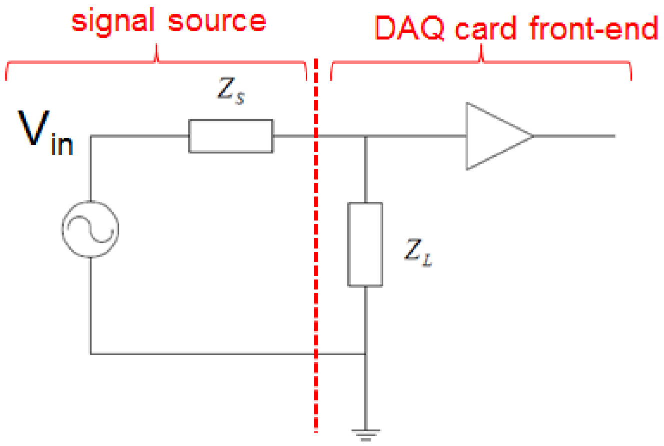
Exam solutions FYS3240/4240 2015

Problem 1

a)



Voltage divider:

$$V_{out} = \frac{Z_L}{Z_s + Z_L} \cdot V_{in}$$

From the voltage divider equation we can se that we need to make sure that the DAQ device has a much higher input impedance Z_L than the signal source impedance Z_S, in order to get Vout ≈ Vin.

Connection options are single-ended or differential signals. Differential connection, if suitable, is usually the best option. Non-References single-ended (NRSE) can also be used.

b)

- DMA enables more efficient use of interrupts, and <u>increases data throughput</u>.
- The CPU can do other work.
- Less likely to get an overflow error in the DAQ card FIFO buffer.
    - that the First In First Out (FIFO) memory buffer onboard the data acquisition card has reached its maximum capacity for storing acquired samples and can no longer accept new samples. An overflow error is symptomatic of a bus transfer rate that falls short of the requested data input rate.

c)
RAID in general:

- RAID = Redundant Array of Independent Drives.
- RAID is a general term for mass storage schemes that split or replicate data across multiple hard drives.
    - To increase write/read performance and/or increase safety (redundancy)

RAID-5:

- Increases write/read performance (but less than for RAID-0) and also increase safety (due to redundancy).
- Parity data distributed on all disk drives (single parity).
- Write overhead because of the additional parity data that has to be created and written to the disk array.
- Can only tolerate one drive failure (array continues to operate with one failed drive).
- As vulnerable to data loss as a RAID-0 array until the failed drive is replaced and its data rebuilt (can be a problem for arrays with large disk drives due to long rebuild time).
- Gives less space for measurement data (due to parity information).
- The minimum numbers of disks are 3.

d)

- Can use <u>unbuffered file I/O</u> to <u>increase write/read speed for RAID systems</u>
  - disable buffering in Windows API
- Optimizes streaming applications.
- Important for RAID systems.
- Must read from or write to the file in integer multiples of the disk sector size (e.g. 512 bytes for a disk sector size of 512)
- The data can span multiple sectors but must fill each sector completely.
- If the data is not a multiple of the sector size, you must pad the data with filler data and delete the filler data before the data is used.

Problem 2

a)
V_{out} = 1.25 V + 0,125* V_{in} (using superposition)

b)
Analog input signals can be unipolar or bipolar. Unipolar signals swing between zero and positive full-scale, while bipolar signal swings above and below some reference point, typically ground.
In battery powered embedded systems **single-supply circuits** are often used to save power. Therefore a level shifting is required to convert a bipolar signal into a unipolar signal.

c)
If a CPU is used the power consumption P is given by:$\mathbf{P = ACV^2f}$ (P is consumed power, A is active chip area, C is the switched capacitance, f is frequency and V is voltage). For this question it is sufficient to state that the power consumption is proportional to the supply voltage squared and the frequency; $P \sim V^2 f$: Information about how CMOS works wrt power consumption is not required, but is relevant.

d)
I2C is a <u>multi-master</u> serial computer bus (but only one master at a time) that uses only <u>two bidirectional</u> lines. SPI on the other hand is a <u>"four-wire" serial bus</u>, and only one master. So, the biggest advantage with I2C is the lower number of signal lines.

e)
Three satellites are need for three distance measurements (Measure “signal time of flight” : s = v * t) The fourth satellites is needed in order to remove clock errors in the receiver unit, since GPS satellites have calibrated atomic clocks but a GPS receiver has only a standard quartz crystal oscillator.

- An error of 1 ms in the receiver clock gives a position error of
  $3*10^8$ m/s * 1 ms = 300 km

A more mathematical explanation (not required):

The measured pseudorange P from a satellite k can be expressed as (since we can assume no clock error in the satellite):

$$\tilde{P}^k = \sqrt{(X^k - x)^2 + (Y^k - y)^2 + (Z^k - z)^2} + d + v$$
$$= \rho^k + d + v$$

where d is the position error due to receiver clock error, (X^k,Y^k,Z^k) is the known position of satellite k, (x,y,c) is the true receiver position, and v is zero mean Gaussian white noise with variance $\sigma^{2.}$ Solving for the unknowns *x, y, z* and *d* requires minimum four equations and therefore minimum for satellites.

f)
IRIG-B is a serial time code, there are both AM and DC versions of the code. (The IRIG-B signal is usually generated from a GPS receiver). IRIG-B gives time accuracies in the order of a few microseconds or better.

g)
All bit in register PORTB.DIR set to ‘1’.
PORTB.DIR = 0xFF; % 0x specifies a HEX number

Problem 3

a)
For instance:

$$\hat{x} = \left(\frac{{\sigma_2}^2}{{\sigma_1}^2+{\sigma_2}^2}\right) z_1 + \left(\frac{{\sigma_1}^2}{{\sigma_1}^2+{\sigma_2}^2}\right) z_2$$

Writing the linear combination of z1 and z2 as:

$$\hat{x} = k_1 z_1 + (1 - k_1) z_2$$

and state that this is a complimentary filter would not give full score alone, since the determination of the coefficients k1 and k2 is then missing.

Coefficients k1 and k2 can relatively easy be remembered by looking at what happens in the two

cases:
$\sigma_1{}^2 = \sigma_2{}^2$, or if σ_1 or σ_2 is equal to zero

b)
The idea behind the complementary filter is to take slow moving signals and fast moving signals and combine them. The filter is based on an analysis in the frequency domain.
The complementary filter fuses the sensor1 and sensor 2 data by passing the former through a 1st-order low pass and the latter through a 1st-order high pass filter and adding the outputs.

(Examples: balance robot and INS/GPS integration).

c)
The Kalman filter is a prediction (time) - correction (measurement) algorithm. The estimated state vector (x_hat), the unknowns, are equal to the sum of the predicted state vector x_bar and the difference between sensor measurements z and the predicted measurements Hx_bar, multiplied with a gain K.

The predicted state vector x_bar is calculated from a mathematical model of how the state vector x varies with time.

If the gain K is zero the estimated state vector is set equal to the predicted state.

The gain K is given by the relationship between the uncertainty in the measurements (measurement noise covariance matrix R) and the uncertainty in the state model (process noise covariance matrix Q).

The states x in the Kalman filter can be time varying (an also the matrices H and K)

d)
A small measurement uncertainty (small R) gives a large gain K, and therefore a fast response to new measurements. (See also question c). Telling the Kalman filter that the measurement uncertainty is small is the same as telling the filter to trust the measurements. If we also said that the model uncertainty Q was large, the filter will give the information from sensor measurements much higher weight than the information from the mathematical state model.