

# SOUND SYNTHESIS

ADDITIVE  
SUBTRACTIVE  
RM-AM  
FM  
PHASE DISTORTION  
WAVESHAPING  
PHYSICAL MODELING  
GRANULAR  
SAMPLING  
WAVETABLE

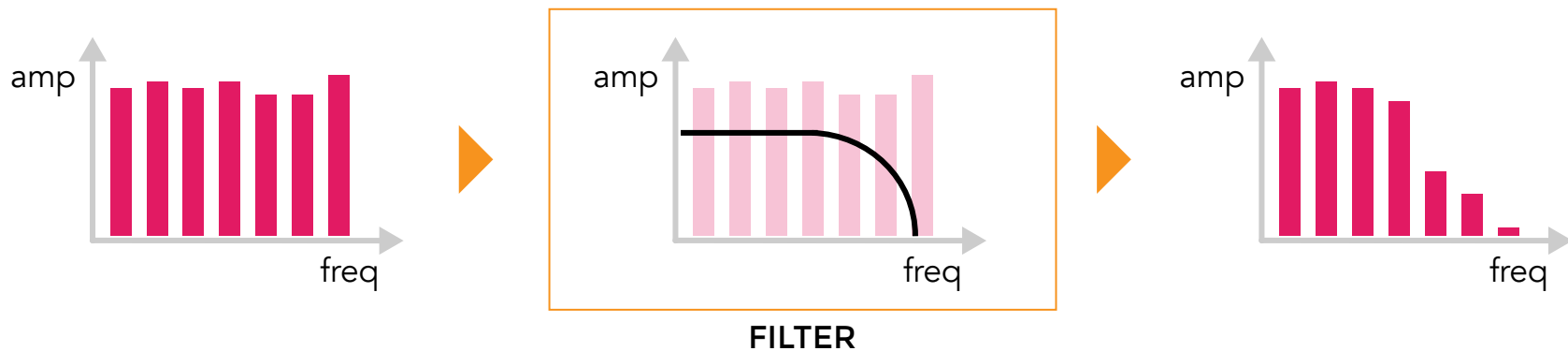


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# Introductory concepts

## FILTERS

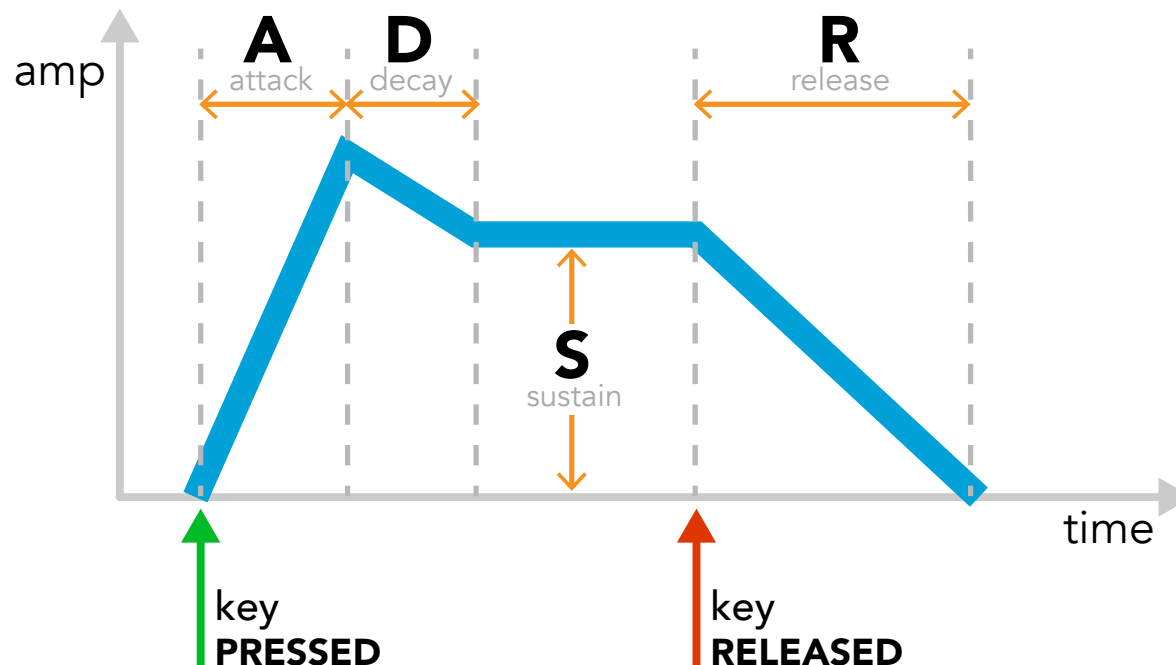
A **filter** is a device that attenuates (sometimes emphasizes) the amplitude and alters the phase of certain frequencies in a sound. Filters can help us refine or sculpt the sound to meet our sound synthesis goals.



# Introductory concepts

## ENVELOPE

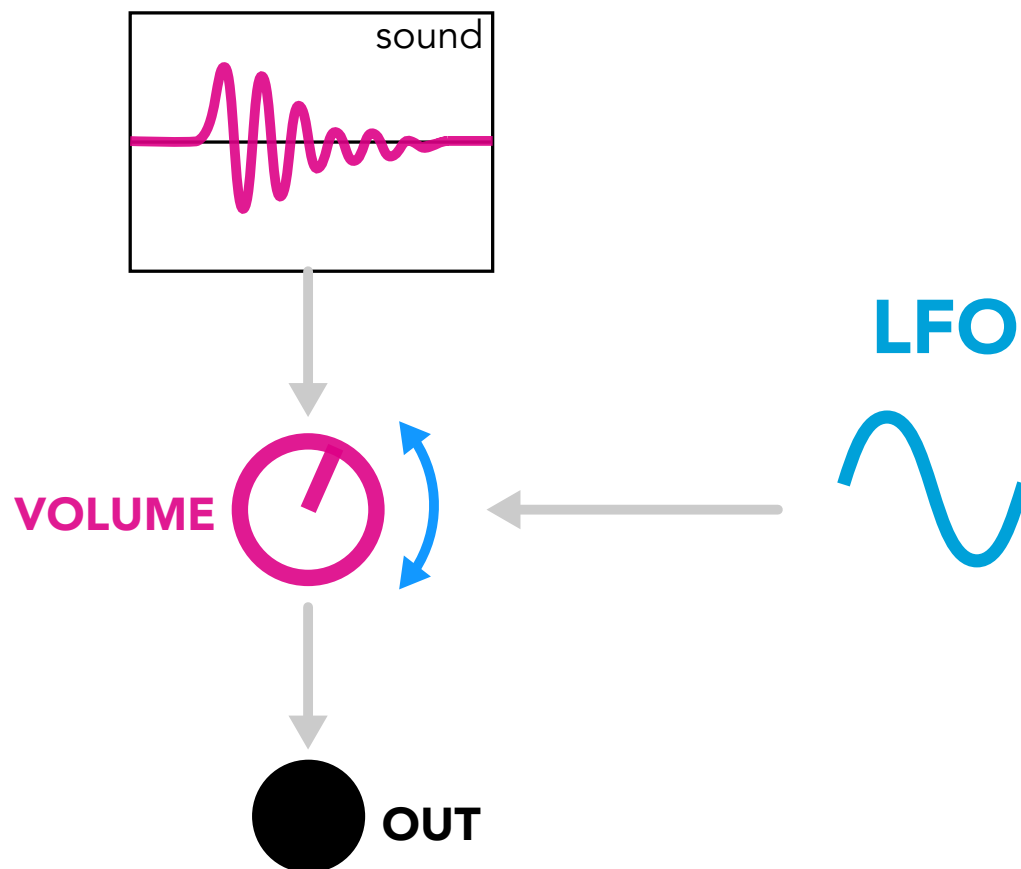
The **envelope** is the trend of an instrument's amplitude (or sometimes other parameters) from the moment it is excited to when the note fades away to nothing.



# Introductory concepts

## LFO (Low Frequency Oscillator)

**LFO (Low-Frequency Oscillator)** is an oscillator that is not used to generate sound but to move certain parameters of the synthesis algorithm.

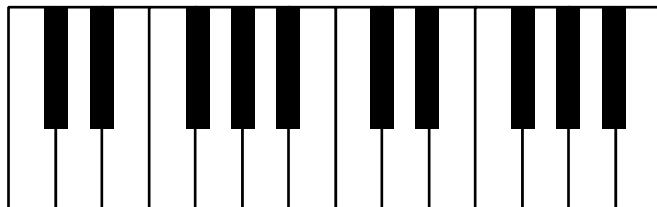




# Introductory concepts

## CV (Control Voltage)

**Control Voltage** is the system that uses the output voltage of analog synths to control parameters.



**VCO** [voltage control oscillator]  
control the pitch of the oscillators

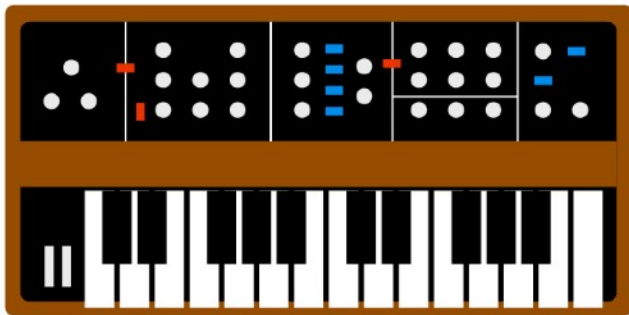
**VCF** [voltage control filter]  
control the filter stage

**VCA** [voltage control amplitude]  
control a dynamic envelope

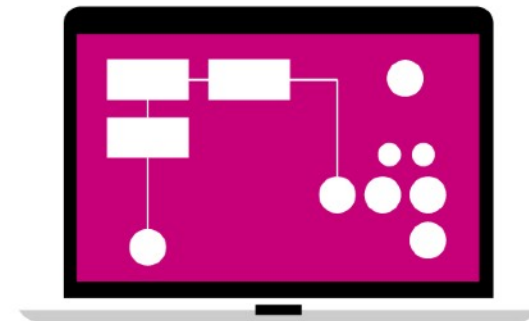
## Hardware or Software

Sound synthesis algorithms can be implemented by **hardware** systems, or **software** programs executed on a computer.

HARDWARE



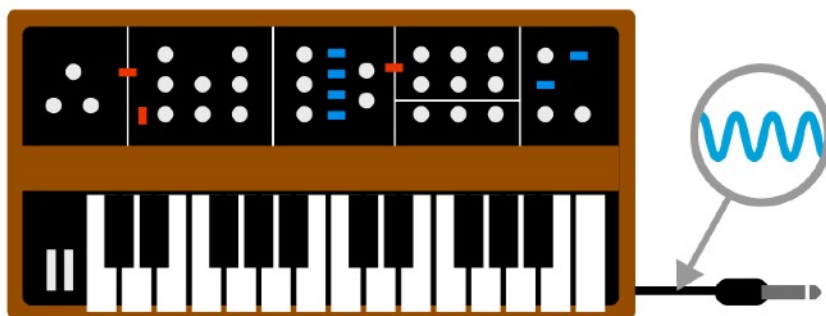
SOFTWARE



## Analog or Digital

In the case of software, the synthesis will be **digital**. Historically, hardware synthesizers only had **analog** circuits, but modern hardware synths can take a hybrid approach with analog and digital circuitry.

ANALOG

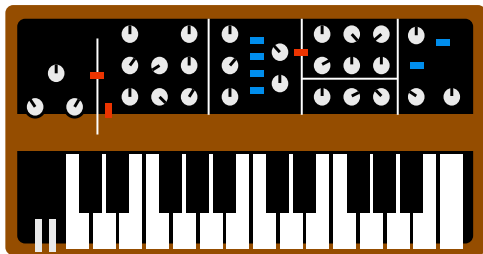


DIGITAL

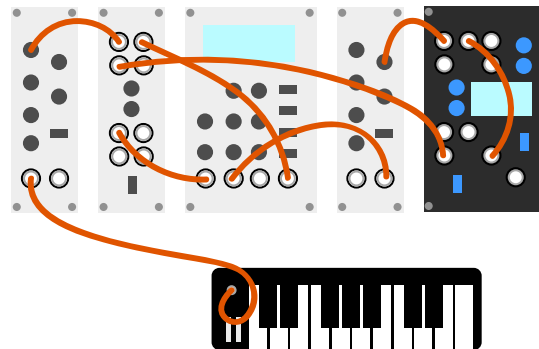


# Hardware or Software, Analog or Digital, Standalone, Modular or Semi-Modular

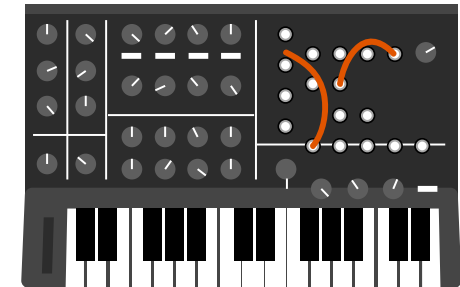
Synthesizers can be classified into **standalone** with fixed internal circuitry, **modular** with customizable module connections, and **semi-modular** that blend standalone operability with modular flexibility.



standalone



modular



semi-modular

## LINEAR techniques

They are based on adding and subtracting processes.

Consequently, the degree of the algorithm's complexity is directly linked to the spectral complexity of the sound produced.

ADDITIVE; SUBSTRACTIVE, GRANULAR

## NONLINEAR techniques

Here the sonic result does not vary in proportion to the signals' complexity. We can generate signals with many harmonic or inharmonic components from just a few initial elements.

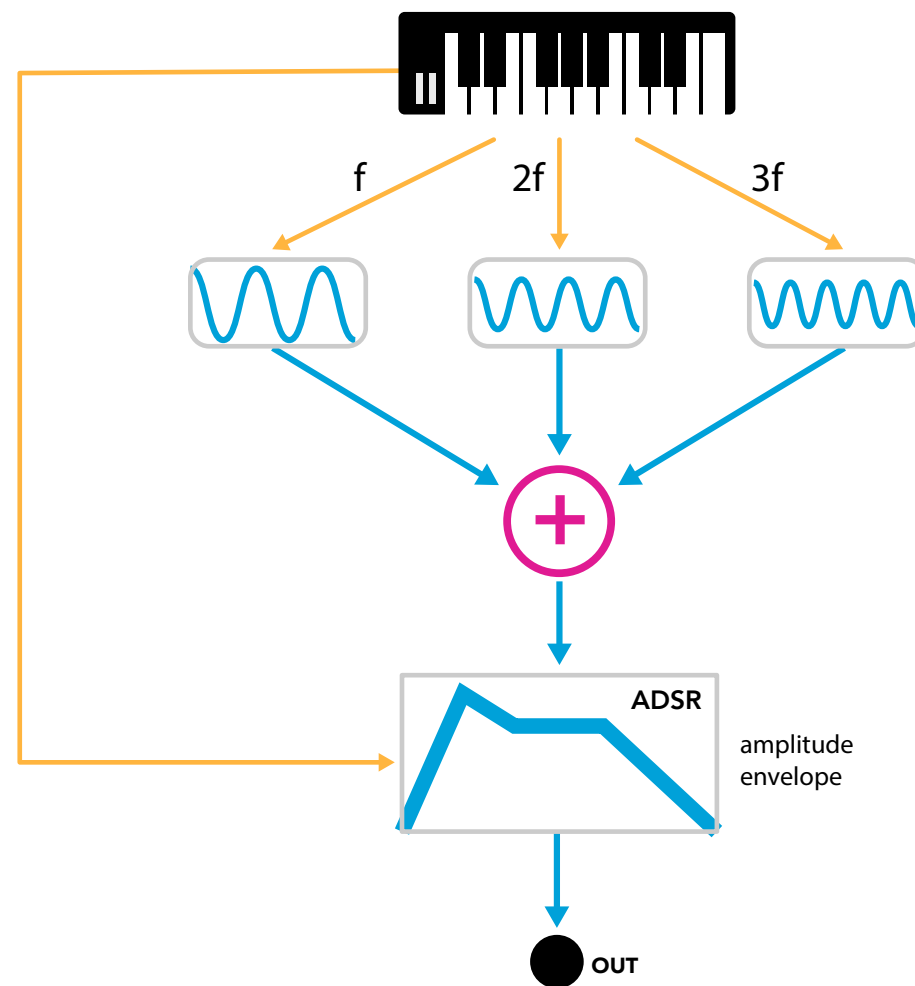
AM, RM, FM, WAVESHAPING



# ADDITIVE SYNTHESIS

A linear sound synthesis technique, additive synthesis operates on the summation of properly tuned sine waves.

Potentially, with additive synthesis, it is possible to reconstruct any timbre and create complex new ones, but it is very computationally expensive.



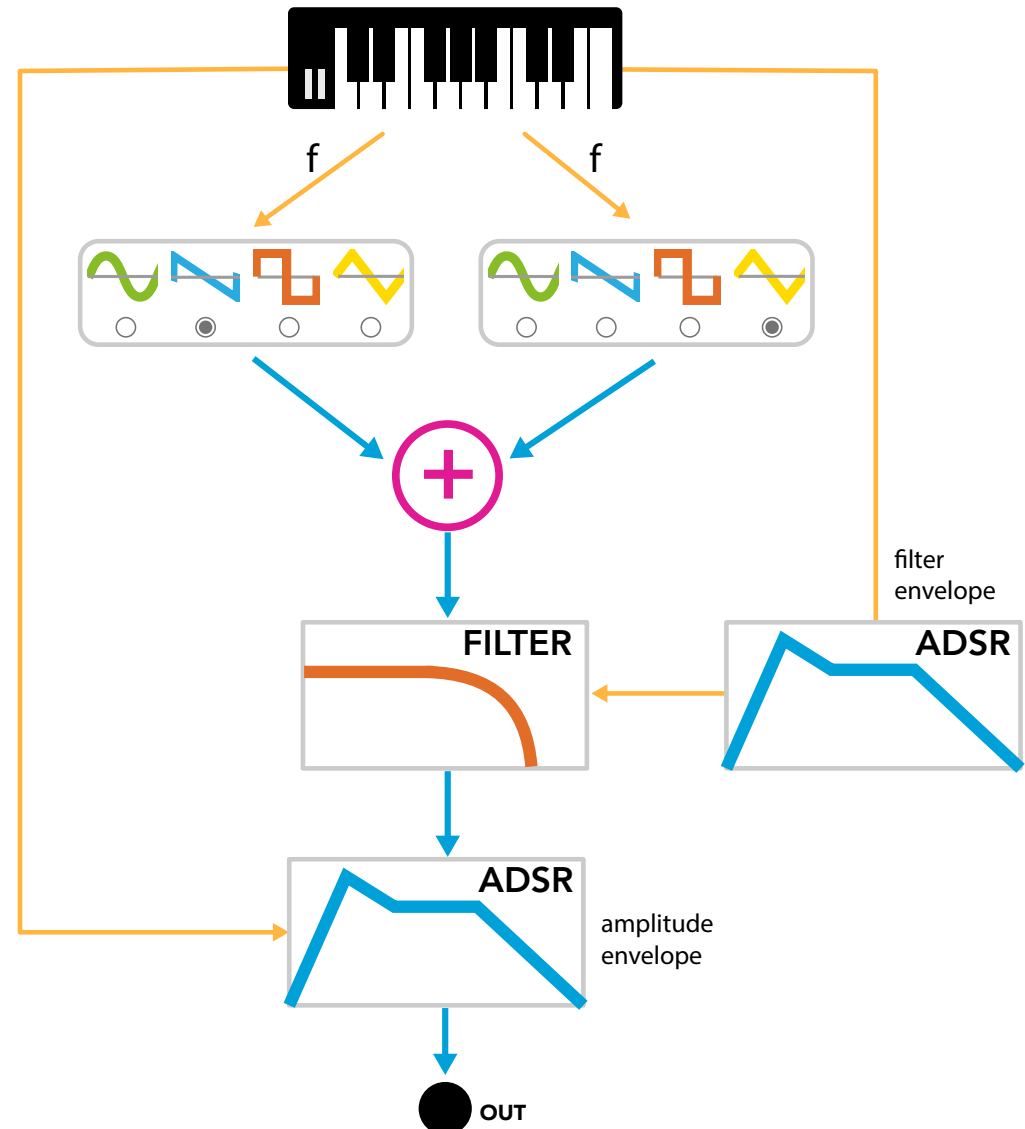
# ADDITIVE SYNTHESIS



Hammond Organ B3 (Model A from 1935)

# SUBTRACTIVE SYNTHESIS

Subtractive synthesis, widely favored for its simplicity and applicability in both analog and digital formats, shapes sound by filtering out frequencies from a complex base sound, much like a sculptor chiseling away marble to reveal a form.





# SUBTRACTIVE SYNTHESIS



Minimoog Model D (1971)

# SUBTRACTIVE SYNTHESIS



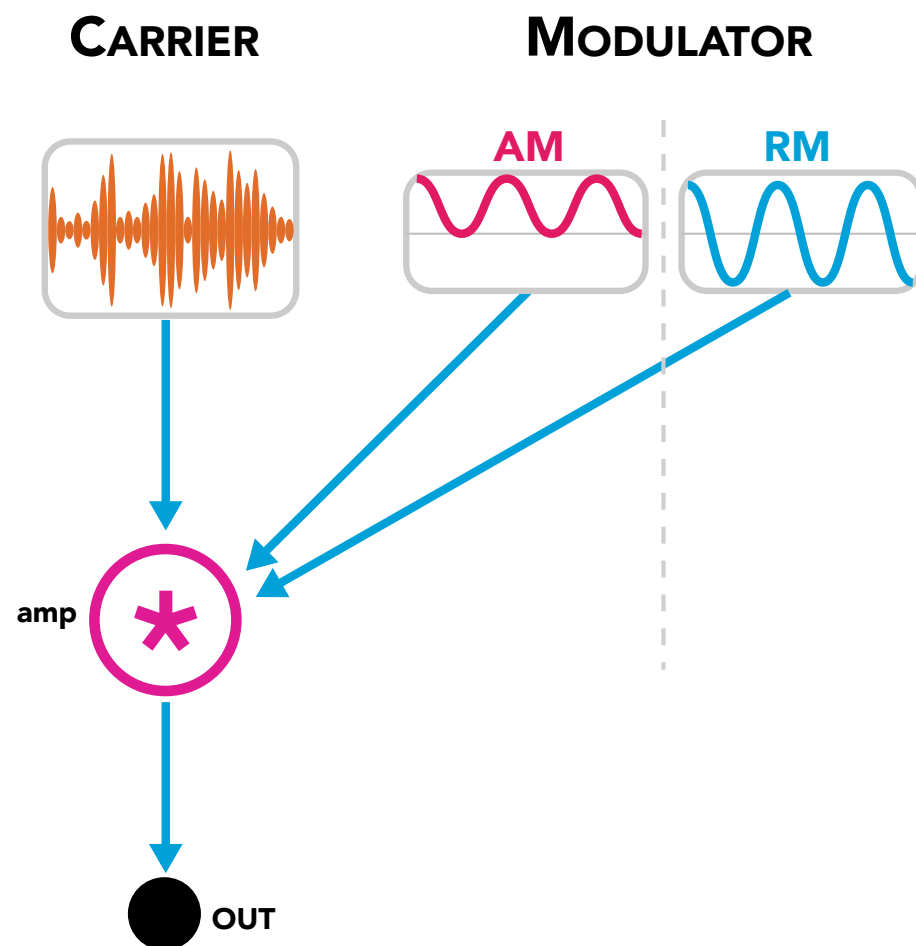
ANALOG in Ableton Live



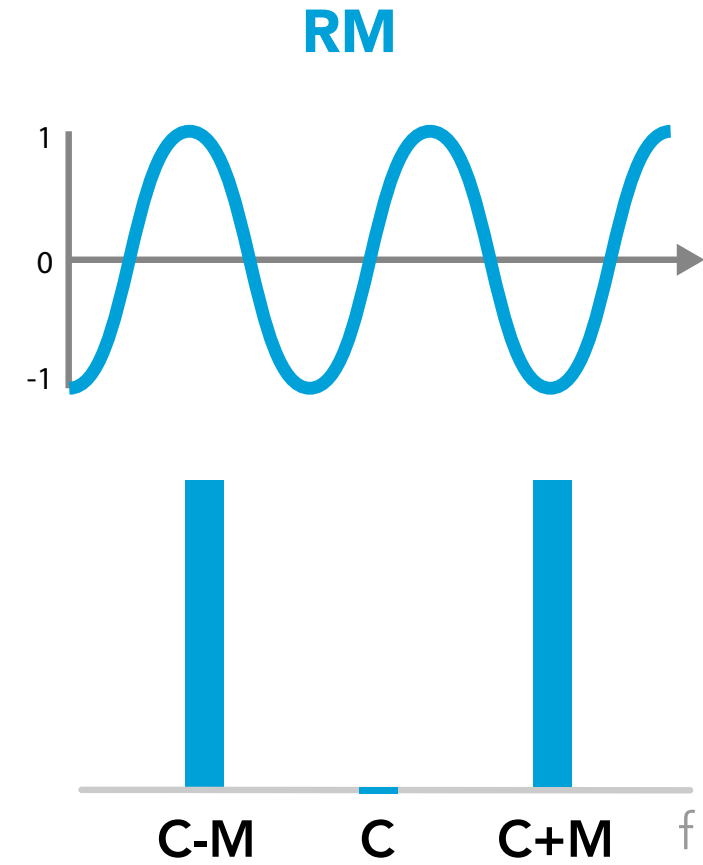
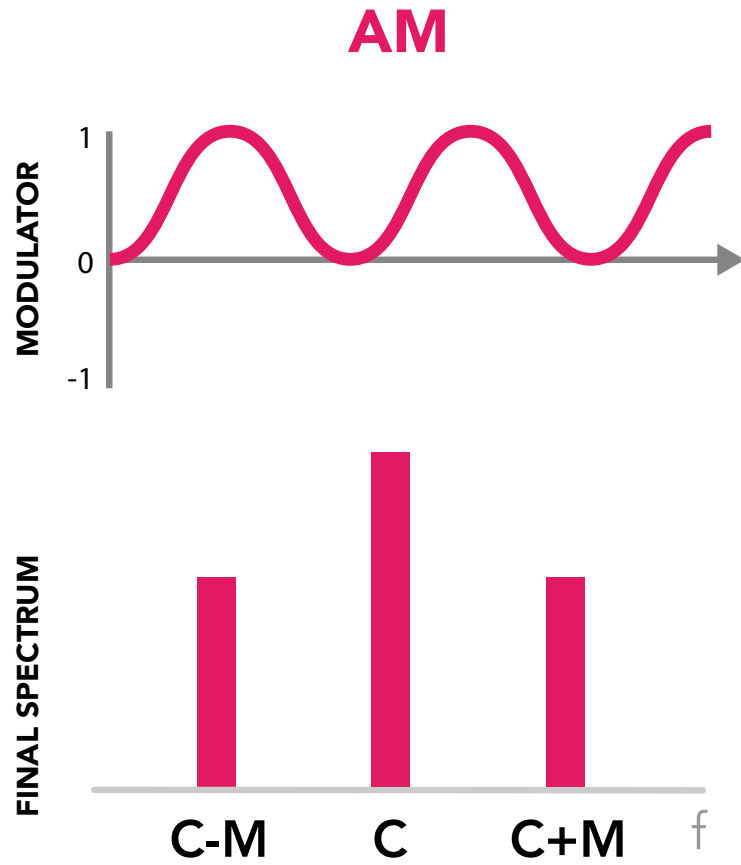
# AM (AMPLITUDE MODULATION) - RM (RING MODULATION)

Using a high frequency (more than 15 Hz) sine wave to control the amplitude of another sound wave generates a richer timbre.

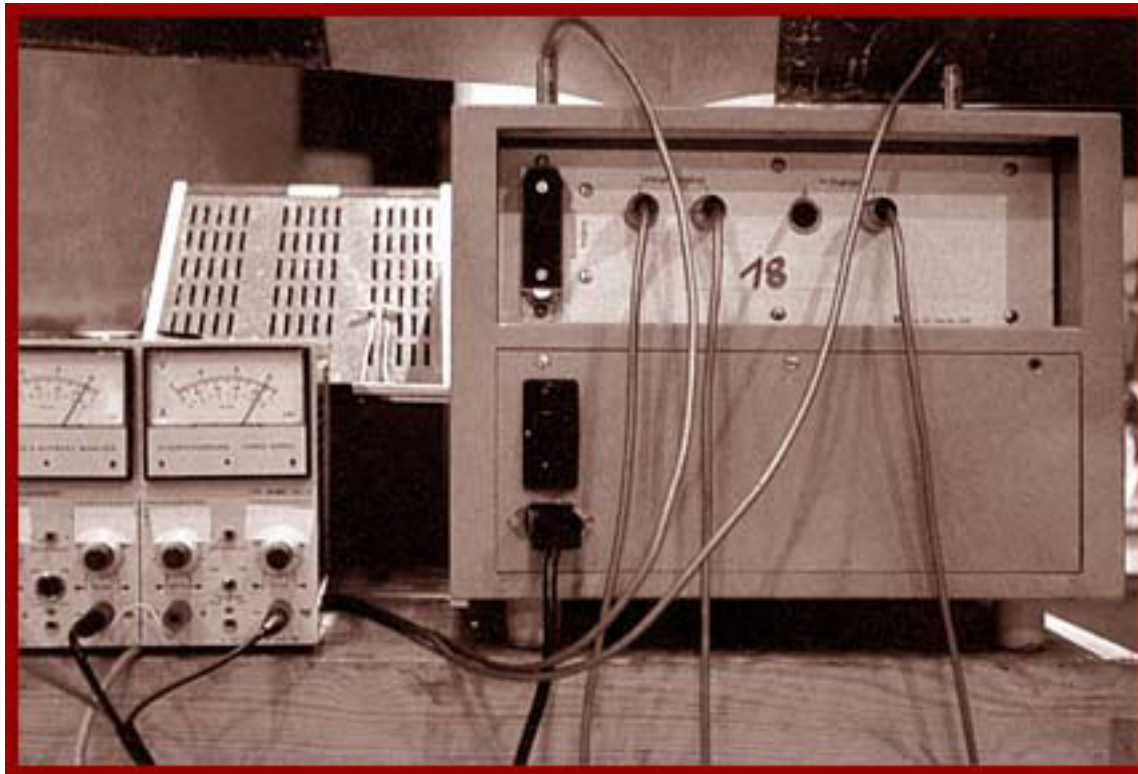
The key difference is that AM uses a unipolar wave (amplitude 0 to 1) for modulation, while RM employs a bipolar wave (amplitude -1 to 1).



# AM (amplitude modulation) - RM (ring modulation)

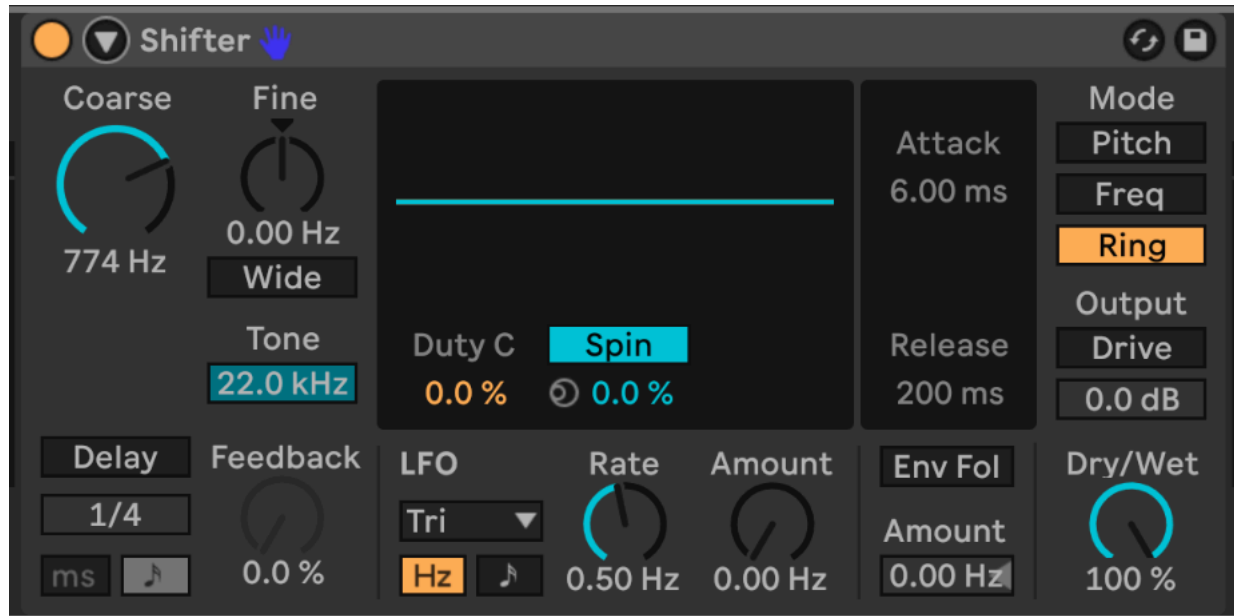


## AM (amplitude modulation) - RM (ring modulation)



Ring Modulator, Studio for Electronic Music of the West German Radio in Cologne (1955)

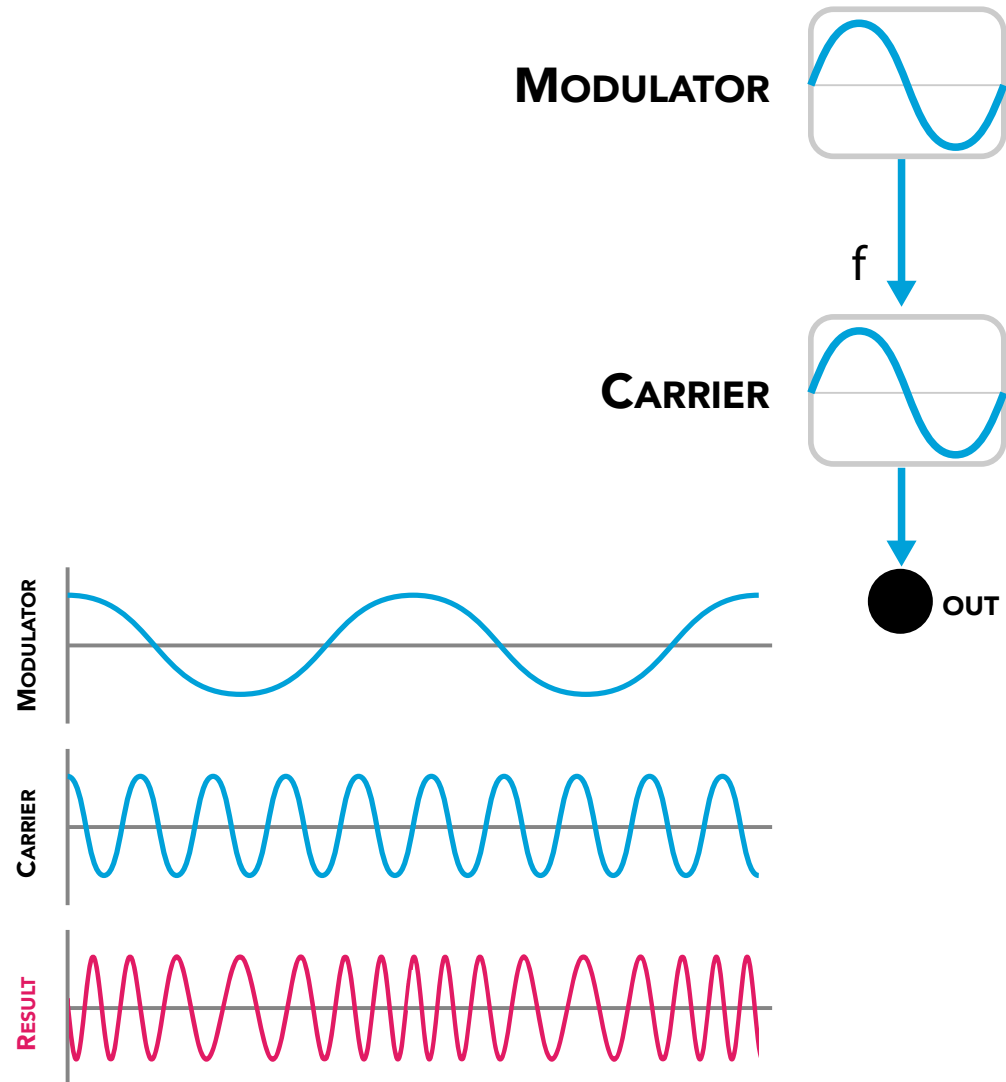
# AM (amplitude modulation) - RM (ring modulation)



SHIFTER audio effect in Ableton Live

# FM (FREQUENCY MODULATION)

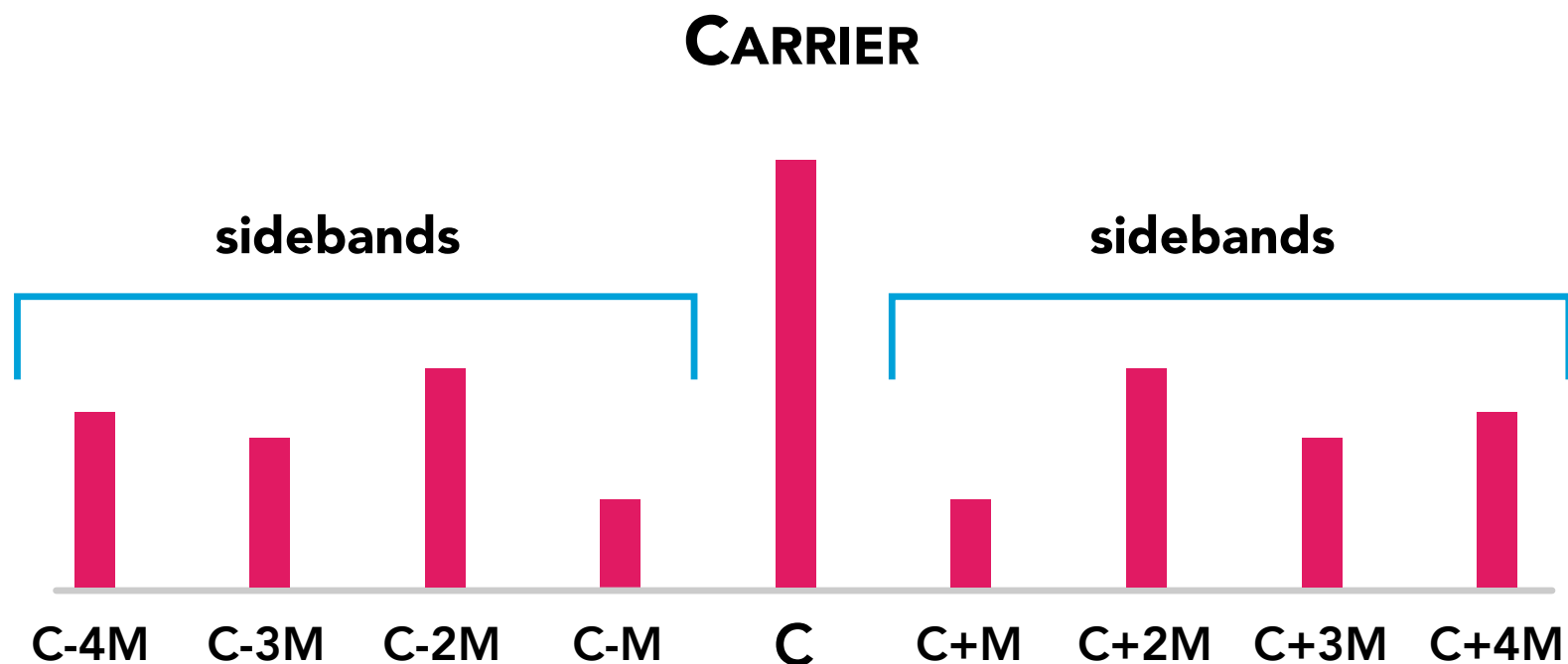
**Frequency Modulation synthesis** is based on an oscillator wave, called a Modulator, that varies the frequency of another oscillator called the Carrier. This process is nonlinear because it can create a complex spectra with many frequencies, all while using just a few elements.





## FM (frequency modulation)

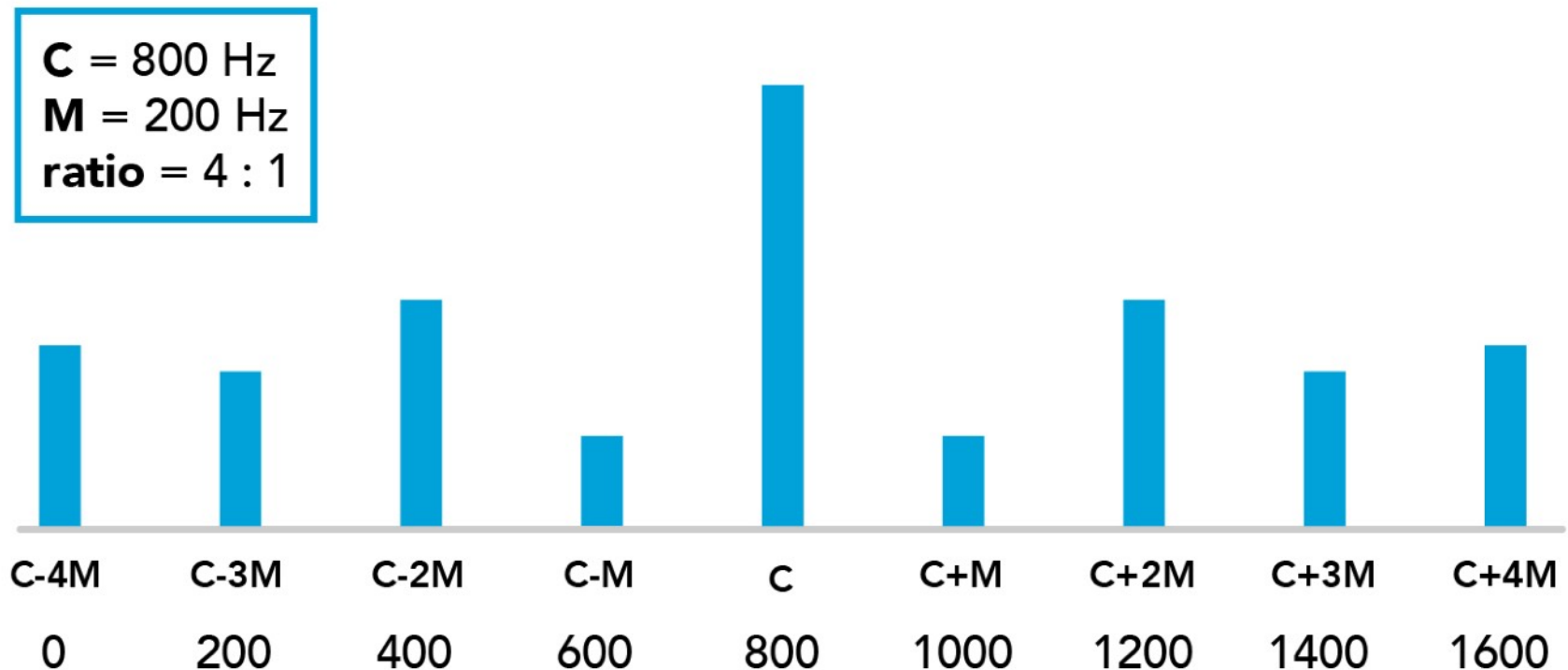
FM synthesis results in a timbre with a series of sidebands centered around the Carrier frequency.



## FM

## POSITION OF THE BANDS

The resultant frequency of the sidebands depends on the ratio between Carrier and Modulator, or the **C:M ratio**. The FM generates a harmonic spectrum when the C:M ratio can be reduced to an integer (e.g., 4:1).

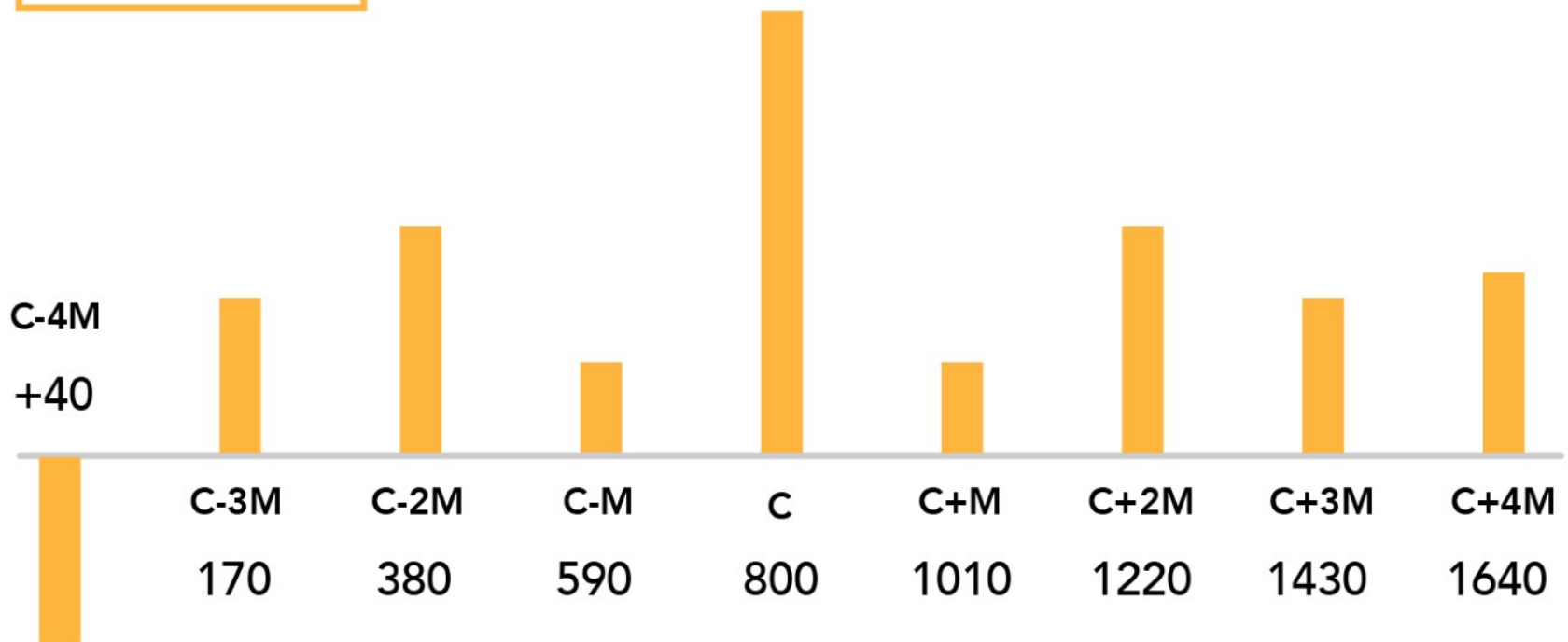


# FM

## POSITION OF THE BANDS

If the **C:M ratio**, on the other hand, is not an integer (e.g., 8:2.1), the resultant sound has an inharmonic spectrum.

**C** = 800 Hz  
**M** = 210 Hz  
**ratio** = 4 : 1,1



# FM

## QUANTITY OF BANDS

As the **modulation index** increases, the timbre becomes more complicated with an increasing number of sidebands.

$$I = \frac{D}{M}$$

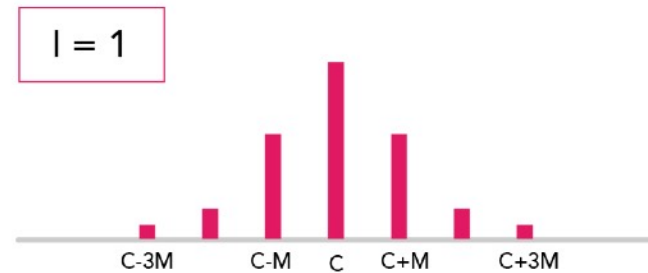
Deviation between C and M  
M frequency

$$I * M = D$$

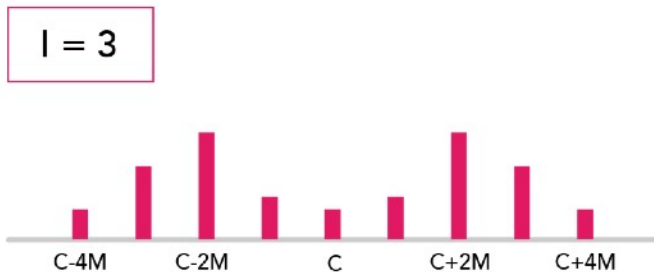
$$I = 0$$



$$I = 1$$



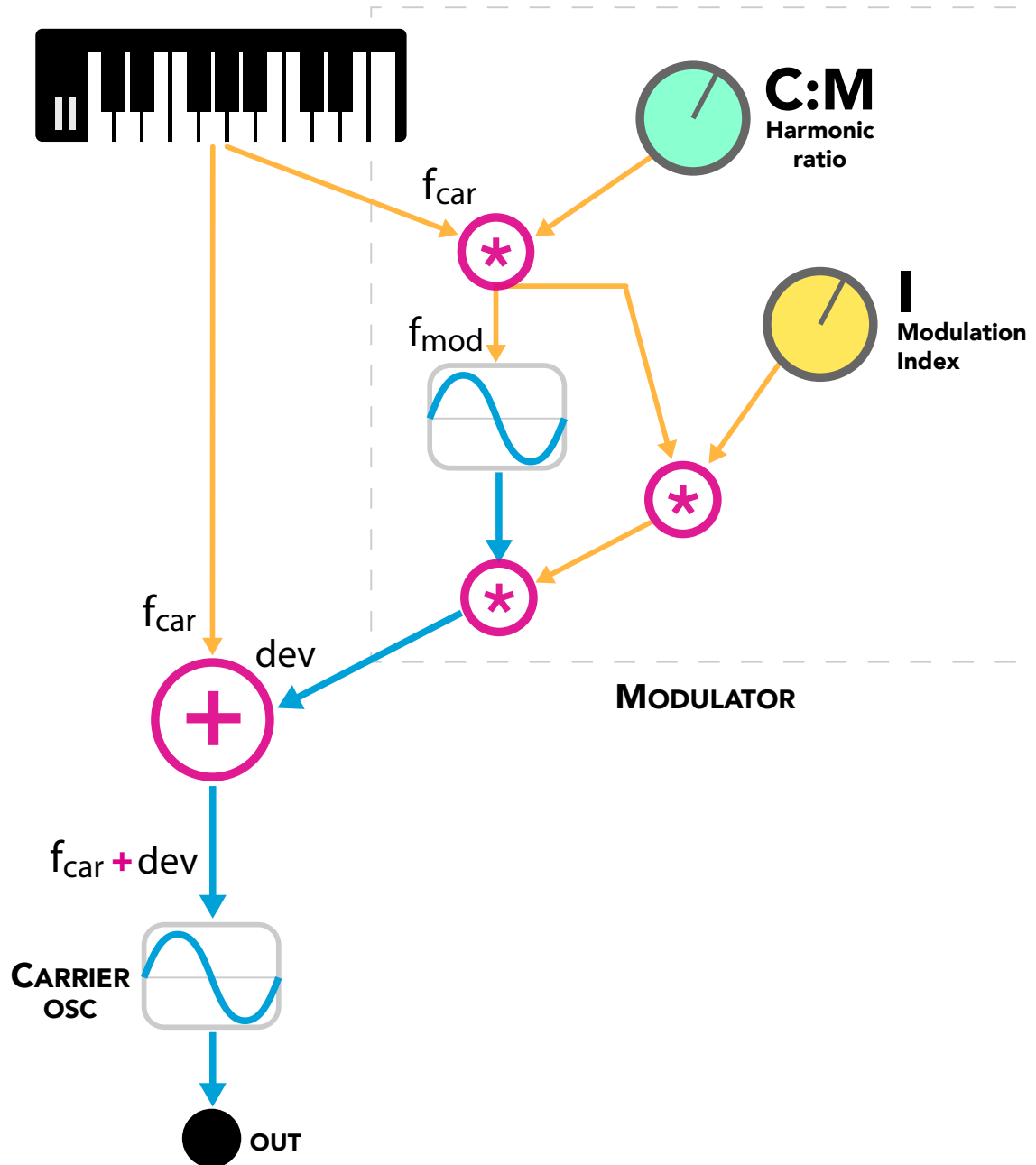
$$I = 3$$



$$I = 4$$



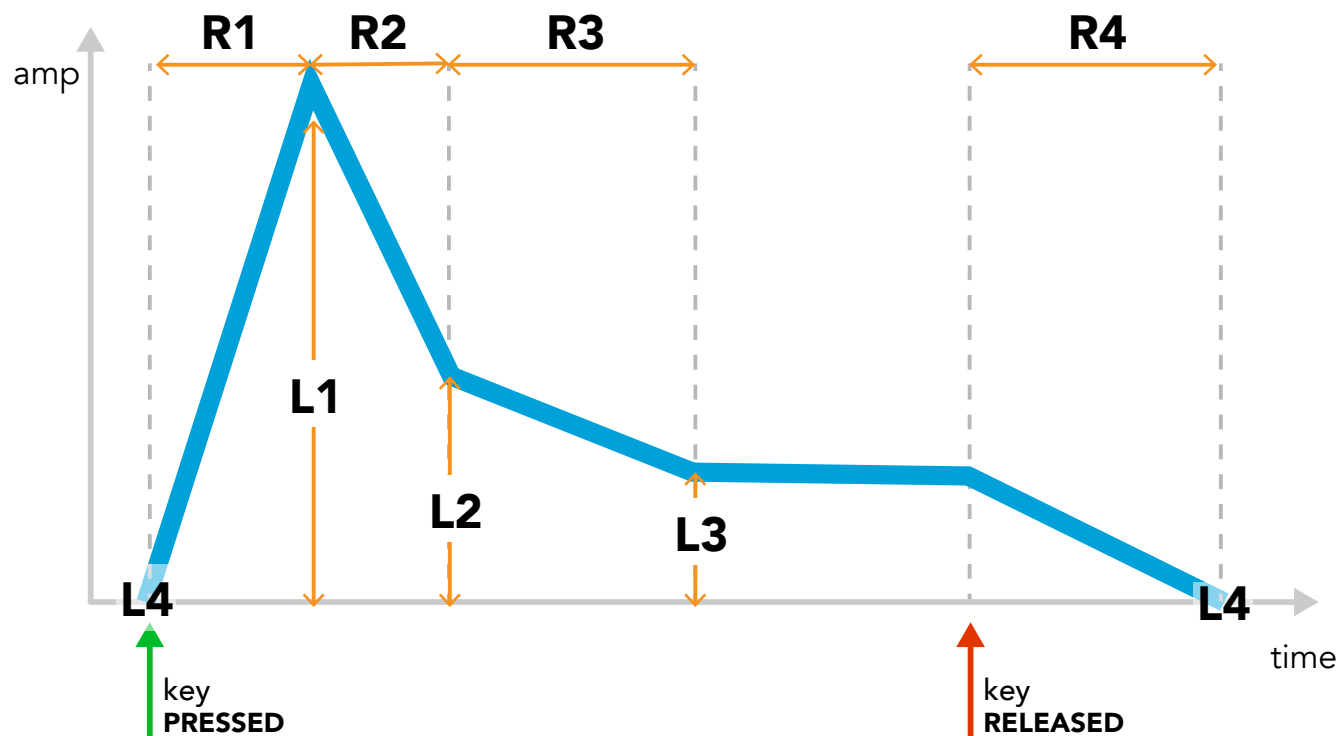
# FM





## FM (frequency modulation)

Envelopes in classic FM is called **EG (envelope generator)** and is composed of 4 amplitude levels and 4 rates that control the slope (and consequentially the time) of the envelope segments.



# FM (frequency modulation)

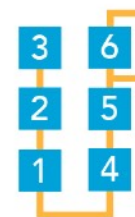
In the classic FM synthesis model, oscillators are called **operators** and an **algorithm** is defined by how we combine, organize, and configure multiple operators together.



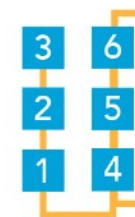
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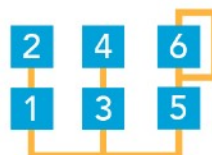
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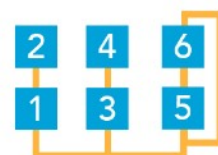
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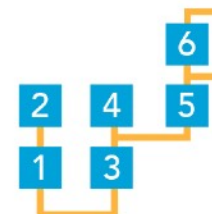
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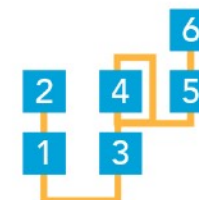
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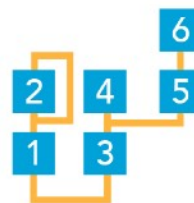
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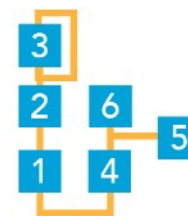
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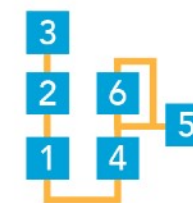
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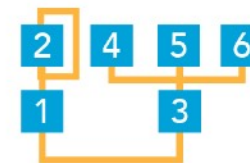
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10

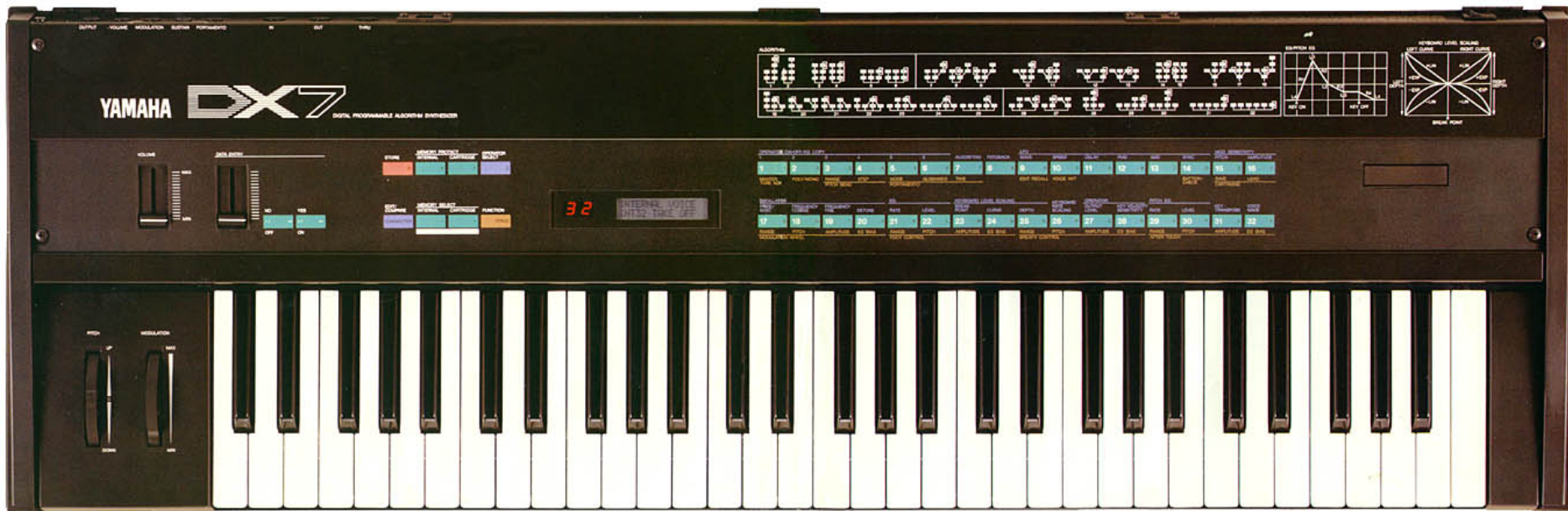


11



12

# FM (frequency modulation)



YAMAHA DX7 (1982)

# FM (frequency modulation)



FM8 VST by Native Instruments

# FM (frequency modulation)

**Operator**

Coarse: 1, Fine: 0, Fixed: , Level: -2.5 dB (D)

Coarse: 8, Fine: 0, Fixed: , Level: 0.0 dB (C)

Coarse: 0.5, Fine: 0, Fixed: , Level: 0.0 dB (B)

Coarse: 1, Fine: 10, Fixed: , Level: -6.0 dB (A)

**Envelope**

Attack	Decay	Release	Time<Vel
0.00 ms	2.96 s	50.0 ms	0 %
Initial	Peak	Sustain	Vel
-inf dB	0.0 dB	-inf dB	0 %
Loop	Key	Phase	Osc<Vel
None	0 %	0 %	0

**Oscillator**

Wave: Sin

Feedback: 0 %

Repeat: Off

**LFO**

Wave: Sine, L: , R:

Rate: 10.00, Amount: 50 %

**Filter**

Filter: Low 12dB

Freq: 500 Hz, Res: 1.00

**Pitch Env**

Pitch Env: 0.0 %

**Spread**

Spread: 67 %

**Transpose**

Transpose: 0 st

**Time**

Time: 0 %

**Tone**

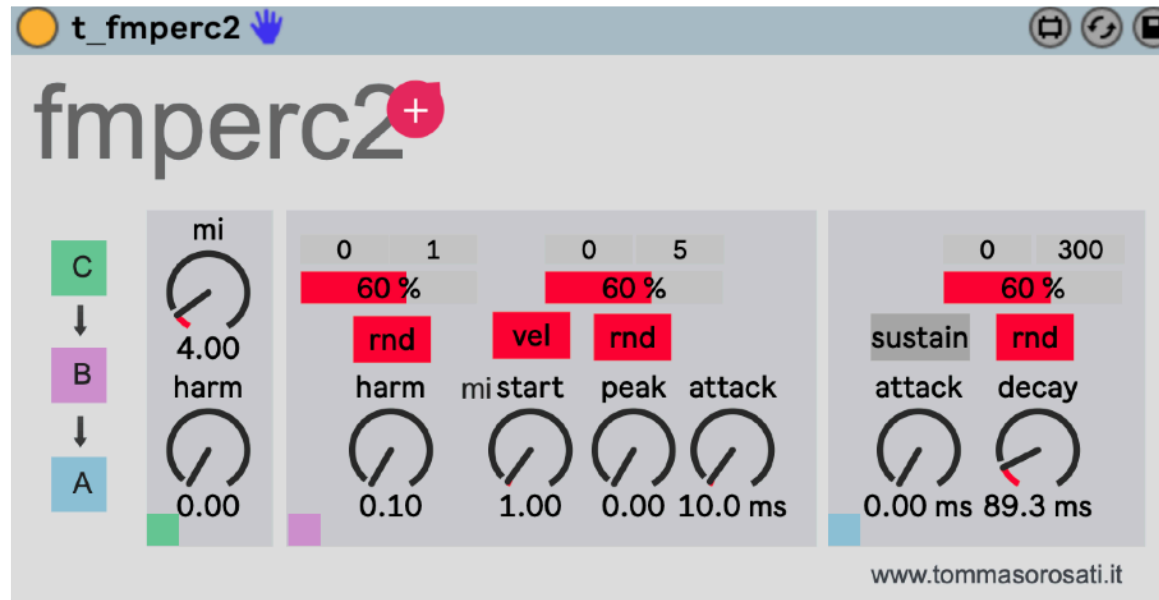
Tone: 70 %

**Volume**

Volume: -12 dB

Operator in Ableton Live

# FM

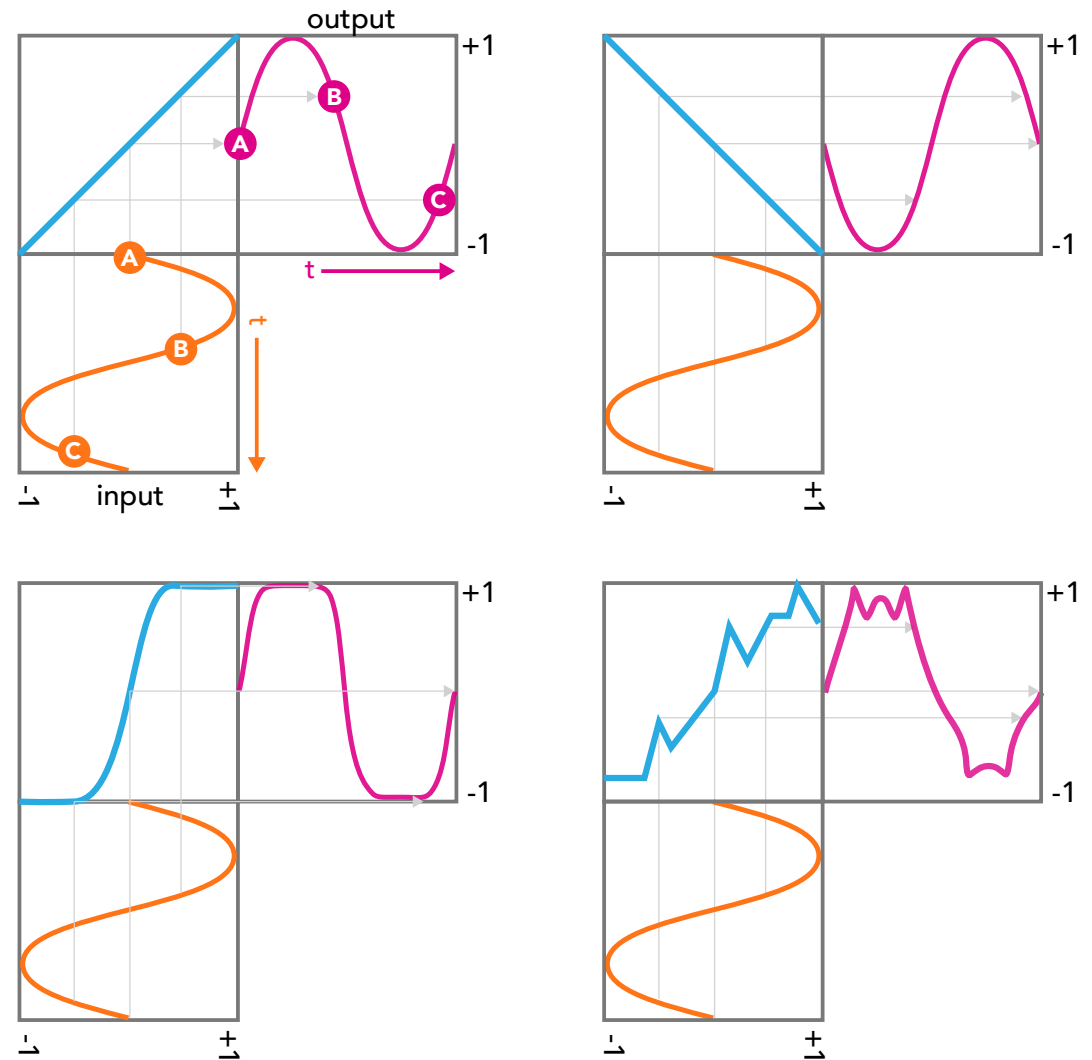


fmperc2 by piymaxforlive



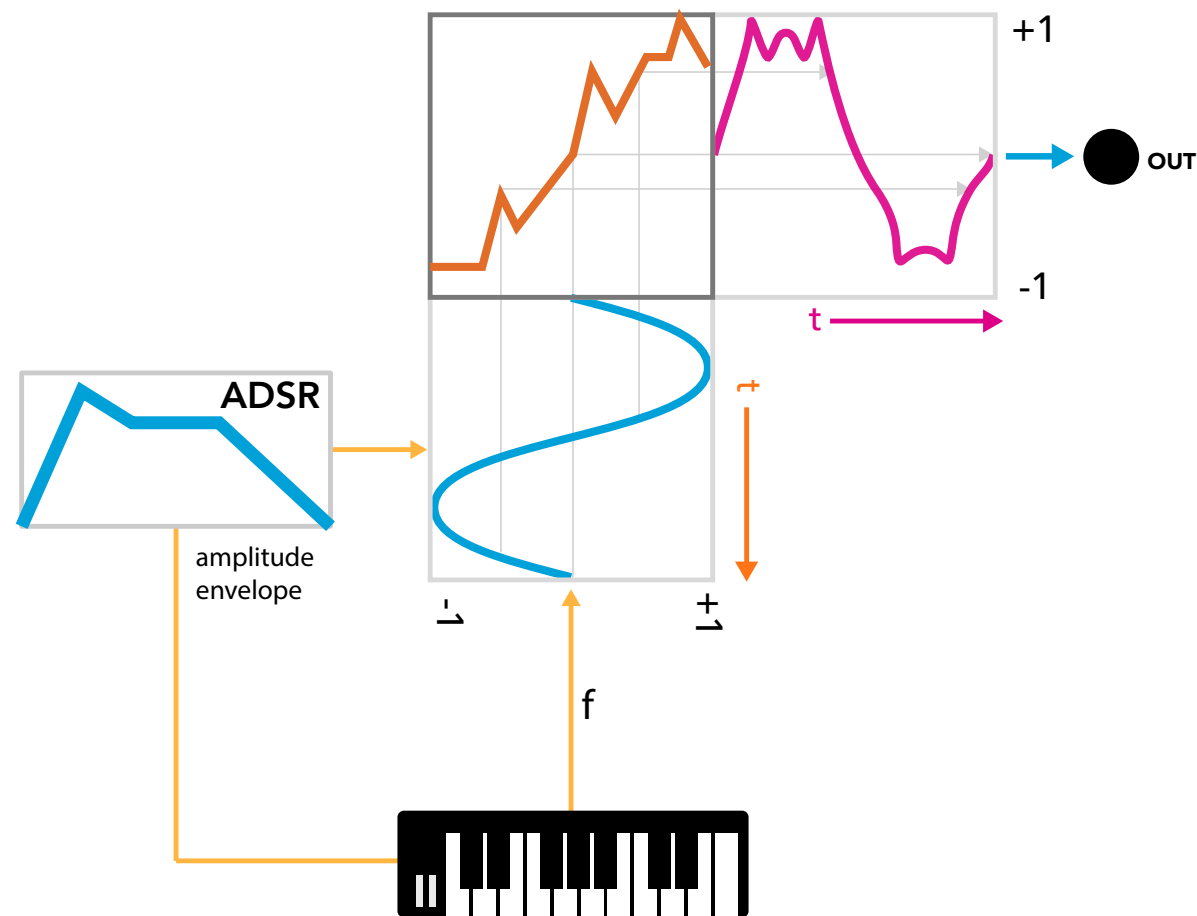
# WAVESHAPING

This algorithm is based on passing one period of an input wave, such as a sine wave, through a transfer function that alters its shape and results in one period of an output wave.



# WAVESHAPING

The input amplitude range determines how much our input "reflects" off the transfer function. If we dynamically control the gain of the input, we can change, in real time, how much of the transfer function is used, thus changing the timbre of distortion of our output.



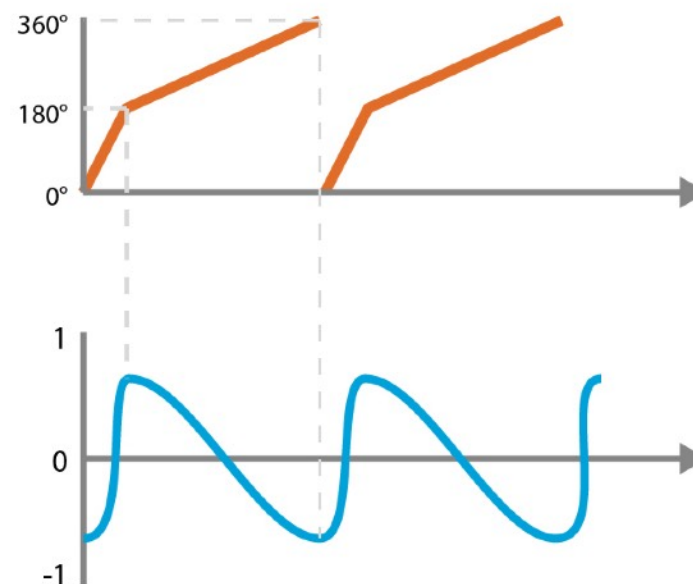
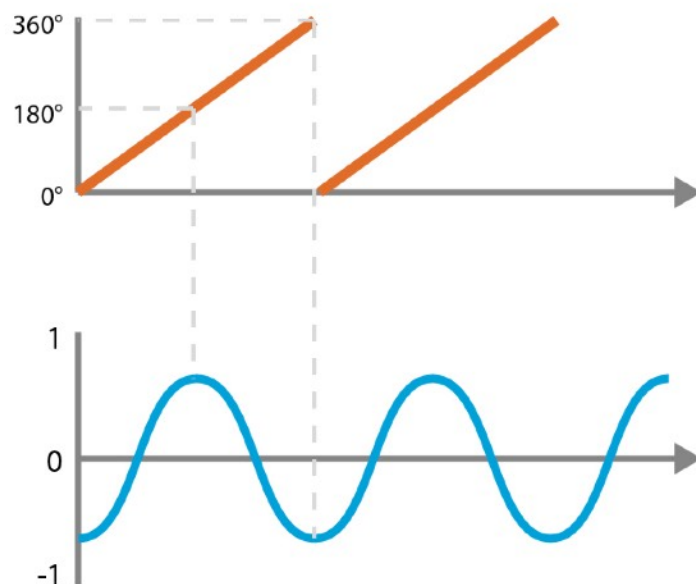
# WAVESHAPING



Cakewalk Z3TA+2

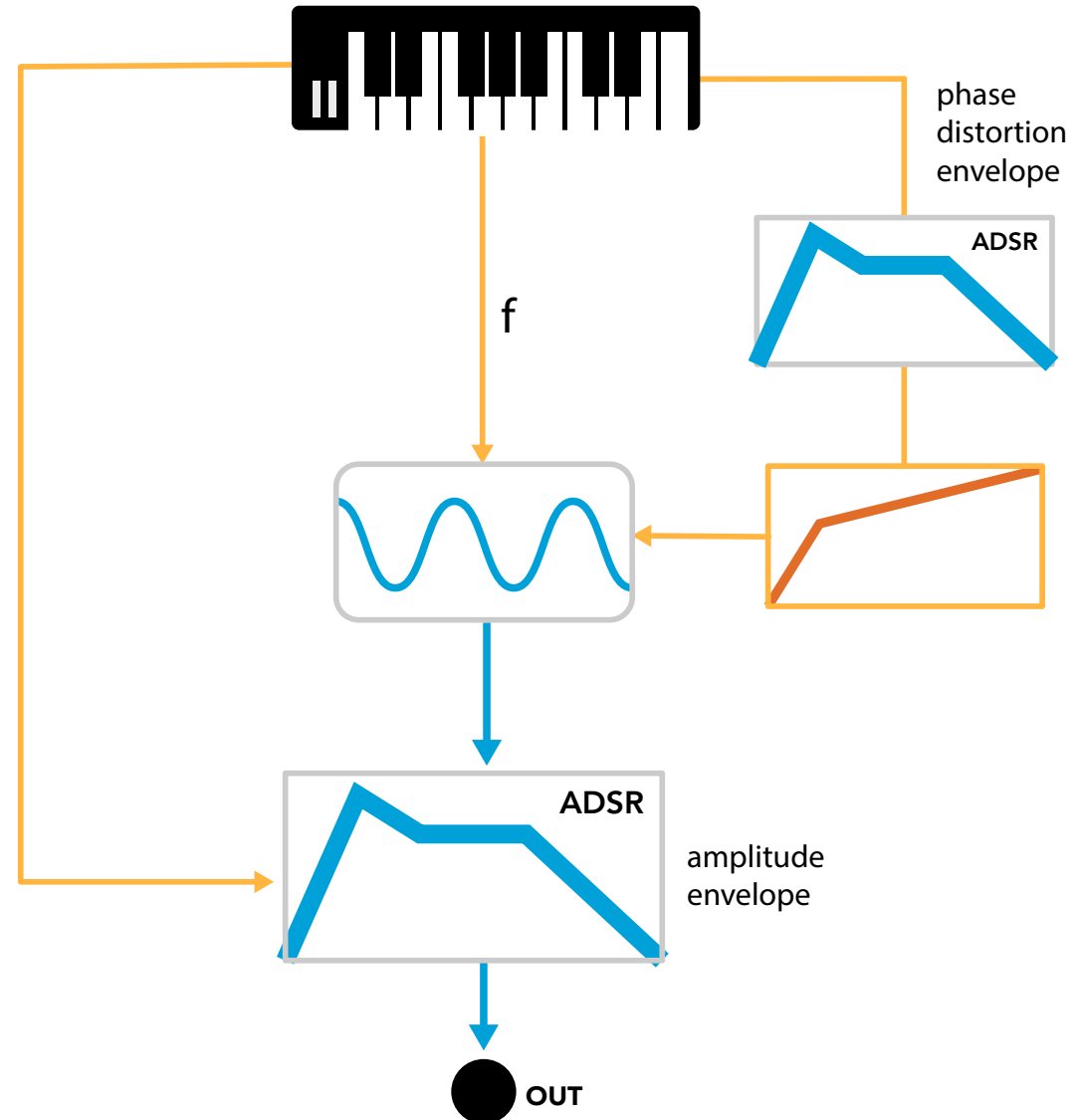
# PHASE DISTORTION

**Phase distortion** is based on reading the values of a sine wave stored in a table at a variable rate. One common phase distortion algorithm reads the wave faster from  $0^\circ$  to  $180^\circ$  (the first half of the full cycle), and then reads slower from  $180^\circ$  to  $360^\circ$  (the second half). The result is that the pitch remains constant on average, but the waveform varies leading to an embellishment of harmonics.



# PHASE DISTORTION

Another important feature is the presence of envelopes for both the table lookup speeds and the final amplitude envelope.





# PHASE DISTORTION



Casio CZ-101 (1984)



# PHASE DISTORTION

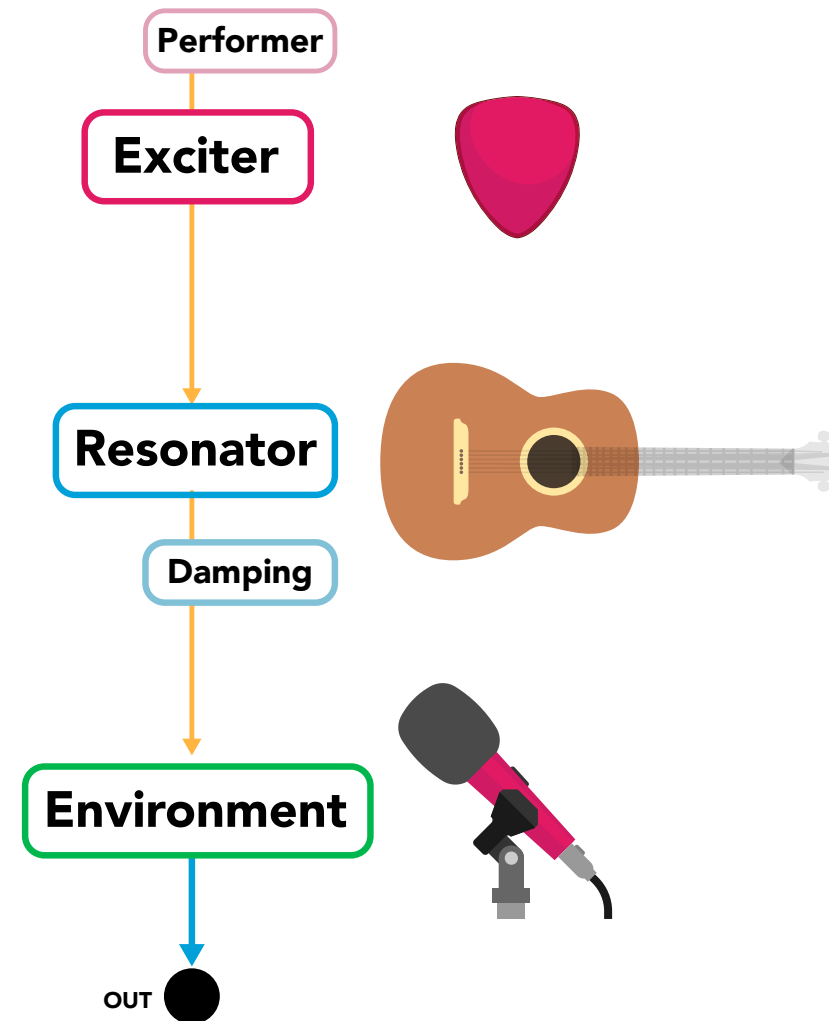


# PHYSICAL MODELING

**Physical modeling synthesis** is not simply a single algorithm, or single concept, but a family of algorithms or concepts that share the same principle:

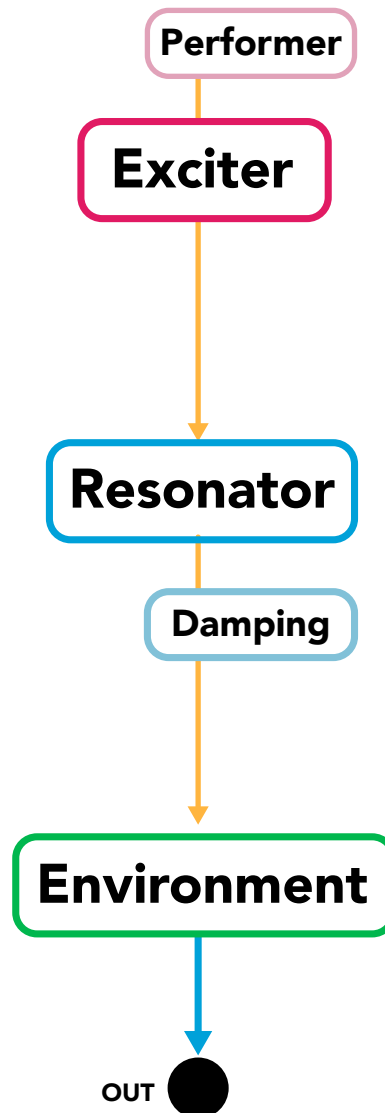
to observe the physical behavior of real acoustic instruments, mathematically model the physical phenomenon, and mimic the physical behavior virtually to recreate the sound.

They usually focus on three stages of sound generation: the **Exciter**, the **Resonator**, and the **Environment**.



# PHYSICAL MODELING

All these stages have a unique set of parameters that, when varied, allow us to model the size, material, and shape of each stage into our algorithm, customizing our final sound.



## Parameters

Physical characteristics performer: mouth, lips, arm...

Type of exciter: plectrum, pizzicato, reed, air column, drumstick...

Size, shape and material of the resonant body.

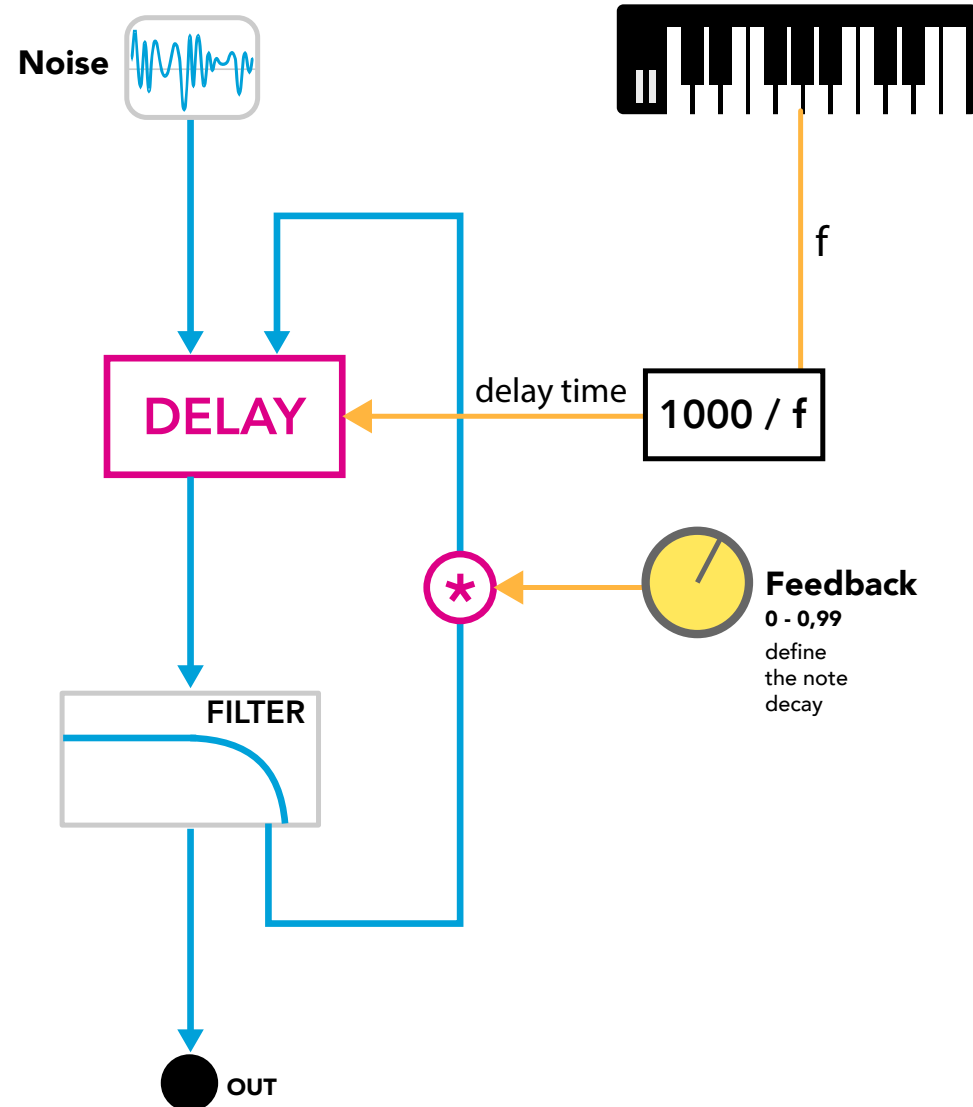
Impedance to sound propagation in the resonator given by size, material and shape...

Type of sound pickup system: pickup, microphone... Distance between microphones, between resonant body and microphones...

# PHYSICAL MODELING

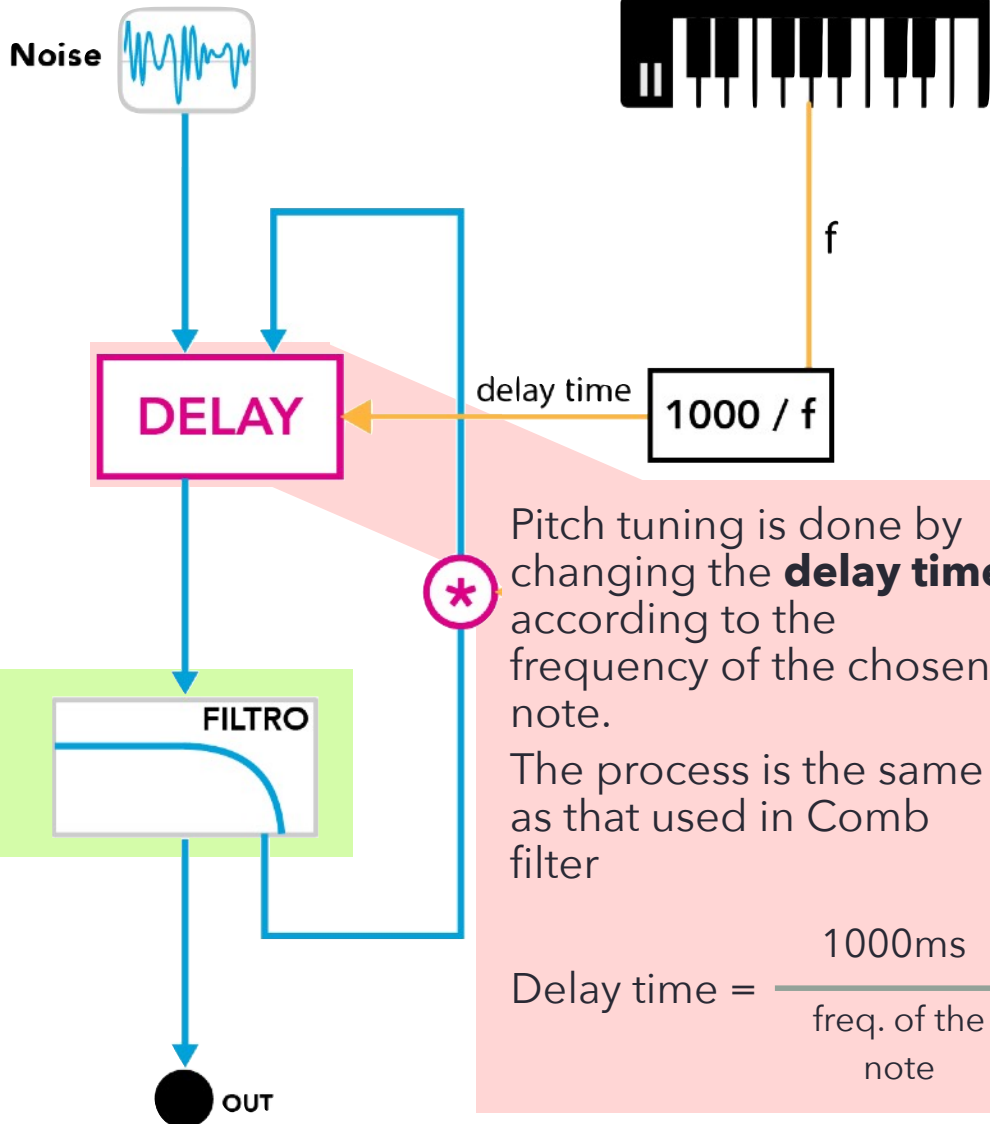
## Karplus - Strong (KS) algorithm

1. White noise is generated
2. This excitation is fed directly into a delay line
3. The output of the delay is input into a filter (usually a first-order low-pass filter)
4. The filtered signal goes to the output and is simultaneously fed back into the delay line, where steps 2-4 are repeated.



# PHYSICAL MODELING

## Karplus - Strong (KS) algorithm



The **low-pass filter (LPF)** removes higher harmonics and creates a sound similar to traditional instruments. As we move away from the fundamental, the harmonics gradually fade.

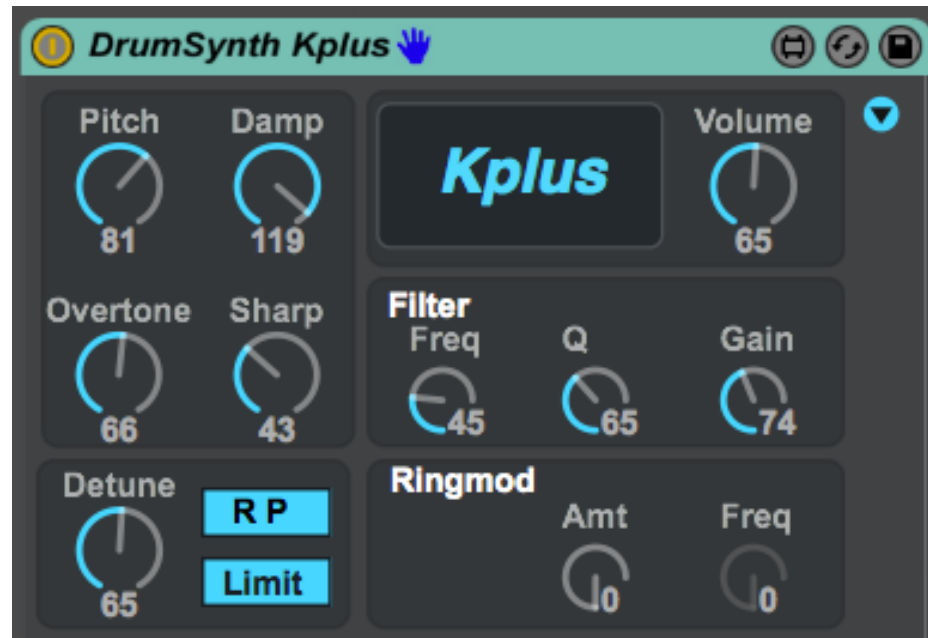
Pitch tuning is done by changing the **delay time** according to the frequency of the chosen note.

The process is the same as that used in Comb filter

$$\text{Delay time} = \frac{1000\text{ms}}{\text{freq. of the note}}$$

# PHYSICAL MODELING

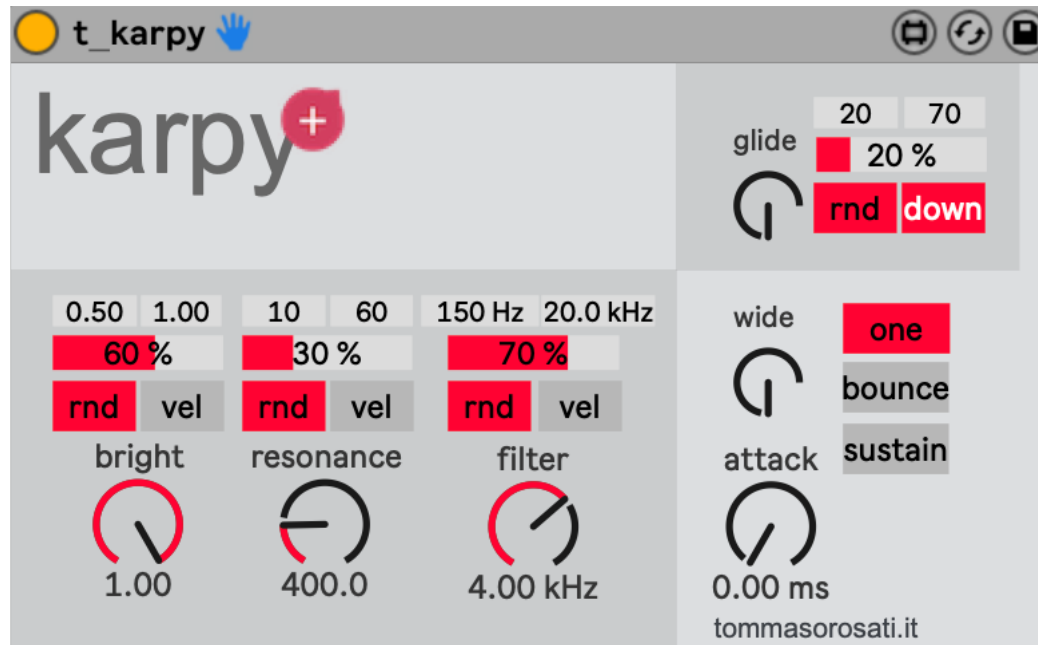
## Karplus - Strong (KS) algorithm



Drum Synth Kplus in Ableton Live

# PHYSICAL MODELING

## Karplus - Strong (KS) algorithm



karpy by piymaxforlive



# PHYSICAL MODELING



YAMAHA VL1 (1993) - Waveguide synthesis

# PHYSICAL MODELING

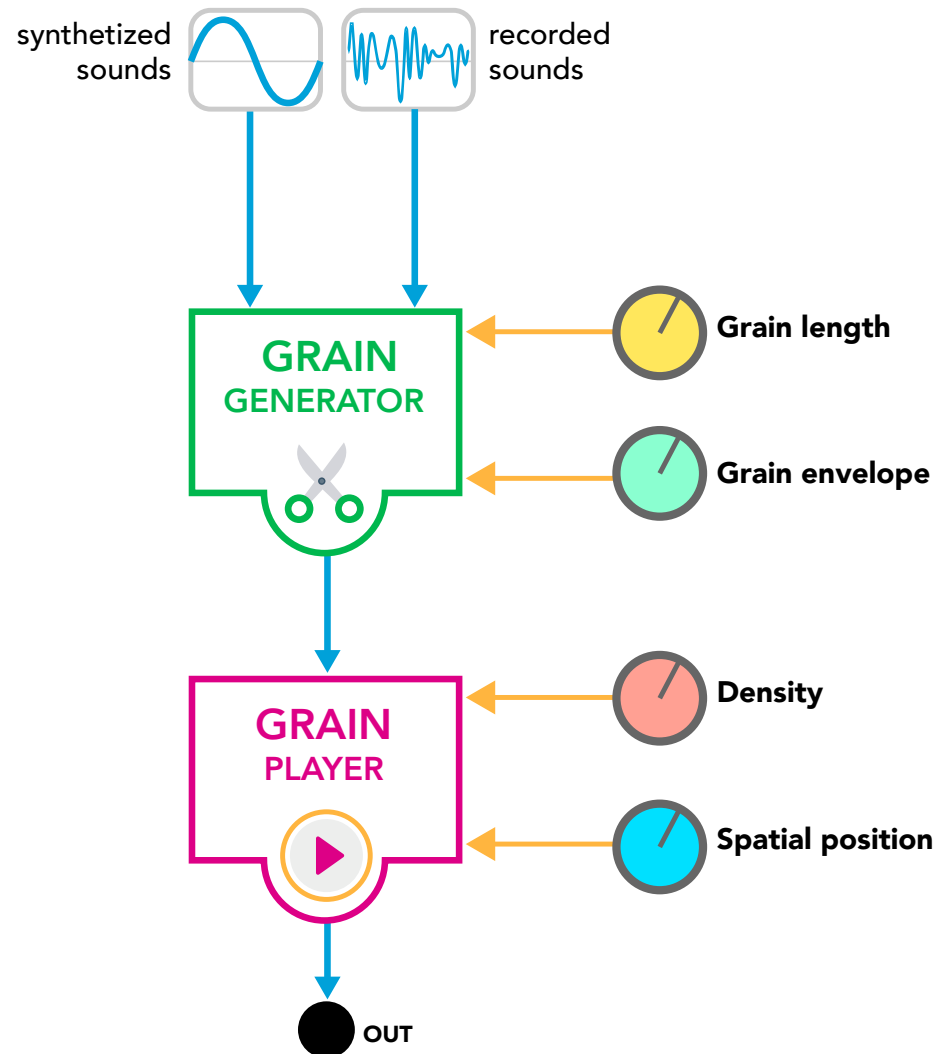


TENSION and COLLISION in Ableton Live

# GRANULAR SYNTHESIS

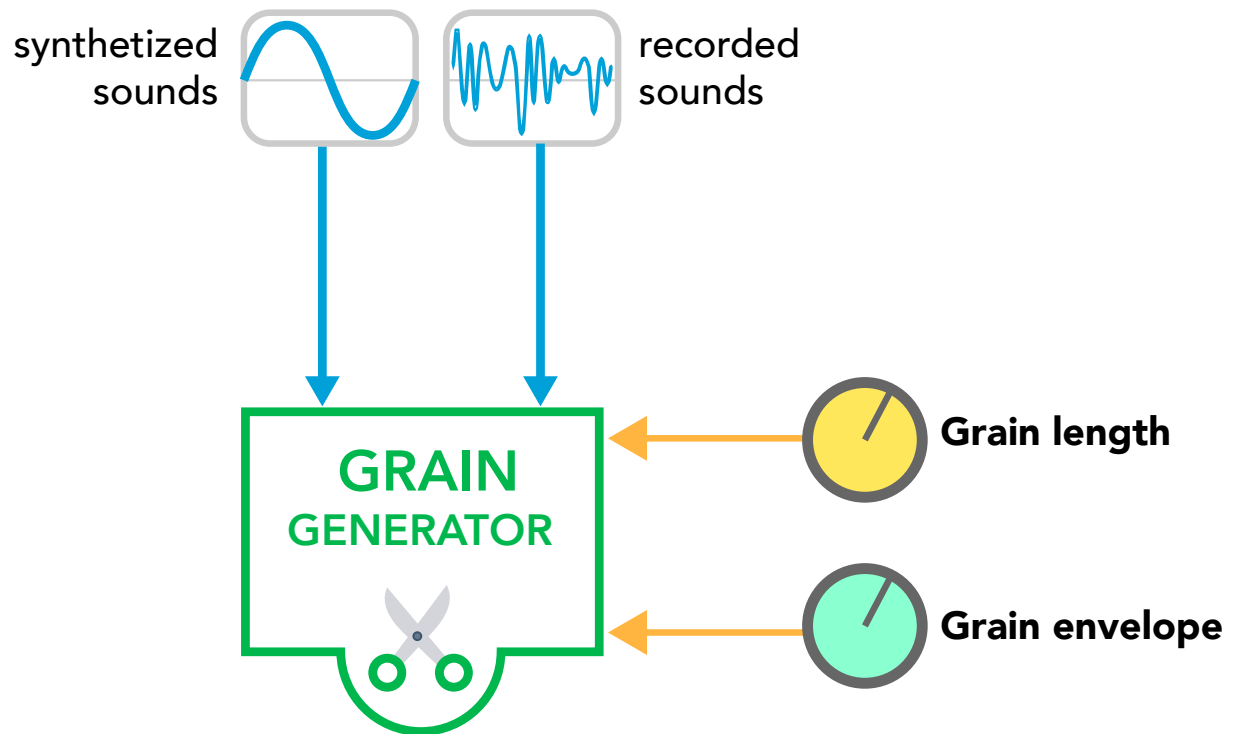
**Granular synthesis** centers on the idea of creating complex sounds from many simple short sounds called grains.

A **grain** is a very small sound fragment that, when combined and played sequentially and/or superimposed at varying speeds, phases, and volumes, generates a single fused timbre.



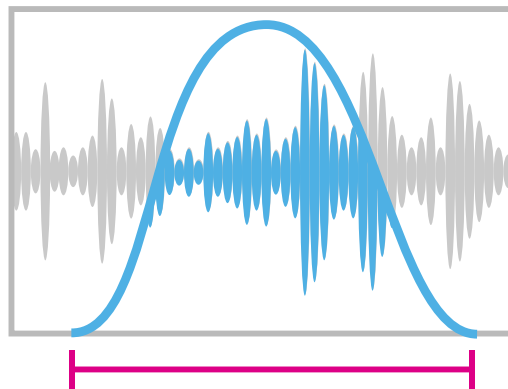
# GRANULAR SYNTHESIS

These sounds grains can be derived from various sources, from recorded sounds to other synthesized sounds.

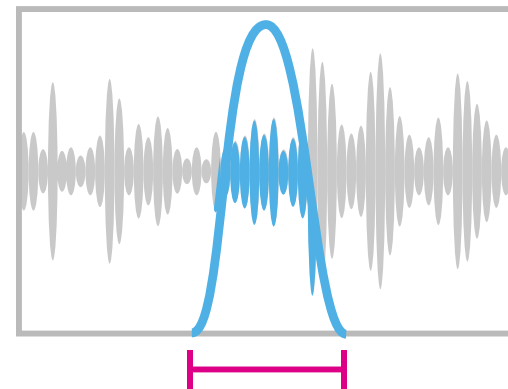


# GRANULAR SYNTHESIS

We set the **length** of the grains (generally between 1 and 100 ms)



grain  
length

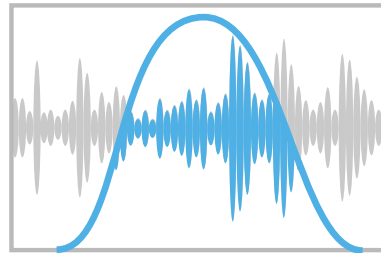


grain  
length

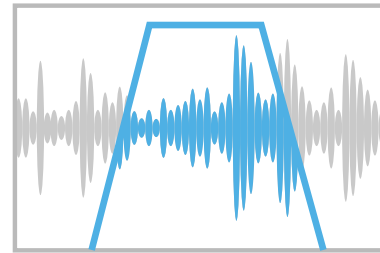
# GRANULAR SYNTHESIS

We apply an amplitude **envelope** to each grain.

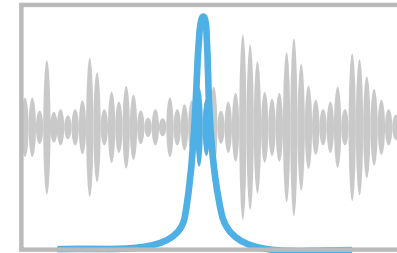
## Symmetrical



**Gaussian**

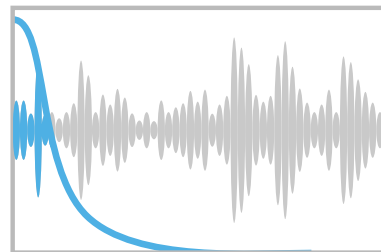


**Trapezoidal**

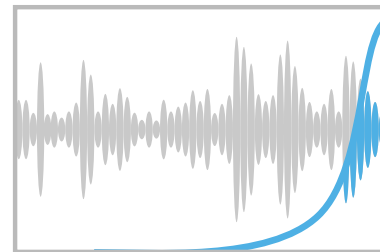


**Pulse**

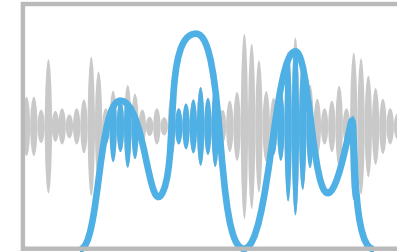
## Asymmetrical



**Fall**



**Rise**



**Noise**



# GRANULAR SYNTHESIS

There are four types of grain playback:

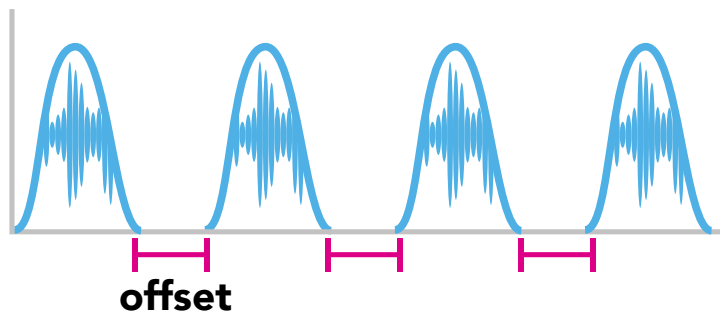
**Synchronous (Synchronous Granular Synthesis):** When playing back a series of grains, the time gap or offset between the grains is constant.

**Quasi-synchronous (QSGS):** The time gap between the grains is almost, but not exactly, constant.

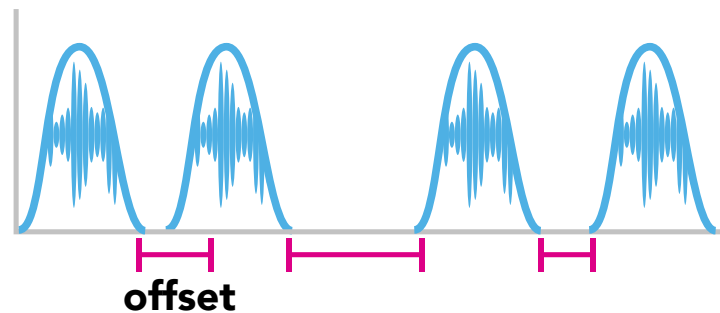
**Asynchronous (ASGS):** The time gap between the grains, during playback, is not constant.

**Pitch-synchronous (Pitch-Synchronous Granular Synthesis):** The time gap between the grains corresponds to a frequency that is synchronized with the pitch of the grains.

## Synchronous



## Asynchronous

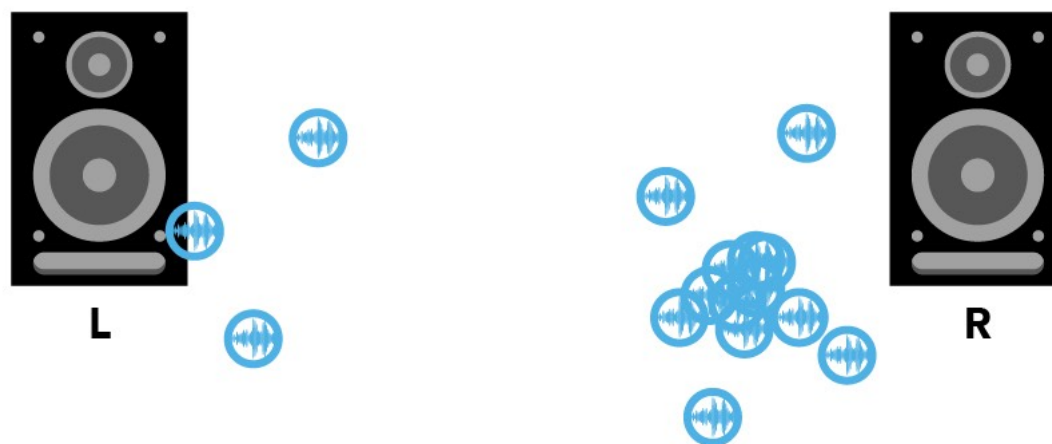




# GRANULAR SYNTHESIS

I can set the following parameters for a grain player:

- **Grain density** is the number of grains that the player plays at the same time.
- **Spatial position** determines where to place the grains the players play within the space.

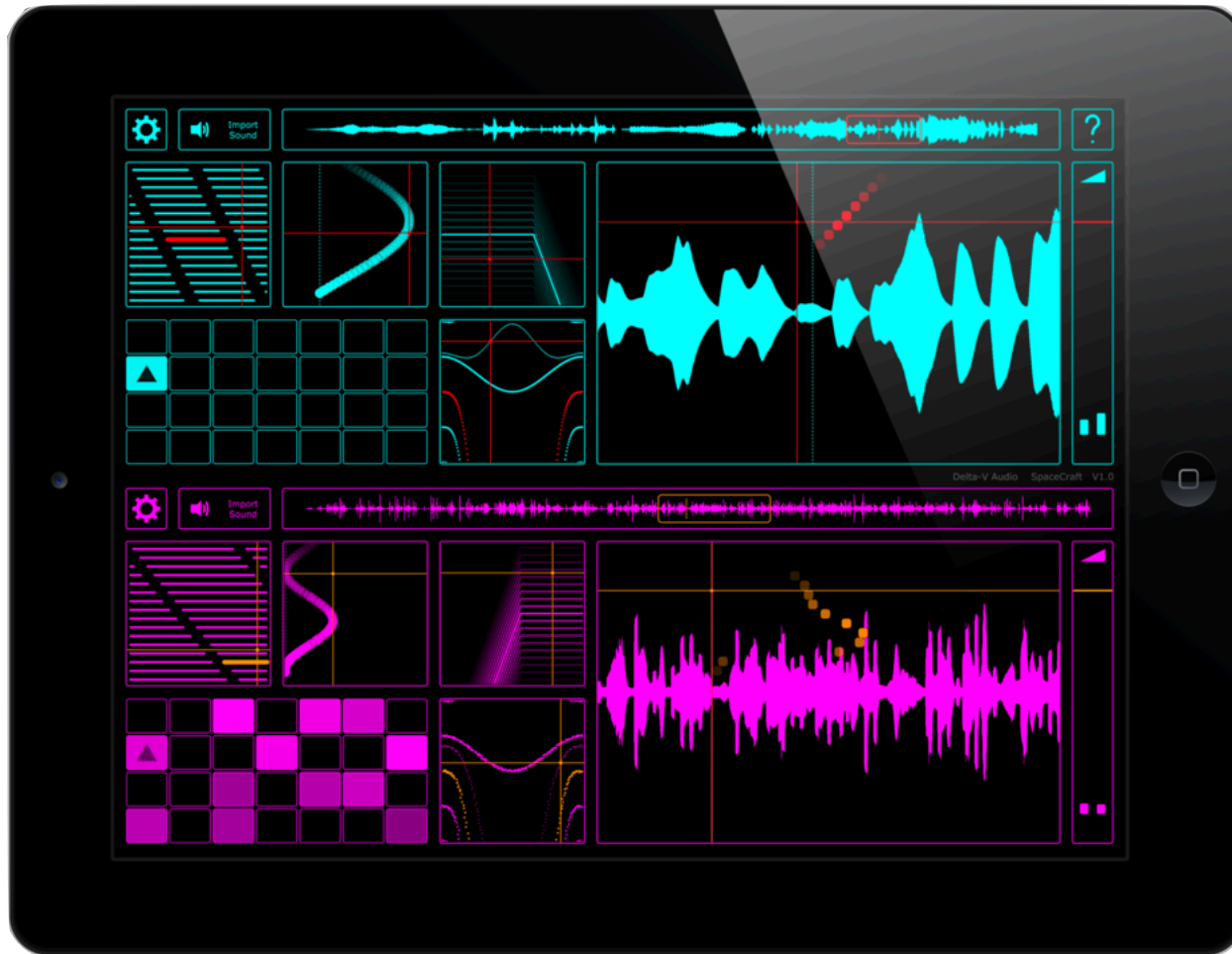


# GRANULAR SYNTHESIS



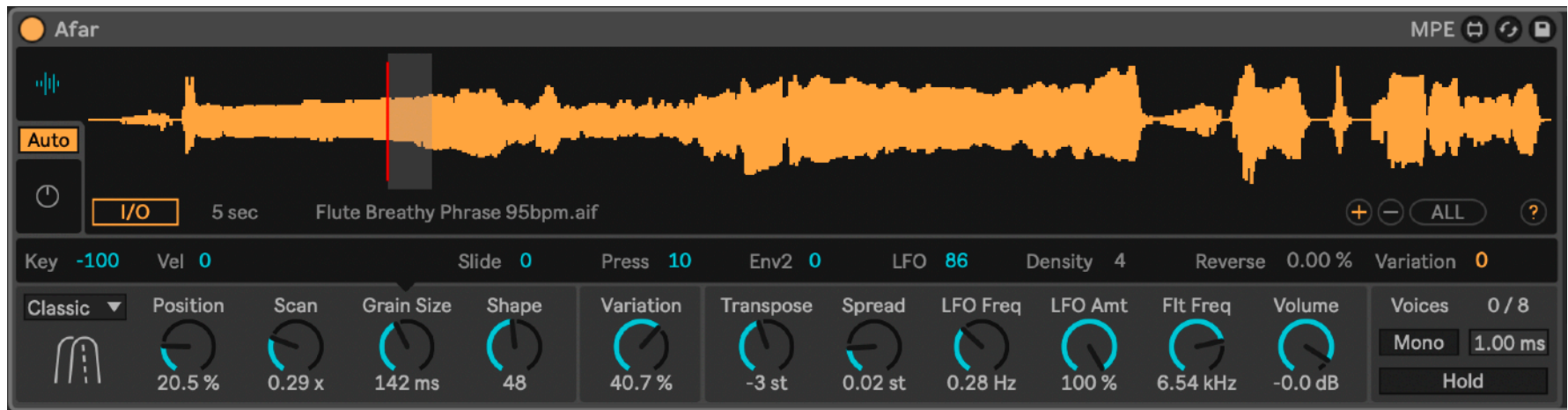
Tasty Chips Electronics GR-1 (2019)

# GRANULAR SYNTHESIS



SpaceCraft app (2018)

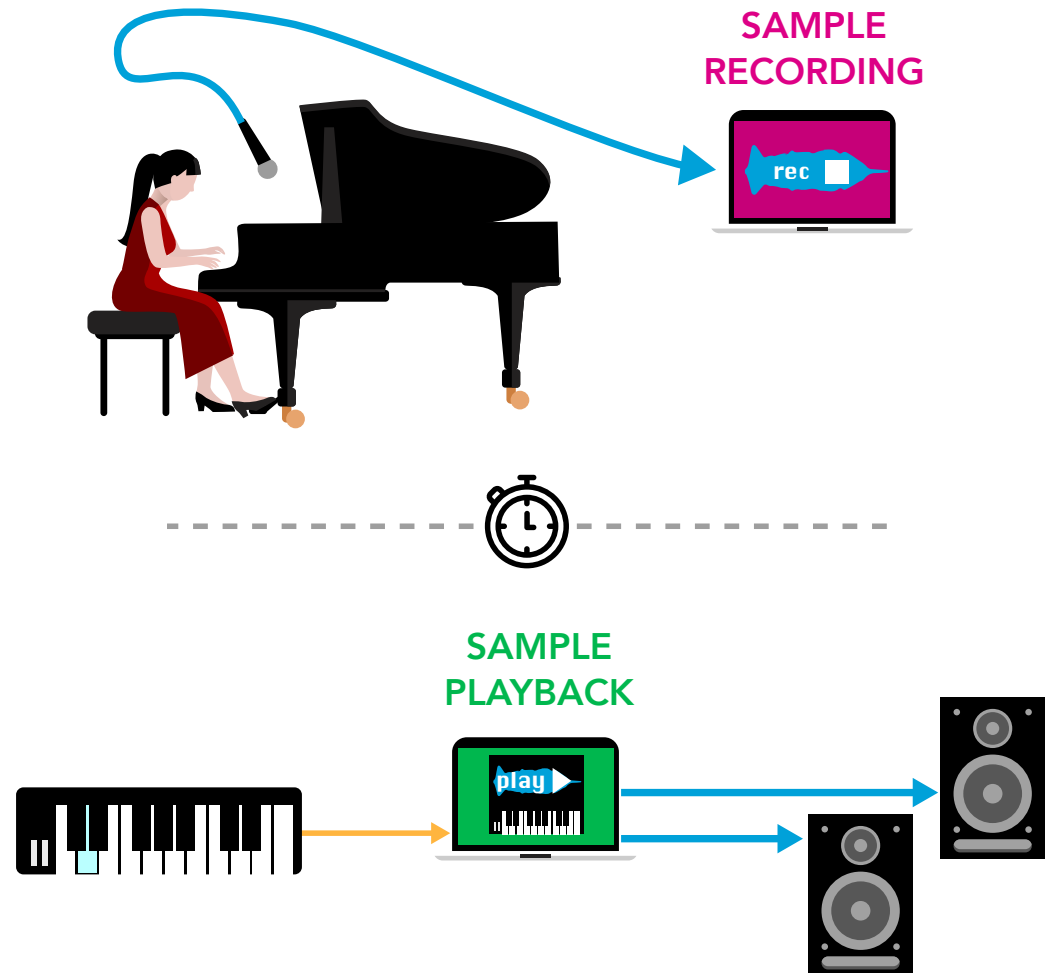
# GRANULAR SYNTHESIS



GRANULATOR III in Ableton Live

# SAMPLING SYNTHESIS

**Sampling synthesis** is based on reproducing previously recorded notes upon pressing, for example, keyboard keys.



# SAMPLING SYNTHESIS

1

## Sample recording

- **Single-sampled**

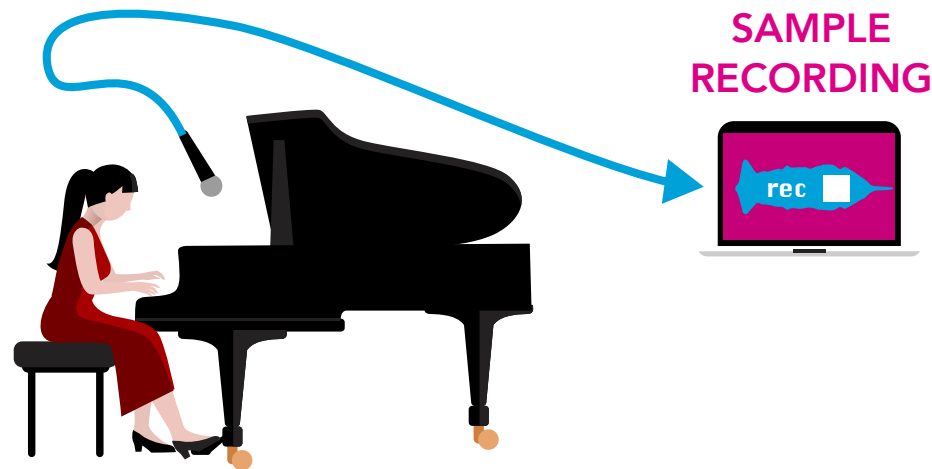
one sample is recorded, which will be used, during playback, to generate all the notes of the instrument

- **Multi-sampled**

a sample is recorded for each note of the instrument

- **MultiLayer-sampled**

not only is every note of an instrument recorded, but each note is also recorded at different dynamics.

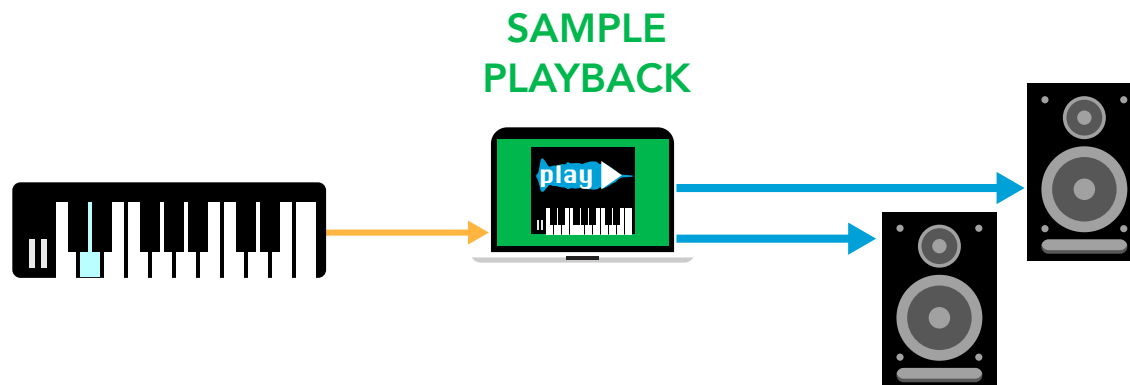


# SAMPLING SYNTHESIS

2

## Sample playback

In a Single-sampled,  
We tell our sampler what note we have recorded as a sample. This way the software takes charge of transposing the note in case other keys are pressed.



# SAMPLING SYNTHESIS

allows you to choose the start and end point of the sample during playback

## Crop



allows you to superimpose an amplitude envelope to each sample

## ADSR



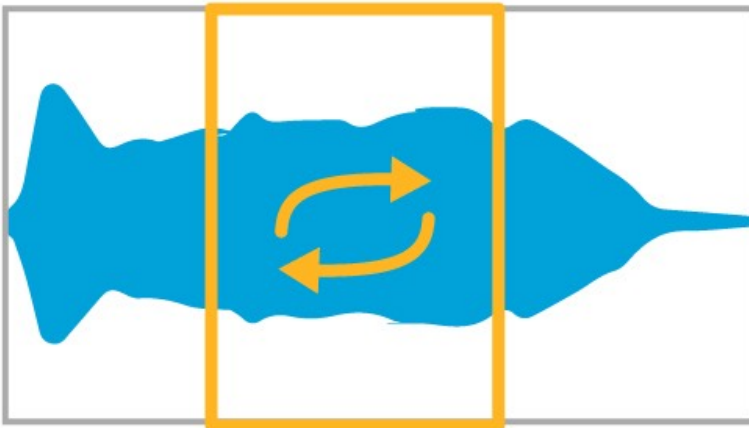


# SAMPLING SYNTHESIS

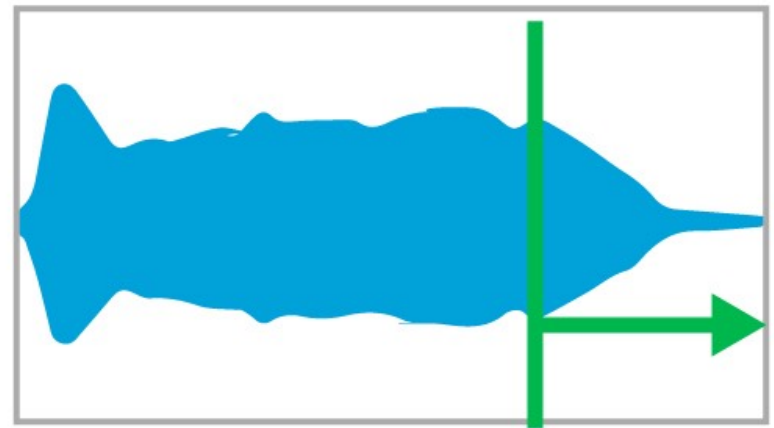
It's the portion of the sample that plays it back continuously as long as my key is pressed and sustained

marks the time point within the sample where to go after we release the key

**Loop**

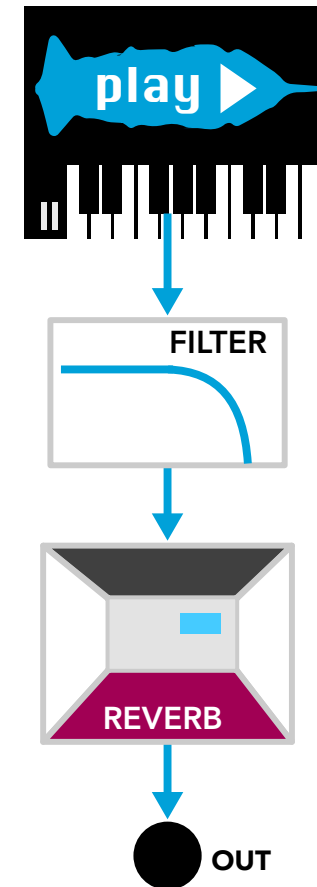


**End point**

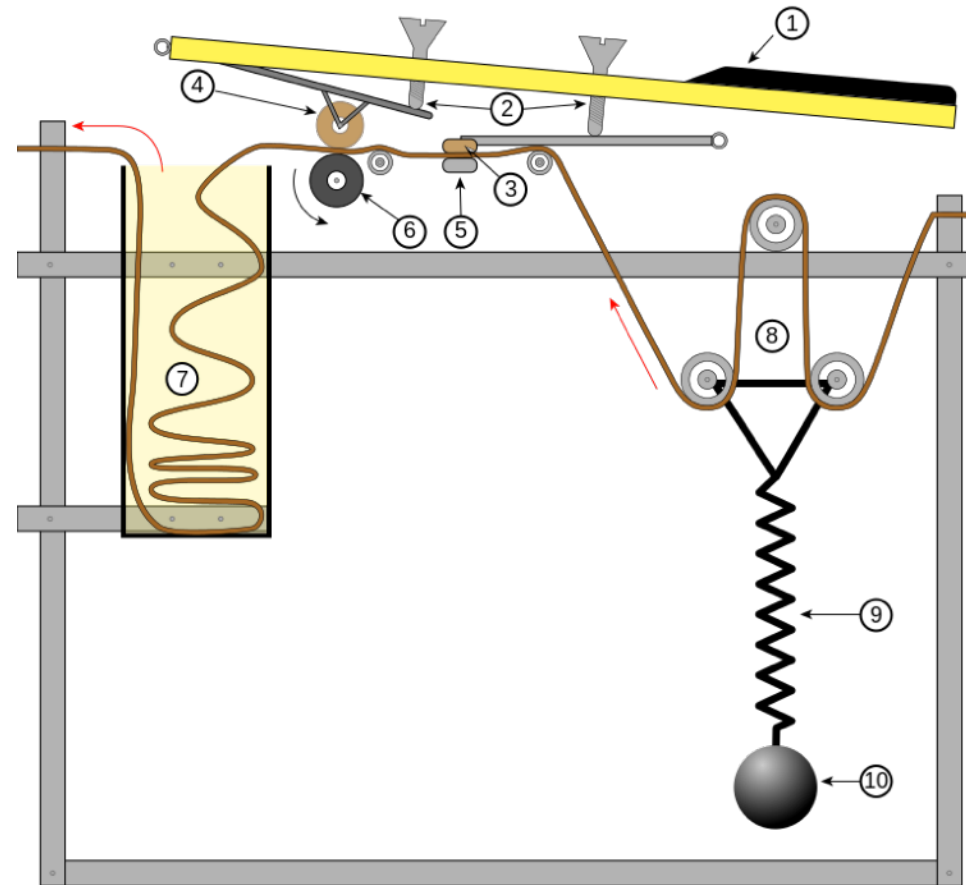


## SAMPLING SYNTHESIS

At the end of the Sampler's workflow, as with other types of synthesizers, we can find one or more **effects** that helps refines the sound we just created.



# SAMPLING SYNTHESIS



MELLOTRON M400 (1970)

# SAMPLING SYNTHESIS



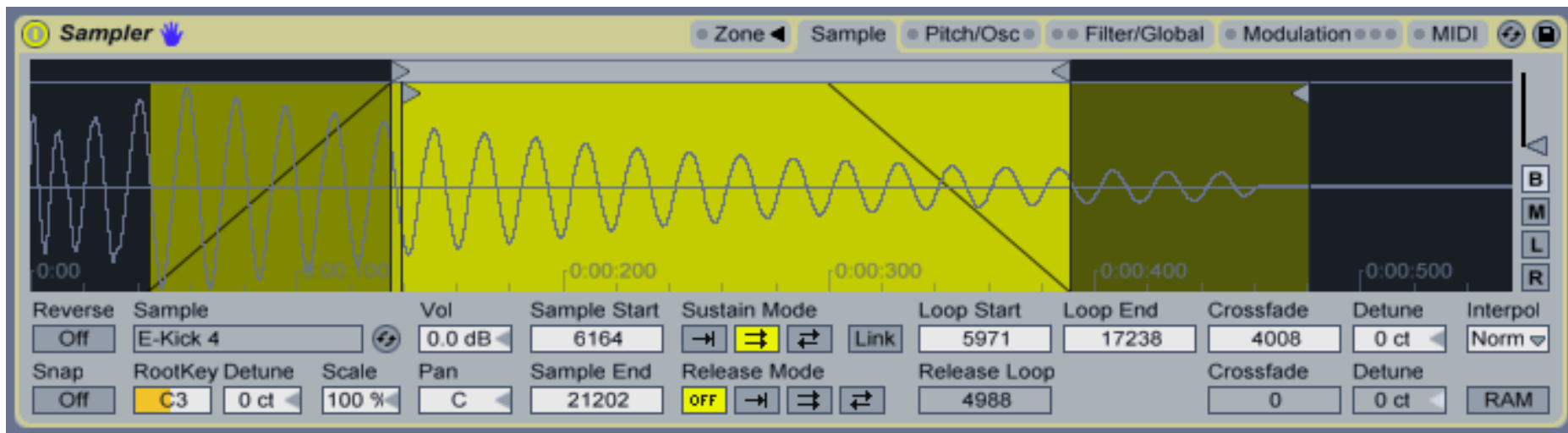
NORD PIANO 4 - Nord (2019)

# SAMPLING SYNTHESIS



KONTAKT by Native Instruments

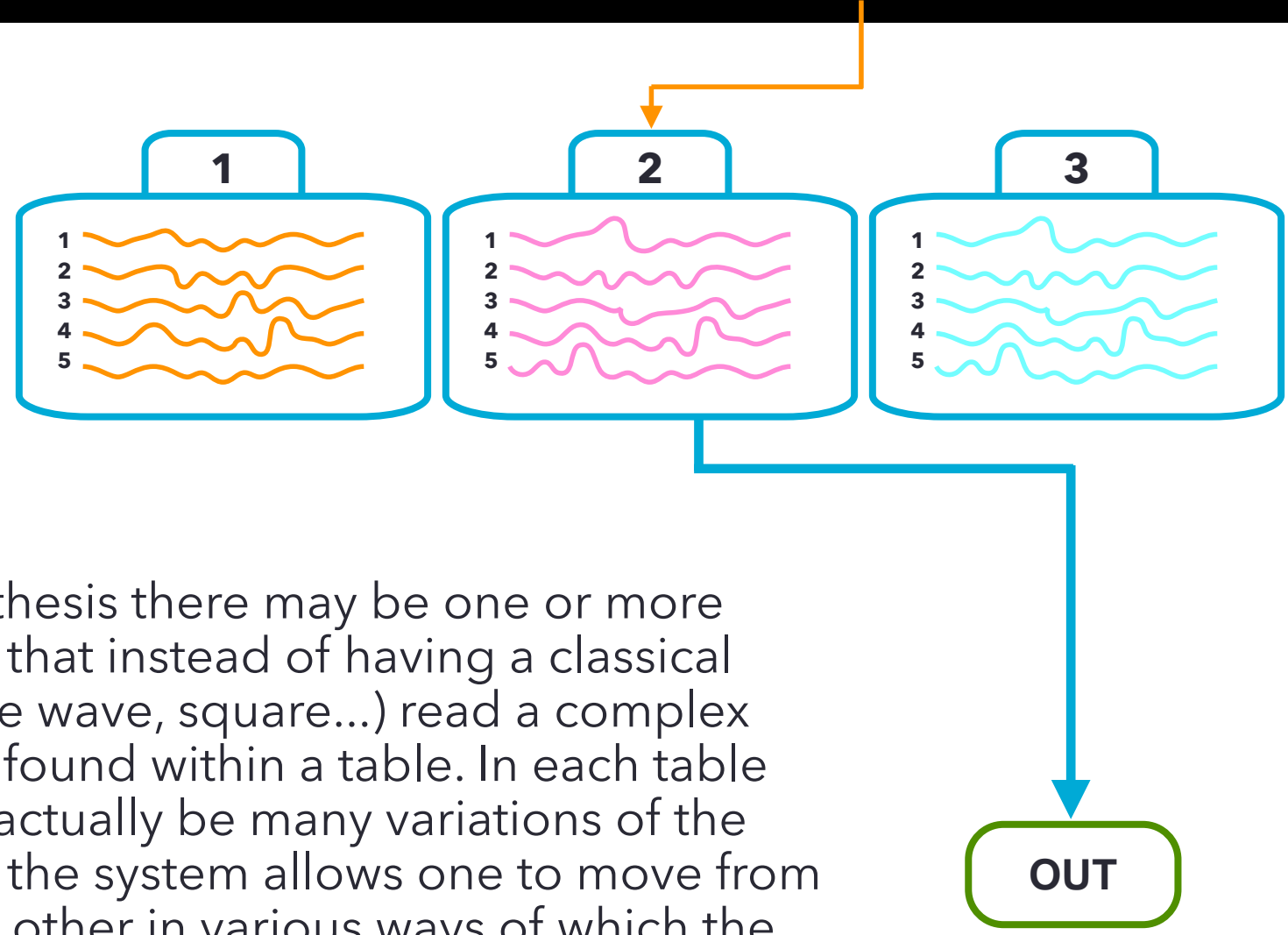
# SAMPLING SYNTHESIS



SAMPLER in Ableton Live



# WAVETABLE SYNTHESIS



In this synthesis there may be one or more oscillators that instead of having a classical shape (sine wave, square...) read a complex waveform found within a table. In each table there can actually be many variations of the wave, and the system allows one to move from one to the other in various ways of which the most important are: stepped and continuous.



# Sintesi Wavetable

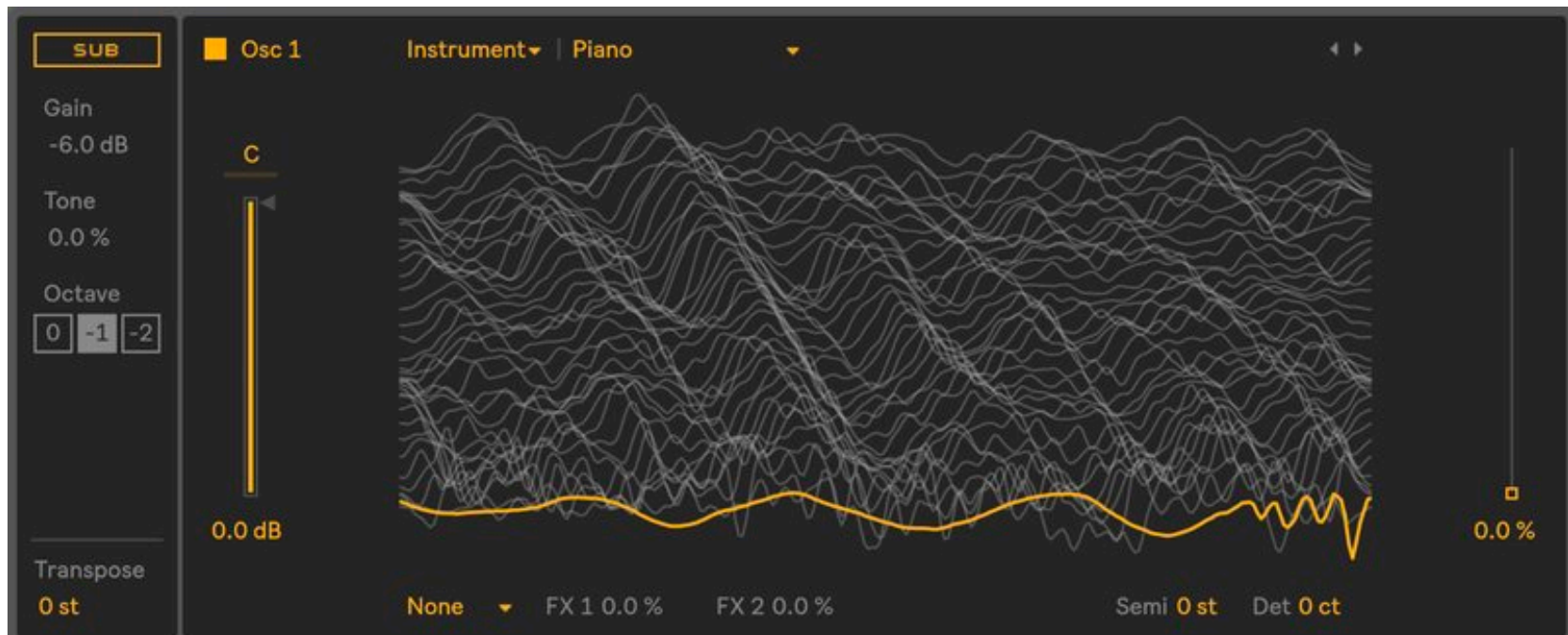


PPG WAVE 2.2 (1981)



KORG WAVESTATION (1990)

# WAVETABLE SYNTHESIS



WAVETABLE in Ableton Live



[www.tommasorosati.it](http://www.tommasorosati.it)