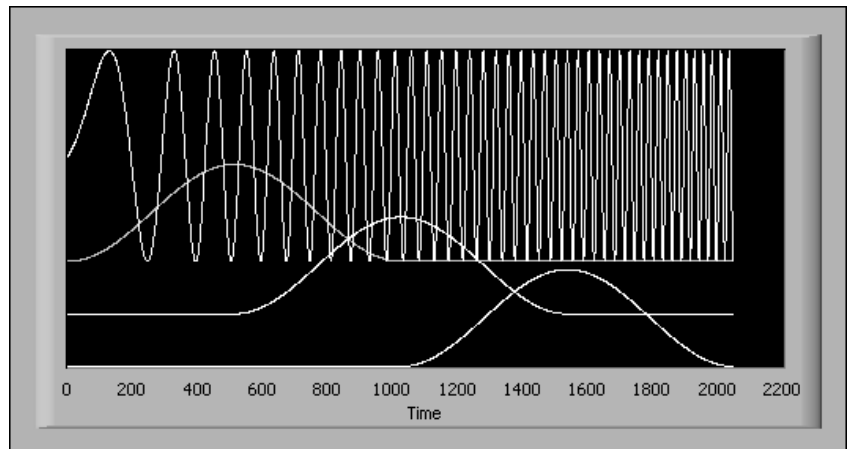


Figure 9-7. 50% Overlap

Figure 9-8 shows the resulting subdivisions when a 50% overlap and a Hamming window are used.

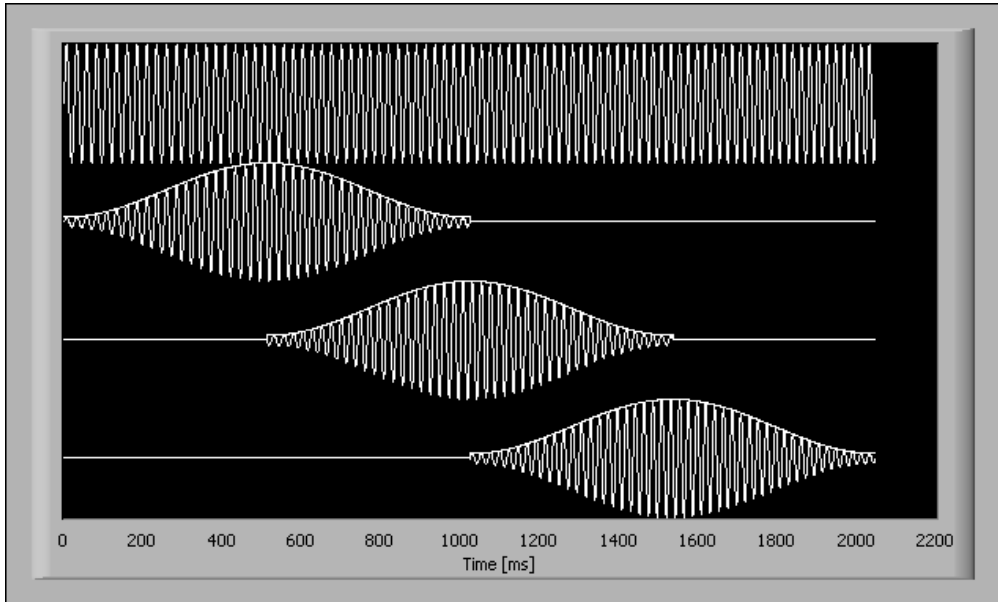


Figure 9-8. Subdivisions of the Time-Domain Waveform

Using the SVT STFT vs Time VI

The following example illustrates how to use the SVT STFT vs Time VI. Figure 9-9 shows the block diagram.

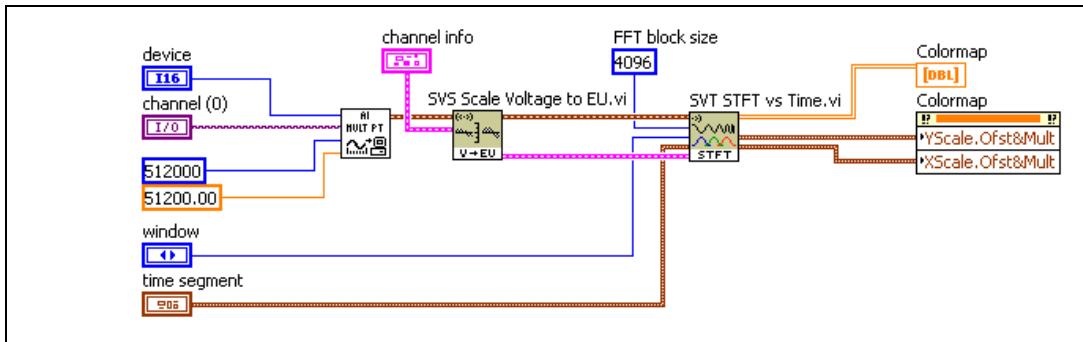


Figure 9-9. Use of the SVT STFT vs Time VI

A high level data acquisition VI is used to acquire 10 seconds of data at a sample rate of 51.2 kHz. This particular example uses the AI Acquire Waveform VI located on the **Functions>Data Acquisition>Analog Input**

palette. After scaling, the signal is sent to the SVT STFT vs Time VI. The result is displayed on an intensity graph, as shown in Figure 9-10.

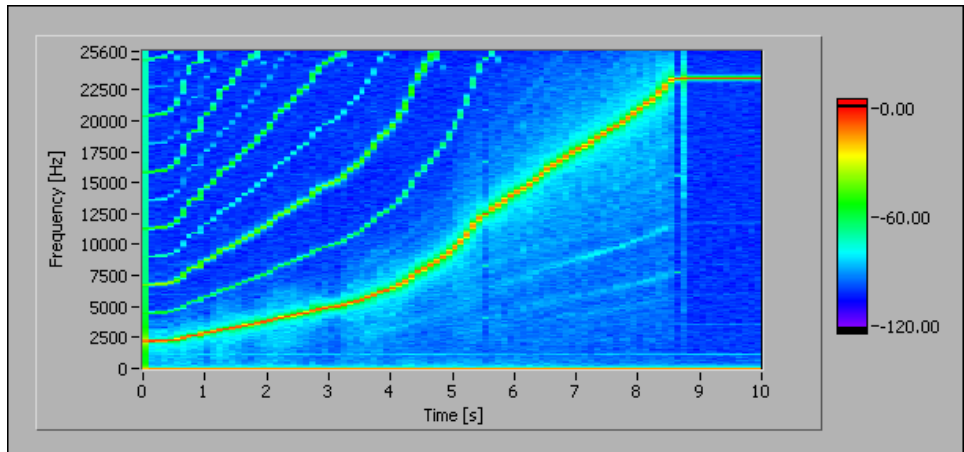


Figure 9-10. STFT vs Time Graph



Note Use the X Scale and Y Scale offset and multiplier properties to properly scale the axes of the intensity graph. In this example, the X Scale range is 0 to 10 seconds. The Y Scale range is 0 to 25,600 Hz. 25,600 Hz is the Nyquist frequency. You can adjust the Z Scale so that only the meaningful part of the signal is displayed. In other words, you can hide noise in the displayed signal by increasing the minimum limit of the z-axis. Refer to the *LabVIEW Help* for information about the offset and multiplier properties.

Performing an STFT Versus Rotational Speed

Analyzing the frequency content as a function of the rotational speed is helpful when dealing with measurements on rotating machinery. Use the SVT STFT versus RPM (analog) VI to analyze the frequency content as a function of the rotational speed.

Converting the Pulse Train to Rotational Speed

Use the SVT Convert to RPM (analog) VI to convert a pulse train acquired by a tachometer or encoder to the rotational speed expressed in rpm.



Note For simplicity, the remainder of this chapter uses the term tachometer to denote both a tachometer and an encoder.

In the following example, an accelerometer is mounted at the test location for an engine run-up. A tachometer is used to measure the speed of the shaft and returns one pulse per revolution as a TTL signal. Use the **tach info** control to specify the characteristics of the pulses generated by the tachometer. Figure 9-11 shows the settings for the **tach info** control. Refer to the *LabVIEW Sound and Vibration Toolset Help* for more information about the **tach info** control.

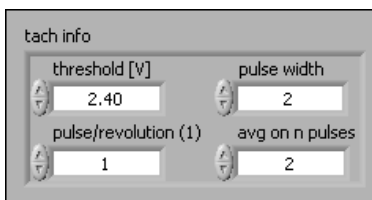


Figure 9-11. Tach Info Control

Figure 9-12 shows the tachometer signal.

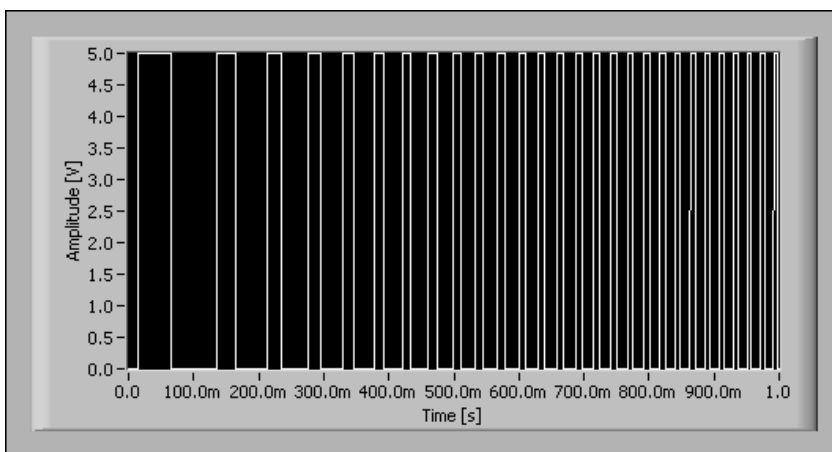


Figure 9-12. Tachometer Signal

You can use the SVT Convert to RPM (analog) VI to measure the rotational speed in rpm as a function of time. Figure 9-13 shows the result obtained with the SVT Convert to RPM (analog) VI and a simulated tachometer signal.

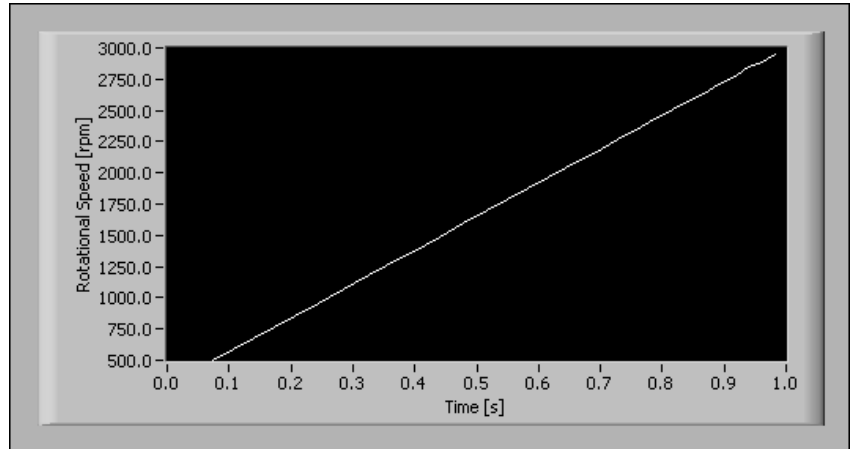


Figure 9-13. Result from SVT Convert to RPM (analog) VI

STFT Versus RPM

You also can display the STFT of an input signal as a function of the rotational speed based on the tachometer signal. Two input signals are needed, the signal of interest and the signal from the tachometer. Again, an engine run-up serves as a good example of computing an STFT as a function of the rotational speed.

During an engine run-up, the sound pressure close to the engine is measured with a microphone. Figure 9-14 shows the signal acquired by the microphone.

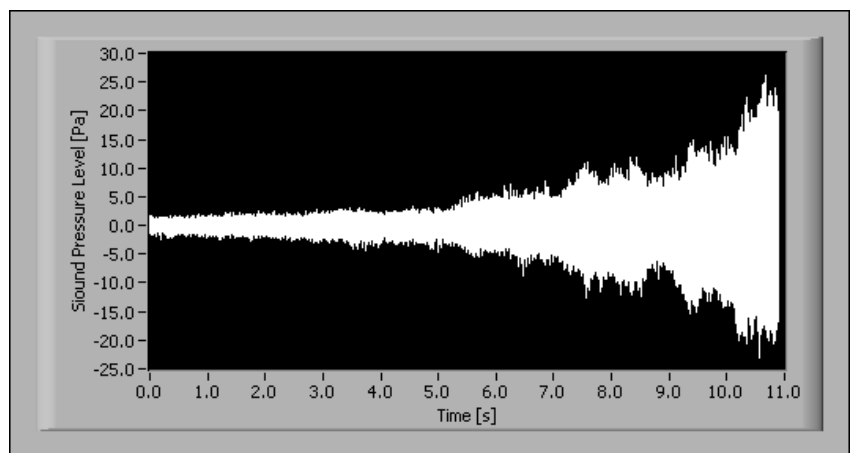


Figure 9-14. Microphone Signal Obtained During Engine Run-Up

The signal from the tachometer also is acquired. The measured tachometer signal is converted to rpm with the SVT Convert to RPM (analog) VI. Figure 9-15 shows the rotational speed as a function of time as computed by the SVT Convert to RPM (analog) VI.

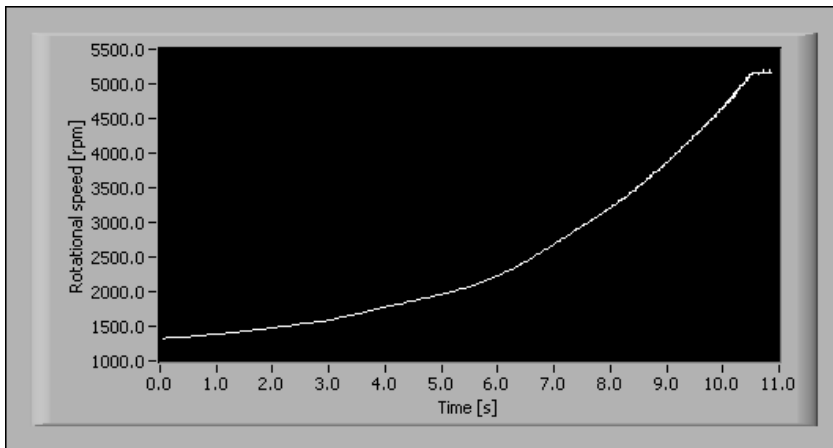


Figure 9-15. Rotational Speed as a Function of Time During Engine Run-Up

Using the SVT STFT vs RPM (analog) VI allows you to measure the frequency content of the signal as a function of the rotational speed of the engine. You can display the results obtained with the SVT STFT vs RPM (analog) VI on an intensity graph, as shown in Figure 9-16.

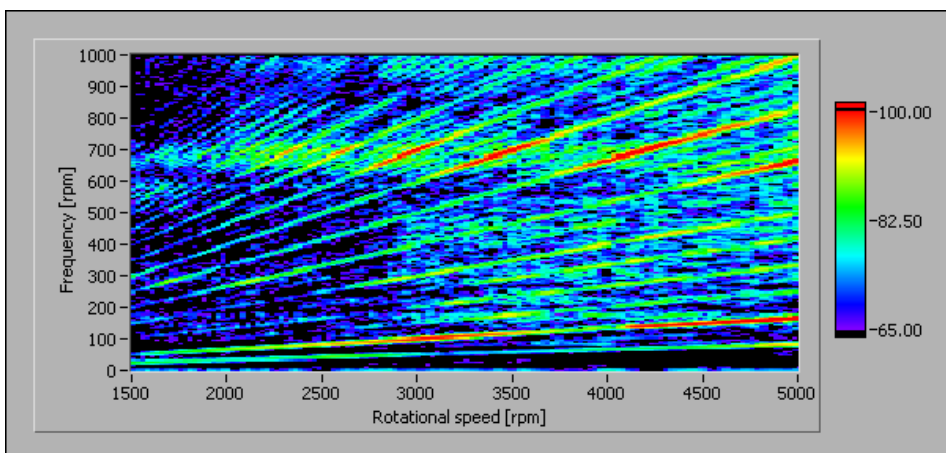


Figure 9-16. Intensity Graph

Waterfall Display

This chapter discusses using the Waterfall Display VIs located on the **Functions»Sound & Vibration»Waterfall Display** palette.

Refer to the *LabVIEW Sound and Vibration Toolset Help* for information about the individual Waterfall Display VIs.

Using the Display VIs

Waterfall display is a visualization technique that permits the visual representation of various analyses of non-stationary signals, such as machine run-up, coast down, transients, and so on.

Use the Waterfall Display VIs to display FFT spectra from frequency analysis or transient analysis and octave spectra from octave analysis in waterfall graphs. Refer to *Front Panel Displays* in the *Sound and Vibration Toolset Help* for more information about displaying octave results.

Specific Waterfall Display VIs display the results of frequency analysis and octave analysis in a waterfall graph. The waterfall display opens in an external window called the **Waterfall** window.

Complete the following basic steps to display results on a waterfall graph.

1. Initialize the display.
2. Send data to the display.
3. Close the waterfall display.

Figure 10-1 shows the **Waterfall** window.

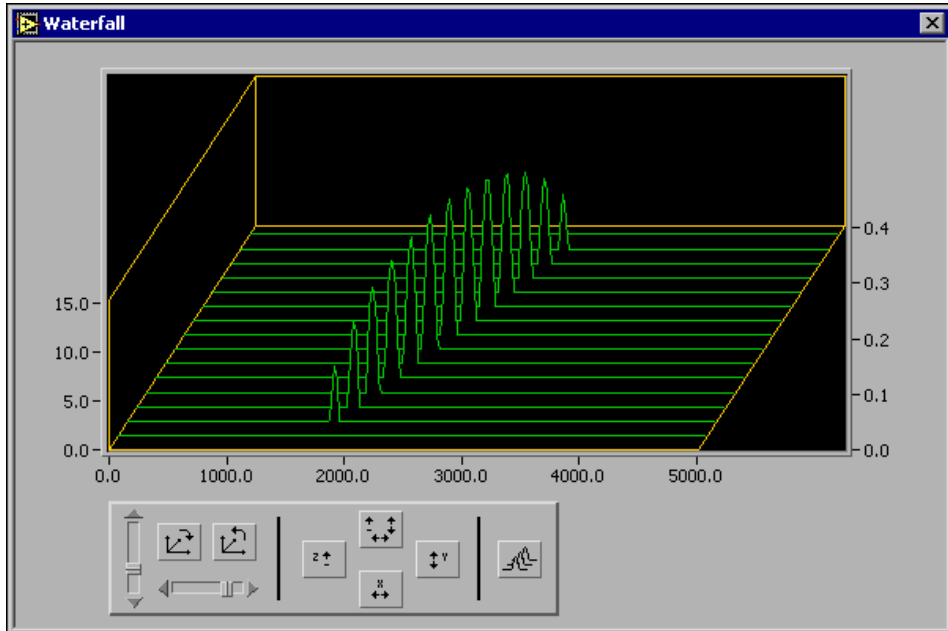


Figure 10-1. Waterfall Display for Frequency Analysis

Initializing the Display

Use the SVT Initialize Waterfall Display VI to create a reference to a waterfall display. If you are displaying octave spectra, use the SVT Initialize Waterfall Display for Octave VI. Both of the initializing VIs also enable you to define graph properties, including the window title, the bounds of the external window, and the colors used in the waterfall display. Refer to the *LabVIEW Sound and Vibration Toolset Help* for information about specifying the graph properties.

Sending Data to the Display

The **Waterfall** window does not open until it receives data sent to it. Use the SVT Send Data to Waterfall VI to send data to a waterfall display. Use the SVT Send Data to Waterfall for Octave VI to send octave data to a waterfall display.

The SVT Send Data to Waterfall VI is polymorphic and accepts an array of spectra, such as that returned by a power spectrum, or a 2D array, such as that returned by a short-time Fourier transform (STFT).

Waterfall Display for Frequency Analysis

The following example shows how to accumulate 20 spectra and display them in a waterfall graph. Figure 10-2 shows the block diagram for the VI.

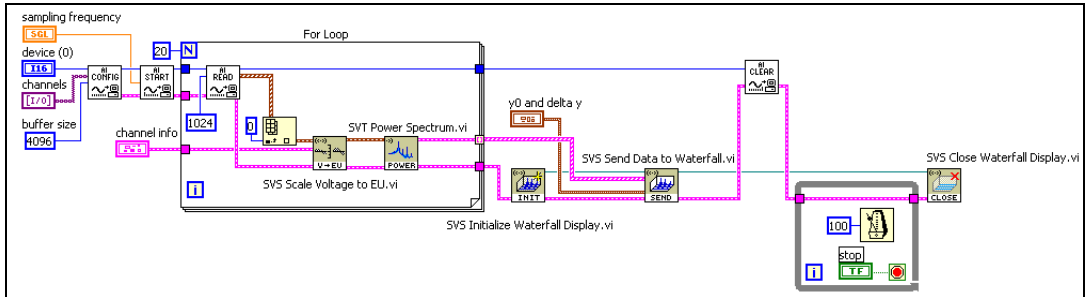


Figure 10-2. Waterfall Display for Frequency Analysis VI Block Diagram

Twenty data blocks of 1,024 points are acquired. The power spectrum is computed on each block. The auto-indexing capability of the For Loop is used to build an array of 20 spectra. From the array, the spectra are sent to the waterfall display. Refer to the *LabVIEW Help* for information about auto-indexing.

Waterfall Display for Transient Analysis

The following example illustrates how to use the waterfall display in conjunction with the Transient Analysis VIs. Figure 10-3 shows the block diagram for the example VI.

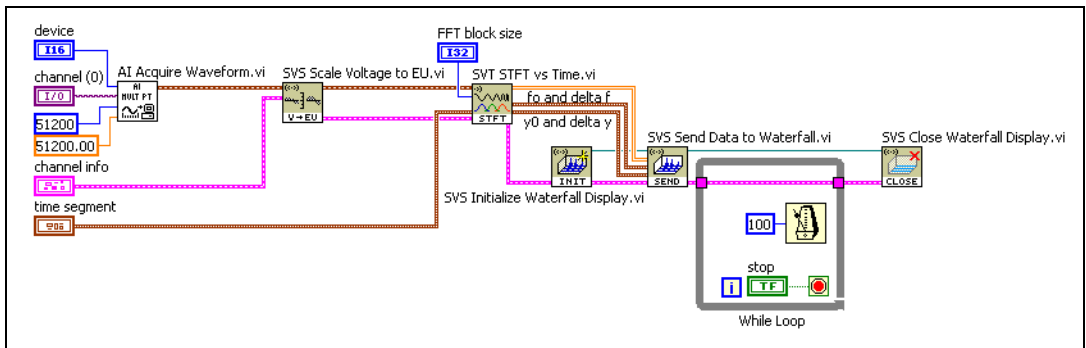


Figure 10-3. STFT VI Block Diagram

The Acquire Waveform VI, a high level data acquisition VI, captures 1 second of data. The data are scaled and sent to the SVT STFT vs Time VI. The SVT STFT vs Time VI returns a 2D array. You can use the results in the 2D array in an intensity graph or connect the 2D array directly to the SVT Send Data to Waterfall VI, as shown in Figure 10-3. The While Loop keeps the waterfall display open until the **Stop** control is set to TRUE.



Note Connect **f0 and delta f** and **y0 and delta y** on the SVS Send Data to Waterfall VI to ensure the graph shows the proper scales.

Figure 10-4 shows the result obtained with the STFT VI illustrated in Figure 10-3.

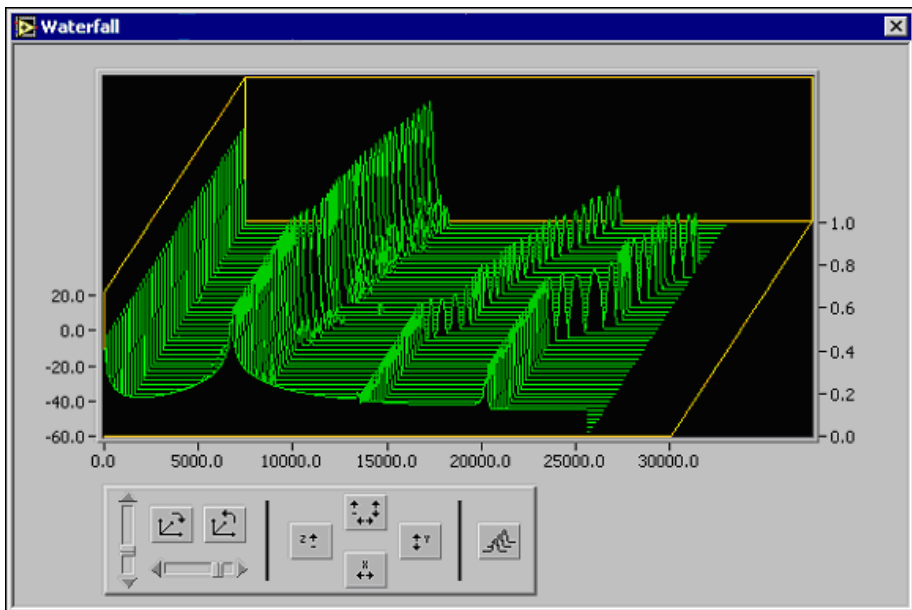


Figure 10-4. STFT Waterfall Display

Refer to Chapter 9, *Transient Analysis*, for information about transient analysis. Refer to the *LabVIEW Sound and Vibration Help* for information about the Transient Analysis VIs.

Waterfall Display for Octave Spectra

To display octave spectra in a waterfall display, use the SVT Initialize Waterfall Display for Octave and SVT Send Data to Waterfall for Octave VIs. Figure 10-5 shows the block diagram for a VI displaying octave spectra in a waterfall display.

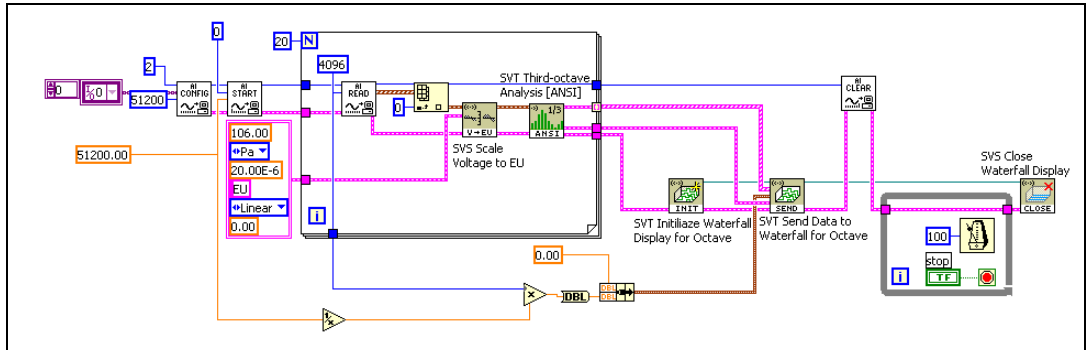


Figure 10-5. Block Diagram for VI Displaying Octave Spectra

Figure 10-6 shows the waterfall display created by the VI shown in Figure 10-5.

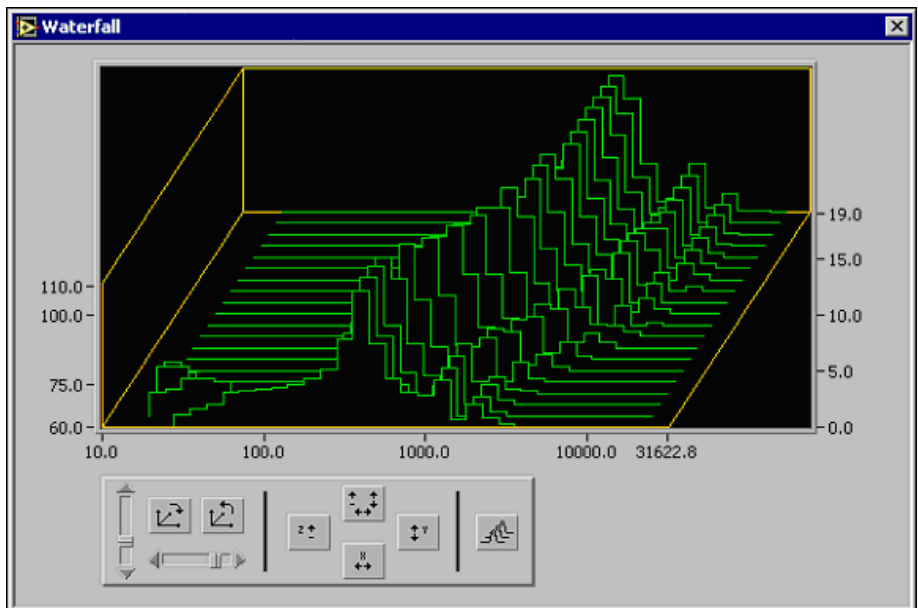


Figure 10-6. Waterfall Display for Octave Spectra Analysis

Customizing the Waterfall Display View

Use the controls shown in Figure 10-7 to customize the view of your waterfall display.

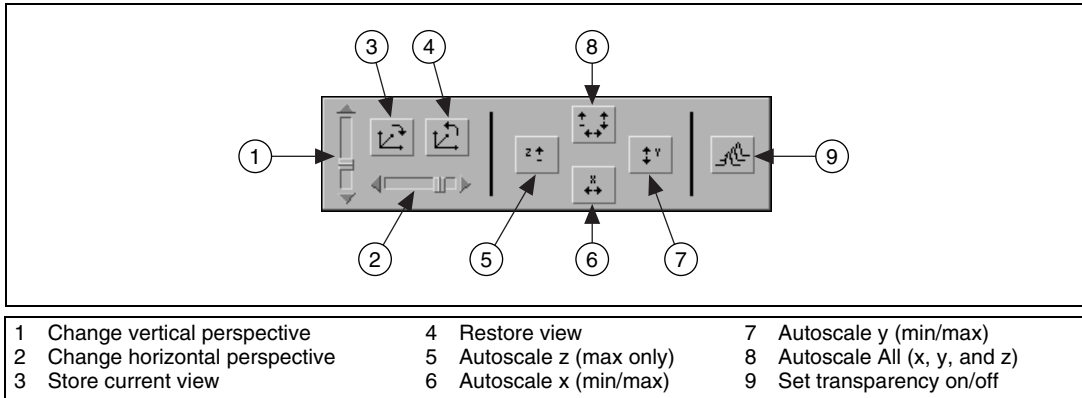


Figure 10-7. Waterfall Graph Controls

You can use the two sliders to change the perspective of the display. You can store and recall any view at a later time and turn transparency on or off. You also can autoscale each axis independently or all three axes simultaneously.

Autoscaling the z-axis differs from autoscaling the x and y axes. Autoscaling the z-axis affects only the maximum value of the z-axis. You must select the minimum value of the z-axis separately because the minimum value of the z-axis defines the baseline of the waterfall graph. You can set the minimum value of the z-axis by double-clicking the z-axis or by specifying the minimum value when you send data to the waterfall display.

Closing the Waterfall Display

The SVT Close Waterfall Display VI, shown in Figures 10-2, 10-3, and 10-5, closes the reference to the waterfall graph and the **Waterfall** window.

References

This appendix lists the reference material used for the Sound and Vibration Toolset for LabVIEW. Refer to the following documents for more information about the theory implemented in this toolset.

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Glossary

Numbers/Symbols

2D	Two-dimensional.
3D	Three-dimensional.
μPa	Micro-Pascal. <i>See also</i> Pa .

A

accelerometer	A sensor mounted on a structure to measure the acceleration at a particular location in one or multiple directions.
aliasing	Aliasing is a phenomenon whereby an analog signal of frequency greater than half the sampling frequency (Nyquist frequency) appears, after sampling, at a frequency less than half the sampling frequency. <i>See also</i> anti-aliasing filter.
ANSI	American National Standards Institute.
anti-aliasing filter	To avoid aliasing, analog lowpass filters are used before A/D conversion to filter out the frequencies greater than half the sampling frequency. Since they are used to prevent aliasing, these analog lowpass filters are known as anti-aliasing filters. <i>See also</i> Nyquist frequency .

B

bandedge frequency	The upper and lower cutoff frequencies of an ideal bandpass filter.
bandpass filter	A filter with a single transmission band extending from a lower bandedge frequency greater than zero to a finite upper bandedge frequency.

C

calibrator	A controlled source generating a known level of excitation used to calibrate a sensor.
------------	--

coherence	Gives a measure of the degree of linear dependence between two signals, as a function of frequency.
coherent output power spectrum	The coherent output power spectrum gives a measure of what part of the (output) power spectrum is fully coherent with the input signal.
colormap	A method of displaying 3-dimensional data on a 2-dimensional plot using color to indicate the value in the third dimension.
crest factor	The ratio of the peak value of a signal to its RMS value. For a sine wave, the crest factor is 1.414. For a square wave, the crest factor is 1.
cross power spectrum	The cross power spectrum of two signals has an amplitude that is the product of the two amplitudes, and a phase that is the difference of the two phases.

D

DAQ	Data acquisition.
dB	Decibels. A logarithmic unit for measuring ratios of levels. If the levels are specified in terms of power, then $1 \text{ dB} = 10 \cdot \log_{10}(P/P_r)$ where P is the measured power and P_r is the reference power. If the levels are specified in terms of amplitude, then $1 \text{ dB} = 20 \cdot \log_{10}(A/A_r)$ where A is the measured amplitude and A_r is the reference amplitude.
DFT	Discrete Fourier Transform—determines the amplitude and phase of frequency components present in a time-domain digital signal.
DSA	Dynamic signal acquisition.
DUT	Device under test.

E

ENBW	Equivalent noise bandwidth.
engineering unit (EU)	Term of data measurement, such as Pa, m/s^2 , g, and so on.

equal confidence	Special exponential averaging mode used for fractional-octave analysis. For equal confidence the time constant for each band is set individually so that the relative confidence in the measurement is equal across all the bands. There is a 68% probability that the results will be within \pm the specified confidence level of the true mean value.
equivalent continuous level (L_{eq})	The energy average level of a signal over a given time interval.
EU	<i>See</i> engineering unit (EU) .
exponential averaging	Time-averaging technique that gives recent data more importance than older data.
F	
Fast	Exponential averaging using a time constant of 125 ms.
FFT	Fast Fourier Transform—an efficient and fast method for calculating the Discrete Fourier Transform. The Fast Fourier Transform determines the amplitude and phase of frequency components present in a time domain digital signal.
FFT block size	The number of samples used to compute an FFT.
FFT lines	The number of FFT lines is related to the FFT block size. Theoretically the number of lines is half of the block size, but it is practically reduced to 80% of that value due to the anti-aliasing filter. For example, a 400 line FFT is based on a block size of 1,024 points.
filter bank	A group of filters.
filtering	A type of signal conditioning that allows you to modify the frequency content of a signal.
fractional-octave	The interval between two frequencies, one of which is a fractional power of two of the other.
frequency response function	Represents the ratio of output-to-input in the frequency domain, and fully characterizes linear, time-invariant systems.
FRF	Frequency response function. <i>See</i> frequency response function.

G

- g Unit for measuring acceleration. One $g = 9.81 \text{ m/s}^2$, the acceleration due to gravity at the surface of the Earth.
- gain The amplification or attenuation of a signal.

H

- H1 The frequency response function computed as the ratio of the cross spectrum to the input autospectrum: G_{xy}/G_{xx} . This technique gives the best performance in the presence of noise for measuring anti-resonances, where the signal to noise ratio tends to be poor. For measurement of resonances, the frequency response function H2, gives a better estimate. In a noise free environment, both techniques give the same result. Since both measurements are based on the same data set, the choice of technique can be made after the data acquisition is completed.
- H2 The frequency response function computed as the ratio of the output autospectrum to the backwards cross spectrum: G_{yy}/G_{yx} . This technique gives the best performance in the presence of noise for measuring resonances, where the signal to noise ratio tends to be best. For measurement of anti-resonances, the frequency response function H1, gives a better estimate. In a noise free environment, both techniques give the same result. Since both measurements are based on the same data set, the choice of technique can be made after the data acquisition is completed.
- H3 The frequency response function computed as an average of H1 and H2.
- Hz Hertz. Cycles per second.

I

- IEC International Electrotechnical Commission.
- Impulse Exponential averaging using a time constant of 35 ms if the signal level is rising and 1,500 ms if the signal level is falling.
- intensity graph A method of displaying three dimensions of data on a 2D plot with the use of color to indicate the value in the third dimension.

L

- L_{eq} See [equivalent continuous level \(Leq\)](#).
- linear averaging Time-averaging technique that weights all data in the average equally.

M

- MAX Measurement & Automation Explorer
- microphone Sensor used to convert sound pressure variations into an electrical signal, usually when the acoustic medium is air.
- midband frequency The center frequency of a bandpass filter, defined as the geometric mean of the bandedge frequencies.
- ms Millisecond.

N

- noise Any unwanted signal. Noise can be generated by internal sources such as semiconductors, resistors, and capacitors, or from external sources such as the AC power line, motors, generators, thunderstorms, and radio transmitters.
- nominal frequency Rounded midband frequency for the designation of a particular fractional-octave filter. This term is used by the IEC standards, but nominal frequencies are identical to the preferred frequencies defined in the ANSI standards.
- nonstationary signal Signal whose frequency content changes within a captured frame.
- Nyquist frequency Half the sampling frequency. Any analog frequency component above the Nyquist frequency will, after sampling, be converted (aliased) to a frequency below the Nyquist frequency. See also [aliasing](#) and [anti-aliasing filter](#).

O

octave	Refers to the interval between two frequencies, one of which is twice the other. For example, frequencies of 250 Hz and 500 Hz are one octave apart, as are frequencies of 1 kHz and 2 kHz.
order	Harmonic of the rotational speed of rotating machinery. As an example, in the case of the shaft rotating at 6000 rpm, the first order component occurs at a frequency of 100 Hz (6000/60), whereas the third order component would occur at a frequency of 300 Hz.
overlapping	A method that uses a portion of the previous data block to compute the FFT of the current data block.

P

Pa	Pascal. International unit of pressure.
peak hold	Peak detection process retaining the maximum value of a signal.
phon	The unit of loudness on a scale corresponding to the decibel scale of sound pressure level with the number of phons of a given sound being equal to the decibels of a pure 1 kHz sine tone judged by the average listener to be equal in loudness to the given sound.
pink noise	Noise for which the spectral energy per octave or any fractional-octave band is independent of the band. The spectrum looks flat on an octave or fractional-octave band display.
pistonphone	Microphone calibrator generating a known sound pressure level, typically at a certain reference frequency.
preferred frequency	Rounded midband frequency for the designation of a particular fractional-octave filter. This term is used by the ANSI standards, but preferred frequencies are identical to the nominal frequencies defined in the IEC standards.
pregain	Any gain applied to a signal by an external device (amplifier, preamplifier, signal conditioning, and so on) before the data acquisition board.
PSD	Power spectral density.

R

reference sound pressure	A reference pressure of $20\text{E-}6$ Pa. This reference pressure was conventionally chosen to correspond to the quietest sound at 1,000 Hz that the human ear can detect.
reverberation time	$T60[s]$. At a point in an enclosure and for a stated frequency or frequency band, the time required for the pressure level to decrease by 60 dB after the source has been stopped.
RMS	Root Mean Square.
RMS averaging	RMS averaging is used to average the energy or power of a signal. RMS averaging reduces fluctuations.
rpm	Revolutions per minute.

S

s	Seconds.
S/s	Samples/second.
sampling frequency	The rate at which a continuous waveform is digitized.
sensor	A device that converts a physical stimulus (such as force, sound, pressure, motion) into a corresponding electrical signal.
settling time	The amount of time required for a signal to reach its final value within specified limits.
Shannon Sampling Theorem	States that to properly sample a signal, the signal must not contain frequencies above the Nyquist frequency.
Slow	Exponential averaging using a time constant of 1,000 ms.
sound level meter	A device used to measure sound pressure levels. Sound level meters usually consist of a microphone, a preamplifier, a set of standardized frequency weighting filters, standardized exponential time weighting circuits, a logarithmic amplitude detector, and a display in decibels.
sound pressure level	In decibels, 20 times the base 10 logarithm of the ratio of the sound pressure, in a stated frequency band, to the reference sound pressure.

spectral leakage A phenomenon whereby the measured spectral energy appears to leak from one frequency into other frequencies. It occurs when a sampled waveform does not contain an integral number of cycles over the time period during which it was sampled. The technique used to reduce spectral leakage is to multiply the time-domain waveform by a window function. *See also* [windowing](#).

STFT Short-Time Fourier Transform.

SVS Sound and Vibration Suite. SVS is a collection of VIs shared by the Sound and Vibration Toolset and other National Instruments Toolsets.

SVT Sound and Vibration Toolset.

T

tach *See* tachometer.

tachometer Device used to measure the rotational speed of a rotating part.

third-octave Ratio between two frequencies, equal to $2^{1/3}$.

time constant A standardized time constant used in exponential time weighting for acoustical analysis. The standard time constants are Slow = 1,000 ms, Fast = 125 ms, and Impulse = 35 ms while the signal level is increasing or 1500 ms while the signal level is decreasing.

transducer *See* [sensor](#).

transient A very short-duration signal. Normally only occurs once, or very infrequently.

TTL Transition-transistor logic—a typical medium speed digital technology. Nominal TTL logic levels are 0 and 5 V.

V

V Volts.

vector averaging Computes the average of complex quantities directly, that is, the real and imaginary parts are averaged separately. Vector averaging eliminates noise from synchronous signals and usually requires a trigger.

W

waterfall	A 3-dimensional plot displaying the amplitude of spectral components as a function of both time and frequency. The frequency spectrum is displayed as a curve for each specified time instant. Several such curves (for different time instants) are displayed simultaneously.
weighting filter	Filter used to reproduce the varying sensitivity of the human ear to sound at different frequencies. Originally, A-weighting was intended to represent the varying sensitivity of the ear to sound pressure levels ranging between 40 and 60 dB ref 20 μ Pa. Subsequently, B-weighting and C-weighting were developed to represent the varying sensitivity of the ear over higher sound pressure level ranges.
white noise	Noise that has the same power spectral density at all frequencies. As an example, the average power of white noise in a 100 Hz bandwidth between 300 Hz and 400 Hz, is the same as the average power of white noise in the 100 Hz bandwidth between 10,000 Hz and 10,100 Hz.
window function	A smooth waveform that generally has zero value at the edges. <i>See also</i> windowing.
windowing	Technique used to reduce spectral leakage by multiplying the time-domain waveform by a window function. The process of windowing reduces the amplitudes of discontinuities at the edges of a waveform, thereby reducing spectral leakage. If the waveform contains an integral number of cycles, there is no spectral leakage. <i>See also</i> spectral leakage .

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