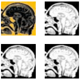

Small minded?

Just because a data acquisition system is portable does not mean it is less capable

by Larry Jacobsen, Campbell Scientific

Recent advances in integrated circuit technology have decreased the size, power consumption, and cost of digital signal processing hardware to a point where portable data acquisition systems now offer capabilities previously reserved for larger, more expensive, AC-powered instrumentation.

The CR9052DC filter module (Figure 1), developed for the CR9000 Measurement and Control System is an example of how high-performance digital signal processing is becoming available in portable, battery powered data acquisition systems. The digital design capabilities of the CR9052DC's have impact on two specific signal processing issues – anti-alias filtering and Fourier-transform spectral analyses – and they cannot be underestimated.

Anti-alias filtering: Aliasing is the erroneous interpretation of high-frequency signals as lower-frequency ones – such misinterpretations are an expected result of making discrete measurements with sampling devices such as analog-to-digital (A/D) converters. Figure 2 shows a continuous 40Hz sinusoid, a continuous 60Hz sinusoid, and the positions of discrete samples taken at a 100Hz measurement rate. As the sample positions indicate, it is impossible to distinguish the sampled 40Hz signal from the sampled 60Hz signal because, at a 100Hz sample rate, the 60Hz signal is aliased on top of the 40Hz signal. Furthermore, aliasing will occur for any signal whose frequency is above one-half of the sample frequency (Nyquist frequency).

This setup affects the data acquisition process in two ways: firstly, it is impossible to discern which frequencies

Figure 1: The CR9000 measurement and control system and CR9052DC (inset photo)

were present in the original signal on the basis of the discrete measurements; and secondly, if it is the 40Hz signal that is of interest, aliasing by the 60Hz signal will corrupt measurement of the 40Hz signal.

Data acquisition systems often use an analog low-pass filter prior to, or in conjunction with, the sampling process to prevent high-frequency signals (or noise) from corrupting lower-frequency measurements. This low-pass filter is called an anti-aliasing filter, and its goal in the example of Figure 2 is to preserve the 40Hz signal, while eliminating the 60Hz signal before it is aliased. Traditionally, data acquisition systems have used high-order analog filters such as four- and eight-pole Butterworth filters in front of the A/D converter in order to achieve anti-alias filtering.

Figure 3 shows the block diagram for a digital approach to anti-aliasing implemented by the CR9052DC (or 9052). The 9052 achieves programmable anti-alias filtering by the combination of a simple, three-pole Butterworth analog filter, a sigma-delta A/D converter, and a digital signal processor (DSP). A DSP is a specialized microprocessor that is well suited to implementing signal processing algorithms. The analog filter and the A/D converter are replicated for each input channel, and a single DSP processes all six channels.

The combination of the three-pole analog filter and the sigma-delta A/D converter provides alias-free, 50kHz samples to the DSP. The DSP applies additional, programmable filtering in the form of finite-impulse-response (FIR) filters, and then decimates these filtered results to the user's desired sample rate. An FIR filter produces an unevenly weighted, moving average of a finite number of A/D samples, while the DSP tailors the frequency response of the FIR filters by changing the relative weights and lengths of the moving average.

There are several advantages of using the digital approach over the traditional analog filter approach. Firstly, digital components are software programmable, whereas in the case of an analog filter, when the user desires to use a different sample frequency, the anti-alias filter must change with it because the sampling frequency determines which frequencies are aliased. A different analog filter is required to attenuate the new aliased frequency components. In contrast, to re-program the digital filter, the DSP simply loads a different set of FIR filter coefficients from its library. The superior programmability of the digital approach also allows users to implement customized low-pass and band-pass filters of their own design, in addition to the standard low-pass anti-alias filters in the 9052's library.

A second advantage of the digital filter approach is that it is not affected by analog component tolerances, component ageing, or component changes caused by temperature. All of these issues affect the analog filter performance, especially near the filter's cut-off frequency (-3dB response). The digital filter is perfectly repeatable from channel to channel, and from module to module.

At first glance, it may appear that the 9052's digital approach depends on analog components, as does the analog filter approach, because it includes a three-pole analog filter preceding the A/D converter. The objective of the three-pole filter is to remove the sigma-delta A/D converter's replicated pass bands at multiples of its fundamental sample rate (3.2MHz, 6.4MHz, 12.8MHz, etc) without compromising the A/D converter's frequency response from 0 to 20kHz. This goal is easy to achieve by setting the three-pole filter's cut-off frequency at 150kHz. This relatively high cut-off frequency greatly minimizes the effects of component variations at 20kHz, and because the 9052 does not change the sigma-delta A/D converter clock rate to achieve the desired frequency response, there is no need to re-program the three-pole analog filter.

The third, and perhaps most important, advantage of the digital approach over the analog approach is the digital filter's rapid transition from its

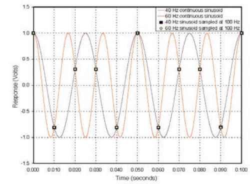

Figure 2: Graph of 40- and 60Hz sinusoids being sampled at 100Hz

pass band to its stop band. This rapid transition gives users a more usable bandwidth for a given sample rate. Figure 4 compares the response of an ideal eight-pole Butterworth filter with the FIR filter of the 9052. Both of the filters in Figure 4 give 95dB of attenuation (about 1/56,000) at frequencies above 25kHz, but the digital filter response is flat within ±0.01dB (flat to about ±0.1 per cent) to frequencies beyond 20kHz. In contrast, the analog filter maintains this same flatness to only 4.3kHz. The frequency response of either of these filters will anti-alias (to -95dB) a 50kHz sample rate, but if users desire amplitude measurements accurate to within ± 0.1 per cent, then the digital filter provides 20kHz of useable bandwidth, while the analog filter only gives 4.3kHz of usable bandwidth. Figure 4 shows the ideal eight-pole Butterworth filter for comparison with the 9052 filter. In real-world applications, component tolerances and operational-amplifier finite bandwidths will degrade analog filter performance from that shown.

As a final comparison, the digital FIR filter offers a linear phase response, or time delay independent of frequency, throughout its pass band. In contrast, the phase response of the eight-pole Butterworth filter is non-linear, causing different time delays for different frequencies. The group delay (negative the slope of phase versus frequency) of an ideal eight-pole Butterworth filter with a 20kHz cut-off frequency changes by over 300 microseconds over a frequency range of 0 to 20kHz. This frequency dependent group delay causes distortion in the filter's output, including substantial ringing in its step response. The 9052 FIR filter delay is a constant 630 microseconds over this same 20kHz range – the FIR filter delay is larger than the analog filter, but this delay does not change with frequency.

Fourier-transform spectral analyses: Although data acquisition systems typically measure signals as a function of time, users often evaluate the resulting data as a function of frequency. The fast fourier transform (FFT) is often used to spectrally decompose a sampled time series into its sinusoidal frequency components. Users record a snapshot of data, then apply the FFT to obtain average amplitude and phase (or alternatively, real and imaginary parts) of the sinusoidal frequency components of that signal.

Because the FFT is computationally intensive, many users must apply it in post processing rather than at the time of data acquisition. The fact that the small number of data acquisition systems with real-time FFT capabilities have traditionally been specific purpose, AC-powered instruments with a limited number of input channels is also limiting.

The DSP in the 9052 can collect real-time, seamless, 2,048-point snapshots of 50kHz data from each of its six A/D converters, and perform FFT analyses on these snapshots to give 1,024-point complex (real and imaginary) spectra. The term 'seamless' means that there are no gaps between consecutive time-series snapshots that are processed into spectra. This factor ensures that the 9052 will not miss transient events that could otherwise occur between snapshot recordings.

Following capture and FFT analyses, the 9052 can convert those complex spectra to amplitude spectra, power spectra, or power spectral density functions; bin average adjacent spectral components; and then output the spectral results to the CR9000 CPU. The DSP can accomplish this data collection, analysis, and output in real-time for all six 50kHz A/D converters.

The FFT gives the average spectral amplitude computed over the entire snapshot length. For this reason, the 9052 allows users to choose FFT lengths

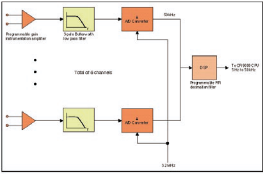

Figure 3: Block diagram of the CR9052's digital approach using a Butterworth filter, sigma-delta A/D converter and DSP with programmable FIR filter

shorter than 2,048 points, thereby shortening the snapshot time and allowing accurate measurement of short-lived signals. The anti-alias filtering and decimation described previously allow the DSP to transform time-series collected at sample frequencies slower than 50kHz. These slower sample rates allow users to trade high-frequency bandwidth for enhanced low-bandwidth resolution.

The optional spectral bin averaging of adjacent frequency components can be programmed for a fixed averaging width, or it can use an averaging width that increases exponentially with increasing frequency. This exponentially increasing bin width provides results known as 1/n octave analyses, where 'n' specifies the number of measured frequency components per octave (an octave is a twofold increase in frequency). The 9052 lets users choose 1/n octave analyses for n values ranging from 1 to 12.

After the 9052 has provided spectra to the CR9000 CPU, CR9000 provides numerous options for further processing. For example, users can save raw spectra, ensemble-averaged spectra, or peak-detected spectra into final storage, and they can also quantify spectral similarities of different channels by computing spectral ratios, co-spectra, cross-spectra, or spectral covariances.

Further benefits include the ability to compute and save the power in a specific frequency band, to compare this computed power to a fixed threshold, and then to save the associated spectra on the basis of that comparison. The device allows the user to manipulate and save spectra on the basis of measurements from other CR9000 input modules.

Lastly, users can benefit from the fact that the 9052's measurements can be processed and output in real-time with the measurements made by other CR9000 modules. These measurements could come from thermocouples, hydraulic pressure transducers, pickup coils, or the stream of CANbus data. ●

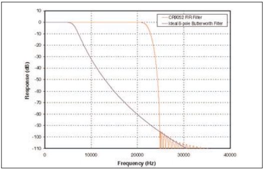

Figure 4: Comparison of the transition band roll-off of an eight-pole Butterworth filter against the CR9052DC's FIR filter