Lab 8: Synthesizing the Sound of a Plucked String

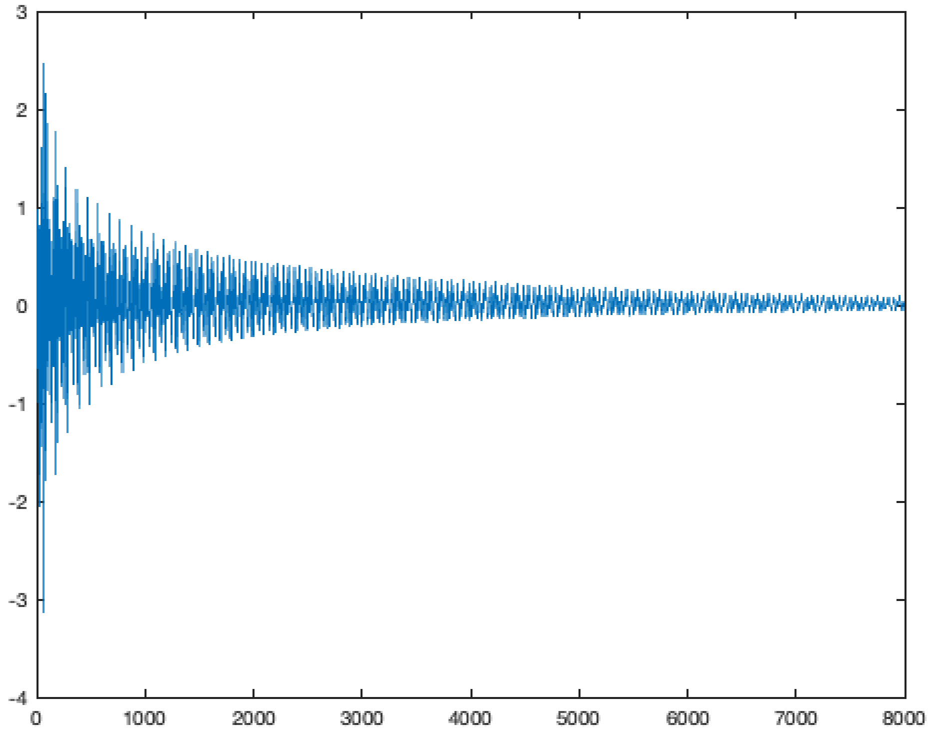
Table of Contents

Questions and Calculations

```
%{

    --(+)--------------------------------------->
       ^       _____     _______     ______     |
       |_____| LPF |---| Delay |---| Gain |<---
               -----     -------     ------

    LPF: H(z) = 0.5 + 0.5z^(-1)
    Delay: D(z) = z^(-N)
    Gain: G(z) = K

%}

%{

Q:  What is the total transfer function of the system
    above? How many poles does it have?
A:  F(z)=z^(-N)*(K)(0.5+0.5z^(-1))
    H(z)=1/(1-F(z))
    There are N+1 poles.

Q:  Difference Equation for the above system.
A:  Y(Z)(1-K(0.5z^(-N)-0.5^z^(-N-1))) = X(Z)(1)
    y(n) = x(n) + K0.5y(n-N) + K0.5y(n-N-1)

Q:  y=filter(b,a,x). What is a,b.
A:  b=[1, zero(1,N)], a=[0.5, 0.5, zeros(1,N-1)]

%}
```

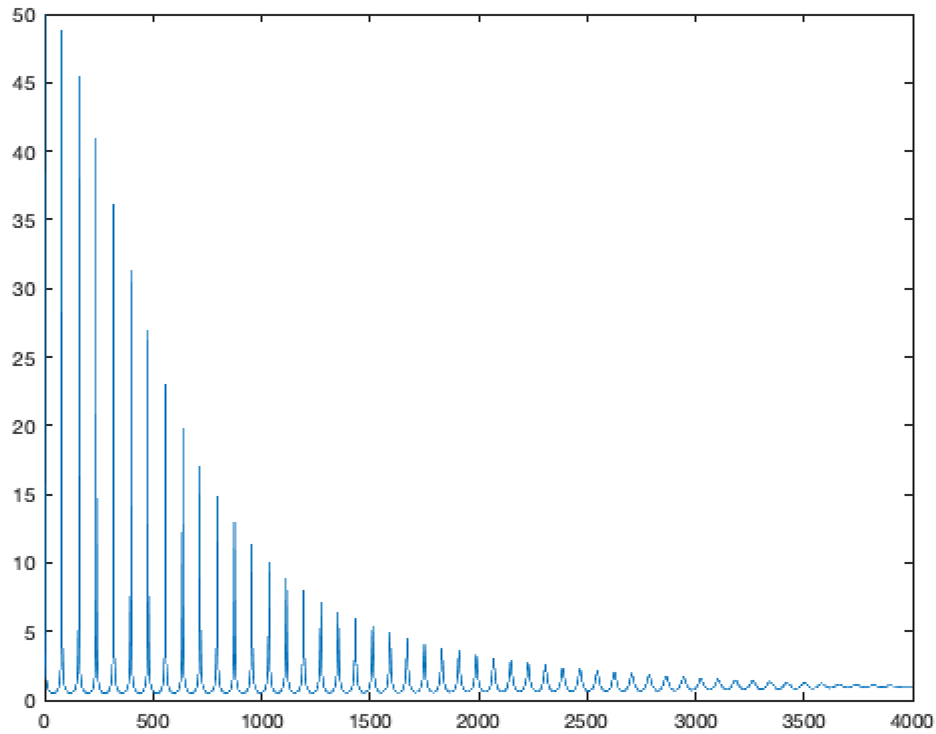
Implementation

```
Fs = 8000;
K = 0.98;
N = 100;
L = 7900;

b=[1];
a=[1, zeros(1,N-1), -0.5*K, -0.5*K];

x = [randn(1,N) zeros(1,L)];

y = filter(b,a,x);
figure %Figure 1
plot(y)
soundsc(y,Fs)

%{
Q:  What is N and L if we want 1 second sound?
A:  have signal y with length N+L points. Since we
    want to sound to last 1 second and are given
    that N=100 and Fs=8000, we have L=7900. Gain
    value K=0.98 for this example.

Observation: The sound is not the same every time
because of the randn that changes the initial data
point of the input signal.
%}
```



Frequency Response and Periodicity

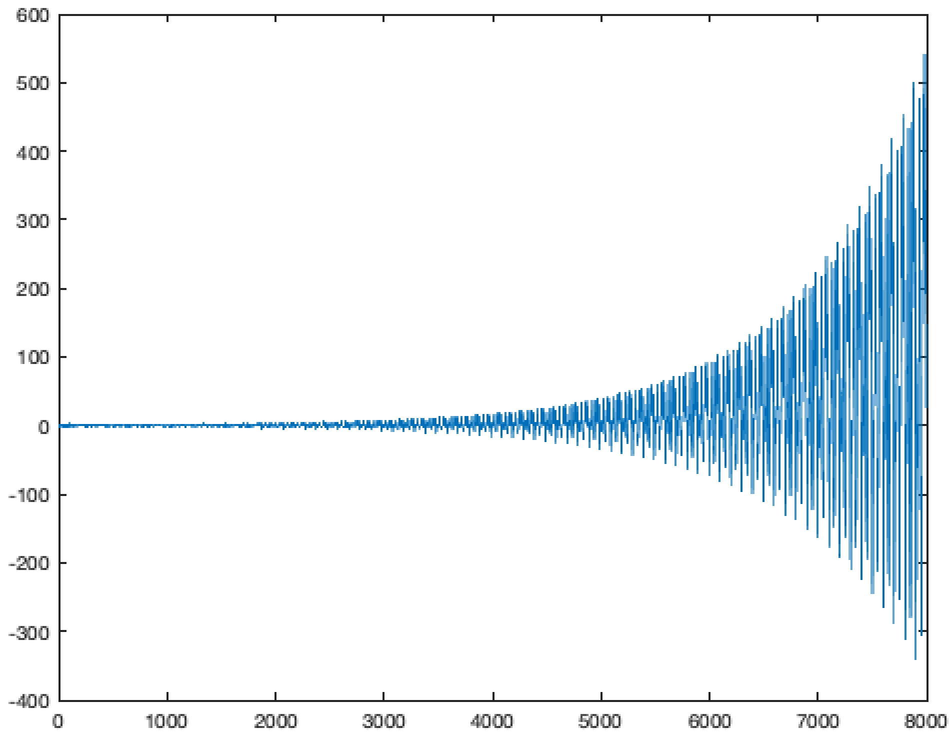
```
Fs = 8000;
K = 0.98;
N = 100;
L = 7900;

b=[1];
a=[1, zeros(1,N-1), -0.5*K, -0.5*K];

x = [randn(1,N) zeros(1,L)];

y = filter(b,a,x);
[H,w] = freqz(b,a,2^16);
figure %Figure 2
plot(w/pi*Fs/2, abs(H))
soundsc(y,Fs)

% Since the frequency response has consistently spaced
% spikes and attenuates frequencies between those spikes,
% it can be assumed that the output signal would be a
% Relatively simple combination of periodic signals, itself
% nearly a periodic signal of diminishing amplitude.
```



Changing the parameters

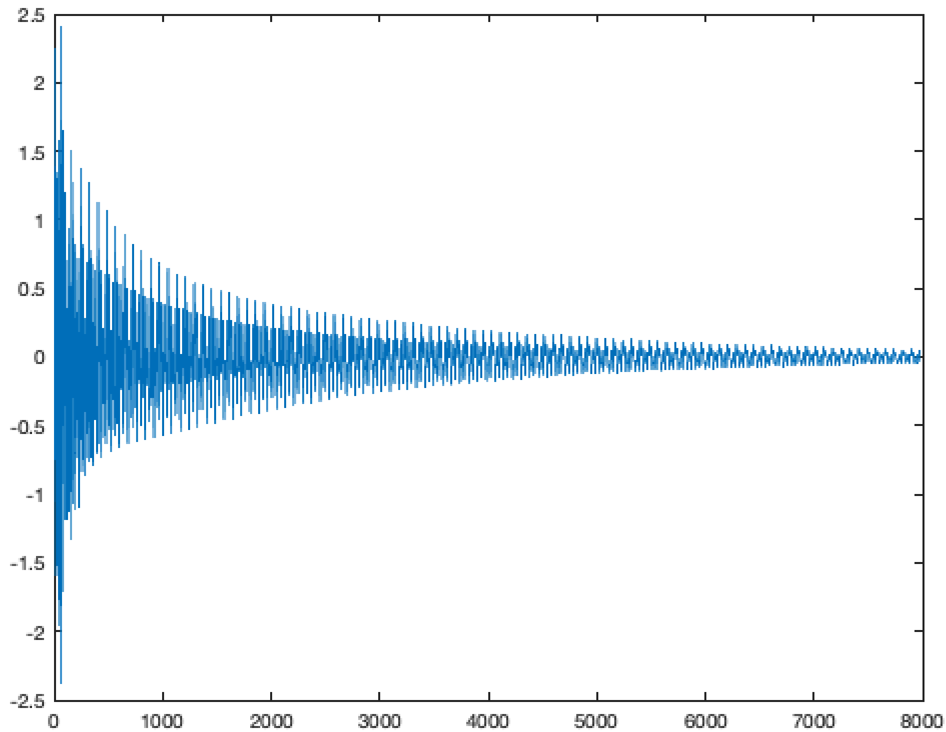
- Changing K, the Feedback Gain

```
Fs = 8000;
K = 0.98;
N = 100;
L = 7900;

% Changing the K value changes the speed at which
% amplitude of the signal decreases. Setting K>1 also
% causes the sound to be "reversed". See figure below.

K=1.1;

x = [randn(1,N) zeros(1,L)];
b=[1];
a=[1, zeros(1,N-1), -0.5*K, -0.5*K];

y = filter(b,a,x);
figure %Figure 3
plot(y)
soundsc(y,Fs)
```



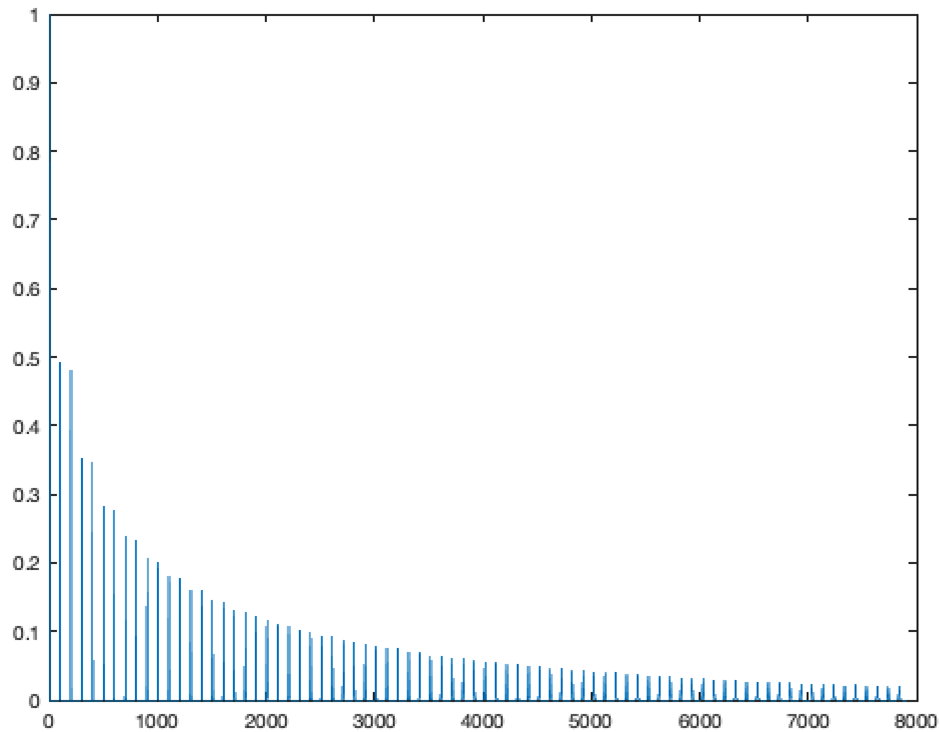
- Changing N, the Delay

```
Fs = 8000;
K = 0.98;
N = 100;
L = 7900;

% Changing the frequency changes the delay time. This
% effectively changes the frequency of the output signal
% and alters the pitch of the sound. Lower N raises the
% frequency and thus the pitch, while increasing N lowers
% the pitch. When frequency is too low, around N-300, the
% frequency is too low and the pitch becomes indescernible.

N=80;

x = [randn(1,N) zeros(1,L)];
b=[1];
a=[1, zeros(1,N-1), -0.5*K, -0.5*K];

y = filter(b,a,x);
figure %Figure 4
plot(y)
soundsc(y,Fs)
```



Changing the Input Signal

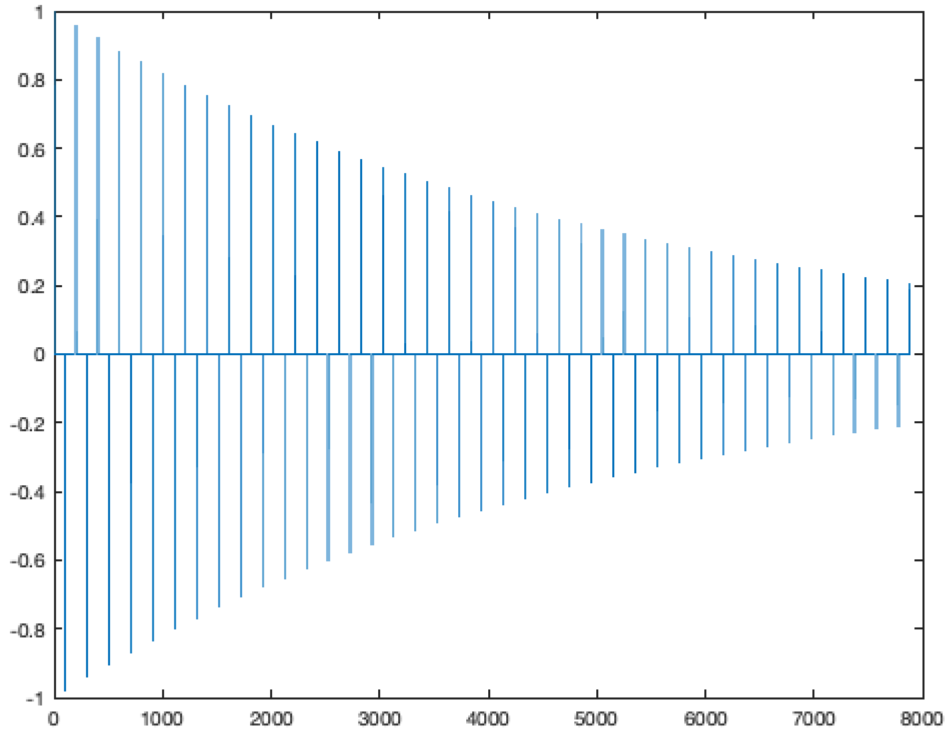
```
Fs = 8000;
K = 0.98;
N = 100;
L = 7900;

b=[1];
a=[1, zeros(1,N-1), -0.5*K, -0.5*K];

x = [1, zeros(1,L)];

y = filter(b,a,x);
figure %Figure 5
plot(y)
soundsc(y,Fs)

% Using the impulse as signal results in a much
% thinner output signal. The sound of this output
% is much less full and, in my opnion, much less
% realistic in tone.
```



The Comb Filter

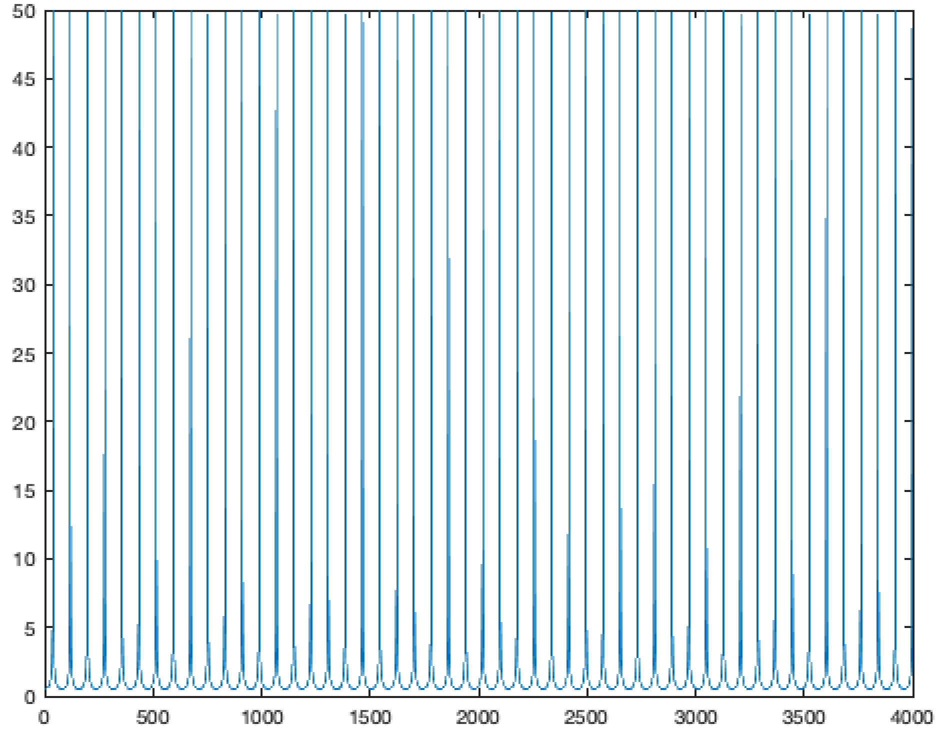
```
%{
A system with only the Gain and Delay is known
as the comb filter.

    --(+)---------------------------->
       ^     _______     ______     |
       |___| Delay |---| Gain |<---
             -------     ------

    Delay: D(z) = z^(-N)
    Gain: G(z) = K

    Transfer Function
    H(z) = 1/(1-F(z))
    H(z) = 1/(1-Kz^(-N))

    Difference Equation
    Y(z)(1-Kz^(-N)) = X(z)
    y(n) = x(n) + Ky(n-N)

%}

Fs = 8000;
```

```
K = 0.98;
N = 100;
L = 7900;

b=[1];
a=[1, zeros(1,N), K];

x = [1, zeros(1,L)];

y = filter(b,a,x);
figure %Figure 6
plot(y)
soundsc(y,Fs)
```



- Frequency Response of the comb filter

```
[H,w] = freqz(b,a,2^16);
figure %Figure 7
plot(w/pi*Fs/2, abs(H))
soundsc(y,Fs)

% The comb filter just allows evenly spaced frequencies
% to pass while attenuating the frequencies in-between.
% Presumably, it is called a comb filter because it looks
% like a comb.
```



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