Lab 5 – Sounds and Music

Objectives: In this lab we will

- Synthesize and listen to some familiar sounds,
- Develop a simple music synthesizer in Matlab.

5.1 Familiar sounds (function + script)

Write a function xt = SumOfSines(A, F, P, td, fs) that generates samples of the signal $x(t)$

$$x(t) = \sum_{k=1}^{N} a_k \sin\left(2\pi f_k t + \phi_k\right)$$

where,

- A – vector of length N consisting of amplitudes a_k
- F – vector of length N consisting of the frequencies f_k
- P – vector of length N consisting of the phases ϕ_k
- td – duration of the generated signal $x(t)$ in seconds
- fs – sampling frequency for generating samples
- xt – the output signal consisting of samples of $x(t)$.

>> Note: for faster code avoid writing for-loop over time variable.

>> Using the above function, write a script for generating the signals below. For each sinusoid below use an amplitude of 0.5 and phase of 0, set the sampling frequency to $f_s = 10\ kHz$, and use the matlab command sound() to listen to the generated signal.

(a) Generate a signal $x_1[n]$ that is sum of two sinusoids with frequencies 350 Hz and 440 Hz in the time interval [0, 4] seconds. Listen to it.

(b) Generate a signal $x_2[n]$ which is composed of **alternating copies** of signals b1 and z1 (of 0.5 second duration each) **repeated four times** where: b1 is the sum of two sinusoids with frequencies 480 Hz and 620 Hz, and z1 is an all zero signal. In matlab, you can simply concatenate two row vectors b1 and z1 to get a longer row vector by writing [b1, z1]. Listen to the signal $x_2[n]$**.**

(c) Generate a signal $x_3[n]$ which is composed of alternating copies of signals b2 and z2 (of 2 second duration each) **repeated four times** where: b2 is the sum of two sinusoids with frequencies 440 Hz and 480 Hz, and z2 is an all zero signal. Listen to it.

(d) Do the above signals sound familiar?

(e) Plot these three signals in a 3x1 figure (plot only the first 500 samples) and observe how they look (use plot() command).

5.2 Creating a signal with harmonics (function + script)

Many musical instruments' sounds are well-modelled as the sum of harmonically related sinusoids. Hence each "note" played by a musical instrument (for ex. note of a piano) can be approximated as

$$x(t) = \sum_{k=1}^{N} a_k \sin\left(2\pi k f_0 t + \phi_k\right)$$

where f_0 is the fundamental frequency of the note and there are N harmonics.

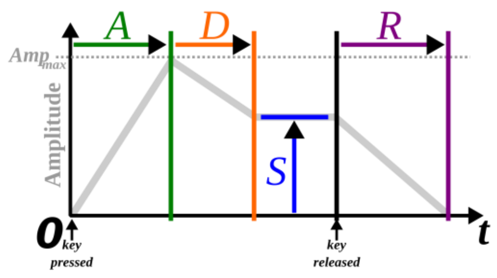
>> Write a function xt = harmonics(A, f0, P, td, fs) which internally calls the function SumOfSines() from part **5.1** to generate a "note" of frequency f0. The remaining parameters are as before.

>> Note: as **precaution** set your computer volume to low before listening to the sounds.

>> Write a script to call the function harmonics() for various inputs and listen to the generated signal using the soundsc() command (this scales your signal before playing). You can set phase P to be all zeros, sampling frequency of $f_s = 10\ kHz$, and "note" duration to be td = 1 second. Try the following:

(a) N = 5, f0 = 50, $a_k = \frac{1}{k}$
(b) N = 5, f0 = 50, $a_k = \frac{1}{k^2}$
(c) Repeat above with N = {10, 15} and f0 = {100, 150, 200}
(d) Repeat above with other amplitude variations such as $a_k = \sin\left(\frac{\pi k}{N}\right)$, $a_k = \cos\left(\frac{\pi k}{N}\right)$, $a_k = k$ and any other function a_k of your choice
(e) Plot the signal (first 500 samples) for different a_k and observe
(f) Repeat (a) and (b) with random phase, P = 2*pi*rand(1,N), note your observations.

5.3 Creating a signal envelope (function + script)



ADSR curve

The ADSR (attack, decay, sustain, release) envelope is used in music synthesizers to model how the amplitude of a note changes over time, see figure above (and read in Wikipedia).

>> Write a matlab function `[t_env,env] = envelope(a,d,s,sd,r,fs)`, that takes inputs

- a – attack duration in seconds

- d – decay duration in seconds
- s – sustain level in [0,1]
- sd – sustain duration in seconds
- r – release duration in seconds
- fs – sampling frequency in Hz

and returns

- t_env – time vector sampled at fs Hz of length a+d+sd+r seconds
- env – the corresponding ADSR envelope.

You can assume that the durations a, d, sd, and r, will be integer multiples of sampling interval (1/fs). See comments in the solution template for how to interpret the parameters. Copy paste the code template given and complete it.

```
function [t_env,env] = envelope(a,d,s,sd,r,fs)
% For each portion of the note, determine the corresponding piece of
time vector and envelope.
% Attack: amplitude linearly increases from 0 to 1 in ‘a’ seconds
tattack = 0:1/fs:a;
env = ...
t_env = tattack;

% Decay: amplitude linearly decreases from 1 to ‘s’ in ‘d’ seconds
tdecay = (a+1/fs):1/fs:a+d;
t_env = [t_env, tdecay];
env = [env, ...];

% Sustain: amplitude stays at ‘s’ for ‘sd’ seconds
tsustain = ...
t_env = [t_env, tsustain];
env = [env, ...];

% Release: amplitude linearly decreases from ‘s’ to 0 in ‘r’ seconds
trelease = ...
t_env = [t_env, trelease];
env = [env, ...];
end
```

>> Make sure that when a+d+sd+r = 1, your code returns t_env & env of length fs.

>> Write a script to listen to the notes with and without ADSR envelop. Compare using soundsc(xt, fs) & soundsc(xt.*env, fs) to hear the effects of applying the envelope:

- use fs = 10000
- for xt use one of the notes generated in part **5.2** i.e. xt = harmonics()
- env is obtained by calling the function envelope(0.2,0.2,0.7,0.4,0.2,fs)
- note that you should have td = a+d+sd+r for this to work
- change envelope by changing values of `a,d,s,sd,r` and listen again.

>> In a 3x1 figure, plot xt, env & xt.*env in the three panels. Repeat this for two different envelopes.

5.4 A simple music synthesizer (function + script)

>> Now you can combine the results of parts **5.2** and **5.3** to create a simple music synthesizer function which constructs M notes in the specified sequence (a tune).
Your synthesizer function should take the following inputs

- A – vector of length N consisting of amplitudes a_k (assumed same for all notes)
- F_notes – length M vector of note fundamental frequencies in Hz
- P – vector of length N consisting of the phases ϕ_k (assumed same for all notes)
- adsr – length 5 vector of (`a,d,s,sd,r`) (assumed same for all notes)
- td_notes – length M vector of note durations in seconds
- fs – sampling frequency in Hz

>> Since each note can be of different duration, we assume that in the input a+d+sd+r = 1 and for each note we accordingly scale them so that the envelop is of required duration.

>> Your function should produce an output signal y, so that sound(y, fs) produces the specified sequence of notes. Edit the following code template to get the output.

```
function y = synthesizer(A,F_notes,P,adsr,td_notes,fs)
% Initialize output as empty
y = [];
% Loop over the notes
for ii = 1:length(F_notes)
    % scale a,d,sd,r so that they sum to required note duration
    ...

    % Compute the time vector and ADSR envelope for this note
    [t,env] = envelope(...);

    % Compute the sum of harmonics for this note
    xt = harmonics(...);

    % Modulate the sum of harmonics with the envelope
    xte = ...;

    % Add the note to the sequence
    y = [y,xte];
end
end
```

>> Write a scrip which calls the synthesizer() function with various inputs and listen to the synthesized signal using soundsc(). You can set N = 5, amplitudes $a_k = \frac{1}{k^2}$, phase P to be all zeros, sampling frequency of $f_s = 10\ kHz$. Try the following:

(a) F_notes = 50:5:100, all notes of same duration (1 second)
(b) F_notes = 100:-10:40, all notes of same duration (1 second)
(c) Random tune: M = 5; F_notes = 50 + 50*rand(1,M); td_notes = 0.5 + rand(1,M);
(d) Use different a_k, F_notes and td_notes till you can synthesize/hear a unique tune. Try and see if you can generate your favourite tune!
(e) Use the command audiowrite() to save a tune (> 10 seconds) as wav file and submit.