Computer music – Input Team

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Our project:

- Record either from microphone or a file above a certain db threshold
- Extracting sound envelopes and doing Fourier analysis in given frequency bands

Preview

Definitions for

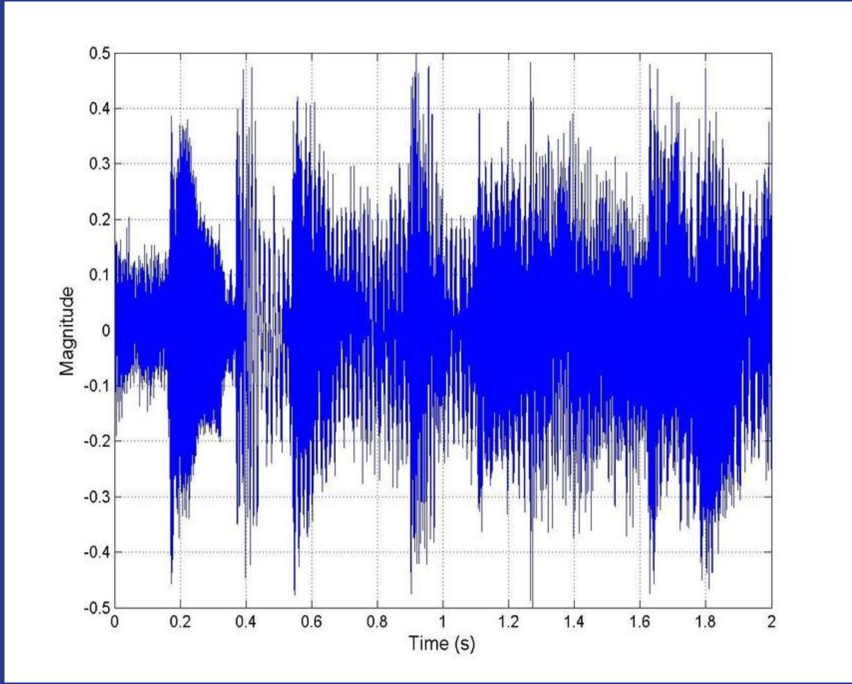
- Db Threshold
- Sound envelopes

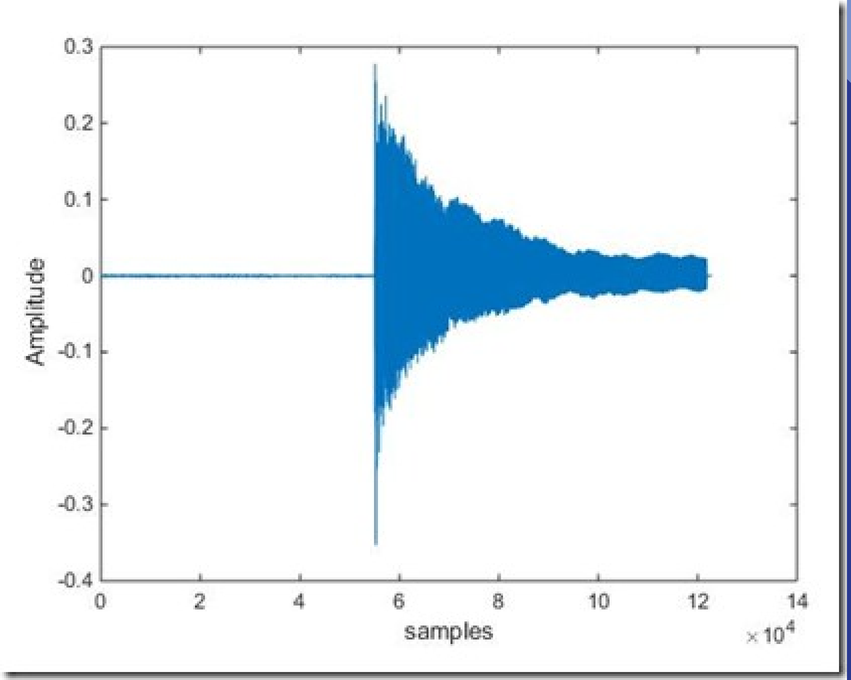
And the algorithms we followed

Obstacles we faced

- No previous work for extracting sound envelopes
  Attack, Decay, Sustain, Release (adsr)
- Not clear definitions for each of those envelopes
- The input signal can be very abnormal and complicated

Some signals





Db Threshold

- In order to capture processable sounds we used a db Threshold

  This means that the final recorded .wav files will be above a certain db level

- When recording from microphone the threshold is altered live by taking the standard deviation of the current amplitude

Db Threshold

- When recording from microphone the threshold is altered live by taking the standard deviation of the current amplitude

By doing this the Threshold changes in a way that we capture only the “special events” that happen in each location

Sound Envelopes (avg_amp)

- Average amplitude
  The average of magnitude of all samples in the signal

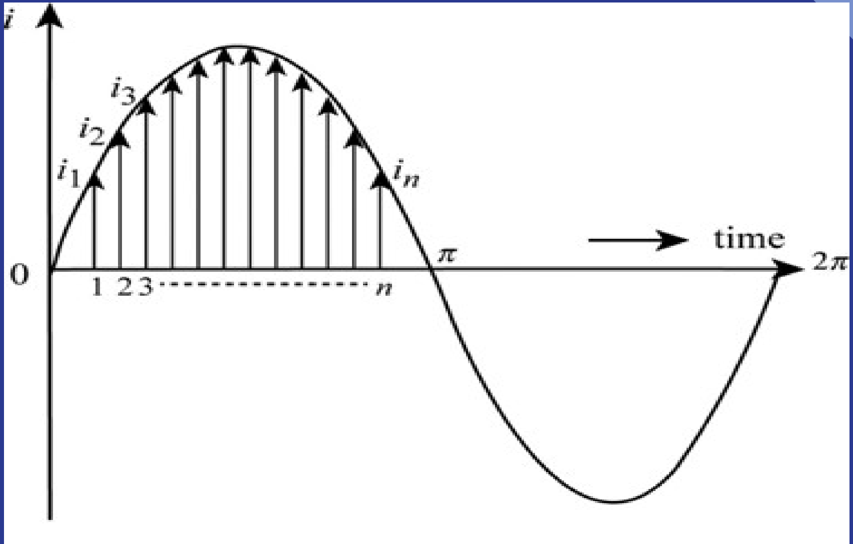


Figure 1

Sound Envelopes (A.D.S.R)

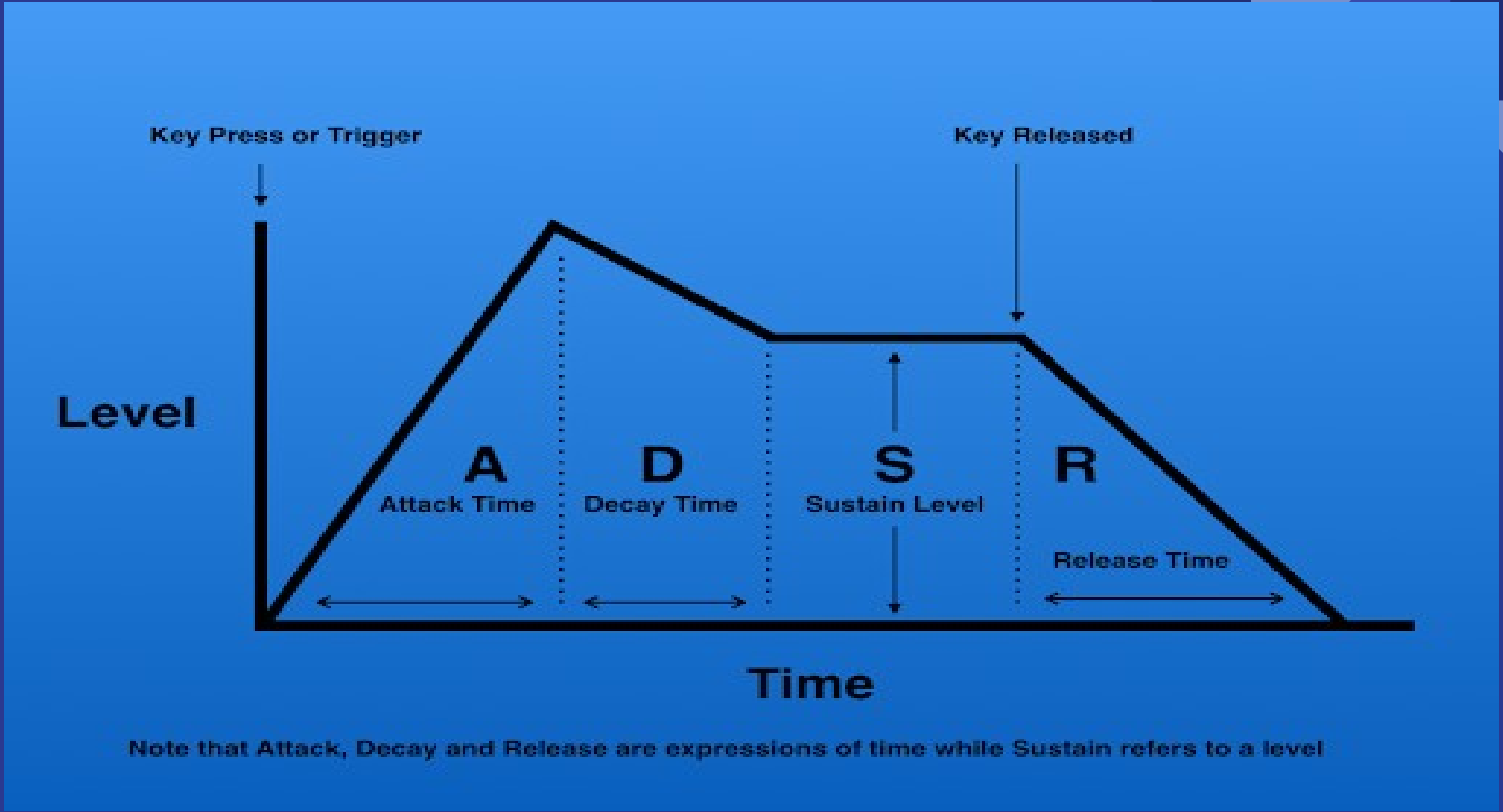
- Attack

  The time it takes for the signal to reach its max amplitude
- Decay

  The time it takes after attack for the amplitude to stop decreasing
- Sustain

  The level of amplitude the sound will stay before it begins fading out
- Release

  The amount of time it takes for the sound to fade out


Key Press or Trigger
Key Released
Level
A
Attack Time
D
Decay Time
S
Sustain Level
R
Release Time
Time
Note that Attack, Decay and Release are expressions of time while Sustain refers to a level

Sound Envelopes (A.D.S.R)

- Attack

  check the amplitude sample by sample until we find max

- Decay

  check when the amplitude stops decreasing

Sound Envelopes (A.D.S.R)

- Sustain
  Instead of the sustain level we check for the sustain time
  The search stops when the difference of the amplitude between two successive samples will be below a certain constant

- Release
  Release time = total_wav_time - (attack + decay + sustain)

Sound Envelopes(avg_amp)

- Average amplitude

  Average amplitude = (sum_of_individual_amplitudes) / (number_of_samples)

Concluding

The characteristics we extracted will be used by other sub-projects that include

- Machine Learning
- Visual Effects
- Midi - Sequencer

Future thoughts

- Implementing these techniques in constructing better music instruments
- Finding the music that fits in each environment

Thank you for your attention!