

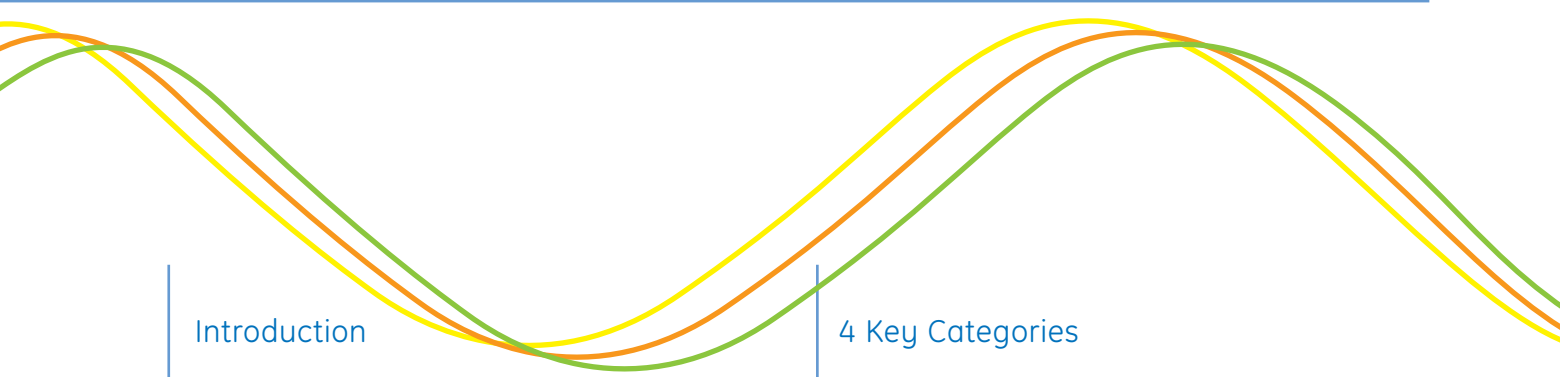


# Understanding Discrepancies in Vibration Amplitude Readings Between Different Instruments

*Part 1 of 2*

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## Introduction

Numerous times each year, customers contact our sales and service professionals with a common question: "I took a vibration reading with a portable diagnostic instrument, but it does not agree with what I am seeing on my Bently Nevada® monitor. Why?"

Often, customers assume that there must be a problem with the monitor, the diagnostic instrument, or both. However, this is rarely the case. Instead, there are some very straightforward reasons why there are often discrepancies in vibration readings – particularly peak-to-peak amplitude readings – between two instruments. In this first installment of a 2-part article, we explore those differences, building on and updating material that was originally published in the article "Monitoring versus diagnostics – why do peak readings differ?" in the March 1994 issue of ORBIT.

## 4 Key Categories

The reasons why readings differ between instruments can generally be divided into four basic categories as summarized below and as shown in Figure 1:

1. Dissimilar inputs
2. Dissimilar signal processing prior to amplitude detection
3. Dissimilar amplitude detection algorithms
4. Calibration/indication problems

This article discusses each of these in order, illustrating how each can contribute to discrepancies in the indicated amplitude of a vibration signal. In part 1, we focus on the first two categories: inputs and signal processing. In part 2, which will appear in the next issue of ORBIT, we examine the last two categories: amplitude detection and indication/calibration.



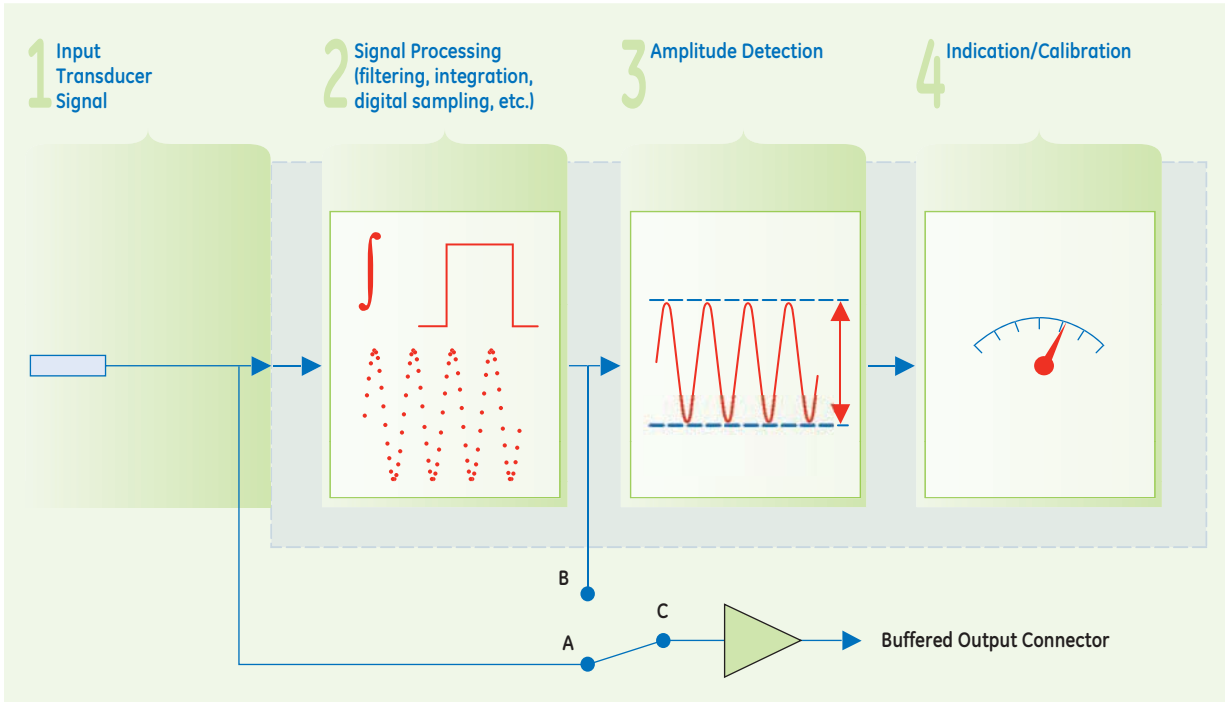


Figure 1 – By considering the four stages of a signal as it travels through an instrument, it is easier to understand and isolate discrepancies in readings between two devices.

Table 1 –

**WAVEFORM AMPLITUDE CONVENTIONS USED IN BENTLY NEVADA INSTRUMENTATION**

Peak-to-Peak (pp)	The difference between the maximum positive-going and negative-going peaks in a periodic waveform during one complete cycle.
Zero-to-Peak (pk)	The pp value of a vibration signal divided by two (pp/2). Also referred to as “true peak.”
Root Mean Square (RMS)	A measurement of the effective energy content in a signal. Mathematically, the RMS value of a waveform $f(t)$ is defined as $A_{RMS} = \sqrt{\frac{1}{T} \int_{t_0}^{t_0+T} f(t)^2 dt}$ <p>where <math>T</math> is the period (one complete cycle) of the waveform*.</p>

\* To accommodate all expected periods in generalized waveform inputs to an instrument,  $T$  is typically chosen for computation purposes to coincide with the lowest frequency measurable by the instrument. Thus, an instrument with a bandwidth down to 0.5 Hz would use  $T = 2$  seconds.

## Definitions

Because Bently Nevada monitoring instruments typically use proximity probe inputs that measure the total back-and-forth displacement of the shaft, peak-to-peak (pp) is the most commonly encountered engineering unit when dealing with our instruments. However, this article will also discuss zero-to-peak (pk) and RMS readings since they are frequently encountered as well, particularly in portable instruments.

One of the primary reasons that confusion arises when making amplitude measurements is that many instruments (such as portable analyzers/data collectors) provide the user with multiple configuration options for peak detection. For example, one well-known manufacturer of portable data collectors provides the following four options on several of their models:

- Digital Overall
- Analog Overall
- True Peak
- Average Peak

Compounding this, different manufacturers use not only different detection algorithms/circuits, but different nomenclature such as “calculated peak,” “peak,” “derived peak,” “pseudo peak,” “overall peak,” and others. Clearly,

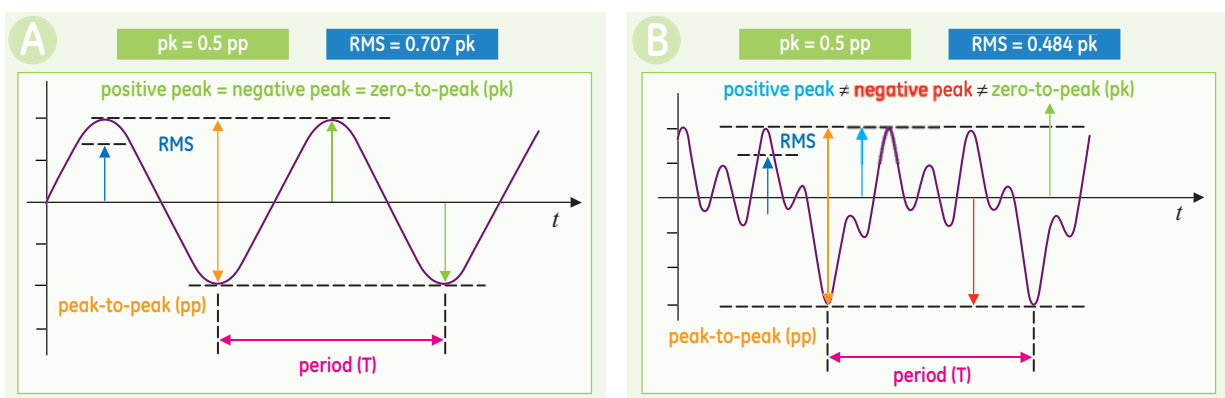
a user needs to understand the specific algorithms and conventions used when interpreting amplitude readings, and in part 2 of this article we will investigate in much greater detail some of the algorithms used for these readings, comparing them to the algorithms used in Bently Nevada instruments.

Throughout this article, we will use the Bently Nevada instrumentation conventions for describing signal amplitudes, which are summarized in Table 1.

Figure 2 illustrates these concepts for two types of waveforms: a pure sine wave and a generalized complex signal consisting of several sinusoids of varying frequencies and phases.

While the pp values are obvious for both waveforms, the pk and RMS values merit further discussion.

As noted in Table 1, the convention used for pk in Bently Nevada instrumentation is simply the pp value divided by two. For the symmetrical sinusoid of Figure 2A, the maximum amplitude traversed by the wave is the same in both the positive and negative directions and is equal to  $pp/2$ . However, the waveform of Figure 2B is non-symmetrical. That is, its positive peak is not equal to its negative peak. Some manufacturers define pk as the larger of either the maximum negative-going or positive-



**Figure 2** – Peak-to-peak (pp), zero-to-peak (pk), and RMS are the most commonly used conventions for expressing the amplitude of vibration waveforms. The conversion between RMS and pk is only equal to  $1/\sqrt{2}$  (0.707) for a pure sine wave (A). For a more complex signal (B), the equation of Table 1 must be used to compute the RMS value. The conversion between pp and pk can differ between manufacturers and becomes apparent if the waveform is asymmetrical (B). When using the Bently Nevada instrumentation convention of  $pk = pp/2$ , the pk value may not be equal to either the positive or negative peaks on an asymmetrical waveform.

going peak. Thus, they would return a peak value for the waveform of Figure 2B as 3.0, while a Bently Nevada instrument would return a peak value of  $5/2 = 2.5$ , resulting in a 20% discrepancy between the readings. There is no right or wrong convention, and when a symmetrical waveform is used, both conventions will provide identical results. Discrepancies occur when the waveform is non-symmetrical, as is the case for most real vibration waveforms except when filtered to a single frequency (such as 1X). Thus, real vibration waveforms may yield different peak readings simply by virtue of the different conventions used between two instrument manufacturers.

RMS measurements are a frequent area of misunderstanding and lead to many of the discrepancies seen in the field. Many practitioners have been taught that pk is simply  $RMS \times \sqrt{2}$  (1.414) and many instruments provide a setting that returns such a value, sometimes referred to as “derived peak” or “calculated peak.” While this relationship between peak and RMS is true for a pure sine wave, it is not true for a generalized waveform with complex frequency content. Thus, use of the word “peak” for such measurements is an unfortunate and potentially misleading choice of nomenclature; it is really more

appropriate to think of these measurements as “scaled RMS.” As examples of the complete independence of true peak values from RMS values, consider the following:

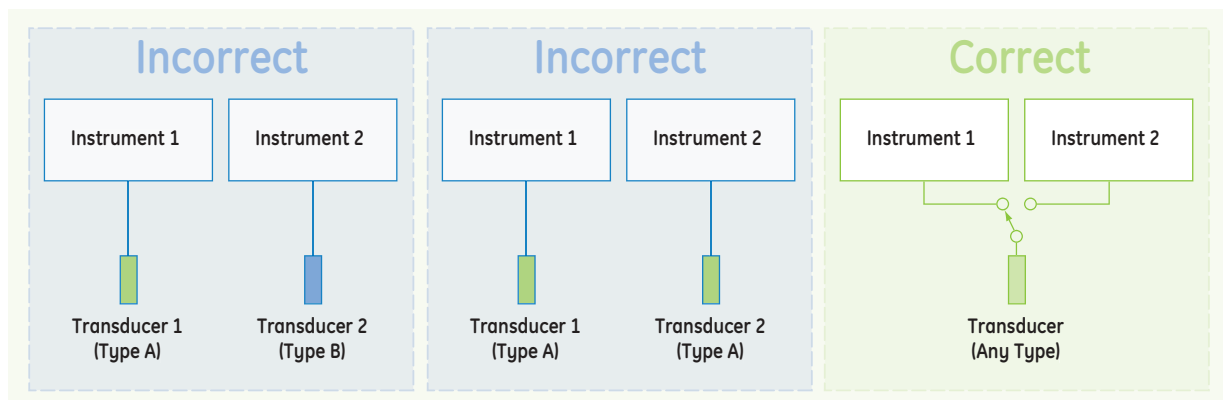
- The RMS value of a square wave is  $1.0 \times pk$ .
- The RMS value of a triangle wave is  $0.577 \times pk$ .
- The RMS value of the waveform in Figure 2B is  $0.484 \times pk$ .

Because RMS and true peak are completely unrelated, instruments capable of accurately providing both readings must employ one type of circuit or algorithm for computing true peak and a distinctly different one for computing RMS.

We will have much more to say about the topic of “derived peak” in part 2 of this article when we discuss and compare the various algorithms and circuits used for computing RMS and pk waveform amplitudes.

### 1. Dissimilar Inputs

The first and most fundamental part of comparing readings from different instruments is to ensure that you are using an *identical* input to each instrument. Until this has



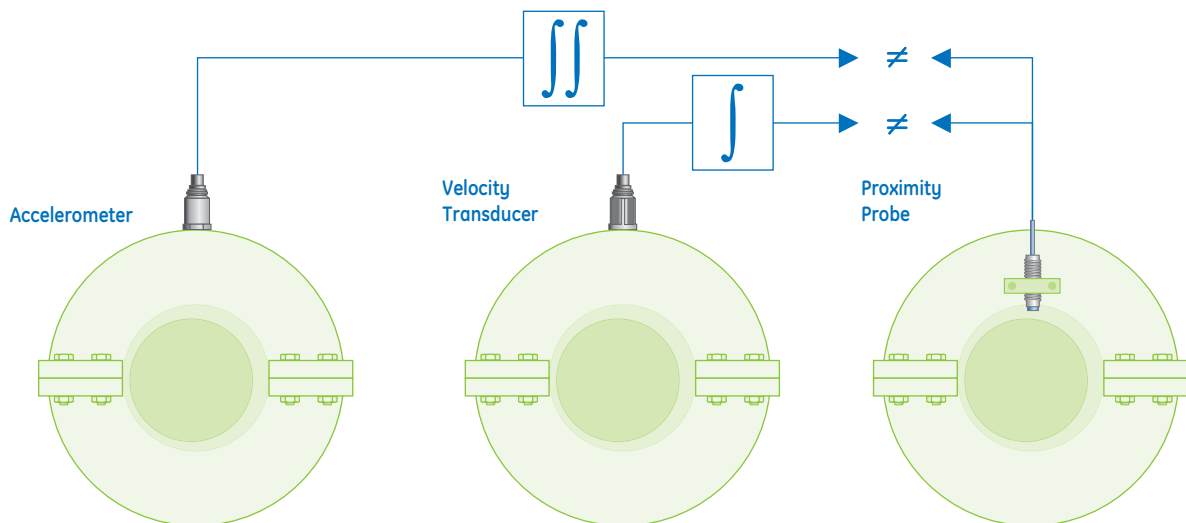
**Figure 3** – When comparing readings from separate instruments, it is imperative to introduce the same transducer signal to each (right). Merely using the same type of transducer (center) is not sufficient and will almost always lead to discrepancies in readings. Nor is it appropriate to use a different type of transducer (left) and simply convert the signal to consistent units, such as via integration.

been thoroughly and completely confirmed, it is pointless to proceed. This simple step accounts for quite a few of the discrepancies in readings seen in the field, because users are not comparing “apples with apples.” Ensure that you are actually using the same transducer to provide an input signal to each instrument. This means not just the same *type* of transducer, but the same physical transducer (Figure 3).

Often (but not always) a Bently Nevada monitoring system will be using eddy-current proximity probes as the input type. These probes measure the relative displacement between the probe’s mounting location (often the bearing housing or machine case) and the shaft. Customers will sometimes suggest that all they have to do is integrate their velocity signal or double-integrate their acceleration signal to obtain displacement units, allowing them to compare casing measurements with those from proximity probes. However, although this may result in readings with the same engineering units, the physical measurements being made are quite different (Figure 4). Seismic transducers measure casing

motion relative to free space. Proximity probes measure shaft motion relative to the probe’s mounting. Integration of a casing-mounted velocity transducer signal merely gives the amount of casing displacement which is *not* the same as the relative displacement between the shaft and the probe’s mounting (usually the machine case). Thus, while comparing these two signals to one another does have a valuable role in machinery diagnostics, it is not appropriate to compare them when attempting to see if two instruments agree with one another.

Even when the input to the permanent monitoring system is a permanently affixed seismic transducer, it is still inappropriate to use a different transducer (such as a hand-held or magnetically mounted seismic sensor) for the other instrument. Enough variation will often exist between the mounting methods of permanent, stud-mounted transducers and a hand-held or magnetically affixed sensor to give noticeable discrepancies in readings. In addition to differences contributed by transducer mounting methods, the differences in mounting *locations* can be pronounced. The vibration response can differ



**Figure 4** – Although integration yields measurements with the same engineering units, they are not equivalent measurements because a proximity probe measures the relative motion between the probe’s mounting surface and the shaft, while seismic transducers measure the absolute motion of the transducer relative to free space.

greatly even when two transducers are located very close together. They must be oriented along the same axis, use the same method of attachment, and be attached to a part of the machine with identical stiffness. For example, many machines are supplied with a specially machined “flat spot” designed to accommodate portable or permanent seismic transducers. However, this location is rarely large enough to accommodate two transducers side-by-side. Placing one transducer on the flat spot and another transducer just a few inches away can result in different readings for the reasons mentioned above.

Next, we examine the second of four categories for dissimilar amplitude readings: differences in signal processing that occur prior to amplitude detection.

## 2. Dissimilar Signal Processing Prior to Amplitude Detection

Once you have confirmed that the identical signal is being introduced to each instrument, you must then understand any intermediate signal processing that is to be performed prior to the amplitude detection portion of the instrument. Typically, this intermediate signal processing may include filtering, integration, and digital sampling.

### Anti-Alias Filtering

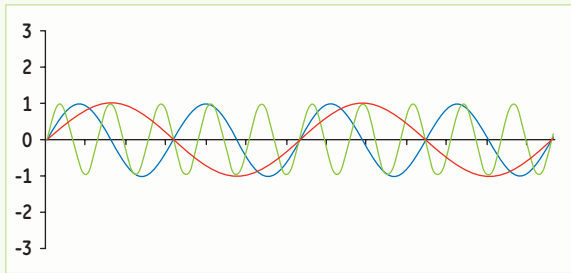
Many instruments today perform their functions digitally. Digital signal processing theory shows that in order for a signal to be represented faithfully, the sampling frequency (known as the Nyquist rate) must be at least twice the highest frequency present in the analog signal. For example, if the sampling frequency of an instrument is 1000 Hz, the highest frequency that can be present in the analog signal is 500 Hz. Digital instruments use an analog filter (known as an anti-alias filter) to limit the high-frequency content of the incoming signal before sampling. In the example above, the anti-alias filter is designed to reject frequencies above 500 Hz, passing only those frequencies below 500 Hz to the digital sampling circuitry of the instrument.

It is important to understand where the anti-alias cutoff frequency occurs for your instrument. Depending on the amplitude and phase of the frequencies that are filtered out, this can have a dramatic effect on the amplitude measured by the instrument. Such filtering will provide a very different waveform amplitude when compared with an analog instrument that does not employ any filtering of the signal, or a digital instrument with extremely broad bandwidth such as a digital oscilloscope capable of capturing frequencies in the Gigahertz range. Also, it is important to note that many instruments provide different anti-alias settings for each measurement. For example, a single incoming signal providing an overall (unfiltered) measurement, bandpass filtered measurement, synchronous waveform measurement, and asynchronous waveform measurement may have different anti-alias settings for each of these four measurements.

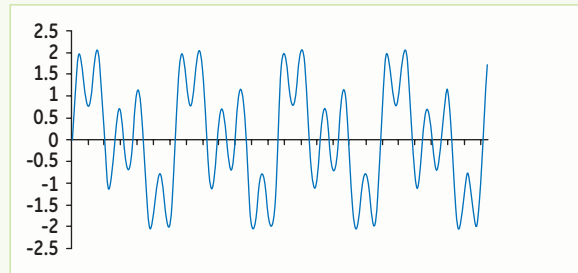
Many times, the sampling frequency in devices such as portable data collectors/analyzers is a user-configurable option, designed to balance storage limits with diagnostic needs (higher sampling rates give larger bandwidth and higher resolution at lower frequencies, but require more memory). By configuring the sampling frequency, the anti-alias filter will automatically be set by the instrument. Thus, you can generally determine the anti-alias cutoff frequency by looking at the instrument’s sample rate setting. For example, if the instrument is set to sample at 10 kHz, the anti-alias filter will be slightly less than one-half of this frequency (approximately 5 kHz or 300,000 cpm). For instruments where the sample rate is not configurable or is unknown, consult the manufacturer.

Some data collectors will also allow the user to independently choose the spectral resolution which affects the amount of memory required for each vibration sample collected. When “reconstructing” the analog waveform from the digital samples, both the spectral resolution and the maximum frequency will affect the results.

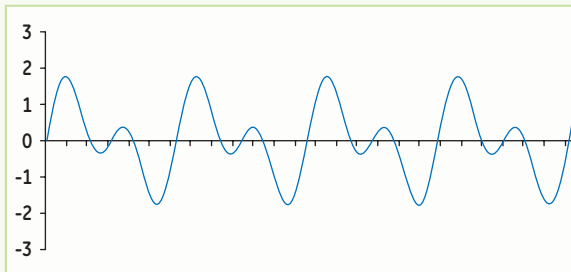
To understand the effect that an anti-alias filter will have when the incoming signal has significant frequency content above the anti-alias cutoff frequency, consider the example of Figure 5. In Figure 5A, sinusoids of 100, 200, and 500 Hz are shown. Each sinusoid is identical in phase



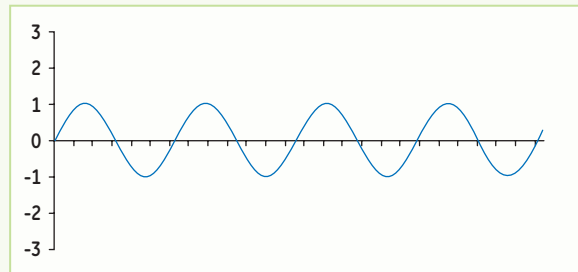
**Figure 5A** – Three sinusoids of equal amplitude and phase, differing only in frequency (100 Hz, 200 Hz, 500 Hz). Amplitude of all sinusoids is 1.0 pk.



**Figure 5B** – The resulting waveform when all three sinusoids of Figure 2A are summed. Amplitude is approximately 2.1 pk.



**Figure 5C** – The resulting waveform when an anti-alias filter with a 400 Hz cutoff frequency is introduced. Only the 100 Hz and 200 Hz components remain and waveform amplitude is reduced to approximately 1.7 pk.



**Figure 5D** – The resulting waveform when an anti-alias filter with a 150 Hz cutoff frequency is introduced. Only the 100 Hz sinusoid is left, with an amplitude of 1.0 pk.

and amplitude (pk = 1.0), differing only in frequency. Figure 5B shows the composite waveform that results when these sinusoids are summed, creating a waveform with pk amplitude of approximately 2.1. Assuming an “ideal” anti-alias filter that completely removes all frequencies above the cutoff and has no effect on the amplitude or phase of frequencies below the cutoff, the introduction of such a filter with a 400 Hz cutoff frequency removes the 500 Hz component and reduces the original waveform pk amplitude by approximately 19% (Figure 5C). Lowering the anti-alias filter to 150 Hz removes both the 200 and 500 Hz components, reducing the original pk amplitude by approximately 52% and leaving only a 100 Hz sinusoid (Figure 5D).

In practice, actual vibration waveforms are generally far more complex, composed of many frequencies, each with a different phase relationship and amplitude.

Depending on the amplitude and phase relationships between frequency components, filtering out a specific frequency may decrease or *increase* the overall amplitude. This is often the cause of much confusion, because users will sometimes reason, “How can I ‘remove’ signal content and end up with a larger amplitude?” The answer lies in the way that waves add. Depending on the phase, waves can add constructively or destructively. Consider the extreme case of two sinusoids, equal in amplitude and frequency, but exactly 180° out of phase. When summed, they will completely cancel each other and the resulting waveform will have zero amplitude. Thus, even though we start with two waveforms with non-zero amplitudes, by adding them together, we end up with a smaller amplitude (in this case, zero). In fact, this is precisely the theory behind the new “noise canceling” headphones now on the market.

As another example, consider a square wave  $f(t)$  with a pk amplitude of  $A$ . This waveform can be mathematically represented as an infinite sum of sinusoids (Equation 1) consisting of a fundamental frequency (1X) and its odd harmonics (3X, 5X, 7X, etc.) that decrease proportionately in amplitude.

$$f(t) = \frac{4A}{\pi} \sum_{n=1,3,5,\dots}^{\infty} \frac{1}{n} \sin nt = \frac{4A}{\pi} \left\{ \sin t + \frac{1}{3} \sin 3t + \frac{1}{5} \sin 5t + \frac{1}{7} \sin 7t + \dots \right\} \quad [1]$$

If we low-pass filter the square wave such that all frequencies higher than 1X are removed and only the fundamental frequency remains, we are left with Equation 2

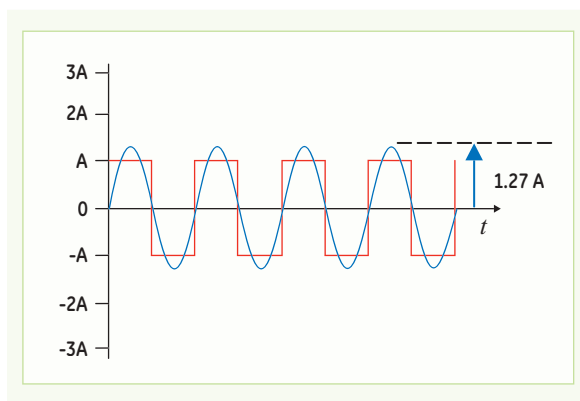
$$f(t)_{\text{filtered}} = \frac{4A}{\pi} \sin t \quad [2]$$

which has peak amplitude  $4A/\pi$ . Notice that this is 27% *larger* than the pk amplitude of the original square wave (see Figure 6). Admittedly, a square wave is not routinely encountered in vibration analysis, but it does serve to show how a waveform and its constituent frequency components sum in a complex fashion, adding in some places and canceling in others. We occasionally see real vibration waveforms in practice that have slightly higher filtered amplitudes than unfiltered amplitudes. Thus, while the removal of frequency components will

generally reduce the overall amplitude, this may not always be the case and does not necessarily indicate a problem with the instrumentation.

### Filtering Effects on Phase

Above, we discussed the effects of entirely removing (or attenuating the amplitude of) certain frequency components from a composite waveform. However, it must be remembered that filters modify not only the *amplitude* of frequency components, but the *phase* as well. These phase relationships of frequency components are just as important as their amplitudes in determining the shape and amplitude of the resulting waveform. Because vibration analysts typically only look at the amplitude of spectrum components, it is easy to forget the importance of phase and its effect on the resulting complex waveform in the time domain. Indeed, two waveforms with identical amplitude spectra can have very different waveform shapes and pp amplitudes. To appreciate this, consider the waveform of Figure 5B. Its amplitude spectrum is shown in Figure 7A, corresponding to the three discrete sinusoids of Figure 5A. Assume now that we have a filter which leaves the amplitude of each frequency component unchanged and merely shifts the phase. The resulting waveform, as shown in Figure 7B, has a very different shape, symmetry, and pp amplitude from the waveform of Figure 5B and serves as a reminder that identical amplitude spectra do not always correspond to identical timebase waveforms.

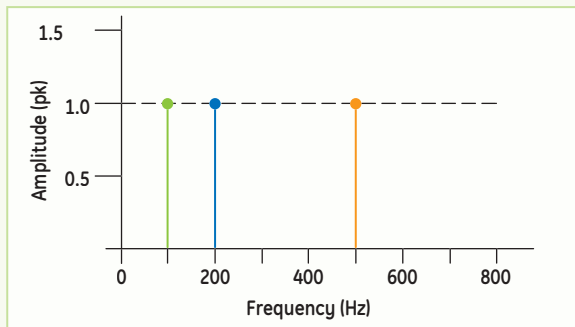


**Figure 6 – The pp amplitude of a square wave is actually less than the pp amplitude of its fundamental (1X) component. In the same way, although not common in vibration diagnostics, a filtered amplitude can occasionally be larger than the unfiltered overall amplitude.**

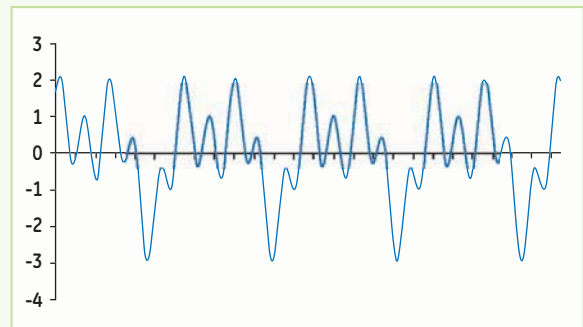
Thus, be aware, as shown in this example, that filtering can affect the amplitude and shape of the original waveform not only by altering the amplitude of frequency components, but the phase as well.

Although certain digital filters with any desired amplitude roll-off can be designed with constant phase shift, this is

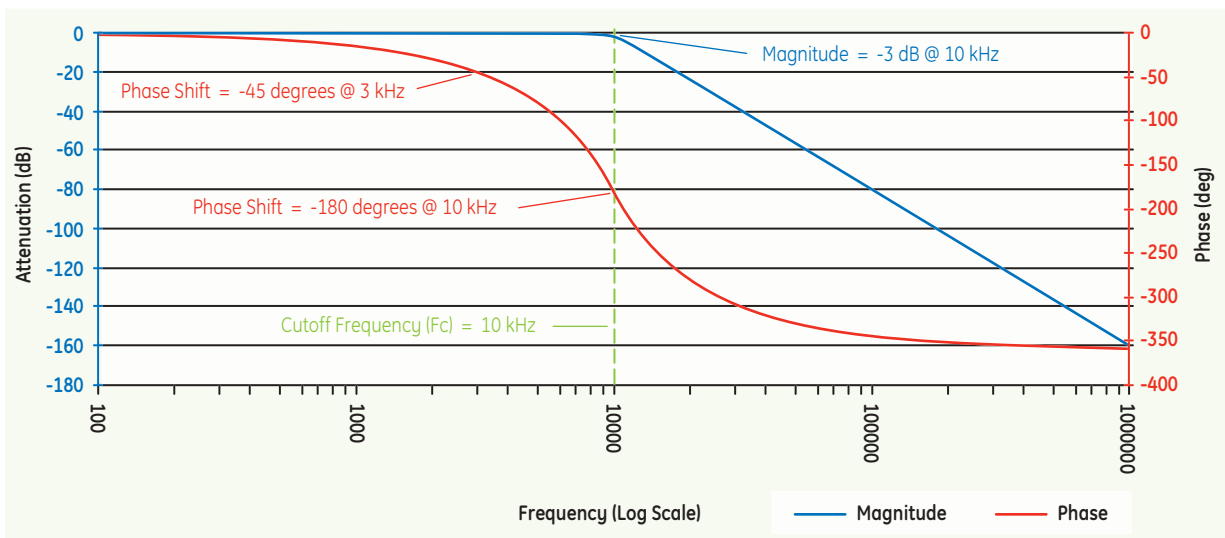
not possible for analog filters. Indeed, with analog filters, the steeper the amplitude roll-off, the more phase distortion that is introduced near the cutoff frequency. A typical 4-pole low-pass filter with a 10 kHz cutoff frequency is shown in Figure 8. The phase distortion exceeds 15° for frequencies above 1 kHz and is a full 180° at the cutoff frequency (-3 dB point).



**Figure 7A** – Amplitude spectrum for the waveforms of Figures 5B and 7B. Note that only the amplitudes of the frequency components are shown, not their phase relationships to one another.



**Figure 7B** – Resulting waveform when the phase of the 200 Hz and 500 Hz frequency components in Figure 5B are shifted by -270° and -315° respectively. Note the changes in waveform shape, pp amplitude, and symmetry that occur.



**Figure 8** – Amplitude and phase response for a 4-pole low-pass filter. Although the cutoff frequency is at 10 kHz, appreciable phase distortion occurs for frequencies above several thousand Hz. For example, a 3 kHz frequency component will undergo a phase shift of approximately 45 degrees, even though its amplitude remains unchanged. Such phase changes can have a significant effect on the filtered waveform, including its peak-to-peak amplitude.

### Other Filtering

Depending on the type of transducer input, additional digital or analog filtering may be performed on the signal besides just anti-alias filtering. This is often the case with accelerometer and velocity transducers. For example, aeroderivative gas turbines typically use high-temperature accelerometers on the machine's casing, and to prevent the extremely high amplitude blade-pass frequencies from "swamping" the rotor-related frequencies, low-pass or band-pass filtering is often done either before the signal enters the monitoring system (such as in the transducer interface module), or inside the monitoring system. Clearly, such filtering affects the waveform shape and amplitude.

### Other Signal Processing

In addition to filtering, integration is a very common type of signal processing performed by vibration measurement instruments. Some instruments allow the user to perform integration either before or after filtering. While it can be shown mathematically for ideal integrators and filters that the order in which these operations are performed does not matter, practical circuit implementations are non-ideal. Thus, integration prior to filtering can yield different results than vice-versa. In particular, integration prior to high-pass filtering may amplify low frequency components that are not directly related to machine condition, thereby reducing overall signal quality. For this reason, with few exceptions, Bently Nevada monitors generally only permit the user to integrate after filtering (see Table 2). When comparing two instruments, ensure that the order in which they filter and integrate are the same.

### Buffered Outputs

All Bently Nevada permanent monitoring systems provide buffered output connectors, generally on the front of the monitoring system. Analysis instruments, such as

portable data collectors and oscilloscopes, are routinely connected to these outputs. In fact, it is recommended that when comparing a portable or bench instrument to your Bently Nevada monitor, you use these buffered output connectors to ensure an identical signal is being introduced to the instrument as to your Bently Nevada monitor. These outputs also have the advantage of being isolated so that any inadvertent wiring problems will not affect the incoming signal to the monitor.

In the vast majority of cases, these outputs represent the unconditioned ("raw") transducer signal coming into the monitor. However, there are some exceptions with seismic monitoring channels on our 3300 and 2201 monitoring systems where these outputs can be configured for either conditioned or unconditioned signals (see Table 2).

Referring back to Figure 1, we illustrate the difference between conditioned and unconditioned outputs. When points A and C are connected, the signal at the buffered output is identical to the incoming transducer signal. This is referred to as "unconditioned." However, when points B and C are connected, the buffered output will reflect any intermediate filtering and/or integration being performed. This is referred to as "conditioned." Thus, be certain that you understand what type of signal is present at the buffered output connectors of your 3300 or 2201 monitoring system seismic channels before assuming that it is identical to the "raw" input transducer signal. If in doubt, consult the documentation supplied with your system or contact our nearest sales and service office for assistance. Also, some applications (particularly those for aeroderivative gas turbines) will pre-filter the transducer signal before it enters the monitoring system, as mentioned above. Occasionally, both the "raw" and "pre-filtered" signals will be brought into the monitoring system and made available via separate buffered output connectors.

Table 2 –

**SUMMARY OF BUFFERED OUTPUT SIGNAL PROCESSING  
FOR SELECTED BENTLY NEVADA CONTINUOUS MONITORING SYSTEMS**

System	Buffered Output Convention	Filtering/Integration Order	Buffered Output Impedance
3500	Unconditioned	Filtering always before integration	550 Ω
3300	User configurable for conditioned or unconditioned*	User configurable*	100 Ω
1701	Unconditioned	Filtering always before integration	200 Ω
2201	User configurable for conditioned or unconditioned**	User configurable**	100 Ω
7200	Unconditioned	Filtering always before integration	100 Ω
9000	Unconditioned	Filtering always before integration	3300 Ω
1900	Unconditioned	Filtering always before integration	300 Ω (1900/55) 500 Ω (1900/25 and 1900/27) 550 Ω (1900/65)
1800	Unconditioned	Filtering always before integration	9090 Ω

\* User configurable for 3300/25, 3300/26, and 3300/55 monitor modules only.

\*\* User configurable for acceleration and velocity channels only.

**Output/Input Impedance**

Another area in which discrepancies can be introduced relates to signal loss when connecting the output of instrument A to the input of instrument B. This occurs because a voltage divider is formed and the signal loss between the two devices is given as

$$\text{Signal Loss (\%)} = \left[ 1 - \frac{Z_{in}}{Z_{out} + Z_{in}} \right] \times 100 \tag{3}$$

where  $Z_{out}$  is the output impedance of instrument A and  $Z_{in}$  is the input impedance of instrument B. To keep the signal loss to 1% or less, the ratio of input impedance to output impedance should be 100:1 or greater.

For example, if the buffered output from a Bently Nevada 1800 series relative vibration transmitter (scale factor = 200 mV/mil) is connected to a device with a relatively small input impedance (say, 100kΩ), the loss of signal can be found using Table 2 and Equation [3] as follows:

$$\text{Signal Loss (\%)} = \left[ 1 - \frac{Z_{in}}{Z_{out} + Z_{in}} \right] \times 100 = \left[ 1 - \frac{100,000}{9090 + 100,000} \right] \times 100 = 8.33\%$$

Thus, an 1800 series transmitter measuring 5 mils of pp vibration would provide an output signal of 5 mils x 200 mv/mil = 1 volt pp. However, 8.33% of this signal will be lost, resulting in only 0.916 volts at the input of the other instrument. Assuming no other sources of discrepancy are introduced, this other instrument would only indicate a vibration amplitude of 4.6 mils pp.

### Summary

Because discrepancies in vibration amplitude readings from different instruments are a common source of questions from customers, understanding how these discrepancies arise is valuable for anyone tasked with interpreting the meaning and severity of such readings. In this article, we have shown that many discrepancies can be grouped into four primary categories, starting with the necessity for ensuring that the input signal to both instruments is truly identical and proceeding to some of the intermediate signal processing that typically occurs prior to amplitude detection. In particular, we have examined the effect of filtering on a signal's content and the ways that both changes in spectral amplitude and spectral phase can alter the shape and overall amplitude of a waveform.

We have also defined the conventions used in Bently Nevada instrumentation for describing a waveform's amplitude and discussed the differences in conventions used by other instrument manufacturers.

In part 2 of this article, slated for the next issue of ORBIT, we will continue with an in-depth examination of the circuits and algorithms actually used in a number of instruments to compute pk and pp amplitude, showing how they can account for dissimilar readings. We then conclude with a discussion of indication and calibration considerations. Armed with this information, vibration analysts and machinery engineers will be better prepared to understand the reasons why amplitude measurements can differ and how to interpret such differences. 