The University of Queensland
School of Information Technology and Electrical Engineering

ELEC3004/7312: Signals, Systems and Controls

EXPERIMENT 3: ECHO FILTERS ON THE NEXYS 2

Aims

In this laboratory session you will:

1. Use VHDL for implementing a simple "echo" filter via a difference equation.
2. Compare IIR and FIR filters
3. Study comb filters
4. Have some fun with echo and reverb!

Introduction

Two common digital filter types are IIR and FIR, each of which has advantages and disadvantages. Filters are available in low pass, high pass, bandpass, notch etc, and can have different responses such as Chebyshev, Elliptic, etc. Another type of filter is the comb filter which works by combining an input signal with a related signal which forms constructive and destructive interference. The two options we will explore involve combining the input signal with a delayed version of either the input (FIR), or the output (IIR). When constructive interference occurs, the output signal reaches a maximum. Conversely, destructive interference causes the output signal to reach a minimum, sometimes reaching zero. The depth of the notch depends on the relationship between maximum and minimum output values. The spacing between the notches depends on the sampling frequency and the delay.

Preparation

<u>Note: preparation will be checked at the start of each laboratory class.</u>

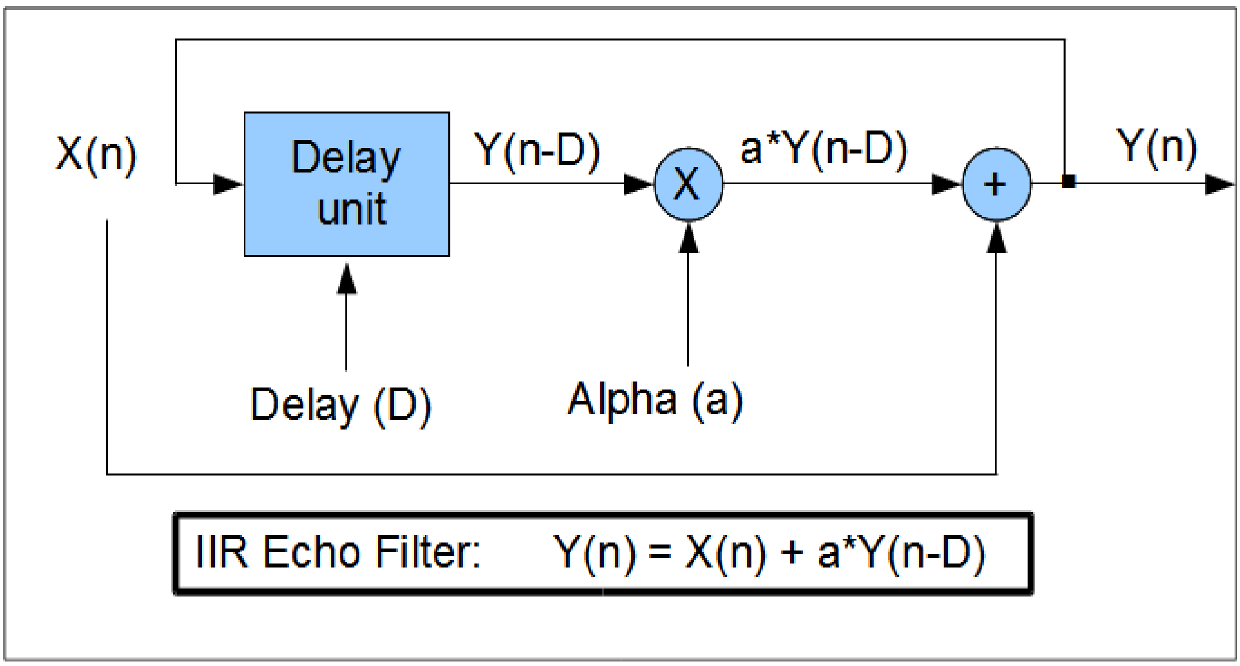

Figure 1: IIR Echo Filter block diagram

Figure 1 shows a block diagram of an IIR Echo Filter consisting of a memory (Delay Unit), a multiplier, adder and feedback path.

Alpha (α) is a value between 0.0 and 1.0, D (delta) is the delay value in samples.

Answer the following questions:

1. A simple echo filter can be defined by the difference equation: $y[n] = x[n] + \alpha x[n - \Delta]$, where $y[n]$ is the output, $x[n]$ is the input of the filter, α is the amplitude of the echo and Δ is the echo delay in samples, n. Clearly, this is an FIR filter (the equation does not have any feedback from the output.)
   a) Implement this difference equation in Matlab and confirm its operation with Handel's hallelujah chorus (see *help sound*). Note, to hear an echo try setting Δ to equal the number of samples in ~100ms, i.e., the sampling frequency divided by 10. You can use a *for* loop to implement this.
   b) Using *zplane* command, sketch the pole and zero locations in the *z*-plane for α = 0.8 and Δ = 6;
   c) Using *freqz,* sketch $|H(\omega)|$, i.e., the magnitude response of this filter from zero to half the sampling frequency. Alternatively, you could also use the *fvtool* command
   (We will be using this command in lab 4 so please familiarise yourself if you can)
   d) What effects do α and Δ have on the shape and number of peaks in the magnitude response?
   e) Suggest how a "fading" echo might be implemented by changing the original FIR filter to be IIR.
   What role do the poles and zeros now play? Confirm your design by implementing this filter in Matlab. *(Hint: IIR filters also have feedback from the output)*

2. Redraw Figure 1 as follows. Delete the feedback connection from Y(n) to the delay unit. Now connect X(n) to the input of the delay unit as well as the adder. Replace the Y(n-D) text with X(n-D). Write down the difference equation. What type of filter is this?

3. Now redraw Figure 1 as follows: Place a 2-position switch at the input of the delay unit so the centre connection goes to the delay unit. Now connect the other two lines, one of which is a connection to Y(n), and the other joins to X(n) as well as X(n)'s connection to the adder. Place the text Z(n) at the input to the delay unit, and replace the Y(n-D) text with Z(n-D). Write down the difference equation. You will use this setup in Part 3.

4. With reference to http://en.wikipedia.org/wiki/Comb_filter write the two equations for Continuous-time comb filters

Equipment

1. PC with Xilinx ISE, Digilent Adept & Matlab;
2. PMOD AD1 and DA2 boards plus 2 PMOD CON4 boards
3. ADC712 and DAC712 Filter boards (fit between ADC/DAC and PMOD CON4)
4. Nexys 2 + JTAG interface cable/s;
5. Oscilloscope (preferably TDS1002);
6. 2 x cable: mono RCA male to mono BNC male, 0.5 - 1 metre long
7. Mono or stereo 3.5mm male to mono or stereo RCA male
8. Mono or stereo Y-adaptor, 3.5mm Male to 2 x 3.5mm Female
9. Signal Generator;
10. External speakers + audio jack cable + power pack;
11. 1 BNC T-adaptor M to 2F (F-M-F);
12. 1 x cable BNC Male to BNC Male, 0.5 - 1 metre long.