DSP in High Performance Oscilloscopes

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Abstract: The digital signal processing revolution began around 1980 as the first dedicated digital signal processors began to appear. Since that time DSP has been incorporated into just about every aspect of modern electronics. Oscilloscopes are no exception. Digital storage oscilloscopes have almost totally replaced analog scopes. They use DSP in acquisition modes, averaging, interpolation, display, bandwidth enhancement filters, optical reference receivers, mask testing, measurements, waveform math, FFT spectral analysis, and other applications. This paper discusses various attributes of bandwidth enhancement, optical reference receivers, and interpolation.

1.0 Introduction

Over the last 25 years digital signal processing has found its way into virtually every field of the electronics industry. The beginning of that trend started around the 1980 time frame as the first really general purpose DSP microprocessors became widely available. Since that time the three largest oscilloscope manufacturers have completely discontinued analog scope designs and only offer DSOs, digital storage oscilloscopes. However, the front-end pre-amplifiers of the DSOs, are by definition analog up to the point of the A/D converters. The bandwidth has traditionally been specified as the analog bandwidth of the pre-amplifier and A/D converter. But as technology improves digital filter algorithms have been applied to extend and shape the magnitude and phase response of the scope channel.

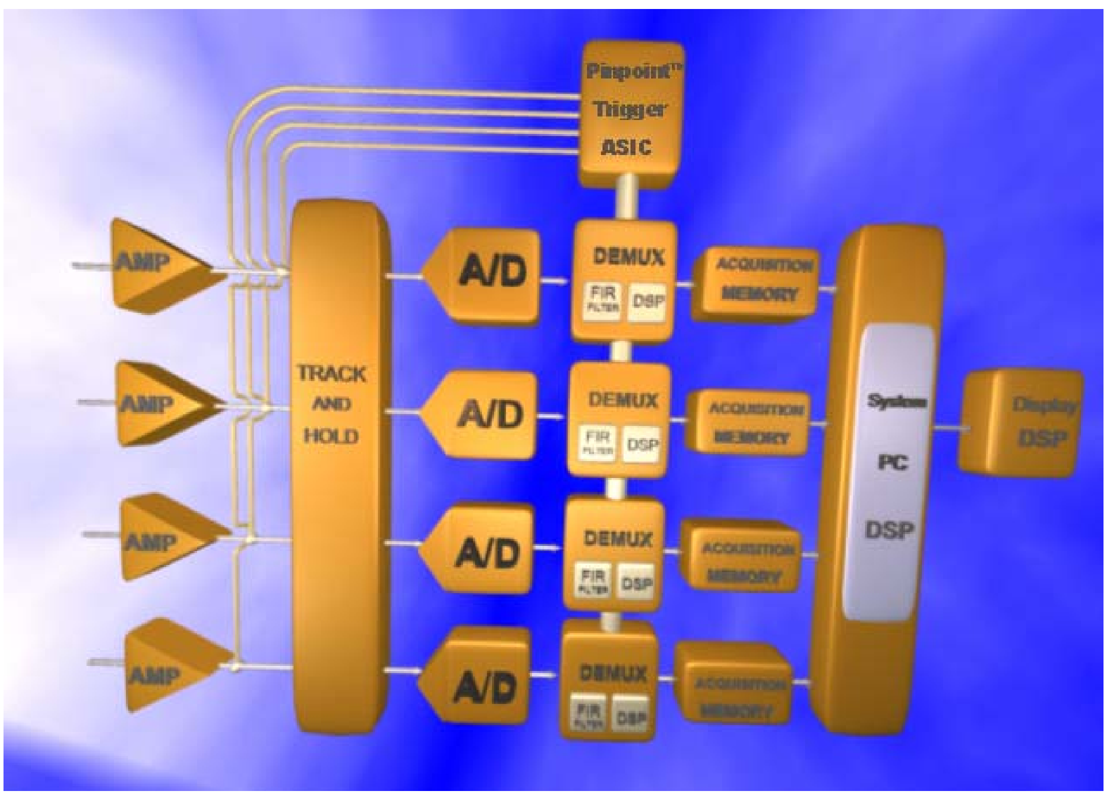

Figure 1. Block diagram of a DSO with four channels illustrates analog and DSP blocks.

Reasons for using DSP: Once the signal is acquired the digital processing does not depend on the effects of HW component tolerance variations, temperature drift, and aging that can be associated with analog circuits. In addition it is possible to implement more complex processing algorithms in the digital domain then in the analog domain. FFTs, and arbitrary filters with large numbers of poles and zeros are specific examples. Arbitrary filters are used for bandwidth enhancement and optical reference receivers. Additional DSP algorithms incorporated in DSOs, are interpolation, averaging, measurements, and waveform math functions, ET acquisition, and display processing, and more.

As will be shown in this paper, if the combination of the analog and digital processing are designed well then the end result is a more ideal acquisition channel.

2.0 Basic Digital Storage Oscilloscope Architecture

A brief description of the DSO architecture is presented here to provide context for the environment in which the DSP algorithms exist. Refer to Figure 1.

The channel input of a typical digital storage oscilloscope has an analog amplifier and attenuator that feed into a track and hold IC. The track and hold receives the analog output from the amplifier on each channel of the scope. Its purpose is to operate as an analog switch to rout the signals to A/D converters in each channel. To obtain four-channel operation each amplifier is connected to the A/D converter for that channel. To obtain interleave operation to increase the sample rate a single channel might be connected to all four A/D converters with their sample clocks skewed. Each A/D always operates at its maximum sample rate and is 8-bits wide in all Tektronix high performance real time oscilloscopes.

The output of the A/D converter feeds into an IC called the demux. This IC is very complex and can perform many different functions. However, its' primary job is to take in samples at the rate of the A/D and write them to acquisition memory at a slower rate. Thus it typically writes 16 or 32 samples at a time to memory. The demux also contains various DSP processing machines implemented in hardware and it contains a dedicated DSP chip. Typical functions provided by this hardware are bandwidth enhancement filtering, HIRES acquisition mode filter, envelope mode acquisition, and trigger position calculations.

The acquisition memory is a different block then the main system processor memory. The demux has circular addressing and control logic and receives triggers from the trigger system. The data stream is continuously written in a circular fashion overwriting past data until a trigger event occurs. After the desired amount of post trigger data is captured the demux ceases to write to memory.

Once the acquisition is acquired the main system processors may then perform DSP operations on the data. Also, data may be passed back through the demux IC to perform additional DSP operations.

3.0 Bandwidth Enhancement

A DSP arbitrary equalization filter can be used to improve the oscilloscope channel response. This filter extends the bandwidth, flattens the scope channel frequency response, improves phase linearity, and provides a better match between channels. It also decreases risetime and improves the time domain step response. The filter is calibrated at the SMA TekConnect® input during manufacturing for each channel and each attenuator setting of 50 mV/div, 100 mV/div, and 200 mV/div. The filter also operates on all of the other volts/div settings. For settings less then 50 mV/div it uses the calibrated filter for the 50 mV setting. For settings greater then 200 mV/div it uses the filter that was calibrated for 200 mV/div. The filter operates at a base sample rate of 20 GS/s and higher ET and interpolated sample rates. It is applied prior to putting data into an ET, equivalent time acquisition record. Therefore, it is still useful with eye diagram displays. The degree of channel matching in step, phase, and magnitude response with bandwidth enhancement sets a new precedence in the oscilloscope market. It was initially offered on the TDS6804B oscilloscope.

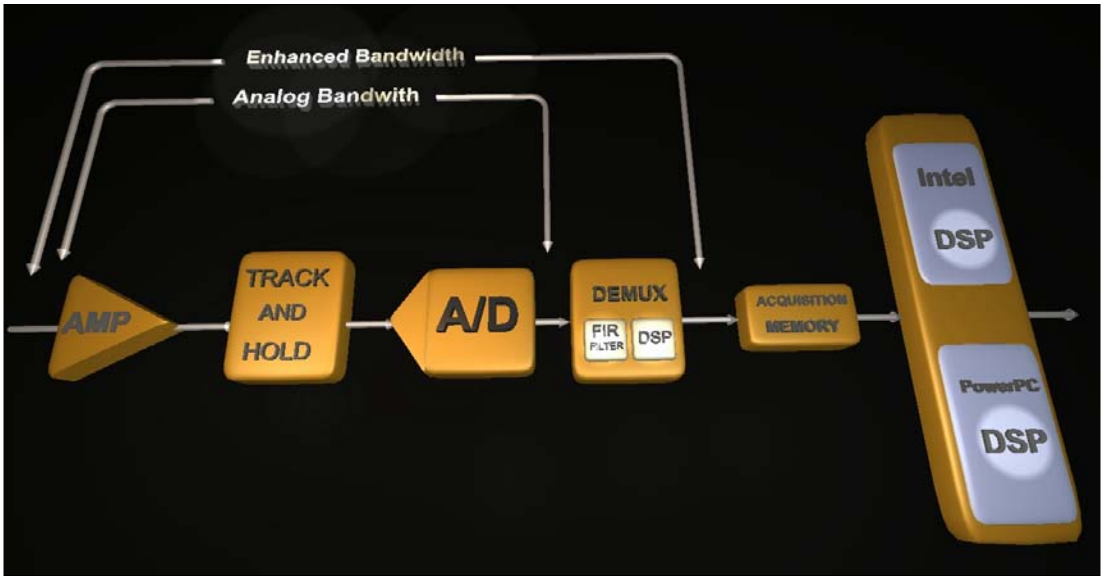

Figure 2. Channel architecture indicates difference between analog bandwidth and DSP enhanced bandwidth.

3.1 Applications for Bandwidth Enhancement Filter

Tektronix allows the user to control whether bandwidth enhancement is enabled or not. The bandwidth enhancement filter, $^{B}W+$, provides better measurement accuracy for many situations. The following list describes areas where the filter is useful:

- **Rise time measurement:** The $^{B}W+$ filter gives the scope channel a faster rise time resulting in more accurate rise time measurement. Figure 3 illustrates the accuracy achieved with the $^{B}W+$ filter in the TDS6804B. This was demonstrated using 75 ps and 100 ps hardware filters as DUTs applied to the output of a 15 ps source and then feed into Channel 1 of the instrument. The TDS6804B reports a more accurate risetime measurement with $^{B}W+$ ON than with the $^{B}W+$ filter turned OFF.

- **Comparison of signals in multiple channels:** More accurate multi channel signal comparison can be made with $^{B}W+$ turned on. This is due to the fact that each channel is specifically calibrated with it's own filter coefficients during manufacturing resulting in a closer match of phase and magnitude response between channels.

- **Measurements such as undershoot or overshoot:** The user probably should leave bandwidth enhance turned on. If the signal being measured has a rise time much greater then 50 ps then good accuracy is obtained for the overshoot measurement. As the input signal rise time approaches 50 ps then the scope really does not have adequate bandwidth to perform the overshoot measurement because the pre and post ringing due to Gibbs' phenomena. i.e. At this point the scope does not have enough bandwidth to accurately observe all the harmonics of the input signal. This is true whether or not a DSP filter is used.

- **Spectral Analysis:** More accurate frequency domain measurements can be made with $^{B}W+$ turned on. This is true for both phase and magnitude.

- **Eye Diagrams:** The $^{B}W+$ filter results in cleaner eye diagrams because it removes some noise and aliasing and reduces the amount of overshoot. Figure 4 shows the affect of applying the bandwidth enhance filter to a 5Gb/s eye diagram. In this case, RT-Eye™ Serial Data Compliance and Analysis software is used to recover the clock and display the eye diagrams.

- **The $^{B}W+$ filter only operates when the base sample rate is 20GS/s.** For a TDS6804B it can operate on all four channels at 20GS/s. It also operates in interpolated or equivalent time sampling up to 2 TS/s. For these modes the filter is applied at the 20GS/s base rate.

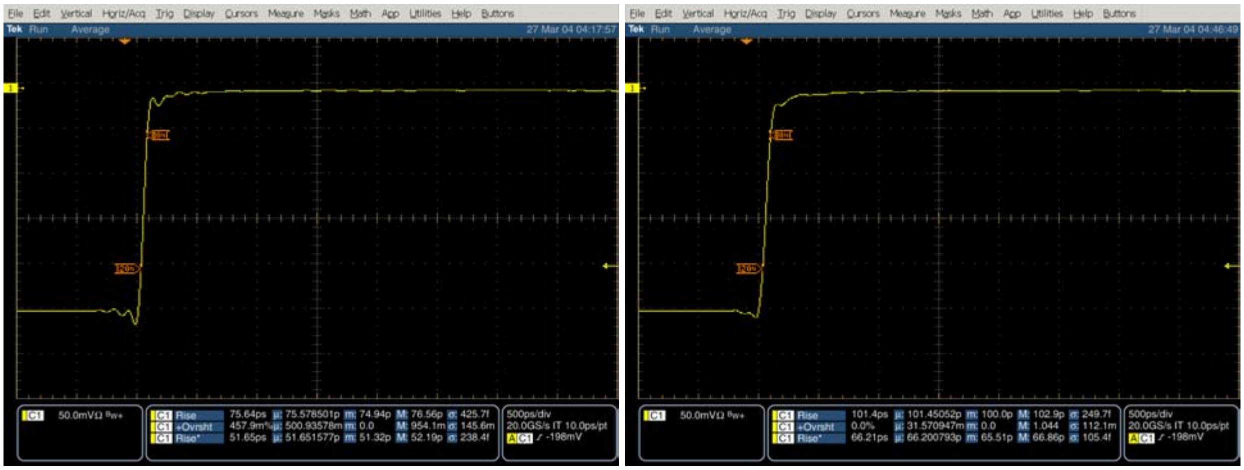

Figure 3. Step response of TDS6804B illustrates accuracy of risetime measurements. The display at left shows 76.64 ps risetime measurement from the 75ps hw filter DUT. The display on the right shows a 101.4 ps ristime measurement on the 100 ps external hw filter DUT. Use of ET mode rather then IT mode would have further reduced error in this measurement.

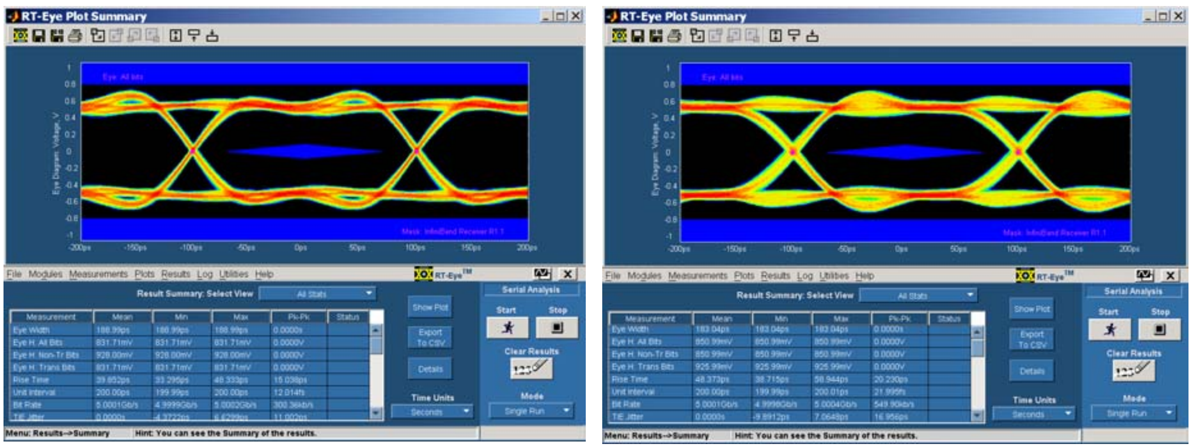

Figure 4. Comparison of 5 Gbit/s eye diagrams. The display at the left is with $^{B}{}_{W}$+ OFF and at the right is with $^{B}{}_{W}$+ ON.

There are some cases when the customer may choose to turn off the bandwidth enhancement filter, these include the following:

- **Impedance Mismatch between device under test and channel input.** If the source impedance of the device under test does not match the nominal 50Ω input of the step generator used to calibrate the filter, then the accuracy of the $^{B}{}_{W}$+ enhanced filter is degraded. The $^{B}{}_{W}$+ enhancement filter is calibrated with a nominal 50Ω source step generator in the manufacturing process. For 50Ω transmission lines such as copper serial data transmission, this provides an optimal solution. However, for environments that are not 50Ω, the $^{B}{}_{W}$+ enhance filter should be used with the impedance mismatch in mind. This is true whether or not DSP is used. The reference plane is at the TekConnect® SMA input connector on the oscilloscope front panel.

- **User may do their own DSP Filters or Interpolation.** The user may want to apply their own DSP algorithms to filter or interpolate the acquired data. In this case, they may prefer the non-processed sample points from A/D.

3.2 Bandwidth Enhancement Filter Characteristics

Oscilloscope channels normally have a passband magnitude response that is flat to within 3 dB. There are variations in the flatness of the frequency response over the passband. Likewise the response of different channels can have some variation with respect to each other. This can translate to 2% amplitude error for deviation of 0.175 dB, 10% amplitude error for deviation of 0.915 dB, 20% error for deviation of 1.938 dB, and 30% error for a deviation of 3.098 dB.

The time domain step response for TDS6804B is shown below in Figure 5. The $^{B}{}_{W}$+ filter reduces overshoot and decreases the risetime. It also adds some pre-shoot that is the direct effect of Gibbs phenomena. That means there is not enough bandwidth to see all of the harmonics of this specific signal. i.e. It is there as a result of correcting a non-linear phase response to be closer to the ideal linear phase response and because of insufficient bandwidth for this signal. However, refer back to Figure 3 to see that accurate risetime measurements can be made.

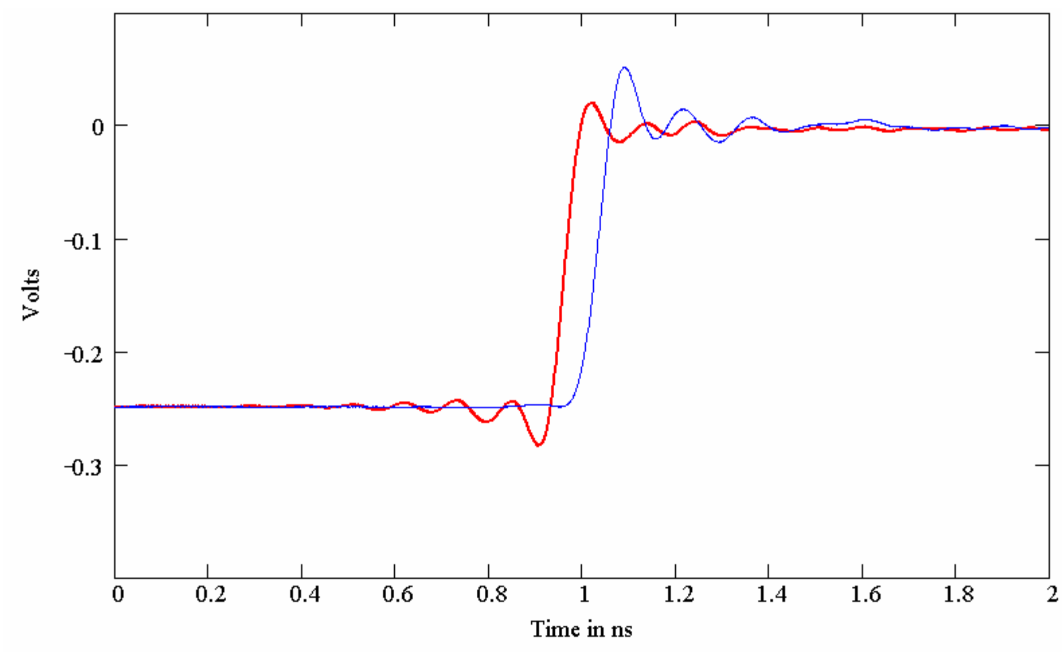

Figure 5. Comparison of step response with $^{B}{}_{W}$+ OFF in blue and $^{B}{}_{W}$+ ON in red. For this example the risetime using ET acquisition mode is approximately 46.8 ps measured using 10% to 90% levels on a 15.5 ps step generator.

3.3 Match Between Channels in the Time Domain

Figure 6 below illustrates that a high degree of channel matching is obtained using the bandwidth enhance, $^{B}{}_{W}$+, filters.

Figure 6. All four channels overlaid with $^{B}{}_{W}$+ turned on and deskew adjustment made to time align. Risetime readout using 10% to 90% is 46.86 ps using a 15.5 ps step generator. An average of 500 was used.

The $^{B}{}_{W}$+ step response of each channel was adjusted for time deskew and saved into a REF memory. Figure 6 shows all four channels stored in REFs 1 through 4 overlaid on top of each other.

An expanded view of the step response of each of the four channels overlaid is shown in Figure 7 below. This is with $^{B}{}_{W}$+ turned OFF. The degree of channel match improvement with $^{B}{}_{W}$+ turned ON is then shown in Figure 8. Both figures have the same vertical and horizontal scale.

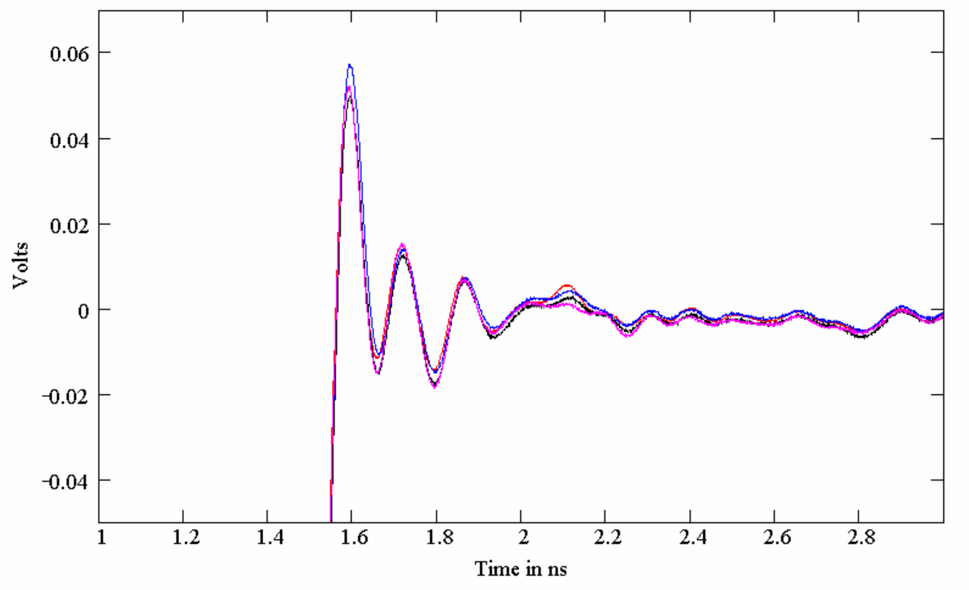

Figure 7. Time domain view of all four channels step response with $^{B}{}_{W}$+ OFF.

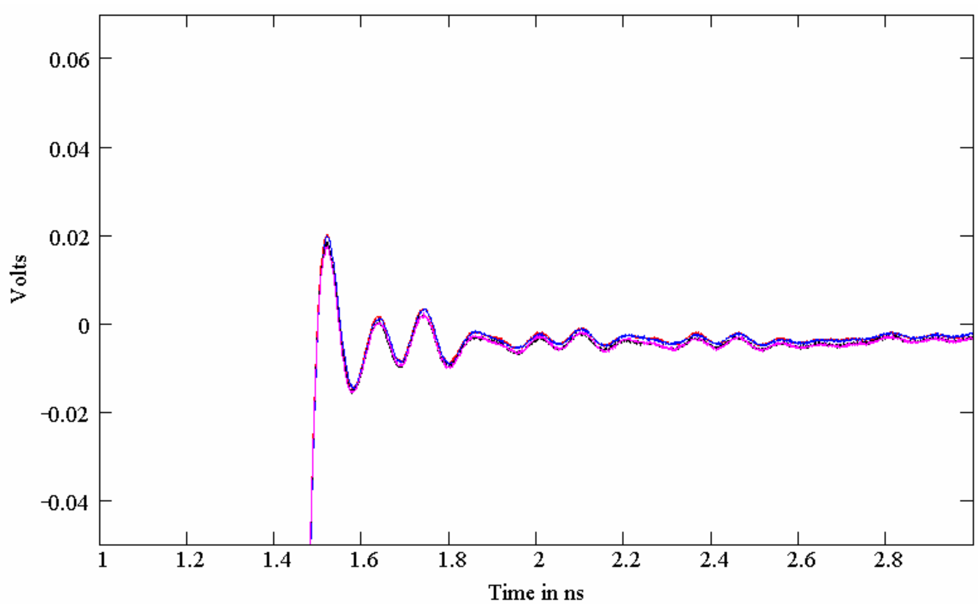

Figure 8. Time domain view of all four channels with $^{B}{}_{W}$+ turned ON.

3.4 Match Between Channels in the Frequency Domain

The frequency domain plot in Figure 9 shows the magnitude response of four channels of the TDS6804B oscilloscope with $^{B}{}_{W}$+ turned OFF. Figure 10 shows the frequency response channel match that is obtained with $^{B}{}_{W}$+ turned ON. The arbitrary FIR filter is able to correct most of the passband to obtain a flatter magnitude and more linear phase response for the channel. These responses were obtained by using a Tektronix 067-1338-00 step generator with 15.5 ps 10-90% risetime. It was connected directly to the TekConnect® SMA input of each channel with no cables. One thousand averages were used.

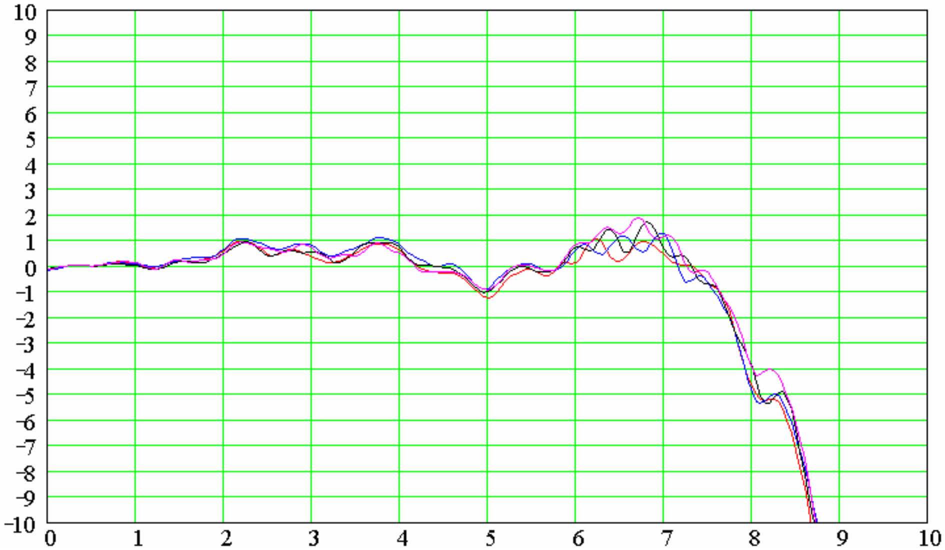

Figure 9. Frequency domain view of all four channels with $^{B}{}_{W}$+ OFF.

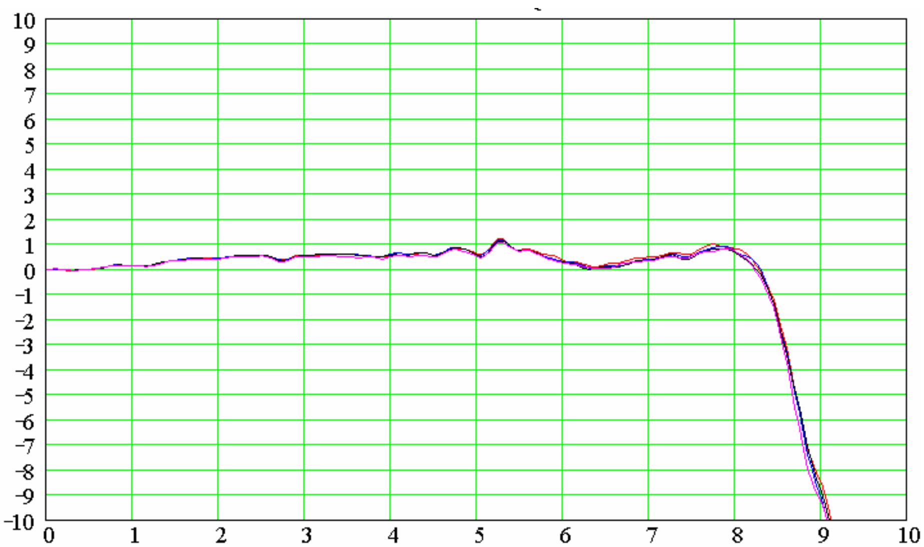

Figure 10. Frequency domain view of all four channels with $^{B}{}_{W}$+ ON illustrating a high degree of CHANNEL MATCH.

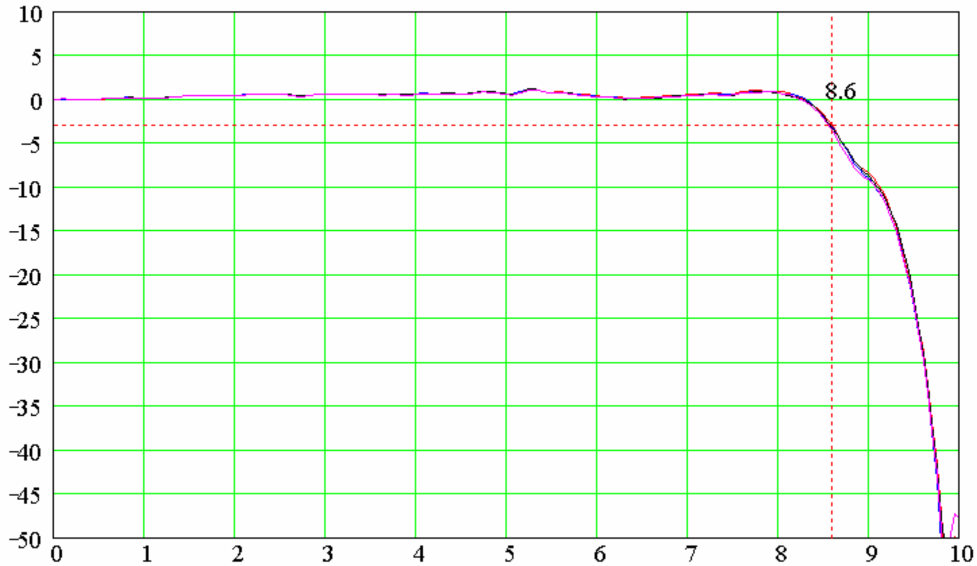

Figure 11. Expanded vertical scale of all four channels with $^{B}{}_{W}$+ On. Note 3dB roll-off at 8.6 GHz.

3.5 The $^{B}{}_{W}$+ Frequency Response Measured with Swept Sine and Power Meter

An example of the frequency response of the TDS6804B when measured with swept sine and power meter is shown below in Figure 12. A two resistor power splitter was connected directly to the TekConnect® SMA input with no cables. The power meter is connected to one side of the splitter and the TekConnect® to the other. The swept sine amplitude was then set to the same power level at each frequency. The resulting response from the scope channel is then measured and graphed below. (This plot was made from an early manufacturing unit showing a cutoff frequency close to 8 GHz. However, succeeding units are actually tuned to cutoff frequency closer to 8.6 GHz even though the published specification is 8 GHz.)

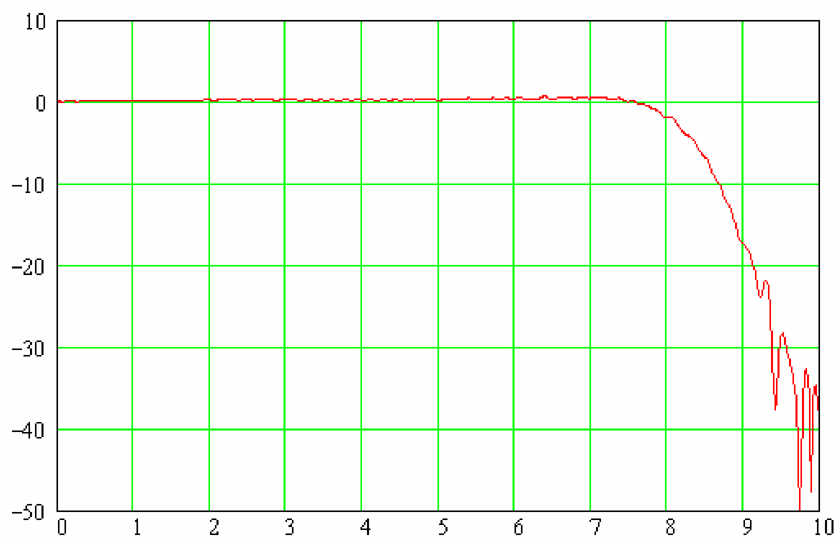

Figure 12. Swept sine and power meter magnitude results of $^{B}{}_{W}$+ when the filter is turned on for a TDS6804B oscilloscope.

3.6 Match between Phase Response Across All Channels

The phase response of all four scope channels of the TDS6804B is shown in Figure 13 with $^{B}_{W}$+ turned off. The response with $^{B}_{W}$+ ON is shown in Figure 14. Note that the enhanced response has improved linearity over the entire passband of the channel to 8.0 GHz. This enhanced linear phase response will result in less pulse distortion through the channel and will enable more accurate spectral analysis measurements involving phase.

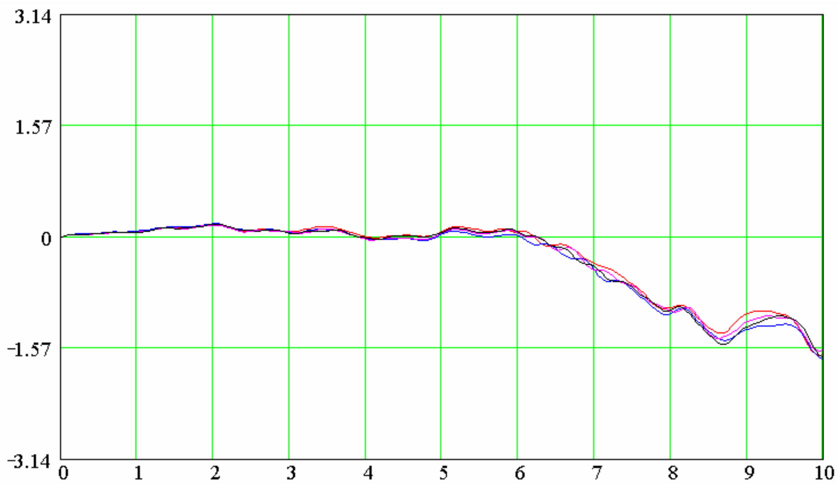

Figure 13. Phase response of all four channels with $^{B}_{W}$+ OFF. The vertical axis is phase in radians and the horizontal axis is frequency in GHz.

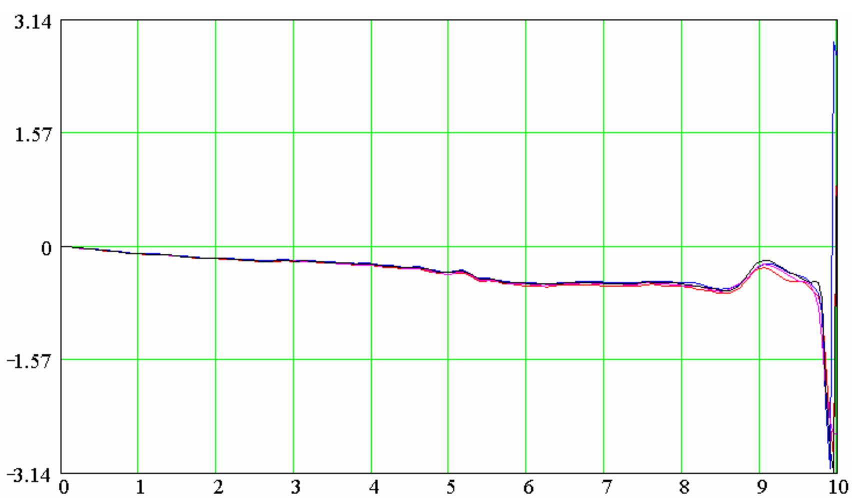

Figure 14. Phase response of all four channels with $^{B}_{W}$+ on indicates high degree of channel match. The vertical axis is phase in radians and the horizontal axis is frequency in GHz.

3.7 Match Between Instruments

Matching is not only achieved from channel to channel, but is also achieved from instrument to instrument. That can be very important if trying to compare test results from different instruments. The TDS6804B was designed to have a 7 GHz analog bandwidth specification. However, as in any design, there is a process distribution in the components used. This distribution is typically a Gaussian distribution with a 6σ minimum of 7 GHz (the warranted specification), 7.5 GHz (mean, or typical specification), and a specified maximum of 8 GHz. While a single instrument is not likely to exhibit this radical of a difference in analog bandwidth, the response of different instruments may. The calibration of every channel to the same 8GHz filter response insures more repeatable signal integrity measurements from instrument to instrument. Thus $^{B}{}_{W}$+ insures smaller deviation in bandwidth, and magnitude and phase response from channel to channel and scope to scope. The calibrated target 3 dB point for bandwidth enhancement filter is ~8.6 GHz while the published specification for it is 8 GHz.

An example of the time domain step response is shown in Figure 5 above. A 15.5 ps risetime step generator was connected through SMA to the scope input channel. (All references to risetime in this paper are measured from the 10% to the 90% levels.) With averaging turned on for 1000 acquisitions the step response is obtained. The step without the filter is blue and the enhanced step is red.

3.8 Gibbs Phenomena

Both pre and post edge ringing is an expected result when high frequency harmonics on a fast edge are removed by a low pass filter. The 8.0 GHz bandwidth of the scope is not wide enough to capture all of the high frequency harmonics for this very fast edge. As a result according to Gibbs phenomena pre and post ringing will occur on the edge if the scope channel response is lowpass and linear phase. One of the primary specifications of the bandwidth enhancement filter was the requirement for linear phase. This means all frequencies pass through the channel with the same amount of time delay. The result of a system with non-linear phase is that some frequencies are shifted in time with respect to others. This will result in pulse shape distortion. As slower rise time edges are input to the scope more of the harmonics will be in the passband and the ringing will decrease. (NOTE: In applications such as optical reference receivers a 4^{th} order Bessel Thompson filter is used. It has a slow roll of rate and has a cutoff frequency specified so that high order harmonics are attenuated from their true value. Thus ringing which would have otherwise showed up with insufficient bandwidth is reduced for the required mask test. For this case the user is not interested in the “actual” signal shape but rather how the actual signal would look when passed through a reference filter channel.)

3.9.0 Bandwidth Enhancement With Interpolated Acquisitions

Interpolated acquisition mode requires the use of a sinx/x interpolation filter that is computed at the same time as the bandwidth enhancement filter. This results in slightly different frequency response for the system as shown below in Figure 15. The red trace is ET mode with $^{B}{}_{W}$+ off. The black trace is IT mode with $^{B}{}_{W}$+ off. Blue trace is ET mode with $^{B}{}_{W}$+ on. Magenta trace is IT mode with $^{B}{}_{W}$+ on. Note that the $^{B}{}_{W}$+ filter rolloff before the 10 GHz Nyquist point is beneficial because it removes some of the aliasing. As can be seen from these graphs the interpolation filter has a minimal influence on the bandwidth enhancement. However, keep in mind that there is still some frequency response energy at around 11 to 12 GHz that is aliased that the filter tends to increase slightly.

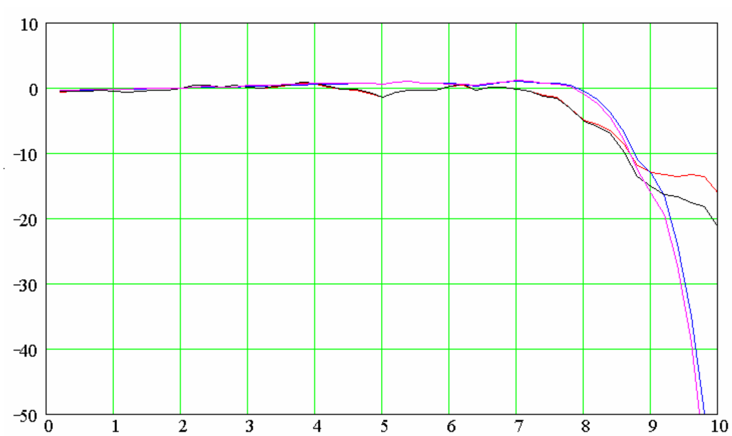

Figure 15. Comparison of bandwidth enhance response in ET versus IT acquisition mode. Vertical axis is magnitude in dB and horizontal axis is Frequency in GHz.

3.9.1 Use of Average Acquisition Mode With BW Enhancement

Average acquisition mode may be used to reduce the noise and the variation in ringing on a step when the BW enhancement filter is in use at its base sample rate of 20GS/s. From one acquisition to the next the samples are in different positions with respect to the analog trigger level crossing point. Thus they are in different positions on the waveform step. The location of these samples affects the amount of ringing when the filter is applied. For example when a sample is centered between high and low on the edge the most symmetrical result occurs with very low ringing. But on other acquisitions as the position of a sample moves higher or lower more ringing will occur. This affect is illustrated in the following 3D surface plot. The X axis is horizontal with Y axis vertical. Steps are shown as voltage with respect to x axis. The Z axis shows successive acquisitions where the samples are located at different positions on the waveform step. As can be seen the ringing is maximum when the sample moves from center close to high or from center close to low.

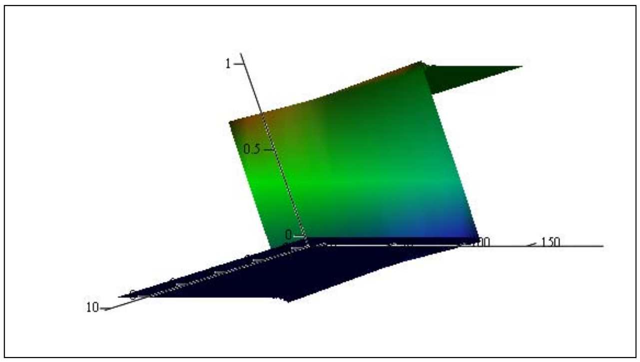

Figure 16. 3D plot of different acquisitions of a step with samples at different position.

Thus with no averaging the user would then see noise on the ringing that varied from almost no ringing as shown in Figure 17 to a maximum ringing as shown at each end of the surface plot shown in Figure 16.

If the waveforms shown in Figure 17 are averaged the result is essentially identical to the response obtained when the samples are centered on the edge. Thus average mode with bandwidth extension filters can remove the ringing effects of the filter. This is when the sample rate is at the base 20 GS/s of the filter. In ET or IT mode this does not occur. The ringing remains in tact.

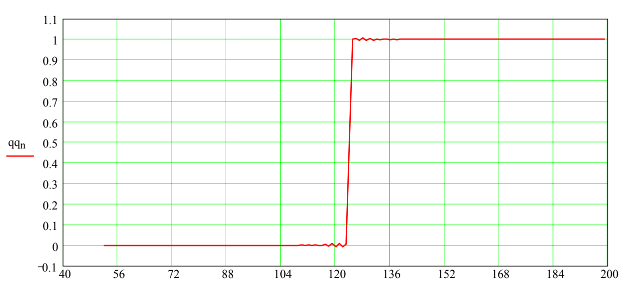

Figure 17. Step response when samples are centered on the rising edge. This is very similar to the response that is seen if acquisitions are averaged.

If averaging is done at the base sample rate it will cause a rolloff in frequency response of -4.01 dB at Nyquist. This is due to the one sample interval uncertainty of where the sample clock is with respect to the analog trigger position on a waveform. However, if higher ET or IT sample rates are used then this effect is reduced. The higher the ET sample rate the less effect average mode has on the bandwidth.

4.0 Description of Tektronix BW Enhancement Filter

The bandwidth enhancement filter is of type FIR, finite impulse response. FIR filters have the advantage over IIR, infinite impulse response filters, in that they are guaranteed to be stable and may have exactly linear phase response. The FIR filter is also more suitable for the implementation of arbitrary equalization of phase and magnitude over almost the entire passband of the channel. This algorithm requires that each scope channel and attenuator setting that supports filtering will be calibrated during manufacturing. A set of FIR filter coefficients is computed based on the measured response of the scope channel at manufacturing for each individual supported attenuator setting and for each individual channel. Thus an unrivaled matching of phase and magnitude response across all four channels is achieved. (NOTE: As mentioned previously, on TDS6804B only three attenuator settings are calibrated for filter response. These are 50 mV/div, 100 mV/div, and 200 mV/div. For settings below 50 mV/div the 50 mV/div filter is used. For settings above 200mV/div the 200 mV/div filter is used.)

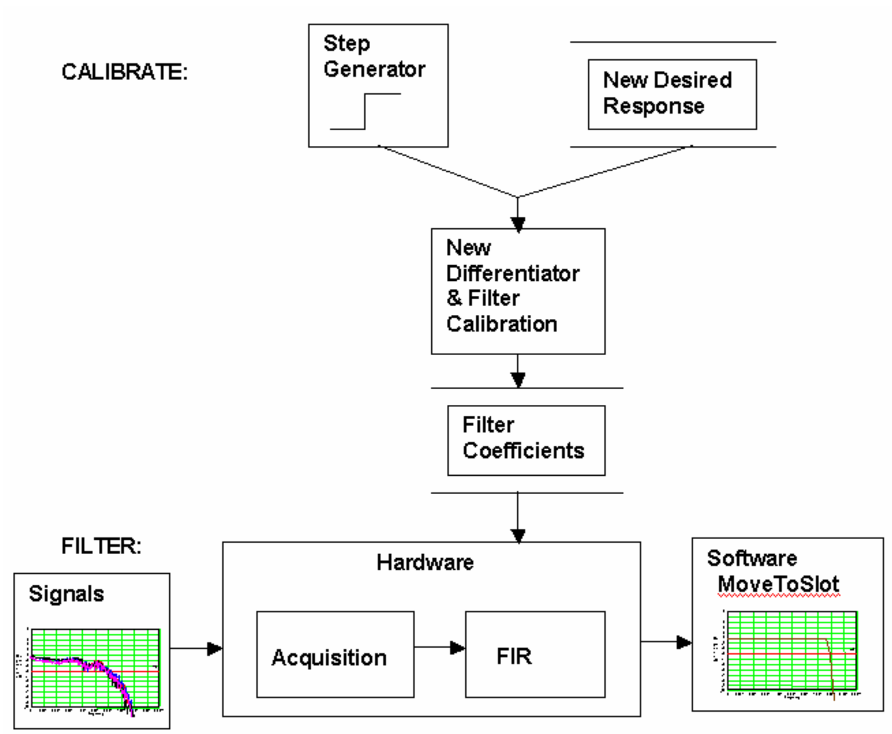

Figure 18. Block diagram showing CALIBRATE that is performed during manufacturing and FILTER that is applied during real time operation of the oscilloscope.

4.1 Description of the Filter Calibration Algorithm

The first stage of implementation of the $^{B}{}_{W}$+ filter was to design the desired response. This would be the frequency and phase response of what the oscilloscope will be once it is filtered. Bessel Thompson filters are commonly used for channel shaping because they most closely approximate a Gaussian filter response. However, it was found that many standard filter types such as this and others did not lend themselves to producing the desired results for a bandwidth enhancement filter. Therefore, Tektronix invented its own filter type using a modified Gaussian function and then assigned it to have exactly linear phase. This filter transfer function was fine tuned for optimal step response. It's step response is stored in the oscilloscope to be used as part of the calibration algorithm performed when the oscilloscope is manufactured. It will be referred to as the "desired response" in the following discussions of the algorithm. A plot of the measured bandwidth enhanced desired response can be seen in Figure 12. When this filter algorithm is used for optical reference receivers the desired response is a 4th order Bessel Thompson filter as specified by SONET and Fiber Channel industry requirements. An example is shown in Figure 20.

The filter coefficients will be convolved with the scope acquisitions during run time to obtain the desired channel response. They will be computed during manufacturing for each supported attenuator and channel setting and stored in the oscilloscope. The filter coefficients are obtained using the block least square algorithm [3]. This method requires as input a vector, d, which contains the impulse response for the arbitrarily specified desired filter and a vector, x, which contains the acquired impulse response of the oscilloscope channel. For bandwidth enhancement the desired filter is linear phase (to avoid pulse shape distortion) using a modified Gaussian function. For optical filters for SONET and Fiber Channel the desired response is a 4^{th} order Bessel Thompson design as specified by industry standards. The calibration process for bandwidth enhancement during manufacturing requires the use of a step generator connected to the input of the SMA input of the oscilloscope channel to obtain the actual response, x. This same algorithm is also used for optical reference receiver filter design but the step generator is replaced with an optical impulse generator connected through the O/E, optical to electrical converter input on a CSA7404 series oscilloscope. Refer to Figure 19 for an example of the input channel and desired impulse responses. NOTE: For $^{B}{}_{W}$+ additional proprietary calibration operations are involved in the filter design process.

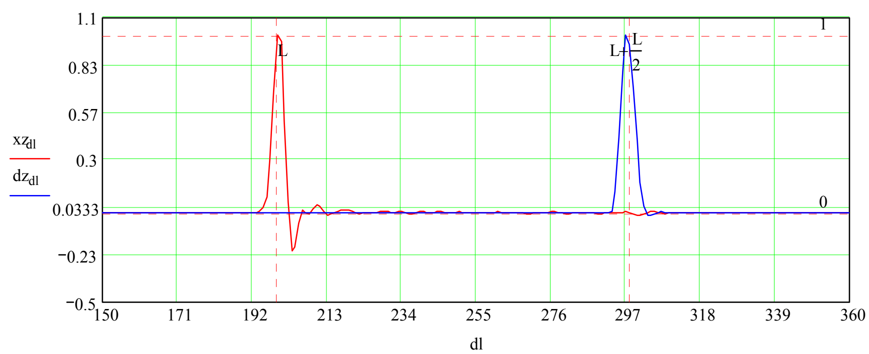

Figure 19. The actual impulse, x, is on the left and the desired impulse, d, is on the right.

Given the actual response, x, the desired response, d, the filter length, L, and the number of points in the autocorrelation, N, then the following equations define the algorithm for computing the filter, W:

k := L … N + L –1
u := 0 … N + L – 1
m := 0 … L –1
n := 0 … L – 1

(1) $$R_{m,n} := \frac{1}{N} \cdot \sum_{k} x_{k-m} \cdot x_{k-n}$$

(2) $$P_{n} := \frac{1}{N} \cdot \sum_{k} d_{k} \cdot x_{k-1}$$

(3) W := R-1 P

The actual response autocorrelation matrix is represented by R in (1) The cross correlation vector between the actual and desired impulse is represented by P in (2). The vector, W, contains the resulting FIR filter coefficients and is computed using equation (3).

6.0 Optical Reference Receivers

The optical reference receivers in Tektronix CSA7404B oscilloscopes use the same filter algorithm for calibration as described in the previous section on bandwidth enhancement filters.

Industry mask testing standards for SONET, Fiber channel, and others require that the measurement channel consisting of an optical to electrical converter abbreviated O/E, and preamplifier, and digitizer exhibit a nominal 4^{th} order Bessel Thompson response [1] [2]. If the magnitude response falls within the standard limits then the measurement channel is referred to as an optical reference receiver. This results in uniform measurements across various test instrumentation setups.

First generation oscilloscope solutions for optical mask testing relied on sampling oscilloscopes with a very high bandwidth of 20 GHz or more for optical mask testing. A sampling oscilloscope collects only one sample for each trigger event. This is still one of the primary choices in the market place for the highest frequency mask standards. The signal path for this solution consists of an optical to electrical converter followed by an analog Bessel Thompson filter followed by the input to the oscilloscope channel followed by the digitizer. This solution requires that the bandwidth of the oscilloscope be much wider than the bandwidth of the analog filter. Thus its primary disadvantage is cost because excessive bandwidth is needed.

An alternative solution is to use a real time oscilloscope to obtain lower cost by using an arbitrary FIR filter that can be calibrated during the manufacturing process. The signal path consists of an optical to electrical converter followed by the input to the oscilloscope channel followed by the digitizer followed by a software FIR filter. The advantage of this approach is that the optical standard bandwidth can be almost equal to the oscilloscope bandwidth. This allows optical mask standards to be implemented at higher frequency and lower cost within a given oscilloscope bandwidth. The filter can easily be calibrated in both magnitude and phase to transform any reasonable oscilloscope response into the desired 4^{th} order Bessel Thompson response with good accuracy. In addition this approach allows all optical mask standards that fit within the oscilloscope bandwidth to be implemented with no additional hardware cost. Additional cost savings result because real time oscilloscopes are useful for a broader range of applications then are sampling oscilloscopes.

6.1 Description of the Desired Filter

The specifications for the 4^{th} order Bessel Thompson filter are published in numerous standards documents [1][2]. The nominal filter response shall have the transfer function:

(4) H(s) = 105 / (s4 + 10s3 + 45s2 + 105s + 105)

The passband tolerance of the filter must be within plus or minus 0.3 dB or 0.5 dB depending on the standard. The tolerance spreads to plus or minus 3 dB at a frequency of 1.5 times the bit rate. These limits are shown in Figure 20. This figure shows a desired filter response that is very close to the roll-off of the combined optical to electrical converter and the oscilloscope channel response. The objective is to design an FIR filter to apply to the O/E oscilloscope response and transform it into the desired response.

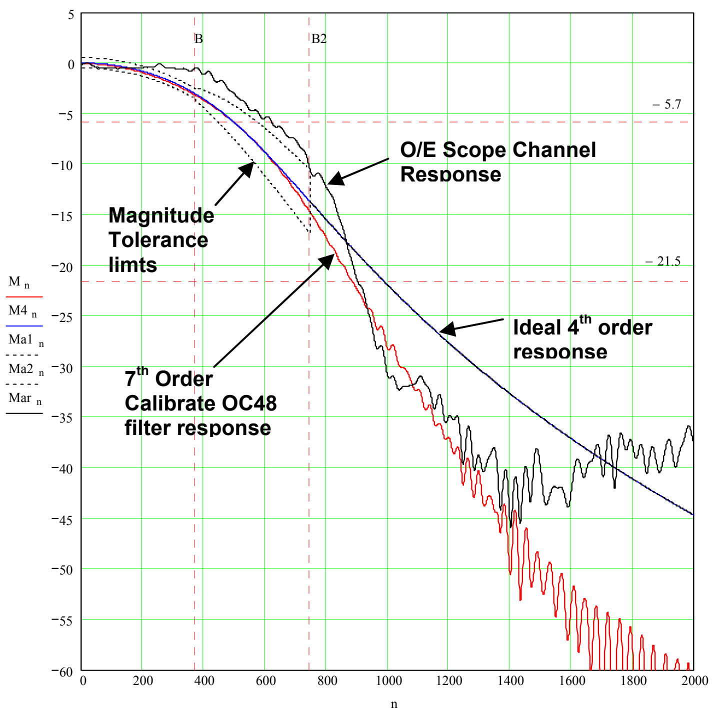

Figure 20. Transforming the O/E oscilloscope channel into the nominal 4th order Bessel Thompson response magnitude limits.

6.2 Application of the Optical Filter in the Oscilloscope

The filter is applied to the acquired waveform data from the O/E converter in a CSA7404B oscilloscope. This is done at the maximum real time sample rate of the oscilloscope. Lower frequency standards are highly over sampled. Therefore, floating point math is used for the filter convolution. This over sampling allows the bandwidth of the oscilloscope channel to operate as an anti-alias filter. From this point the filtered data is placed into a waveform database to create the source for the eye diagram that will appear on screen with the optical mask standard as shown in Figure 3. The data placement is done using an equivalent time sampling mode algorithm that results in, sample rates much higher than the base sample rate. Therefore, the eye diagram requires many triggered acquisitions to accumulate enough samples to obtain good measurement results.

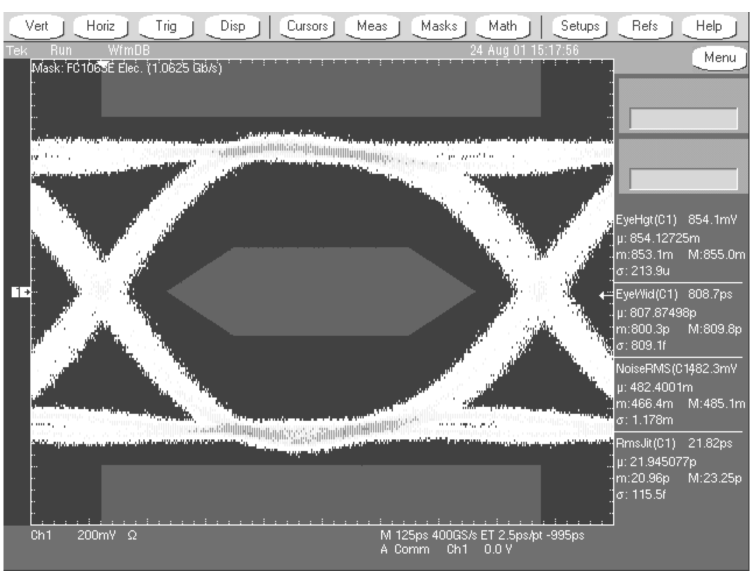

Figure 21. Example of optical mask standard and eye diagram from the optical reference receiver.

7.0 Acquisition Interpolation for Higher Sample Rate

There are three different acquisition modes in an oscilloscope that may be used for increasing the sample rate of the waveform to be higher then the A/D converters used to acquire the data. These are interleaved acquisitions, interpolated acquisitions, and Equivalent time sampling. (ET is an abbreviation for equivalent time.)

7.1 Time Interpolator

Before looking at these acquisitions modes some explanation of the time interpolator is needed. The sample clock in an acquisition is asynchronous with respect to the trigger position of the waveform. Therefore, the samples in an acquisition will vary randomly in position with respect to the trigger position from one acquisition to the next. They may vary by as much as one sample interval. The oscilloscope incorporates a time interpolator that measures the time from the analog trigger position to the first sample after that point in time. The resulting TTOFF value is the time normalized with respect to one sample interval and thus has a range of values of 0.0 to 1.0. This value is then used in conjunction with ET or interpolated acquisition modes to increase the sample rate of the waveform to higher rates than the base A/D converter. The TTOFF value determines the correct time position for the waveform for interpolated acquisitions. For ET acquisitions it determines which time slot, referred to as a bin, to place each acquisition. The time interpolator resolves time measurements to a small fraction of a base sample interval at the base sample rate of the A/D converter.

7.2 Over Sampling

Even though the bandwidth of an oscilloscope may be on the order of 8 GHz with a base sample rate of 20 GS/s there is an advantage to be gained by over sampling at a higher rate such as 2.0 TS/s. This can be accomplished with either interpolated acquisition mode or with ET, equivalent time, acquisition mode. The time interpolator is what provides this advantage by providing smaller time position resolution of events in the waveform. Note that this does not allow the user to see any additional detail in the waveform. Wider bandwidth is required for that. Another benefit of over sampling is a reduction in aliasing.

7.3 Interleaved Acquisition Mode:

Interleaved acquisition mode is used to increase the sample rate by using multiple A/D converters and skewing their sample clocks. The data from each A/D converter is then interleaved with the data from other A/D converters to obtain a sample rate that is higher then the base sample rate of a single A/D converter. Past architectures had one A/D converter per channel. Interleaving would, therefore, reduce the number of available channels. More recent oscilloscope architectures use interleaved A/D converters on each individual channel to obtain a higher base sample rate. In addition multiple channels can also be interleaved.

7.4 Equivalent Time, ET, Acquisition Mode

In this mode the A/D converter operates at the base sample rate such as 20 GS/s or at higher interleaved sample rates. The resulting waveform is built with multiple triggered acquisitions to fill in the waveform at a higher ET sample rate such as 2 TS/s. The time interpolator determines what time position, bin, each acquisition will be placed into for the ET waveform. Since the time interpolator value is random from one acquisition to the next the waveform will be filled in after multiple acquisitions. This mode of acquisition requires a repetitive waveform. However, for eye diagrams the waveform may be triggered at different positions in order to build the eye. NOTE: When ET mode is in operation the bandwidth enhancement filter is applied to the data before it is placed in the ET record. As a result the Tektronix oscilloscope is able to display eye diagrams in ET mode while bandwidth enhancement is enabled.

7.5 Sinx/x Acquisition Interpolation

Interpolation is another means by which the oscilloscope increases the sample rate of the acquired waveform. This has the advantage of operating on single shot as well as repetitive acquisitions. An extensive description of sinx/x interpolation is given in [4]. This interpolation process uses a low pass filter to perform the interpolation. It involves taking the original waveform data and adding zeros in between samples. The number of zeros depends on the desired interpolation ratio. The process of adding these zeros to the waveform data results in shifted images of the harmonic content in the frequency domain up to the new Nyquist point at the interpolated sample rate. Applying a low pass filter that only passes the original spectral content and rejects all the images results in the zeros taking on the values of the interpolated points. The time interpolator value, TTOFF, is then used to correctly position the interpolated points in time with respect to the trigger position from one acquisition to the next.

A Mathcad example of linear interpolation will be used here to illustrate some effects of interpolation. This first example uses linear interpolation as shown below in Figure 22. Note that from one acquisition to the next in an oscilloscope the samples occur at different positions with respect to the trigger position. Thus the linear interpolations between these different sample positions causes variations in the interpolated wave shape.

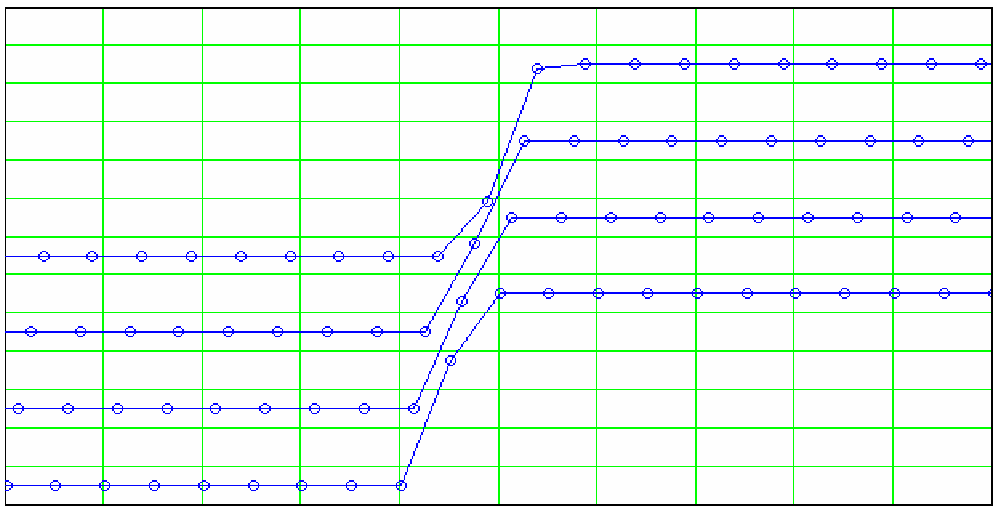

Figure 22. Linear interpolated Mathcad simulated acquisition showing from one acquisition to the next.

The average of 10 simulated steps is shown below in red below in Figure 23. Notice that this average tends towards the original wave shape.

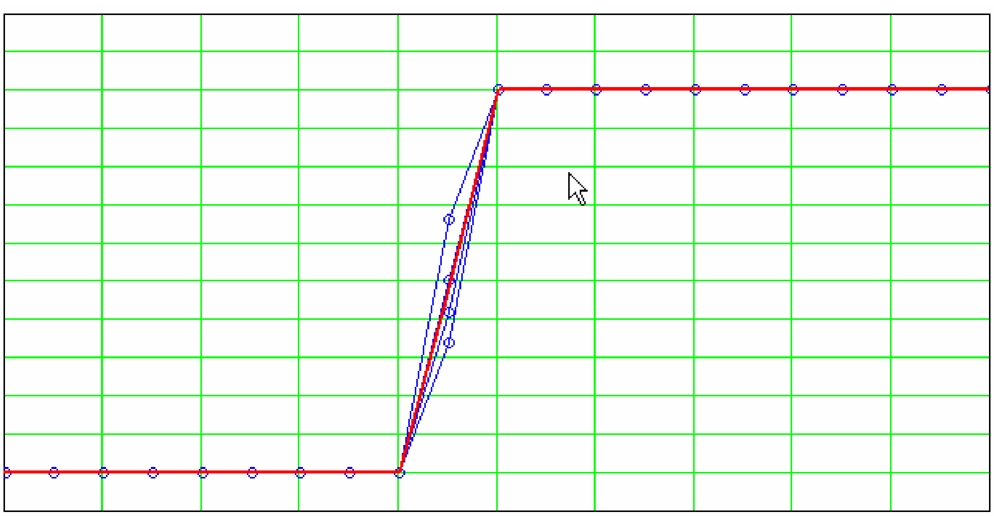

Figure 23. Average of multiple linearly interpolated waveforms.

Also, notice that the average process has time shifted all of the individual acquisitions so that their samples are aligned. This results in the famous frequency response roll-off of –4.01 dB at Nyquist caused by averaging. The most practical way to avoid that roll off effect is to highly over sample the waveform that is being averaged.

Another way to avoid the –4.01 dB rolloff for averaging is to interpolate the waveforms before they are averaged and realign them according to the TTOFF value and then average them. However, that approach would have to be done offline by the user.

Pspline interpolation in Mathcad was then applied to the same original data as in the above example. The result is shown in Figure 24. This example illustrates again how the interpolation for each acquisition may be different due to the different sample positions from one acquisition to the next. (NOTE: Tektronix oscilloscopes use sinx/x or linear interpolation. MathcadTM pspline was used for this example only as a matter of ease to illustrate the desired topics. Sinx/x is better suited for waveform interpolation then is pspline.)

However, when these 10 acquisitions in this example are averaged together the solid red trace is obtained. Notice that because the peaks of the ringing are shifted slightly their average on the red trace is a lower peak ring.

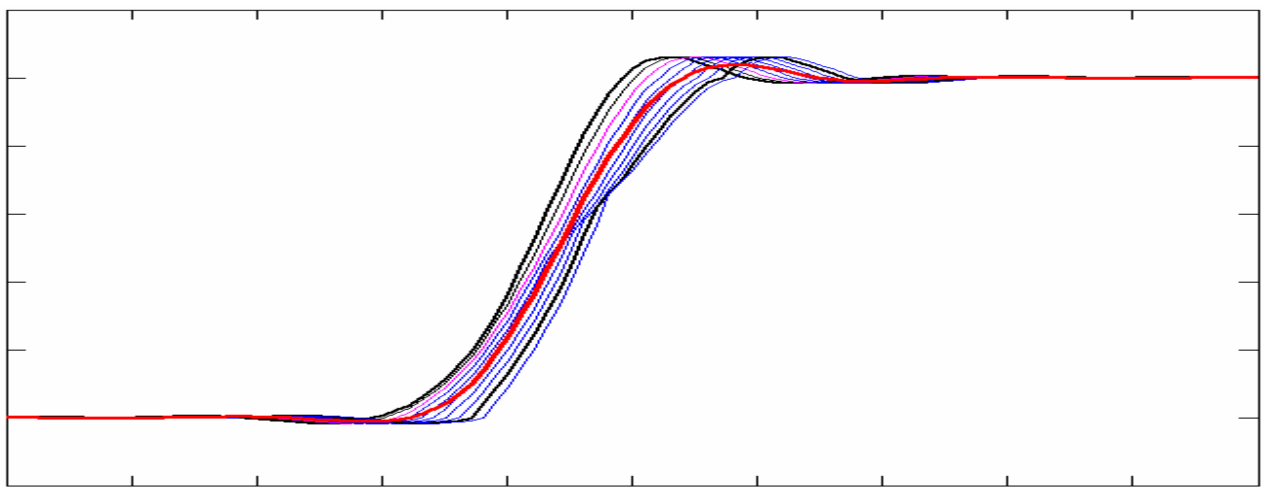

Figure 24. Multiple acquisitions using Mathcad pspline interpolation.

7.6 Interpolation Distortion

Interpolators are not perfect and do introduce small distortion. In general linear interpolation will show more distortion then sinx/x. This is especially true when there are not many samples covering a rapid change in voltage. Sinx/x interpolation will have less distortion then linear for this situation. That assumes that a good interpolation filter was used.

There are numerous factors that will affect the amount of distortion when using sinx/x interpolation. One factor is the quality of the interpolation filter such as passband ripple, stopband attenuation, transition band slope and so on. Another factor is the interpolation ratio. The higher the ratio the more distortion is likely to be introduced. The TDS7000 series oscilloscopes have introduced an improved sinx/x filter system compared to its predecessor, the TDS700 series. This interpolator has excellent performance producing averaged waveforms that are very similar to averaged waveforms produced in ET mode. The most noticeable difference is a slight decrease in risetime that would be on the order of approximately 1 ps out of 75.

7.7 Comparison of interpolation to ET Acquisition Mode

One may wonder why both interpolated acquisitions and equivalent time acquisitions are needed in order to increase the sample rate and decrease the timing resolution. The most obvious difference between the two modes is that interpolation can be performed on a single shot acquisition while ET mode requires numerous repetitive acquisitions to fill in the waveform.

It might appear that the effects of interpolation described in the previous section would be a disadvantage compared to ET mode. Although ET mode does not have that effect it has another effect due to binning of the acquisitions. Some error in placement of data into ET bins results in high frequency noise that is above the bandwidth of the oscilloscope. Averaging or smoothing can be used to remove this out-of-band noise. An interesting point is that the average of interpolated waveforms and the average of ET mode waveforms produce virtually identical result! This tends to contradict misconceptions that sin(x)/x interpolation does not accurately reproduce high speed digital signals.

8.0 Conclusion

This paper has focused on some of the DSP that is performed in modern digital storage oscilloscopes. Tektronix offers competitive advantage in terms of arbitrary filter design capability used for bandwidth enhancement and for optical reference receivers. This is especially true because these types of filters are calibrated during manufacturing so that the specific channel response is transformed by the filter into a desired channel response. As bandwidths and sample rates are pushed to higher performance levels DSP will continue to play increasing rolls in enhancing and improving the accuracy for the oscilloscope response. These complex algorithms are typically well suited to DSP methodologies and impractical to implement in analog circuitry. In addition this paper provided basic discussions of DSP used for optical reference receivers. It also provided information on interpolation and other acquisition modes such as interleave, and equivalent time.

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