Designing High-Speed Data Acquisition Systems on VMEbus

Before the benefits of modern digital signal processing techniques can be applied to high-speed VMEbus systems, the dynamic signals coming from a predominantly analog world must be acquired and converted to digital form.

This is done by a high-speed data acquisition stage similar to the one shown in Figure 1. The primary component is the A/D (analog-to-digital) converter, also sometimes called a "digitizer". Other circuitry includes a signal-conditioning amplifier, an anti-aliasing filter, and a sample-and-hold, which is usually integrated within the A/D. Figure 1 forms the basis for discussing the function of each block.

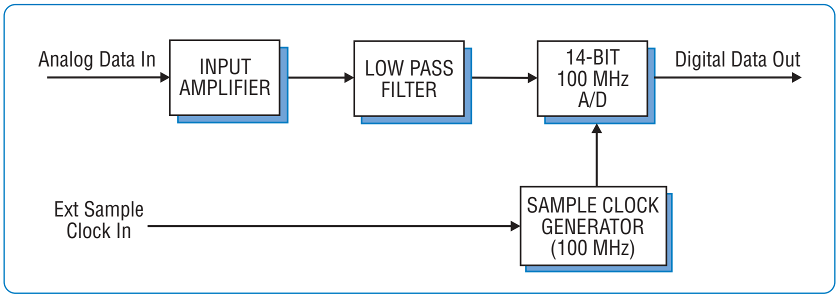

Figure 1. Typical high-speed A/D converter for data acquisition.

Signal Conditioning

The need to apply some conditioning to the incoming signal stems from various factors, such as:

Is the signal expected to drive the A/D to nearly full-scale some of the time, or do we need an amplifier to boost it to the recommended range, so that we don't sacrifice A/D dynamic range? Alternatively, do we need to attenuate the signal, so that it does not drive the A/D into saturation, thereby generating undesirable distortion products?

Is the small differential signal of interest riding on top of a large ground-referenced signal that does not convey any information? In this case, we not only need an amplifier, we need one with differential inputs and high "common mode rejection" to attenuate the unwanted common mode signal.

Other characteristics of the amplifier, such as input impedance and bandwidth, are dictated by the nature of the input signal and our frequency band of interest.

In short, any data acquisition system is likely to include an input signal conditioner. If the system is to handle more than one channel of data, we can expect it to employ multiple signal conditioners, one for each channel. In some cases, these may be followed by an analog multiplexer, so that we need to use only one A/D converter.

Sampling

Having settled on the specifications of the signal conditioner, the next task at hand is to pick the sampling frequency of the A/D converter. As anyone familiar with sampled data theory knows, we must pick the sampling frequency to be at least twice the input signal bandwidth, i.e. the highest expected frequency of the input signal, which is also known as the Nyquist or folding frequency.

But how do we know what might be the highest frequency ever expected in our input signal?

Is the input signal bandlimited by its own nature? What about noise? What about noise infecting a low-level signal because of ground loops, heavy traffic on the VMEbus, or pickup from adjacent boards? Is this noise bandlimited too, or is it wideband?

And what would happen if we disregard these warnings and just hope for the best?

Aliasing

If we do nothing to alleviate these concerns, we are guaranteed to have aliasing. Simply stated, aliasing generates false signals which we will process with our DSP system, and obtain false and erroneous measurements.

Aliasing can be illustrated in both the time and frequency domains. As we can readily see in Figure 2A, any number of time waveforms can result in the same set of sampled data points and will be indistinguishable to the DSP processor. In the frequency domain, shown in Figure 2B, we can readily see that aliasing generates folding, or spectrum overlap. The only way to eliminate it from our processing is by using a low-pass filter whose attenuation at the lowest alias frequency meets our dynamic range requirements.

Anti-Aliasing Filters

An anti-aliasing filter is always required to ensure that input signal and noise are indeed bandlimited, so that any aliases generated in the subsequent sampling will meet the dynamic range specification.

The example shown in Figure 3, depicts an anti-aliasing filter suitable for processing signals with up to 40 MHz bandwidth. This filter is a high-order elliptic with

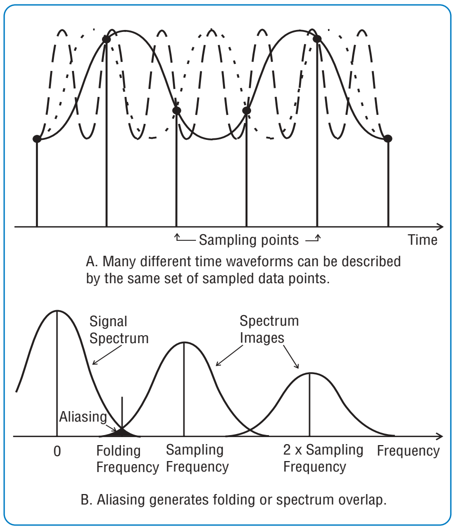

Figure 2. Aliasing as seen in both, the time (A) and frequency (B) domains.

minimum out-of-band rejection of 80 dB. This rejection is first reached at 60 MHz and the subsequent typical elliptic "bouncebacks" in its response remain below 80 dB.

To process the 40 MHz information bandwidth, the sampling frequency should be set at 100 MHz, rather than the 80 MHz minimum dictated by the Nyquist criterion. Using a higher frequency than the theoretical value merely acknowledges the fact that there is no such thing as the "ideal" low-pass filter with infinite rolloff rate. In this case, the critical folding frequency is 100 – 40 = 60 MHz, rather than 40 MHz. This is the lowest frequency where the filter attenuation meets the 80 dB specification, thus assuring an alias-free analysis band from DC to 40 MHz.

Types of Anti-Aliasing Filters

The filter whose response is shown in Figure 3 may be implemented in one of two ways: a passive-LC filter, or an active-RC filter.

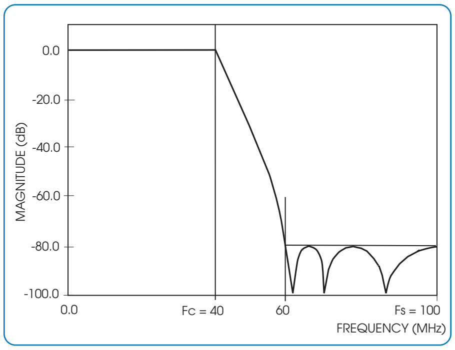

Figure 3. Typical anti-aliasing filter response.

These implementations involve an analog filter ahead of the A/D. What about a digital filter after the A/D instead? We may implement one as part of our digital signal processing system, but we still need an anti-aliasing filter before the A/D. If aliasing should occur during the sampling process, it just cannot be eliminated by subsequent digital filtering.

Sample-and-Hold and A/D

Going back to our previous numerical example, we specified the response of the anti-aliasing filter, and we set the sampling frequency at 100 MHz rather than 80 MHz. Using the "rule of thumb" of 6 dB per bit, we conclude that we should use a 14-bit A/D, if we are to realize 80 dB of alias-free dynamic range.

We have now completed the specifications for the input amplifier, the anti-aliasing filter, the sampling frequency, and the A/D converter. The function of the sample-and-hold is to hold the input to the A/D at a constant value until the end of the conversion process. In many cases, it is an integral part of the A/D. If a separate one is used, its specifications are determined by the maximum sampling frequency, number of bits of accuracy, and A/D conversion time specification.

A New Kind of A/D

In low frequency applications (i.e. less than 100 kHz), aliasing problems have been solved by the sigma-delta (or delta-sigma) A/D converter. By incorporating a very fast oversampling front end and a high-order digital filter, analog filter complexity is dramatically reduced. We can easily visualize how a simple-RC filter can provide sufficient aliasing rejection for a 10 kHz input signal, if the signal is oversampled at a 10 MHz rate. The internal digital filter eliminates all signals above 10 kHz and provides a decimated output sampling rate of just over 20 kHz.

Noise, the Clear and Present Danger

Having 80 dB of alias-free dynamic range does not guarantee 80 dB overall dynamic range in our high-speed VMEbus data acquisition system. Factors which can degrade performance include noise picked up or generated after the filter, such as: digital noise from various clocks coupled through the backplane; noise picked up from adjacent boards; noise from improper system grounding; and noise or line-related spurious components from the card cage power supply.

Power supply problems can be solved with decoupling and filtering, regulators, and/or isolation with DC/DC converters on the data acquisition board. Noise from a switching power supply would not be a problem, if the switching frequency were outside the data bandwidth. This, however, is not the case with high-speed systems, so in all likelihood we will have to use linear rather than switching power supplies in the VME card cage.

The best way to avoid backplane noise is to bring signals into the data acquisition board through separate front panel connectors, rather than using the VMEbus connectors. This may not be practical and is reserved for the most demanding requirements.

Noise from adjacent boards can be reduced with more physical separation and shielding. It just takes experience and persistence to solve noise problems due to improper system grounding. For best results, try killing noise at the source, rather than letting it spread around and then having to cure it in a dozen places. ❑