Music 320: Lab 2: Synthesis with sinusoids.

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1 Sinusoids

Try Sinedrill test to test your skills in recognising amplitude, frequency and phase of a sinusoid.

2 Additive synthesis

- Intuitions: decomposition of a waveform as a sum of sinusoids.

Constructing a square wave from harmonic sinusoids.

```
n=[0:1/44100:1];

sine1=sin(2*pi*10.*n);
sine2=0.5*sin(2*pi*30.*n);
sine3=0.25*sin(2*pi*50.*n);
sine4=0.125*sin(2*pi*70.*n);
sine5=0.0625*sin(2*pi*90.*n);

figure(1)
subplot(5,1,1);
plot(n,sine1)
subplot(5,1,2);
plot(n,sine1+sine2);
subplot(5,1,3);
plot(n,sine1+sine2+sine3);
subplot(5,1,4);
plot(n,sine1+sine2+sine3+sine4);
subplot(5,1,5);
plot(n,sine1+sine2+sine3+sine4+sine5);
```

2.1 Creating more interesting sounds using envelopes

Using additive synthesis it is pretty straightforward to reproduce the steady state portion of most sounds.

The attack portion and the decay, however, are usually more complicated to detect and model. In computer music it is typical to use the so-called ADSR envelopes (which stands for attack, decay sustain release). Figure 1 shows a typical ADSR envelope.

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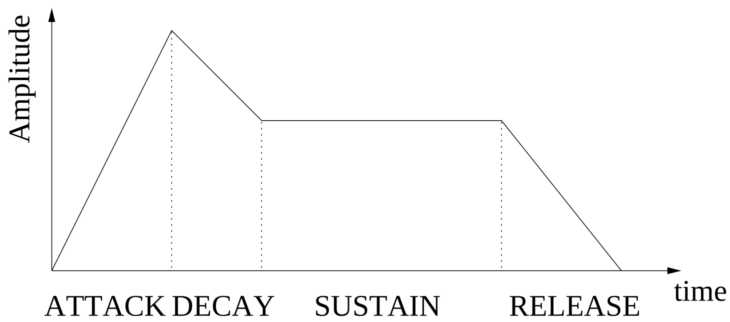

Figure 1: A typical ADSR envelope.

2.2 Sinewave speech

Sinewave speech is an experimental technique that tries to simulate speech with just a few sinewaves, in a kind of primitive additive synthesis.

Try the tools here: http://www.ee.columbia.edu/∼dpwe/resources/matlab/sws/

A problem of additive synthesis is that it is very computationally expensive. The following synthesis techniques are much cheaper.

3 Modulation synthesis

Modulation means that one aspect of a signal (the carrier) varies according to an aspect of another signal (the modulator).

When the frequency of modulation is bigger than 20 Hz audible modulation products of sidebands begin to appear.

3.1 Ring modulation

Ring modulation multiplies one signal by another. Usually the second signal is a cosine wave. Ring modulation produces frequencies which are not in a harmonic relationship to the original tone.

It is a form of amplitude modulation.

$$RingMod_t = C_t M_t$$

3.2 Amplitude modulation

Check the following matlab example of modulation:

```
fs = 22050;                    % sampling rate
Ts = 1/fs;                     % sampling period
Tmax = 2.0;                    % signal duration
t = [0:Ts:Tmax];               % time vector

x = 0.4*cos(2*pi*198*t) + 0.4*cos(2*pi*202*t);

disp('Sum of two sinusoids');

plot(t,x);
axis([0 0.5 -1 1])
xlabel('Time (seconds)');
ylabel('x');
```

```
soundsc(x,fs)

disp('paused ... multiplication of two sinusoids next');
pause

x = 0.8*cos(2*pi*200*t).*cos(2*pi*2*t);

plot(t,x);
axis([0 0.5 -1 1])
xlabel('Time (seconds)');
ylabel('x');

soundsc(x,fs)

disp('paused ... AM next');
pause

x = 0.1*(5 + 2*cos(2*pi*30*t)).*cos(2*pi*200*t);

plot(t,x);
axis([0 0.5 -1 1])
xlabel('Time (seconds)');
ylabel('x');

soundsc(x,fs)
```

3.3 Example of tremolo

```
function y=tremolo(x, amp, f, mix, Fs)
%y=tremolo(x, amp, f, Sf)
%Tremolo and ring modulation effect, sinusoidal
%x      - input
%amp    - the amount of constant versus variable amplitude,
%         0=only variable, 1=constant amplitude, -1=ring modulator
%f      - the tremolo frequency
%mix    - mix of modulated and original sound, mix=1 - only modulated
%Sf     - sampling frequency


step=1:length(x);
y(step)=(amp+(1-amp)/2*(1-cos(2*pi*(f/Fs)*step))).*x(step)';
y=mix*y'+(1-mix)*x;
```

A signal is multiplied by a sinusoidal curve.
Test it with a sinusoids:

```
Fs = 44100;
s =sin(2*pi*100*(1:2^18)/Fs)';
amp=0.4;
f = 2;
mix = 1;
```

```
out = tremolo(s,amp,f,mix,Fs);
sound(out,fs);
```

3.4 Frequency modulation

- CCRMA history: Chowning and the DX7
- computational economy relative to additive synthesis
- perceptual economy: limited set of strong parameters controlling spectral content

(from DSP first)

Given a signal $x(t) = A\sin(\psi(t))$ FM, meaning "frequency modulation," refers to the fact that the frequency of a signal $x(t)$ changes according to the oscillations of $\psi(t)$. This is useful for synthesizing instrument sounds because the proper choice of the modulating frequencies will produce a fundamental frequency and several overtones, as many instruments do.

The general equation for an FM sound synthesizer is:

$$x(t) = A(t)\sin[2\pi f_c t + I_0 \sin(2\pi f_m t + \phi_m) + \phi_c]$$

In this equation, $A(t)$ is the signal's amplitude. It is a function of time so that the instrument sound can be made to fade out slowly or cut off quickly. Such a function is called an envelope. The frequency $\omega_c = 2\pi f_c$ is called the "carrier" frequency. Note that when you take the derivative of $\psi(t)$ to find $\omega_i(t)$, ω_c will be a constant in that expression. It is the frequency that would be produced without any frequency modulation. The parameter ω_m is called the "modulating" frequency. It expresses the rate of oscillation of $\omega_i(t)$. The parameters ϕ_m and ϕ_c are arbitrary phase constants. The function $I(t)$ is called the "modulation index envelope."

For more details see the paper by Chowning: John M. Chowning, "The Synthesis of Complex Audio Spectra by means of Frequency Modulation," Journal of the Audio Engineering Society, vol. 21, no. 7, Sept. 1973, pp. 526–534.

Here are some examples of sounds that can be synthesized with the appropriate choice of the parameters.

- brass
- clarinet
- bell
- knocking

These sounds were originally synthesized by Robbie Griffin.