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## **Signal Processing for Digitizers**

Modern digitizer support software incorporates many signal processing features. These include ensemble and boxcar averaging, Fast Fourier Transform (FFT), and advanced filtering functions. These signal processing functions allow the user to extract extremely useful information from a simple measurement or allow enhancement of the measurement itself. This article will focus on the key signal processing functions; averaging, FFT, and filtering.

### **Averaging**

Averaging is a signal processing tool used to reduce the effects of noise and non-synchronous periodic waveforms on acquired signals. It requires multiple acquisitions and a stable trigger. Signal components that are not synchronous with the trigger timing, including random noise, are reduced in amplitude. The degree of reduction is dependent on the waveform characteristics and the number of acquisitions added to the average.

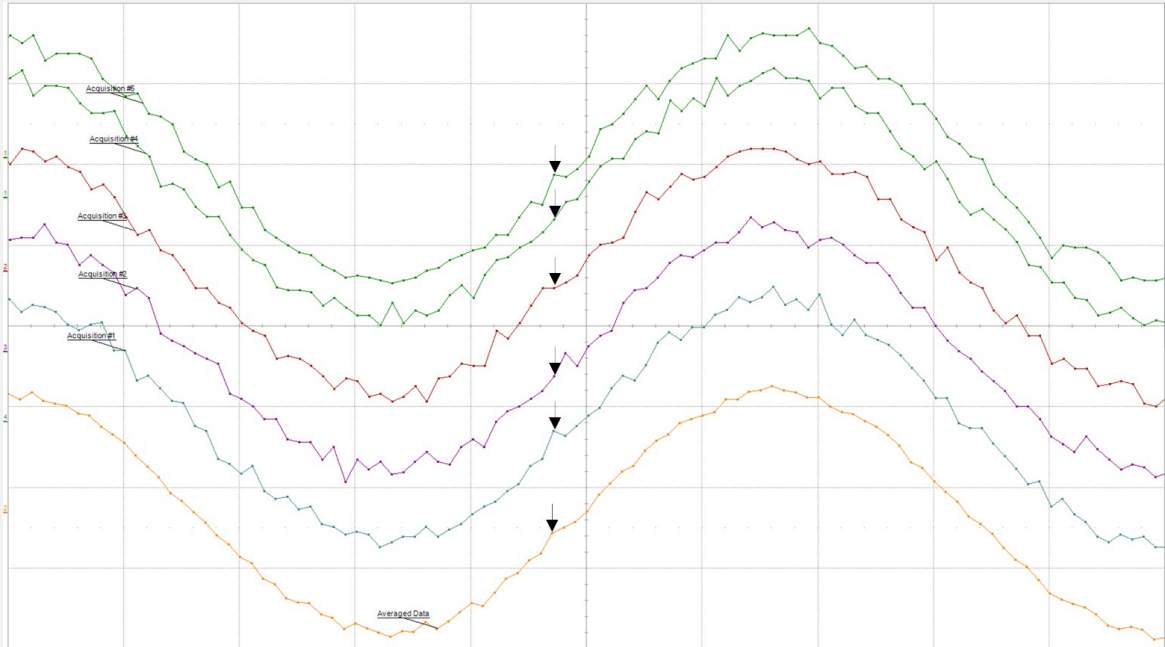
Spectrum's SBench 6 software, used in this article, and most oscilloscopes perform ensemble averaging, meaning that the same sample location in multiple acquisitions are averaged together. If a stable trigger is available, the resulting average has a random noise component lower than that of a single-shot record.

### **Summed Averaging**

Summed Averaging uses a fixed number of acquisitions and is the repeated addition, with equal weight, of the same sample locations from successive waveform acquisitions. Whenever the maximum number of sweeps is reached, the averaging process either stops or is reset to begin again.

Figure 1 shows the concept of a summed ensemble average:

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**Figure 1: Summed ensemble averaging adds the nth point of multiple acquisitions and then divides the sum by the number of acquisitions to determine the averaged value for the nth point.**

In Figure 1 the arrows indicate the nth point. The amplitude value of the nth point of each acquisition is summed with those of the other acquisitions. The sum is then divided by the number of acquisitions to determine the nth value of the average. This takes place for all sample points in the acquisition group. The resultant averaged waveform has the same number of points as each acquired waveform.

Averaging is supported for both normal acquisition and for multiple (segmented) acquisitions. Multi averaging calculations permit the average of consecutive segments of the multiple recording acquisition.

### **What Improvement Can You Expect?**

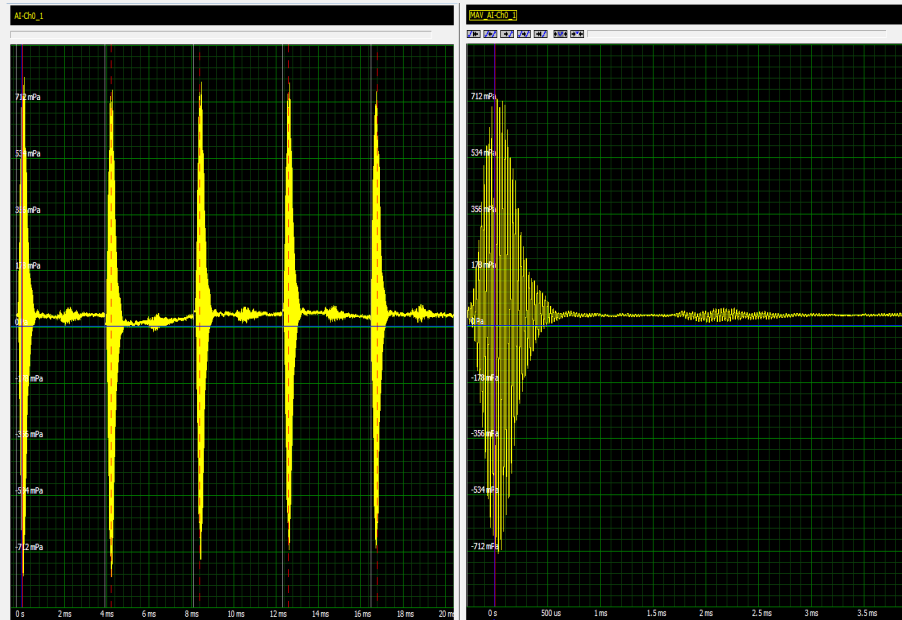
When a signal is averaged additive broadband Gaussian noise will be reduced by the square root of the number of averages. So averaging four acquisitions can improve the signal to noise ratio by two to one. Similarly, non-synchronous periodic signals will be reduced in the average. The degree of reduction depends on the phase variation of the interfering signal from acquisition to acquisition. Signals synchronous to the trigger, such as distortion products, will not be reduced in amplitude by averaging.

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**Example of Averaging**

Figure 2 shows a typical example where averaging is useful. The signal from an ultrasonic range finder is acquired using a broadband instrumentation microphone and a Spectrum M4i series 14 bit digitizer. The range finder transmits five 40 kHz pulses for each measurement. The digitizer acquires all five pulses in the acquisition memory using a multiple (segmented) mode acquisition.

The five acquired traces are shown in the left hand display grid. The transmitted pulse is obvious on the left side of each acquired segment. The low amplitude reflections are harder to see, combined with noise and spurious background signals running through the acquired signal. The waveform 'thickening' is due to noise and high frequency pickup. The baseline shaping and offset are due to low frequency pickup of ambient noise in the room where the measurement was made.



**Figure 2: The average of five acquired segments from an ultrasonic range finder improves signal to noise ratio and helps extract low amplitude reflection signals.**

The average of the five segments is shown in the right hand grid. Noise has been reduced. Extraneous low frequency pickup has also been reduced as evidenced by the flatter baseline. The reflection from the target surface is more easily discernable. Averaging has improved the signal to noise ratio by reducing the effects of noise and spurious signals not synchronous with the averaging trigger.

The principle limitation of the summed ensemble average is that it requires multiple repetitive waveforms with a stable trigger.

**Moving Average**

The moving average, sometimes called a 'boxcar' average or smoothing, takes an average of a user defined number of symmetrically placed adjacent samples. For a sample size of five the process is defined mathematically by the following equation:

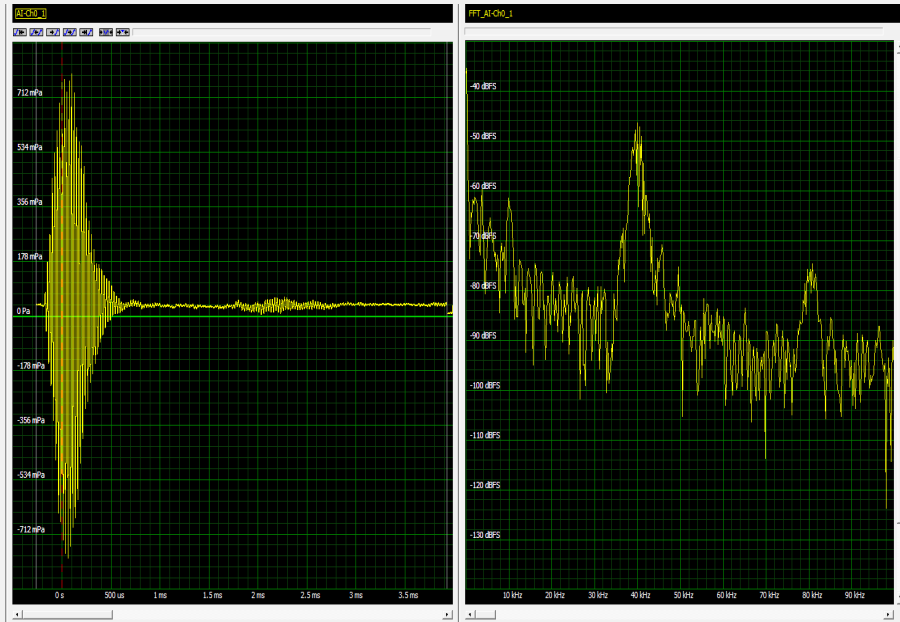
$$\text{Averaged Sample} = [\text{sample } (x-2) + \text{sample } (x-1) + \text{Sample } (x) + \text{sample } (x+1) + \text{sample } (x+2)] / 5$$

The number of samples used in the average must be matched to the period of variations in the waveform otherwise the moving average can reduce the amplitude of narrow features.

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Figure 3 provides an example of using a moving average of 15 adjacent samples, shown in the left hand grid. Note the smoothing and elimination of noise compared to the acquired waveform shown in the right hand grid

The samples are uniformly weighted and the average is taken running along the samples of the acquisition. The advantage to a moving average is that the signal need not be repetitive. The tradeoff is that in creating a smoothed waveform there is a corresponding loss of high-frequency information. Care must be exercised in setting the number of samples averaged.



**Figure 3: An example of using a moving average using 15 adjacent samples (left hand grid) showing the smoothing of the acquired waveform shown in the right hand grid.**

## Fast Fourier Transform

The Fast Fourier Transform (FFT) maps the acquired waveform from the time domain (amplitude versus time) into the frequency domain spectrum (amplitude versus frequency). This allows observation of the frequency components which make up the signal. The FFT does not improve signal quality directly but it shows the structure of the signal and provides information on how to remove undesirable components. Looking at the spectrum of our ultrasonic signal, shown in Figure 3:

The FFT allows us a better understanding of the elements that make up this signal. The primary signal is the 40 kHz burst which is clearly the frequency component with the highest amplitude. There is an 80 kHz signal which is the second harmonic of the 40 kHz component. Its amplitude is about 25 dB below the 40 kHz signal component. There are also a lot of low frequency components between 0 Hz and 10 kHz. The highest components, those near DC, are ambient noise found in the room where the device was used. Averaging may be able to reduce those components but it may take a much larger number of acquisitions. Another thing to consider is to use filtering to pass only the 40 kHz component. That leads us to our next signal processing tool.

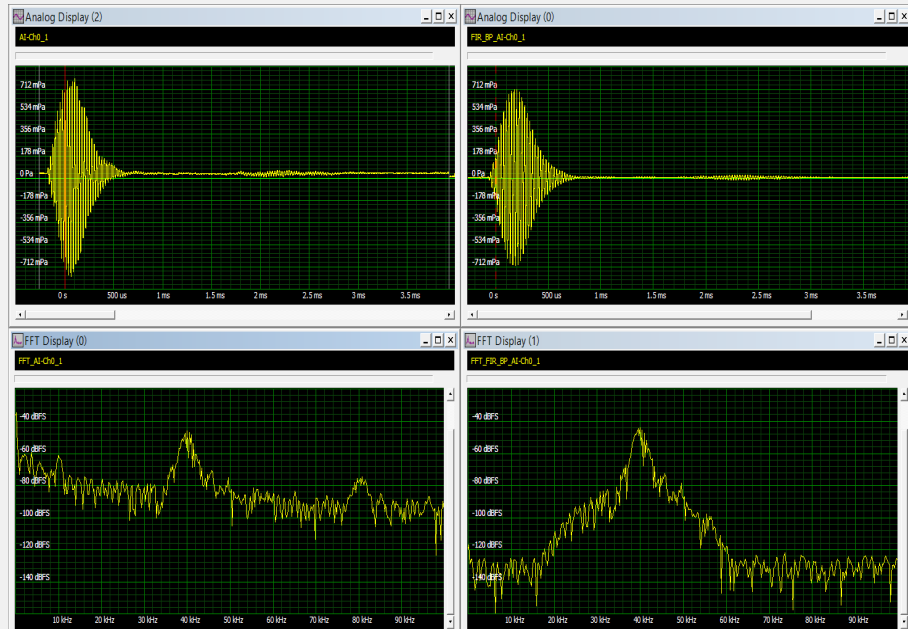
## Filtering

The SBench 6 professional software offers a choice of finite impulse response (FIR) or infinite impulse response (IIR) digital filters in low pass, band pass, or high pass configuration. These filters can be applied to the acquired signal and we can compare the results with the raw or averaged acquisitions. In Figure 4 an FIR band pass filter with cutoff frequencies of 30 and 50 kHz has been applied to the acquired signal.



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The upper left grid contains the raw waveform. Below that is the FFT of the raw signal which we have seen before. The upper right grid contains the band pass filtered waveform. The FFT of the filtered signal is in the grid on the lower right. Note that the band pass filter has eliminated the low frequency pickup and the 80 kHz second harmonic. The time domain view of the filtered signal has a flat baseline. The reflections are clearly discernible, which is the goal of the processing. Again the FFT provides greater insight into the filtering process.



**Figure 4: Showing the effects of applying an FIR band pass filter with cutoff frequencies of 30 to 50 kHz to the 40 kHz ultrasonic signal. The raw waveform and its FFT are on the left side of the display. The filtered signal and its FFT are on the right side. Note the flatness in the filtered baseline, the result of eliminating the low frequency pickup.**

## Conclusions

We have investigated the application of several signal processing tools including averaging, FFT, and filtering. Each of these tools has its role in the analysis of acquired signals. These tools are available in digitizer support software such as Spectrum's SBench 6. They can also be applied using third party math or system integration software.