Examination
SSY130 Applied Signal Processing

14:00-18:00, Jan. 14, 2015

Instructions

- *Responsible teacher:* Tomas McKelvey, ph 1788. Teacher will visit the site of examination at approximately 14:45 and 16:30.
- Score from the written examination will together with course determine the final grade according to the Course PM.
- Solutions are published on the course home-page latest on Monday, January 19.
- Your preliminary grade is reported to you via email.
- Exam grading review will be held in room 7430 on level 7 at 12:00-13:00 on Wednesday, January 28.

Allowed aids at exam:

- L. Råde and B. Westergren, Mathematics Handbook (any edition, including the old editions called Beta or copied sections from it), Formulaires et tables Mathématiques, Physique, Chimie, or similar.
- Any calculator
- One a4 size single sheet of paper with written notes on both sides.

Other important issues:

- The exam consists of 5 numbered problems.
- The ordering of the questions is arbitrary.
- All solutions should be well motivated and clearly presented in order to render a full score. Unclear presentation or adding, for the problem in questing, irrelevant information render a reduction of the score.
- Write solutions to each problem on a *separate* sheet of paper.
- The maximum score is 52 points.

Problems

1. In the following subproblems you will get 1 point if the answer is correct and 1 point if the answer is correctly motivated. All frequency scales are relative to the sampling frequency, i.e. 1 corresponds to the sampling frequency.

   (a) The magnitude of the DTFT of two FIR filters designed with the Parks-McClellan optimal equiripple FIR filter design method are shown in Figure 1 (top left). Which of the two filters have the longest length. (2pt)

   (b) Two high-pass filters of the same length are designed using the window method. Filter A is designed using the window function A and filter B is designed with window function B. The DTFT of the two window functions are shown in Figure 1 (top right).

      i. Which of the two filters will have the best stop-band attenuation? (2pt)

      ii. Which of the two filters will have the largest transition band? (2pt)

   (c) The magnitude of the DTFT of the impulse response of a filter is shown in Figure 1 (bottom left.

      i. Describe the type of filter (HP,LP,BP or BS) (2pt)

      ii. What is/are the crossover frequency/frequencies in Hertz if we assume the sample rate of the filter is 20 kHz. (2pt)

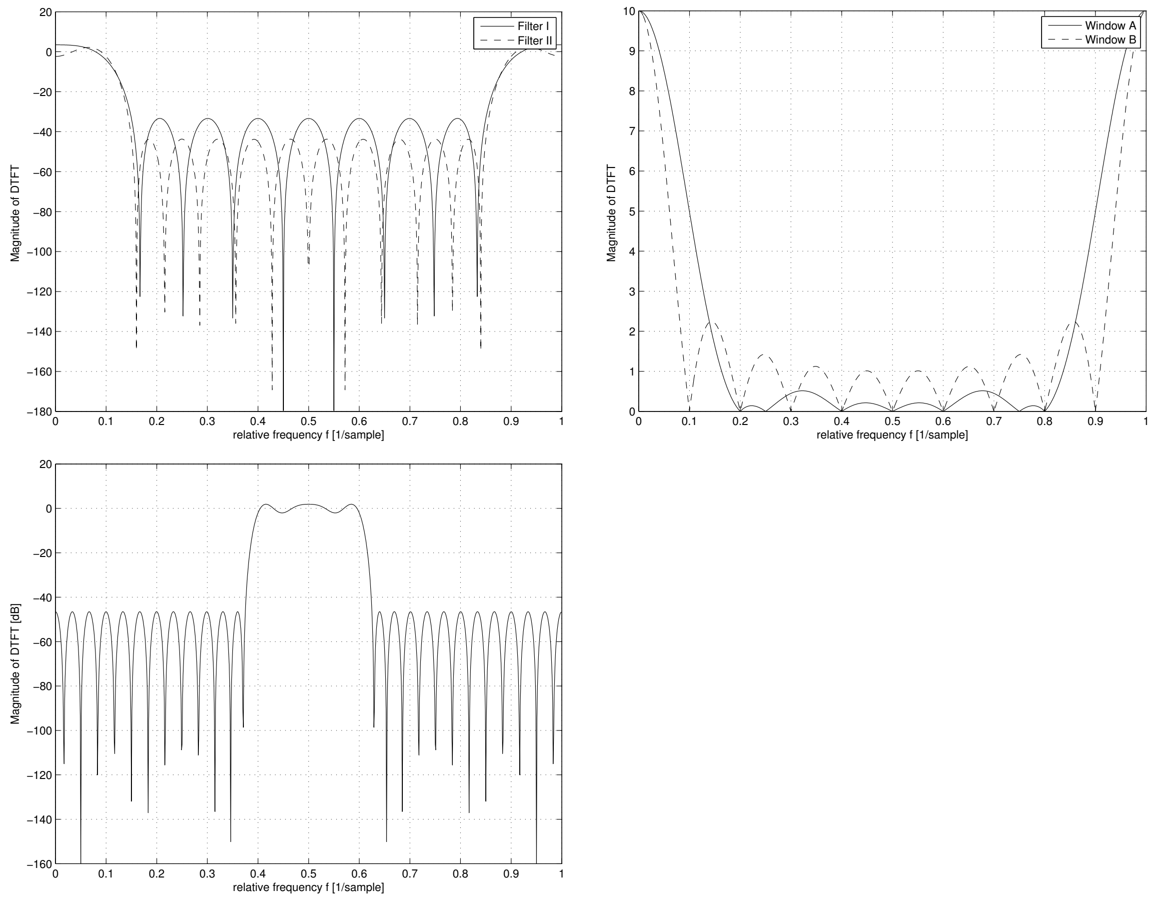


Figure 1: Magnitude DTFT of two FIR filters (top left). Magnitude DTFT of two window functions (top right). Magnitude DTFT of the impulse response of a filter (bottom left).

2. This problem is about signal interpolation and it's implementation.

   (a) Describe the two basic signal processing operations found in a linear signal interpolation function and what their purpose are. (3pt)

   (b) Assume the original real signal has a sample rate of 1kHz. Describe an interpolation design which will change the sample rate to 5 kHz but keep the original signal bandwidth. Specify the edge frequencies of the passband of the filter involved. (3pt)

   (c) Describe what a polyphase filter representation is and how it can be applied to signal interpolation. (3pt)

3. In telephone applications such as, conference telephones, speaker phones or mobile handsfree operation, a common feature is that the loudspeaker and the microphone is located in the same acoustic environment. The microphone of the receiver not only picks upp the voice of the person localized in the same room as the telephone (local caller) but *also* the sound originating from the remote caller. If this effect is not mitigated (reduced) by some technique the remote caller will hear his own voice with some delay depending on the transmission delay to and from the local caller and it often sounds like an echo.

   (a) Construct a signal block diagram explaining the setup described above. Use the following notation:

      $s_r(n)$ is the signal from the remote caller at the remote site

      $G(\omega)$ is the frequency function modeling the transmission path between the local site and the remote site. For simplicity we assume it is symmetric, i.e. the frequency function is equal in both directions.

      $s_l(n)$ is the signal from the local caller

      $H(\omega)$ is the frequency function between the local loudspeaker and the local microphone

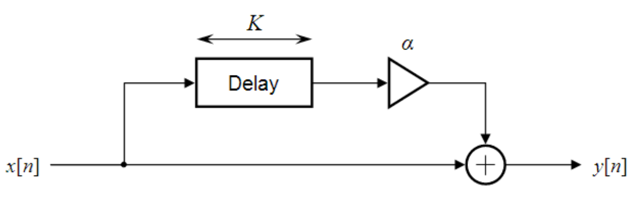
      $z_r(n)$ is the signal the remote caller hears at the remote site. (5pt)

   (b) To mitigate the echo effect add an adaptive filter to the local site. Make a new block diagram which includes the frequency function of the adaptive filter and describe which signals are used to update the filter. (4pt)

   (c) Under what conditions will the echo be perfectly cancelled? (3pt)

   (d) Assume both the local caller and the remote caller are listening to the same radio channel during the phone call with the adaptive filter in operation. Describe what will happen. (5pt)

4. The comb filter is a filter with the general structure as illustrated below



   which means that the filtered output is the present input added with the input delayed K samples, where K is a positive integer, and scaled with α.

   (a) Explain why the filter is particularly inexpensive to implement if $\alpha = \pm 1$ (2pt)

   (b) Derive the frequency function for the comb filter (3pt)

   (c) Which choice $\alpha = +1$ or $\alpha = -1$ would yield a filter where the lowest frequencies will pass through the filter? (2pt)

(d) Assume $\alpha = -1$. At which frequencies will the frequency function have magnitude zero? *Hint:* Recall the complex solutions to the algebraic equation $x^K = 1$ (3pt)

5. The DFT and IDFT pairs are defined as:

$$\text{DFT:} \quad X(k) = \sum_{n=0}^{N-1} x(n) e^{-j2\pi kn/N}, \quad \text{IDFT:} \quad x(n) = \frac{1}{N} \sum_{k=0}^{N-1} X(k) e^{j2\pi kn/N}$$

The Fast Fourier Transform (FFT) is a computational method to efficiently calculate the DFT. Assume the FFT implementation of DFT is available as a function FFT in some computer language (or hardware). Show how you can use the FFT function to also calculate the IDFT by introducing some elementary operations on the input/output data. *Hint:* Use conjugation! (6pt)

END - Good Luck!