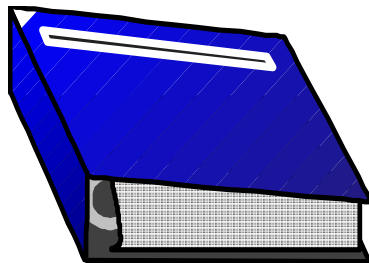
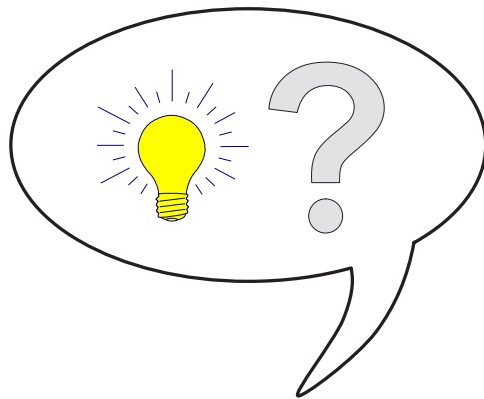


Theory Manual

Polytec Scanning Vibrometer



**PSV
PSV-3D
MSA
MSV**

As Of Software 8.4

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Appendix A: Introduction to Complex Numbers

Index

1 Introduction

Polytec's Scanning Vibrometers measure and analyze vibrations. If you are familiar with the theoretical background of signal processing and analysis, you can use the data acquisition system more efficiently. For further details refer to the specialist literature.

This manual explains

- how harmonic vibrations are described,
- how a time signal is transformed into a frequency spectrum by means of Fourier transformation,
- how alias rejection works,
- the effects of window functions,
- the effects of averaging,
- the calculation of frequency response and coherence,
- the choice of the appropriate excitation signal,
- the advantages of multiple reference analysis (MIMO),
- how to use digital filters to condition time signals,
- the theoretical background for PC-based digital demodulation (VDD)
- how to optimize the measurement setup for the PSV,
- how to work with 3D geometries and
- the introduction to complex numbers.

2 Description of Harmonic Vibrations

Vibrations can be described with trigonometric functions or complex numbers. Based on the special case of harmonic vibrations, the following describes the basics as needed for the following chapters.

2.1 Harmonic Vibrations described with Trigonometric Functions

Harmonic vibrations have a sinusoidal shape. A harmonic vibration is described by frequency, amplitude and zero phase angle as follows

$$U(t) = A \cos(\omega t + \varphi_0) \quad \text{Equation 2.1}$$

where

$$\omega = 2\pi f \quad \text{Equation 2.2}$$

A... Amplitude
 ω ... Angular frequency
 φ_0 ... Zero phase angle
 f... Frequency.

By means of the addition theorems, we can rewrite equation 2.1 as

$$U(t) = A \cos \varphi_0 \cos \omega t - A \sin \varphi_0 \sin \omega t \quad \text{Equation 2.3}$$

or

$$U(t) = A_{\cos} \cos \omega t - A_{\sin} \sin \omega t \quad \text{Equation 2.4}$$

where

$A_{\cos} = A \cos \varphi_0$... Amplitude of the cosine part
 $A_{\sin} = A \sin \varphi_0$... Amplitude of the sine part.

2 Description of Harmonic Vibrations

A harmonic vibration can thus be described as a superposition of a sinusoidal and cosinusoidal vibration whose amplitudes depend on the zero phase angle.

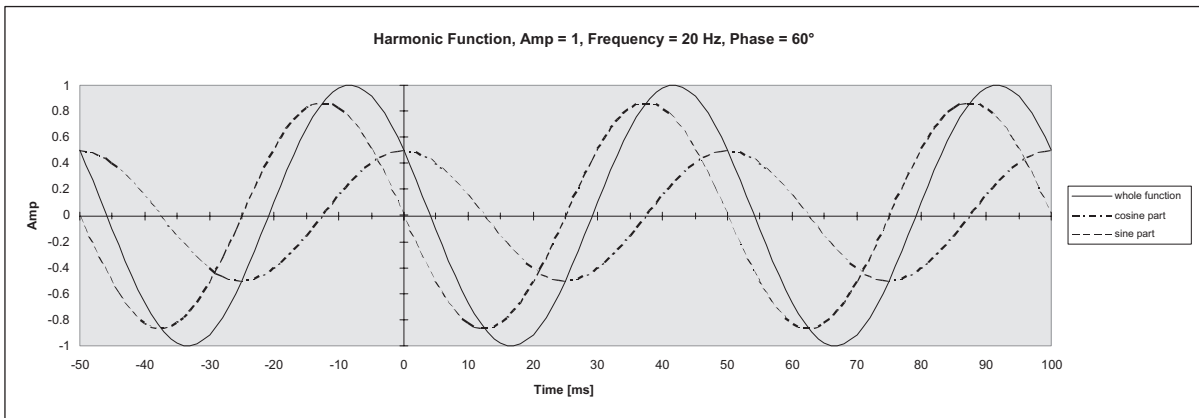


Figure 2.1: Harmonic vibration, split into cosinusoidal and sinusoidal part

2.2 Harmonic Vibrations described with Complex Numbers

Instead of trigonometric functions, you can also use complex numbers to describe harmonic vibrations. This is advantageous for mathematical analysis and allows clear presentation in the complex number plane. If you are not familiar with complex numbers, you will find a short introduction to the calculus with complex numbers in [appendix A](#).

Complex numbers are underlined in this section.

With complex numbers, a harmonic vibration $\underline{U}(t)$ can be written as

$$\underline{U}(t) = A e^{i(\omega t + \varphi_0)}. \quad \text{Equation 2.5}$$

The argument of the exponential function comprises a time-dependent part $i\omega t$ and a constant part $i\varphi_0$.

The vibration can be made clearer in the complex pointer diagram. It corresponds to a complex pointer which revolves around the origin of coordinates with angular frequency ω .

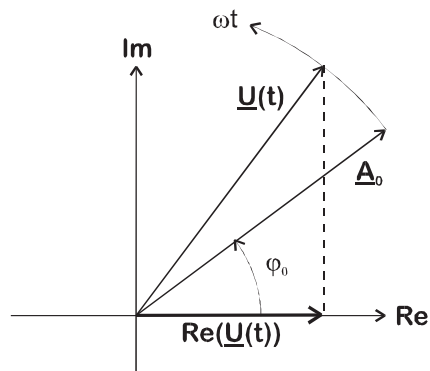


Figure 2.2: Vibration in the complex pointer diagram

The projection of the pointer to the real axis corresponds to the time-dependent instantaneous value of the real vibration.

$$\begin{aligned} \operatorname{Re}[Ae^{i(\omega t + \varphi_0)}] &= A\operatorname{Re}[\cos(\omega t + \varphi) + i\sin(\omega t + \varphi)] \\ &= A\cos(\omega t + \varphi) \end{aligned} \quad \text{Equation 2.6}$$

Equation 2.5 can be rearranged as follows

$$\begin{aligned} \underline{U}(t) &= Ae^{i\varphi_0} e^{i\omega t} \\ &= \underline{A_0} e^{i\omega t} \end{aligned} \quad \text{Equation 2.7}$$

where

$$\underline{A_0} = Ae^{i\varphi_0} \dots \text{Complex amplitude.}$$

The complex amplitude contains information on the amplitude and the zero phase angle of the vibration.

2 Description of Harmonic Vibrations

3 Transformation of the Time Signal to the Frequency Spectrum

3.1 Fourier Transformation

Periodic functions can be described as a sum of trigonometric functions (Fourier series). For example, a rectangular vibration can be described by the sum of the fundamental oscillation ($\sin\omega_0t$ = 1st harmonic) and its odd harmonics. This results in

$$U_{\text{rec}}(t) = \sin\omega_0t + \frac{1}{3}\sin3\omega_0t + \frac{1}{5}\sin5\omega_0t + \dots$$

$$\frac{1}{7}\sin7\omega_0t + \dots$$

Equation 3.1

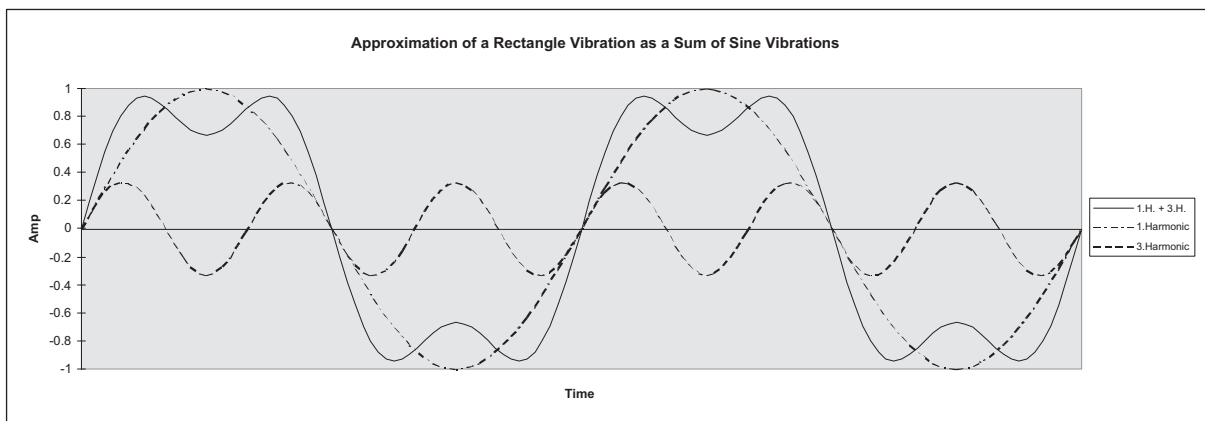


Figure 3.1: Fundamental oscillation (1st harmonic) superimposed with the 3rd harmonic

Non-periodic functions can be described as an integral of trigonometric functions (Fourier integral). An example is a single rectangular pulse.

The Fourier transformation splits any time signal into a sum of vibrations with different frequencies. All information which the time signal contains remains available. The original time signal is obtained by an inverse transformation.

3.1.1 Frequency Spectrum of a Rectangular Vibration

In the FFT spectrum of a rectangular vibration, lines appear at odd harmonics of the fundamental frequency f_0 . This corresponds to splitting the rectangular vibration into sinusoidal vibrations (refer to [section 3.1](#)).

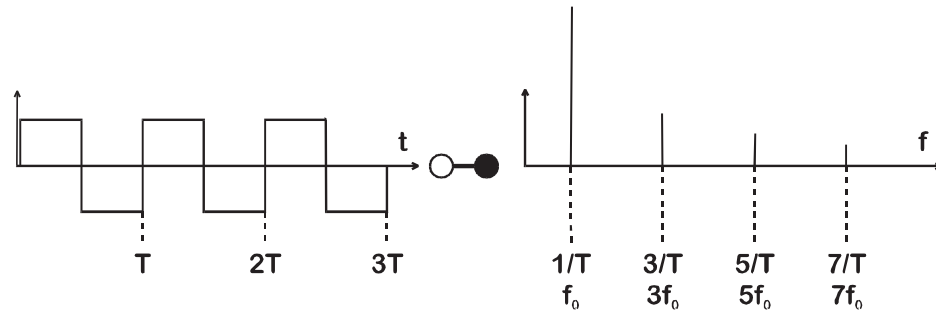


Figure 3.2: Rectangular vibration and its frequency spectrum

3.1.2 Frequency Spectrum of a Harmonic Vibration

A harmonic vibration with the frequency f_0 generates one line with frequency f in the FFT spectrum.

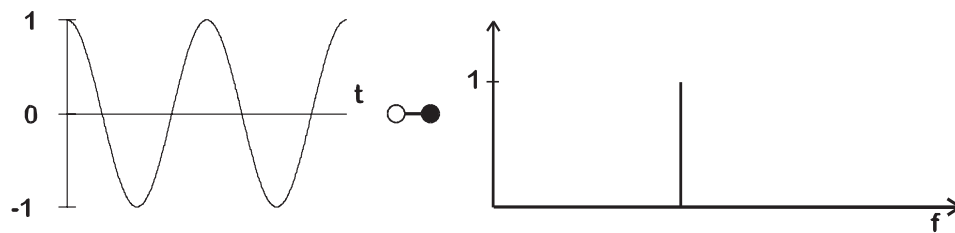


Figure 3.3: Harmonic vibration (Dirac Pulse) and its frequency spectrum

3.1.3 Frequency Spectrum of a Delta Function

The Delta Function is infinitely short and infinitely high. The integral of the Delta Function has the value 1. The frequency spectrum of the Delta Function has the value 1 at all frequencies.

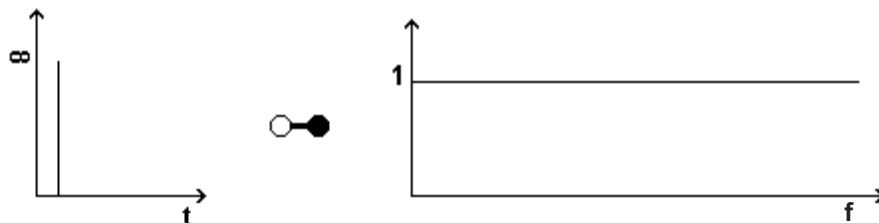


Figure 3.4: Delta Function and its frequency spectrum

An approximation of the Delta Function is a hammer blow.

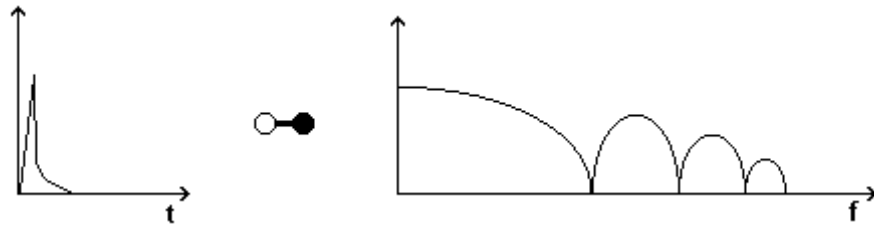


Figure 3.5: Hammer blow and its frequency spectrum

With a hammer blow a constant excitation of each frequency no longer takes place. This must be considered when using a hammer blow to excite vibrations. The FFT spectrum of a single rectangular pulse is similar to the hammer blow spectrum.

3.2 Generation of the Frequency Spectrum in the Software

The recorded time signal consists of a discrete number of samples. The software uses the FFT procedure (Fast Fourier Transformation) to generate the corresponding frequency spectrum.

Each frequency which appears in the time signal generates a spectral line in the FFT spectrum.

The number of samples which are processed by the FFT must be a power of two.

The FFT is normalized in the software such that a harmonic vibration in the time signal with amplitude 1 generates a line with amplitude 1 in the frequency spectrum if the harmonic vibration is periodic in the time window (refer to [chapter 5](#)).

3.3 Calculation of 1/3 Octave Bands in the Software

1/3 octave band analysis is widely used in acoustics. 1/3 octave band data is calculated from the FFT data by means of averaging the square of the FFT data and extracting the root of this value. The time window is considered with a correction factor. The display of 1/3 octave bands is also possible for signals calculated from a combination of channels (Vibrometer and Reference) like FRF, H1, H2 and Coherence. Please note that in this case first the combined signals are calculated (refer to [chapter 7](#)) and then the averaging in the 1/3 octave frequency bands is done.

3 Transformation of the Time Signal to the Frequency Spectrum

Center frequency For the center frequency f_m of the nth band we have

$$f_{m,n} = 10^{0,1n} \quad \text{Equation 3.2}$$

n... Integer.

For the output, the calculated values are rounded according to the National Standard DIN EN 60651.

Band limits With a logarithmic frequency scale, the bands limits are symmetric with respect to the center frequency. The following applies to the lower limit $f_{l,n}$

$$f_{l,n} = \frac{f_{m,n}}{10^{0,05}} \quad \text{Equation 3.3}$$

and the upper limit $f_{u,n}$

$$f_{u,n} = f_{m,n} \cdot 10^{0,05} \quad \text{Equation 3.4}$$

As there is no standard for rounding band limits, we use our own scheme for this and thus obtain the results in [table 3.1](#).

Table 3.1: Rounding of 1/3 octave band limits

f_m [Hz]	f_l [Hz]	f_u [Hz]
1.00	0.89	1.12
1.25	1.12	1.40
1.60	1.40	1.80
2.00	1.80	2.25
2.50	2.25	2.80
3.15	2.80	3.50
4.00	3.50	4.50
5.00	4.50	5.60
6.30	5.60	7.10
8.00	7.10	8.90
10.0	8.90	11.2

Display On condition that a band contains at least two FFT lines, 1/3 octave bands are displayed approximately from the 8th FFT line onwards.

Integration and differentiation Integration and differentiation of 1/3 octave bands are possible and are performed by means of the center frequency. The maximum theoretical error for this amounts to ± 1.0 dB.

4 The Alias Effect

The software records and digitally processes analog time signals. Under certain conditions, the spectrum of digitally sampled signals may be falsified by the alias effect. In [section 4.1](#) is described how alias rejection works in the MSA, MSV and PSV. If you are not familiar with the alias effect, read [section 4.2](#).

4.1 Alias Rejection

All MSA, MSV, and PSV systems as well as the PSV 200 with an upgraded data management system provide built-in alias suppression. You do not need to take any measures to suppress aliases.

- ☞ If you use the high-frequency decoder VD-05 or DD-300, alias rejection is only available for bandwidths above 4MHz or sample frequencies above 10.24MHz. With lower bandwidths or sample frequencies high-frequency signal portions and noise levels are not suppressed and interfere the measurement signal. In this case, we recommend to use the broadband decoder.

4.2 Background Information on the Alias Effect

You can picture an actual analog time signal as a series of an infinite number of samples with an infinitely short distance. To digitally process analog time signals, they must be sampled for a certain period with a sampling frequency f_{Sample} . The number of samples becomes limited, which naturally means information is lost. It must therefore be made sure that the digital samples within the required accuracy provide the same information as the analog time signal.

If the time signal is sampled k times, the following applies

$$Wt = \frac{K}{f_{\text{Sample}}} \quad \text{Equation 4.1}$$

Wt ... Length of a measured time signal = time window

K ... Number of samples

f_{Sample} ... Sampling frequency.

For correct digital signal processing, the sampling theorem must be fulfilled. This is guaranteed when

$$f_{\text{Sample}} > 2f_{\text{Signal}} \quad \text{Equation 4.2}$$

f_{Sample} ... Sampling frequency

f_{Signal} ... Maximum signal frequency.

4 The Alias Effect

Signals with frequencies $f_{\text{Signal}} > f_{\text{Sample}}/2$ generate artificial lines at frequencies $f_{\text{Alias}} < f_{\text{Sample}}/2$. This is called the **alias effect**. The spectrum is falsified. To avoid infringement of the sampling theorem, signal frequencies which are higher than half the sampling frequency must be suppressed or the sampling frequency must be high enough.

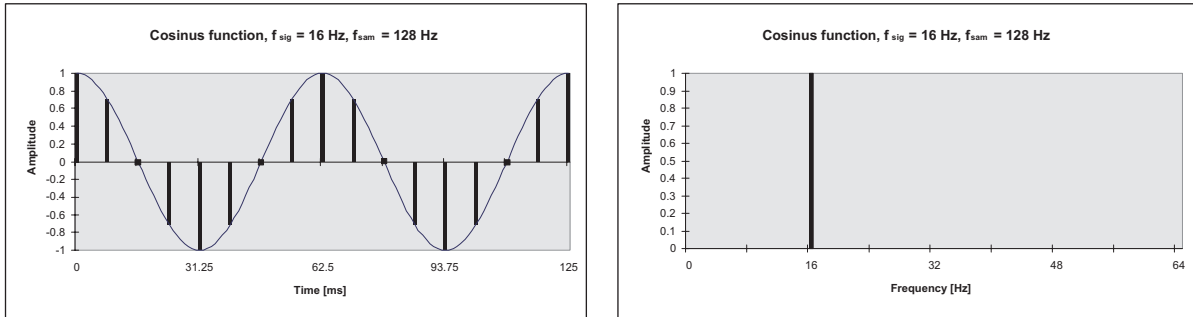


Figure 4.1: Sampling of a cosine function, sampling theorem fulfilled

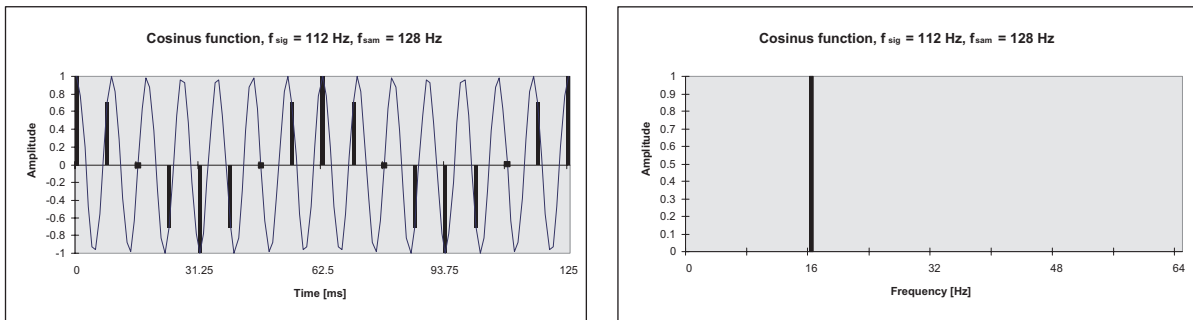


Figure 4.2: Sampling of a cosine function, sampling theorem infringed

Example

Cosine function, sampled with $K=16$ and $f_{\text{Sample}}=128\text{ Hz}$.

With $f_{\text{Signal}}=16\text{ Hz}$, a spectral line is generated at 16 Hz. With $f_{\text{Signal}}=112\text{ Hz}$, the spectral line is also generated at 16 Hz. The spectra of $f_{\text{Signal}}=16\text{ Hz}$ and $f_{\text{Signal}}=112\text{ Hz}$ cannot be distinguished. The artificial spectral line (alias) with $f_{\text{Signal}}=112\text{ Hz}$ developed because the Nyquist theorem was infringed.

5 Window Functions

5.1 Basics

The result of the FFT is a discrete spectrum. Only frequencies which fall exactly on an FFT line are shown correctly in the frequency range. If there are frequencies in the time signal which do not fall on an FFT line, the information in the spectrum is distributed over the neighboring FFT lines. This is known as the leakage effect.

The leakage effect can be reduced by applying window functions. The time signal is weighted with the window function before calculating the FFT. For most window functions, the weighted time signal becomes zero at the start and the end of the time window which thus avoids signal jumps at the edges of the time window.

The most simple window function is the rectangular window. The multiplication of a time signal with the rectangular window is synonymous with cutting off of the time signal outside the time window.

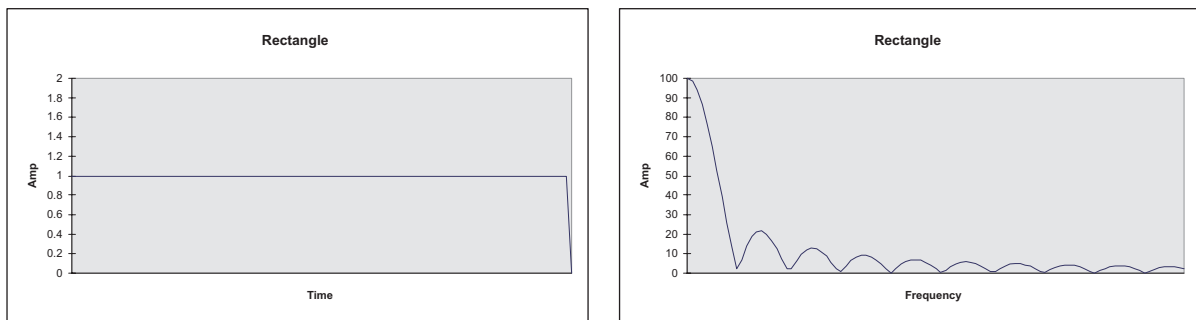


Figure 5.1: Rectangular window and its spectrum

The following applies to a periodic time signal:

- If the time signal is **periodic in the time window**, the periodic continuation of the windowed signal corresponds to the actual time signal (see [figure 5.3](#), rectangle, signal without leakage).
- If the time signal is **non-periodic in the time window**, the periodic continuation of the windowed signal no longer complies with the actual time signal. This cannot be interpreted properly by the FFT and the frequency spectrum is falsified (see [figure 5.3](#), rectangle, signal with leakage).

Generally, in actual signals the periodic continuation of the windowed signal is not the same as the actual time signal. To minimize leakage, most window functions artificially minimize the amplitudes at the beginning and the end of the time window. Thus the phase shifts are minimized in the periodic continuation of the windowed time signal. Hanning, Hamming, Blackman Harris, Bartlett and Flat Top are among these window functions.

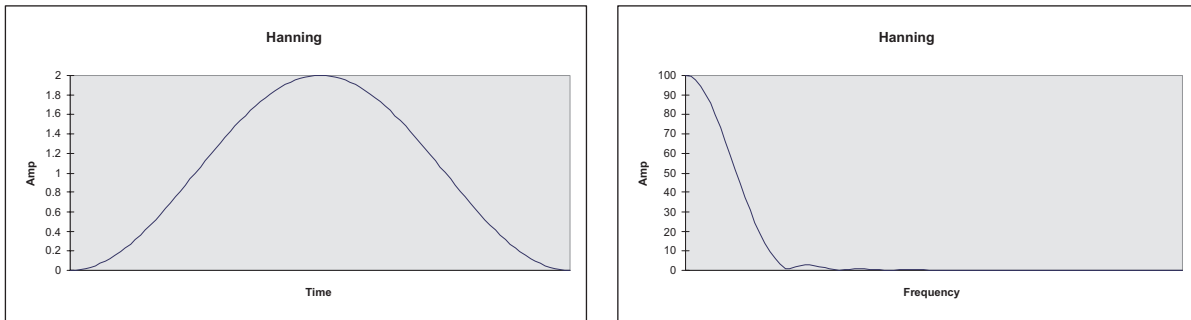


Figure 5.2: Hanning window and its spectrum

5.2 Attenuation and Ripple

All windows influence time signals which are non-periodic in the time window. A window function changes line shape and amplitude of the spectral lines in the frequency spectrum in actual measurements. Based on the spectrum of a window function, you can estimate how the window function influences the measured spectrum.

The spectrum of the rectangular window consists of a main maximum and secondary maxima. This applies to all window functions which artificially minimize amplitudes at the beginning and the end of the time window (Hanning, Hamming, Blackman Harris, Bartlett, Flat Top).

Attenuation

A measure of the frequency resolution of a window function is the frequency value by which a sharp spectral line in the windowed spectrum widens. A deciding factor is at which attenuation of the spectral line the line widening may appear.

If **low** attenuation is required, the frequency resolution depends on the width of the main maximum of the window spectrum. Half the width of the main maximum of the rectangular window ($= 2\pi(f_{\text{Sample}}/K)$) serves as a means of comparison for the various window functions. The required attenuation is already achieved within the main maximum of the window spectrum and must be lower than the attenuation of the amplitude of the highest secondary maximum.

If **higher** attenuation is required, the frequency resolution depends on the amplitude of the highest secondary maximum in the window spectrum (highest side lobe attenuation). The required attenuation is achieved at the highest secondary maximum.

If **very high** attenuation is required, the frequency resolution depends on how fast the amplitudes of the secondary maxima decrease in the window spectrum (roll-off rate). The required attenuation is not yet achieved at the highest secondary maximum.

Normally, higher attenuation is required to clearly define spectral lines. The frequency resolution of a window function increases with the length of the time window.

With increasing length of the time window,

- the width of the main maximum decreases in the window spectrum,
- the amplitude of the secondary maxima in the window spectrum falls to lower values in most window functions as the window spectrum is longer.

The following applies

$$Wt = \frac{K}{f_{\text{sam}}} > \frac{n_K}{\Delta f_0} \quad \text{Equation 5.1}$$

Wt... Time window = length of the measured time signal

K... Number of samplings

f_{Sample} ... Sampling frequency

n_K ... Index of the secondary maxima with required attenuation with regard to the main maximum

Δf_0 ... Maximum permissible one-sided widening of a spectral line.

All window functions are conceived for certain measurement tasks. With a window function which reaches a certain attenuation with a smaller n_K , you receive a higher frequency resolution with the same window width. Select a window function according to your measurement conditions.

Ripple

Ripple describes the quality of the reproduction of the amplitude of a window function. A windowed sinusoidal time signal with a frequency in the center of the bandwidth is observed. Ripple is the maximum attenuation of the frequency amplitude of this signal which is non-periodic in the time window. Windows with a very wide main maximum have the least Ripple. The frequency resolution decreases with increasing width of the main maximum.

5.3 Comparison of the Rectangular Window and the Hanning Window

The rectangular window does **not** falsify the spectrum of signals which are periodic in the time window, see [figure 5.3](#).

The rectangular window **strongly** falsifies the spectrum of signals which are non-periodic in the time window.

The Hanning window falsifies the spectrum of signals which are

- periodic in the time window **more** than the rectangular window,
- non-periodic in the time window **less** than the rectangular window.

5 Window Functions

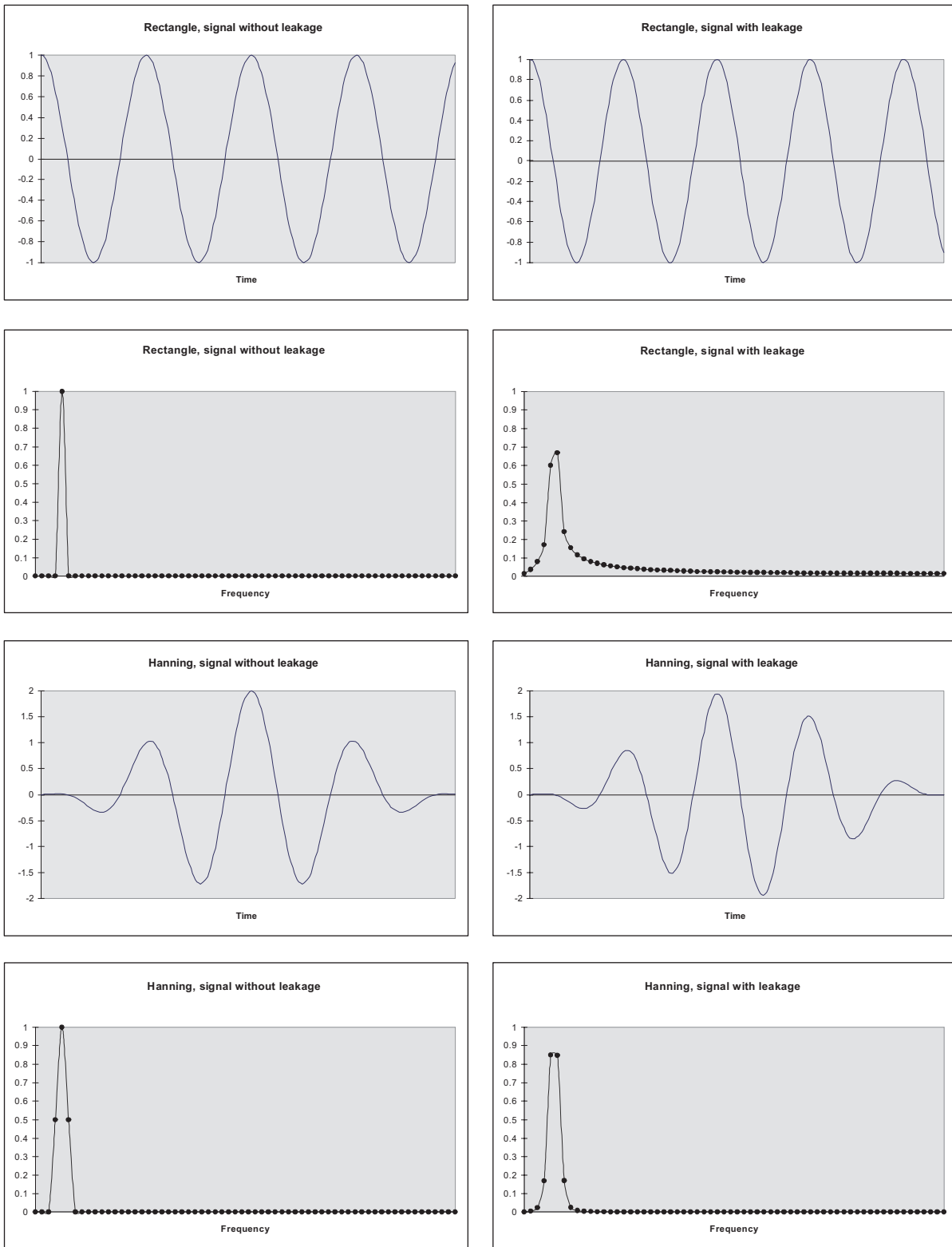


Figure 5.3: Comparison of the rectangular window with the Hanning window

5.4 Window Functions in the Software

The window functions which are implemented in the software are normalized as follows: A sinusoidal time signal which is periodic in the time window and with a frequency in the center of the bandwidth is considered. The rectangle window function reproduces the frequency amplitude of this signal in a pure manner. In order to also apply this to the window functions Hanning, Hamming, Blackman Harris, Bartlett and Flat Top, these window functions are multiplied with a scaling factor.

The individual window functions are described in detail in the following. In [section 12.8](#) you will find a brief overview of the most important parameters for the window functions implemented.

Rectangle

The main maximum in the spectrum of the rectangular window is the narrowest in comparison with other window spectrums.

Ripple: 3.92 dB, Highest side lobe attenuation: 13.3 dB.

The rectangular window is **highly** suitable

- for signals which are periodic in the window,
- for non-periodic signals which decrease to zero within the time window (e.g. hammer blows).

The rectangular window is **not** suitable

- for higher and very high frequency resolutions, as the amplitude attenuation of the highest secondary maxima is low in the spectrum (highest side lobe attenuation is low) and the amplitudes of the secondary maxima only decrease slowly (roll-off rate is low).

Hanning

The Hanning window has the shape of an inverse, lifted cosine function.

With noisy measurement signals occurring in a noise excitation, a window function should display even-sized attenuating behavior in the complete window spectrum.

Ripple: 1.42 dB, Highest side lobe attenuation: 31.5 dB.

The Hanning window is **highly** suitable for strong noisy measurement signals as

- the amplitudes of the secondary maxima decrease rapidly (roll-off rate is high),
- the main maximum is only double the width compared to that in the rectangular window spectrum,
- the attenuation is constant.

Hamming

The Hamming window is a combination of the Hanning window and the rectangular window.

Ripple: 1.75 dB, Highest side lobe attenuation: 42.6 dB.

In comparison to the Hanning window, the Hamming window is **more** suitable

- for the frequency resolution of spectral lines in attenuations above the highest side lobe attenuation.

The Hamming window is **not** better suited than the Hanning window

- for very noisy measurement signals and the frequency resolution of spectral lines in an attenuation beneath the highest side lobe attenuation of the Hamming window, as the amplitudes of the secondary maxima only decrease slowly (roll-off rate is low) and the main maximum is double the width as compared to the rectangular window spectrum.

Blackman Harris

It is a good compromise for a window with hardly any influence on the line shape and amplitude of the spectral lines.

Ripple: 0.83 dB, Highest side lobe attenuation: 92.1 dB.

The Blackman Harris window is **highly** suitable

- for the frequency resolution of strongly attenuated spectral lines, as the highest side lobe attenuation is very large and the secondary maxima decrease quickly at the beginning,
- for periodic measurement signals.

As the main maximum in the Blackman Harris window spectrum is four times as wide as in the rectangular window spectrum, the Blackman Harris window is **not** better suited

- than the Hanning window for very noisy measurement signals as the attenuating behavior is very irregular in the Blackman Harris window spectrum,
- for the frequency resolution of weakly attenuated spectral lines.

Bartlett

The Bartlett window has the shape of an isosceles triangle.

The features of the Bartlett window lie between the rectangular window and the cosinusoidal windows Hanning, Hamming and Blackman Harris.

Ripple: 1.82 dB, Highest side lobe attenuation: 26.5 dB.

Flat Top

The Flat Top window is based on the cosine function. The main maximum in the Flat Top window spectrum is five times wider than in the rectangular window spectrum.

Ripple: 0.02 dB, Highest side lobe attenuation: 68.3 dB.

The Flat Top window is **highly** suitable for

- a good reproduction of the amplitude height, e.g. for calibration purposes, as the ripple is very small,
- for measurements with sinusoidal excitation.

The Flat Top window is **not** suitable

- for very noisy measurement signals, as the attenuation is very irregular,
- for measurements which require a detection of immediate spectral lines and a good reproduction of the line shapes of spectral lines.

Exponential

Unlike the other window functions, the exponential window does not attenuate signals at the beginning of the window.

The exponential window increases the attenuation proportionally. An exponentially decreasing signal is attenuated to the e -th part ($e=2.718\dots$) of its initial value according to the time constant τ .

The exponential window is **highly** suitable

- for measurement signals which are excited with pulses (hammer blow) and are only attenuated weakly. With such measurement signals, the amplitude is high at the beginning of the time window and decreases slowly in an exponential manner. If the amplitude is not nearly zero at the end of the time window, the measurement is falsified. This effect is prevented by the exponential window and it receives the characteristic exponential decrease of the measurement signal.

The exponential window is **not** suitable

- for unattenuated periodic measurement signals.

6 Averaging

Measured signals are always superimposed by noise. To minimize the noise level, you can average in the software. This chapter describes averaging in time and frequency domain.

6.1 Time Domain Averaging

In time domain averaging a sequence of time traces is collected and averaged. A time trace is a contiguous sequence of samples. As time traces are acquired at different times and combined to the average result, the measurement conditions should be repeatable and the same for each time trace. This can be achieved for example by synchronizing the measurement to the excitation with the help of a trigger.

To use time domain averaging switch the acquisition mode to Time on the General page of the A/D Settings dialog, choose averaging mode Time and select the number of averages that contribute to the average result.

The time traces are collected one after the other and have the same number of samples. A single sample of the average result at time $t = t_0 + \Delta t$ after the first sample of the result is calculated from the samples at times $t_i = t_{0i} + \Delta t$ of the time traces i . Here t_{0i} is the start of time trace i . Therefore the average result has the same number of samples as the time traces. See [figure 6.1](#) for an illustration. The illustration shows that it is necessary to trigger the data taking of every time trace synchronized to the shown sine wave. Otherwise you would not get meaningful results from time domain averaging.

6 Averaging

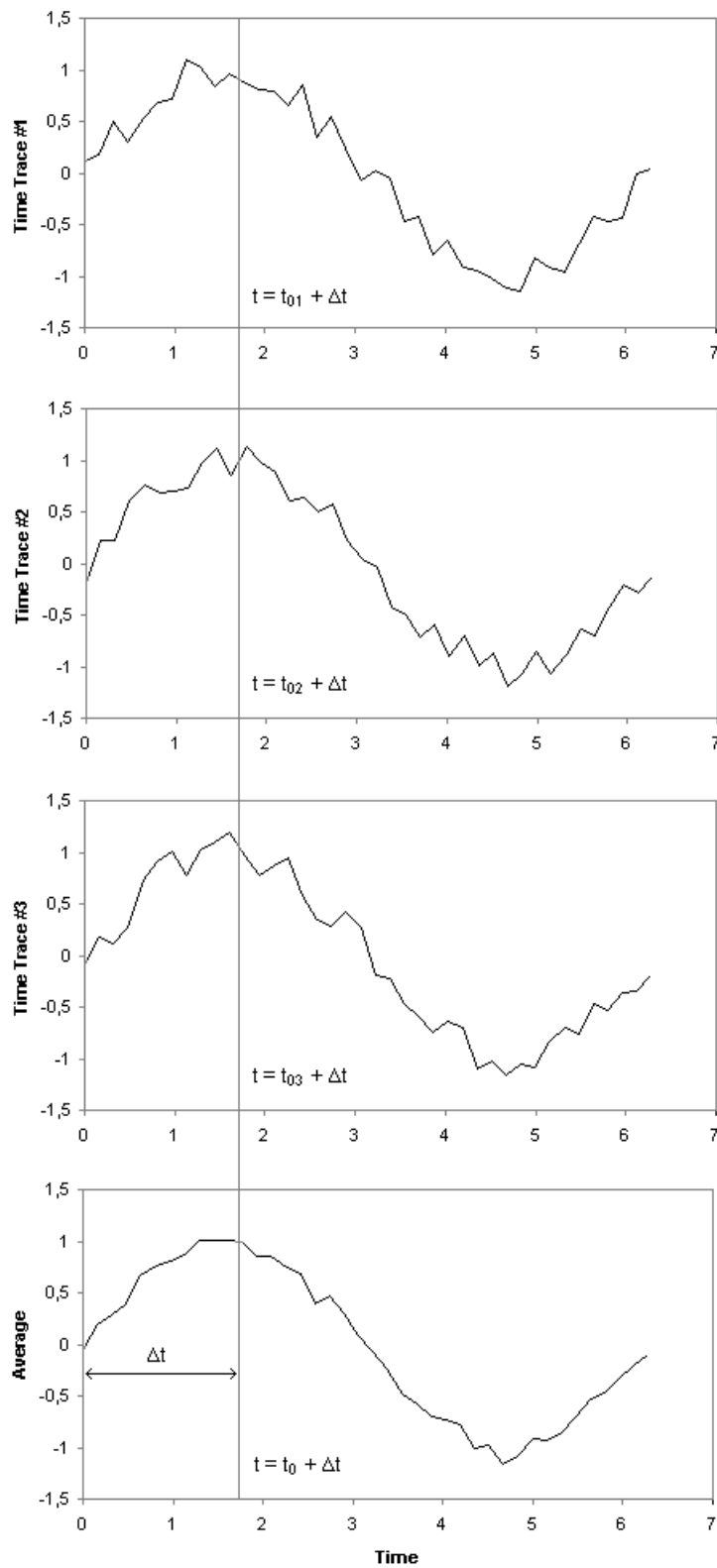


Figure 6.1: Principle of time averaging

There are two different modes available in time domain averaging: mean and median averaging. To select median averaging for an acquisition channel, check the corresponding box on the SE (Signal Enhancement) page of the A/D Settings dialog.

In mean averaging mode the average is calculated by adding up the samples and dividing by the number of samples:

$$\bar{T} = \frac{1}{N} \cdot \sum_{n=1}^N T_n \quad \text{Equation 6.1}$$

In median averaging mode the samples at a given time after start of the traces are first sorted and then the middle value of the sorted values is used as the average value. If the number of traces is even, the mean of the two values adjacent to the middle is calculated. Median averaging mode is especially useful to suppress low probability, high amplitude noise, such as dropouts of the vibrometer signal.

The following table demonstrates the differences between median and mean averaging mode. Column A shows an example for 3 measurement values to be averaged. Both, the mean and median average give the same result. Column B shows the same sequence, except the middle value, which has been altered by a dropout in the vibrometer signal. The dropout clearly distorts the mean average, whereas it has little effect on the median average.

Trace / Sample	#1	#2
#1	3	3
#2	2	20
#3	4	4
Mean average	3	9
Median average	3	4

6.2 Frequency Domain Averaging

As in time domain averaging, in frequency domain averaging a sequence of time traces is collected. Each time trace has the same number of samples. From each time trace a spectrum is calculated by the means of an FFT. All spectra have the same number of FFT lines. The averaged spectrum is obtained by averaging all values at each frequency.

The following sections describe the different modes available in frequency domain averaging. There are some general rules for selecting complex or magnitude averaging mode for different applications:

If the phase between output and input signal is stable you can use complex averaging to reduce noise that is not phase correlated to the input signal. If the phase between output and input signal is not stable you can use magnitude averaging to obtain an average amplitude.

For example, complex averaging is used for deterministic excitations like a shaker driven with a periodic chirp signal whereas magnitude averaging is used for stochastic excitations like in a wind tunnel.

6.3 Complex Averaging

With complex averaging, N real parts and N imaginary parts of N measurement values are added and divided by N.

$$\bar{S} = \frac{1}{N} \sum_{n=1}^N S_n = \frac{1}{N} \cdot \left(\sum_{n=1}^N \text{Re}(S_n) + i \cdot \sum_{n=1}^N \text{Im}(S_n) \right) \quad \text{Equation 6.2}$$

This corresponds to the pointer in figure 6.2: $\frac{1}{2} \cdot (S_1 + S_2)$.

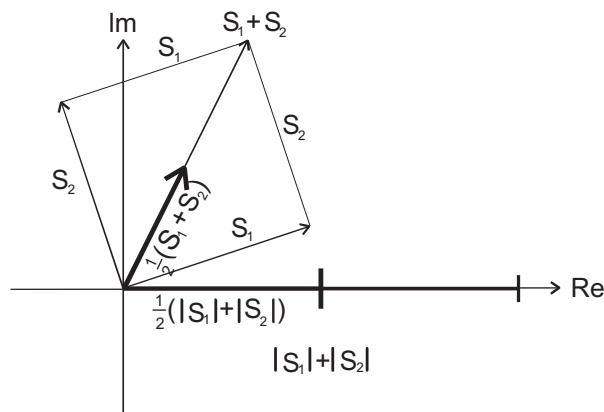


Figure 6.2: Complex and magnitude averaging of signals S₁ and S₂ (number of averages 2)

- ☞ If you perform measurements without a trigger the start of the measurement and therefore the phase is arbitrary. Nevertheless there is a fixed relation between the phase of the reference channel and the vibrometer channel. Prior to complex averaging the software relates the phase of all channels to that of the reference channel with the lowest number (see [table 12.2](#)).

6.4 Magnitude Averaging

With magnitude averaging, first the magnitudes of the complex values are calculated, then these N magnitude values are added and divided by N.

$$\overline{|S|} = \frac{1}{N} \cdot \sum_{1=n}^N |S_n| \quad \text{Equation 6.3}$$

This corresponds to the length of $\frac{1}{2}(|S_1| + |S_2|)$ in [figure 6.2](#).

- ☞ For crosspower signals complex averaging is used even if magnitude averaging mode is selected on the A/D Settings general page, refer to [chapter 7](#).

When using magnitude averaging, all phase information is lost. In order to allow an animation of the operating deflection shape, the phase of the crosspower signal with the reference channel with the lowest number is assigned to the averaged result of the other channels (see [table 12.2](#)). The animation only makes sense if there is some phase correlation between reference and vibrometer channel.

6.5 Peak Hold Averaging

Select Peak Hold for averaging if you want to calculate and display the respective maximum of the spectra over the set number of spectra.

The reference channel plays a special role with Peak Hold, as the measurement values of this channel decide which measurement values are to be used in the Peak Hold average spectrum for all channels. If several reference channels have been selected, then the one with the lowest channel number is used. If no reference channel has been selected, then the vibrometer channel with the lowest channel number is used. In the following, only the term reference channel is used.

During the measurement the spectra of the reference channel are analyzed. If, at a certain frequency, the magnitude of the current spectrum exceeds the maximum magnitude saved until now, then the measurement values of **all channels** at this frequency are saved as Peak Hold values. If there are complex measurement values, then they are also saved as complex values; however only the magnitude of the reference channel is used as the comparative criteria.

The way this algorithm works can be shown using as an example a Sweep excitation which is connected to the reference channel. At the current Sweep frequency, the spectrum of the reference channel shows a peak. The magnitude of this peak is usually larger than that of all previous spectra. For this reason, the measurement values of all channels at this Peak frequency are copied into the Peak Hold average spectra. Once the Sweep has finished, the Peak Hold spectra from all channels are made up of data which corresponds to a peak magnitude of the reference.

This approach has several advantages. The Peak Hold spectra contain the required data, i.e. the data at which excitation was at a maximum. Apart from that, it is ensured that the measurement values from all channels at a given frequency come from spectra acquired at the same time. Therefore the H1, H2 and coherence signals can be calculated in the usual consistent way (refer to [chapter 7](#)). Furthermore, a brief drop out of the vibrometer signal in a single spectrum only affects the frequencies around the current Sweep frequency, but not the whole frequency range.

To further improve the signal quality, you can activate Signal Enhancement for the relevant channels. Then the combined signal quality of these channels is analyzed. The quality of the averaging spectra is compared to a threshold and as a result of this, the measurement values are either discarded for all channels or are used for calculating the Peak Hold spectra. In a sequence of measurements, the threshold automatically adapts to the average quality of the signals.

- ☞ If you select Peak Hold averaging, you cannot make any further quality adjustment for Signal Enhancement (Fast ... Best).
- ☞ If you want to use a sweep excitation together with peak hold averaging, the sweep time should correspond to the measurement time. As a rule of thumb, the number of averages should equal to the number of measured FFT lines. To calculate the measurement time, multiply the number of averages with the sample time (displayed on the A/D settings dialog's frequency page) and 100% minus the used overlap (in percent). The measurement time should be slightly larger than the sweep time to ensure that all frequencies of the sweep contribute to the result.
- ☞ Disable all reference channels if you want to perform the Peak Hold based on the signal of the vibrometer channel.

7 Calculation of Frequency Response and Coherence

The frequency response of a vibrating system is the magnitude and phase response of the system at all frequencies of interest. The frequency response is defined as the Fourier Transform of the Impulse Response of a system.

Frequency response measurements require the excitation of the vibrating system with energy at all measured frequencies. The fastest way to perform the measurement is to use a broadband excitation signal that excites all frequencies simultaneously, and use FFT techniques to measure at all of these frequencies at the same time.

The coherence function is a measure of the power in the output signal caused by the input. If the coherence is 1, then all the output power is caused by the input. If the coherence is 0, then none of the output is caused by the input.

In PSV and VibSoft the frequency response (FRF, H1, H2) and coherence functions (COH) are calculated in several steps. First FFTs are calculated of all active channels. Then for each channel the auto power spectra (AP) and for every possible combination of a Vibrometer and reference channel the cross power spectra (CP) are calculated:¹

$$AP_{SS} = S \cdot S \quad \text{Equation 7.1}$$

$$CP_{RV} = R \cdot V \quad \text{Equation 7.2}$$

$$CP_{VR} = V^* \cdot R = CP_{RV}^* \quad \text{Equation 7.3}$$

Here S is the spectrum at any channel, V the spectrum of a Vibrometer channel (output) and R the spectrum of a reference channel (input). * denotes the complex conjugate (refer to [section A.4](#)).

Without averaging the FRF is calculated from the cross power and auto power spectra:

$$FRF_{VR} = \frac{CP_{RV}}{AP_{RR}} = \frac{V}{R} \quad \text{Equation 7.4}$$

¹ All variables in this chapter (AP, CP, S, V, R, FRF, H1, H2, N) are functions of the frequency f. For better readability the argument (f) is omitted in the functions.

With averaging the response functions H1 and H2 and the coherence function COH are calculated:

$$H1_{VR} = \frac{\overline{CP_{RV}}}{\overline{AP_{RR}}} \quad \text{Equation 7.5}$$

$$H2_{VR} = \frac{\overline{AP_{VV}}}{\overline{CP_{VR}}} \quad \text{Equation 7.6}$$

$$COH_{VR} = \frac{H1_{VR}}{H2_{VR}} = \frac{\overline{CP_{RV}} \cdot \overline{CP_{VR}}}{\overline{AP_{VV}} \cdot \overline{AP_{RR}}} \quad \text{Equation 7.7}$$

Here the overlined values denote averaged values. As the CP signals are always averaged in complex averaging mode and AP signals are real, H1, H2 and COH do not depend on the averaging mode.

H1 and H2 are estimators for the frequency response function FRF_{sys} of the measured system. While the phase is equal for H1 and H2, the magnitude differs. H1 is mainly affected by noise in the input signal (reference signal), whereas H2 is mainly affected by noise in the output signal (vibrometer signal).

☞ Without averaging H1 and H2 have the same result as FRF_{VR} in equation 7.4. The coherence has always the value 1. Therefore without averaging FRF is displayed instead of H1 and H2. The coherence is not displayed.

The following approximations can be made (the higher the number of averages, the better the approximation):

if $N_R^2 \ll AP_{RR}$:

$$H1 = FRF_{sys} \left(1 - \frac{N_R^2}{AP_{RR}} \right) \quad \text{Equation 7.8}$$

if $N_V^2 \ll AP_{VV}$:

$$H2 = FRF_{sys} \left(1 + \frac{N_V^2}{AP_{VV}} \right) \quad \text{Equation 7.9}$$

Here, N_R^2 is the power of the noise in the input signal and N_V^2 is the power of the noise in the output signal.

It can be seen that with the presence of input noise H1 is smaller than FRF_{sys} and with the presence of output noise H2 is bigger than FRF_{sys} .

As the coherence function is $H1/H2$, it can be approximated as:

$$\text{COH} = \frac{1 - \frac{N_R^2}{\overline{AP}_{RR}}}{1 + \frac{N_V^2}{\overline{AP}_{VV}}} \quad \text{Equation 7.10}$$

A further approximation for $N_R^2 \ll \overline{AP}_{RR}$ and $N_V^2 \ll \overline{AP}_{VV}$ results in:

$$\text{COH} = 1 - \left(\frac{N_R^2}{\overline{AP}_{RR}} + \frac{N_V^2}{\overline{AP}_{VV}} \right) \quad \text{Equation 7.11}$$

Example 1

A coherence value of 0.99 indicates that the sum of the relative input noise power and the relative output noise power $\left(\frac{N_R^2}{\overline{AP}_{RR}} + \frac{N_V^2}{\overline{AP}_{VV}} \right)$ is approximately 0.01. Coherence values close to 1 indicate a good signal-to-noise ratio.

Example 2

A coherence value of 0.1 indicates that the relative input noise power or the relative output noise power or both of them are high. The approximation formula is not valid anymore as $N^2 \ll AP$ is not fulfilled. Coherence values close to zero indicate a bad signal-to-noise ratio.

☞ If the number of averages is increased:

- $H1$ converges to a smaller value than FRF_{sys} , depending on the input noise.
- $H2$ converges to a bigger value than FRF_{sys} , depending on the output noise.
- COH converges to a smaller value than 1, depending on the input and output noise.

☞ For scanning measurements, normally the output (vibrometer) noise is bigger than the input (reference) noise, at least at some scan points. For visualizing an operating deflection shape (where all scan points are displayed simultaneously), $H1$ is the better choice than $H2$ because $H1$ is less sensitive to output noise.

8 Excitation Signals

The quality of a vibration measurement depends strongly on the choice of the excitation signal. The principal criterion for the selection of the excitation signal is the purpose of the measurement. In case of data acquisition for modal analysis typically a complete spectrum must be measured at each scan point. Not all measurement tasks require a modal analysis. Questions like "where on the structure is the highest vibration amplitude?" can be answered much faster by measuring the deflection shapes at the dominant resonance frequencies.

The linearity of the system also plays an important role for the selection of the excitation signal. If the device under test has non-linearities it is often desirable to get a linear approximation. This is particularly important for doing a modal analysis, as the used parameter estimation schemes are based on linear system models. Only certain excitation signals have the ability to average out non-linearities. With other signals non-linear distortions are clearly visible in the spectrum.

An important difference between excitation signals is the RMS-to-peak ratio, which is closely coupled to the obtained signal-to-noise ratio of the measurement. A higher RMS-to-peak ratio denotes a signal with higher energy content for a given peak amplitude. Generally, the higher the excitation energy, the lower the relative noise, at least insofar as the system behaves linearly.

Some excitation signals generate leakage effects in the spectrum calculated by FFT, which are an additional noise source (refer to [chapter 5](#)). However, a variety of signals exists which do not generate leakage effects intrinsically.

A further criterion is the reproducibility of the measurement. A scanning vibrometer measures all points sequentially. If the excitation is the same for all scan points and the test object and the test setup remain unchanged during the measurement, the sequential measurement is equivalent to a parallel one. If the excitation signal is exactly the same for each scan point, the vibrometer signal can be displayed as area data. If the excitation signal is different from scan point to scan point, the frequency response function must be used to visualize area data.

Available excitation signals

The signals available in the PSV software can be grouped into four classes: harmonic, periodic in the time window, random and transient. Some signals belong to more than one class (see [table 8.1](#)).

Table 8.1: Available excitation signals and their classes

Signal	Class	RMS-to-peak	Leakage	Recommended window	Suitable for MIMO
Sine	Harmonic	0.7	no ¹	Rectangle ¹	No
Sweep	(Pseudo) Harmonic	0.7	yes	Hanning	No
Pseudo Random	Periodic	0.25	no	Rectangle	No
Periodic Chirp	Periodic	0.5	no	Rectangle	No
Burst Chirp	Transient	0.35	no ²	Rectangle	No
White Noise	Random	0.25	yes	Hanning	Yes
Burst Random	Transient, Random	0.2	no ²	Rectangle	Yes

¹ If frequency corresponds to an FFT-line

² If response dies out within the time window

- **Harmonic** signals have the best signal-to-noise ratio and allow fast measurements at single frequencies.
- Signals which are **periodic** in the time window are best suited for fast measurements of complete spectra as they excite all frequencies simultaneously and do not generate leakage effects.
- If a linear approximation of a non-linear system is required, random signals must be used. Only with **random** signals non-linear distortions are averaged out.
- **Transient** signals excite a short vibration which dies out. The time window should be selected long enough to record the complete vibration signal until it died out. No leakage occurs.

☞ External generators do not offer all of the described signals nor does the internal generator of PSV-300-U or PSV-400-B systems.

Sine: Sine excitation is used to measure the deflection shape at one particular frequency. The main advantages of Sine excitation are the very good signal-to-noise ratio and the possibility to perform a scan in a very short time. For scanning with sine excitation only the values at the excitation frequency need to be measured and saved. This can be done most efficiently using the least square method regression (FastScan), which is faster than an FFT and never suffers from leakage effects.

☞ Non-linearities can be detected by looking for harmonics in the spectrum. For that purpose measure in the FFT mode.

Sweep: Sweep denotes a sine signal, whose frequency is slowly altered, according to a ramp function. Slowly swept sine can be used for single point measurements, in combination with the peak hold function.

With Sweep excitation the resonance of a very lightly damped structure can be found which might remain undetected with other excitation signals.

The measurement time is proportional to the square of the frequency resolution. This means that normally the measurement is very slow and therefore is hardly ever used for scanning.

Periodic Chirp: Periodic Chirp is designed to excite all FFT lines of the measured spectrum. The time signal is generated out of the spectrum by an inverse Fourier transformation. Typically the magnitude is set for all frequencies to the same value. The phase is generated by an algorithm which maximizes the energy for a given maximum amplitude.

After waiting for steady state conditions the excitation and the response are measured without leakage effects. As all frequencies of interest are excited simultaneously no averaging is required. This is very useful in order to do fast measurements. However, for precise measurements averaging can be used in order to increase the signal-to-noise ratio.

The use of the rectangle window is required. Any other window will lead to major distortions.

Pseudo Random: Like Periodic Chirp, Pseudo Random is generated in the frequency domain. The only difference is the phase which is calculated as a uniformly distributed random number, giving the signal its random sound. The time signal is generated out of the spectrum by an inverse Fourier transformation. The properties of Pseudo Random are the same as for Periodic Chirp, the main difference is that the RMS-to-peak ratio is smaller for Pseudo Random.

- ☞ Although Pseudo Random has the word "random" in its name, it is no random signal. Once generated using random numbers for phase, it is repeated over and over again.
- ☞ If a MIMO measurement (refer to [chapter 9](#)) has to be performed, Pseudo Random is not suitable as excitation signal. Although a different random number sequence is generated for every channel, the principal component analysis cannot be performed, as those signals are repeated over and over again. Use Burst Random or White Noise excitation for MIMO measurements.

For linear systems the FRFs are the same for Periodic Chirp and Pseudo Random excitation. If measuring non-linear systems, Periodic Chirp and Pseudo Random both generate response signals with different distortions. Therefore, comparing the results with both excitation signals is a good method to check for non-linearity.

Burst Chirp: Burst Chirp is a fast swept sine signal followed by a zero signal. The data acquisition must be synchronized to the excitation by connecting the SYNC output of the generator to the external trigger input. In order to avoid leakage, the sample time (displayed on the frequency page) has to be chosen such that the response signal has decayed to zero at the end of the time window. Moreover the rectangle window has to be used.

To allow for time to reactivate the trigger, the complete chirp signal (including pause) is 20% longer than the selected time window.

For Burst Chirp the start and the end frequency is specified. To reverse the signal the start frequency can be set higher than the end frequency. The start (delay to SYNC) and duration of the chirp is specified in % of the sample time (displayed on the frequency page). Burst Chirp is generated in the time domain.

Burst Random: Burst Random is a random signal followed by a zero signal. Each individual burst is a different random signal, which excites non-linearities differently for each single measurement. When using a sufficient number of averages (~25), a good linear approximation of the system can be obtained. Burst Random excitation is suitable for MIMO measurements (refer to [chapter 9](#)).

The data acquisition must be synchronized to the excitation by connecting the SYNC output of the generator to the external trigger input. In order to avoid leakage, the sample time (displayed on the frequency page) has to be chosen such, that the response signal has decayed to zero at the end of the time window. Moreover the rectangle window has to be used.

To allow for time to reactivate the trigger, the complete burst signal (including pause) is 20% longer than the selected time window.

The start (delay to SYNC) and duration of the burst is specified in % of the sample time (displayed on the frequency page). Burst Random is generated in the time domain.

White Noise: White Noise is a random signal with a flat spectrum adapted to the measured bandwidth. In difference to Pseudo Random, White Noise is not periodic in the time window but generated with a practically new sequence of random numbers over and over again.

White Noise is neither periodic nor does it allow the response to die out within the time window, therefore leakage occurs. In order to reduce the leakage effect, windowing - typically with the Hanning window - is applied.

As the input signal is always delayed by the measured system, only a part of the response correlates with the excitation. At the beginning of the sample period the response contains information caused by an excitation prior to the sample period. A part of the response excited in the sample period will occur only after the sample period. Both facts generate additional noise which can be reduced by a long sample time and averaging.

It is recommended to use the Hanning window and about 25 averages. The measurement can be sped up using 50% overlap, provided your system supports overlap. When using a sufficient number of averages (~25), a good linear approximation of the system can be obtained. White Noise excitation is suitable for MIMO measurements (refer to [chapter 9](#)).

Square, Triangle and Ramp: Those signals are available for special applications.

User defined: User defined signals are offered to allow for signals which are not implemented in the software.

9 Multiple Reference Analysis (MIMO)

9.1 What Is MIMO?

MIMO means Multiple Input Multiple Output and is a measuring technique where multiple inputs (excitations, references) are used to excite the measurement object at the same time and multiple outputs (responses) of the vibration are measured to obtain a matrix of FRFs between all excitation and response points that describes how the vibrating system reacts to an excitation.

9.2 Why Using Multiple References?

Setting up a multiple reference measurement is not an easy task and requires a good degree of careful preparation, understanding of the measurement and cross checks of the underlying assumptions to be successful. Nevertheless multiple reference tests are an important tool for the following reasons:

- Using multiple inputs allows to put energy into the structure under test at several positions. Especially for large structures it is easier to provide enough energy with multiple inputs compared to a single input to excite global modes.
- If the structure under test exhibits closely coupled modes or repeated roots, the only way to insure that all modes can be identified is to measure with multiple references.

9.3 Example Application Using PSV

This section describes step by step the set up and analysis of a MIMO test using a scanning vibrometer. A typical application for MIMO is to find the modal shapes and mode properties of a large structure, such as a car body in white.

Using a PSV-400-H4 or PSV-300-H system, up to three shakers can be used to excite the structure at different positions. With White Noise or Burst Random, an uncorrelated broadband random excitation generated by the 3 output channels of the PSV's internal generator is used to drive the shakers. Generally we recommend to use a burst random signal combined with a rectangle window function for MIMO tests. Activate the check box Multiple Channels on the page Generator in the dialog Acquisition Settings.

Force cells are used to measure the force fed into the structure. The force cells are connected to the reference channels of the PSV system. The channel properties are set according to the calibration factors of the cells. Activate the check boxes of all reference channels in the Ref column of the page Channels in the dialog Acquisition Settings.

At this time, you might want to check if the forces measured by the cells are uncorrelated. To do this do the following:

1. On the page **General** in the dialog **Acquisition Settings**, select the desired FFT measuring mode (except **FastScan**), magnitude averaging (see [section 6.4](#)) and the number of averages. You have to provide at least $M+2$ averages where M is the number of reference channels.
2. Check the box **Principal Component Analysis (MIMO)** to let the software calculate the MIMO FRFs and coherences during data acquisition.
3. Switch the generator on.
4. Start a continuous measurement in acquisition mode.
5. Have a look at the coherence signals between the various reference channels. The coherence signals should be near to zero over the whole frequency range, as the internal generator provides uncorrelated random signals on its three output channels. Nevertheless, the coupling of the shakers via the structure might lead to a remaining correlation between the driving forces. You might want to optimize the positions of the shakers to minimize this effect.

Although PSV offers just a single response channel, scanning the structure with the laser provides fast and efficient means to perform a sequence of MISO (multiple input, single output) measurements with a roving output. Due to the steady state condition and because of the inputs are measured at every scan point, these MISO measurements can be combined to one MIMO data set.

Now you can set up the geometry of the scan points to cover the parts of interest of the structure to the desired resolution. If you intend to use the measured set of FRFs for modal analysis, you **should** measure the responses in the excitation points (DOFs) to obtain the driving point function. You can enter the indices of the driving points as reference point indices of the reference channels on the page **Channels** in the dialog **Acquisition Settings**. You should also set the direction of the forces and the measured vibration for every active channel on this page according to the geometry of your test set up. Furthermore, we recommend to enter the distance from the scanning head to the object at this time if you plan to export the data for further modal analysis (see below). Now start the scan.

After the scan has finished you can use the PSV presentation mode to analyze the deflection shapes at the resonances of the structure. You might want to export the FRFs and analyze them further with a modal analysis software package like Vibrant's ME'Scope to find the modal parameters and shapes of the structure.

As you have set the reference number indices, the directions of the inputs and outputs and the distance to the object before starting the measurement, no additional information has to be entered at the time of export apart from the choice which data to export.

9.4 The MIMO Model

This section describes the dynamic model that is used by PSV to obtain the FRFs and coherence signals during a MIMO test. The MIMO model is a linear frequency domain model where spectra (Fourier transforms) of multiple inputs are multiplied by elements of an FRF matrix to yield spectra of multiple outputs. The FRF matrix model can be written as:

$$V_N(f) = H_{NM}(f) \cdot R_M(f) \quad \text{Equation 9.1}$$

Here M is the number of inputs (references), N the number of outputs (responses), f the frequency, $V_N(f)$ the vector of outputs, $R_M(f)$ the vector of inputs and $H_{NM}(f)$ the matrix of the FRFs describing the system. If $H_{NM}(f)$ is known, the response of the system to a set of inputs can be predicted.

$H_{NM}(f)$ is a result of the type of MIMO test described above, where N is the number of scan points and M the number of the reference channels. To obtain $H_{NM}(f)$ the software performs the following steps at every scan point:

1. A series of time traces is acquired. The number of traces corresponds to the number of averages specified. The signals at the response channel (vibrometer channel) and the reference channels are acquired simultaneously.
2. The window functions are applied to the time traces and a FFT (fast Fourier transform) is calculated for every channel and for every trace.
3. The auto power signals of all channels and the cross power signals of all combinations of response and reference channels are calculated and averaged over the spectra. For the cross power signals complex averaging is used (see [chapter 7](#)).
4. The principal component analysis is performed on this set of cross power and auto power signals to obtain the FRFs of this scan point:

$$H_{iM}(f) = (V_i(f) \cdot R_M(f)^{T*}) \cdot (R_M(f) \cdot R_M(f)^{T*})^{-1} \quad \text{Equation 9.2}$$

Here i is the index of the current scan point, $V_i(f)$ the response measured by the vibrometer at this point and $R_M(f)$ the vector of the references measured. R^{T*} denotes the transposed complex conjugate of the vector R, $(V_i(f) \cdot R_M(f)^{T*})$ is the 1 by M vector of the cross powers between the responses and references and $(R_M(f) \cdot R_M(f)^{T*})^{-1}$ the inverted M by M matrix of the auto powers and cross powers of the references.

5. The coherence signal is calculated for the current scan point. The coherence is a measure of the input energy visible at the output of the system. The software calculates two types of coherences, the partial coherence and the multiple coherence. The multiple coherence is available for every response channel and is common for all reference channels. Only this signal can potentially reach a value of 1. The partial coherences between the output channel and the individual input channels do in general never reach a value of 1. The Multiple Coherence is a sum of the partial coherences over all reference channels.
6. The coherences between the input channels are calculated. This allows for checking the correlation between the inputs. These coherences should be near to zero.

After the scan has finished, all elements of the FRF matrix $H_{NM}(f)$ are available for analysis.

To be able to solve equation 9.2, the following conditions have to be met:

1. The inputs R_M have to be uncorrelated between channels.
2. The inputs R_M have to be uncorrelated between time traces that contribute to the averages.
3. For all inputs R_M and all outputs $V_N(f)$ the same window function has to be used.
4. The number of averages has to be at least $M+2$.

When you activate the Principal Component Analysis (MIMO) check box on the page General in the dialog Acquisition Settings and you are using the internal generator of the PSV-300-H or PSV-400-H4 system, the software automatically checks the above conditions.

☞ The software is not limited to a single output channel. You could use two of the acquisition channels as output and two as input by checking the corresponding check boxes in the Ref column on the page Channels in the dialog Acquisition Settings. The software then correctly calculates the matrix $H_{22}(f)$ and the two multiple coherences of the output channels. This is the case with the PSV-3D where you measure three outputs simultaneously.

10 Digital Filters

Digital filters are used to condition time signals, predominantly if the evaluation is done in the time domain. As spectra are calculated from the time signals, digital filtering always also changes the spectra. If spectra are all that is to be evaluated, then as a general rule digital filters are not necessary.

10.1 Filters to Limit the Bandwidth

Without using digital filters, the signal contains frequencies between 0Hz and half of the sample frequency (refer to [chapter 4](#)). This is optimal for evaluating spectra. As the FFT divides the signal up into the individual frequencies, the different frequencies do not disturb each other. In the time domain, signal portions of all frequencies are visible simultaneously. This means that the signal to be evaluated can under certain circumstances be covered by other signals. Digital filters are used to suppress these undesired signal portions.

10.1.1 Operating Principle

Zero phase FIR filters are used. These filters only change the amplitudes of the spectral components, the phases remain unchanged.

The following example demonstrates the operating principle of the filters:

FIR filter, length 3

1/4	1/2	1/4
-----	-----	-----

Low frequency signal (approximately sinusoidal):

-4	0	4	7	9	10	9	7	4	0	-4
----	---	---	---	---	----	---	---	---	---	----

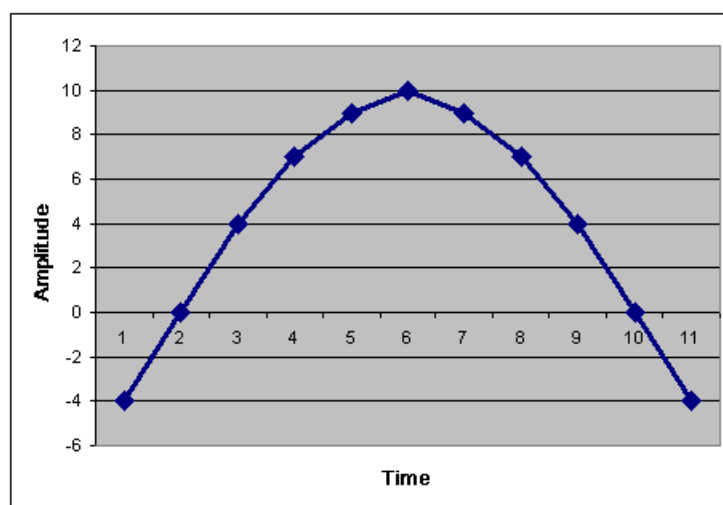


Figure 10.1: Graphic representation of a signal with a low frequency

To calculate the output signal, respectively three neighboring data points of the input signal are multiplied with the filter coefficients and the results are added together.

1st data point:

-4	0	4	7	9	10	9	7	4	0	-4
*	*	*								
1/4	1/2	1/4								
-1	0	1								
+										
	0									

2nd data point:

-4	0	4	7	9	10	9	7	4	0	-4
	*	*	*							
	1/4	1/2	1/4							
	0	2	1.75							
+										
	0	3.75								

3rd data point:

-4	0	4	7	9	10	9	7	4	0	-4
		*	*	*						
		1/4	1/2	1/4						
		1	3.5	2.25						
+										
	0	3.75	6.75							

4th data point:

-4	0	4	7	9	10	9	7	4	0	-4
			*	*	*					
			1/4	1/2	1/4					
			1.75	4.5	2.5					
+										
	0	3.75	6.75	8.75						

...

Result:

	0	3.75	6.75	8.75	9.5	8.75	6.75	3.75	0	
--	---	------	------	------	-----	------	------	------	---	--

The low-frequency input signal is emitted nearly unchanged. The maximum deviation to the input signal is 5%.

High frequency signal

-8	0	8	-8	0	8	-8	0	8	-8	0
----	---	---	----	---	---	----	---	---	----	---

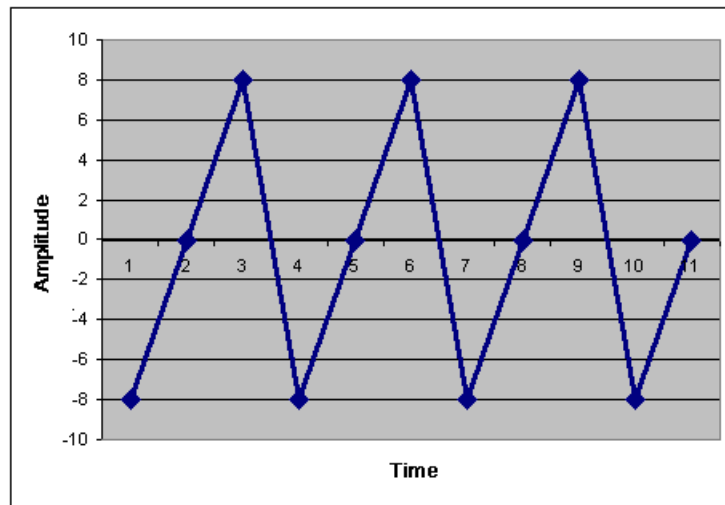


Figure 10.2: Graphic representation of a signal with a high frequency

1st data point:

-8	0	8	-8	0	8	-8	0	8	-8	0
*	*	*								
1/4	1/2	1/4								
-2	0	2								
+										
	0									

2nd data point:

-8	0	8	-8	0	8	-8	0	8	-8	0
	*	*	*							
	1/4	1/2	1/4							
	0	4	-2							
	+									
	0	2								

3rd data point:

-8	0	8	-8	0	8	-8	0	8	-8	0
		*	*	*						
		1/4	1/2	1/4						
		2	-4	0						
		+								
	0	2	-2							

...

Result:

	0	2	-2	0	2	-2	0	2	-2	
--	---	---	----	---	---	----	---	---	----	--

The filter attenuates the high-frequency signal in this example by a factor of 4.

As a conclusion, this example is a low pass filter which allows low frequencies to pass through and attenuates high frequencies. The example is just for explanation purposes, in the software longer filters are used. The longer the filter, the steeper the edges in the filter characteristic function.

The examples also make the following properties of the selected filtering clear:

- The input and output signal are not shifted against each other in time, i.e. the phase is not affected.
- The output signal is shorter than the input signal by the filter length - 1.

For the output signal to have the desired length, the input signal has to be extended by the filter length - 1. This is done automatically by the software. The extended input signal can not be detected by the user, apart from a slightly longer measurement time. When using the trigger, the trigger time is adjusted so that it applies to the filtered signal. When using several channels with different filter settings, the software takes care that the time signals are synchronous.

Filter characteristic

The properties of the digital filters are shown in a filter characteristic diagram. Attenuation is plotted over the frequency. Attenuation of 0dB does not change the input signal. The range with 0dB attenuation is called the pass band. In the stop band, the input signal is attenuated by the stop band attenuation. Between the pass band and the stop band there is the transition band. The cutoff frequency selected by the user is approximately in the middle of the transition band.

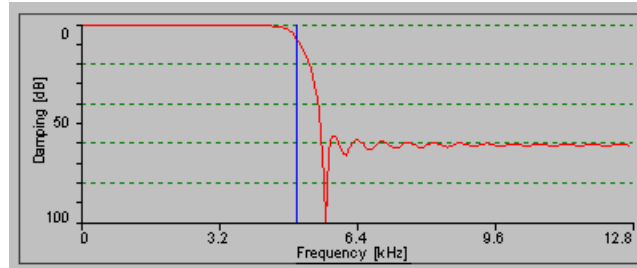


Figure 10.3: Filter characteristic

10.1.2 Filter Parameters

The following filter types are available:

- Low pass
- High pass
- Band-pass
- Band stop

The individual types are described below.

The parameter Quality affects both the length of the filters and also the approximate attenuation in the stop band. For illustration of the filter length the example filter $[\frac{1}{4} | \frac{1}{2} | \frac{1}{4}]$ described above has a filter length of 3.

Quality	Length of filters	Stop band attenuation
very low	21	30 dB
low	41	40 dB
medium	61	50 dB
high	81	60 dB
very high	101	70 dB

Low pass

Low pass filters allow low frequencies to pass and attenuate high frequencies.

High pass

High pass filters allow high frequencies to pass and attenuate low frequencies.

Band-pass

Band-pass filters allow medium frequencies to pass and attenuate high and low frequencies.

Band stop Band stop filters allow low and high frequencies to pass and attenuate medium frequencies.

10.2 Filters for Integration or Differentiation

Filters for integration and differentiation are used to convert the input signals displacement, velocity and acceleration into each other. The same applies to angle, angular velocity and angular acceleration.

☞ When evaluating the spectrum, it is possible to convert the signal during analysis.

With a time data evaluation, the signal already needs to be converted during the measurement. Converting the signal in the frequency and time domain can show different effects.

10.2.1 Integration

Integration is used to convert velocity into displacement or acceleration into velocity. Integrating an acceleration signal twice to generate a displacement signal is not possible.

Operating principle

Integration is explained here using a velocity signal as an example which is converted into a displacement signal through integration. At the start of the time domain, it is arbitrarily defined that the displacement is to be zero. Therefore the first value of the displacement signal is 0. The second value of the displacement signal is determined by multiplying the first velocity value with the sample interval. The sample interval is the reciprocal of the sample frequency (sample interval = $1/\text{sample frequency}$). The third value of the displacement signal is determined by multiplying the second velocity value with the sample interval and then adding the result to the second value of the displacement signal etc. The process is explained with the aid of [figure 10.4](#) and [figure 10.5](#).

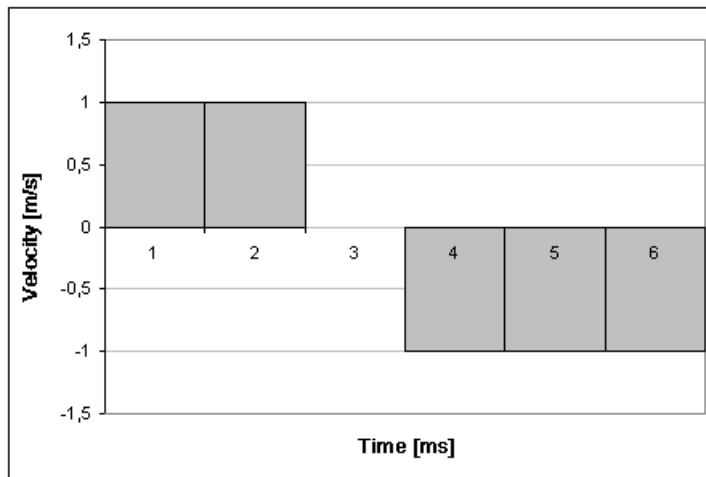


Figure 10.4: Measured velocity signal

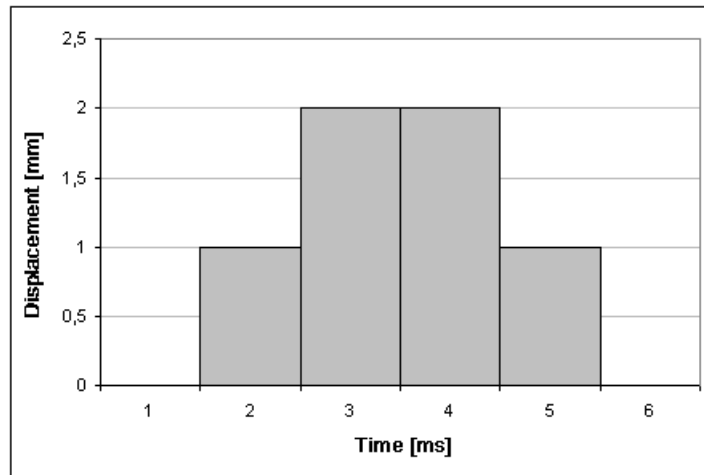


Figure 10.5: Calculated displacement signal

The displacement signal is generated, based on the assumption that the current velocity value remains unchanged until the next velocity value. In the case of constant velocity per section, it can be said that:

$$s_i = s_{i-1} + v \cdot \Delta t \quad \text{Equation 10.1}$$

This assumption only leads to correct results with frequencies which are small in comparison to the sample frequency. In the case of frequencies which are nearly half the sample frequency, the approximation of a constant velocity per section can no longer be made. It would lead to errors of up to 30%. These errors are compensated for by a special FIR filter (refer to [section 10.1](#)). For frequencies up to 40% of the sample frequency (equivalent to 80% of the bandwidth), the resulting error is less than 0.5%.

Properties

Integration in the time domain usually only works satisfactorily if the time signal does not have a DC offset. An offset causes a ramp in the integrated signal. This ramp is often undesired, in particular for low sample frequencies.

The offset in the time domain can be suppressed in many cases by inserting a high pass filter. However, it must be ensured that this does not cut off any portions of the wanted signal.

During integration, a time delay of half a sample interval occurs (sample interval = 1/sample frequency). This leads to a frequency-dependent phase shift.

☞ If the time signal in the time window is periodic and the signal does not have a DC offset, then the integration in the time domain corresponds to the integration in the frequency domain. Otherwise the two results do not match exactly.

An integrated time signal as a general rule shows a DC offset because the first time value is arbitrarily set to zero. Because integrating a time signal with DC offset does not give any meaningful results, the software does not offer double integration.

10.2.2 Differentiation

Differentiation is used to convert displacement into velocity or velocity into acceleration. By differentiating twice, displacement can be converted into acceleration.

Operating principle

Differentiation is explained here using a displacement signal as an example which is converted into a velocity signal through differentiation. From two respective neighboring values of the displacement signal, the average velocity is determined by dividing the difference between the displacement values by the sample interval. The sample interval is the reciprocal of the sample frequency (sample interval = $1/\text{sample frequency}$). The process is explained by the aid of [figure 10.6](#) and [figure 10.7](#).

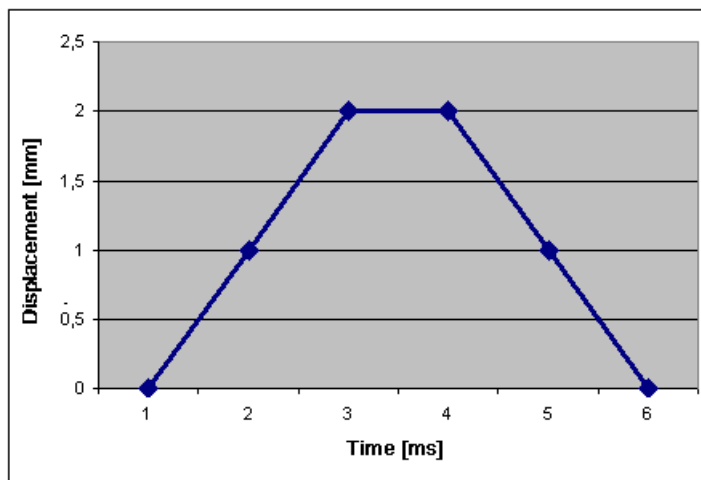


Figure 10.6: Measured displacement signal

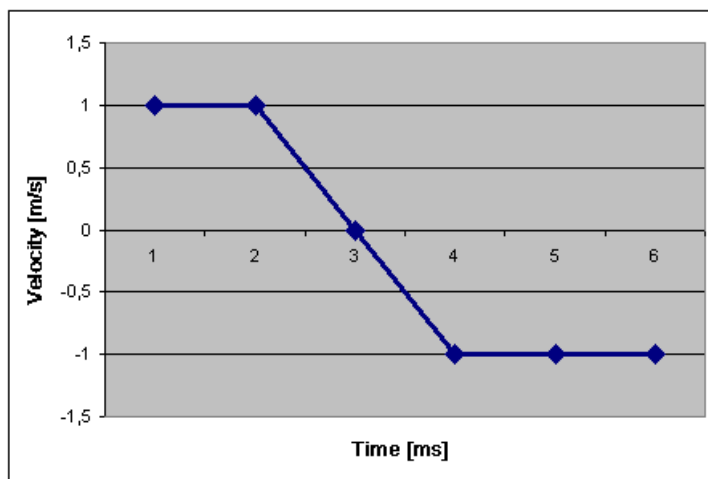


Figure 10.7: Calculated velocity signal

Thus it is presumed that the velocity between two neighboring samples is constant. In this case, it can be said:

$$v = (s_i - s_{i-1}) / \Delta t \quad \text{Equation 10.2}$$

This assumption leads to correct results only with frequencies which are small in comparison to the sample frequency. In the case of frequencies which are nearly half the sample frequency, the approximation of a constant velocity per section can no longer be made. It would lead to errors of up to 30%. These errors are compensated for by a special FIR filter (refer to [section 10.1](#)). For frequencies up to 40% of the sample frequency (equivalent to 80% of the bandwidth), the resulting error is less than 0.5%. When differentiating twice, a maximum error of 1% can occur.

Properties

Differentiation in the time domain usually does not present any difficulties. Only a time delay of half a sample interval occurs during differentiation (sample interval = 1/sample frequency). This leads to a frequency-dependent phase shift.

☞ In the case of double differentiation there is no time delay. If the time signal is periodic in the time window, then the differentiation in the time domain corresponds to the differentiation in the frequency domain. Otherwise the two results do not match exactly.

11 Theoretical Background for PC-based Digital Demodulation (VDD)

In the sensor head a heterodyne interferometer generates a high-frequency carrier signal with the aid of a Bragg cell. For the digital demodulation, the carrier signal is converted into two quadrature signals I and Q in a mixing process. The quadrature signal pair carries the same information as the carrier signal, however it lies in the base band and is thus easy to process electronically.

Ideally the quadrature signals I and Q are sinusoidal, have the same amplitudes and are phase-shifted 90° apart. In [figure 11.1](#) we can see that in an I-Q diagram they depict a rotating pointer whose angle of rotation is equal to the interferometric phase difference $\Delta\phi$. The sense of rotation corresponds to the direction of movement of the object under investigation.

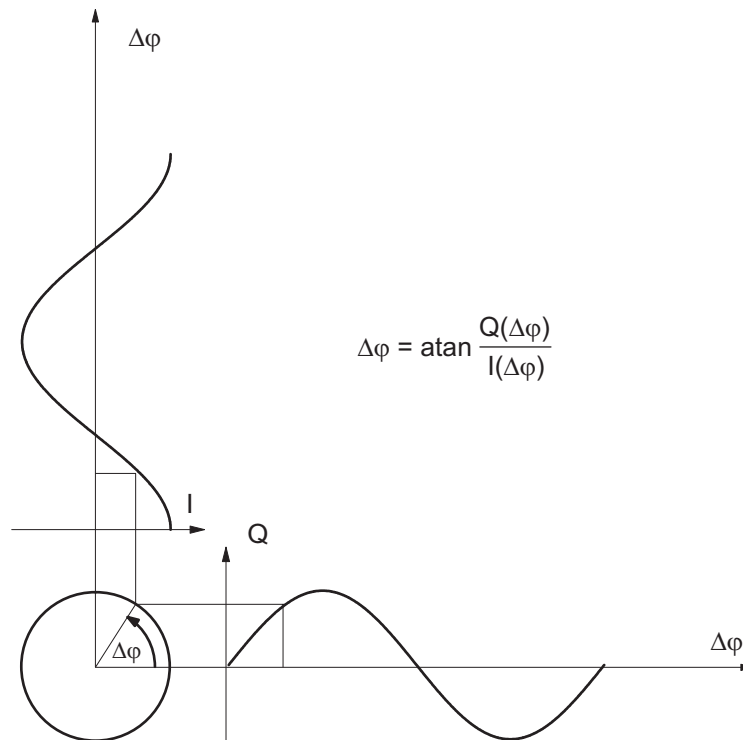


Figure 11.1: Graphical representation of the quadrature signals I and Q

The phase difference $\Delta\phi$ is calculated using the following trigonometric relationships. We have

$$I(\Delta\phi) = A \cos \Delta\phi \quad \text{Equation 11.1}$$

$$Q(\Delta\phi) = A \sin \Delta\phi. \quad \text{Equation 11.2}$$

We divide equation 11.2 by 11.1 and obtain

$$\frac{Q(\Delta\varphi)}{I(\Delta\varphi)} = \frac{\sin\Delta\varphi}{\cos\Delta\varphi} = \tan\Delta\varphi. \quad \text{Equation 11.3}$$

After applying the inverse tangent function we finally get

$$\Delta\varphi = \text{atan} \frac{Q(\Delta\varphi)}{I(\Delta\varphi)}. \quad \text{Equation 11.4}$$

The phase difference $\Delta\varphi$ is proportional to the displacement Δx of the object according to

$$\Delta\varphi = \frac{4\pi}{\lambda} \Delta x. \quad \text{Equation 11.5}$$

In practice it is important to generate the quadrature signals with a high level of stability and spectral purity. Furthermore a suitable A/D converter is required to sample both the quadrature signals with sufficient resolution. After sampling, the remaining errors – which are mainly caused by the analog components in the mixing process – are corrected using special algorithms.

12 Working Efficiently with the PSV

This chapter explains how

- the PSV records measurement data of a vibrating system,
- you can optimize the measurement setup,
- you select suitable settings for your measurement.

12.1 Recording Measurement Data in a Vibrating System

A typical measurement setup with the PSV is shown in [figure 12.1](#).

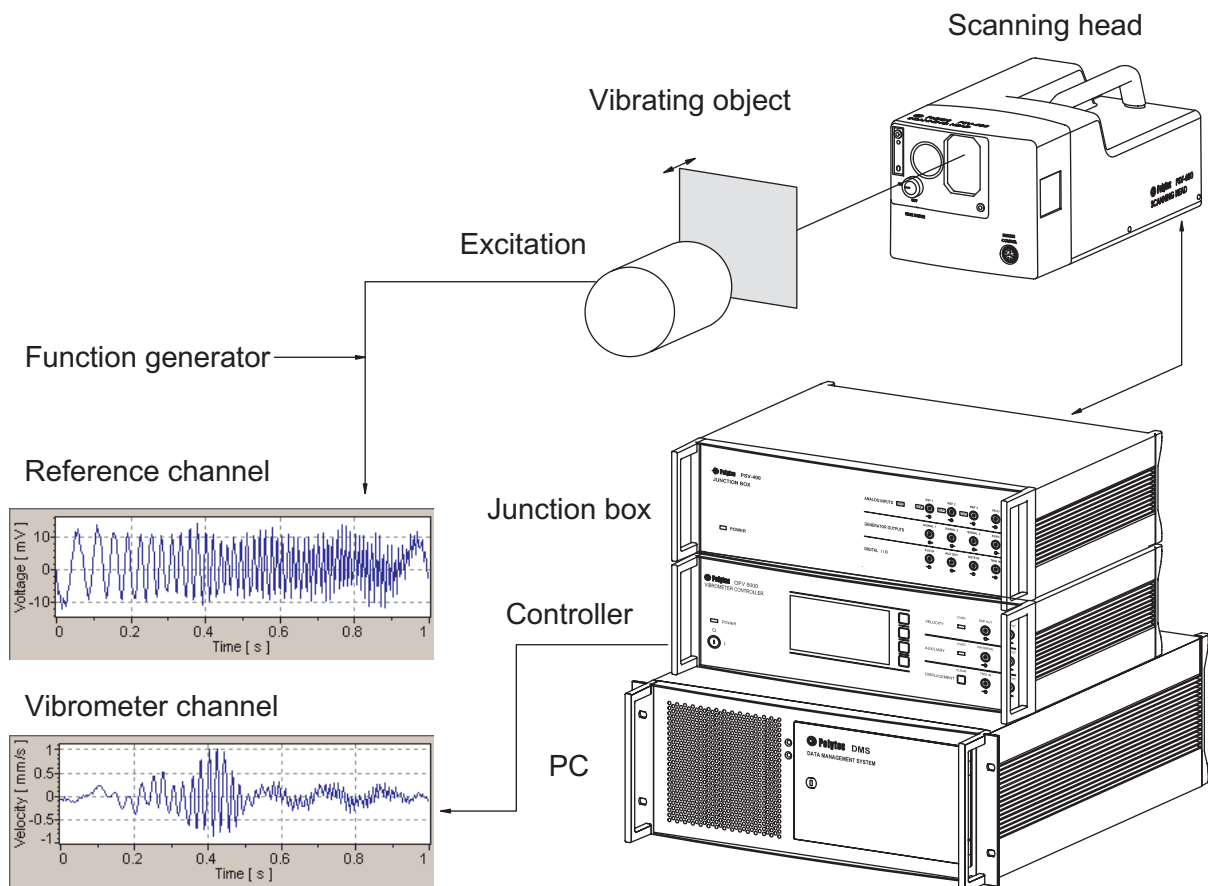


Figure 12.1: Principle of data recording with the PSV

- An excitation (shaker, loudspeaker, hammer etc.) causes the object under investigation to vibrate.
- The measurement beam from the interferometer in the scanning head is positioned to a scan point on the object by means of mirrors and is scattered back.
- The backscattered laser light interferes with the reference beam in the scanning head.
- A photo detector records the interference.
- A decoder in the controller provides a voltage which is proportional to the velocity of the vibration parallel to the measurement beam.
- The voltage is digitized and processed as vibrometer signal.

It is important to select the kind of excitation which allows you to receive the best results for your measurement task. The type of excitation also decides on the measurement parameters. There are the following types of excitation:

- Periodic i.e. with a repeating signal (sinusoidal vibration, periodic chirp, pseudo random, etc.)
- Transient i.e. with a pulse (e.g. rectangular pulse, hammer blow etc.)
- Stochastic (random) i.e. with noise (e.g. noise generator, self-excited)

You can use the excitation as a reference signal or for triggering. You can also use the excitation force measured by a force cell as a reference signal. The reference signal is processed in the same way as the vibrometer signal.

12.2 Optimizing the Measurement Setup

Position the scanning head

The PSV measures velocity components parallel to the laser beam. Position scanning head and object such that

- the laser beam can cover the complete area to be scanned,
- the front panel of the scanning head and the center of the area to be scanned are located opposite each other,
- the longitudinal axis of the scanning head is positioned perpendicular to the area to be scanned.

Optimize the signal level

Test the vibrometer signal level (in the scanning head control) at several scan points, especially at scan points with unfavorable backscattering behavior. The higher the level, the better the signal-to-noise ratio will be. The PSV often also provides useful measurement results at low levels.

If the vibrometer signal level is too low:

1. Check the focus of the laser beam.

Use **Assign Focus** during **Scan** if one focus value does not apply for all scan points.

Check the distance between the front panel of the scanning head and the measured object:

2. Increase the distance if the laser beam hits the lateral points of the scan field at a great angle.
3. Try to place the object in a maximum of the laser intensity. The distances between the maxima are 204 mm (= length of the laser resonator). In between there are minima of the laser intensity where strong temporal fluctuations of the signal intensity can appear. You will find more detailed information on visibility maxima in your hardware manual.

4. If necessary, treat the surface of the area to be scanned. Highly reflective and transparent surfaces usually do not scatter enough laser light back to the scanning head. Evenly matt surfaces are ideal, as for example achieved by means of a sandblaster. A useful tip is to cover a large object with a coat of paint using water-soluble white wall paint or Spanish white. You can also use Ardrex developer spray or reflective film.

12.3 Optimizing Resolution and Measurement Time

The total time needed for a scan is:

$$t_T = t_W \cdot n_p \cdot n_a + t_{add} \quad \text{Equation 12.1}$$

t_T ... Total time for a scan

t_W ... Length of the time window

n_p ... Number of scan points

n_a ... Number of averages

t_{add} ... Additional times

Additional times for a scan are, for example

- time to wait for a trigger,
- time for remeasurement,
- time for waiting until the scanner mirrors are at rest,
- time for updating measurement data in analyzers,
- computing time.

You can optimize the resolution and the time needed for a scan as follows:

1. Select a sufficient number of FFT lines to obtain an appropriate number of frequency values in the set bandwidth.
2. Select the number of scan points according to the size and structure of your scan field.
3. Select the number of averages depending on the noise level.
4. Select **Overlap** if available (refer to [section 12.6](#)).
5. Avoid triggering if possible.
6. Close analyzers which you do not need.

12.4 Noise Reduction with the Tracking Filter

The following applies only for analog velocity decoders: When using the digital velocity decoder VD-07/VD-08, the tracking filter does not have any effect.

Principle

It is important to select a suitable tracking filter setting (dialog Acquisition Settings, page Vibrometer). Due to physical principles (speckle effect), the back scattered laser light can intermit for a short time. The tracking filter bypasses these intermittences and decreases the noise of the input signal according to the PLL Principle (Phase Locked Loop).

If the tracking filter is activated, an oscillator is coupled to the input signal. The oscillator has the effect of a flywheel.

The oscillator follows the input signal

- if velocities and accelerations which are not too high appear on the scan field. The acceleration is proportional to the product of velocity and frequency of the measured vibration.
- if the input signal only intermits for a short period only.

The oscillator does not follow the input signal and the output signal is of no use

- if very high velocities and accelerations appear on the scan field.
- if the input signal intermits for a long period.

Settings

You will find detailed information on selecting suitable tracking filter settings in your hardware manual.

12.5 Optimizing Input Signals with Signal Enhancement and Speckle Tracking

For a scan it is desirable to achieve similar signal quality - i.e. similar noise level – for all scan points. This is, however, made difficult by two problems:

- The speckle nature of the back scattered light
- Differing surface qualities, curved surfaces as well as varying distances for different scan points

The functions Signal Enhancement and Speckle Tracking (dialog Acquisition Settings, page SE) remedy or reduce these problems. This is achieved as follows:

- The noise level is estimated for each individual measurement. In the PSV-3D the worst noise level in the Cartesian coordinate directions x, y, or z is used for the estimate. If averaging is switched on (page General), the individual measurements are averaged and weighted according to their noise level. Measurements with low noise level are weighted higher. The noise level of the averaged spectrum is estimated.

- During a scan the noise level of each scan point is compared with previous scan points. If the noise level is relatively high, additional averages are carried out until the typical noise level is achieved (up to five times the manually set averages). If speckle tracking is activated, the position of the beam is slightly altered (typically 50 µm at 1m stand-off distance) between the individual measurements. At the end of this procedure a scan point with a relatively low noise level receives the status Optimal. Other scan points receive the status Valid.
- At the end of a scan the signal quality of all scan points is compared. Few scan points may change their status.

With Remeasure activated on the page General, all non-optimal scan points are automatically remeasured (up to ten times) after a scan is finished. The remeasurement starts at scan points with a noise level immediate to optimal scan points.

- ☞ You can stop a measurement any time. Stopped measurements are evaluated and saved in exactly the same way as completed measurements.

12.6 Overlap

For PSV-3D, PSV 400-B and -H4 (also PSV 300-U and -H) in the measurement modes FFT and Zoom FFT you can set overlap (dialog Acquisition Settings, page Frequency). The software only takes the parameter into account for measurements with averaging and without trigger. With overlap, the software evaluates the data more effectively. For small bandwidths the time needed for a measurement can thus be reduced.

Principle

The measured time signal is cut into blocks for the FFT. With an overlap of zero, the blocks are right next to each other. If the value is greater than zero, the blocks overlap by this amount. Thus the software records less samples for a measurement.

Example

The following settings have been selected:

- 3 averages
- 100 FFT lines, i.e. 256 samples per block.

The software calculates the FFT for this measurement from the blocks as detailed in [table 12.1](#).

Table 12.1: Calculation of FFT blocks with and without overlap

FFT	Without overlap	With 50% overlap
1st block	1st to 256th sample	1st to 256th sample
2nd block	257th to 512th sample	129th to 384th sample
3rd block	513th to 768th sample	257th to 512th sample

In this example, only 512 Samples are acquired with 50% overlap instead of 768 Samples without overlap.

If you are measuring with overlap, pay attention to the following:

- The noise reducing effect of the averaging is less, particularly if you use the window function Rectangle or Tapered Hanning.
- For bandwidths above approximately 5kHz, the time for a measurement is determined by the computing time. In this case, overlap does not give you a time advantage.
- If Signal Enhancement is active, the probability of getting more scan points with the status Optimal in the same time is higher. The best setting in this case is 50% overlap and twice the number of averages than you would set without overlap.

12.7 Selecting a Reference and Trigger

If possible, you should always measure with a reference signal or trigger. Then you obtain useful phase information.

The phase relation of the vibrometer signal to the excitation is normally not constant at the individual scan points, but depends on the location of the scan points. Therefore the vibrometer signal must not be used as a trigger.

Reference

As a reference signal, you can use the excitation of the vibration, for example.

Trigger

As a trigger, you can use

- the excitation itself,
- a periodic or transient signal that comes from the object and that is in a fixed phase relation with the excitation.

If the message NO TRIGGER appears in the status bar, the trigger signal is either missing or below the threshold (value in Level, dialog Acquisition Settings, page Trigger).

1. Check in an analyzer the signal on which you are triggering.
2. Adapt the level to the signal with which you are triggering.

The phase of the vibrometer signal is calculated by different methods, according to the measured channel, averaging and trigger as detailed in [table 12.2](#).

Table 12.2: Calculation of the vibrometer signal

Averaging	Trigger	Ref available	Phase Vibrometer (V)	Phase Reference (R)
Off	Off	No	0	not available
Off	Off	Yes	Phase V referred to phase R	0
Off	On	No	Phase V referred to start of time window	not available
Off	On	Yes	Phase V referred to start of time window	Phase R referred to start of time window
Magnitude	Off	No	0	not available
Magnitude	Off	Yes	Phase R*V (crosspower spectrum) assigned after averaging	0
Magnitude	On	No	0	not available
Magnitude	On	Yes	Phase R*V (crosspower spectrum) assigned after averaging	0
Complex	Off	No	settings not possible	settings not possible
Complex	Off	Yes	Phase V referred to phase R prior to averaging	0
Complex	On	No	Phase V referred to start of time window	not available
Complex	On	Yes	Phase V referred to start of time window	Phase R referred to start of time window

See [section 6.3](#) and [section 6.4](#) for more details.

12.8 Windowing

The most important parameters to distinguish between the window functions are:

- Maximum amplitude error (ripple)
- Highest side lobe attenuation
- Roll-off rate
- Increase of the resolution bandwidth.

In [table 12.3](#) a summary of the values for these parameters is shown. You will find a detailed description of the parameters and also general information on windowing and the various window functions in [chapter 5](#).

Table 12.3: Parameters for window functions

Window function	Max. amplitude error %	Highest side lobe attenuation dB	Roll-off rate dB/dec	Increase of the resolution bandwidth
Rectangle	-36	13	20	1.00
Hanning	-15	32	60	1.50
Hamming	-18	43	20	1.36
Blackman	-12	58	60	1.73
Blackman Harris	-9	92	20	2.00
Bartlett (triangle)	-19	27	40	1.33
Flat Top	-0.2	68	20	3.77
Tapered Hanning (Parameter=10)	-35	12	20	1.08

The **exponential window** has not been included in [table 12.3](#) as it is not suitable for windowing continuous signals. It is used for impulse excitation if the answer signal does not decay in the time window. An important parameter for the exponential window is the time constant which can be set in the software.

13 Working with 3D Geometries

The PSV Software supports three dimensional geometries for the scan points. This means that in addition to the two dimensional position on the live video image a three dimensional coordinate in space is assigned to every scan point. After the vibration data has been measured at the scan points it can be displayed on the three dimensional geometry of the object under investigation.

13.1 Why Using a 3D Geometry ?

For PSV-3D systems using 3D geometry is mandatory. The software needs to know the positions of the three scanning heads in the coordinate system of the object under investigation to be able to calculate the directions of the laser beams at the current scan point. Only with this information the vibration components measured in the directions of the laser beams can be transformed to 3D data with the vibration components in x, y and z direction.

When using a 3D geometry the software knows the position of the scanning head in the coordinate system of the object under investigation. For every individual scan point the distance to the scanning head can be calculated. This can be used by the PSV software to automatically focus the laser beam at the current scan point. Before, select `Scan > Assign Focus Fast` in acquisition mode.

Knowing the position of the scanning head and the 3D coordinates of the scan points allows for calculating the scanning mirror angles such that the laser beam precisely hits the scan points in space. Especially for objects that are not flat but quite deep this method is superior to the method used to position the laser beam by using the positions on the live video image.

If you want to measure at nodes for example of a finite element model of your object you can import the coordinates of the nodes into the PSV software and measure exactly at these points. Comparing measurement results with calculation results of the finite element software becomes much easier when you use the same positions of the scan points and the finite element nodes.

Using a common 3D coordinate system for a sequence of measurements from different positions and/or directions of the scanning head allows for combining the measurement results into a single result on the combined geometry of the object under investigation.

13.2 Alignment Methods

The laser beam is deflected in the scanning head by mirrors controlled by the PSV software. There are two different procedures that allow for positioning the laser beam on the scan points: standard alignment (2D) and 3D alignment. Both methods allow to calculate the mirror angles for a given scan point either from its position on the live video image (standard alignment) or from its 3D coordinates (3D alignment).

Standard alignment (2D)

Using this function the software can position the laser beam if you click on a position on the live video image. Moreover the laser beam is able to follow the mouse cursor. Scan points can be defined on the live video image.

During standard alignment a sequence of positions of the laser beam are marked on the live video image. After setting an alignment point at the laser spot on the live video image, the beam is moved to a different position and the next alignment point is marked by clicking the position. During this process the software stores a pair of values for every alignment point: the position on the live video image (expressed in video coordinates) and the angles of the horizontal and vertical scanning mirrors. From these pairs of values the software calculates the parameters of a regression function that maps video coordinates to angles and vice versa.

For three dimensional objects this works well only if the camera is positioned very near to the scanning mirrors compared to the distance to the object. Otherwise the different view positions of scanning head and camera lead to parallax effects that can not be compensated by the standard alignment function. For objects on which not all points have the same distance to the scanning head, deviations between the point marked on the live video image and the laser position can occur. Those deviations become larger with greater depth of the object and shorter distance between object and scanning head. They can be avoided using the coaxial unit for the scanning head.

For the PSV-3D the camera used is not parallel to all of the scanning heads. Therefore, the deviations between the point marked on the live video image and the laser position are generally larger. In order to increase the accuracy for the PSV-3D, higher order polynomials are used to calculate the regression if at least 10 points are used for the standard alignment.

3D alignment

Using the 3D alignment, the software can calculate the mirror angles that let the laser beam hit a point in space from the 3D coordinate of this point. This is used by the software during a scan to position the laser beam to points with known 3D coordinates.

During 3D alignment at least 3 alignment points with known distances to the scanning head or at least 4 alignment points for which the distances are not known are used. For every alignment point the 3D coordinates have to be known. The coordinate system of these points defines the coordinate system of the object under investigation (object system). Together with the mirror angles that let the laser beam hit the alignment points, the software calculates the position and direction of the scanning head in the object system.

With the optional geometry scan unit the software is able to measure the distance to the point on the object, where the laser points to. This is used during 3D alignment to measure coordinates of alignment points of a defined type (e.g. origin of the object system, point on a coordinate axis, etc.).

13.3 Coordinate Systems

Video coordinates

The video coordinate system is used to place the scan points on the live video image in acquisition mode and on the video snapshot in presentation mode. The video coordinate system is used internally only.

Scanning head system

The scanning head system is a Cartesian coordinate system. The origin of this system is the position where the laser beam with $0^\circ/0^\circ$ mirror angles would hit the front panel of the scanning head. The orientation of the coordinate system depends on the setting of the direction of the vibrometer channel on the page Channels in the dialog Acquisition Settings. The different orientations of the coordinate axes are summarized in the following table.

Vib Direction	x-axis	y-axis	z-axis
+X	to head	up	to the left
+Y	to the right	to head	down
+Z	to the right	up	to head
-X	from head	up	to the right
-Y	to the right	from head	up
-Z	to the left	up	from head

The scanning head system is used if no 3D coordinates are known for the scan points. In this case, you can enter the distance to the object under investigation and the software will calculate the x-, y- and z-coordinates of the scan points from the mirror angles assuming that all scan points are placed on a plane perpendicular to the $0^\circ/0^\circ$ direction of the laser beam placed at the given distance to the scanning head. As the origin of the scanning head system is placed on the front panel of the scanning head, all scan points will have either the same x-, y- or z-coordinate, depending on the orientation of the coordinate system and therefore on the direction of the vibrometer channel. These 3D coordinates will be displayed for the selected scan point in acquisition and presentation mode below the element 2D Point.

During export of the scan point coordinates to Universal File format, ME'Scope structure files or ASCII files these coordinates will be used as 3D coordinates of the scan points. The coordinate that is equal to the distance to the scanning head will be set to 0 during export for all scan points.

Object system

The object system is the Cartesian coordinate system used for the alignment points during 3D alignment. The coordinates of the alignment points define the object system.

During import of 3D geometries from Universal File format, ASCII format or ME'Scope structure files the 3D coordinates will be interpreted to be in the object system of the current 3D alignment.

During a geometry scan the measured 3D coordinates will be measured in the object system. Therefore you need a valid 3D alignment to perform a geometry scan.

In APS point mode, you can define single scan points. The 3D coordinates are either calculated by measuring the distance to the scan point with the geometry scan unit or via triangulation of the three laser beams that hit this point (only possible with PSV-3D). In both cases, the object system will be used for the 3D coordinates of the defined scan points. Therefore you have to have a valid 3D alignment to define points with 3D coordinates.


The 3D coordinates of the scan points will be displayed in the object system below the element 3D Point when you select a scan point in acquisition or presentation mode. These coordinates will be used during export to Universal files, ME'Scope structure files and ASCII files without further modification.

13.4 Sources of a 3D Geometry

This section describes the different methods for getting a 3D geometry into the PSV software. Every method has its special advantages and drawbacks: Knowing them, you can choose the method suited best for your needs.

Import of geometry data (PSV-3D, optional for PSV-1D)

If you have a valid 3D alignment, i.e. if you have defined an object coordinate system, you can import geometries from Universal File format or ASCII format geometry files or from Vibrant's ME'Scope structure files.

To do so, change to APS point mode and click  or in the menu bar select Scan > Import Geometry. You can then navigate to the geometry file and import the geometry from this file.

The geometry definition files contain the 3D coordinates of a set of measurement points and only for Universal File format or ME'Scope structure files lists of connections between these points defining surface and line elements of the geometry. The latter are used to display a surface of the object under investigation in presentation mode.

The 3D coordinates of the measurement points have to correspond to the coordinate system that you have used during 3D alignment. With the 3D alignment the software can position the laser beam to hit the measurement points. The precision of the laser position depends mainly on the quality of the 3D alignment.

The measurement points will be placed on the live video image using the standard alignment. These positions are used for displaying purposes only, they do not influence in any way the measurement results. The precision of the measurement point positions on the live video image (i.e. how well the points are positioned on the corresponding laser spots) depends on the quality of the standard alignment. For deep objects or if the camera position is not at the position of the scanning head (e.g. if you use the center camera of PSV-3D), the quality of the standard alignment might not be satisfactory. The software offers to correct the positions on the live video image automatically. To do so, the software successively

positions the laser to every measurement point using the 3D alignment and tries to find the laser spot on the live video image. If it succeeds, this position will be used instead of the position calculated via the standard alignment.

- ☞ Please pay attention that during this action no movement should occur in the live video image. Keep the personal from walking through the field of view.

The imported geometry might be larger than the range that can be covered by the live video image and/or the range of the scanning mirrors. Nevertheless, the software will import the complete geometry, placing the points on the live video image if possible. The other points will be visible around the zoomed out live video image. The software marks the points that can not be reached by the scanning mirrors with a color code. These not reachable points will be excluded from the scan automatically. If points are hidden by any parts of the object under investigation, i.e. if there is no direct line of sight from all scanning heads, these points are marked with a different color code. These hidden points will be scanned after a warning message.

Once you have imported a geometry you are free to edit it. You can delete or add points (adding with PSV-1D needs the geometry scan unit) or modify the connections between the points. You can move points around on the live video image and you can modify the 3D coordinates of the selected points to optimize how well the three lasers hit the same point on the object under investigation.

Triangulation (PSV-3D only)

If you define your geometry in APS standard or point mode, the PSV software at first has information on the positions of the measurement points on the live video image only. For PSV-3D it is mandatory to have 3D coordinates for the measurement points. Therefore, the software calculates the 3D coordinates from the positions on the live video image by a method called triangulation. Using the standard alignment for all three scanning heads (Top, Left and Right), the scanning mirror angles for every scanning head and for every scan point are calculated. From the scanning head positions and orientations known from the 3D alignment the software calculates the laser beam paths for every scanning head. The 3D coordinate of the point in space where these three beams come closest to each other is used as the 3D coordinate of the respective scan point.

As triangulation uses the standard alignment which is not precise in all cases, the 3D coordinates calculated by this method might not be precise enough. To check this, you can select the scan points one after the other and check if the laser spots hit the object under investigation at the same spot and at the desired position. Here the 3D alignment is used to calculate the mirror angles from the 3D coordinate. These mirror angles are slightly different to the mirror angles calculated from the standard alignment before.

Modify 3D coordinates

To modify the 3D coordinates, you can select a single or more points and right click one of the selected points. Select **Modify 3D Coordinates** in the pop-up menu. The dialog **Modify 3D Coordinates** appears where you can modify the 3D coordinates of the selected points by changing the coordinates directly or by a slider. Moving the slider will move the 3D

coordinate along the direction of the laser beam path. For the PSV-3D the 3D coordinate is moved on a line that corresponds to the average direction of the three laser beam paths. You can use the slider to minimize the distance between the laser spots on the object under investigation. All 3D coordinates of all selected points will be changed by the same amount when closing the dialog.

Geometry scan unit (PSV-3D, optional for PSV-1D)

With the geometry scan unit PSV-A-420 that is attached to the PSV-I-400 scanning head, the software can measure distances from the scanning head to any point on the object under investigation that can be reached by the laser. The geometry scan unit contains a laser with a microcontroller that allows for measuring distances. The laser is fed through the optics of the scanning head such that the same scanning mirrors are used to position the geometry laser and the vibrometer laser. The scanning head optics is aligned so that the laser spot of the geometry laser hits the same point on the object under investigation as the vibrometer laser if the same scanning mirror angles are used. You can switch between the two lasers any time in the software by selecting **Scan > Geometry Laser**. In addition you can move a filter into the optical path of the geometry laser either by using the mechanical switch at the geometry scan unit or by activating the check box **Filter** in the scanning head control of the software (depends on the version of the geometry scan unit). This can be used to dim the geometry laser if too much light is scattered back from the object which can overexpose the sensor of the geometry scan unit.

While the geometry laser is active, you will see a signal strength indicator for the geometry laser in the scanning head control. It is recommended that you check the signal strength for the part of the object under investigation that you want to scan. The indicator should show some signal strength and the LED **OVER** should be green. If applicable, modify either the filter setting or change the distance of the scanning head to the object. As you might have to change the distance of the scanning head to the object, do this before you perform the standard or 3D alignment because otherwise you would have to repeat the alignments. You can move the laser without a valid standard alignment either with the icons **Position** in the scanning head control or by pressing and holding down the middle mouse button (mouse wheel) while moving the mouse.

- ☞ If the measured distance is about 2 m, the sensitivity of the geometry laser has a minimum. In order to increase the signal strength, the scanning head can be moved to a shorter or larger distance (this applies only if the filter switch is on the geometry scan unit).

Using the PSV software you can perform a geometry scan. First you have to perform a standard alignment, so that the software can position the laser using the position of the measurement points on the live video image. Then you have to perform a 3D alignment to define the object coordinate system that is used for the 3D coordinates measured during the geometry scan. After this you can define your geometry as usual using one of the APS modes. Having left APS mode, you can start the geometry scan. To do so, select **Scan > Geometry Scan**. Note that this command is only enabled if you have a valid standard and 3D alignment. The coordinate system of the 3D coordinates is defined by the coordinate system of the alignment points of the 3D alignment (object system).

If you want to check the distance measurement at some points before performing the geometry scan, position the laser to the desired points and select **Scan > Geometry Point**. The software will measure the distance to the point on the object under investigation and display a dialog where you can read the measured distance and the 3D coordinate of the point in the object system.


During the geometry scan the software positions the geometry laser beam to the scan points calculating the scanning mirror angles from the positions of the scan points on the live video image using the standard alignment. Therefore the precision on how well the positions on the live video image match to the positions on the object depends on the quality of the standard alignment.

☞ If you are using a PSV-3D system, we recommend using the video camera of the scanning head **Top** to display the live video image because this allows for higher quality in the standard alignment of the scanning head **Top** which is used for the geometry scan.

If the distance measurement fails for some scan points, the software will display a message and offers to repeat the measurement or interpolate the coordinates for the remaining points. During interpolation the distance of a scan point is interpolated from the distances of the neighbors of this point. Then the coordinate is calculated from this distance and the position of the point on the video image. If the point has no neighbors with coordinates, the coordinate remains unmeasured and the dialog is displayed again. If you click **Cancel** in this dialog, the remaining points will receive the point status **No 3D coordinate**.

The distance measurement might fail for the following reasons:

- There is too much light scattered back to the geometry scan unit. You can switch the filter of the geometry scan unit on before repeating the geometry scan for the remaining points (only if the filter switch is on the geometry scan unit).
- There is too little light scattered back to the geometry scan unit. You can switch the filter of the geometry scan unit off before repeating the geometry scan for the remaining points (only if the filter switch is on the geometry scan unit).
- If the amount of light is still too much or little after applying/removing the filter, you can try to paint the object, attach a piece of paper to it or find some other means to change the reflective properties of the object's surface. You might even want to remove the specific points from your geometry by converting the geometry to a point geometry and edit it in APS point mode. Repeat the geometry scan after you have done this.
- Not the whole laser spot hits a single part of the object. You might want to convert your geometry to a point geometry by entering the APS point mode and move this point. Then repeat the geometry scan.

If you want to add or remove some points of the geometry you can switch to APS point mode. This will keep the 3D coordinates of the points. You can add new points by positioning the laser on the object and click  or select **Scan > Point at Laser Position**. The software will automatically perform a distance measurement to this point, calculate the corresponding 3D coordinate and add the point to the geometry.

13.5 3D Geometry and Vibration Data

After you have acquired a 3D geometry you can perform a vibration measurement at the defined scan points. The software will position the vibrometer laser to the measurement points by calculating the scanning mirror angles from 3D coordinates via the 3D alignment. After that you can switch to presentation mode and analyze the vibration data.

There is a 2D and a 3D mode to display the data. In 2D mode the geometry and measurement results are displayed on the live video snapshot taken at the beginning of the vibration measurement scan. You can use color-coding and different view styles to analyze the vibration data.

In 3D mode the vibration data is displayed on a projection of the 3D geometry. Using a virtual camera, the 3D geometry is projected to the view area of the presentation window. By using the mouse and other controls of the software you can move the camera around the object and get an impression of the geometry from different view angles.

In addition to color-coding on the surface of the object, the vibration data is displayed as displacement from the zero position of the geometry. The 3D geometry is deformed by the vibration data. You can display the zero position together with the deformed geometry of the vibration data. To do so, mark the box *Zero Position* on the page *3D View* in the dialog *Display Properties*.

During phase animation, the instantaneous values are calculated and the deformed geometry and color-coding are updated accordingly. During time domain animation the deformed geometry and color-coding are updated according to the measurement values corresponding to the current time after trigger.

Please note that the displacements used to visualize the vibration data do not represent real displacements of the object under investigation. You can not only visualize displacement data like this but also data in other physical units like velocity, acceleration or FRFs. Even when you display displacement data the displacement in general is exaggerated by a large number of orders of magnitude. The PSV usually measures displacements in the μm range that are displayed as displacements in the m range.

Visualize one dimensional data (PSV)

The geometry is deformed along the coordinate axis, that corresponds to the direction of the acquisition channel. If e.g. the direction of the channel is $-X$ then positive values are displayed as displacement along the negative x-coordinate axis and negative values along the positive x-coordinate axis. If the vibrometer channel direction is $+Z$ then positive values are displayed along the positive z-coordinate axis and negative values along the negative z-coordinate axis.

☞ An angle correction is available to correct the measured vibration data for the incident angle of the laser beam at the surface of the object under investigation, because the vibrometer measures the vibration component along the laser direction. On the page **Scanning Head** of the dialog **Preferences** you can activate a correction for the horizontal and vertical scanning directions. The algorithm used for this assumes a planar object surface perpendicular to the $0^\circ/0^\circ$ direction of the laser beam and a pure out-of-plane vibration. If this is a bad approximation to your object under investigation, you might consider to switch off the angle correction altogether, displaying the uncorrected vibration component along the laser beam.

Visualize three dimensional data (PSV-3D)

For three dimensional data measured with the PSV-3D the color codes correspond to the value

$$A = \sqrt{A_x^2 + A_y^2 + A_z^2} \quad \text{Equation 13.1}$$

where A_x , A_y and A_z correspond to the components measured in the corresponding coordinate axis directions of the object coordinate system. You can switch off any combination of components by deselecting the corresponding icons in the toolbar of the presentation window. In this case the corresponding colors are calculated from the remaining active components only. All displayed values are positive, as can be seen from equation 13.1. The phase display is not available for 3D data. To display single direction components with sign information or phase, switch to 2D display mode and select the corresponding vibrometer channel **Vib X**, **Vib Y** or **Vib Z**.

The position of the deformed geometry is calculated from the individual A_x , A_y and A_z components by adding the scaled components with sign information to the 3D coordinate of the measurement point. Note that only those components are taken into account, that are active. It is not possible to display values that are positive or negative only as a deformed geometry, e.g. it is not possible to display magnitude, magnitude dB or magnitude dB(A) as deformed geometry, because those would be displayed as a misleading displacement towards the positive or negative coordinate axes only.

13.6 Best Practices

This section provides a list of practices that you might find useful when you are working with 3D geometries.

To set up the measurement, proceed as follows:

1. Check the signal strength of the geometry laser.

2. Carry out a standard alignment.
3. Carry out a 3D alignment.
4. Define the geometry.
5. Make a geometry scan.
6. Select Scan > Assign Focus Fast.
7. Make a vibration measurement scan.

If you have a 3D geometry, using Assign Focus Fast is the best choice. As the software knows the distances to all scan points it is enough to determine the laser focus values at a very small number of scan points and interpolate the rest of the focus values using the known distances. In case of a 3D geometry the Assign Focus Fast method is as good as Assign Focus Best for nearly all practical situations but it is much quicker.

If the Assign Focus Fast command fails, you might want to exclude scan points from being used for the algorithm. To do this, disable these scan points by selecting them, right click one of them and select Disable in the pop-up menu. The Assign Focus Fast algorithm will then only use the points that are not disabled. Nevertheless, if it succeeds, the determined laser focus values are assigned to all points, including the disabled points. Enable the points you have disabled again, so that they are included in the measurement.

If you are using the geometry scan unit with the PSV-3D, use the video camera in the scanning head Top to perform the standard alignment. This minimizes effects caused by the different view directions and positions of the camera and the scanning head that can not be fully compensated by the standard alignment for deep objects. A better standard alignment quality allows for a better match of the measurement point positions on the live video image to the measurement points on the object.

If your object under investigation is too large to be scanned from a single position of the scanning heads or if you want to scan different sides of the object, you can stitch a sequence of measurements. In this case, you have to use a common coordinate system for all measurements, i.e. the reference points of all 3D alignments have to be defined in the very same coordinate system. To achieve this, proceed as follows:

1. Open the page Devices in the Preferences dialog and select the junction box PSV-E-400-3D (1D).
2. Open the page Channels in the Acquisition Settings dialog and deactivate the channels Vibrometer Left and Vibrometer Right.
3. Position the scanning head Top with the geometry scan unit attached at a position where you have a direct line of sight to a large part of your object under investigation.

4. Perform a standard and 3D alignment for the scanning head Top only. For the 3D alignment either use reference points that you have determined before (see below) or define your object coordinate system with one of the coordinate definition modes of the 3D alignment. Use the geometry scan unit during alignment.
5. Mark reference points on the object under investigation. Mark the points with a unique number. For every position of the scanning heads during the later measurements, you have to have at least four reference points that can be directly reached by all scanning heads from their position. It is recommended to use at least five points for optimal quality of the alignment.
6. Switch to APS Point Mode and define scan points at the reference points. Position the laser of the Top scanning head on the reference points and use the command Point at Laser Position at every reference point. Make sure that the indices of the points match those of the marks on the object under investigation. You can change the indices of the points, if necessary. The position of the points on the live video image will not be used, only the 3D coordinates of the points are important.
7. Repeat steps 3 to 6 until you have reference points on all parts of the object under investigation that you want to scan. When you change the 3D alignment in step 4, the software will ask you if the scan point positions on the live video image should be recalculated. Click No, because in this case the positions are irrelevant. You can even move the points out of the way on the live video, if you want to. This will not change the 3D coordinates of the points. When you perform the 3D alignment select the Points with free Coordinates mode and use reference points with known coordinates.
8. Leave APS Point Mode and export the scan points to a Universal file. To do so, in acquisition mode, select File > Export > Universal File.
9. Open the page Devices in the Preferences dialog and select the junction box PSV-E-400-3D.
10. Position all scanning heads for the first measurement.
11. Open the dialog 3D Alignment. Deactivate the box Auto. Import the coordinates of the reference points by clicking Import. A list of the reference points with their unique numbers and their coordinates will be imported into the table at the lower part of the 3D Alignment dialog.
12. Define a new alignment point at one of the reference points marked on the object under investigation. Position all laser beams on this point and assign the laser positions to the alignment point.
13. Select the reference point in the coordinate list and assign its 3D coordinate to the alignment point. Use the unique number of the reference point to identify the entry in the list.
14. Repeat steps 12 and 13 for more reference points.
15. Calculate the 3D alignment, define the geometry and perform the scan.

16. Repeat steps 10 to 15 for all measurements in the measurement sequence
17. Change to presentation mode and stitch all measurements by creating a new combined file.

If the coordinate definition modes available during 3D alignment with the geometry scan unit do not suite your needs, you might want to do the following: Start with either *Origin*, *Axis*, *Plane* or *Three Points on Axes* by defining the alignment points. Calculate the 3D alignment. This will enter the actual coordinates of the points into the coordinate list. Switch to mode *Points with free Coordinates* and modify the coordinates of the alignment points. To do so, click the alignment point in the live video image with the right mouse button, choose *Points with free Coordinates* and modify the coordinates in the appearing dialog. By adding an offset to the coordinates you can e.g. move the origin of the coordinate system by a specific amount.

If you are working with PSV-3D and the geometry scan unit and if you want to use the 3D alignment in *Points with free Coordinates* mode but the position and/or coordinates are not known precisely, use the points with the free coordinates for the scanning head *Top* only. Add additional alignment points for the scanning heads *Left* and *Right*. You have to position the laser *Top* to these alignment points, too, because the scanning head *Top* is used to measure the distances to this points, but during alignment the points are only used for the scanning heads *Left* and *Right*. As a result you obtain a 3D alignment where the object system is defined with the same accuracy as the free coordinates you used for the scanning head *Top*. However, the accuracy of the relative positions of three scanning heads to each other is higher, which is important for the accuracy of the laser positioning and the coordinate transformation during the vibration measurement scan.

If you are using the geometry scan unit to acquire the 3D geometry of the object, first define the "raw" geometry using APS mode standard. Make sure that all points are well apart from edges of the object under investigation and are fully hit by the geometry laser. Then perform the geometry scan. After this switch to APS point mode and add the more delicate points near to the edges of the object under investigation. Note that you can use the hand set PSV-Z-051 or the PSV-A-PDA to position the laser beam directly at the object and teach in the points. This minimizes the effects of a non perfect standard alignment on the acquisition of the 3D geometry.

If you repeatedly work with the geometry scan unit to measure distances, e.g. while defining alignment points in 3D alignment or teaching in geometry points in APS point mode, switch to the geometry laser by selecting *Scan > Geometry Laser*. This saves time because now the software does not switch from and to the vibrometer laser any more for every distance measurement.

Use the hand set or the PDA and select *Scan > Auto Align* in the menu bar to define alignment points with the hand set or the PDA in 3D alignment. With the hand set or the PDA you are free to walk around, go to the object under investigation, position the laser beam there and define new alignment points. With the hand set or the PDA you can position the laser beam more precisely than by looking at the live video image.

Appendix A: Introduction to Complex Numbers

A.1 Representation of Complex Numbers

Complex numbers are underlined in this section.

A complex number consists of a real part and an imaginary part

$$\underline{Z} = \text{Re} + i \cdot \text{Im} \quad \text{Equation A.1}$$

Re... Real part of \underline{Z}

Im... Imaginary part of \underline{Z}

i... Imaginary unit, $i = \sqrt{-1}$.

As an alternative to the presentation with a real and an imaginary part, complex numbers can also be described with an absolute value and a phase (Euler's formula)

$$\underline{Z} = |\underline{Z}|e^{i\varphi} \quad \text{Equation A.2}$$

$|\underline{Z}|$... Absolute value of \underline{Z}

φ ... Phase of \underline{Z}

e... Euler's number ($e = 2.718\dots$).

The exponential function with an imaginary argument can be replaced by trigonometric functions

$$e^{i\varphi} = \cos\varphi + i \sin\varphi. \quad \text{Equation A.3}$$

This results in

$$\underline{Z} = |\underline{Z}| \cos\varphi + i|\underline{Z}| \sin\varphi \quad \text{Equation A.4}$$

$$\text{Re} = |\underline{Z}| \cos\varphi \quad \text{Equation A.5}$$

$$\text{Im} = |\underline{Z}| \sin\varphi. \quad \text{Equation A.6}$$

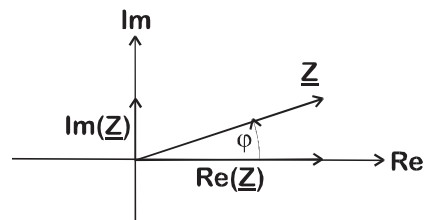


Figure A.1: Graphical presentation of a complex number

The complex plane is a coordinate system with real numbers in x direction and imaginary numbers in y direction.

A complex number \underline{Z} is depicted as a pointer in the complex plane. Real part $\text{Re}(\underline{Z})$, imaginary part $\text{Im}(\underline{Z})$, absolute value $|\underline{Z}|$ and phase φ can be taken from the pointer diagram. The phase is 0 if the pointer is in x direction and 90° if the pointer is in y direction.

A.2 Addition and Subtraction of Complex Numbers

Addition of complex numbers:

$$\underline{Z}_1 + \underline{Z}_2 = \text{Re}_1 + \text{Re}_2 + i(\text{Im}_1 + \text{Im}_2) \quad \text{Equation A.7}$$

Subtraction of complex numbers:

$$\underline{Z}_1 - \underline{Z}_2 = \text{Re}_1 - \text{Re}_2 + i(\text{Im}_1 - \text{Im}_2) \quad \text{Equation A.8}$$

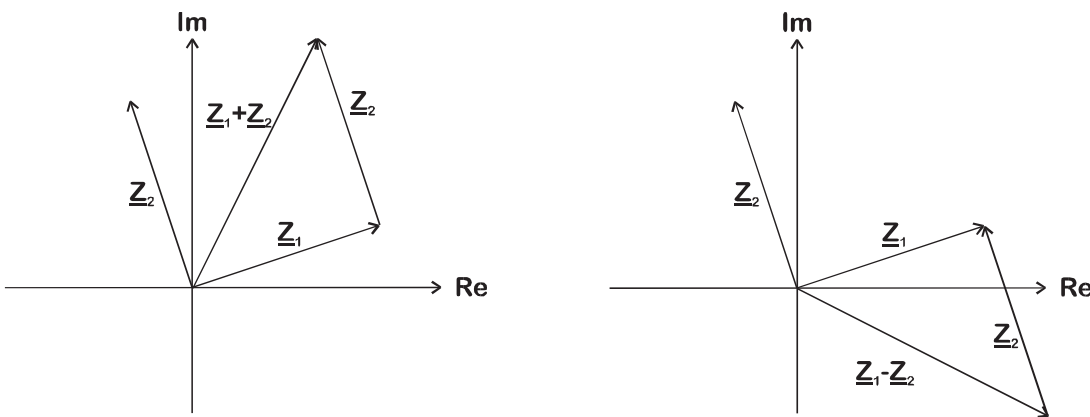


Figure A.2: Addition and subtraction of complex numbers

In the left pointer diagram two complex numbers are added by vector addition.

In the right pointer diagram the pointer \underline{Z}_2 is subtracted from the pointer \underline{Z}_1 by vector subtraction.

A.3 Multiplication and Division of Complex Numbers

Multiplication of complex numbers:

$$\underline{Z}_1 \cdot \underline{Z}_2 = \text{Re}_1 \cdot \text{Re}_2 - \text{Im}_1 \cdot \text{Im}_2 + i(\text{Re}_1 \cdot \text{Im}_2 + \text{Re}_2 \cdot \text{Im}_1) \quad \text{Equation A.9}$$

The multiplication is much simpler if the complex numbers are described with an absolute value and phase

$$\underline{Z}_1 \cdot \underline{Z}_2 = |\underline{Z}_1| \cdot |\underline{Z}_2| \cdot e^{i(\varphi_1 + \varphi_2)} \quad \text{Equation A.10}$$

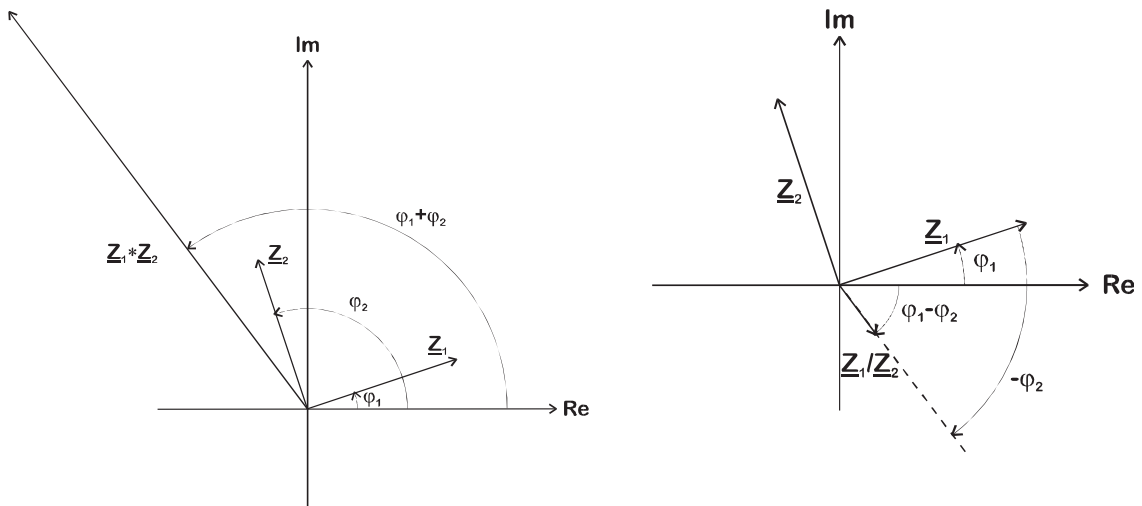


Figure A.3: Multiplication and division of complex numbers

Two complex numbers are multiplied by multiplying the absolute values and adding the phases

$$\frac{\underline{Z}_1}{\underline{Z}_2} = \frac{|\underline{Z}_1|}{|\underline{Z}_2|} \cdot e^{i(\varphi_1 - \varphi_2)} \quad \text{Equation A.11}$$

Two complex numbers are divided by dividing the absolute values and subtracting the phases.

A.4 The Complex Conjugate

The complex conjugate of Z is

$$\underline{Z}^* = \text{Re} - i \cdot \text{Im} . \quad \text{Equation A.12}$$

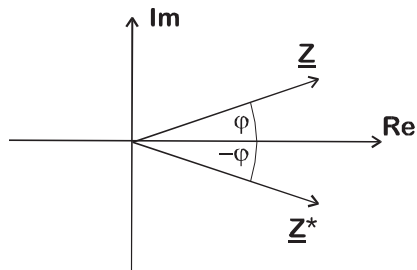


Figure A.4: Conjugated complex of \underline{Z}

In the pointer diagram the complex conjugate develops through reflection on the real axis.

According to the multiplication rules the following applies

$$\underline{Z}^* \underline{Z} = |\underline{Z}| \cdot |\underline{Z}| \cdot e^{i(\varphi - \varphi)} = |\underline{Z}|^2 . \quad \text{Equation A.13}$$

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