Signal processing boosts digitizer performance

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Engineers use oscilloscopes and modular digitizers to acquire signals, analyze their characteristics, and make decisions. Often, those decisions affect designs or give an indication of a pass/fail condition in production. Working with raw data doesn't always tell you what you need to know. Bench oscilloscopes often have some signal processing included. Digitizers often come with software that lets you view and analyze digitized waveforms. You can also use third-party software to analyze digitized signals in real time or offline.

Common signal-processing applications include ensemble and boxcar averaging, FFT (Fast Fourier Transform), and digital filtering. You can use these functions to extract useful information from a simple measurement or enhance the measurement itself.

Averaging
Averaging can reduce the effects of noise and non-synchronous periodic waveforms on acquired signals. It requires multiple acquisitions and a stable trigger. Averaging will reduce the amplitude of signal components that aren't synchronous with the trigger timing, including random noise. 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 and other oscilloscope PC software applications 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. 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.

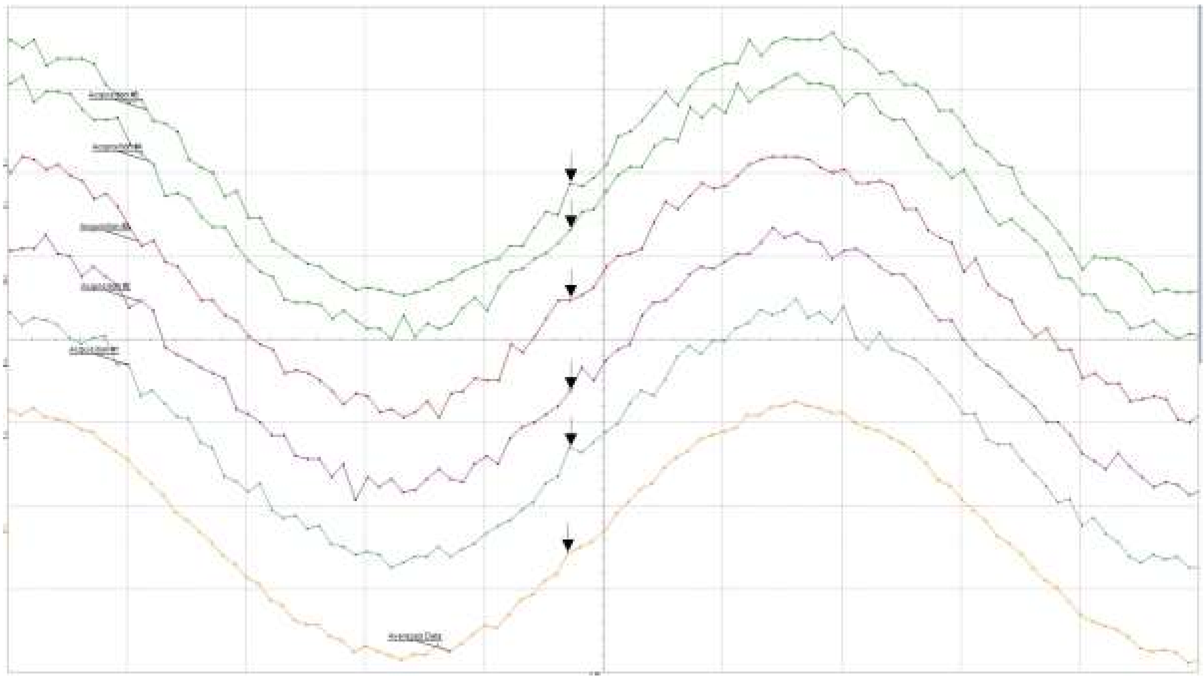

Figure 1. Summed ensemble averaging adds the nth point of multiple acquisitions and then divides the sum by the number of acquisitions to determine the averaged value for the nth point.

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.

When a signal is averaged additive broadband Gaussian noise will be reduced by the square root of the number of averages. Thus, averaging four acquisitions can improve the signal to noise ratio by 2:1. 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.

Averaging example

Figure 2 shows a typical example where averaging is useful. The signal from an ultrasonic range finder is acquired using a broadband instrumentation microphone. 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.

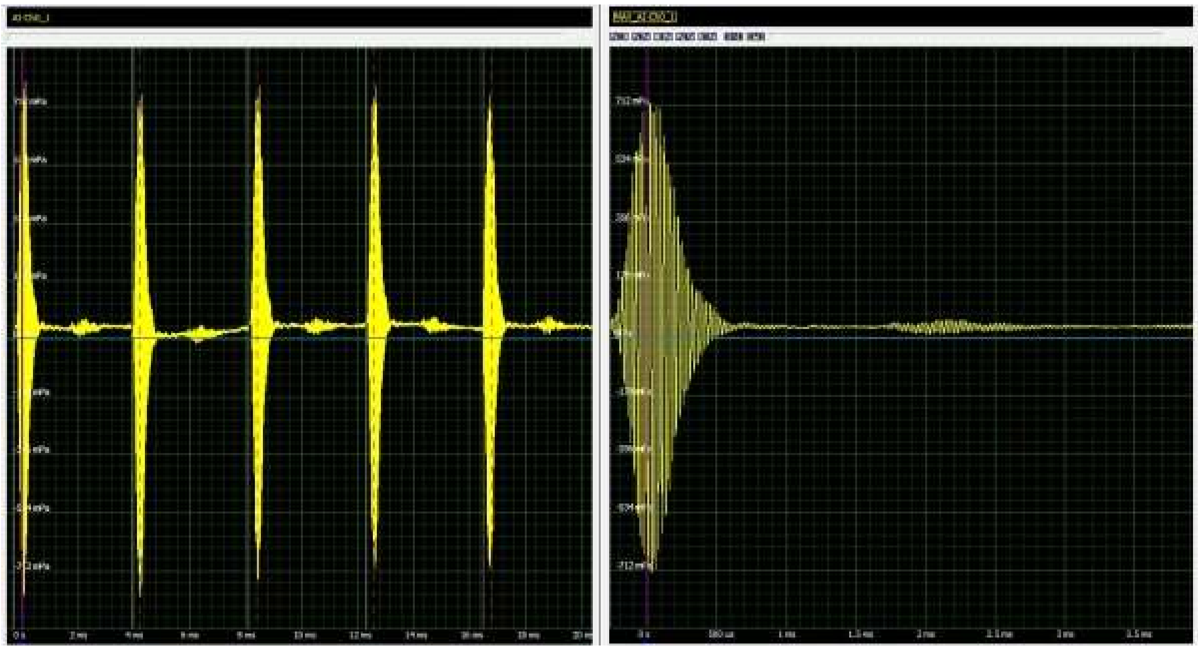

Figure 2. The average of five acquired segments from an ultrasonic range finder improves signal to noise ratio and helps extract low amplitude reflection signals.

The five acquired traces are shown in the left-hand grid in Figure 2. 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 caused by 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.

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 SNR (signal-to-noise ratio) by reducing the effects of noise and spurious signals not synchronous with the averaging trigger. Remember that the summed ensemble average 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:

$$Averaged\ Sample = \frac{\mathrm{sample}\ (x-2)\ +\ \mathrm{sample}\ (x-1)\ +\ \mathrm{Sample}\ (x)\ +\mathrm{sample}\ (x+1)\ +\ \mathrm{sample}\ (x=2)}{5}$$

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. **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.

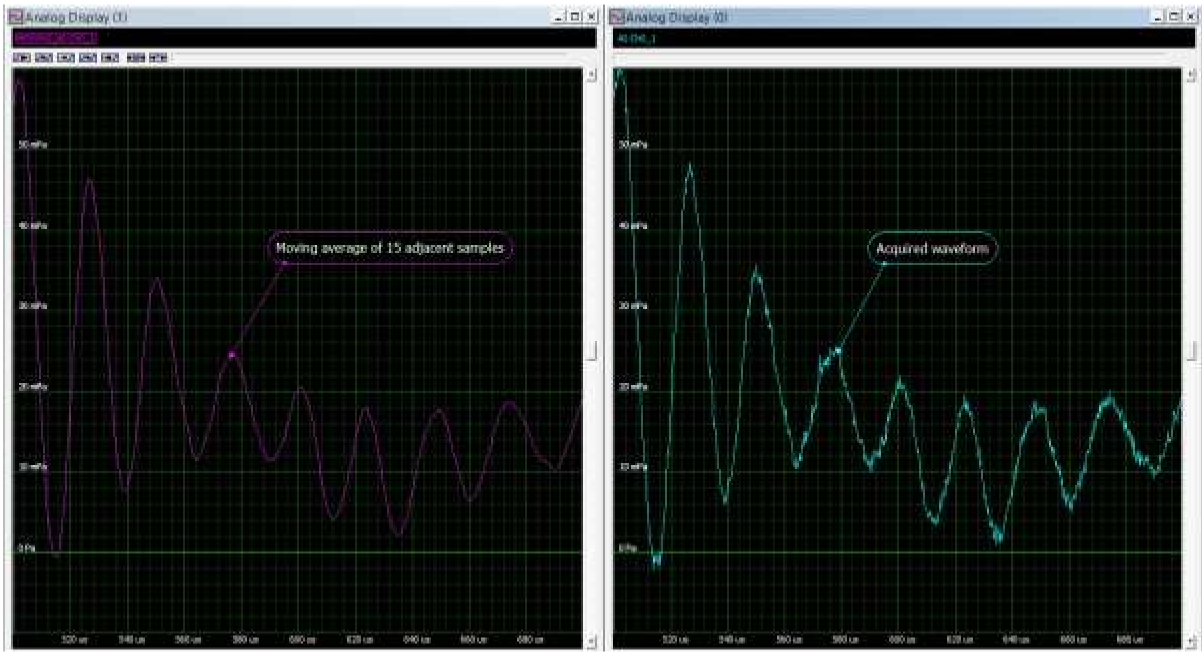

Figure 3. An example of using a moving average using 15 adjacent samples (left hand grid) showing the smoothing of the acquired waveform shown in the right hand grid.

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.

FFTs and filters

Fast Fourier Transform

The FFT maps the acquired waveform from the time domain (amplitude versus time) into the frequency domain spectrum (amplitude versus frequency), which lets you observe a signal's frequency components. The FFT won't directly improve signal quality, but it shows the structure of the signal and provides information on how to remove undesirable components. **Figure 4** shows spectrum of our ultrasonic signal.

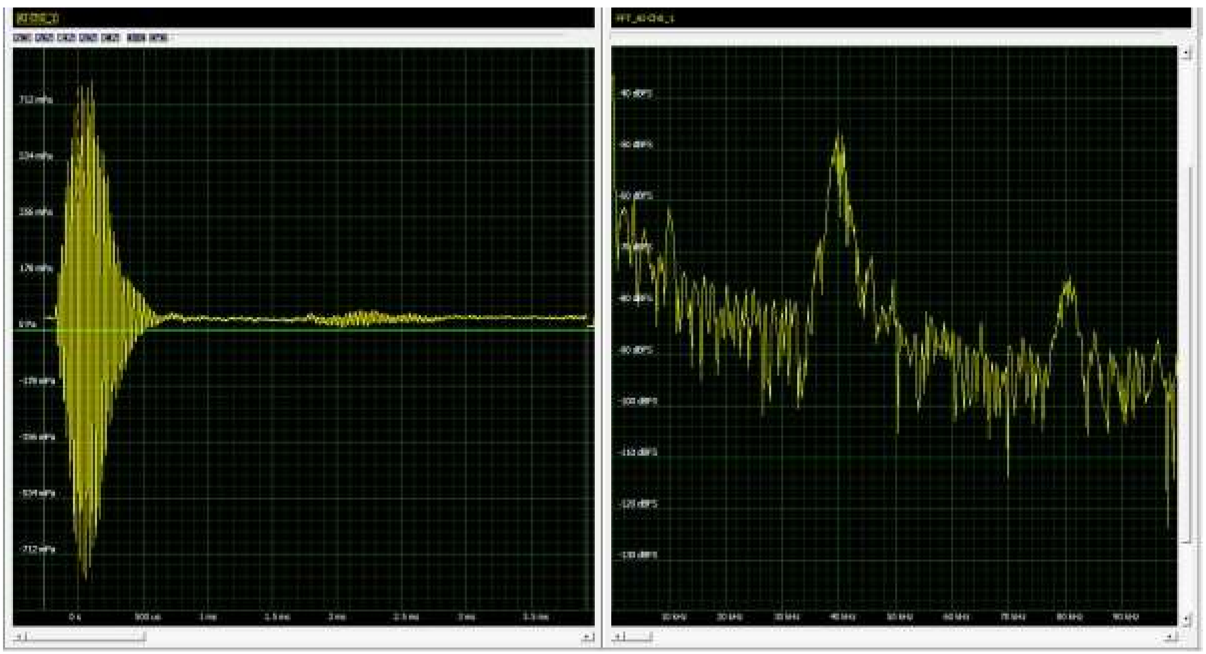

Figure 4. The FFT of the acquired ultrasonic signal shows the main signal centered at 40 kHz, a second harmonic at -25 dB below the fundamental, and significant low frequency components between DC and 10 kHz.

The FFT provides 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 fundamental frequency. 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. You should also consider using a filter to pass only the 40 kHz component. That leads us to our next signal processing tool.

Filtering

Software used with many digitizers often provides a choice of FIR (finite impulse response) or IIR (infinite impulse response) 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 5**, an FIR band pass filter with cutoff frequencies of 30 and 50 kHz has been applied to the acquired signal.

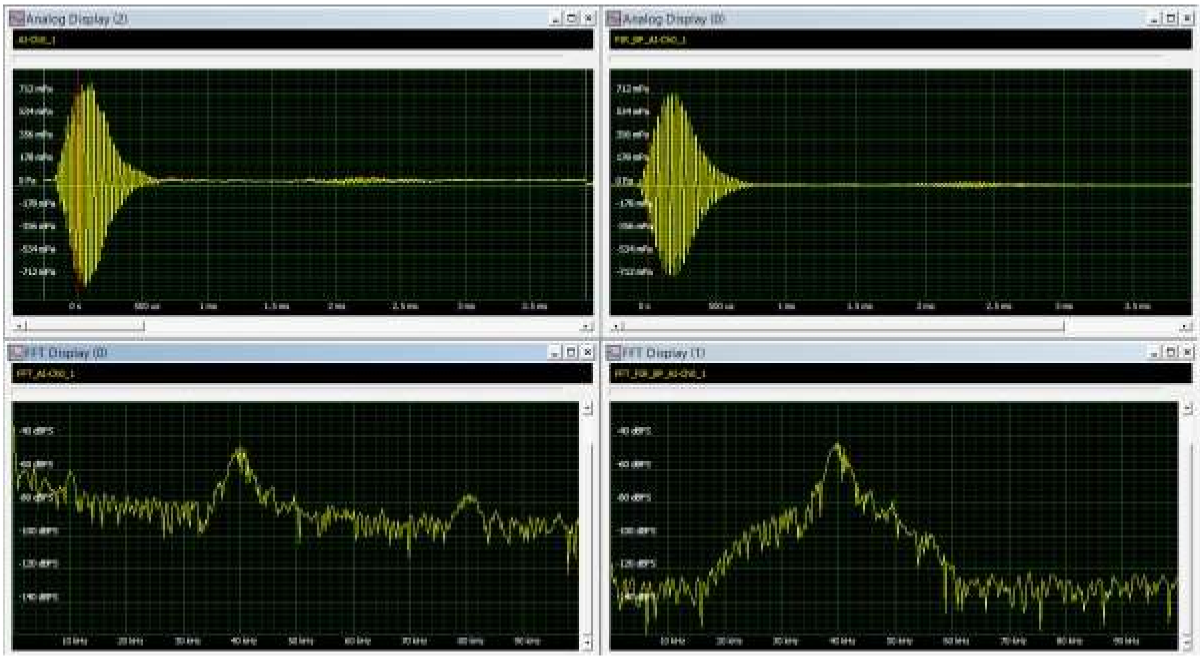

Figure 5. Applying FIR band pass filter with cutoff frequencies of 30 to 50 kHz to the 40 kHz ultrasonic signal. The raw waveform and its FFT are on the left side of the display. The filtered signal and its FFT are on the right side. Note the flatness in the filtered baseline, the result of eliminating the low frequency pickup.

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.

Conclusion

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. They can also be applied using third-party math or system-integration software.

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