

MarjiWave

Final major project

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What is Marjiwave and what does it do?

Marjiwave is a combination of 3 modules (subtractive synthesizer, blocks technique device and resonant body simulator), which can be used for compositional and sound design purposes. Below you can find a brief explanation of the features and qualities of each device.

Subtractive synth:

It is a polyphony synth that is driven by two oscillators able to produce classic waveforms (sawtooth, triangle, square) along with white noise. The output of these oscillators can be mixed together with the option of detuning in order to produce beats. The frequency and velocity at which the sound is generated can be controlled using the internal keyboard, the step sequencer or an additional external keyboard. By changing the duty cycle the square waveform can be altered to produce different harmonic spectra. The hard sync button can also be used to synchronize the cycle of two oscillators even when they are detuned and have different frequencies. The produced sound is then processed with an amplitude envelope that outlines its attack, decay, sustain and release. After that the signal is sent to a lowpass filter whose cutoff frequency can be fixed or can be varying dependent on either the note played or the amplitude envelope using the “key follow” and “env depth” parameter. The filter has also a Q parameter that specifies the resonant factor for the cutoff frequency. Moreover, there is a high pass filter with a gentle slope of 6 dB per octave with an assignable cutoff frequency that is not affected by the note or amp envelope. Four parameters in this synth (amplitude, pitch, cutoff frequency and Q) can be controlled using “Low Frequency Oscillators” also known as LFOs with different waveforms (sine wave, square wave, sawtooth wave and random wave) whose depth and rate can be changed. For the square and sawtooth LFOs there are smooth dials that can be used to smooth the sudden jump present in the LFO signals. There is also a stereo panning modulation that can be used to vary the pan of the sound using sine wave and square wave LFOs with assignable rates.

When the step sequencer is off the duration of the sound can be assigned either in millisecond or in note values whose duration depends on the tempo. Turning the step sequencer on, the pitch, velocity and duration for each note can be assigned by the values produced on the step sequencer.

Blocks technique:

It is a compositional system based on the control of reading points in a sampled sound. It works by fragmentation of a sound and reconstruction of the fragments based on four methods of reading a sampled sound.

- Original playback direction for the blocks & incrementing reading points
- Original playback direction for the blocks & decrementing reading points
- Reverse playback direction for the blocks & incrementing reading points
- Reverse playback direction for the blocks & decrementing reading points

The duration of the fragments (blocks) can be between a tenth of a second and three seconds. The sampled file can be performed based on a score indicating which blocks to play, in which direction and order and also when to play them. These features can all be controlled with four parameters as followed:

- Direction of playback for individual cues
- Metronome intervals
- Increment/decrement value of cues
- Duration of cues

Resonant bodies:

It is a synthesizer that can generate sounds, which are the simulation of the vibrations emitted by an object when it is bounced on a flat surface. This device makes use of a number of band pass resonant filters with very high Q values that are fed by impulses and as a result treat like resonant bodies. The two multi-sliders can control the frequencies present in the sound and the duration of the resonances for those frequencies. Changing the number of resonators, the distribution of frequencies in the first multi-slider and the duration of resonances in the second multi-slider, a lot of different timbres can be produced.

Getting started guide

First within you Max application please select the “File preferences” item in the options menu. Click on + in the lower left corner of the opened window and then press the choose button. After that you should find the “virtualsound macros” folder inside the FMP folder and select it.

Open the mixer by clicking the open mixer button, turn the audio output on and increase the volume for any of the devices that you want to work with.

Subtractive synthesizer:

Turn on the subtractive synth with the red button at the top left corner of the synth.

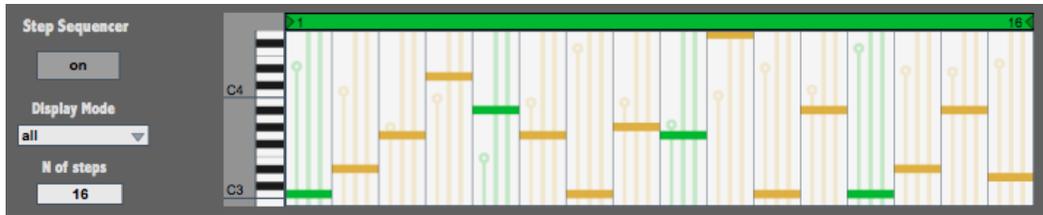
You can first turn on the step sequencer to easily monitor the sound of the synth. Then choose a waveform for each oscillator and mix them using the mix button. If one or both of the waveforms you have chosen are square wave you can use the duty cycle knob to produce different harmonic spectra. To modify the intonation of the oscillators you can detune them up to ∓ 12 semitones. It is also possible to set hard sync between the two oscillators.

In the filter section change the cutoff frequency and hear the effect it has. You can increase the key follow or env depth to associate the cutoff frequency to the note that is played or the amplitude.

In the amp envelop section, press the button of “clear envelop” and then draw a new envelope of your choice.

Turn on any of the LFOs you want and choose a waveform from the pop up menu. You can then increase the rate of the LFO using the knobs labeled the name of the waveforms. After that raise the depth sliders to hear the effect of the LFO on sound.

In the step sequencer, you can modify the pitch, velocity and the duration for the notes by first choosing an item from the menu to change the display mode and then using the mouse to modify the values in the display. The number of steps can be changed between 1 to 64 steps. Besides, dragging the ends of the loop ruler (the green strip at top of the display) can also change the number of steps.



Try different presets and see how the changes in parameters affect the sound. You can also save your own presets by holding shift and click one of the holes in the preset section.

Blocks technique device:

Import your own audio sample or you can also listen to the audio that has already been imported. Choose between the backward or forward direction for playback of the individual blocks. Set an interval time for the metronome, which means how often a new block of sound will be played. You can set the cue point value both in positive and negative. Positive value means the playback of the sample will start from the beginning of the audio and for negative value the playback will start from the end of the audio. The cue value indicates the point in which the sample is played back each time the metronome sends a bang. If for instance the cue point is set at 5ms, the playback starts from the beginning of the sampled file and after the duration of the metronome interval, a new playback starts at the 5 ms point of the audio file, next time at 10 ms and so on. You can then change the duration of each playback by either the number box labeled “duration of cue or the knob next to it. The ranges of the knobs can be changed using the number boxes at the right column of the device.

If you want to use an external MIDI controller to control the parameters, choose your device port from the pop up menu in the MIDI configuration section and click the open button to see the configurations available for the channel and controller number.

Try different available presets and then make some new presets of your own.

Resonant body simulator:

First turn on the device clicking the red button at the top left corner and try different available presets to see how the changes in multi sliders can generate completely new sounds. Make sure to turn on the button labeled preset playback to be able to hear the

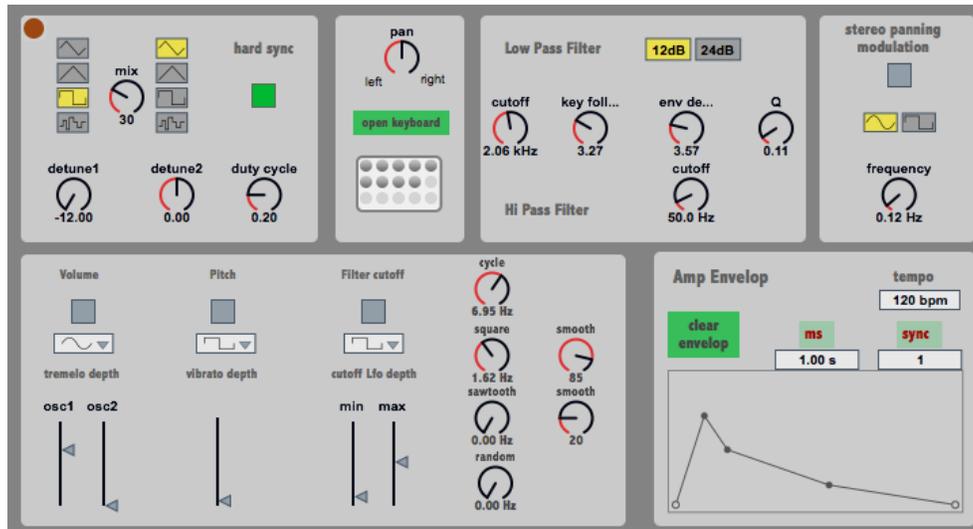
sounds while changing the presets. You can also change the ranges of notes and duration of resonances using the min and max knobs at the sides of the multi sliders. The number of steps value can be increased up to 40. This value indicates the number of sliders in multi sliders and as a result specifies the number of resonances in the sound.

The rebound start knob specifies the number of bounces at the beginning and this number will be increased to reach the rebound end value during the time that can be set using the duration knob.

Turn on the random generator and the bouncing sound will be generated in random intervals, with random initial bounce, random amplitude and random panning. You can then change the height of the multi slider, every time a new bounce is created.

More details about the modules of Marjiwave

Subtractive synthesizer:



A subtractive synthesizer is a sound synthesis technique that applies signals as source, which are rich in harmonics and filters them to attenuate some portions to alter the timbre of the sound. On this synthesizer two waveforms can be chosen with the option of mixing them together.

Among the waveforms, the square wave can be altered using the duty cycle knob that shifts the ratio between the pulse duration and the period of the wave. Changing the duty cycle, different harmonic spectra can be produced.

The oscillators generate their waveforms at the frequency of the notes being played. Detuning one or both the oscillators will shift their frequency up to ± 12 semitones. Choosing float numbers you can shift the frequency at intervals between semitones.

The hard sync means synchronization between the cycles of two waveforms. Even if the frequency of the waves is different, turning on the hard sync every time the first oscillator starts its cycle the second oscillator will be forced to restart disregarding its frequency.

There are two types of filter available one with the attenuation of 12 dB per octave and another with 24 dB per octave. The mixed sound of the oscillators will then pass

through the chosen filter whose cutoff frequency is derived from the sum of three parameters:

- 1- Fixed cutoff frequency
- 2- Key follow multiplied by the frequency of the note
- 3- The envelope multiplied by the env depth factor

The cutoff frequency can also be modulated via the filter LFO choosing a waveform, giving it a rate and increasing the depth. The depth of the LFO corresponds the amplitude of its wave and on this device it is dependent on the frequency of the played note and the cutoff frequency of the filter. The minimum and maximum of the depth can be adjusted on the filter LFO. Setting the min slider to its minimum amount and the max slider to its maximum amount has the highest effect on the cutoff frequency. Be aware that the effect is directly controlled by the frequency of the note so playing different notes will give different effects.

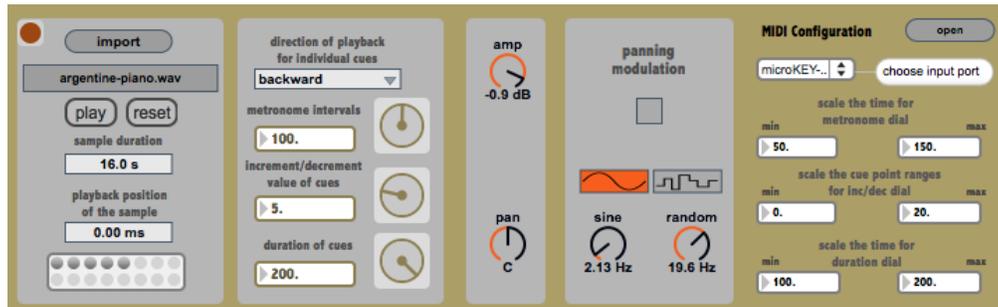
There are also two other LFOs available for controlling the amplitude and pitch of the sound. For amplitude you can have choose different amount of effect for the oscillator 1 waveform and oscillator 2 waveform, which can result in different interesting tones.

Since the square and sawtooth waveform have a sharp transition, using the smooth dials you can adjust how much you want to smooth this transition.

To play notes, there are three options:

- The step sequencer with which you can draw notes and specify the duration and velocity for each.
- The internal keyboard that can be opened pressing the open keyboard button
- The external MIDI controller keyboard

Blocks Technique device:



The block technique is a composite technique developed by Alessandro Cipriani using the PODX composition system. This technique is based on creating a loop from a sampled audio file whose starting point and duration can be incremented or decremented with each successive repetition. The sound file is divided into blocks and the duration of these blocks can be changed between a tenth of a second and three seconds. These portions of the file can be read in 4 different ways:

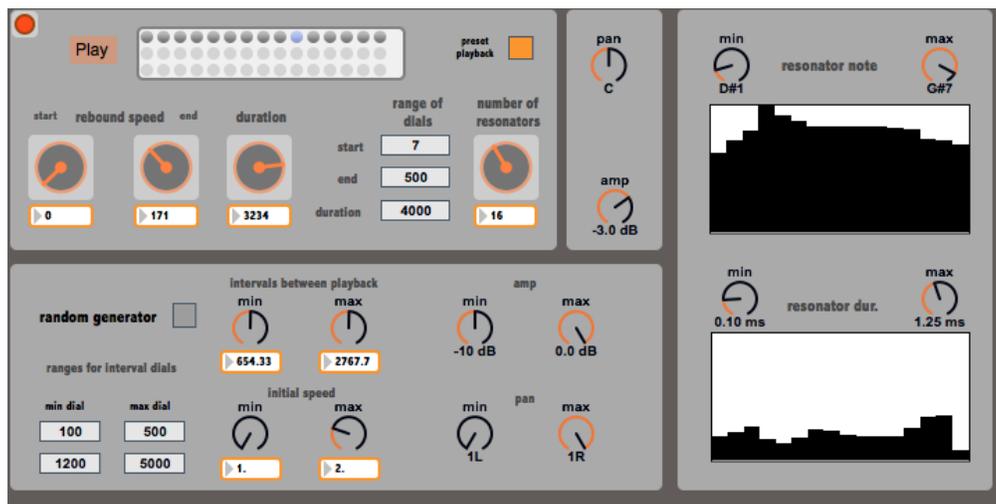
- The blocks are played in original direction and the cue point is incrementing meaning that the blocks will be played back from the beginning of the file and with each repetition the cue point will be incremented.
- The blocks are played in original direction and the cue point is decrementing meaning that the blocks will be played back from the end of the file and with each repetition the cue point will be decremented.
- The blocks are played in reverse direction and the cue point is incrementing meaning that the blocks will be played back from the beginning of the file and with each repetition the cue point will be incremented.
- The blocks are played in reverse direction and the cue point is decrementing meaning that the blocks will be played back from the end of the file and with each repetition the cue point will be decremented.

Exploiting this technique, you can explore the timbre of the sound by reconstructing the timbral variations in a single sound through the use of rhythm. The idea is to create complex sounds by overlapping and mixing a large number of grains from a sound file to

generate a completely new sound. This technique is very similar to the granulation technique. However, the grains in granular synthesis have much shorter duration.

The panning modulation section can be used to control the stereo position of the sounds through low frequency sine wave for rhythmic effect and low frequency random wave for a random effect.

Resonant body simulator:



This simulator is using up to 40 band pass resonant filters that are fed by impulse type signals. A waveform similar to a unipolar sawtooth wave has been used but to have the impact of the impulse, the wave rises sharply from 0 to 1 at the beginning and then ramps downward back to 0 (opposite the sawtooth wave).

For the purpose of simulating resonances of different objects, the unitary impulses could be fed to filters but here the sawtooth type signal has been chosen since compared to unitary impulses it has less energy in high frequency range and therefore, is more effective for resonances in the low frequency range.

In a resonant filter, the response time of the filter is directly related to its Q. Setting a very high Q can spoil the decay and attack of the sound specially the short sounds with rich spectra such as percussive sounds, which results in a noticeable lengthening, generating a resonant oscillation. Exploiting this effect, a bandpass filter can be transformed to a sound generator by feeding it a source that contains impulses with a

rich frequency spectrum. The duration of the resonances depends directly on the Q and inversely on the center frequency: The higher the Q value the longer the duration and also with a fixed Q, the higher the center frequency the lower the duration.

On this device, the Q values in bandpass filters are calculated by multiplying the center frequencies and the durations of the resonances in the multi sliders.

Changing the values in the multi sliders, you can simulate different resonant objects that are made of various materials such as metal, wood, glass and skin. For instance, the metallic sounds consist of medium-high frequencies with long resonance duration or sounds made of wood objects have medium frequencies with short resonances.

The rate of the impulses that feed the filters is controlled via an exponential curve that starts with the bound start value and accelerates to the bound end value over the duration that is set using the duration knob or number box.

Postscript: To build Marjiwave, the volume I and II of the book “Electronic and sound design theory and practice with MAX and MSP” has been my best source to learn the different methods of sound synthesis and signal processing.

Bibliography:

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