Digital Signal Processing

Assignment 4: audio filtering – notch filter

As we saw in the previous assignment a simple low pass filter is not very efficient at removing a single frequency interference. A better method to filter out a single frequency is by using a notch filter. A common application of notch filters is to remove any interference at 50Hz caused by power lines.

The transfer function is of a notch filter is as follows:

$$H(z) = \frac{1 - 2\cos(\omega_n)\, z^{-1} + z^{-2}}{1 - 2a\cos(\omega_n)\, z^{-1} + a^2 z^{-2}}$$

Where

- ω_n normalised frequency to null out
- a determines width of the frequency dip, value between 0 and 1

The normalised frequency to null out is:

$$\omega_n = 2\pi \frac{f_{interference}}{f_{sample}}$$

The transfer function does not have unit gain for DC signals. To ensure a unit gain for DC signals the transfer function is multiplied by a constant K. Remember that DC has zero frequency and in the z-domain this corresponds to $z = 1$. To calculate this factor K the transfer function is evaluated at $z = 1$.

$$K = \frac{1}{H(z=1)} = \frac{\sum coefficients\ denominator}{\sum coefficients\ numerator}$$

A value of 0.9 is a safe choice for a.

Tasks

This assignment is a continuation of the previous assignment, hence we use the same audio sample and the sampling frequency is also still the same.

1. Plot the frequency response of the filter using MATLAB's *freqz()* function
2. Derive the difference equation
3. Implement the filter using a for loop again
4. Show the single sided spectrum of the filtered audio signal
5. Determine the attenuation, in dB, of the interference
6. Enjoy ☺