Precision Time Interval Measurements with the E1437

Howard Hilton, Hewlett-Packard, Lake Stevens Division

The traditional approach to time interval measurement attempts to derive the event arrival time strictly from the instant at which the input signal crosses a given voltage threshold. The measurement instrument then associates the event arrival with a time resolution interval, usually determined by a high frequency clock and counter. Thus, a 1GHz internal clock would produce 1ns time interval measurement resolution. For precision time interval measurement, this approach relies on a wide bandwidth comparitor, very high clock frequencies with low jitter, and a high speed counter.

An alternate approach is to use a general purpose digitizing front-end preceded by an alias protection filter. This approach collects the entire bandlimited signal rather than just the threshold crossing time. From this more complete information it is possible to determine the arrival time of an event to much higher resolution than the digitizing sample frequency. Using this technique the E1437A, a 20MSample/s digitizer, has been used to achieve sub-nanosecond resolution single shot time interval measurements. This is not an academic, special case example. In fact the technique described in this paper has significant benefits in many applications.

The principle of the digitizing approach can be demonstrated with an example. Assume that the event in question is a positive voltage step and that the input signal contains not only this step, but added random noise as well. Figure-1 shows an actual measurement of such a signal digitized at 20MSamples/sec on an E1437A. To make the noise large enough to be noticeable, the step size was made very small compared to the ±10.24V input range used for this measurement. Note that the -100mV to 100mV step is approximately 1% of the input range.

Figure-2 shows this same signal sampled at 20MHz, using the ±160mV range and an expanded time scale. The lower input range was used to show how small the E1437A input noise can be made because of its exceptional dynamic range. The expanded time scale shows the problem with using an unfiltered input signal. The step could have occurred any time between the 150ns sample and the 200ns sample. The straight line drawn on the graph between these two samples is not an accurate representation of the actual input signal because we have no information about what the signal is really doing in this interval. The best that can be done is to say the step arrived between 150ns and 200ns after the measurement was triggered. Thus, the timing resolution is 50ns.

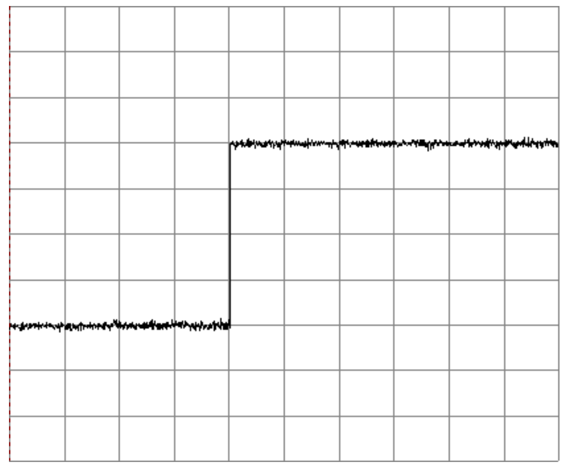

Figure-1, Unfiltered step input, 50mV/div vertical, 5us/div horizontal

Figure-2, Unfiltered step input, 50mV/div vertical, 150ns/div horizontal

By applying a lowpass filter to the input signal before digitizing, the arrival time ambiguity can be removed, provided that the bandwidth of the filter is narrower than ½ the sample rate. This filter, known as an alias filter or Nyquist filter, assures that the digitized samples contain sufficient information to completely reconstruct the underlying continuous bandlimited signal. The analog alias filter built into the E1437A is an 8MHz wide 11-pole elliptic filter. Because of the filter's non-linear phase characteristics, the resulting step response is not symmetric. Figure-3 shows the sampled step response with the alias filter enabled. Note the ringing response caused by the filter. At first glance this may look no better than the previous graph for extracting timing information. In reality, the step timing can now be determined much more accurately because we know what the input signal looks like between samples. We can reconstruct this bandlimited signal completely.

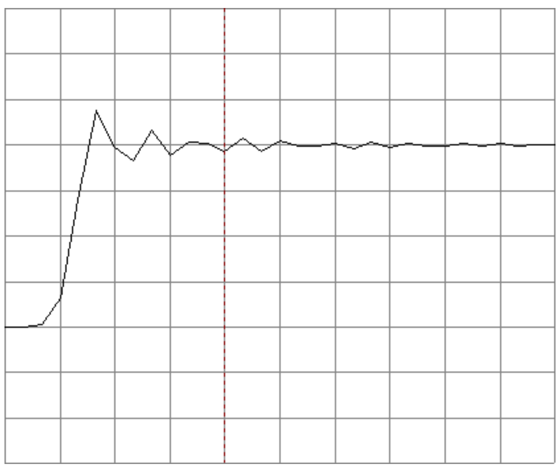

Figure-3, Filtered step input, 50mV/div vertical, 150ns/div horizontal

The E1437A is shipped with a small library of useful signal processing functions. This HPDSP library includes a function called hpdsp_resample( ) which accomplishes two things. It applies additional filtering to the signal to achieve a non-ringing response which settles more quickly, and it computes output samples at a higher rate than the original sample rate. In order for this function to work properly the original sampled signal must meet the Nyquist criteria. Applying the resampling reconstruction function to non-alias protected data will give incorrect and misleading results. Figure-4 shows resampled actual data taken from the E1437A. The output resample rate was increased to 160MSamples/sec, although it could just as easily have been resampled at 1GHz or higher.

Once the signal has been resampled at least eight times the Nyquist rate, the straight lines connecting adjacent samples become more representative of the actual band limited signal. In fact a straight line interpolation between samples is extremely accurate. Thus, we can now compute a very accurate measurement of the time at which the filtered signal crosses a given voltage threshold. However, a better technique is often not to look for a particular threshold crossing time, but to look for the time where the slope is at its maximum. The latter approach eliminates the arrival time dependence on step size. It is less sensitive to low frequency additive noise which shifts the baseline. It also avoids the problem of having to select the threshold voltage relative to a potentially unknown DC offset.

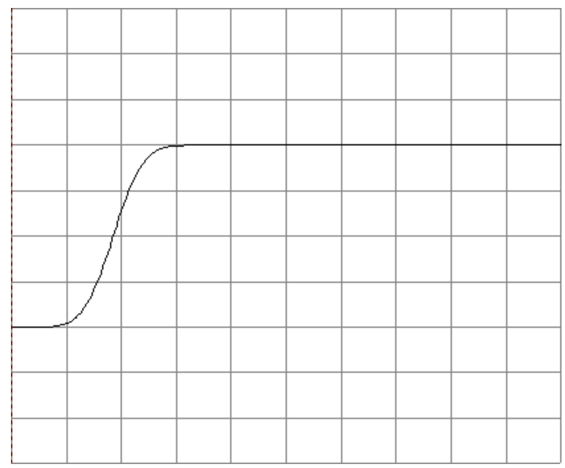

Figure-4, Step resampled at 160MS/s, 50mV/div vertical, 150ns/div horizontal

The specific algorithm is as follows: **(1)** Start a measurement and read a record of data long enough to contain two events. **(2)** Resample the data to 10x the original ADC sample rate using the hpdsp_resample function. **(3)** Approximate the time derivative by forming a record consisting of the differences between subsequent samples, recording the peak difference, Vp, in the process. **(4)** Scan the derivative record until a sample less than Vp/2 is encountered. **(5)** Continue scanning record until the signal exceeds Vp/2. **(6)** Continue scanning until a relative maximum is found (the next sample is smaller than the current sample). **(7)** Use the sample at this relative maximum and the two adjacent samples to fit a quadratic polynomial to the data. **(8)** Calculate the peak time of the quadratic to determine the actual peak of the derivative. **(9)** Starting at the first peak, repeat steps 4 through 8 to find the second peak. **(10)** Subtract to get the time between peaks and scale the result using the ADC sample frequency.

The technique described above was used to measure single shot time intervals between subsequent rising edges of a 1MHz square wave. The square wave was produced by an HP 3326A synthesized function generator, and the measurement was made with the HP E1437A VXI digitizer. Figure-5 shows a histogram of period measurements on a scale of 100ps/div. The jitter shown in this histogram is dominated by the jitter inherent in the 3326A. This fact can be demonstrated by using a 3325B function generator instead. The resulting jitter histogram is shown in figure-6. Even this is dominated by the jitter of the source, a fact that will become evident in the next example.

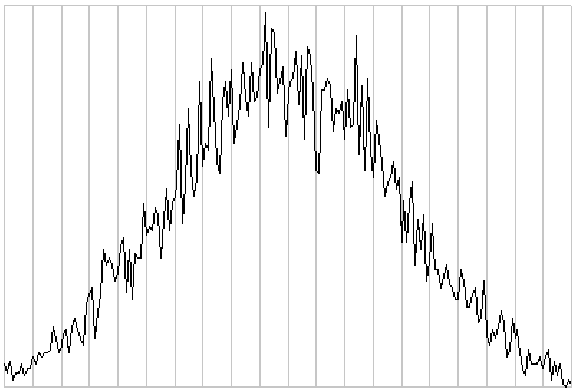

Figure-5, Interval histogram of 3326A 1MHz square wave, 100ps/div horizontal, 1us center

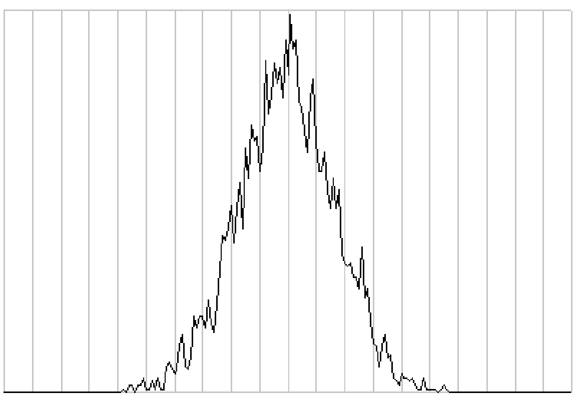

Figure-6, Interval histogram of 3325B 1MHz square wave, 100ps/div horizontal, 1us center

Although the preceding example used voltage step events, it is possible to make accurate time interval measurements with any repeatable waveform, provided that individual events are separated by sufficient time to not overlap after filtering is applied. The detection algorithm can use any characteristic of the event as the definition of arrival time.

Because the E1437A was designed to operate in multi-channel synchronous applications, it is also possible to measure relative time of arrival of events on two separate channels. This actually makes a better example to show the timing resolution of the E1437A since the measurement is no longer dominated by the source jitter. In the following example the same voltage step was applied to two E1437A modules via a coaxial "T". The previously described algorithm was used to determine the step arrival time at each channel. A histogram of the resulting time differences is shown in figure-7. The standard deviation for this measurement was 16.1ps. Since this measurement is based on two independent timing measurements, one on each channel, the standard deviation for each channel is $1/\sqrt{2}$ of this value, or 11.4ps. This is the same standard deviation expected from an ideal quantized time measurement with 40ps resolution. It would require a 25GHZ clock and counter to achieve the same results using the traditional approach.

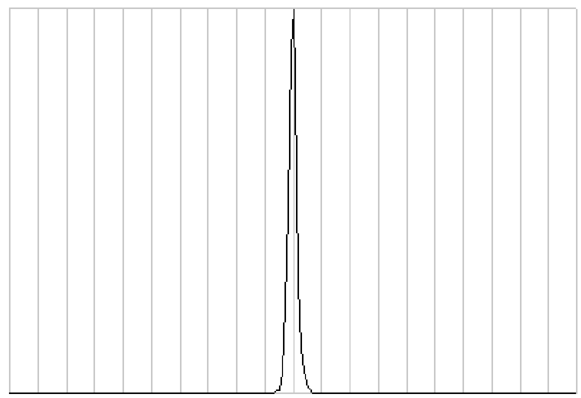

Figure-7, Two channel relative arrival time distribution, 100ps/div

There are several key points to remember about using the E1437A for precision timing measurements.

- Timing resolution is NOT limited to the ADC sampling period.
- Accurate interpolation requires an analog alias filter and Nyquist sampling.
- The 11.4ps RMS event timing resolution is based on SINGLE-SHOT measurements without averaging.
- The excellent SNR and linearity of the E1437A are important for accurate timing results.