```
% ZERO PHASE FILTERING : FIRST ORDER BUTTERWORTH FILTER %


fc = 1000; %Cut off Frequency  = 1000 Hz
fs = 16000; %Sampling  Frequency = 16000 Hz
t=0:1/fs:0.05;
x = sin(2*pi*fc*t);

%[b,a] = cheby1(1,3,fc/(fs/2));
[b,a] =  butter(1,(2*fc)/fs);


samples = size(x,2); %Determining the size of the input
index = 1; %Pointer to keep track of peak and low samples
blk = 100; %Block Size for Block Processing




%Preprocessing the input signal
blk_no = samples/blk;

%To check if Number of Blocks is an Integer or not
check_int = isinteger(blk_no);
if(check_int==0)
    blk_no = ceil(blk_no); %Convert block size to an integer value
end


mod_op = mod(samples,blk);
if(mod_op)
    append_zero = blk - mod_op;
    x = [x,zeros(1,append_zero)]; %Appending Zeros to the input signal to make the↙
block size dynamic
end


%Block Processing of Input Data for Peak Amplitude Prediction
z=[];
temp=[];
start =1;
endb=blk;
zi=0; %Setting initial condition as zero

for j = 1 :blk_no
    temp = x(start:endb); %Block of Data
    [y,zf]=filter(b,a,temp,zi); %Filtering the Block of data
    y=fliplr(y); %Reversing the filtered signal
    zi=zf;
    [y,zf]=filter(b,a,y,zi); %Filtering the Reversed Signal
    y=fliplr(y); %Flipping the signal to get the final output
```

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    zi=zf; %Setting the initial condition for the next block
    z=[z y]; %Concatenating the signal to get the output

    start = start+blk; %Incrementing to next block
    endb = endb+blk;
end


y_filt= filtfilt(b,a,x); %Finding the filtfilt output
y_reg = filter(b,a,x); %Filtering to check the phase response
%Plotting the input sample
figure
subplot(3,1,1)
plot(x);
grid on
xlabel 'Number of samples'
ylabel 'Amplitude';
legend('Input Signal');


%plotting the regular filter output
subplot(3,1,2)
plot(y_reg,'r');
grid on
xlabel 'Number of samples'
ylabel 'Amplitude';
title 'First Order Butterworth Filter';
legend('Regular Filtered Output');

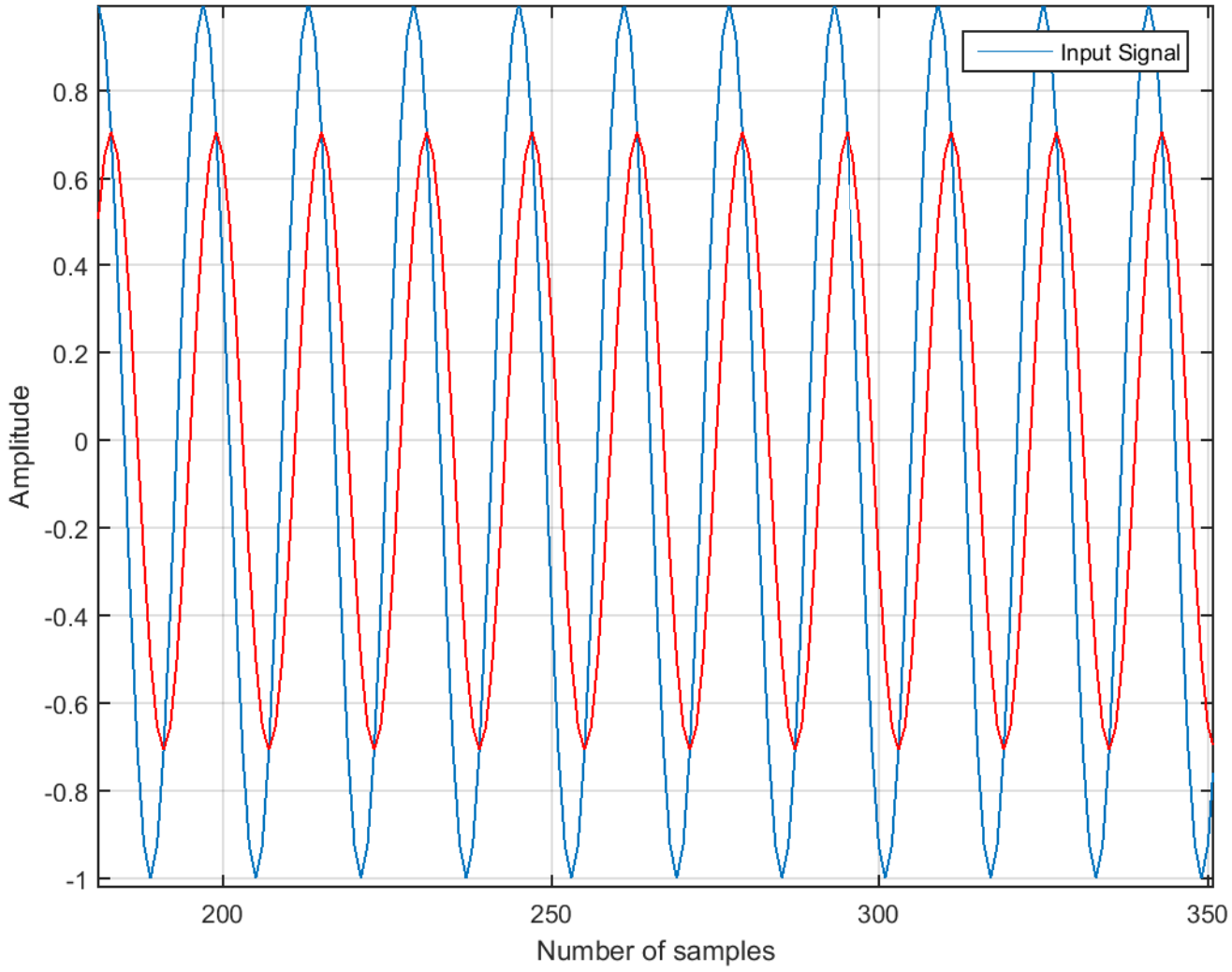
%Plotting the zero phase filtered output
subplot(3,1,3)
plot(x)
hold on
plot(z,'r-');
plot(y_filt,'black');
grid on
hold off
xlabel 'Number of samples'
ylabel 'Amplitude';
title 'Zero Phased Filtered Output';
legend('Input Signal','Processed Signal','FiltFilt Output');

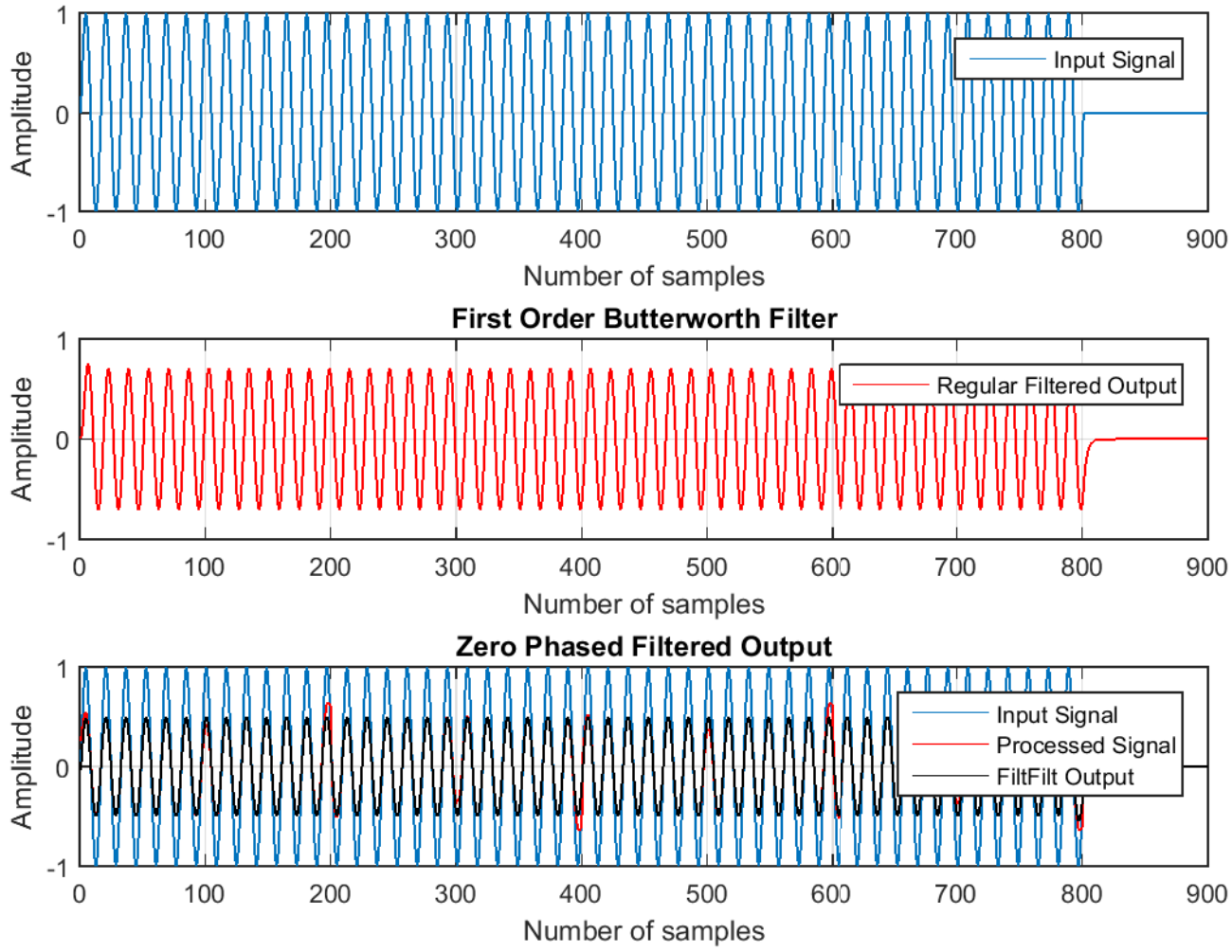
%Plotting the zero phase output
figure
plot(x)
hold on
plot(z,'r-');
plot(y_filt,'black');
grid on
hold off
xlabel 'Number of samples'
```

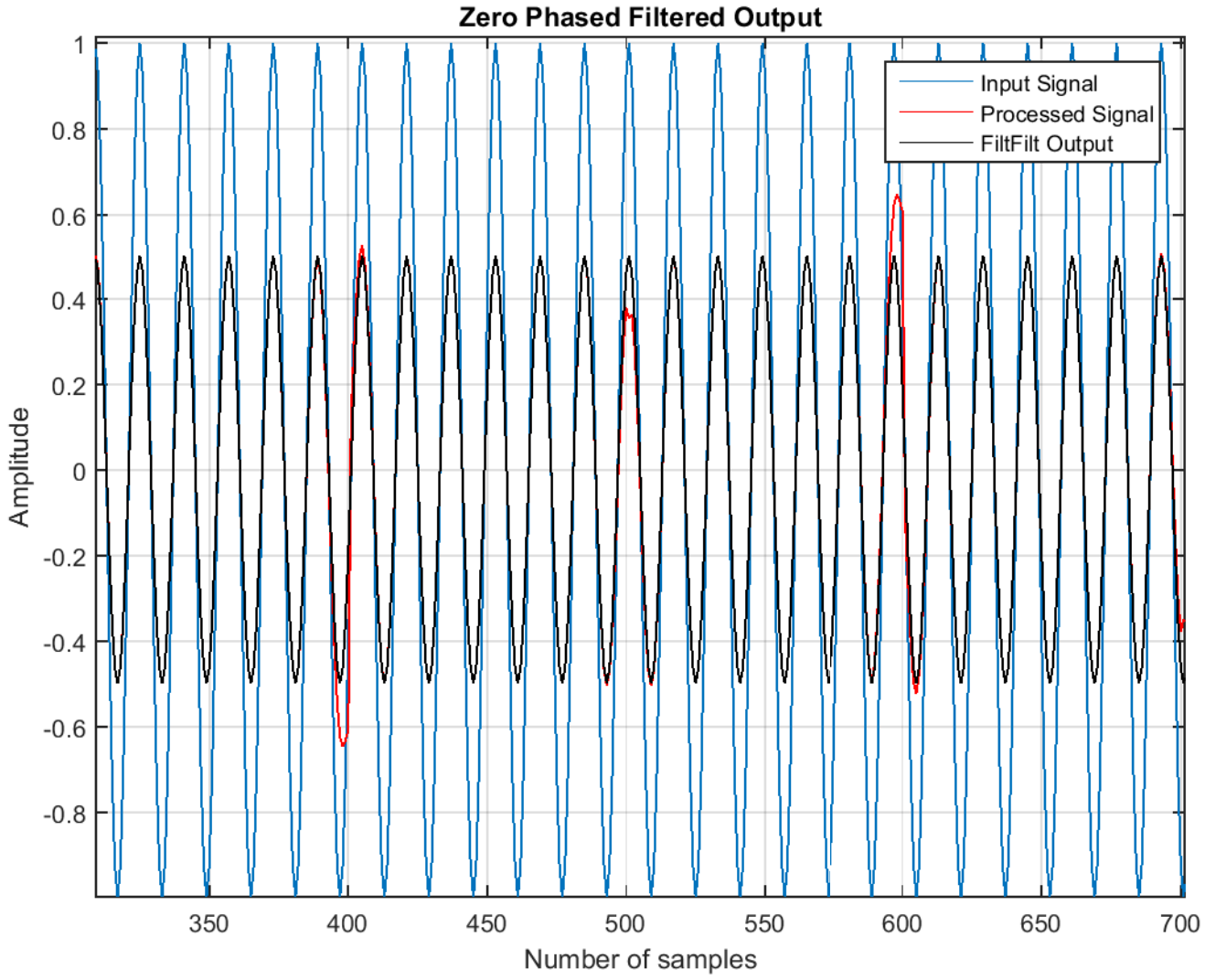
```
ylabel 'Amplitude';
title 'Zero Phased Filtered Output';
legend('Input Signal','Processed Signal','FiltFilt Output');

%Plot to check the phase difference in signals
figure
plot(x);
hold on
plot(y_reg,'r');
grid on
xlabel 'Number of samples'
ylabel 'Amplitude';
legend('Input Signal');


h=freqz(y_reg);
figure
plot(h);
```


Input Signal
0.8
0.6
0.4
0.2
0
-0.2
-0.4
-0.6
-0.8
-1
Amplitude
200
250
300
350
Number of samples


Input Signal
Amplitude
Number of samples
First Order Butterworth Filter
Regular Filtered Output
Amplitude
Number of samples
Zero Phased Filtered Output
Input Signal
Processed Signal
FiltFilt Output
Amplitude
Number of samples


Zero Phased Filtered Output
Input Signal
Processed Signal
FiltFilt Output
Amplitude
Number of samples
1
0.8
0.6
0.4
0.2
0
-0.2
-0.4
-0.6
-0.8
350
400
450
500
550
600
650
700

```
% ZERO PHASE FILTERING : FIRST ORDER TYPE-I CHEBYSHEV FILTER %


fc = 1000; %Cut off Frequency  = 1000 Hz
fs = 16000; %Sampling  Frequency = 16000 Hz
t=0:1/fs:0.05;
x = sin(2*pi*fc*t);

[b,a] = cheby1(1,3,fc/(fs/2));
%[b,a] =  butter(1,(2*fc)/fs);


samples = size(x,2); %Determining the size of the input
index = 1; %Pointer to keep track of peak and low samples
blk = 100; %Block Size for Block Processing




%Preprocessing the input signal
blk_no = samples/blk;

%To check if Number of Blocks is an Integer or not
check_int = isinteger(blk_no);
if(check_int==0)
    blk_no = ceil(blk_no); %Convert block size to an integer value
end


mod_op = mod(samples,blk);
if(mod_op)
    append_zero = blk - mod_op;
    x = [x,zeros(1,append_zero)]; %Appending Zeros to the input signal to make the↙
block size dynamic
end


%Block Processing of Input Data for Peak Amplitude Prediction
z=[];
temp=[];
start =1;
endb=blk;
zi=0; %Setting initial condition as zero

for j = 1 :blk_no
    temp = x(start:endb); %Block of Data
    [y,zf]=filter(b,a,temp,zi); %Filtering the Block of data
    y=fliplr(y); %Reversing the filtered signal
    zi=zf;
    [y,zf]=filter(b,a,y,zi); %Filtering the Reversed Signal
    y=fliplr(y); %Flipping the signal to get the final output
```

```
    zi=zf; %Setting the initial condition for the next block
    z=[z y]; %Concatenating the signal to get the output

    start = start+blk; %Incrementing to next block
    endb = endb+blk;
end


y_filt= filtfilt(b,a,x); %Finding the filtfilt output
y_reg = filter(b,a,x); %Filtering to check the phase response
%Plotting the input sample
figure
subplot(3,1,1)
plot(x);
grid on
xlabel 'Number of samples'
ylabel 'Amplitude';
legend('Input Signal');


%plotting the regular filter output
subplot(3,1,2)
plot(y_reg,'r');
grid on
xlabel 'Number of samples'
ylabel 'Amplitude';
title 'First Order Butterworth Filter';
legend('Regular Filtered Output');

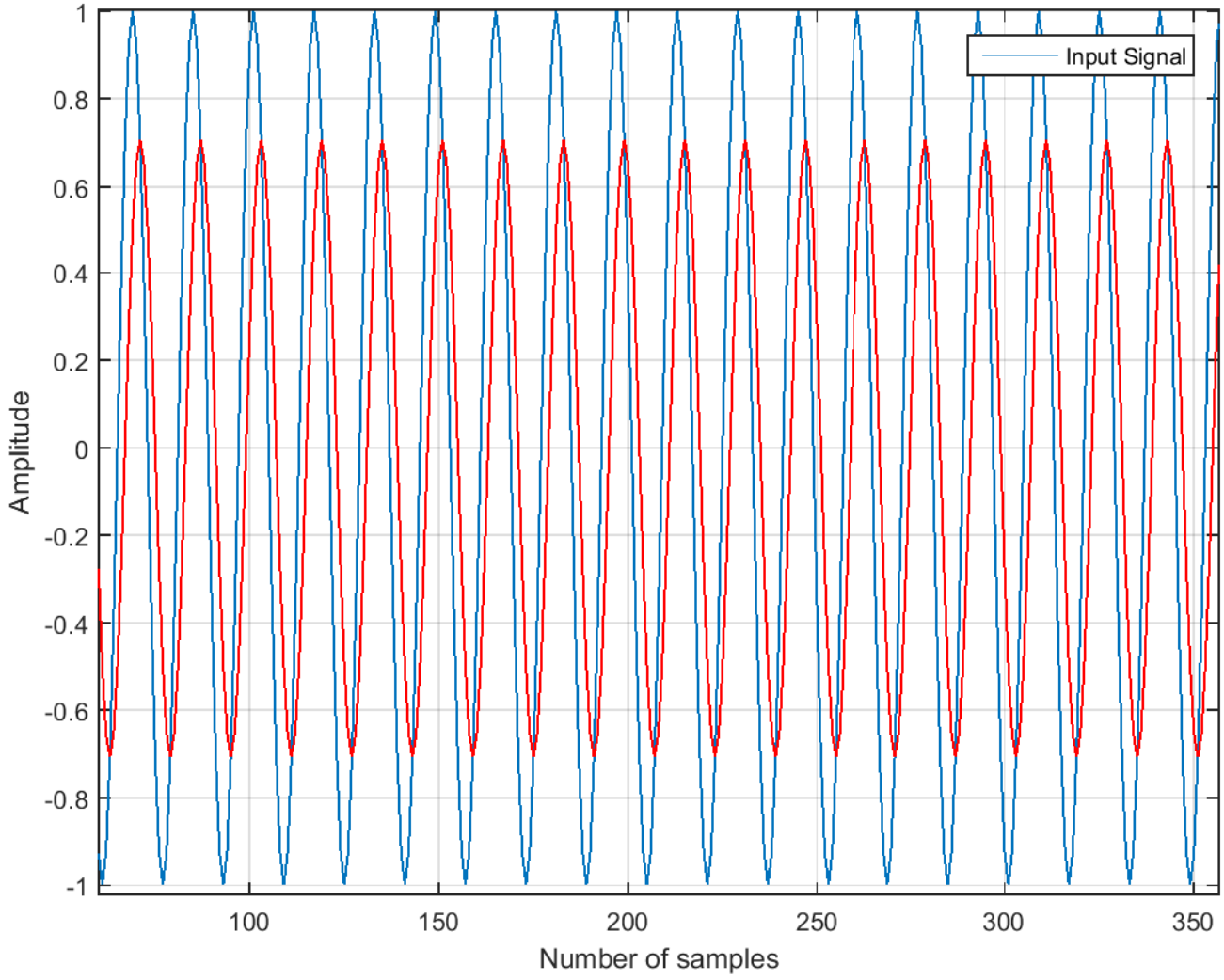
%Plotting the zero phase filtered output
subplot(3,1,3)
plot(x)
hold on
plot(z,'r-');
plot(y_filt,'black');
grid on
hold off
xlabel 'Number of samples'
ylabel 'Amplitude';
title 'Zero Phased Filtered Output';
legend('Input Signal','Processed Signal','FiltFilt Output');

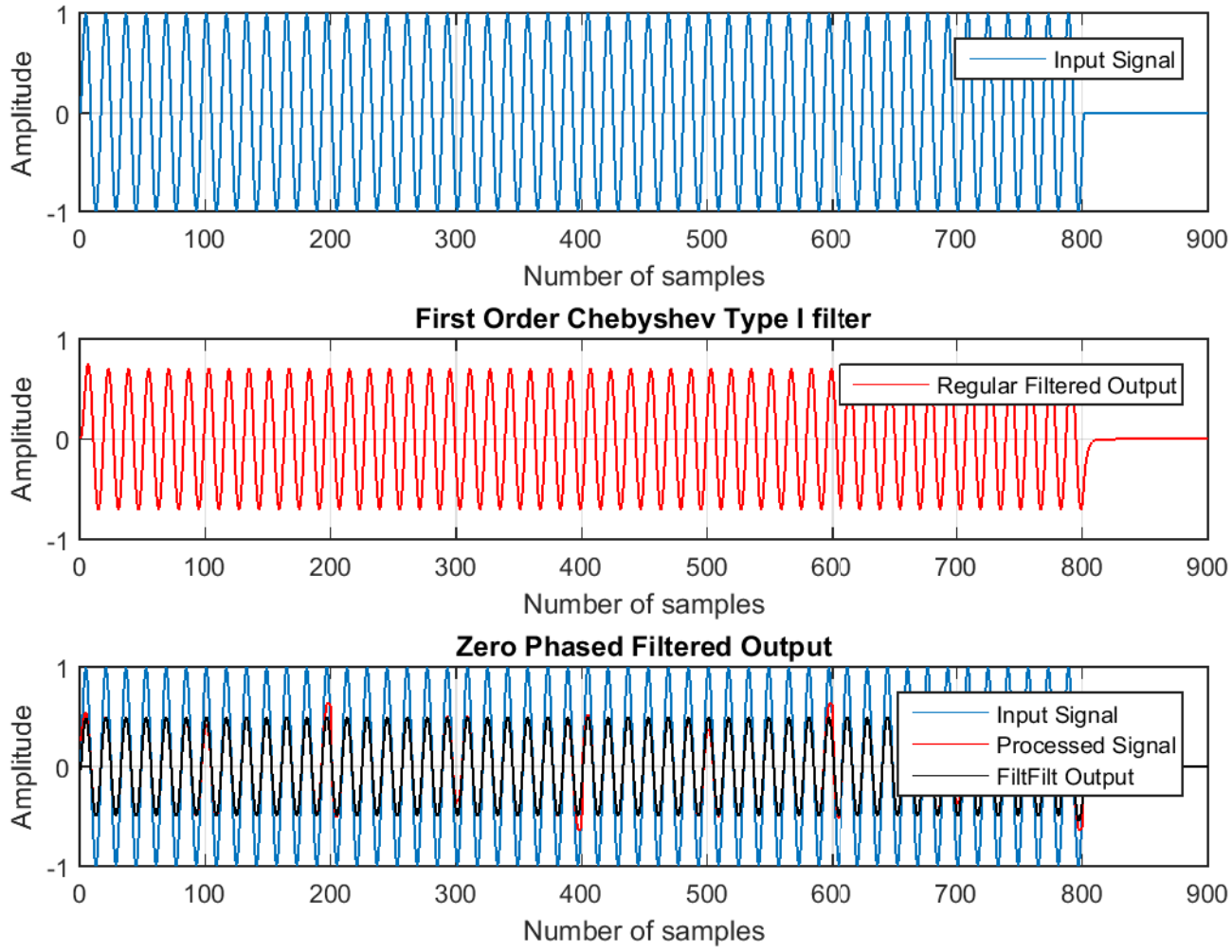
%Plotting the zero phase output
figure
plot(x)
hold on
plot(z,'r-');
plot(y_filt,'black');
grid on
hold off
xlabel 'Number of samples'
```

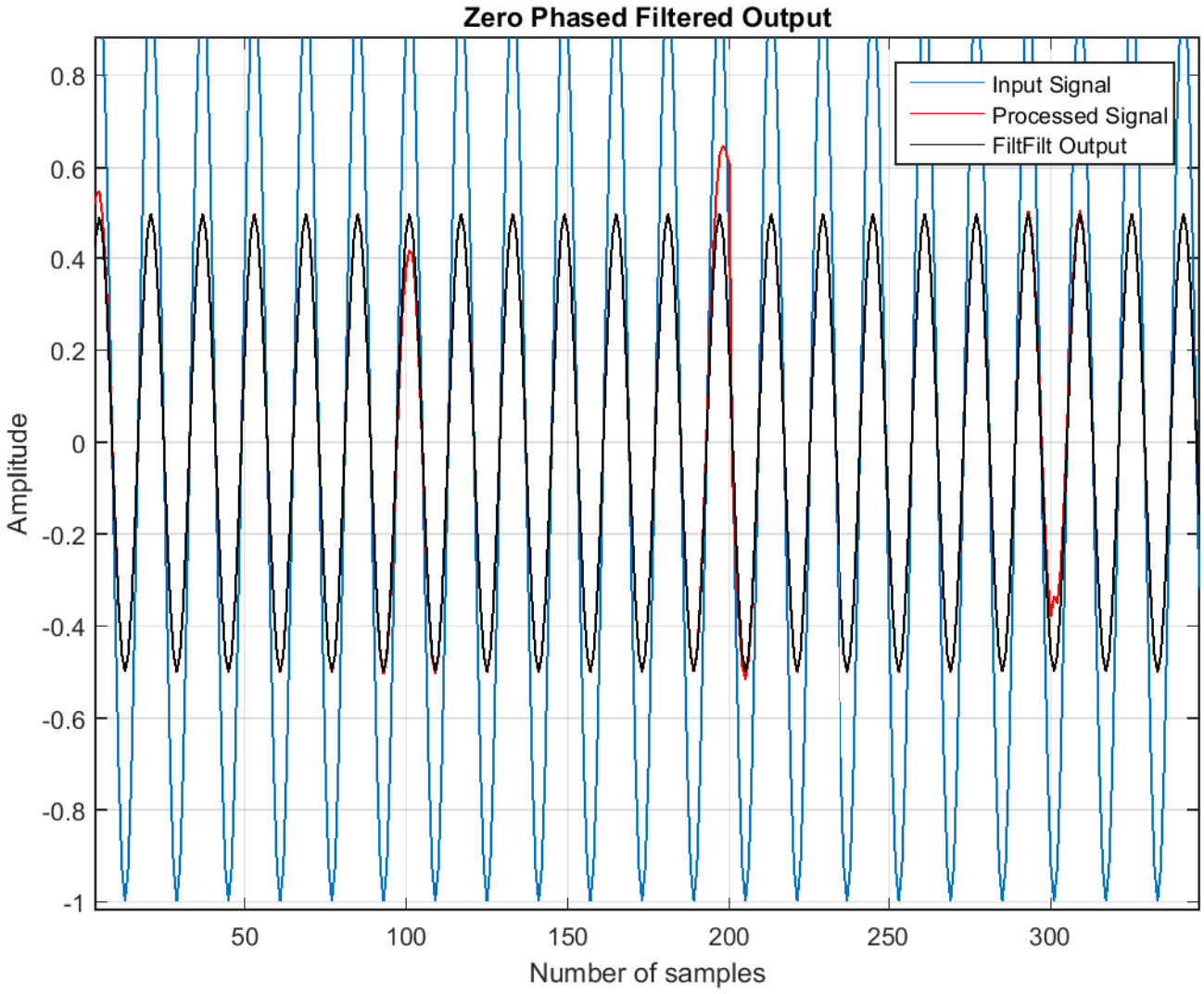
```
ylabel 'Amplitude';
title 'Zero Phased Filtered Output';
legend('Input Signal','Processed Signal','FiltFilt Output');

%Plot to check the phase difference in signals
figure
plot(x);
hold on
plot(y_reg,'r');
grid on
xlabel 'Number of samples'
ylabel 'Amplitude';
legend('Input Signal');


h=freqz(y_reg);
figure
plot(h);
```


Input Signal
Amplitude
Number of samples
1
0.8
0.6
0.4
0.2
0
-0.2
-0.4
-0.6
-0.8
-1
100
150
200
250
300
350


Input Signal
Amplitude
Number of samples
First Order Chebyshev Type I filter
Regular Filtered Output
Amplitude
Number of samples
Zero Phased Filtered Output
Input Signal
Processed Signal
FiltFilt Output
Amplitude
Number of samples


Zero Phased Filtered Output
Input Signal
Processed Signal
FiltFilt Output
Amplitude
Number of samples
0.8
0.6
0.4
0.2
0
-0.2
-0.4
-0.6
-0.8
-1
50
100
150
200
250
300