TECH NOTE :: Analog to Digital Data Path in QuantumX Modules

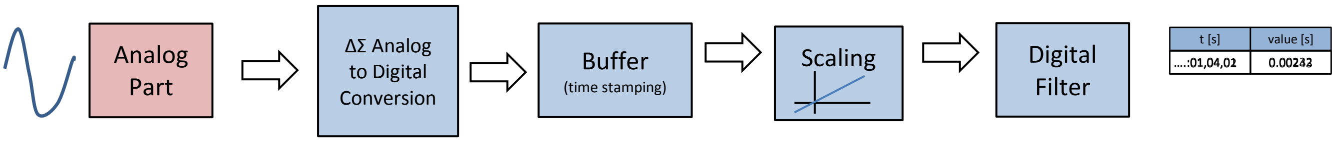
Version: 2016-07-29
Author: Christof Salcher, Product Manager Test & Measurement, HBM Germany
Status: public

Abstract

This Tech Note describes in general the digitalization path in QuantumX data acquisition modules, explaining anti-aliasing filter, analog to digital conversion, digital filtering, scaling, datarate and distribution of data within the data acquisition system or to a data sink whether this is a storage medium or gateway.

The Analog to Digital Data Path

What does digitalizing of analog data mean and what is it for?
The major advantages are as soon of digital information over time is that it can be transferred or copied to different platforms and can be stored, archived and analysed in a quicker way, simulation or real-time control. But digitalization also comes along with the disadvantage of getting discrete signals in amplitude and over time too. But these effects are well known and thus can be handled in a proper way.



Analog to Digital Conversion in General

An analog-to-digital converter (ADC, A/D or A to D) is an electronic circuit part that converts a continuous voltage signal to a digital number. An extra math operation scales the digitalized value to its physical representative. Digital filters allow focusing on a certain frequency spectrum according to the system response I am interested to analyze. The most common way to digitalize analog data is acquiring equidistant values over time with a fix sample-rate and a given resolution.

There are a high number of different ADC technologies available in the market.

- Flash
- Successive-Approximation Register (SAR)
- Sigma-Delta (ΔΣ)
- Ramping
- Pipeline

All technologies are different in its characteristics so finally the **target application** selects the ADC type.

The following characteristics need to be considered for a proper ADC selection:

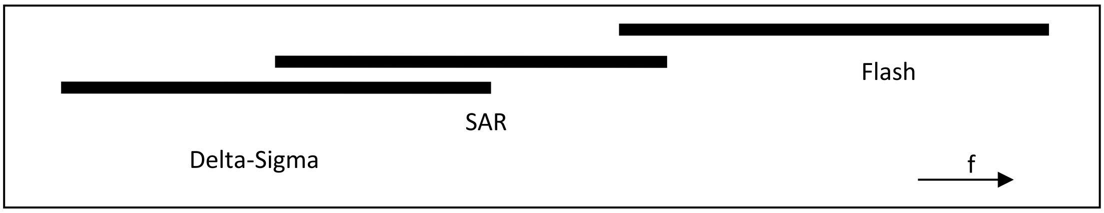
- **Resolution:** number of bits in total
- **Dynamic range**: influenced by many factors, including resolution (i.e. 24 Bit) and accuracy and jitter. The dynamic range of an ADC is often summarized in terms of its effective number of bits (ENOB). As a quick example – the Universal High-Speed Module QuantumX MX410B offers 19 ENOBs, outstanding in the data acquisition market.
- **Settling time and thus Latency**: time between input and data transfer
- **Power consumption**

Let's go into details. We leave out the two special types ramping and pipeline:

- **Flash ADC**: offers a bank of comparators sampling the input signal in a single very fast step in parallel in the gigahertz range. They are regularly power hungry and their analog circuit count increases exponentially with the number of bits, so its usage is typically limited to low-resolution applications such as video, wideband or communication or transient testing.
- **Sigma-delta ADC:** use high oversampling rates and noise shaping techniques to create a high-speed bit stream whose density accurately represents the input signal. The digital representation of the input signal results from the accumulation of a long period of the bit-stream, yielding a relatively slow effective conversion rate. This characteristic makes them useful especially for high-resolution, low- to moderate-speed applications.
- **SAR ADC:** digitizes the input signal step-by-step by running a successive approximation algorithm and determining each bit one by one until the complete conversion is finalized. This architecture reuses the same hardware for each step; therefore, it requires a high clock rate to complete the conversion before the next sampling instant.

ADC Type	Latency	Speed	Resolution	Power consumption
Flash	Low	Very high	Low	High
Delta-Sigma	High	Slow-medium	High	Medium
SAR	Low	High	Medium-high	Low

Table 1: comparison of ADC types



Picture 1: comparison in speed (datarate)

QuantumX Analog to Digital Conversion (ADC)

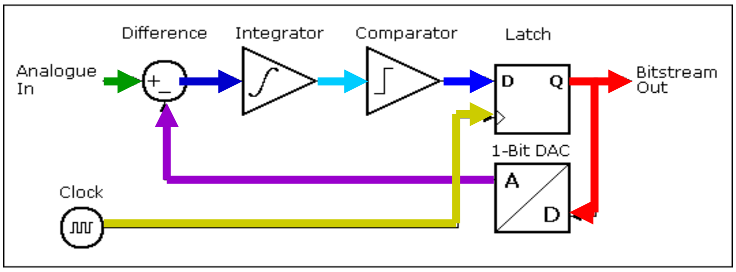
One basic approach in the QuantumX family has been that we wanted to use as less analog components as necessary as they have significant effect on temperature and the modules are all rated -20…65°C (). So in an environment with extreme temperature ranges, analog circuits requiring a high degree of stability can be very complex or require local temperature control.

All data acquisition modules basically offer the same analog to digital data path. In some modules digital filtering and scaling are realized in FPGA technology of software running on the main processor. All inputs are digitalized with a **Delta-Sigma ADC** (ΔΣ, also sigma delta – ΣΔ ADC). Why?

Using Delta-Sigma in QuantumX is the perfect compromise of:

1. High accuracy (ENOB)
2. High speed and dynamic
3. Low power for small distributable boxes with max. 12 Watt

The conversion in a ΔΣ ADC is done by using error feedback, where the difference (Δ) between the two signals is measured and used to improve the conversion. This type of ADC first encodes the analog signal into a one bit stream. The pulsed bit stream goes in a feedback loop and is subtracted in a known interval (voltage x time) from the input; the result is integrated and controls over a comparator the single bit. This fast bit stream is then digitally filtered and forms a higher-resolution digital output but with less bandwidth. The ADC offers 24 Bits which means 2^{24} or 16 777 216 discrete steps over range. The clock frequency in a QuantumX ADC data path is around 5.8 MHz.



The pros and cons of a ΔΣ ADC:

- \+ no passive anti-aliasing filter necessary
- \+ Few passive analogue components (voltage divider) - perfect long term (aging) and temperature stability. No need for gain setting
- \+ High continuous sampling rate with for example 5,8 MHz for MX410B:
  60 times oversampled compared to output or datarate of maximum 100 kS/s. For example, to sample a 30 kHz sine signal, a sample rate greater than the Nyquist rate, which is > 60 KHz, would be theoretically sufficient. However, the ΔΣ ADC in a QuantumX MX modules is using a sampling rate of 6 MHz which means a 100 times higher rate and thus far better anti-aliasing
- \+ Excellent dynamic range (high bandwidth)
- \+ Identical transfer-function channel to channel – early digitalization lead to comparable results
- \+ high accuracy or signal-to-noise-ratio (SNR) with > 100 dB
- — longer conversion time compared to sample-and-hold: > 100 µs and thus not ideal for real-time applications
- — no ideal flat top signal throughput

As the conversion rate and thus the time shift between true analog input and digital output of a ΔΣ ADC is higher than for example a sample-and-hold ADC this technology is not perfect for high speed control applications with much lower than 1 ms turnaround loop and where fast reaction is necessary. This has to be taken in mind when working with the analog inputs. It clearly focusses on high quality of data for online or post process analysis.

The continuous data stream of the ADC output is immediately time stamped and filled in a buffer. This buffer hosts the maximum data rate possible and is the base for digital filtering. After the digital filter the selected data rate is pulled out of the filled buffer. Time stamping is shifted accordingly. Example MX410B: 100 kS/sec max. data rate. Selected data rate of 20 kS. Every 5the value will be taken.

QuantumX – Efficient Number of Bits Calculation (ENOB)

The IEEE standard 1241-2010 defines test methods for ADCs [1]. The Effective Number of Bits (ENOB) can be seen as dynamic performance, true resolution and quality of an instrument [2].

According to this standard the ENOB is calculated in the following way:

$$\mathrm{ENOB} = 0.5\log_2(\mathrm{SINAD}) - 0.5\log_2(1.5) - \log_2\left(\frac{A}{V}\right)$$

With the following definitions:

- V: full-scale range of the device under test
- A: peak to peak amplitude of the sine wave fitted to the output
- SINAD: signal to noise and distortion ratio.

$$\mathrm{SINAD} = \frac{P_S}{P_{NAD}}$$

The following definitions apply:

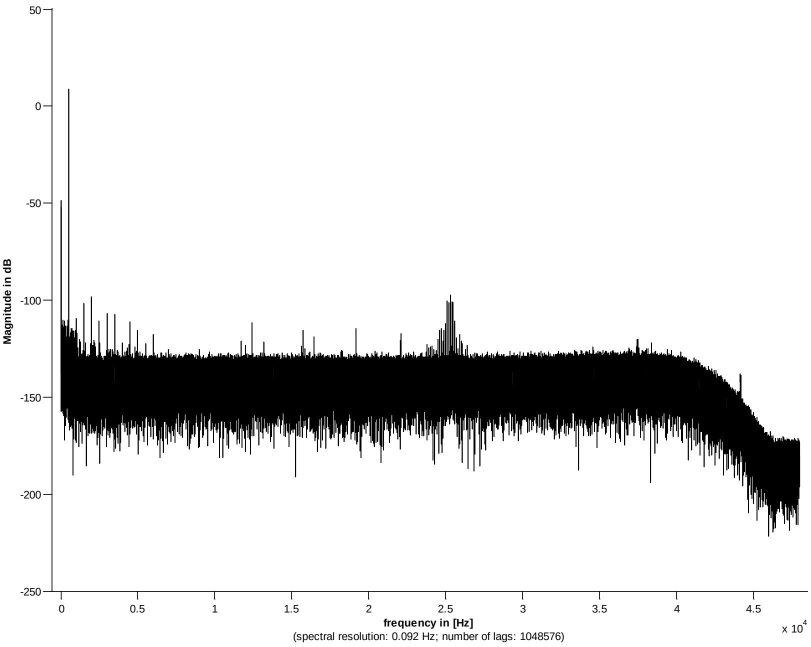
- P_S: signal power; power in the FFT bin corresponding to the input frequency
- P_{NAD}: noise and distortion power; sum of powers in all other frequency bins
- Excluding the 0 frequency bin, up to and including the bin at Nyquist frequency.

SNR and SINAD are ratios of “RMS” (Root Mean Square) values.

Time or frequency domain analysis can be used as methods for calculating SINAD.

Evaluation has been done outside of HBM and based on QuantumX MX410 with the following parameters:

Amplifier	MX410	
Input	Range	± 5 V
	Coupling	DC
	Sampling rate	100 kS/s
	Filter characteristic	Bessel
	Measurement time	10s
	Source	SRS Generator DS360
Input values	Impedance	50 Ohm
	Impedance adapter	50 Ohm BNC adapter
	Signal	Sine
	Amplitude	± 4 V (8 Vpp)
	Frequency	500 Hz



The sine has only limited harmonics. The noise floor is around **-130dB**. There is only a small disturbance around 25 kHz in the spectrum. The signal is bandwidth limited to 40 kHz.

QuantumX – Scaling

QuantumX is used in Test & Measurement applications in the large field of **mechanical, hydraulics, thermal, electrical, noise or mixed systems**. Typical **physical sensor information** is **force, strain, torque, pressure, temperature, displacement, speed, position, acceleration, flow, voltage, current, noise** and many more. All this information can be measured by different sensor or transducer types based on **voltage, current, bridge** (resistive, strain gages or inductive), **piezos (IEPE), LVDT, PT100/PT1000, thermocouple** and many more. Some sensors deliver **digital signals** which need to be decoded like **pulses**, **frequency** or **absolute encoder signals (SSI)**.

However all inputs to the data acquisition modules need to be scaled or linearized. Both expressions are often used as synonyms in the Test & Measurement domain.

Scaling is used to calculate a physical value out of an electrical value.
Example:

Electrical value	Physical value
0 mV/V	0 kN
2 mV/V	10 kN

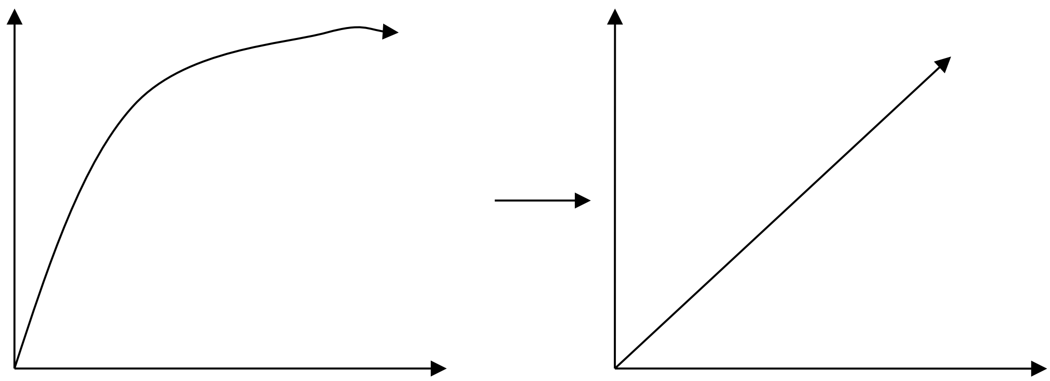
Sensors or transducers do have linear or non-linear behavior.
Some examples for linear behavior within the measurement range and in constant ambient temperature:

- Strain based sensors (load cells, strain)
- Acceleration based on IEPE, piezo-resistive, MEMS
- Temperature based on PT100, PT1000
- Direct voltage measurement (i.e. 1000 V)

Just two points or gain and offset need to be entered into the sensor database.

Some sensors do have **non-linear** behavior within the measurement range:

- Temperature based on thermocouple type K, N, E, T, ….



The following scaling mechanisms are most common:

- Linear
  - Zero point and full scale
  - Gain and offset (ax + b)
- Non-linear or party linear
  - Table based scaling (x1/y1, x2/y2, x3/y3, ….) for load cells measuring pull and push forces
  - Polynomial: $a + bx + cx^2$ in certain sections typical for thermocouples

Modern DAQ systems like QuantumX offer TEDS functionality. This IEEE1451.4 standard identifies the sensor automatically as the sensor datasheet is stored electronically in the transducer or in the plug which parameterizes the input automatically and quick. TEDS offers the following:

- Identification number
- Transducer manufacturer, model and serial number
- Characteristic transducer data
- User settings
- IEEE bridge sensor template
- Physical measured quantity
- Physical unit
- Characteristic curve settings

- HBM inductive displacement
- transducer template
- Physical measured quantity
- Physical unit
- HBM signal conditioning template
- Filter settings
- Zero balance
- HBM user channel name template Comments
- HBM unit conversion template Conversion into display units not allowed for per IEEE standard.

QuantumX – Digital Filter

Using filters is common in test and measurement applications but not in control or automation applications.

The anti-aliasing (AA) filter is set according to the true sample rate of the sigma delta ADC and shall not be considered here..

QuantumX has a pure digital path and thus digital filters.

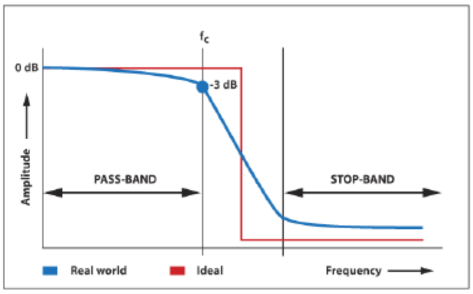
HINT: It is not recommended to work without digital filter. The digital filter is part of the signal path. For dynamic or frequency oriented signal analysis it is recommended to use the highest Butterworth filter, for MX410B 20 kHz.

Digital filters are used for

- Suppressing mains hum or power line hum (50 or 60 Hz)
- Suppression of white noise (i.e. thermal or resistive contact noise)
- Suppressing high frequency noise injected into the lead wires
- Suppression of non-acceptable frequencies (i.e. harmonic vibration / swinging)
- Smoothing signal behavior and improving resolution
- Avoidance of anti-aliasing

There are different types of filters available:

- Low pass
- High pass
- Combination of those as band pass and band stop



In this TECH NOTE we only consider low pass filters.
QuantumX offers the following digital filter characteristics:

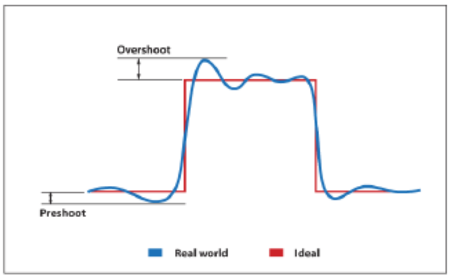
- Bessel
- Butterworth
- Linear Phase

Typically filters are shown in double logarithmic scale in frequency and amplitude.

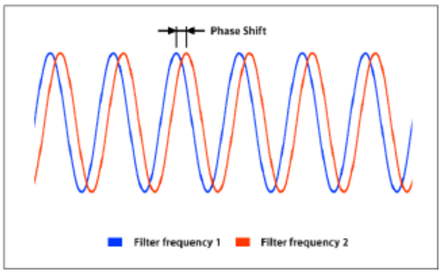
The so called cut-off frequency is defined as the point where the amplitude is reduced by -3 dB (70,7 % of the amplitude and thus ~ 30% reduction). This value has been established internationally. For true measurement specialists the -1 dB point is the value to look at (89% of the amplitude and thus already 10% reduction).

The impulse response behavior is a common criterion for filters.

Impulse response
Depending on the type of filter, effects like pre-shoot and overshoot may appear. These effects appear around a sudden change in the signal like a square wave corner or a fast transient. This is also called an impulse response. Some filter types have zero to none overshoot and pre-shoot; they are said to have good impulse response. Other filters have significant (up to 30%) impulse response. It is very critical to minimize overshoot when doing time domain analysis.



Phase shift
Each filter delays the signal by a certain amount of time, called the phase shift of the filter. The phase shift is a function of the input frequency. In case all channels are filtered the same way, the phase shift is the same. Different cut of frequencies and filter types cause phase mismatch. When doing frequency domain analysis with multiple devices phase shift shall be minimized or equalized. The known phase shift can be compensated by the software (i.e. catman) which allows you true, parallel and phase aligned time based data analysis.



IIR Filter = Infinite Impulse Response do have memory.

Most likely are the characteristics Bessel and Butterworth.
The Butterworth filter

+ is designed to have maximally flat magnitude response in the pass-band.
+ Good all-around performance
+ Rate of attenuation better than Bessel
- Some overshoot in step response.

Typical application is material testing machines with sine based load profiles

Bessel in return

+ comes along with a constant run time
+ best step response-very with very little overshoot
- Slower initial rate of attenuation beyond the pass-band than Butterworth.

Typical applications are measurement and analysis of unknown signal behavior over time. The impulse response is excellent with only little overshoot but with a significant phase shift.

IIR filter allow a high ratio between data rate and cut-off frequency (up to 20,000).

FIR Filter = Finite Impulse Response settle to zero in finite time when stressed by an impulse.

FIR filter do have a fix run time of approximately sample rate multiplied by length of coefficients divided by two.

FIR filters…

- require no feedback
- are inherently stable
- can be easily designed to be linear phase
- are good to use when sampling rate and corner frequency are not far from each other (factor 3…30)

The FIR based Butterworth filter can be used in time and frequency domain work. It offers great slope and phase match, but adds some overshoot / pre-shoot errors to a transient or single shot event.

Linear Phase filter = is usually used in frequency domain but can used in time domain work as well and offers great slope and phase match but adds an overshoot error.

The **steepness** of a filter can be seen in its "order" which in return defines the number of poles of the digital filter. Most typical is 4th, 6th or 8th order (n).
The steepness in the damped section is n*-6 dB/octave.

Example: filter with 6th order and 1000 Hz corner frequency:
- 2,000 Hz signal is 36 dB smaller, which is around 1,6% of amplitude
- 10,000 Hz signal is 120 dB smaller, which is around 10^-6 of amplitude

All digital filters implemented in QuantumX and its curve plots are shown in the datasheets at the very end.

Transfer Function and Frequency Response

If a system behaves dynamically correct the system has to be traced back correctly as well. As the sensor is the interface to the process and without fixing all side conditions, this part always remains nonlinear. To investigate the overall topic HBM took part in the EMRP IND09 Project "Traceable Dynamic Measurements of Mechanical Quantities". EMRP stands for “European Metrology Research Program”, carried out in the framework of EURAMET, the “European Association of National Metrology Institutes”.

The expectations to an amplifier in the frequency domain are quite simple - it is expected that the conditioning amplifier is a “constant” in the system, no matter at what frequency (in a defined bandwidth) the desired signal occurs – means a flat frequency response.

Thus the biggest progress HBM made in the project was in the field of data acquisition, as HBM now provides this system what can claim to be dynamically suitable.

The QuantumX MX410B has been under detailed investigation and after four years of joint research, finally found appropriate by the Physikalisch-Technische Bundesanstalt (PTB), Germanys National Metrology Institute in Brunswig, Germany. One can summarize that QuantumX MX410B and MX430B are suitable for dynamic calibration. More information about this can be found in [3].

Datarate and Aliasing

A signal with a certain frequency digitalized with equidistant data cannot be shown with only double the data rate (Nyquist frequency). The Nyquist frequency is sometimes also called the folding frequency of a sampling system.

Best practice approach
Measure and analyze with maximum datarate and with maximum filter settings. Now analyze the overall frequency spectrum by running a FFT. Then decide what type of filter you need and where the cut-off frequency shall be.
When taking IIR filters select minimum a 6 times higher data rate to the cut-off frequency.
When taking FIR filters take minimum factor 3.

Another interesting approach is this: if you analyze mechanical system dominated by harmonic sine based signals a factor of 10 between cut-off frequency and datarate when analyzing in time and frequency domain is OK. When analyzing electrical signals such as voltage or current of 3-phase engine a factor of minimum 20 or even more is good, as a rule of thumb.

-- end

Literature

[1] IEEE Standard for Terminology and Test Methods for Analog-to-Digital Converters, IEEE Standard 1241-2010
[2] The Effective Number of Bits (ENOB) of my R&S Digital Oscilloscope, Technical Paper, Rohde & Schwarz, 04-2011
[3] Optimal transfer function (frequency response) of conditioning amplifier MX410B, from André Schäfer, HBM