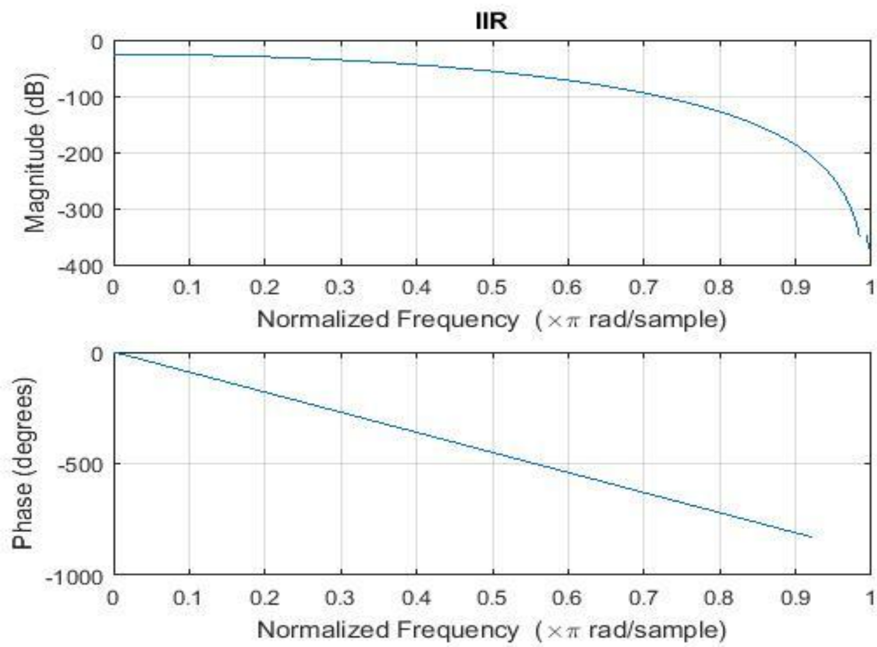
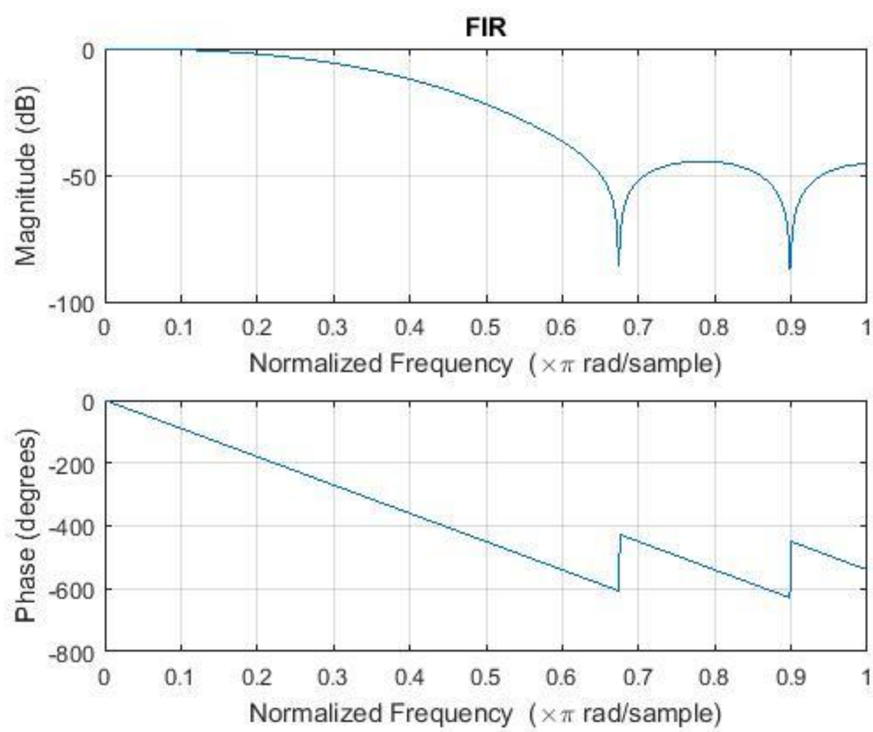
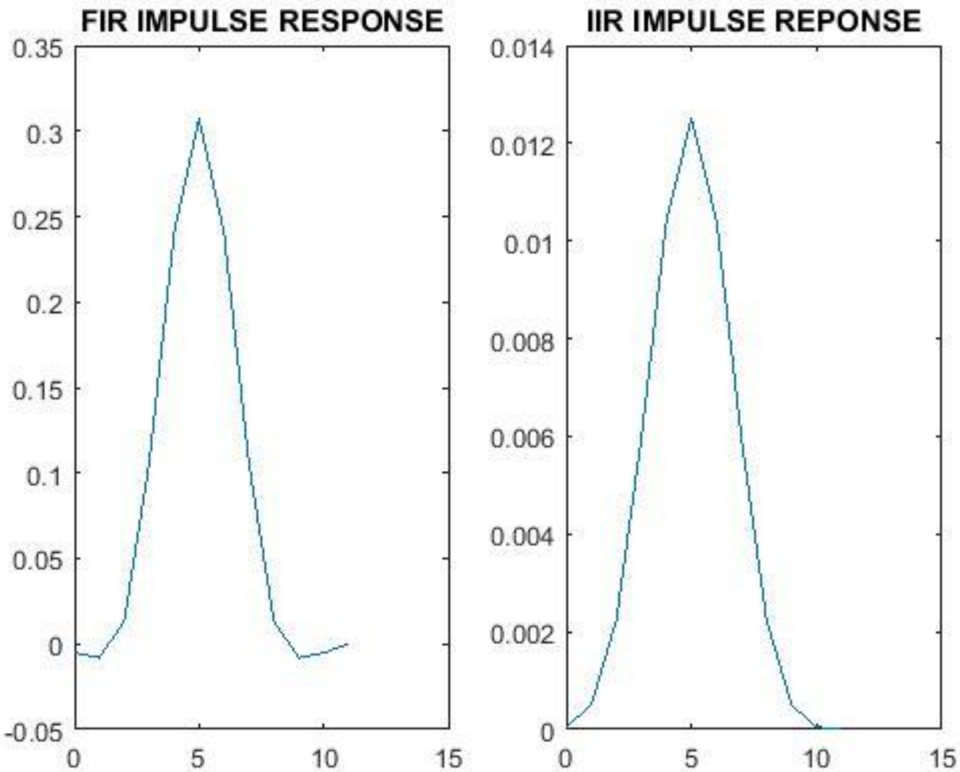
Lab 3: Digital Filtering

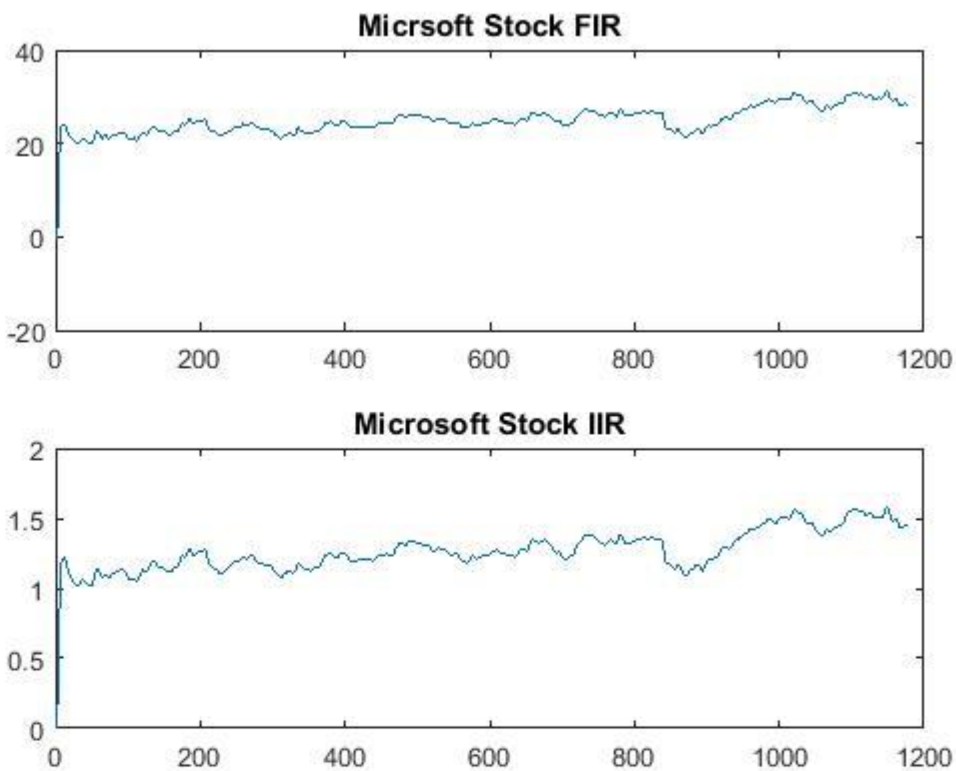
Zhimingyuan Liu

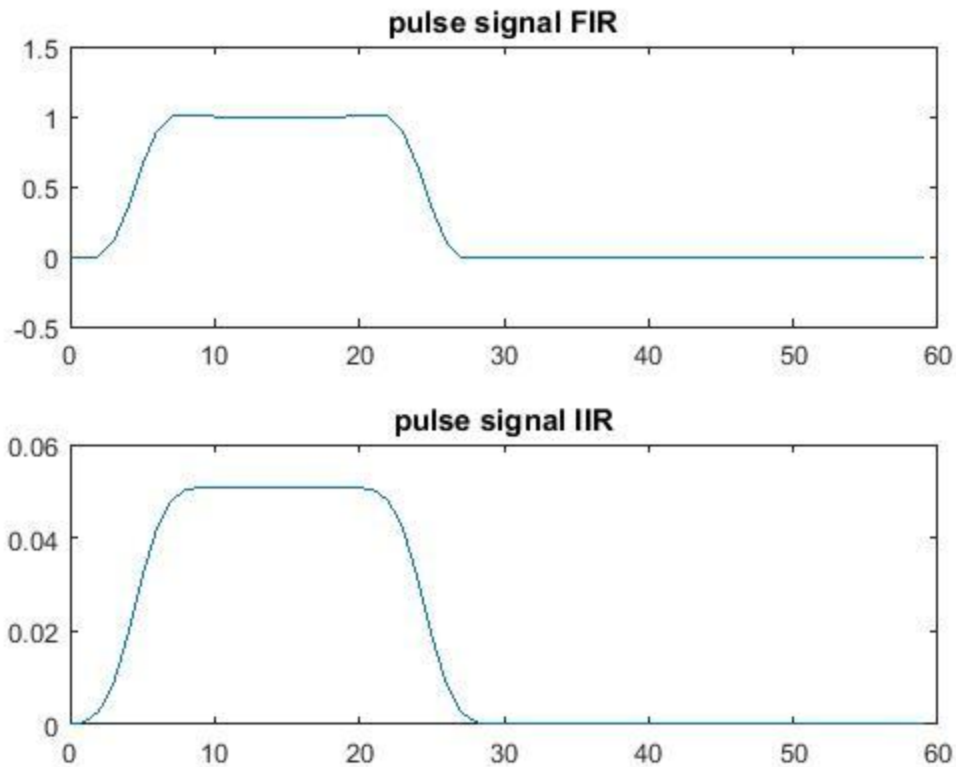
In this part 1, we learn how to use filter method and convolution method to design the filter and using impulse response to find the impulse function, then we learn how to use freqz to plot magnitude and phase.






FIR IMPULSE RESPONSE
0.35
0.3
0.25
0.2
0.15
0.1
0.05
0
-0.05
0
5
10
15
IIR IMPULSE REPONSE
0.014
0.012
0.01
0.008
0.006
0.004
0.002
0
0
5
10
15

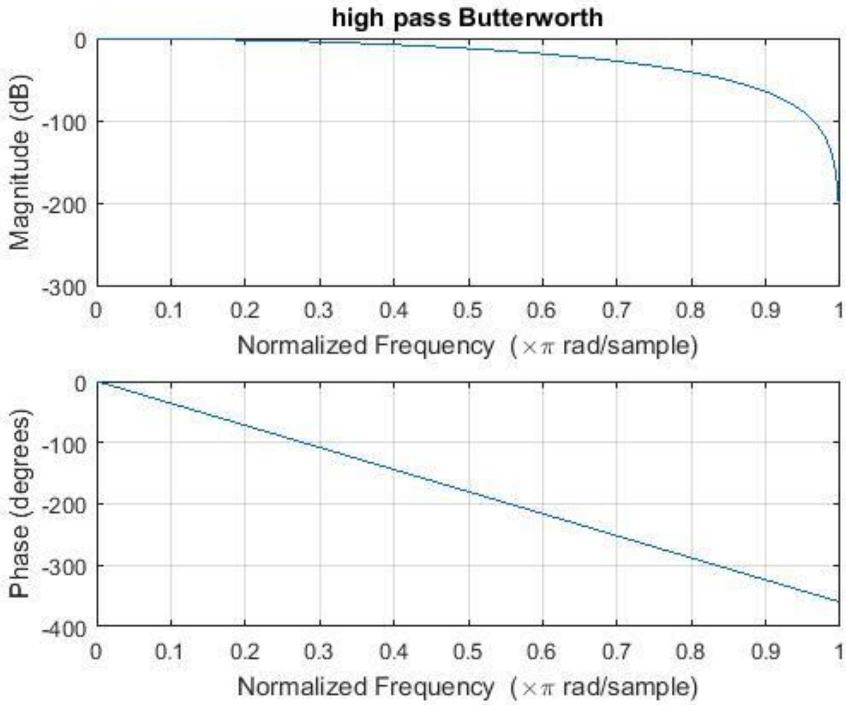

Micrsoft Stock FIR
40
20
0
-20
0
200
400
600
800
1000
1200
Microsoft Stock IIR
2
1.5
1
0.5
0
0
200
400
600
800
1000
1200



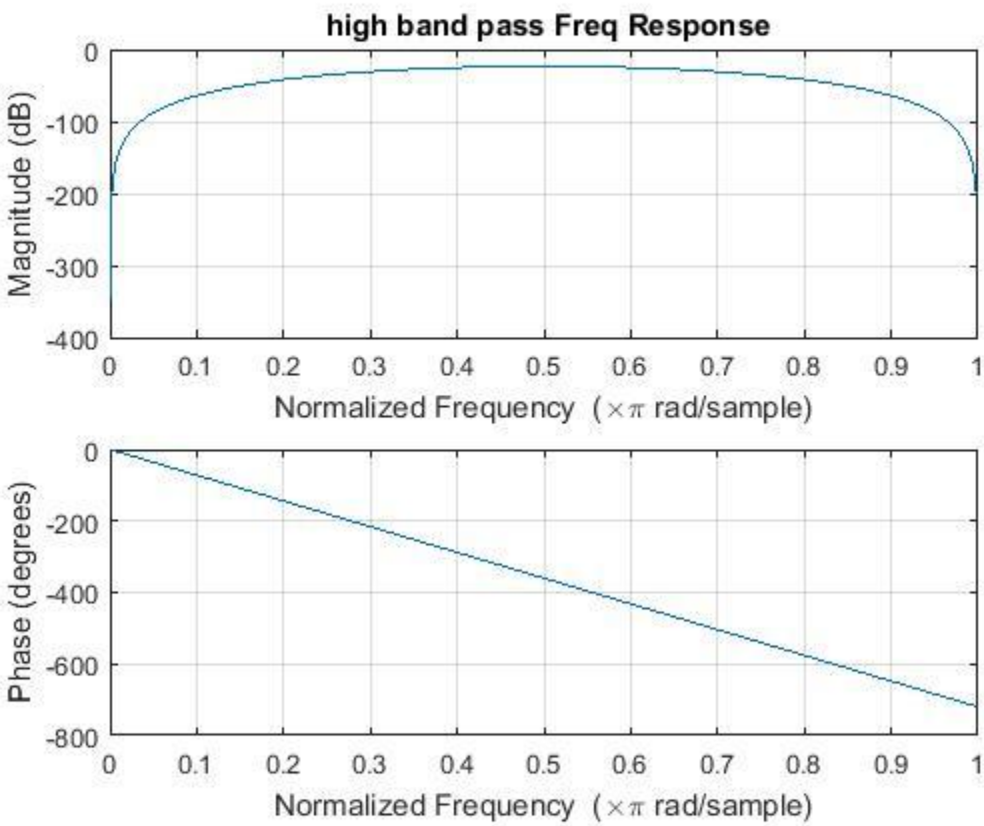
FIR and IIR both relative 0 DB at low frequency, but when the frequency becomes high, the FIR signal fluctuates. Instead IIR will decrease without any fluctuation. For the phase FIR and IRR both has same negative slope when the frequency is high, but FIR will fluctuate a high frequency. The output for IIR is smoother that FIR, due to the fluctuation.

FIR filter is better, because FIR filters have an amazing property called linear phase, Linear phase implies that the phase is linear function of frequency, it ensures that signal of all frequencies are delayed by the same amount of time, thereby eliminating the possibility of phase distortion. From the stock graph, we can see that the range is from 0 to 20, which is close to real value. For pulse signal, they both have flat response at low frequency, and IIR has more flat than FIR, but FIR has the magnitude close to one, so FIR is better.

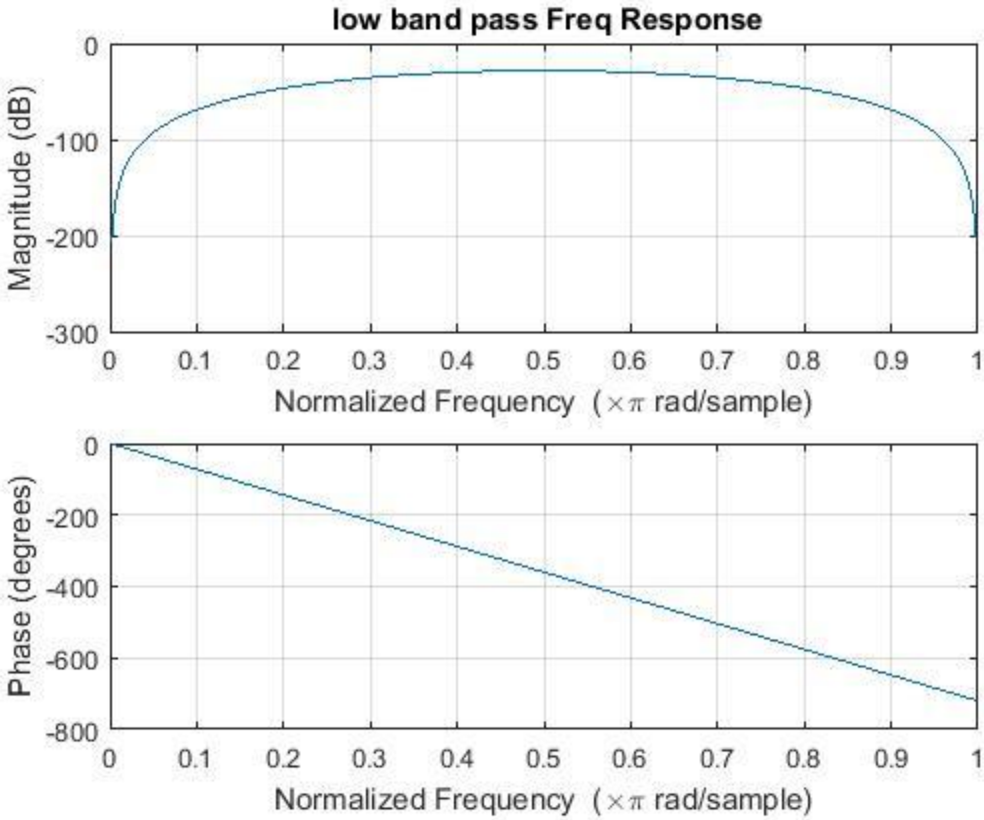
Part 3



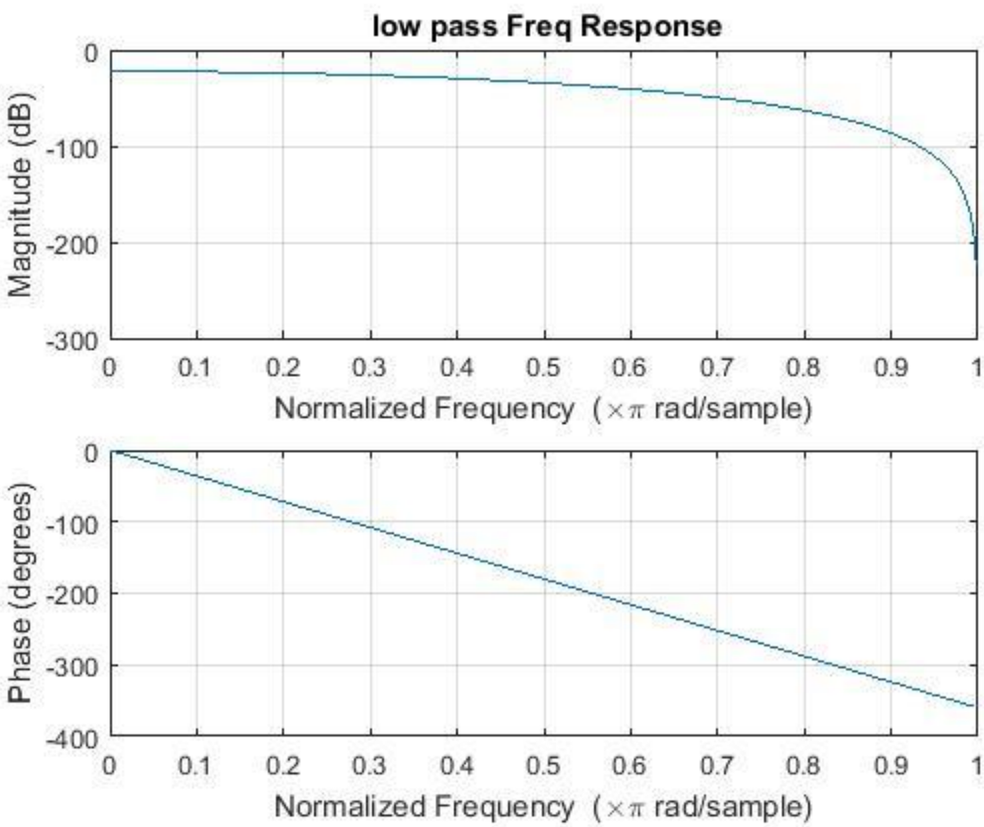
high pass Butterworth
Magnitude (dB)
0
-100
-200
-300
0 0.1 0.2 0.3 0.4 0.5 0.6 0.7 0.8 0.9 1
Normalized Frequency (×π rad/sample)
Phase (degrees)
0
-100
-200
-300
-400
0 0.1 0.2 0.3 0.4 0.5 0.6 0.7 0.8 0.9 1
Normalized Frequency (×π rad/sample)



high band pass Freq Response
Magnitude (dB)
0
-100
-200
-300
-400
0 0.1 0.2 0.3 0.4 0.5 0.6 0.7 0.8 0.9 1
Normalized Frequency (×π rad/sample)
Phase (degrees)
0
-200
-400
-600
-800
0 0.1 0.2 0.3 0.4 0.5 0.6 0.7 0.8 0.9 1
Normalized Frequency (×π rad/sample)



low band pass Freq Response
Magnitude (dB)
0
-100
-200
-300
Phase (degrees)
0
-200
-400
-600
-800
0 0.1 0.2 0.3 0.4 0.5 0.6 0.7 0.8 0.9 1
Normalized Frequency (×π rad/sample)



low pass Freq Response
Magnitude (dB)
0
-100
-200
-300
Phase (degrees)
0
-100
-200
-300
-400
0 0.1 0.2 0.3 0.4 0.5 0.6 0.7 0.8 0.9 1
Normalized Frequency (×π rad/sample)

There are four sections for this filter, which are low pass(0, 1/5), low band pass(1/5, 2/5), high band pass(2/5, 2/3), and high pass filter(2/3, 1). The reason we use Butterworth filter because it is close the ideal filter produce less filter coefficient. So I multiply each filter with amplitude [`0.5,5,5,0.5`], so convert it to the DB is (-6db, 18db, 18db,-6db).

```
h1 = butter(4, 1/5);
h2 = butter(4, [1/5 2/5]);
h3 = butter(4, [2/5 2/3]);
h4 = butter(4, 2/3);
```