

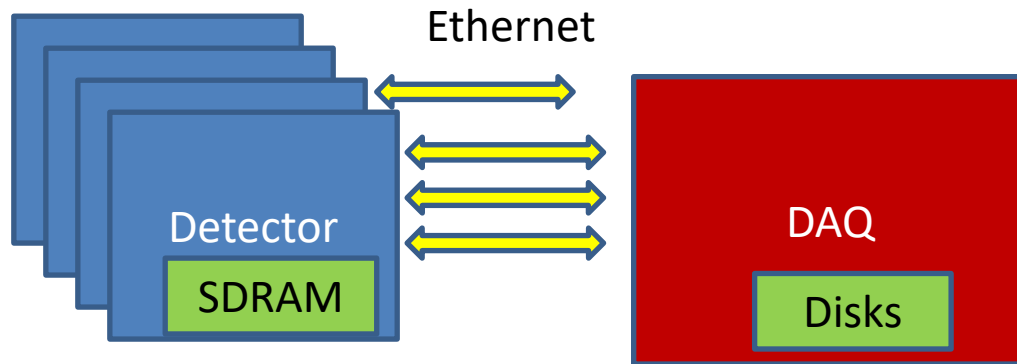
# ECAL Compression Techniques

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# Need for compression

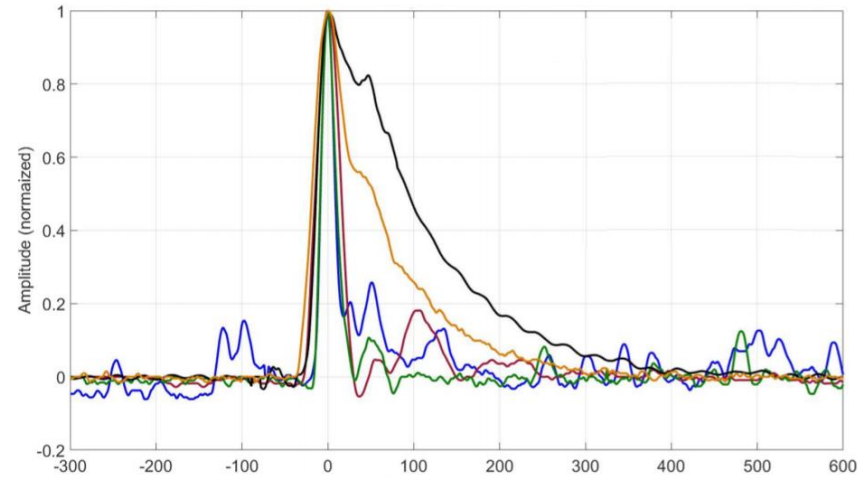
- Saving disk space for the archiving
- Limited bandwidth between detectors and the data acquisition system (DAQ)
- Saving RAM capacity in detector modules



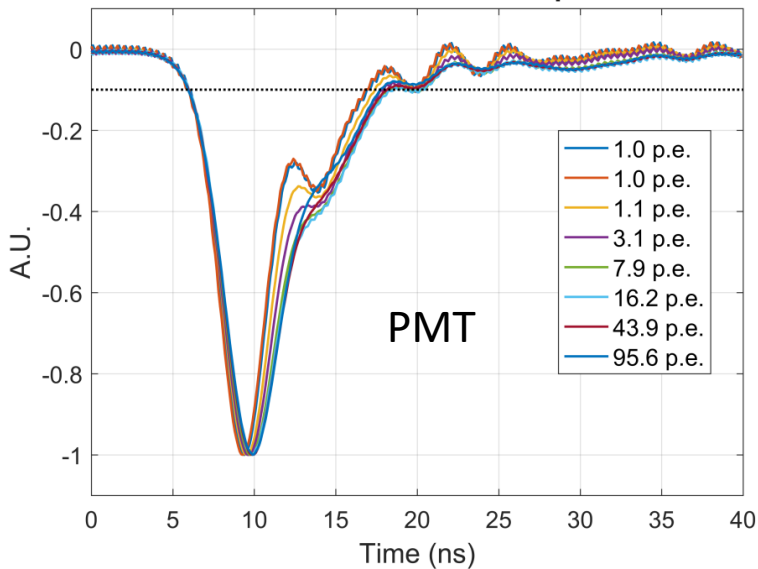
- Constraints on resources and power

# Input Signals

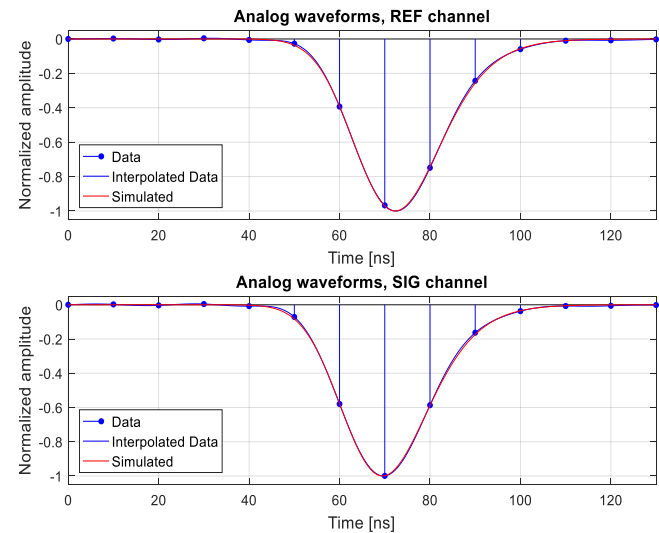
- Acquired PMT and SIPM signals:
  - Similar to some extent,
  - Stability is limited,
  - Shaping changes original signal.



Normalized Waveform Templates

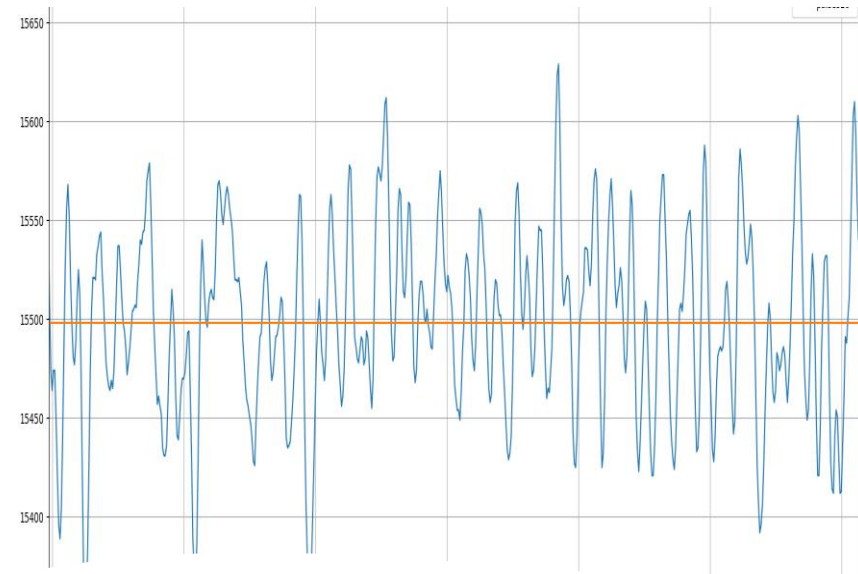
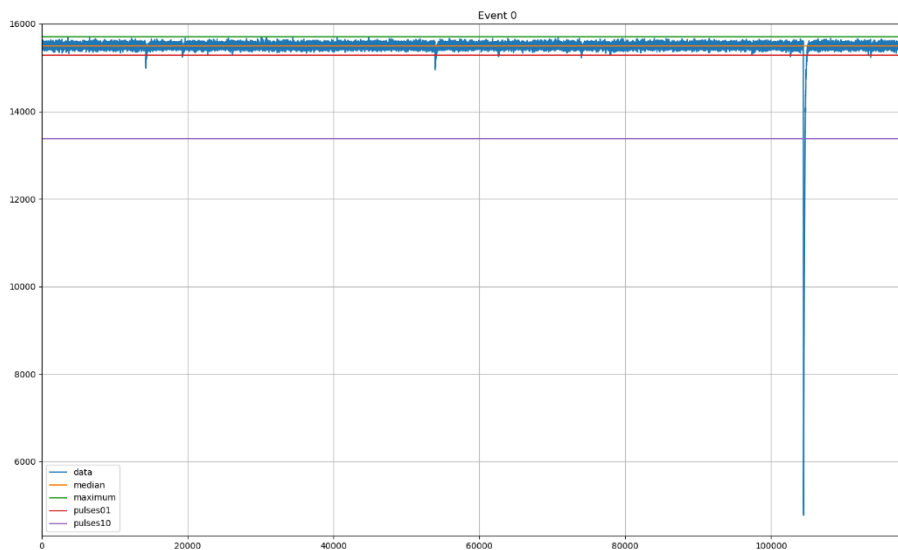


shaping



# Compression scenarios

- Lossless or lossy compression:
  - Allowable losses in processing should be small to preserve key signal features
- Compression of all samples or pulse selection:
  - signals without selection
  - Compression of pulses with high sensitivity
  - Compression of pulses with low sensitivity (large pulses)
  - Number of selected samples around pulses
- Sampling frequency vs. bandwidth



# Selected compression scenario

- In the AMBER experiment, available MSADC boards feed new Carrier Cards

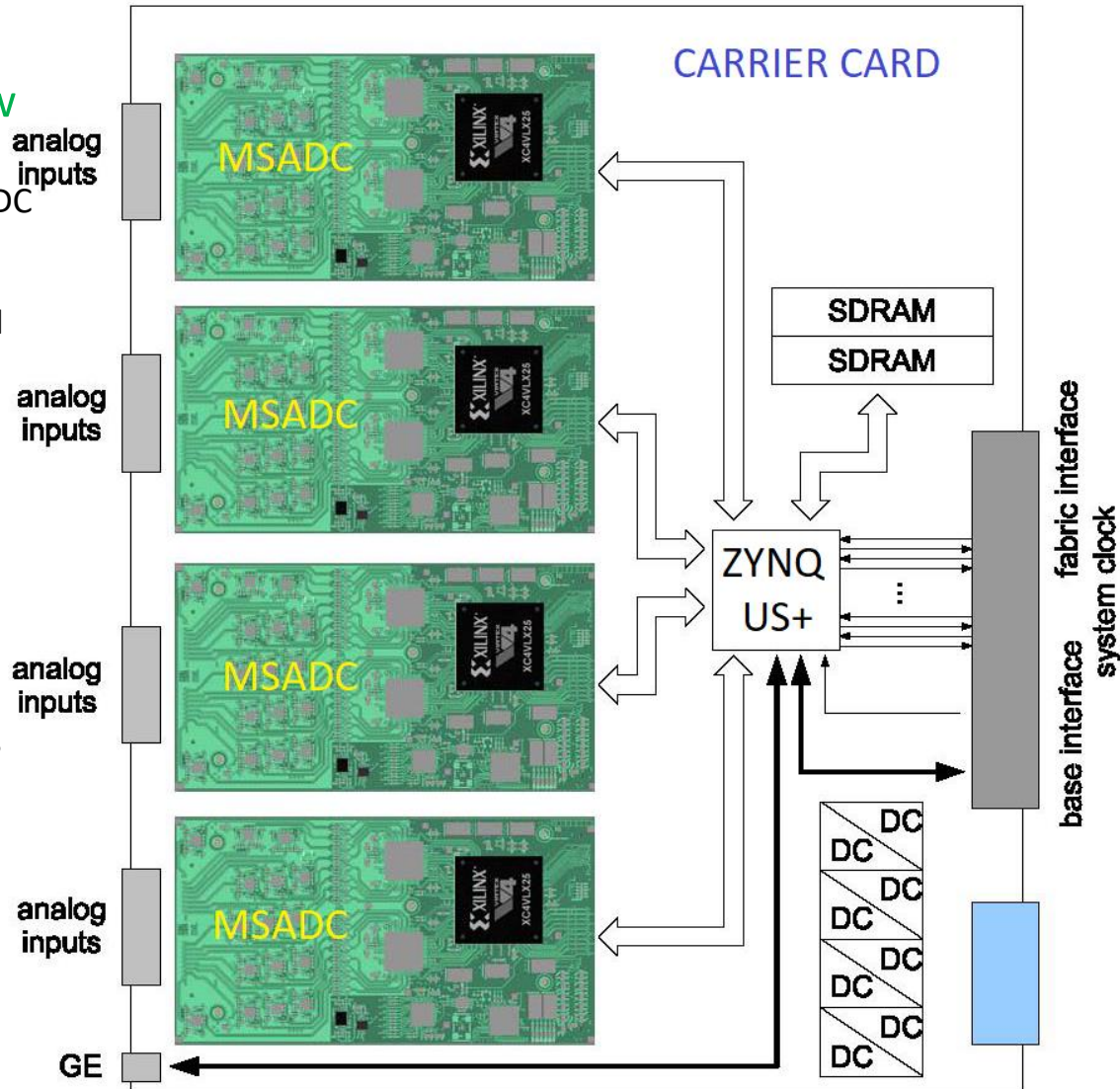
- FPGA Virtex-4 devices in available MSADC boards have limited amount of free resources
- New Carrier Card is equipped with Zynq UltraScale+ FPGA to extract time and charge
- a limited throughput on interfaces between FPGAs

- Low-complexity compression indispensable

- lossless
- Selection with conservative selection – thresholds close to signal noise
- Wide selection window, e.g. 4k samples

- Development conducted for available SiPM waveforms

- Waiting for decoded data from COMPASS experiment
- Methodology is the same - only pulse shapes and SNRs change to some extent



# Compression Methods

- Modeling

- Linear Prediction LPC
- Signal Models
- Transforms



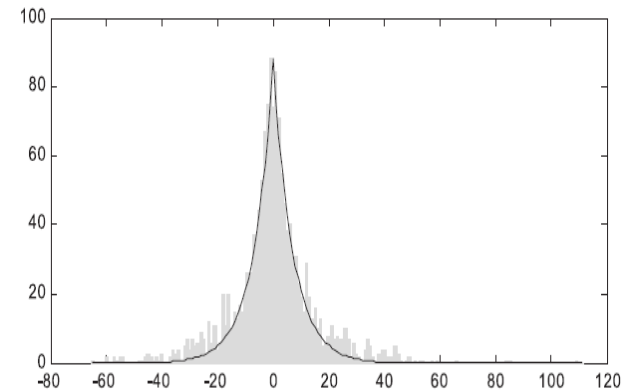
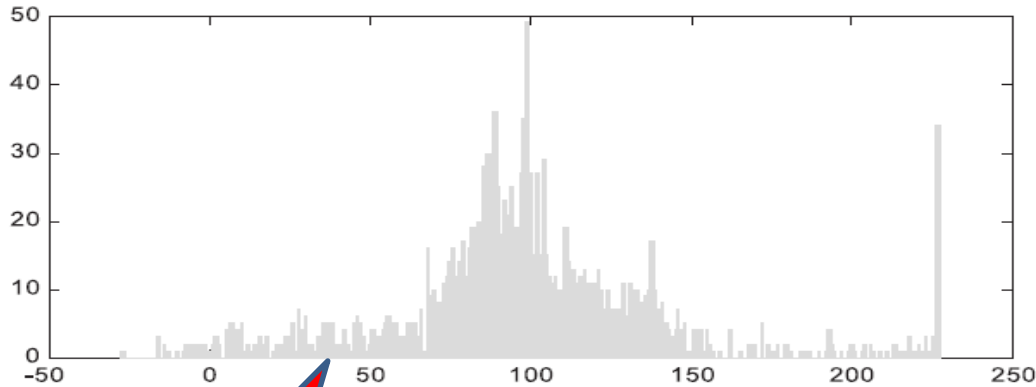
- Quantization (lossy compression)

- Scalar quantization
- Vector quantization – using signal models

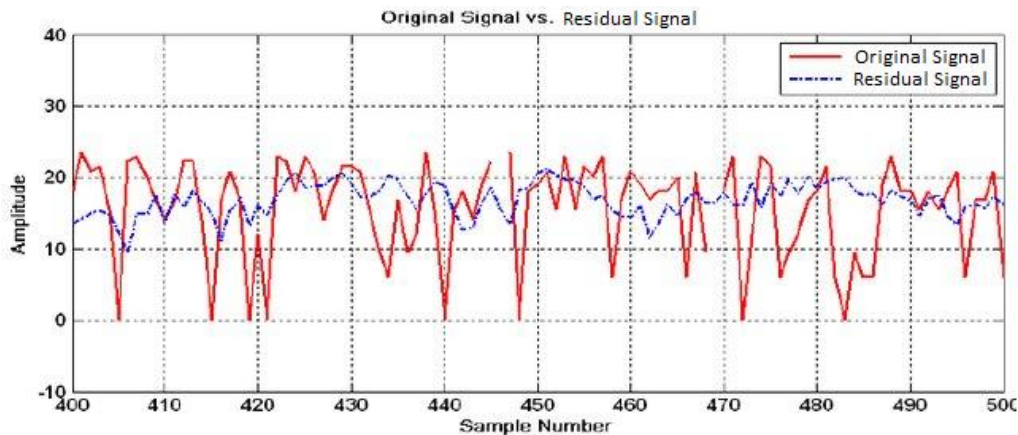
- Entropy Coding

- Variable length coding
- Arithmetic coding – more complex and better compression

# Signal Modelling



Values distributed  
in a wide range



Values concentrated  
around zero

- Predictions and/or Transformations decrease the dynamics
- Distributions of residual signal concentrated around zero
- Signal reconstruction using reverse operations

# Signal Models

- Set of representative sample sequences are compared with acquired samples to find the best matching in terms of SAD or MSE

**SAD**  $i = \arg \min \left( \sum_t |x[t, i] - x[t]| \right)$

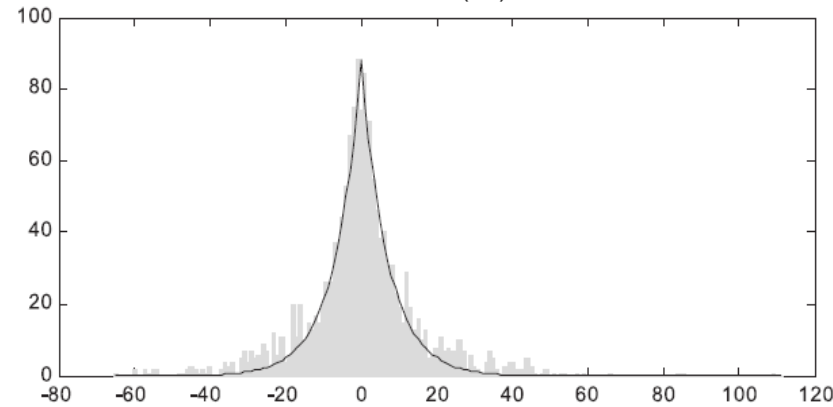
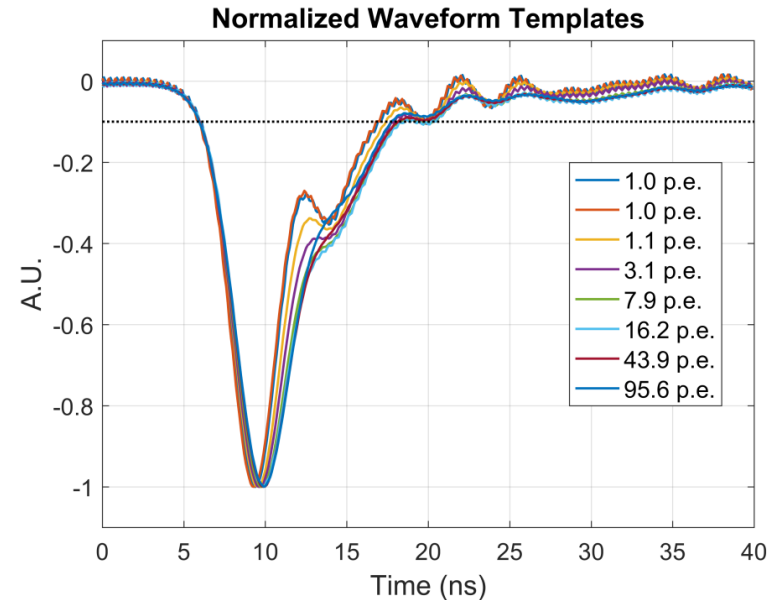
**MSE**  $i = \arg \min \left( \sum_t (x[t, i] - x[t])^2 \right)$

$$x_{predicted}[t] = x[t, i]$$

- Residuals (equal to difference between input samples and their predictions) have much lower values and energy

Compression efficiency similar to LPC

Higher computational cost





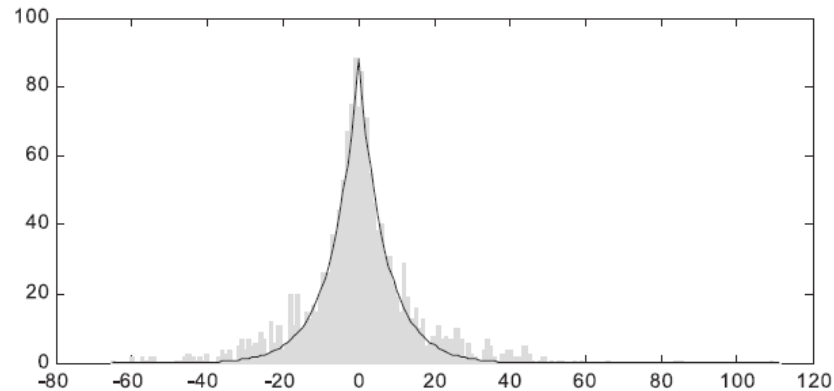
# Linear Predictive Coding (LPC)

- Prediction as a sum of previous samples multiplied by coefficients

$$x_{\text{predicted}}[t] = \sum_{i=1}^N a_i x[t-i]$$

- Residuals (equal to difference between input samples and their predictions) have much lower values and energy

$$\Delta x[t] = x[t] - \sum_{i=1}^N a_i x[t-i]$$

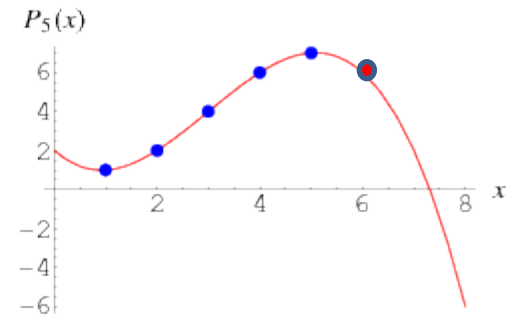
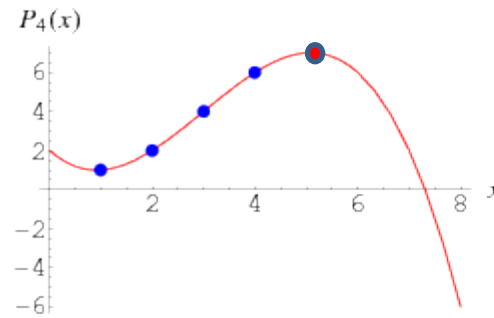
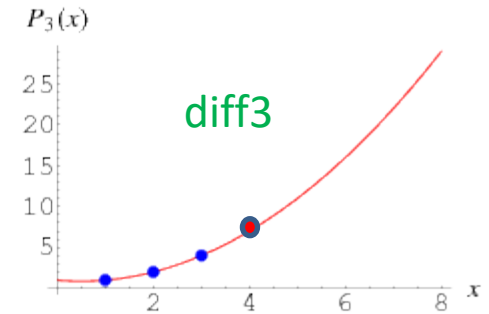
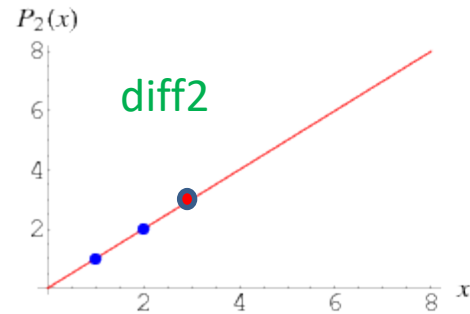


- Coefficients  $a_i$  must be known at the decoder
  - precomputed or sent with residuals

- Error energy: 
$$E = \sum_{t=0}^T (\varepsilon[t])^2 = \sum_{t=0}^T \left( x[t] - \sum_{i=1}^N a_i x[t-i] \right)^2$$

# Prediction based on derivatives

- Polynomial model based on directly preceding samples
- Applied separately or jointly with LPC
- Diff1 Improves slightly LPC by up to 0.05 bps

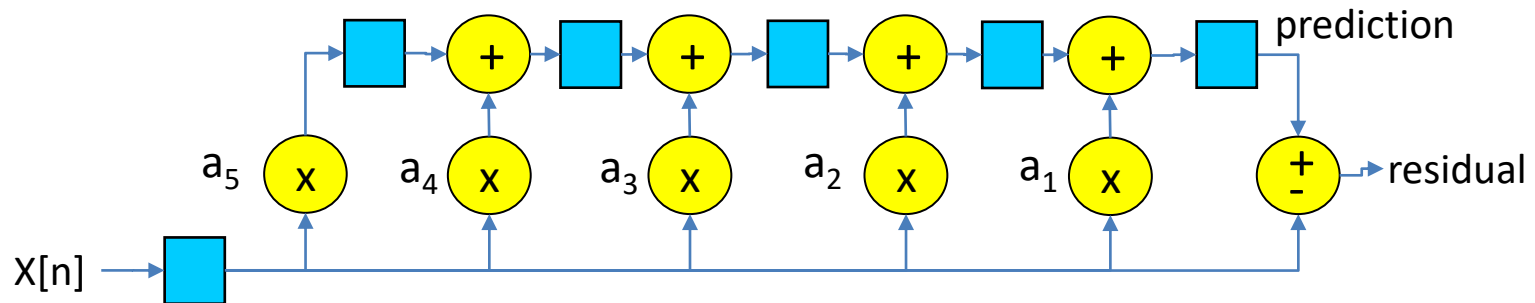


- diff0: signal:  $x[i]$
- diff1:  $x[i] - x[i-1]$
- diff2:  $x[i] - 2*x[i-1] + x[i-2]$
- diff3:  $x[i] - 3*x[i-1] + 3*x[i-2] - x[i-3]$

Assure unbiased predictor for LPC

# LPC architecture

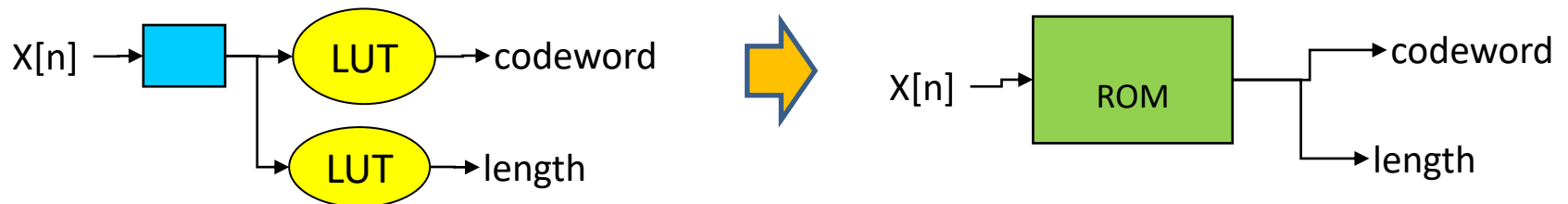
- LPC is  $n$  as a sum of previous samples multiplied by coefficients
- Filter architecture suitable for LPC



- Coefficients  $a_i$  are successive values of the covariance function
- Increasing filter order improves compression efficiency
  - Four-tap filter is usually sufficient to get bit-rates about 5.5 bps
  - Changing order from 4 to 16 reduces bit-rate by up to 0.2 bps

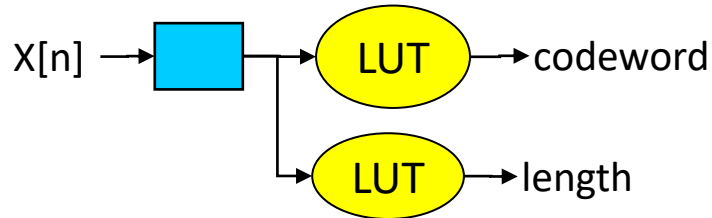
# Variable Length Coding (1)

- Assignment of input values to codewords
- Codewords with variable lengths inversely proportional to probabilities
- Bit rate greater than the information entropy by a small fraction of bit per sample ( $\sim 0.03$ - $0.09$  bps)
- Variable Length Coding is simple in implementation

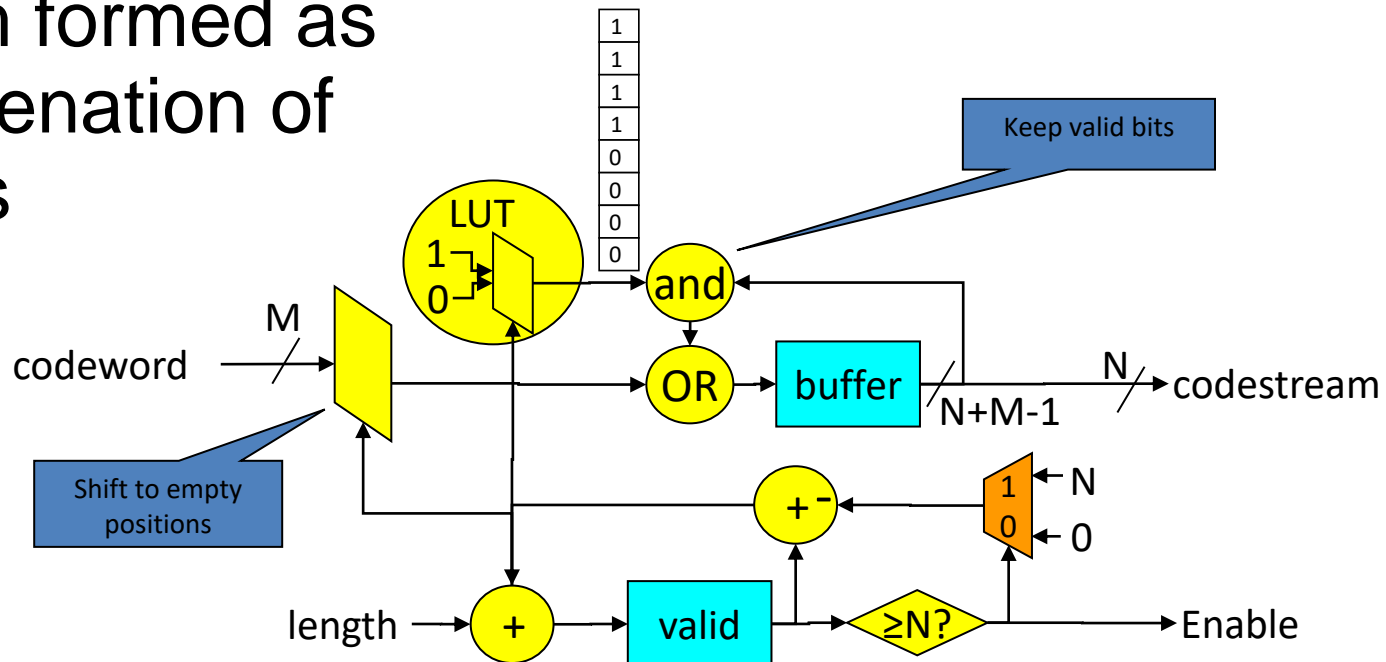


# Variable Length Coding (2)

- Input value directly mapped to codewords and their lengths



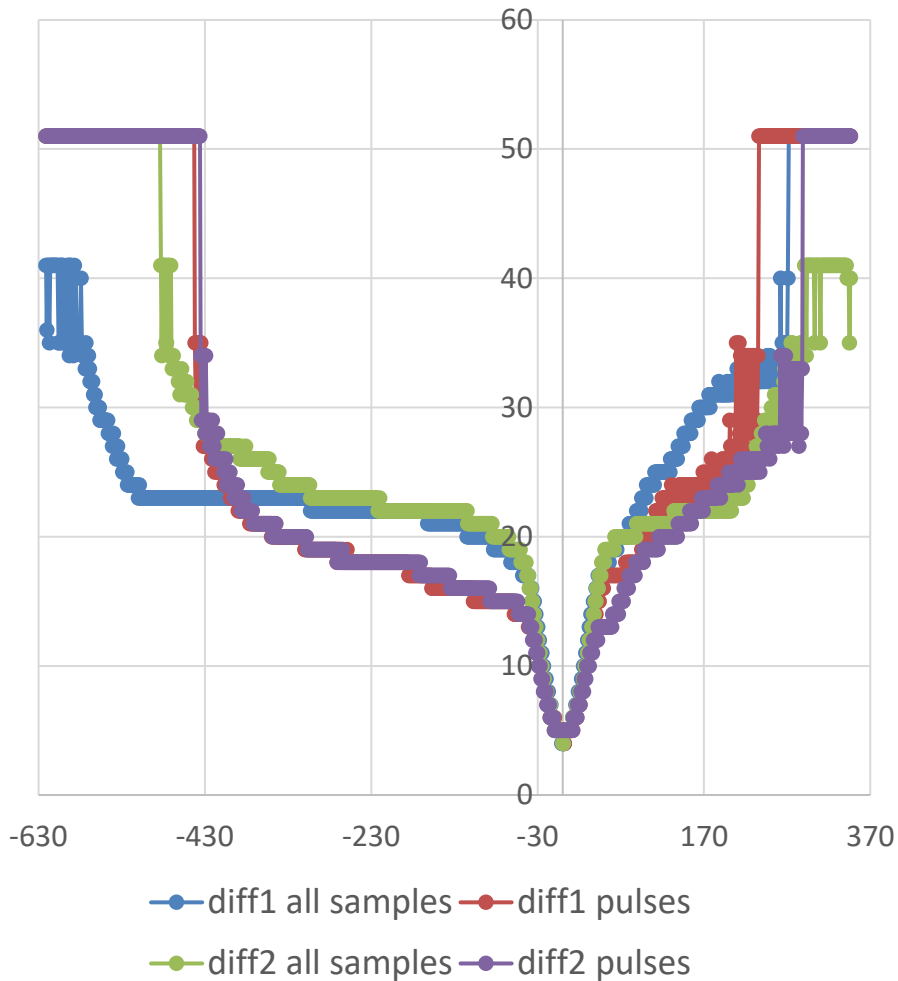
Code-stream formed as the concatenation of codewords



Hardware cost: ~500 LUTs in FPGA

# Variable Length Coding (3)

Example codewords lengths for SiPM



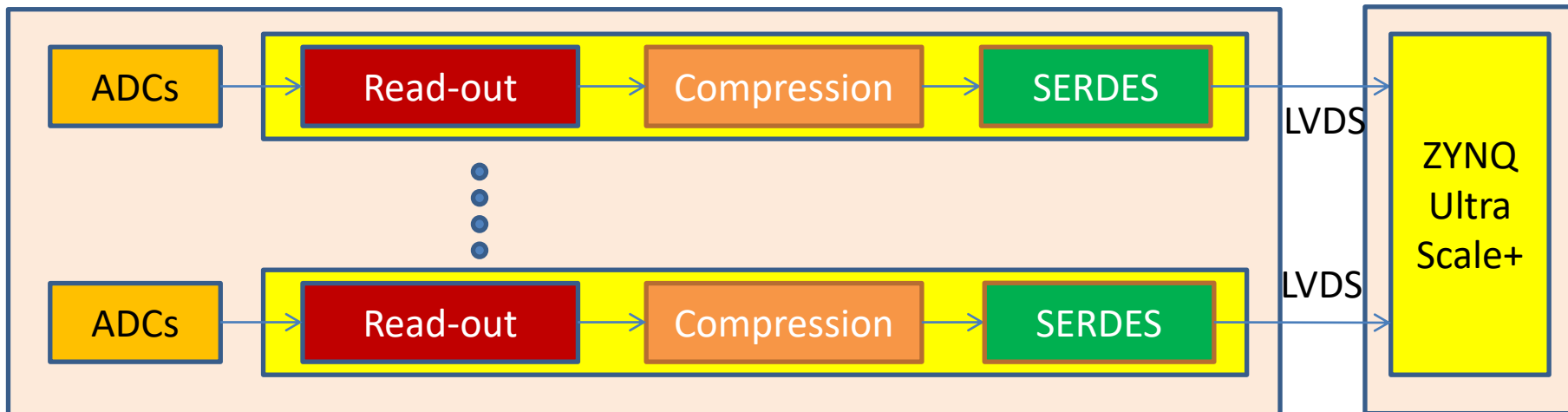
- Short codewords for most probable values close to 0
- Very small probable values can be coded on the fixed number of bits
- Possible division into subranges with separate Golomb codes – suitable for logic
- 1k RAM/ROM sufficient to map all samples

# Compression Efficiency

- Lossless Coding of waveforms
  - Compression ratio: 2 to 3
  - About 5.3 bits per sample (bps) for 14-bit ADC for SiPM
  - It corresponds to input noise std. deviation  $\sim 50$
  - Expected smaller values for 12-bit ADC for ECAL
- Selection of pulses strongly limits the amount of data to compress
  - Compression ratio depends on selection rate

# FPGA – architecture

- Compression in basic version is not complex
  - LPC filter order determines the number of utilized multipliers in DSP units
  - Variable length coding is implemented as LUTs mapped to logic resources or on-chip RAM
  - Estimated resources are below 1k 6-bit-input LUTs per channel
- Parallel processing for 16 PMTs proportionally increases requirements on resources and power





# Summary

- Waveform compression utilizes Linear Prediction Coding (LPC) and Huffman coding
- Lossless compression allows about 5.3 bps for SiPM
- Similar compression ratios for scenarios with all samples and pulses
  - Ratio depends on noise
  - Slightly higher bit-rates for pulses
- Amount of resources is moderate for multichannel implementations
  - Limitations results from multipliers in LPC filters
- Future works:
  - Verify the compression for PMTs/ECALs
  - Hardware design and implementation
  - Optimization of variable length coding by contexts