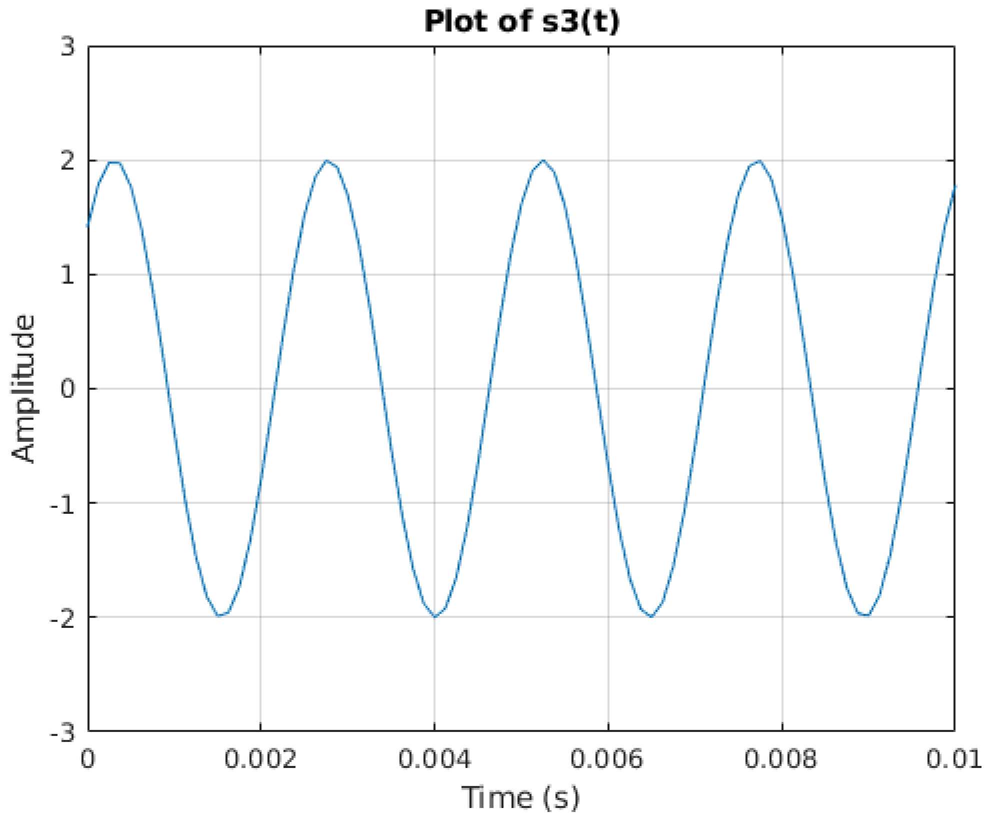
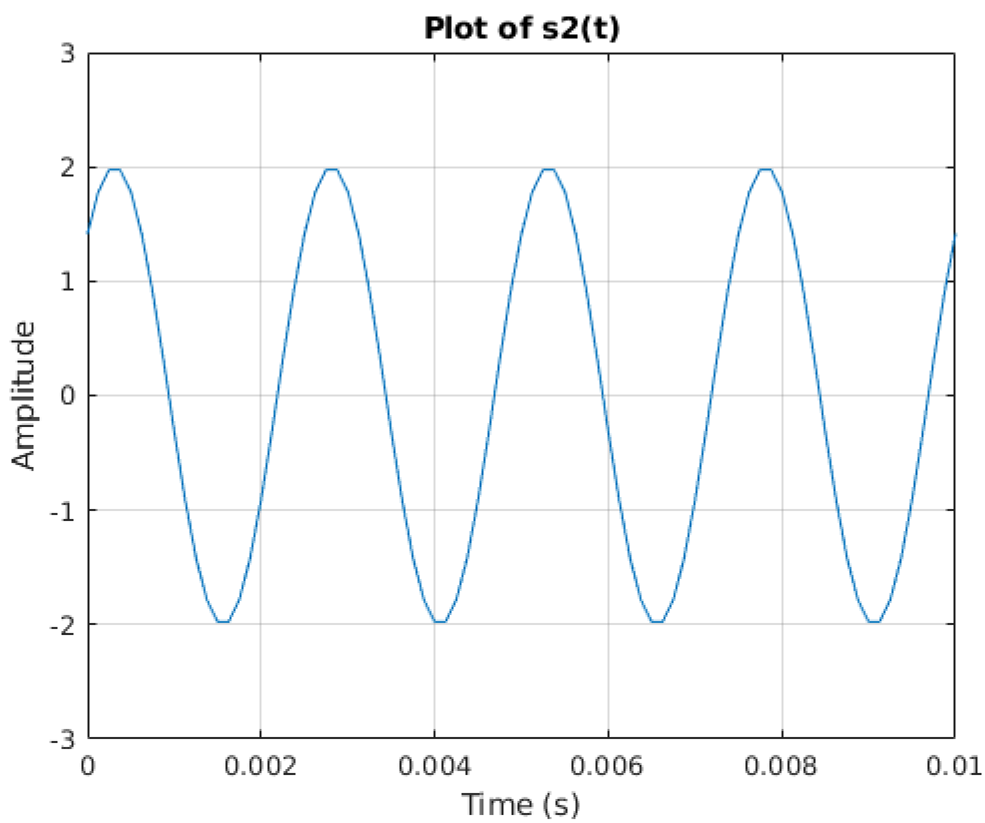
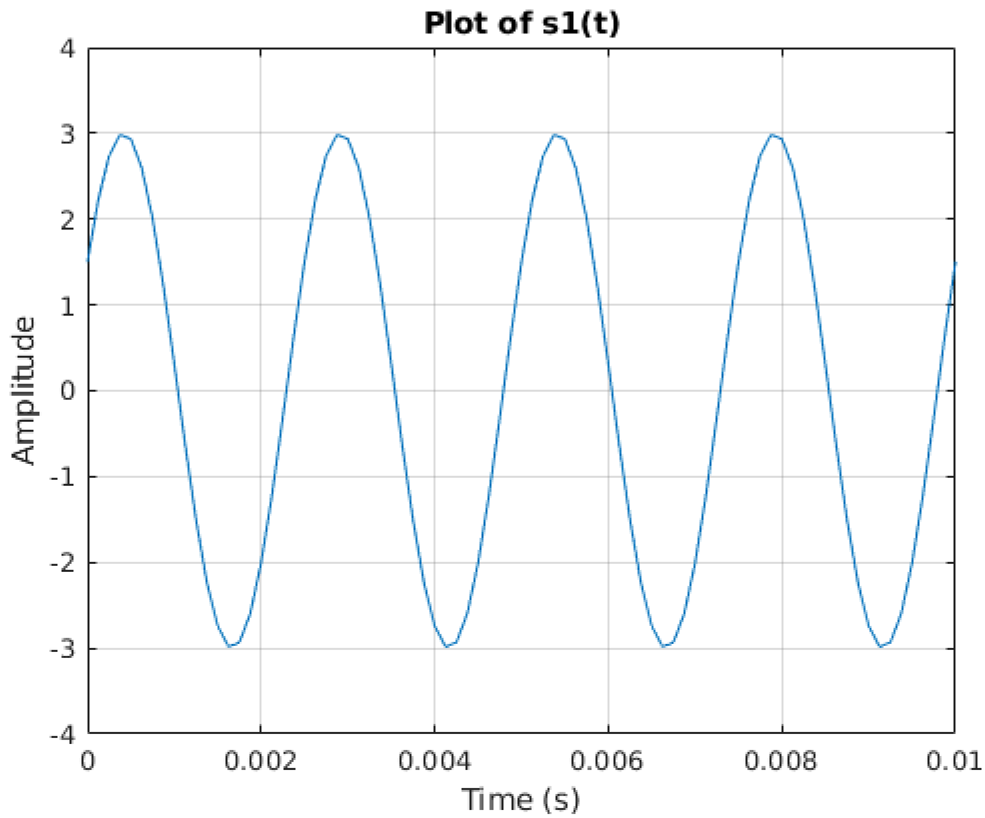
Exercise 2.1

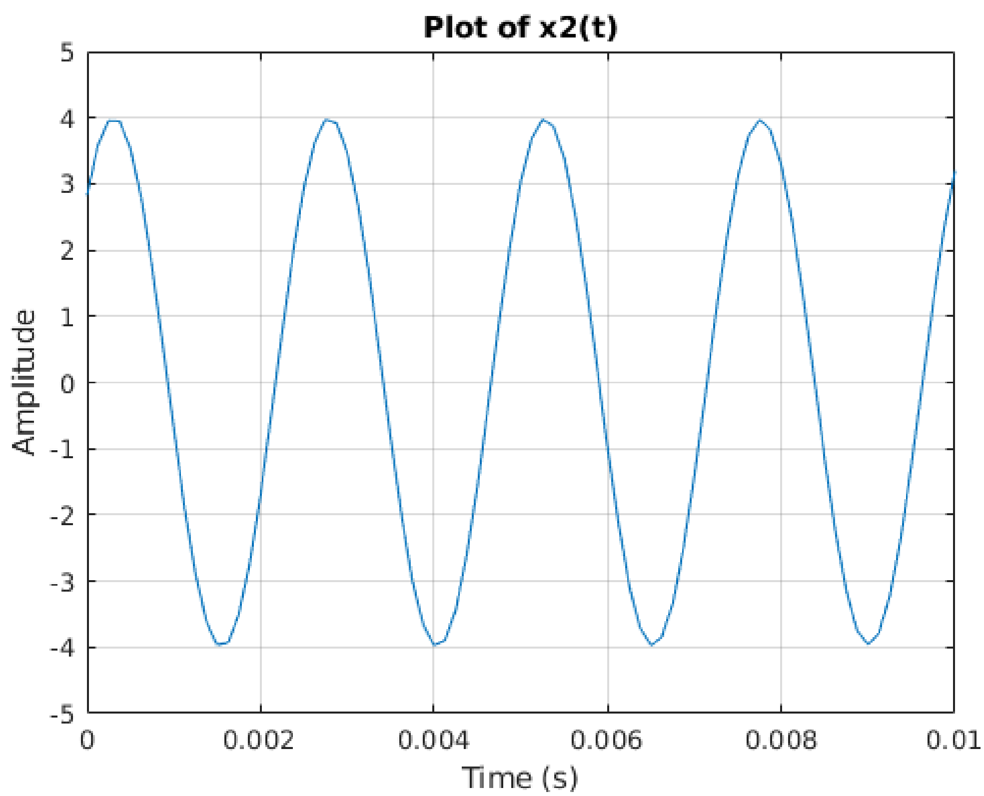
```
%%2.1(a)
tt = 0:1/8000:1;
s1 = 3*cos(800*pi*tt - pi/3);
s2 = 2 * cos(800 * pi * tt - pi/4);
s3 = 2 * cos(810* pi * tt - pi/4);
figure(1);
plot(tt,s1);
xlabel('Time (s)');
ylabel('Amplitude');
title('Plot of s1(t)');
grid on;
axis([0 0.01 -4 4]);
% There are 400 periods of the sinusoid s1 in a second
figure(2);
plot(tt,s2);
xlabel('Time (s)');
ylabel('Amplitude');
title('Plot of s2(t)');
grid on;
axis([0 0.01 -3 3]);
% There are 400 periods of the sinusoid s2 in a second
figure(3);
plot(tt,s3);
xlabel('Time (s)');
ylabel('Amplitude');
title('Plot of s3(t)');
grid on;
axis([0 0.01 -3 3]);
% There are 405 periods of the sinusoid s3 in a second

%%2.1(b)
s1_scaled = s1/max(abs(s1));
s2_scaled = s2/max(abs(s2));
s3_scaled = s3/max(abs(s3));

audiowrite('s1.wav',s1_scaled,8000);
audiowrite('s2.wav',s2_scaled,8000);
audiowrite('s3.wav',s3_scaled,8000);

%%2.1(c)
soundsc(s1_scaled,8000);
pause(2);
soundsc(s2_scaled,8000);
pause(2);
soundsc(s3_scaled,8000);
pause(2);

% The sound from s1 and s2 is indestinguishable. The sound from s3 is
% slightly different
```

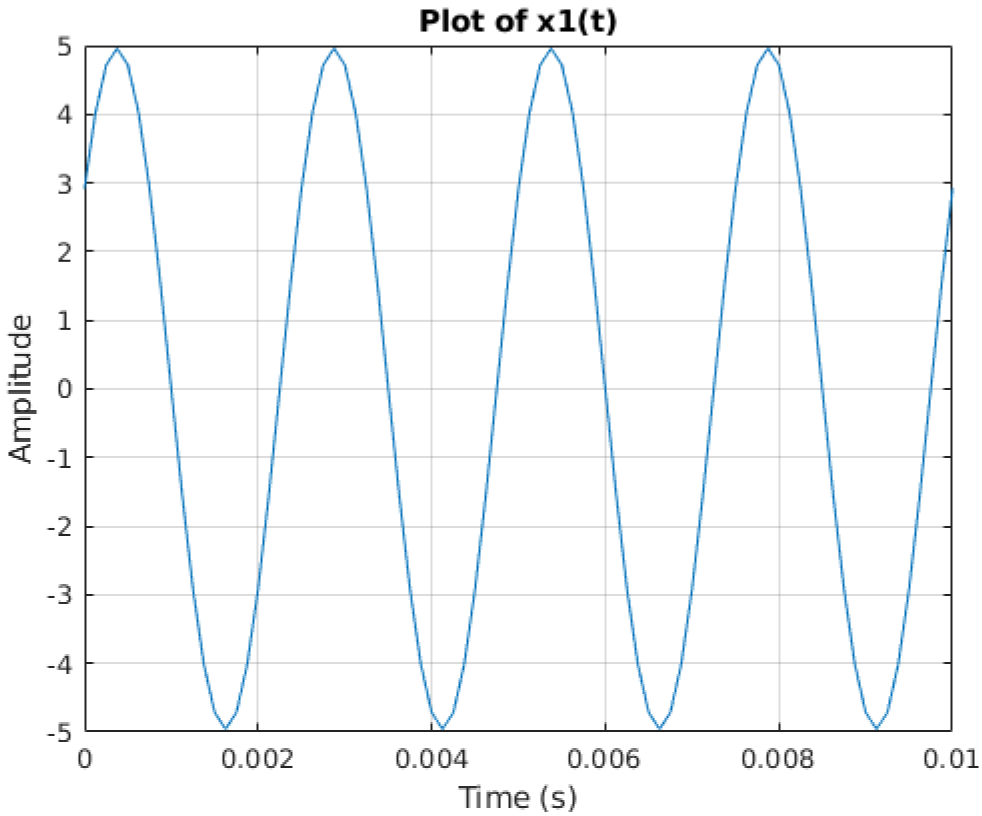
```
%%2.1(d)
x1 = s1 + s2;
% x1 = [e^j(pi/3) + e^j(pi/4)]e^j(800pi)t
figure(4);
plot(tt,x1);
xlabel('Time (s)');
ylabel('Amplitude');
title('Plot of x1(t)');
grid on;
axis([0 0.01 -5 5]);
% the plot does match the calculated equation


%%2.1(e)
% the fundemental frequency is 5 HZ
x2 = s2 + s3;
figure(5);
plot(tt,x2);
xlabel('Time (s)');
ylabel('Amplitude');
title('Plot of x2(t)');
grid on;
axis([0 0.01 -5 5]);
x2_scaled = x2/max(abs(x2));
soundsc(x2_scaled,8000);
pause(2);
% The fundemental frequency matches and is correct

%x2 sounds different from s2 and s3, x2's sound resonates while s2 and
 s3
%sound like pure tones
```


Plot of s3(t)
3
2
1
0
-1
-2
-3
Amplitude
0
0.002
0.004
0.006
0.008
0.01
Time (s)


Plot of s2(t)
3
2
1
0
-1
-2
-3
Amplitude
0
0.002
0.004
0.006
0.008
0.01
Time (s)


Plot of s1(t)
4
3
2
1
0
-1
-2
-3
-4
Amplitude
0
0.002
0.004
0.006
0.008
0.01
Time (s)


Plot of x2(t)
5
4
3
2
1
0
-1
-2
-3
-4
-5
Amplitude
0
0.002
0.004
0.006
0.008
0.01
Time (s)



Exercise 2.2

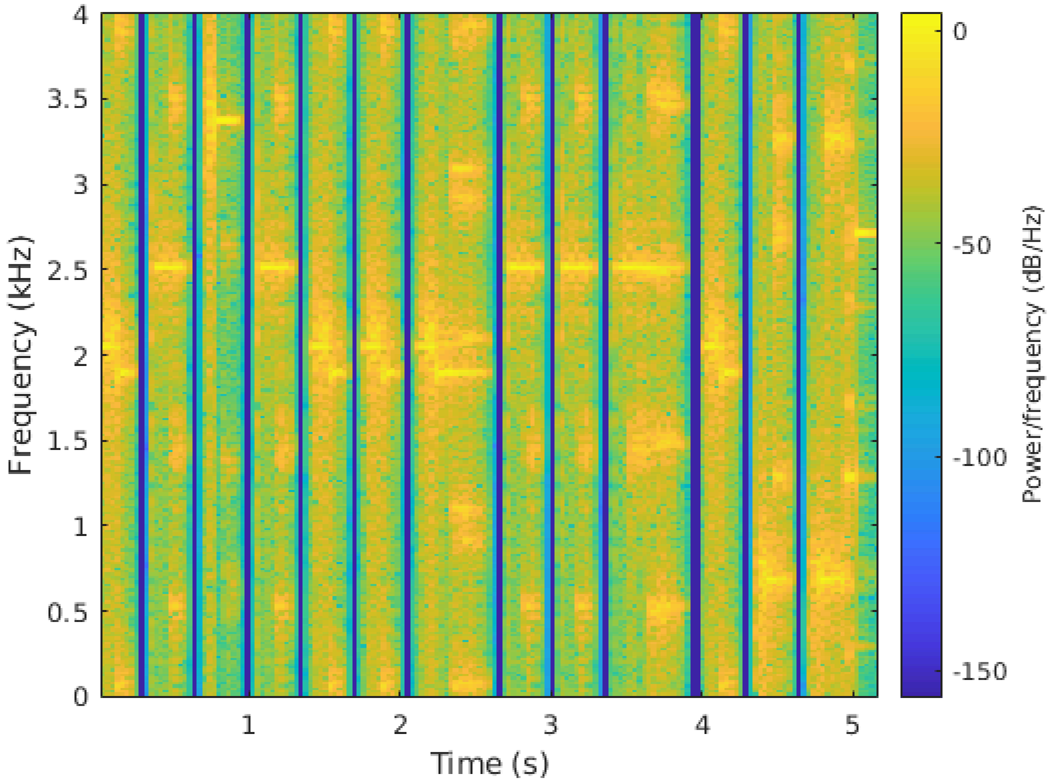
```
%%2.2(a)
%Functions below

%%2.2(b)
%Functions below

keys = [64 62 60 62 64 64 64 62 62 62 64 67 67];
Xs = [1 1 1 1 1 1 1 1 1 1 1 1 1];
start_times = [0 0.35 0.7 1.05 1.4 1.75 2.1 2.7 3.05 3.4 4 4.35 4.7];
end_times = [0.25 0.6 0.95 1.3 1.65 2 2.6 2.95 3.3 3.9 4.25 4.6 5.2];
fs = 8000;
x = build_song(keys, Xs, start_times, end_times, fs);
soundsc(x, 8000);
pause(2);
audiowrite('mary.wav',song,8000);
% Besides the tune, the song sounds garbled and the transitions
 between
% notes is not smooth.

%%2.2(c)
figure(4);
spectrogram(x,512,256,512,fs,'yaxis');
```

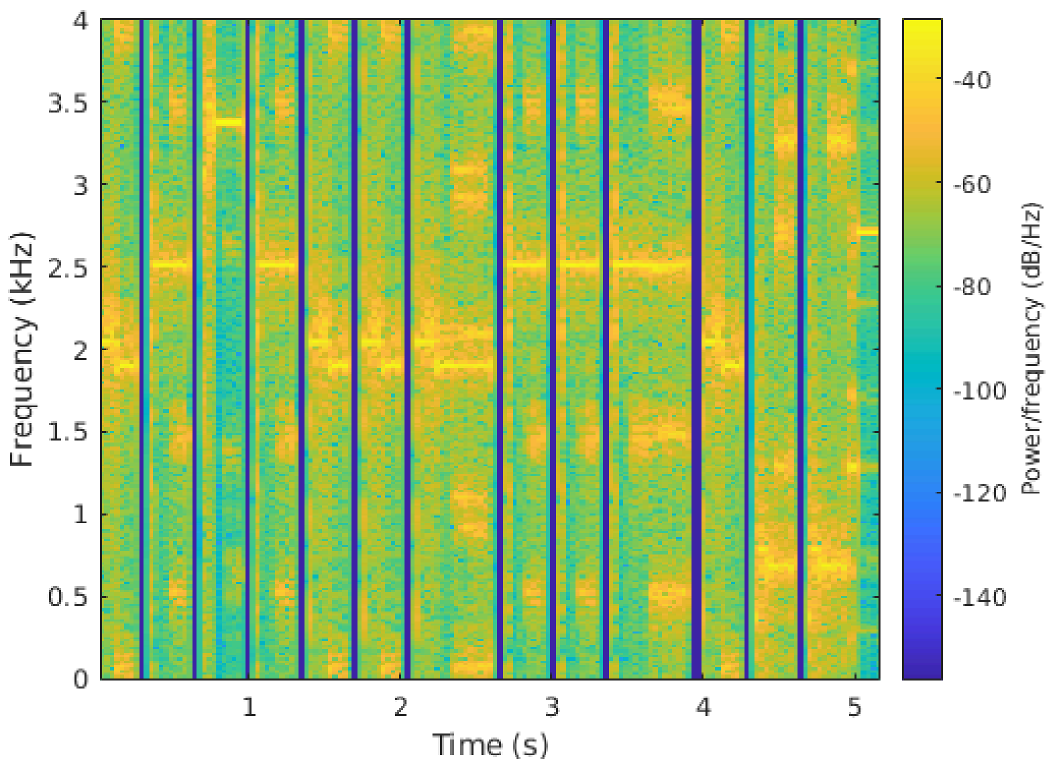
```
% The sinusoids in the spectrogram are represented by the sectioned
 off
% "yellowish" areas of the spectrogram. Some oddities seem to be the
 "blue"
% areas of high power over the entire frequency range. These seem to
 occur
% between consecutive notes, and likley correlate to the rough
 transitions
% between notes.

%%2.2(d)
Xs = [0.1155 * exp(j * -2.1299)...
, 0.3417 * exp(j * 1.6727), 0.1789* exp(j * -2.5454) ...
, 0.1232 * exp(j * 0.6607), 0.0678 * exp(j * -2.0390) ...
,0.0473 * exp(j * 2.1597), 0.0260 * exp(j * -1.0467) ...
, 0.0045 * exp(j * 1.8581), 0.0020 * exp(j * -2.3925)];

x = build_song_wo_adsr(keys, Xs, start_times, end_times, fs);
audiowrite('mary_trumpet.wav',x,8000);
figure(5);
spectrogram(x,512,256,512,fs,'yaxis');
% This spectrogram shows much more defined and clear lines for the
% frequencies (much less noise) for each sinusoid

%%2.2(e)
figure(6);
[ note tt ] = key_to_note (69 , 1 , 0.25 , fs ) ;
plot ( tt , note ) ;

hold on ;
plot ( tt , adsr ( note ) , 'r -') ;
hold off ;

x = build_song(keys,Xs,start_times,end_times,fs);
audiowrite('mary_trumpet_adsr.wav',x,8000);
figure(7);
spectrogram(x,512,256,512,fs,'yaxis');
% The ADSR makes the notes fade in and fade out. This makes the
 transitions
% between notes less abrupt and helps the song sound smoother. This is
% shown in the spectrogram by the fading power of each signal near its
% start and end times.


%%ALL FUNCTIONS SUPPORTING THIS CODE

type key_to_note
type build_song
type build_song_wo_adsr
type key_to_musical_note
type adsr.m


function [ x,t ] = key_to_note ( key , X , dur , fs )
```

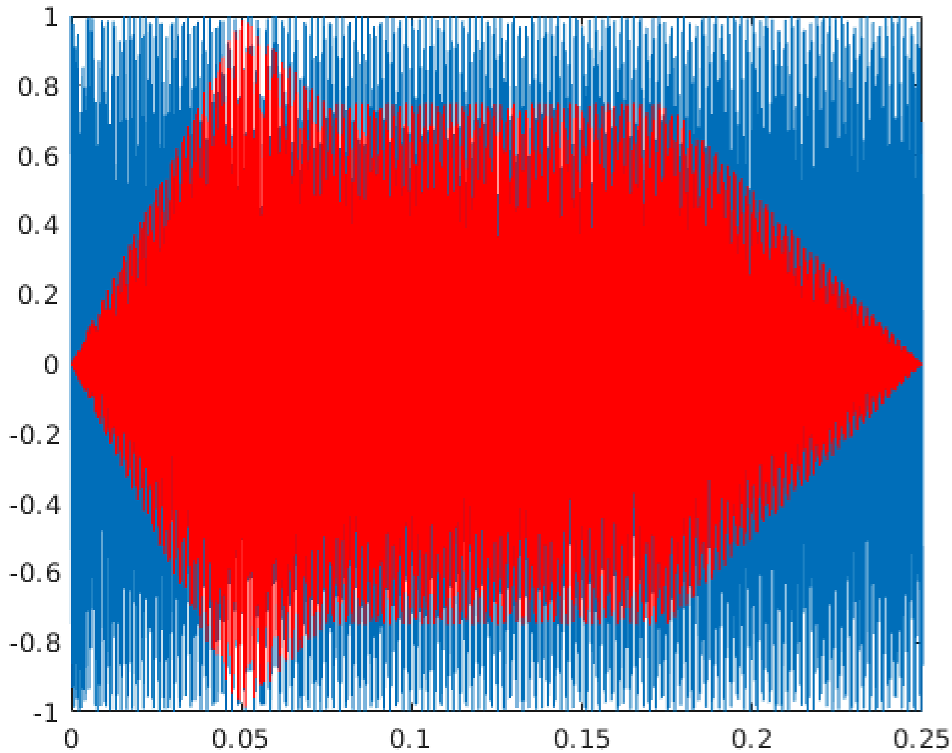
```
% key_to_note : Produces a sinusoidal waveform corresponding to a
%given piano key number
% Input Args :
%   key : number of the note ( key ) on piano keyboard
%    X : phasor of sinusoid
%   dur : duration of note ( in seconds )
%   fs : A scalar indicating the sampling rate ( in Hz )
%
% Output :
%   x : sinusoidal waveform of the note
%   t : optional time vector

t = 0:1/ fs : dur ; % Time vector
f = 440 * 2^(key -69/12); % <========== complete this line
x = real (X* exp(j*2 *pi*f*t)); % <========== complete this line
end


function x = build_song ( keys , Xs , start_times , end_times , fs )
% build_song : This function takes in the input parameters used to
 describe
% a song, and outputs a sampled vector representing the signal of the
 song
%
% Input Args :
%   keys : A length - N vector of key / note numbers , where
%   N = number of notes in song
%   Xs : A length - N vector of phasors
%   start_times : A length - N vector of start times of notes
%                   ( in seconds )
%   end_times : A length - N vector of end times of notes
%               ( in seconds )
%   fs : A scalar indicating the sampling rate ( in Hz )
%
% Output :
%   x : A vector that holds the signal samples of the song built

len_in_samples = ceil(max(end_times)*fs) + 1; % the ceil function
 ensures a
% whole number is returned to define the size;
x = zeros(1,len_in_samples);

for i = 1: (length(keys))
    note = key_to_musical_note(keys(i), Xs, end_times(i) -
 start_times(i), fs);
    note = adsr(note);
    start_in_samples = round(start_times(i) * fs) +1;
    end_in_samples = start_in_samples + length(note) -1;
    x(start_in_samples:end_in_samples) =
 x(start_in_samples:end_in_samples) + note;
end
end
```

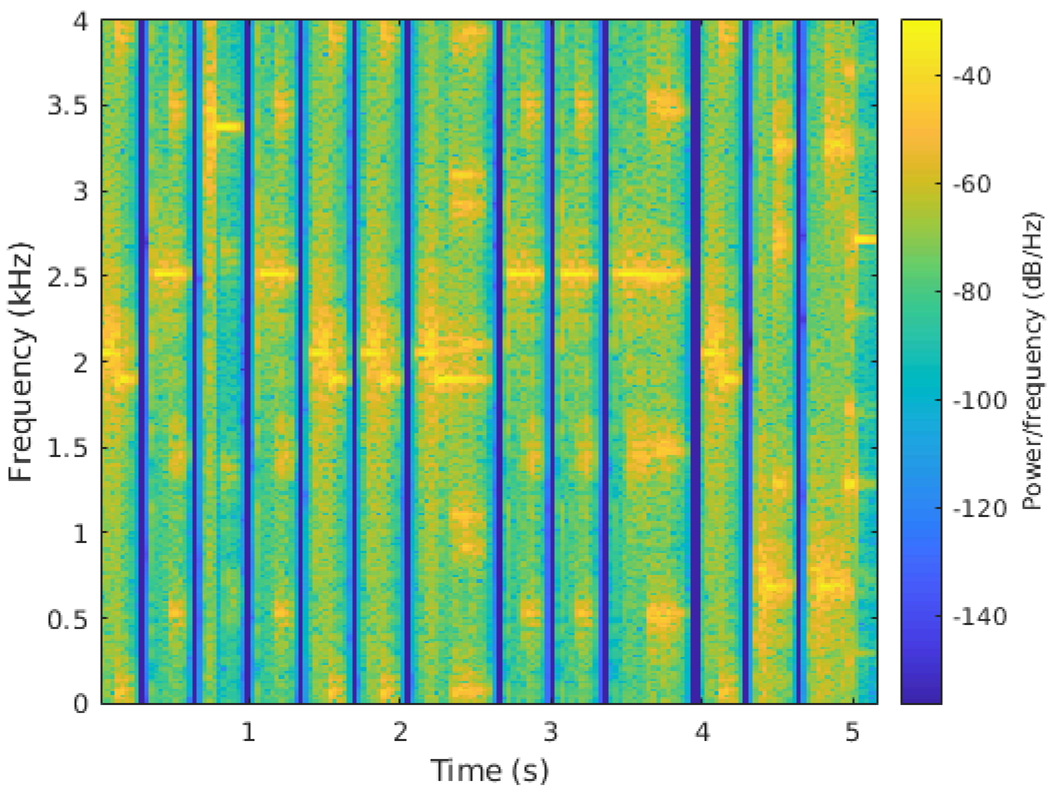
```
function x = build_song_wo_adsr ( keys , Xs , start_times ,
 end_times , fs )
% build_song : This function takes in the input parameters used to
 describe
% a song, and outputs a sampled vector representing the signal of the
 song
%
% Input Args :
%   keys : A length - N vector of key / note numbers , where
%   N = number of notes in song
%   Xs : A length - N vector of phasors
%   start_times : A length - N vector of start times of notes
%                  ( in seconds )
%   end_times : A length - N vector of end times of notes
%              ( in seconds )
%   fs : A scalar indicating the sampling rate ( in Hz )
%
% Output :
%   x : A vector that holds the signal samples of the song built

len_in_samples = ceil(max(end_times)*fs) + 1; % the ceil function
 ensures a
% whole number is returned to define the size;
x = zeros(1,len_in_samples);

for i = 1: (length(keys))
    note = key_to_musical_note(keys(i), Xs, end_times(i) -
 start_times(i), fs);
    start_in_samples = round(start_times(i) * fs) +1;
    end_in_samples = start_in_samples + length(note) -1;
    x(start_in_samples:end_in_samples) =
 x(start_in_samples:end_in_samples) + note;
end
end



function [x , t ] = key_to_musical_note( key , Xs , dur , fs)
% key_ t o _m u s ic a l _n o t e : Produces a musical node
 corresponding
% to a given piano key number
%
% Input Args :
%   key : number of the note ( key ) on piano keyboard
%   Xs : A vector contains the phasors of harmonics starting
%        from the 1 st harmonic
%   dur : duration of note ( in seconds )
%   fs : A scalar indicating the sampling rate ( in Hz )
%
% Output :
%   x : waveform of the note
%   t : optional time vector
```

```
t = 0:1/ fs : dur ; % Time vector
f = 440 * 2^(key -69/12);%
x = zeros (1 , length ( t ) ) ; % create 0s vector for the sampled
 signal
num_harmonics = length ( Xs ) ; % number of harmonics
% instantiates the signal x with the sum of the harmonics
for n = 1: num_harmonics
x = x + real (Xs(n) * exp(j * 2 * pi * f * t));
end
end

function env_note = adsr ( note )
% This function smoothes out the transitions between the notes by
 adding
% decays and attack to subsequent notes

% Below are the parameters that specify the ADSR envelope
Pattack = .2; % Length of attack ( proportion )
Pdecay = .1; % Length of decay ( proportion )
Prelease = .3; % Length of release ( proportion )
Vattack = 1; % Attack maximum value
Vsustain = 0.75; % Sustain value

L = length ( note ) ; % Length of note signal
Lattack = floor ( L * Pattack ) ;% Length of attack
Ldecay = floor ( L * Pdecay ) ; % Length of decay
Lrelease = floor ( L * Prelease ) ; % Legnth of release
Lsustain = L - Lattack - Ldecay - Lrelease ; % length of sustain


% Generate the ADSR portions of the time weights
attack = linspace (0 , Vattack , Lattack ) ;
decay = linspace ( Vattack , Vsustain , Ldecay ) ;
sustain = linspace ( Vsustain , Vsustain , Lsustain ) ;
release = linspace ( Vsustain , 0 , Lrelease ) ;
% Concatenate to get time weight vector
weight = [ attack , decay , sustain , release ];
env_note = weight .* note; % Apply ADSR envelop to note
end
```


4
3.5
3
2.5
2
1.5
1
0.5
0
Frequency (kHz)
1
2
3
4
5
Time (s)
0
-50
-100
-150
Power/frequency (dB/Hz)


4
3.5
3
2.5
2
1.5
1
0.5
0
Frequency (kHz)
1
2
3
4
5
Time (s)
-40
-60
-80
-100
-120
-140
Power/frequency (dB/Hz)


1
0.8
0.6
0.4
0.2
0
-0.2
-0.4
-0.6
-0.8
-1
0
0.05
0.1
0.15
0.2
0.25


Frequency (kHz)
4
3.5
3
2.5
2
1.5
1
0.5
0
1
2
3
4
5
Time (s)
-40
-60
-80
-100
-120
-140
Power/frequency (dB/Hz)

Published with MATLAB® R2020a