LabVIEW™

Sound and Vibration Toolkit
User Manual

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About This Manual

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

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.
monospace bold	Bold text in this font denotes the messages and responses that the computer automatically prints to the screen. This font also emphasizes lines of code that are different from the other examples.
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 Help*, available by selecting **Help»VI, Function, & How-To Help** in LabVIEW

- *Getting Started with LabVIEW*

- *LabVIEW User Manual*

- *LabVIEW Measurements Manual*

1

Introduction

This chapter introduces the Sound and Vibration Toolkit, the toolkit functions and controls palettes, and where to find examples to help you get started.

Sound and Vibration Toolkit

The Sound and Vibration Toolkit is a collection of virtual instruments (VIs) for LabVIEW that you can use to perform typical measurements required by audio, acoustics, or vibration applications. Use the Sound and Vibration Toolkit 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 swept-sine measurements
- Performing single-tone measurements
- Performing limit and mask testing
- Performing fractional-octave analysis
- Performing frequency analysis
- Performing transient analysis
- Performing distortion analysis
- Displaying results

You can use the Sound and Vibration Toolkit 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.
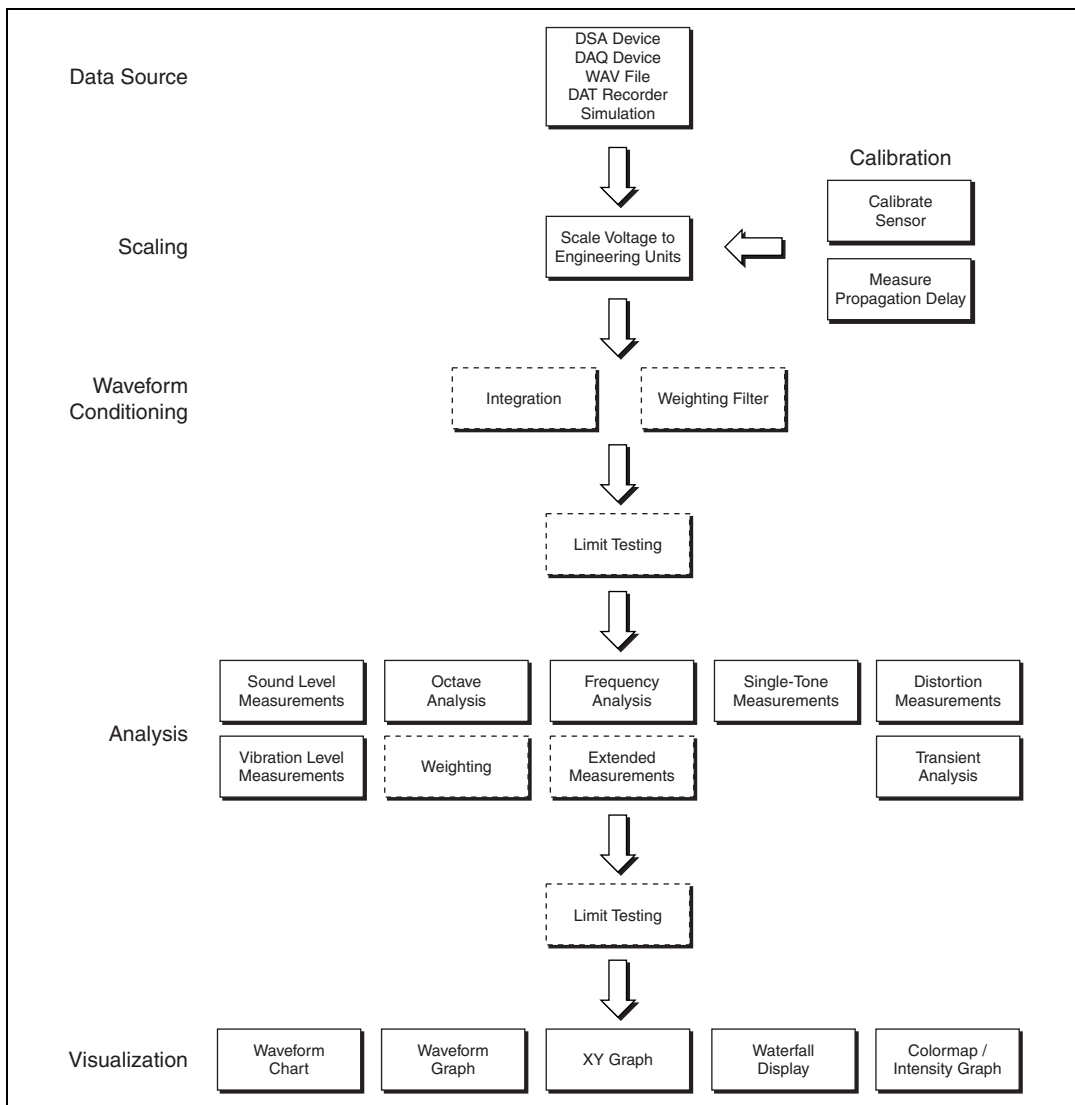
Figure 1-1. Sound and Vibration Toolkit Overview

Toolkit Palettes

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

All the high-level VIs in the Sound and Vibration Toolkit 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 SVL Scale Voltage to EU VI allows you to scale the original signal to engineering units. The SVL Scale Voltage to EU VI is part of the Sound and Vibration Library (SVL). The SVL is a collection of VIs shared by the Sound and Vibration Toolkit and other National Instruments (NI) toolkits. Refer to Chapter 3, *Scaling and Calibration*, and to the *LabVIEW Help* for more information about the SVL Scale Voltage to EU VI.

Calibration

The Calibration VIs allow you to perform an end-to-end calibration on a selected channel and measure the propagation delay of the measurement device. The Calibration VIs are part of the SVL. Refer to Chapter 3, *Scaling and Calibration*, for information about the calibration process. Refer to the *LabVIEW Help* for information about the individual Calibration VIs.

Limit Testing

The SVT Limit Testing VI allows you to apply limit analysis to any type of measured result produced by the Sound and Vibration Toolkit. Refer to Chapter 4, *Limit Testing Analysis*, for more information about using limit testing to analyze measurement results. Refer to the *LabVIEW Help* for information about the SVT Limit Testing VI.

Weighting

The Weighting VIs allow you to apply A-, B-, or C-weighting filters on the time-domain signal. Additionally, ITU-R 468-4 and Dolby filters are available for radiocommunications applications, and C-message and CCITT filters are available for telecommunications applications. Refer to Chapter 5, *Weighting Filters*, for more information about applying weighting to a signal. Refer to the *LabVIEW Help* for information about the individual Weighting VIs.

Integration

The SVT Integration VI allows you to perform single or double integration on the time-domain signal. Refer to Chapter 6, *Integration*, for information about the integration process. Refer to the *LabVIEW Help* for information about the SVT Integration VI.

Generation

The SVT Pink Noise Waveform VI allows you to generate a continuous pink noise waveform. The **Generation** palette also contains a subpalette linked to the standard **Waveform Generation** palette in LabVIEW. Refer to the *LabVIEW Help* for information about the Generation VIs.

Vibration Level

The Vibration Level VIs 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 7, *Vibration-Level Measurements*, for more information about performing vibration level measurements. Refer to the *LabVIEW Help* for information about the individual Vibration Level VIs.

Sound Level

The Sound Level VIs offer typical sound-level measurements, including equivalent continuous averaging (L_{eq}), exponential averaging, and peak hold. Refer to Chapter 8, *Sound-Level Measurements*, for information about performing sound-level measurements. Refer to the *LabVIEW Help* for information about the individual Sound Level VIs.

Octave Analysis

The Octave Analysis VIs 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 9, *Fractional-Octave Analysis*, for information about performing octave analyses. Refer to the *LabVIEW Help* for information about the individual Octave Analysis VIs.

Frequency Analysis

The Frequency Analysis VIs 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 10, *Frequency Analysis*, for information about performing frequency analyses. Refer to the *LabVIEW Help* for information about the individual Frequency Analysis VIs.

Transient Analysis

The Transient Analysis VIs offer two techniques for obtaining information about transient signals. Use the short-time Fourier transform (STFT) to extract frequency information as a function of time or rotational speed. Use the shock response spectrum (SRS) to evaluate the severity of a shock signal. Refer to Chapter 11, *Transient Analysis*, for information about performing transient analyses. Refer to the *LabVIEW Help* for information about the individual Transient Analysis VIs.

Waterfall Display

The Waterfall Display VIs 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 12, *Waterfall Display*, for information about displaying analysis results in a waterfall graph. Refer to the *LabVIEW Help* for information about the individual Waterfall Display VIs.

Swept Sine

The Swept-Sine VIs allow you to characterize the frequency response of a device under test (DUT). The swept-sine measurements include dynamic measurements for stimulus level, response level, frequency response (gain and phase), total harmonic distortion (THD), and individual harmonic distortion. Refer to Chapter 13, *Swept-Sine Measurements*, for information about performing swept-sine measurements. Refer to the *LabVIEW Help* for information about the individual Swept-Sine VIs.

Distortion

The Distortion VIs allow you to measure the harmonic, intermodulation, and broadband noise components due to nonlinearities in the DUT. Refer to Chapter 14, *Distortion Measurements*, for information about performing distortion analyses. Refer to the *LabVIEW Help* for information about the individual Distortion VIs.

Single-Tone

The Single-Tone VIs allow you to perform single-tone measurements, defined as a group of measurements where the excitation is a single tone. These measurements are often used to measure the linear response, nonlinear distortion, and noise of audio devices. Refer to Chapter 15, *Single-Tone Measurements*, for information about performing single-tone measurements. Refer to the *LabVIEW Help* for information about the individual Single-Tone VIs.

Front Panel Displays

The customized graphs located on the **Sound & Vibration** controls 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 Toolkit Help* for information about the individual displays.

Examples

The Sound and Vibration Toolkit includes examples to help you get started using the toolkit. Select **Help»Find Examples** in LabVIEW to launch the NI Example Finder. Select **Toolkits and Modules»Sound and Vibration** in the **Browse** tab to view all of the available examples, or use the **Search** tab to locate a specific example. The examples demonstrate the following Sound and Vibration Toolkit capabilities:

- Display
- Frequency analysis
- Integration
- Level measurements
- Octave analysis
- Scaling
- Transient analysis
- Weighting filters
- Swept-Sine measurements
- Audio measurements
- Limit testing

2

Dynamic Signals

This chapter discusses how to obtain data to analyze with the Sound and Vibration Toolkit, as well as issues that can affect the quality of your data. You can simulate data with the generation VIs located on the Generation palette as well as with other VIs.

Refer to the *LabVIEW Help* for more information about acquiring and simulating data.

Acquiring and Simulating Dynamic Signals

This section discusses obtaining data and some key issues when acquiring or simulating dynamic signals to ensure valid measurement results. The three techniques that allow you to obtain data are as follows:

• Acquire data with a data acquisition (DAQ) device system

• Read data from a file

• Simulate data with a generation VI or other source

Figure 2-1 illustrates how the data source, either acquired or simulated, fits into the sound and vibration measurement process.
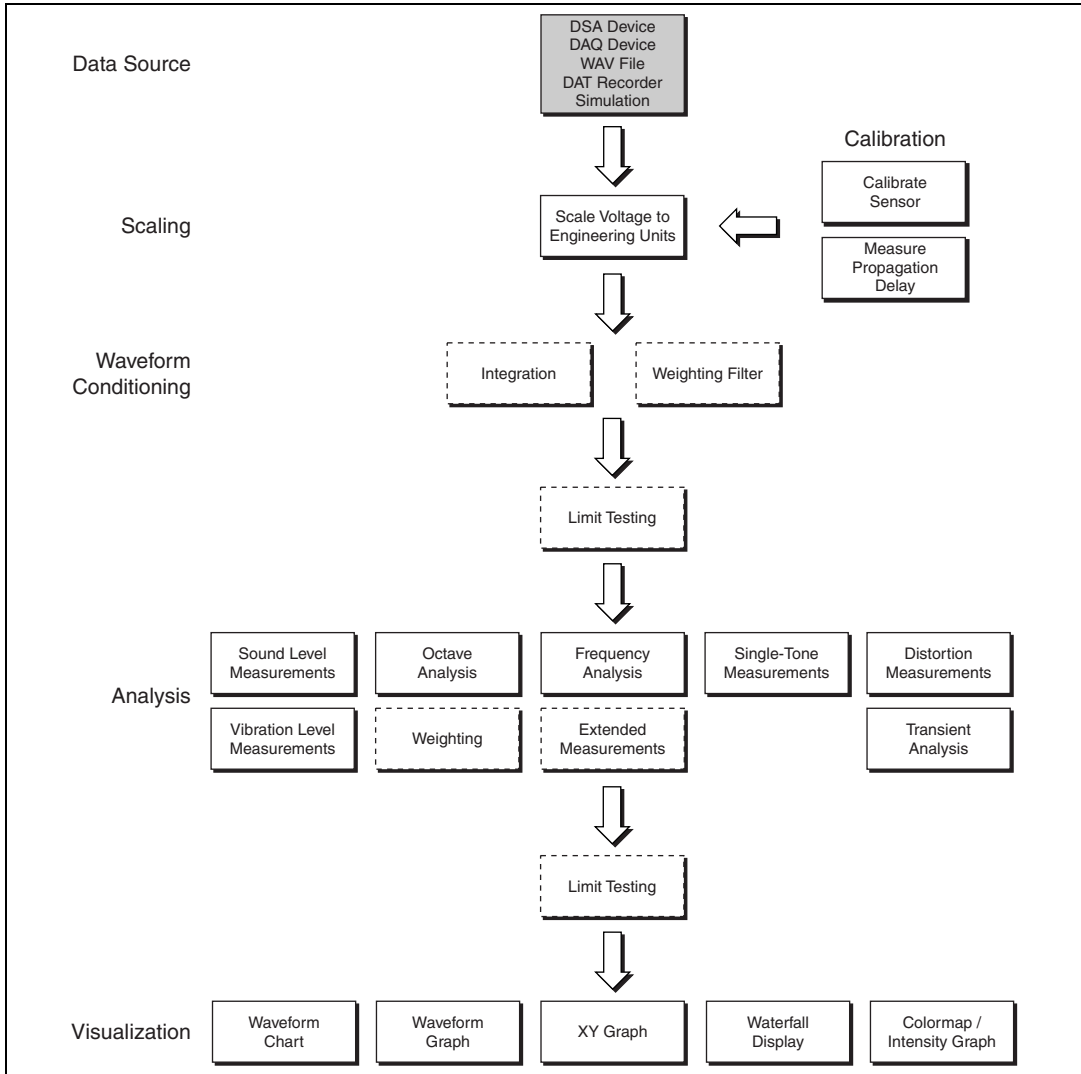
Figure 2-1. Relationship of the Data Source to the Sound and
Vibration Measurement Process

It is important that you keep certain considerations in mind when you
obtain your data. The measurement and analysis VIs in the Sound and
Vibration Toolkit do not compensate for inaccurate data. Therefore, the test
equipment and test procedure should be calibrated to ensure accurate
results. Generally, the test equipment should have specifications at least
10 times better than those of the DUT. Use a verifiable and repeatable test
procedure to get accurate results.

Whether you are obtaining the data from a DAQ system, reading the data from a file, or simulating the data, aliasing and time continuity are common issues which you should consider in your measurement analysis.

Aliasing

When a dynamic signal is discretely sampled, aliasing is the phenomenon in which frequency components greater than the Nyquist frequency are erroneously shifted to lower frequencies. The Nyquist frequency is calculated with the following formula:

$$f_{\text{Nyquist}} = sample\ rate/2$$

When acquiring data with an NI Dynamic Signal Acquisition (DSA) device, aliasing protection is automatic in any acquisition. The sharp anti-aliasing filters on DSA devices track the sample rate and filter out (attenuate) all frequencies above the Nyquist frequency.

When performing frequency measurements with an NI E Series DAQ device, you must take steps to eliminate aliasing. These anti-aliasing steps can include the following actions:

- Increasing the sample rate
- Applying an external lowpass filter
- Using an inherently bandlimited DUT

Simulated data also can exhibit aliasing. The signals often are generated according to a time-domain expression and, therefore, have high-frequency components that are aliased in the discretely sampled data. Figure 2-2 shows an example of this aliasing for a simulated square wave.
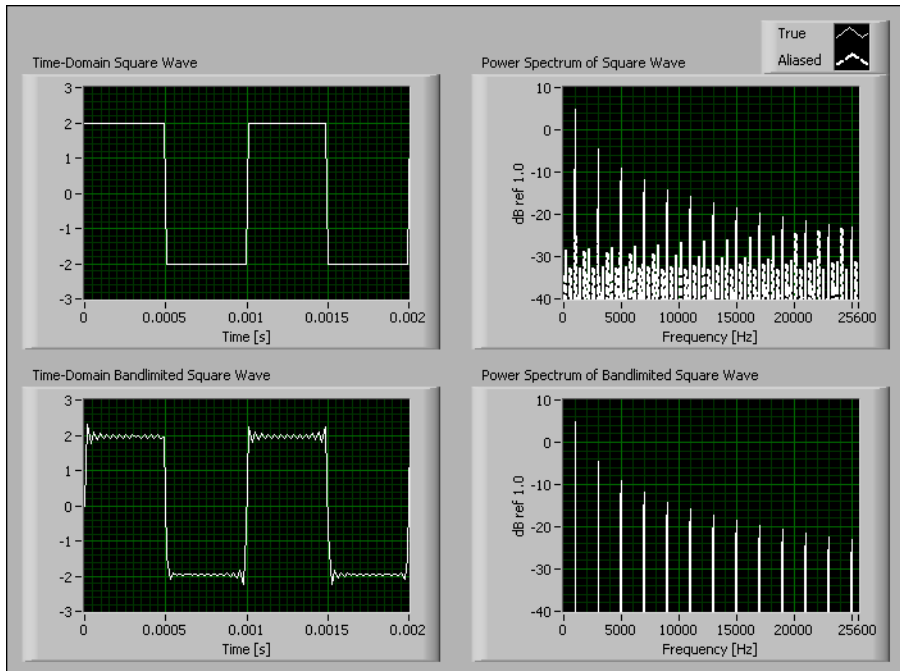
Figure 2-2. Simulated Data Aliasing

The only way to protect data from aliasing is to apply appropriate aliasing protection before the data are generated or acquired. Aliasing occurs when the data are generated or sampled, and it is not possible to remove aliased components from the data without detailed knowledge of the original signal. In general, it is not possible to distinguish between true frequency components and aliased frequency components. Therefore, accurate frequency measurements require adequate alias protection.

Time Continuity

When you acquire data in a continuous acquisition, you can use the **t0** parameter in the waveform datatype to ensure there are no gaps between successive blocks of waveforms returned by sequential calls to the DAQmx Read VI or AI Read VI. When signals are generated with one of the waveform generation VIs in the Generation palette or the Waveform Generation palette, the **t0** of the current waveform is one sample period later than the timestamp of the last sample in the previous waveform. Continuity is enforced in this way until the generation is reset.

The waveform datatype is integral for testing time continuity in the Sound and Vibration Toolkit. If you read data from a file or simulate a signal using one of the VIs in the Signal Generation palette, wire a **t0** that meets the continuous timestamp condition to the waveform datatype connected to the measurement analysis VIs. This action prevents unexpected resets of the measurement analysis due to detected discontinuities in the input signal.

3

Scaling and Calibration

This chapter discusses using the SVL Scale Voltage to EU VI located on the **Scaling** palette to scale a signal to engineering units (EU) and using the Calibration VIs located on the **Calibration** palette.

Refer to the *LabVIEW Help* for more information about the SVL Scale Voltage to EU VI and the Calibration VIs.

Scaling to EU

This section discusses scaling data to the appropriate EU so you can perform measurement analysis.

Figure 3-1 illustrates how scaling and calibration fit into the sound and vibration measurement process.

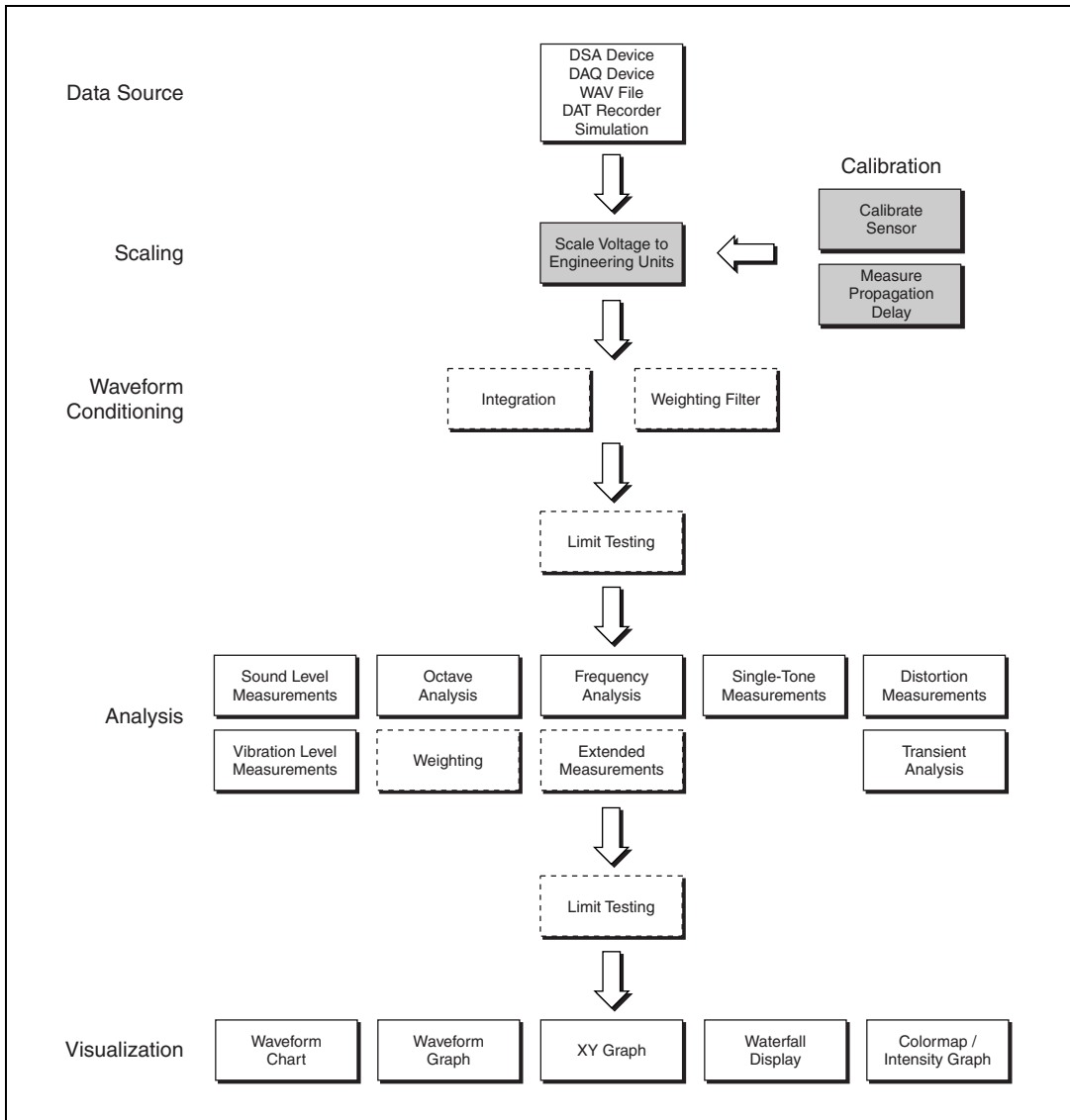**Figure 3-1.** Relationship of Scaling to the Sound and Vibration Measurement Process

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

All measurement VIs in the Sound and Vibration Toolkit expect input signals and return results with the appropriate units, such as time-domain

signals in the correct EU, frequency spectra in decibels with the proper reference, 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 EU.

Note If you use any method outside of the Sound and Vibration Toolkit to apply scaling to a waveform, do *not* use the SVL Scale Voltage to EU VI. NI provides several tools and methods to apply scaling to a waveform. These include, but are not limited to, NI-DAQmx tasks or global channels created with Measurement & Automation Explorer (MAX), the DAQ Assistant, or the DAQmx Create Virtual Channel VI.

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 SVL Calibrate Microphone VI. If you are calibrating an accelerometer, consider using the SVL Calibrate Accelerometer VI. These VIs are very similar to the general-purpose SVL Calibrate Sensor VI, but they 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.

Propagation Delay Calibration

The Sound and Vibration Toolkit provides VIs for calibrating the propagation delay of the measurement system. National Instruments DSA devices like the NI PXI-4461 and NI PCI-4451 can acquire and generate signals on the same device. The input and output channels have analog and digital circuitry, such as anti-aliasing and anti-imaging filters, that introduce a certain delay to the signal. The propagation delay is the number of samples from the time a sample is first written to the output channel to when that sample is digitized on the input channel, assuming there is no delay from the output channel to the input channel. This delay varies by DSA device.

There are two ways to determine the propagation delay of the DSA device. You can refer to the documentation for the DSA device to find the propagation delay specifications, also referred to as group delay. You also can measure the propagation delay in samples with the SVL Measure Propagation Delay VIs. The SVL Measure Propagation Delay VIs allow you to measure the delay introduced in the input and output circuitry for a

specific device at the desired sample rate. Connect the DSA device output channel directly to the input channel, as displayed in Figure 3-2, to measure the device propagation delay.
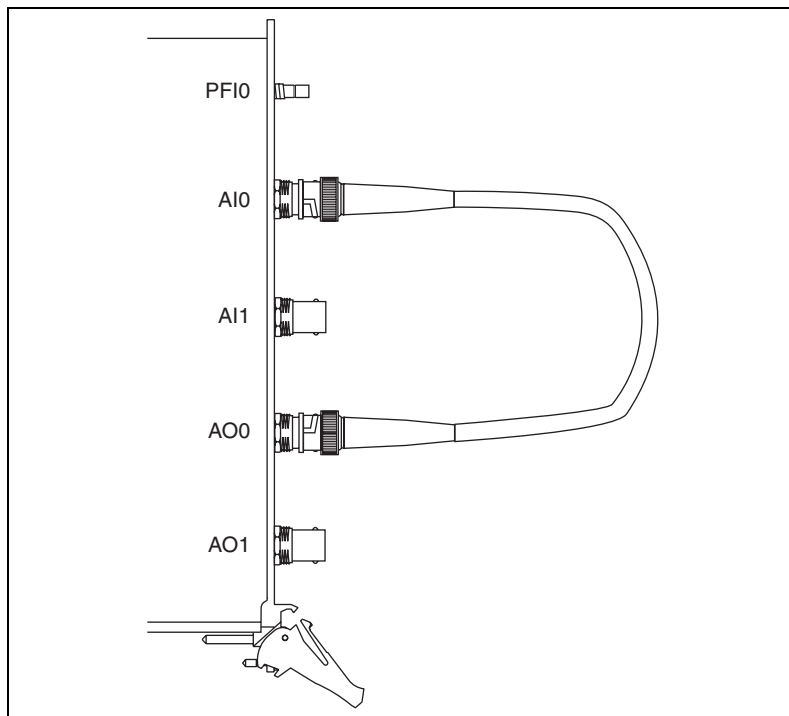
Figure 3-2. Measuring the Device Propagation Delay

✎ **Note**   Do *not* put a DUT in the signal path when measuring the propagation delay for the DAQ device.

For an E or S Series DAQ device from NI, you should expect to measure a one-sample propagation delay due to the time required for the signal to traverse the signal path between the D/A converter (DAC) on the analog output channel and the A/D converter (ADC) on the analog input channel. Figure 3-3 shows the time domain data for the propagation delay measurement of an NI PCI-6052E.
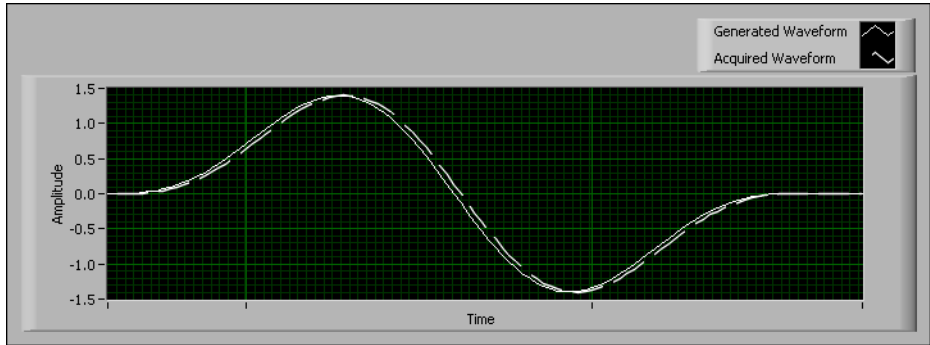
Figure 3-3. Propagation Delay Measurement of an NI PCI-6052E

For DSA devices, or any other device which has onboard filtering on either the input, output, or both channels, you should expect to measure a propagation delay consistent with the sum of the delays specified for the onboard filters on the input and output channels. Figure 3-4 shows the delay of a smooth pulse generated and acquired by an NI PXI-4461 with a 204.8 kHz sample rate.

Figure 3-4. NI PXI-4461 Propagation Delay with a 204.8 kHz Sample Rate

Not all DSA devices have a constant propagation delay across the entire range of supported sample rates. For example, the NI PXI-4461 propagation delay is dependent on the output update rate. Figure 3-5 shows the total propagation delay versus sample rate relationship for the NI PXI-4461 from output to input as a function of the sample rate.

Figure 3-5. NI PXI-4461 Propagation Delay versus Sample Rate

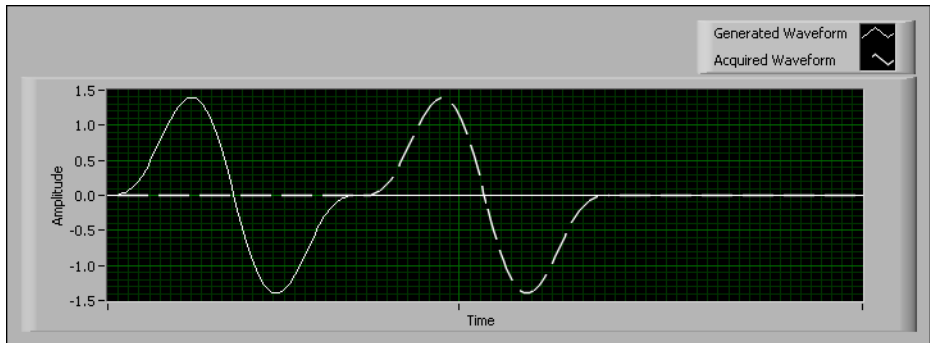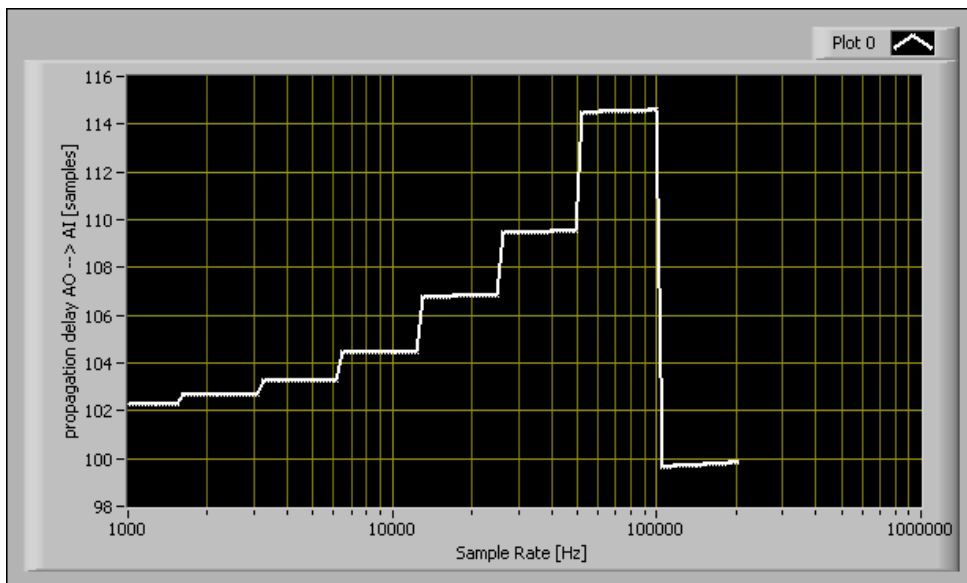As illustrated by Figures 3-3, 3-4, and 3-5, the propagation delay can vary significantly with different sample rates and devices. To ensure measurement accuracy in your I/O applications, determine and account for the propagation delay of the DAQ device at the same sample rate used in your application.

It is important to remove the effects of the delay due to the data acquisition system for two reasons. First, there is always a delay between the generated output signal and the acquired input on the device even when the output and input channels are hardware synchronized. Second, the anti-imaging and anti-aliasing filters of the device introduce additional delays. You must account for this delay to perform accurate dynamic measurements. Use the **device propagation delay [samples]** input on the examples found in the LabVIEW program directory under \examples\Sound and Vibration\Audio Measurements\ to remove the delay due to the DAQ device.

The anti-imaging and anti-aliasing filters have a lowpass filter effect on the data. This effect results in a transient response at sharp transitions in the data. These transitions are common at the start and stop of a generation, at a change in frequency (swept sine), and when the amplitude changes (amplitude sweep). The swept-sine analysis and audio measurements

examples in the Sound and Vibration Toolkit account for this transient behavior in the device response to achieve the highest degree of accuracy.

The propagation delay of the DUT is also an important specification in some applications. For example, the propagation delay for the DUT is a required input when performing audio measurements and when measuring the frequency response using swept sine. If the DUT and the propagation medium can successfully pass the pulse signal used by the SVL Measure Propagation Delay VIs without excessive attenuation, then this measurement also applies when measuring the propagation delay of the DUT and the propagation medium. Figure 3-6 shows the wiring diagram for this configuration.
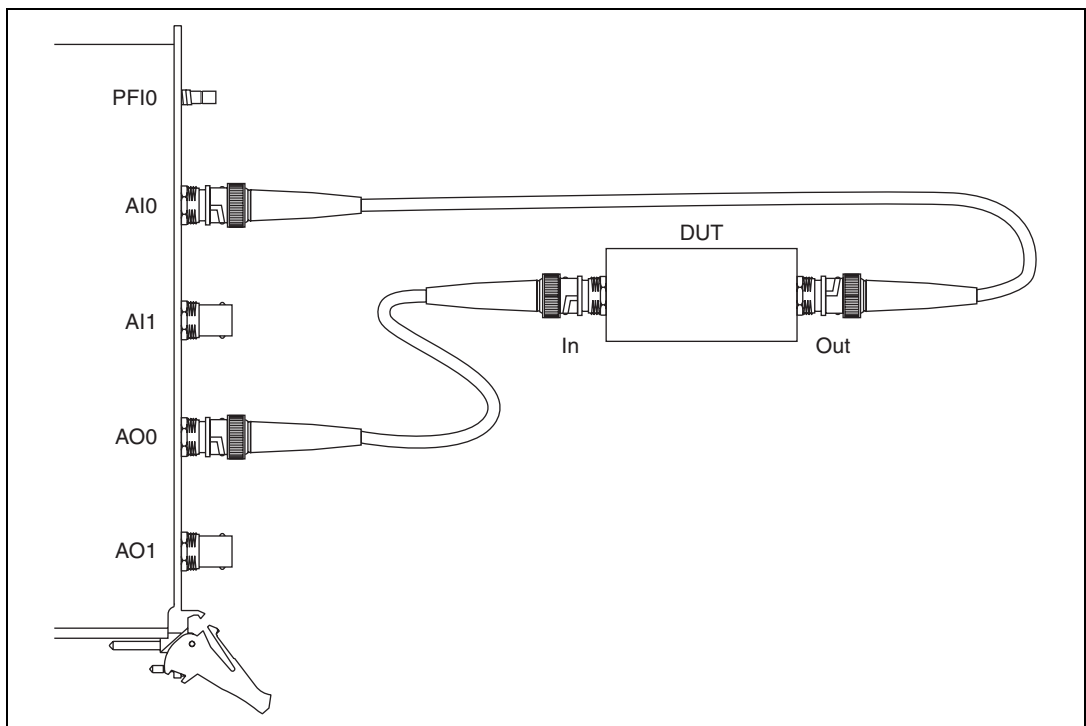
Figure 3-6. Measuring the DUT Propagation Delay

The DUT propagation delay is the delay of the entire system minus the device delay. Remember to measure the device delay without the DUT connected.

The propagation delay for an analog DUT is a constant time delay rather than a delay of samples. Use the following equation to convert the measured delay in samples to the equivalent delay in seconds:

$$delay[s] = delay[samples]/sample\ rate[Hz]$$

Note   The swept sine VIs expect the DUT propagation delay measurement in seconds and use the equation to convert the delay in seconds to samples.

4

Limit Testing Analysis

This chapter discusses using the polymorphic SVT Limit Testing VI located on the Limit Testing palette.

Use the SVT Limit Testing VI to perform analysis on any type of measured result produced by the Sound and Vibration Toolkit, including the following measurements:

- Waveform
- Spectrum
- Peak
- Octave
- Swept sine
- Scalar

Refer to the *LabVIEW Help* for information about the individual SVT Limit Testing VI instances.

Limit Testing Overview

Figure 4-1 illustrates how limit testing fits into the sound and vibration measurement process.
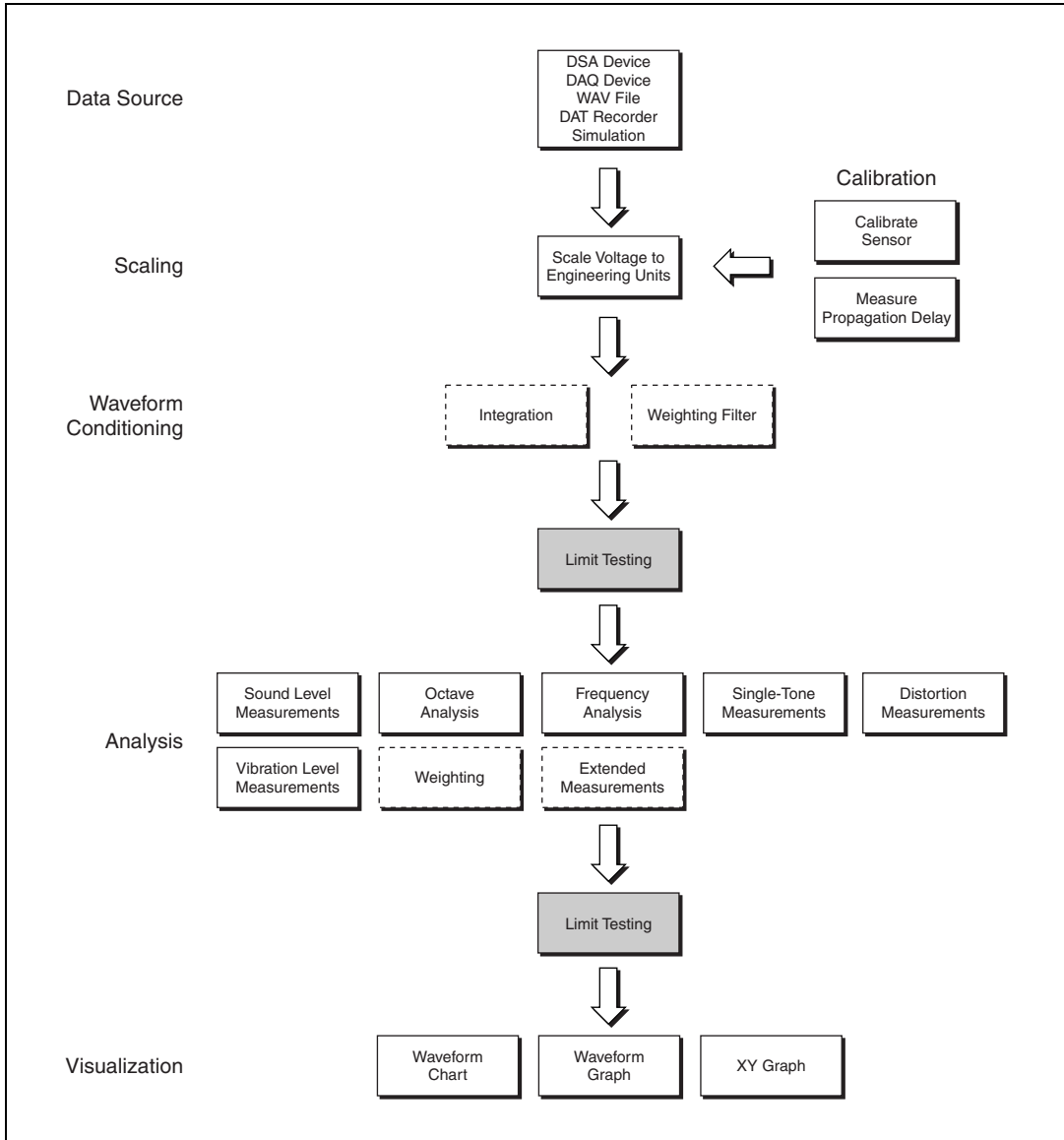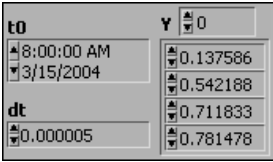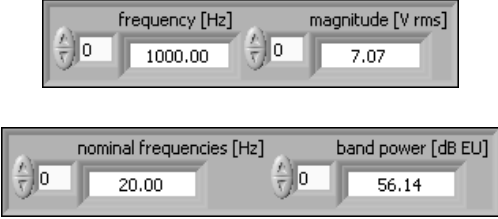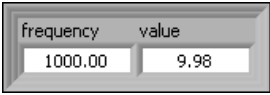
Figure 4-1. Relationship of Limit Testing to the Sound and
Vibration Measurement Process

You can use the SVT Limit Testing VI to analyze almost any measured
result produced by the Sound and Vibration Toolkit. Refer to Table 4-1 for
examples of datatypes supported by the SVT Limit Testing VI and VIs that
generate supported datatypes.

Table 4-1. Compatible Data Types for SVT Limit Testing VI

Datatype		Output VIs
Waveform Measurement		AI Read, DAQmx Read, Waveform Generation, Weighting, Integration, Vibration Level, Sound Level
Frequency Spectrum Measurement		Baseband FFT, Baseband Subset FFT, Zoom FFT, Extended Measurements
XY Data		ANSI and IEC Octave, Swept Sine
Peak Measurement		Distortion, Single-Tone, Extended Measurements
Scalar Measurement		Calibration, Vibration Level, Sound Level, ANSI and IEC Octave, Distortion, Single-Tone, Extended Measurements

Using the SVT Limit Testing VI

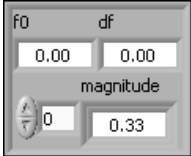Limit testing allows you to specify an envelope around the data to define a pass range. You can enter a scalar to the upper limit, lower limit, or both to specify a constant ceiling and floor for the data to perform tests such as range detection. You can enter an upper limit mask, lower limit mask, or

both to the SVT Limit Testing VI to define a pass range that varies in shape and level based on acceptable results at any given point in the measurement. You also can create a discontinuous mask which allows you to perform limit testing on only a part of the results while ignoring the rest.

You must enter at least one limit, or the SVT Limit Testing VI returns an error. You can visually display the input signal, failures, upper limit, and lower limit by creating an indicator from the **output values** terminal.

The upper limit and lower limit inputs to the SVT Limit Testing VI must be compatible with the input signal. Table 4-2 lists the criteria that must be met for each input signal type that is compatible with the SVT Limit Testing VI.

In Table 4-2, the following abbreviations apply:

- **dt** is the time spacing, in seconds, between elements
- **df** is the frequency spacing, in hertz, between elements
- N is the number of elements in the array
- $f(i)$ is the ith frequency element
- $_S$ is the signal
- $_U$ is the upper mask limit
- $_L$ is the lower mask limit

Table 4-2. Criteria for Upper and Lower Limits

Input Signal Type	Criteria on Input Limit Masks
Waveform Data Type (**t0**, **dt**, [signal])	$\mathbf{dt} > 0$ $\mathbf{dt}_S = \mathbf{dt}_U = \mathbf{dt}_L$ $N_S = N_U = N_L$
Frequency Spectrum (**f0**, **df**, [spectrum])	$\mathbf{f0}_S = \mathbf{f0}_U = \mathbf{f0}_L$ $\mathbf{df}_S = \mathbf{df}_U = \mathbf{df}_L$ $N_S = N_U = N_L$
Octave Spectrum, Swept Sine Spectrum, XY data ([X], [Y])	$[X]_S = [X]_U = [X]_L$ $N_S = N_U = N_L$
Identified Peaks, Harmonic Components, Multitone Phases [(frequency, amplitude)]	$f(i)_S = f(i)_U = f(i)_L$ $N_S = N_U = N_L$

Limit testing covers a broad range of data testing from range detection to discontinuous mask testing of a swept-sine frequency response spectrum. Figures 4-2, 4-4, 4-6, and 4-8 illustrate some, but not all, of the different ways you can use the SVT Limit Testing VI in your application.
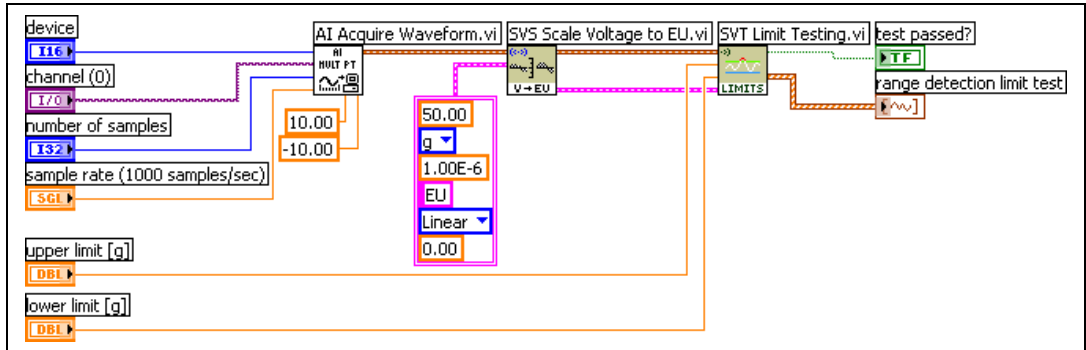
Figure 4-2. Range Detection Performed in Engineering Units

Figure 4-2 illustrates a range-detection test. Scaled waveform data and upper and lower limits are input to the SVT Limit Testing VI. The VI checks that the data falls within the envelope specified by the upper and lower limits. Figure 4-2 shows the output results for the range detection test.
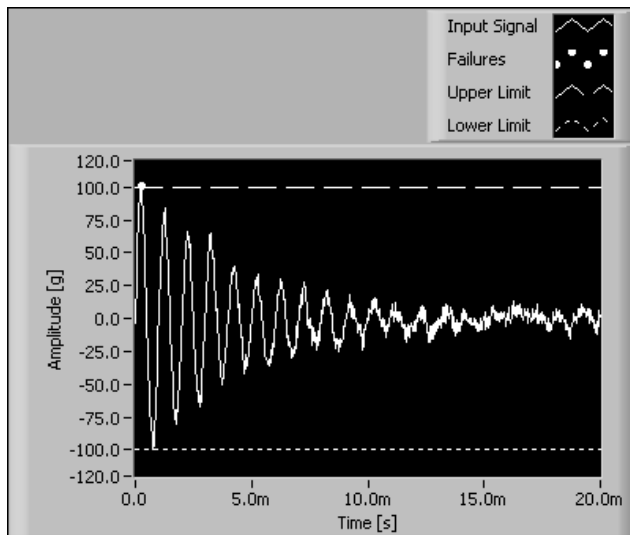
Figure 4-3. Range Detection Test on a Time-Domain Signal
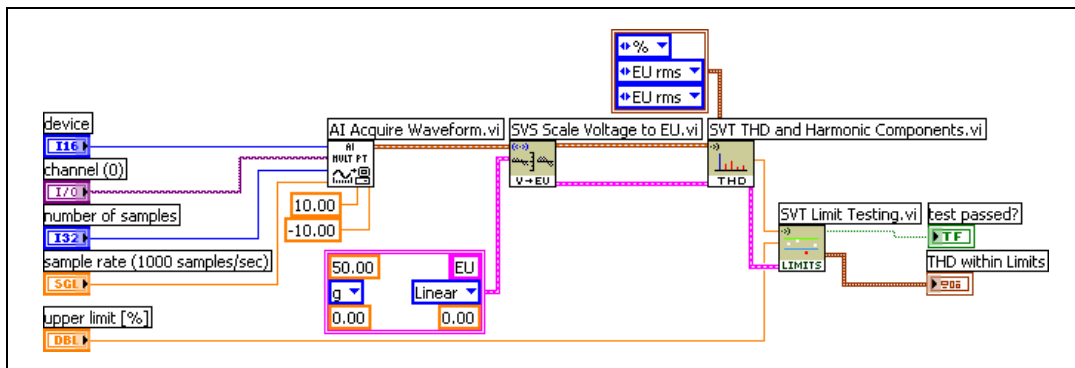
Figure 4-4. Test Scalar Measurement

Figure 4-4 shows a pass/fail test on the measured THD. This test only checks the upper limit of the measurement, therefore, only the upper limit is wired to the VI. The upper limit should have the same units as the input measurement. In this case both THD and the upper limit are expressed as percentages. Figure 4-5 shows the THD test output results.
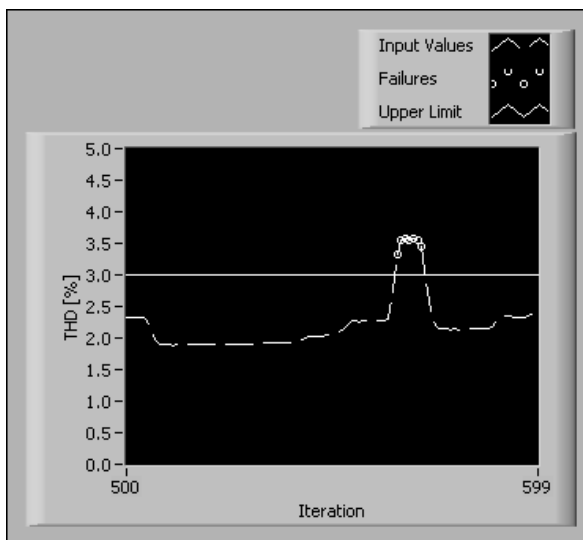
Figure 4-5. Limit Testing on THD Measurements
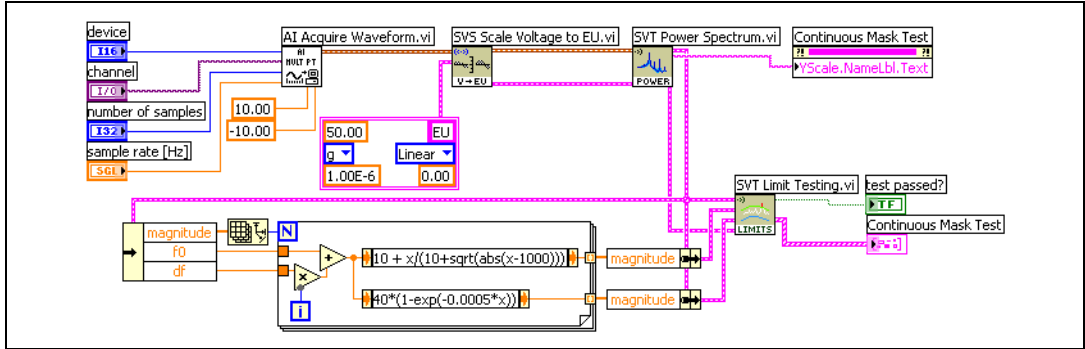
Figure 4-6. Perform Continuous Mask Test on Power Spectrum

Figure 4-6 shows a continuous mask test on a power spectrum. Formula nodes define both the upper and lower limits in this VI, making this a more complex test than the one in Figure 4-4. Figure 4-7 shows the output graph for the power spectrum continuous mask test.
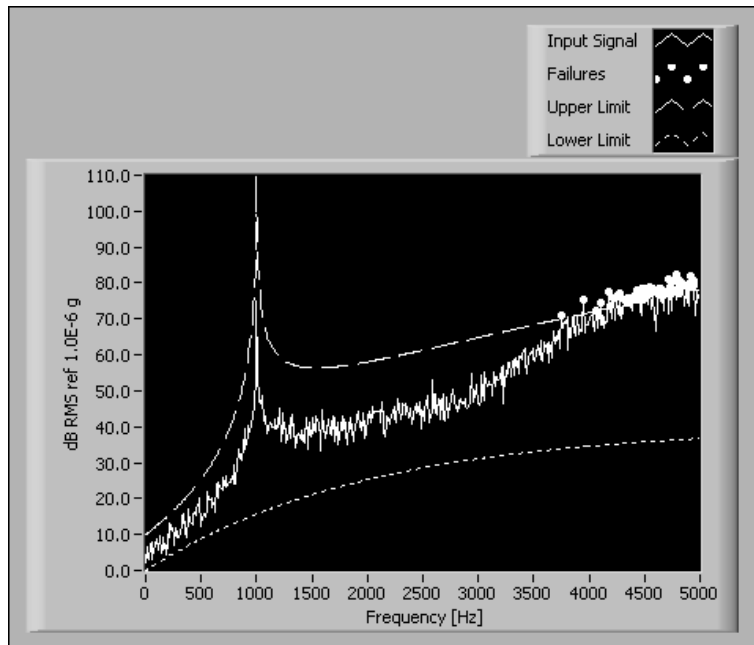
Figure 4-7. Continuous Mask Test on a Power Spectrum
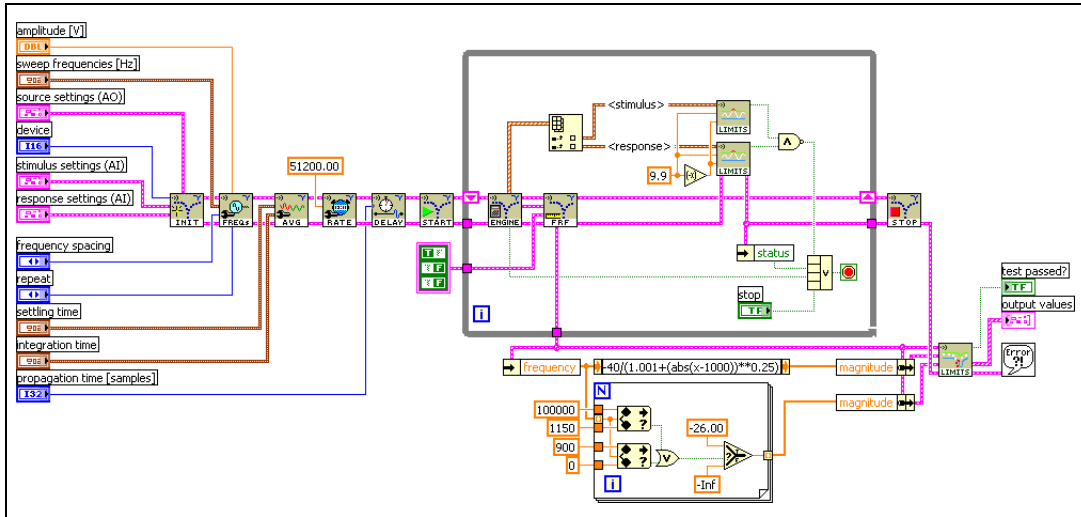
Figure 4-8. Discontinuous Mask Test on Swept-Sine Frequency Response

Figure 4-8 shows a discontinuous mask test on a swept-sine frequency response. A discontinuous mask test can track and test the results at different magnitudes and ranges, as well as stop testing at defined intervals. For example, you might use the envelope defined by the upper and lower limit masks in this example for a DUT such as a notch filter. Figure 4-9 shows the output graph for the discontinuous mask test.
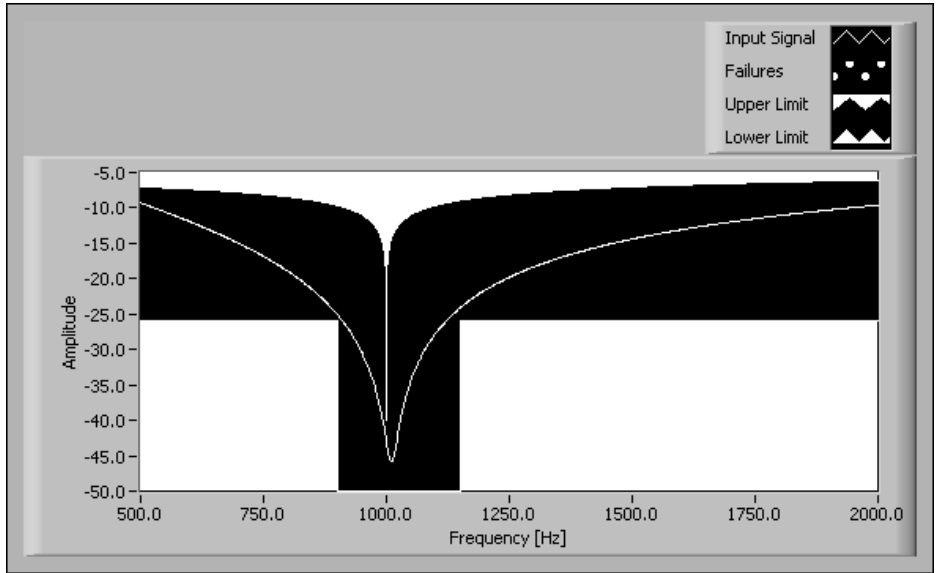
Figure 4-9. Discontinuous Mask Test on a Swept-Sine Frequency Response

5

Weighting Filters

This chapter discusses using weighting filters in sound and vibration analysis, including describing 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 Toolkit palettes contain weighting filter VIs:

- Weighting
- Octave Analysis»Extended Measurements
- Frequency Analysis»Extended Measurements

Refer to the *LabVIEW 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 psophometric weighting filters, are used to account for these nonlinearities. A typical microphone frequency response is shown in Figure 5-1 and serves as a good example of the linear response of a sensor.
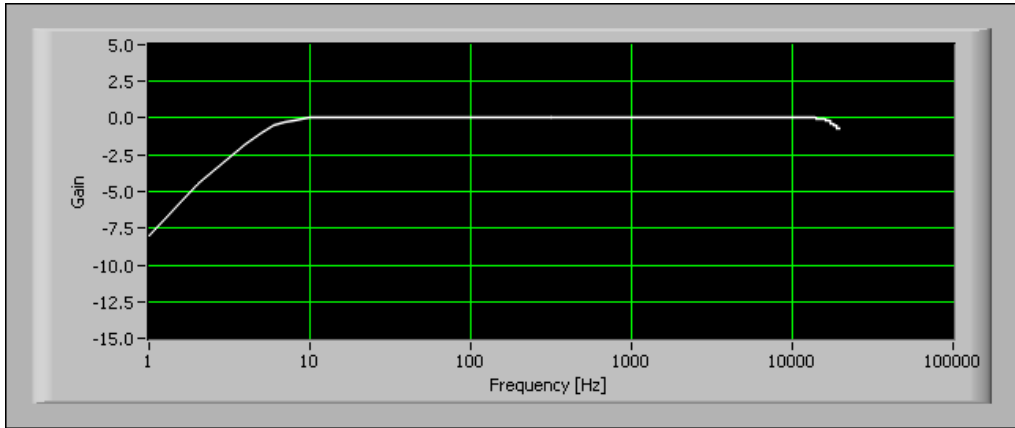
Figure 5-1. Frequency Response of a Typical Instrumentation-Grade Microphone

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 5-2.
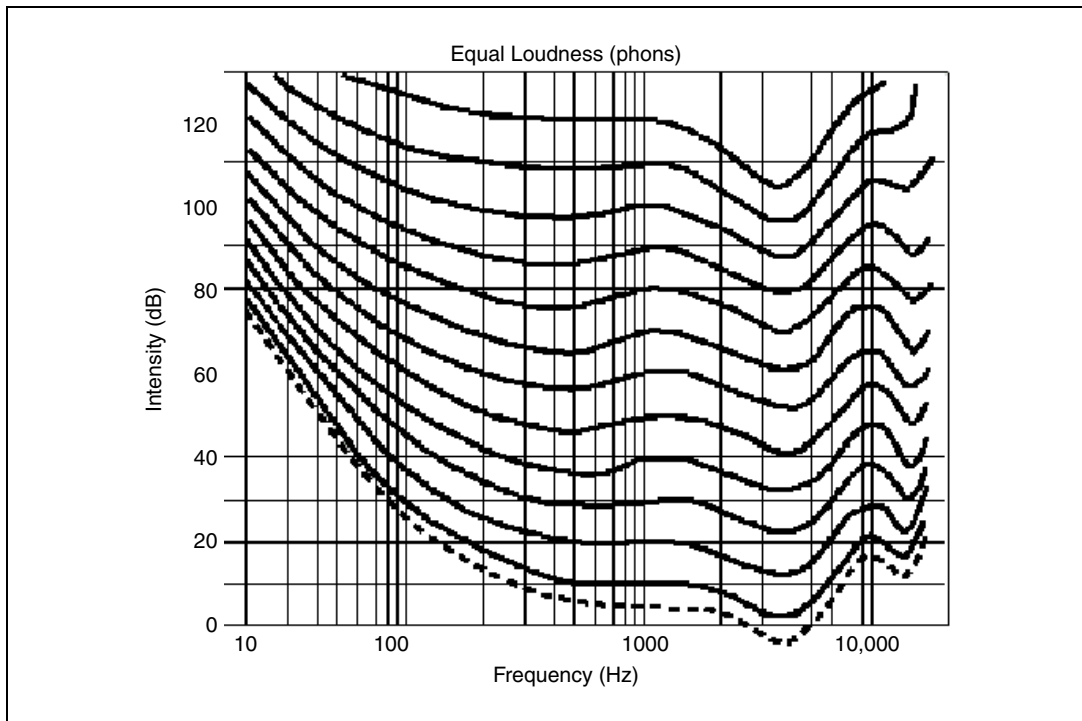
Figure 5-2. Equal Loudness Curves

Loudness is a subjective indicator of the perceived noise level expressed in phons. The loudness level in phons is the sound pressure level in decibels 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 referenced to 20 µPa. However, a signal frequency of 100 Hz requires a sound pressure level of 44 dB referenced to 20 µPa to provide the same loudness level.

Psophometric Weighting Filters

This section discusses the psophometric weighting filters available in the Sound and Vibration Toolkit.

A-, B-, and C-Weighting Filters

A-, B-, and C-weighting filters are 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 are standardized according to the ANSI S1.4 standard. Figure 5-3 shows the relative attenuation defined for A-, B-, and C-weighting filters.
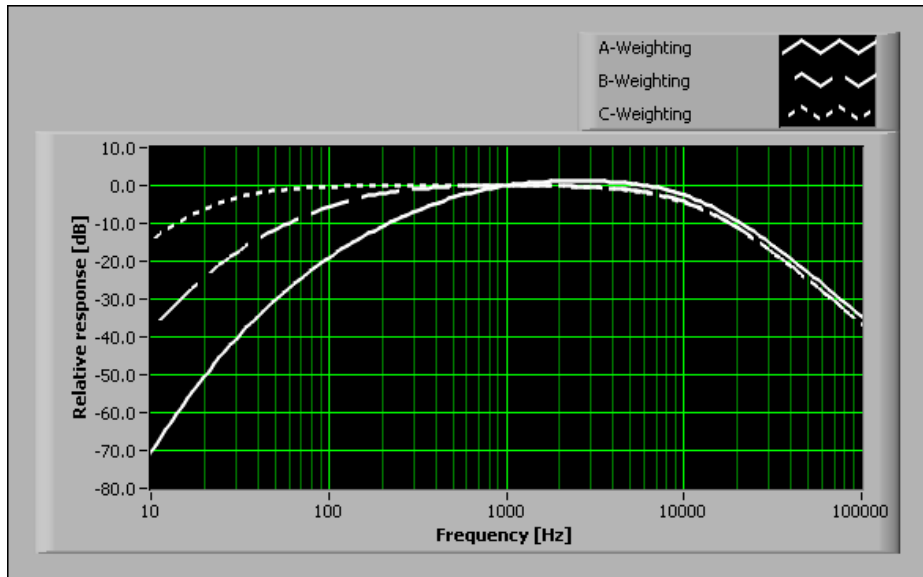
Figure 5-3. Relative Attenuation of A-, B-, and C-Weighting Filters

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

Radiocommunications Weighting Filters

ITU-R 468-4 and Dolby weighting filters are bandpass filters used to emphasize the response to the types of impulsive noise that often couple into audio cables. Typically, these filters are used to measure audio frequency noise in broadcasting, sound-recording systems, and sound program circuits.

The frequency response of the ITU-R 468-4 filter is standardized by ITU-R Recommendation 468-4. Figure 5-4 shows the relative attenuation defined for the ITU-R 468-4 and Dolby weighting filters.
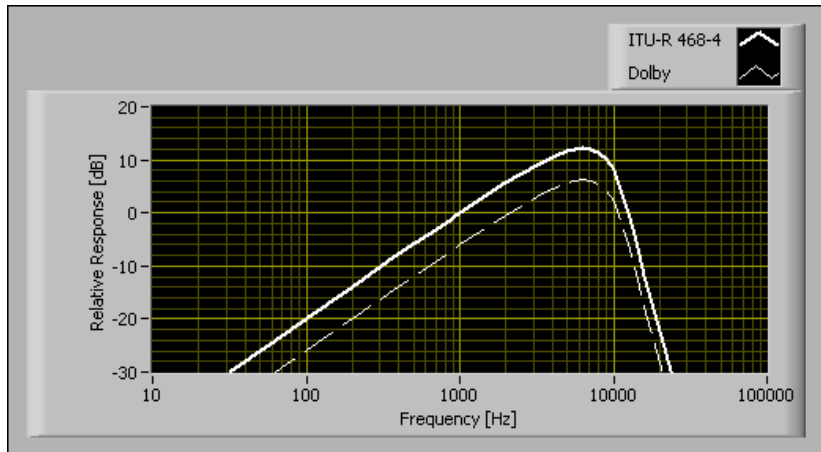
Figure 5-4. Relative Attenuation of Radiocommunications Weighting Filters

Telecommunications Weighting Filters

CCITT and C-message weighting filters are bandpass filters used to measure audio-frequency noise on telephone circuits. The CCITT (ITU-T) filter is used for international telephone circuits. The C-message filter is typically used for North American telephone circuits.

The frequency response of the CCITT and C-message weighting filters are specified in the ITU-T O.41 standard and Bell System Technical Reference 41009, respectively. Figure 5-5 shows the relative attenuation defined for the CCITT and C-message weighting filters.
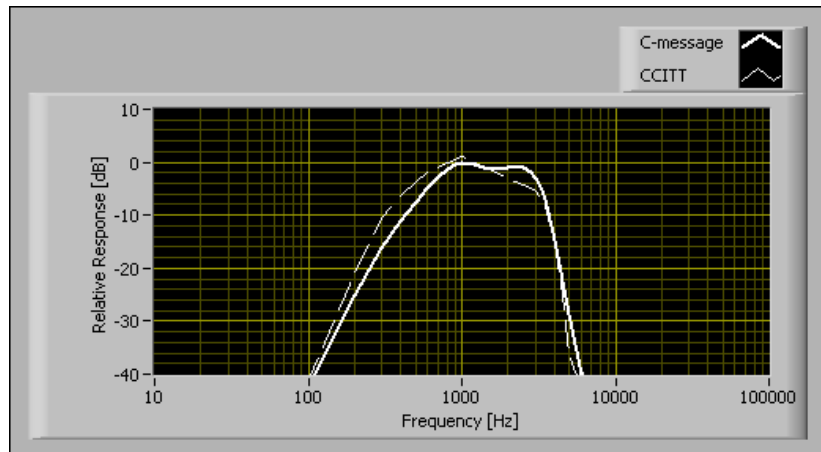
Figure 5-5. Relative Attenuation of Telecommunications Weighting Filters

Applying Weighting Filters

Use the Sound and Vibration Toolkit to apply weighting on time-domain signals, on FFT-based spectra, or on fractional-octave spectra. Figure 5-6 illustrates how applying weighting fits into the sound and vibration measurement process.
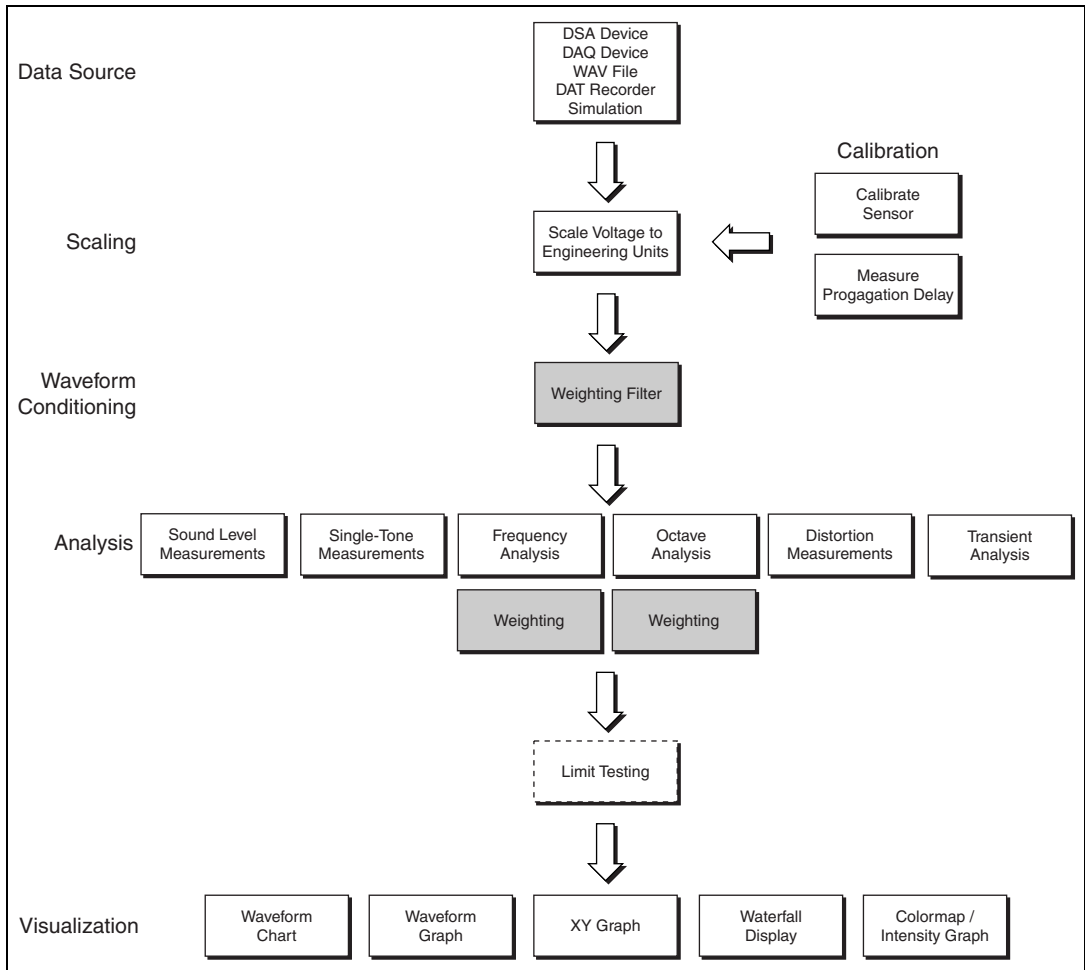
Figure 5-6. Relationship of Weighting to the Sound and Vibration
Measurement Process

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 SVL Scale Voltage to EU VI to ensure proper
display of the selected units. Also, the weighting VIs use the **channel info**
parameter to report an error if the application attempts to apply additional
weighting to a previously weighted signal.

✍ **Note**   The **weighting filter** parameter in the SVL Scale Voltage to EU VI **channel info** control assigns the correct units to the waveform, but it does not cause the VI to perform any filtering.

Refer to Chapter 3, *Scaling and Calibration*, for information about scaling a signal.

Applying Weighting to Time-Domain Data

This section discusses the two different approaches to applying weighting in the time domain offered in the SVT, as well as how each approach complies with the appropriate standards. This section also discusses how to perform A-weighted sound level measurements, and the importance of selecting an appropriate sampling frequency when using the arbitrary sample rate approach to time-domain weighting.

Use the Weighting VIs on the Weighting palette to apply weighting to time-domain signals. These VIs use pre-designed filters to apply the desired psophometric weighting. The predesigned filter nature means they only support a finite set of sample rates. Table 5-1 lists the specific supported sample rates.

For A, B, or C-weighting, you can use the VIs on the Weighting (Arbitrary Rate) palette to apply weighting for sample rates not listed in Table 5-1.

Table 5-1. Supported Psophometric Filter Rates

Sample Rates	Supported Filters		
	A, B, C-weighting	**ITU-R 468-4/Dolby**	**CCITT/C-message**
4 kHz to 20 kHz 4 kHz, 8 kHz, 10 kHz, 11.025 kHz, 12.8 kHz,	Yes	—	Yes
20 kHz to 1 MHz 20 kHz, 22.05 kHz, 25.6 kHz, 40 kHz, 44.1 kHz, 48 kHz, 50 kHz, 51.2 kHz, 80 kHz, 96 kHz, 100 kHz, 102.4 kHz, 192 kHz, 200 kHz, 204.8 kHz, 500 kHz, 1 MHz	Yes	Yes	Yes

NI recommends using the fixed-rate weighting filter VIs if the VIs support the desired sample rate. These VIs offer two advantages over the arbitrary rate VIs: compliance with the appropriate standards over the entire frequency range and slightly faster execution due to precomputed filter coefficients.

Note The filter design algorithms used by the fixed and arbitrary rate weighting approaches are different. Using a fixed-rate weighting filter with a supported frequency or using the equivalent arbitrary rate filter at the same sample rate achieve different results. Each implementation offers compliance with the appropriate standard over the frequency range specified in the *Standards Compliance* section.

Standards Compliance

This section discusses the standards to which the various weighting filter VIs comply.

A-, B-, and C-Weighting Filters

Use the SVT A, B, C Weighting Filter (Fixed Rates) VI or the SVT Weighting Filter VI to apply an A-, B-, or C-weighting filter to time-domain signals.

ANSI Standards

When combined with any DSA device, the weighting filter used by the SVT A, B, C Weighting Filter (Fixed Rates) VI or designed by the SVT Weighting Filter VI complies with the following standards:

• ANSI S1.4-1983

• ANSI S1.42-2001

The SVT Weighting Filter VI accommodates any sample rate greater than 4 kHz 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. This frequency is the maximum frequency within tolerances. Use the SVT Max Frequency Within Tolerances [ANSI] VI located on the **Weighting (Arbitrary Rate)** palette to determine the maximum frequency within tolerances.

The SVT A, B, C Weighting Filter (Fixed Rates) VI supports the sample rates listed in Table 5-1. For all supported sample rates the VI achieves Type 0 compliance to the Nyquist frequency.

ISO/IEC Standard

Use the SVT A, B, or C Weighting Filter (Fixed Rates) VI or the SVT Weighting Filter VI to apply an A-, B-, or C-weighting filter to time-domain signals. When combined with any DSA device, the weighting filter used by the SVT A, B, or C Weighting Filter (Fixed Rates) VI or designed by the SVT Weighting Filter VI complies with the IEC 61672-1:2002 standard.

The SVT Weighting Filter VI accommodates any sample rate greater than 4 kHz and designs the filter coefficients to target the attenuation curves defined by the IEC standards. Given the selected sampling frequency, compliance with a particular filter type, either Class 2 or Class 1, is ensured up to a specific frequency. This frequency is the maximum frequency within tolerances. Use the SVT Max Frequency Within Tolerances [IEC] VI to determine the maximum frequency within tolerances.

The SVT A, B, C Weighting Filter (Fixed Rates) VI supports the sample rates listed in Table 5-1. For all supported sample rates the VI achieves Class 1 compliance to the Nyquist frequency.

Radiocommunications Weighting Filters

When combined with any DSA device, the weighting filter used by the SVT Radiocommunications Weighting Filter (Fixed Rates) VI complies with the ITU-R 468-4 standard.

The SVT Radiocommunications Weighting Filter (Fixed Rates) VI accommodates the sample rates listed in Table 5-1. For all supported sample rates, the VI achieves compliance to the Nyquist frequency.

Telecommunications Weighting Filters

When combined with any DSA device, the weighting filter used by the SVT Telecommunications Weighting Filter (Fixed Rates) VI complies with the following standard and reference:

- ITU-T O.41
- Bell System Technical Reference 41009

The SVT Telecommunications Weighting Filter (Fixed Rates) VI supports the sample rates listed in Table 5-1. For all supported sample rates, the VI achieves compliance to the Nyquist frequency.

Performing A-Weighted Sound Level Measurements

The block diagram in Figure 5-7 shows a VI that uses weighting filters to perform an A-weighted sound level measurement with a slow time constant (L_{AS}) on a simulated signal. The application applies weighting to the time-domain signal with the SVT A, B, C Weighting Filter (Fixed Rates) VI because the signal is simulated with a sample rate supported for A-weighting, as shown in Table 5-1.
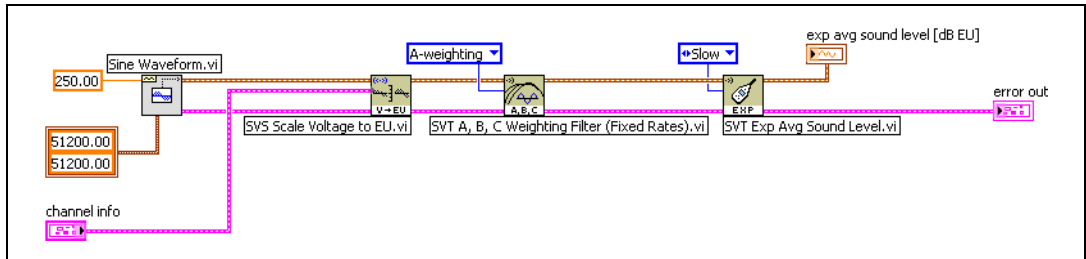
Figure 5-7. A-Weighting of a Simulated Time-Domain Signal Block Diagram

Time-domain data is simulated and scaled before being sent to the SVT A, B, C Weighting Filter (Fixed Rates) VI. The weighted signal is sent to the Sound Level Measurement VI.

Figure 5-8 shows the time-domain input and output waveforms when a 250 Hz sine wave is sent to the SVT A, B, C Weighting Filter (Fixed Rates) VI using the A-weighting filter.
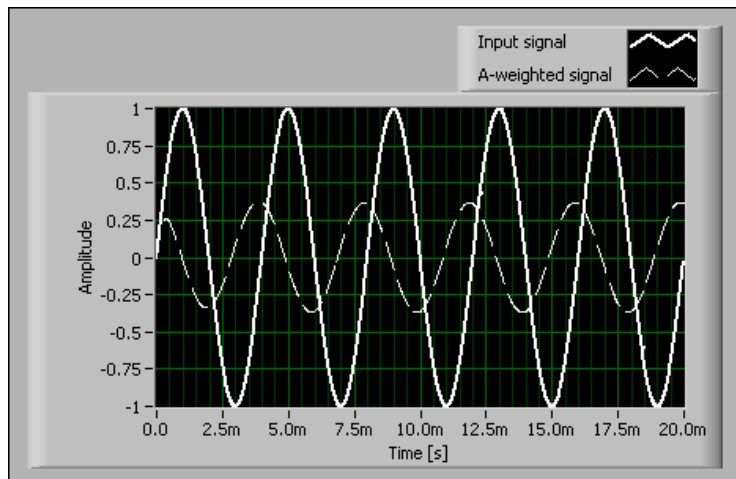
Figure 5-8. A-Weighting Response in the Time Domain

There is a phase difference between the input and output signals because a filter applies the time-domain weighting. The transient behavior at the beginning of the filtered waveform corresponds to the filter settling time.

Applying Weighting to an Octave Spectrum

When performance, such as CPU usage, is an issue, applying weighting in the frequency domain can improve the efficiency of the measurement process. The block diagram in Figure 5-9 uses the SVT Weighting Filter (octave) VI located on the Octave Analysis»Extended Measurements palette to apply weighting to a third-octave spectrum. The weighting method illustrated in Figure 5-9 requires less processing time than applying weighting in the time domain and subsequently computing the third-octave spectrum.
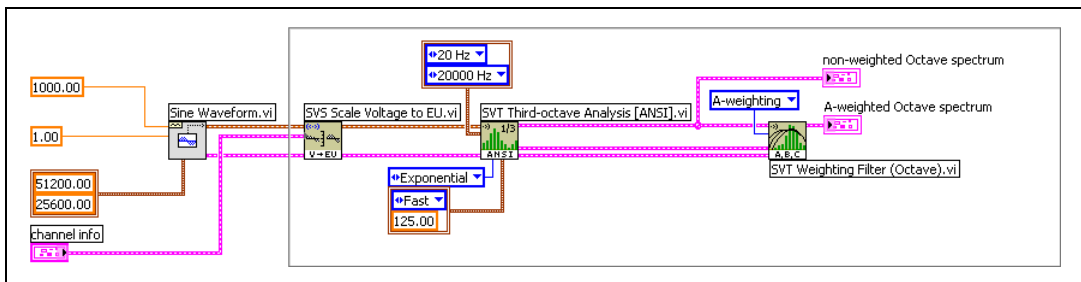
Figure 5-9. Applying Weighting to a Third-Octave Spectrum

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 does not have a continuous 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. Figure 5-10 shows how the slope of the A-weighting curve can contribute to potential measurement errors at low frequencies.
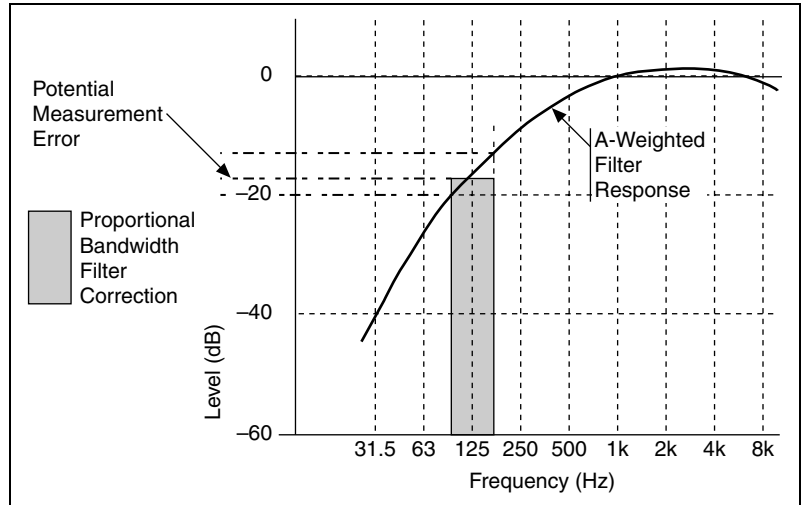
Figure 5-10. Potential Measurement Error for A-Weighting at Low Frequency

✎ **Note**   The same type of measurement error in Figure 5-10 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, especially when you need to 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 Frequency Analysis»Extended Measurements palette to apply weighting in the frequency domain.

Figure 5-11 shows a weighting filter in a frequency measurement.
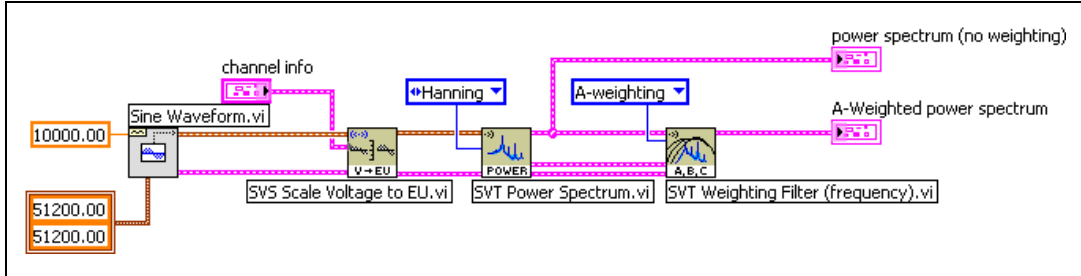
Figure 5-11. Applying Frequency Weighting to a Power Spectrum

Figure 5-12 shows a different implementation based on applying the weighting filter on the time-domain signal and then computing the power spectrum.
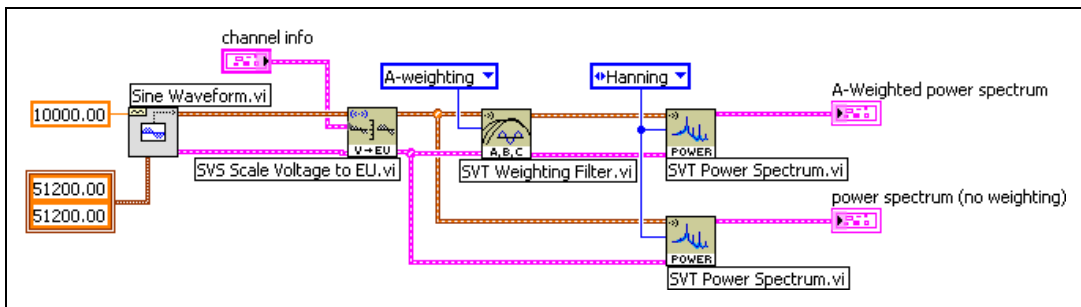
Figure 5-12. Applying Weighting before Computing the Power Spectrum

In Figure 5-11, a single power spectrum is computed. In Figure 5-12, 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 in Figure 5-11, you can decrease CPU usage and processing time.

6

Integration

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

The Sound and Vibration Toolkit contains the following integration VIs:

- SVT Integration VI located on the Integration palette for time-domain integration
- SVT Integration (frequency) VI located on the Frequency Analysis»Extended Measurements palette for frequency-domain integration

Refer to the *LabVIEW 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:

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

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

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 or perform a double integration with respect to time to compute the displacement.

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:

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

$$v = -\frac{A}{\omega} \cos(\omega t) = \frac{A}{\omega} \sin\left(\omega t - \frac{\pi}{2}\right) \qquad (6\text{-}1)$$

$$d = -\frac{A}{\omega^2} \sin(\omega t) = \frac{A}{\omega^2} \sin(\omega t - \pi) \qquad (6\text{-}2)$$

Note Initial condition is arbitrarily set to zero in Equations 6-1 and 6-2.

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 6-1 illustrates the relationship between acceleration, velocity, and displacement.
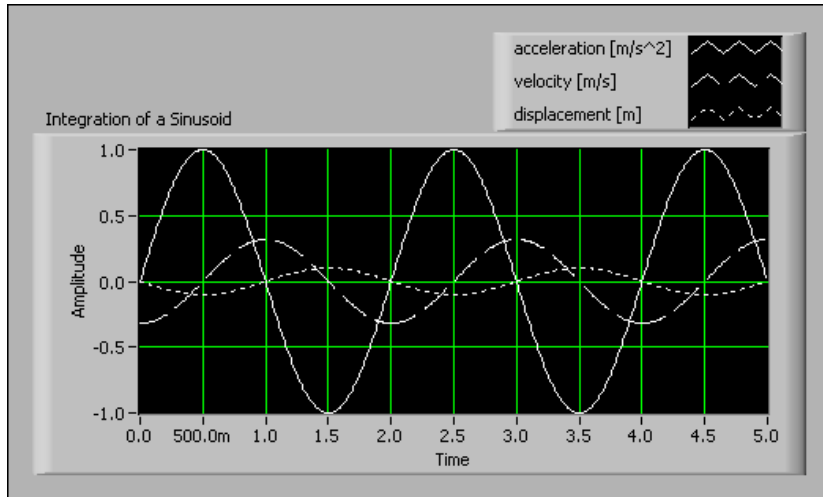

Figure 6-1. Integration of a 0.5 Hz Sine Wave

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 in Figures 6-9 and 6-10.

Implementing Integration

If you need to perform measurements on velocity or displacement data when you have only acquired acceleration or velocity data, respectively, integrate the measured signal to yield the desired data. You can perform integration either in the time domain as a form of signal conditioning or in the frequency domain as a stage of analysis. When performed in the frequency domain, integration is one of the extended measurements for frequency analysis.

Figure 6-2 illustrates how integration fits into the sound and vibration measurement process.
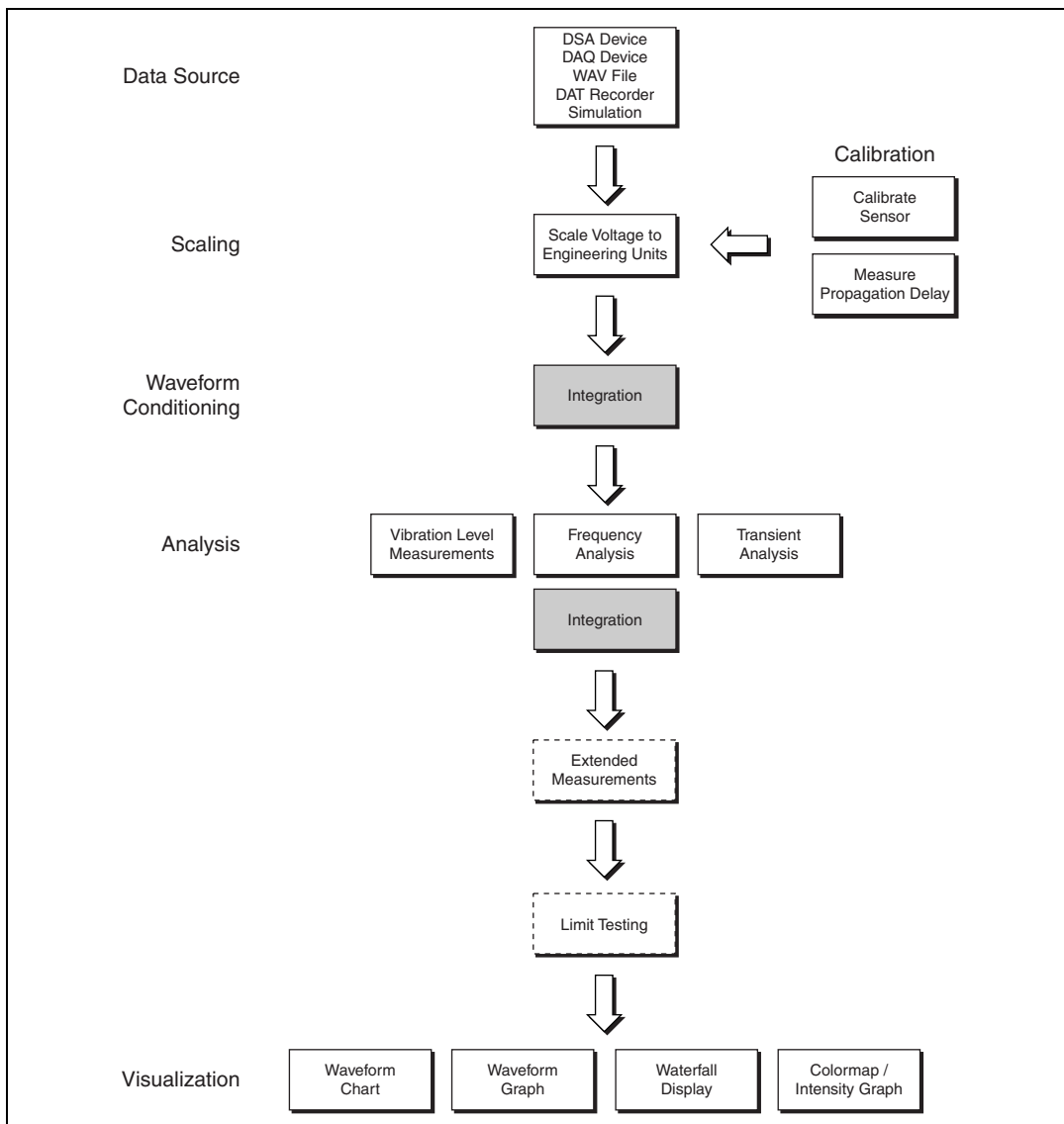
Figure 6-2. Relationship of Integration to the Sound and Vibration
Measurement Process

Challenges When Integrating Vibration Data

Converting acceleration data to velocity or displacement data presents a pair of unique challenges. Measured signals typically contain some unwanted DC components. Refer to the *DC Component* section for information on this issue. The second challenge is the fact that many transducers, especially vibration transducers, have lower-frequency limits. A transducer cannot accurately measure frequency components below the lower-frequency limit of the transducer. Refer to the *Transducers* section for information on this issue.

DC Component

Even though a DC component in the measured signal might be valid, the presence of a DC component indicates that the DUT has a net acceleration along the axis of the transducer. For a typical vibration measurement, the DUT is mounted or suspended in the test setup. The net acceleration of the DUT 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. Closeness 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 Toolkit

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 is double integrated. Figure 6-3 shows the block diagram for the VI.
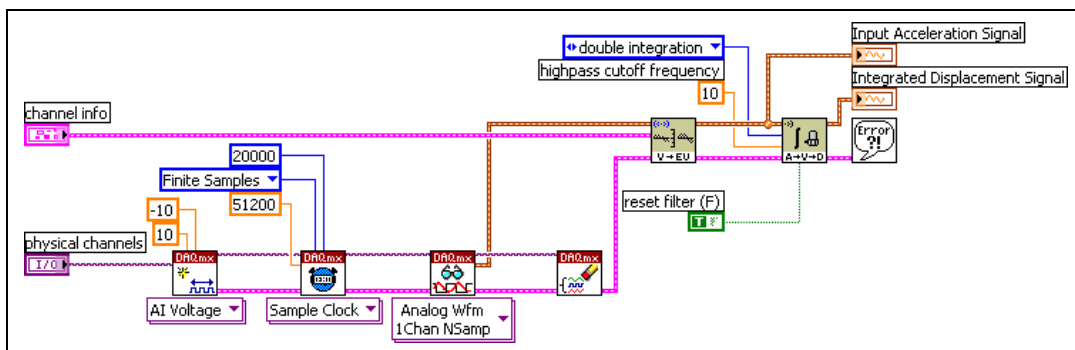
Figure 6-3. Block Diagram for Single-Shot Acquisition and Integration

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 6-4 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 6-4. 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 6-5 shows the block diagram for a VI designed for continuous acquisition and integration.

Figure 6-5. Continuous Acquisition and Integration
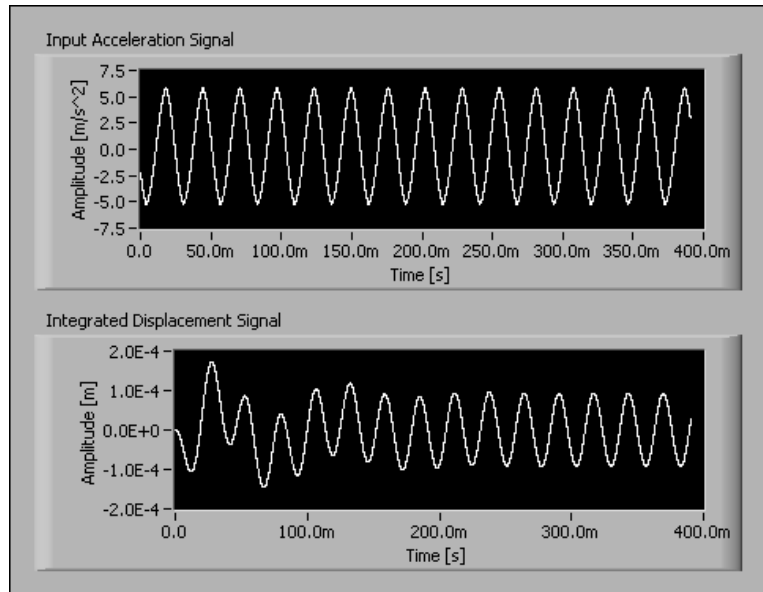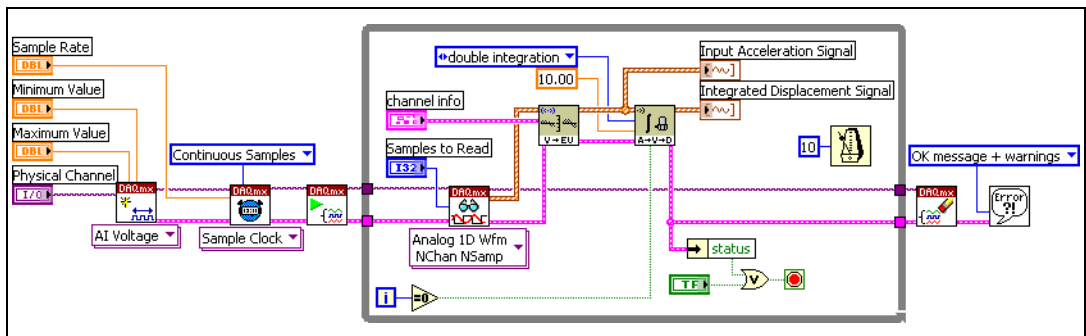
In this example, the highpass cut-off frequency used for the integration is 10 Hz. Additionally, the integration is explicitly reset in the first iteration of the VI and performed continuously thereafter. In this example, this additional wiring is optional because the SVT Integration VI automatically resets the first time it is called and runs continuously thereafter.

If you use the block diagram in Figure 6-5 in a larger application that requires starting and stopping the data acquisition process more than once, NI 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 6-6 shows the results of the continuous acquisition and integration of the same 38 Hz sinusoid used in the single-shot acquisition and integration example.
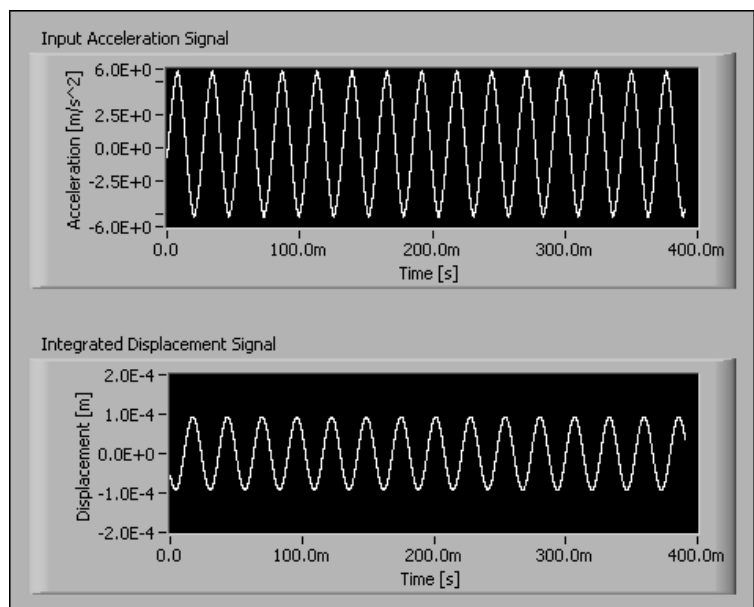
Figure 6-6. Settled Response of Continuous Acquisition and Integration

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 settle, you can use the integrated signals for additional analysis.

Figure 6-7 shows the frequency response for time-domain single integration. Figure 6-8 shows the frequency response for time-domain double integration.
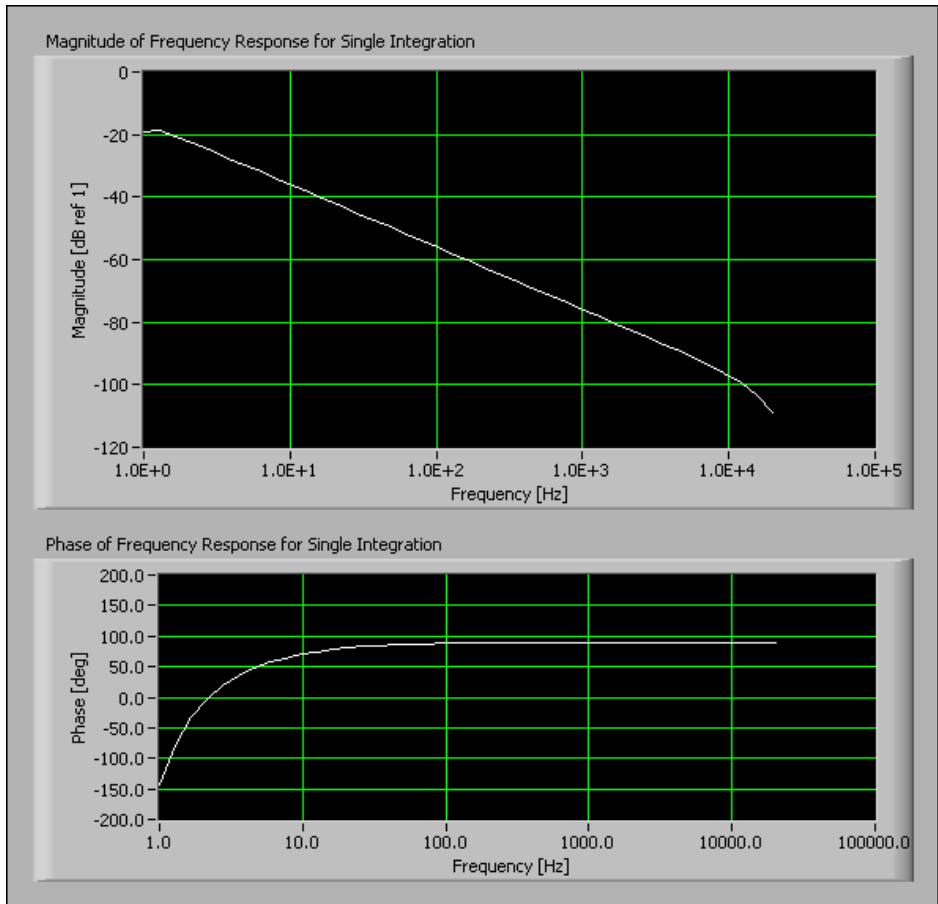
Figure 6-7. Frequency Response for Single Integration
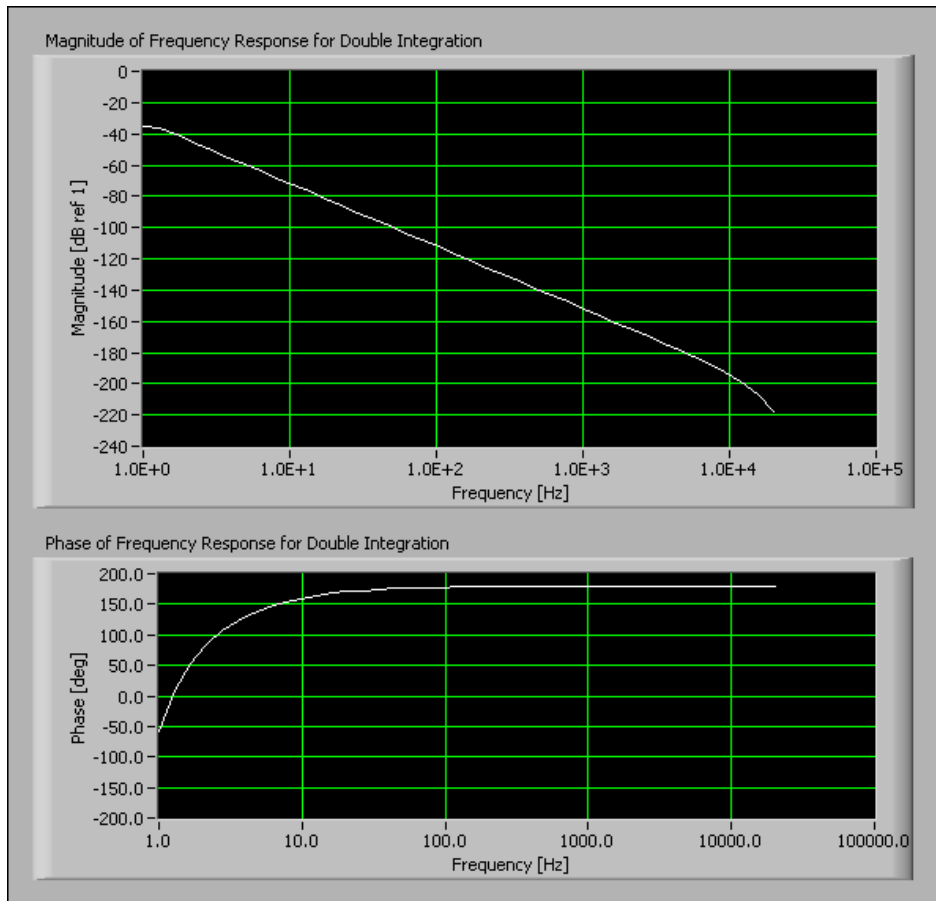
Figure 6-8. Frequency Response for Double Integration

In Figure 6-7, you can see the characteristic 20 dB per decade roll-off of the magnitude response of the single integration. In Figure 6-8, 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 Hz to 6.6 kHz for double integration. The accuracy ranges change with the sampling frequency and the highpass cut-off 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. 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 6-9 shows the block diagram for the example VI.
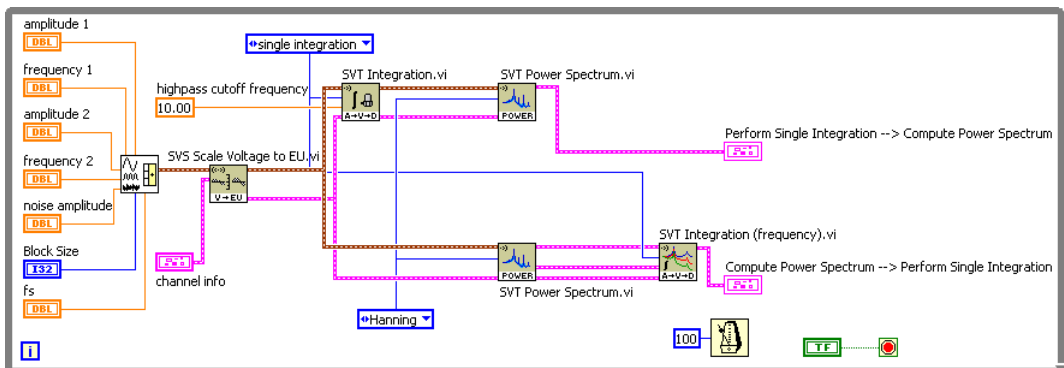
Figure 6-9. Integration in the Time Domain and in the Frequency Domain

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 (–Inf) if the spectrum scale is in decibels.

Figure 6-10 shows the results of integrating in the time and frequency domains.
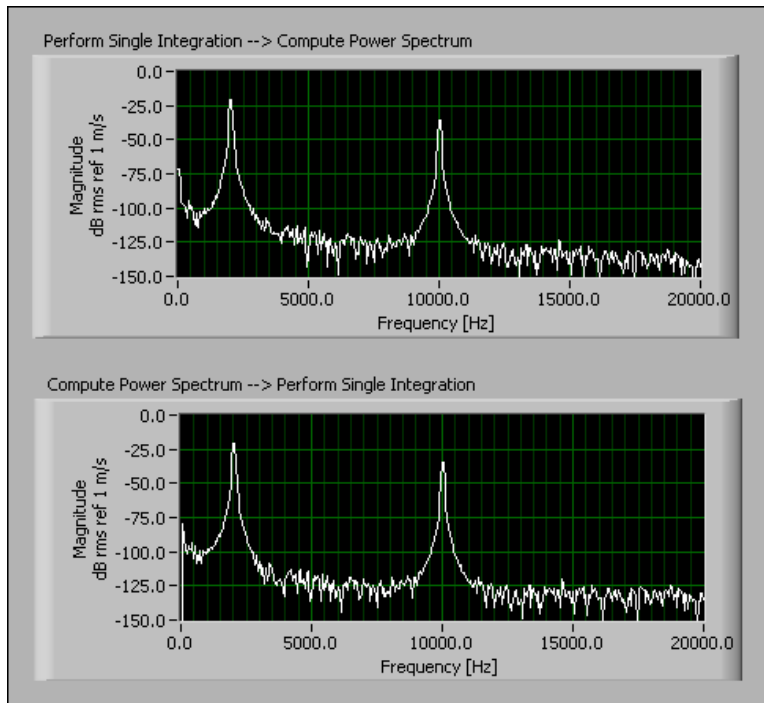
Figure 6-10. Power Spectra of the Integrated Signal

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

7

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 Vibration Level palette to perform vibration-level measurements.

Refer to the *LabVIEW Help* for information about individual vibration level VIs.

Figure 7-1 illustrates how vibration-level measurement fits into the sound and vibration measurement process.

Figure 7-1. Relationship of Vibration-Level Measurements to the Sound and Vibration Measurement Process

Measuring the Root Mean Square (RMS) Level

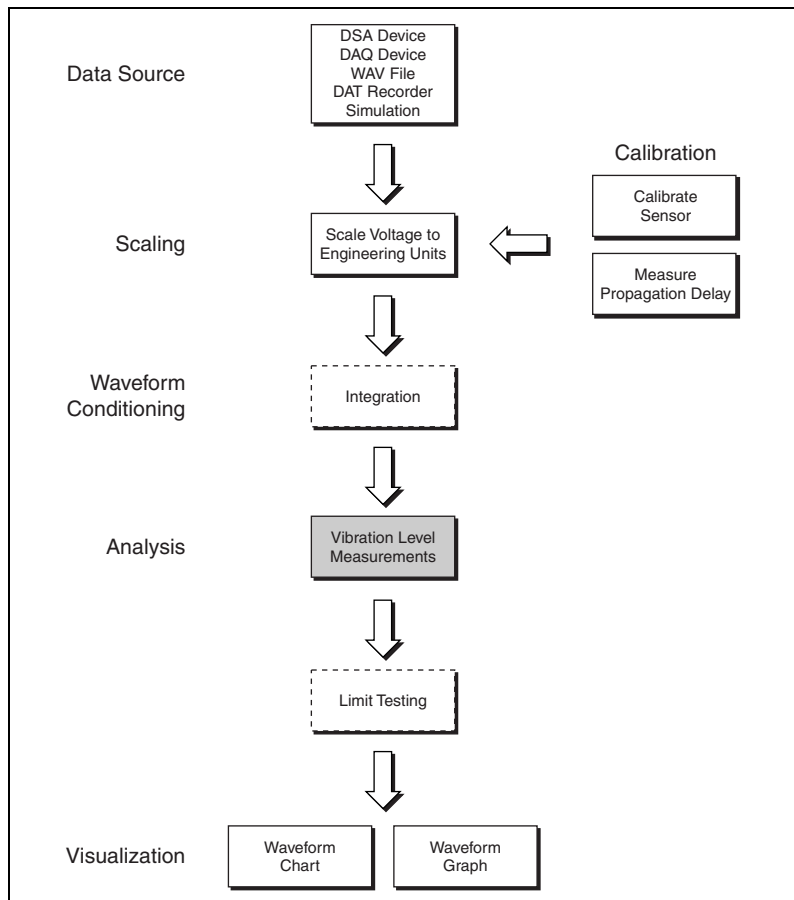A basic requirement of vibration measurements is measuring the level of the signal returned by an accelerometer. The level of the accelerometer signal generally is expressed in root-mean-square acceleration (g_{rms}).

Single-Shot Buffered Acquisition

The block diagram in Figure 7-2 illustrates a VI designed to perform a single-shot acquisition and compute the RMS levels.
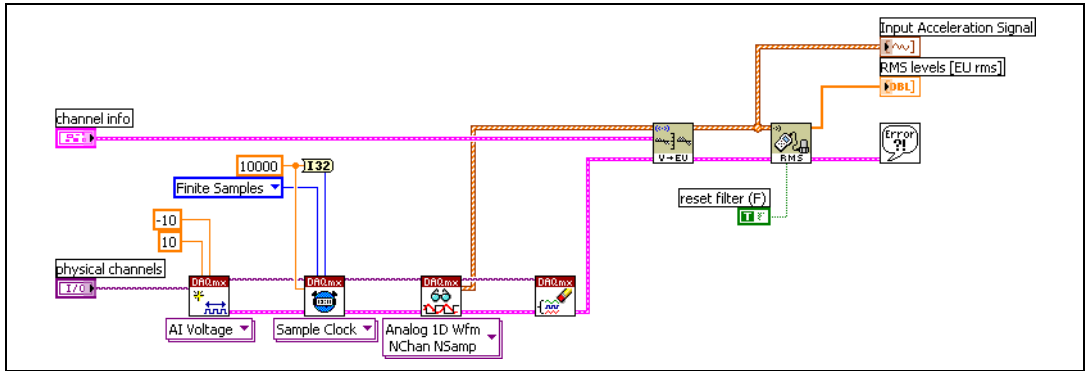
Figure 7-2. Single-Shot Buffered Acquisition and RMS Level VI

The sampling frequency is 10 kS/s. A buffer containing 1 s of data is returned by the read VI.

Continuous Signal Acquisition

You can use the block diagram in Figure 7-2 with a While Loop to continuously acquire signals from an accelerometer and display the vibration level in a chart. The block diagram in Figure 7-3 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.
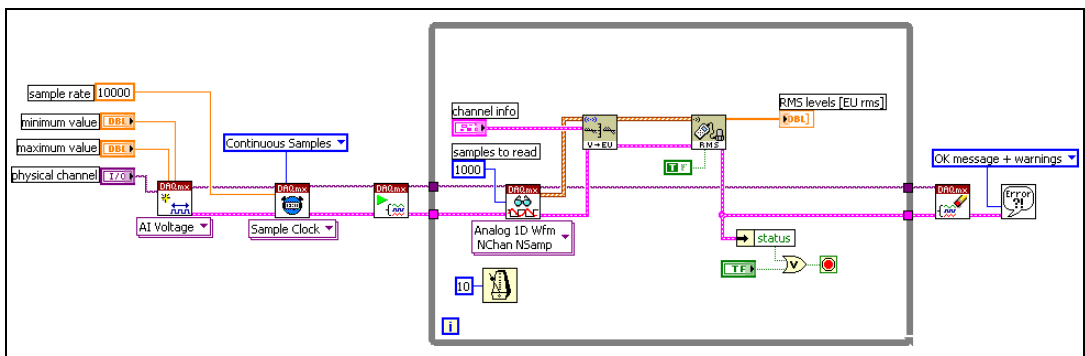
Figure 7-3. Continuous Data Acquisition and RMS Level VI

> ✏ **Note** Set the **restart averaging** control on the SVT RMS Level VI to TRUE. Otherwise, the SVT RMS Level VI accumulates intermediate results 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, which is the integration time. The block diagram in Figure 7-4 illustrates an application using the SVT Running RMS Level VI. The sampling frequency is 10 kS/s. The read VI reads 1,000 samples at a time.
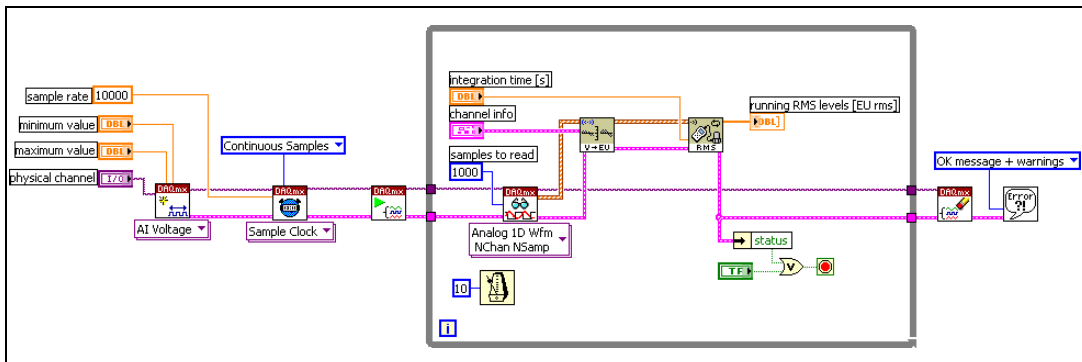
Figure 7-4. Running RMS VI

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 8, *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 7-5 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.
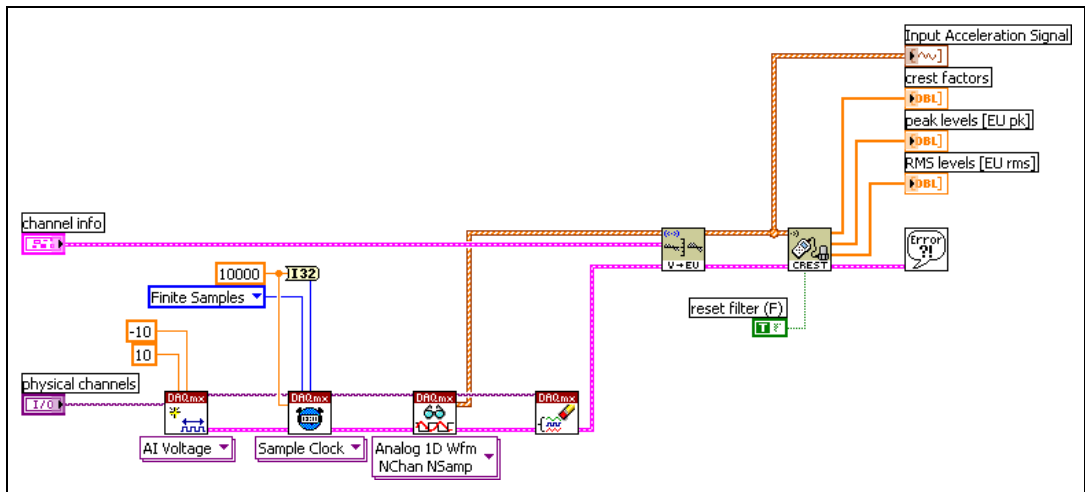
Figure 7-5. Crest Factor VI

8

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 Sound Level palette to perform sound-level measurements. You can combine different sound-level measurements and use them simultaneously to provide flexibility with acoustic measurements.

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

Figure 8-1 illustrates how sound-level measurement fits into the sound and vibration measurement process.
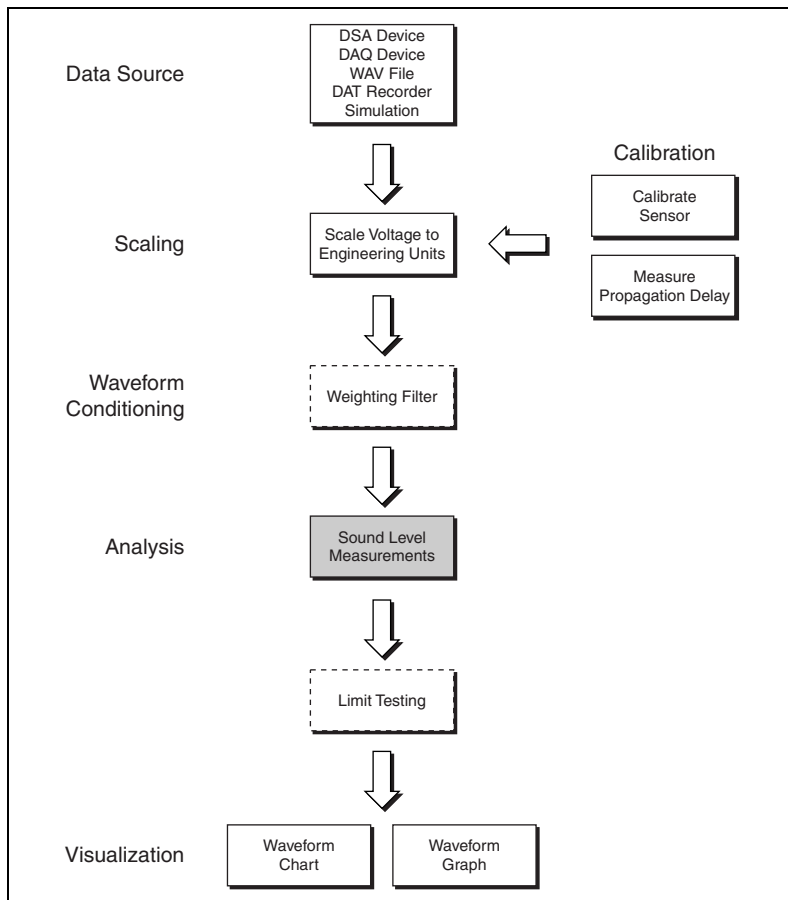
Figure 8-1. Relationship of Sound-level Measurements to the Sound and Vibration Measurement Process

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 decibel 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.

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

where

P_0 is the reference pressure of 20 μPa for acoustics.

Single-Shot Linear Averaging

Figure 8-2 illustrates an example of linear averaging on a single-shot acquisition.
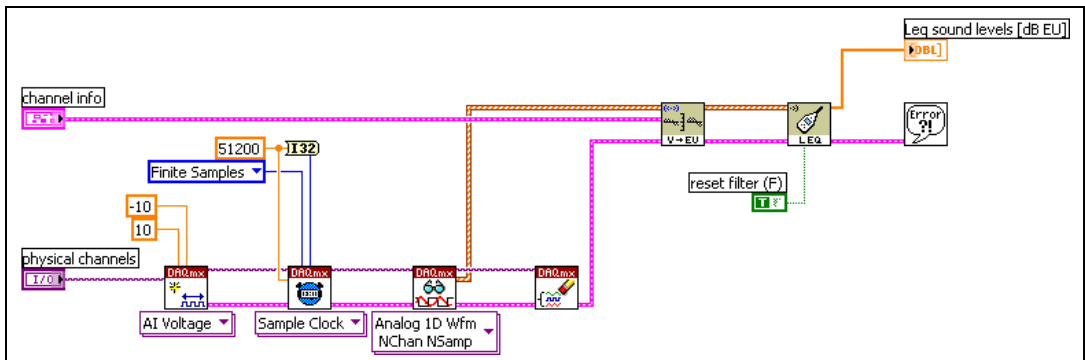

Figure 8-2. Single-Shot L_{eq} VI

The single-shot acquisition 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 at the specified sampling frequency.

Figure 8-3 displays the resulting L_{eq} measurement and the instantaneous sound pressure level.
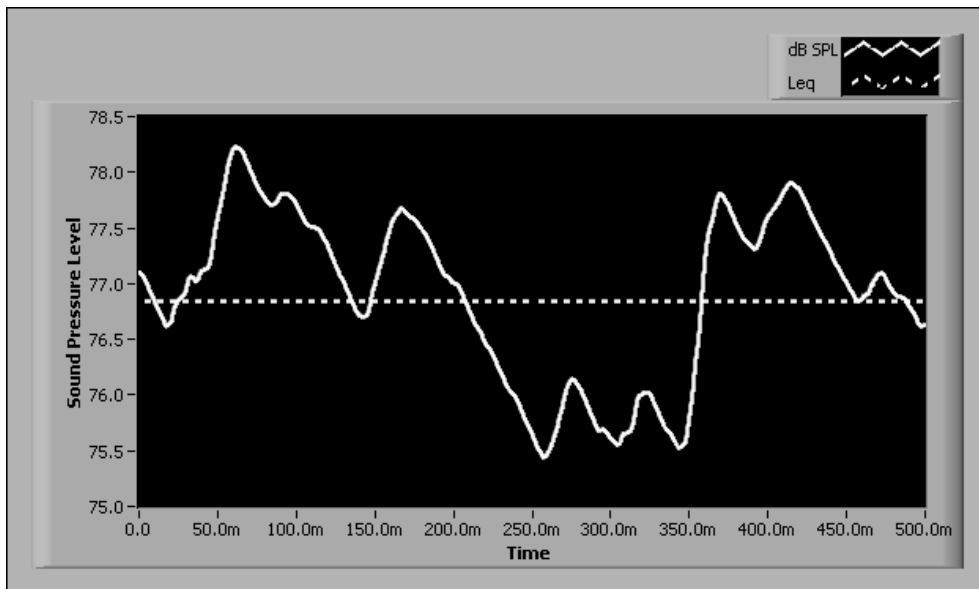
Figure 8-3. L_{eq} and Instantaneous Sound Pressure Level versus Time

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} in this case, 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, 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 when working with data acquisition devices with aliasing protection, a sampling frequency of 51.2 kS/s is used to perform measurements up to 20 kHz.

For one hour of data with a sampling frequency of 51.2 kS/s, you need to acquire more than 184 million samples, as calculated by 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 alternate 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 Toolkit to perform this alternate method of L_{eq} measurement. Figure 8-4 shows the block diagram for a VI designed to compute the L_{eq} for a one-hour period.
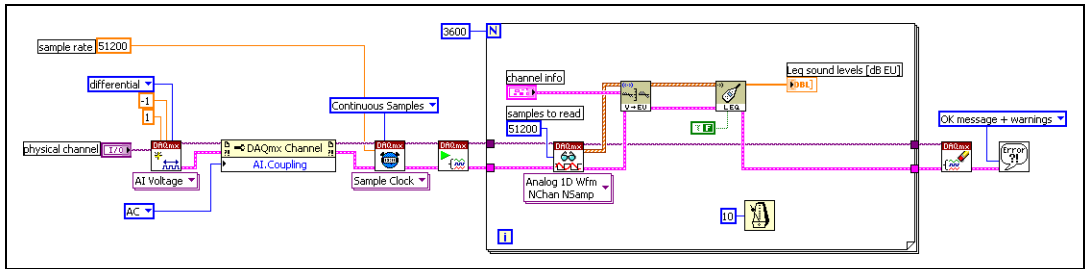
Figure 8-4. One Hour L_{eq} VI

The VI in Figure 8-4 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 the measurement over four seconds and repeat it 900 times, and so forth.

Restart Averaging and Advanced Concepts

Some applications require you to perform L_{eq} measurements continuously, using a specific integration time. For example, you may need to measure a reverberation time that requires the application 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 8-5.
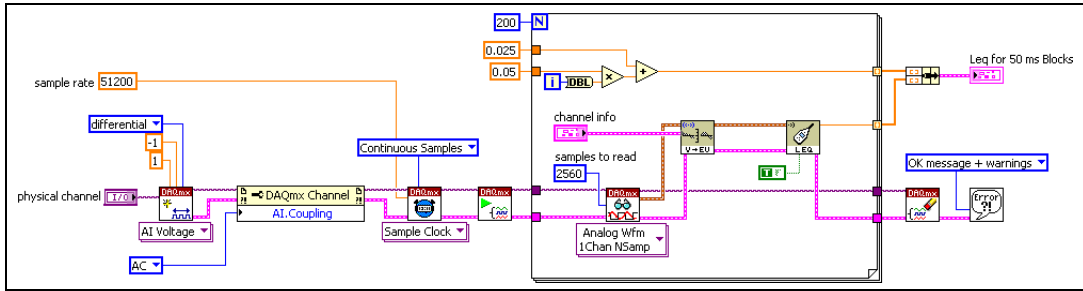
Figure 8-5. Block Diagram for Reverberation VI

Because you divide the 10 s 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 8-6 shows the results of performing a reverberation time measurement for 10 s.
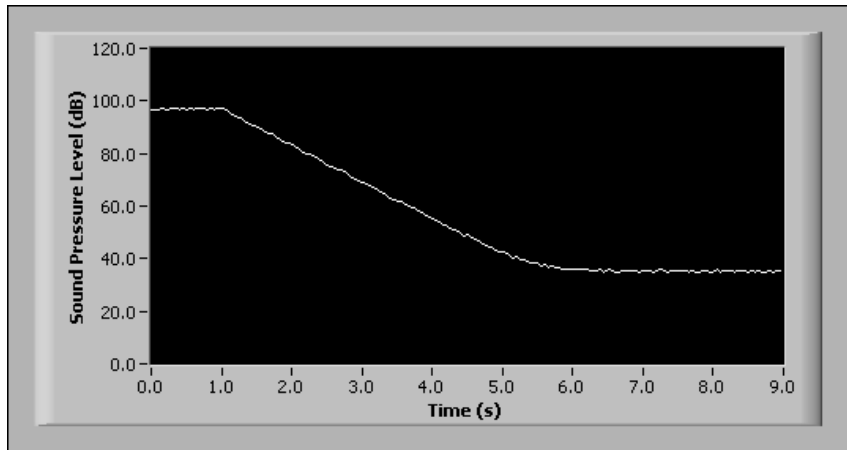
Figure 8-6. 10-Second Reverberation Time Measurement

Performing a Running L_{eq}

Use the SVT Running Leq VI to perform a running L_{eq} over a specified period of time, which is the 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 with sound pressure levels that vary slowly.

- **Fast** uses a time constant of 125 ms. Fast averaging is useful for tracking the sound pressure of signals with sound pressure levels that 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 increases in the sound pressure level and recording the increases so that you have a record of the changes.

- **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 Help* for information about the SVT Exp Avg Sound Level VI and the SVT Decimated Exp Avg Sound Level VI.

Figure 8-7 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.
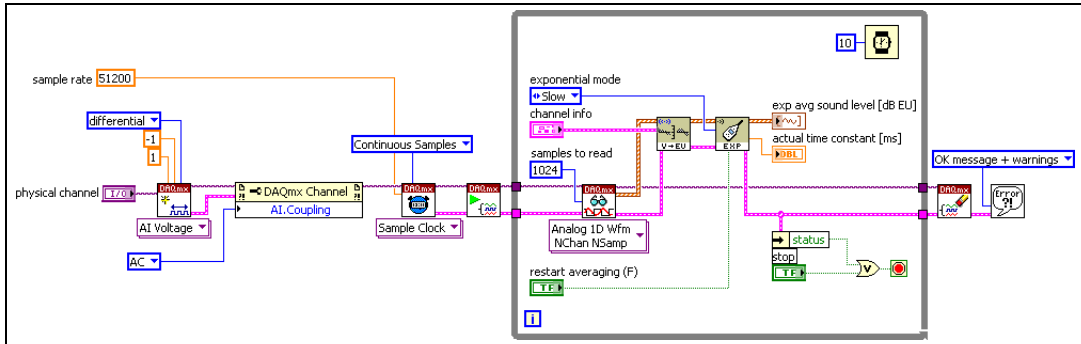
Figure 8-7. Exponential Averaging with Slow Time Constant VI

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. 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 the 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 5, *Weighting Filters*, for information about weighting filters.

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

9

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 Octave Analysis palette to perform the following analyses:

- Full-octave
- Third-octave
- Fractional-octave, 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 Help* for information about the individual Octave Analysis VIs.

Figure 9-1 illustrates the relationship of fractional-octave analysis to the sound and vibration measurement process.
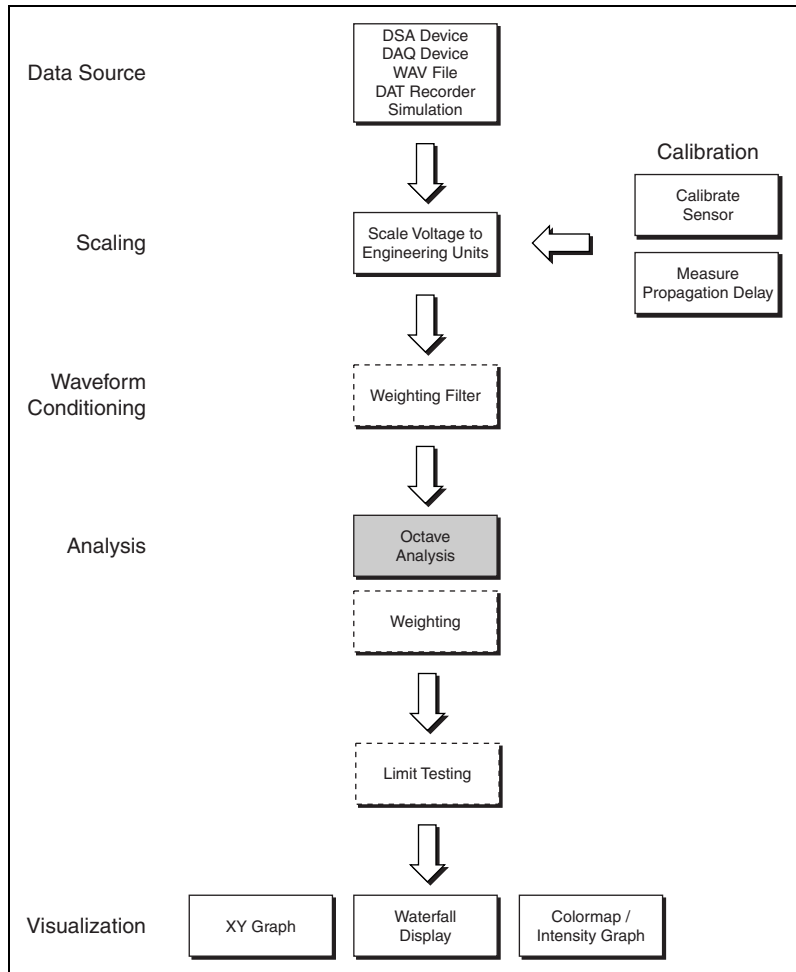
Figure 9-1. Relationship of Fractional-Octave Analysis to the Sound and
Vibration Process

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* bandpass 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.

Full-Octave Analysis in the 31.5 Hz–16 kHz Band

The following example is a simple illustration of full-octave analysis. In this example, an input signal acquired by a microphone is sent through 10 bandpass filters centered at the frequencies listed in Figure 9-2.

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 9-2. 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. Figure 9-3 shows this bar graph.
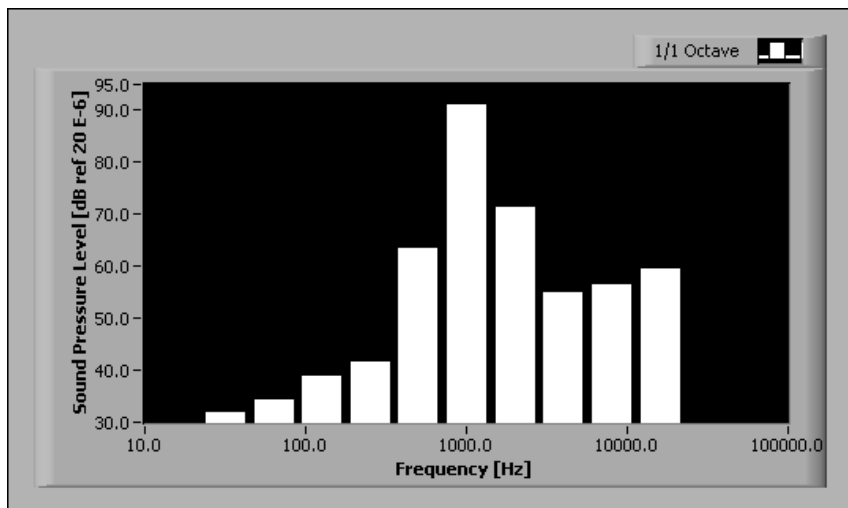
Figure 9-3. Full-Octave Spectrum

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. Figure 9-4 shows an octave filter, a filter whose passband covers one octave.
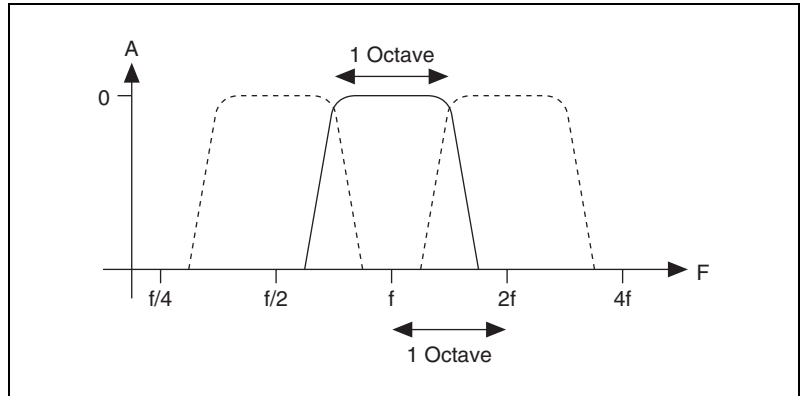
Figure 9-4. Octave Filter

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

The quality constant Q is defined as the ratio of the bandwidth over the center frequency of the filter. Q remains constant across all octave bands for octave filters. For example, an octave filter with a center frequency of 1,000 Hz leads to the following bandedge frequencies:

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

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

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

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

where

f_1 and f_2 are bandedge frequencies.

BW is the bandwidth.

Q is the quality constant.

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

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

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

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

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

where

f_1 and f_2 are bandedge frequencies.

BW is the bandwidth.

Q is the quality constant.

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

Octave filter resolution is limited, being that there are 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 9-5 shows the 1/3 octave response at frequencies of 500 Hz, 630 Hz, and 800 Hz.
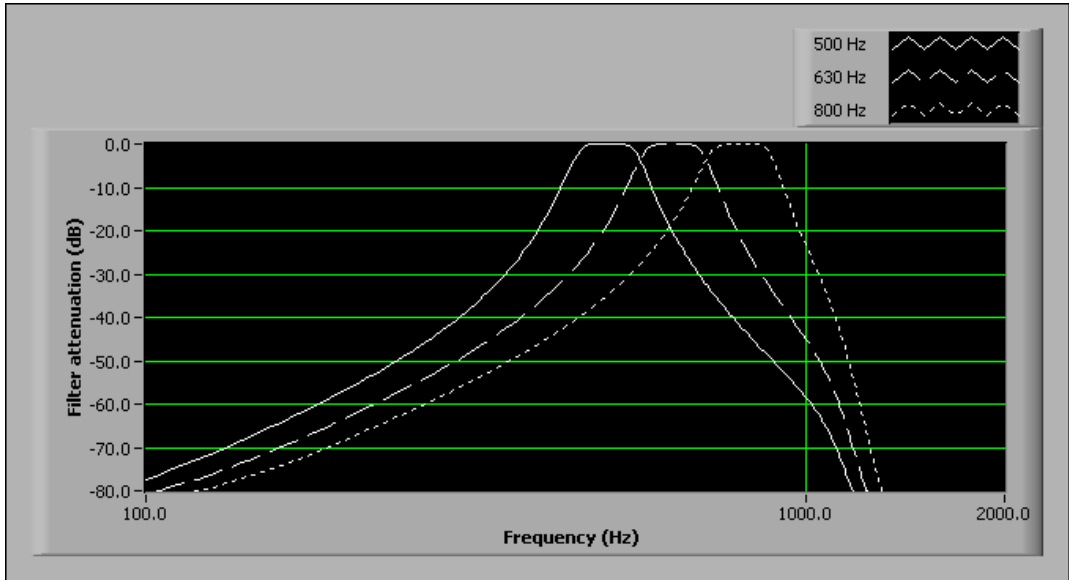
Figure 9-5. Response of 1/3 Octave Filters

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 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 you can consider the complete fractional-octave measurement valid. In the Sound and Vibration Toolkit, settling time is defined as five divided by the bandwidth of the filter used for the lowest frequency band. The **filter 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 averaging

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 8, *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 9-6 illustrates the block diagram for a VI performing 1/3 octave analysis in the 20Hz–20kHz range using fast exponential averaging.

Figure 9-6. 1/3 Octave Analysis with Fast Exponential Averaging VI

Note In order to use exponential averaging, you must connect the **exp avg settings** parameter to a control or constant. Refer to the *LabVIEW Help* for information about the **exp avg settings** control.

Refer to Chapter 8, *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 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 8,
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 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 Hz–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
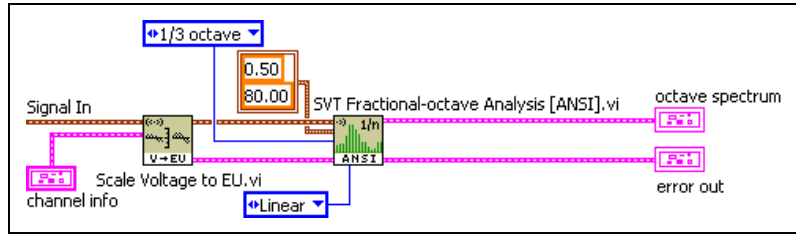bandwidth equal to 1/3 octave, as illustrated in Figure 9-7.

Figure 9-7. Performing 1/3 Octave Analysis Outside Standard Bands

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 bandpass filter is defined by the following equation.

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

where

f_i is the center frequency of the ith 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 bandpass 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 ith 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 device, 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 bandpass filter is 3, which leads to a Type 1-D filter.

For octave filters, choose a sampling frequency at least three 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 Sound & Vibration controls palette contains two graphs designed to display octave results, the Octave Graph and the Multiplot Octave Graph. Refer to the *Sound and Vibration Toolkit 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.

To modify an XY graph to display octave results, complete the following steps:

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.

7. Right-click the Plot Legend.

8. Select **Interpolation** and the style with no line connecting the points.

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 *LabVIEW Help* for more information about the SVT Weighting Filter (octave) VI. Refer to Chapter 5, *Weighting Filters*, for more information about weighting filters.

10

Frequency Analysis

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

Refer to the *Sound and Vibration Toolkit 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 Fourier Transform (FFT) sizes.

Note For simplicity, the remainder of this document uses the term FFT to denote both the FFT and the DFT.

Figure 10-1 illustrates the relationship of frequency analysis to the sound and vibration measurement process.
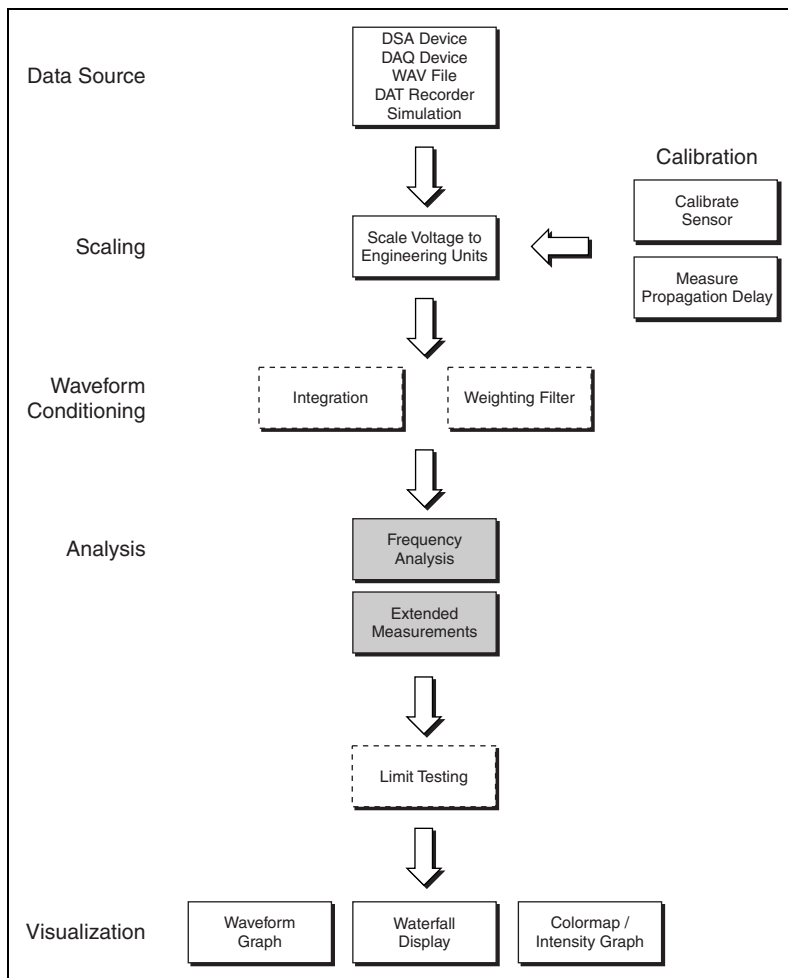
Figure 10-1. Relationship of Frequency Analysis to the Sound and Vibration
Measurement Process

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. Compute T with 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.

Compute N with 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*. Calculate the frequency resolution with 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 time increases.

Maximum Resolvable Frequency

The sampling rate of the time waveform determines the maximum resolvable frequency. According to the Shannon Sampling Theorem, the maximum resolvable frequency must be half the sampling frequency. To calculate the maximum resolvable frequency use the following equation:

$$f_{max} = f_{Nyquist} = \frac{f_s}{2}$$

where

f_{max} is the maximum resolvable frequency.

$f_{Nyquist}$ is the Nyquist frequency.

f_s is the sampling frequency.

Minimum Resolvable Frequency

The minimum resolvable frequency is 0 (DC). The term *baseband analysis* is often used to describe analysis from 0 to $f_{Nyquist}$.

Number of Spectral Lines

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

Directly specify the number of lines in the spectrum for the Zoom FFT VIs in the Sound and Vibration Toolkit. Specify the number of data samples to control the number of spectral lines for the Baseband FFT and the Baseband Subset VIs.

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

Table 10-1 summarizes the relationship of time-domain specifications to frequency-domain parameters.

Table 10-1. Time-Domain Specifications to Frequency-Domain Parameters

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

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 in Table 10-2 if you prefer to specify the spectrum parameters and determine the required data-acquisition parameters from these specifications.

Table 10-2. Frequency-Domain Specifications to Time-Domain Parameters

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

where

> f_{max} is the maximum resolvable frequency.
>
> E_b is the effective bandwidth.
>
> *# lines* is the number of lines in the spectrum.
>
> *df* is the frequency resolution.

In Tables 10-1 and 10-2, 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. Acquire these same 1,024 samples on an instrument with a standard $\frac{1}{2.56}$ effective bandwidth, then use the equation in Table 10-1 to find the expected number of alias-free lines in the computed spectrum. Complete the necessary calculations with the following equations:

$$\# \, lines \; = \; E_b \cdot N$$

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

E_b is entirely a hardware property. However, mathematically, you can use the FFT to compute the frequency spectrum up to the Nyquist frequency. Remember to 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. Use the frequency range to limit the analysis to the alias-free region of the spectrum with FFT subset and zoom FFT measurements. Use the SVT Get Spectrum Subset VI to limit the analysis to the alias-free region of the spectrum with baseband FFT measurements.

✎ **Note**   Table 10-1 shows that 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. However, many stand-alone instruments are operated by specifying the frequency range of interest and the number of lines in the FFT. Table 10-2 shows how a stand-alone instrument uses the frequency range of interest and the number of lines in the FFT to determine an appropriate sampling frequency and block size.

Increasing Frequency Resolution

Increasing the frequency resolution helps you distinguish two individual tones that are close together. For example, if you analyze a signal that contains two tones at 1,000 Hz and 1,100 Hz, use a sampling frequency of 10,000 Hz. Acquire data for 10 ms and the frequency resolution is 100 Hz. Figure 10-2 shows the results of this analysis. In this particular case, you cannot distinguish one tone from the other.
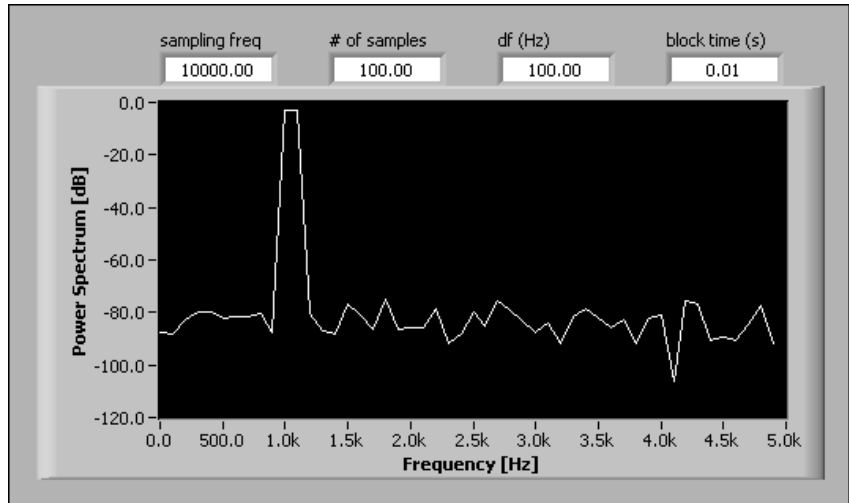
Figure 10-2. Power Spectrum Obtained with an Acquisition Time of 10 ms

Increase the acquisition time to 1 s to achieve a frequency resolution of
1 Hz. Figure 10-3 shows the results obtained with a 1 s acquisition time.
You can distinguish the individual tones with the increased acquisition
time.
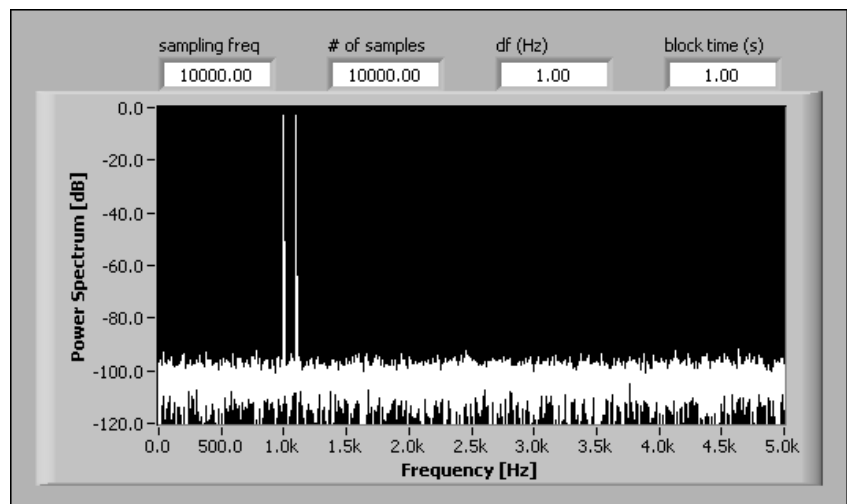
Figure 10-3. Power Spectrum Obtained with an Acquisition Time of 1 s

The following strategies achieve a finer frequency resolution:

- Decrease the sampling frequency, f_s. Decreasing f_s usually is not practical because decreasing f_s reduces the frequency range.

- Increase the number of samples, *N*. Increasing *N* yields an increased number of lines over the original frequency range.

Implement the decreased f_s strategy with zoom FFT analysis. 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.

Refer to the *Zoom FFT Analysis* section and the *Subset Analysis* section for 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 Toolkit.

Zoom FFT Analysis

In some applications, you need to obtain the spectral information with very fine frequency resolution over a limited portion of the baseband span. In other words, you must zoom in on a spectral region to observe the details of that spectral region. 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 you acquire the required number of points. Because the transform operates on a decimated set of data, you only need to compute a relatively small spectrum. The data is accumulated, so 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

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

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

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.

Acquire a sine wave at 1,390 Hz with an National Instruments DSA device and the VI displayed in Figure 10-4.
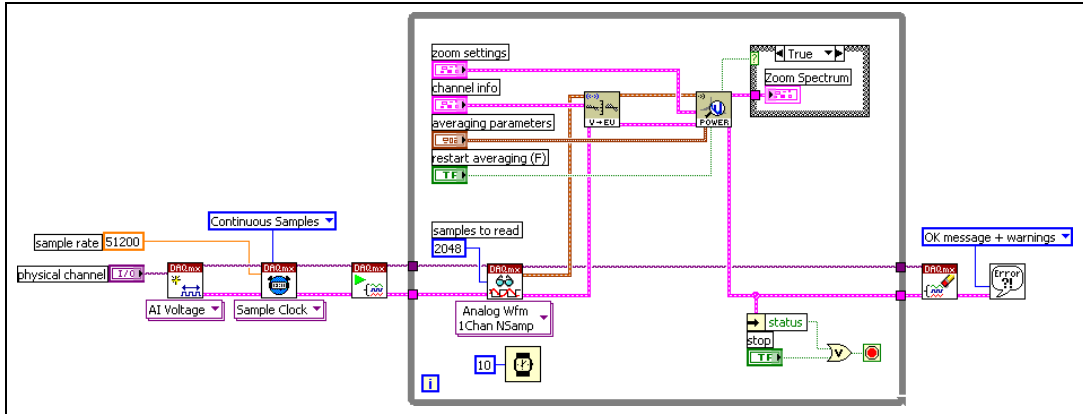
Figure 10-4. Performing a Zoom Power Spectrum Measurement

Acquire the signal at 51.2 kHz. The VI reads the data in blocks of 2,048 samples. Compute the frequency resolution of this measurement using baseband analysis with the following equation:

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

Use the SVT Zoom Power Spectrum VI located on the Zoom FFT palette to analyze a narrow band with a much finer frequency resolution. Figure 10-5 shows the result of limiting the measurement to the frequency band between 1 kHz and 2 kHz and computing 400 lines. Derive the frequency resolution of the computed spectrum with the following equation:

$$\frac{2,000 - 1,000 \text{ Hz}}{400 \text{ lines}} = 2.5 \text{ Hz}$$
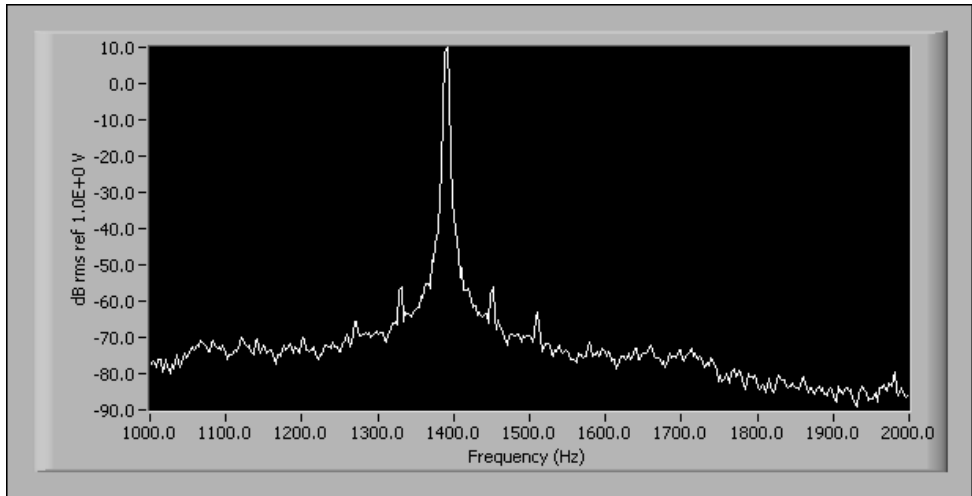
Figure 10-5. Zoom Power Spectrum Measurement Results

Zoom Settings

Figure 10-6 shows the **zoom settings** control used to acquire the zoom measurement results displayed in Figure 10-5. Use this control to specify the frequency range, window, number of lines, and percent overlap used in the zoom analysis. Refer to the *LabVIEW Help* for information about the **zoom settings** control.
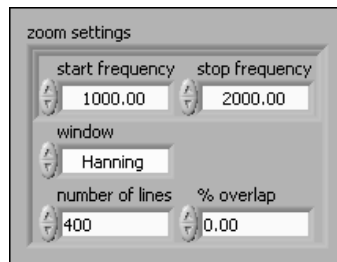
Figure 10-6. Zoom Settings Control

Subset Analysis

The Baseband Subset VIs located on the Baseband Subset palette allow you to compute a subset of the baseband FFT measurement. Subset analysis uses the 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 Baseband Subset VIs algorithm 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**. The **frequency range** parameter 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 f_{max} yields the same spectrum as the corresponding Baseband FFT VI. If you set the stop frequency to –1, the baseband subset VIs return the Nyquist frequency as the highest frequency in the computed spectrum.

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.
- The analysis of a narrow subset of the baseband span requires better processing 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:

- Baseband FFT
- Baseband Subset
- 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 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 measurement 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 measurement 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

Use the Frequency Analysis VIs to perform a single-shot or continuous acquisition, as illustrated by the block diagrams in Figures 10-7 and 10-8.
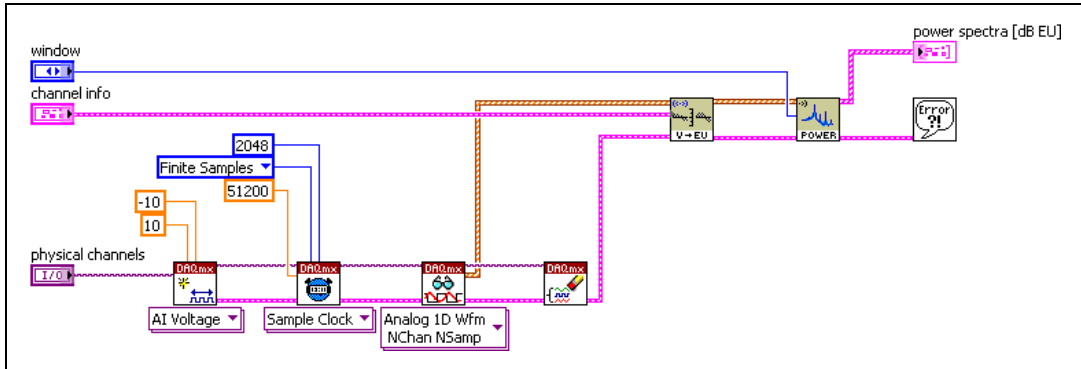
Figure 10-7. Single-Shot Acquisition
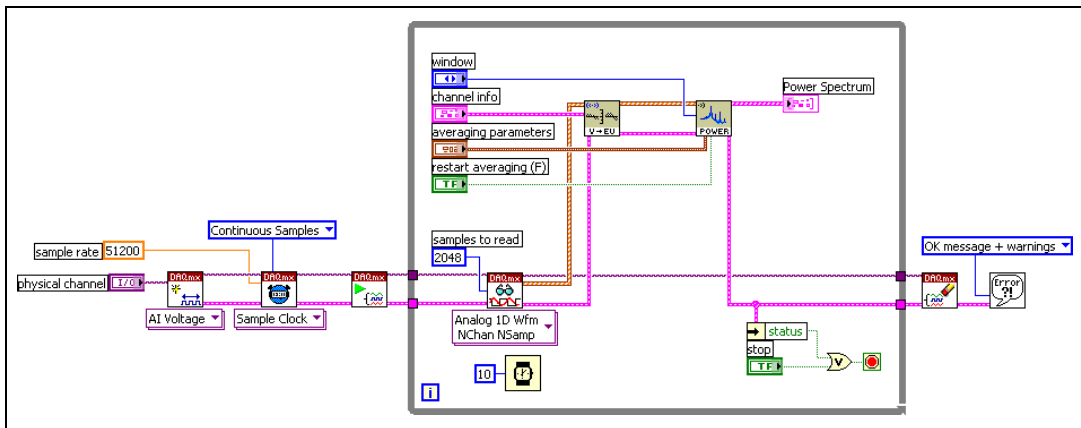
Figure 10-8. Continuous Acquisition

Note   In this example, the output of the read VI is an array of waveforms. You can wire the output array from the read VI directly to the SVL Scale Voltage to EU VI. If the read VI output array contains only a single waveform, use the instance of the polymorphic read VI that returns only a single channel before wiring the waveform to the SVL Scale Voltage to EU 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, as well as others.

Use the Frequency Analysis VIs to make the following dual-channel measurements:

- Frequency response computes the transfer function of the response to the stimulus. 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. Use the cross spectrum in some advanced analyses. Dynamic data often is stored in terms of cross spectra. 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. 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 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 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. Several stimulus signals are valid sources. 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 PXI-4461 DSA device is used to generate the stimulus signal. The stimulus signal is white noise. The DUT is a notch filter centered at 1 kHz. Figure 10-9 illustrates the connection scheme used in this example.
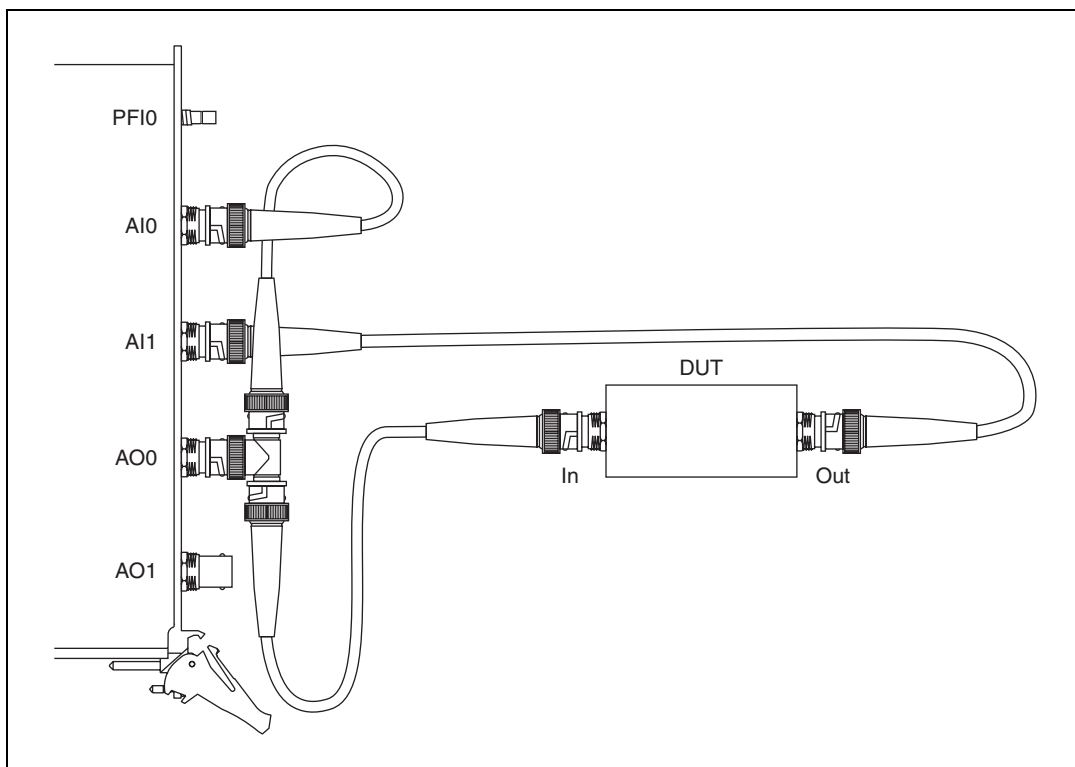
Figure 10-9. Connection Scheme for FRF Measurement with an NI 4461 Device

The DSA device converts the stimulus signal from digital to analog. Analog output channel 0, AO0, sends the stimulus signal to the DUT. Analog input channel 0, AI0, receives the stimulus signal. Analog input channel 1, AI1, receives the response of the DUT.

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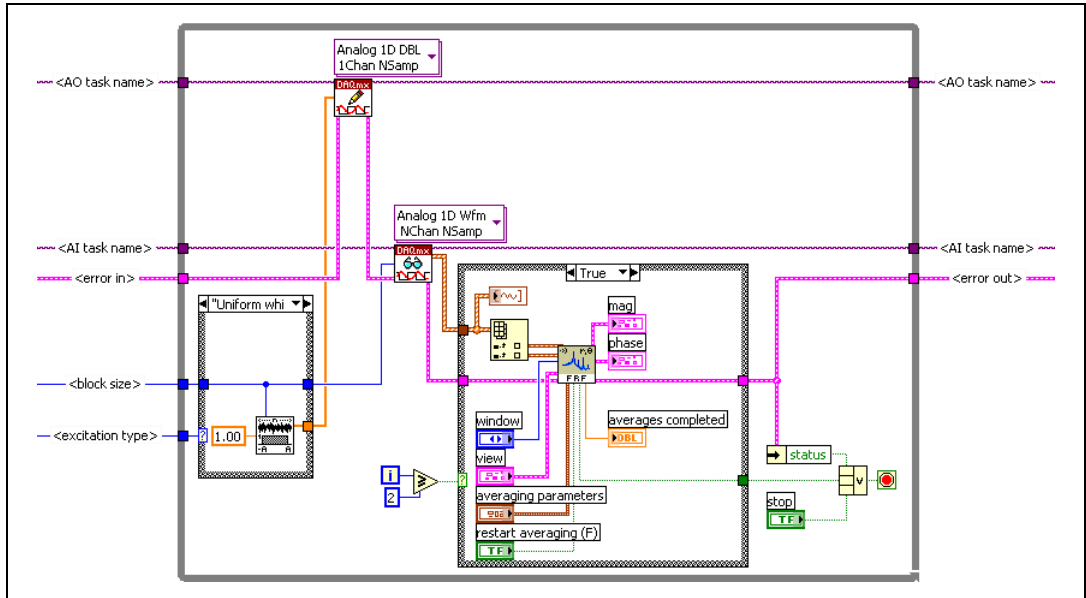
Figure 10-10. FRF Block Diagram

The **AO physical channel** control is set to Dev1/ao0. The **AI physical channels** control is set to Dev1/ai0:1. 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 automatically configured by NI-DAQmx.

The While Loop in Figure 10-10 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, AO0 in Figure 10-9. Analog input channel 0, AI0 in Figure 10-9, acquires the same number of samples as the buffer that is generated. Simultaneously, analog input channel 1, AI1 Figure 10-9, acquires the response signal from the DUT. Figure 10-11 shows the measured time-domain stimulus and response signals.
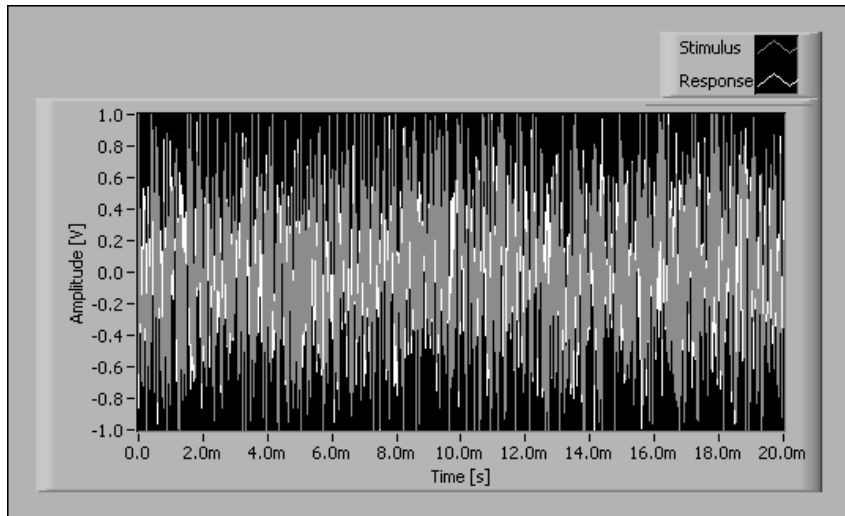
Figure 10-11. Time-Domain Stimulus and Response Signals

It is difficult to interpret the time-domain signals in Figure 10-11. Looking at the results in the frequency domain provides more insight into the DUT dynamic response.

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 10-10 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 400 alias-free spectral lines up to 20 kHz.

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 10-12 shows the frequency response of the notch filter measured without averaging.
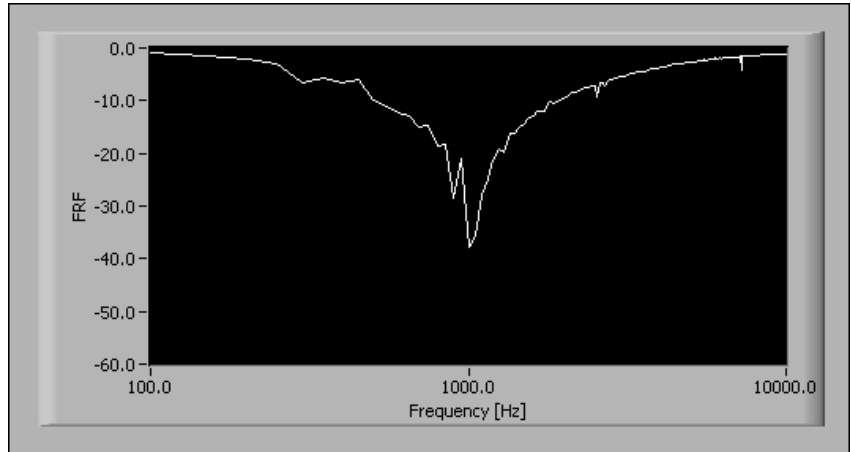
Figure 10-12. Frequency Response without Averaging

FRF with Averaging

You can use averaging to improve the FRF measurement. Figure 10-13 shows the frequency response obtained by applying RMS averaging over 100 measurements.
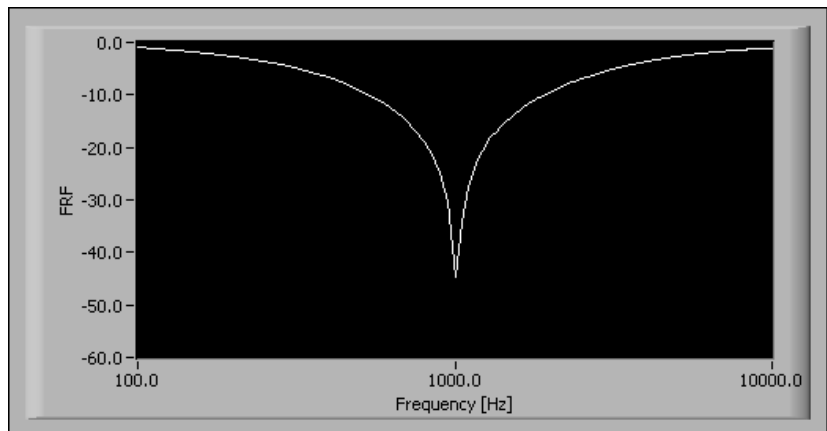
Figure 10-13. Frequency Response with Averaging

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

Refer to the *Averaging Parameters* section 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 meant 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 10-13 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. Figure 10-14 displays the averaged measurement obtained with the new block size.
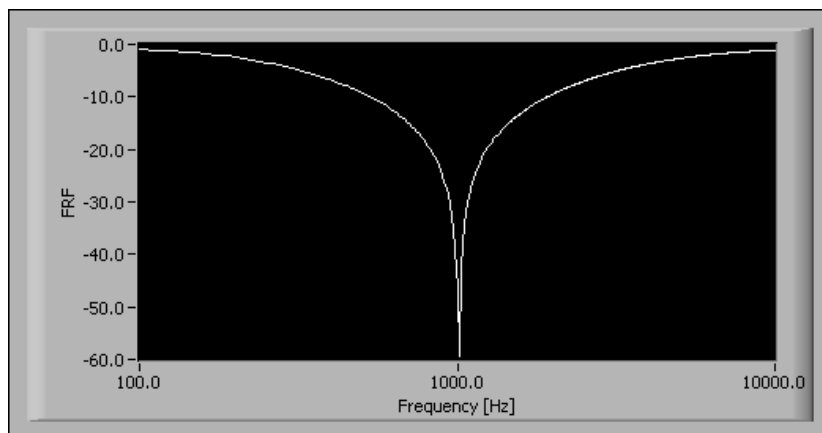
Figure 10-14. Frequency Response with 4,096 Samples

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 10-15 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.
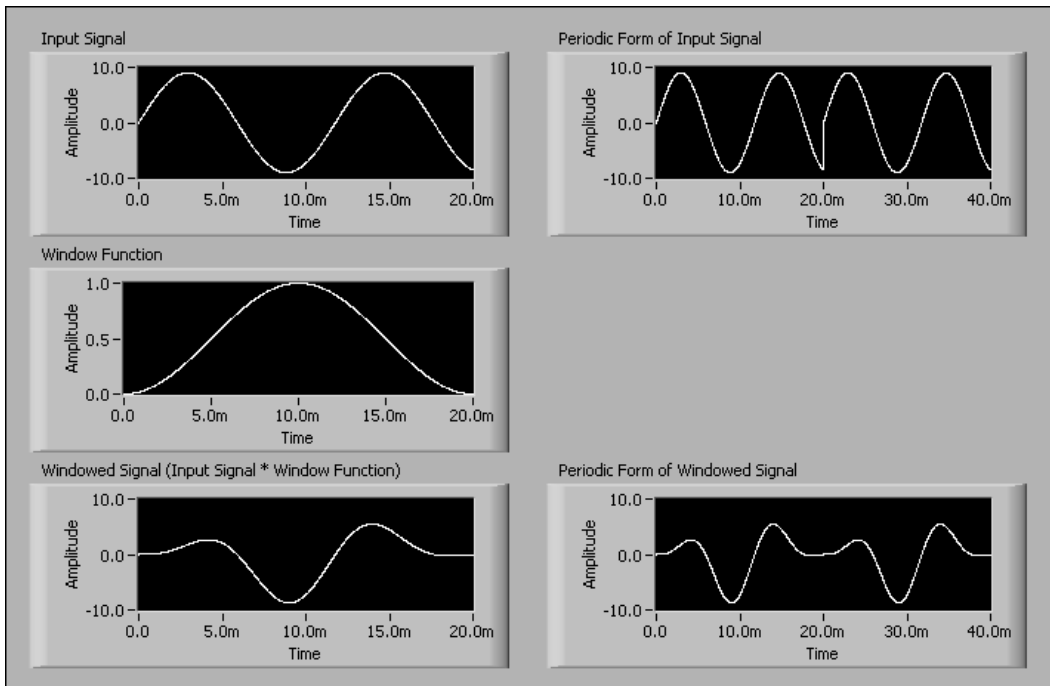
Figure 10-15. Effect of Windowing on Periodicity

The windows supported by the Sound and Vibration Toolkit and their equivalent noise bandwidths (ENBW) are listed in Table 10-3. ENBW is a property of the window applied to the signal.

Table 10-3. 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
4 Term Blackman-Harris	2.00
7 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 in Figure 10-16 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.
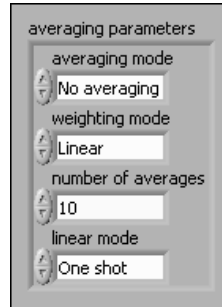
Figure 10-16. Averaging Parameters Control

Refer to the *LabVIEW 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 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:

FFT spectrum $\sqrt{\langle X^* \cdot X \rangle}$

power spectrum $\langle X^* \cdot X \rangle$

cross spectrum $\langle X^* \cdot Y \rangle$

frequency response $H1 = \dfrac{\langle X^* \cdot Y \rangle}{\langle X^* \cdot X \rangle}$

$H2 = \dfrac{\langle Y^* \cdot Y \rangle}{\langle Y^* \cdot X \rangle}$

$H3 = \dfrac{(H1 + H2)}{2}$

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.

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, using 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:

FFT spectrum $\langle X \rangle$

power spectrum $\langle X^* \rangle \cdot \langle X \rangle$

cross spectrum $\langle X^* \rangle \cdot \langle Y \rangle$

frequency response $\dfrac{\langle X \rangle}{\langle Y \rangle}$ (H1 = H2 = H3)

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 10-17 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 1 V_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.
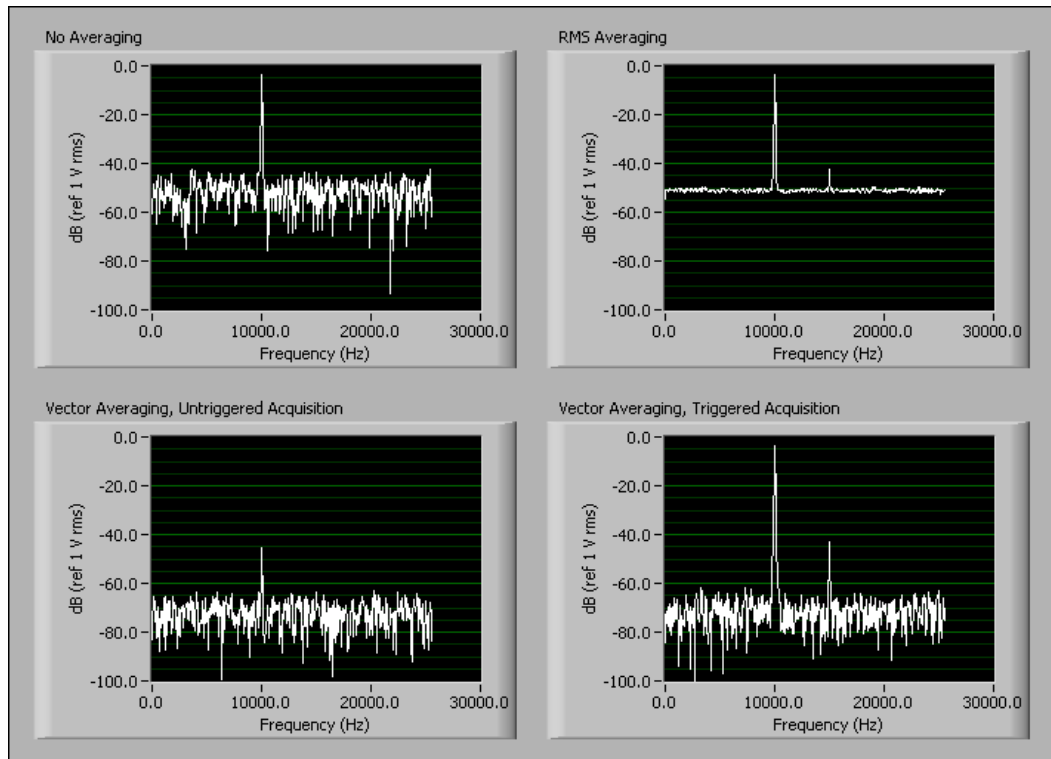
Figure 10-17. Effect of Averaging Mode on Averaged Spectrum

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, Untriggered Acquisition underestimates the energy present at 10 kHz. Also, the tone at 15 kHz is indistinguishable from the noise.

Vector averaging, Triggered Acquisition 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 record. 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:

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

power spectrum $\qquad$ $MAX(X^* \cdot X)$

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 ith 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 (COP) only support RMS averaging and are computed according to the following equations.

Coherence

$$\gamma^2 \;=\; \frac{\left|\langle X^* \cdot Y\rangle\right|^2}{\langle X^* \cdot X\rangle \langle Y^* \cdot Y\rangle}$$

Coherent output
power

$$\text{COP} \;=\; \gamma^2 \langle Y^* \cdot Y\rangle$$

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 Frequency Analysis»Extended Measurements palette to perform the following measurements:

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

Refer to Chapter 4, *Limit Testing Analysis*, for information about limit testing.

Refer to Chapter 5, *Weighting Filters*, for information about weighting.

Refer to Chapter 6, *Integration*, for information about frequency-domain integration.

Refer to the *LabVIEW 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 10-4 shows the equations for computing power in band based on the type of input spectrum.

Table 10-4. Power in Band Equations Based on Input Type

Type of Input Spectrum	Equation for Power in Band
Power Spectrum (PS)	$$\dfrac{\displaystyle\sum_{\text{start frequency}}^{\text{stop frequency}} \text{PS}}{\text{ENBW}}$$
Power Spectral Density (PSD)	$$\left(\sum_{\text{start frequency}}^{\text{stop frequency}} \text{PSD} \right) \cdot df$$
Magnitude Spectrum (MS)	$$\dfrac{\displaystyle\sum_{\text{start frequency}}^{\text{stop frequency}} (\text{MS})^2}{\text{ENBW}}$$

where

ENBW is the equivalent noise bandwidth.

df is the frequency resolution of the measurement.

Spectrum Peak Search

The SVT Spectrum Peak Search VI operates on magnitude and power spectra. The VI estimates the frequency and amplitude of the tones that satisfy the search criteria specified in the **peak search settings** input parameter. For the VI to identify a tone, the spectrum must exceed the specified threshold. Specify the threshold in the same units as the input spectrum.

The SVT Spectrum Peak Search VI uses an algorithm to estimate the actual frequency and amplitude of each tone that meets the search criteria. The algorithm also estimates the frequency and amplitude of tones that are not exactly periodic within the measurement window. The ability to identify nonperiodic tones that are not exactly on-bin allows you to use a smaller block size, therefore a smaller FFT, and still accurately identify tones present within the spectrum.

Unit Conversion

The SVT Unit Conversion VI operates on complex, magnitude, and power spectra. You can use the VI to convert the complex spectra produced by distortion and single-tone measurements to a magnitude or power spectrum more suitable for display on a waveform graph. You also can use the conversion utility to switch between magnitude and power spectra, switch between linear and logarithmic scaling, change the decibel reference for the spectrum, and change the peak units.

Note   You cannot use the SVT Unit Conversion VI to scale the data to engineering units or change the spectrum physical units. Use MAX, the DAQ Assistant, or the SVL Scale Voltage to EU VI to scale your data to the proper engineering units.

When using the SVT Unit Conversion VI, input the **desired scaling** parameters as they should apply to the **converted spectrum** parameter. You do not need to perform units conversions on the decibel reference. The decibel reference always has the same units as the converted spectrum. For example, if you have a magnitude spectrum in V_{rms}, you can convert this to a power spectrum in decibels referenced to 10 V_{pk} by specifying the **desired scaling** parameters shown in Table 10-5.

Table 10-5. Desired Scaling Parameters

desired scaling	
linear/decibels	dB
decibel reference (0: no change)	10
power/magnitude	power
view	peak

11

Transient Analysis

This chapter discusses performing transient analysis with the Transient Analysis VIs located on the Transient Analysis palette.

Refer to the *LabVIEW Help* for information about individual Transient Analysis VIs.

Figure 11-1 illustrates the relationship of transient analysis to the sound and vibration measurement process.
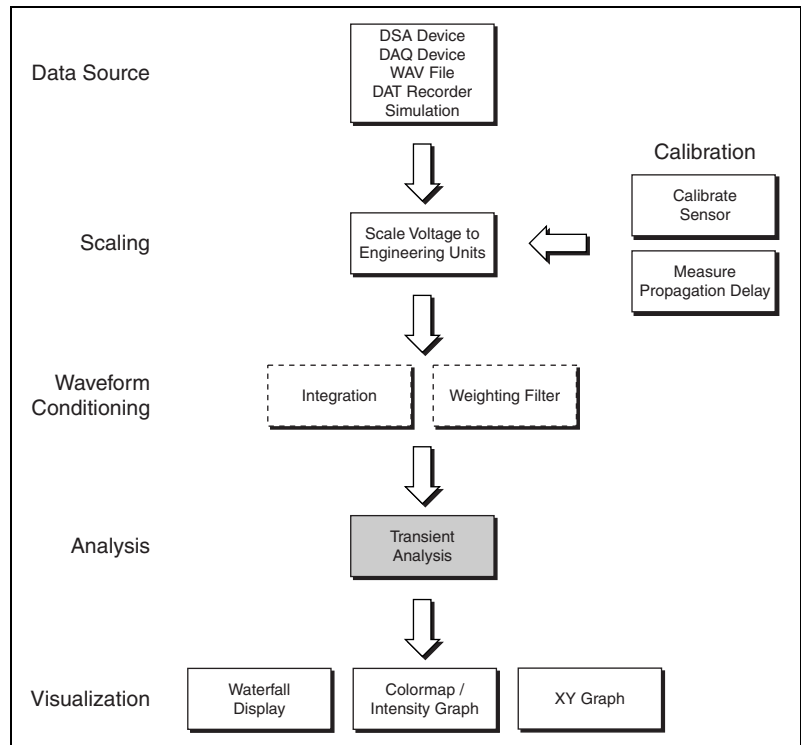
Figure 11-1. Relationship of Transient Analysis to the Sound and Vibration Measurement Process

Transient Analysis with the Sound and Vibration Toolkit

Transient analysis is the analysis of nonstationary signals. The Transient Analysis VIs offer two different techniques for extracting information about transient signals. Use the STFT for signals in which the frequency content changes relatively slowly with time. Use the shock response spectrum (SRS) for shock signals.

You can use the STFT VIs to extract frequency information as a function of time directly from the signal of interest. Additionally, in the case of a rotating machine where a tachometer signal is simultaneously acquired with the signal of interest, the STFT 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 STFT VIs return the information needed to properly scale the axes of the displays. You can pass the information directly to a Waterfall Display VI. Use property nodes for the colormap display. Refer to Chapter 12, *Waterfall Display*, for more information about the waterfall display. Refer to the *LabVIEW Help* for information about the colormap display and property nodes.

You can use the SVT Shock Response Spectrum VI to evaluate the severity of a shock signal. The results generated by the SRS are typically displayed on an XY graph.

Note Other LabVIEW toolkits are available that provide additional transient analysis capabilities. The Order Analysis Toolkit is designed for rotating machinery analysis and monitoring. The Signal Processing Toolkit 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 Toolkit can compute multiple Fourier transforms on the time-domain signal with or without overlapping.

For example, analyze a waveform containing 10 s 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 11-2 shows the signal corresponding to the first 200 ms of the waveform.
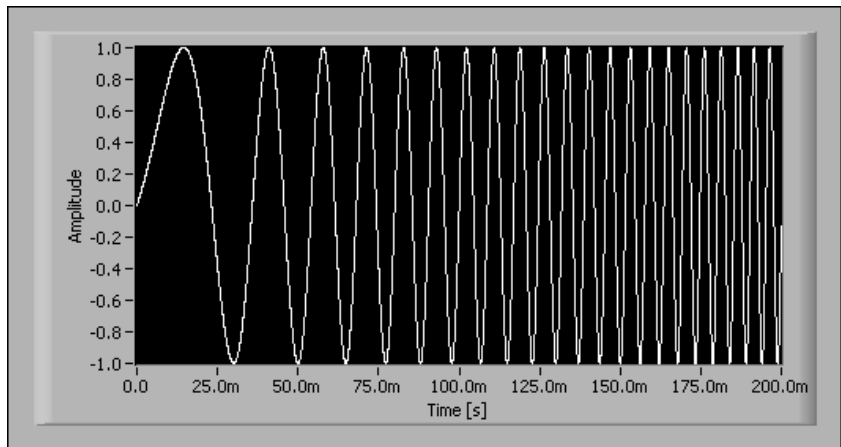
Figure 11-2. Chirp Signal

Figure 11-3 shows the result of applying a baseband FFT on the entire waveform.
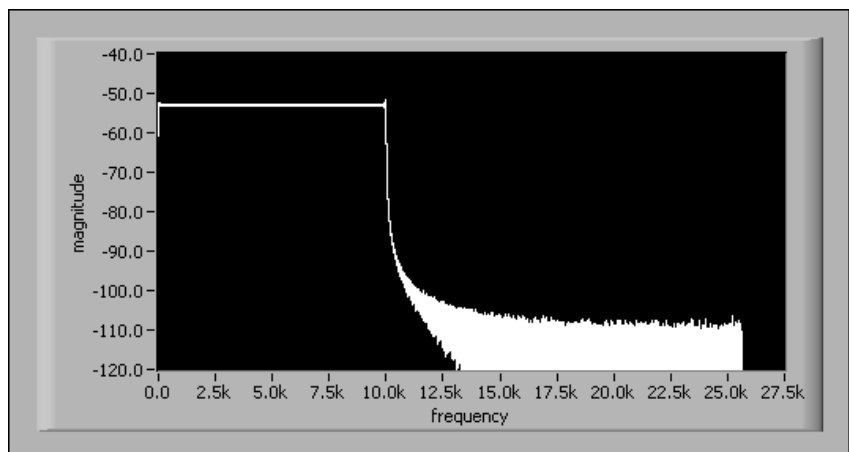
Figure 11-3. Baseband FFT on a Chirp Signal

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 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 versus Time VI.

Subdivide the time-domain signal by configuring the **time segment** control displayed in Figure 11-4.
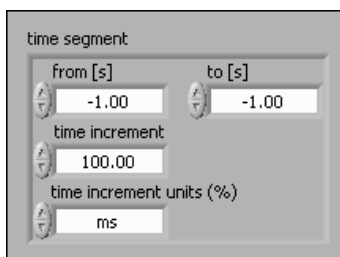
Figure 11-4. Time Segment Control

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.

Create 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 versus 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 512 lines (400 alias-free 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 11-5 shows the result obtained with a 1,024 sample FFT.
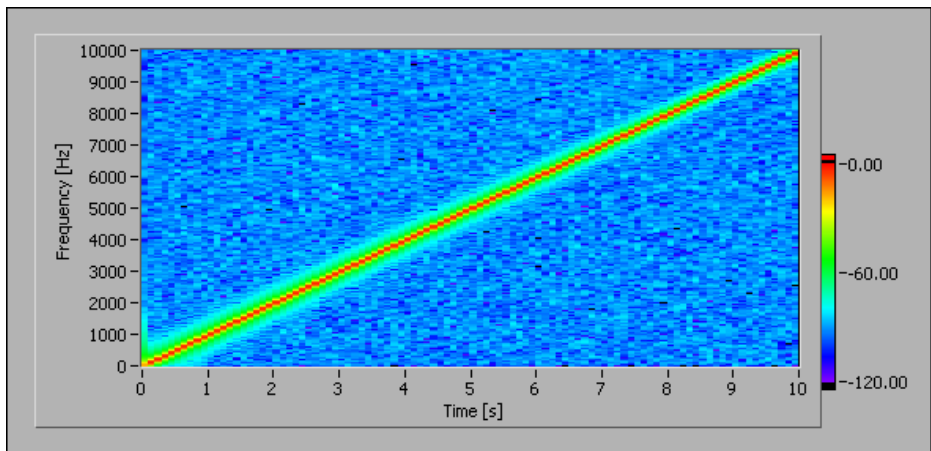
Figure 11-5. STFT Using a 1,024 Sample Block Size

If you select an FFT Block size of 4,096 samples, or 1,600 alias-free lines,
the resolution improves, as illustrated in Figure 11-6. However, the
increased resolution comes with the expense of extra processing.
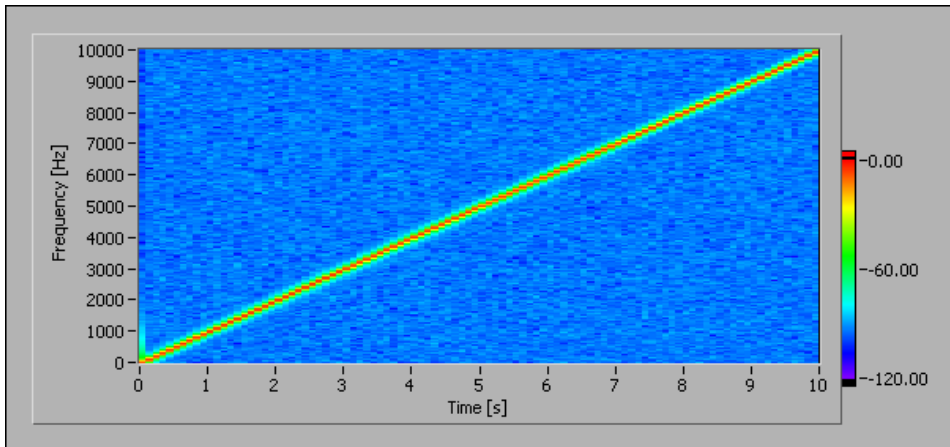
Figure 11-6. STFT Using a 4,096 Sample Block Size

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   Set the desired overlapping rate by specifying **%** in the **time increment units (%)** in the **time increment** control. Refer to Figure 11-4 for the location of this control. 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 11-7 and Figure 11-8 illustrate the overlapping process. Figure 11-7 shows a 50% overlap.
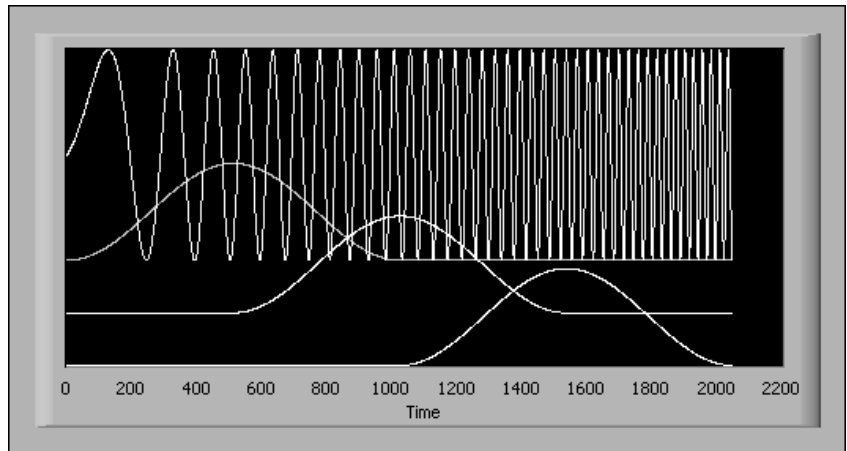
Figure 11-7. 50% Overlap

Figure 11-8 shows the resulting subdivisions when you use a 50% overlap and a Hamming window.
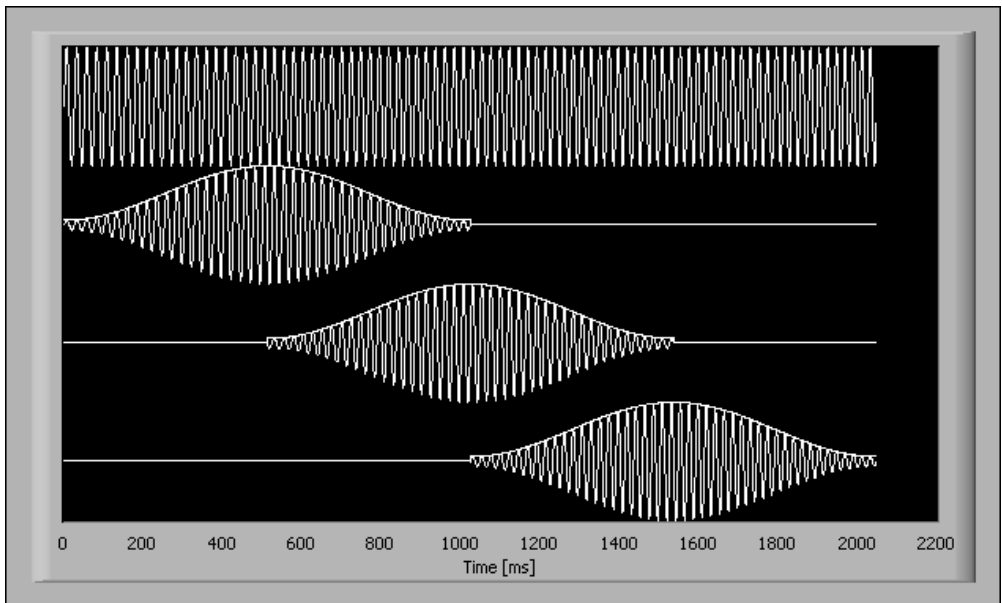
Figure 11-8. Subdivisions of the Time-Domain Waveform

Using the SVT STFT versus Time VI

The following example illustrates how to use the SVT STFT versus Time VI. Figure 11-9 shows the block diagram.
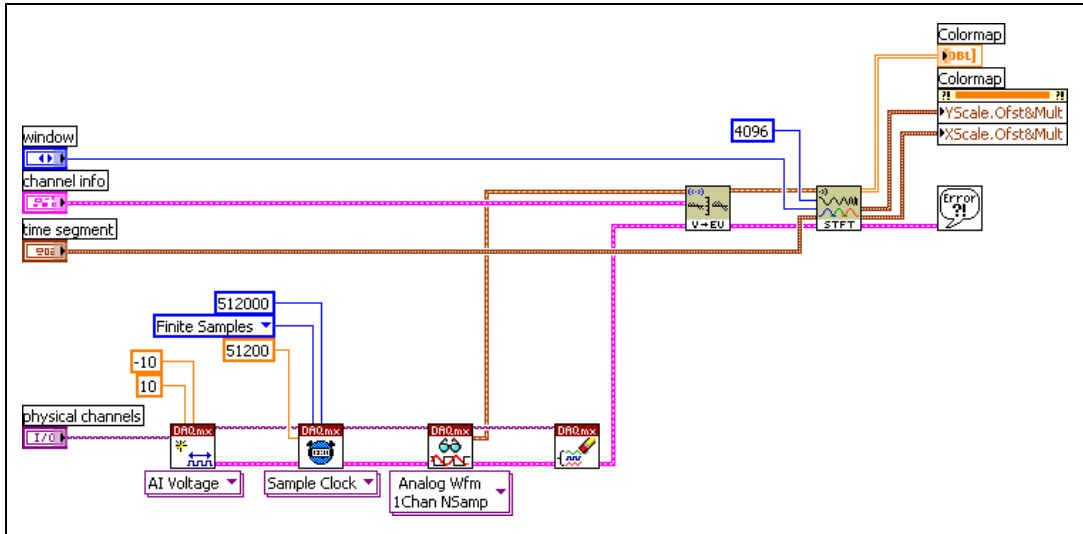
Figure 11-9. Use of the SVT STFT versus Time VI

The example in Figure 11-9 acquires 10 s of data at a sample rate of 51.2 kHz. After scaling, the signal is sent to the SVT STFT versus Time VI. The result is displayed on the intensity graph in Figure 11-10.
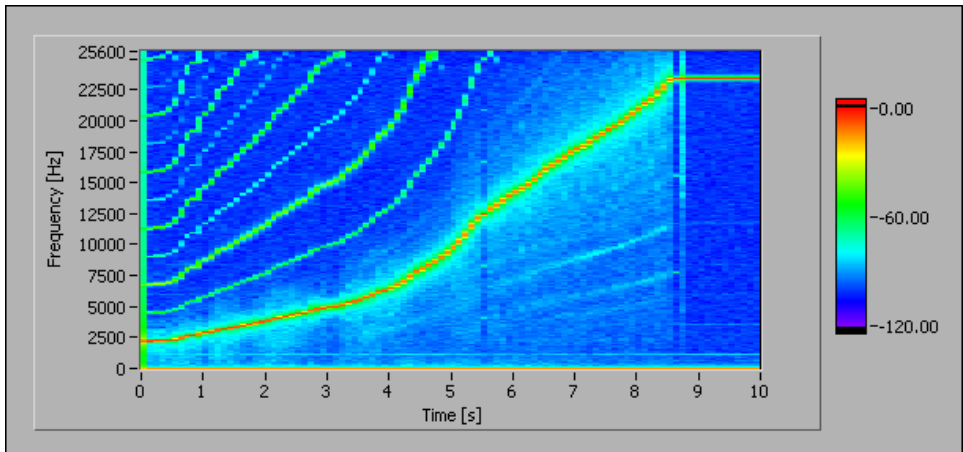
Figure 11-10. STFT versus Time Graph

> **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 s. 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 relevant 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 for graph controls.

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 rotations
per minute (RPM).

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

In this 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 transistor-transistor logic (TTL)
signal. Use the **tach info** control to specify the characteristics of the pulses
generated by the tachometer. Figure 11-11 shows the settings for the **tach
info** control. Refer to the *LabVIEW Help* for more information about the
tach info control.
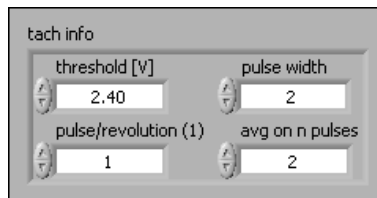
Figure 11-11. Tach Info Control

Figure 11-12 shows a simulated tachometer signal.
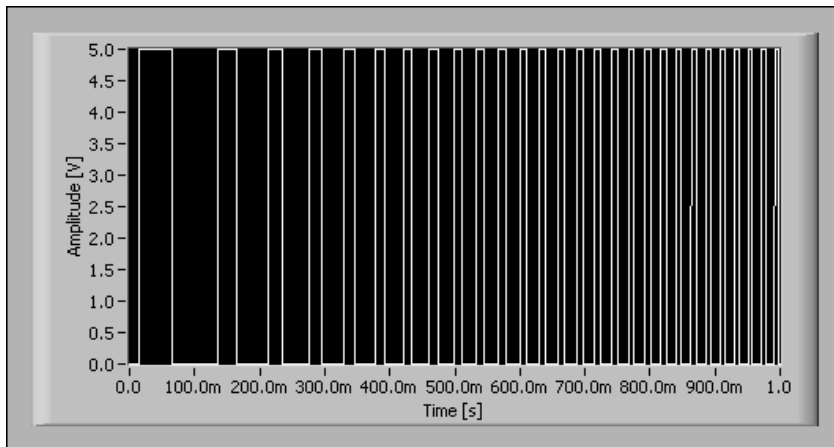
Figure 11-12. Tachometer Signal

You can use the SVT Convert to RPM (analog) VI to measure the rotational speed in RPM as a function of time. Figure 11-13 shows the result obtained with the SVT Convert to RPM (analog) VI and a simulated tachometer signal.
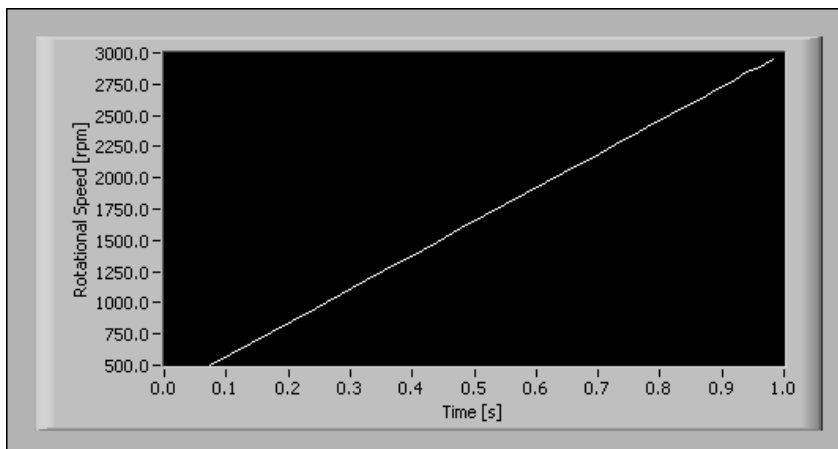
Figure 11-13. Result from SVT Convert to RPM (analog) VI

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 11-14 shows the signal acquired by the microphone.
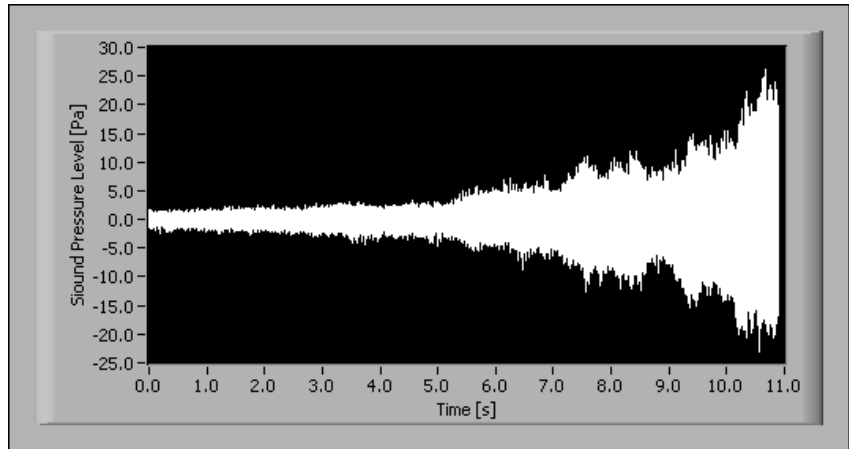
Figure 11-14. Microphone Signal Obtained during Engine Run-Up

The signal from the tachometer is also acquired. The measured tachometer signal is converted to RPM with the SVT Convert to RPM (analog) VI. Figure 11-15 shows the rotational speed as a function of time as computed by the SVT Convert to RPM (analog) VI.
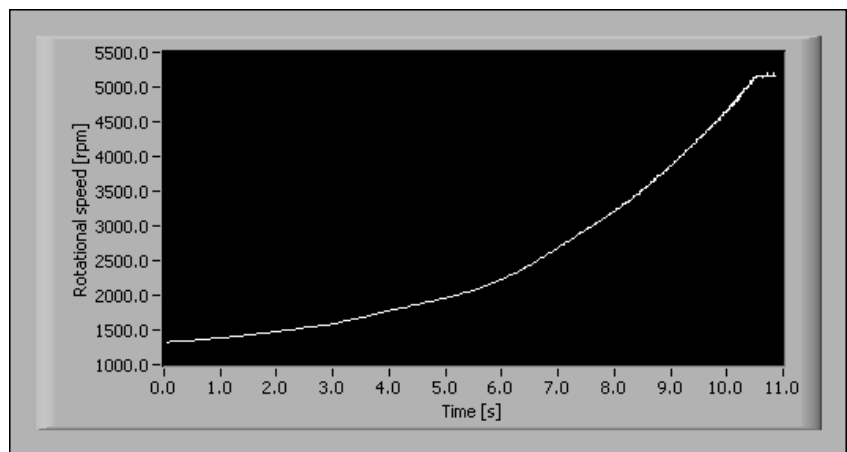
Figure 11-15. Rotational Speed as a Function of Time during Engine Run-Up

Using the SVT STFT versus RPM (analog) VI allows you to measure the frequency content of the signal as a function of the rotational speed of the engine. Figure 11-16 displays the results obtained with the SVT STFT versus RPM (analog) VI on an intensity graph.
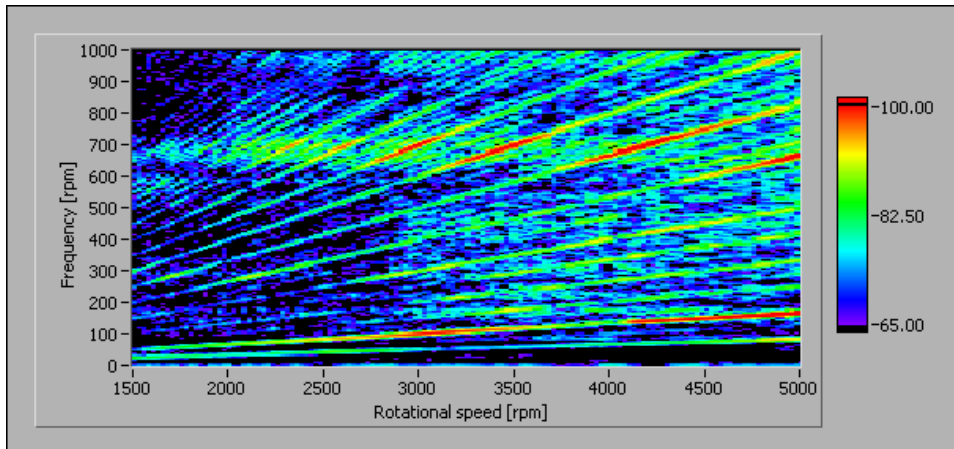
Figure 11-16. Intensity Graph of Sound Pressure Level for an Engine Run-Up

Measuring a Shock Response Spectrum

Obtain the SRS by applying the acquired shock pulse to a series of single degree of freedom (SDOF) systems. Plot the system maximum response as resonance frequency of the system.

An SDOF mechanical system consists of the following components:

- Mass, whose value is represented with the variable m
- Spring, whose stiffness is represented with the variable k
- Damper, whose damping coefficient is represented with the variable c

The resonance frequency, f_N, and the critical damping factor, ζ, characterize a SDOF system, where:

$$f_N = \frac{1}{2\pi}\sqrt{\frac{k}{m}}$$

$$\zeta = \frac{c}{2\sqrt{km}}$$

For light damping where ζ is less than or equal to 0.05, the peak value of the frequency response occurs in the immediate vicinity of f_N and is given by the following equation, where Q is the resonant gain:

$$Q = \frac{1}{2\zeta}$$

Figure 11-17 illustrates the response of a SDOF system to a half-sine pulse with a 10 g acceleration amplitude and 10 ms duration. The top graph shows the time-domain acceleration. The middle graph is the SDOF system response with a 50 Hz resonance frequency. The bottom graph is the SDOF system response with a 150 Hz resonance frequency. In both cases, ζ is 0.05.
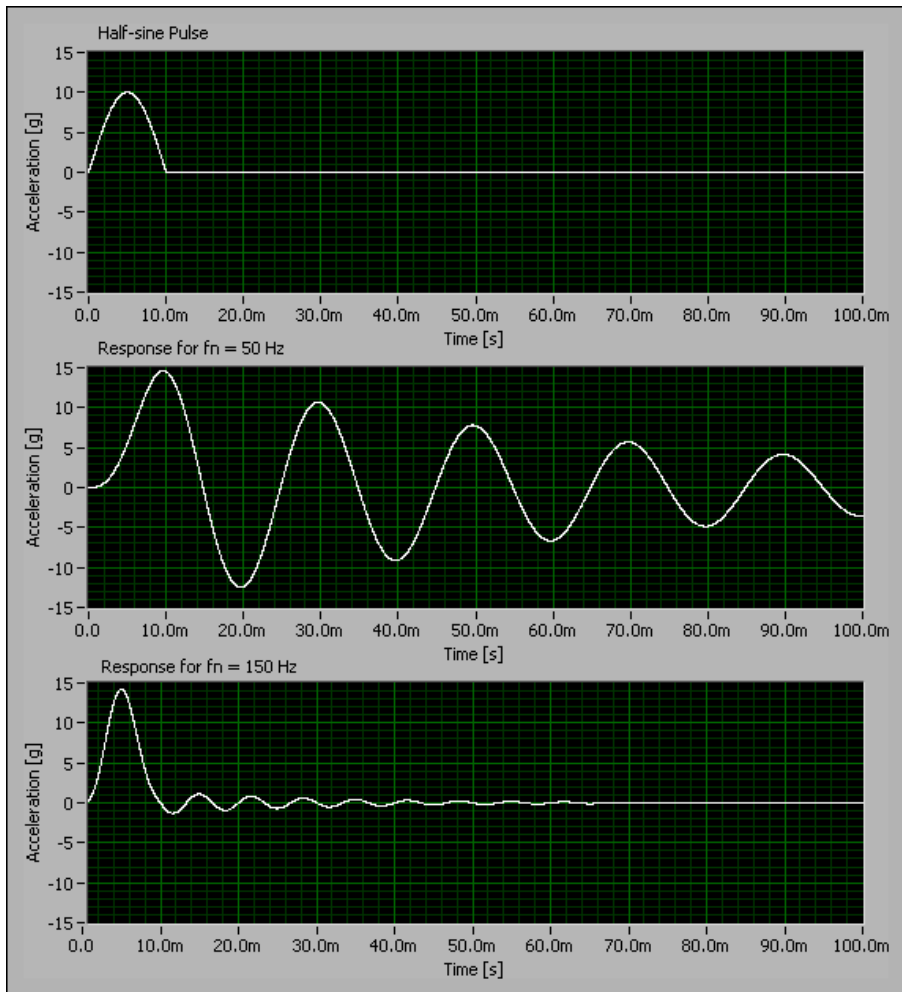
Figure 11-17. SDOF System Response to a Half-Sine Shock

Use the signals shown in Figure 11-17 to construct the SRS. For example, the *maximax*, the absolute maximum response of the calculated shock response signal over the entire signal duration, uses the absolute maximum system response as a function of the system natural frequency. Figure 11-18 illustrates the maximax SRS for the same half-sine pulse.

Note Each computed SRS is specific to the pulse used to perform the measurement.
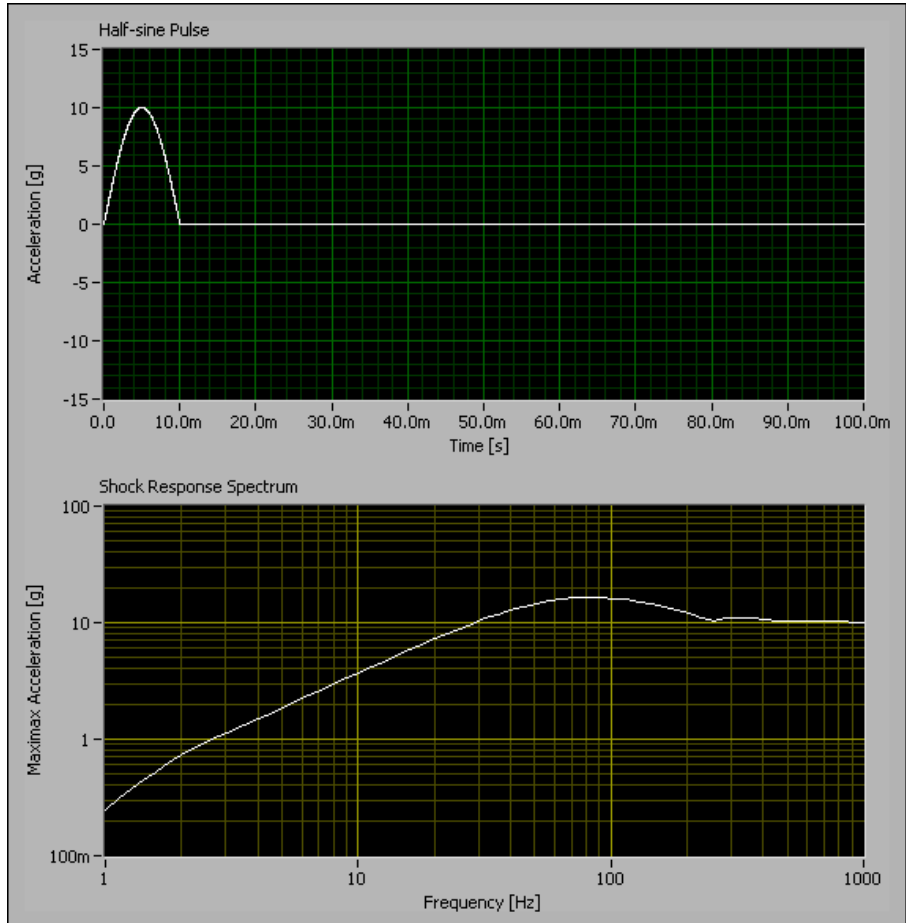
Figure 11-18. Half-Sine Pulse SRS (Maximax)

You can use other types of shock spectra depending on the application. These spectra include the initial shock response from the system response over the pulse duration or from the residual shock spectrum from the system response after the pulse. You can use the positive maximum, the negative maximum, or the absolute maximum response signal value.

The Sound and Vibration Toolkit uses the Smallwood algorithm to compute the SRS. The SVT Shock Response Spectrum VI also offers the ability to preprocess the time-domain signal to improve SRS results. You can remove any DC component or apply a lowpass filter with a selectable cut-off frequency.

The SVT Shock Response Spectrum VI can compute the SRS from the absolute acceleration response or from the relative displacement response. Use the **model** control on the SVT Shock Response Spectrum VI to select the appropriate response.

Figure 11-19 shows how to use the SVT Shock Response Spectrum VI. The example in Figure 11-19 acquires 1,000 samples of data from an accelerometer during a shock. The shock signal triggers the acquisition. The program stores 100 samples before the trigger to properly capture the entire shock signal.
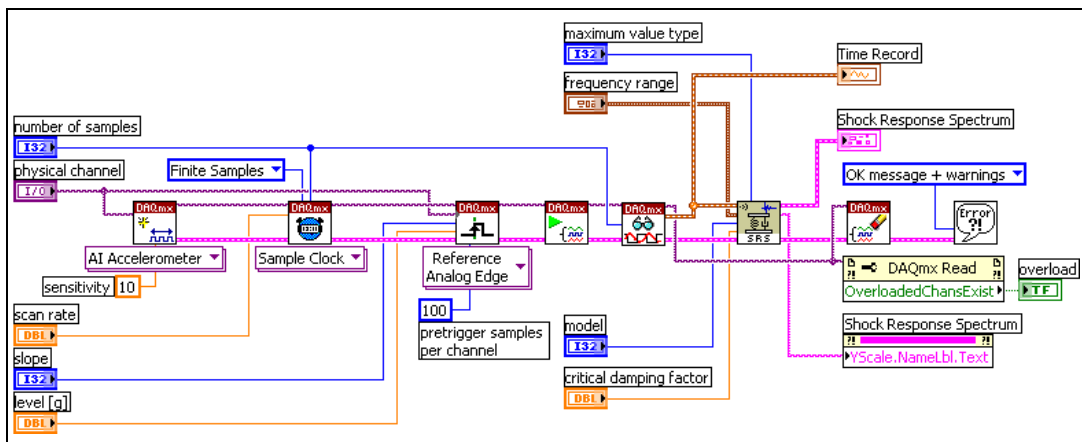
Figure 11-19. Using the SVT Shock Response Spectrum VI

Figure 11-20 displays the acquired time-domain signal and the computed SRS.
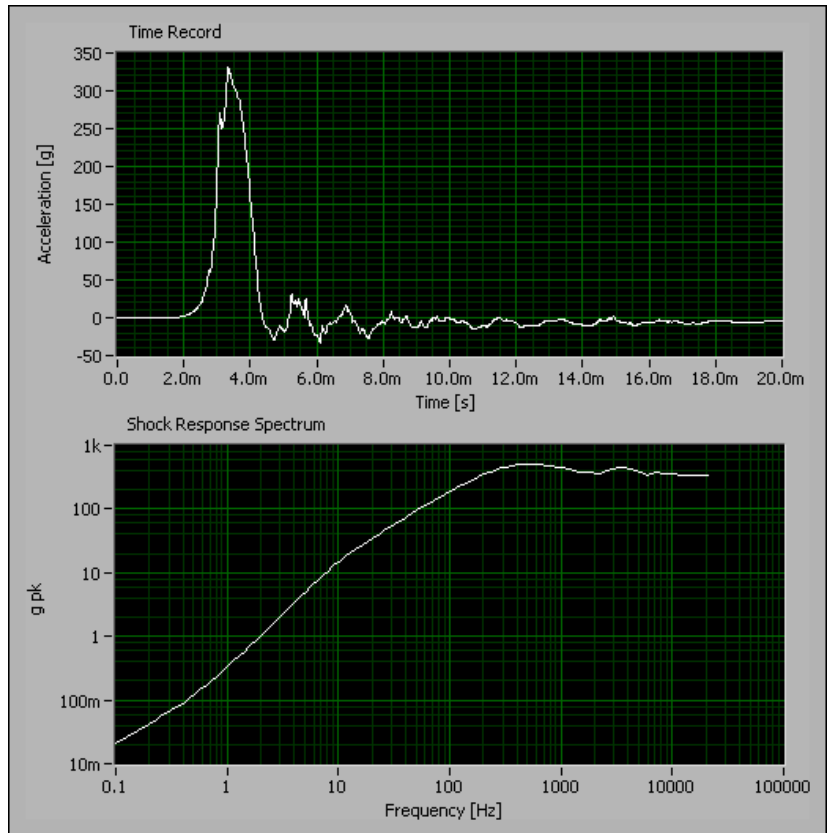
Figure 11-20. Acquired SRS (Maximax)

12

Waterfall Display

This chapter discusses using the Waterfall Display VIs located on the Waterfall Display palette.

Refer to the *LabVIEW 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 nonstationary signals, such as machine run-up, coast down, transients, as well as others.

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 *LabVIEW 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 12-1 shows the **Waterfall** window.
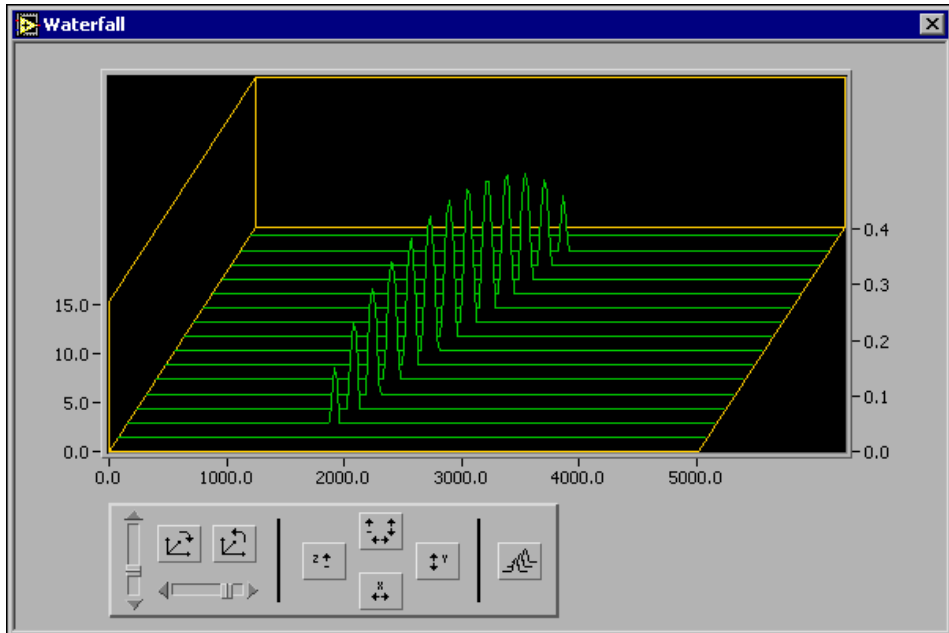
Figure 12-1. Waterfall Display for Frequency Analysis

Initializing the Display

Use the SVL 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 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 SVL 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 SVL Send Data to Waterfall VI is polymorphic and accepts an array of
spectra, such as that returned by a power spectrum, a 2D array, or 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 12-2 shows the block diagram for the VI.
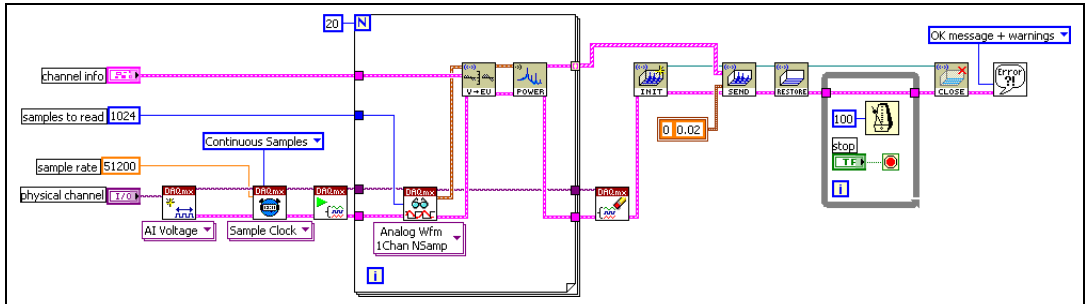
Figure 12-2. Waterfall Display for Frequency Analysis VI Block Diagram

Twenty data blocks of 1,024 samples 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. The array or spectra is sent to the waterfall display. Refer to the *LabVIEW Help* for information about auto-indexing.

Waterfall Display for Transient Analysis

This example illustrates how to use the waterfall display in conjunction with the Transient Analysis VIs. Figure 12-3 shows the block diagram for the example VI.
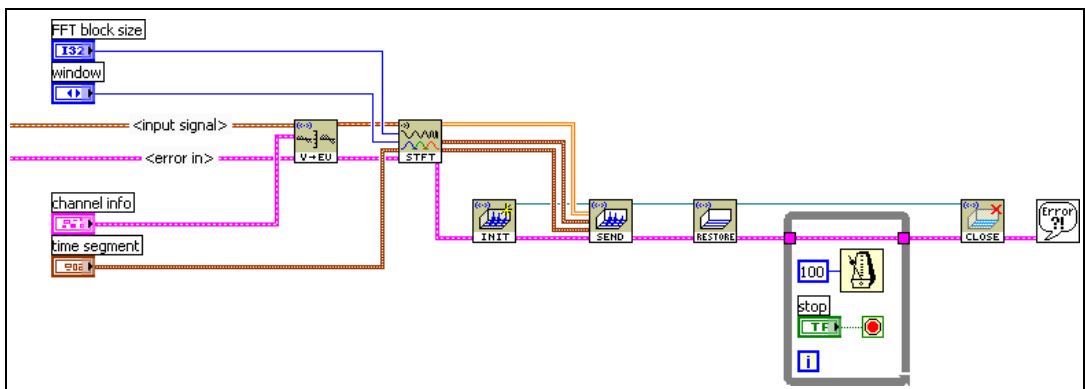
Figure 12-3. STFT VI Block Diagram

The data are scaled and sent to the SVT STFT versus Time VI. The SVT STFT versus 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 SVL Send Data to Waterfall VI. Figure 12-3 shows the 2D array connected directly to the Waterfall VI. 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 SVL Send Data to Waterfall VI to ensure the graph shows the proper scales.

Figure 12-4 shows the result obtained with the STFT VI illustrated in Figure 12-3.
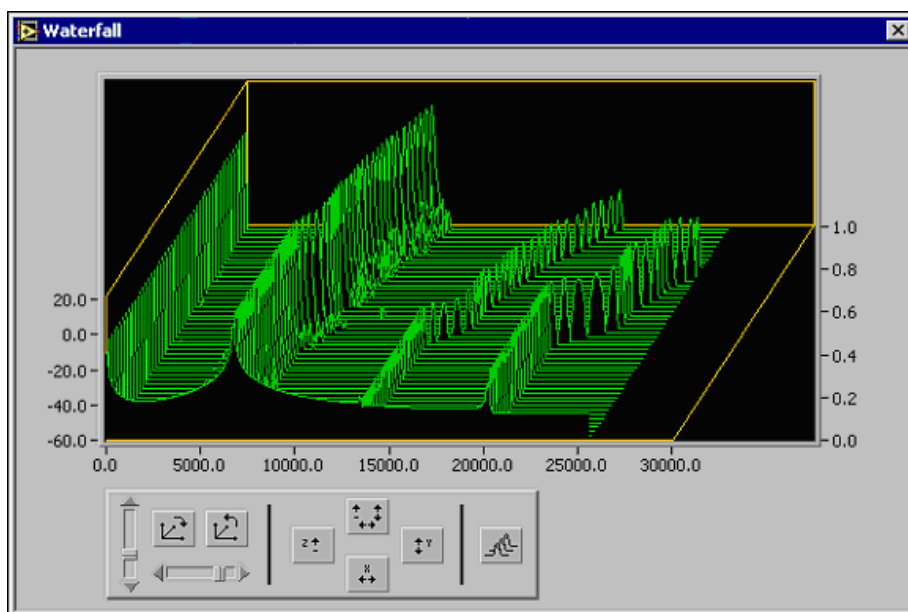
Figure 12-4. STFT Waterfall Display

Refer to Chapter 11, *Transient Analysis*, for information about transient analysis. Refer to the *LabVIEW 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 12-5 shows the block diagram for a VI displaying
octave spectra in a waterfall display.
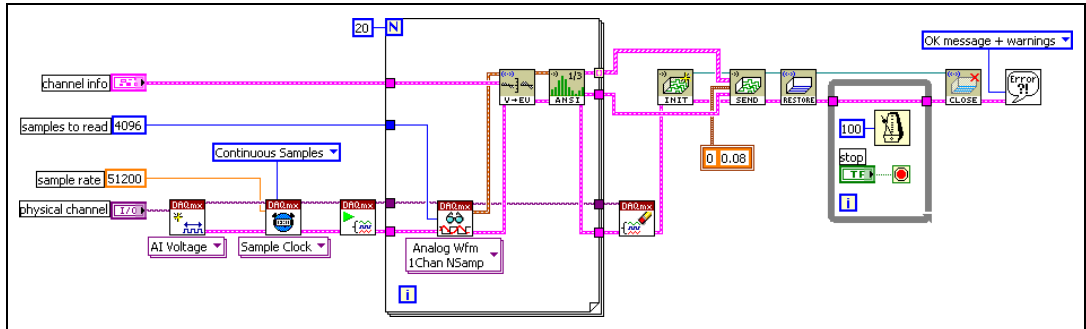
Figure 12-5. Block Diagram for VI Displaying Octave Spectra

Figure 12-6 shows the waterfall display created by the VI in Figure 12-5.
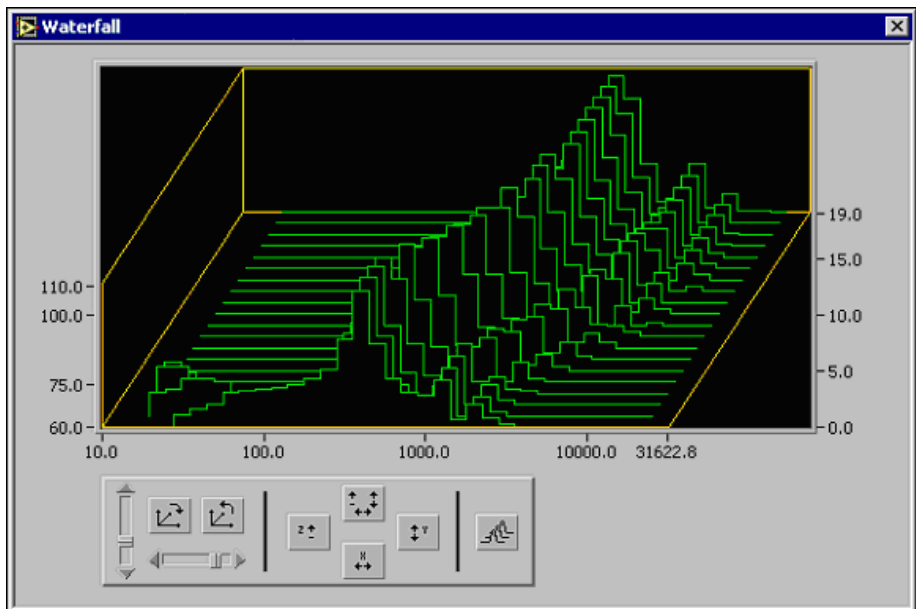
Figure 12-6. Waterfall Display for Octave Spectra Analysis

Customizing the Waterfall Display View

Use the controls in Figure 12-7 to customize the view of your waterfall display.

1	Change vertical perspective	4	Restore view	7	Autoscale y (min/max)
2	Change horizontal perspective	5	Autoscale z (max only)	8	Autoscale all (x, y, and z)
3	Store current view	6	Autoscale x (min/max)	9	Set transparency on/off
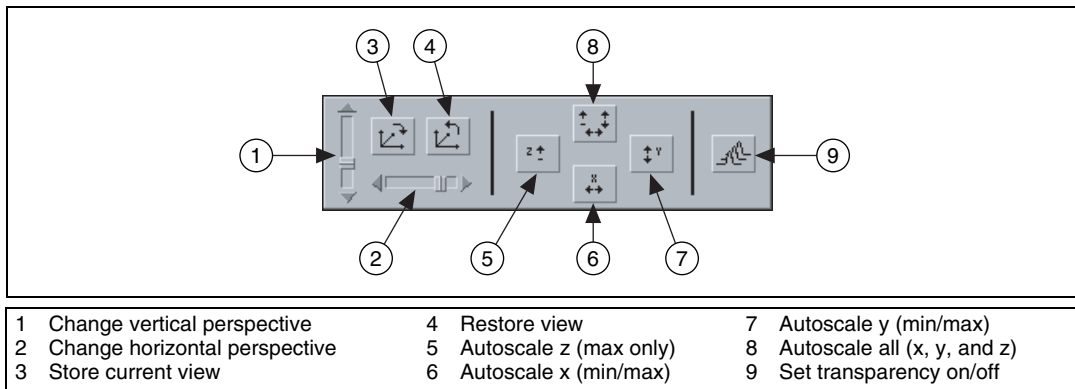
Figure 12-7. Waterfall Graph Controls

You can use the *Change vertical perspective* and *Change horizontal perspective* 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 SVL Close Waterfall Display VI in Figures 12-2, 12-3, and 12-5, closes the reference to the waterfall graph and the **Waterfall** window.

13

Swept-Sine Measurements

This chapter discusses using the Swept-Sine VIs located on the Swept Sine palette.

Refer to the *LabVIEW Help* for information about the individual Swept-Sine VIs.

The swept-sine measurements include dynamic measurements for stimulus level, response level, frequency response (gain and phase), THD, and individual harmonic distortion.

Figure 13-1 illustrates the application flow used when performing measurements with swept sine.
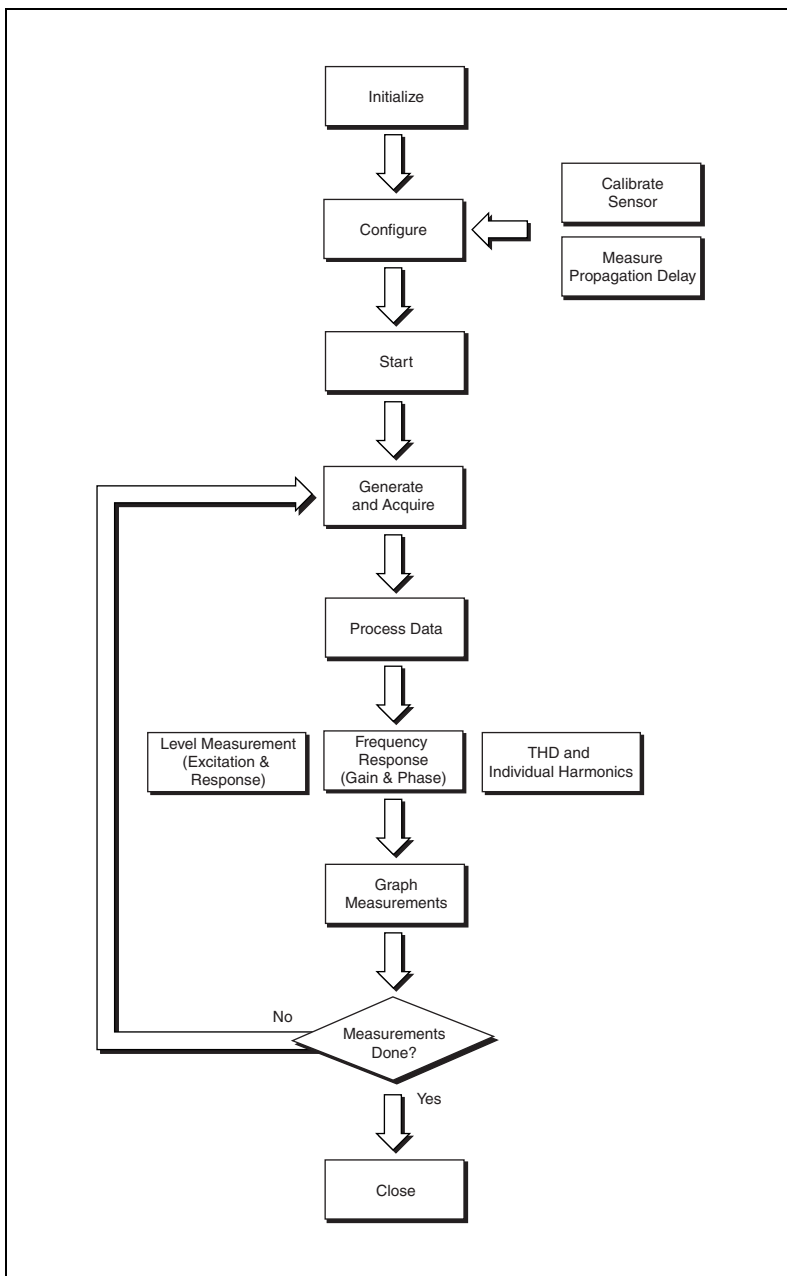
Figure 13-1. Swept-Sine Measurement Flowchart

Swept Sine Overview

Swept sine is a technique for characterizing the frequency response of the DUT. Two techniques are commonly used in swept-sine measurements. The first technique slowly sweeps through a range of frequencies in a manner similar to a chirp. Figure 13-2 shows an example of the excitation signal for this form of swept-sine measurement.

The second technique steps through a range of frequencies. Figure 13-3 shows an example of the excitation signal for this form of swept-sine measurement. The swept sine implemented in the Sound and Vibration Toolkit generates an excitation signal that steps through a range of test frequencies, similar to the signal in Figure 13-3. Both techniques can yield similar results. However, they require very different measurement analysis processes.
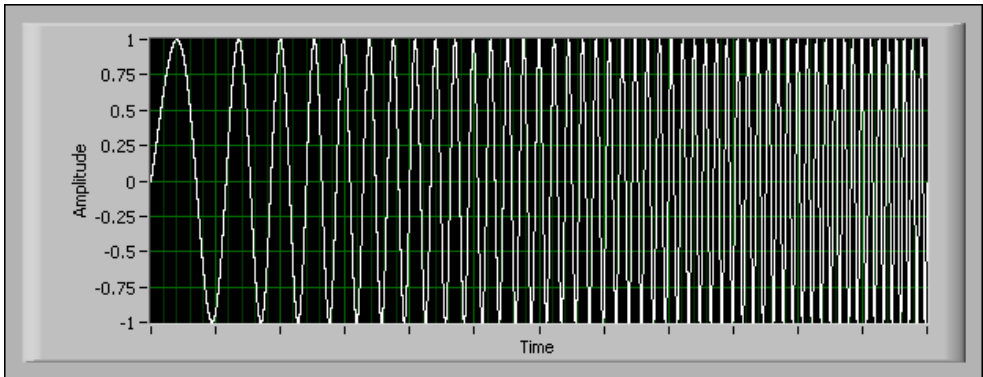
Figure 13-2. Sweeping Swept Sine Example
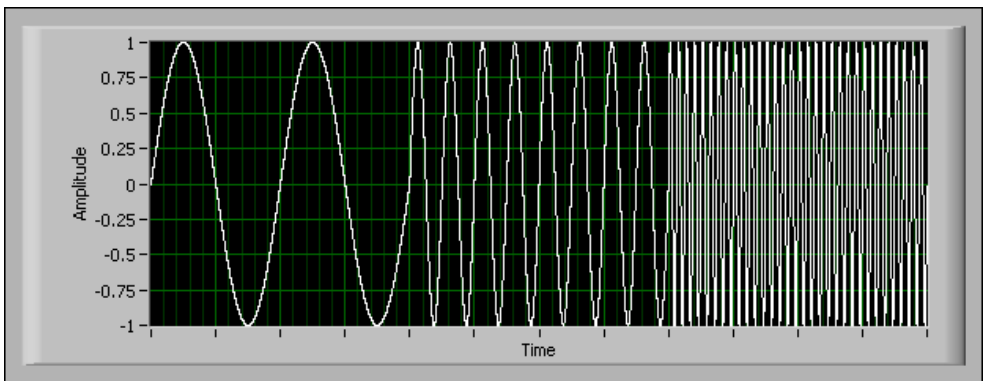
Figure 13-3. Stepping Swept Sine Example

Swept-sine frequency-response measurements compare a response signal to the stimulus tone in order to compute the FRF of the DUT. The magnitude of the FRF is equivalent to gain and represents the ratio of the output level to the input level for each test frequency. The phase of the FRF is equivalent to the phase lag introduced by the DUT for each test frequency.

Swept-sine measurements require a signal source. The stimulus signal is always a single tone that excites the DUT at the test frequency. Since the stimulus is a single tone, swept-sine analysis can measure the harmonic distortion while simultaneously measuring the linear response.

Choosing Swept-Sine versus FFT Measurements

The frequency response of the DUT is a useful tool. The Sound and Vibration Toolkit provides two distinct techniques to measure the frequency response. The swept-sine technique performs single-tone measurements at each test frequency. The FFT-based technique measures the response over the entire acquisition bandwidth. Table 13-1 lists the basic differences between swept-sine and FFT-based techniques for measuring frequency response.

Table 13-1. Swept Sine and FFT Differences

Swept-Sine Frequency Response	FFT-based Frequency Response
Single-tone excitation	Broadband excitation
Can measure harmonics	Cannot measure harmonics
Arbitrary test frequencies	Linearly spaced frequency resolution
Longer test time for many test frequencies	
Better dynamic range possibility	

Swept-sine measurements offer superior dynamic range over FFT-based measurements because you can optimize the signal level and input ranges at each test frequency. FFT-based techniques must specify a signal level and input ranges appropriate for the maximum broadband response.

Figure 13-4 shows the simulated frequency response function for a four DOF system. The peak at 17.6 Hz has a magnitude roughly 1,000 times larger than the peak at 5.8 Hz. To use an FFT-based technique, use broadband excitation to excite the entire frequency range of interest to measure the frequency response. This situation forces you to set the input range so that the overall response does not overload the DUT or the acquisition device. Therefore, when you measure the response at 5.8 Hz, you lose 60 dB of measurement dynamic range. The swept-sine technique allows you to tailor the excitation amplitude to the specific test frequency, preserving the full measurement dynamic range.
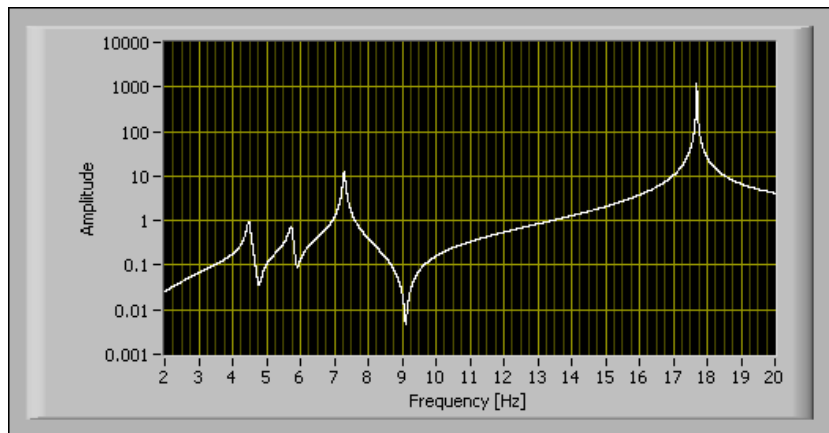
Figure 13-4. Swept-Sine and FFT Measurements

FFT-based measurements are limited to a linear-spaced frequency resolution determined by the sample rate and the block size. Refer to Chapter 10, *Frequency Analysis*, for more information on FFT-based measurements. When the response changes rapidly, this frequency resolution may not yield enough information about the dynamic response. Also, a linear resolution may yield an excessive amount of information in frequency regions where the dynamic response is relatively constant. Swept-sine analysis has the ability to test arbitrary frequency resolutions that are linear, logarithmic, or adapted to the dynamic response of the DUT. When the frequency resolution is adapted to the DUT dynamic response, you can test more frequencies in regions where the dynamic response is of interest to your application and less where it is not.

The main benefit of swept-sine analysis is the ability to measure harmonic distortion simultaneously with linear response. FFT-based analysis offers a speed advantage for broadband measurements with many test frequencies.

Taking a Swept Sine Measurement

Use the SVT Initialize Swept Sine VI to create a new swept-sine task for the designated device, source channel settings, and acquisition channel settings. Swept sine in the Sound and Vibration Toolkit only supports measurements on a single device with output and input capabilities.

Use the Configure Swept Sine VIs in the Configure Swept Sine palette to configure the scaling, test frequencies, averaging, delays, and other measurement settings. These configuration VIs allow control over basic and advanced measurement parameters. The order in which you place the configuration VIs is important, as it allows you to customize a swept-sine measurement. For example, you can easily generate 100 logarithmically spaced test frequencies in the audio range, then apply inverse A-weighted scaling to the excitation level by adding code similar to that in Figure 13-5 into your swept-sine application.
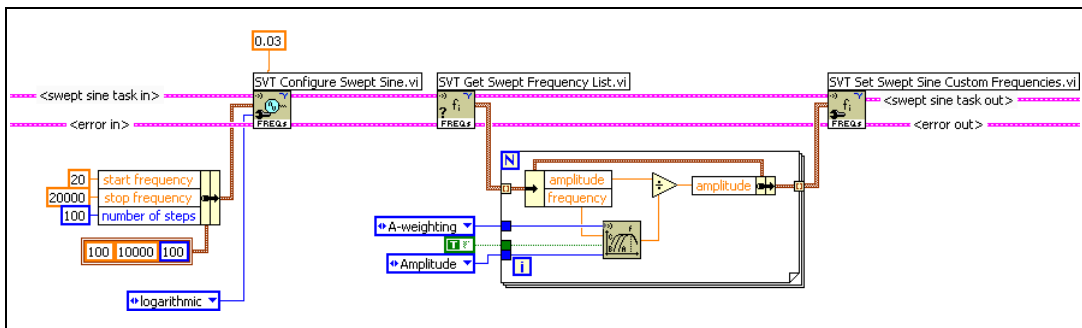
Figure 13-5. Customizing a Swept Sine Measurement

You can use the swept-sine configuration VIs to customize your swept-sine application. For example, to speed up a swept-sine measurement, reduce the settling or integration time specified by the SVT Set Swept Sine Averaging VI. You also can configure the device integrated electronic piezoelectric (IEPE) excitation with the SVT Set Swept Sine Coupling and IEPE Excitation (DAQmx) VI. You also can reduce the **block duration** input to SVT Set Swept Sine Block Duration VI.

Note   The minimum block duration is limited by the capabilities of the computer processing the measurement. A very small block duration can result in a loss of continuous processing, causing the swept-sine measurement to stop and return an error.

Use the SVT Start Swept Sine VI to begin the generation and acquisition. Initially, the VI fills the device output buffer with zeros before writing the first test frequency excitation.

The SVT Swept Sine Engine VI continually acquires data and processes it to remove samples acquired during delays, transitions, and settling periods. The SVT Swept Sine Engine VI performs measurement analysis on samples acquired during integration periods. The SVT Swept Sine Engine VI updates the excitation to excite the DUT at the next test frequency after it integrates sufficient data at the current test frequency.

Note   The transition to the next excitation tone, both frequency and amplitude, always occurs at a zero crossing to minimize transients introduced to the DUT.

Use the Read Swept Sine Measurements VIs in the Read Swept Sine Measurements palette to read the raw measurements, scale the measurements, and perform additional conversions to display and report the data in the desired format.

Use the SVT Close Swept Sine VI to stop the generation and acquisition, and clear the swept-sine task.

Swept Sine Measurement Example

This example of a swept-sine application measures the frequency response and harmonic distortion of a notch filter. In this example an NI PXI-4461 generates the excitation signal and acquires the stimulus and response signals.

Figure 13-6 illustrates the connection scheme used in this example to measure the dynamic response of the DUT using a swept-sine measurement. The acquired stimulus signal on analog input channel 0, AI0, is the generated excitation signal from analog output channel 0, AO0.
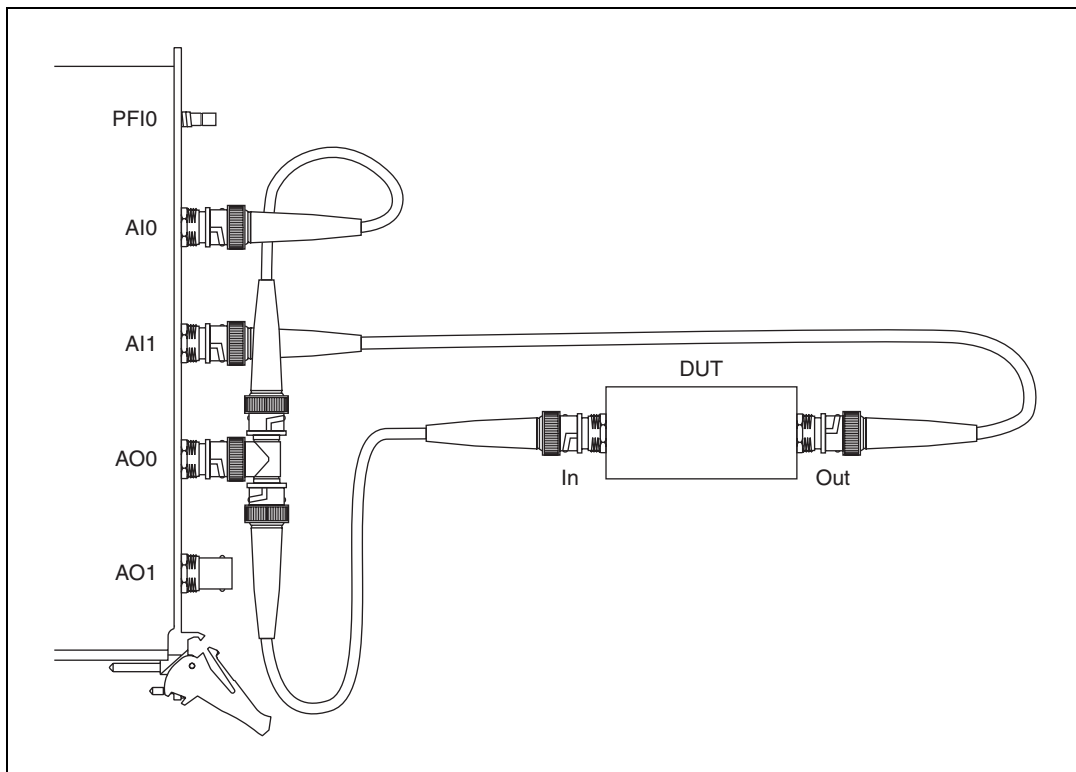
Figure 13-6. Swept-Sine Measurement Connection Diagram

The NI PXI-4461 converts the desired stimulus signal from digital data to an analog signal and outputs that signal on AO0. The excitation signal is connected to both the stimulus input channel AI0 and the input terminal of the DUT. The response signal is connected from the output terminal of the DUT to the response input channel AI1.

The DUT for this example is a notch filter centered at 1 kHz.

Figure 13-7 shows the block diagram of the example SVXMPL_Swept Sine FRF DAQmx VI, which ships with the Sound and Vibration Toolkit.
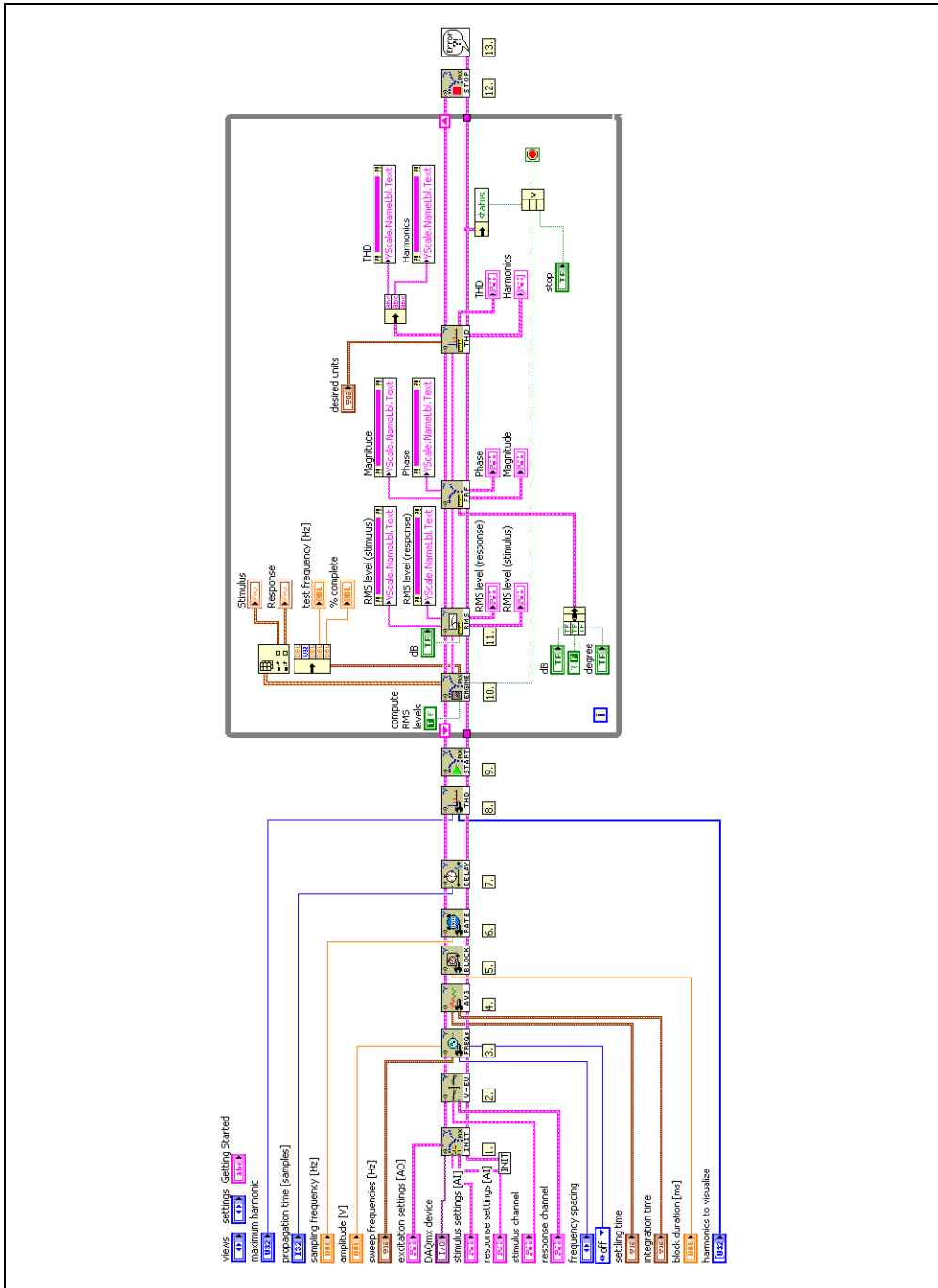
Figure 13-7. Block Diagram of SVXMPL_Swept Sine FRF DAQmx VI

Table 13-2 documents the actions performed by the VIs in Figure 13-7. Some steps are required and must be done for the VI to function correctly. The optional steps allow you to customize your measurement.

Table 13-2. Swept Sine Measurement Steps

Step Number	Description	Optional or Required
1	Initialize a swept-sine measurement by specifying the hardware device and channel settings.	Required
2	Specify the scaling that will be applied to the acquired stimulus and response data.	Optional
3	Configure the source by specifying the test frequencies, amplitude, and whether or not the sweep automatically restarts after completion.	Required
4	Set the settling and integration parameters to allow sufficient time for the DUT to settle before the measurement is performed at the new test frequency and sufficient integration time to achieve the desired level of accuracy.	Required
5	Set the **block duration** input terminal for the measurement to be small enough to give a reasonable test time and large enough so it does not put the test computer at risk of being unable to continuously generate and read the signals. The smaller the block size, the faster the swept sine can transition from one test frequency to the next.	Optional
6	Explicitly set the sample rate for the measurement. The rate is automatically selected if this VI is not used. The same rate is used for input and output channels.	Optional
7	Specify the **propagation time** terminal input specific to the DAQ device being used for the measurement. You can measure the device propagation time using the SVL Measure Propagation Delay VI. Refer to Chapter 3, *Scaling and Calibration*, for more information.	Optional

Table 13-2. Swept Sine Measurement Steps (Continued)

Step Number	Description	Optional or Required
8	Configure the harmonic distortion measurement by specifying the maximum harmonic to use in the computation of the THD. Only those harmonics specified in the **harmonics to visualize** array return individual harmonic components.	Required if performing distortion measurements
9	Start the swept sine to perform the hardware configuration and start the output and input tasks. Channel synchronization is performed internally in this VI.	Required
10	Generate the excitation and acquire the stimulus and response data at each test frequency.	Required
11	Convert the raw data to the specified format in order to display and report measurement results.	Required
12	Stop the swept-sine measurement and clear the output and input tasks to release the device.	Required

The While Loop in Figure 13-7 controls the synchronized generation and acquisition. Display controls and measurement indicators are updated inside the While Loop. This loop allows for the monitoring of intermediate results.

Many of the steps in Table 13-2 are configuration steps. Through the Sound and Vibration Toolkit swept-sine configuration VIs, you can specify numerous configuration parameters to achieve fine control of the swept-sine measurement parameters. For many applications two or three configuration VIs are sufficient.

It is important to allow for the propagation delay of the DAQ or DSA device. This delay is specific to the device used to perform the measurement. To determine the device propagation delay, refer to the device documentation or measure the delay with the SVL Measure Propagation Delay VI. Refer to Chapter 3, *Scaling and Calibration*, for more information about the SVL Measure Propagation Delay VI.

Figures 13-8 to 13-11 display measurement results obtained with the SVXMPL_Swept Sine FRF DAQmx VI example program. Figure 13-8 shows the time-domain stimulus and response signals for the 138.49 Hz

test frequency. From the time-domain data, you can see that the notch filter has attenuated the signal and introduced a phase shift.
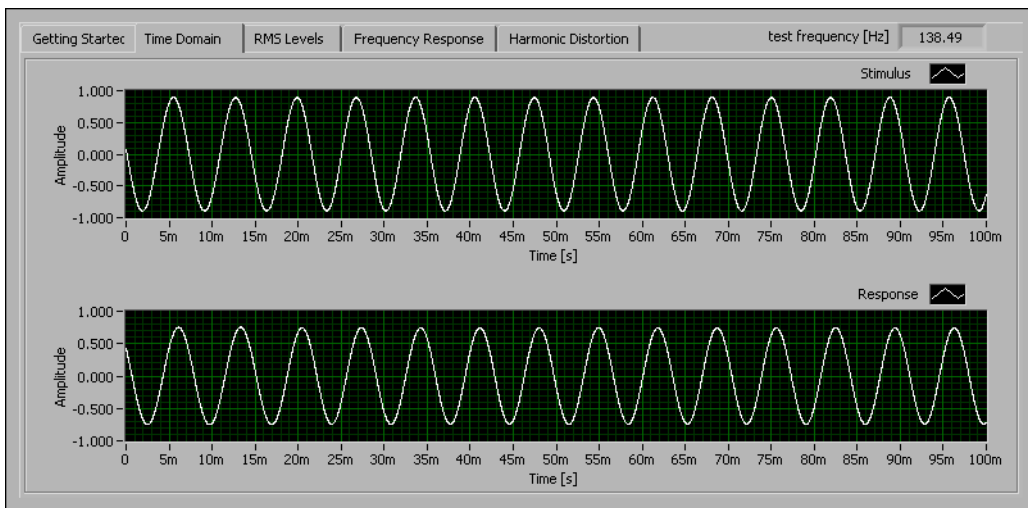
Figure 13-8. Time Domain Results

Figure 13-9 shows the magnitude and phase responses of the notch filter at all the test frequencies in the magnitude and phase spectra in the Bode plot.
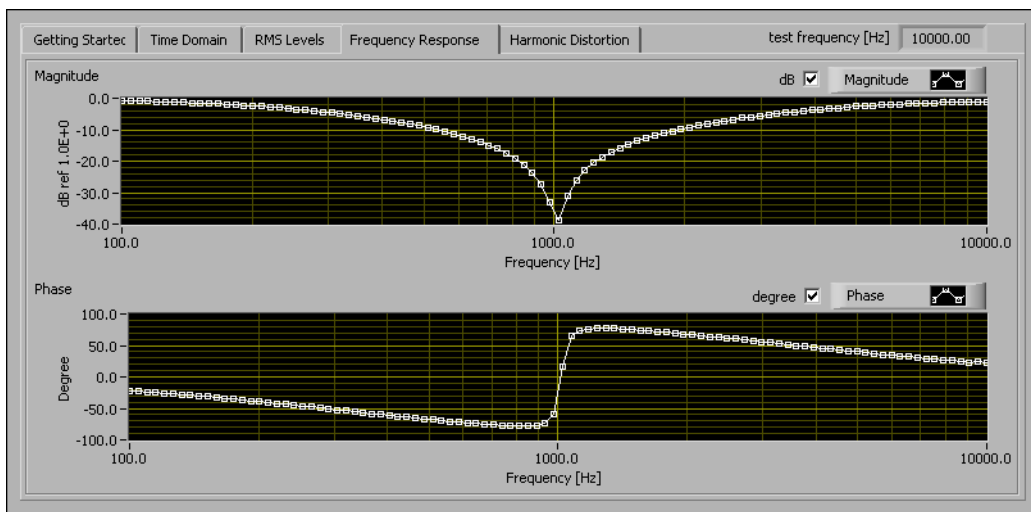
Figure 13-9. Magnitude and Phase Response of a 1 kHz Notch Filter

In addition to measuring the frequency response, this example simultaneously measures the harmonic distortion at each test frequency. Figure 13-10 shows the graph of THD versus frequency.
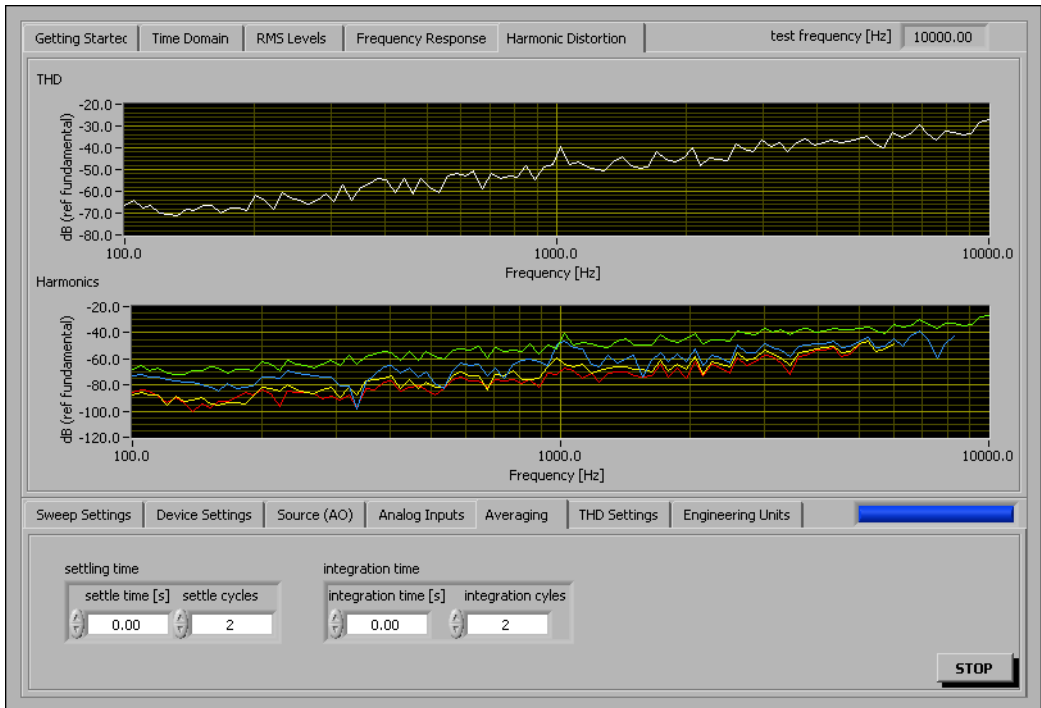
Figure 13-10. THD versus Frequency

You expect to see a peak in the THD at the notch frequency. The peak occurs because the fundamental frequency is attenuated at the notch frequency. However, the graph indicates that this measurement has failed to accurately identify the power in the harmonic distortion components. For the example in Figure 13-10, the number of **integration cycles** is two. More integration cycles must be specified to perform accurate harmonic distortion measurements. If you change the number of integration cycles to 10 and rerun the example, you obtain the THD versus frequency results displayed in Figure 13-11.
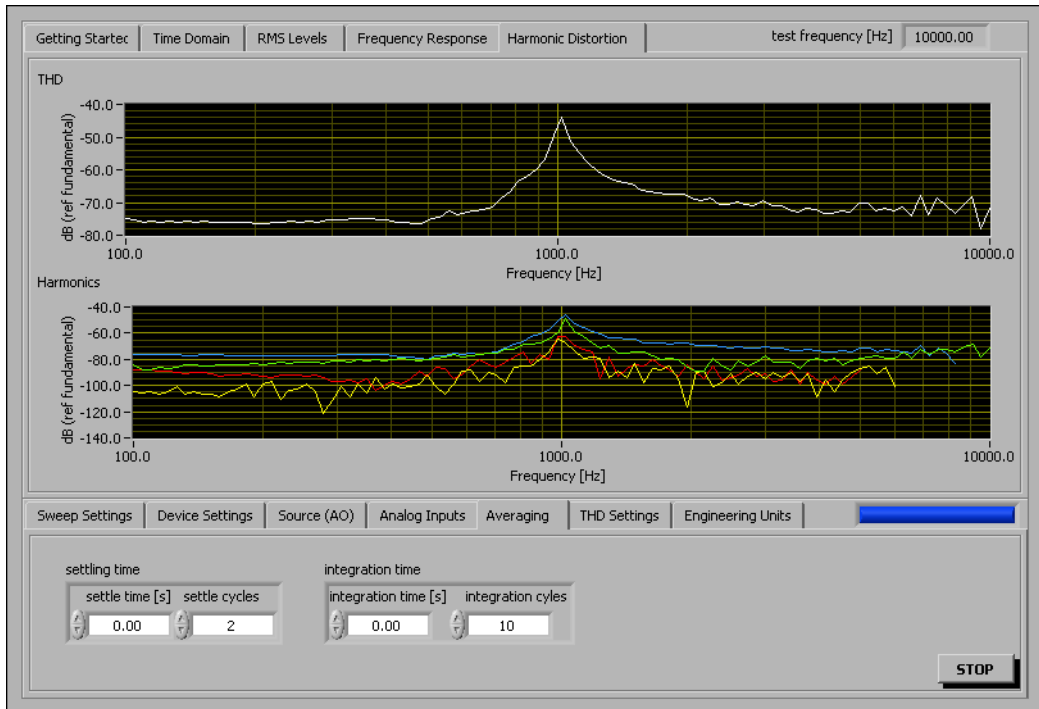
Figure 13-11. THD versus Frequency Results

Now, with a sufficient number of integration cycles specified, you can see the characteristic peak in the THD at the center frequency of the notch filter.

14

Distortion Measurements

This chapter discusses using the Distortion VIs located on the Distortion palette.

Refer to the *LabVIEW Help* for information about the individual Distortion VIs.

Variable Definitions

In the equations in this chapter, the variables are defined as:

F is the fundamental tone root-mean-square (RMS) level

K is the maximum harmonic in the measurement bandwidth

H_i is the ith harmonic RMS level, where i is an integer <2..K>

N is the noise RMS level

$$\sum P_{\text{Harmonics}} \;=\; \sqrt{(H_2)^2 + (H_3)^2 + ... + (H_K)^2}$$

$$\sum P_{\text{Harmonics}} + \sum P_{\text{Noise}} \;=\; \sqrt{(H_2)^2 + (H_3)^2 + ... + (H_K)^2 + (N)^2}$$

$$\sum P_{\text{Total}} \;=\; \sqrt{(F)^2 + (H_2)^2 + ... + (H_K)^2 + (N)^2}$$

Figure 14-1 shows where distortion analysis fits into the sound and vibration measurement process.
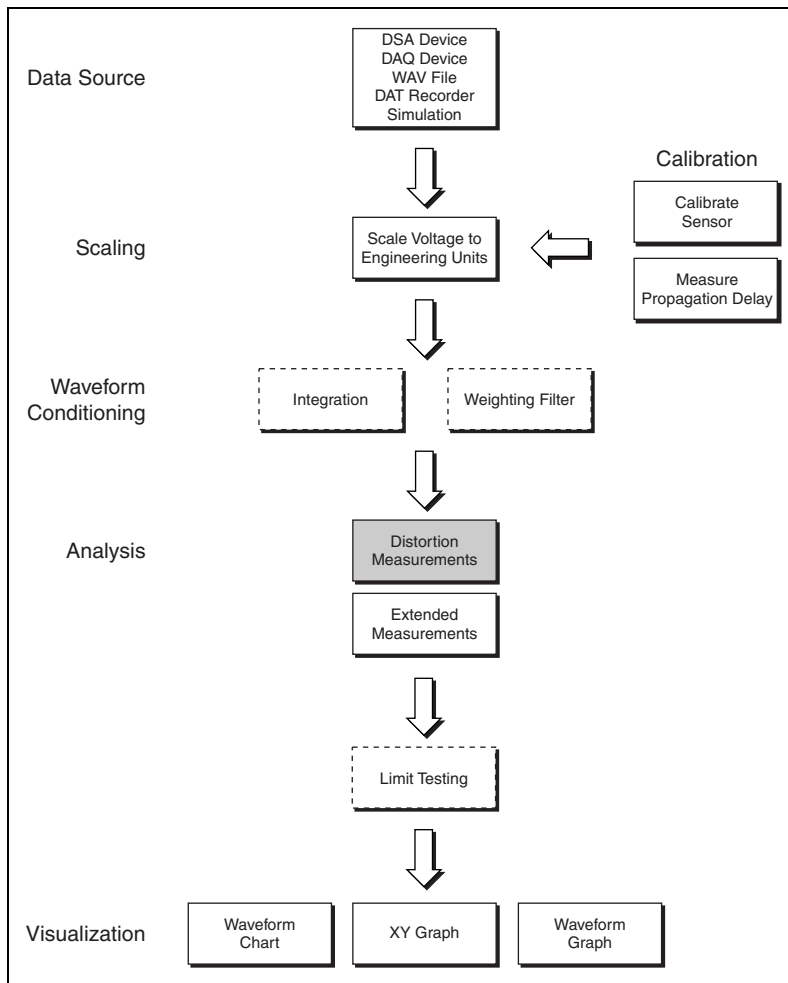
Figure 14-1. Relationship of Distortion Analysis to the Sound and Vibration Measurement Process

Signal in Noise and Distortion (SINAD)

This section provides more detail on the SVT SINAD VI.

SINAD is defined by the following equation:

$$\text{SINAD} = \frac{\sum P_{\text{Total}}}{\sum P_{\text{Harmonics}} + \sum P_{\text{Noise}}}$$

The SVT SINAD VI identifies the fundamental tone and effectively removes the fundamental tone from the input signal to measure the power in the harmonics and noise. The VI automatically identifies the tone with the highest amplitude as the fundamental tone unless you wire a value to the **expected fundamental frequency [Hz]** terminal.

You can input a frequency range to refine the measurement. The measured power in the total signal is distributed across the entire measurement bandwidth as a result of the noise in the measured signal. You can limit the measurement to a particular frequency range, exclude low frequency noise, or exclude high frequency noise by selecting an appropriate frequency range. For example, if you want to specifically exclude DC energy from the measurement, you should set the start frequency to at least 8(df), where df is the frequency resolution of the FFT used by the SVT SINAD VI. If there are frequency components of interest below 8(df) that are not DC, reduce df by increasing the measurement duration. Use the following equation to determine the relationship between df and the measurement duration, where T is the measurement duration:

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

The example illustrated in Figures 14-2 and 14-3 shows how you can effectively remove DC energy from the measurement and still measure SINAD with low-frequency test signals.
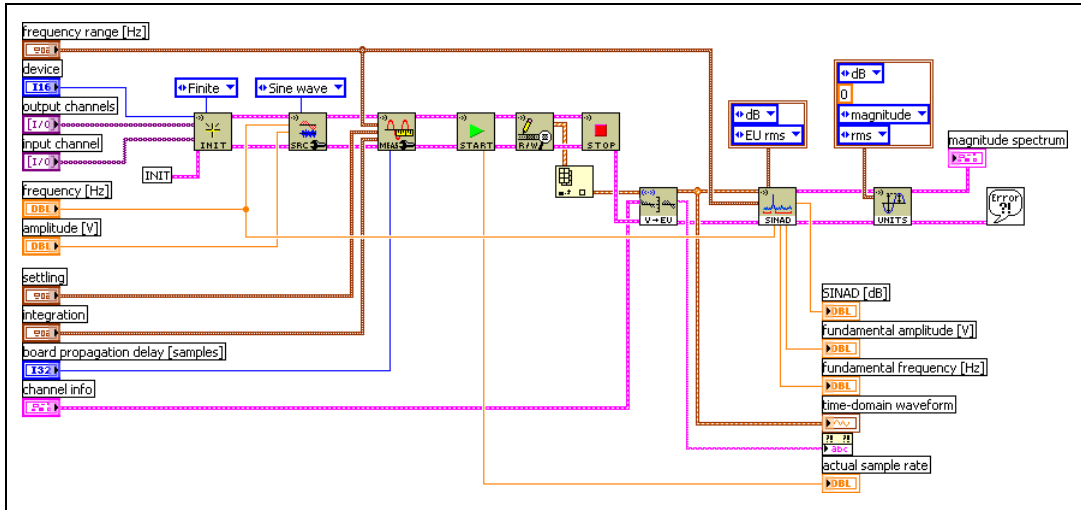
Figure 14-2. Removing DC Voltage from Low-Frequency SINAD Measurement

The VI in Figure 14-2 is a modified version of the SVXMPL_One Shot Gain and Distortion (Traditional DAQ) audio example VI in the Sound and Vibration Toolkit. Figure 14-2 uses the same single-tone excitation signal and finite acquisition used in the example VI. However, the SVT SINAD VI replaces SVT Gain and SVT THD and Harmonic Components VIs the used in the example. The added frequency range control allows you to control the bandwidth of the SINAD measurement.

You can use the VI in Figure 14-2 to measure the SINAD for the left channel of a device like a two-channel audio equalizer. In this example, the left channel level settings are set to 0 dB for all available octave bands. The measurement result displayed in Figure 14-3 has a 1 s integration time that results in a frequency resolution of 1 Hz. The start and stop frequencies are set to 20 Hz and 20 kHz, respectively. The tone frequency used in the test is 400 Hz. With these settings, the SINAD measurement includes the entire audio band and prevents DC energy from giving an inaccurately low result.
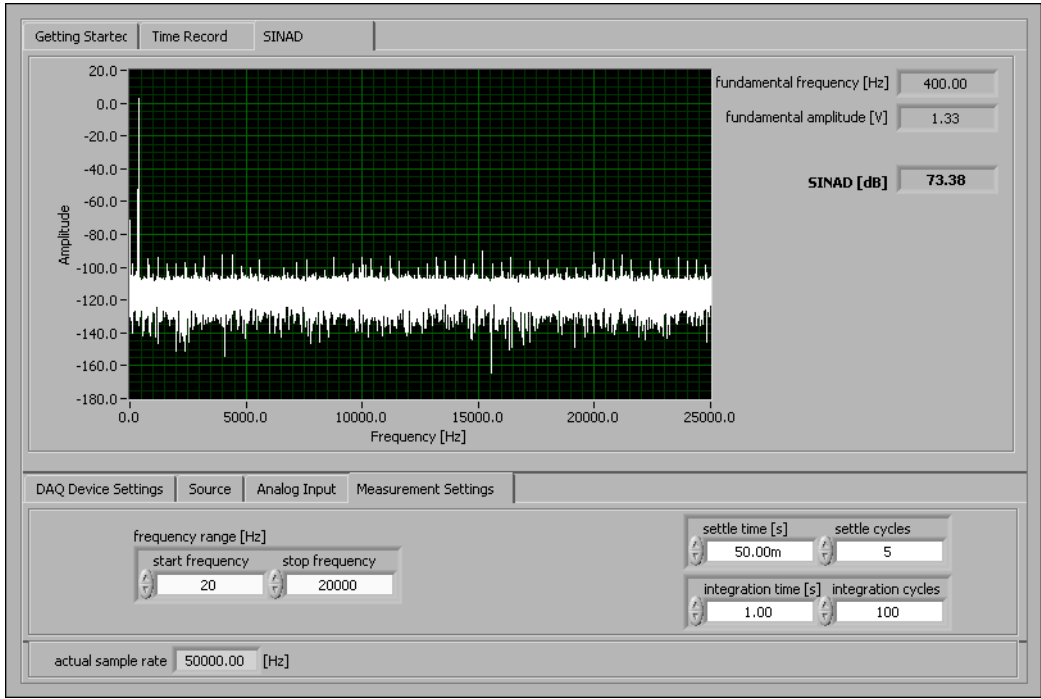
Figure 14-3. Front Panel of SINAD Measurement VI

The reported SINAD for the left channel of the audio equalizer is SINAD=73.4 dB, 20 Hz–20 kHz, test amplitude 2 V_{pk}, test frequency 400 Hz, and unity gain.

Total Harmonic Distortion Plus Noise (THD+N)

You can define THD+N with the following equation:

$$\text{THD+N} = \frac{\sum P_{\text{Harmonics}} + \sum P_{\text{Noise}}}{\sum P_{\text{Total}}} \qquad (14\text{-}1)$$

Alternatively, other sources define THD+N with the following equation:

$$\text{THD+N} = \frac{\sum P_{\text{Harmonics}} + \sum P_{\text{Noise}}}{F} \qquad (14\text{-}2)$$

When the fundamental tone dominates the signal, these two definitions are equivalent. The Sound and Vibration Toolkit uses the definition in Equation 14-1. In this equation the power in the harmonic components and noise is divided by the total signal power to compute the THD+N.

Note that when we use the definition in Equation 14-1, the THD+N is the inverse of the SINAD. When these measurements are expressed in decibels, the THD+N is the negative of the SINAD.

The Sound and Vibration Toolkit THD+N function identifies the fundamental tone and effectively removes the fundamental tone from the input signal to measure the power in the harmonics and noise. The VI automatically identifies the tone with the highest amplitude as the fundamental tone unless you wire a value to the **expected fundamental frequency [Hz]** terminal.

Input a frequency range to refine the measurement. The measured power in the total signal is distributed across the entire measurement bandwidth as a result of the noise in the measured signal. You can limit the measurement to a particular frequency range, exclude low frequency noise, or exclude high frequency noise by selecting an appropriate frequency range. For example, if you want to specifically exclude DC energy from the measurement, you should set the start frequency to 8(*df*). If there are frequency components of interest below 8(*df*) that are not DC, reduce *df* by increasing the measurement duration.

Total Harmonic Distortion (THD)

This section provides more information on total harmonic distortion (THD). THD is the level of undesired harmonics present in an audio signal, usually expressed in decibels. The following formula calculates THD:

$$\text{THD} = \frac{\sum P_{\text{Harmonics}}}{F}$$

The SVT THD and Harmonic Components VI identifies the fundamental tone and effectively removes the tone from the input signal before identifying harmonics in the signal. Removing the tone before identifying harmonics allows accurate measurement of the signal amplitude in the midst of the harmonics. The VI automatically identifies the tone with the highest amplitude as the fundamental tone unless you wire a value to the **expected fundamental frequency [Hz]** terminal.

THD only includes the energy in the harmonics and does not include the energy in the broadband noise. Therefore, the measured THD is less than the measured THD+N for the same input signal.

The SVT THD and Harmonic Components VI allows you to specify the **maximum harmonic** used to compute the THD. You can wire –1 to this input to include all the harmonics up to the Nyquist frequency. If the **maximum harmonic** input to the VI is greater than the Nyquist frequency, the function uses all the harmonics up to the Nyquist frequency and returns a warning with code **1947**. This warning indicates a coerced maximum harmonic.

You can view any of the measured harmonics by specifying the desired harmonics as an array to the **harmonics to visualize** terminal. If you wire a single-element array with a –1 in the first element to this terminal, all the measured harmonics will be output to the **harmonic components** terminal.

Figure 14-5 and Table 14-1 show the magnitude spectrum, measured gain, THD+N, and THD of the left channel of the same two-channel audio equalizer used for the example in Figure 14-3.

The VI in Figure 14-4 uses the SVXMPL_One Shot Gain and Distortion example VI modified by the addition of the SVT THD+N VI. In the measurement analysis section of this VI, the audio measurements are applied sequentially to the same channel. The Scale to EU VI scales the time-domain data to engineering units. The Gain VI computes the complex spectrum and measures the gain. The SVT THD+N VI then uses the spectrum to compute the THD+N. The SVT THD and Harmonic Components VI uses the same spectrum to compute the THD, and the Unit Conversion VI converts the complex spectrum to a magnitude spectrum for display.

In this way you can perform several audio measurements sequentially to take advantage of the polymorphic VIs found in the Distortion and Single-Tone Measurements palettes. These VIs accept a waveform or a complex spectrum, and their polymorphic features allow multiple measurement analysis VIs to act sequentially on the data without recomputing any additional FFTs.
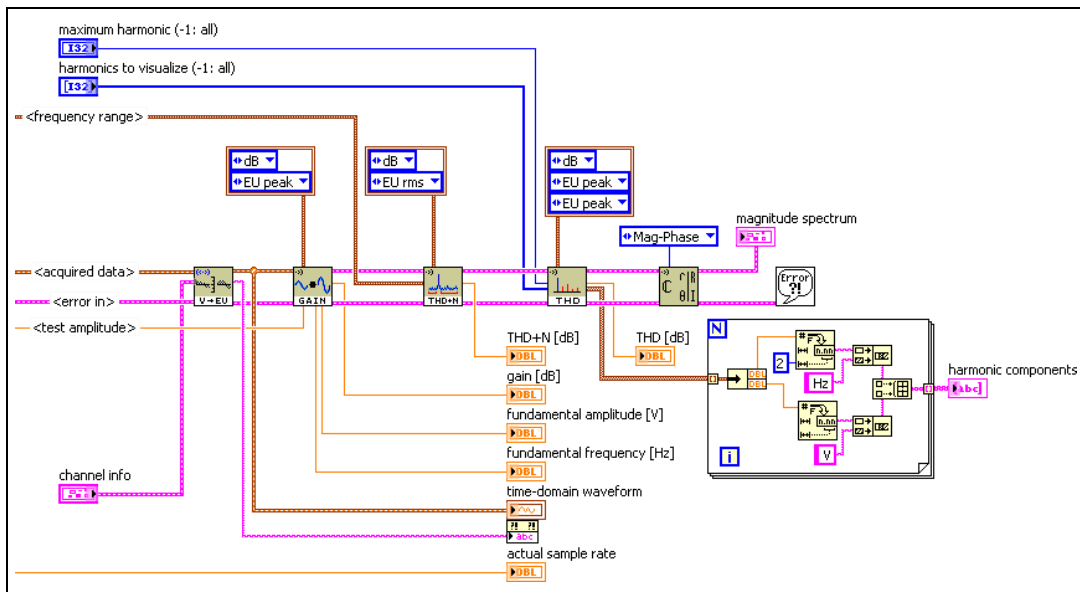
Figure 14-4. Distortion Test VI

Figure 14-5 displays the distortion test measured magnitude spectrum. The fundamental frequency of 1,000 Hz is clearly visible followed by harmonics in multiples of 1,000. Refer to Table 14-1 for more information.
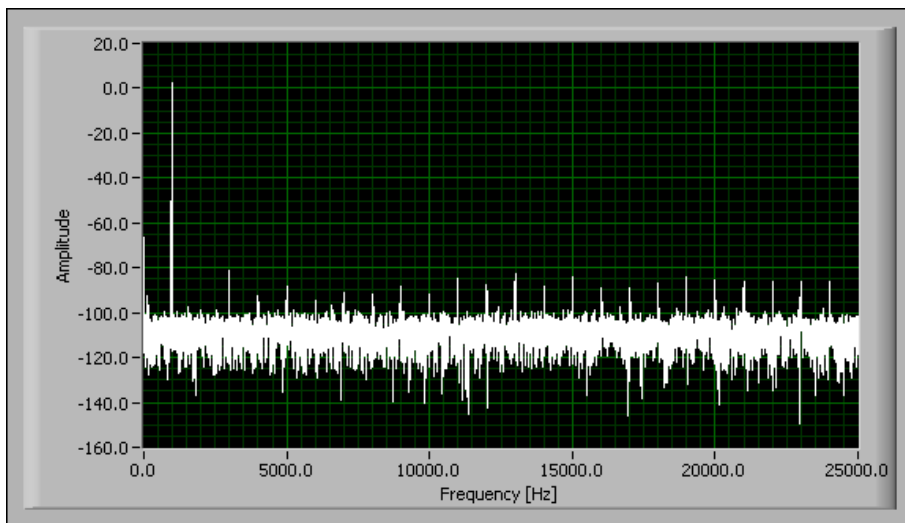
Figure 14-5. Distortion Test Measured Magnitude Spectrum

Table 14-1 lists the distortion test measurement results.

Table 14-1. Distortion Test Measurement Results

Variable	Result
Fundamental Frequency	1,000.00 Hz
Fundamental Amplitude	1.82 V_{pk}
gain	−0.80 dB
THD+N	−71.69 dB
THD	−76.16 dB
h_2	0.000021 V_{pk}
h_3	0.000120 V_{pk}
h_4	0.000040 V_{pk}
h_5	0.000059 V_{pk}

Figure 14-5 illustrates that the measured THD is lower than the THD+N over the same bandwidth of 20 Hz to 20 kHz. This relationship always holds true because the THD+N measurement includes the energy in the harmonics as well as the energy in the noise.

Intermodulation Distortion (IMD)

Intermodulation Distortion (IMD) is another measure of distortion due to nonlinearity in the DUT. IMD measurements use a dual-tone test signal composed of a low-frequency tone at f_1 and a high-frequency tone at f_2. When a dual-tone signal is input to a nonlinear device, the device output will contain intermodulation components of f_1 and f_2 given by $m(f_2) \pm n(f_1)$ where m and n are integers.

IMD is often used to measure the distortion of the DUT near the high-frequency limit of the DUT or the measurement system. You also can arrange the test so that many IMD components occur within the measurement bandwidth.

The IMD measurement is dictated by the IMD standard used for the measurement. There are several standard configurations for IMD measurements, and these configurations use one of two types of IMD test signals.

In the first type of IMD test signal, with a low-frequency modulation tone and a high-frequency carrier tone, the intermodulation components appear as side bands around the high-frequency carrier tone.

The Figure 14-6 shows the side bands around the 8 kHz carrier tone, as computed by the following equation:

$$f_2 \pm n \bullet f_1$$
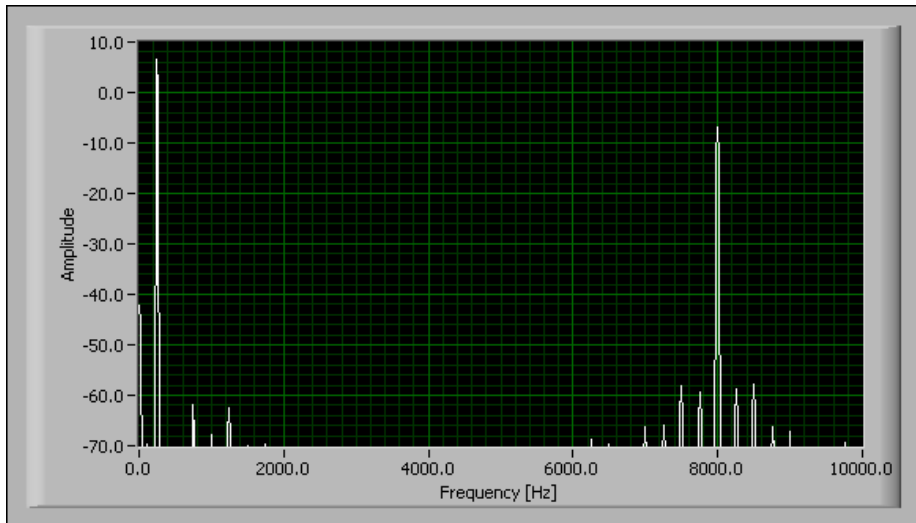
Figure 14-6. IMD Test Signal with High-Frequency Carrier Tone Sidebands

In the second type of IMD test signal, with two closely spaced tones near the high-frequency limit of the measurement bandwidth, the intermodulation components appear at multiples of the difference frequency calculated by the following equation:

$$m \bullet (f_2 - f_1)$$

Figure 14-7 shows a typical IMD test spectrum. The intermodulation components are found at multiples of 1,000 Hz, the difference frequency between the twin 11 kHz and 12 kHz tones.
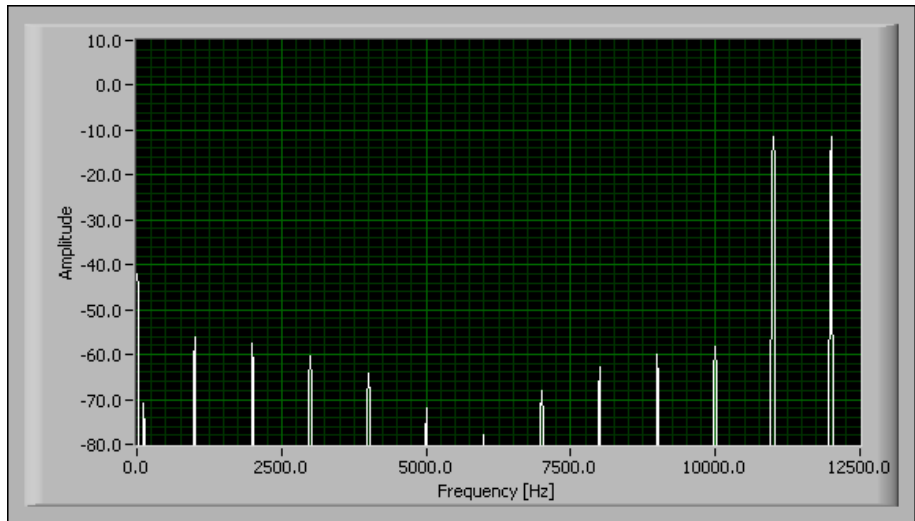
Figure 14-7. IMD Components as Multiples of the Difference Frequency

IMD measurements are defined by several standards. Each of these standards have particular advantages and disadvantages when trying to reveal distortion. The standards, associated test signals, and typical use cases are listed in Table 14-2.

Table 14-2. IMD Standards and Typical Applications

IMD Standard	Test Signal ($f_1, f_2,$ ratio)	Typical Applications
SMPTE/DIN	(60, 7000, 4) (250, 8000, 4) others	Excite low-frequency distortion mechanisms (such as thermal distortion in power amplifiers) Disk recording and film
ITU-R (CCIF)	(11000, 12000, 1) (14000, 15000, 1) (19000, 20000, 1) others	ADC and DAC Slope-induced distortion

The frequencies f_1 and f_2 are the frequencies, in hertz, of the low- and high-frequency tones, respectively. The ratio is the ratio of the amplitude of the low-frequency tone to the amplitude of the high-frequency tone.

Do *not* expect to get the same measured IMD when performing the measurement with different standards. It is important that you specify the test signal used to perform the measurement when you report IMD measurement results.

Phase Linearity

Phase linearity is one way to measure the amount of group delay distortion introduced by the DUT. The phase linearity measurement uses a multitone signal to measure the relative delay of each component through the DUT. The multitone test signal should be constructed with zero relative phase between the individual frequency components. In addition, the signal should be constructed with evenly spaced frequency components given by the following equation:

$$f_i = i \bullet f$$

This test signal has a high crest factor. It is important that you construct the multitone signal with frequency components appropriate for the DUT. In general, f should be low enough to measure the group delay introduced by the DUT, as calculated by the following equation:

$$group\ delay \leq \frac{1}{f}$$

The phase linearity is defined as the maximum deviation from a straight line of the plot of the phase versus frequency. This measurement is performed in steps. First, the multitone signal is generated, passed through the DUT, and acquired on the input channel. The phase of each tone in the multitone signal is measured and plotted versus frequency to yield a graph similar to the one in Figure 14-8.
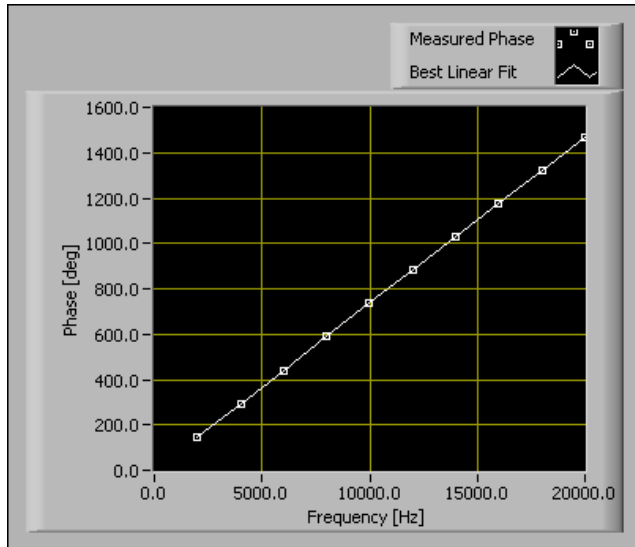
Figure 14-8. Phase Linearity Graph

The SVT Phase Linearity VI returns the group delay. The group delay is the slope of the best linear fit to the plot of phase versus frequency. Next, the VI computes the difference between the measured phase and the best linear fit. You can plot this difference against frequency to show the deviation of the measured phase from the best fit line. Figure 14-9 shows this plot.
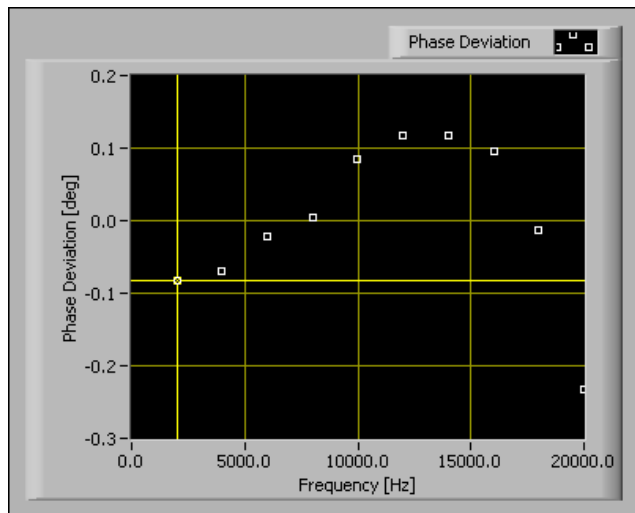
Figure 14-9. Deviation of the Measured Phase

The maximum absolute value of the phase deviation is the value returned as the phase linearity. In this case the maximum phase deviation value is 0.215°. Phase linearity measurements are always dependent on the bandwidth of the measurement. To completely report the result, specify that the phase linearity is 0.215° with a 20 kHz bandwidth, 1 V_{pk} test amplitude, and unity gain set for the DUT.

15

Single-Tone Measurements

This chapter discusses using the Single-tone Measurement VIs located on the **Single-tone Measurements** palette.

Refer to the *LabVIEW Help* for information about the individual single-tone measurement VIs.

Single-Tone Measurement Overview

Single-tone measurements are defined as a group of measurements where the excitation is a single tone. These measurements are often used to measure the linear response, nonlinear distortion, and noise of audio devices. These measurements fall into two broad categories, single-channel and dual-channel measurements.

Examples of dual-channel measurements include measurements of gain, phase, and crosstalk. Dual-channel measurements require that the excitation signal be connected to one of the input channels, as illustrated in Figure 15-1.
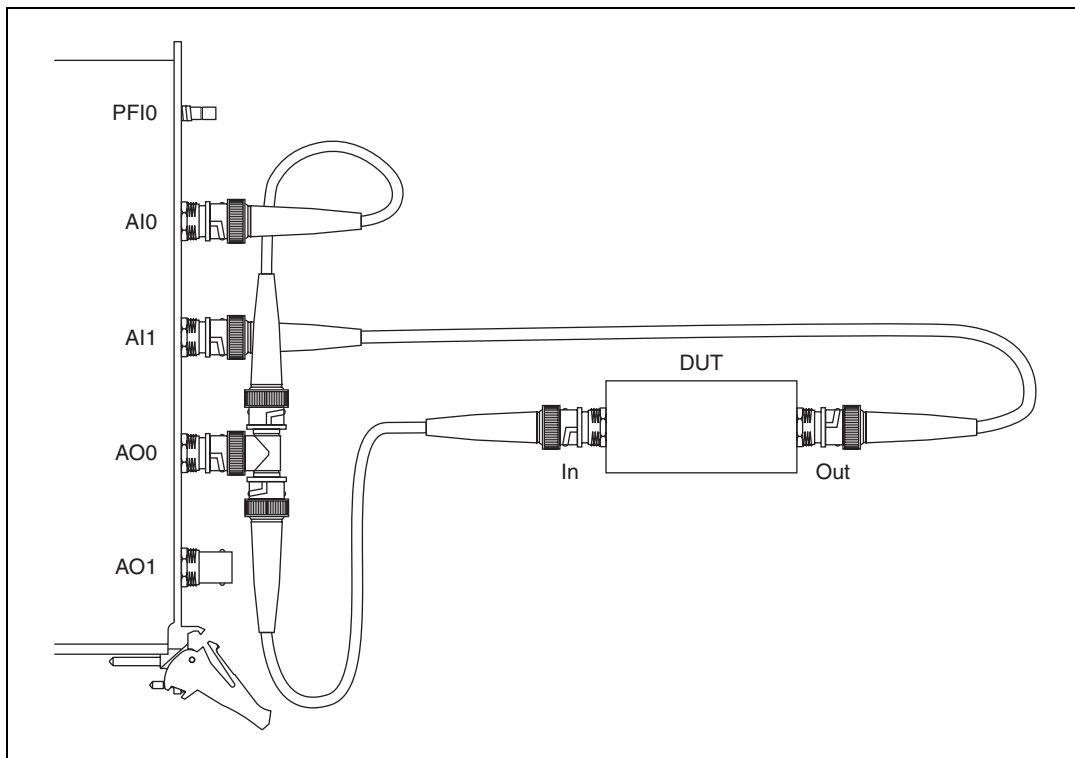
Figure 15-1. Dual-Channel Measurement

Typically, you know the frequency and amplitude of the excitation signal. The real advantage of connecting the excitation signal to one of the input channels is the ability to measure the relative phase between the stimulus and response signals. In a two-channel measurement configuration the input channel connected to the excitation signal is the stimulus channel, and the input channel connected to the output of the DUT is the response channel.

Examples of single-channel measurements are measurements of gain, idle-channel noise, dynamic range, and spurious-free dynamic range. Single-channel measurements do not require that the excitation signal be connected to an input channel of the acquisition device. Figure 15-2 illustrates the connection scheme for single-channel measurements.
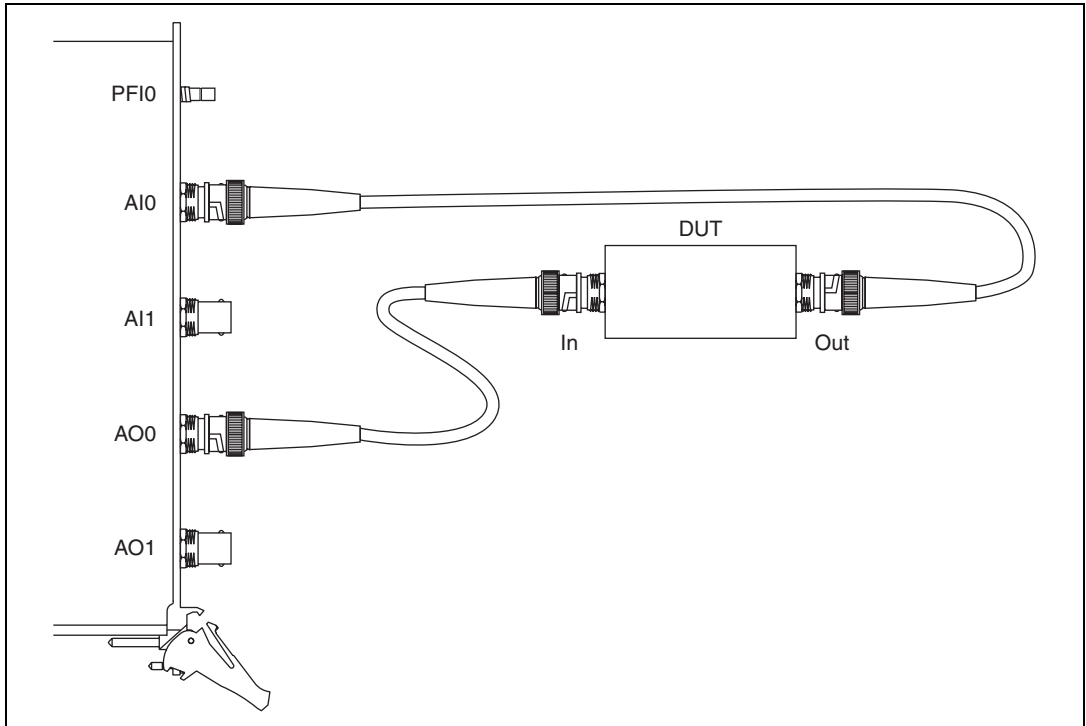
Figure 15-2. Single-Channel Measurement

Figure 15-3 illustrates where single-tone measurements fit into the sound and vibration measurement process.
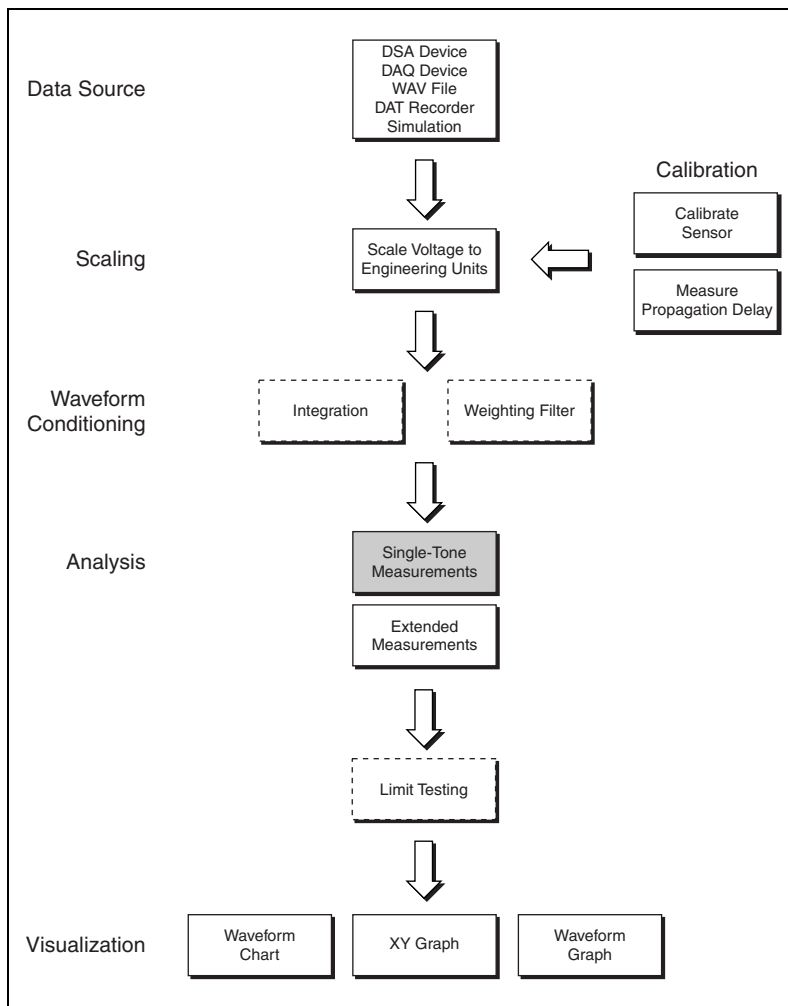
Figure 15-3. Relationship of Single-Tone Measurements to the Sound and
Vibration Measurement Process

Gain and Phase

The SVT Gain and Phase VI measures the amplification and phase lag
between the stimulus channel and the response channel at the test
frequency. The SVT Gain and Phase VI identifies the detected tone
amplitude and phase on the stimulus channel. The stimulus tone amplitude
and phase are treated as references when specifying the response channel
tone relative amplitude and relative phase. The SVT Gain and Phase VI

searches for a response channel tone at the same frequency as the stimulus tone. The SVT Gain and Phase VI is a dual-channel measurement which requires you to wire both stimulus and response signals. A single-channel measurement reads only the response signal.

You have the option of generating the excitation signal from the acquisition device, a separate output device, or an external source since the stimulus and response measurements are acquired. If you are using a DSA device, NI recommends you perform a dual-channel versus a single-channel measurement when possible. Performing a dual-channel measurement takes full advantage of the excellent interchannel gain mismatch and simultaneous sampling of the analog input channels. Refer to the DSA device specifications for more information. For example, in single-channel mode the measurement uncertainty when using an NI PXI-4461 is 0.1 dB, in dual-channel mode the measurement uncertainty is typically 0.01 dB.

Crosstalk

The SVT Crosstalk VI measures the amplification and phase lag between the reference channel and the idle response channel at the test frequency. The SVT Crosstalk VI identifies the detected tone amplitude and phase on the stimulus channel. The stimulus tone amplitude and phase are treated as references when specifying the idle response channel tone relative amplitude and relative phase. The SVT Crosstalk VI searches for an idle response tone at the same frequency as the stimulus tone.

The measurement analysis is equivalent to that performed for a gain and phase measurement. The difference between these two measurements is that the gain and phase measurement is applied on the active response channel, while the crosstalk measurement is applied to the idle response channel. Figure 15-4 illustrates this difference.
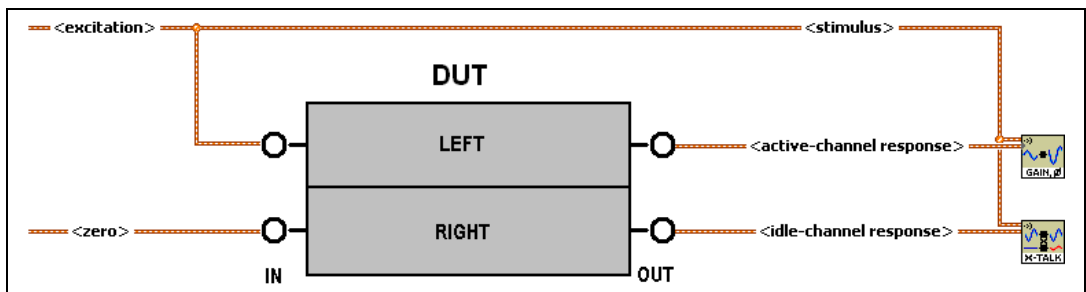
Figure 15-4. Difference between Gain and Phase Measurement and Crosstalk Measurement

Gain

The SVT Gain VI measures the amplification between the amplitude of the identified tone in the input signal and the reference amplitude. The VI automatically identifies the tone with the highest amplitude as the fundamental tone, unless you wire a value to the **expected fundamental frequency [Hz]** terminal. You must connect the amplitude of the excitation tone to the **amplitude** input terminal regardless of whether you use the output of the acquisition device or an external source. Figure 15-5 shows the required connection.
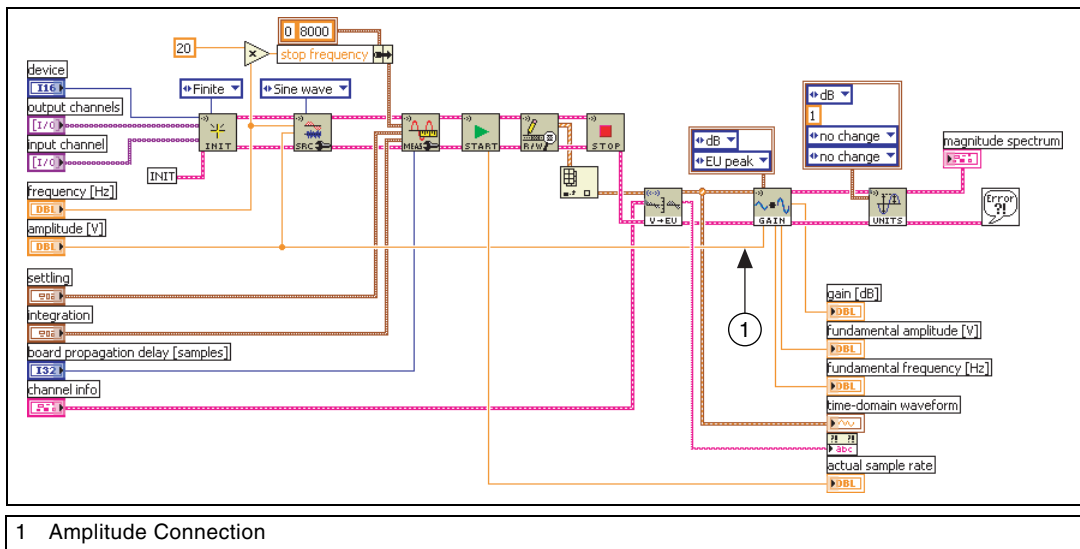
1   Amplitude Connection

Figure 15-5. Connecting the Amplitude of the Excitation Tone to the Amplitude Input

The accuracy of this measurement is dependent on the absolute amplitude accuracies of the output channel and the input channel. When generating and acquiring the signal with an NI PXI-4461, the expected measurement uncertainty is less than 0.3 dB.

Idle-Channel Noise

The SVT Idle Channel Noise VI measures the total power within the specified frequency range. For practical purposes, the SVT Idle Channel Noise VI is a power-in-band measurement that can operate on waveform data or the complex spectrum output by the other audio measurements. You can measure idle-channel noise with the input channel to the DUT floating, terminated, or excited with a zero amplitude signal.

Unlike gain, phase, and crosstalk measurements, idle-channel noise is a broadband measurement. A broadband measurement is one in which the fundamental frequency is expected to be zero. Use the **frequency range** control to limit the bandwidth of the measurement. To exclude DC power, set the start frequency to 8(*df*), where *df* is the frequency resolution of the spectrum. Increase the start frequency in the frequency range control to exclude additional low-frequency components from the measurement. You can enter the default values of –1 for the **start frequency** and –1 for the **stop frequency** to perform the measurement over the entire acquisition bandwidth.

Dynamic Range

A dynamic range measurement quantifies the distortion and noise found in the DUT output when a low-level signal is provided. Typically, the dynamic range is measured with the amplitude of the excitation signal set to –60 dB FS, where FS is the DUT input channel full-scale amplitude.

The connection scheme for a dynamic range measurement follows that of the single-channel measurement in Figure 15-2.

Configure the measurement by specifying the measurement settings in the SVXMPL_Settled Dynamic Range (DAQmx) VI. It is important to allow for the propagation delay of the DAQ or DSA device. This delay is specific to the device used to perform the measurement. To determine the device propagation delay, refer to the device documentation or measure the delay with the SVL Measure Propagation Delay VI. Refer to Chapter 3, *Scaling and Calibration*, for more information about the SVL Measure Propagation Delay VI.

The SVXMPL_Settled Dynamic Range (DAQmx) VI performs a continuous generation and acquisition. The measurement continues until the measured dynamic range is settled according to the tolerances specified under the **Settling Parameters** tab. For the left channel of this DUT,

a two-channel audio equalizer, a typical measurement configuration results in a dynamic range of 83 dB FS, 20 Hz–20 kHz, test amplitude –60 dB FS, test frequency 1.0 kHz, and unity gain.

Figure 15-6 shows the final measured spectrum for the dynamic range measurement.
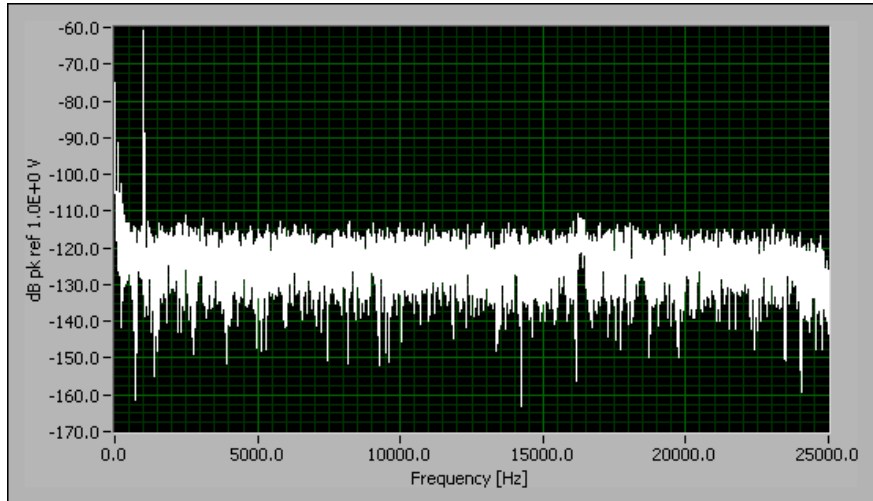
Figure 15-6. Measured Spectrum from Dynamic Range Test

Spurious Free Dynamic Range (SFDR)

An SFDR measurement quantifies the lowest level peaks identifiable as actual components of the signal instead of DUT artifacts. Typically, the SFDR is measured with the amplitude of the excitation signal set to 0 dB FS or –1 dB FS. The measured SFDR is then the difference in amplitude, in decibels, between the excitation tone and the largest spur. Figure 15-7 shows the SFDR test spectrum on the left channel of the two-channel audio equalizer.
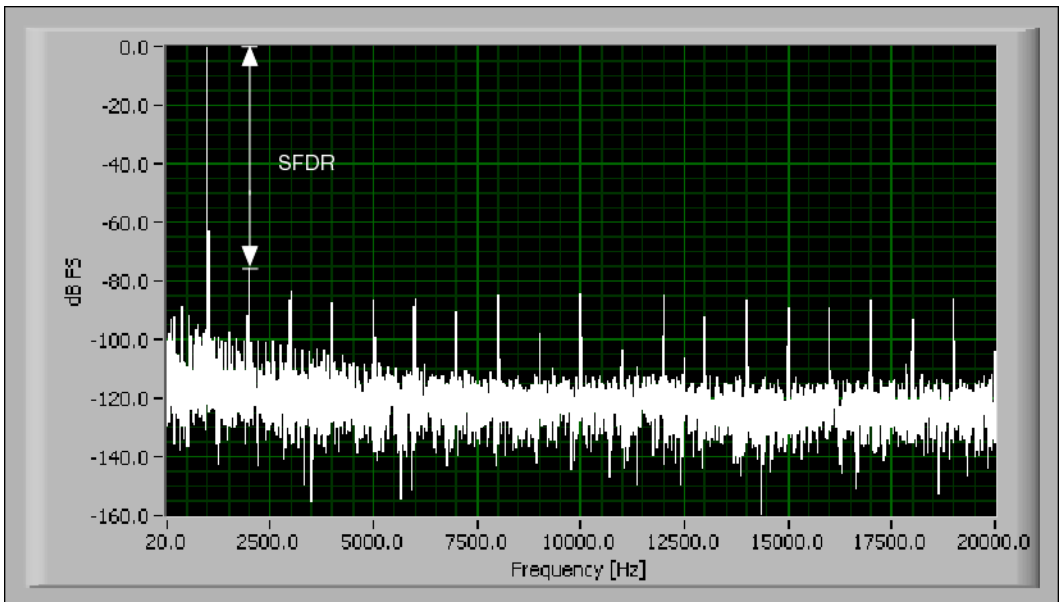
Figure 15-7. SFDR Test Spectrum

A

References

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

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B

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Glossary

Numbers/Symbols

2D	Two-dimensional.
3D	Three-dimensional.
μPa	Micro-Pascal. *See also* Pa.

A

acceleration	The rate of change of velocity.
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. *See also* 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. *See also* 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.
baseband analysis	The range of frequencies extending from 0 Hz (DC) to a maximum frequency, often the Nyquist 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.
constant Q filter	The type of filters used in fractional-octave measurements.
COP	Coherent output power.
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.
cut-off frequency	The frequency at which the filter attenuates the input 3 dB, or half of its original power.

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 dB = 10*log10 (P/Pr) where P is the measured power and Pr is the reference power. If the levels are specified in terms of amplitude, then 1 dB = 20*log10(A/Ar) where A is the measured amplitude and Ar is the reference amplitude.
Distortion	The production of signal components not in the original signal due to non-linearities in the system or transmission path.
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.
Dynamic range	The ratio of the largest signal level a circuit can handle to the smallest signal level it can handle (usually taken to be the noise level), normally expressed in decibels.
DZT	Discrete Zak Transform.

E

ENBW	Equivalent noise bandwidth of the time-domain window applied to the signal in order to perform frequency analysis.
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 ± 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	*See* 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. *See* frequency response function.
fundamental frequency	The frequency of the dominant (highest amplitude) tone in the signal. For harmonic analysis the fundamental frequency is the greatest common divisor of the harmonic frequencies.

G

g	Unit for measuring acceleration. One $g = 9.81$ 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: Gxy/Gxx. 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: Gyy/Gyx. 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.
Harmonic	A signal whose frequencies are integer multiples of the input signal.
Harmonic distortion	A form of distortion in analog circuits that generates harmonics. It is calculated as the ratio of the level of a single harmonic to the level of the original signal.
Hz	Hertz. Cycles per second.

I

IEC	International Electrotechnical Commission.
IEPE	Integrated Electronic Piezoelectric.
IMD	*See* intermodulation distortion.
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.
intermodulation distortion	The distortion that arises as the result of the modulation between two or more signals.

L

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

M

MAX	Measurement & Automation Explorer.
maximax	The absolute maximum of the calculated shock response signal over the entire signal duration.
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.
periodicity	One of the basic assumptions made in FFT-based frequency analysis.
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 device.
PSD	Power spectral density.

R

reference sound pressure	A reference pressure of 20E-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 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.
SDOF	Single degree of freedom.
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.
SFDR	*See* spurious free dynamic range.
Shannon Sampling Theorem	States that to properly sample a signal, the signal must not contain frequencies above the Nyquist frequency.
shock response spectrum	A processing method which evaluates the severity of a shock signal.
signal in noise and distortion	The ratio of the input signal to the sum of noise and harmonics.
SINAD	*See* signal in noise and distortion.

single-shot acquisition	An acquisition that runs only one time.
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.
spurious free dynamic range	The dynamic range from full-scale deflection to the highest spurious signal in the frequency domain.
SRS	*See* shock response spectrum.
STFT	Short-Time Fourier Transform.
SVL	Sound and Vibration Library. SVL is a collection of VIs shared by the Sound and Vibration Toolkit and other NI Toolkits.
SVT	Sound and Vibration Toolkit.

T

tach	*See* tachometer.
tachometer	Device used to measure the rotational speed of a rotating part.
THD	*See* total harmonic distortion.
THD+N	*See* total harmonic distortion plus noise.
total harmonic distortion plus noise	The ratio, in decibels, of the sum of noise and harmonics to the input signal.

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.
total harmonic distortion	The ratio of the sum of harmonics to the fundamental tone.
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.
velocity	A vector quantity whose magnitude is a body's speed and whose direction is the body's direction of motion.

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 µPa. 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. *See also* 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. *See also* spectral leakage.

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