Tektronix

Application Note

11800/CSA803 Using Filtering to Control Risetime

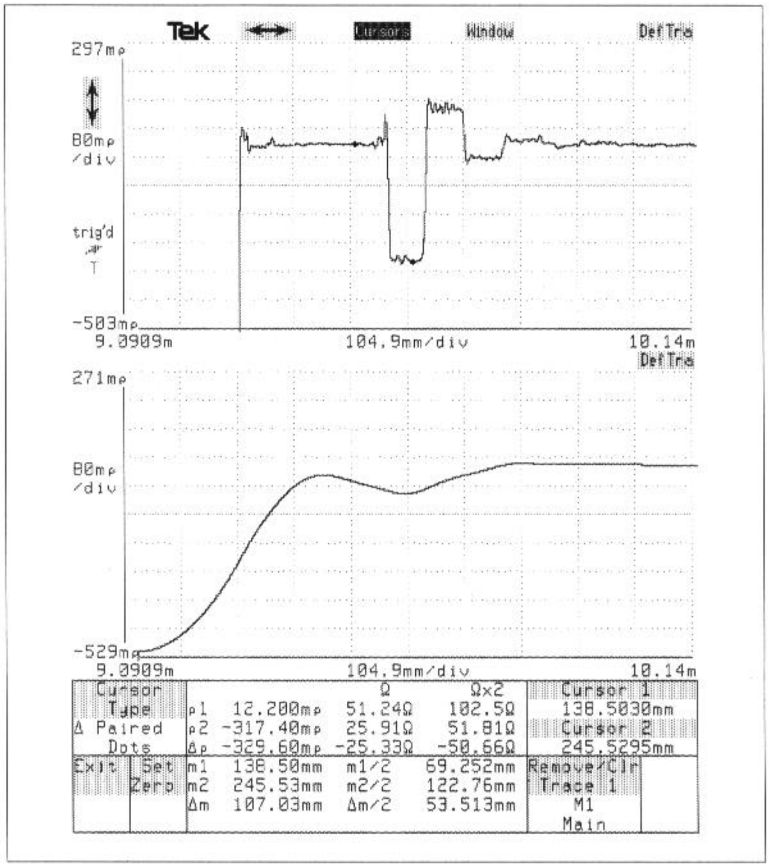


Figure 1. TDR trace of a 50 Ω transmission line with a one inch unterminated stub.

In most TDR and device characterization applications, more is better when it comes to acquisition bandwidth and TDR step speed. However, there are some cases where the "more is better" rule doesn't apply. For example, measuring impedance variations of circuit board runs in an ECL system doesn't require the high-speed stimulus of the 11800 family of digital sampling oscilloscopes. In fact, the 35-picosecond step from the 11800 with an SD-24 sampling head will often show much more information than you really need.

Typical ECL outputs deliver edges with rise times of between a few hundred picoseconds and one nanosecond. With one nanosecond rise time signals, many small discontinuities can be ignored. For example, Figure 1 shows the effect of a 1-inch unterminated stub on a 50-ohm transmission line. This stub might be a short unterminated section of circuit board run to an ECL gate, for example. The upper trace shows the TDR display for such a transmission line, driven with the 35-picosecond edge from the 11800/SD-24. The bottom trace shows the same transmission line driven with a one-nanosecond edge, more typical of ECL rise times. In actual operation, an ECL gate driving this transmission line would see the reflections of the lower trace, not the much larger reflections shown in the upper trace.

Until recently, however, it was difficult to see the response of a transmission line to actual stimulus unless you were fortunate enough to have a pulse or step generator of the appropriate rise time or appropriate filters to slow down the generator or acquisition.

With the advent of modern digital oscilloscopes and powerful on-board real-time signal processing, this type of measurement has become much easier. Both the 11801A Digital Sampling Oscilloscope and the CSA803 Communications Signal Analyzer from Tektronix offer a powerful real-time "Filter" function that can quickly mathematically compute the response of a system with a user-selectable rise time.

Setting up the Filtered Trace. Setting up a filtered trace on the 11801A or CSA803 is simple. Just connect the device under test or signal to be acquired to a sampling head input and use the Define Trace **(DefTra)** function to create a filtered trace. Use the following steps as a guide:

1. Touch the **DefTra** icon in the upper left corner of the 11800 or CSA display. A full-screen popup menu will appear.
2. Select the Filter function from the center section of the popup menu.
3. The first parameter for the filter function is a channel, so select the channel you want to display from the channel selection part of the popup in the upper left corner. Notice that the expression you are building shows up on the top line of the popup menu.

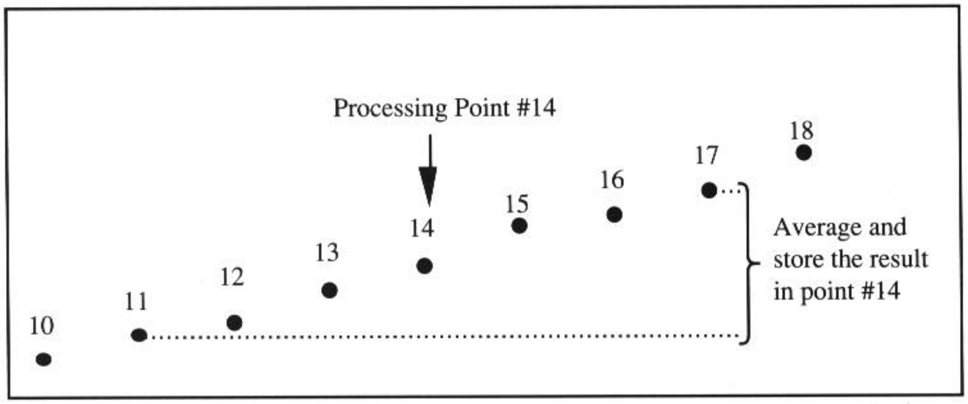


Figure 2. The moving average or "smoothing" algorithm takes a group of points surrounding the point to be processed, averages the group of points, and stores the result in the waveform point. This process is repeated for all points in the waveform.

4. Enter a comma (near the bottom of the menu) to separate the channel number from the rise time parameter.
5. Next, enter a rise time on the numeric keypad in the upper-right corner of the menu. The rise time is in seconds, so if you want to enter nanoseconds, for example, you need to use the **EEX** (Enter Exponent), the minus sign and "9" in standard scientific notation
6. Finally, touch the closing parenthesis ")" to close the expression and **Enter Desc** (Enter Description) to create the new waveform.

A new trace will appear on the display and it will show the results of filtering the input channel with a filter of the rise time you specified. The trace is continuously recomputed and updated so any changes in your device under test or stimulus signal will show up immediately.

The filtered trace is autoscaled to fill the display, so the scale factor may be different than a non-filtered trace on the same channel. You can change the vertical scale factor using the vertical icon and knobs.

Editing the Filter Rise Time. You can easily modify the filter function's rise time parameter by editing the trace description. To edit a description use the following steps:

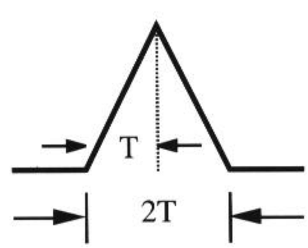
1. If you have more than one trace on the screen, make sure the trace whose description you want to edit is the "selected" trace. The selected trace is easy to identify because the graticule outline and readouts are the same color as the selected trace. To select a trace, simply touch it on the display.
2. Once you've selected the Filter trace, touch the **Vertical Desc** (Vertical Description) selector in the Waveform major menu. Touching this selector brings up the same popup you saw when you created the Filter trace, but instead of creating a new trace, it allows you to edit the description of the selected trace. Notice that the Filter trace expression appears at the top of the popup.
3. Use the Backspace selector to erase the closing parenthesis and current rise time parameter. Then enter the new value for rise time.
4. Enter the close parenthesis and touch **Enter Desc** as you did when you created the trace.When the popup disappears, you'll see the filtered trace with the new rise time value you entered.

Filter Applications. The filter function is useful in a variety of applications. The primary application is in TDR measurements, where it is common to measure impedance variations and reflection coefficients for lower speed systems than the 11800's 35-picosecond rise time. However filtering can be applied to any acquired waveform. For example, crosstalk measurements can be made at any desired rise time, since crosstalk is heavily influenced by the rate of change of the driving signals.

Understanding How it Works. The filter function is implemented using a moving average (sometimes called a "box-car" or "smoothing") algorithm. The moving average takes a group of waveform points centered around each point to be filtered and averages them together, placing the averaged results in that point (See Figure 2). The number of points included in the averaged group is determined by the filter's rise time. Slower rise times average together larger groups of points, while faster rise times average fewer points. This process is equivalent to convolving the unfiltered waveform with a square pulse whose width is equal to the number of points in the average.

The 11800 filter algorithm runs two successive moving average passes over the waveform data. This is equivalent to convolving the signal with a triangle-shaped pulse whose width is proportional to the rise time. This filter provides very good results over a wide variety of rise times.

When you enter a filter expression, the instrument automatically converts the rise time you enter to a number of samples that will be included in the two successive moving averages. The relationship between the filter's rise time and the number of samples in the moving averages is based on the filter's impulse and step response. The filter's impulse response is a triangle pulse. Integrating the impulse response gives the step response and the time from the 10% to the 90% point of the step response is the filter's rise time.



For the sake of this discussion, assume that the filter's impulse response (the triangle pulse) is 2T wide at it's base (where T = the smooth filter's length in seconds). The height of the triangle pulse is 1/T. At time T, the impulse response has a magnitude of 1/T (the peak of the triangle). For simplicity, let's consider only the first half of the impulse reponse (up to time T). For any time t up to the peak of the triangle, the impulse response is given by:

$$f(t)=(1/T^2)t$$

Integrating this function over the interval from 0 to t yields:

$$h(t)=(1/T^2)\int \tau \ d\tau = t^2/2T^2$$

Since the impulse response is symmetrical, we can find the 10% to 90% rise time by finding the 10%to 50% time and doubling it. We already know that the time at the 50% point (the peak of the impulse response) is T. By setting the above expression equal to 0.1 (10%), we can find the time at the 10%point:

$$\begin{aligned} t^2/2T^2 &= 0.1 \\ t^2 &= .2T^2 \\ t_{(10\%)} &= .447T \\ t_{(50\%-10\%)} &= T-0.447T \\ &= 0.5528T \\ t_{(90\%-10\%)} &= 2(0.5528T) \\ &= 1.11T \end{aligned}$$

Thus, the width of the moving average (in seconds) is the user-entered rise time divided by 1.11. The instrument converts the width in seconds to a number of points by dividing by the equivalent-time sample interval (i.e. the time between sample points in the waveform record).

For example, a one-nanosecond filter requires a moving average width of:

$$T_{width}=\frac{T_{rise}}{1.11}=\frac{1ns}{1.11}=904.5ps$$

At 500 ps/div with 512 points, the sample interval is 10 ps/sample. Thus, the moving average width in points is:

$$\frac{904.5ps}{10ps/point}=90.45points$$
$$\approx 90points$$

Note that whenever the time/division or record length changes, the sample interval changes, so this last computation must be repeated to change the number of samples included in the moving average. The instrument takes care of this automatically whenever you change the time/division of a filtered waveform.

While the time-domain analysis above gives more precise results, we can also compute the bandwidth (and thus approximate the rise time) of the filter using frequency-domain analysis. The filter can be described in terms of the Fourier transform for the triangular pulse:

$$x(f)=\frac{Sin^2\left(\frac{\omega T}{2}\right)}{\left(\frac{\omega T}{2}\right)^2}$$

the –3dB bandwidth of the filter can be found by setting the above expression equal to the filter's magnitude at the –3dB point $\left(\sqrt{\frac{1}{2}}\right)$

$$\sqrt{\tfrac{1}{2}}=\frac{Sin^2\left(\frac{\omega T}{2}\right)}{\left(\frac{\omega T}{2}\right)^2}$$

A transcendental solution for ω yields: $\omega(-3dB)=\dfrac{2.004}{T}$

Substituting 2 πF for ω gives:

$$Frequency(-3dB)=\frac{.3189}{T}$$

For a gaussian filter, the rise time is related to the bandwidth by the approximation:

$$T_{rise}=\frac{.35}{Freq(-3dB)}$$

Using this approximation, the rise time of the filter is $T_{rise=}1.097T$.

This result is within 1% of the time-domain solution.

Choosing a rise time. In most cases, you can simply enter the rise time you desire to see in the filter expression. However, remember that the actual rise time is a function of both the filter and the rise time of the unfiltered stimulus. The observed rise time is approximately:

$$T_{r(observed)}\approx\sqrt{T_{r(stimulus)}{}^2+T_{r(filter)}{}^2}$$

As long as the filter's rise time is significantly longer than the stimulus rise time, the stimulus rise time has negligible effect on the observed rise time. For example, when you are using the SD-24 TDR/Sampling head, which has a TDR rise time of 35 picoseconds, a 200 picosecond filter produces an output waveform with a rise time of about 203 picoseconds (about 1.5% error). For rise times shorter that about 6 times the stimulus you should include the stimulus rise time in the value you set for the filter rise time. You can calculate the filter required by simply rearranging the above expression to compute the filter rise time given the observed and stimulus rise times:

$$T_{r(filter)}\approx\sqrt{T_{r(observed)}{}^2-T_{r(stimulus)}{}^2}$$

If you're using a stimulus source other than the SD-24, be sure to use it's rise time in this expression.

End-Point issues. The moving average algorithm averages a number of points surrounding each point it processes. This can lead to a problem near the ends of the record where there may not be sufficient points before or after the point being processed to form the normal averaging group. In these cases, the instrument simply repeats the value of the last valid point prior to the end of the record as many time as required to complete the averaging group. As a general rule, it's a good idea to use time/division settings of approximately 1/2 the rise time or slower and keep the events you're interested in within the center 6 divisions of the screen. This avoids errors that can arise due to this end-point effect.

WHEN IS POST FILTERING THE SAME AS SLOWING DOWN THE INCIDENT RISETIME

With the filter function built into the 11800 and CSA, one can apply a fast step to a device, and then mathematically filter the reflected TDR signal in the oscilloscope. In the examples shown in this application note, the post filtering, after reflection, apparently gives the same response the device would have given if the incident, fast, step was directly filtered before the device. In general, this is only true for linear, time invariant systems (LTI). If the system is LTI, then one can express the output waveform in terms of the input step as:

$$out(t) = step(t) * dut(t) * filter(t)$$

In the above expression, step(t) is the incident step waveform, dut(t) is the impulse response of the device under test and the filter(t) is the impulse response of the filter function. The * corresponds to convolution in the time domain. Most people are not familiar with convolutions, so it is easier to transfer the above equation to the frequency domain, where convolutions become multiplications: $Out(f) = Step(f) \bullet Dut(f) \bullet Filter(f)$

In the above expression, each of the terms are Fourier transforms of the terms in the first equation. Dut(f) and Filter(f) are commonly called the transfer functions for the device and filter respectively. Since multiplication is commutative, the above equation can be reordered as:

$$\begin{aligned} Out(f) &= Step(f) \bullet Dut(f) \bullet Filter(f) \\ &= Step(f) \bullet Filter(f) \bullet Dut(f) \\ &= [Step(f) \bullet Filter(f)] \bullet Dut(f) \\ &= [SlowerStep(f)] \bullet Dut(f) \end{aligned}$$

The commutative property of multiplication and the above reordering prove that post filtering after measurement is the same as filtering the incident step (for LTI systems).

What types of systems are linear, time invariant systems? Many systems meet this qualification, basically anything you can conceive of building with inductors, capacitors, resistors and transmission lines is a LTI system. You can even add dependent voltage and current sources as long as they are linearly dependent on the input current or voltage (don't make a voltage source dependent on the square of the input voltage). If you do not know what the SPICE equivalent circuit is for your system, but you know the frequency domain transfer function, then the system is LTI. The TDR capability of the 11800/CSA oscilloscope is often used for measuring IC packages and interconnects. In almost every case, these can be considered LTI devices. This is true even if the package contains lossy and/or dispersive transmission lines.

What types of systems are not LTI? Active devices used in large signal mode are often nonlinear. For example, the collector current, IC, in a bipolar transistor is related to the base-emitter voltage, Vbe, through $I_c = I_{ss} e^{qV_{BE}/kT}$ the exponential in this equation is nonlinear, so post filtering, in this case, would not be equivalent to slowing down the incident risetime. Note however that if the applied signal were small enough, the exponential could be approximated by a linear section and a small signal linear model could then be developed at each bias point. In rare cases, IC packages and interconnects could be nonlinear, this might occur for an interconnect used in commercial power distribution where 10,000 volts could be applied to the interconnect. With this large of voltage, there could be some nonlinear effects in the dielectric properties of the material itself which makes up the interconnect. In this, somewhat extreme case, an interconnect which would be considered LTI for computer applications would not be LTI for high voltage applications.

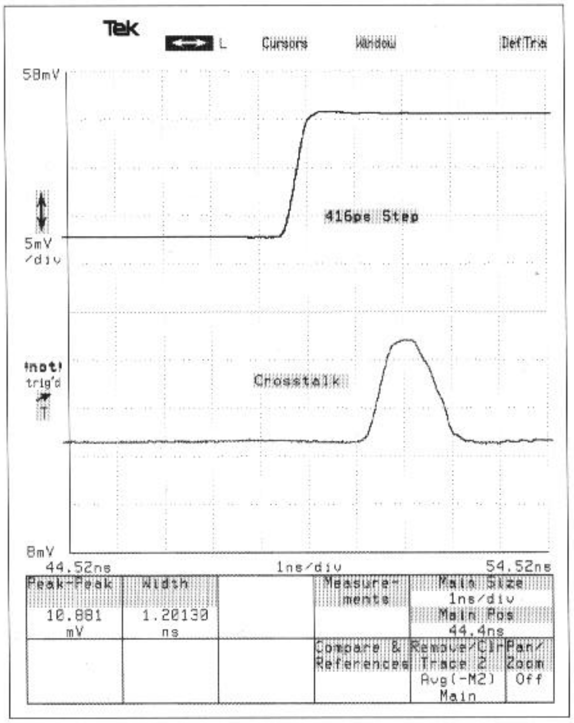


Figure 4. Crosstalk measurement made with a 479 ps pulse generator.

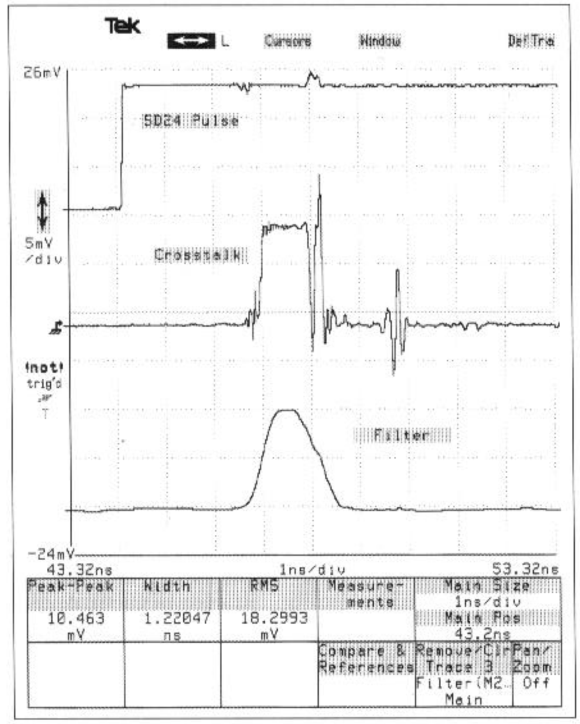


Figure 5. Crosstalk measurement using an SD-24 step generator and a 479 ps filtered trace.

Some Example Results. Figures 4 and 5 show two measurements of crosstalk taken with an 11801A Digital Sampling Oscilloscope. The top trace in each figure is the stimulus signal driving a trace on a circuit board. The bottom trace in each figure shows the crosstalk induced on an adjacent trace. Figure 4 shows the measurement made without filtering, using a pulse source whose rise time is measured to be 479 picoseconds. Figure 5 shows the same measurement using the SD-24's internal TDR generator. The top trace shows the unfiltered stimulus signal, while the bottom trace shows a trace created with the filter function set for a rise time of 479 picoseconds. All traces were acquired using the SD-24 for acquisition.

	Actual	Filter
Peak to Peak	10.88 mV	10.46 mV
Width	1.20 ns	1.22 ns

Table 1. Comparison of crosstalk.

Table 1 shows the results of a peak-to-peak crosstalk measurement and a width measurement on the crosstalk-induced pulse. The results are virtually identical.

Close examination of the crosstalk waveforms shows minor differences in aberrations on the waveforms, but these differences are due to imperfections in the slow pulse generator's output.

Filter and Phase. One difference between the filtered trace response and the response through an actual low-pass filter should be kept in mind. A low-pass filter injects a phase shift (i.e. delay) as well as slowing down the rise time of the signal. The filter function does not introduce this phase shift. For symmetrical waveforms, the filtered and unfiltered responses have the same delay at the 50% point.

This phase difference is not a problem in most applications if the same filter is also applied to the time reference waveform. This is automatically the case for TDR measurements, since the stimulus and response measurements are made on the same waveform.

For further information, contact:

U.S.A., Africa, Asia, Australia, Central & South America, Japan
Tektronix, Inc.
P.O. Box 500
Beaverton, Oregon 97077-0001
For additional literature, or the address and phone number of the Tektronix Sales Office or Representative nearest you, contact: (800) 426-2200

Belgium: Brussels
Phone: 32(2) 725 96 10
FAX: 32(2) 725 99 53

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FAX: (705) 737-5588

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Phone: 45 44 53 54 55
FAX: 45 44 53 07 55

Eastern Europe, Middle East, and Austria
Tektronix Ges.m.b.H.
Doerenkampgasse 7
A-1100 Vienna, Austria
Phone: 43(222) 68-66-02-0
FAX: 43(222) 68-66-00

Finland: Helsinki
Phone: 358(0) 7282 400
FAX: 358(0) 7520033

France and North Africa
Tektronix S.A.
ZAC Courtaboeuf, 4 Av du Canada
B.P.13
91941 Les Ulis Cedex, France
Phone: 33(1) 69 86 81 81
FAX: 33(1) 69 07 09 37

Germany: Koeln
Phone: 49 (221) 96969-0
FAX: 49 (221) 96969-362

Italy: Milan
Phone: 39(2) 84441
FAX: 39(2) 8950-0665

Japan: Tokyo
Phone: 81(3) 3448-4611
FAX: 81(3) 3444-0318

The Netherlands: Hoofddorp
Phone: 31(2503) 13300
FAX: 31(2503) 37271

Norway: Oslo
Phone: 47(2) 165050
FAX: 47(2) 165052

Spain: Madrid
Phone: 34 (1) 404.1011
FAX: 34 (1) 404.0997

Sweden: Stockholm
Phone: 46(8) 29 21 10
FAX: 46(8) 98 61 05

Switzerland: Zug
Phone: 41(42) 219192
FAX: 41(42) 217784

U.K.: Marlow
Phone: 44 (0628) 486000
FAX: 44 (0628) 47 4799

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