Goal

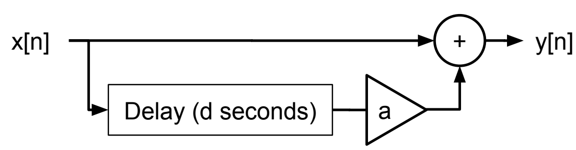
The goal of this assignment is to use digital filters to add sound effects to music. Done properly, this assignment can be a lot of fun: you get to play music producer! For the most part, we'll be doing a time-based analysis (as opposed to frequency-based). Note that any audio signal can be listened to in Matlab using the `soundsc` command. Assuming your audio signal is named `sig` and has a sampling frequency of `fs`, you would type:

```
soundsc(sig,fs);
```

In this project, the sampling frequency will be 44.1kHz in all cases. You don't need to manually enter that, though - it is saved directly into the data file you'll be using. Note that most of the sound effects in this project will be more noticeable with earbuds or headphones.

Background

You can use FIR and IIR filters to add different kinds of echo into an audio signal. A generic FIR echo filter might have the following block diagram:



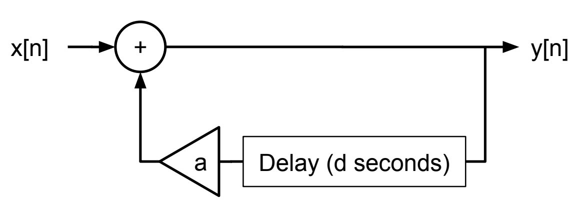
It's impulse response `h[n]` would be something like

```
h[n] = [1, "d seconds worth of zeros", a]
```

You can imagine that the output signal `y[n]` would contain the original signal `x[n]` followed by a scaled copy of `x[n]` occurring `d` seconds later. As long as `a` is between zero and one, that scaled copy should sound like an echo. You can implement this filter in Matlab with the following commands:

`d =` *some value of your choosing*
`a =` *some value of your choosing*
```
b = [ 1 zeros(1, d*fs ) a ];
y = filter(b,1,x);
```
Note that the `1` in the last step effectively makes the denominator of the transfer function equation $H(\Omega)$ equal to 1 and therefore makes the filter an FIR.

An IIR filter works mostly along the same lines, but instead of using a copy of the input signal to create the echo, it uses a copy of the output signal to create the echo:



It's impulse response `h[n]` would be something like

```
h[n] = [1, "d seconds worth of zeros", a, "d seconds worth of zeros", a^2, "d
seconds worth of zeros", a^3, ...]
```

This is a very different echo than the FIR case, as we'll see. Recall that the filter will only converge when $|a| < 1$. You can create it in Matlab by doing:

```
b = 1;
a = [ 1 zeros(1, d*fs ) a ];
y = filter(b,a,x);
```

Another form of echo is *stereo* echo. We create this by introducing a delay between the sounds heard in the left and right ears, respectively. The brain interprets the delay to mean the sound reached one ear before another which in turn would mean that the sound came in at an angle. These left-right delays are therefore the basis for surround sound stereo. In Matlab, you can build a stereo signal by letting your signal have two columns. The first column will correspond to the left speaker and the second column to the right speaker. You can use your FIR and/or IIR filters to introduce a delay between the two ears. Assuming `x` is some row vector audio signal, we might say:

```
sig_out(:,1) = x; % assign the left channel
sig_out(:,2) = filter(b,a,x); % introduce an echo to the right channel using an FIR
or IIR filter
```

Things to Do

1. Load `hw5Data.mat` and plot the signal `beep` using the command `plot(t_beep,beep)`. Listen to it using the `soundsc` command.
2. Create an FIR-echo of `beep`. Play around with different values of `d` and `a` till you hear something that is noticeable and sounds fun or interesting. Create a plot of your echoed `beep` signal. Can you see the echo in the plot?
3. Create an IIR-echo of `beep`. Play around with different values of `d` and `a` till you hear something that is noticeable and sounds fun or interesting. Be sure to keep `|a|<1`. Create a plot of your echoed `beep` signal.
4. How are the FIR and IIR echoes different? Explain.
5. Try creating an IIR-echo of beep with `|a|>1`. What happens to the time plot? What happens to the sound? Why is it a bad idea to have `|a|>1`?
6. Using the two sound clips `audio1` and `audio2`, experiment with creating some echo sound effects into the music. What values of `d` and/or `a` tend to sound good, bad, or unnoticeable? Try to give a sense of how these options relate to the quality of the sounds you are creating.
7. Using either of the `audio` tracks, create a stereo echo effect by introducing an FIR or IIR echo into one of the two channels. What values of `a` and `d` give a good sounding echo? Explain what you are hearing. What sounds better? FIR or IIR?
8. *Honors Students:* Create a discrete-time filter to remove as much of Mariah Carey's voice as you can from `audio3`. You can use `fdatool` to build a discrete-time filter and `mySpectrogram` to identify which frequencies you might want to be removing.

Write a paper (using the IEEE template) of any length that explains your methods and findings. Include any observations you make along the way and try to demonstrate that you have some understanding of how the filters affect the sounds you are creating. Submit your paper (MSWord format only) along with audio of your echo signals from steps 6, 7, and 8 in a single zip file via the link on Canvas. The due date is Monday April 30th at 5pm.

Finally!
For one of my favorite examples of signal processing in music, visit this site:
`http://www.npr.org/2010/08/03/128935865/queens-brian-may-rocks-out-to-physics-photography`
and scroll to the part titled, "On The 'Stomp-Stomp-Clap' Section Of 'We Will Rock You' ". You might be surprised to learn that the guitarist for one of the best known bands in history has a PhD in Astrophysics!