Dept. of Computer Science and Engineering

CSE4210 – Architecture and Hardware for DSP

Lab 4

Audio effects using TMS320C6713 DSK and Code Composer Studio

Objective

In this lab, you will study how to implement some audio effects using the TMS320C6713 board.

Introduction

In the previous lab, you used delay to simulate the effect of echo. A more interesting audio effects can be obtained by combining several delays. One example of that is the FIR comb filter. For more information about this topic see [1]. The output of the filter is defined as

$$y_n = x_n + ax_{n-D} + a^2 x_{n-2D} + a^3 x_{n-3D}$$

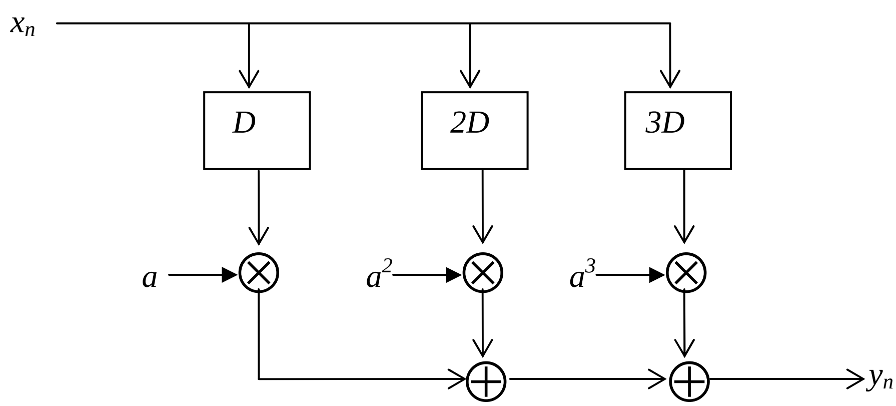
The transfer function of this filter is

$$H(z) = 1 + az^{-D} + a^2 z^{-2D} + a^3 z^{-3D}$$

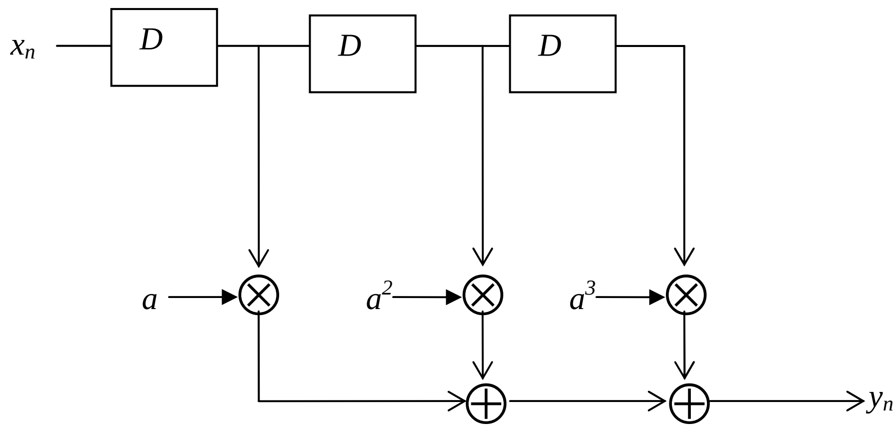
The impulse response of this filter is

$$h = [1, \underbrace{0, 0, \cdots, 0}_{D\text{-}1 \text{ zeros}}, a, \underbrace{0, 0, \cdots, 0}_{D\text{-}1 \text{ zeros}}, a^2, \underbrace{0, 0, \cdots, 0}_{D\text{-}1 \text{ zeros}}, a^3]$$

A direct implementation of this as an FIR is shown below



A more efficient implementation will use one delay line of length 3D and tap out the signal after D and 2D. In this case we used 3D delay elements (buffers in your program) instead of 1+2+3=6D delay elements



Lab Procedure

a) Write a C program to implement the above filter. In your program using a circular buffer similar to the one you used in the previous lab in order to reduce the number of shifts you perform in your program.

b) Use a sampling rate of 8 KHz; choose the delay D to be 4000. This way every D is corresponding to 0.5 sec. Also use a to be 0.5.

c) Build and run the program. Listen to the impulse response of the filter by tapping on the microphone.

d) Speak in the microphone and listen through the loudspeaker.

e) Connect as input your favorite song, listen through the loudspeaker.

f) Change a to 0.2 and to 1, repeat (d) and (e) above

References

[1]. S. J. Orfanidis, Introduction to Signal Processing, online book, 2010. Available from: http://www.ece.rutgers.edu/~orfanid/intro2sp