ADQ14-FWATD firmware Datasheet

PXIe

PCIe

MTCA.4

USB3.0

Never before detectable faint pulses can now be captured with 14 bit resolution at up to 2 GSPS via the Advanced Time Domain firmware option (–FWATD) for the ADQ14 digitizer. This market leading performance is based on SP Devices' noise reduction technology for pulse data systems consisting of the following core elements:

- *A filter function for linear identification of pulse data*
- *A threshold function for a non-linear suppression of noise*
- *A waveform averaging for suppressing noise by repeated measurements.*

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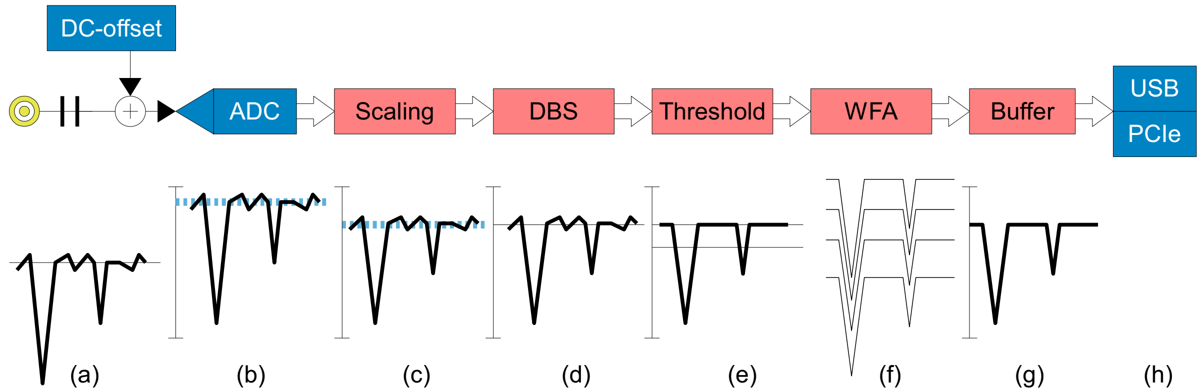
1 Principle of operation

The purpose of the ADQ14-FWATD firmware option is to enhance signal to noise ratio in a pulse application measurement. The firmware contains a baseline adjustment and three levels of noise suppression; linear filtering; non-linear threshold; and waveform averaging which reduces noise by repeating a measurement several times.

In an ADQ14 with several channels, there is one ATD unit per channel. They operate synchronously with the same settings on waveform size and number of accumulations. The threshold properties are individual per channel.

The principle structure of the ADQ14-FWATD is shown in **Figure 1**.



#	DESCRIPTION	USER CONTROL	REF
a	The analog signal is a unipolar pulse (negative in this example)		2
b	To fully use the range of the ADC an analog DC-offset can be added to the signal.	*SetAdjustableBias*	2
c	Digital scaling of gain and offset.	*SetGainandOffset*	3
d	The Digital Baseline Stabilizer, DBS, is an SP Devices' proprietary algorithm for baseline stabilization in pulse data systems. The DBS output is a controlled baseline level set at a user defined value.	*SetupDBS*	4
e	The threshold operation is a noise suppression function that sets noise below a threshold to a defined level. There is a linear FIR filter for signal enhancement and threshold level for non-linear noise suppression.	*ATDSetupThreshold* (TBC) *ATDSetupFilter* (TBC) (TBC)	5
f	Waveform averaging accumulates a large number of waveforms for noise suppression by repeated measurements. (TBC)	*ATDSetupWFA* (TBC)	6
g	Data transfer to host PC. Double buffer means dead-time free transfer even for a large WFA.		7
h	Data transfer to host PC over USB3.0 or PCIe	See ADQ14 manual	7

Figure 1: ADQ14-FWATD architecture

2 Analog input

The analog signal is unipolar. This means that the signal is built up by a baseline at a DC level and a signal pulse, **Figure 1** (a). The DC-level of the signal does not have to be zero. The DC-coupled versions, ADQ14DC, will let the DC level of the input signal through. If required, the AC-coupled version, ADQ14AC, can be used for removing the input DC level.

To fully use the symmetrical input range of the ADC, an analog DC-offset can be added to the signal. This will place the baseline of the signal near one rail in the signal range. The peaks can then cover all the signal range. This DC-offset effectively doubles the resolution for unipolar signals, **Figure 1** (b). The DC-offset level is controlled from software. The control range is rail to rail, but preferably a margin of about 10% is left to allow for overshoot.

3 Zero level adjustment and signal scaling.

When using a DC-offset for optimal analog signal range, baseline is near one end of the signal range. If requested, it is possible to use the digital gain and offset adjustment block to scale the signal.

4 Digital Baseline Stabilizer, DBS

The analog DC-level is accurate for most common straight forward signals analysis. However, for optimum performance of non-linear threshold operation, it is an advantage if the baseline is locked to a determined value, This is achieved in the Digital Baseline Stabilizer, DBS, which is placed right after the A/D Conversion. DBS is a proprietary technology from SP Devices, which analyses the data, finds and adjusts the baseline to 22 bits precision to a target value. **Figure 2**.

The target value is set by the user. The value is preferably close to the analog DC-offset (if that is applied).

The DBS is a blind identification process which operates in the background. Since DBS is always active, it will constantly monitor and follow variations in the baseline and correct for time-invariant behavior like temperature changes.

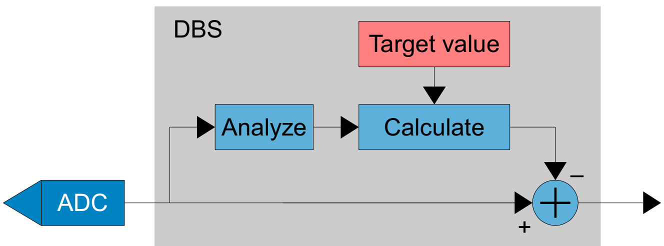


Figure 2: DBS principle of operation.

5 Advanced threshold

5.1 Overview

The threshold function in a basic model is one of the following (depend on polarity of the unipolar pulse), **Figure 3**:

- For a system with positive pulse, samples below a threshold values is set to the a reference value.
- For a system with negative pulse, samples above a threshold values is set to the a reference value.

The result is that values are considered signals and kept and low values are considered noise and are removed.

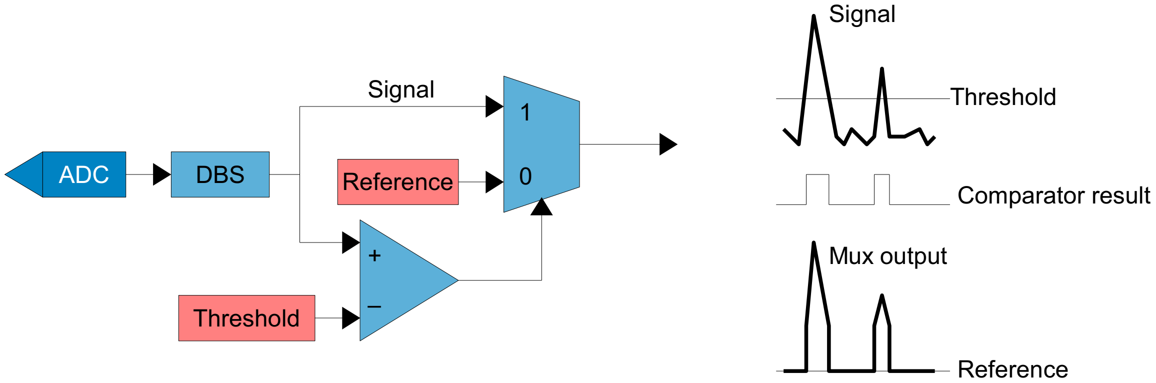


Figure 3: Standard threshold operation on a positive pulse.

5.2 Linear noise suppression in a filter

When the pulse has some property that can be used for separating the pulse from the background noise, a filtering function can be used for enhancing that property. Then the ratio between the signal power and noise level can be increased and the threshold precision can be improved. A filter is a general form of correlation. To be able to perform a filtering or correlation which has an effect, the pulse has to be represented by more than one sample.

Example; low pass filter will then discriminate the high frequency content of the noise and thus reduce the peak amplitude of the noise.

Example; the pulse shape of the weakest pulses is more or less known. Apply a filter that correlates to that shape to amplify this type of signal and suppress other shapes, that is, noise.

The filter coefficients are given as 16 bits two's complements words. In order to represent 1, there are 14 bits decimal values. Thus $0100\ 0000\ 0000\ 0000_{BIN} = 1_{DEC}$ and $1100\ 0000\ 0000\ 0000_{BIN} = -1_{DEC}$.

If the filter output overflows it will saturate.

Note that the filter only affects the selection process. The output data is not filtered. The placement of the filter is seen in **Figure 4**.

5.3 Applying threshold

A block diagram is in **Figure 4**. This is the non-linear part of the noise suppression. The principle is that a large amplitude is considered a signal and a small amplitude is noise.

The threshold is applied to the filtered signal and positions where there are pulses can be identified. On positions with pulses, the data from the original data is saved. Note that the threshold is applied to the filtered signal, whereas data from the original non-filtered signal are selected as output. This is to preserve the shape of the original signals. There is a delay line to match the timing of the branches.

Positions where there are no pulses are replaced by the reference level. The reference level is ideally the same as the target value for DBS.

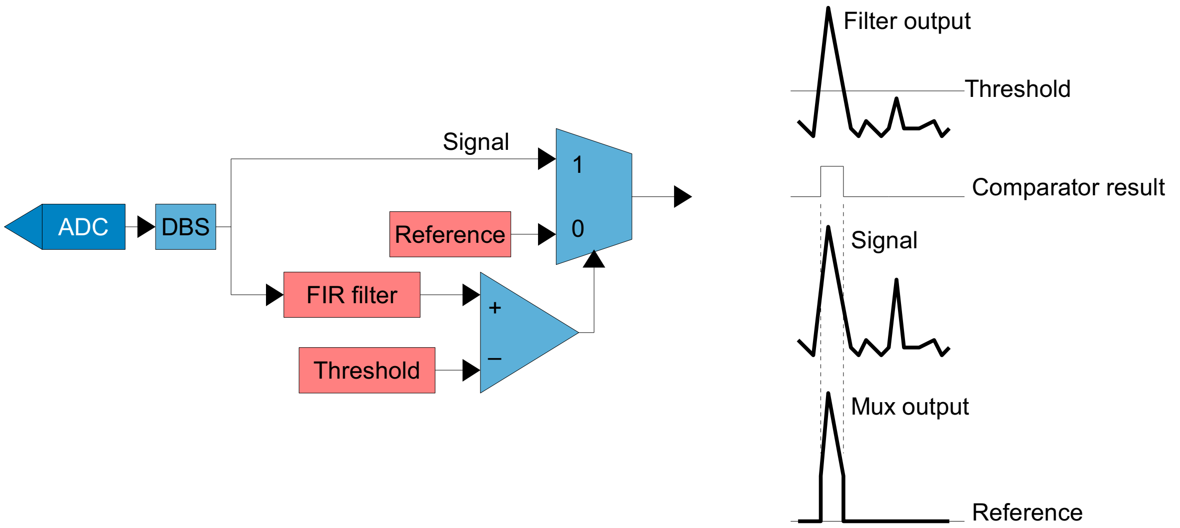


Figure 4: Data flow. Filtered signal is used for selecting samples. Selection is done from original samples.

5.4 Accuracy

The precision of the threshold is determined by the accuracy of DBS, that is 22 bits. DBS will adjust the base line to a 14 bits integer with a measurement error on the 22 bits level on average. As long as the threshold and the reference is in the same format as the data, there will be no rounding errors.

6 Waveform Averaging (WFA)

6.1 Accumulator data word

The waveform averaging is an accumulator that adds waveforms to each other. The first sample in each waveform is added to the first sample of all other waveforms. The result thus has the same length as the waveform but the size of each data word is 32 bits. Since input 14 bits data is MSB aligned into 16 bits words, there is 16 bits available headroom for the additions. This means that 2^16 = 65 536 waveforms may be added before there is a risk of overflow. The data format in the WFA is 2's complement.

An averaging function by definition should contains a division with the number of waveforms. However, this division is not suitable for FPGA implementation and is left for the user to perform in software.

6.2 Accumulator vector length

The waveform averaging uses the on-board DRAM as accumulator. The waveforms may thus be very large, up to 1 MSamples.

6.3 Trigger a waveform

Each waveform is started with a trigger. All triggers used for acquisition may be selected as triggers for the waveform.

A system using waveform averaging is preferably a repeated measurement. Then the trigger is either the external trigger starting the acquisition from an external equipment or the internal trigger where the ADQ14 controls the external equipment.

There are also cases where certain pulse shapes can be analyzed and the level trigger is used for triggering several times on a repeated pulse.

The dead-time at end of each waveform is about 20ns. This is the time period from where the last sample was take until the system may accept a new trigger.

7 Data read-out to host PC

Since the data read out speed to the PC is lower than the acquisition data rate, there has to be an extra buffer on the ADQ14 to store a waveform result before sending it to the PC. The data rate requirements to the PC is rather low since it is reduced with the number of waveforms in the accumulator. But after the last waveform has been accumulated, the accumulator is blocked for readout and cannot accept new waveforms during a period of time. This is solved by introducing an extra buffer on the ADQ14, that holds the result while reading out and not blocking the accumulator.

A new accumulation may start after 20ns.

*Example: The WFA will accumulate 100 waveforms of 50 kSamples each. The sample rate is 2 GSPS and the accumulator size is 4 bytes, The trigger rate is 10 KHz. This means that the accumulator will contain 50 k * 4 = 200 kBytes. Since a double buffer is available, the readout may continue for an entire accumulation, that is 100 / 10 kHz = 10 ms. To read out 200 kbytes in 10 ms requires a data rate of only 20 MBytes/s.*

The data read-out puts requirements on the number of waveforms to be accumulated, ***NOFWAVEFORMS***. For a USB3.0 host connection, the NOFWAVEFORMS has to be

2 GSPS X 2 channels * 4 bytes/sample / 200 MByte/s = 80 waveforms (1)

For a PCIe Gen2 by 8 lanes link, the same calculation gives

2 GSPS X 2 channels * 4 bytes/sample / 3200 MByte/s = 5 waveforms (2)

Below these limits, the dead-time has to increase. Note that these numbers are independent of the length of the waveform.

8 Performance summary

Table 1: Performance

	ADQAC14-1X, ADQ14DC-1X MIN	TYPICAL	MAX	DEFAULT
Thresbold				
Filter length		16		
Filter coefficients	–2	0	$2-2^{-14}$	0, first tap = 1
Filter coefficients resolution		2^{-14}		
WFA				
Number of accumulations			65 536[1]	
Waveform length –2A –4A –2C –4C [Samples]	512		1 Mi	
Waveform length –1X –2X [Samples]	1024		2 Mi	
Waveform length [ms]			1	
Length setting granularity [Samples]	8			
Dead time waveform [ns]			20	
Dead time accumulation [ns]			20	
Registers				
DC-offset level [codes]	-32768	+/- 26000	+32767	0
DBS target [codes]	-32768	+/- 26000	+32767	0
Threshold level [codes]	-32768	+/- 26000 -/+ T	+32767	0
Reference level [codes]	-32768	+/- 26000	+32767	0

1. Guaranteed safe scaling.

9 Appendix on simulation of the advanced threshold

Two pulses on different phase compared to the sampling clock and one noise spike[1] is shown in blue in **Figure 5**. A filter is applied which result in the red curve. The key point here is that the peak of the signal pulse is more than 1 sample wide. The noise is as narrow as 1 sample in the peak. This is typical for Gaussian noise.

Figure 6 shows two pulses and the spike from **Figure 5** with added Gaussian noise. The blue curve is before the filter and the red curve after the filter. The red curve at the bottom is selections from the original signal based on the thresholded applied to the filtered signal.

The filter coefficients was set as:

B = [0 0 1 2 3 7 7 3 2 1 0 0]; B = B./sum(B); (MATLAB syntax) (3)

The filtering method is then more like a correlation, since the coefficients are selected to look like a pulse. A low pass filter generated with a standard FIR generator gives a similar result.

This simulation reduced noise peak value by 3 dB and power by 3 dB. **Figure 7** shows the histogram of the noise before and after the filter.

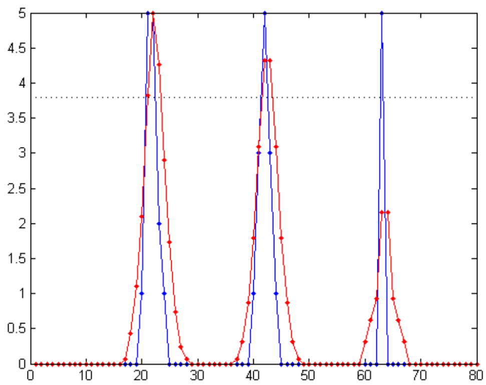


Figure 5: Two pulses and one noise spike before and after filter.

1. Gaussian noise has spikes above the threshold line, but they are very rare. For illustration purpose one such spike is forced into sample 64.

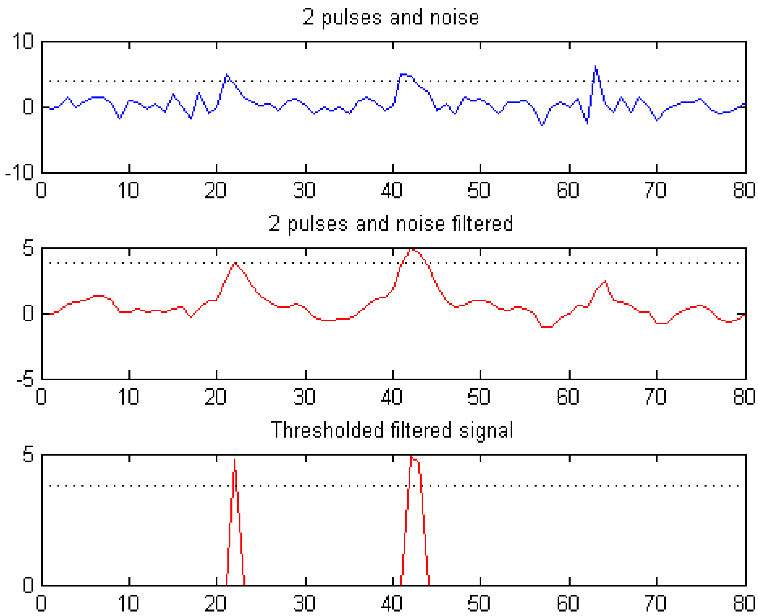


Figure 6: Two pulses and one noise spike + random noise before and after filter. Threshold applied and only the two pulses are left.

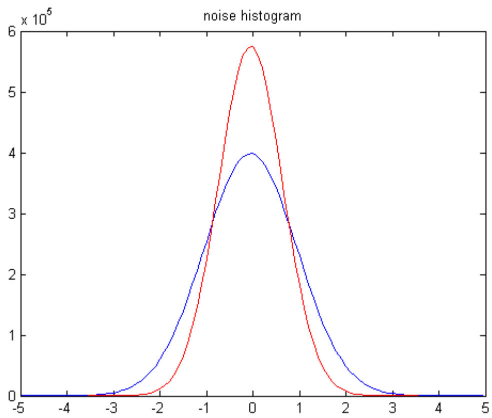


Figure 7: The histogram of the noise before and after the filtering shows that the amplitude is reduced.

10 Beta release

The beta release limitations:

support ADQ14–1X and ADQ14–2C only.

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