

Department of Electrical & Electronics Engineering

**Abdullah Gül University**

**EE3001 DSP Lab 5**

**Submitted on: 05.23.24**

**Submitted by:**

**Oğuzhan Alasulu**

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**Lab Instructors: Oğuzhan Ayyıldız**

**Grade: / 100**

**LAB 5**

**3.1 Nulling Filters for Rejection**

**A)**

**A number and number equation

Description automatically generated with medium confidence**

**B)**

A graph of a function

Description automatically generated with medium confidence

**C)**

A screenshot of a computer screen

Description automatically generated

**D & E)**

A graph with numbers and numbers

Description automatically generated

The input signal consists of three sinusodial components

where:

The first filter attenuates the frequency at and the second filter attenuates the frequency at . The coefficients for the filters are stated above and they are:

After filtering, just is stayed. From here output signal would be only :

For , the output signal can be written as

where:

* **Magnitude:** 5
* **Frequency:** 0.3π
* **Phase:** 0

The displayed how the filters removed the other two sinusoidal components are filtered and how the component remained. Other than that, we can see that output signal for is pure cosine with a frequency of with a magnitude 5, and phase 0.

**F)**

The output signal is not same for the first points due to the transient response or "start-up" attribute of the filters. This start-up behavior exists since the filters necessiate time to be balanced and attain to the steady-state after the input signal is applied.

In the FIR filters domain, the start-up points are the first points in the output signal where the filter response has not yet totally balanced. Points are affected by the initial situations and transient impacts of the filtering process.

For the cascade of two length-3 FIR filters designed in part (a), each single filter has a length of 3 coefficients. However, when cascaded, the length of the combined filter is the sum of the lengths of the filters minus one. Due to that, when convolving two filters, the length of output is the sum of the individual filter lengths minus one.

Hereby, for two length-3 FIR filters cascaded each other, the length of the combined filter is . Thus, there are 5 start-up points in the output signal, which is relevant with the lengths of the filters designed in part (a). These start-up points correspond to the first transient attributes of the filtering process, and they converge to the steady-state response.

**3.2 Simple Bandpass Filter Design**

**A)**

**A screen shot of a computer

Description automatically generated**

**B)**

A screenshot of a computer

Description automatically generated

The impulse response of the bandpass filter is designed using cosine function for given filter length and center frequency and, the impulse response and frequency response is given by:

Substituting h[n] to frequency response:

can be written also with the euler’s formula of

The frequency response become like (1) and at the final form we get (2):

In the next step, we need to examine the passband determination and main lobe width in order to determine the passband width of the main lobe. To simplify the process, let’s just focus the first element due to filter is symmetric:

Let’s say x. The width of main lobe of (3) is inversely proportional to . The main lobe width can be approached as:

That shows us passband width can be changed with the . Also we know, passband width has the interval of where passband width at 0.707 level at peak value (:

Finally, let’s investigate relationship between and passband width:

**C)**

The selectivity of a filter touch on its features to pass particular frequencies while filtering or rejecting others. For the bandpass filter located at , its selectivity can be explained by the attributes of its frequency response.

Inspecting frequency response plot, the filter display a peak in magnitude in the neighborhood of the center frequency of . This peak is the passband region where frequencies located neighborhood points of center frequency are admitted to pass. It means, the filter "passes" frequencies close to .

At frequencies lower () or higher () than the center frequency, the magnitude of the frequency response reduces reasonably. This filtering represents the filter's properties to reduce or reject frequencies other than the passband region.

The algorithm of this selectivity depends on the model of the filter's impulse response. By structuring the filter with a certain impulse response that indicates the desired frequency component at while attenuating others, the filter effectively behaves as a bandpass filter, admitting only the requested frequency to "pass through."

**D)**

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**E)**

A group of blue lines

Description automatically generated

Explanation:

* Input signal at subplot(2,2,1) and subplot(2,2,3) exhibits the original input which is sum of three sinusoids.
* Subplot(2,2,2) displays the filtered signal using the convolution method. Frequencies within the passband are kept while outside of passband frequencies attenuated.
* Subplot(2,2,4) displays the filtered signal through the “fftfilt” method. The output is related with “conv”, demonstrating that FFT-based filtering.

**F)**

**A graph with a line

Description automatically generated**

Explanation:

• The frequency response of a filter, demonstrated as explains how the filter impacts sinusoidal elements of different frequencies.

• The magnitude of the frequency response at a certain frequency shows how filter diminished or amplifies sinusoidal elements at that frequency.

• The magnitude response at desired frequency could be utilized to establish the size of sinusoidal elements in the output signal. Sinusoidal components at frequency peaks in the magnitude response would experience less filtering and will be more attenuated in the output signal for elements at frequencies on the dips or nulls in the magnitude response.

• Hence, by inspecting the frequency response plot, we can conclude which frequencies are amplified or filtered by the filter, and at the end, how the filter impacts the size of each sinusoidal elements in the output signal.