1.1 IIR filters

1.

The order of the filter is 13, the phase is what I expect for.

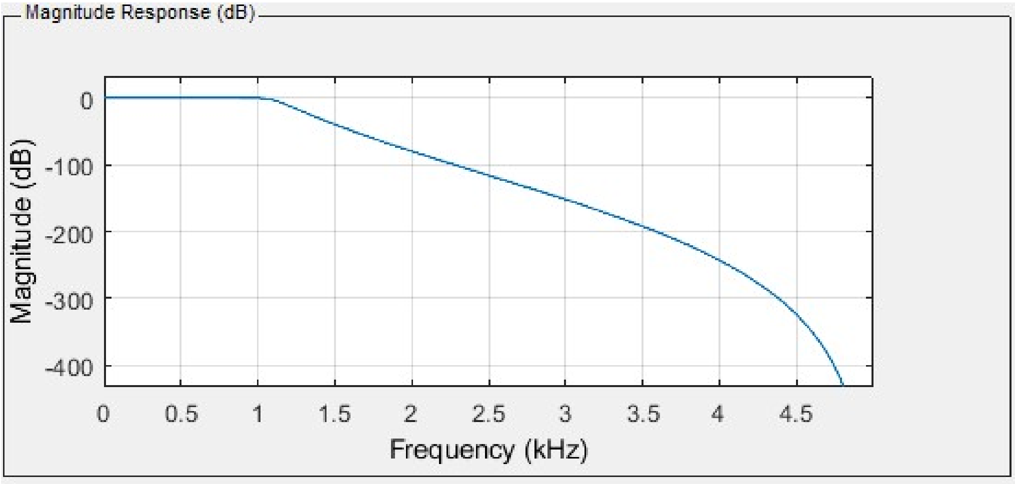

Figure 1. Magnitude Response

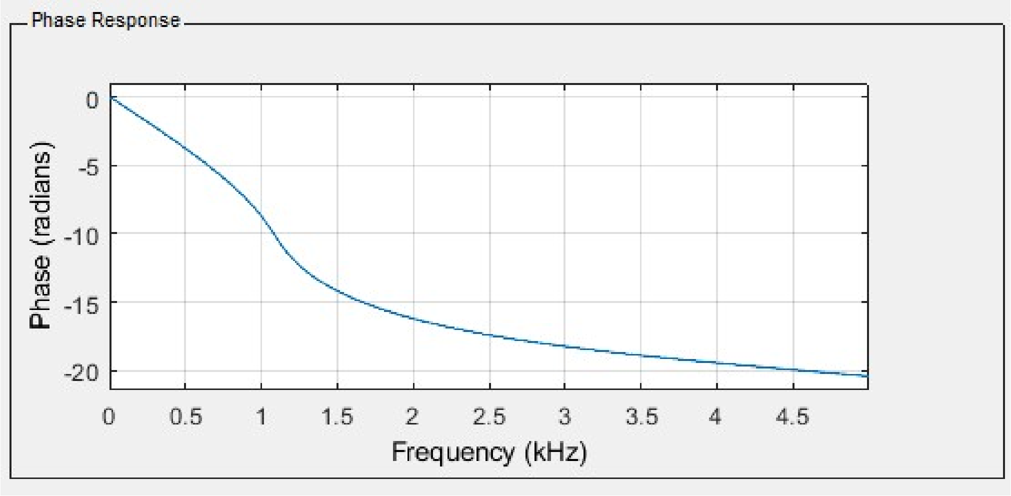

Figure 2. Phase Response

2.

Change the transition region to a narrow one, Fpass = 1 kHz, Fstop = 1.5kHz, the order increase to 22.

Change the transition region to a widen one, Fpass = 1 kHz, Fstop = 2.5kHz, the order decrease to 9.

When we narrow the transition region, we will get better performance in transition region, but we will increase the order of the filter which means consuming more resources. When we widen it, the order will decrease. We need to balance the performance and resource consumption at the same time.

3.

As the result of the calculation, the minimum order of the filter should be 1.64, we use 2. We create the filter as follow and the minimum order is 2.

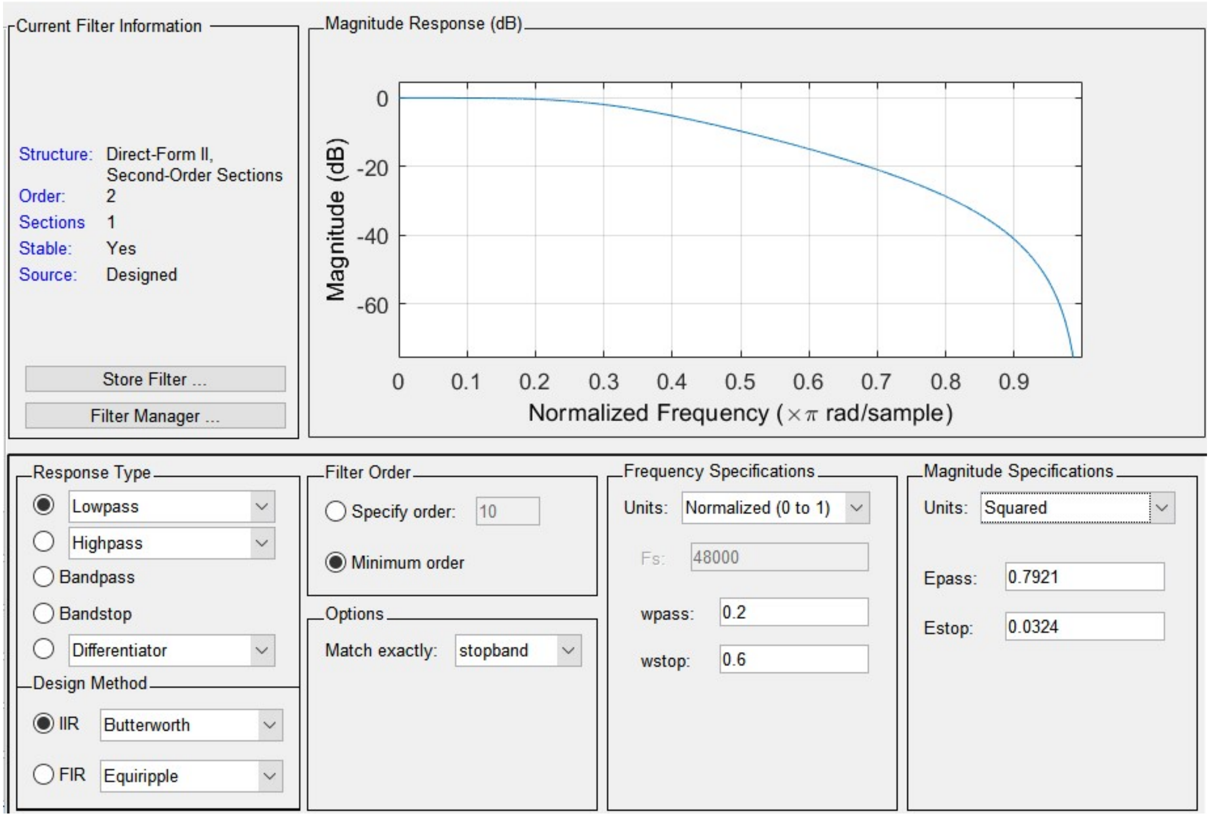

Figure 3. Verification of theoretical result

4.

NO, Order 7 will be OK if we calculate it with impulse invariance but in MATLAB, the software will calculate the order with bilinear transform, and we will get the order 5.

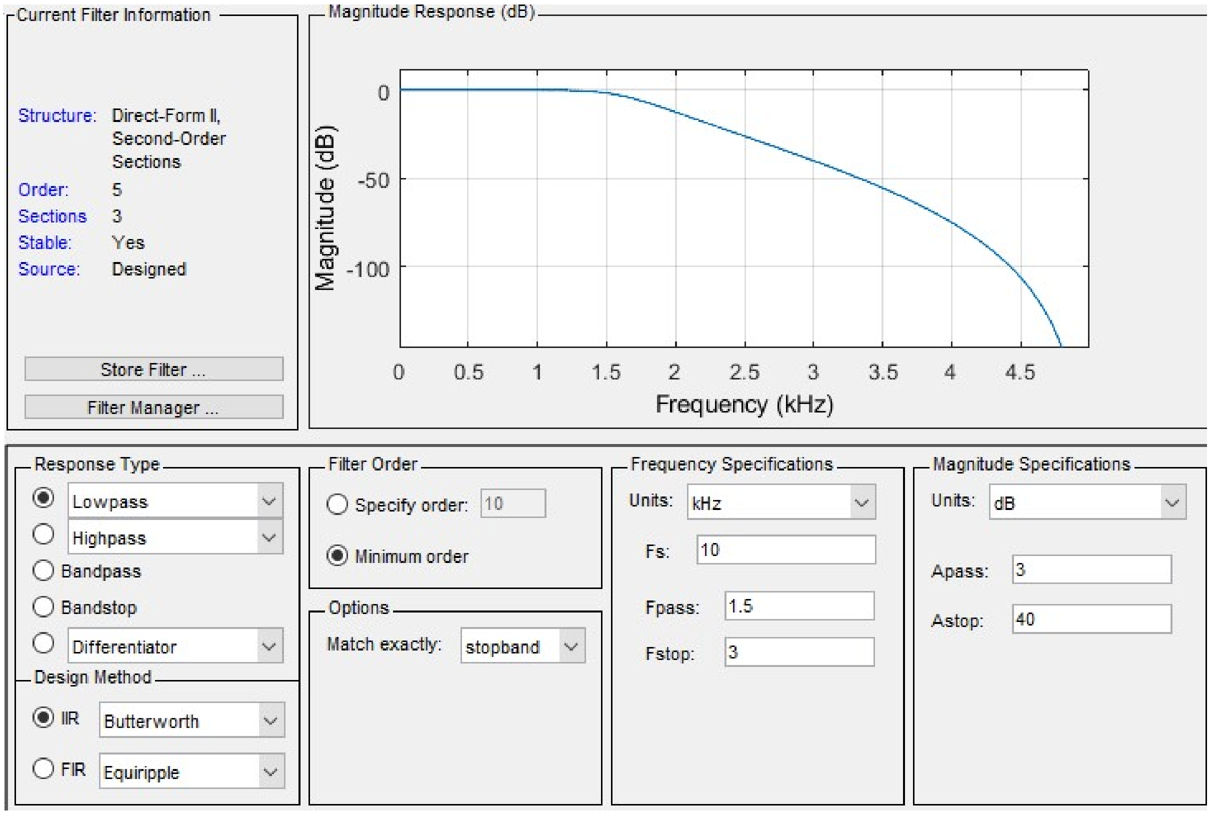

Figure 4. Order 5 low pass filter

5.

No, it does not match the figure which found in the impulse invariance method and the reason is that if we calculate it with impulse invariance but in MATLAB, the software will calculate the order with bilinear transform, and we will get the order 5.

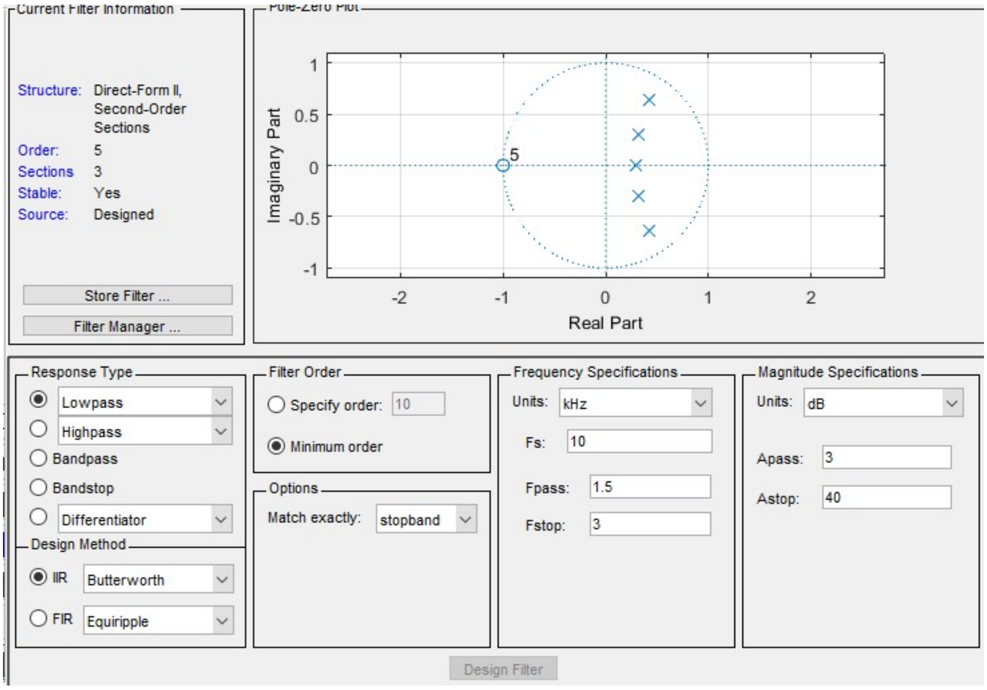

Figure 5. Pole-Zero Plot of the filter

6.

Compare with Butterworth filter:

1. Butterworth: no ripples in passband and stopband, smooth in transition region.
2. Chebyshev : equiripple ripples in passband and stopband. Sharper roll-off in the transition region.

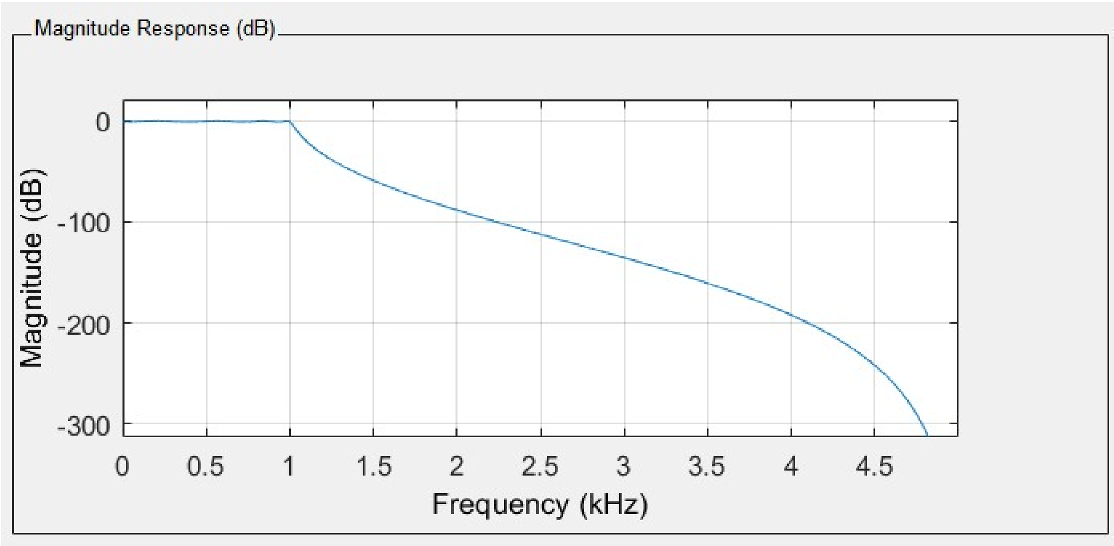

Figure 6. Magnitude Response of Chebyshev Type 1

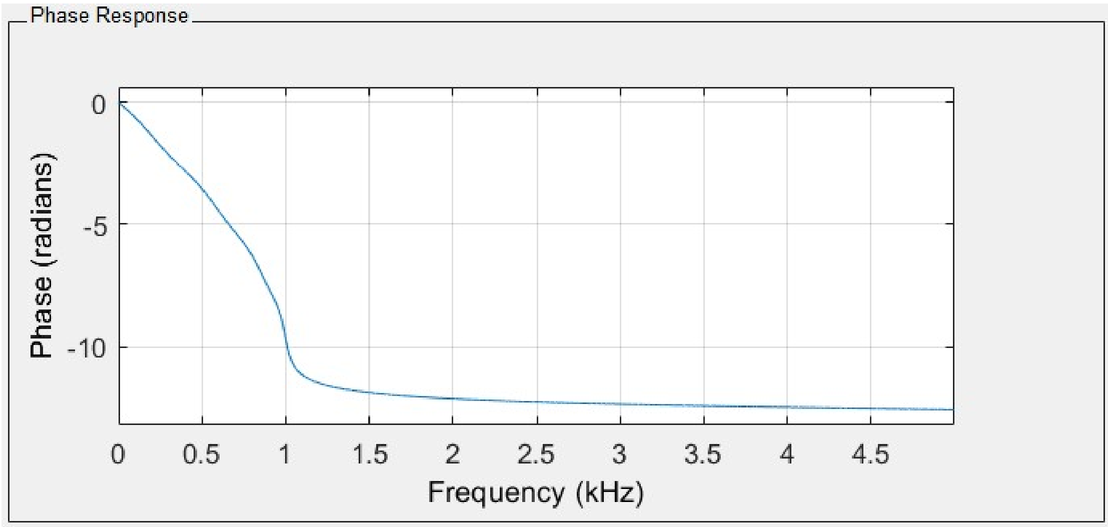

Figure 7. Phase Response of Chebyshev Type 1

The order of Chebyshev Type I is 8.

7.

Consider the filter designed before, we can conclude the pros and cons of these two filters as below.

Table 1. Compare two filters

	Chebyshev	Butterworth

filter order	8(lower)	13(higher)
Pass-band response	equiripple	no ripple
Stop-band response	equiripple	no ripple
transition region	narrow	wide
phase response	non-linear	linear
Stop-band attenuation	high	low
Group Delay	non-constant	constant

1.2 FIR filters

1.

The order of the filter should be 36, the magnitude response and phase response are as below.The phase response of IIR filter is not linear, but the one of FIR filter is linear and stable. The FIR filter has better phase response, but meanwhile, it needs higher level in orders and the ripples in the stop-band will be much worse.

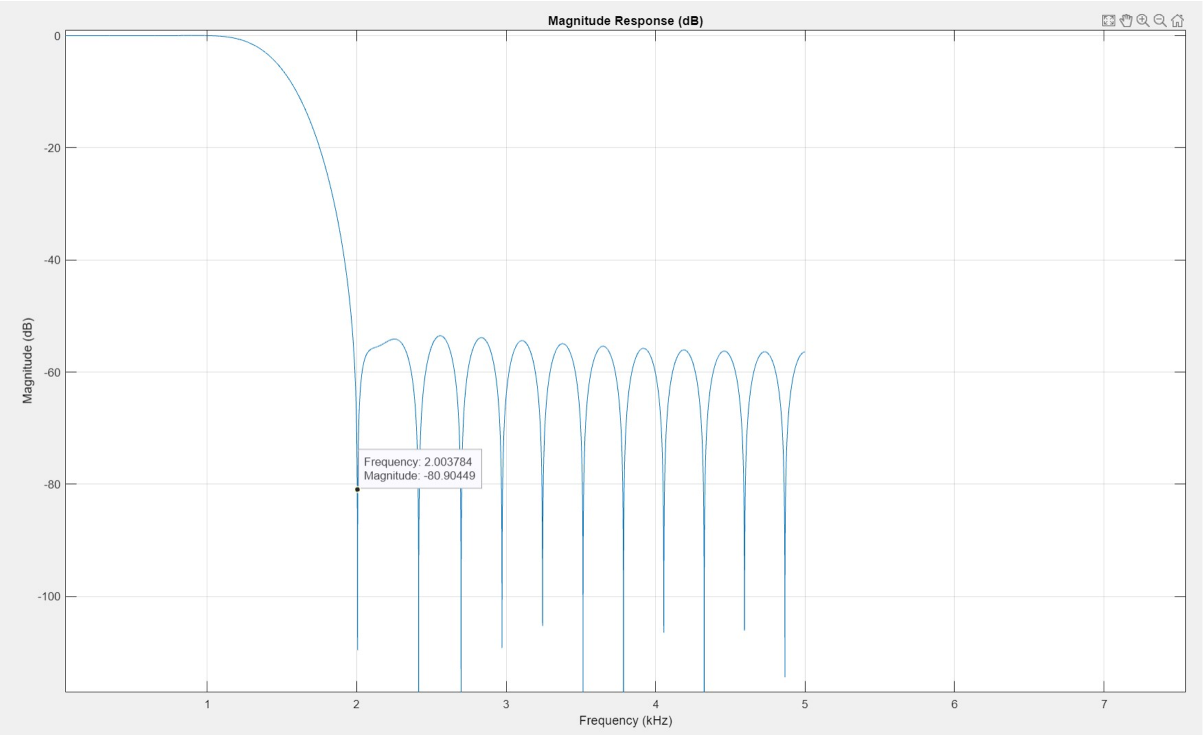

Figure 8. Magnitude Response of FIR filter

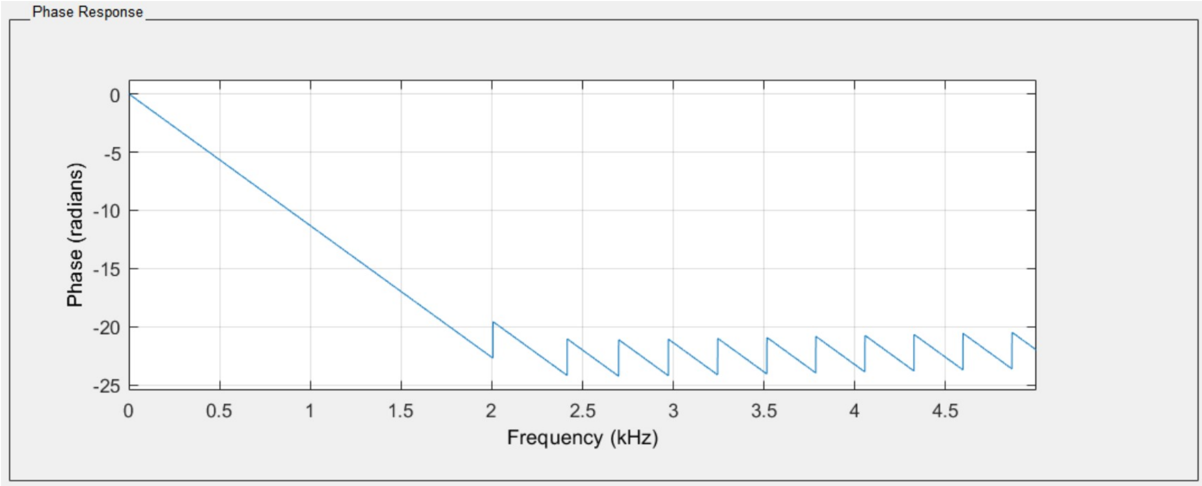

Figure 9. Phase Response of FIR filter

The impulse response of the filter is Type I, because it is axis symmetric and has an odd number of points.

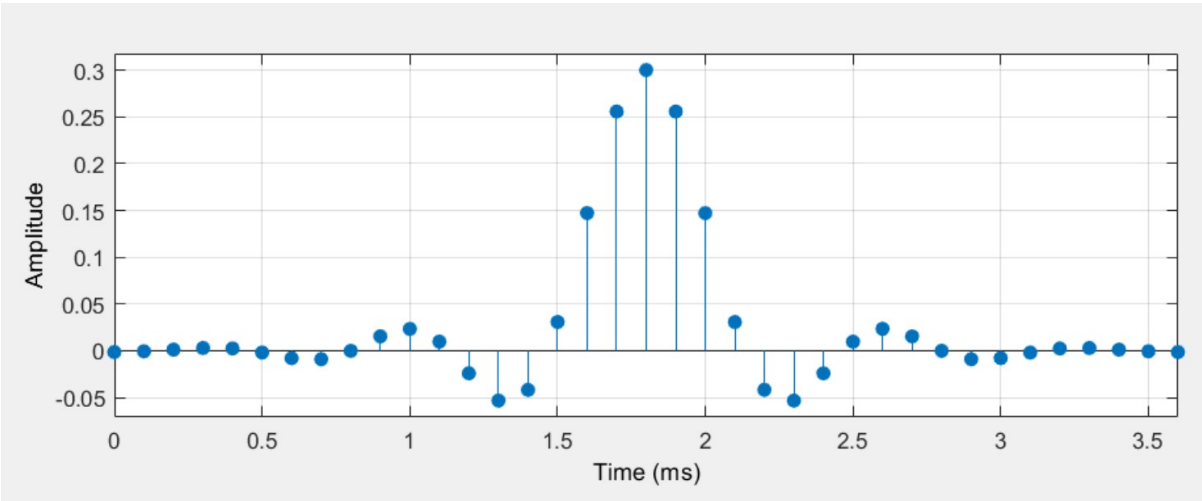

Figure 10. Impulse Response of FIR filter

2.

Using **Barlett window**, the order 6 will be necessary. The magnitude response is as follow. It has minimum order to achieve the goal that 40dB attenuation before 15kHz. But meanwhile it has a wide range of transition region in order to decrease the order of the goal which is not so ideal.

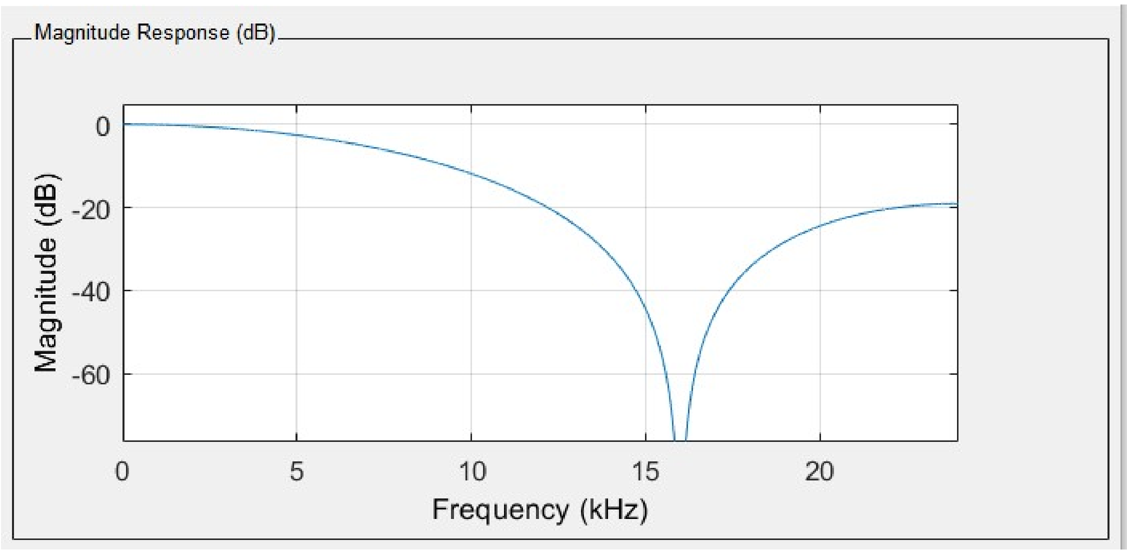

Figure 11. Magnitude Response of Barlett window

Using **Kaiser window** with same order(29), the stop-band is more like an equal ripple and get better attenuation. When I tune the beta to 5, we can get a better response to meet the requirement.

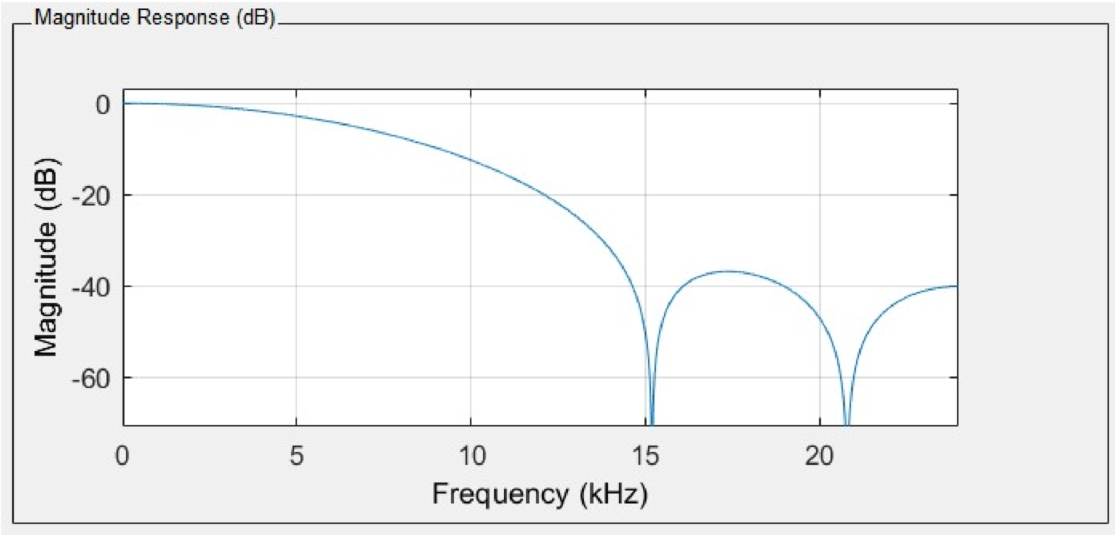

Figure 12. Magnitude Response of Kaiser Window(beta = 5)

The phase response and impulse response are as below. The phase response of kaiser FIR filter is linear which can keep the phase relationship between different signals. The impulse response of this filter belongs to Type I impulse response which means shorter length, less oscillating and more smooth waveform, it means the filter will get more stable features and more reliable.

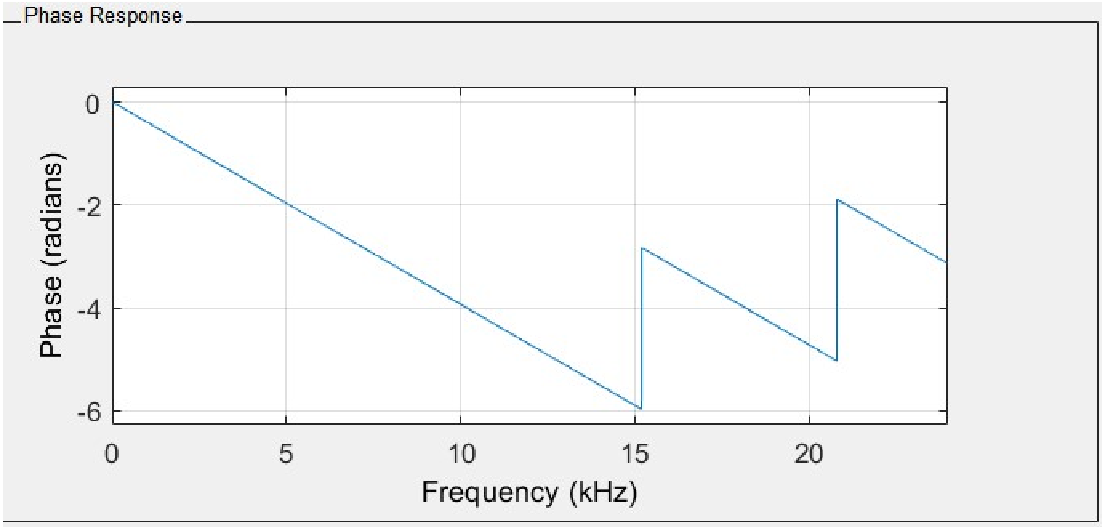

Figure 13. Phase Response of Kaiser Window(beta = 5)

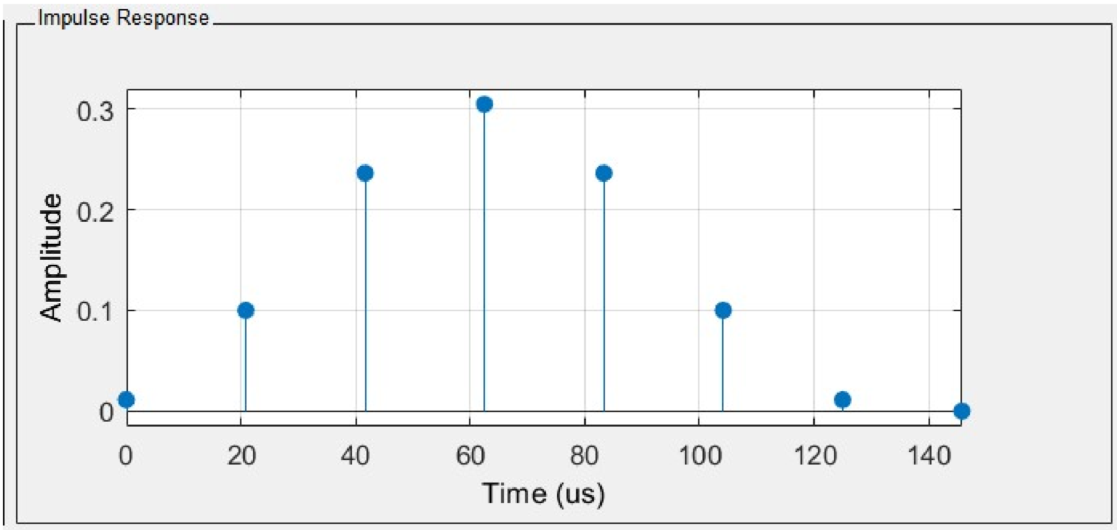

Figure 14. Impulse Response of Kaiser Window(beta = 5)

3.

Calculate with the given formula we can find that N = 18.2, so N = 19 for better performance. Try to design the Kaiser filter to meet the requirement. We found that the minimum order the filter is 23 when using Kaiser filter, 14 with Equiripple method and 15 with Generalized Equiripple method and all of them are around 19 which means the approximation is valid.

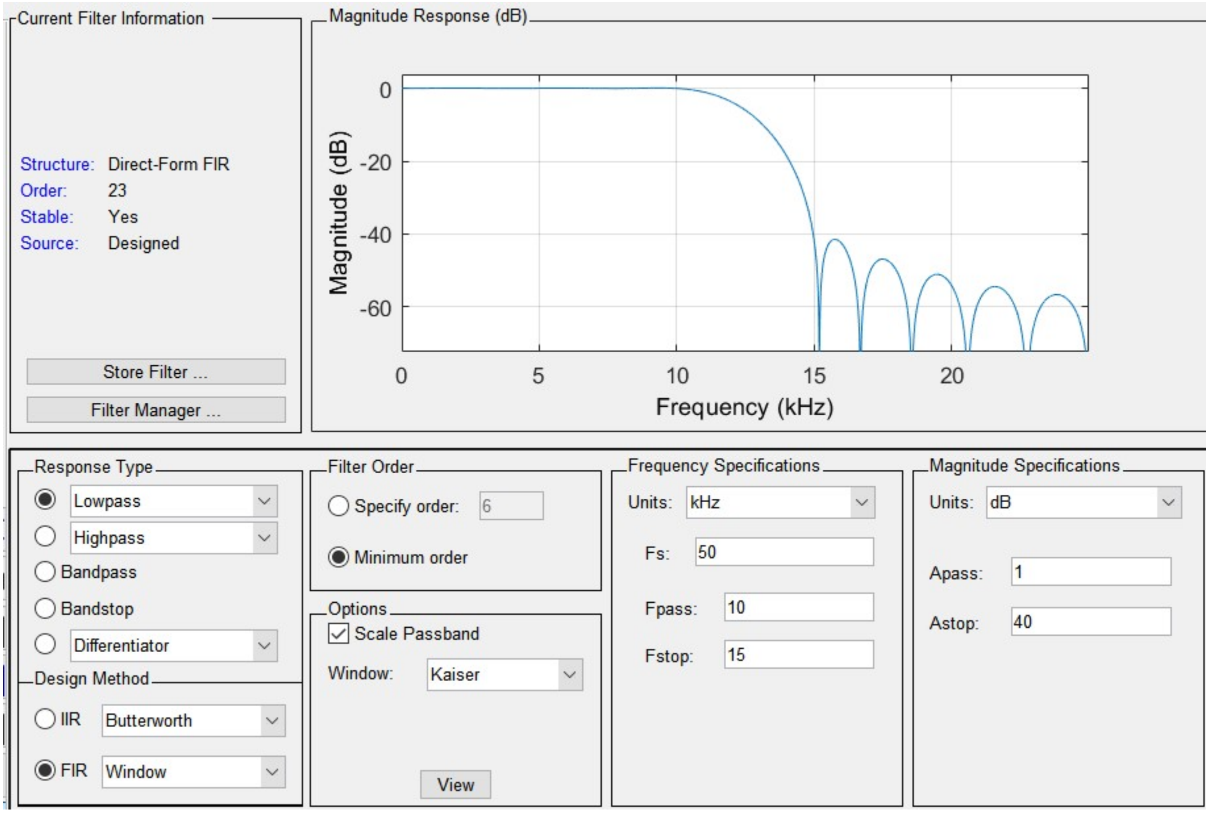

Figure 15. Kaiser filter to meet the requirement

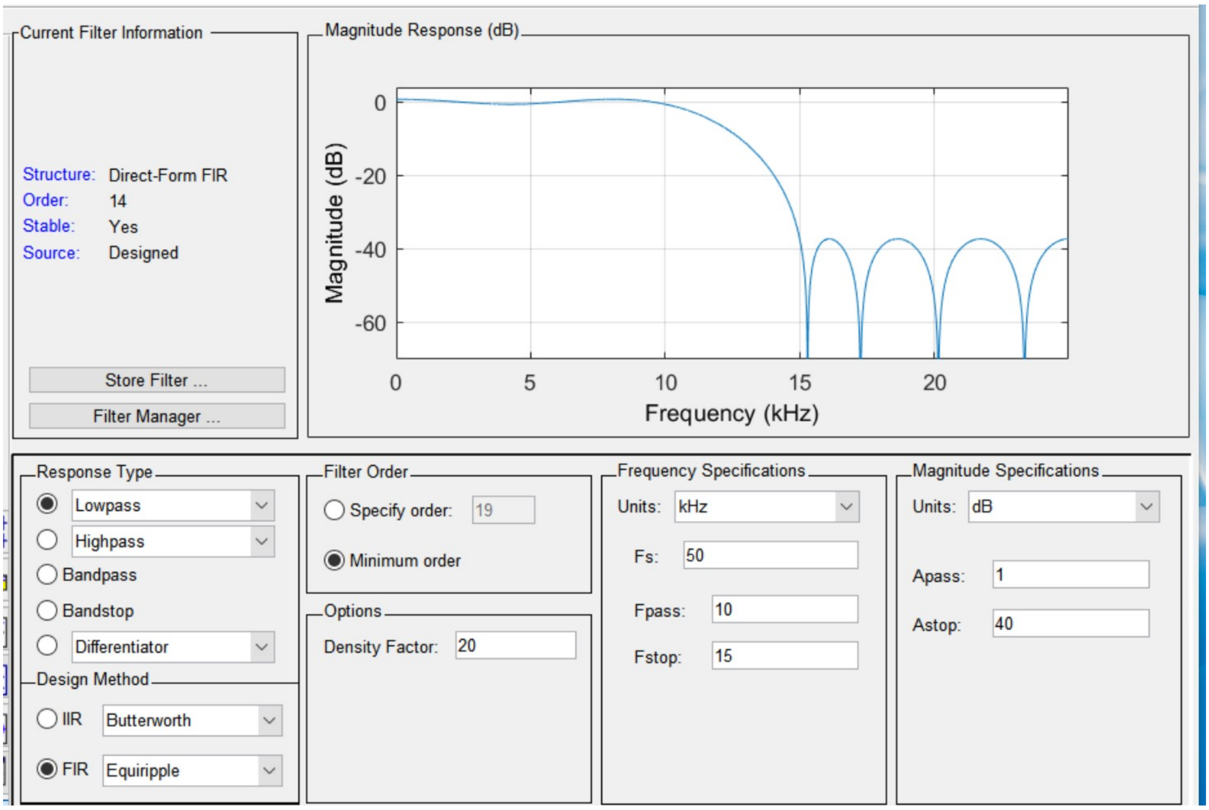

Figure 16. Equiripple filter to meet the requirement

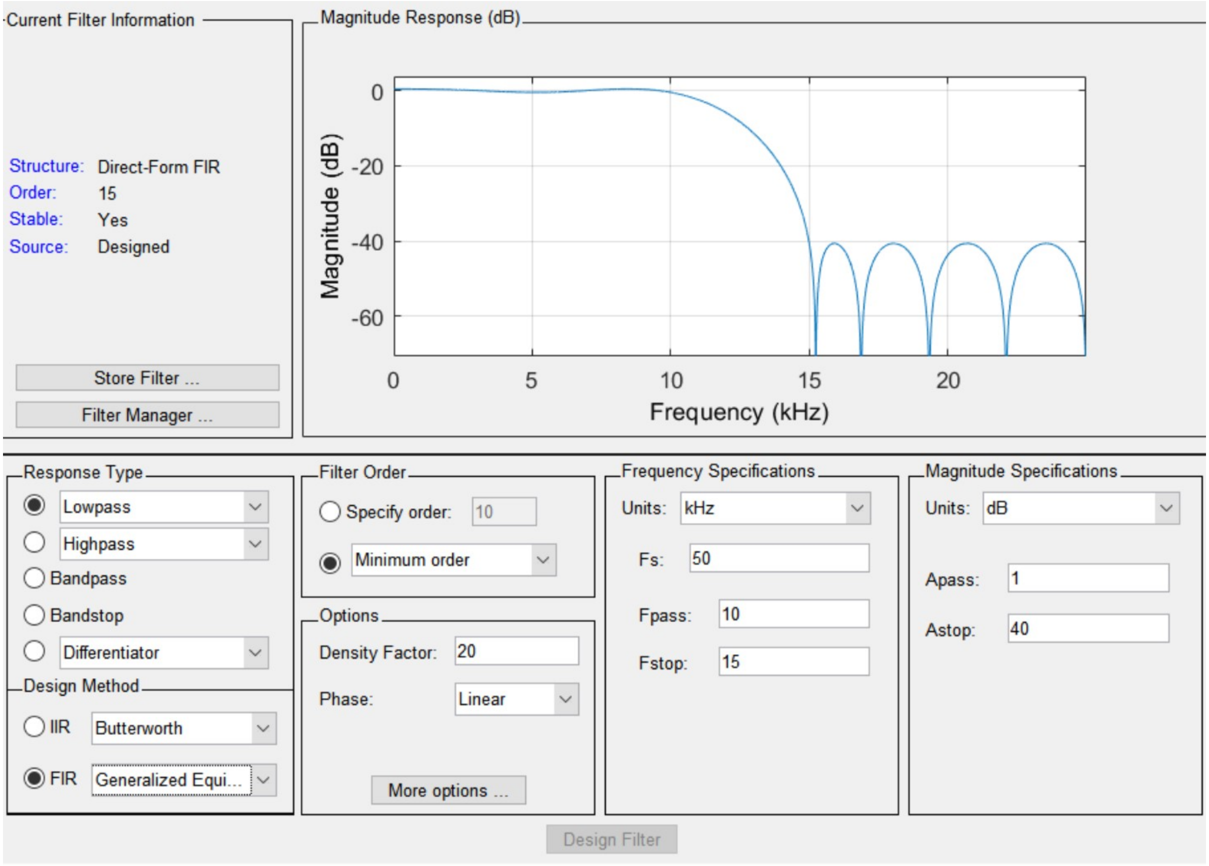

Figure 17. Generalized Equiripple filter to meet the requirement

1.3 Restore a speech file with a suitable filter

2.

Using Butterworth filter with parameters as below, the MSE has been reduced from 0.1115 to 0.0017.

```
% parameter of the filter
filter_type = 'butter';
Fpass = 1000; % First Passband Frequency
Fstop = 3000; % Second Stopband Frequency
Apass = 1; % Passband Ripple (dB)
Astop = 80; % Second Stopband Attenuation (dB)
match = 'stopband'; % Ban d to match exactly
```

To achieve the stretch goal, I built up a function to try different parameters of the filter.By listening to the result and observe the MSE of the filter, I finally chose the filter which has the parameter marked in green.

Table 2. Compare performance

type	Fpass	Fstop	Astop	MSE	Order

butter	800	3500	60	0.00071	3
butter	1000	3000	60	0.001125	5
butter	1000	3500	60	0.000437	4
butter	1000	3500	70	0.000702	4
butter	1000	3500	80	0.001069	4
butter	1000	3800	60	0.000169	3
butter	1100	3500	60	0.000437	4
cheby1	1000	3500	60	0.000652	3

The method and the reason why I choose this parameter can be concluded as below. Considering the MSE performance and the listening experience, I made this choice.

Table 3. Result of different parameters

	lower	current	higher
Fpass	poor MSE	1000	higher filter order
Fstop	poor MSE	3500	better MSE, but poor listening experience because introduce note noise
Astop	better MSE, but poor listening experience because cant remove noise effectively	60	poor MSE
filter type	cheby1 poor in MSE	Butterworth	cheby1 poor in MSE

3.

Insert distortion with note 30 and process with a high pass filter, the waveform are as below. We got MSE = 0.0030 with the parameters below.

```
filter_type = 'butter';

Fpass = 300; % First Passband Frequency

Fstop = 50; % Second Stopband Frequency

Apass = 1; % Passband Ripple (dB)

Astop = 80; % Second Stopband Attenuation (dB)

match = 'stopband'; % Ban d to match exactly
```

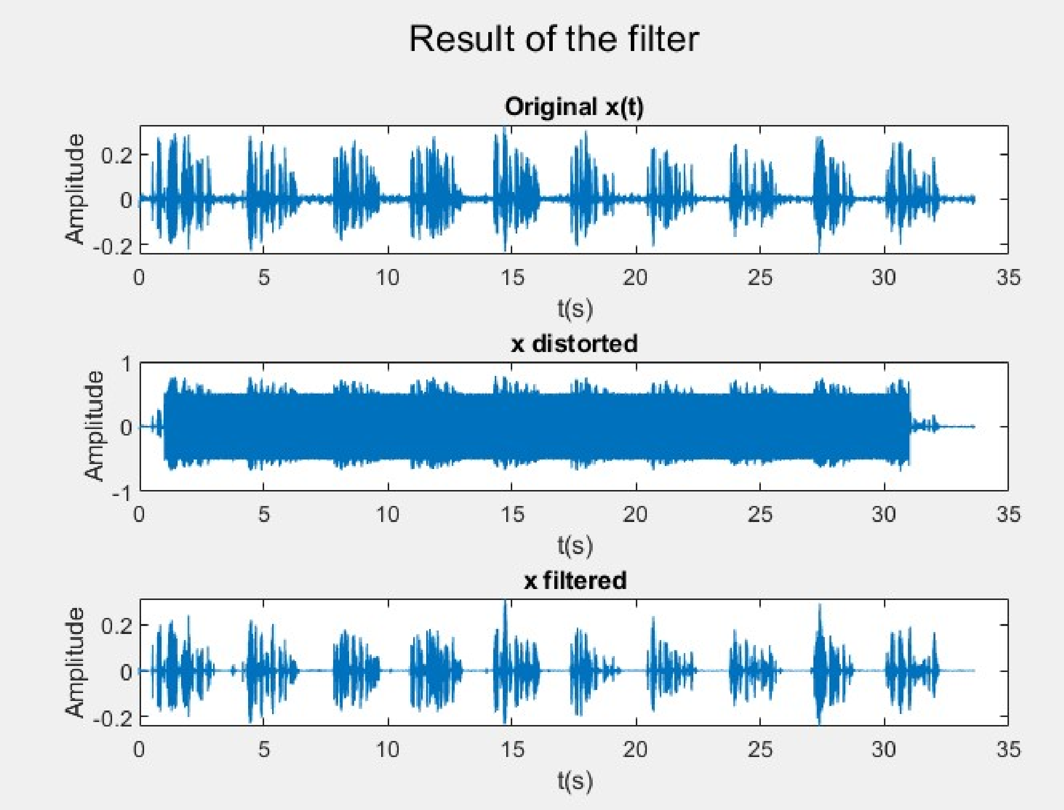

Figure 18. Process with high-pass filter(note = 30)

4.

Insert distortion with note 30 and 105 and process with a high pass filter and a low pass filter, the waveform are as below. We got MSE = 0.0027 with the filter parameters like below.

```
% highpass filter

filter_type = 'butter';

Fpass = 300; % First Passband Frequency

Fstop = 50; % Second Stopband Frequency

Apass = 1; % Passband Ripple (dB)

Astop = 80; % Second Stopband Attenuation (dB)

match = 'stopband'; % Ban d to match exactly


% lowpass filter

filter_type = 'butter';

Fpass = 1000; % First Passband Frequency

Fstop = 3500; % Second Stopband Frequency

Apass = 1; % Passband Ripple (dB)
```

```
Astop = 60; % Second Stopband Attenuation (dB)
match = 'stopband'; % Ban d to match exactly
```

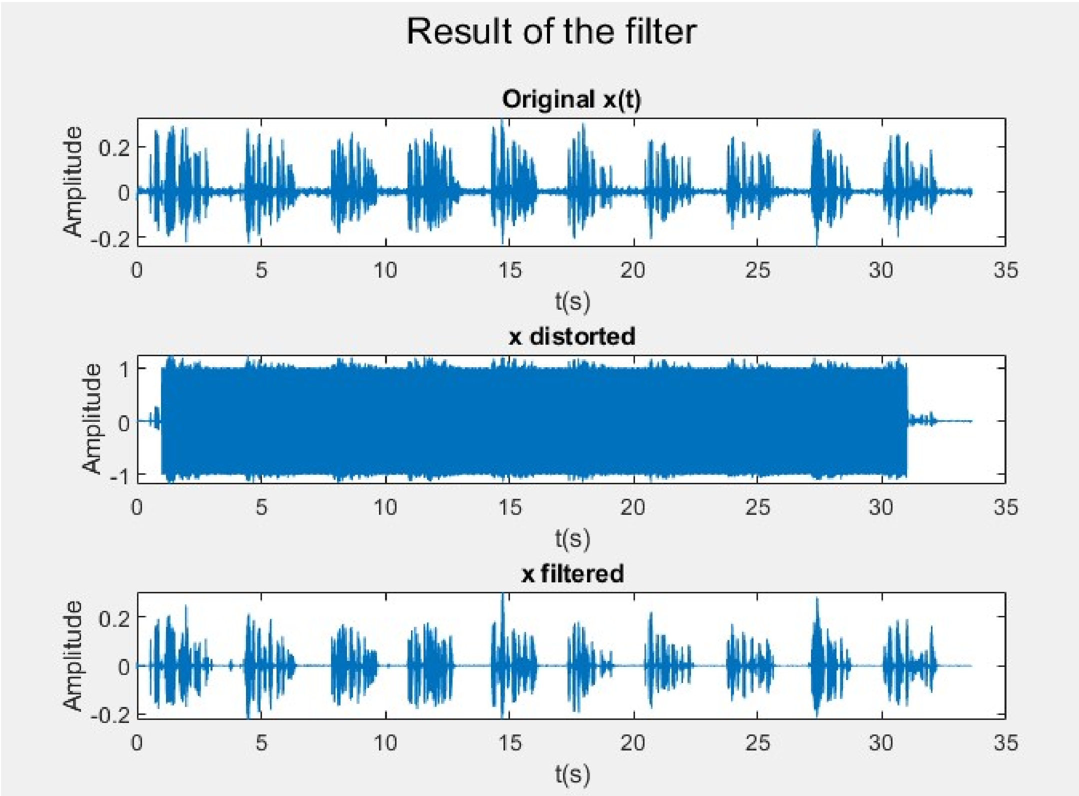

Figure 19. Process with low-pass filter and high-pass filter (note = 30, 105)