

SOUND

WHAT IS SOUND?

- Waves (vibrations/disturbances) traveling through a medium (air, water, bone)
- Wave types
 - Impulse – single pulse
 - Noise – random wave, ongoing but no repetition
 - Pitched – repeating (oscillating waveform)
 - Sine waves/sinusoidal – simplest, building blocks for more complex sounds

SOUND WAVES

- Amplitude – How compressed the peaks are (distance between them) – related to loudness
- Wavelength (λ) – How far apart the peaks are
- Speed (c) – How fast the peaks move through the medium
 - In air – 343 meters/second, 768 mph, about 5 seconds per mile (at 20°C)
 - In water – 1484 m/s, about 4.32 times faster
 - In bone – 2650 m/s (Why bone?)
- Frequency ($f = c / \lambda$) in Hertz (Hz) (cycles/sec) – related to pitch
- Concert A = 440 Hz (to tune an orchestra)
 - $\lambda = c / f = (343 \text{ meters/sec}) / (440 \text{ Hertz}) = 0.78 \text{ meter}$
 - f stays put in different mediums $\Rightarrow \lambda$ changes with c

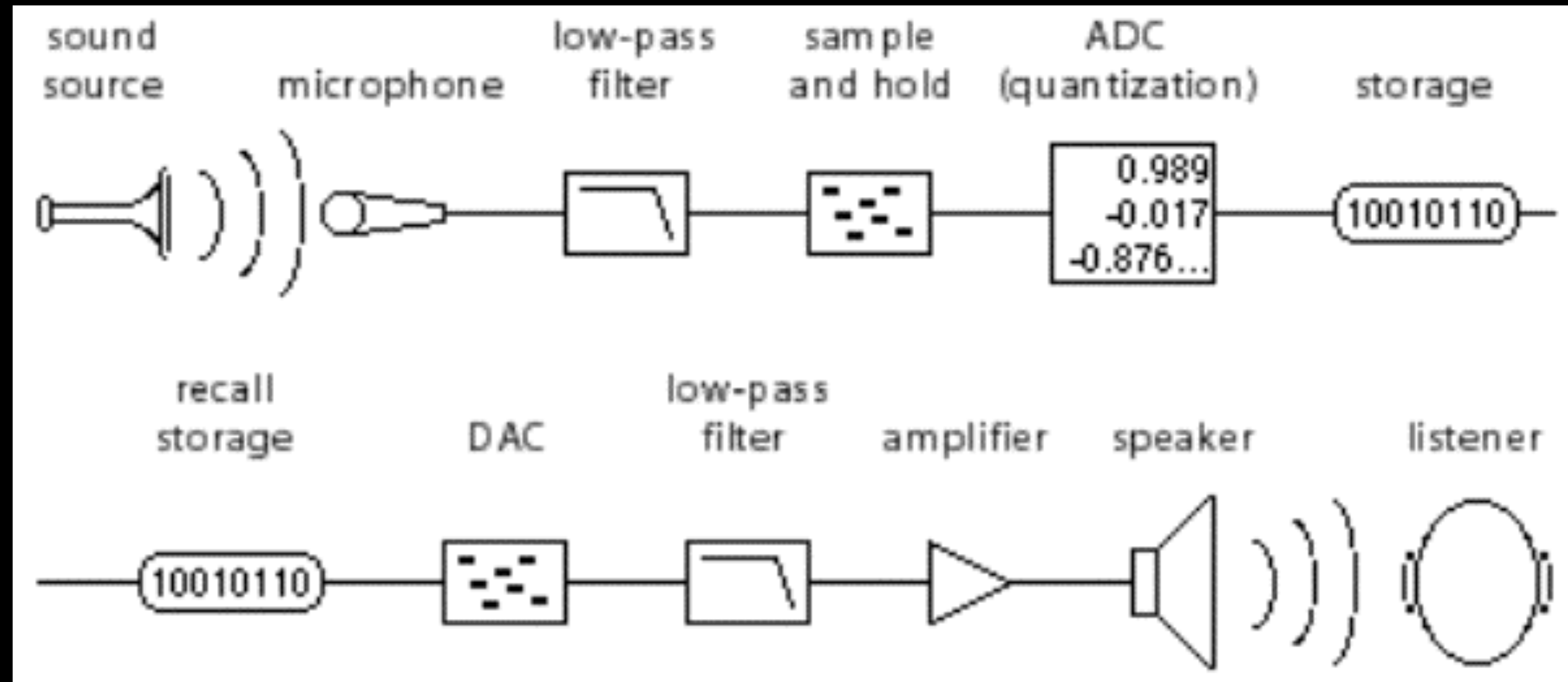
DIGITAL AUDIO

Capture, storage and retrieval of sound usually happens digitally (these days)

- Digital – Variations in air pressure are converted to voltages then **sampled** at discrete times and stored on tape, disk, flash memory as a sequence of discrete numbers - conversion of a continuous-time signal to a discrete-time signal.
- Mechanical (Analog) – Needles making scratches on moving tinfoil, wax, shellac, vinyl
- Electric (Analog) – Continuous voltage curves stored as magnetic fields on moving wire, cylinder, tape

DIGITAL AUDIO

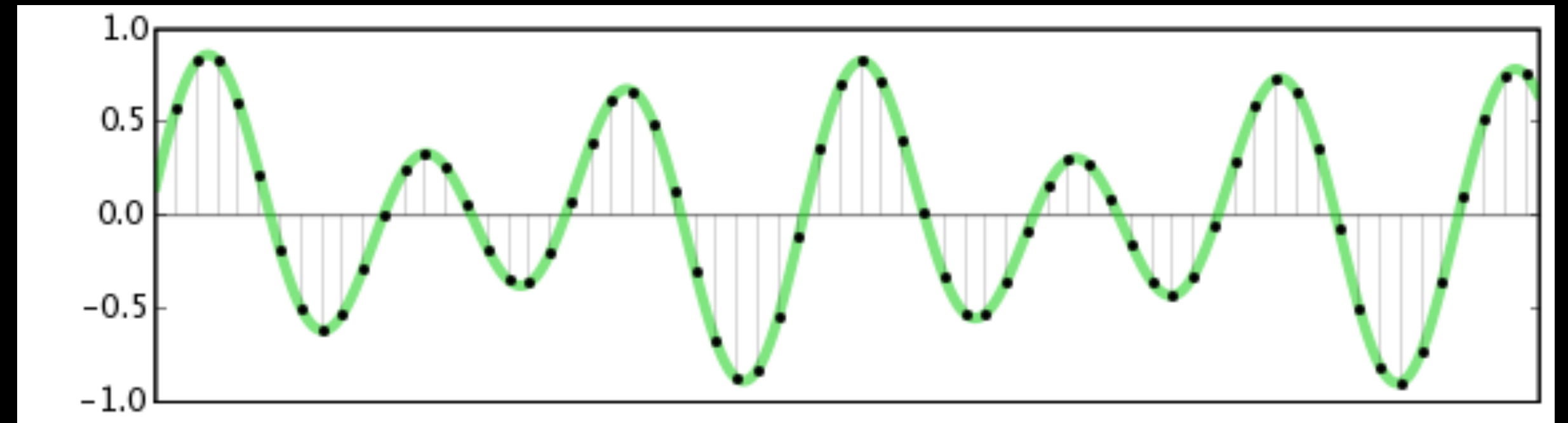
- A2D / D2A



More info: https://docs.cycling74.com/max8/tutorials/02_mspdigitalaudio

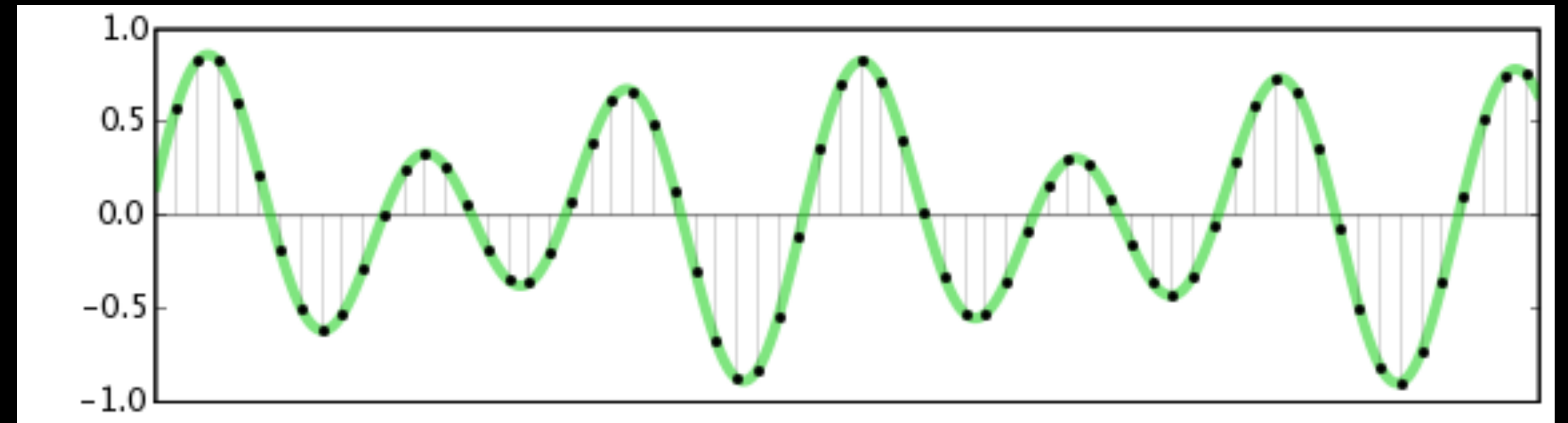
DIGITAL AUDIO

- Sampling:
 - Sampling Rates: how often you are taking "snapshots" of the signals (44.1kHz, 48kHz, 96kHz, etc)
- Bit Depth: number of bits of information in each sample (resolution). 16-bit (CD), 24-bit (DVD/Blu-ray). Affects dynamic range.



DIGITAL AUDIO

- Sampling:
- An audio sample is simply a small portion of stored sound (e.g. from a file or a data structure like an array) that can be accessed, played back and otherwise manipulated.



Processing Sound library uses the
SoundFile and AudioSample
classes for this

SOUND SYNTHESIS

Sound synthesis is the technique of generating sound “from scratch”, using electronic (analog or digital) hardware or software

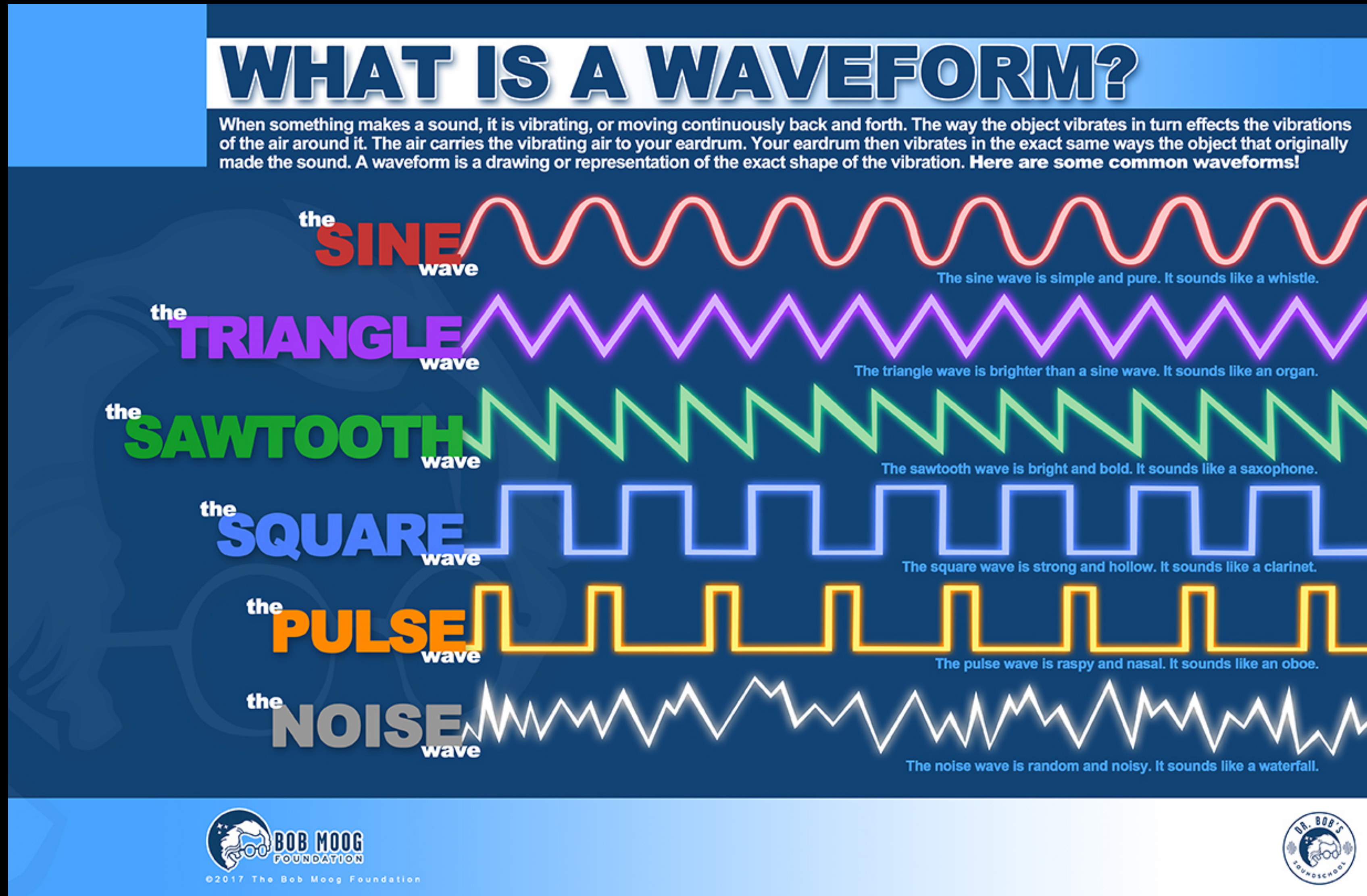
- Many types:
 - Subtractive
 - Additive
 - Granular
 - FM (Frequency Modulation)
 - Wavetable
 - and many more...

SOUND SYNTHESIS

Components of a typical sound synthesis system

- Oscillator/Waveforms
- Envelope generator
- Filter
- LFO (Low frequency oscillator, acts as modulator)
- Amplifier

SOUND SYNTHESIS



Oscillator: generates these repeating or periodic waveforms

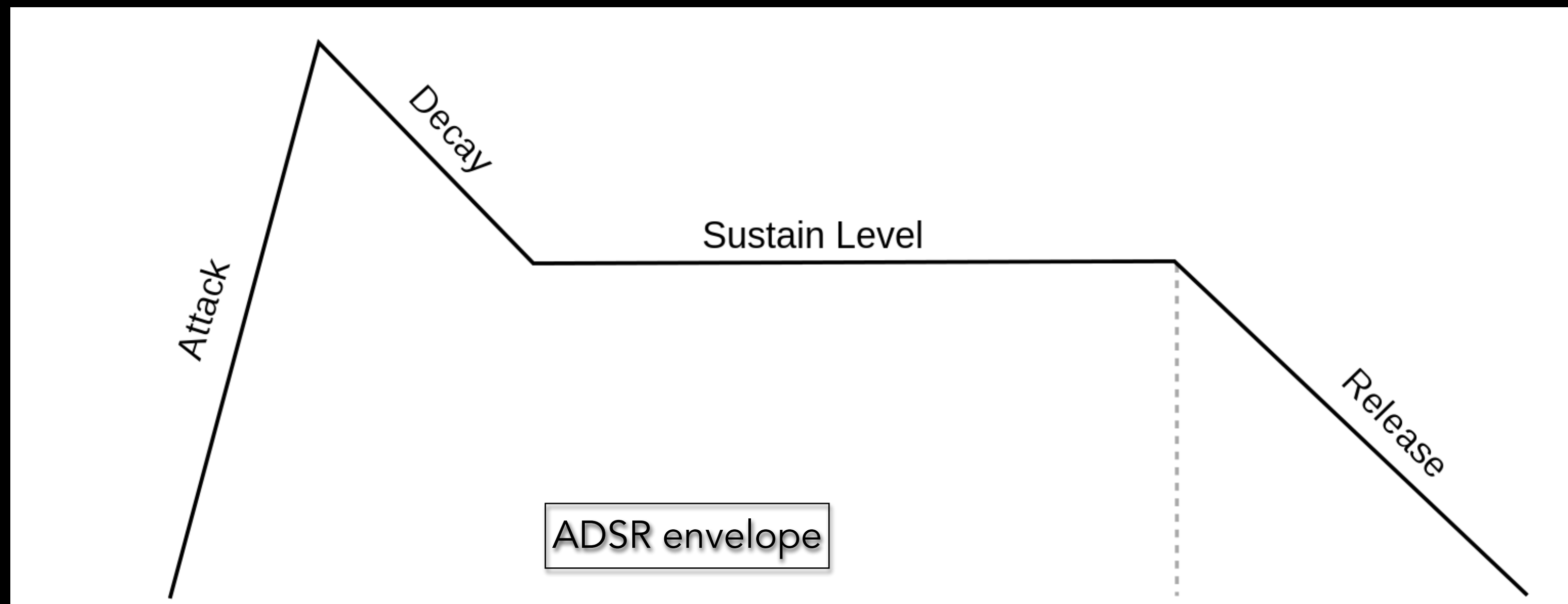
Processing & p5.js Sound libraries have classes for generating all of these

← Arduino tone() function produces these only

SOUND SYNTHESIS

Envelope: control amplitude shape over time.

Can be applied to volume (e.g. of an amplifier, filter cutoff and more)



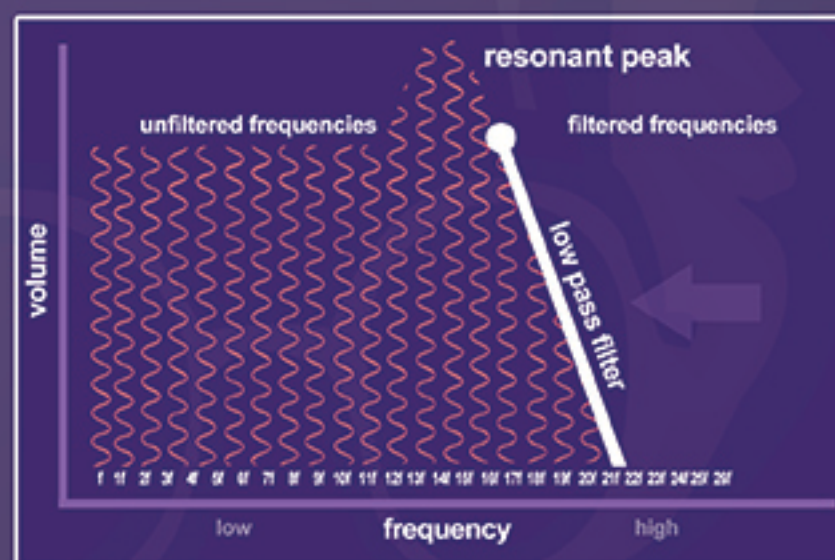
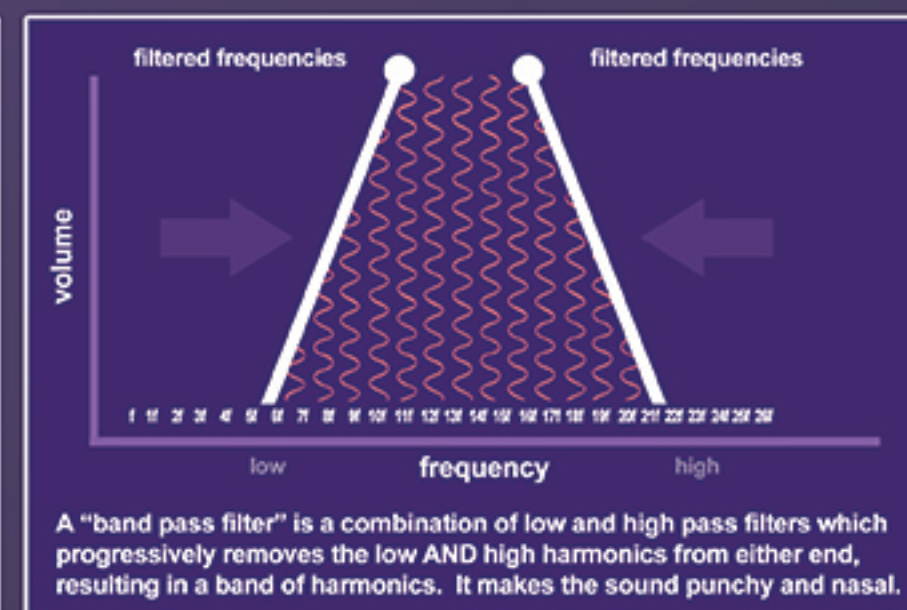
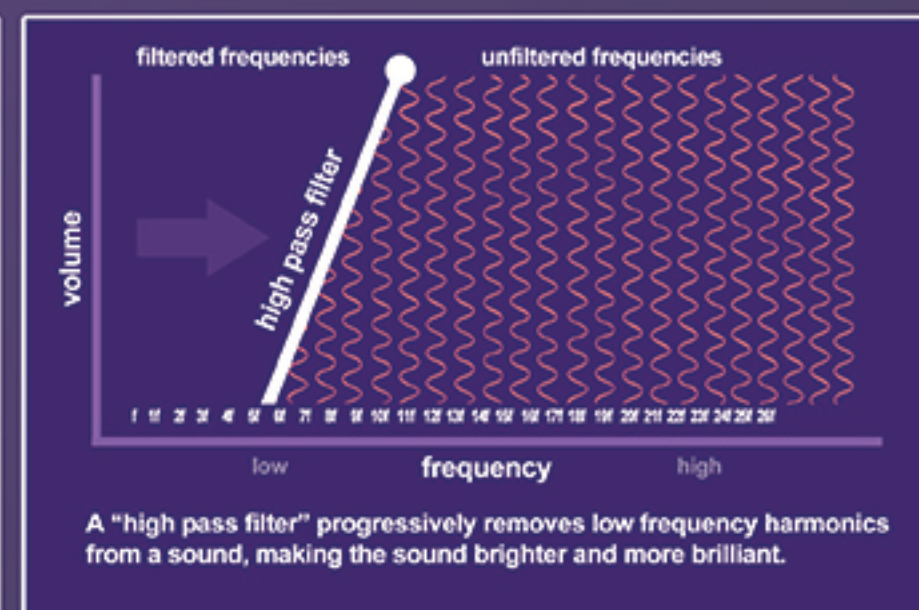
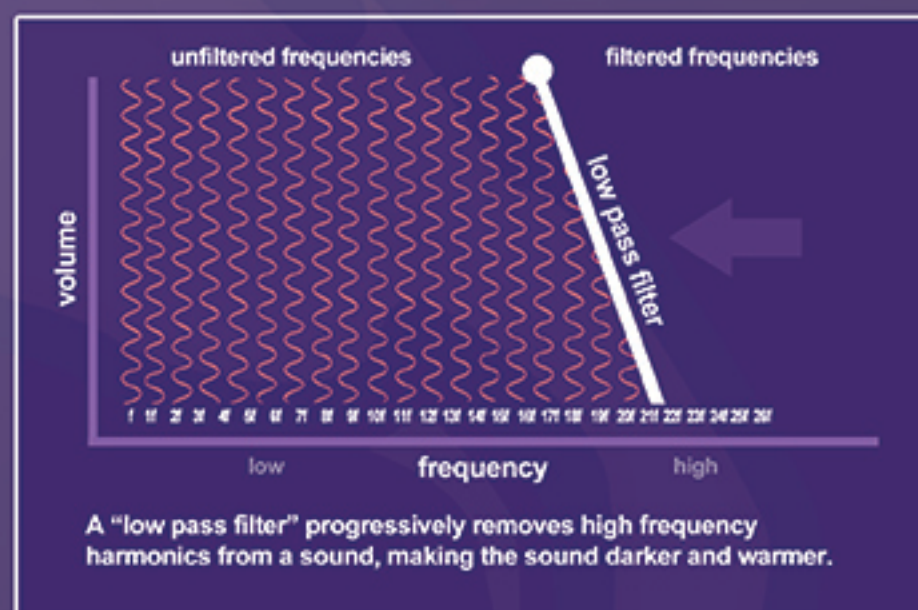
note: Processing sound library only offers ASR envelope

SOUND SYNTHESIS

WHAT IS A FILTER?

A filter is an electronic component which helps the user shape the timbre (tone color) of a musical tone by changing its harmonic content. Any musical tone is made up of the note you hear and a lot of other notes you don't hear, which are called "harmonics." They define the timbre (tone color) of the sound. The filter affects them in ways that change the timbre of your musical tone.

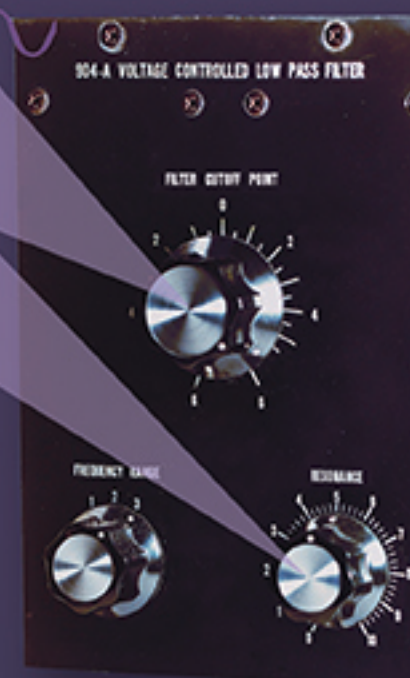
The purpose of the filter is to remove or accentuate harmonics in the tones produced by the oscillator(s). The oscillator produces tones which are full of harmonics. By removing some of these harmonics with a filter, we can attain a wide variety of timbres.



The **"filter cutoff point"** is the point where harmonics begin to be affected by the filter. It is represented in the graphics above by a white ball. It allows you to control how many of the harmonics are filtered out by the filter.

The **"resonance"** function amplifies the harmonics that exist at the cutoff point of the filter. The result is a more intense sound with a boost around the mid-range (depending where you have the cutoff point set). Sometimes the boost accentuates a certain frequency so much that it begins to become audible, resulting in a sine wave tone. That tone can be tuned by adjusting the cutoff frequency to the pitch you want.

The filter's cutoff point can be controlled by other parts of the synthesizer, resulting in harmonic content changing over time. Synthesizer filters often have controls which allow the synthesizer's envelope, LFO, or even oscillator "modulate (change over time)" the cutoff point of the filter.



Filter: shapes timbre (tone color) by altering spectral/harmonic content (typically by removing certain range of frequencies)

Processing and p5.js Sound libraries have classes for lowpass, highpass & bandpass filters

SOUND SYNTHESIS

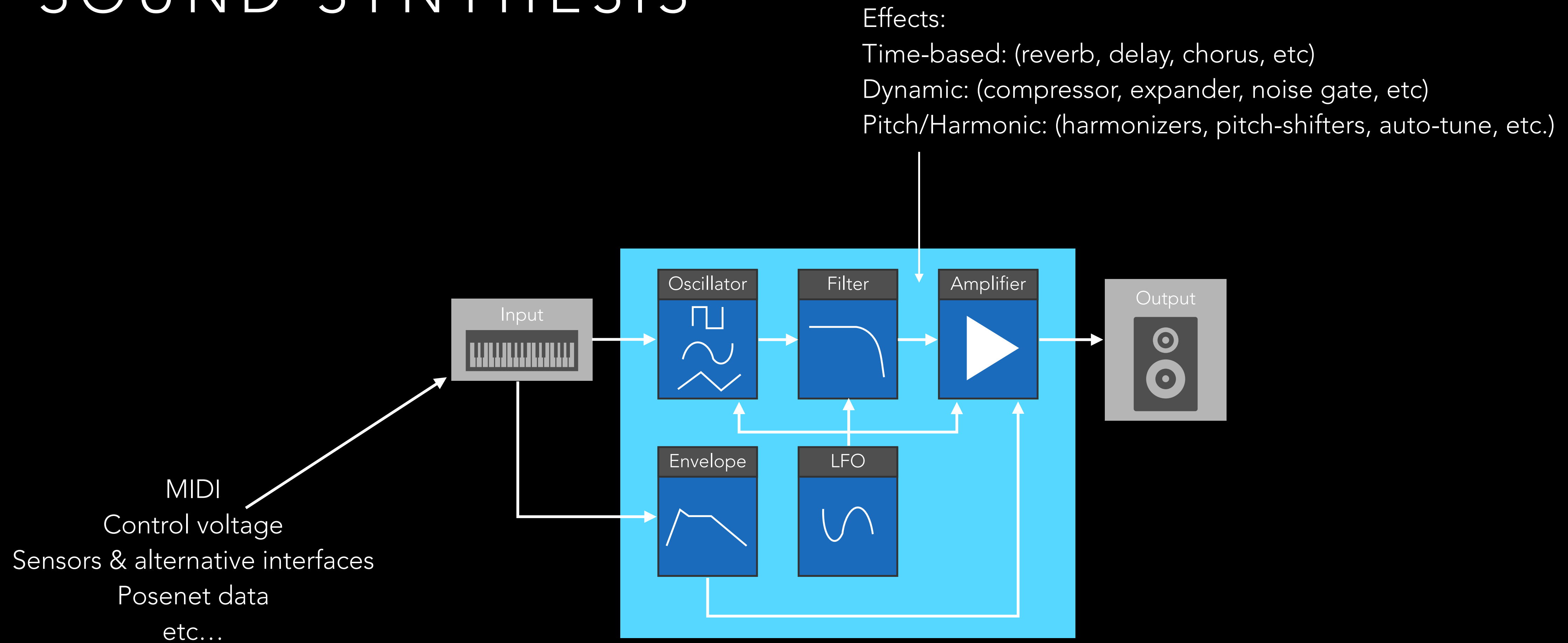
LFO (Low Frequency Oscillator): repeating waveform of very low pitch (usually below range of human hearing) used to alter the shape or “modulate” the pitch or amplitude of your sound or some other part of your synth.

Basically you are using a waveform oscillator not as a sound source but a modulation source

SOUND SYNTHESIS

Amplifier: Volume control

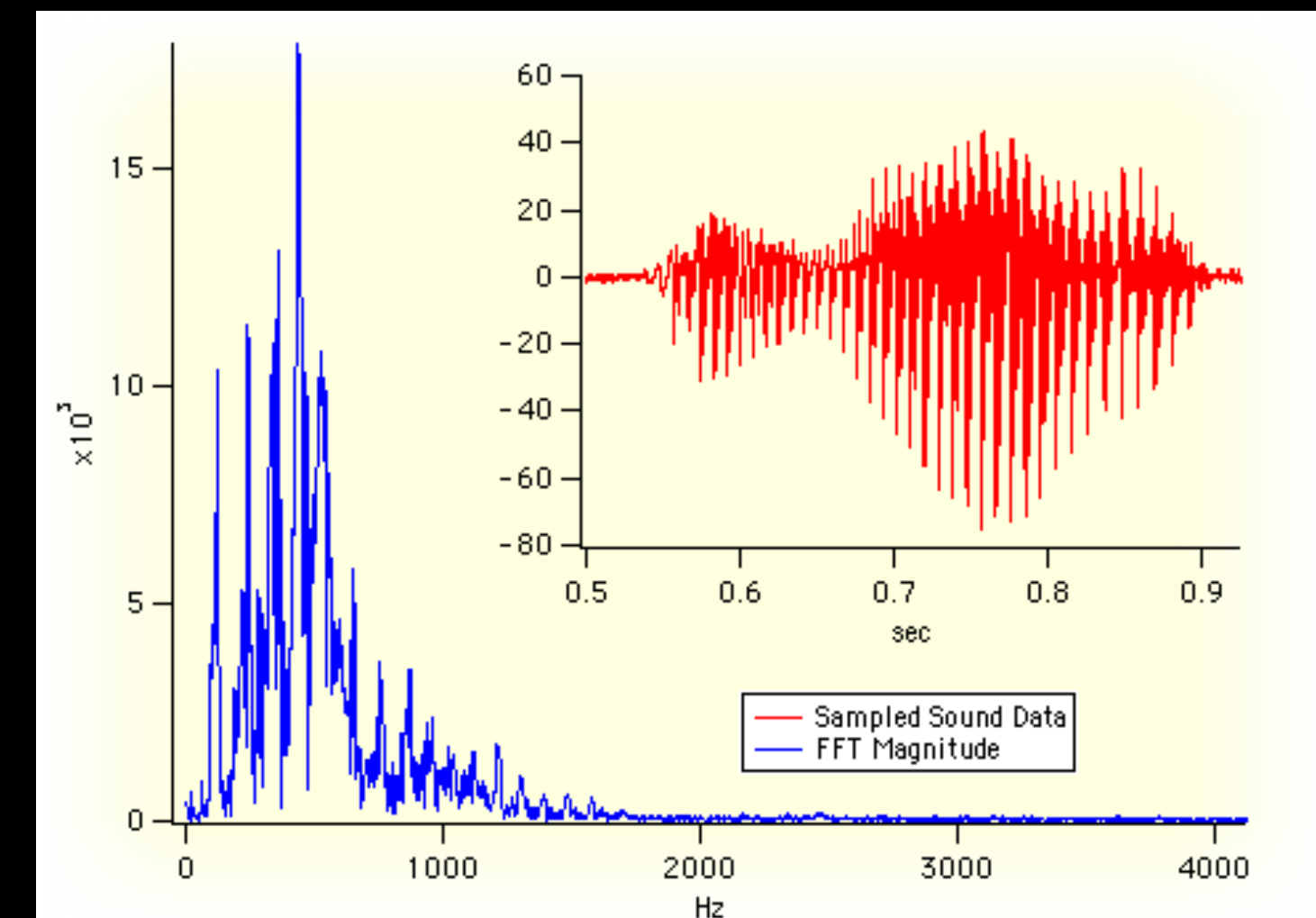
SOUND SYNTHESIS



Typical synthesis set-up

TIME DOMAIN & FREQUENCY DOMAIN

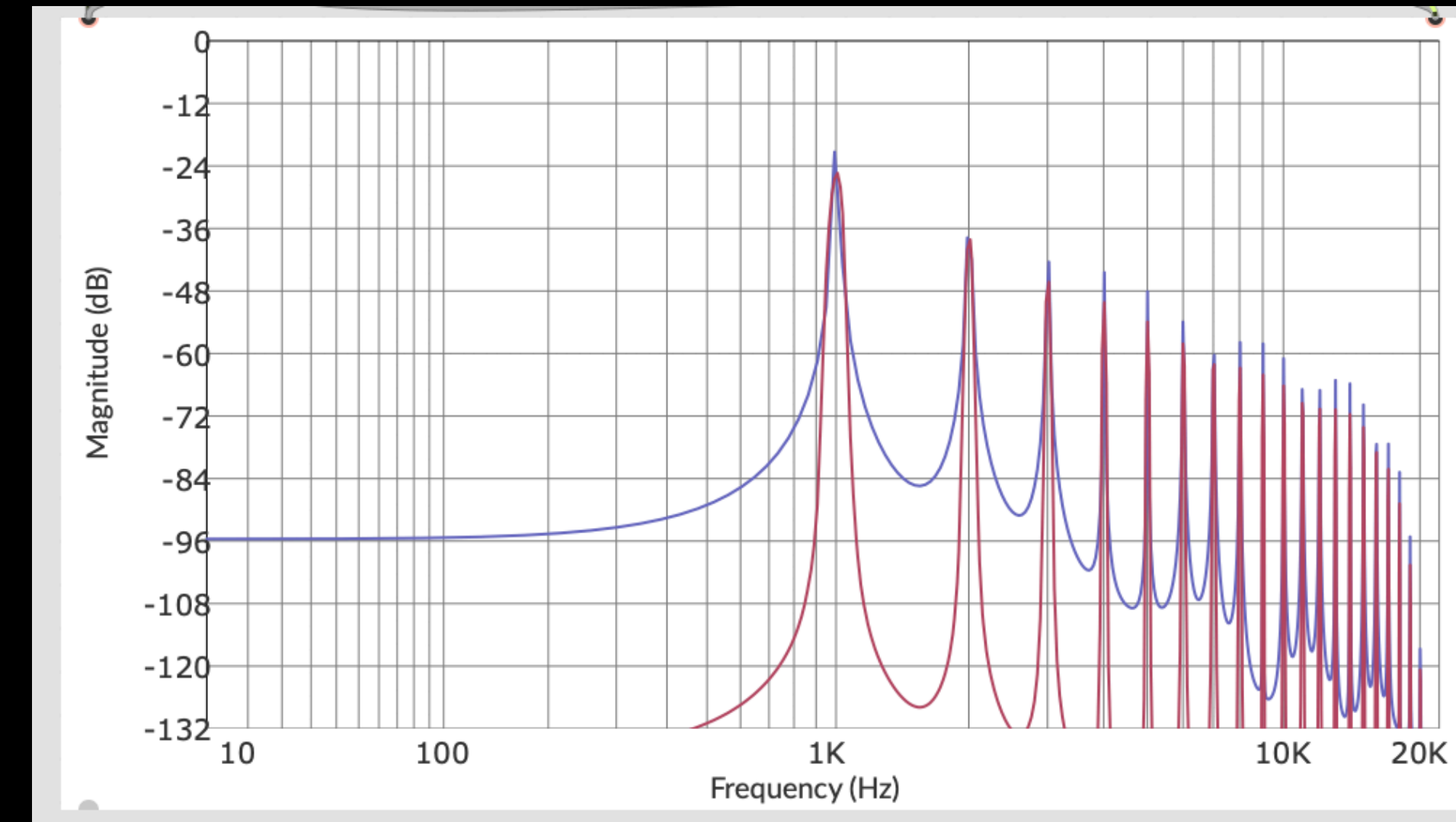
- Waveform: time domain (X axis is time, Y axis is amplitude at that time)
- Spectrum: frequency domain (X axis is frequency, Y is amplitude at that frequency over the course of the segment being analyzed.)



From <http://www.wavemetrics.com/products/igorpro/dataanalysis/signalprocessing/fouriertransforms.htm>

TIME DOMAIN & FREQUENCY DOMAIN

- Spectrogram is a series of spectra (spectrums) (usually done over time or in real-time)
 - Each spectrum taken on a segment of the waveform
 - Use a sliding window to define the segments
 - Windows usually overlap
 - Windows are enveloped (ramp amplitude up and down)
- Usually accomplished via FFT (Fast Fourier Transform)



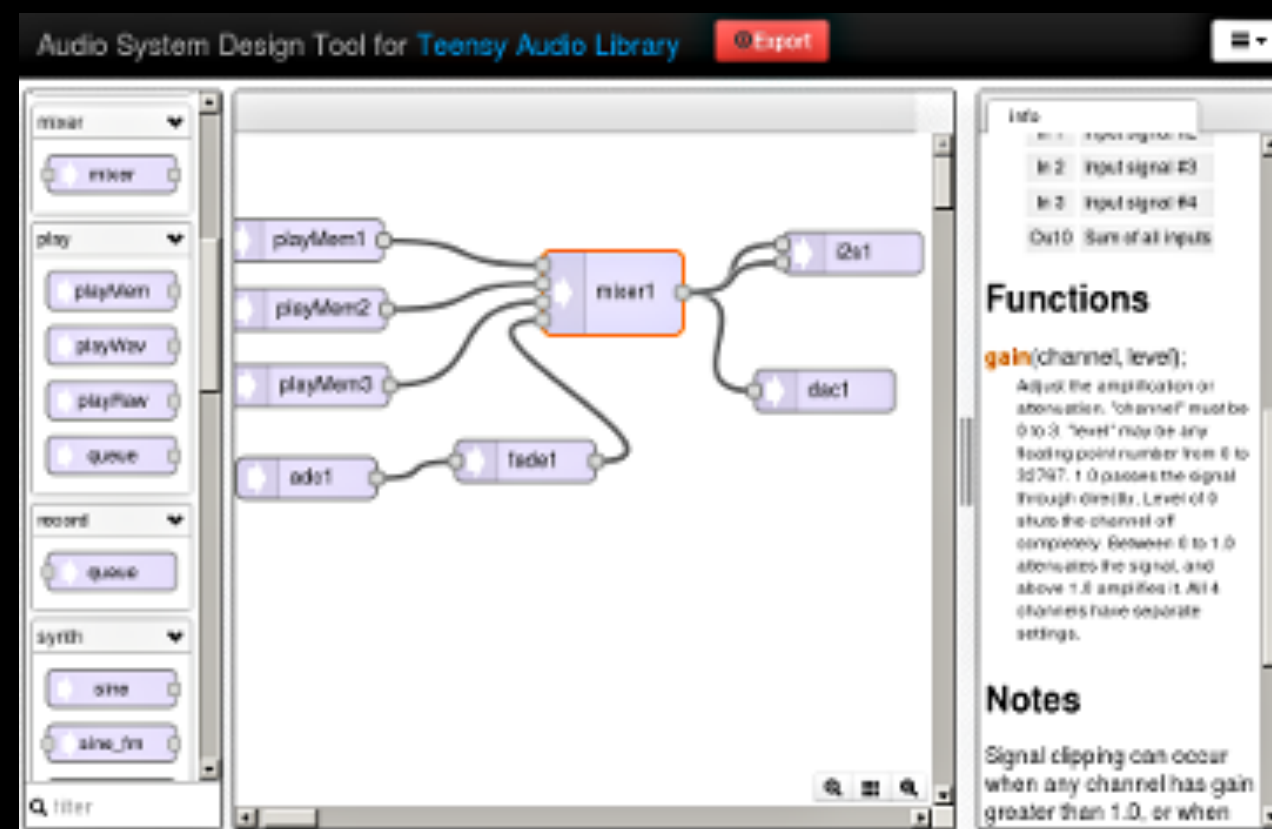
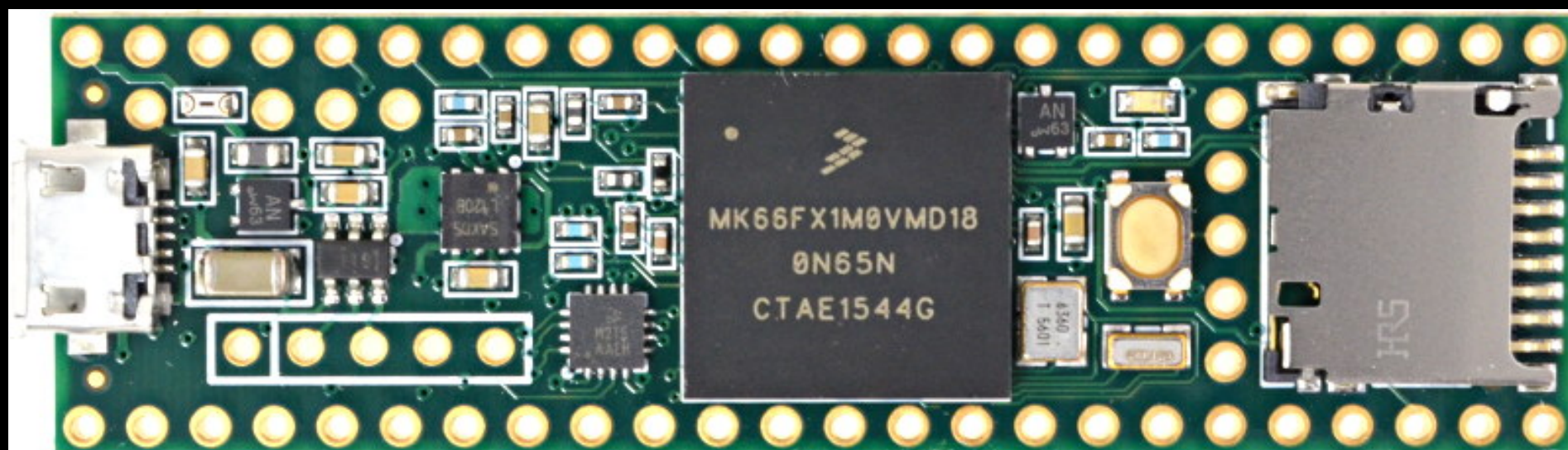
FFT spectrum of a sawtooth wave at 1kHz

SOFTWARE TOOLS FOR SOUND

- [Processing Sound Library](#)
- [Minim for Processing](#)
- [p5.js Sound Library](#)
- [openFrameworks addons](#) (too many to list)
- [Max](#)
- [PD \(Pure Data\)](#)
- [SuperCollider](#)
- [FoxDot](#) (Python library and programming abstraction layer for SuperCollider)
- [Chuck](#)
- [Csound](#)
- [Faust](#)
- [JUCE](#)

MICROCONTROLLERS FOR SOUND

Teensy Audio Library



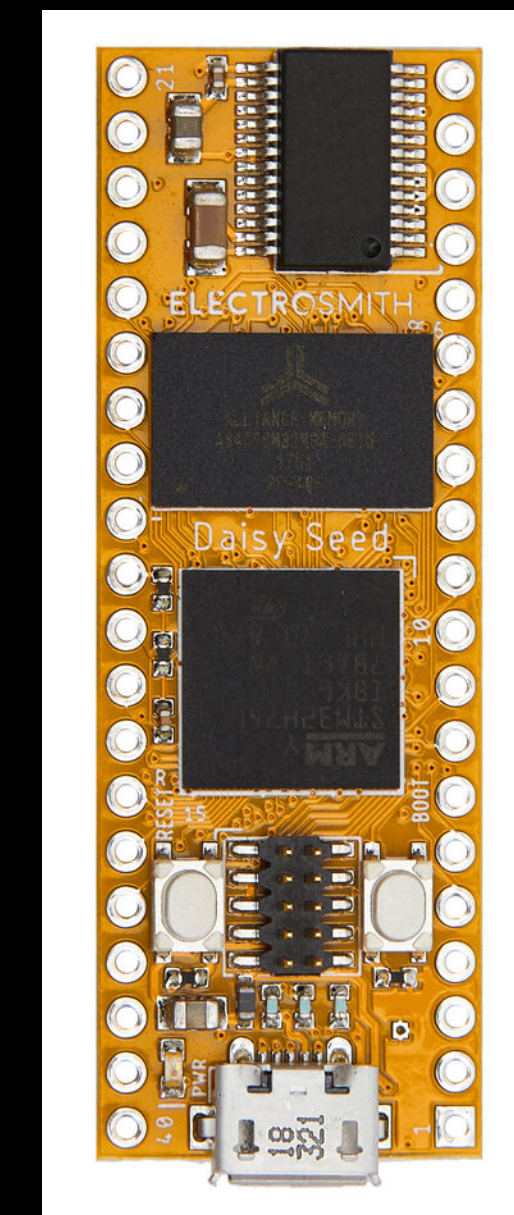
https://www.pjrc.com/teensy/td_libs_Audio.html

Bela



<https://bela.io>

Electrosmith Daisy



<https://www.electro-smith.com/daisy>

REFERENCES

- How Digital Audio Works
- Synthesis Fundamentals (Bob Moog Foundation)
- The Foundation of Synthesis [video series] (Bob Moog Foundation)
- Handmade Electronic Music: The Art of Hardware Hacking, Nicholas Collins (2009), Routledge, <https://www.nicolascollins.com/handmade.htm>
- The Computer Music Tutorial, Curtis Roads (1996), MIT Press.