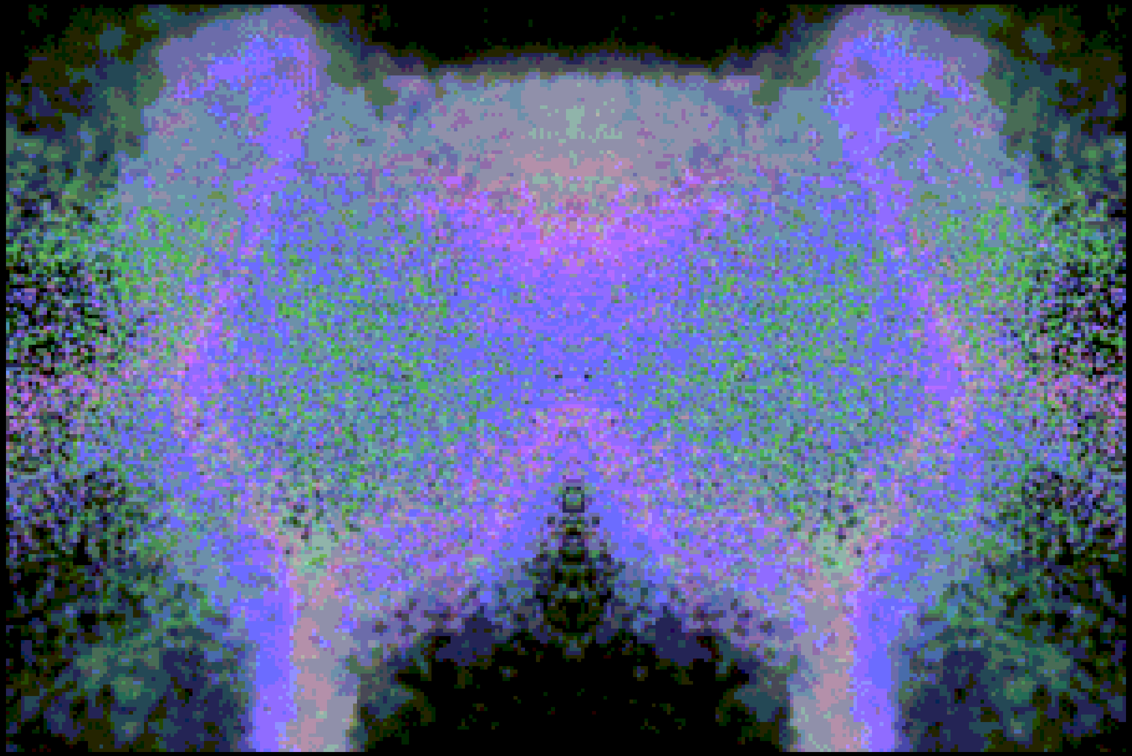
AUDIO REACTIVITY AND COMPUTER SOUND



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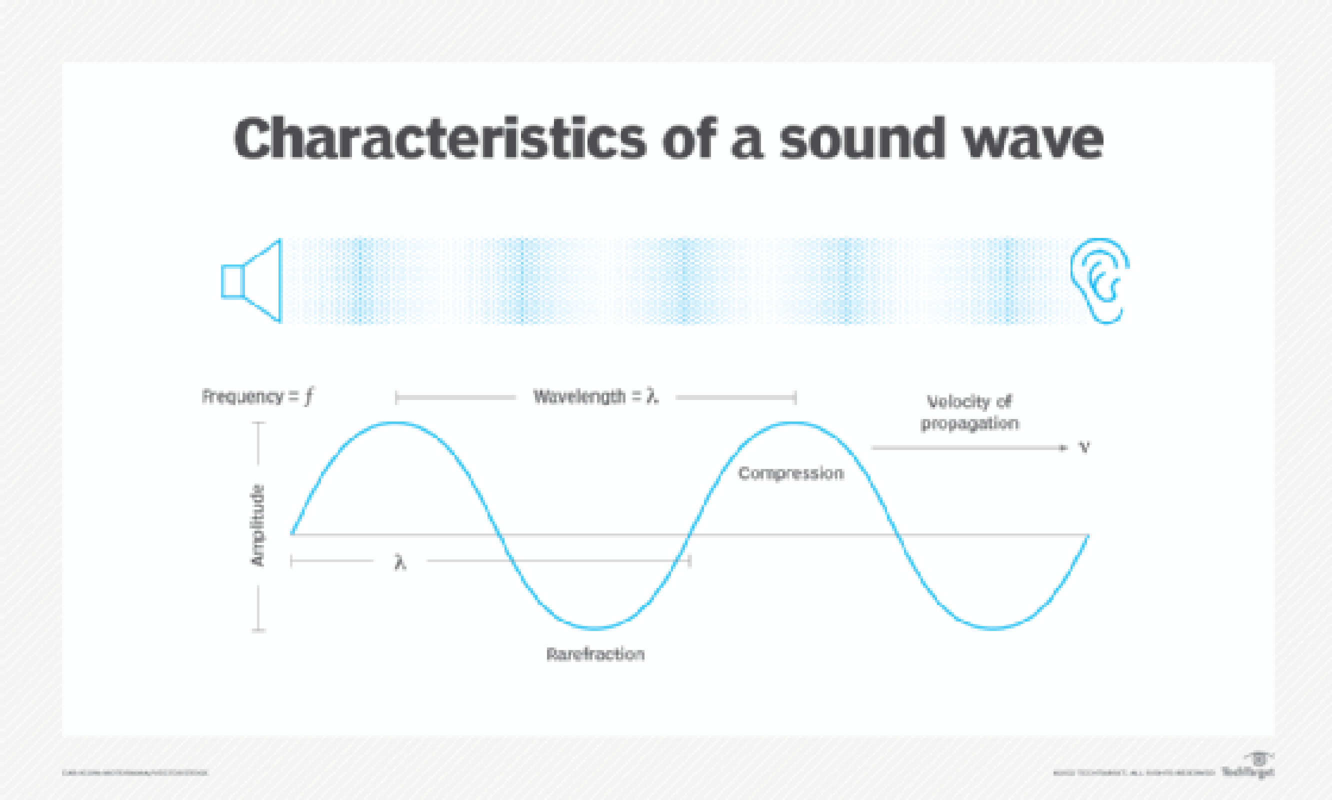
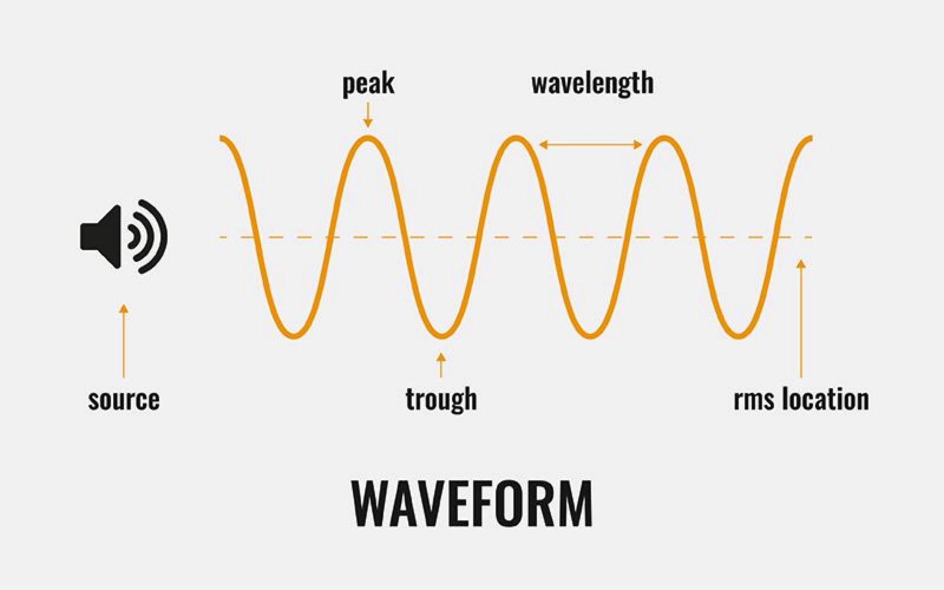
A SMALL INTRODUCTION ON SOUND

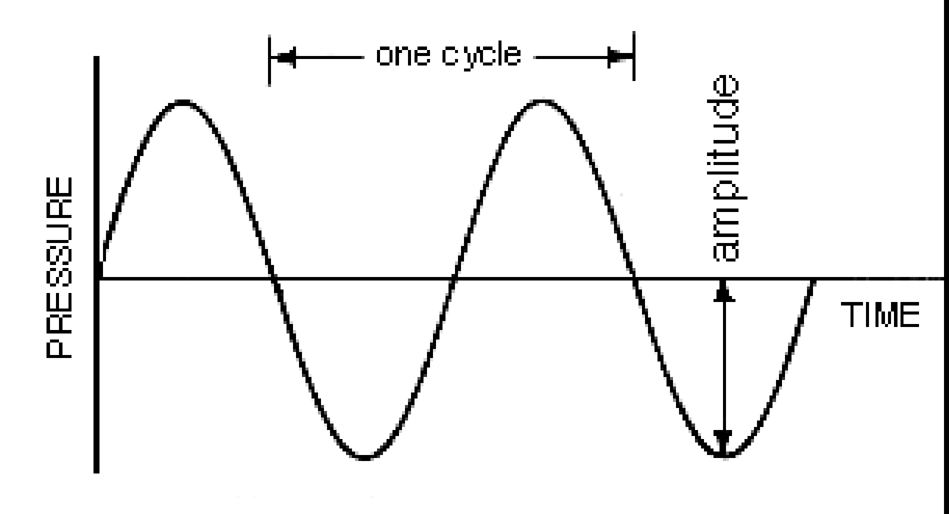


BY NATURE SOUND IS TIME BASED



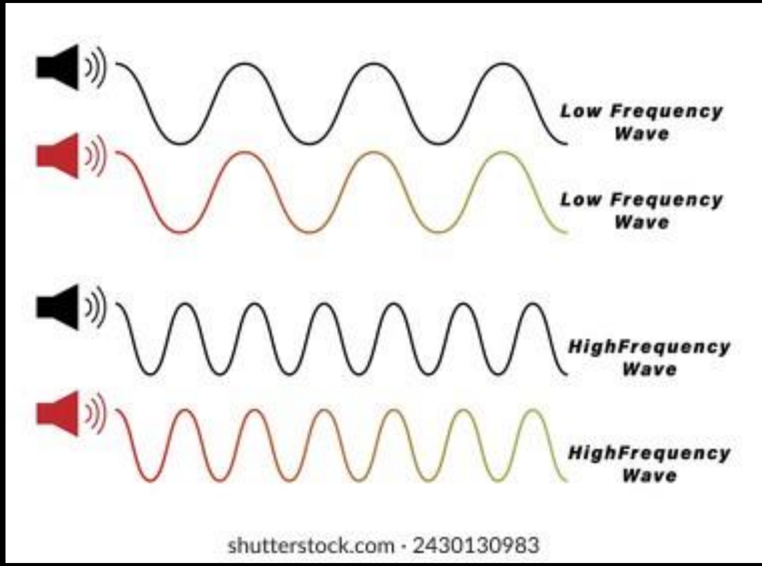
- Sound is the change of air pressure over time.
- A fluctuating amount of pressure changes the amount of presence
- A fluctuance in pressure events changes the pitch or the tone of sound.
- Sound is a wave and not a particle

TIME, CYCLES AND AMPLITUDE



- You start hearing sound when at least one cycle of air pressure has passed on the timeline.
- The change from a low air pressure to a high pressure and back to low actively sends a wave through the atmosphere which can be perceived by the ear when in hearing range.

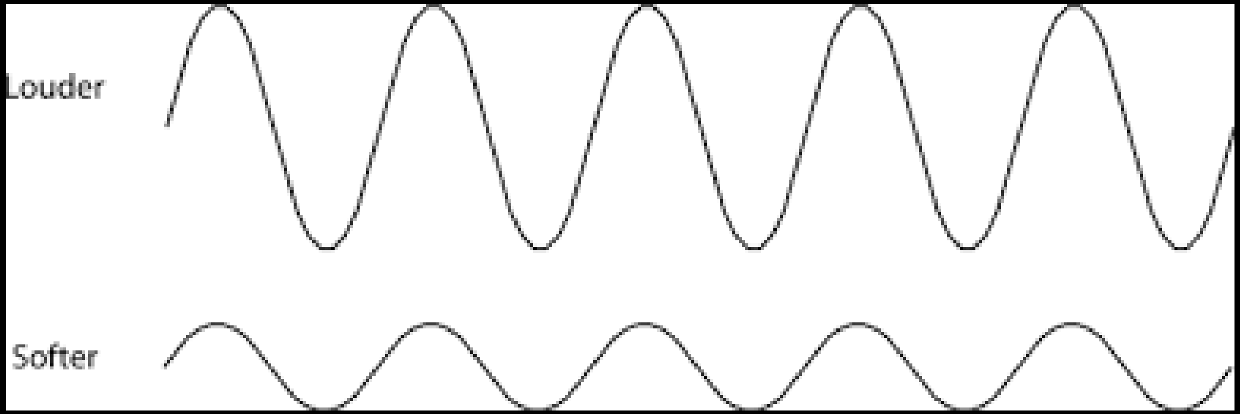
PITCH -> TONE -> FREQUENCY -> WAVELENGTH



- Pitch, tone, frequency and wavelength all refer to the same thing within sound.
- It is the amount of distance carrying between two peaks on the plot. More distance means a greater wavelength resulting in a lower tone because it takes longer to complete the wave cycle.
- The tone, pitch or frequency is expressed in HERZ(Hz)
- However it does not change the amplitude or the volume of the sound. Although a lower pitch will die out faster then a higher pitch sound.

AMPLITUDE

- Amplitude refers to the 'height' of the sound wave.
- A higher amplitude means the sound source had more power when creating the sound.
- Therefor the sound is perceived louder due to the change in air pressure being stronger.
- Amplitude is also expressed in decibels(dB)

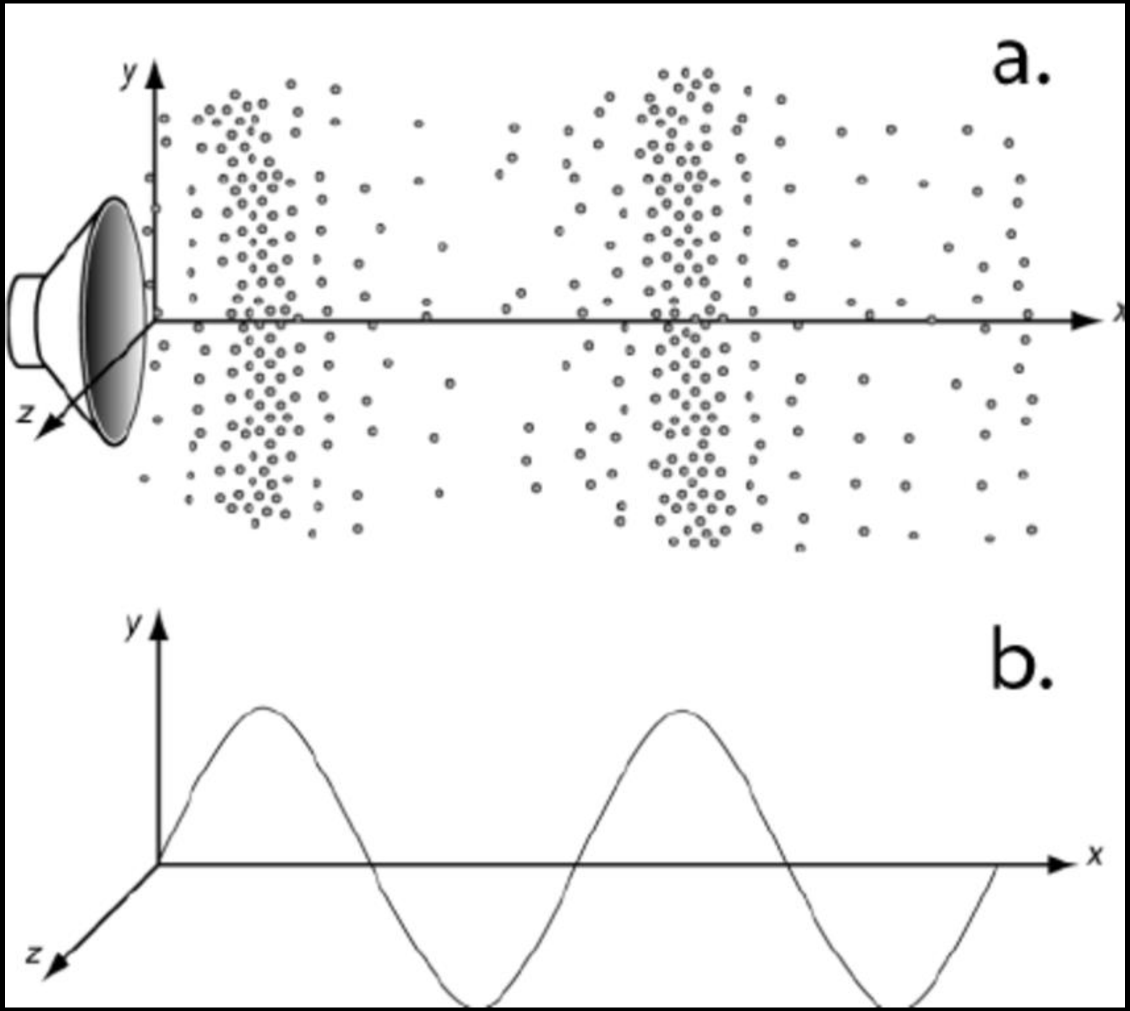


TIME FOR SOME CONFUSION!

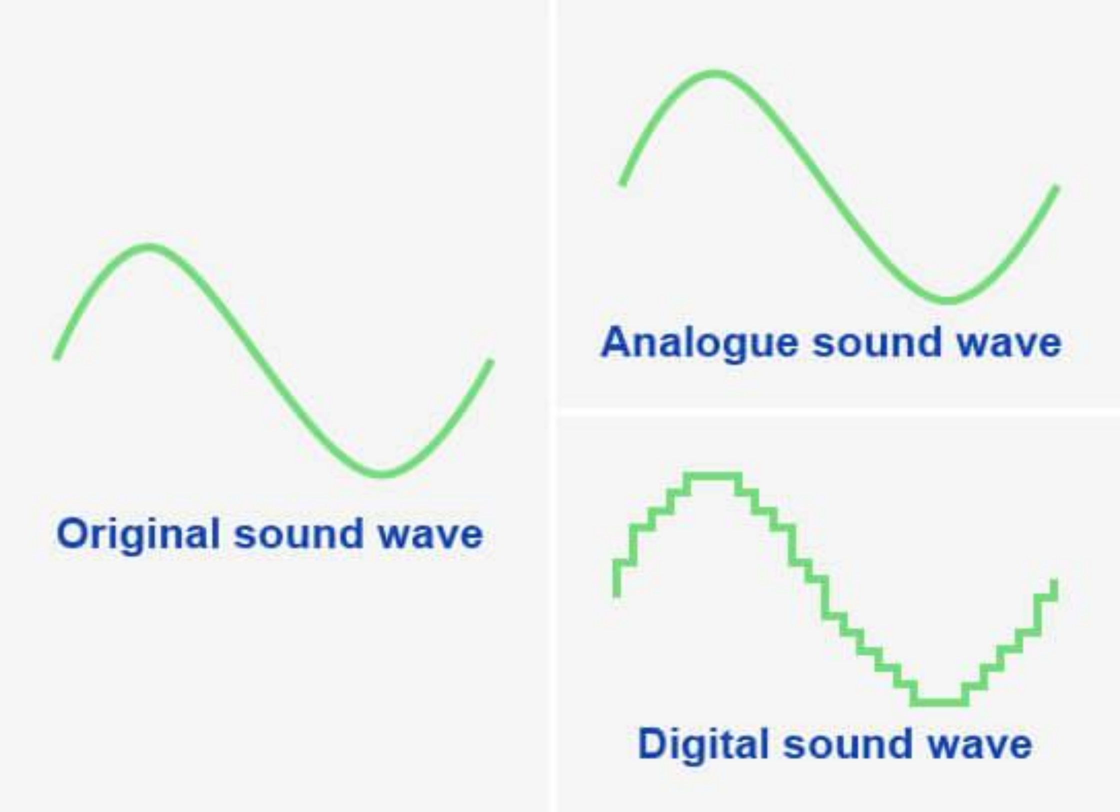
When a speaker cone completes one cycle it sends out a change in air pressure. This air pressure meets the eardrum wherein movement gets translated into electrical signals which the brain perceive as sound.

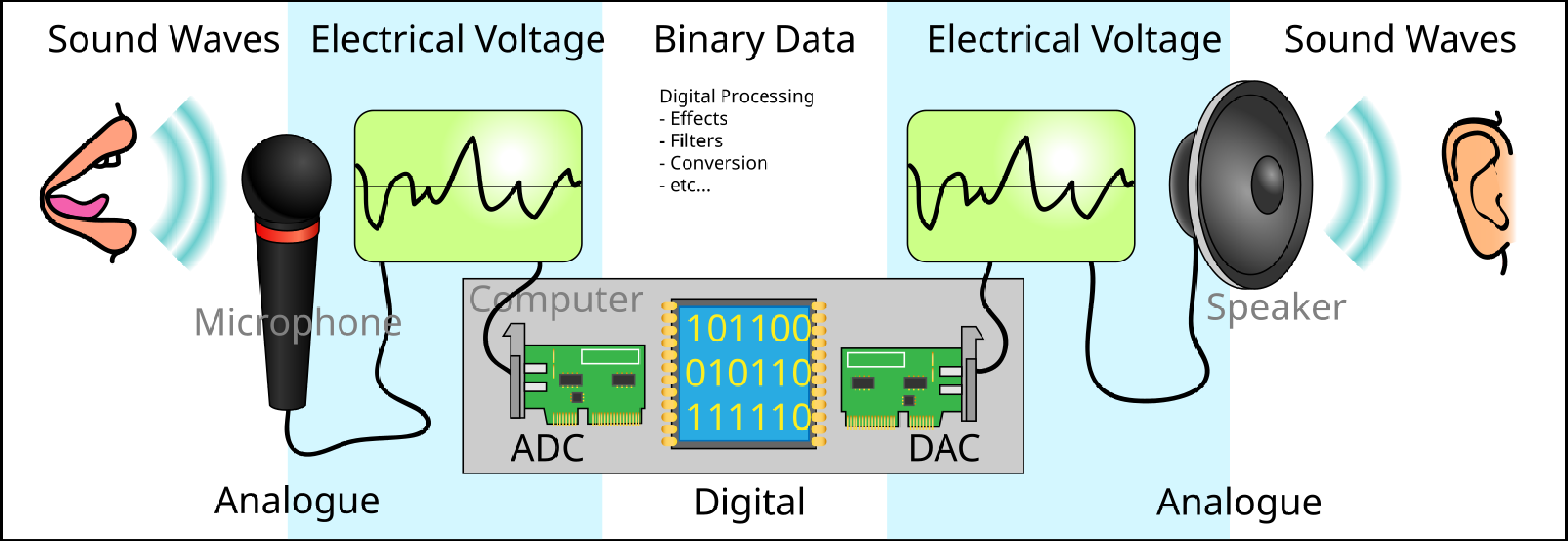
Although the change of pressure in the air can only exist due to a displacement of air molecules. When the pressure is high the presence of air molecules is denser. So when a higher air pressure is reaching your ear, also a denser field of air molecules travel along with it.

Hence, there are physicists that say that sound is not a wave but is made out of particles.



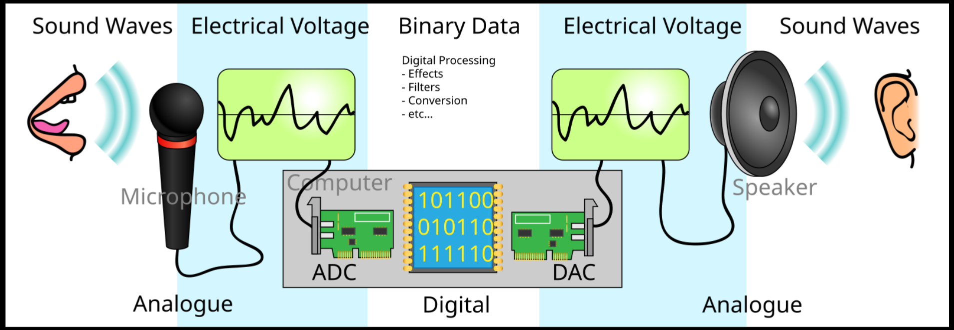
ALRIGHT, THAT WAS ANALOG AUDIO, HOW ABOUT DIGITAL?




Sound Waves
Electrical Voltage
Binary Data
Electrical Voltage
Sound Waves
Digital Processing
- Effects
- Filters
- Conversion
- etc...
Computer
Microphone
Speaker
101100
010110
111110
ADC
DAC
Analogue
Digital
Analogue

THE GENERAL FLOW OF SOUND IN THE DIGITAL DOMAIN

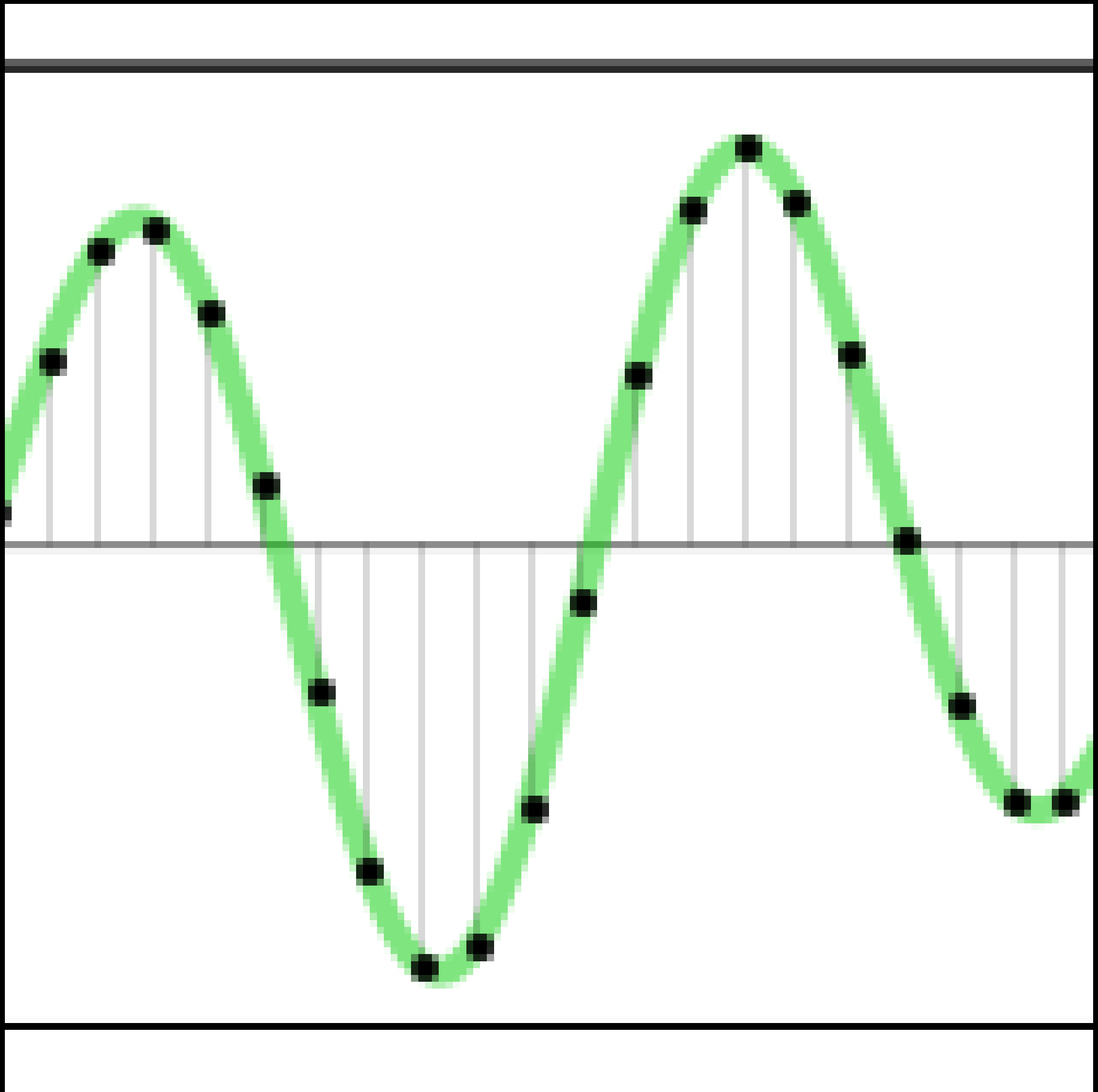
- Mic/sensor to capture the sound
- ADC
- Binary data / digital audio
- DAC
- Your ears, speakers, etcetera



ADC

ADC stands for 'Analog to Digital Conversion.

It usually is a chip that captures snapshots of the analog soundwave and converts that into digital data that represents audio.



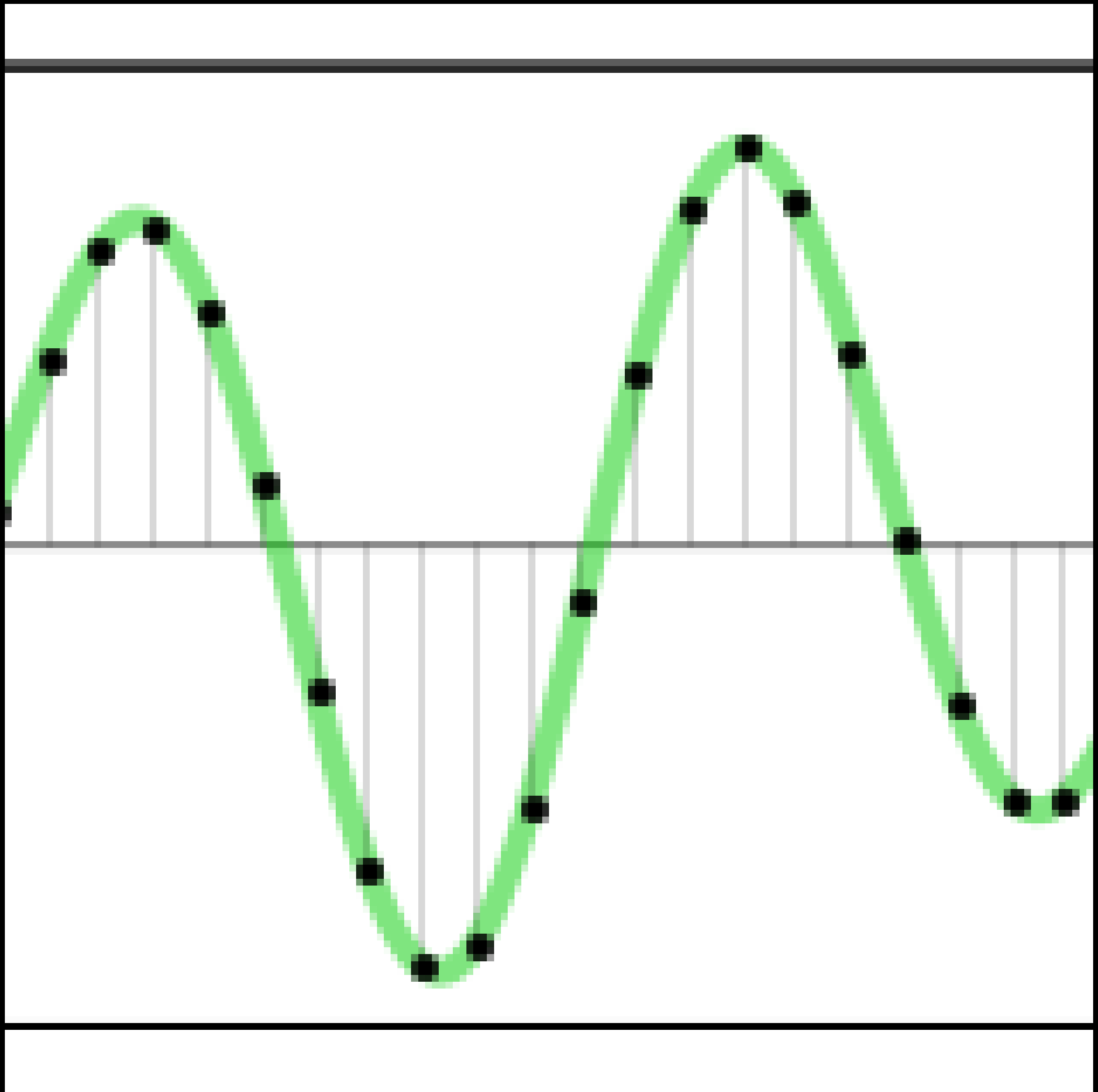
RESOLUTION

The snapshots that the ADC-chip makes is called a sample. The quality, or resolution of your sound depends on roughly two things.

Samplerate

And

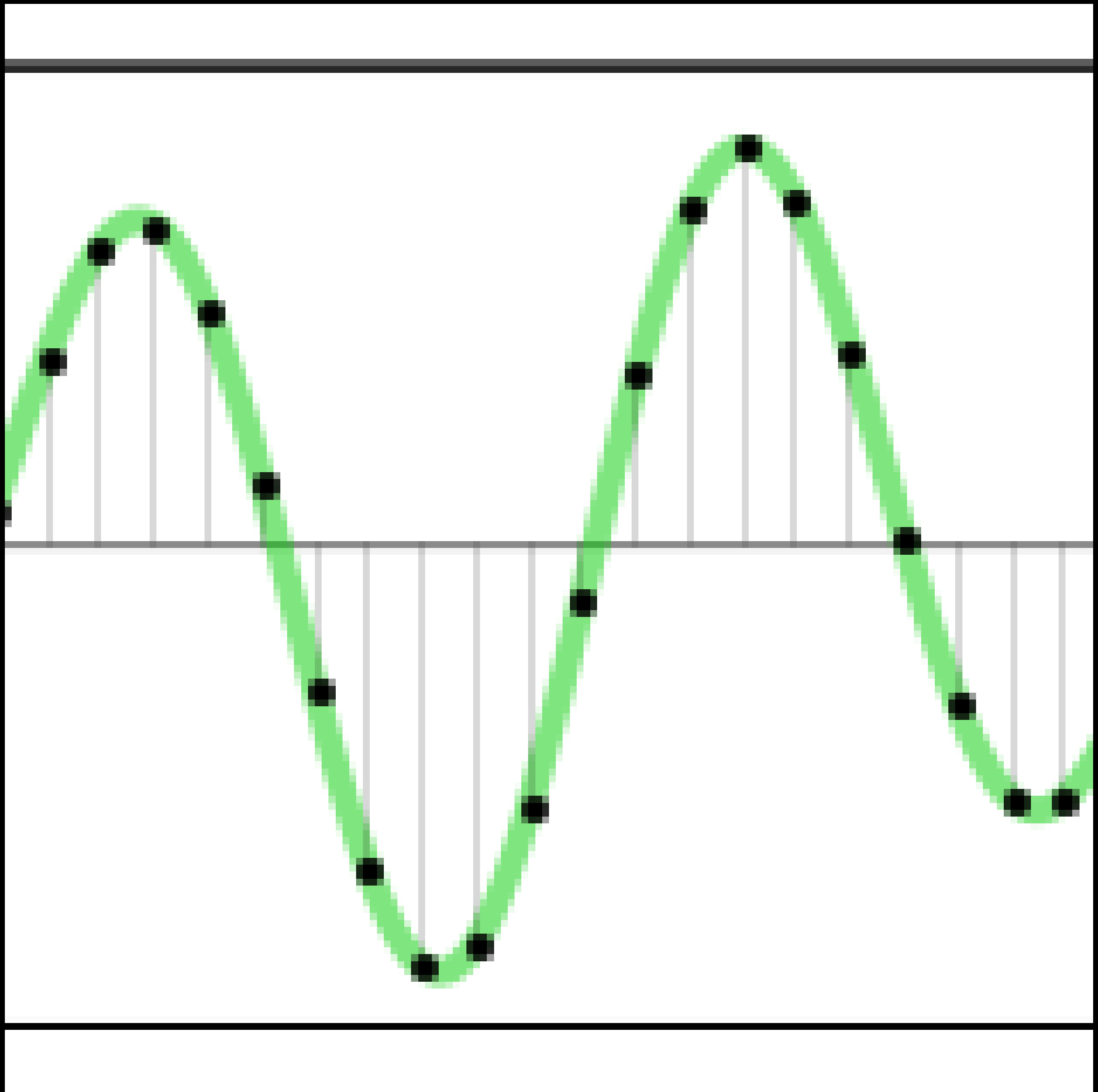
Bitdepth



SAMPLERATE

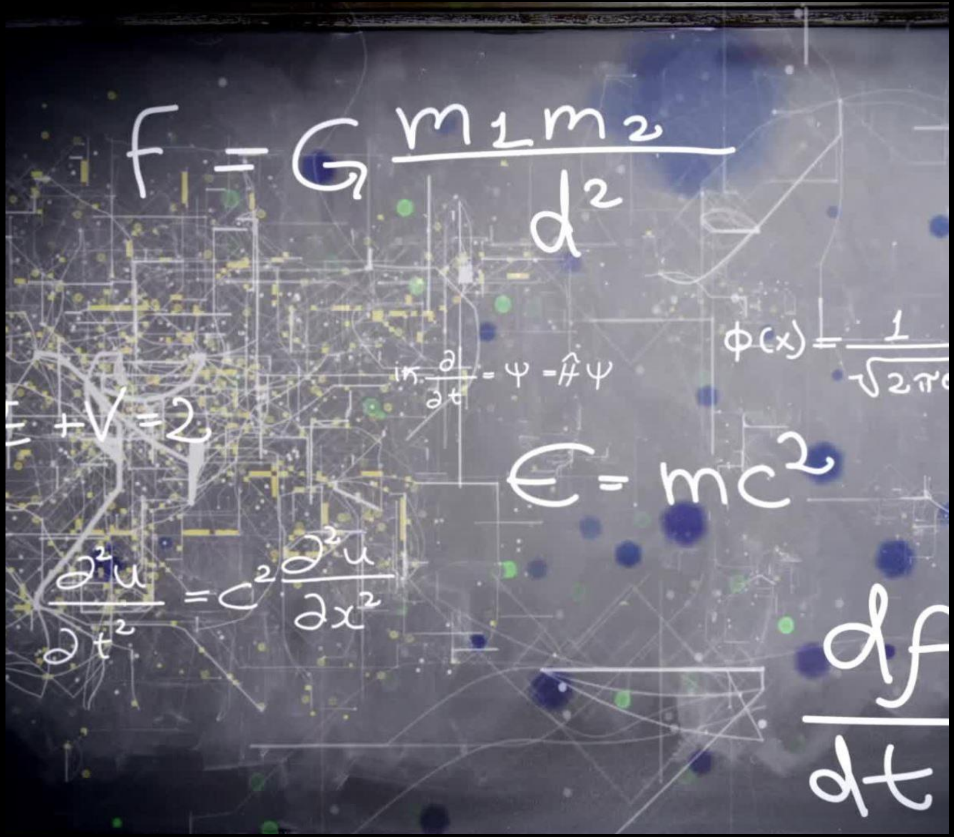
Samplerate refers to the frequency, or how often the analog to digital converter takes a sample, or snapshot, of the current wavelength passing through the converter chip.

Common samplerates are 44100, 48000, 88200, 96000, 192000. These numbers refer to the amount of samples per second.



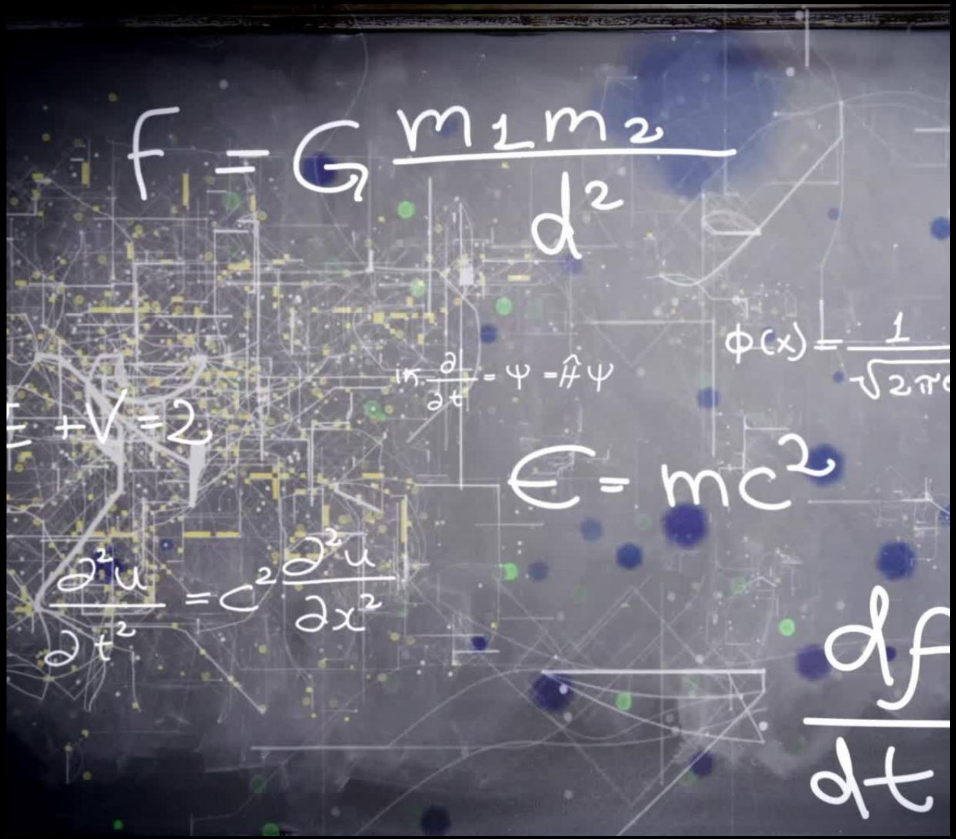
NYQUIST THEOREM

- The samplerate actually detemines the maximum height of frequency that you can capture with your recorder. It is complicated maths but boils down to:
- Half of the samplerate is the heighest frequency.
- 44100 / 2 = 22050 Herz, which is roughly 2050 herz above human hearing rate, so that would be enough, right?



NYQUIST THEOREM

- Why would you want to use higher samplerates if they result in frequencies well above the human hearing threshold and into ultrasonic territory. We make music for bats?
- No, well, maybe, but the extra ultrasonic frequencies give 'headroom' to our recordings. Meaning that you can do complex filtering of effects like pitchshifting without losing any of the fundamental elements of your sound like transients, timbres or partials.

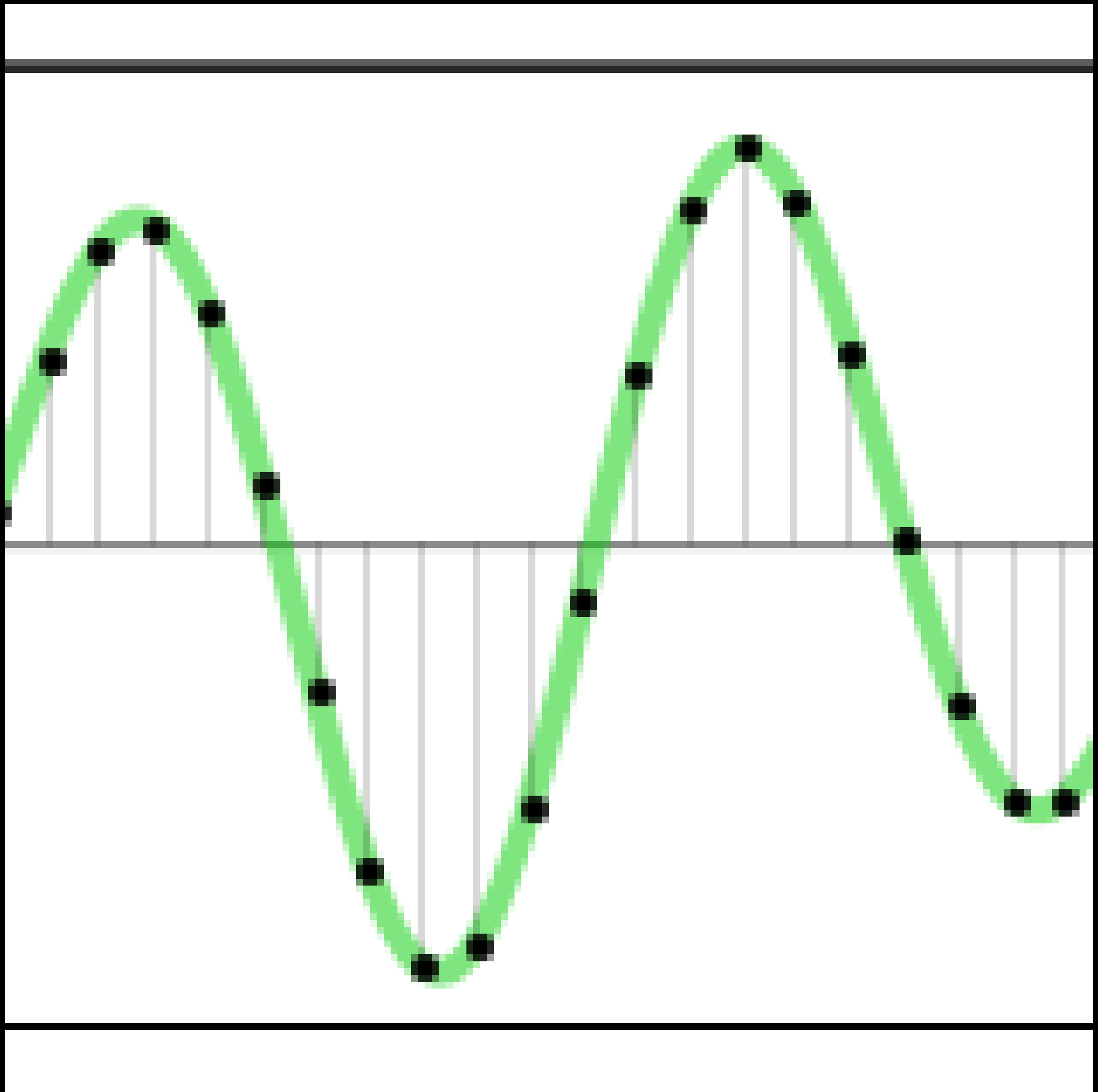


BITDEPTH

Alright, back to this story.

We know samplerate, which determines the amount of samples in a second. But it tells us nothing of how accurate the conversion is.

That is where bitdepth comes into play. Higher the bitdepth number the more precise the converter tries to estimate the wave in time.



8-bit Audio

- **Total values:** 256 (2^8)
- **Dynamic range:** ~48 dB
- **Usage:** Vintage computers, lo-fi sampling, retro game audio
- **Signal-to-Noise Ratio:** ~48 dB
- **Precision:** Very limited, noticeable quantization noise

16-bit Audio (CD Quality)

- **Total values:** 65,536 (2^16)
- **Dynamic range:** ~96 dB
- **Usage:** CD audio, MP3, most consumer audio
- **Signal-to-Noise Ratio:** ~96 dB
- **Precision:** Sufficient for most listening scenarios, below the threshold of human hearing in good conditions

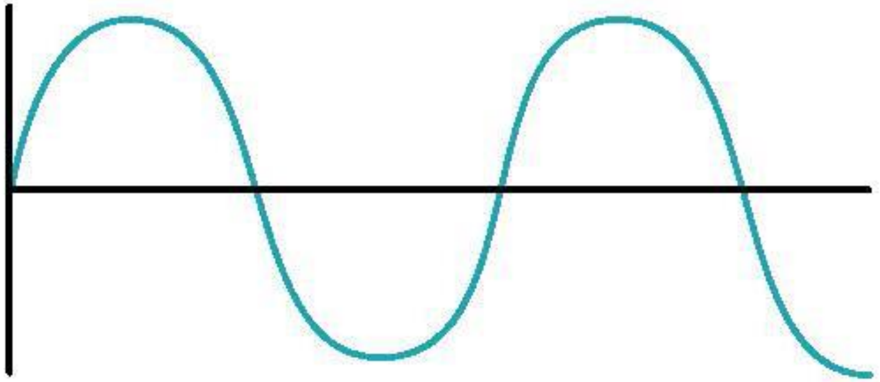
24-bit Audio (Professional/Studio)

- **Total values:** 16,777,216 (2^24)
- **Dynamic range:** ~144 dB
- **Usage:** Professional recording, mixing, mastering, high-end audio interfaces
- **Signal-to-Noise Ratio:** ~144 dB
- **Precision:** Exceeds human hearing capabilities (~120 dB), provides headroom for processing

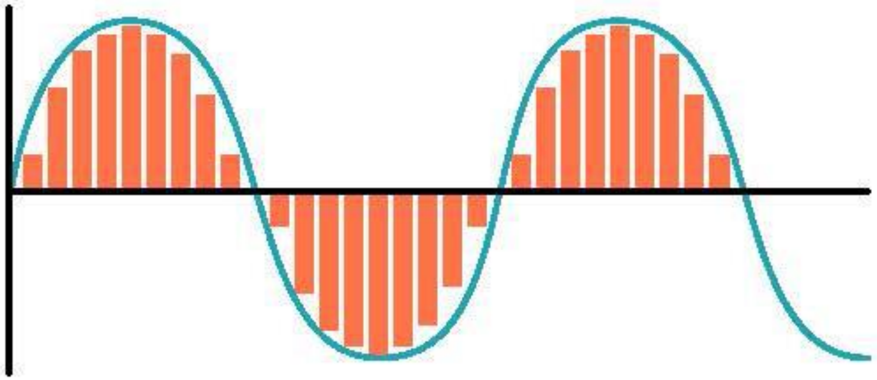
32-bit Integer Audio

- **Total values:** 4,294,967,296 (2^32)
- **Dynamic range:** ~192 dB
- **Usage:** Rare in practice, mostly theoretical or specialized applications
- **Signal-to-Noise Ratio:** ~192 dB
- **Precision:** Far beyond any practical audio requirement

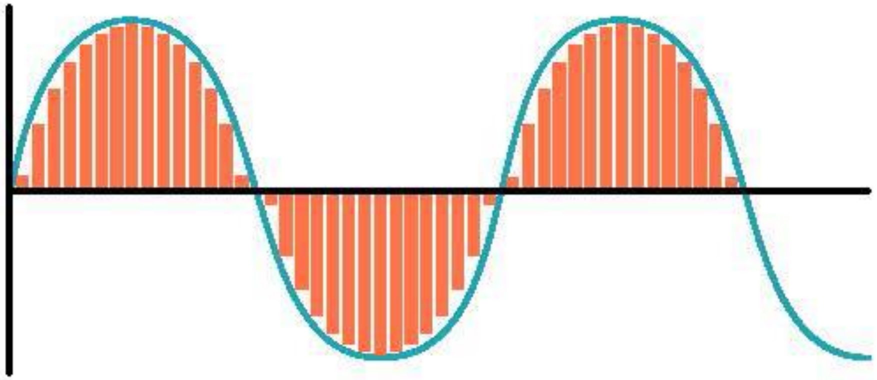
Format	Bit Depth	Dynamic Range	Typical Use	Values Below 0 dBFS	Values Above 0 dBFS
8-bit int	8	48 dB	Lo-fi/retro	256 steps	❌ Clips
16-bit int	16	96 dB	CD, consumer	65,536 steps	❌ Clips
24-bit int	24	144 dB	Professional recording	16.7M steps	❌ Clips
32-bit int	32	192 dB	Rare/speciali zed	4.29B steps	❌ Clips
32-bit float	32	~1,528 dB	DAW processing	✅ ~24-bit precision	✅ Can exceed without clip
64-bit float	64	~1,800 dB	Scientific/res earch	✅ ~53-bit precision	✅ Virtually unlimited



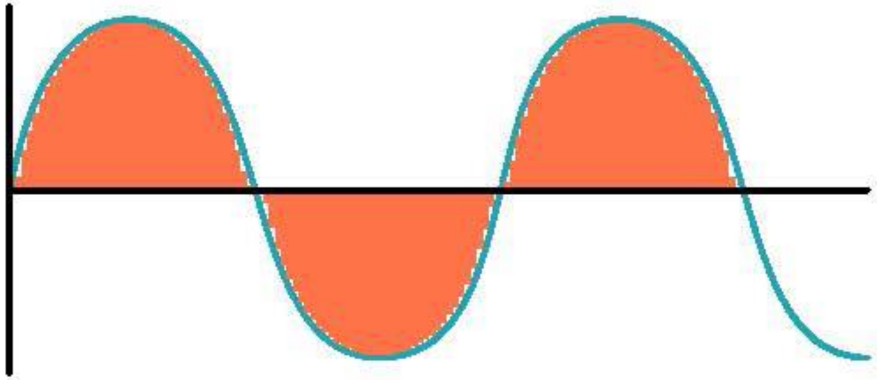
Analog Audio Signal



CD Quality Audio: 16-bit/44.1 kHz



High-resolution Audio: 24-bit/96 kHz



High-resolution Audio: 24-bit/192 kHz

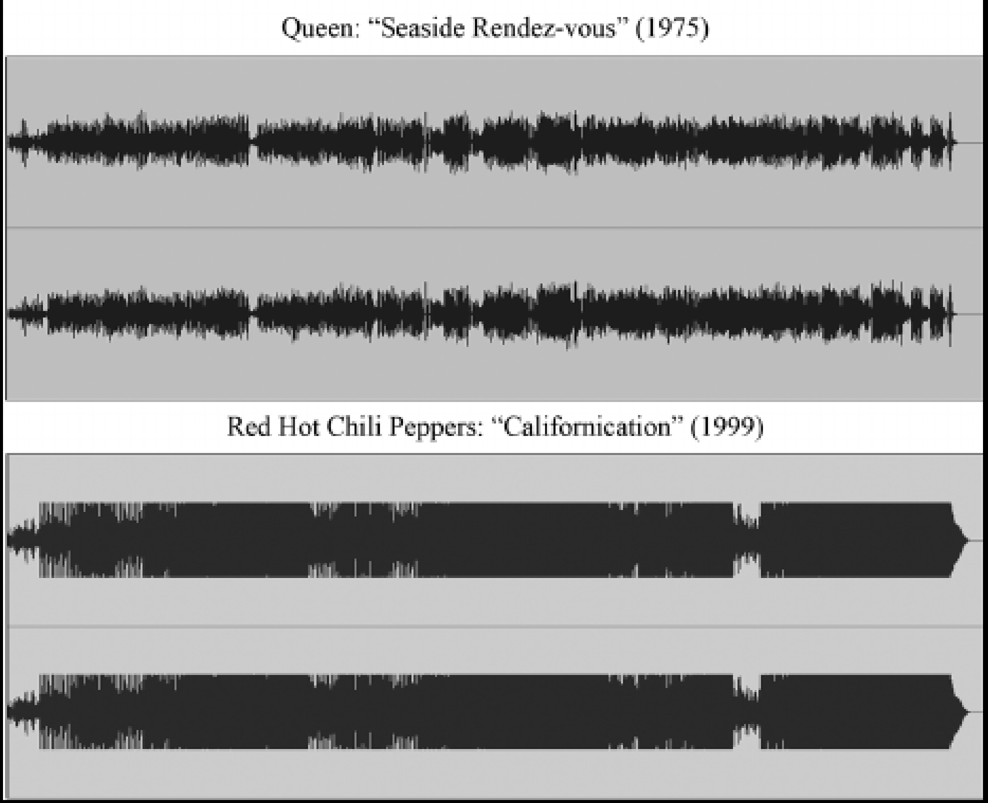
DIGITAL AUDIO IS FRIGGING WILD

- The sheer amount of data being processed per second, or actually per frame, which is sub-time, it is smaller than a microsecond, is intense and should be appreciated.
- Even working with the lowest consumer settings in digital audio it means that 44100 times per second a floating point value is saved into a buffer file that is accurate to the range of 65,536th. Which, I believe, is insane, magical and powerful.

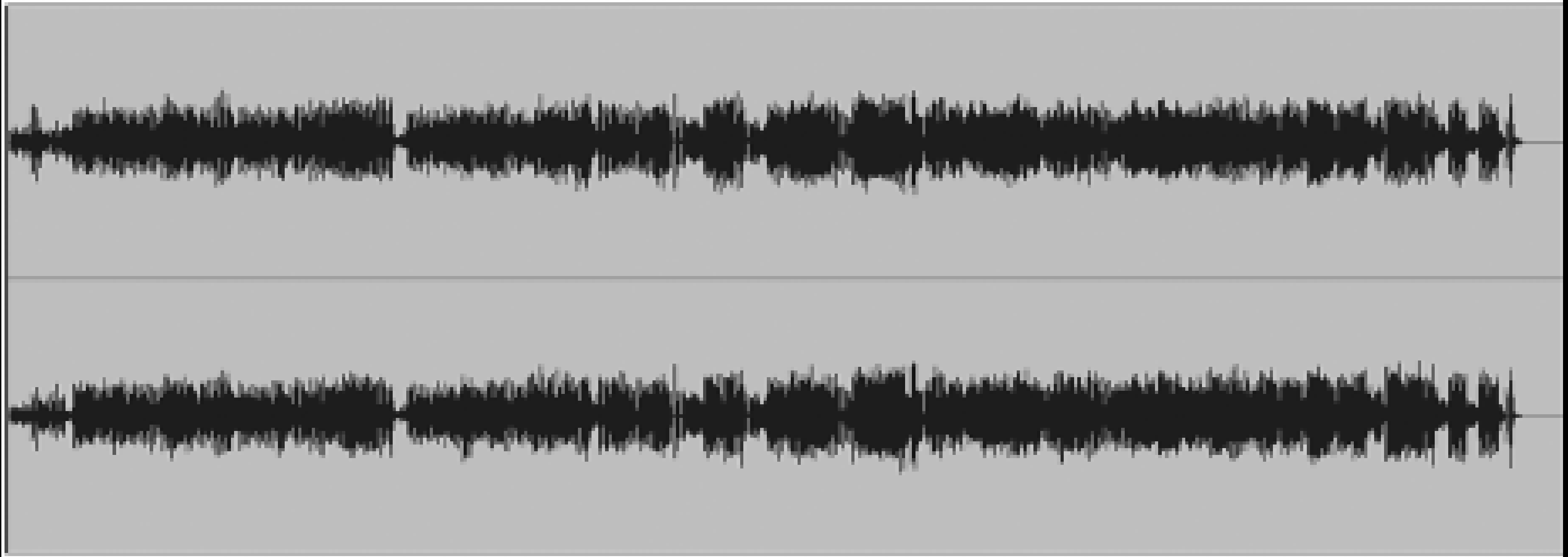
SOOOO, WHAT DO I END UP WITH?

- A digital audio file is made up of a couple of factors.
- Time
- Frequency
- Amplitude
- So basically a digital audio file is a collection of sampledata chronologically ordered where each sample holds frequency and amplitude information. This ordering allows you to scrub through a sound file where something tied to analog time is freed from the time constraint and can be processed or altered to your leisure.

LET'S TAKE A CLOSER LOOK TO A SONG

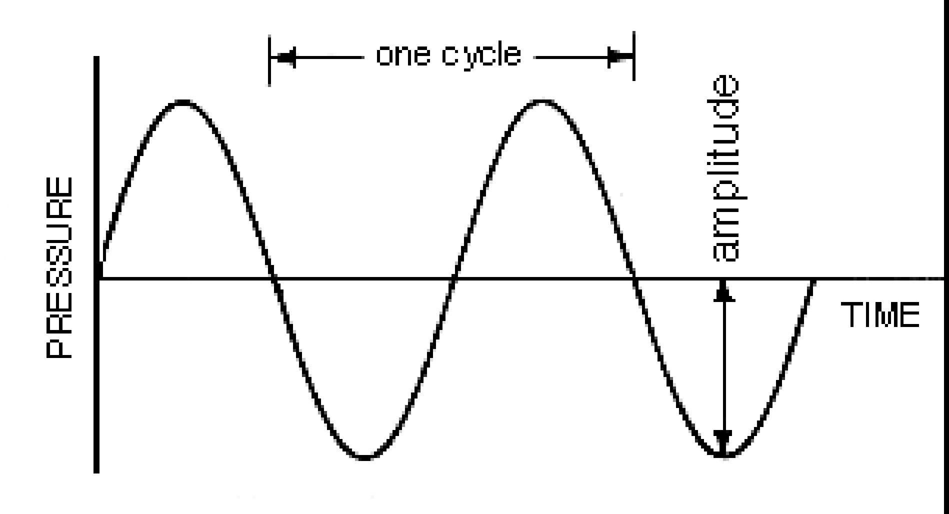


Queen: “Seaside Rendez-vous” (1975)

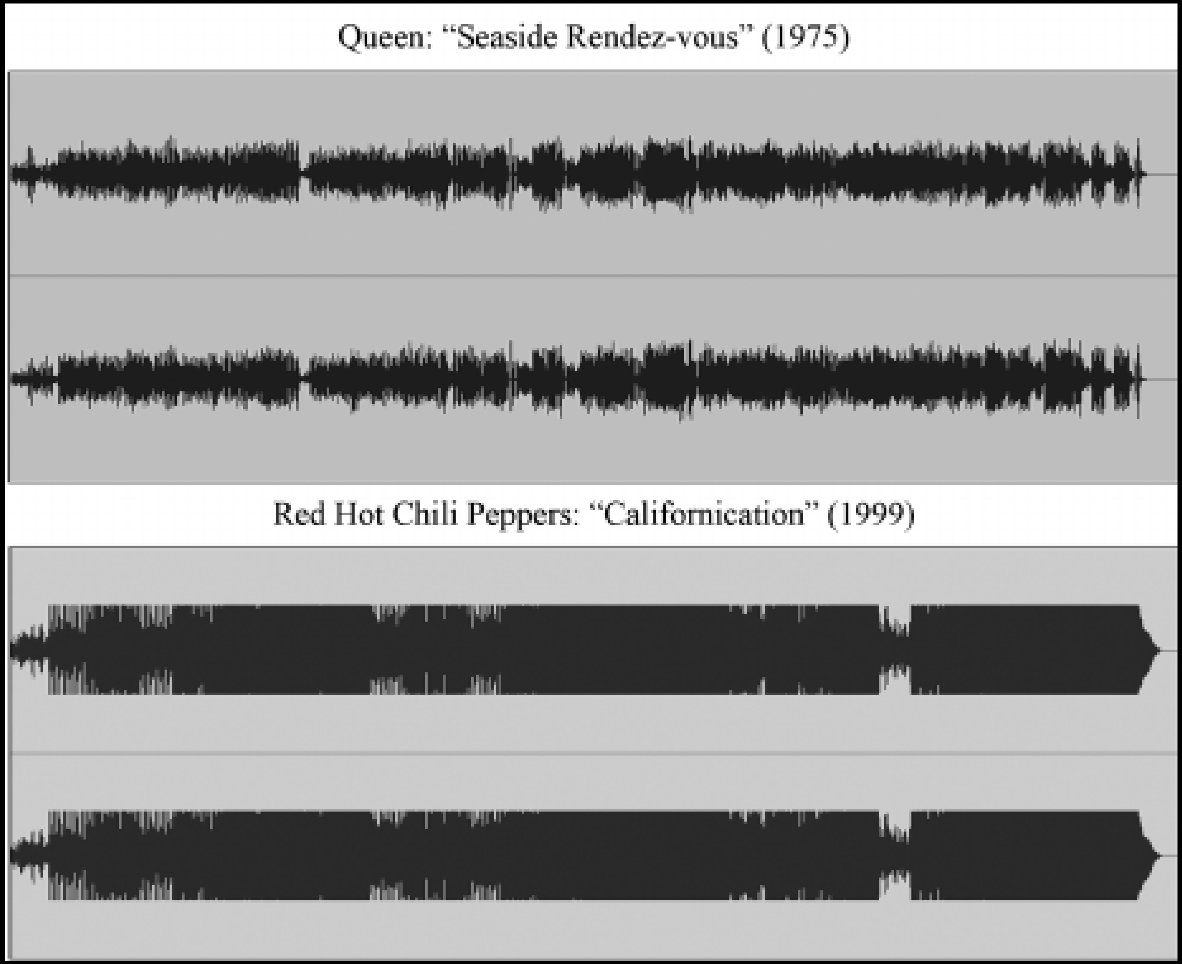


Red Hot Chili Peppers: “Californication” (1999)

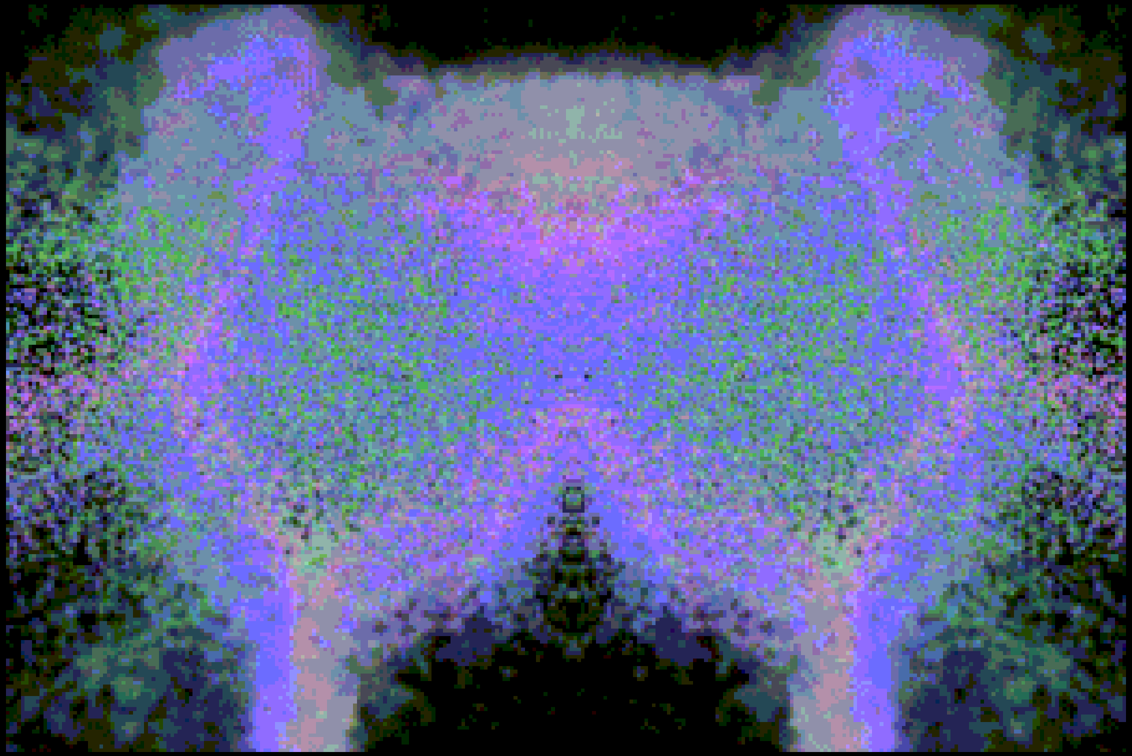
SIMPLE SINGLE FREQUENCY



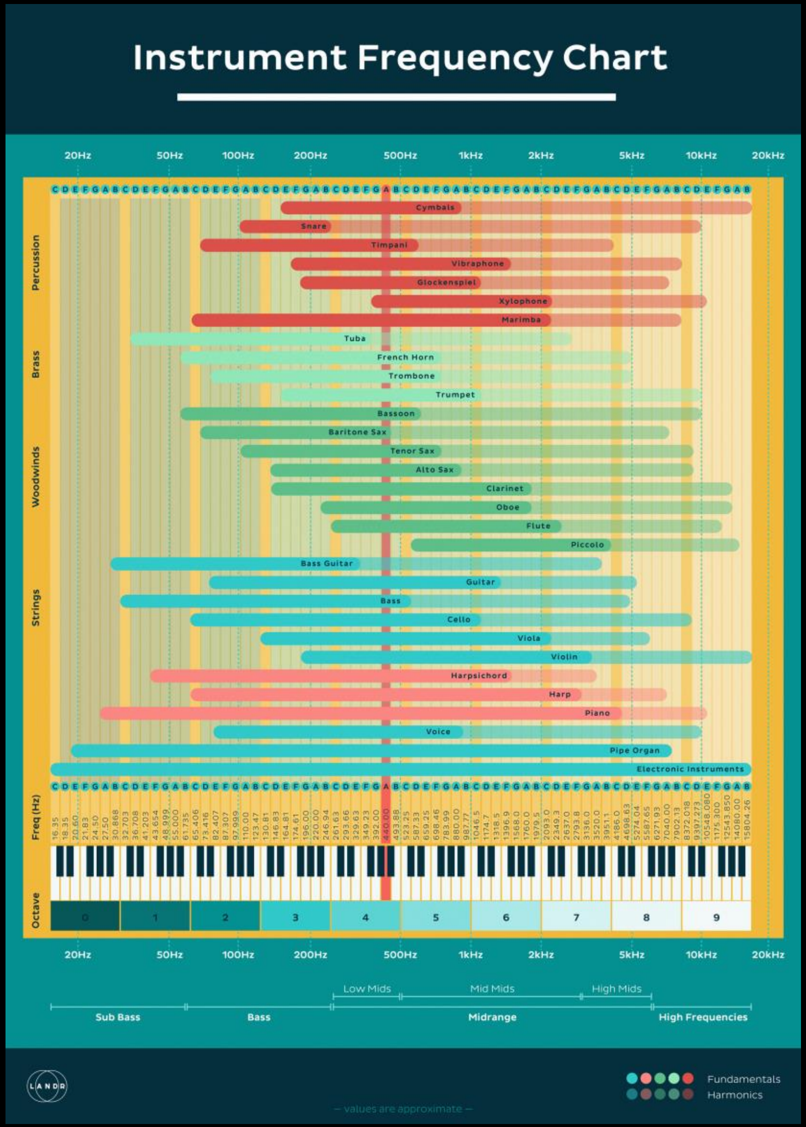
COMPLEX GROUP OF FREQUENCIES

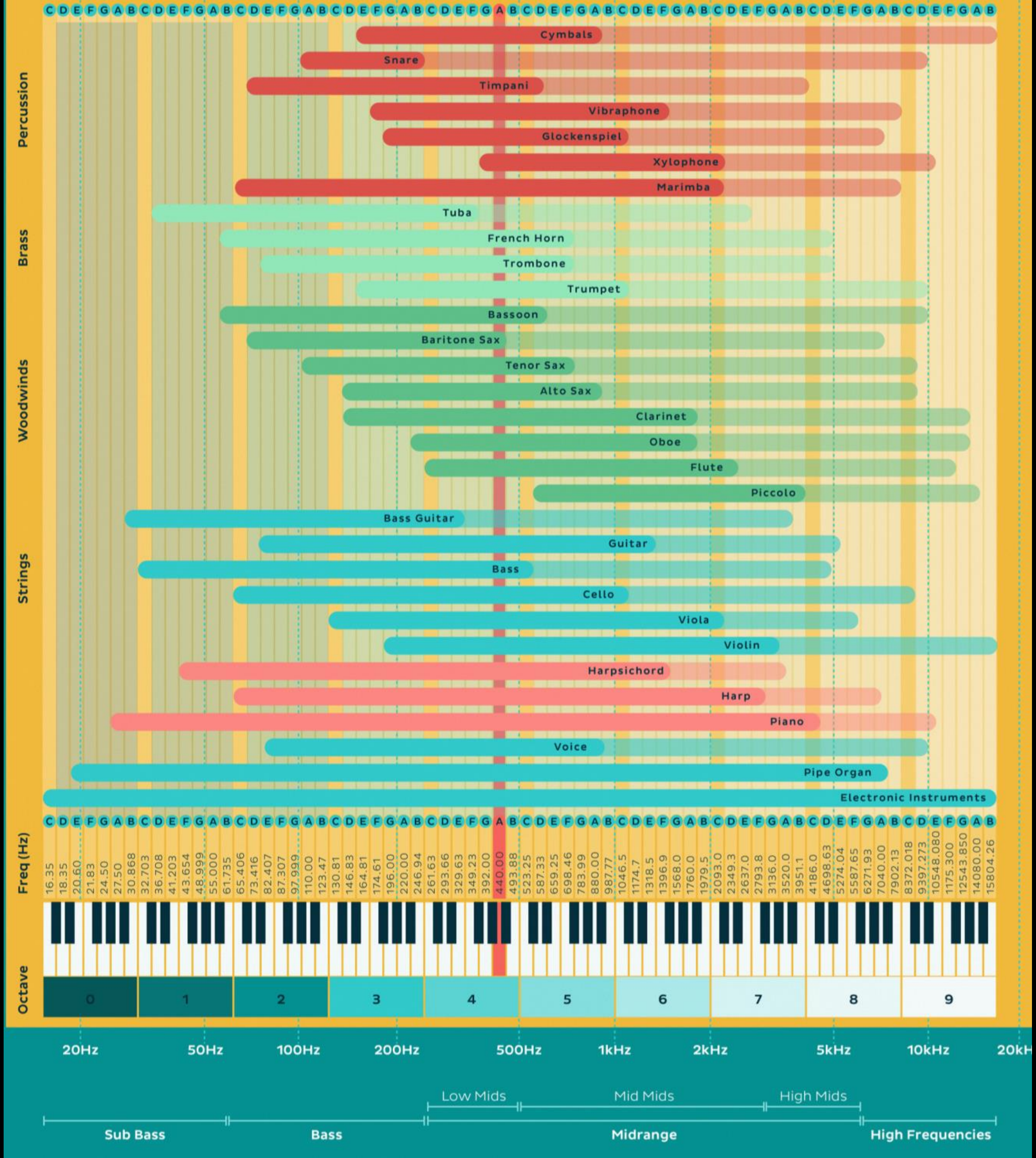


- If we want to make things audio reactive we tend to want to do that with a musical piece instead of a single frequency.
- Inside the complex group of frequencies we can find all the frequencies that are in a song. Most songs are standardized to a certain range of frequencies.
- The more frequencies you can distill from a song, the more responsive the animation will look like.



A LOOK AT FREQUENCIES




Percussion
Cymbals
Snare
Timpani
Vibraphone
Glockenspiel
Xylophone
Marimba
Brass
Tuba
French Horn
Trombone
Trumpet
Woodwinds
Bassoon
Baritone Sax
Tenor Sax
Alto Sax
Clarinet
Oboe
Flute
Piccolo
Strings
Bass Guitar
Guitar
Bass
Cello
Viola
Violin
Harpsichord
Harp
Piano
Voice
Pipe Organ
Electronic Instruments
Freq (Hz)
Octave
0
1
2
3
4
5
6
7
8
9
20Hz
50Hz
100Hz
200Hz
500Hz
1kHz
2kHz
5kHz
10kHz
Low Mids
Mid Mids
High Mids
Sub Bass
Bass
Midrange
High Frequencies

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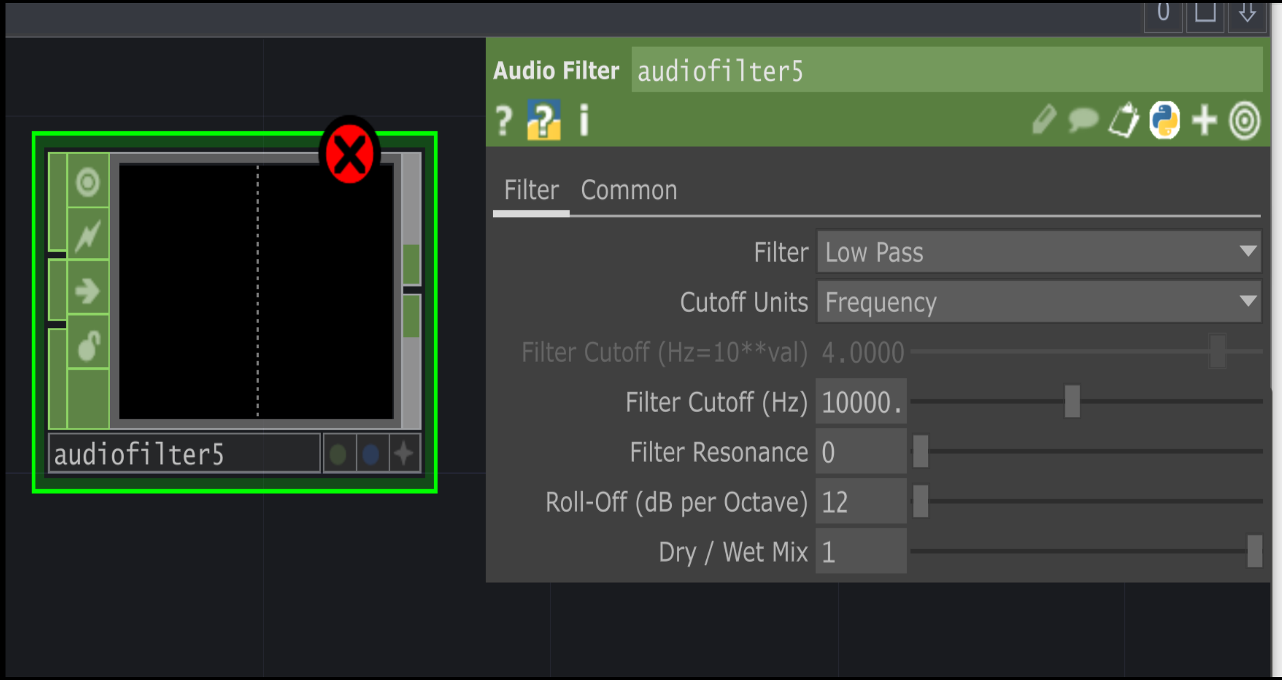
HOW DO WE ACCESS THESE FREQUENCIES?

FILTERS

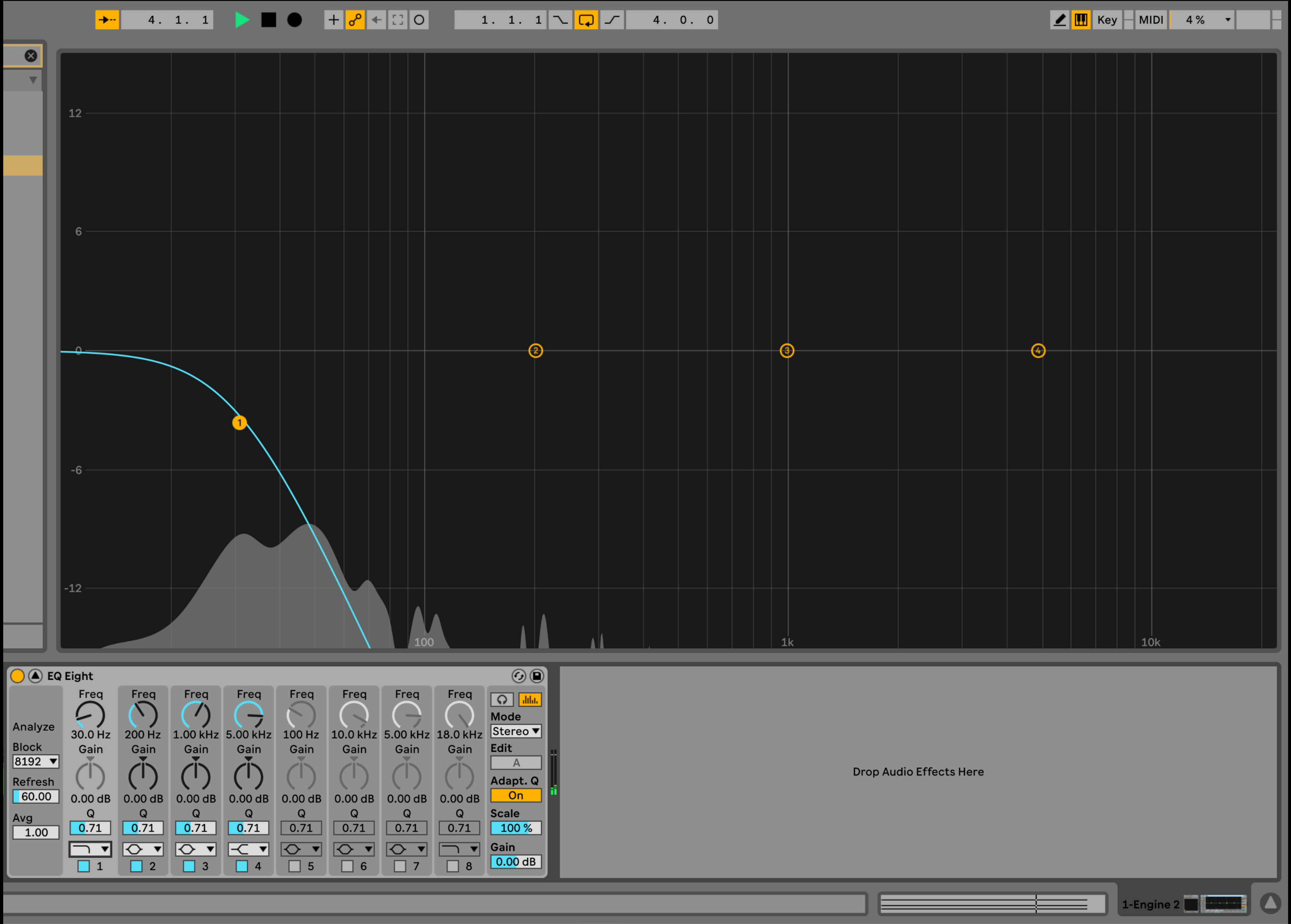
- To access certain frequencies in Touchdesigner we will work with Audio Filter CHOP
- There are *HIGH PASS FILTERS*
- There are *LOW PASS FILTERS*
- There are *BAND PASS FILTERS*
- We work within the CHOP domain because digital audio has to be calculated for every sample passed on the internal clock of the computer. In audio software it is calculated with 44.100 samples per second. In Touchdesigner we say that one sample is 1/60th of a second. So the resolution is a bit less then working with audio sample rates but still more than fast enough to make credible audio reactive animations.

LOW PASS

- A *LOW PASS* filter sets a threshold frequency and every frequency below that can travel through the filter



4. 1. 1
1. 1. 1
4. 0. 0
Key
MIDI
4 %
12
6
0
-6
-12
100
1k
10k
EQ Eight
Analyze
Block
8192
Refresh
60.00
Avg
1.00
Freq
30.0 Hz
200 Hz
1.00 kHz
5.00 kHz
100 Hz
10.0 kHz
5.00 kHz
18.0 kHz
Gain
0.00 dB
Q
0.71
1
2
3
4
5
6
7
8
Mode
Stereo
Edit
A
Adapt. Q
On
Scale
100 %
Gain
0.00 dB
Drop Audio Effects Here
1-Engine 2



Key
MIDI
4 %
12
6
0
-6
-12
100
1k
10k
EQ Eight
Analyze
Block
8192
Refresh
60.00
Avg
1.00
Freq
30.0 Hz
200 Hz
1.00 kHz
5.00 kHz
100 Hz
10.0 kHz
5.00 kHz
18.0 kHz
Gain
0.00 dB
Q
0.71
Mode
Stereo
Edit
A
Adapt. Q
On
Scale
100 %
Gain
0.00 dB
Drop Audio Effects Here
1-Engine 2

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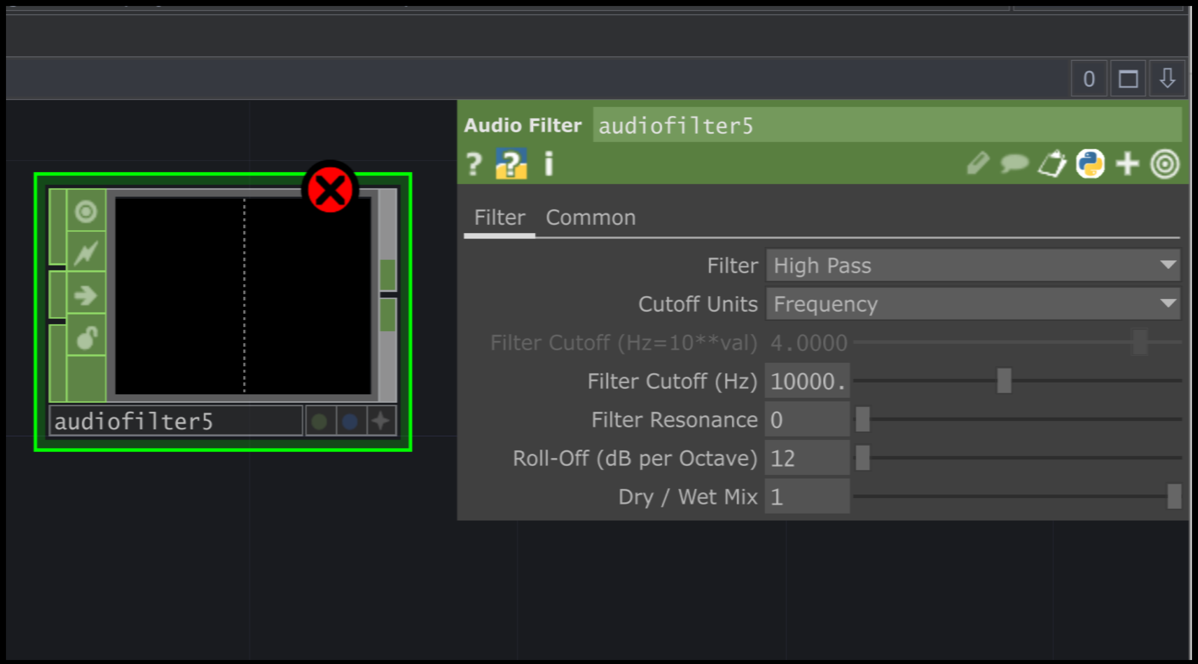
HIGH PASS

- A *HIGH PASS* filter sets a threshold frequency and every frequency above that can travel through the filter




12
6
0
-6
-12
100
1k
10k
EQ Eight
Analyze
Block
8192
Refresh
60.00
Avg
1.00
Freq
30.0 Hz
200 Hz
1.00 kHz
5.00 kHz
100 Hz
10.0 kHz
5.00 kHz
18.0 kHz
Gain
0.00 dB
Q
0.71
Mode
Stereo
Edit
A
Adapt. Q
On
Scale
100 %
Gain
0.00 dB
Drop Audio Effects Here
Key
MIDI
4 %
1-Engine 2

11-11-2025

2. 4. 2
1. 1. 1
4. 0. 0
Key
MIDI
4 %
12
6
0
-6
-12
100
1k
10k
1
2
3
4
EQ Eight
Analyze
Block
8192
Refresh
60.00
Avg
1.00
Freq
30.0 Hz
200 Hz
1.00 kHz
5.00 kHz
100 Hz
10.0 kHz
5.00 kHz
18.0 kHz
Gain
0.00 dB
Q
0.71
1
2
3
4
5
6
7
8
Mode
Stereo
Edit
A
Adapt. Q
On
Scale
100 %
Gain
0.00 dB
Drop Audio Effects Here
1-Engine 2

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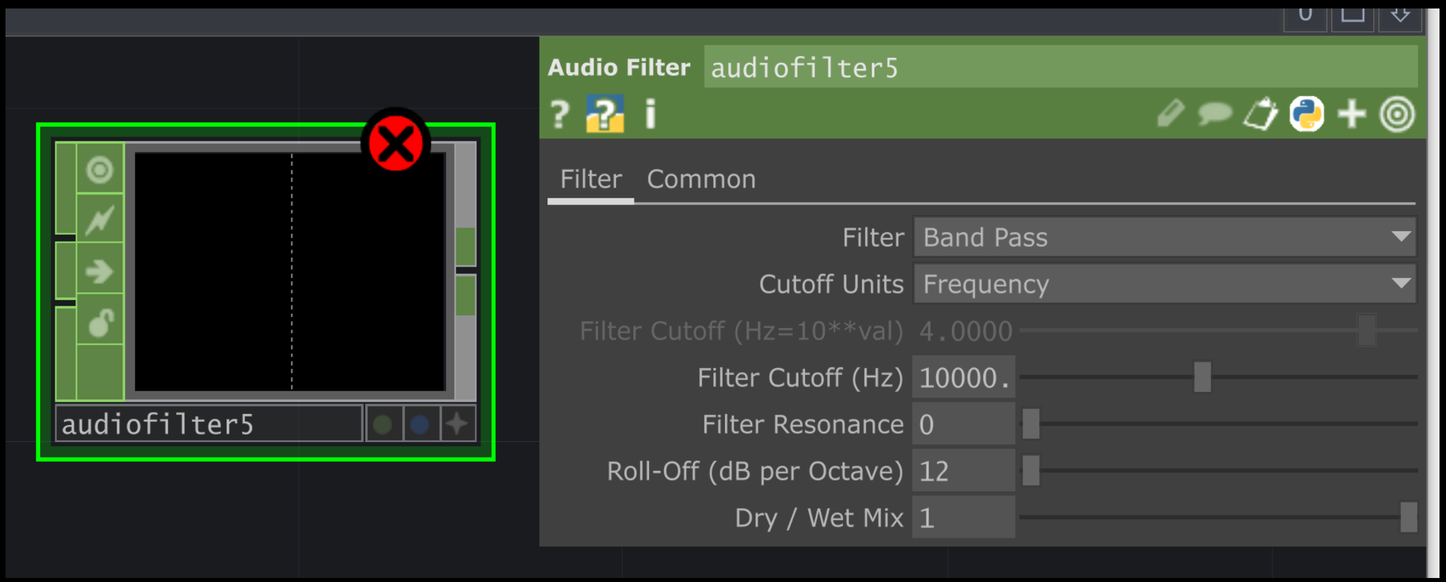
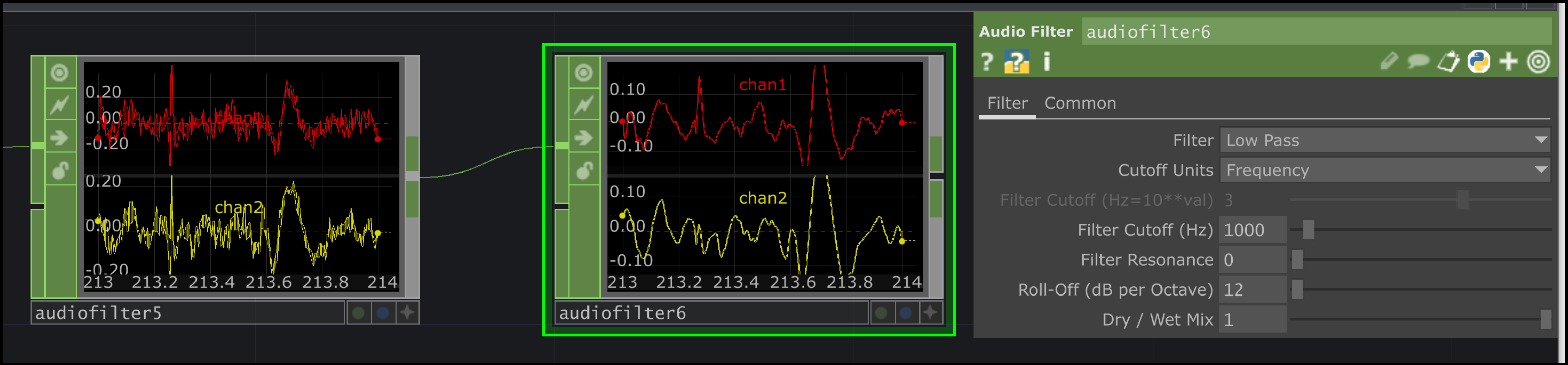
BAND PASS

- A *BAND PASS* filter sets two threshold frequencies and every frequency between those can travel through the filter
- The band pass in Touchdesigner doesn't work so well



WE CAN ALSO CHAIN A HIGH AND LOW PASS TO CREATE A BAND PASS



First we place a high pass, thresholding the lower end of the band
Second we place the low pass, thresholding the higher end of the band.

So if we want a frequency band of 400 to 600 hz it looks like so:

High pass threshold: 400   Low pass threshold: 600

12
6
0
-6
-12
100
1k
10k
EQ Eight
Analyze
Block
8192
Refresh
60.00
Avg
1.00
Freq
30.0 Hz
200 Hz
1.00 kHz
5.00 kHz
100 Hz
10.0 kHz
5.00 kHz
18.0 kHz
Gain
0.00 dB
Q
0.71
Mode
Stereo
Edit
A
Adapt. Q
On
Scale
100 %
Gain
0.00 dB
Drop Audio Effects Here
Key
MIDI
4 %
1-Engine 2

11-11-2025

3. 2. 4
1. 1. 1
4. 0. 0
Key
MIDI
4 %
12
6
0
-6
-12
489 Hz
B3
-7.97 dB
100
1k
10k
EQ Eight
Analyze
Block
8192
Refresh
60.00
Avg
1.00
Freq
30.0 Hz
83.9 Hz
130 Hz
5.00 kHz
100 Hz
10.0 kHz
5.00 kHz
18.0 kHz
Gain
0.00 dB
0.06 dB
0.00 dB
0.00 dB
0.00 dB
0.00 dB
0.00 dB
0.00 dB
Q
0.71
0.71
0.75
0.71
0.71
0.71
0.71
0.71
1
2
3
4
5
6
7
8
Mode
Stereo
Edit
A
Adapt. Q
On
Scale
100 %
Gain
0.00 dB
Drop Audio Effects Here
1-Engine 2

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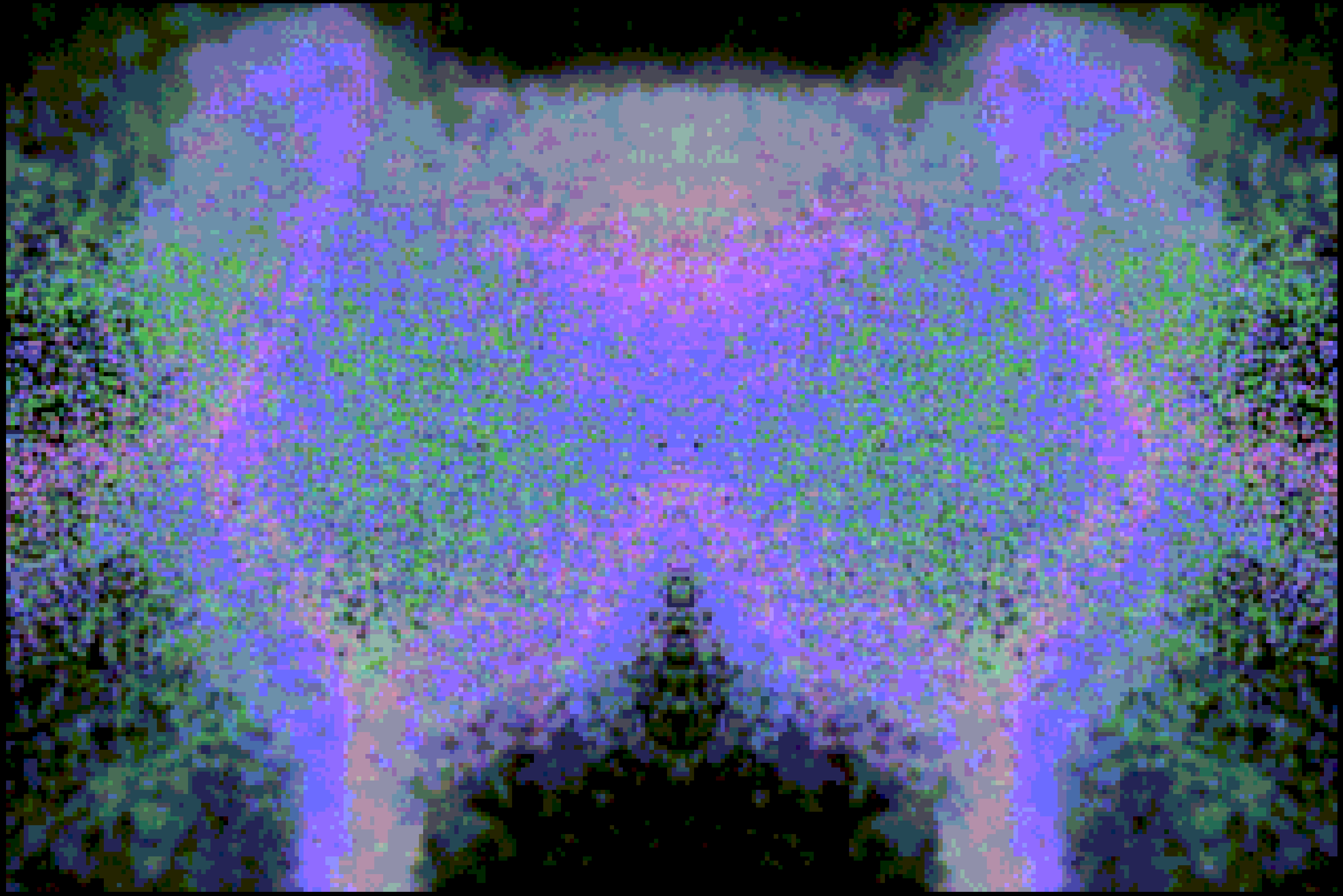
CASE: AUDIO REACTIVE PARTICLE CLOUD



CASE: AUDIO REACTIVE PARTICLE CLOUD

- Audio file
- Audio analysis network
- Particle system
- Post processing