Istanbul Technical University
Faculty of Computer and Informatics

BLG438E Digital Signal Processing Lab
Experiment 5

Cem Yusuf Aydoğdu
150120251

In this experiment, a Finite Impulse Response (FIR) filter was implemented on the TMS320 C5515 DSP kit. Equation of the filter is given below. In this equation, output depends on the past two values of the input, so this equation shows that the filter is a second order filter.

$$Y_n = A_0 X_n + A_1 X_{n-1} + A_2 X_{n-2}$$

Transfer function of the filter is:

$$H(z) = Y(z)/X(z) = A_0 + A_1 z^{-1} + A_2 z^{-2}$$

Coefficients of the filter was found from Matlab for 100 Hz frequency as $A_0 = 0.038$, $A_1 = 0.2$ and $A_2 = 0.0038$. Block diagram of the filter is given below:

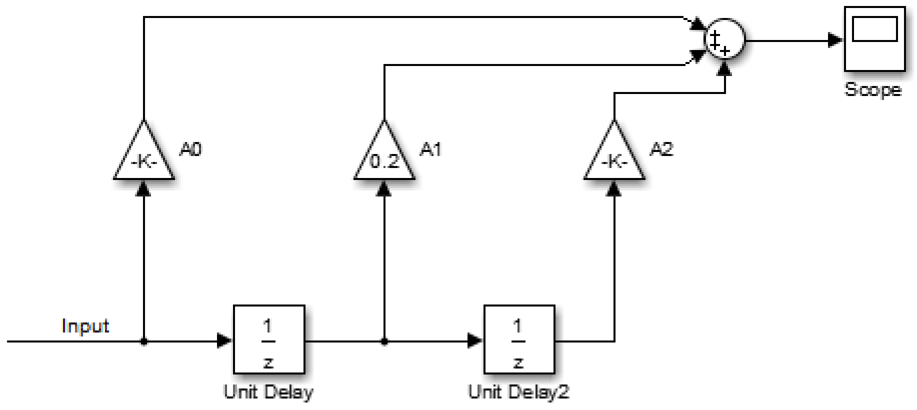

Figure 1 Block diagram of the filter

Following Matlab script is written in order to obtain bode diagram of the filter. First, transfer function is obtained using numerators and denumerators. Then, bode diagram is obtained from the transfer function. Corresponding Matlab code and bode diagram is shown below:

```
num = [0.038, 0.2, 0.0038];    %% A0, A1, A2
denum = [1];
transfer_func = filt(num, denum);
bode(transfer_func);
```

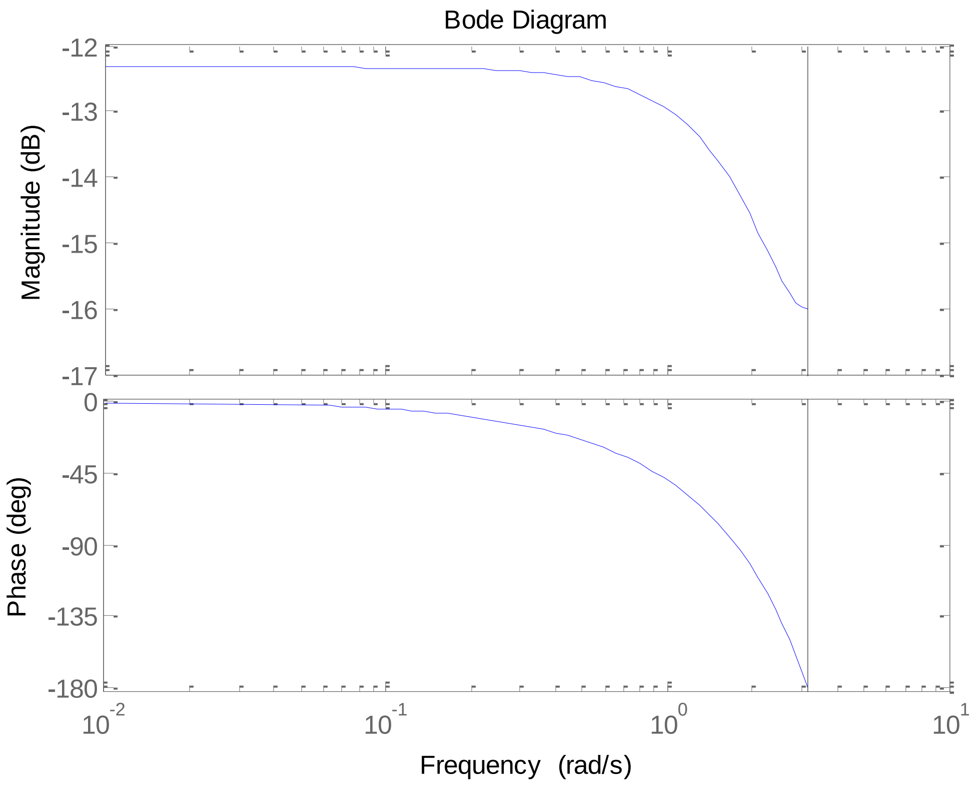

Figure 2 Bode diagram of the filter

Code implementation of the given FIR filter for TMS C5515 DSP kit is given below. This code runs in an interrupt.

```
float A0 = 0.038, A1 = 0.2, A2 = 0.0038;

float x[3] = {0.0, 0.0, 0.0};          // x, xz^-1, xz^-2

while(1)
{
      while((XmitR & I2S0_IR) == 0);//Wait for transmit interrupt to be pending

       x[2] = x[1];
       x[1] = x[0];
       x = I2S0_W0_MSW_R;

       //write to output
       I2S0_W0_MSW_W = (Int16)(A0*x[0] + A1*x[1] + A2*x[2]);
       I2S0_W0_MSW_L = 0;
       I2S0_W1_MSW_W = (Int16)(A0*x[0] + A1*x[1] + A2*x[2]);
       I2S0_W1_MSW_L = 0;

}
```