In the Name of God



University of Tehran

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Digital Signal Processing

Computer Assignment #2

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Abstract

Goal of the computer assignment is to get familiar with Filter Designer Tool of matlab. Designing filters in this environment is the main goal. It takes look at frequency response of images and voices either. At last we get familiar with kernels and what is the result of using them.

Question one

Part A & B

Sampling frequency is equal to 11025.

$$f_s = 11025$$

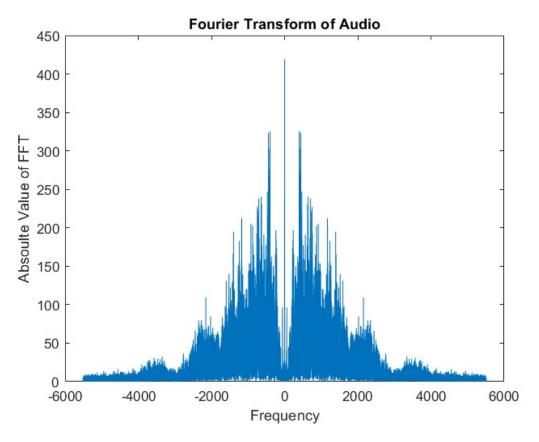


Figure 1, Fourier Transform of Audio

As We can see bandwidth of signal is:

$$BW = 5500 \ hz$$

Part C & D & E

From definition of autocorrelation function, we have:

$$\begin{split} R_{y}[m] &= \sum y[n]y[n-m] = \\ &\sum (x[n] + \alpha x[n-k_{1}] + \beta x[n-k_{2}])(x[n-m] + \alpha x[n-m-k_{1}] + \beta x[n-m-k_{2}]) \\ &= R_{x} + \alpha^{2}R_{x} + \beta^{2}R_{x} + \alpha R_{x}[m+k_{1}] + \alpha R_{x}[m-k_{1}] + \beta R_{x}[m-k_{2}] + \beta R_{x}[m+k_{2}] + \alpha \beta R_{x}[m-k_{2} + k_{1}] + \alpha \beta R_{x}[m+k_{2} - k_{1}] \end{split}$$

So, from eq. above and xcorr function output which shown in Figure 2 We can understand that:

$$\begin{array}{l} at \; m = k_1 = 5500 \rightarrow \alpha R_x[0] = 306.9 \\ at \; m = k_2 = 9000 \rightarrow \beta R_x[0] = 211.4 \\ at \; m = k_2 - k_1 = 3500 \rightarrow \alpha \beta \; R_x[0] = 66.26 \end{array}$$

After solving eq:

$$\alpha = \frac{66.26}{211.4} = 0.31 \approx 0.3; \ k_1 = 5500$$
$$\beta = \frac{66.26}{306.9} = 0.22 \approx 0.2; \ k_2 = 9000$$

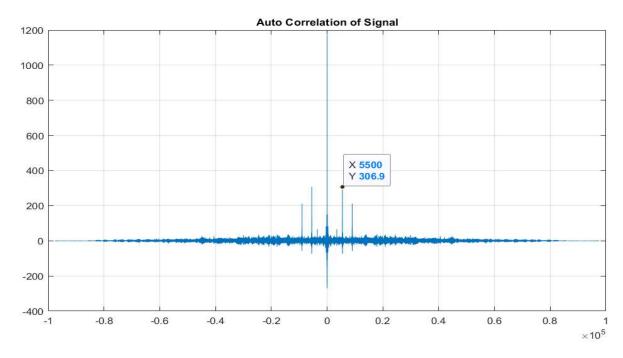


Figure 2, Autocorrelation of audio sample

Part F, Finding Impulse Response of Echo System

we know that : x * h = y

and from eq we have:

$$y[n] = x[n] + 0.3 x[n - 5.5k] + 0.2 x[n - 9k]$$

$$\rightarrow h[n] = \delta[n] + 0.3\delta[n - 5.5k] + 0.2 \delta[n - k]$$

Part G, Removing Echo from Audio Signal

Filter function in MATLAB requires coefficient of transfer function to filter signal, therefore we find z-transform of Echo System impulse response then, inverse the response to remove Echo From audio.

$$Z-Transform\ of\ impulse\ respone: 1+0.2\ z^{-9k}+0.3z^{-5k}$$

$$Inverse\ form\ of\ singal=\frac{1}{1+0.2\ z^{-9k}+0.3z^{-5k}}$$

So coefficient for filter function is:

$$b = 1;$$

$$a = [1, zeros(1,5499), 0.3, zeros(1,3499), 0.2];$$

After performing filter function and listening to audio we can understand that Echo has been completely removed from audio.

The audio has been recorded as: "y Without Echo.wav" and it can be found with other files.

Part H, Kaiser Filter in Filter and Design App MATLAB

Kaiser Filter has been designed as figure 4, and its frequency response has been shown in figure 3.

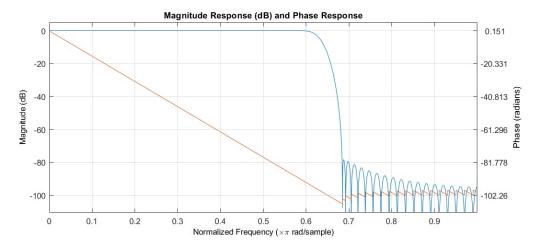


Figure 3, Frequency Response of Kaiser Filter

File Edit Analysis Targets View Window Help

Current Filter Information

Magnitude Response (dB)

Structure: Direct-Form FIR Order: 100
Stable: Yes Source: Designed

Store Filter...

Filter Manager...

Response Type

Filter Order: 100

Specify order: 100

Minimum order

Scale Passband

Window: Kaiser

Design Method

For better noise limitation, parameter F_C has been changed to 3500

View

Figure 4, Kaiser Filter Designing

Part I

The last

As we can see in figure 5 Hd filter has limited the noise and reduced bandwidth of signal to $(f_c = 3500)$

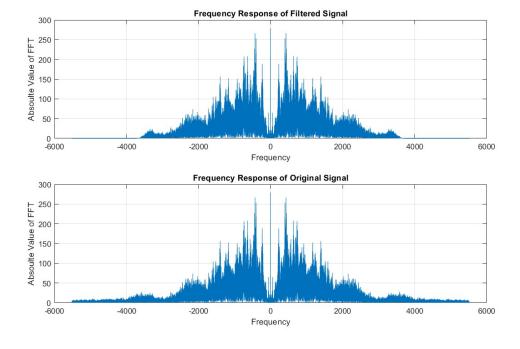


Figure 5, Frequency Response Original and Filtered Signal

Part J

Here is Signal in Time Domain, for having a closer look time has been limited in figure 6 to understand difference between filters, as we can see high frequencies has been cut from Filtered Signal and it looks smoother.

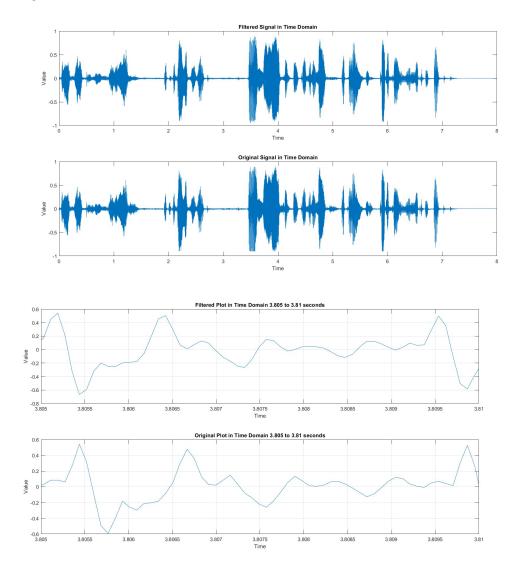


Figure 6, Signals in Time Domain

Part K, Designing another FIR Filter

Chebyshev filter is used and its frequency response is as below.

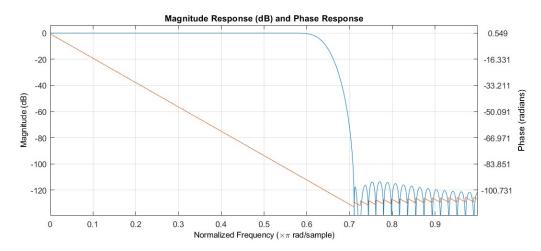


Figure 7, Chebyshev Magnitude and Phase Response

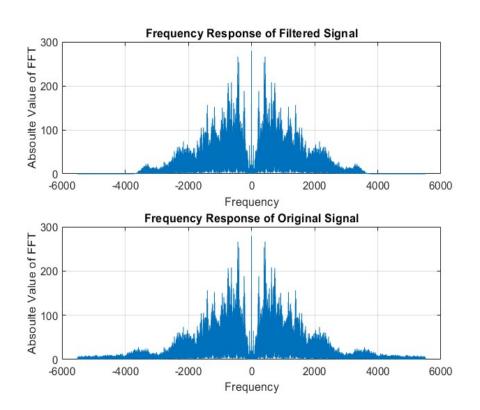


Figure 8, Frequnecy Response of Signal

As We can see Kaiser filter is a little better than Chebyshev in same order.

Question Two

The transfer function of previous part is used to creat echo and it has been removed with same method.

$$Z-Transform\ of\ impulse\ respone: 1+0.2\ z^{-9k}+0.3z^{-5k}$$

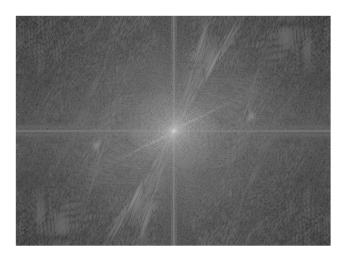
$$Inverse\ form\ of\ singal=\frac{1}{1+0.2\ z^{-9k}+0.3z^{-5k}}$$

The Voices Has been recorded and saved, The file Will Be Uploaded With Project.

Question Three

In This question the goal is to implement different filters on pictures, the original image has shown below





 $Figure \ 9 \ , Original \ Grayscale \ Image$

Analyzing Different Filters:

Gaussian: returns a rotationally symmetric Gaussian lowpass filter. It has two parameters, sigma which indicates power and mean which indicates offset.



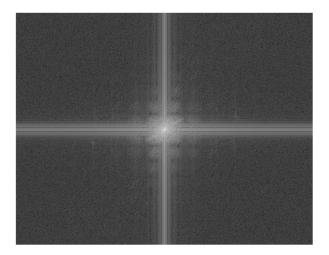


Figure 10 Guassian Filter Output, With Frequency Response

Average: returns an averaging filter, the parameter indicates average window size as it increases picture tends to be more blurry.



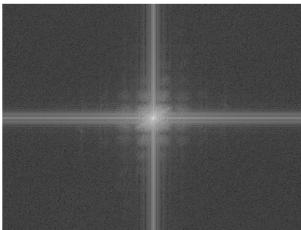


Figure 11, Average filter

The output is very similar to Guassian

Disk: returns a circular averaging filter, it is same as before but average is circular. The parameter R shows radius of circle which is being averages as it increases picture is more blurry.



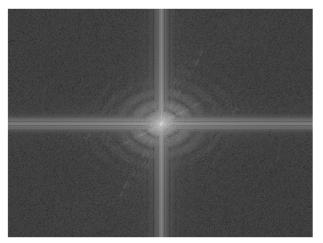


Figure 12, Disk Filter

These three filters (disk, average, guassian) have lowpass noise as it is clear from pictures .

Sobel: emphasizes horizontal edges using the smoothing effect by a vertical gradient. it has no other parameter. It just keep the edges and other parts will be removed.



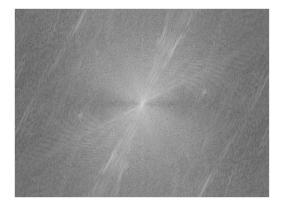
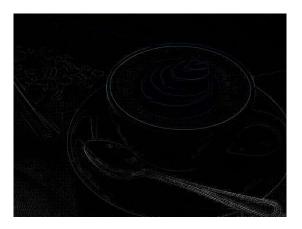


Figure 13, Sobel Filter

Laplacian: It is same as Laplacian operator, returns a 3-by-3 filter approximating the shape of the two-dimensional Laplacian operator, alpha controls the shape of the Laplacian. As it increases white pixel reduces. Laplacian differentiate picture two times and find the edges of picture



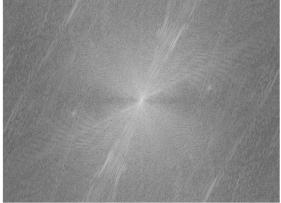


Figure 14, Laplacian Filter

These two filters find edges, they are high pass and its clear from their frequency response

Question 4

Part A

An image kernel is a small matrix used to apply effects such as blurring, sharpening, edge detection and more.

Here is a glimpse to how they work, the kernel convolves with image for instance I show mechanism for sharpen kernel:

Sharpener Kernel:
$$\begin{pmatrix} 0 & -1 & 0 \\ -1 & 5 & -1 \\ 0 & -1 & 0 \end{pmatrix}$$

Below, for each 3x3 block of pixels in the image on the left, we multiply each pixel by the corresponding entry of the kernel and then take the sum. That sum becomes a new pixel in the image on the right. Hover over a pixel on either image to see how its value is computed.¹

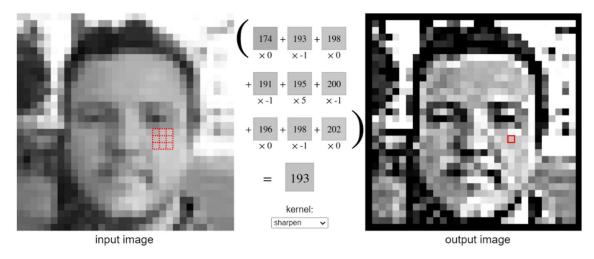


Figure 15

¹ https://setosa.io/

Part B $^{\rm 1}$ Edge Detection: It emphasis on edges and set other pixels to zero.



Figure 16, Edge Detector Kernel

Sharpen: Same as previous, This filter's focus is on edges of picture but it keeps other pixels and it just make image sharper.



Figure 17, Edge Detector

¹ Two methods has been used to show the result: conv2 and imfilter, Here imfilter results are shown. Output of the other method can be seen in code.

Gaussian Blur: it blurs image by applying gaussian function, it makes picture more blurry.



Figure 18, Guassian Blur

Identity: Does not change the image as we can see in figure 19.



Figure 19, Identity

Part C

Kernels are also used in machine learning for 'feature extraction', a technique for determining the most important portions of an image. By this technique image will be moved in higher dimension for better analysing.

Appendix: Instruction in Running Code

Each Question has its own name in code documentary.

For question 1 it is necessary to import Kaiser filter which is designed as an object to workspace.

Necessary explanation has been written in comments.