Data acquisition and processing

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Content

• Introduction
• Analogue approach
• Digital approach
• Digital signal processing
Introduction

✓ Discuss approach for processing signals from detector for detector R&D

✗ Analysis of “sample related properties”, like d-spacing or $S(Q, \omega)$ type data analysis

✗ “Back-end” electronics / computer interface

✗ Varies too much from institute to institute
Introduction

- Which properties of the detector do we want to measure during detector R&D?

1) Neutron detection efficiency
2) Gamma sensitivity
3) Position resolution
4) Count-rate capability (dead time)
5) Stability
6) Timing resolution?
Introduction

• How do we measure these properties?

1) Pulse height spectrum
2) Counter/scaler
3) Time to Digital Converter (TDC)
4) Digitiser/Oscilloscope
Introduction

• What is a pulse height spectrum?

- Histogram of the distribution of signal amplitudes
Introduction

• Neutrons usually produce a peak in a pulse height spectrum

• Noise, gammas and cosmic radiation usually produce a exponentially decaying tail.

➤ Pulse height spectrum tells how easy it is to discriminate neutron from unwanted background ⇒ pulse height analysis one of most important tools.
Introduction

Pulse height spectra for GS20 scintillator directly coupled to a photomultiplier tube

Counts/300sec

Threshold

“rubbish”

neutron

(PMT) noise

Gammas

Neutrons

Pulse height (V)
Introduction

- Position spectrum: tells where the neutrons were detected
Analogue approach

• Typical analogue data acquisition chain (text book):

  “detector”

  preamplifier

  shaping amplifier

  discriminator

  pulse height analyser

  scaler/counter, TDC

  pulses: \[\frac{\text{pulse}}{\text{signal}}\]
Analogue approach

• For fast timing some people use this circuit:

  (Personally) I don’t like it because the pulse height spectrum doesn’t tell you anything about how the set the threshold for the discriminator.
Analogue approach

- Different type of measurements require different types of electronics ⇒ have to buy a lot of equipment

- and .......
Analogue approach

• Pulse height analysers, a.k.a. Multi-Channel Analysers (MCAs)

• Realize that pulse height analysers are mainly used for spectroscopy applications ⇒ dealing with slow signals:

pre-amplifier signal:

shaping amplifier signal:
Analogue approach

- Can they handle this:
- Answer: not very well

200 ns
Analogue approach

- Peaking time: nearly all MCAs require peaking time >=250ns

- MCA in:
  - What will happen?
  - Pulse height spectrum:

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Analogue approach

• Let’s slow down signal so that peaking time >250ns

• Shaping amplifier in:

  200 ns

  Counts

  Pulse height (V)

• Shaping amplifier out:

  500 ns

  Counts

  Pulse height (V)
Introduction

- Much less stable with incorrect threshold

Pulse height spectra for GS20 scintillator directly coupled to a photomultiplier tube

Counts/300/sec

Pulse height (V)

using long shaping time

Threshold

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Neutron detectors
Analogue approach

• Have you ever wanted to veto the MCA processing an analogue signal like this?

• MCA in:

• Veto signal:

• Veto signal is too late.

• Very hard to delay fast analogue signals for microseconds ⇒ cannot veto MCA.
Analogue approach

• Other things to keep in mind when using MCAs:

• Most MCAs can handle only positive signals

  OK:  

  not:  

• MCAs usually have \(~1k\,\Omega\) input impedance \(\Rightarrow\) need 50\,\Omega\ termination resistor at input.
Analogue approach

- My opinion:

- Generation:

(for neutron detector development)
Digital approach

- Typical digital data acquisition chain:

```
detector ─── preamplifier ─── sampling ADC + computer interface (digitiser) ─── computer

software
```
Digital approach

• That is all the equipment you need for whatever type of measurement you want to do

• All analysis is done in software!

• Extremely flexible and versatile. Compact.

• Hardware is cheaper. But, to make use of full potential, software development needs resources. Do not underestimate this.

• Digitisers with dedicated signal processing hardware are commercially available
Digital approach

- Could do analysis in dedicated hardware for equipment “permanently” installed on beam line.
Digital approach

• Sampling Analogue to Digital Converter (ADC).

• ADC measures the amplitude of the signal at fixed intervals (T sampling period)
  • Normally specified as sampling rate (1/T)

• ADC gives amplitude in discrete values
  • Number of values = $2^{\text{number of bits of ADC}}$
Digital approach

• Two “digitisation” errors:
  1) Time
  2) Amplitude

• Offset errors

• Time digitisation/sampling

• Nyquist–Shannon sampling theorem:

  Perfect reconstruction of a signal is possible when the sampling frequency is greater than twice the maximum frequency of the signal being sampled (Nyquist rate)
Digital approach

• If the sampling rate is less than the Nyquist rate, aliasing will occur.

• Time domain:

• Frequency domain: (sampling frequency $f_s$)
Digital approach

• Equivalent for spatially undersampling image: Moiré pattern
Digital approach

• Aliases hard/impossible to remove with digital signal shaping filters (they are really present in signal).

• Detector signals are intrinsically bandwidth limited ⇒ do not really need anti-aliasing filter.

• Usually better to chose sampling rate ~2 times higher than Nyquist rate because of digital signal filtering.

• To give you an idea: for resistive wire gas tubes we use a 33MHz sample rate for the ADCs

➢ Time digitization errors are easy to deal with in practice
Digital approach

• Amplitude digitisation

• Needs some attention

• Number of bits.

• 10 effective bits (not actual number of bits) is good enough for most applications.

• Effective number of bits is less than actual number of bits because of noise and other error sources.

• Most errors specified in Least Significant Bit (LSB) $\Rightarrow$ helps to have ADC with higher number of bits.
Digital approach

- Differential Non-Linearity (DNL):

- “Short range” non-linearity (how much it is varies from neighbour to neighbour)
Digital approach

- AD9648: Sampling ADC, 14-bit, 125 MHz maximum sampling rate

- Usually specified for slow signals. Non-linearity noticeably higher for fast signals
Digital approach

- Integral Non-Linearity (INL):
  - Deviation between actual output value and perfect/theoretical output value.

- AD9648:
  - See as medium to long range effects
Digital approach

- Effect of non-linearity on pulse height spectrum:
- Dips usually appear at predictable bins (128, 256, 512)
- Dips/peaks will be greatly reduced by using low pass filter with (very) short RC time
Digital approach

- Offset voltages

- ADC will give non-zero reading even with 0V at input.

- Intrinsic in ADC ⇒ not good enough to remove DC component from analogue signal before ADC.

- Originate from analogue parts of ADC ⇒ don’t get better (in absolute value) for ADCs with more bits.

- Offsets devastating for resistive wire gas tube electronics.
Digital approach

• Offsets vary with temperature ⇒ one off calibration is not good enough.

✓ Offsets are easily removed with digital signal processing.
I will discuss only digital signal filtering.

Two types of digital (signal) filters:
1) Finite Impulse Response (FIR)
2) Infinite Impulse Response (IIR)
Digital signal processing

• Infinite Impulse Response (IIR)

• General formulae (time domain):

\[ y[n] = \sum_{i=0}^{P} b_{i} x[n - i] + \sum_{i=1}^{Q} a_{i} y[n - i] \]

• Current value of output of filter \( y[n] \) depends on:
  1) the previous input values of the filter \( x[n-i] \)
  2) the previous output values of the filter \( y[n-i] \) (feedback) \( \Rightarrow \) recursive

• P and Q are the order of the 2 parts
Digital signal processing

- IIR filters are equivalent of analogue filters (RC, CR, Butterworth and so on) and are usually designed starting from their analogue counterpart.

- They are easy to implement in software and hardware.

- Because of feedback, signals are “infinitely” long, like the capacitor in an RC filter will never get completely discharged.

- Can be unstable, but 1st order RC filter is unconditionally stable, provided that $0 < a_1 < 1$
Digital signal processing

• One (theoretical) complication:

• Maximum frequency in analogue systems is infinite.

• Maximum frequency in digital systems is the Nyquist frequency (finite).
Digital signal processing

- Bilinear transform is simplest way to map (infinite) analogue frequency range onto (finite) digital frequency range (frequency warping).
Digital signal processing

- I prefer to make my digital filters by cascading 1\textsuperscript{st} order RC and CR filters.

- 1\textsuperscript{st} order filter: \(P=1\) and \(Q=1\)

\[
y[n] = \sum_{i=1}^{Q} a_i \ y[n-i] + \sum_{i=0}^{P} b_i \ x[n-i] \quad \Rightarrow
\]

\[
y[n] = a_1 \ y[n-1] + b_1 \ x[n-1] + b_0 \ x[n]
\]

- This web-site can calculate coefficients for most common filters (warped and unwarped): http://www.cs.york.ac.uk/~fisher/mkfilter
Digital signal processing

• Problem for first sample of trace \( (n=0) \) in:
  \[
y[n] = a_1 \ y[n-1] + b_1 \ x[n-1] + b_0 \ x[n]
  \]

  \[
y[0] = a_1 \ y[-1] + b_0 \ x[0] \quad \text{where } y[-1] \text{ is undefined}
  \]

• Simple solution: start with second sample \( (n=1) \) and assign \( y[0] = y[1] \) after calculating \( y[1] \)

• Issue for calculating \( y[1] \): have to use \( y[0] \) which is not calculated previously (filtered) \( \Rightarrow \) filter needs some samples to start up.
Digital signal processing

- Effect:

- Leave enough samples in trace before start of signal!!

“start up time”

- Leave enough samples in trace before start of signal!!
Digital signal processing

• How to implement a simple (unwarped) 1\textsuperscript{st} order low-pass RC filter (integrator)

\[ a_1 = e^{-T/\tau} \quad b_0 = 1 - e^{-T/\tau} \quad b_1 = 0 \]

• where \( T \) is sampling period and \( \tau \) the RC time of the analogue RC filter.

• Remember that \( \tau = \frac{1}{2 \pi f_c} \) with \( f_c \) cutoff frequency

(mkfilter web-site wants \( f_c \) and not \( 2\pi f_c \))
• How to implement a simple (unwarped) 1st order high-pass RC filter (differentiator)

\[ a_1 = e^{-T/\tau} \quad b_0 = 0.5 (1 + e^{-T/\tau}) \quad b_1 = -0.5 (1 + e^{-T/\tau}) \]

• where T is sampling period and \( \tau \) the RC time of the analogue RC filter.
Digital signal processing

- Finite Impulse Response (IIR)

- General formulae (time domain):
  \[ y[n] = \sum_{i=0}^{P} b_i \ x[n - i] \]

- \( P \) is the order of the filter

- Current value of output of filter \( y[n] \) only depends on the input values of the filter \( x[n-i] \).
Digital signal processing

- FIR filters are always stable (no feedback).

- FIR filters tend to be of high order (~20) \(\Rightarrow\) computational intensive.

- They don’t have direct analogue equivalents.

- More complicated to calculate coefficients.

- FIR filtered signals return to baseline much faster because of finite length (number of terms).
Digital signal processing

• Most interesting FIR filters:
  1) Moving average filter
  2) Trapezoidal filter

• Moving average filter: \[ y[n] = \frac{1}{P+1} \sum_{i=0}^{P} x[n-i] \]

• Low pass filter (integrator type).

• Order of filter determines rise/fall time of signal.

• Very efficient implementation using recursion.
Digital signal processing

• Example 5\textsuperscript{th} order filter

\[
y[n] = \frac{1}{6} \left( x[n] + x[n-1] + x[n-2] + x[n-3] + x[n-4] + x[n-5] \right)
\]

\[
y[n+1] = \frac{1}{6} \left( x[n+1] + x[n] + x[n-1] + x[n-2] + x[n-3] + x[n-4] \right)
\]

• after rewrite:

\[
y[n+1] = y[n] + \frac{1}{6} \left( x[n+1] - x[n-5] \right)
\]

✓ Need only 1 addition and 1 subtraction, independent of order of filter.
Digital signal processing

- Once $y[n]$ gets corrupted output of filter will be corrupted for ever $\Rightarrow$ best to add redundancy.

- Trapezoidal filter

- Band pass filter (combined low pass and high pass).

- Good timing resolution and energy resolution (spectroscopy).
Digital signal processing

- Works best for signals with long fall time.

- Very hard to find good filter coefficients for fast signals.
Digital signal processing

• Implementation: Basically 2 time-shifted moving average filters.

• Assume moving average filter: 
  \[ MA[n] = \frac{1}{P+1} \sum_{i=0}^{P} x[n-i] \]

• Trapezoidal output is given by:
  \[ y[n] = MA[n] - MA[n-P-G] \]

• where G is the width of the flat part (in samples).

• Find it hard/impossible to suggest values for P and G.
Digital signal processing

• Pros and cons of IIR compared to FIR.

✔ IIR does require much fewer terms (additions and multiplication) than most FIR.

✔ More intuitive/easier to design since there are direct equivalents of analogue filters.

✗ Require a long time to get back to baseline.

➢ I normally use IIR filters for my “electronics”.

• Neutron detectors

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Summary

• Current generation of fast sampling ADC is good enough to be able to use digital approach.

• Very flexible, if you have the data analysis software.

• Signal filtering in digital electronics has many advantages.

• Modern Field Programmable Arrays (FPGAs) are capable of implementing signal processing algorithm that was optimised in software.