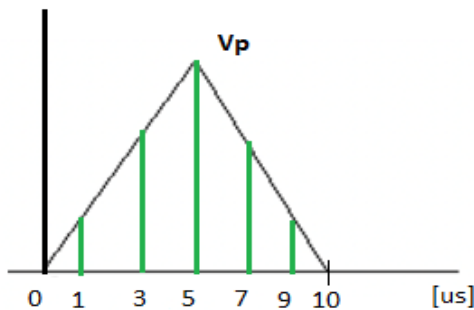


Problem 1

- 1) The Fourier transform of the triangular signal is a square sinc with the first zero in $1/T_p = 1/5\mu s = 200\text{kHz}$. Most of the frequency content of the signal is within 200kHz. To be conservative, we can use a preamplifier with $f_{pa} = 10 * 200\text{kHz} = 2\text{MHz}$. A reasonable value for input referred voltage noise is $\sqrt{S_V} = 10\text{nV}/\sqrt{\text{Hz}}$. Given the preamplifier bandwidth limitation, the output noise has a Lorentzian spectrum with $\tau = 80\text{ns}$. This time is much lower than the minimum sampling time of the microcontroller so noise samples are always **uncorrelated**. To implement a GI in the digital domain, we must give the same weight to each sample. Exploiting the maximum sampling frequency of the microcontroller, we can acquire 5 samples. To maximize the SNR we must acquire the signal peak, so we must acquire the samples corresponding to the green lines shown in the following figure:



The first half of the signal can be described as follows:

$$S_{\text{ANALOG}} = V_p/T_p * t \text{ for } 0 < t < T_p$$

$$S_{\text{DIGITAL}} = V_p/T_p * k * T_s/2 \text{ for } k \in \{0, 1, 2, 3, 4, 5\}$$

Therefore, giving the same weight A to all samples we get

$S = V_p/T_p * A * T_s/2 * (1^2 + 3^2 + 5^2)$, where 1 and 3 have been doubled to take into account the second (descending) part of the signal.

$$N = \sqrt{5 * A^2 * S_V * \frac{\pi}{2} f_{pa}}, \text{ where 5 is the number of acquired samples.}$$

As a result we obtain $V_{pmin} = 15.2\mu V$.

- 2) Being able to choose any weight for each sample, we can design a filter that has the same shape of the signal in order to implement a digital version of the optimum filter. In this case the weighting function is equal to 1, 3, 5, 3, 1 for the 5 samples.
 $S = V_p/T_p * T_s/2 * (1^2 * 2 + 3^2 * 2 + 5^2) = V_p/T_p * T_s/2 * 45$
 $N = \sqrt{(1^2 * 2 + 3^2 * 2 + 5^2) * S_V * \frac{\pi}{2} f_{pa}}$
 In this case, $V_{pmin} = 13.2\mu V$.
- 3) The digital weighting function tends to its analog counterpart as the sampling frequency tends to infinite. See theory for more details.

Problem 2

- 1) Wheatstone bridge configuration with 4 Strain Gauges (SGs): 2 SGs are used to collect as much signal as possible, while the remaining 2 are used for temperature compensation (see theory for the placement of the SGs on the component under test).

With a SG featuring a $R_0=100\Omega$, we can use a bias $V_a=10\text{mV}$ exploiting the whole power budget.

A typical Gauge factor G for the SG is 2.

The differential signal at the input of the preamp is $V_a * G/2 * \epsilon$, where ϵ is the applied strain.

- 2) A filter is necessary. Being the signal intrinsically modulated (it's a square wave centered around zero) we can use a LIA. The best solution is demodulating with ± 1 square wave ($f_{\text{demod}}=500\text{Hz}$, duty cycle 50%) and a LPF with $f_p=10\text{BW}_s = 100\text{Hz}$. As a result, $\epsilon_{\text{MIN}} = 5$ microstrain.

- 3) $1/f$ noise affects the first and the third harmonic of the signal. To minimize its effect we must change the bias of the Wheatstone bridge to modulate the signal, i.e. $f_a=20\text{kHz}$ to avoid $1/f$ noise; by using $\pm V_a$ we keep the same power dissipation.

In this case we need two nested LIAs to extract the signal: the first one with ± 1 demodulation and $f_{\text{demod}}=20\text{kHz}$ followed by the same acquisition scheme of point b) thus obtaining the same ϵ_{MIN} of point b).