VA336 interactive sound

Week 4

Sound Synthesis

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Thessia Machado

Working in analogue and interested in sound’s physicality, Thessia Machado creates kinetic, sculptural, music-making machines for installations and performances. She once made an absolutely terrifying composition using a found Little Tikes toy, but her invented instruments also make use of more mature materials, like customized turntables, LCD screens, and >cassette decks.

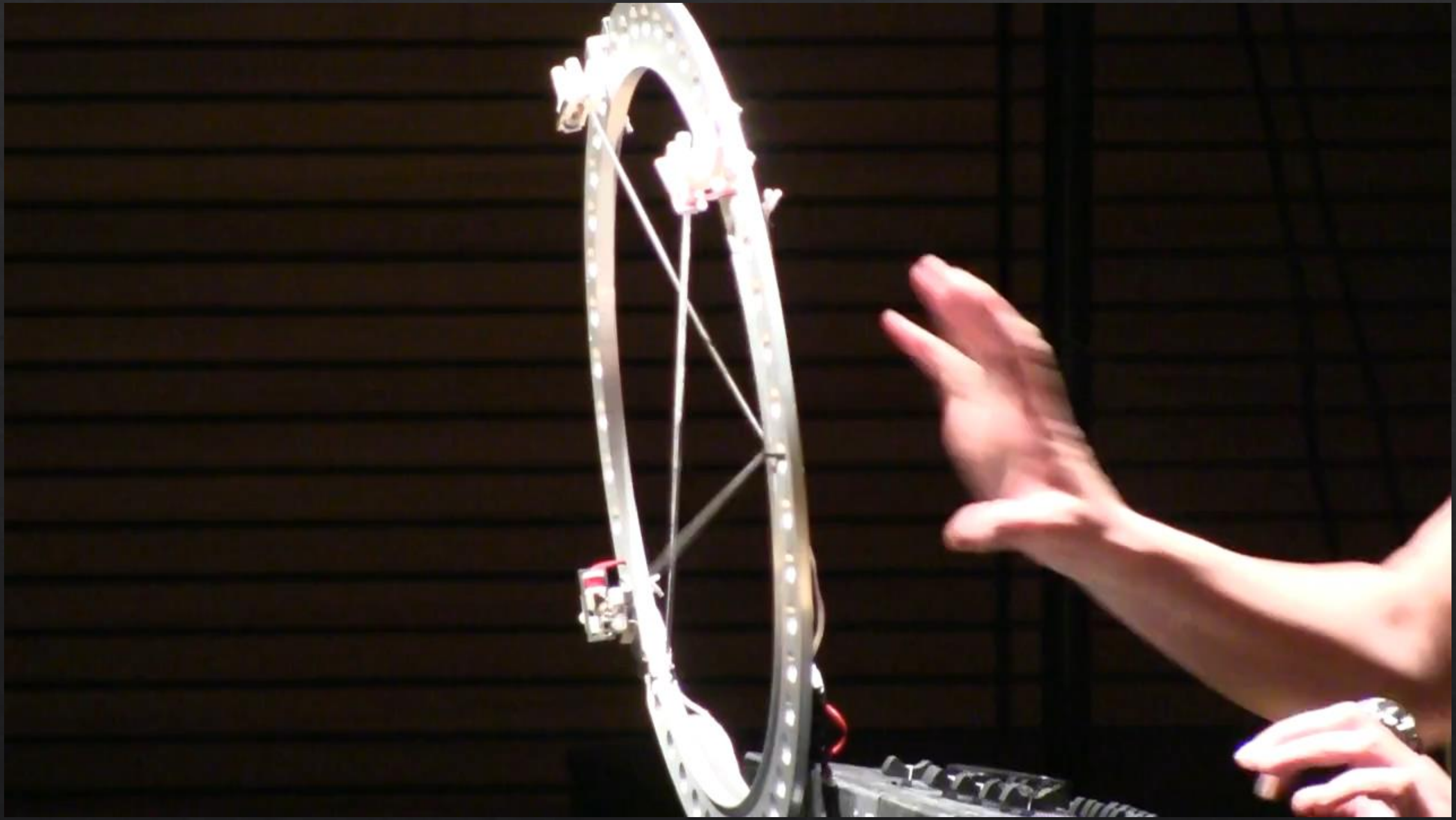
In 2017, the artist will have a residency at the American Academy in Berlin, and presumably continue her Jean Tinguely-esque foray into machine building.

Thessia Machado

Laetitia Sonami

Laetitia Sonami is a sound artist, performer and researcher. Born in France, she settled in the United States in 1975 to pursue her interest in the emerging field of electronic music and studied with Eliane Radigue, Joel Chadabe, Robert Ashley and David Behrman.

Laetitia Sonami



Sound Synthesis Methods

Substractive

Wavetable

FM

Additive

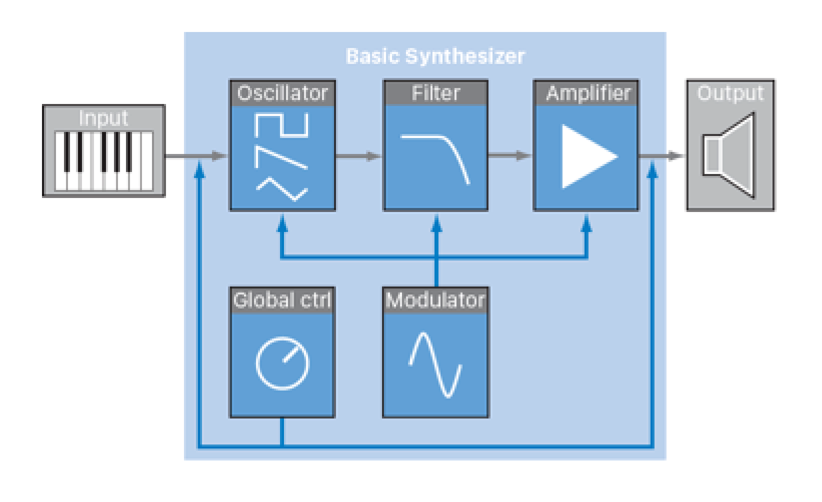
Today’s Challenge

let's make a Max patcher for converting temperatures: specifically, converting Fahrenheit to Celsius.

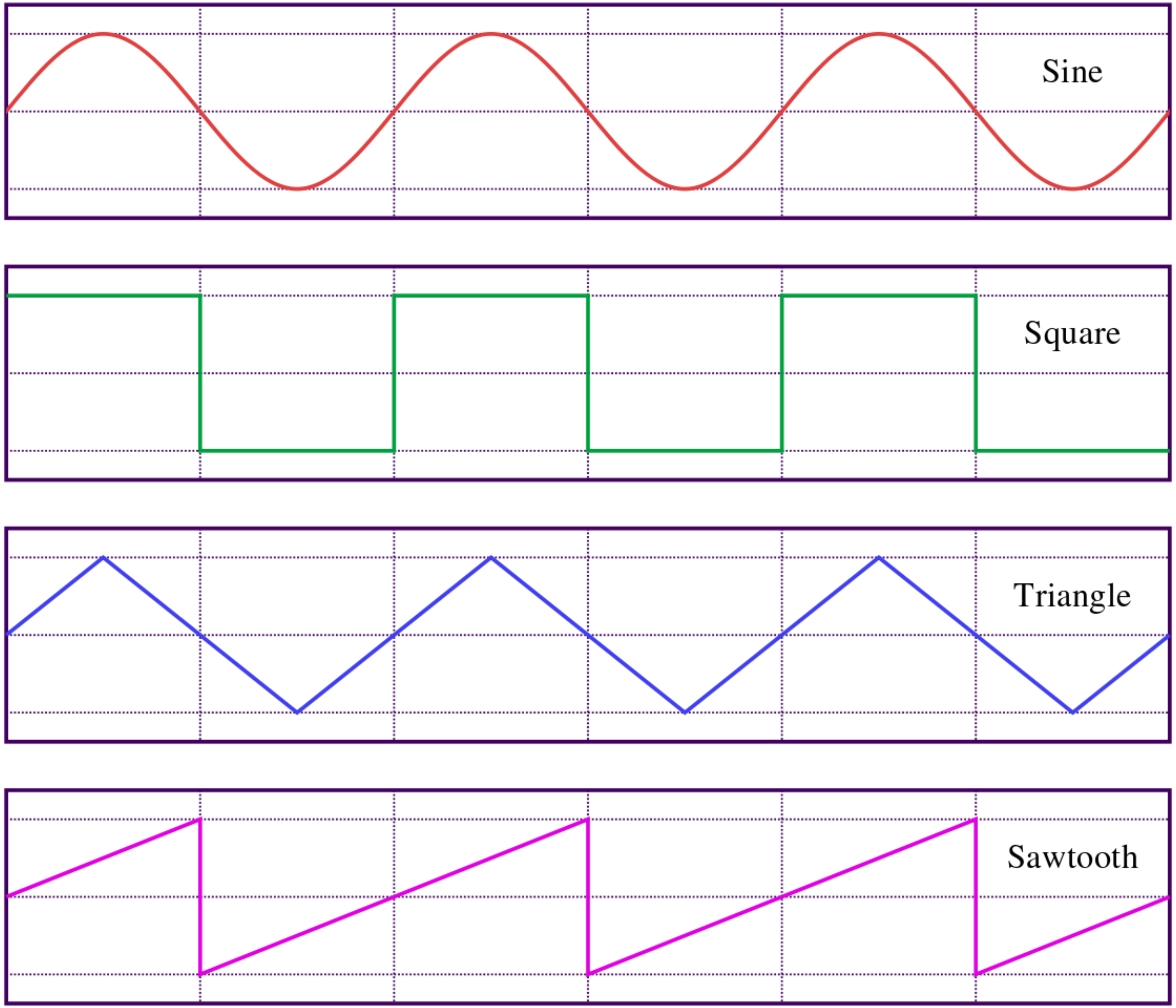
$$°C = (°F - 32) * (5/9)$$

Subtractive Synthesis

Use an existing basic waveforms (oscillator) and apply modifications, effects, filters to manipulate its sonic character



OSC → FILTER → ENVELOPE

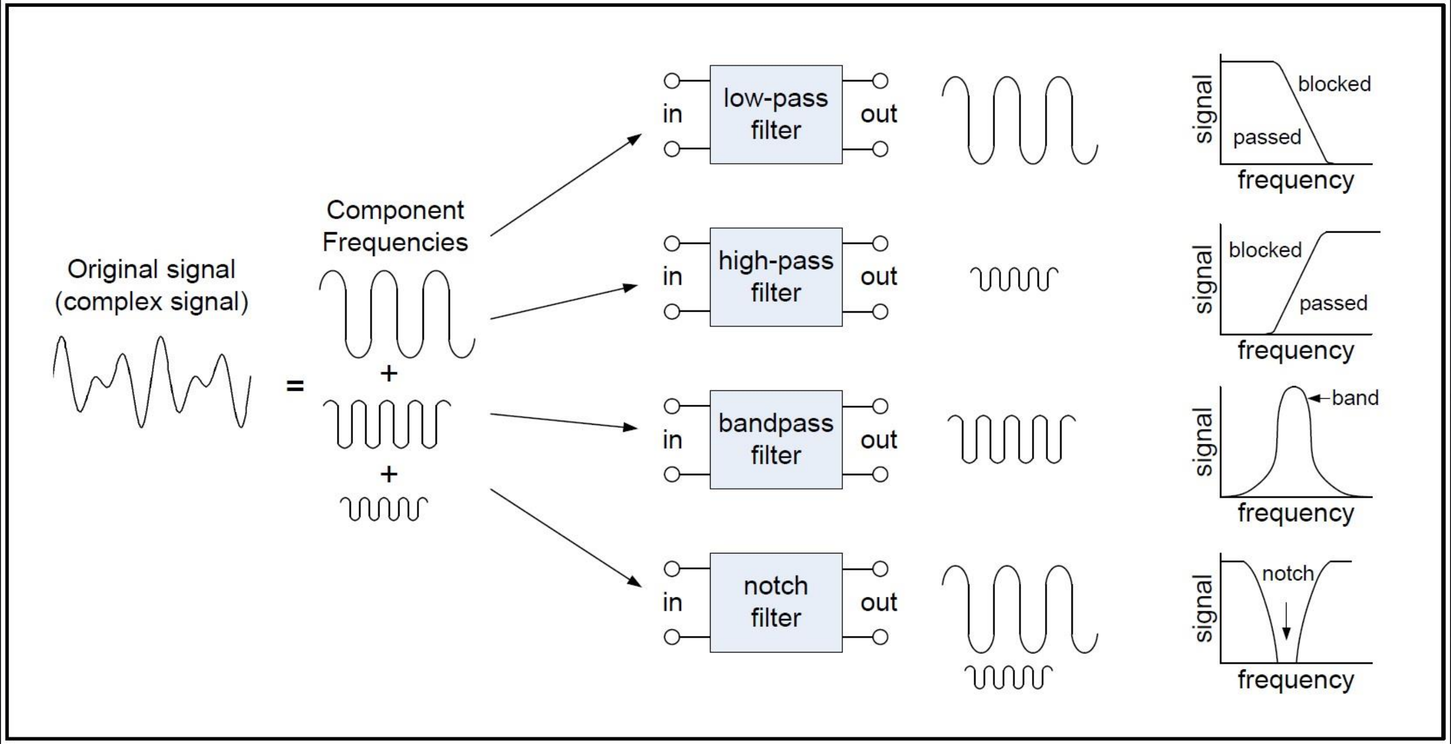

Sine
Square
Triangle
Sawtooth

Subtractive Synthesis

Let’s hear them (sine, saw, triangle, square)

Basic Filters

The four primary types of filters include the low-pass filter, the high-pass filter, the band-pass filter, and the notch filter (or the band-reject or band-stop filter).



Filters explained

A filter is a circuit or software routine that can change the spectral shape of a signal; that is, it will change the amplitude of some frequency regions and leave others alone.

The simplest example is the bass control found on most audio systems—it will increase the low end of the music, or if that's not your style, turn it down. Similarly, the treble control will afect the high end.

Both leave the middle as it is. There are many kinds of filter available; some are named by what they do, others are named by the technique used to implement them.

Ref: https://docs.cycling74.com/max7/tutorials/08_filterchapter01

Filters explained

When we talk about filters, we will make use of the following terms:

Any audio system can be described by its frequency response, which is a graph of the amplitude change across the audio spectrum (20-20,000 Hz). in an ideal system, the graph is a straight line, indicating a flat frequency response. To talk about filters, we look at the frequency response.

The frequency region that is unaffected by a filter is the passband.

A highpass filter affects signals lower than a specified frequency.

A lowpass filter affects signals higher than a specified frequency.

Most filters have a gradual transition from the passband to the rejected region. The shape of this transition is called the slope, which is specified in dB per octave.

Ref: https://docs.cycling74.com/max7/tutorials/08_filterchapter01

Filters explained

The frequency at which a filter becomes effective is called the cutoff frequency. It is actually the frequency at which the signal is reduced by 3dB. (A just noticeable difference in level.)

A bandpass filter affects signals above and below a specified center frequency.

Obviously, a bandpass filter has two cutoff frequencies. The difference between these is the bandwidth.

The ratio of center frequency to bandwidth of a bandpass filter is known as the quality factor or Q of the filter. A filter with high Q will have a narrow passband. A filter with a high Q will also, depending on design, tend to resonate. This has led to an association of Q and a feature called resonance, at least in synthesis circles. Strictly speaking, resonance is a feature of some low and high pass designs, but some authors (and manufacturers) use the terms interchangeably.

The opposite of a bandpass filter is a notch filter, which rejects a band in the middle of the spectrum.

In addition to modifying amplitude of each frequency, filters also modify phase. A plot of phase change vs. frequency is the phase response. An ideal phase response would also be a flat line.

Ref: https://docs.cycling74.com/max7/tutorials/08_filterchapter01

Envelope

ADSR is a four-segment envelope generator. This means that it provides control over four independent parameters of the envelope signal.

Attack (A) : Attack is a time-based function. It determines the time taken by the signal to reach the internally determined maximum value.

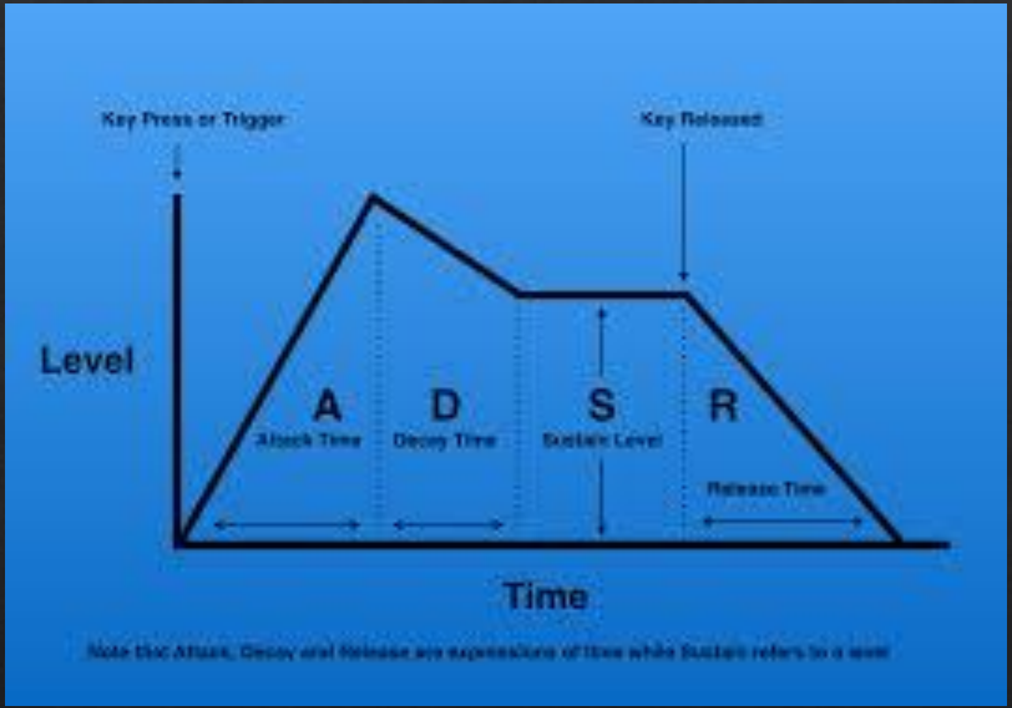
Decay (D) : Decay is a time-based function. It determines the time in which the volume goes from max to the programmed sustain level.

Sustain (S) : It is a level-based function. It determines the level at which the signal sustains.

Release (R) : It is a time-based function. It determines the time that the signal takes to get to zero value once the gate has been turned off.

Ref: https://soundbridge.io/adsr-envelope-and-gate-signals/

Envelope



Ref: https://theproaudiofiles.com/synthesis-101-envelope-parameters-uses/

LFO

Abbreviation for Low Frequency Oscillator.

An oscillator primarily used as a modulator for other things. Low frequency oscillators may or may not operate exclusively below 20 Hz (the lower limit of typical human hearing), but by definition they are not designed to be used as sound generating elements, even though they can be in some synthesizers.

When you bring in modulation (vibrato) in a keyboard, or a chorus in an effects processor you are using an LFO to generate the waveform that produces the variance. In synthesizers and more advanced effects units LFO's can often be routed to many different parameters (often simultaneously).

They can sometimes generate different waveforms (sine, sawtooth, square, random, etc.) for different types of modulation effects.

Ref: https://soundbridge.io/adsr-envelope-and-gate-signals/

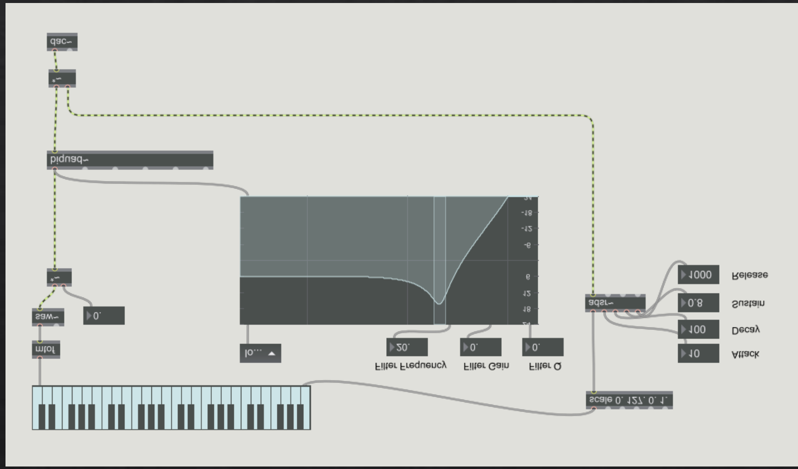
LFO explained

Ref: https://www.youtube.com/watch?v=YEHnd9b79Uc

Subtractive Synthesis Explained

Building Subtractive Synthesis Patch

OSC ➞ FILTER ➞ ENVELOPE



Assignment

Build your own substractive synthesis

Features required

- Selection of oscillator (sine, saw, square, tri)

- ADSR Filter

- LFO

Recommended resources:
https://theproaudiofiles.com/the-fundamentals-of-subtractive-synthesis/