SYN semesteral project

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In this document I will briefly describe used techniques and created instruments. All samples used in this project were downloaded from webpage https://freewavesamples.com/ and can be found in samples directory. Some samples in this directory contain only clipped pitch period, but can be still retrieved from this webpage under the same name. The only sample I recorded myself is sample My-Lovely-Yamaha-C4. In the following sections I will describe the used techinques in this work. In the next section I will specify for each instrument, which concrete techniques were used to synthesise the instrument.

1 Techniques

1.1 Wavetable Synthesis

To create instruments with wavetable synthesis, I extracted one pitch period of signal at zero crossings in Audacity, normalized it and used provided tabsynth matlab function. The function took samples from the extracted pitch period, to match required frequency and repeated those signals, to match required duration.

1.2 Additive Synthesis

Instruments created with additive synthesis were at first analyzed using Fourier transformation. More precisely, I downloaded a sample of real instrument, analyzed its frequency spectrum and retrieved coefficients of k = 1, ..., n frequencies. After that I refined the selection of frequencies, by adjusting the lower and upper value of the frequency array. Finally I synthesised the final instrument by summing sine waves for k = l, ..., u; but now with the required frequency, duration and sampling rate. All of the sine waves, were multiplied by an exponential envelope with some amplitude and tau values.

1.3 LPC synthesis

LPC synthesis was the one and only tool for synthesising percussion sounds. I used the built-in matlab lpc function to find linear predictor coefficients. These coefficients were then used as a transfer function to filter an input signal of choice. In these project I used dirac and exponentially damped white noise as inputs.

1.4 Formant Synthesis

Formant synthesis is based on modelling sounds and voices based on physical structure of instrument. Using some base frequency, which resonates up to 4 filters in cascade, one can produce a huge variety of sounds. Such as the aaah voice or saxophone.

1.5 Cross Synthesis

One can combine the residual prediction error of a voice and LPC analysis of an instrument using a windowing function to obtain a voice overlayed by an instrument. I used this technique especially to create the choir aaah voicing.

1.6 Effects

Applying filters to created instrument can create more interesting sounds by equalizing or delaying the signal in various ways, to add more full feel of an instrument played in a room. In this section I will briefly describe types of filters used in this project.

- lowpass filter - This type of filter can filter out high frequencies in order to create a deeper, bassy sound.
- reverb - To create reverb I used provided coefficients for FIR and IIR filters. In general I found out, that IIR filter produces a deeper feel, meanwhile FIR is more clean.
- distortion - Distortion effect was created by adding a sine wave to the signal, where it's absolute value is bigger then some constant value.
- chorus - Chorus effect was achieved by repeating the signal, but with some small delay in the start of the signal.
- vibrato - Vibrato effect was achieved by adding a small variation of each harmonic frequency in time. The frequency of vibrato effect and its impact were tunable parameters for each instrument.

1.7 Envelopes and Normalizing

After using one of the described synthesis, I applied ADSR, exponential or both envelopes to produce sound that reproduces natural instrument. ADSR or attack-delay-sustain-release envelope defines 5 points and interpolates them, in order to modify amplitude of the produced signal. Exponential envelope works in the same manner, but instead of specifying 5 points, the signal is just multiplied by the exponential with peak y=1 at x=0 and tau, that specifies the damping of the signal. The smaller the tau value, the faster the signal damps. After applying envelope to the signal, it is multiplied by the amplitude and normalized, so that all instruments are at the same level when combined together.

2 Instruments

- n01 acoustic grand piano - wavetable synthesis + reverb + exp + adsr
- n03 electric grand piano - sawtooth wave + lowpass fir + exp + adsr
- n21 reed organ - wavetable synthesis + reverb + exp + adsr
- n31 distortion guitar - sine wave + first 1/10 replaced with white noise + vibrato + lowpass iir + adsr + exp
- n34 electric bass - wavetable synthesis + distortion + reverb + exp + adsr
- n48 timpani - dirac - LPC synthesis (2 samples for each tone) + exp
- n49 string ensemble 1 - wavetable synthesis + distortion + reverb + exp + adsr
- n53 aaah - formant synthesis + cross synthesis + chorus + reverb + exp + adsr
- n56 orchestra hit - wavetable synthesis (violin) + adsr + LPC synthesis (electric tom) + volume mastering
- n61 french horn - additive synthesis
- n63 synth brass - square wave + lowpass fir + exp + adsr
- n66 alto sax - sine wave + vibrato + formant synthesis + adsr
- n70 english horn - additive synthesis
- p35 bass drum - dirac + LPC synthesis + interpolation for short
- p40 snare drum - exponentially damped white noise + LPC synthesis + interpolation for short
- p41 electric tom - dirac + LPC synthesis + interpolation for short
- p42 hi hat closed - exponentially damped white noise + LPC synthesis + interpolation for short
- p43 floor tom - dirac + LPC synthesis + interpolation for short
- p45 & p47 low tom - dirac + LPC synthesis + interpolation for short
- p46 hi hat open - exponentially damped white noise + LPC synthesis

- p49 & p57 crash - exponentially damped white noise + LPC synthesis
- p51 ride - exponentially damped white noise + LPC synthesis + interpolation for short
- p52 & p55 chinesse cymbal - exponentially damped white noise + LPC synthesis + interpolation for short
- p81 triangle - exponentially damped white noise + LPC synthesis