Data acquisition and processing

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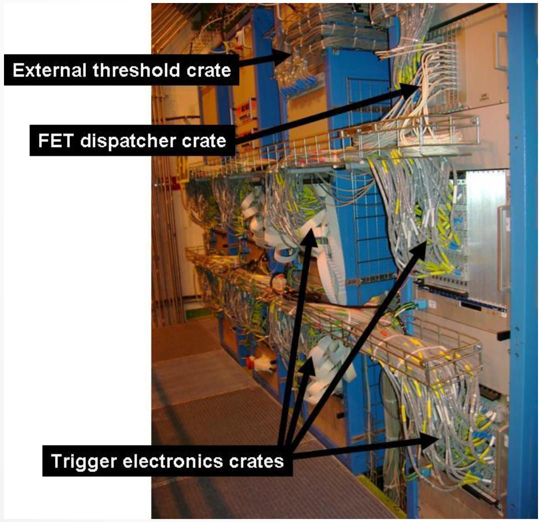
Content

- Introduction
- Analogue approach
- Digital approach
- Digital signal processing

Introduction

- ✓ Discus approach for processing signals from detector for detector R&D
- ✗ Analysis of "sample related properties", like d-spacing or S(Q, ω) 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?

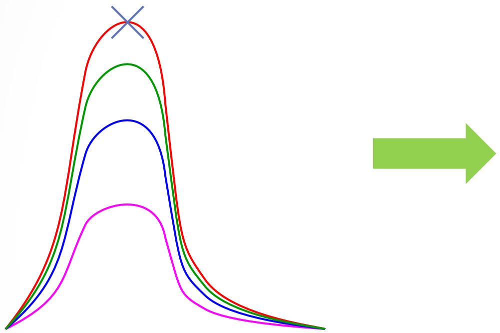
Introduction

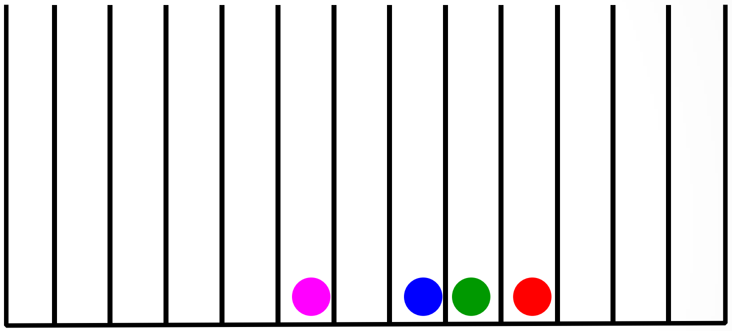
- How do we measure these properties?

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

Introduction

- What is a pulse height spectrum?





Histogram

- Histogram of the distribution of signal amplitudes

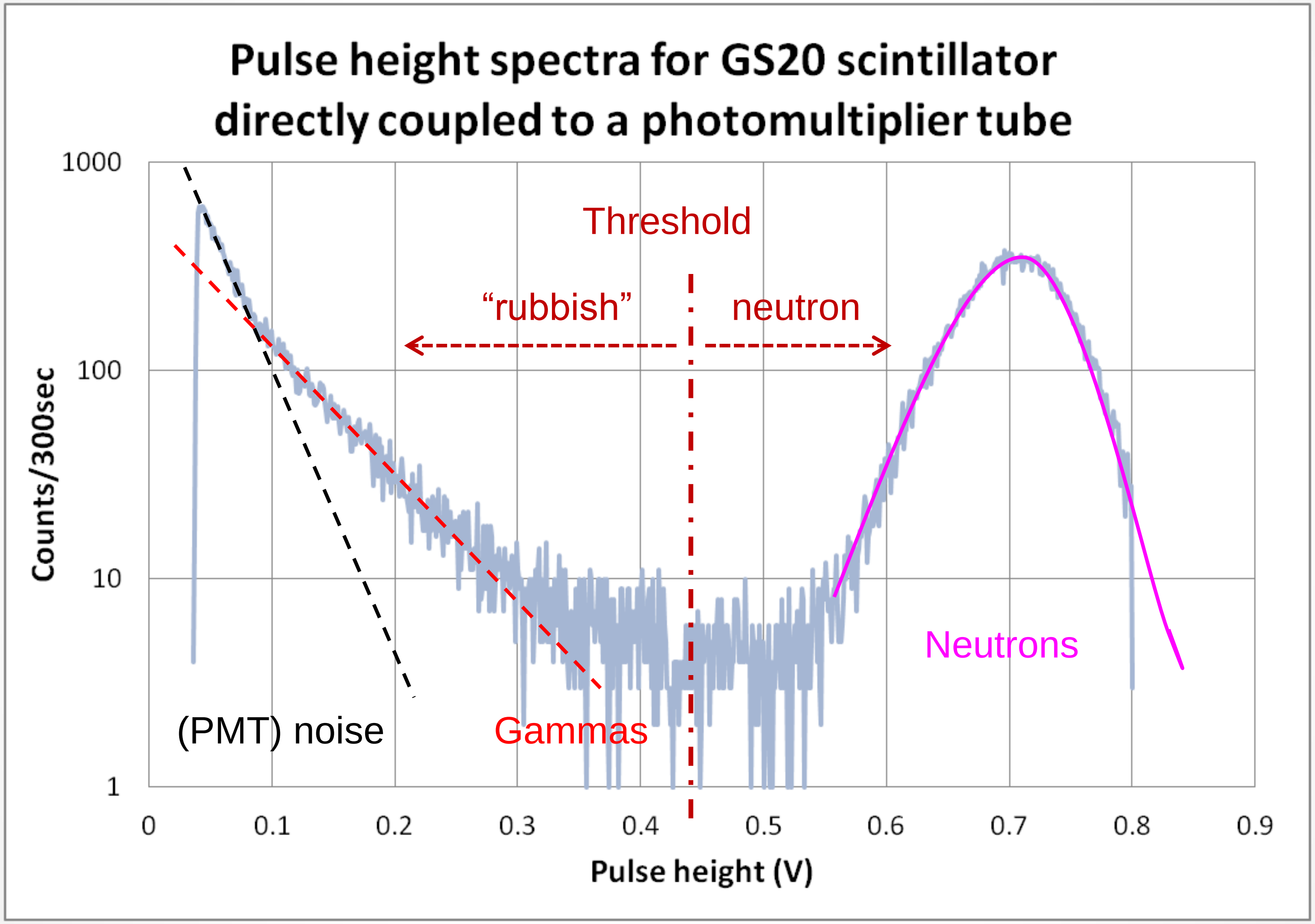
Introduction

- Neutrons usually produce a peak in a pulse height spectrum

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

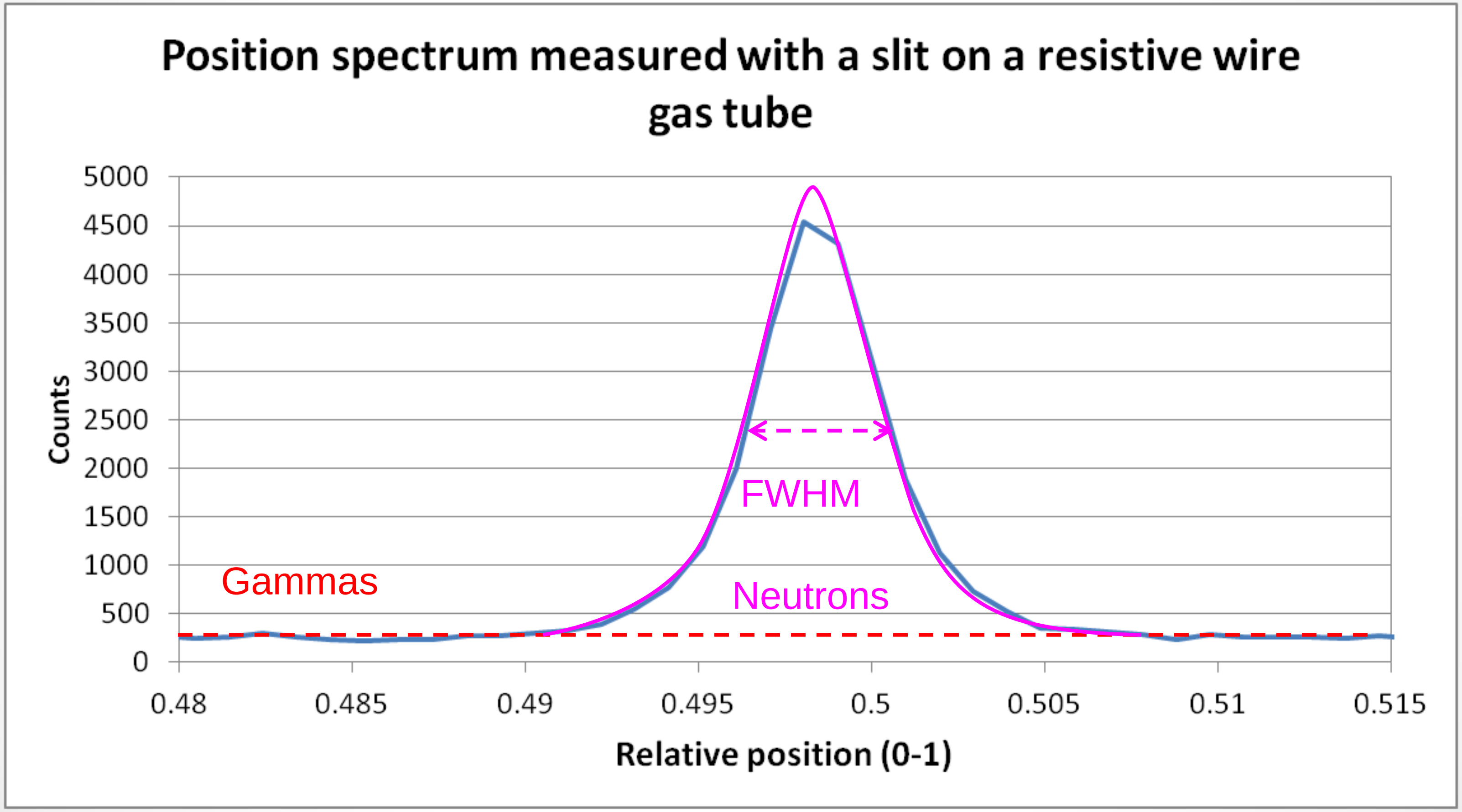
➢ Pulse height spectrum tells how easy it is to discriminate neutron from unwanted background ⇒ pulse height analysis one of most important tools.

Introduction



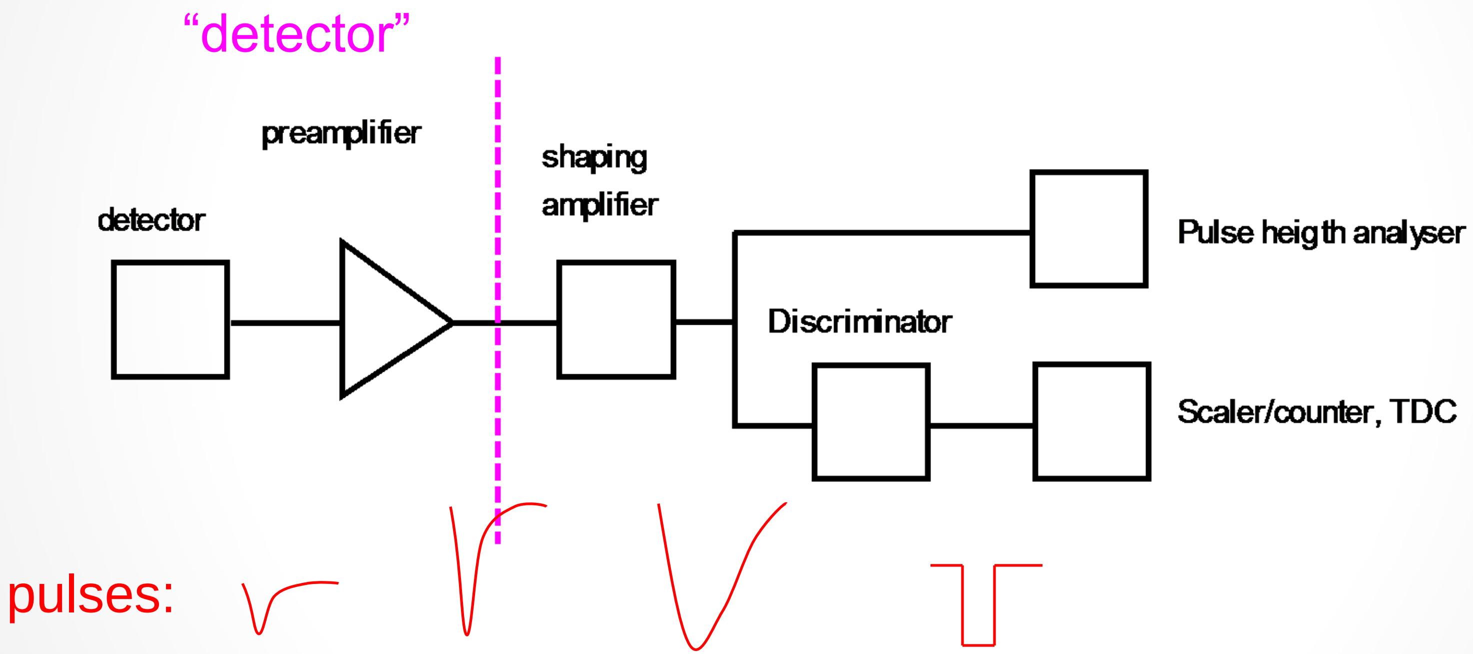
Introduction

- Position spectrum: tells where the neutrons were detected



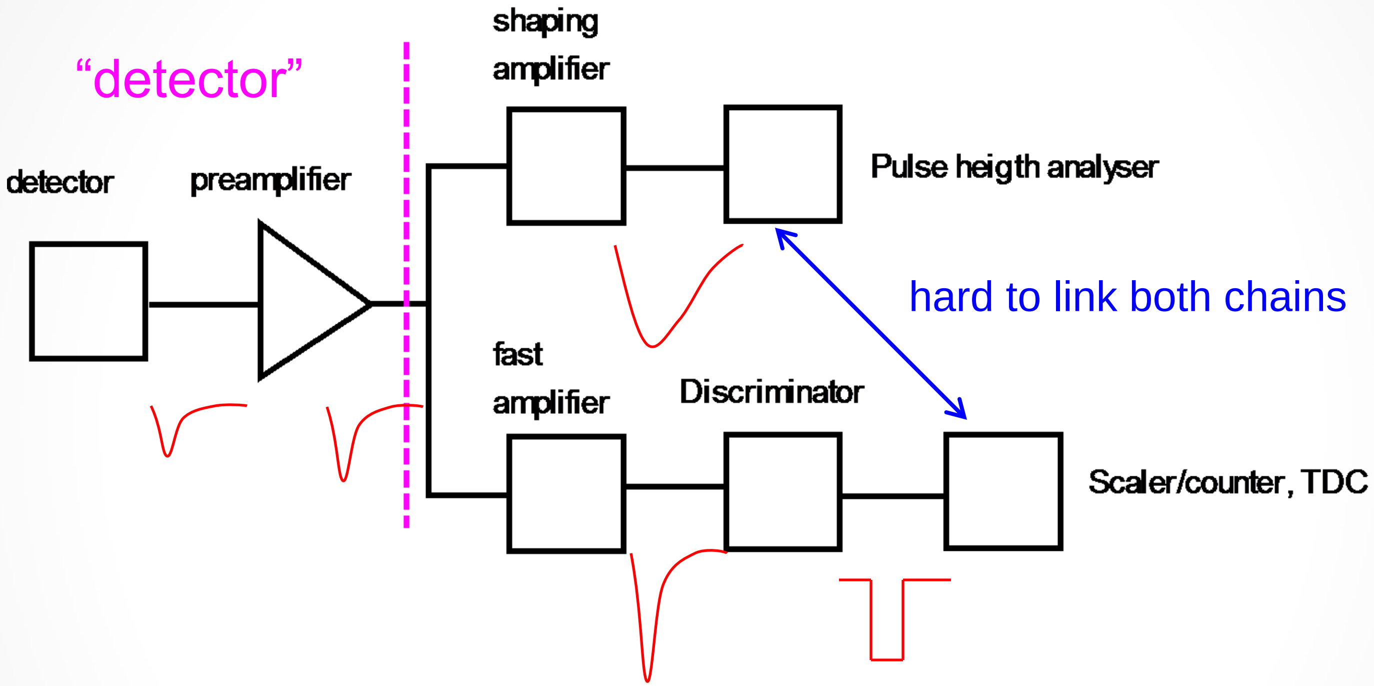
Analogue approach

- Typical analogue data acquisition chain (text book):



Analogue approach

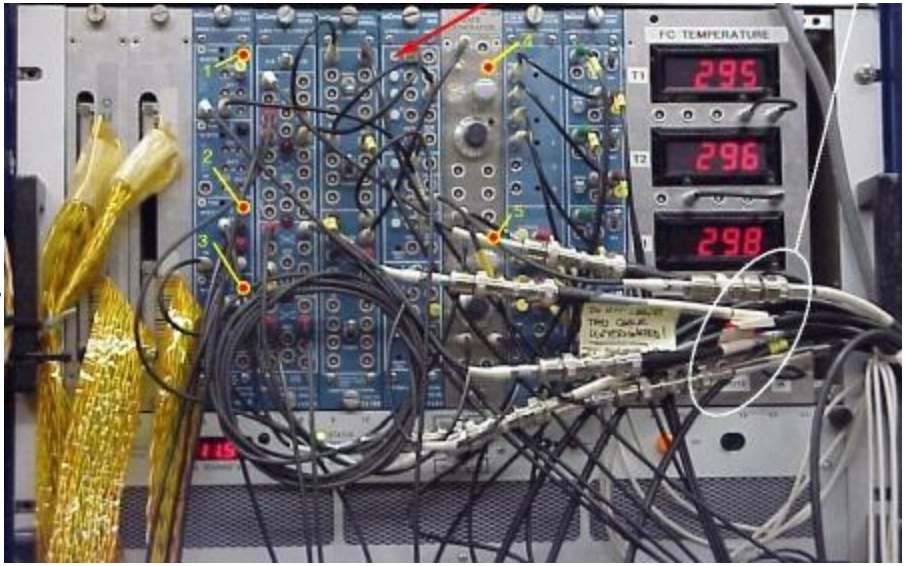
- For fast timing some people use this circuit:



- (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 $\Rightarrow$ have to buy a lot of equipment

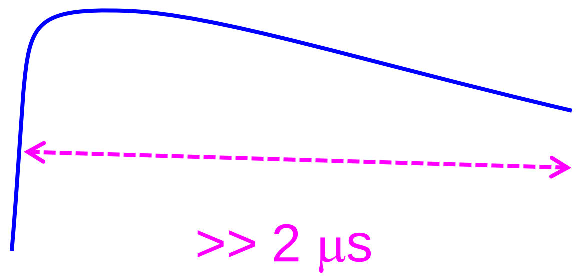
- and .............

Analogue approach

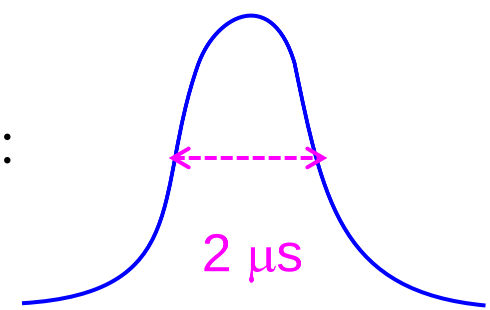
- Pulse height analysers, a.k.a. Multi-Channel Analysers (MCAs)

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

pre-amplifier signal:

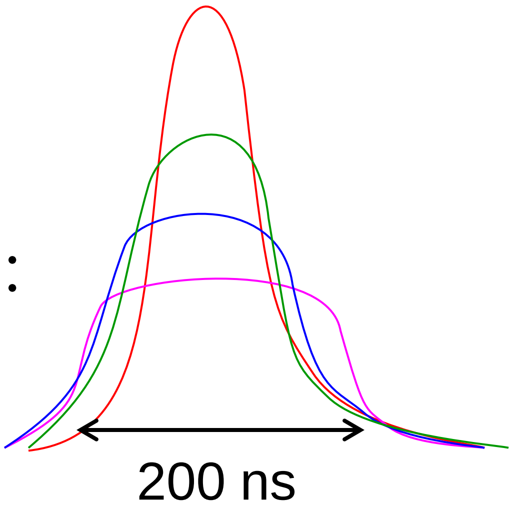


shaping amplifier signal:



Analogue approach

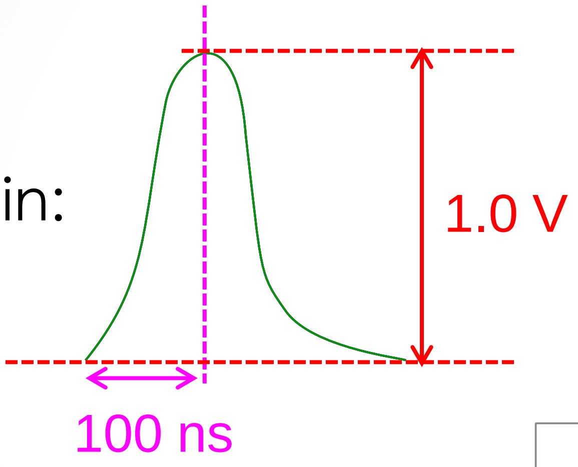
- Can they handle this:



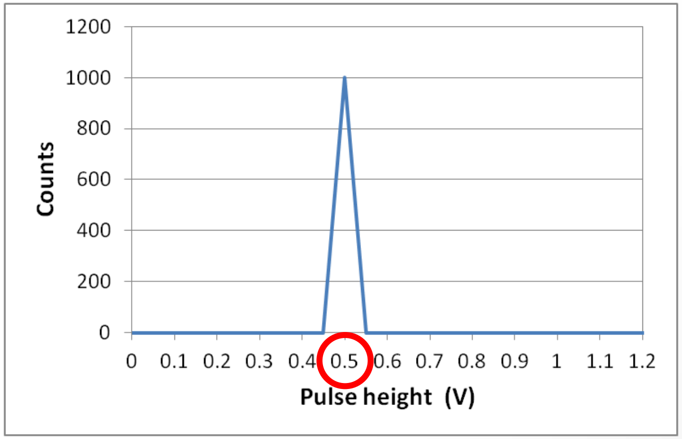
- Answer: not very well

Analogue approach

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

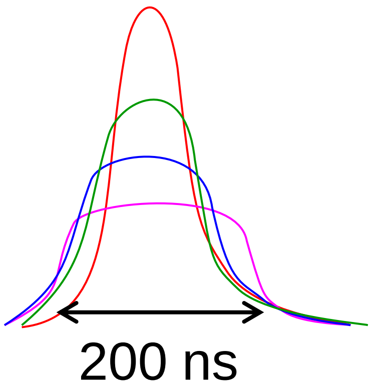


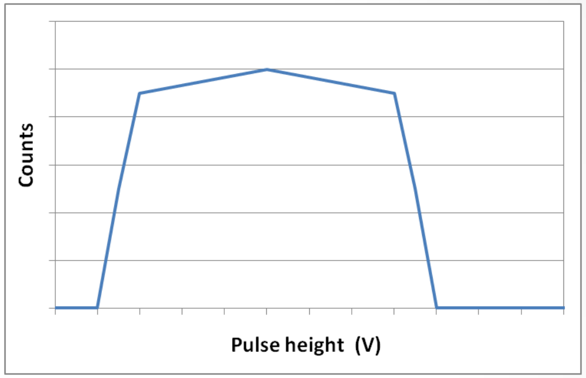
- What will happen?
- Pulse height spectrum:



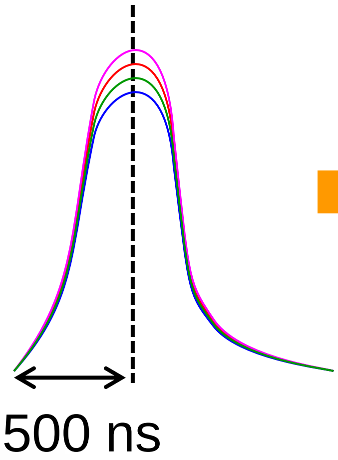
Analogue approach

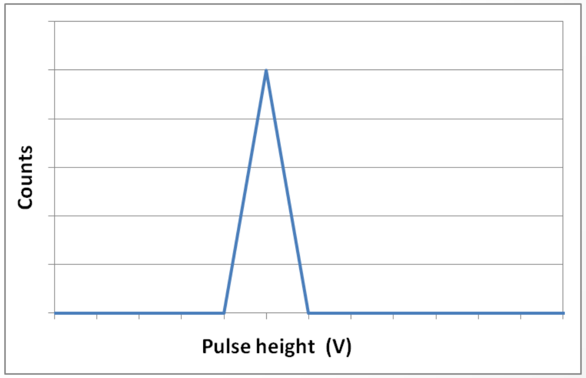
- Let's slow down signal so that peaking time >250ns
- Shaping amplifier in:



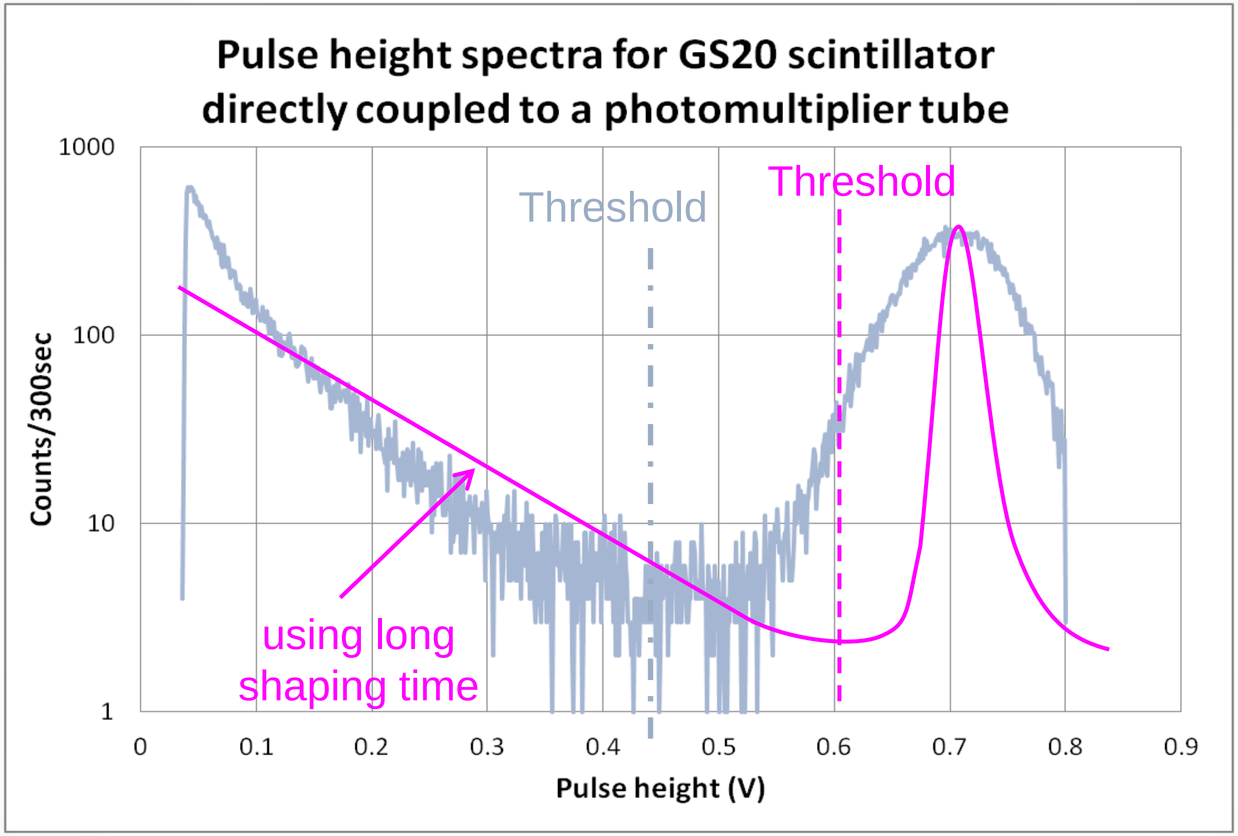


- Shaping amplifier out:





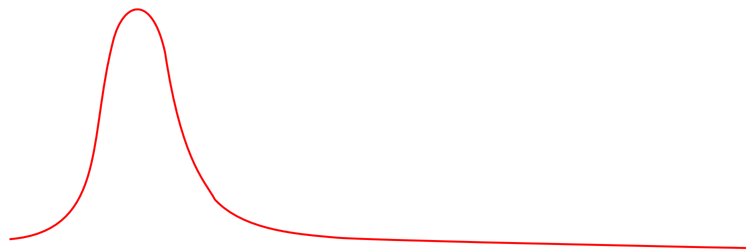
Introduction



- Much less stable with incorrect threshold

Analogue approach

- Have you ever wanted to veto the MCA processing an analogue signal like this?
- MCA in:



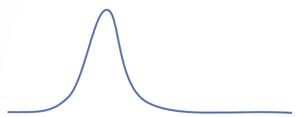
- Veto signal:

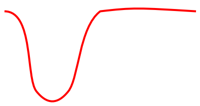
- Veto signal is too late.
- Very hard to delay fast analogue signals for microseconds $\Rightarrow$ cannot veto MCA.

Analogue approach

- Other things to keep in mind when using MCAs:

- Most MCAs can handle only positive signals

OK: 

not: 

- MCAs usually have ~1k Ω input impedance $\Rightarrow$ need 50 Ω termination resistor at input.

Analogue approach

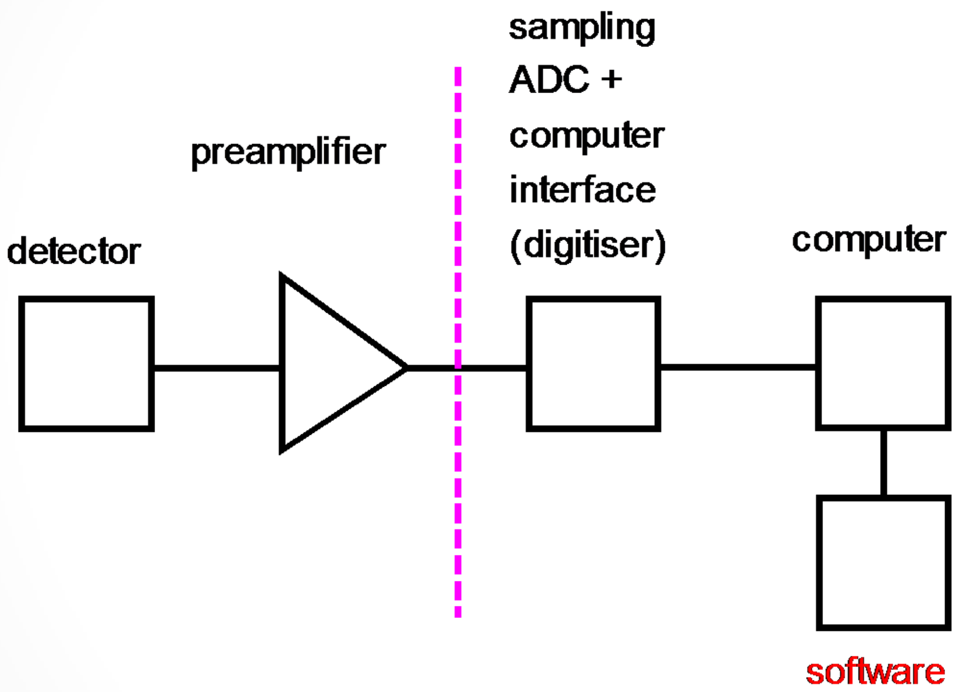
- My opinion:

- Generation:

(for neutron detector development)

Digital approach

- Typical digital data acquisition chain:

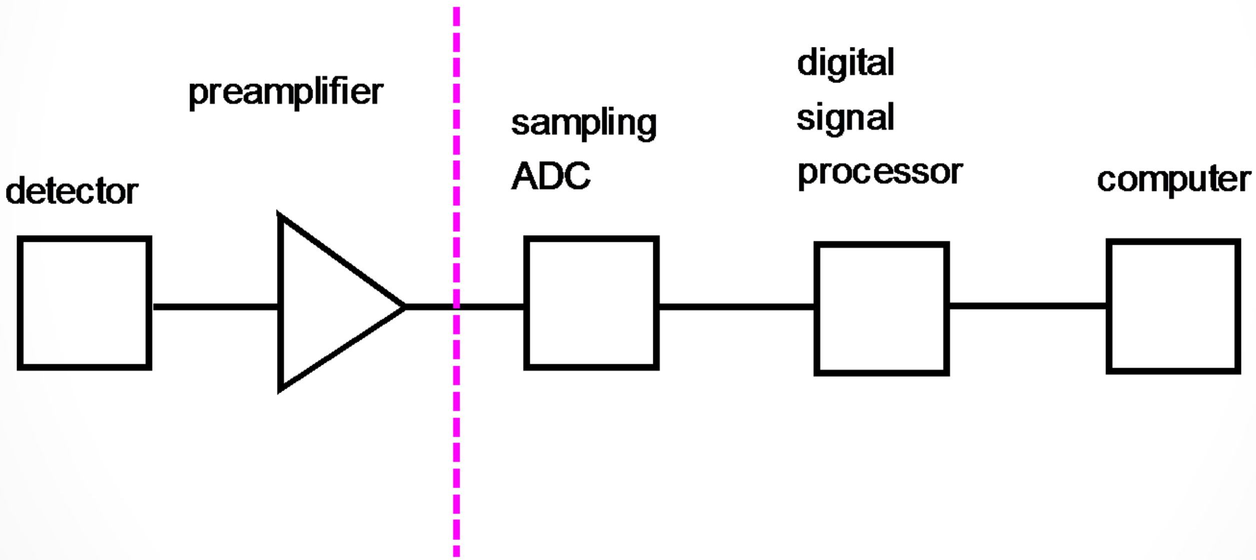


Digital approach

- 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

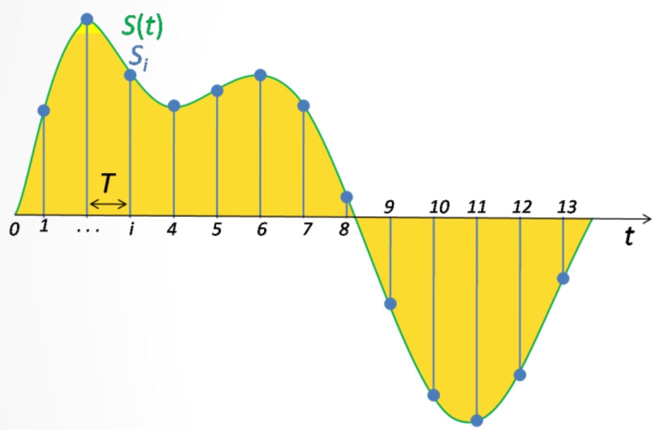
Digital approach

- 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

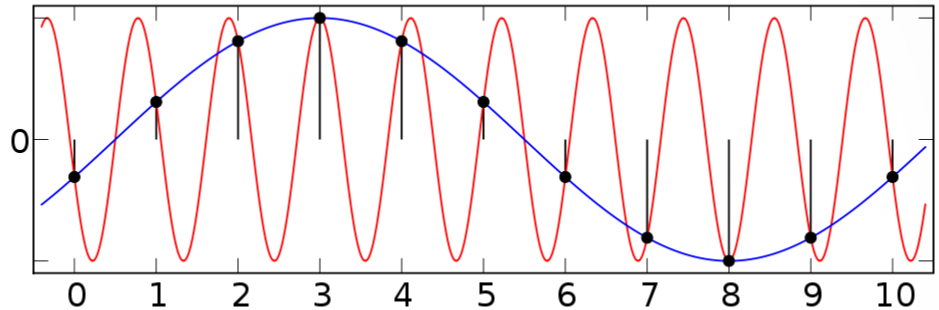
- Nyquist–Shannon sampling theorem:

  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

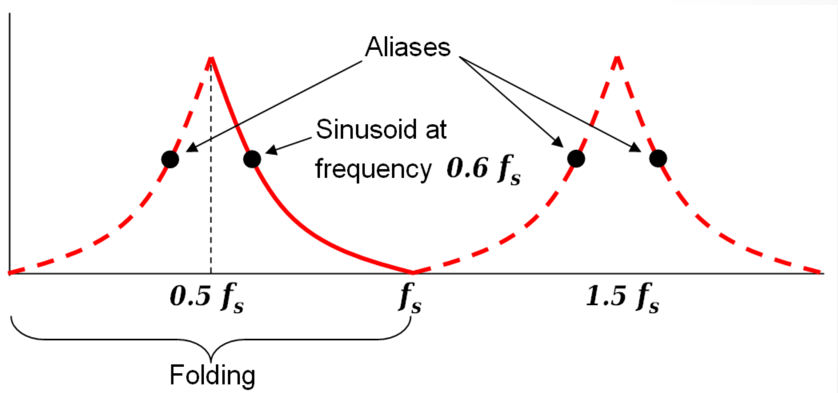
- If the sampling rate is less than the Nyquist rate, aliasing will occur.

- 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 $\Rightarrow$ 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

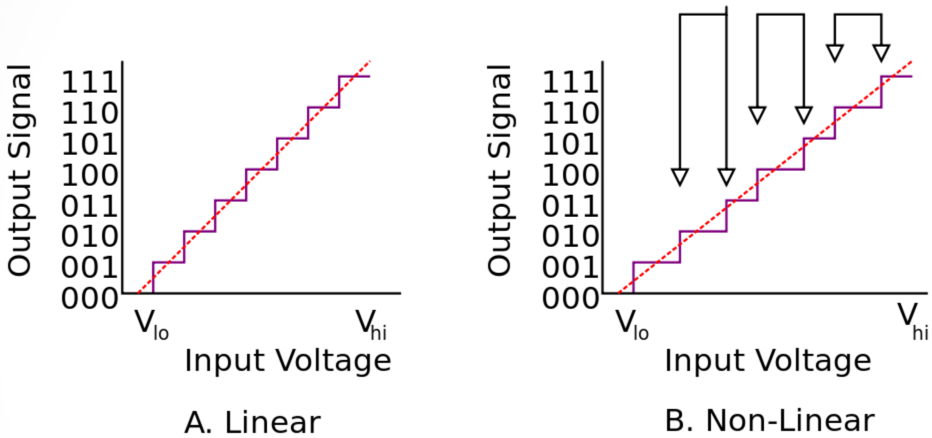
➢ Time digitization errors are easy to deal with in practice

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 bits.

Digital approach

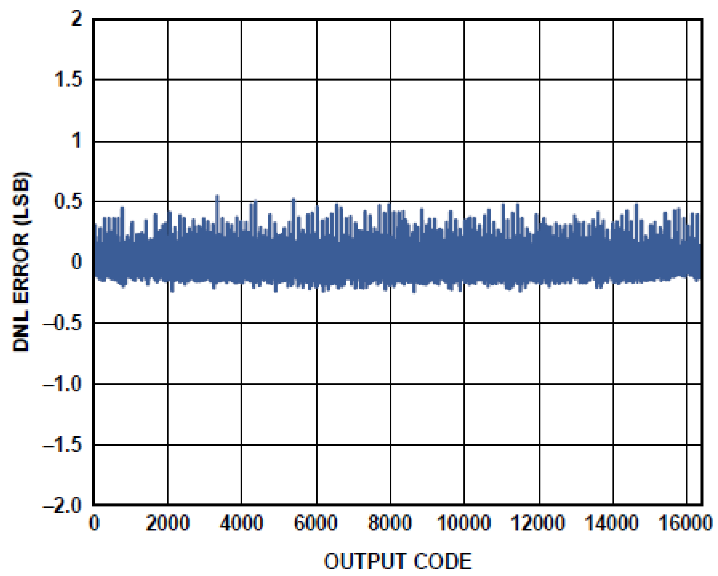
- Differential Non-Linearity (DNL):



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

Digital approach

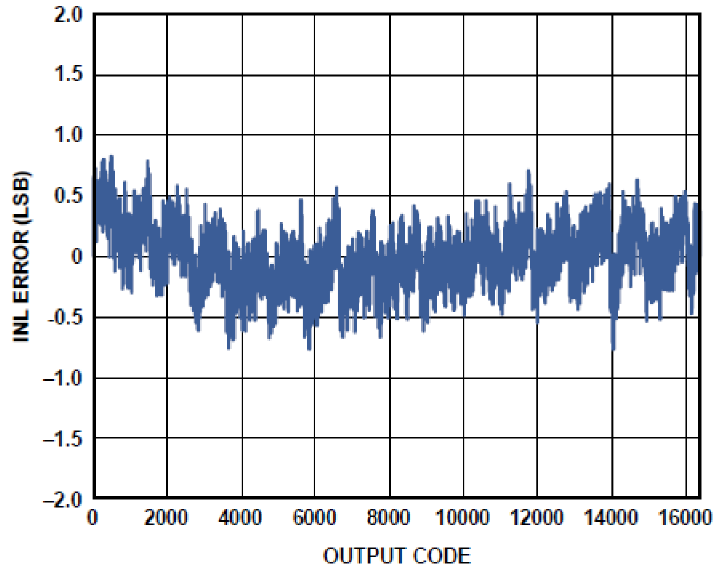
- AD9648: Sampling ADC, 14-bit, 125 MHz maximum sampling rate



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

Digital approach

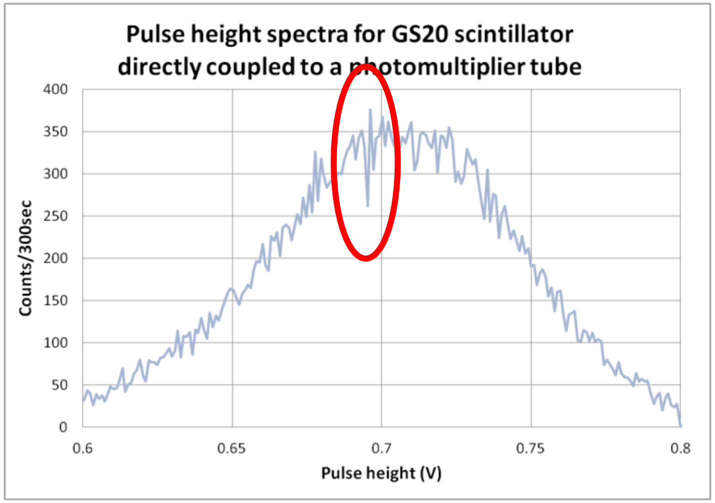
- Integral Non-Linearity (INL):
- Deviation between actual output value and perfect/theoretical output value.
- AD9648:



- 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.

INSIS

Digital approach

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

✓ Offsets are easily removed with digital signal processing.

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

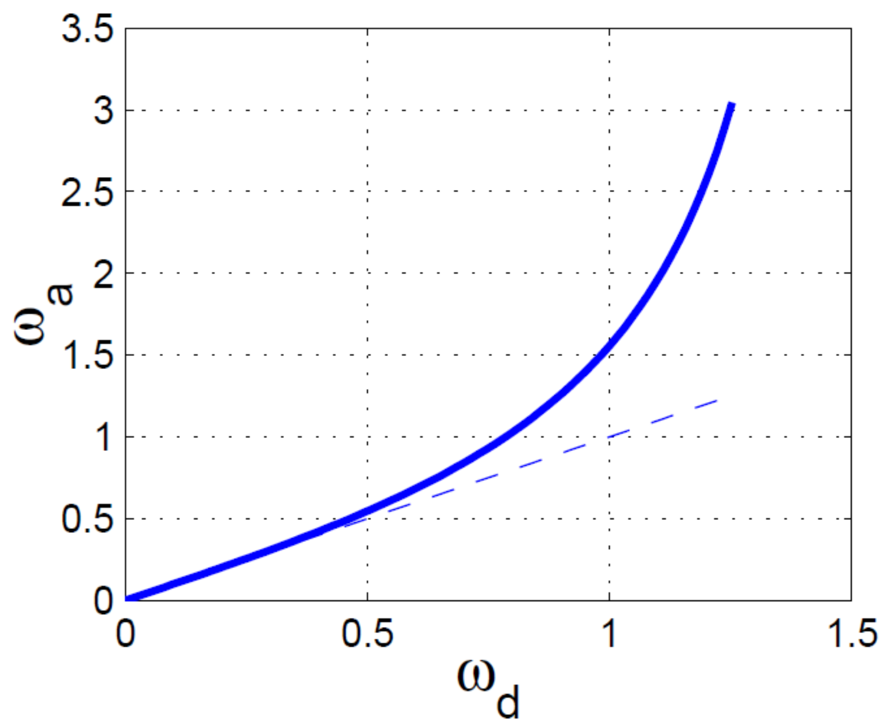
- IIR filters are equivalent of analogue filters (RC, CR, Butterworth and so on) and are usually designed starting from their analogue counterpart.
- They are easy to implement in software and hardware.
- Because of feedback, signals are "infinitely" long, like the capacitor in an RC filter will never 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 1st order RC and CR filters.

- 1st order filter: P=1 and Q=1

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

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

- 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:

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

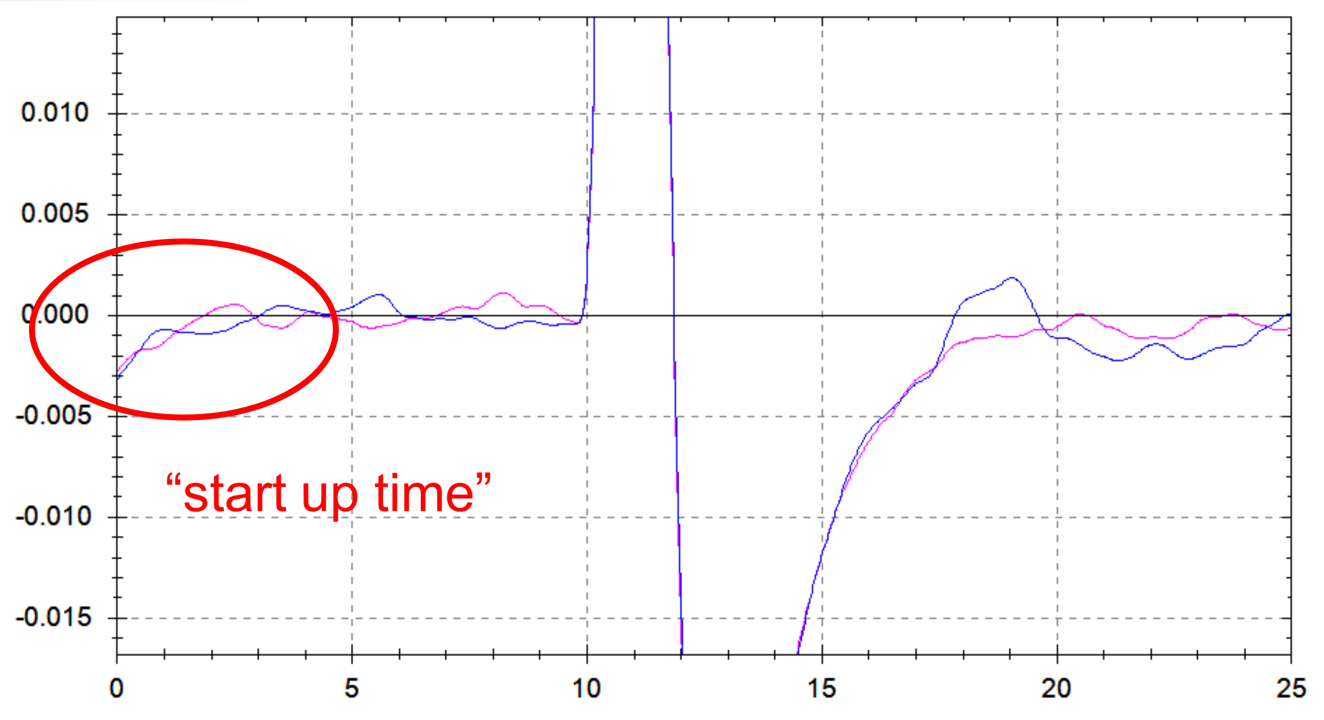
$y[0] = a_1\ y[-1] + \ b_0\ x[0]$ where y[-1] is undefined

- 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!!

Digital signal processing

- How to implement a simple (unwarped) 1st order low-pass RC filter (integrator)

$$a_1 = e^{-T/\tau} \qquad b_0 = 1 - e^{-T/\tau} \qquad b_1 = 0$$

- where T is sampling period and τ the RC time of the analogue RC filter.

- Remember that $\tau = \dfrac{1}{2\pi f_c}$ with f_c cutoff frequency

  (mkfilter web-site wants f_c and not $2\pi\ f_c$)

Digital signal processing

- 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 τ 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 5th 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.

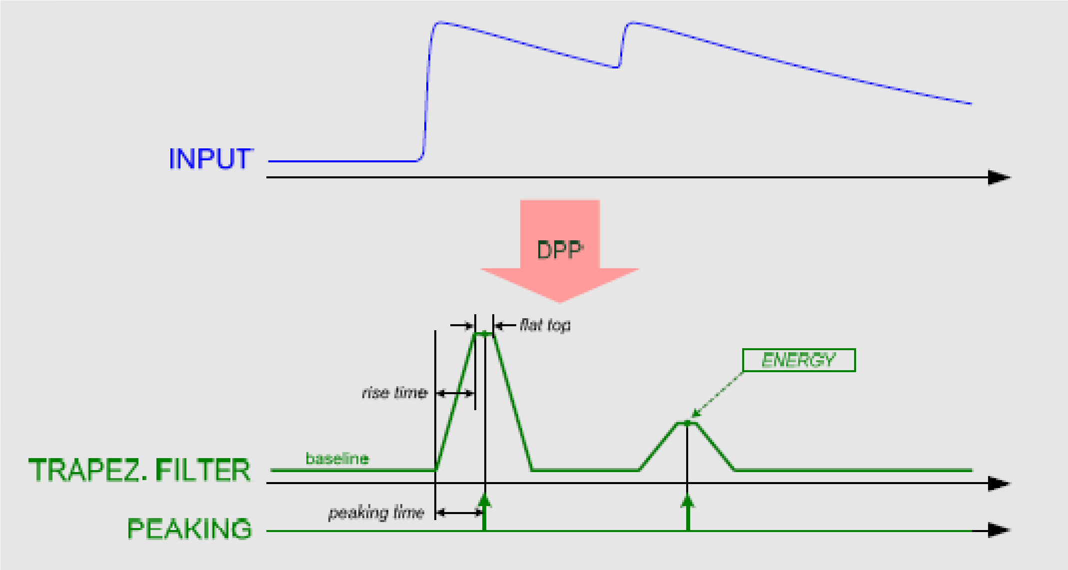
- Trapezoidal filter

- 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".

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.

The End