Semester: [SS/WS] 20__ Lab group (DCL/01/02): Team name/number:	Performed tasks: ☐ Hands-on hardware ☐ Hands-on software (Matlab/Simulink) ☐ Presentation	Protocol manager:
Date of exercise:		Other participants:
Professor:	Attestation:	

Digital Communications - Experiment DCL-1 SHN/01.04.2023

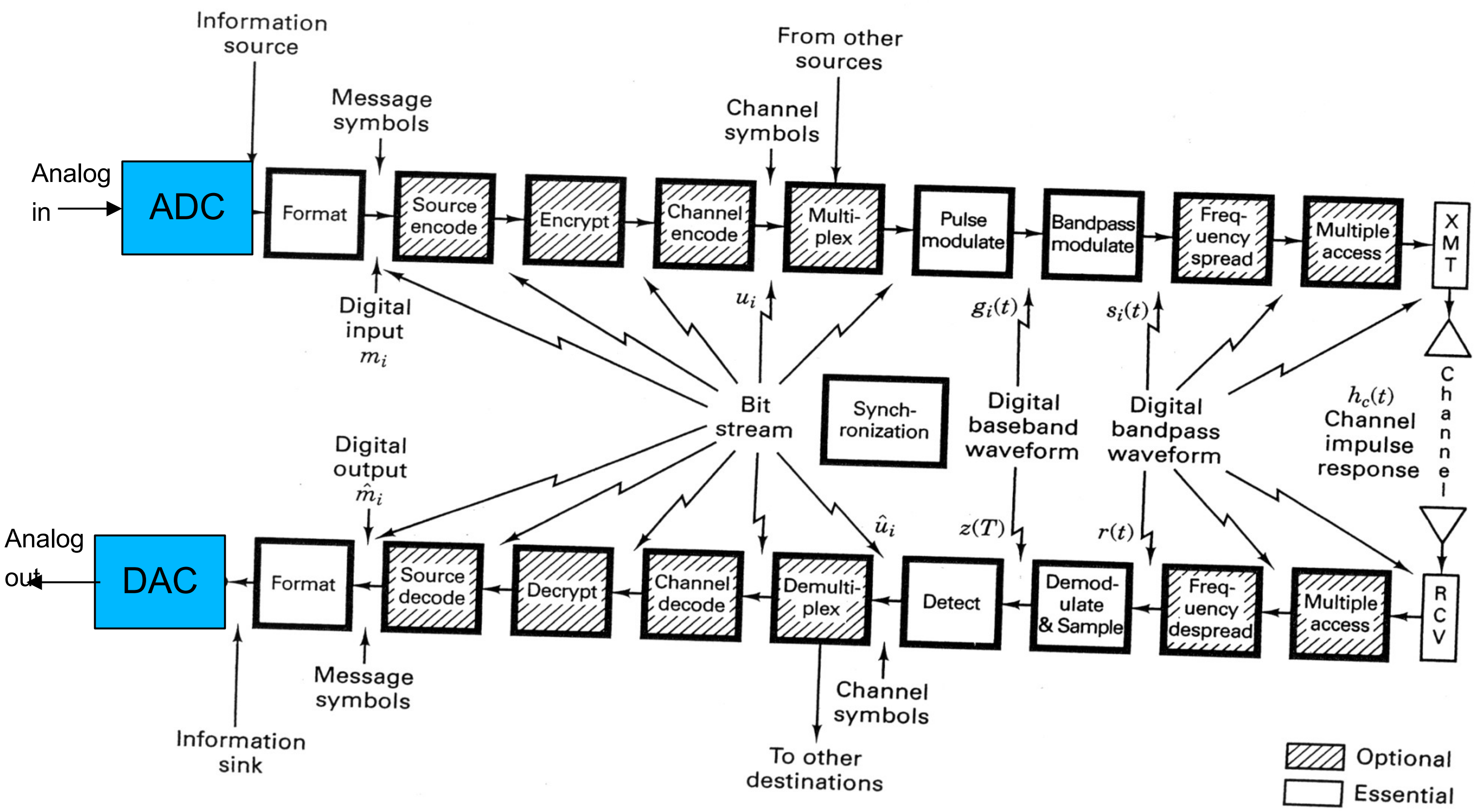
Sampling and Quantization (ADC+DAC)

- ## Introduction

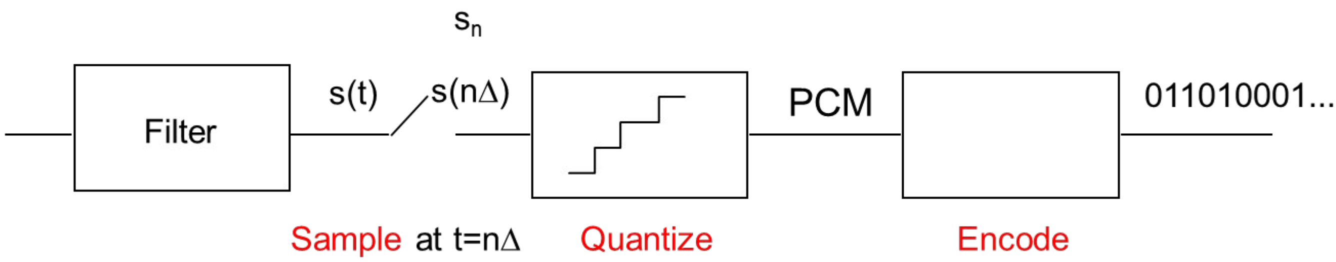
According to the block diagram of digital communication systems, the interface between the analog into the digital domain is realized by an analog-to-digital converter (A/D converter, ADC) and the reconstruction from digital to analog is implemented by a digital-to-analog converter (D/A converter, DAC). An ADC performs two discretization tasks

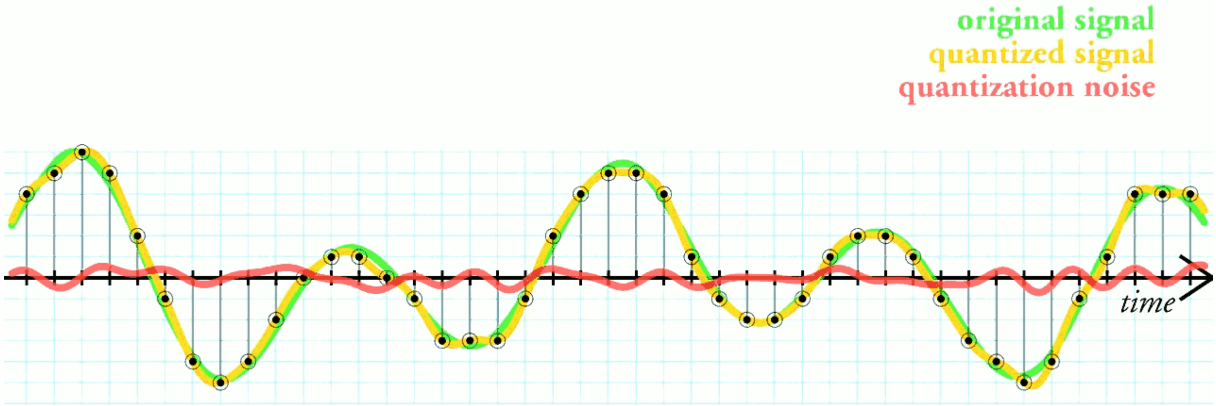
- Sampling (continuous time to discrete time),
- Quantization (continuous amplitude to discrete amplitude, or steps).

Both change the set of possible values from quasi-infinite (floating point values) to integer numbers.



This lab experiment focuses on the following components on the ADC side:





Hardware experiment (electronics) [1/2 of the groups]

A part of this lab exercise is performed using real hardware. Each group should have the chance to work with real hardware at least 50% of the time. Depending on the group size two, three or four people work as a team on one workbench with hardware. Due to the limited nature of the hardware resources, all the other time some experiments are performed in software (see below).

Software experiment (Matlab Simulink) [optional, 1/2 of the groups]

The rest of the time when the (limited resource) hardware is not available for the group, the experiments will be performed using a simulation software toolkit. In this lab we use Matlab Simulink which is very common in all industries and research organizations. Make a copy of the provided files into a work directory on your computer, set the Matlab include path to it and run the initialization script “init_digital_communications.m”,
and then the GUI “digital_communications_gui.m”.
Then follow the steps provided after clicking the correct button of your lab exercise.

You have to make the correct settings for the blocks. <u>Don’t use constant numbers</u> but use variables defined in the file init_digital_communications.m, e.g., for c.sampling_frequency. Be careful of the units: If a frequency is asked for in the unit “radiants/second”, that is $\omega=2\pi f$, not f [Hz]!

Sampling in time (ADC)

Prepare an analog input signal which you will sample during this exercise. Use a sine wave generator (function generator) with adjustable output voltage and verify its output shape, frequency and amplitude with an oscilloscope. If the available generator is the "Agilent 33500B", then beware that the displayed amplitude (voltage) is Û as in U(t)=Û·cos(ωt), and not Vpp (which means peak-to-peak).

a) Adjust the output voltage to match the valid input range of the ADC we are going to use. Hint: It must be Û=2.5V to allow a voltage swing of -2.5V to +2.5V, i.e. 5V total swing.

b) Adjust the frequency to stay inside the base band bandwidth discussed with the staff. Hint: Suitable are 1kHz, 2kHz, 3kHz.

1.1. Sample-and-hold block (if available; subject to cancellations)

a) If available, attach a sample-and-hold unit (S&H) to the source. Use input E1 and output PAM1. On the S&H box, the clock ("Takt") setting should be on 4=external (1=4kHz, 2=8kHz, 3=16kHz, 4=ext., 5=ext/64).

b) Apply a sampling clock (TTL signal, square wave) with a frequency *fs* discussed with the staff (*fs* = 8kHz). You may use the generator "PNG02" with a clock ("Takt") output of 8kHz.

c) Measure its input and output shape on an oscilloscope, as well as the sampling clock (2-3 inputs to the oscilloscope). On which signal do you trigger? Which time base is useful? ____ seconds/unit ⇔ ____ seconds/totalhorizontally.

d) You may adjust the pulse width setting ("τ") and observe the output (1=3μs, 2=6μs, 3=30μs, 4=60μs, 5=120μs).

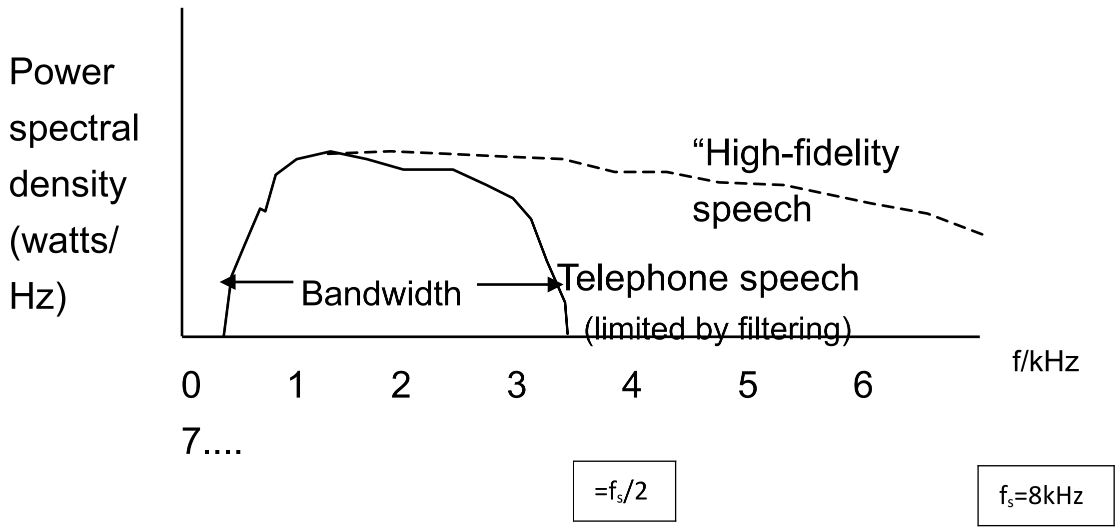
e) Measure its spectrum on a spectrum analyzer, if available, or FFT function of the advanced digital oscilloscope. Where is the dominant input frequency and why is it so close to DC? Can you determine its frequency? What needs to be changed in order to see it better in the FFT?

f) Listen to the output of the sample-and-hold block using headphones or active loudspeakers. You may provide you own audio source later (CD/MP3-player, laptop etc.) or use the sound output of the PC on your workbench.

g) Change the frequency of the input signal to go beyond half of the sample frequency (to 1·*fs* or 2·*fs*) and back. Listen to the output while changing the input frequency and observe it on the oscilloscope. Explain the effect.

h) Change the frequency back to be below *fs* /2 and the signal shape from sine wave to “ramp” (linear increase with time). Verify the shape. Listen to the sampled output.

i) Use another input source which represents a real audio signal like music or voice (e.g. from your Walkman, MP3 player, smartphone, laptop or the lab PC). The following diagram shows the typical bandwidth of a telephony signal. For music you should assume frequencies up to 20kHz.

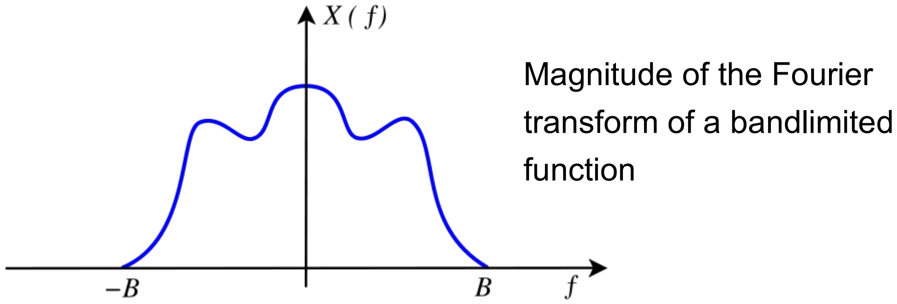


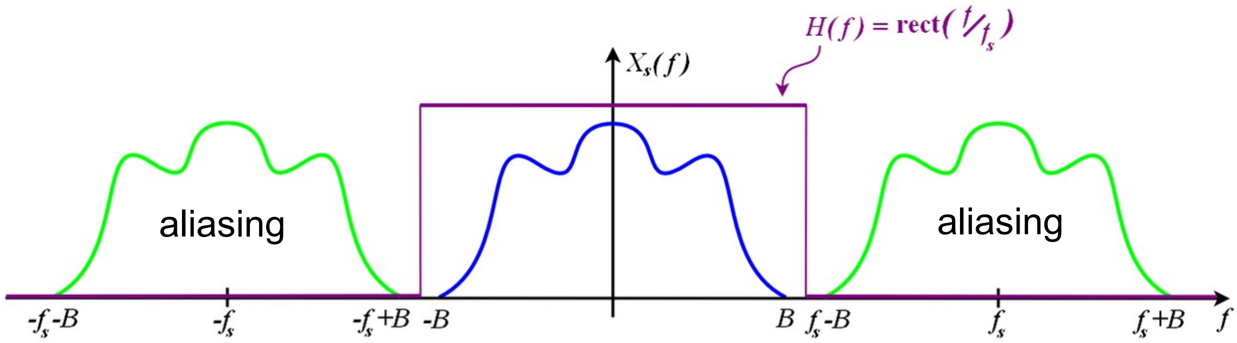
j) Observe the signal on an oscilloscope with channel_A=input and channel_B=output of the sample-and-hold block. Try to use a stationary input signal (using a very long sound sample of one musical instrument or using “A-B-repeat” out of a song).

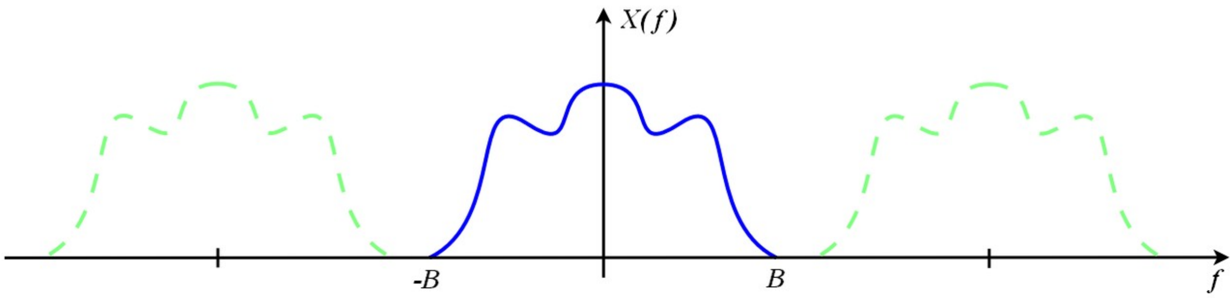
k) Try to get the (power) spectrum of the input and output signal using a spectrum analyzer or the FFT function). Explain the observed effects.

Aliasing effect and Nyquist-Shannon criterion.

> Sampling theorem:
>
> If a function x(t) contains no frequencies higher than B Hz,
> it is completely determined by giving its ordinates at a series of points spaced T=1/(2B) seconds apart.
>
> A sufficient sample-rate is therefore fs=2B samples/second, or anything larger.
> Conversely, for a given sample rate fs the bandlimit for perfect reconstruction is B ≤ fs/2 .







Antialiasing (low-pass) filter

Now use a low-pass (LP=TP) filter ("Butterworth") in front of the sample-and-hold circuit and adjust (if possible) the cutoff frequency such that the input signal is attenuated to at least -40 dB (voltage) or -20 dB (power) at *fs*/2. You may use the box "Butterworth TP" with (filter order) setting n=20 for it.

a) Repeat the experiments above now with the active anti-aliasing filter.

b) Explain the measurement observations and what you hear while listening.

STOP> (breakpoint!)

Discuss your results with the lab staff before continuing to the next step.

Attach the real quantizing ADC

If not done before, attach the real quantizer (ADC) instead of the sample-and-hold unit. It must have (at least) 8 digital outputs plus clock. If possible, set the resolution to the maximum number of bits. How many discrete steps would be there now in the output? Can you measure and observe it? For the (bit) clock you must now 8 times the sample frequency, so fs=8kHz and f_{bit}=64kHz ! Provide this clock to “Bitclock_in”. Observe and verify 8kHz at the output “word clock” (not an input !).

Check the output phases of the bit clock and word clock on an oscilloscope. Trigger on the word clock. Where does the byte start? Which bit is the MSB, and which bit is the LSB ?
Hint: Use a sawtooth=ramp input waveform. Why is the input and sampled signal moving on the oscilloscope?

Reconstruction (DAC)

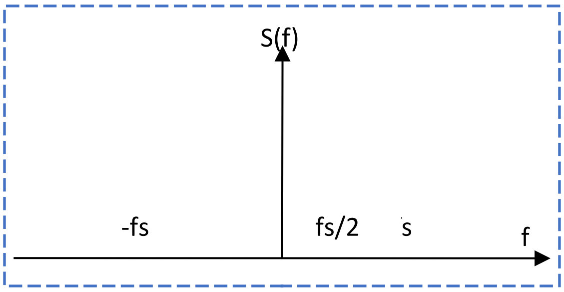
Attach the full-resolution ADC and DAC into the signal path. Connect them using the parallel cable (flat ribbon) connector between “digital out parallel” and “digital in parallel”. When connected via parallel cable, the output “digital out seriell” doesn’t have any signal.
Check the polarity (flank) of the clock and sample signals. Which one is *fs*? What are their frequencies and why are they different?

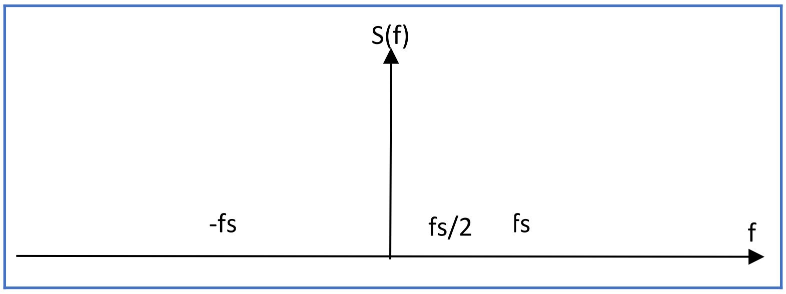
Use a sine wave and/or sawtooth input signal. Not the music input.

Verify that the output of the DAC looks similar to the input of the ADC. Correct if necessary.

Periodic spectrum of discrete-time signals

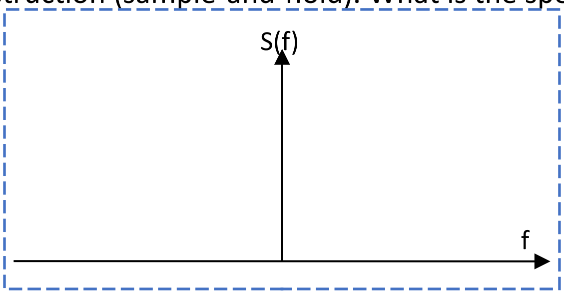
Sketch the spectrum S(f) of the input cosine wave s(t) (frequency *fi*) into this diagram and sketch the spectrum S_s(f) of the discrete time signal s(k) in the diagram next to it:





Sample-and-hold reconstruction

Verify with an oscilloscope that the output of the DAC has the shape of a rect(*t/Ts*) reconstruction (sample-and-hold). What is the spectrum O(f) of the output signal o(t) now like?



Sin(x)/x reconstruction

If possible, apply a different reconstruction/interpolator function now and measure the spectrum again.

Reconstruction by low-pass filter

Use the sample-and-hold output again (rect) and attach a low-pass filter (box “Butterworth-TP”, n=20, fc=4kHz) with cutoff before *fs* in order to reconstruct the signal. Observe the output, measure the spectrum, and explain the results.

Amplitude Quantization (ADC)

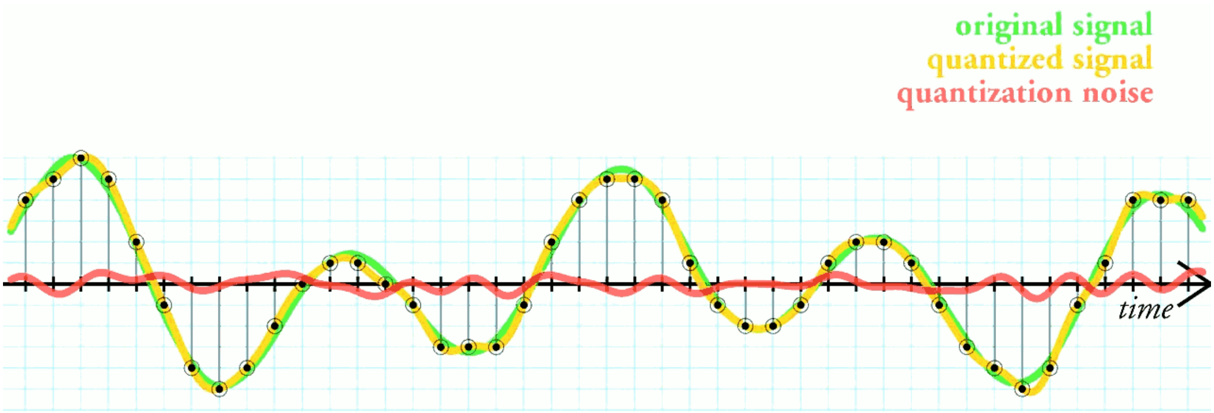


FIGURE 1: THIS GRAPH ABOVE IS WHAT YOU SHOULD EXPECT TO OBSERVE DURING THE EXPERIMENTS.

Effect of exceeding the tolerable input voltage range (optional)

- Increase the input voltage so that the suitable range of the ADC is exceeded ("clipping"). Observe the signal with an oscilloscope and listen to it.
- Change the input to music now and try the same.

Effect of limited quantization steps

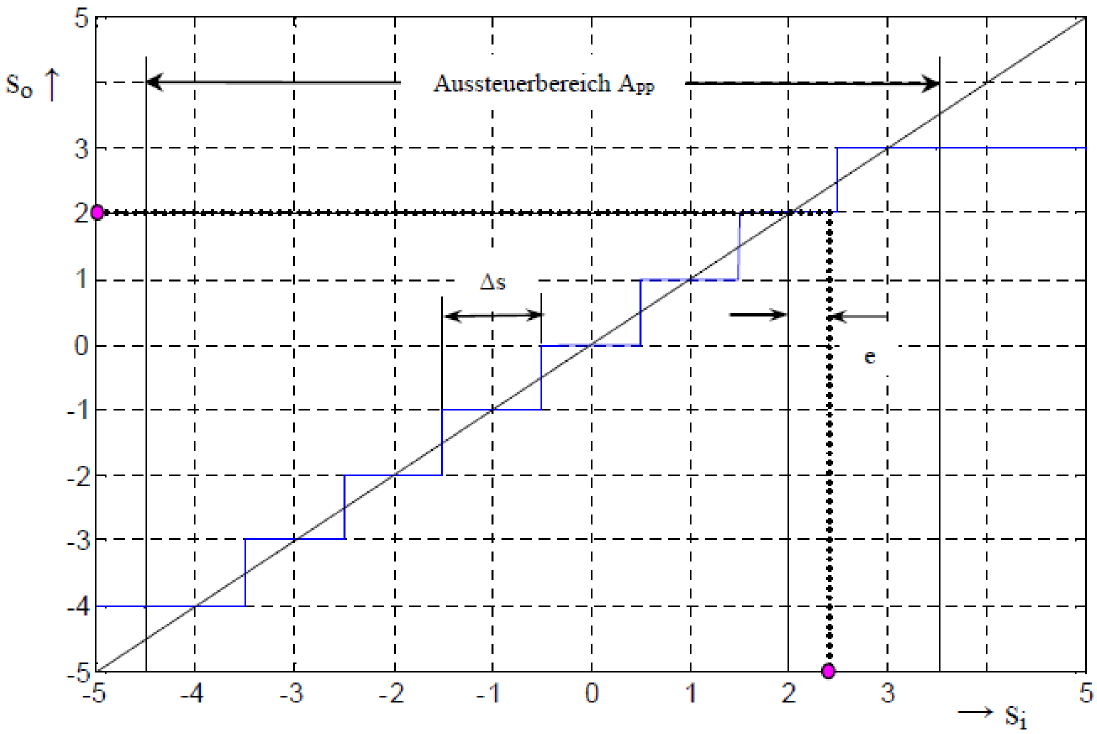


FIGURE 2: INPUT-OUTPUT CHARACTERISTIC DIAGRAM OF A UNIFORM 3-BIT QUANTIZER

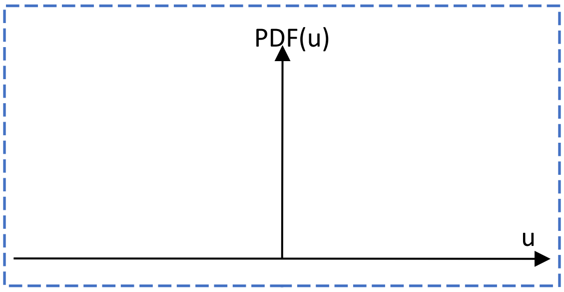
- Now limit the quantizer resolution to 3 bits by switching off the LSBs.
  Hint: Switches D0..D4 down, D5..D7 up

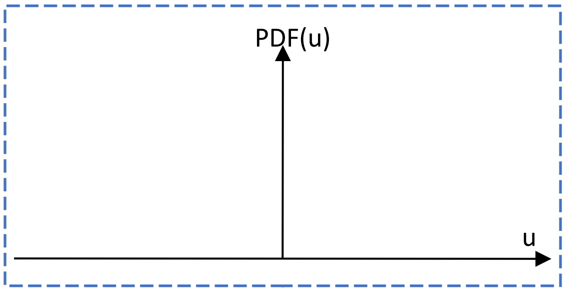
- Try to show the above diagram by using an input sawtooth waveform and the X-Y-diagram of the oscilloscope.

(Probability) distribution function (PDF) of the input signal amplitude

Can you sketch and/or measure the PDF of the input signal in case of

- Sawtooth
- Sine wave
- Gaussian noise?





Can the oscilloscope help calculating the PDF, or histogram?

(Probability) distribution function (PDF) of the quantization noise amplitude

How does the typical quantization error signal look like in time domain?

Measure it. What is the maximum voltage? In dBmV?

How is the PDF of its magnitude distributed? Explain why.

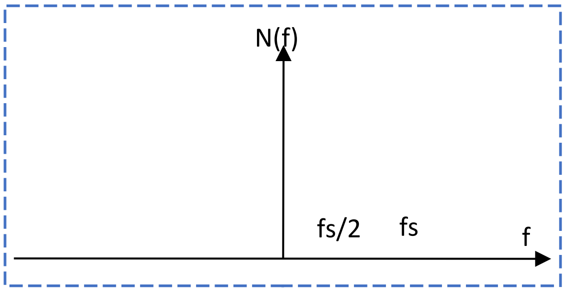
What is the noise power level then? In mW and dBm please, for the 8 bit resolution ADC. Assume R=1Ω.

Spectral distribution of the quantization noise

How does the spectrum of the quantization noise look like?

Sketch it here.

Can you measure it?



Oversampling and digital low-pass filtering (optional)

What happens if the sampling frequency is doubled (from *fs* to 2*fs*)? Does the SNR change?

What if we low-pass filter the discrete signal s(k) to a cut-off frequency of 2*fs*/4 now?

How does the SNR change? Explain!

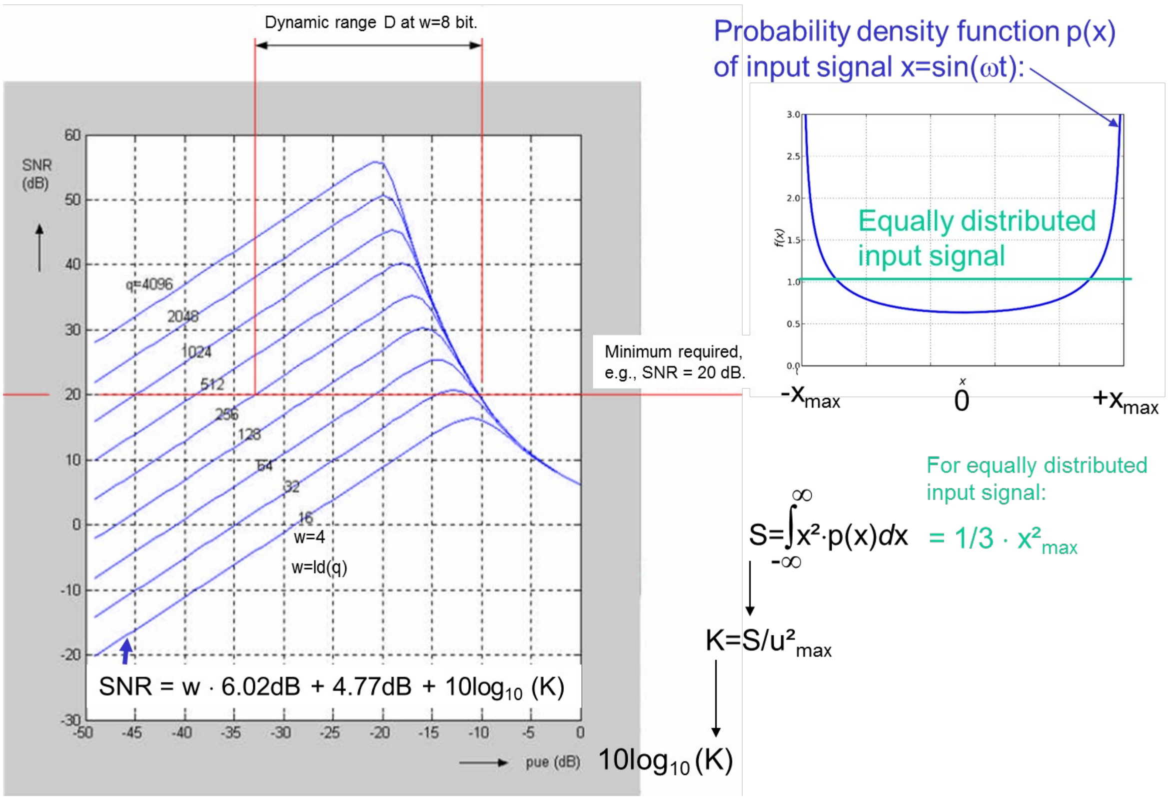
Measuring the distortion (or harmonic) factor or signal-to-noise ratio (SNR)

Design a circuit to measure the SNR of the quantized signal. Which tools can be used for that task?

Supply a sine wave input of maximum tolerable amplitude.

Measure how the DAC output changes with different quantization steps from 1 to 8 bits.

Where in the diagram did you measure precisely? Verify your measurements with the theoretic curve shown below.



Hint: You may use the "Klirrfaktor-Analyser" and the provided band-pass and band-stop filters which are matched=adjusted to the input frequency. One output provides the signal undistortedly (because filtered), while the other output provides all the rest except the dominant frequency.
Take care with the scaling of the outputs. The 0dB ("no attenuation") output might not have the correct amplitude but instead provide a -40dB version (amplitude 1/100) of the distortion.

Write your results into the report and return it before the next lab appointment.