

ASIA PACIFIC UNIVERSITY TECHNOLOGY & INNOVATION

EE004-4-3 DIGITAL SIGNAL PROCESSING OPEN-ENDED LAB WORK FREQUENCY DOMAIN ANALYSIS & FILTER DESIGNS

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INTRODUCTION:

Basically, digital filters are used in processing different signal data with a view to making changes in the signals. The changes referred to here can be in the areas of telecommunications, audio processing, and biomedical engineering, among other fields. Digital filters can be mainly classified into two categories: Infinite Impulse Response (IIR) and Finite Impulse Response (FIR). Their primary function is to allow through only those parts of a signal that are intended to be allowed through by the filter and to reject everything else.

IIR filters, like the Butterworth filter, are designed with feedback mechanisms where the past outputs are mixed with current inputs to form the filter's output. Such a configuration results in a filter that infinitely responds to an input signal, and hence its name of "infinite impulse response." The Butterworth filter is particularly acknowledged for having a flat and smooth frequency response in the passband, hence suitable for applications that require very minimal distortion of signals within the passband.

FIR filters, in contrast, are designed to always process a fixed number of input points to compute each of their output points, and their response to an impulse is of finite duration. The FIR filter design by an FIR filter using a Hamming window is performed in such a manner that it shapes the spectral leakage in the frequency response of the FIR filter. This effect is very important if the frequency characteristic of the filter is required to be accurately controlled.

Both types of filters can be designed for many purposes, including high-pass, low-pass, band-pass, and band-stop filtering. High-pass filters are good at removing lower frequencies and preserving higher frequencies, such as in rumble removal in audio processing. On the other hand, a band-stop filter is designed to remove certain specific frequency bands and finds a lot of use in filtering out noise or interference in many applications.

IIR FILTER BUTTERWORTH FILTER

Characteristics: The Butterworth high-pass filter would be designed to pass the higher frequencies and to attenuate those lower than a cut-off frequency. This is greatly appreciated for its smooth, flat frequency response over the passband that is free of ripples. This makes it particularly useful when it is necessary not to distort the high-frequency components to which the signal may be critical; for instance, in audio processing, it is required not to let its high-frequency components be distorted in order to remove hum or rumble from low frequencies.

Applications: It is applied in most audio and electronic equipment for filtering lower frequencies that are perceived as noise or carrying information not relevant to the application at hand, for instance, biomedical signal processing involving DC offsets or movement artefacts.

The **Butterworth design a Band-Stop filter** follows a filter configuration that eliminates a band of frequencies between two specific cut-off points, allowing frequencies outside this range to pass with minimal attenuation. Similar to the high-pass filter, it provides flat responses in pass bands, which thereby allows the least distortion of signal components that lie out of the stop band

Applications: This filter is particularly useful for applications where a pre-determined range of frequencies is known to be of troublesome nature, such as the elimination of power line interference at 50/60 Hz in recorded audio tracks or biomedical signals.

FIR HAMMING FILTER

Characteristics: A high-pass FIR filter is designed to attenuate frequencies lower than some desired cut-off and transmit higher frequencies. The Hamming window helps in controlling the characteristics of the transition band, obtained by reducing side lobes of the filter frequency response and hence minimizing the effect of leakage in the stop band. Some applications of FIR high-pass filters with a Hamming window include certain communication systems used by digital signal processing; audio processing tasks with the exact need for low-frequency noise removal, but its phase must be saved.

Hamming window band-stop filter Features: A Hamming windowed band-stop FIR filter is developed to reject a specified range of frequencies and pass the other frequencies with at least

ripple. The use of the Hamming window in controlling the sidelobes of the filter makes better attenuation in the stop band, and artefacts are reduced in the frequency response.

Applications: suited for applications where certain frequencies or bands need to be eliminated, for example, vocal frequencies from background music, or in telemetry systems to filter out a certain frequency interference.

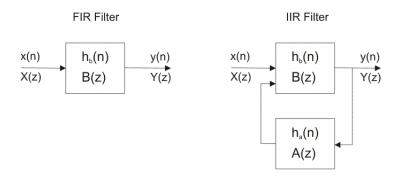


Figure 1: FIR and IIR Filter

Hence, the fundamental difference between FIR and IIR filters, from their block diagrams, is that the IIR filter has feedback, given by A(z), where the past output values influence the current output value and all the following ones. With this feedback, an IIR filter can be designed to occupy less space, holding a given filter response, than in an FIR filter, for fewer coefficients.

OBJECTIVES

- 1. Build appropriate Infinite-Impulse Response (IIR) and Finite-Impulse Response (FIR) filter.
- 2. Perform frequency-domain analysis of discrete-time signals.

FILTER TECHNIQUES IMPLEMENTED.

In this section, the implemented techniques are going to be show in MATLAB code:

INPUT SIGNAL PARAMETERS

Figure 2: Input Parameters High Pass

Sampling Rate (samplingRate): This parameter is set for 2000 Hz, which means that the system will capture 2000 samples in one second.

High-pass Cutoff Frequency (highPassCutoff): The parameter is at 350 Hz and represents the high pass cut off frequency at which the filter passes signals efficiently. All the other signals below such a frequency are considerably attenuated.

Order of the Filter (orderOfFilter): In simple words, the order of the filter can be defined as the steepness of the filter response around the cutoff frequency. The order of the filter is set to 4 in the system.

Normalized High-pass Frequency (normalizedHighPassFreq): The cutoff frequency is normalized with respect to the Nyquist frequency, and MATLAB needs this in terms of a value between 0 and 1 for the Butterworth filter using the design function, butter.

Filter Coefficients (highPassCoeffs, highPassPoles): These are implemented using MATLAB's butter function, which is adapted to create a high pass filter.

The mentioned parameters are reflected in the next figure:

GENERATING INPUT SIGNAL

Figure 3: Generating the Signals

Here, this part of the code is designed to produce test signals with different frequency components to verify high-pass filter effectiveness: The parameter are the next ones:

timeVector: Generates a sequence of time points from 0 to 29, used to construct discrete-time signals.

Frequency Components: Three different frequencies are defined—100 Hz, 300 Hz, and 500 Hz. These represent low, mid, and high frequency components respectively.

Signal Generation:

signalLow: Sinusoidal signal with a frequency of 100 Hz.

signalMid: 300 Hz sinusoidal signal.

signalHigh: A 500Hz sine signal.

These signals are generated from the sine function and modulated by the respective frequencies and sampling rate for them to fit appropriately into the defined digital timeline.

Combine the Signals: Low-, mid-, and high-frequency signals are added to become one composite signal, combinedSignal. This combined signal will be useful during testing in quantifying how effectively the high-pass filter is in separating out and attenuating various components based on their frequencies.

BUTTERWORTH HIGH PASS CODING

```
% Filter Configuration Parameters
                         samplingRate = 2000;
highPassCutoff = 350;
orderOfFilter = 4;
                                                                                         % Sampling rate in Hz
% High-pass cutoff frequency in Hz
% Order of the filter
                        Normalize the high-pass cutoff frequency
normalizedHighPassFreq = highPassCutoff / (samplingRate / 2);
% Compute filter coefficients using Butterworth design
                         [highPassCoeffs, highPassPoles] = butter(orderOfFilter, normalizedHighPassFreq, 'high');
% Generating Test Signals with Distinct Frequencies
timeVector = 0:29;
% Sample index vector
10
11
12
13
14
                         freqLow = 100;
freqMid = 300;
                                                                                                                                          % Low frequency component
% Middle frequency component
                        Trequiu = 300;
freqdigh = 500;
% High frequency consignalLow = sin(2 * pi * freqNid / samplingRate * timeVector);
signalHid = sin(2 * pi * freqNid / samplingRate * timeVector);
signalHigh = sin(2 * pi * freqNid / samplingRate * timeVector);
% Combine individual frequency components into a single signal combinedSignal * signalLow + signalHid + signalHigh;
% Silten posilection to the combined signal
15
16
17
                                                                                                                                           % High frequency component
18
19
20
21
22
23
24
                         % Filter application to the combined signal
filteredOutput = filter(highPassCoeffs, highPassPoles, combinedSignal);
                         % Frequency response analysis
                        % Frequency response analysis
[freqResponse, freqVector] = freqz(highPassCoeffs, highPassPoles, 512, samplingRate);
amplitudeResponse = 20 * log10(abs(freqResponse)); % Conversion to decibels
phaseResponse = angle(freqResponse); % Phase response in radians
```

Figure 4: Code for Butterworth High Pass Filter

High Pass Butterworth filter design and verification will be performed in the figure above. Firstly, it will determine filter parameters. Then, the source will generate test signals with different frequencies and use the filter to separate high frequencies leaving the output of the filter signals only. At the end, it will also undertake a frequency response analysis of this filter, selecting the frequencies being attenuated and those that will be kept. This is the main reason why high-frequency signals are used in the applications where there is a need for a very clear reception of signals.

BUTTERWORTH BAND STOP CODING

```
% Filter Configuration Parameter
              samplingRate = 2000;
                                                % Sampling rate in Hz
                                                % Lower edge of the stop band in Hz
% Upper edge of the stop band in Hz
              lowerEdgeFreq = 250;
             upperEdgeFreq = 350;
                rderOfFilter = 4;
             % Normalize cutoff frequencies for the band-stop filter
normalizedFreqs = [lowerEdgeFreq / (samplingRate / 2), upperEdgeFreq / (samplingRate / 2)];
10
12
13
             % Compute filter coefficients using Butterworth design
             [bandStopCoeffs, bandStopPoles] = butter(orderOfFilter, normalizedFreqs, 'stop');
14
15
16
17
18
19
20
21
22
             % Generating Test Signals with Distinct Frequencies
             timeVector = 0:29:
                                                                             % Sample index vector
             freqLow = 100;
freqMid = 300;
                                                                             % Low frequency component
                                                                             % Middle frequency component
            frequin = 500; % High frequency component freqHigh = 500; % High frequency component signalLow = sin(2 * pi * freqLow / samplingRate * timeVector); signalMid = sin(2 * pi * freqMid / samplingRate * timeVector); signalHigh = sin(2 * pi * freqHigh / samplingRate * timeVector);
23
24
25
26
27
28
29
30
             % Combine individual frequency components into a single signal
             combinedSignal = signalLow + signalMid + signalHigh;
             % Filter application to the combined signal
             filteredOutput = filter(bandStopCoeffs, bandStopPoles, combinedSignal);
                Frequency response analysis
             [freqResponse, freqVector] = freqz(bandStopCoeffs, bandStopPoles, 512, samplingRate);
amplitudeResponse = 20 * log10(abs(freqResponse)); % Conversion to decibels
31
32
33
             phaseResponse = angle(freqResponse);
                                                                                      % Phase response in radians
```

Figure 5: Code for Butterworth Band Stop Filter

The MATLAB scrip coding in the next figure below is made for the examination of the proposed design of Butterworth band-stop filter. The filter is established by the sequence of

cutoff frequency, sampling frequency, and square root of the ratio between frequencies above and below the cutoff frequency; frequencies in the range of cutoff frequency will be attenuated. The filter's nominal frequency is configured by the cut-off frequency, which sets the filter response to be sharp or flat.

The normalization of the cut-off frequencies relative to the Nyquist frequency is defined according to the requirements of the digital filter impact in the code. Hence, those calculated values are used to compute the coefficients in the filter setup that will enable to suppress the frequencies in the interval between the lower and the upper edges and transmit through all the other frequencies.

To evaluate how each filter performs, the code first makes test signals at low, medium, and high frequencies, then combines them into one and finally filters the signal. Results shown that the signal is quite suppressed at the mid-frequency, where the magnitudes in the stop band are strong, which shows good filtering property. Calculating the frequency response of the filter is now done to establish the level of absorption in the given frequency band.

HAMMING HIGH PASS FILTER CODING

```
% Sampling and Filter Parameters
                                            % Sampling frequency in Hz
            samplingRate = 2000;
highPassCutoff = 400;
% High-pass cutoff frequency in Hz
             filterOrder = 30:
                                                       % Order of the filter
            % Time Vector for Signal Generation
             sampleIndex = 0:29;
                                                        % Sample index vector
             % Frequencies for the Test Sine Waves
             freqLow = 150; % Low frequency component in Hz
freqMid = 350; % Mid frequency component in Hz
            freqHigh = 550;
                                                        % High frequency component in Hz
            % Generating Sine Waves sineWave1 = sin(2 * pi * freqLow / samplingRate * sampleIndex); sineWave2 = sin(2 * pi * freqMid / samplingRate * sampleIndex); sineWave3 = sin(2 * pi * freqMigh / samplingRate * sampleIndex);
            % Combining Sine Waves into a Composite Signal
compositeSignal = sineWave1 + sineWave2 + sineWave3;
            % Creating the High-Pass Filter using Hamming Window
            highPassCoeffs = fir1(filterOrder, normalizedHighPassFreq, 'high', hamming(filterOrder + 1));
            % Applying the Filter to the Composite Signal
            % Frequency Response Analysis
            % Frequency Response Analysis
[freqResponse, freqVector] = freqz(highPassCoeffs, 1, 512, samplingRate);
amplitudeResponse = 20 * log10(abs(freqResponse)); % Convert magnitude to
            phaseResponse = unwrap(angle(freqResponse));
                                                                                   % Phase response in radians
```

Figure 6: Code for Hamming High Pass Filter

The MATLAB code above shows how to design and develop a high-cut filter by using a Hamming window. The filter permits only frequencies higher than the designated cutoff frequency to pass while reducing those frequencies breeds.

Above all, this code defines the main parameters such as the sample rate, high pass cut-off frequency and filter order that gives the filter response. It is followed by the beat frequency system which consist of three-time vectors that are used to build signals frequency varying from low, mid, and high.

Using the sine signal function generator the composited signals are joining up to form 1 signal which is intended to be the escaping function input. First, the Hamming window is employed to make the filter that will then enhance the frequency response and in turn, further reduce the side lobes and eventually set the ripples in the passband to zero.

Having done that, the signal is then passed through the filter using a band-pass hamming window, and the latter is run against a frequency response graph to ensure whether the newly implemented filter had been successful in eliminating the low frequency responses, but still allowing the high ones.

HAMMING BAND STOP CODING

```
% Sampling Configuration
"'--Enan = 2000; % Sampling frequency in Hz
samplingFreq = 2000;
              % Cutoff Frequencies
              bandStopFreqs = [200 400]; % Lower and upper cutoff frequencies in Hz
             % Filter Specifications
                                                   % Order of the filter
             % Time Vector for Signal Generation timeSamples = 0:29; % Sample
                                                    % Sample index vector
              % Frequencies for the Sine Waves
              freq1 = 150;
                                                    % Frequency of the first sine wave in Hz
                                                    % Frequency of the second sine wave in Hz
% Frequency of the third sine wave in Hz
              % Generating Sine Waves sineWave1 = \sin(2 * pi * freq1 / samplingFreq * timeSamples); sineWave2 = \sin(2 * pi * freq2 / samplingFreq * timeSamples); sineWave3 = \sin(2 * pi * freq3 / samplingFreq * timeSamples);
              % Combining Sine Waves into a Composite Signal compositeSignal = sineWave1 + sineWave2 + sineWave3;
              % Normalize the Cutoff Frequencies
              % Creating the Band-Stop Filter using Hamming Window
              bandStopCoeffs = fir1(filterOrder, normalizedBandStopFreqs, 'stop', hamming(filterOrder + 1));
              % Applying the Filter to the Composite Signal
              filteredSignal = filter(bandStopCoeffs, 1, compositeSignal);
               % Frequency Response Analysis
              [frequencyResponse Malaysia: [frequencyWector] = freqz(bandStopCoeffs, 1, 512, samplingFreq); amplitudeResponse = 20 * log10(abs(frequencyResponse)); % Conversion to decibels phaseResponse = unwrap(angle(frequencyResponse)); % Phase response in radians
```

Figure 7: Code for Hamming Window Band Stop Filter

These MATLAB code samples build a band-stop filter with the help of a Hamming window that will reduce frequencies in one band. Those frequencies in the other bands will let other frequencies to go through.

The procedure starts with the setting of the sampling rate and then the choice of the cut-off frequencies of the band-stop filter which define the range that will be suppressed. The one major action of the filter to consider is its order; this defines how sharp the filter is or how well it works between these critical cut-off points.

It then creates a time vector and uses three sinusoid signals with low-, mid-, and high-frequency components of the values. These are combined into a composite signal; therefore, the frequency spectrum will be rich and will be used to test the filter performance.

First, filters are designed with the fir1 function where the sample type is used to make it a stopped filter, and by use of the Hamming window when the side-lobe effects are reduced which improves the filter response. The formulated filter is then used as a point filter to oversample the mixture signal. It is used to know the extent to which the selected midfrequency band is attenuated against the other frequencies.

Lastly, the filtered signal is sent to a frequency response test to see how it attenuates the specified band and to make sure about its phase response. In the end, the filter phase response and frequency response are analysed to ensure that the specified band will be fully suppressed and to verify the phase response.

PLOTTING THE GRAPH

```
% Composite Input Signal Plot
subplot(5,1,1);
35 stem(timeVector, combinedSignal, 'MarkerSize', 4, 'LineWidth', 1.5);
36 title('Composite Input Signal, 'FontSize', 12);
xlabel('Sample Index (a)', 'FontSize', 10);
37 ylabel('Signal Amplitude', 'FontSize', 10);
38 grid on;
39 grid on;
40
41 % Output After High-Pass Filtering Plot
subplot(5,1,2);
43 stem(timeVector, filteredOutput, 'MarkerSize', 4, 'LineWidth', 1.5);
44 title('Filtering Graph', 'FontSize', 12);
45 xlabel('Sample Index (a)', 'FontSize', 10);
47 ylabel('Filtered Amplitude', 'FontSize', 10);
48 grid on;
48
49 % Magnitude Spectrum of the Filter Plot
subplot(5,1,3);
51 plot(freqWector, abs(freqResponse), 'LineWidth', 1.5);
52 title('Magnitude Spectrum of the Filter', 'FontSize', 12);
xlabel('Frequency (Hz'), 'FontSize', 10);
53 ylabel('Amplitude', 'FontSize', 10);
54 grid on;
55 % Phase Spectrum of the Filter Plot
subplot(5,1,4);
59 plot(freqWector, phaseResponse, 'LineWidth', 1.5);
61 title('Phase Spectrum of the Filter', 'FontSize', 12);
xlabel('Frequency (Hz'), 'FontSize', 10);
grid on;
62 ylabel('Amplitude', 'FontSize', 10);
63 grid on;
64 % Normalized Frequency Response Plot
subplot(5,1,5);
67 plot(freqWector / (samplingRate / 2), amplitudeResponse, 'LineWidth', 1.5);
68 title('Normalized Frequency (Rr and/sample)', 'FontSize', 10);
70 ylabel('Gain (dB)', 'FontSize', 10);
71 grid on;
71 grid on;
72 ylabel('Gain (dB)', 'FontSize', 10);
73 grid on;
74 prid on;
75 ylabel('Gain (dB)', 'FontSize', 10);
76 grid on;
77 grid on;
78 plote (Gain (dB)', 'FontSize', 10);
78 grid on;
78 prid on;
79 prid on;
70 prid on;
70 ylabel('Gain (dB)', 'FontSize', 10);
71 grid on;
71 prid on;
71 prid on;
72 prid on;
73 prid on;
74 prid on;
75 prid on;
75 prid on;
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79 prid on;
7
```

Figure 8: Plotting graphs

The MATLAB script above in figure 8 will plot the following in one large figure window: a signal in time and frequency domain with various effects of a high pass filter:

Figure Setup: A large figure window is created, which will contain all subsequent plots that will enable the results to be viewed in more detail and compared.

Composite Input Signal Plot: This is the plot of the initial composite signal into the system. This is a stem plot to present variation of the signal among different sample indices.

The Output After High Pass Filtering: Another stem plot to show the signal after it was high pass filtered. The intention is to point out that the low frequencies are significantly damped, while the high frequencies are passed.

Magnitude Spectrum Plot of the Filter: A magnitude spectrum plot of the filter is a line plot that will explain to us how much the filter would amplify or attenuate at different frequencies.

Phase Spectrum of the Filter Plot: Data on this plot is the information about the phase response of the filter at the different frequency.

Normalized Frequency Response Plot: The gain response of the filter versus normalized frequencies; filter performance is shown in dB over the scaled range of frequencies.

All the plots are labelled, and grid lines are put in place to guarantee the development of an accurate understanding of the effect that the filter has on the signal in the various domains.

RESULTS

BUTTERWORTH HIGH PASS

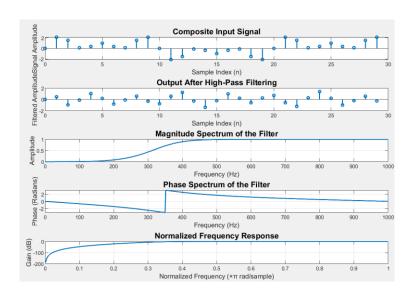


Figure 9: Butterworth High Pass highPassCutoff = 350Hz

In the figure 9, the graph or plot reflects the performance and properties graphs of a high-pass Butterworth filter designed and implemented on a composite signal made up of three independent frequency components.

Composite Input Signal: The graph in figure 9 represents the composite input signal obtained by the sum of three different frequencies: 100 Hz, 300 Hz, and 500 Hz. From this plot, the amplitude of the signal in samples can be seen, representing the periodicity of the signal and signal superposition with several frequencies.

High-Pass Filtering Result: This plot (figure 9) is representative of the signal after high-pass filtering at 350 Hz. In comparison with the plot of the input signal, it is obvious that the amplitude of the 100 Hz and 300 Hz components is way lower, while the 500 Hz component is not that well attenuated. This means that the filter is relatively good at attenuation, especially at frequencies lower than the cut-off frequency.

Magnitude Spectrum of the Filter: the plot in figure 9 represents the magnitude response of the filter up to a 1000 Hz frequency range. It only really begins to rise dramatically beyond a cutoff frequency of 350 Hz, which represents where the filter will begin to pass the frequency.

Filter Phase Spectrum: The phase response of the filter over the same frequency range is following a relatively flat path up to near the cut-off frequency. In this cut-off region, a

remarkable phase shift is showing, typical for a high-pass filter, that substantially changes the phase relationship of the output signal with respect to its input.

Normalized Frequency Response: This is a plot of the gain of the filter in dBs versus the normalized frequency with respect to the Nyquist frequency. We see that this design provides large attenuation, i.e. a sharp decline in gain, for frequencies below about 0.35, or a 350 Hz cutoff frequency with respect to a 1000 Hz Nyquist frequency. The filter has high attenuation at low frequencies.

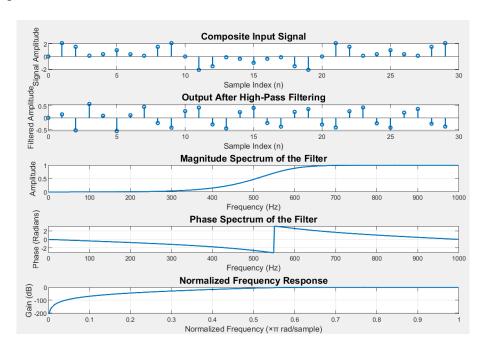


Figure 10:Butterworth High Pass highPassCutoff = 550Hz

The above plots are representations of the effect of an improved Butterworth high-pass filter used to operate on the composite signal. The high-pass cutoff frequency of the filter used is 550 Hz.

Original composite input signal plot: The plot in the figure shows the original signal which consists of 3 different frequencies of sinusoidal components namely 100 Hz, 300 Hz and 500 Hz stem plot. Each point represents a sample index from 0 to 29.

High-pass Filtering Plot output: Its output is shown in this plot, which is the processed signal after applying the high-pass filter. As shown in the plot, all the frequencies below half the cutoff of 550 Hz, which are 100 and 300 Hz, are highly attenuated, and some attenuation for the 500 Hz component is achieved. This plot emphasizes how the filter attenuates all the signals that fall below the cutoff.

Magnitude Spectrum of the Filter Plot: It shows the magnitude spectrum of the filter which depicts how the filter behaves as a function of frequency up to 1000 Hz. The plot reveals the magnitude spectrum increase at the cutoff frequency where the signal can pass more freely.

Phase Spectrum of the Filter Plot: The graph shows how the filter shifts phase as frequency changes. The phase stays the same until close to the cutoff frequency, then there is a big shift, like with high-pass filters.

Normalized Frequency Response Plot: This graph displays how much the filter boosts or cuts the volume in decibels at different frequencies (relative to half the sampling rate, known as the Nyquist frequency). It shows a steep decrease in volume for frequencies lower than about 0.55 normalized frequency, which is the cutoff frequency. It also demonstrates how the filter lessens the volume of lower frequencies while letting higher ones through with less lessening.

Composite Input Signal Composite Input Signal Composite Input Signal Sample Index (n) Output After Band-Stop Filtering Sample Index (n) Magnitude Spectrum of the Filter Phase Spectrum of the Filter Phase Spectrum of the Filter Phase Spectrum of the Filter Frequency (Hz) Normalized Frequency (Response

BUTTERWORTH BAND STOP

Figure 11:Butterworth Band Stop Filter lowerFreq=250Hz, higherFreq=350Hz

These graphs in figure 11 show how a band-stop Butterworth filter behaves in the case of a composite signal. Below is the explanation of each graph:

Composite input signal: spikes on different time instances represent the signal which is composed of three frequencies: 100 Hz, 300 Hz, and 500 Hz. Every point where spikes are accumulated or close to each other represents some combination of the above-mentioned frequencies. You may observe that everywhere there is a signal the major component is the frequency of 100 Hz and 500 Hz.

Output signal after band-stop filtering: you see that the 300 Hz component is reduced while the signal of 100 Hz and 500 Hz is not changed. It is known that a band-stop filter is allowing certain frequencies to pass from another, and the stop band is around 300 Hz, for this reason, the amplitude of signal which is composed of 300 Hz frequency only should be reduced.

Magnitude Spectrum of the Filter: How does the spectrum plot show the filter? The magnitude of the filter frequency response is shown, and the large valley around 300 Hz is a dip. This means that the slope of the filter frequency, representative of the stopband, gets to zero at 300 Hz. The dip is exactly at 300 Hz, and the magnitude is just about zero, which means that the stopband is very much damped. However, the stopband magnitude is almost a flat line from zero frequency to the frequency of the signal.

Phase Spectrum of the Filter: What does it show about the filter? In the frequency spectrum, a filter frequency response phase shows a change in the phase of the stop band between 250 and 350 Hz. The phase magnitude from the stop band is lightly low, and hence, there is a little phase distortion.

Normalized Frequency Response: It represents the normalization of the view of gain in dBs across the frequency spectrum. This is depicted on the normalized frequency axis by the sharp valleys located at 250 Hz and 350 Hz; they show the ability of the filter to attenuate the frequencies in that range.

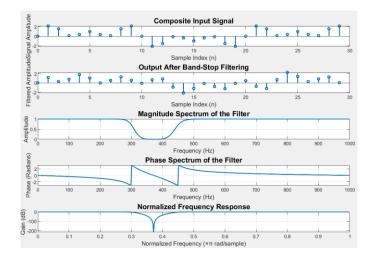


Figure 12: Butterworth Band Stop Filter lowerFreq=300 Hz, higherFreq=450 Hz

In the figure 12, there are significant characteristic of the graph that are going to be discussed as follow:

Composite Input Signal: The plot that is given in the time domain for the combination signal of these sinusoidal components is 100, 300, and 500 Hz. The variation in amplitude is due to the superposition of these frequencies.

Band-Stop Filter Input Output: The realization of the band-stop filter causes its output signal to take a form in which the 300 Hz component is strongly attenuated because it lies at the lower edge of the frequency stop band designed into this filter. There is no significant difference seen for the 500 Hz signal since 450 Hz is very close to 500 Hz and may exhibit low attenuation for this reason. The 100 Hz component is not affected much by this.

Magnitude Spectrum of the Filter: The magnitude response displays a well-defined low-pass filter: deep attenuation in the filtering band, ranging from 300 Hz to 450 Hz. That is, the filtering of the filter. You can also notice more gradual transitions near 450 Hz and up to 500 Hz, which have very small influence on the 500 Hz part.

Filtered Phase Spectrum: The phase response changes noticeably at the stop band edges, which is at 300 Hz and 450 Hz. On the other hand, this graph shows how the filter affects the phase of frequencies, where it gives a greater emphasis on those inside and near the stop band. Frequency Response: The graph of dB gain confirms that the filter works acceptably for the very sharp fall in attenuation at the normalized frequencies corresponding to 300 Hz to 450 Hz. The sharp drops validate the suppression of these frequencies.

HAMMING HIGH PASS

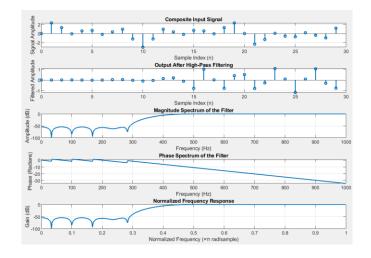


Figure 13: Hamming High Pass Filter highPassCutoff = 400 Hz

Here is a brief description of each graph using the updated values for your MATLAB script that has now implemented a high-pass FIR filter with a Hamming window.

Composite Input Signal: displays a signal consisting of three sine waves at frequencies of 150 Hz, 350 Hz, and 550 Hz. Its presence is revealed by the peaks of these combined amplitudes.

Output After High-Pass Filtering: The output reveals the signal for the high-pass filtering. It is quite noticeable how, at low amplitude, the 150 Hz component is being cut down by reduction because this frequency falls within the cutoff set at 400 Hz. The component of 350 Hz is also very near the cutoff, and thus only a bit is being attenuated, while the component of 550 Hz is retained mostly, providing a fine filtering effect.

Magnitude Spectrum of Filter: This is the filter frequency response. There is huge attenuation below the 400 Hz cutoff, but above the cutoff, the gain gently raises for higher frequencies, showing good high-pass filter behaviour.

Phase Spectrum of the Filter: The phase shift introduced by the filter for various frequencies is shown. It is a smooth, steady shift at the various frequencies, which is one of the characteristics of FIR filters implemented under a Hamming window to represent a linear phase behaviour in the passband.

Normalized Frequency Response: Figure 13 presents the gain plot in dB, which shows a clear high-pass characteristic as a function of normalized frequency. This is a response that dips well below 0.4 of normalized frequency, ensuring the filter is working in the success of attenuating lower frequencies while still passing the higher ones.

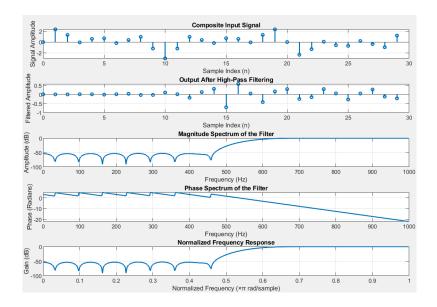


Figure 14: Hamming High Pass Filter highPassCutoff = 580 Hz

Effects of the high-pass filter on the composite signal and its frequency response for a high-pass cutoff frequency of 580 Hz would be:

Composite input signal: This is the graph of the original signal that contained the frequencies 150 Hz, 350 Hz, and 550 Hz. There is no change because it is the graph at the unfiltered state.

Output After High-Pass Filtering: It can be seen from here that now all components of all frequencies have started getting attenuated, as the cutoff frequency has now become above the 550 Hz component. The 550 Hz component, which was supposed to be more prominent in the previous case, is now showing a lot of attenuation but has not been completely cut off; thus, this proved the fact that cutoff slope does not exactly stop its operation at 580 Hz, but it starts to attenuate the frequencies a bit below this threshold level.

Magnitude Spectrum of the Filter: This is significant attenuation indicated by the magnitude response just below 580 Hz, meaning frequencies close to the cutoff are affected and not those above the cutoff. At the 550 Hz frequency, this is attenuation to quite a level of amplitude, while at low frequencies, this is total.

Phase Spectrum of Filter: The phase response shows a continuous shift of phase while it approaches and passes beyond the cutoff frequency. This is a general characteristic of the phase transition around the cutoff frequency for high-pass filters.

Normalized Frequency Response: This normalized frequency response serves only to highlight the filter's performance on a normalized scale, with 1 as the Nyquist frequency, and

in this case, 1,000 Hz. It is suggesting deep attenuation for frequencies below about 0.58 normalized to cutoff, so it assures the reader visually that there is no use below such a value.

Composite Input Signal Compos

HAMMING BAND STOP

Figure 15: Hamming Band Stop Filter lowerFreq=300 Hz, higherFreq=450 Hz

Composite Input Signal: It is the composite of the three sine waves of 150Hz, 350Hz, and 550Hz. The signal comprises the superimposition of the three frequencies. It shows the periodic peaks in the process of constructive combination of phases.

Output Following Band-Stop Filtering: It is clear from the output that the 350 Hz component, which falls within the stipulated stop band of 200-400 Hz, is evidently attenuated. The 150 Hz and 550 Hz components, which are outside the stop band, are practically left unchanged. It is this evidence that testifies to the effectiveness of the filter in middle-frequency attenuation while leaving the passband unchanged.

Magnitude Spectrum of the Filter: The magnitude spectrum shows that a great depression in the frequency response occurs within 200 Hz and 400 Hz; that is, this region is the stop band of the filter. In that respect, quite a considerable amount of attenuation occurs in that region, which means that in the real sense, the filter can quash frequencies in that range.

Phase Spectrum of the Filter: The phase spectrum is relatively flat, which gives light on low phase distortion outside the stopband. This linear trend in the phase response around the

stopband frequencies is quite indicative of the fact that phase shifts by the filter are mostly applied in the region where the suppression of frequencies takes place.

Normalized Frequency Response: The normalized frequency response indicates that the deep valleys are seen in the stop band at the normalized frequencies of 0.2 to 0.4; that is, the frequencies of 200 Hz and 400 Hz. The response outside these frequencies seems to be smooth and flat, indicating good passband performance.

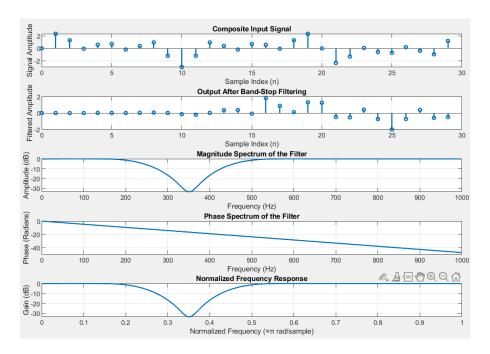


Figure 16: Hamming Band Stop Filter lowerFreq=250 Hz, higherFreq=450 Hz

Taking the new cut-off frequencies at 250 Hz and 450 Hz, the resulting graphs have the following behaviours:

Composite Input Signal: The input signal is a superposition of three sine waves at frequencies 150 Hz, 350 Hz, and 550 Hz. The graph looks periodic, as we expect for a sum of three sinusoids.

Output After Band Stop Filtering: Here the output is self-explanatory regarding the performance of the filter. A 350 Hz component, lying in a new stop band of 250-450 Hz, shows a massive attenuation. Also, a 150 Hz frequency component is highly not influenced since it is smaller than the stop band; and a 550 Hz component is also not affected much since it is higher than the stop band.

Magnitude Spectrum of the Filter: The new stop-band region has the limits of 250 Hz and 450 Hz; therefore, deep attenuation is observed from 250 to 450 Hz in the magnitude response

graph. It is evident that this filter is highly effective in the frequency region between 250 Hz and 450 Hz.

Phase Spectrum of Filter: The phase spectrum describes a slow phase shift, appearing mainly in the stop band. Such shift looks like a typical feature of FIR filters, in which the phase response is commonly linear; hence, phase distortion is small at the frequencies lying outside of the stop band.

Normalized Frequency Response: The response is further plotted in normalized form, where the filter is seen to exhibit a very sharp response in the range between the normalized frequencies of 250 Hz and 450 Hz. The graph shows that the filter is highly responsive in the rejection range specified, whereas other frequencies are very easily passed with little or no type of attenuation.

DISCUSSION

Analysis of the Butterworth and Hamming Window Filter from the values obtained in results section:

Butterworth high Pass Filter

350 Hz High-Pass Filter

Composite Input Signal: The signals at 100 Hz, 300 Hz, and 500 Hz in the input signal require the attenuation of low frequencies and the passing of high frequencies.

Analysing the Output Signal: The output obtained following the filtering demonstrates that the 300 Hz and 100 Hz components are effectively suppressed, indicating that the filter is below the cutoff frequency. As would be expected, the 500 Hz component stays nearly unchanged because it is well above the cutoff.

Magnitude Spectrum: The magnitude spectrum shows the filter's frequency response, with a gain increase immediately before the very slow cut-off that signals the start of frequency passing.

Phase Spectrum: Until it approaches the cutoff frequency, when a discernible shift occurs, the phase response is largely constant. This kind of characteristic shift is highly significant because

it may cause phase distortion close to the cutoff, which must be considered in applications where phase is crucial.

550 Hz High-Pass Filter

Characteristics of the Output Signal: For instance, by raising the cutoff frequency to 550 Hz, the filter will need to reduce the 500 Hz component because it was unchanged at a cutoff of 350 Hz. This indicates that the filter has much more control over the frequency range that it allows to pass.

Magnitude and Phase Response: Above 550 Hz, the filter only begins to become more liberal in the passband. The magnitude response moves incredibly, showing the filter's level of fall. When designing filters for applications that call for extremely fine-grained frequency separation, this is crucial.

Normalized Frequency Response: The normalized frequency below 550 Hz at which the gain drops more sharply indicates excellent design, ensuring that there is very little leakage of undesirable frequencies—a crucial component of any digital signal processing equipment. Analysis of Butterworth Band-Stop Filter

Butterworth Band Stop Filter Analysis

250-350 Hz Band-Stop Filter

Filter Effectiveness: It is discovered that the filter's ability to reduce the 300 Hz frequency component falls inside the designated stopband. The fact that the signals at 500 Hz and 100 Hz are still present suggests that the filter precisely attenuates the designated frequency band.

Magnitude and Phase Spectra: A filter's stop band, which largely mutes this frequency while only slightly affecting other frequencies, is analogous to the magnitude spectrum's deep dip at 300 Hz. The phase response demonstrates very little disturbance outside of the stop band, indicating that the filter was well-tuned to preserve all the input signal's phase characteristics outside of the stop band.

300-450 Hz Band Stop Filter

Broadened Stopband: Reducing the stopband from 300 to 450 Hz only represents a wider window for frequency suppression, which has a significant impact on the signal's 300 and 400 Hz portions. A broader dip that indicates a larger stopband in the magnitude spectrum and the output signal both demonstrate this greater suppression.

Phase Behaviour: Due to the phase shifts the filter introduces, the phase spectrum changes more noticeably over this wide stopband. Phase coherence-related applications will suffer from the larger stopband's potential to produce more pronounced phase distortions.

High Pass Hamming Window Filter

400 Hz High Pass Filter

Composite Input and Output Analysis: It is also effective, as one can show by a Hamming-window high-pass, 400-Hz filter. The output makes it evident that components at the frequencies less than the cutoff are greatly suppressed or practically eliminated. The component just above that of, say, 400 Hz, remains, but with a smaller amplitude.

Magnitude Spectrum: One can see quite clearly from the above spectrum how the filter cutoff works. It begins to pass frequencies just a little below 400 Hz—the smooth roll-off nature of the Hamming window. This gentle slope benefits applications requiring a more supple frequency transition, with no problems of ringing artifacts due to sharp cut-offs.

Phase Response: The phase plot shows very little phase shift until very close to the cut-off frequency, at which point an abrupt change occurs. This continuous-phase change is a characteristic of FIR filters designed with windows, such as Hamming, and offers a trade-off between minimum phase distortion and linear phase characteristics.

580 Hz High-Pass Filter:

Output Signal Characteristics: At 580Hz, the filter does a much better job of reigning in and knocking out components clear into the 500Hz range. This shift to the higher cutoff allows a cleaner distinction in filtering the frequencies that are closer to the midpoint between typical voice and higher-frequency signals.

Magnitude and Phase Response: After the cutoff, the magnitude plot goes steeper, properly attenuating frequencies below 580 Hz, and then quickly recovering the gain for higher frequencies. This is indicative of the fact that the filter can separate the higher frequencies more sharply. The phase spectrum is quite well-behaved, as would be expected for applications in which the phase integrity of the signal is of importance.

Hamming Window Band-Stop Filter Analysis

300-450 Hz Band-Stop

Filter Performance: The shape clearly shows that the filter suppresses the frequency components lying between 300 and 450 Hz because there is a significant attenuation in the output signal for this range. This, for sure, is the driving aspect when removing unwanted noise or interference in the given frequencies, which are typical in most practical applications of signal processing.

Magnitude and Phase Spectra: The magnitude spectrum clearly demonstrates a high level of depth in the stopband with a clear notch being established between 300 and 450 Hz. It has perfectly matched with the desired suppression band. The phase response shows very small deviations within the stopband. Deviations in the phase are mostly normal and tolerated in band-stop filtering since the main objective is to cause amplitude attenuation.

250-450 Hz Band-Stop

Characteristics of a Broad Stopband: With the broadening of the stopband to commence at 250 Hz, the filter yields an increased level of suppression, and it interferes with a more extended scope of the input signal. In case of environments where a broader band of noise or interference needs to be taken away, this is an excellent characteristic.

Magnitude Response: A more pronounced and wider dip from 250 Hz to 450 Hz is visible in the magnitude response of the filter. A wider suppression may be important for achieving a cleaner output for such noise signatures, which are inherently complex.

Phase Spectrum Analysis. The phase fluctuations are strictly limited to the stopband. This is a very strong point of using an FIR filter with a Hamming window because the time-domain behaviour of the signals in the passband is henceforth not significantly affected by the process.

CONCLUSION

This DSP Lab Report presents the design and analysis of IIR and FIR filters using MATLAB in view of Butterworth and Hamming filter designs. Several filter configurations are then tested including high-pass and band-stop filters and their effect on a combination signal containing different frequency components. The functioning of each type of filter is tested and analysed according to graphical presentation of the frequency response and phase responses. The report effectively illustrates the strengths and weaknesses of all the filter designs to reject undesired frequencies within the signals, providing support to the importance of selecting the proper type of filter according to the application demands.

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