

Homework #3 Child of Filter Analysis, Simulation, and Design

Assigned Sunday, Feb. 25, 2024, and due Friday, Mar. 1, 2024, by 11:59pm

Homework submitted after 11:59pm will be subject to a penalty of 2 points per minute late.

Through midterm #1, here are key sections from JSK's *Software Receiver Design*, and signals & systems textbooks Oppenheim & Willsky's *Signals & Systems* (2nd ed); McClellan, Schafer & Yoder's *Signal Processing First* and Lathi & Green's *Linear Systems & Signals* (3rd ed). **Bold are topics for this homework.**

Topic	<u>Lect.</u>	JSK	O&W	SP First	L&G	<u>Handouts</u>
Introduction	<u>0</u>	Ch. 1				
Bandwidth	<u>1</u>	2.2	4.3-4.4	11-4 to 11-8	6.3-1, 7.2, 7.3, 7.9	Fourier Transforms: Intro , Dictionary and w vs. f
Sinusoidal generation	<u>1</u>	3.2				
Upconversion and downconversion	<u>1&4</u>	2.3-2.6, 3.6; Ch. 5; 6.1-6.4	8.1-8.4	11-8.2	7.3, 7.7	Sinusoidal Ampl. Modulation Example & Summary
Communication sys.	<u>1</u>	Ch. 1-2	Ch. 8		7.7	
Basic CT signals	<u>3</u>	2.10, 4.3	1.3-1.4	2-3, 2-5, 4-4 & 9-1	1.4	Common Signals in Matlab
CT system properties	<u>3</u>		1.6	5-5	1.7	LTI System Properties
Basic DT signals	<u>3</u>		1.3-1.4	4-2.1 & 5-3.2	3.3	Common Signals in Matlab ; DT Periodicity ; Chirp Signals
DT system properties	<u>3</u>		1.6	9-4	3.4-1	Time-Invariance ; LTI System Properties
Fundamental Theorem of Linear Systems	<u>3&5</u>	3.5	3.2	6-1 & 10-1	2.4-4 & 3.8-2	Fundamental Theorem of Linear Systems
Sampling theorem	<u>4</u>		7.1	4-1 4-2 4-5	8.1	
Sampling and aliasing	<u>4</u>	2.8, 3.4, 6.1	7.3 & 7.4	12-3	8.2	Sampling Unit Step Signal
Bandpass sampling	<u>4</u>					
DT-to-CT conversion	<u>4</u>	2.10 & 6.4	7.2	4-4	8.2	
CT convolution	<u>5</u>	4.4 & 4.5	2.2	9-6 & 9-7	2.4	Convolution Example
DT convolution	<u>5</u>	4.4 & 4.5	2.1	5-3.3 & 5-6	3.8	Convolution Example and Four ways to filter a signal
Z-transforms	<u>5&6</u>	App. A.4 & F	10.1-10.3, 10.5	7-1 to 7-5	5.1-5.2	
Transfer functions	<u>5&6</u>	4.5	10.7 & 10.8	8-3, 8-4 & 8-9	5.3	Designing Averaging Filters ; LTI Systems & Freq. Resp.
Relationship between z & Fourier transform	<u>5</u>	App. F.2	10.4	7-6, 8-5, 8-6, 8-10	5.5	
DT FIR filter design and implementation	<u>5</u>	2.12, 3.3, 4.2	10-9	7-7	5.4	Four ways to filter a signal ; Designing Averaging Filters
DT FIR filter analysis	<u>5</u>	Ch. 7 & App.G		7-7 to 7-9		Designing Averaging Filters
DT stability of LTI systems	<u>5&6</u>		10.7.2	8-2.4, 8-4.2, 8-8	3.9 & 3.10	Bounded-Input Bounded-Output Stability
DT IIR filter design by pole-zero placement	<u>6</u>		10.4	8-9 & 8-10	5.6	All-pass Filters
DT classical IIR filter design methods	<u>6</u>				5.10	Elliptic IIR filter design
IIR filter implement.	<u>6</u>		10-9	8-9	5.4	Realizations of IIR Filters

Please read the [homework hints](#)

Office hours for the teaching assistants and Prof. Evans; **bold** indicates a 30-minute timeslot. Prof. Evans' weekly coffee hours are on Fridays 12-2pm.

<i>Time Slot</i>	<i>Monday</i>	<i>Tuesday</i>	<i>Wednesday</i>	<i>Thursday</i>	<i>Friday</i>
10:30am	Evans (ECJ 1.312)		Evans (ECJ 1.312)		
11:00 am	Evans (ECJ 1.312)		Evans (ECJ 1.312)		
12:00 pm					Evans coffee hours (EER Cafe)
1:00 pm					Evans coffee hours (EER Cafe)
2:00 pm	Evans (EER 6.882 & Zoom)		Evans (EER 6.882 & Zoom)		
3:00 pm	Evans (EER 6.882 & Zoom)		Evans (EER 6.882 & Zoom)	Barati (EER 1.810)	
3:30 pm				Barati (EER 1.810)	
4:00 pm			Barati (EER 1.810)	Barati (EER 1.810)	
4:30 pm			Barati (EER 1.810)	Eun (EER 1.810)	
5:00 pm			Barati (EER 1.810)	Eun (EER 1.810)	Eun (Zoom)
5:30 pm				Eun (EER 1.810)	Eun (Zoom)
6:00 pm					Eun (Zoom)
6:30 pm					

Please submit any MATLAB code that you have written for the homework solution. In your solutions, please put all work for problem 1 together, then all work for problem 2 together, etc.

As stated on the course descriptor, “Discussion of homework questions is encouraged. Please be sure to submit your own independent homework solution.”

3.1. Using Filtering to Improve Signal Quality. 27 points.

Johnson, Sethares & Klein, exercise 4.21, on page 78. *9 points for each of the two parts.*

For the simulation of the narrowband interferers in parts (a) and (b) please use a chirp signal.

In addition, please complete the following third part: (c) for the narrowband interference occurring inside the transmission band that you simulated in part (b), design a notch filter to use after the bandpass filter. For the notch filter, please use a second-order infinite impulse response (IIR) filter. Write the Matlab code to apply the notch filter to the output of the bandpass filter used to reduce out-of-band interference, and compute the change in SNR due to your notch filter. *9 points.*

Please read the [homework hints](#)

3.2. Quadrature Amplitude Modulation. 27 points.

Johnson, Sethares & Klein, exercise 5.14 on page 92, parts (a) and (b).

3.3. Infinite Impulse Response (IIR) Filter Design for Treatment of Tinnitus Loudness. 46 points.

This problem asks you to design a discrete-time digital IIR filter for the treatment of tinnitus loudness. *This problem is a sequel to homework problem 2.3.*

Tinnitus, a.k.a. “ringing of the ears”, is a symptom due to an underlying condition in the auditory system. It could have resulted from injury, infection, or other causes. People with tinnitus hear a tone, clicking, hiss, roaring or buzzing when no external sound is present [1][2]. The tinnitus sound could be at low, medium or high audible frequencies, and may occur in one ear or both ears. The tinnitus sound might be temporary or chronic. Those suffering from chronic tinnitus would hear the same sound in the same frequency range each time. The tinnitus sound has a principal frequency that can be determined through auditory testing. Hearing sound that contains the principal frequency and frequencies close to the principal frequency is particularly painful.

This problem asks you to design a discrete-time filter to alleviate the loudness of tinnitus:

"Maladaptive auditory cortex reorganization may contribute to the generation and maintenance of tinnitus. Because cortical organization can be modified by behavioral training, we attempted to reduce tinnitus loudness by exposing chronic tinnitus patients to self-chosen, enjoyable music, which was modified (“notched”) to contain no energy in the frequency range surrounding the individual tinnitus frequency. After 12 months of regular listening, the target patient group ($n = 8$) showed significantly reduced subjective tinnitus loudness and concomitantly exhibited reduced evoked activity in auditory cortex areas corresponding to the tinnitus frequency compared to patients who had received an analogous placebo notched music treatment ($n = 8$). These findings indicate that tinnitus loudness can be significantly diminished by an enjoyable, low-cost, custom-tailored notched music treatment, potentially via reversing maladaptive auditory cortex reorganization.” [3]

The proposed treatment for tinnitus [3] alters participants' favorite music to remove an octave of frequencies around the tinnitus frequency f_c . An octave means a range of frequencies from f_l to $2f_l$. Since f_c would be in the middle of the octave, $f_l = (2/3)f_c$. After 12 months of listening to the filtered music, patients reported lessening of tinnitus loudness.

A good rule of thumb in filter design is that the transition region is about 10% of the passband width. In this case, the passband width is $(2/3)f_c$.

Here are the bandstop filter specifications for your design:

- For frequencies 0 Hz to $0.6f_c$, the passband ripple should be no greater than 1 dB.
- For frequencies $(2/3)f_c$ to $(4/3)f_c$, the stopband attenuation should be at least 80 dB.
- For frequencies above $1.4f_c$, the passband ripple should be no greater than 1 dB

Please use a tinnitus frequency f_c of 3000 Hz and a sampling rate f_s of 44100 Hz.

(a) Design IIR filters using the Butterworth, Chebyshev type I, Chebyshev type II, and Elliptic (Equiripple) design methods. For each design method, find the filter of smallest order to meet the specifications. The filter order is the number of poles. Turn in plots of the magnitude and phase responses for each IIR filter you have designed to meet the specifications. Describe the passband and stopband response for each filter design as either monotonic or rippling. **24 points.**

Please read the [homework hints](#)

- (b) List the filter orders required for filters to meet the specification. Which IIR filter family gives the lowest filter order? Plot the pole-zero diagram for the IIR filter with shortest filter order you designed in (a). Because the poles and zeros are close to the unit circle and separated in angle, poles indicate the passband and zeros indicate the stopband. Submit the feedforward and feedback coefficients for the filter with minimum order. **6 points.**
- (c) Use the Elliptic filter you designed in (a) for this part. Assuming that the input data samples and the IIR filter coefficients are stored in 32-bit IEEE floating point format,
1. How many instruction cycles on the TMS320C6700 DSP family would it take to compute one output value for each input value if the IIR filter routine were handcoded in assembly for optimal performance? **6 points.**
 2. How much storage in bytes would it take to store the IIR coefficients and the circular buffer for the current and past inputs and the circular buffer for past outputs? **4 points.**
- (d) How would you advocate using either an FIR filter or IIR filter for this application? In your answer, please include the design method for which you are advocating. **6 points.**

A rule-of-thumb for filter design is that the distance from the (maximum) passband frequency to the (minimum) stopband frequency should be about 10% of the passband bandwidth.

For this problem, please use the filter design and analysis tool, *fdatool*, in Matlab. This tool is particularly useful when exploring different filter structures for implementation.

Hints

(a) The automatic order calculations for an IIR filter to meet piecewise magnitude specifications are very accurate. I don't recall seeing an incorrect calculation before.

By looking at the magnitude response plots, validate that the designed filters meet the specifications. In particular, be sure to check the graphical views of the passband and stopband responses, i.e. zoom in on the magnitude response near the passband frequency and near the stopband frequency. If you are not absolutely sure from visually inspecting the magnitude response, then you can compute the magnitude response at a particular frequency using the `freqz` function in Matlab. The first two arguments are the numerator coefficients of the transfer function and the denominator coefficients of the transfer function. Type `help fdatool` to find out how to export the filter design to Matlab. Then, you can take the magnitude of the result. Finally, you'll need to convert the magnitude to dB using

$$20 \log_{10} \text{magnitude}$$

(c) Since an IIR filter could be calculated as a tapped delay line applied to current and previous input samples, and a tapped delay line applied to previous output samples, Appendix N in the course reader might be helpful in calculating the number of instruction cycles needed to compute one output value.

References

- [1] R. A. Levine and Y. Oron, "Tinnitus", *Handbook of Clinical Neurology*, vol. 129, pp. 409–431, 2015. [doi:10.1016/B978-0-444-62630-1.00023-8](https://doi.org/10.1016/B978-0-444-62630-1.00023-8).
- [2] "[Tinnitus](http://www.nidcd.nih.gov/health/hearing/Pages/tinnitus.aspx)". Retrieved Feb.17, 2017. <http://www.nidcd.nih.gov/health/hearing/Pages/tinnitus.aspx>
- [3] H. Okamoto, H. Stracke, W. Stoll and C. Pantev, "[Listening to tailor-made notched music reduces tinnitus loudness and tinnitus-related auditory cortex activity](http://www.pnas.org/content/107/3/1207)", *Proceedings US National Academy of Sciences*, vol. 17, no. 3, pp. 1207-1210, 2010. <http://www.pnas.org/content/107/3/1207>