ECE 4750: Digital Signal Processing
Project 5
Echo Cancellation

You must prepare a report containing the answers to each question, include the figures and results. Also, you will need to submit your code scripts.

In this project you will work on removing an echo from a recording of a speech signal. A simple echo has two main parameters: The first parameter is the delay: D seconds (or N samples). The second parameter is the amplitude of the echo:α.

Main Objectives:

1. **Create the Echo Signal.** You will add an echo to a speech signal.
2. **Remove the Echo.** You will be given an echo-corrupted speech signal. Remove the echo from the speech signal. In this part you learn about the simple model of echoed signal with parameters D and α.
3. **Estimating Unknown Echo Parameters for a Simple Echo Model.**

   You will try to determine the echo parameters N and α from a speech signal, and then remove the echo.

Tasks:

1. Work with the Signal

a) Load and read the 'humanvoice.wav' speech signal with the command

```
audioread('humanvoice.wav');
```

b) For convenience, lets define this speech as a variable 'x'.

```
>> x = audioread('humanvoice.wav');
```

c) For simplicity, let's work with a chunk of this signal.

```
>> x = x(10000:50000); % working with a chunk of signal.
```

d) You can listen to the signal 'x' with the command 'soundsc(x,Fs)'. The sampling frequency Fs= 8000 Hz.

```
>> Fs = 8000;
>> soundsc(x,Fs);
>> pause(10); % for the later purposes in your code.
```

e) Make a plot of the signal x versus time (in SECONDS). You will need the sampling frequency Fs. How do you find the exact duration of this signal in seconds?

```
>> plot(0:1/Fs:1/Fs*(length(x)-1),x)
```

Explain why it works?

2. Create the Echo-Corrupted Signal

a) We can create an echo by filtering the signal with the following difference equation:

$$y[n] = x[n] + \alpha x[n-N] \quad (1)$$

where $x[n]$ is the echo-free speech signal, which has been delayed by N samples and added back in with its amplitude scaled by α. This is a reasonable model for an echo resulting from the signal reflecting off an absorbing surface like a wall. The delay parameter D to in seconds is N/Fs where Fs is the sampling frequency. (Can you explain why?)

b) What is the impulse response of this system?

c) Let's add an echo to the signal with a delay of D = 1 second and an amplitude of $\alpha = 0.8$. What is the delay N in samples?

d) Based on the difference equation (1), create an echo version of the signal x using the convolution, i.e., $y[n] = x[n] * h[n]$. What should we define for h?

```
>> y = conv(x,h);
```

e) Listen to the output signal 'y'. Do you hear the echo?

```
>> soundsc(y)
>> pause(10)
```

f) Another way of creating the echoed signal is by using the command "filter". The filter command has three main arguments. Generate the same echo-corrupted signal using filter command and listen to the output. Explain why 'a' and 'b' are defined as follows.

```
>> % Generating echoed sound synthetically.
>> alpha = 0.8; % |alpha|<1
>> D = ?; % set it.
>> b=1;
>> a=[ 1 , zeros(1,D) , alpha ];
>> y  = filter(b,a,x);
>> soundsc(y)
>> pause(10)
```

3. Remove the Echo.

a) The difference equation (1) for modeling the echo is an LTI system. To remove the echo, we need to implement the <u>inverse system</u>. What is the impulse response of the inverse system?

b) What is the duration of this impulse response?

c) For the echo parameters you used above, plot the impulse response of the inverse system. Make the horizontal axis in your plot in units of Seconds.

d) Implement the inverse of the echo system to perform echo cancellation. You can use the command

x_hat = filter(a,b,y). What are the vectors b and a?

e) Listen to the result. Is the echo removed?

4. Remove the Echo by Approximating the Echo Model Parameters via Trial-and-Error.

a) Import the given echo-corrupted signal by loading the '**x_echoed2.wav**' speech signal with the command 'audioread'. Fs=8000 Hz.

```
>> y = audioread('x_echoed2.wav');
```

You can listen to the signal with the command

```
>> soundsc(y)
>> pause(10)
```

b) The echoed signal (x_echolded2 or say y) was generated via filtering an echo-free signal 'x' with the following difference equation:

$$y[n] = x\,[n] + \alpha\, x\,[n-N]\ (2)$$

where $x\,[n]$ is the echo-free speech signal, which has been delayed by N samples and added back in with its amplitude scaled by α. The delay parameter D in seconds is N/F_s where F_s is the sampling frequency.

c) The parameters of the echo are unknown. All we know is that the echo is created based on the difference equation (2). The echoed signal y has been generated via (2) and hs been implemented via

y = conv(h,x) or y=filter(b,a,x), where a and b should be defined appropriately from equation (2). What are h, a, and b in terms of α *and* N?

d) *Experimental echo cancellation:* Set initial values for the filter parameters, i.e., a and b. To remove the echo, we need to perform the inverse behavior of the echo generated model. This task can be done by swapping the places of a and b in the filter function, i.e.,

x_hat=filter(a,b,y)

This is the implementation of the inverse of the echo system to perform *echo cancellation*. What are the vectors a and b? By trial and error, approximate the echo parameters α and N, redefine the filter parameters a and b, generate the output 'x_hat', and listen to the sound. Repeat this process till you hear no more echo in 'x_hat. Explain what values to came up with using trial-and-error method. Include your codes.

5) Estimating Unknown Echo Parameters Using Autocorrelation

In a real problem, you will probably not know the echo parameters. You will need to estimate them from the echo signal itself.

Suppose you are given the data $y[n]$ (which is corrupted by an echo). However, you do not know the value of the delay N *and* α. We want to determine a method of estimating N based on the autocorrelation function of $y[n]$. Let $r_{yy}[n]$ be the autocorrelation of $y[n]$,

$$r_{yy}[n] = y[n] * y[-n].$$

a) Import the echo-corrupted signal by loading the '**x_echoed2.wav**' speech signal with the command 'audioread'.

```
>> y = audioread('x_echoed2.wav');
```

b) Compute and plot $r_{yy}[n]$) vs time (in Seconds). $r_{yy}[n]$ is the autocorrelation of 'y'. To compute in Matlab the autocorrelation of a signal you can use the command

```
>> ryy =  conv(y,y(end:-1:1));
```

or the command xcorr

```
>> ryy = xcorr(y);
```

Plot the autocorrelation ryy. You should see that it has a peak in the middle.

c) Can you determine what the value of N is from the plot of $r_{yy}[n]$? (Look at the peaks of $r_{yy}[n]$.)

d) The amplitude α is harder to estimate than N. Once you have estimated N, remove the echo by trying out different values of α and listening to your result. What N and α doyou find work best? Are you able to successfully remove the echo?

e) What if an echo has two components?

$$y(n) = x[n] + \alpha_1\ x[n - N_1] + \alpha_2\ [n - N_2]$$

Discuss what system is required for echo cancellation. How would you find the echo parameters?