EE110B Lab 5

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1) Assume that the system between the voice $x[n]$ from your mouth and the acoutsic signal $y[n]$ picked up by a microphone is linear and time-invariant, and hence

$$y[n] = h[n] * x[n] = \sum_{l=1}^{L} h[l]x[n-1]$$

To see the effect of the echo distortion, assume

$$x[n] = \sin\left(\frac{\pi}{5}n\right)(u[n] - u[n-10])$$

and also chose the echo coefficents $h[l]$ for all $0 \le l \le 10$ randomly. Here L = 10.
Plot $x[n]$ and $y[n]$ for $0 \le n \le 20$. Discuss the impact of $h[n]$ on $y[n]$ in relation to $x[n]$.

```
%n goes from 0 -9 due to the unit step function.
n = 0:1:9;
x_n = sin(pi/5*n);
% Plot the convolution
L = 10;
l = 0:1:L+1;
%Control random number generator so that we get the same random number
%everytime.
rng(0)
h_l = randn(1,L+1);
y_n = conv(x_n, h_l);
fig = figure;
fig.Position = [100,100,1000,300];
```

Plot of $x[n]$

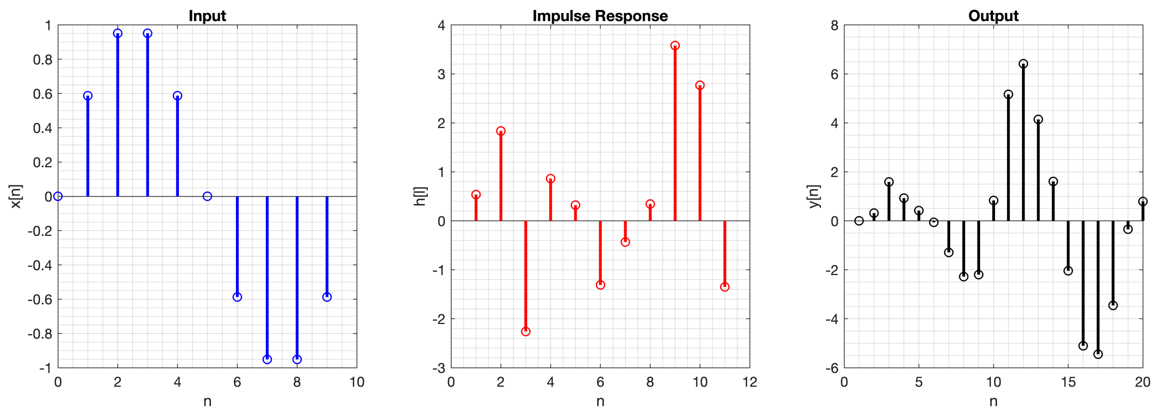
```
subplot(1,3,1);
stem(n,x_n, 'b', 'LineWidth',2);
xlabel('n');
ylabel('x[n]');
title('Input');
grid on;
grid minor;

subplot(1,3,2);
stem(h_l,'r','LineWidth',2);
xlabel('n');
ylabel('h[l]');
title('Impulse Response');
grid on
```

```
grid minor
```

Plot of $y[n]$

```
subplot(1,3,3);
stem(y_n, 'k', 'LineWidth',2);
xlabel('n');
ylabel('y[n]');
title('Output');
grid on;
grid minor;
```



Discuss the impact of $h[n]$ on $y[n]$ in relation to $x[n]$.

In relation to $x[n]$ the output $y[n]$is heavily distorted. I know this because the output is not an exact replica of the input signal $x[n]$. This distortion is due to varying amplitudial lengths of the impulse response. We know that in the frequency domain, every discrete time signal is periodic with a periodicity of 1. however, here we see that the signal repreats itself twice. The second half is the echo. I surmise that the differences in amplitudes represent the varying anounts of intensities of the echos.

2) Assume that the same $x[n]$ in (2) but $h[n] = 0.98^n u[n]$ which is the impulse response of a first-order feedback system.

a) Compute and plot $y[n] = x[n] * h[n]$ for $0 \leq n \leq 20$ (with a large L such that $h[L]$ is negligible). Discuss the distortion effect by $h[n]$

```
%h
L = 1000;
l = 0:1:L;
h_l = 0.98.^l;
```

Plot of $y[n]$

```
y_n = conv(x_n, h_l);
```

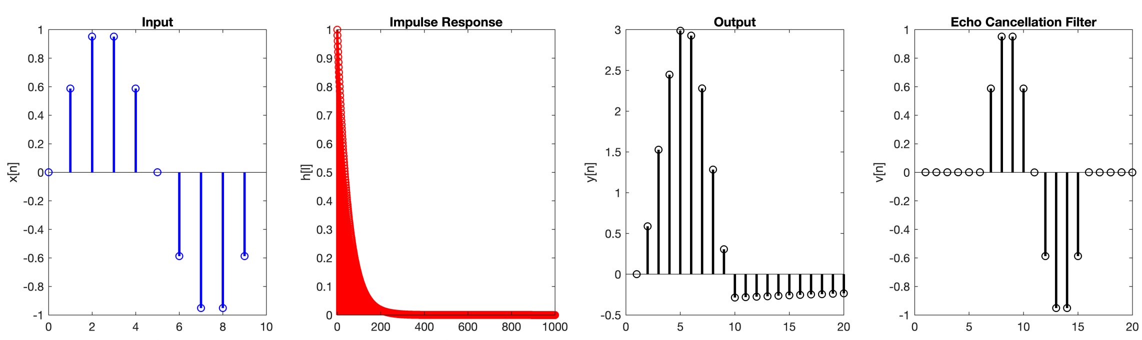
```
fig = figure;
fig.Position = [100, 100, 1200, 300];
subplot(1,4,1);
stem(n, x_n, 'b','LineWidth',2);
xlabel('n');
ylabel('x[n]');
title('Input');

subplot(1,4,2)
stem(l, h_l, 'r','LineWidth',2);
xlabel('n');
ylabel('h[l]');
title('Impulse Response');

subplot(1,4,3)
stem(y_n(1:20),'k','LineWidth',2);
xlabel('n');
ylabel('y[n]');
title('Output');
```

b) Compute and plot $v[n] = g[n] * y[n]$ for $0 \leq n \leq 20$ where $y[n] = \delta[n-5] - 0.98\delta[n-6]$. Discuss the quality of $g[n]$ as an echo cancellation filter

```
% echo cancellation
g_n = zeros(1, 100);
%Remember that arrays are indexed at 1, thus the value for the delta function equation
g_n(6) = 1;
g_n(7) = -0.98;
v_n = conv(g_n,y_n);
subplot(1,4,4)
stem(v_n(1:20),'k','LineWidth',2);
xlabel('n');
ylabel('v[n]');
title('First Order Feedback Echo Cancellation Filter');
```



Discuss the distortion effects of $h[n]$

We can see the output is distorted still. We know this to be true because the output does not match the input. This distortion is due to varying ampltudial lengths of the impulse response. If the impulse had been a delta function, the output would have been a perfect copy of the signal just shifted based on the location of the delta function. Since h[n] is not a delta function but a train of delta functions with different amplitudes we have distortion.

Discuss the quality of $g[n]$ as an echo cancellation filter

the g[n] looks exactly like the original signal with a delay of 6 units. Therefore, as an echo cancelation filter, I do not think that this works very well. The receiver will hear an echo due to the delayed time that is taken to transmit the message. If we want $g[n]$ to act more efficiently as a cancellation filter, we need less delay so that it is on par with the speakers message.

3) If $h[n] = a^n u[n] - b^n u[n]$ (which is the impulse response of a second-order feeback system), then its DTFT is $H(f) = \frac{1}{1 - \mathrm{a}e^{-\mathrm{j}2\pi f}} - \frac{1}{1 - \mathrm{b}e^{-\mathrm{j}2\pi f}} = \frac{(a-b)e^{-\mathrm{j}2\pi f}}{1-(a+b)e^{-\mathrm{j}2\pi f} + \mathrm{ab}e^{\mathrm{j}4\pi f}}$. A good inverse filter of $H(f)$ has the frequency response $G(f) = 1-(a+b)e^{-\mathrm{j}2\pi f} + \mathrm{ab}e^{-\mathrm{j}4\pi f}$. Assume the same $x[n]$ in (2) but $h[n] = 0.98^n u[n] - (-0.95)^n u[n]$.

a) Compute and plot $y[n] = x[n] * h[n]$ for $0 \le n \le 20$ (with a large L such that $h[L]$ is negligible). Discuss the distortion effect by $h[n]$

```
%h
L = 1000;
l = 0:1:L;
h_l = 0.98.^l-(-0.95).^l;
```

Plot of $y[n]$

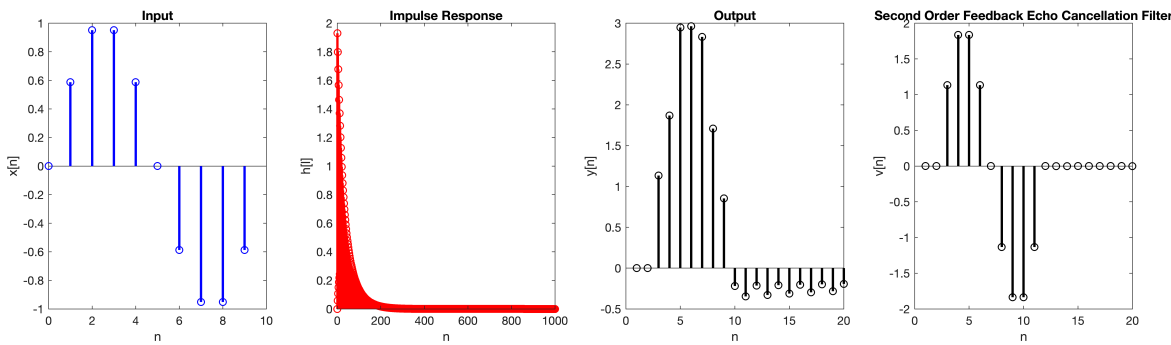
```
y_n = conv(x_n, h_l);
fig = figure;
fig.Position = [100, 100, 1200, 300];
subplot(1,4,1);
stem(n, x_n, 'b','LineWidth',2);
xlabel('n');
ylabel('x[n]');
title('Input');

subplot(1,4,2)
stem(l, h_l, 'r','LineWidth',2);
xlabel('n');
ylabel('h[l]');
title('Impulse Response');

subplot(1,4,3)
stem(y_n(1:20),'k','LineWidth',2);
xlabel('n');
ylabel('y[n]');
title('Output');
```

b) Compute and plot $v[n] = g[n] * y[n]$ for $0 \leq n \leq 20$ where $y[n] = \delta[n] - (a+b)\delta[n-1] + \mathrm{ab}\delta[n-2], a = 0.98$ and
Discuss the quality of $g[n]$ as an echo cancellation filter

```
% echo cancellation
g_n = zeros(1, 100);
%Remember that arrays are indexed at 1, thus the value for the delta function equation
a = 0.98;
b = (-0.95);
g_n(1) = 1;
g_n(2) = -(a+b);
g_n(3) = (a*b);
v_n = conv(g_n,y_n);
subplot(1,4,4)
stem(v_n(1:20),'k','LineWidth',2);
xlabel('n');
ylabel('v[n]');
title('Second Order Feedback Echo Cancellation Filter');
```



Discuss the distortion effects of $h[n]$

We can see the output is distorted still. We know this to be true because the output does not match the input. This distortion is due to varying ampltudial lengths of the impulse response. If the impulse had been a delta function, the output would have been a perfect copy of the signal just shifted based on the location of the delta function. Since h[n] is not a delta function but a train of delta functions with different amplitudes we have distortion. However, the distortion is less than in 2 because the amplitude of the impulse response for this one sided exponential function is greater than 1.

Discuss the quality of $g[n]$ as an echo cancellation filter

the g[n] looks exactly like the original signal with a delay of around 3 units. Therefore, as an echo cancelation filter, I do think that this works much better than the first-order feedback system. The receiver might hear a slight echo due to the delayed time that is taken to transmit the message. If we want $g[n]$ to act more efficiently as a cancellation filter, we need less delay so that it is on par with the speakers message. This can be done by changing the parameters of a and b.

EE110B Lab 5

In a small room with concrete walls (or other similar environment), we often notice acoustic echoes. If you do not talk very closely to a microphone (on a cell phone for example) in such an environment, the microphone will pick up echoes. The echoes are distortion and could cause many problems. In this lab, you will evaluate the effect of echoes and test a method for echo cancellation.

So what is

1) Assume that the system between the voice $x[n]$ from your mouth and the acoustic signal $y[n]$ picked up by a microphone is linear and time-invariant, and hence

h(n) must be fast enough

$$y[n] = h[n] * x[n] = \sum_{l=1}^{L} h[l]x[n-l]. \quad (1)$$

← in the ideal case, the h(n) should be one delta function

To see the effect of the echo distortion, assume

$$x[n] = \sin(\frac{\pi}{5}n)(u[n] - u[n-10]) \quad (2)$$

y(n) will be the distorted a echo'd version of x[n] because y[n] is a summation of a delayed versions of x(n)

and also choose the echo coefficients $h[l]$ for all $0 \leq l \leq 10$ randomly. Here $L = 10$. Plot $x[n]$ and $y[n]$ for $0 \leq n \leq 20$. Discuss the impact of $h[n]$ on $y[n]$ in relation to $x[n]$.

2) Assume the same $x[n]$ in (2) but $h[n] = 0.98^n u[n]$ which is the impulse response of a first-order feedback system.

impulse response of a first order feed back system 1st ur

ways to cancel distortion

   a) Compute and plot $y[n] = x[n] * h[n]$ for $0 \leq n \leq 20$ (with a large L such that $h[L]$ is negligible). Discuss the distortion effect by $h[n]$. ← perform another convolution to ~~retur~~ see if we can recover original signal
   b) Compute and plot $v[n] = g[n] * y[n]$ for $0 \leq n \leq 20$ where $g[n] = \delta[n-5] - 0.98\delta[n-6]$. Discuss the quality of $g[n]$ as an echo cancellation filter.

3) If $h[n] = \boxed{a^n u[n] - b^n u[n]}$ ← general form of a second order feedback system (which is the impulse response of a second-order feedback system), then its DTFT is $H(f) = \frac{1}{1-ae^{-j2\pi f}} - \frac{1}{1-be^{-j2\pi f}} = \frac{(a-b)e^{-j2\pi f}}{1-(a+b)e^{-j2\pi f}+abe^{-j4\pi f}}$. A good inverse filter of $H(f)$ has the frequency response $G(f) = 1 - (a+b)e^{-j2\pi f} + abe^{-j4\pi f}$. Assume the same $x[n]$ in (2) but $h[n] = 0.98^n u[n] - (-0.95)^n u[n]$.
   a) Compute and plot $y[n] = x[n] * h[n]$ for $0 \leq n \leq 20$ (with a large L such that $h[L]$ is negligible). Discuss the distortion effect by $h[n]$.
   b) Compute and plot $v[n] = g[n] * y[n]$ for $0 \leq n \leq 20$ with $g[n] = \delta[n] - (a+b)\delta[n-1] + ab\delta[n-2]$, $a = 0.98$ and $b = -0.95$. Discuss the quality of $g[n]$ as an echo cancellation filter.

x(n) comes from mic

there's a channel in between the microphone and my voice

h(n) is the impulse response of the channel

The output of this L.T.I ~~time~~ system can be found via the convolution

Impulse Response of a first-order feedback system

2

a) $y[n] = x[n] * h[n]$ (1) equation

$$\big\downarrow \mathcal{F}$$

$$Y(f) = X(f)H(f)$$

$v[n] = y[n] * g[n]$ (2) equation

$$\big\downarrow \mathcal{F}$$

$$V(f) = Y(f)G(f)$$

Substituting (1) into (2):

Here, $G(f)$ is the frequency response of the filter

$$\underline{V(f)} = \underbrace{X(f)H(f)G(f)}_{=1}$$

Here, $H(f)$ is the impulse response of the channel that caused the echos in the frequency domain

To retrieve the original signal $X(f)$ the product of the frequency response of the channel, and the frequency response of the filter must be 1.

• Therefore, whatever echo cancellation device I am designing should have the following

$$G(f) = \frac{1}{H(f)}$$

$$h[n] = 0.98^n u[n]$$

$$\big\downarrow \mathcal{F}$$

$$H(f) = \frac{1}{1 - 0.98e^{-j2\pi f}}$$

$$\mathcal{F}^{-1}[G(f)] = \mathcal{F}^{-1}[1/H(f)]$$

$$g[n] = \mathcal{F}^{-1}[1 - 0.98e^{-j2\pi f}]$$

$$g[n] = \delta[n] - 0.98\delta[n-1]$$

Thus, if the designed cancellation filter is as the one above we should not have any distortion and should be able to recover the original signal.