

Written exam in  
Signal processing, 5 ECTS

Monday, January 9, 2017  
9.00 – 13.00

**Read carefully:**

- Remember to write your **full name and study number on every sheet** you return!
- Write **legible** with a pen that allows your answers to be scanned electronically.
- The answers concerning analog filters, digital filters and spectral estimation have to be written on **separate sheets**.
- Problems are weighted as listed.
- Results **without sufficient explanations will not give full credits!**
- All ordinary tools may be used e.g. books, notes, programmable calculators and laptops.
- All electronic communication devices **must be turned off** at all times.
- **Communication with others is strictly prohibited.**

**EIT5/ITC5 Signal Processing / Analog Filters****Written examination Jan. 09 2017**

Note: You may use the following equations:

$$\cosh(x) = \frac{e^x + e^{-x}}{2}, \quad \cosh^{-1}(x) = \ln\left(x + \sqrt{x^2 - 1}\right), \quad \sinh(x) = \frac{e^x - e^{-x}}{2}, \quad \sinh^{-1}(x) = \ln\left(x + \sqrt{x^2 + 1}\right).$$

**A.1 (Weight 12%)**

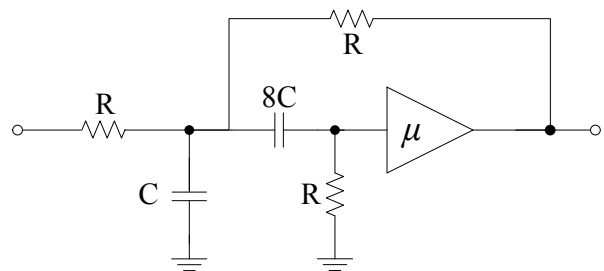
A band-pass filter has to fulfill the requirements:

- Chebyshev characteristic with 1.0 dB ripple.
  - Max. 1.0 dB attenuation at frequencies between 900 and 1000 Hz.
  - Min. 40 dB attenuation at 800 Hz.
  - Min. 30 dB attenuation at 1100 Hz.
- a. Make a sketch (graph) of the band-pass filter.
  - b. Find the necessary order of the filter using the LP  $\leftrightarrow$  BP frequency transformation.
  - c. Find the actual attenuation at 800 Hz.
  - d. Find the actual attenuation at 950 Hz. (Note: The Chebyshev Cn-function expressions are different in the passband and stopband)

**A.2 (Weight 9 %)**

The band-pass filter shown has the transfer function:

$$H(s) = \frac{s \frac{\mu}{RC}}{s^2 + s \frac{25 - 8\mu}{8RC} + \frac{1}{4R^2C^2}}$$



The component values are:

- $R = 5 \text{ k}\Omega$
  - $C = 2 \text{ nF}$
  - Gain of the gain block,  $\mu = 3$ .
- a. Find the centre radian frequency,  $\omega_0$ , and the Q-value.
  - b. Find the gain at the centre radian frequency.

**A.3 (Weight 12 %)**

A band-pass circuit has the transfer function:

$$H(s) = \frac{s \frac{\mu}{4RC}}{s^2 + s \frac{12 - 2\mu}{6RC} + \frac{1}{4R^2C^2}}$$

- a. Find an expression for the sensitivity of the Q-value with respect to changes in the gain block gain  $\mu$ ,  $S_\mu^Q$ , and its value for  $Q = 5$ .
- b. Find a simple expression of the transfer function,  $H(j\omega_0)$ , at the centre radian frequency.
- c. Find an expression for the sensitivity of the transfer function at the centre radian frequency with respect to changes in the gain block gain  $\mu$ ,  $S_\mu^{H(j\omega_0)}$ , and its value for  $Q = 5$ .

**Problem B.1 (weighted with 10% - Digital filters)**

Design a high-pass finite impulse response (FIR) filter with linear phase using the window method. Use a sampling frequency  $f_s=10\text{kHz}$ , a -3dB cutoff frequency of  $f_c=2.5\text{kHz}$ , a rectangular window and an order of  $M=5$ .

Questions:

- Determine the filter coefficients.
- What is the gain in dB at 0 Hz (DC)?
- Draw a signal flow graph for the filter showing the actual coefficients and delays
- Draw a transposed version of the signal flow graph
- Determine the initial 6 values of the step response for the transposed filter

**Problem B.2 (weighted with 14% - Digital filters)**

A digital filter is characterized by:

- Two complex conjugate poles at:  $p_1, p_2 = 0.1848 \pm j0.4021$
- Two real zeros at:  $z_1, z_2=1$

The sampling frequency is:  $f_s=10\text{kHz}$

Questions:

- Make the pole-zero plot for the filter
- Sketch the amplitude response by using vector considerations. Describe how you arrive at the sketch.
- Determine the transfer function  $H(z)$
- Scale the filter such that the gain at 5kHz (half of the sampling frequency) is 0 dB
- Draw a signal flow graph for the filter as a single second order section. Include all coefficients in the figure.
- Determine the initial 5 samples of the impulse response

**Problem B.3 (weighted with 10% - Digital filters)**

An analog second order Butterworth low pass filter with a -3dB cut-off frequency of 2kHz is described by the transfer function:

$$H_a(s) = \frac{\Omega_c^2}{s^2 + \sqrt{2}\Omega_c s + \Omega_c^2}$$

Use the bilinear transformation to determine a digital filter with the same cut-off frequency. The sampling rate is  $f_s=10\text{kHz}$ .

Questions:

- Determine  $H(z)$
- Compute the DC gain
- Verify that the -3dB cut-off frequency for  $H(z)$  is 2 kHz

**Problem C.1 (weighted with 4% - Spectral estimation)**

A digital filter  $H$  is implemented at the sampling frequency  $f_s=10$  kHz and has the impulse response  $h[n]$  with the length  $N=4$ . By using a DFT the following frequency response is achieved:  $H[k]=[0 \ 1+j \ 0 \ 1-j]$ .

Find the sequence  $h[n]$ .

**Problem C.2 (weighted with 3% - Spectral estimation)**

Draw the discrete sequence  $H[k]$  for each  $k$  and put frequency values in Hz on the x-axis.

**Problem C.3 (weighted with 4% - Spectral estimation)**

We now consider a continuous sinusoid signal with frequency  $f=1$  kHz,  $s(t)=\cos(\Omega t)$ , where  $\Omega=2\pi f$  and  $t$  is the time. The signal is sampled at  $f_s$  in order to make a sequence  $x[n]$ .

Find  $x[n]$  using a rectangular window of length  $L=4$ .

**Problem C.4 (weighted with 4% - Spectral estimation)**

The sampled sine signal  $x[n]$  is now sent through the filter  $H$  so that an output sequence  $y[n]$  is obtained.

Find  $y[n]$ .

**Problem C.5 (weighted with 5% - Spectral estimation)**

On your drawing above (see C.2) 1 kHz is not represented, so one might think that the filter doesn't let through this frequency. However,  $y[n]$  is clearly not a zero-sequence.

Explain why  $y[n]$  is not just a sequence of zeros.

**Problem C.6 (weighted with 4% - Spectral estimation)**

Calculate the amplification (gain) of the filter  $H$  at 1 kHz.

**Problem C.7 (weighted with 4% - Spectral estimation)**

Sketch the continuous frequency response  $|H(\omega)|$  and put frequency (in Hz) marks on the x-axis and gain on the y-axis.

**Problem C.8 (weighted with 5% - Spectral estimation)**

The continuous signal  $s(t)$  is now sampled again at  $f_s$  but this time utilizing a Blackman window with the length  $L=256$ . Hereafter the signal is analyzed with a 256-points FFT.

What is the "effective frequency resolution"  $\Delta f$  using the analysis above, i.e. its ability to separate tones (frequencies) in close proximity to each other?