

Introduction to Audio Programming

Meet the Masters, SAE Bochum

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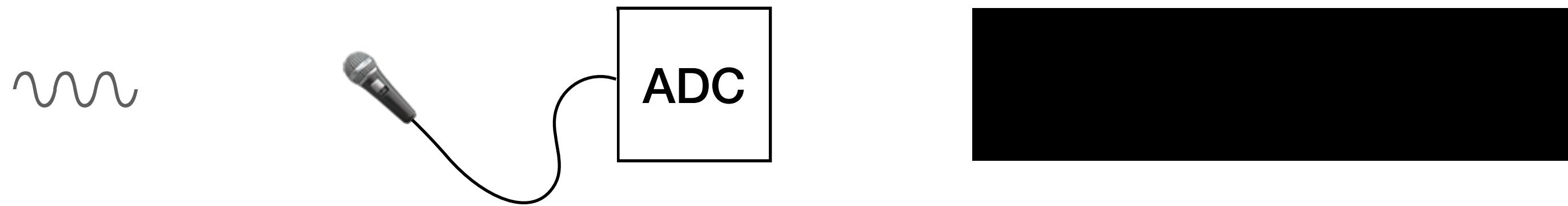


@marcusficner

What is audio programming?

What is digital audio?





Sampling Frequency: F_s

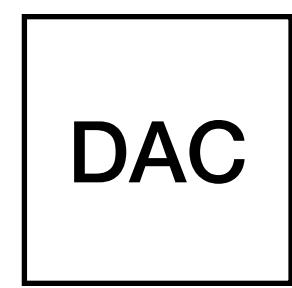
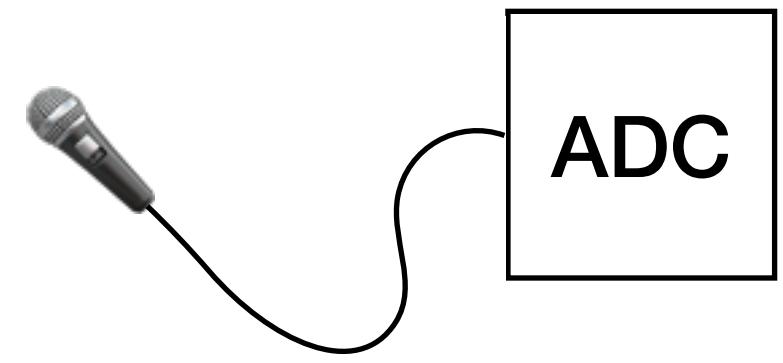
Resolution: Q

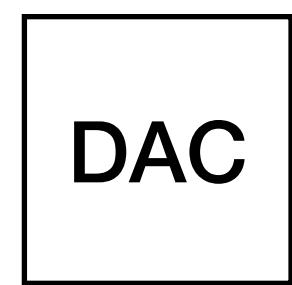
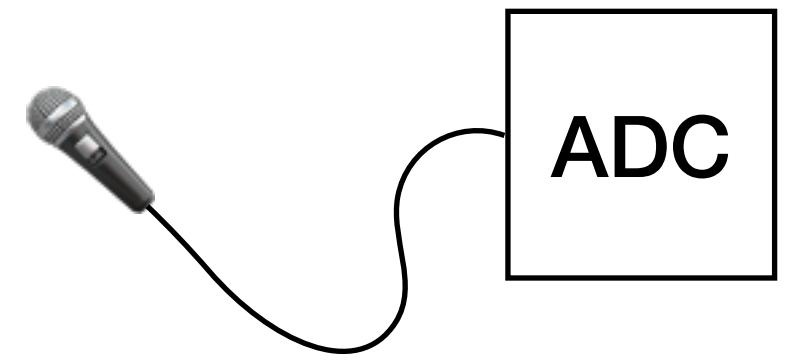


Human hearing range: ~ 20 Hz - 20 kHz

$F_s = 44100 \text{ Hz}$

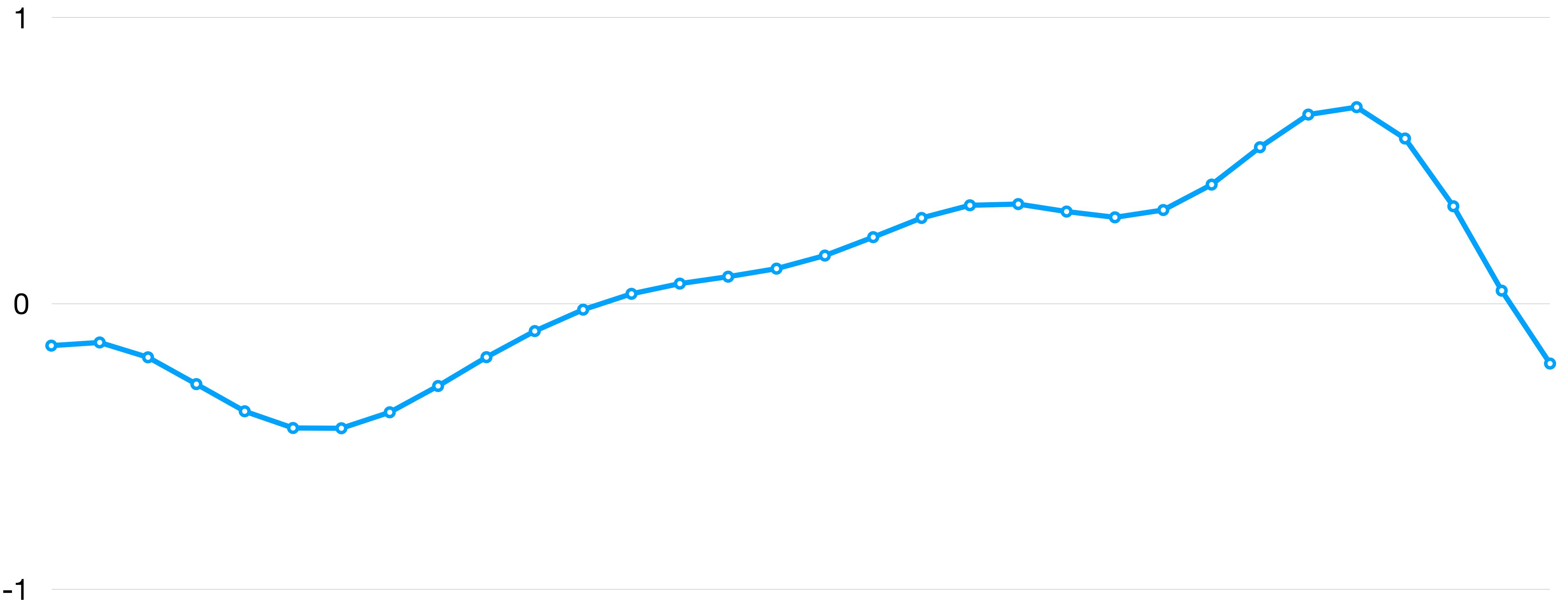
$Q = 16 \text{ Bit}$



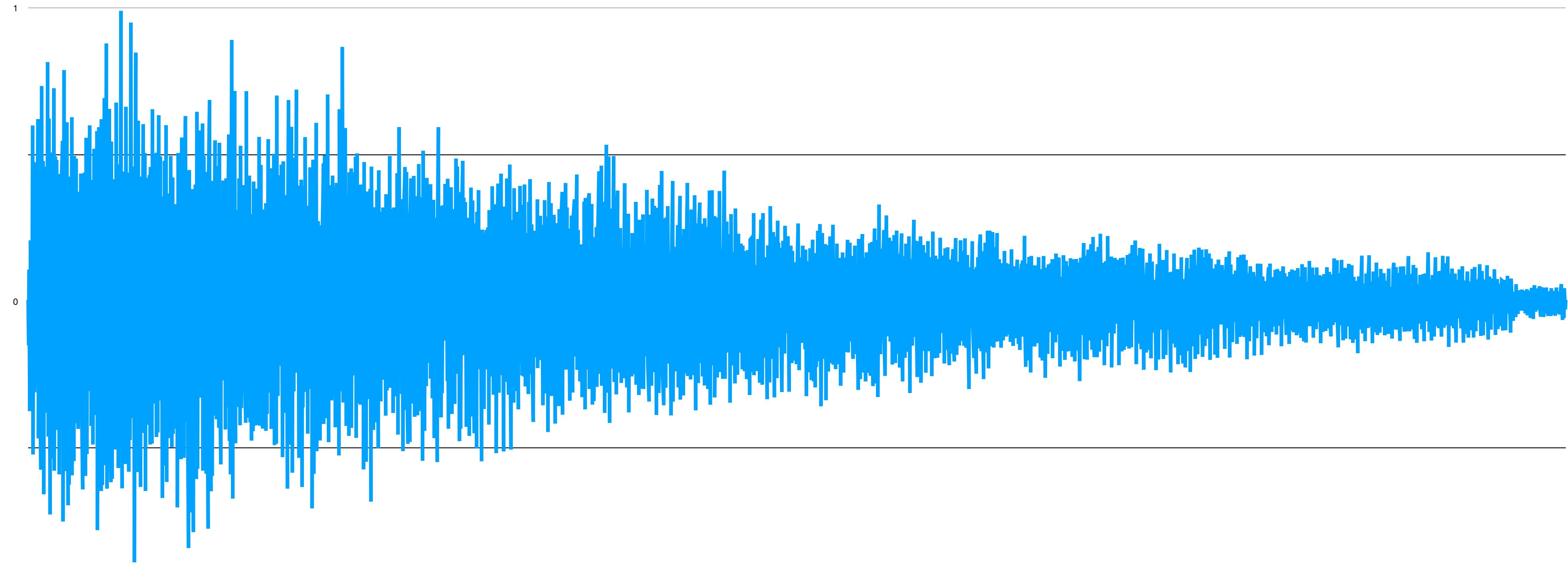


Digital Audio

-0.15	-0.14	-0.19	-0.28	-0.38	-0.44	-0.44	-0.38	-0.29	-0.19	-0.10	-0.02	0.03	0.07	0.09	0.12	0.17	0.23	0.30	0.34	0.35	0.32	0.30	0.33	0.42	0.55	0.66	0.69	0.58	0.34	0.04	-0.21
-------	-------	-------	-------	-------	-------	-------	-------	-------	-------	-------	-------	------	------	------	------	------	------	------	------	------	------	------	------	------	------	------	------	------	------	------	-------

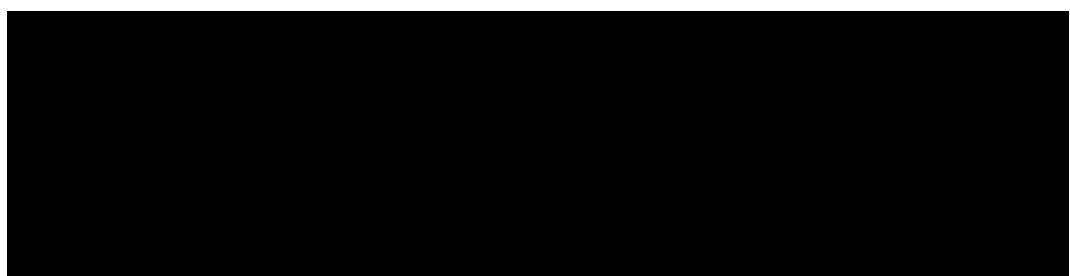
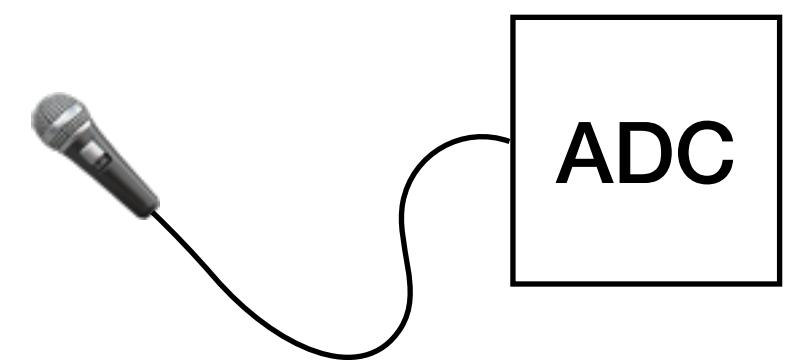


Digital Audio





Visualize



Store



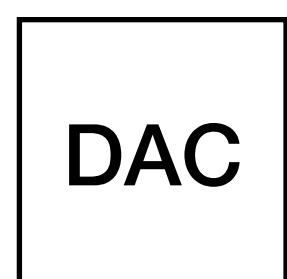
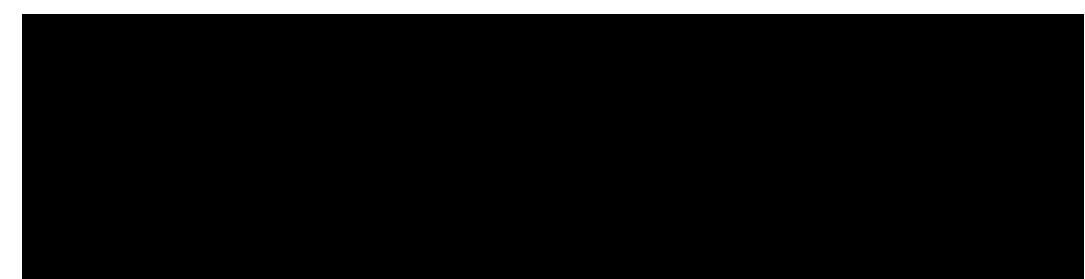
Transmit



Process



Synthesize



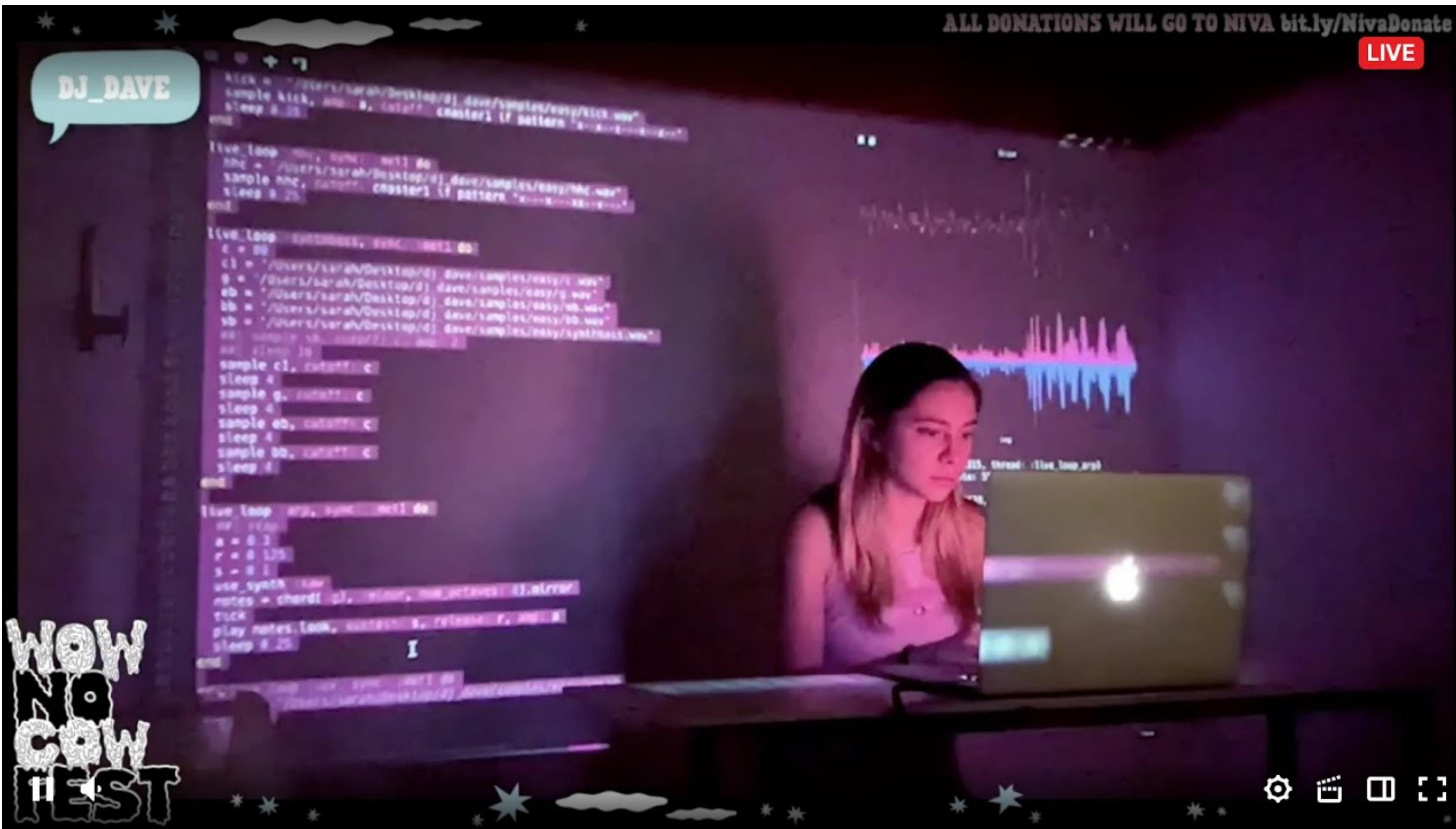
Load Receive

What is audio programming?

Audio Programming

- Live Coding / Algo raves
- Game Audio
- Audio hardware / embedded programming
- Information extraction
- Machine Learning
- Composing
- Music Production

Live coding / Algo Rave



<https://www.sarahcdavis.com/dj-dave>

Live coding / Algo Rave

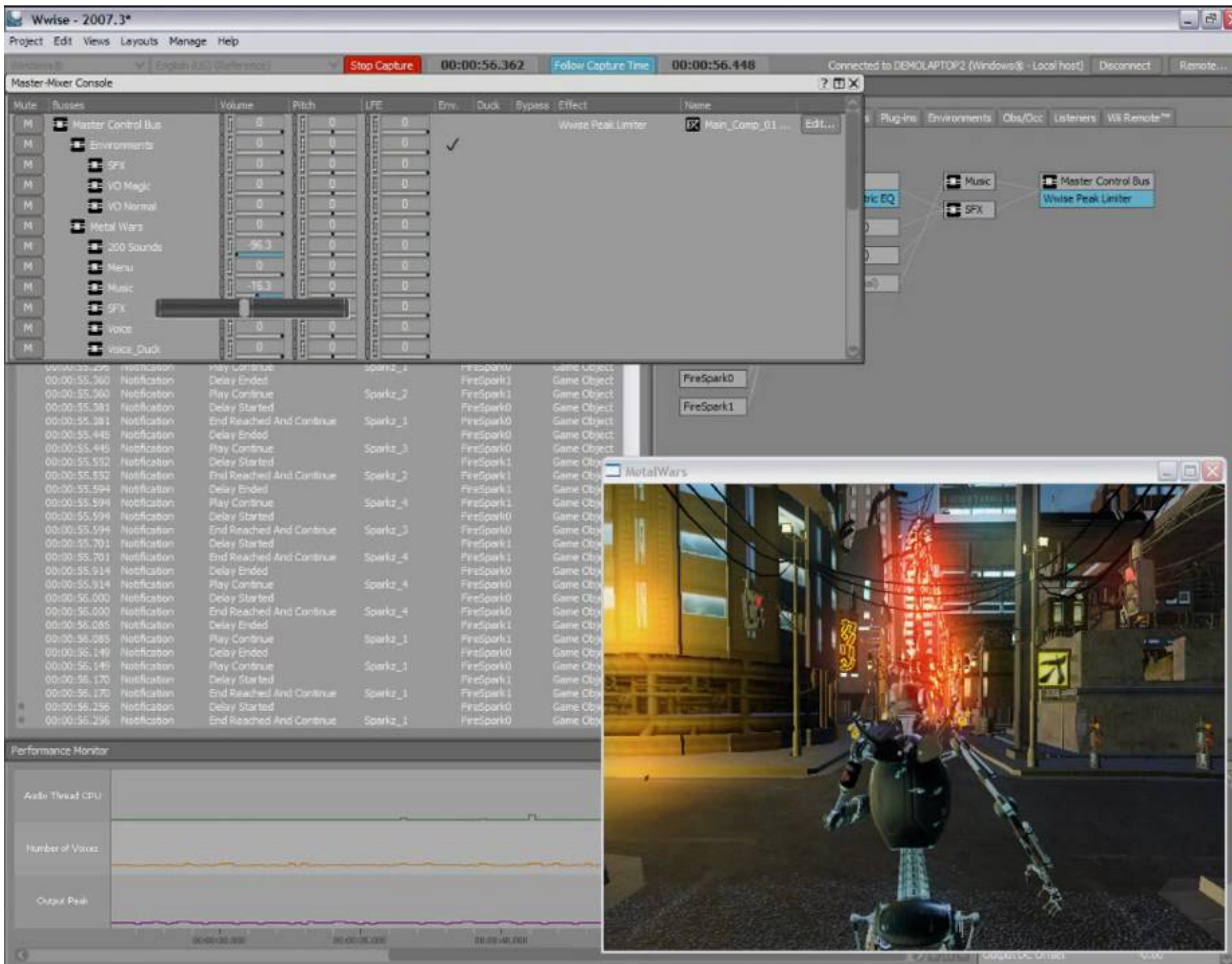


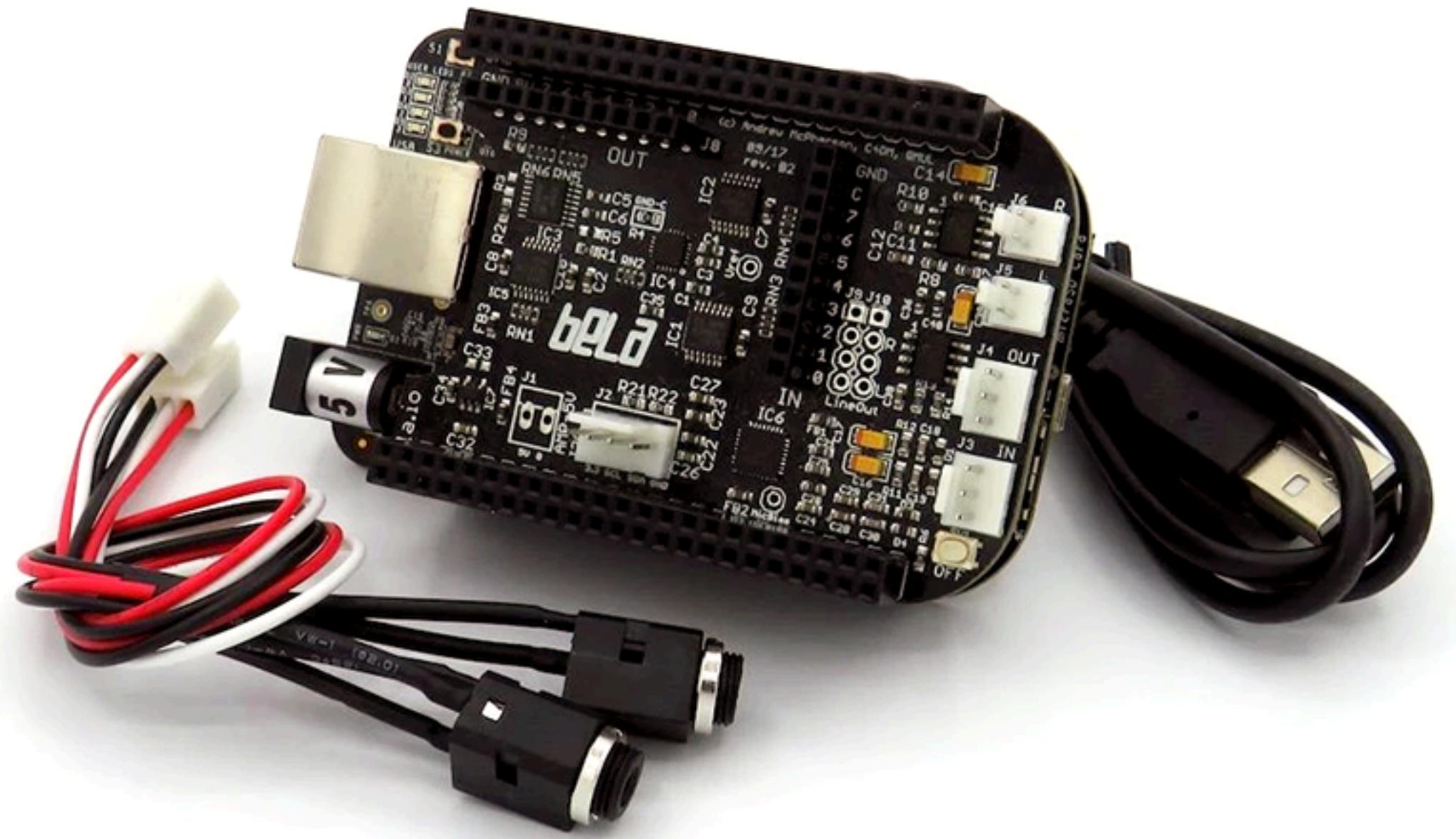
<https://sonic-pi.net>



Game Audio

Game Audio





Audio Hardware



Voice recognition

Music classification

Acoustic fingerprint

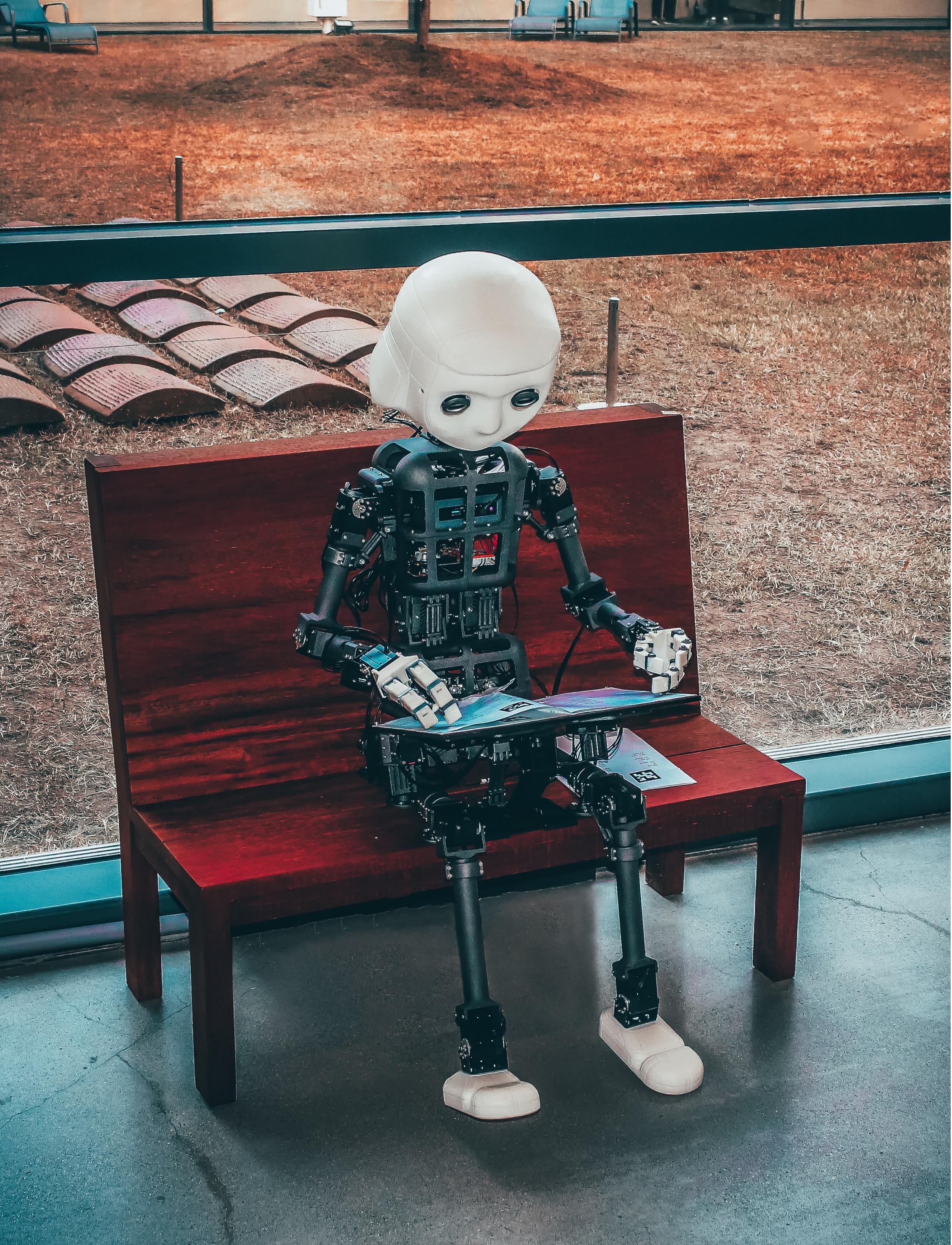
Information extraction

Automatic music transcription

BPM / Beat detection

Music source separation and instrument recognition

Machine Learning



Palety Rozšířená

Přidat palety Více

Klíče

Správce

Prvek

- Viditelný Barva: [Solid Black]
- Automatické umístění
- Nejmenší odstup: 0,00 sp
- X: 0,00sp
- Posun: Y: 0,00sp
- Pořadí vrstvy (Z): 2000

Segment

- Místo vlevo: 0,00 sp

Akord

- X: 0,00 sp
- Posun: Y: 0,00 sp

Kacionál č. 982 - Bože chválíme tebe

Kacionál č. 982

Bože chválíme tebe

Olomoucká verze: BC 1938

Adagio

1. Bo - že, chvá - lí - me te - be, Pa - ne, moc tvou

Composing

Palety Rozšířená

Více

Ozdobné noty

Notové hlavičky

Linky

Taktové čáry

Arpeggia a glissanda

Tremola

Text

Tempo

Dynamika

Prstoklady

Opakování a skoky

Hmatové diagramy

Typ velocity: Posun

Velocity: 0

Vybrat

Tečka 1 Tečka 2 Tečka 3 Tečka 4

Nožička Praporek Trámeč Nepravidelná rytmická skupina

všem tvým skut - kům se di - ví - me; když se vše v svě -
ne - be ze - mě za - stu - po - vé, ode všech jsi
Ne - be, zem, po-vě-tří, vo - dy pl - né jsou cti,
po - dle na - še - ho do - u - fá - ní. Kdo v tě do - u - fá.

10

tě mě - ní, ty sám jsi bez pro - mě - ny.
na - zý - ván: Sva - tý, Sva - tý, Sva - tý Pán.
chvá - ly tvé, neb vše tvo - je dí - lo je.
sa - mé - ho ne - o - pus - tíš žád - né - ho



Music Production



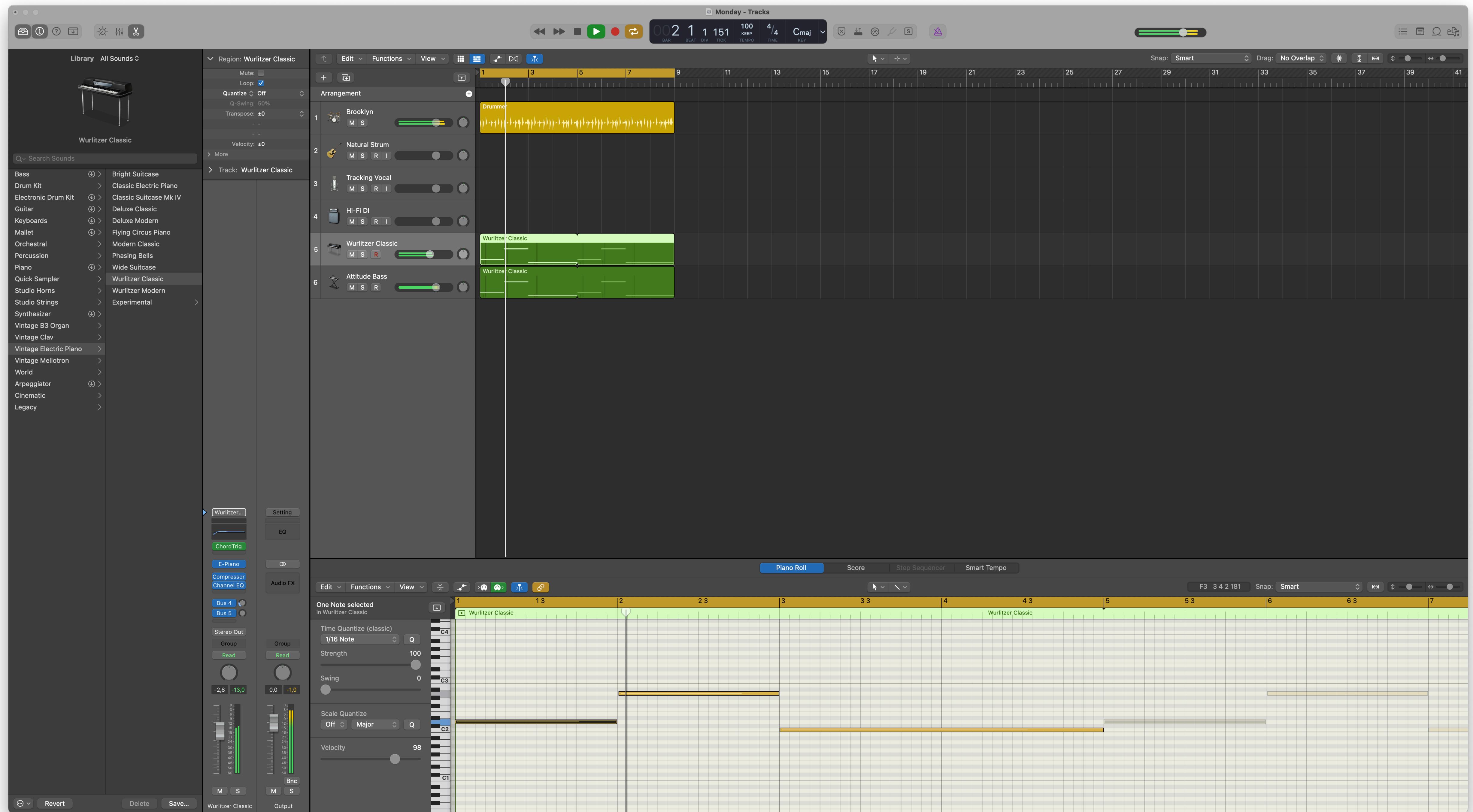
Audio Plug-ins

The screenshot shows the Logic Pro X interface with the following sections visible:

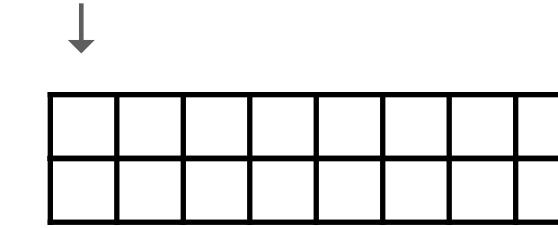
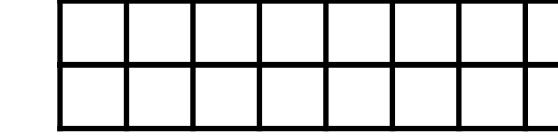
- SYNTH:** Pitch section with Tune and Fine knobs, Filter 1 (LP 12dB Lush (Fat)) with Cutoff, Reso, and Drive knobs, Filter 2 (LP 12dB Lush (Fat)) with Cutoff, Reso, and Drive knobs, and AMP section with Volume and Pan knobs.
- MOD MATRIX:** A table showing modulation assignments:
 - Velocity (Source) to Sample Select (Target) with +100.0% Amount.
 - Ctrl #64 (Source) to Env 1 Release (Amp) (Target) with +436 ms Amount.
 - Key (Source) to Volume (Target) with -50.8% Amount.
 - Key (Source) to Filter 1 Cutoff (Target) with +6.0% Amount.
- MODULATORS:** ENV 1 AMP, ADSR 2, ADSR 1, LFO 1, and LFO 2 sections.

The screenshot shows the Wurlitzer Classic audio plug-in interface with the following sections:

- EFFECTS:** Bass Boost and Volume knobs.
- DETAILS:** EQ (Bass and Treble), Drive (Gain and Tone), and Chorus (Rate and Intensity) sections.

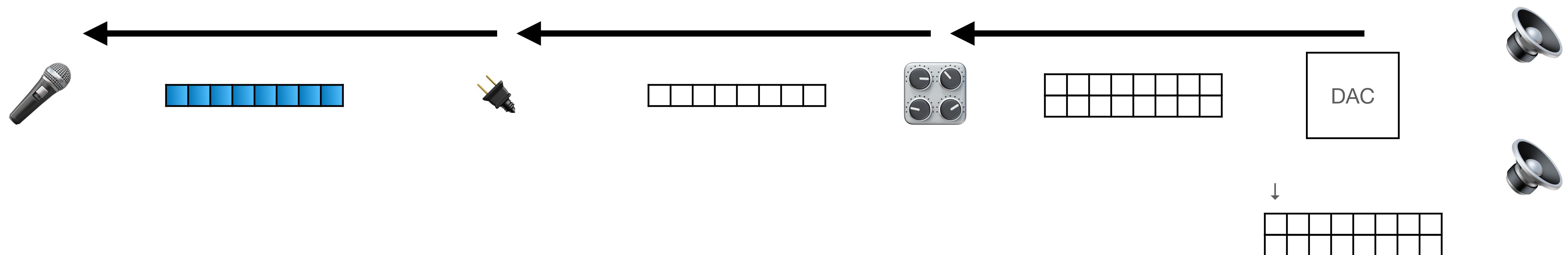


Audio Graph



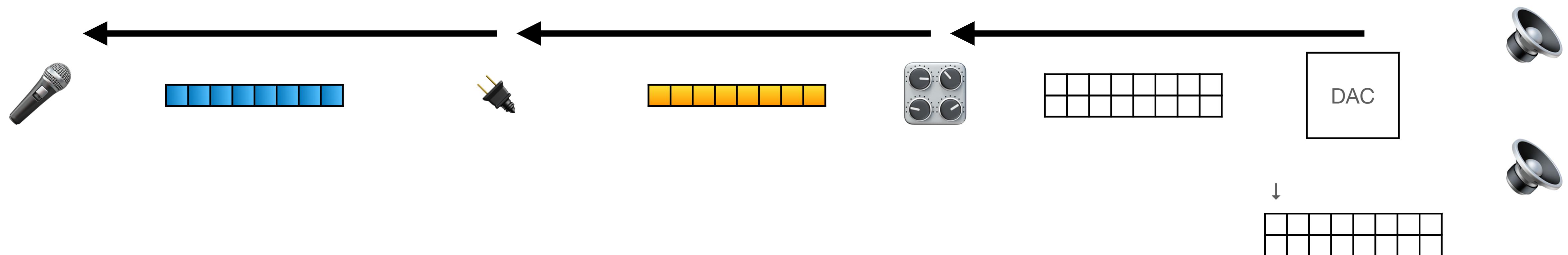
Audio Graph

Pull Model



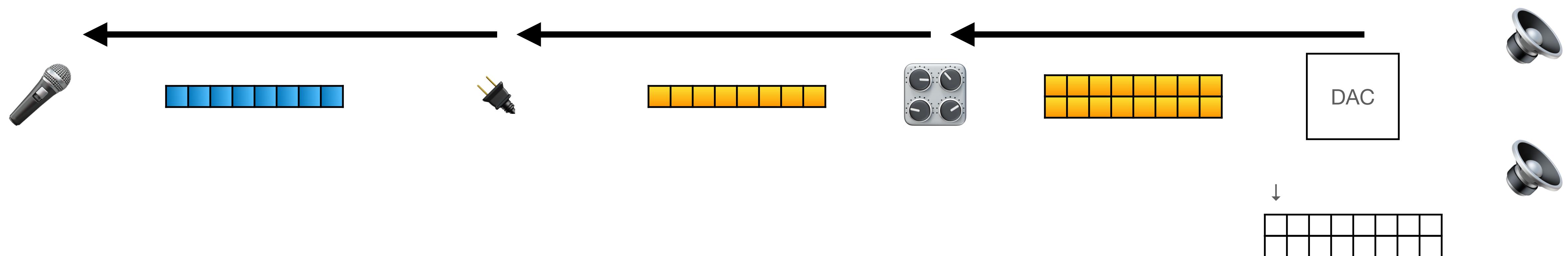
Audio Graph

Pull Model



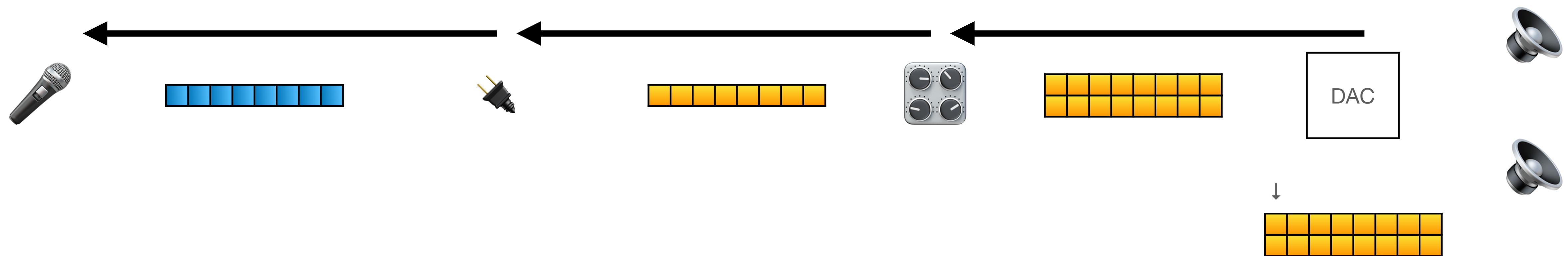
Audio Graph

Pull Model

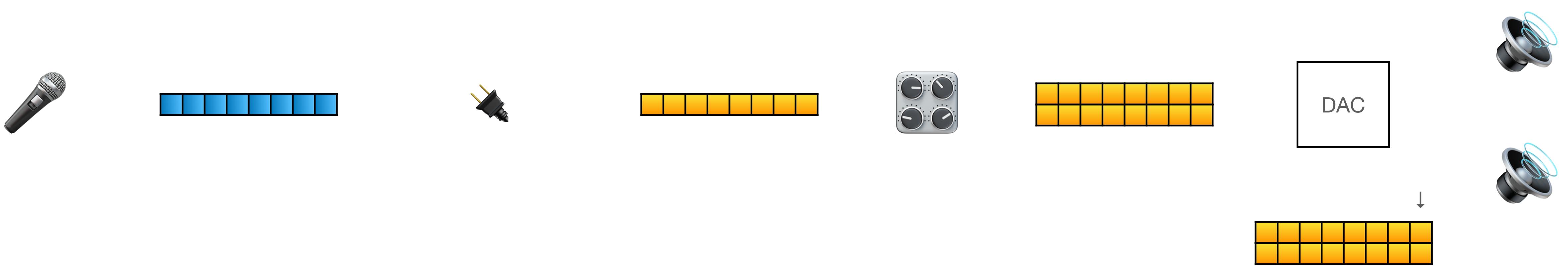


Audio Graph

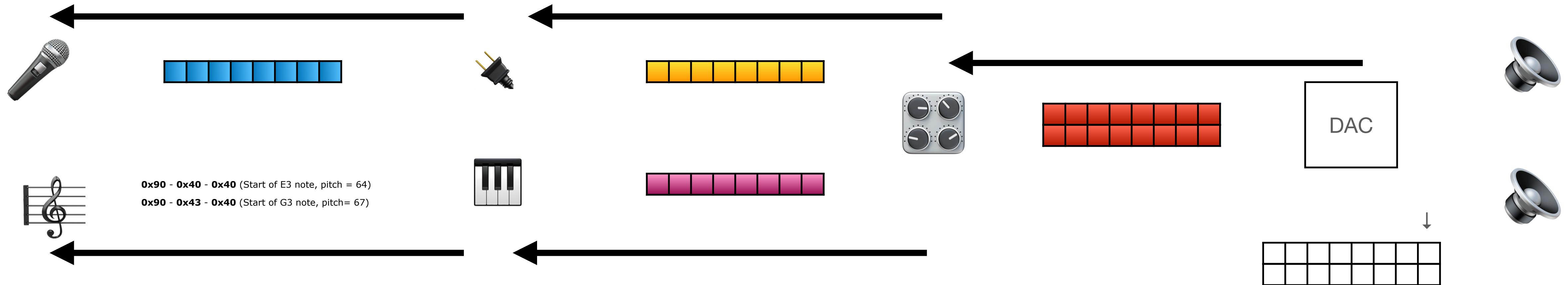
Pull Model



Audio Graph



Audio Graph



Demo



JUCE

Why JUCE?



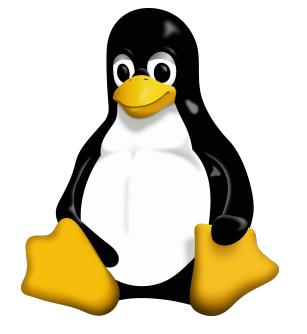
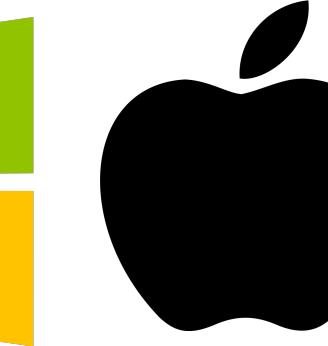
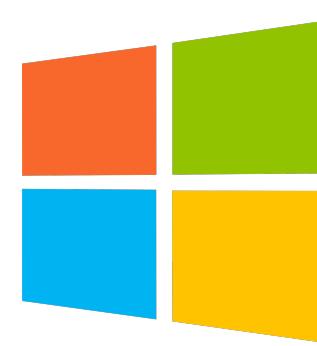
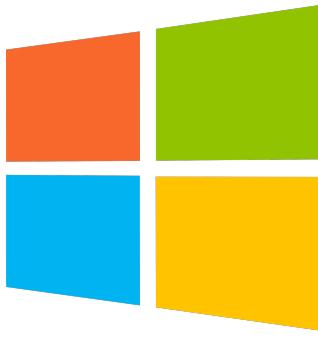
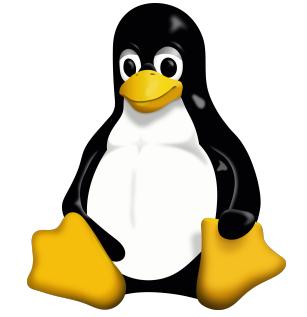
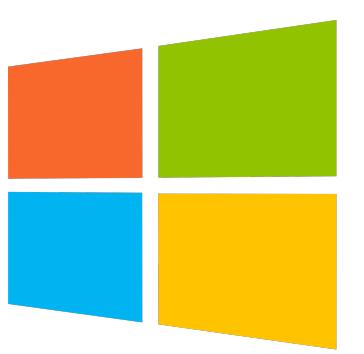
Audio Units



RTAS



AAX



**each framework and operating system
offers a different API and programming model**

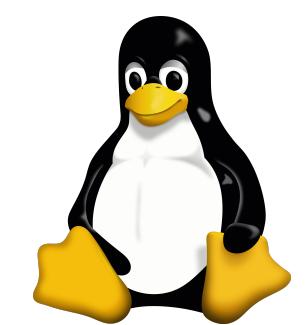
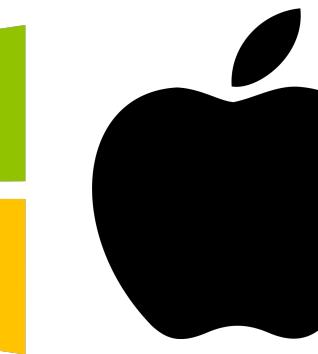
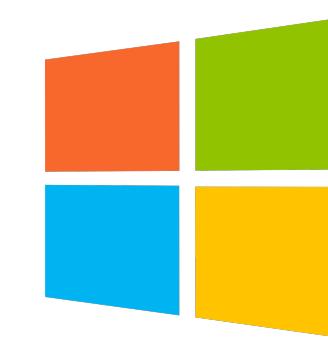
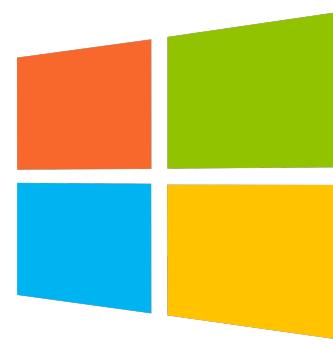
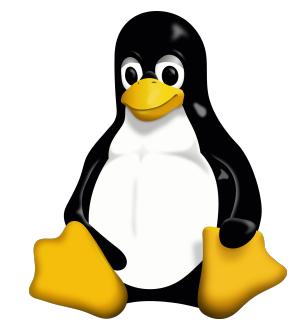
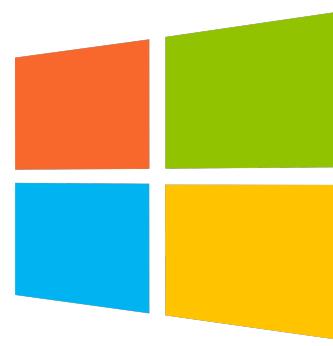
Cross platform development



Audio Units



RTAS



iPlug2

<https://github.com/iPlug2/iPlug2>

in-house frameworks



AudioKit

<https://audiokit.io>

Audio Callback

```
void SimpleDelayAudioProcessor::processBlock (juce::AudioBuffer<float>& buffer, juce::MidiBuffer& midiMessages)
{
    juce::ScopedNoDenormals noDenormals;
    auto totalNumInputChannels = getTotalNumInputChannels();
    auto totalNumOutputChannels = getTotalNumOutputChannels();

    // This is the place where you'd normally do the guts of your plugin's
    // audio processing...
    // Make sure to reset the state if your inner loop is processing
    // the samples and the outer loop is handling the channels.
    // Alternatively, you can process the samples with the channels
    // interleaved by keeping the same state.
    for (int channel = 0; channel < totalNumInputChannels; ++channel)
    {
        auto* channelData = buffer.getWritePointer (channel);

        for (int i = 0; i < buffer.getNumSamples(); ++i)
        {
            channelData[i] *= 0.1f;
        }
    }
}
```

Real-time audio programming 101: time waits for nothing

Anatomy of an audio plug-in

Audio Processor

User Interface

Parameters

Presets

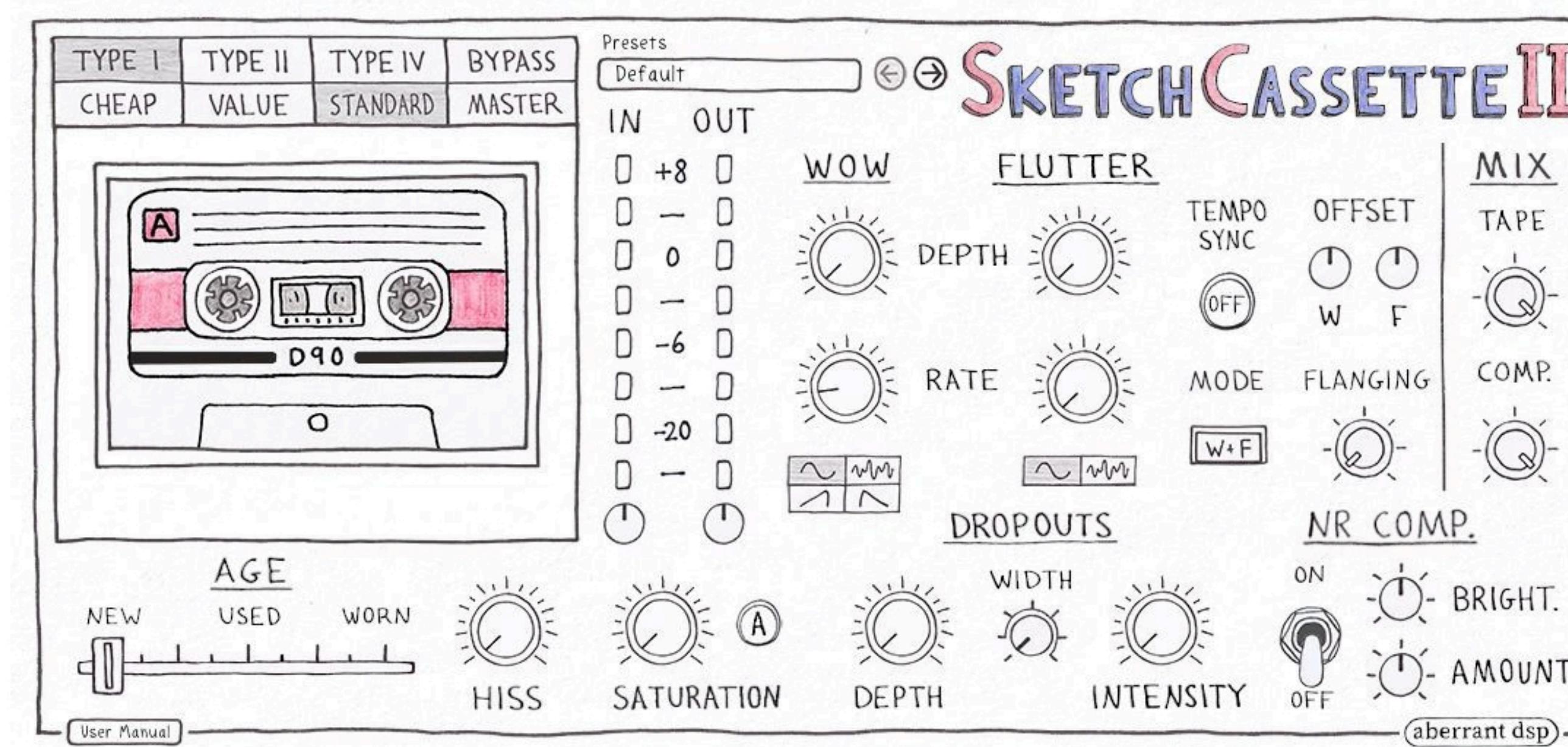
UI Components



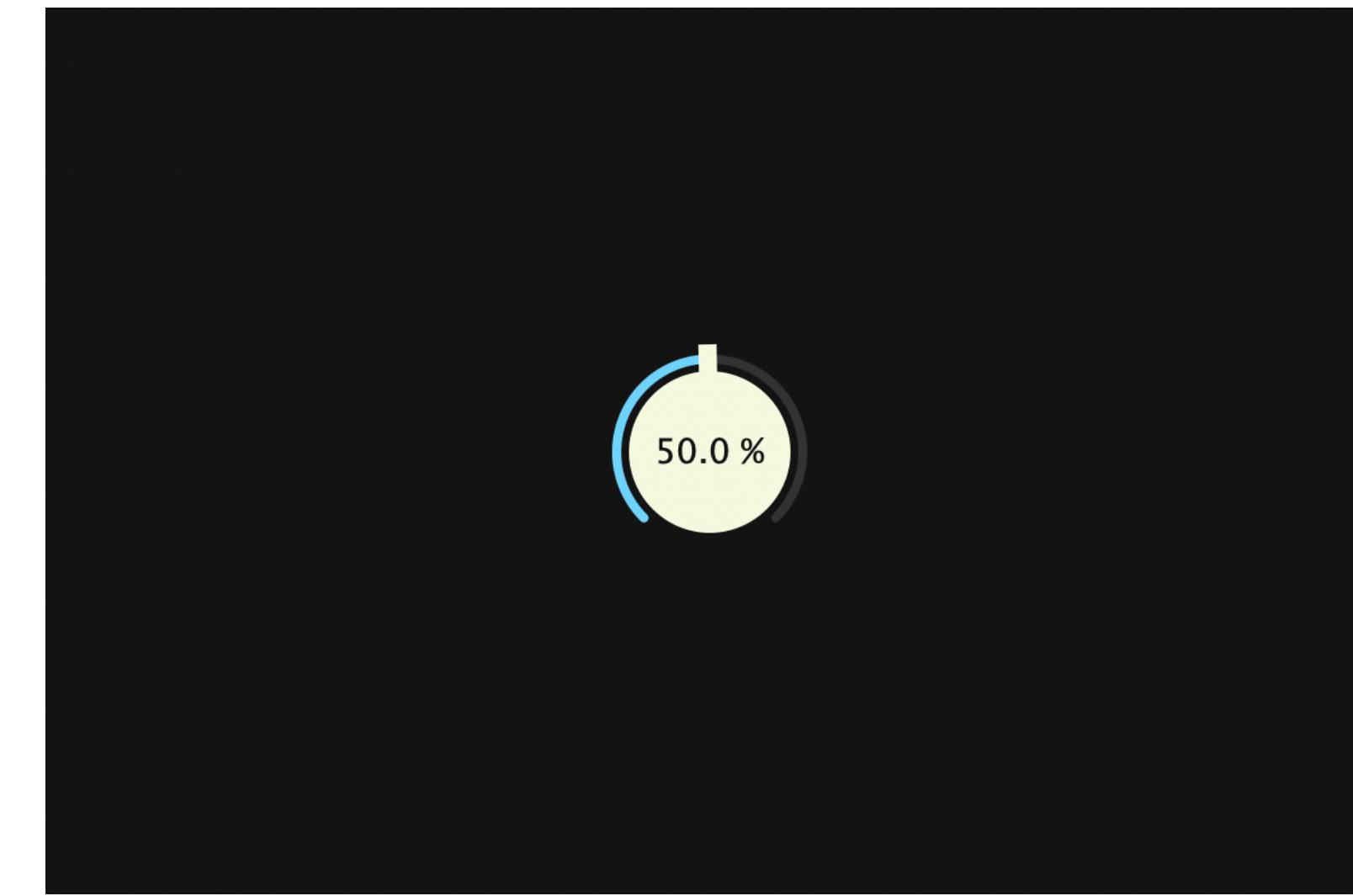
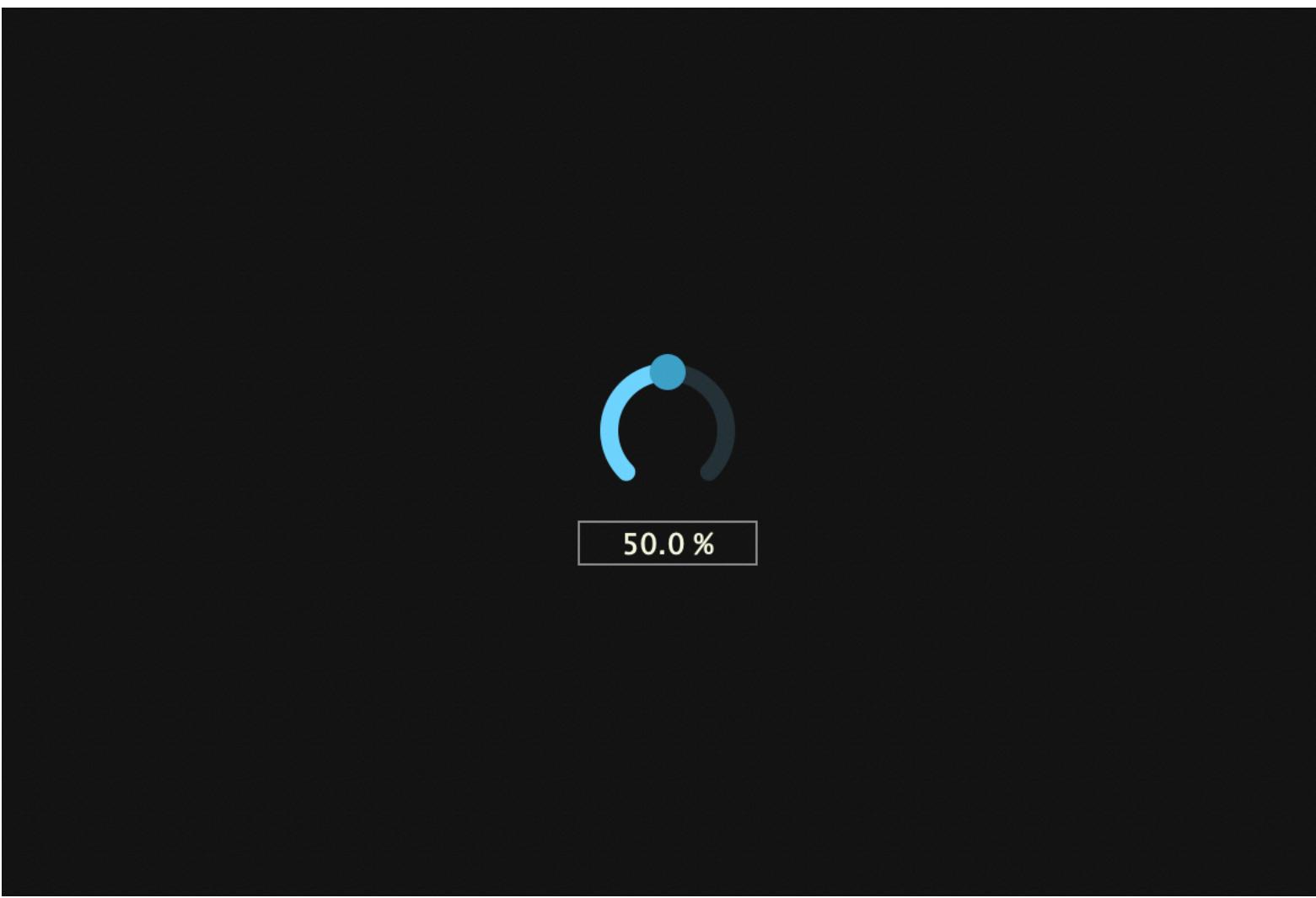
UI Components



UI Components



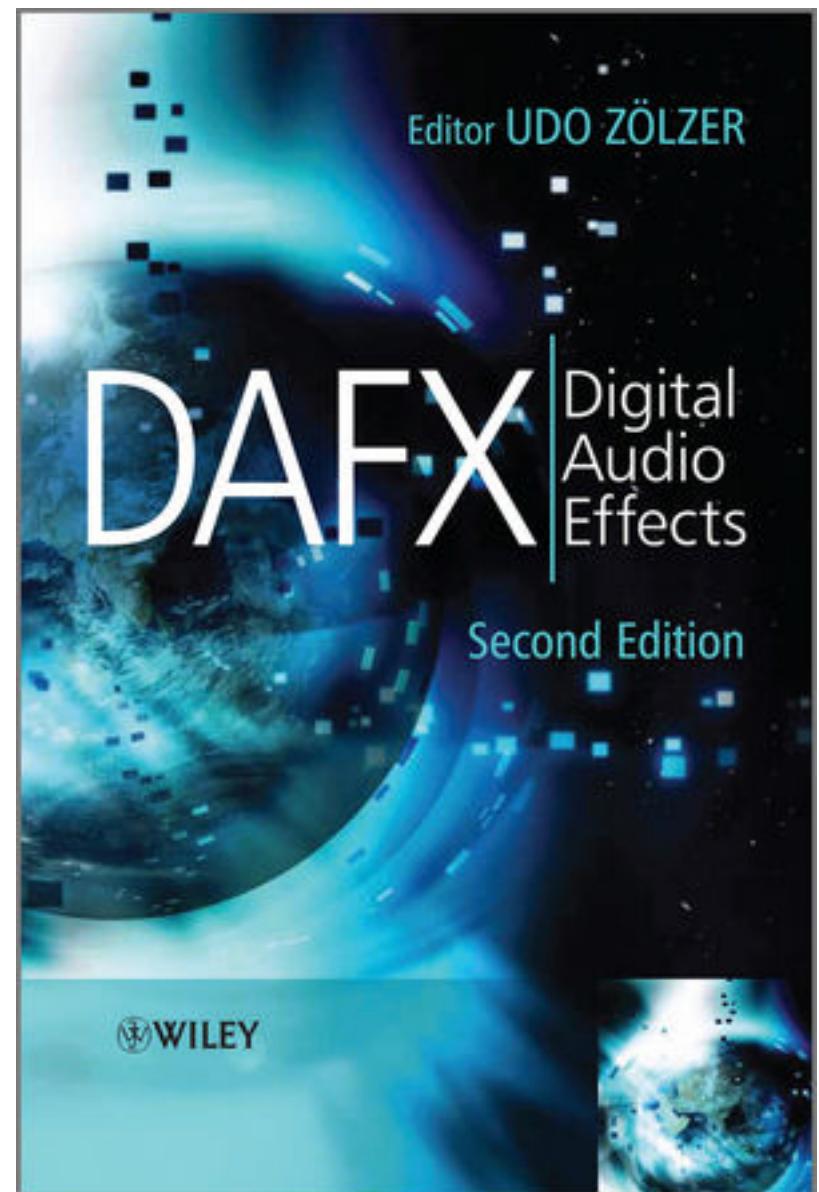
UI Components



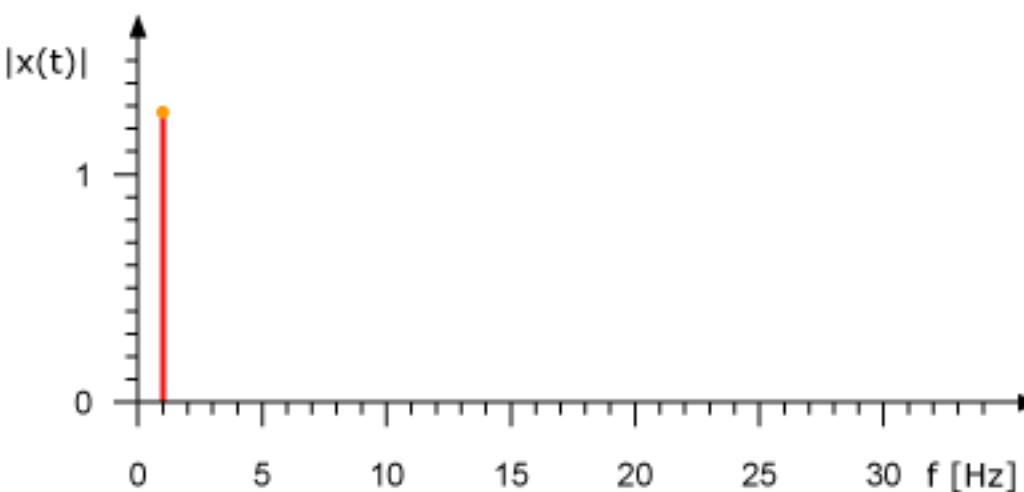
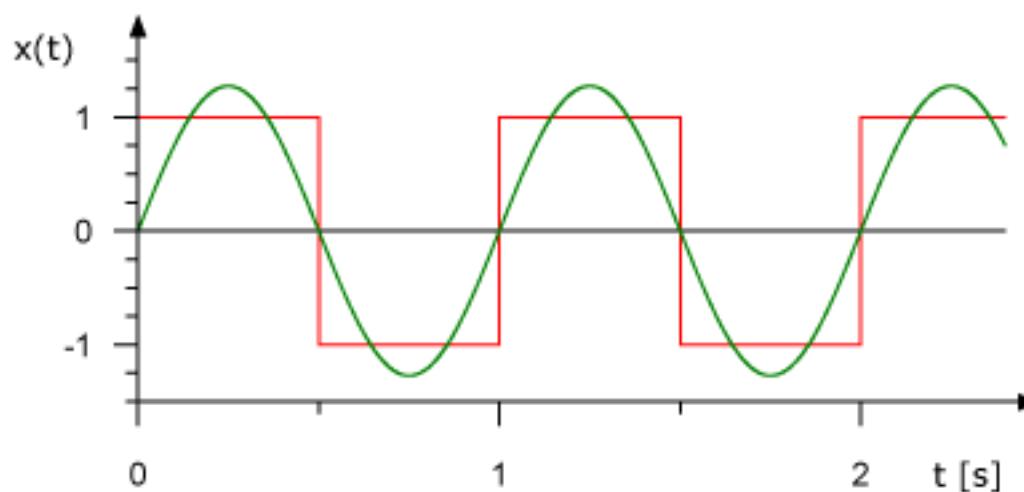
<https://suzuki-kengo.dev/>

Classification of audio effects

- Filters and delays (resampling)
- Modulators and demodulators
- Non-linear processing
- Spatial effects
- Time-segment processing
- Time-frequency processing
- Source-filter processing
- Adaptive effects processing
- Spectral processing
- Time and frequency warping
- Virtual analog effects
- Automatic mixing
- Source separation



Fourier transform

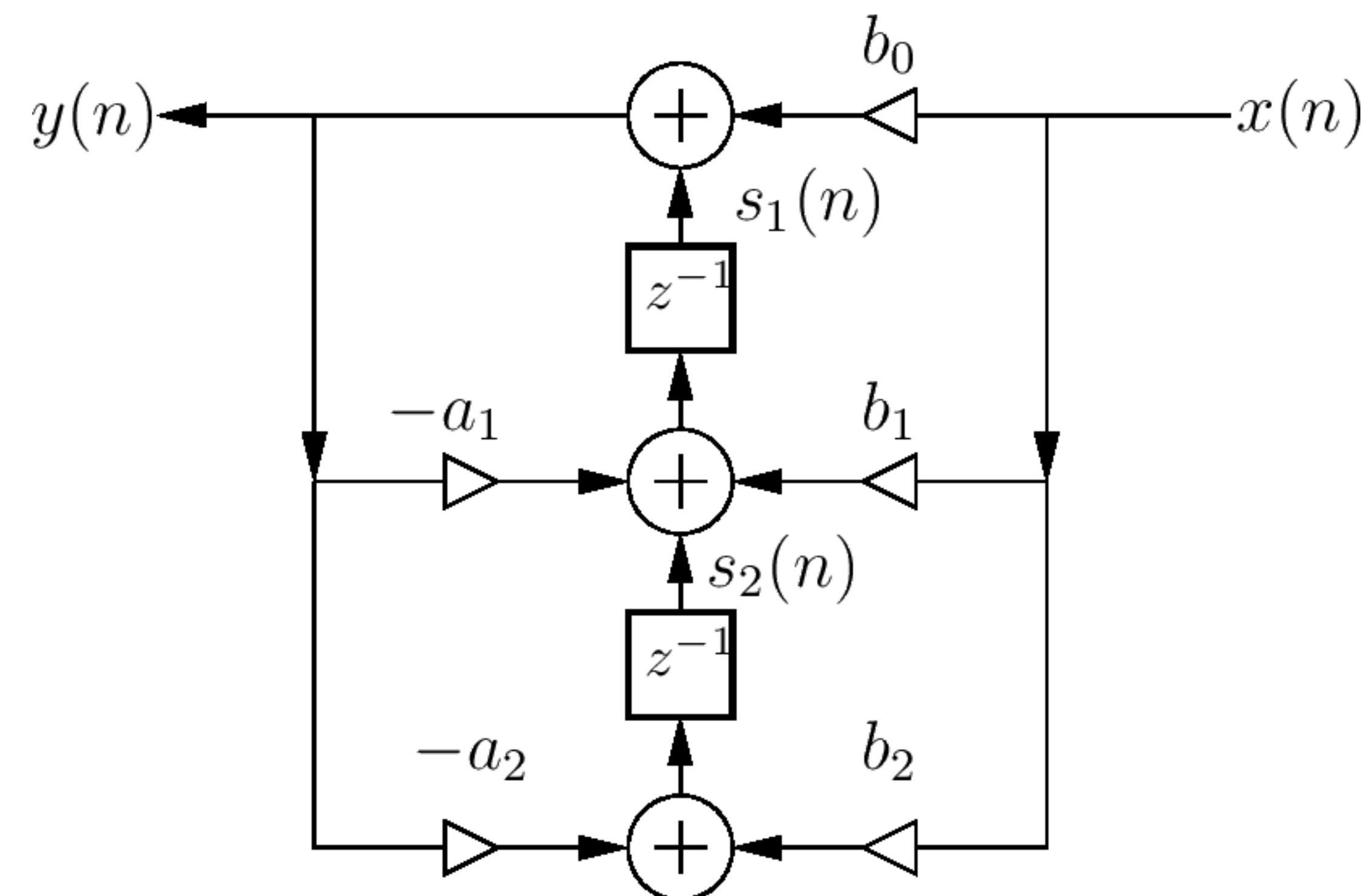


Prototype plug-ins

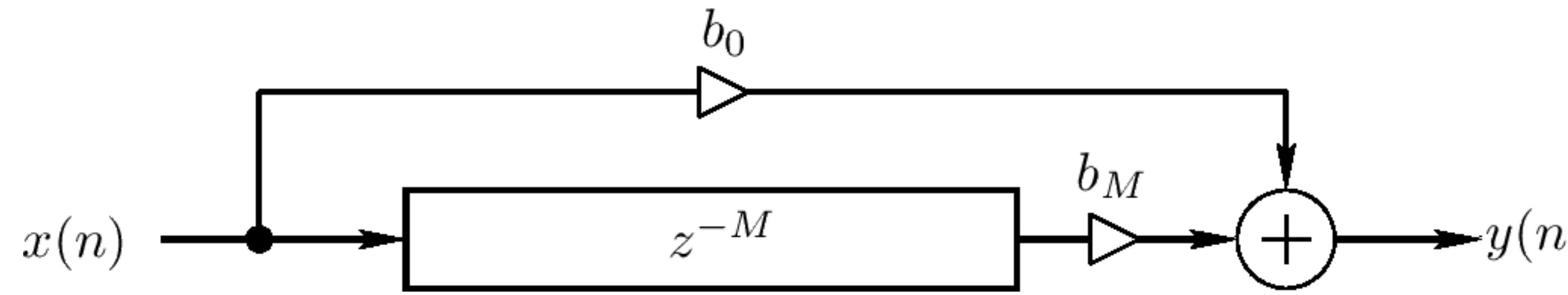
Waveshaping

Waveshaping #2

Block diagrams

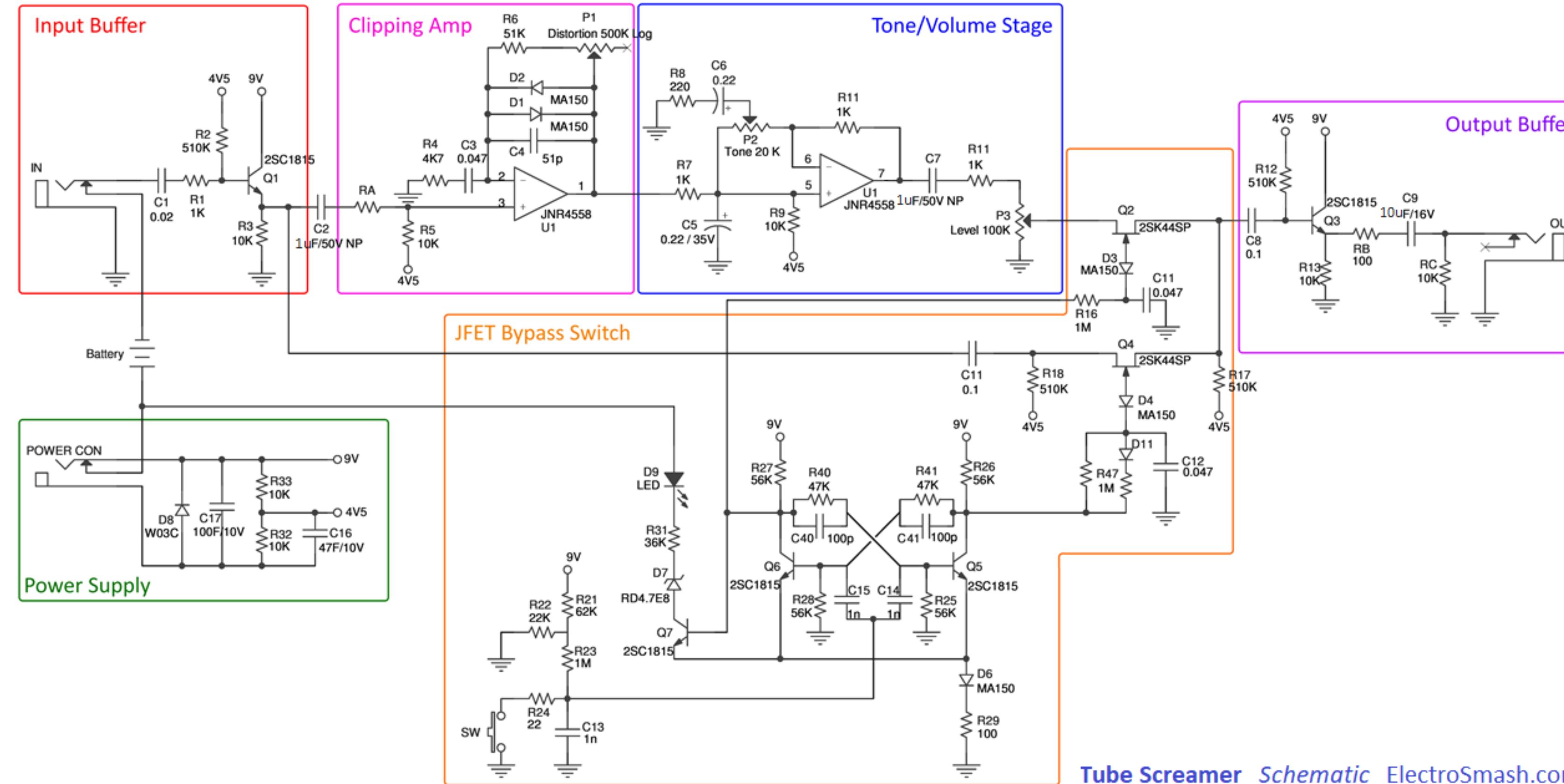


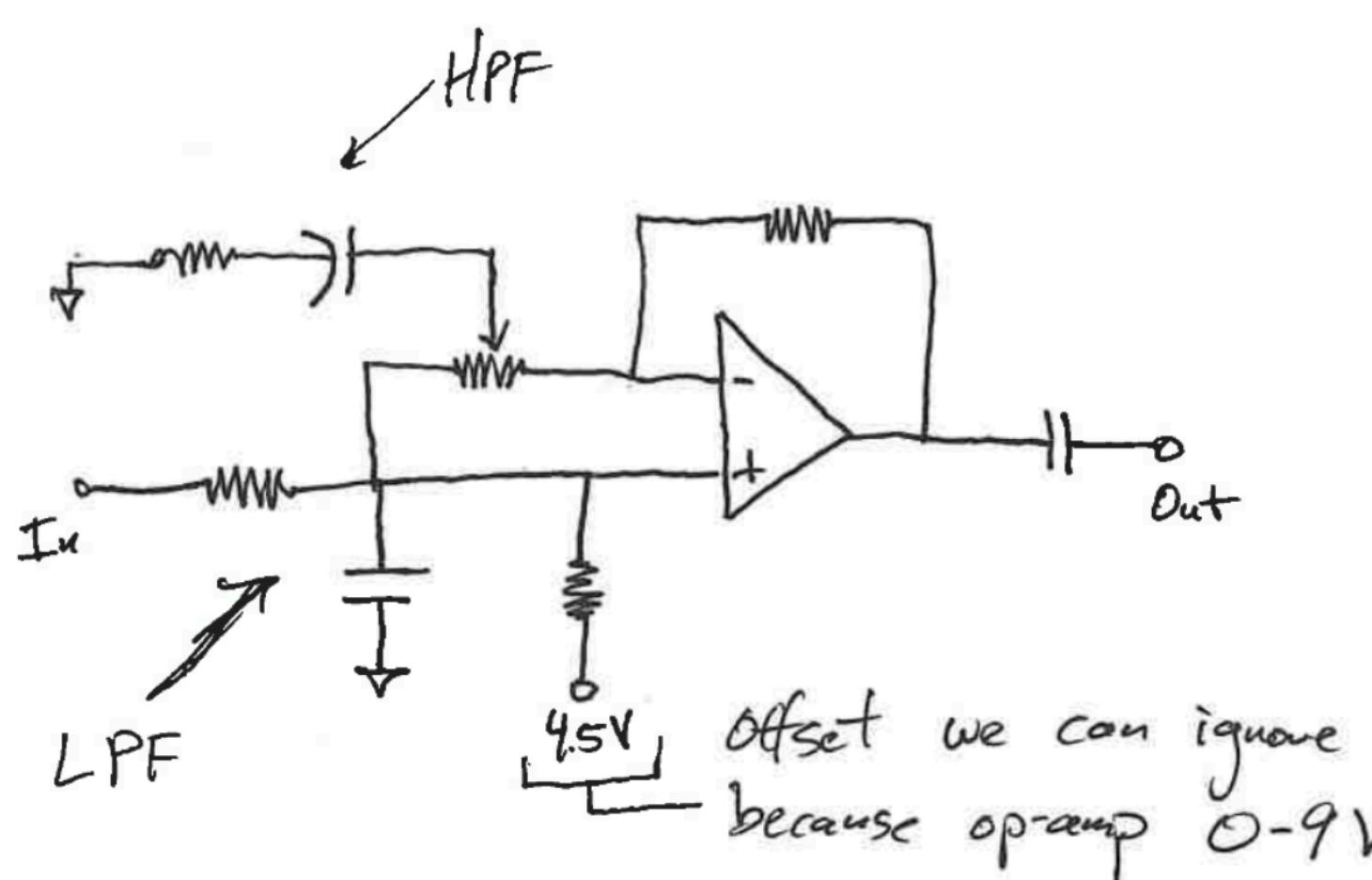
Feedforward Comb Filter



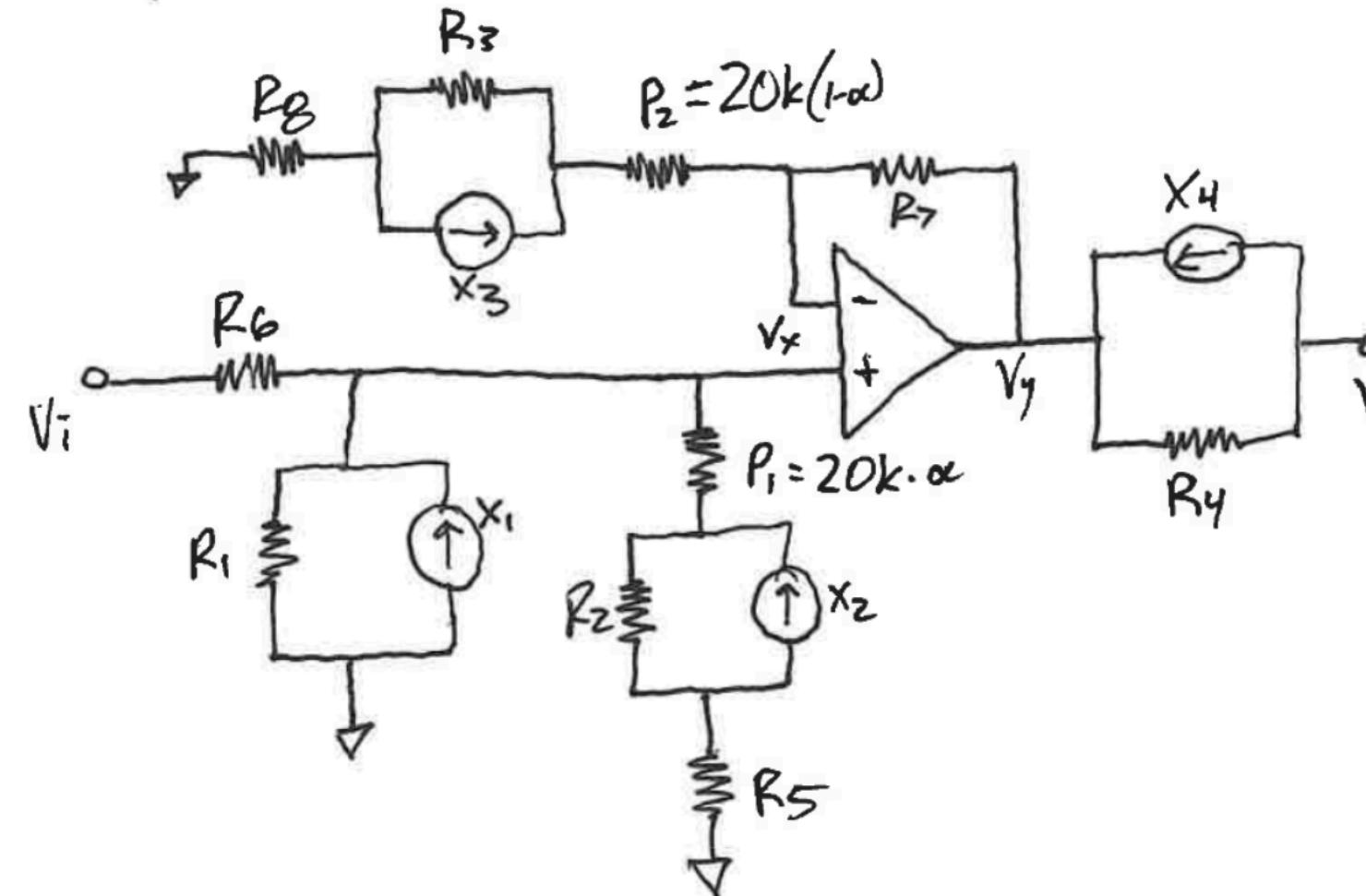
gen~ to JUCE

Analog Modelling





Potentiometer: Tone control single knob
Resistor between "-" & "+" terminals
Assumption no current flow through L/between terminals.



④① ④② Start with sub-circuit
Want current "i" as function of \$V_x\$ & \$X_1\$.

$$V_x = V_{p_1} + V_{p_1} + V_{r_5}$$

same current "i" through each stage.

$$i = \frac{V_{p_1}}{P_1} \quad i = \frac{V_{r_5}}{R_5} \quad i = \frac{V_{r_2}}{R_2} - X_2$$

Find each voltage as a function of \$V_{p_1}\$

$$\frac{V_{r_5}}{P_1} = \frac{V_{p_1}}{P_1} \quad V_{r_5} = \frac{R_5}{P_1} \cdot V_{p_1}$$

$$\frac{V_{r_2} - X_2}{R_2} = \frac{V_{p_1}}{P_1} \quad \frac{V_{p_2}}{R_2} = \frac{V_{p_1}}{P_1} + X_1 \quad V_{r_2} = \frac{R_2}{P_1} V_{p_1} + R_2 X_2$$

Plug in above

$$V_x = V_{p_1} + \frac{R_2}{P_1} V_{p_1} + R_2 X_2 + \frac{R_5}{P_1} V_{p_1}$$

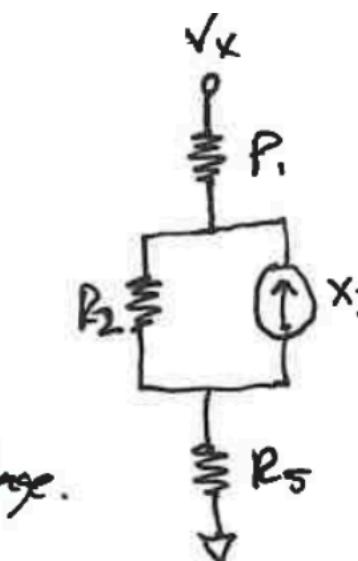
$$V_x - R_2 X_2 = V_{p_1} \left(1 + \frac{R_2}{P_1} + \frac{R_5}{P_1} \right)$$

$$V_{p_1} = \frac{V_x - R_2 X_2}{G_2} \quad \text{Therefore } i = \frac{V_{p_1}}{P_1} = \frac{V_x - R_2 X_2}{G_2 \cdot P_1}$$

use this for current on this path.

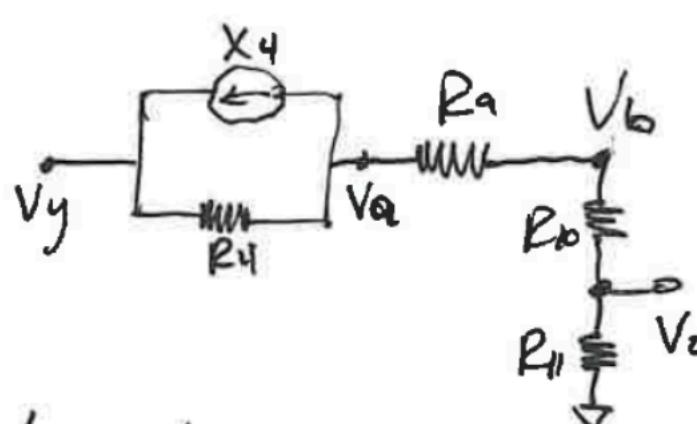
Repeat same process

$$i = \frac{V_x - R_3 X_3}{G_3 \cdot P_2} \quad G_3 = \left(1 + \frac{R_3}{P_2} + \frac{R_8}{P_2} \right)$$



Output Branches

$$V_y \leftarrow \text{II} \text{ am} \quad V_o$$



Current going through each component is equal.
Write everything as a function of V_o for difference equation.

$$\frac{V_b - V_o}{R_{10}} = \frac{V_o}{R_{11}} \quad \frac{V_b}{R_{10}} = V_o \left(\frac{1}{R_{10}} + \frac{1}{R_{11}} \right) \quad V_b = V_o \left(1 + \frac{R_{10}}{R_{11}} \right)$$

$$\frac{V_a - V_o}{R_9} = \frac{V_o}{R_{11}} \quad V_a - V_b = \frac{R_9}{R_{11}} V_o \quad V_a - \frac{R_9}{R_{11}} V_o = V_b \uparrow$$

$$V_a - \frac{R_9}{R_{11}} V_o = V_o \left(1 + \frac{R_{10}}{R_{11}} \right) \quad V_a = V_o \left(1 + \frac{R_{10}}{R_{11}} + \frac{R_9}{R_{11}} \right)$$

$$\frac{V_y - V_a}{R_4} - X_4 = \frac{V_o}{R_{11}} \quad \frac{V_y - V_a}{R_4} = \frac{V_o}{R_{11}} + X_4 \quad \uparrow$$

$$V_y - V_a = \frac{R_4}{R_{11}} V_o + R_4 X_4 \quad V_y - \frac{R_4}{R_{11}} V_o - R_4 X_4 = V_a$$

$$V_y - \frac{R_4}{R_{11}} V_o - R_4 X_4 = V_o \left(1 + \frac{R_{10}}{R_{11}} + \frac{R_9}{R_{11}} \right)$$

$$V_y = V_o \left(1 + \frac{R_{10}}{R_{11}} + \frac{R_9}{R_{11}} + \frac{R_4}{R_{11}} \right) + R_4 X_4$$

\uparrow Save for later.

Now we need to connect branches together.

(43) (44)

Node V_x "-" Terminal current flows $V_y \rightarrow V_x$

$$\frac{V_y - V_x}{R_7} = \frac{V_x}{G_3 P_2} - \frac{R_3}{G_3 P_2} X_3 \quad (\text{From previous page})$$

Isolate V_y to plug in

$$V_y - V_x = \frac{R_7}{G_3 P_2} V_x - \frac{R_7 R_3}{G_3 P_2} X_3$$

$$V_y = V_x + \frac{R_7}{G_3 P_2} V_x - \frac{R_7 R_3}{G_3 P_2} X_3$$

\nwarrow plug in V_y

$$V_o (G_o) + R_4 X_4 = V_x \left(1 + \frac{R_7}{G_3 P_2} \right) - \frac{R_3 R_7}{G_3 P_2} X_3$$

Isolate V_x

$$V_o G_o + R_4 X_4 + \frac{R_3 R_7}{G_3 P_2} X_3 = V_x \left(1 + \frac{R_7}{G_3 P_2} \right)$$

$\underline{\underline{G_x}}$

$$V_x = \frac{G_o}{G_x} \cdot V_o + \frac{R_3 R_7}{G_x G_3 P_2} + \frac{R_4}{G_x} X_4$$

\nwarrow save

now we need to connect input \rightarrow output
"+" terminal to "-"

Node V_x "+" terminal

$$\frac{V_i - V_x}{R_b} = \frac{V_x - X_1}{R_1} + \frac{V_x - R_2 X_2}{G_2 P_1} \quad (\text{from previous page})$$

Isolate V_x

$$\frac{V_i}{R_b} + X_1 + \frac{R_2 X_2}{G_2 P_1} = V_x \left(\frac{1}{R_1} + \frac{1}{G_2 P_1} + \frac{1}{R_b} \right)$$

$$V_x = \frac{V_i}{R_b G_2} + \frac{X_1}{G_2} + \frac{R_2 X_2}{G_2 G_z P_1} \quad \begin{array}{l} \text{set equal to} \\ \text{other } V_x \end{array}$$

$$\frac{V_i}{R_b G_2} + \frac{X_1}{G_2} + \frac{R_2 X_2}{G_2 G_z P_1} = V_o \frac{G_o}{G_x} + \frac{R_3 R_7}{G_x G_3 P_2} X_3 + \frac{R_4}{G_x} X_4$$

Solve for V_o for difference equation

$$\frac{V_i}{R_b G_2} + \frac{X_1}{G_2} + \frac{R_2}{G_2 G_z P_1} X_2 - \frac{R_3 R_7}{G_x G_3 P_2} X_3 - \frac{R_4}{G_x} X_4 = V_o \frac{G_o}{G_x}$$

$$V_o = \frac{G_x}{G_o R_b G_2} V_i + \frac{G_x}{G_o G_2} X_1 + \frac{G_x R_2}{G_o G_2 G_z P_1} X_2 - \frac{R_3 R_7}{G_o G_3 P_2} X_3 - \frac{R_4}{G_o} X_4$$

$$b_0 = \frac{G_x}{G_o R_b G_2} \quad b_1 = \frac{G_x}{G_o G_2}$$

$$b_2 = \frac{G_x R_2}{G_o G_2 G_z P_1} \quad b_3 = \frac{-R_3 R_7}{G_o G_3 P_2} \quad b_4 = \frac{-R_4}{G_o}$$

(45) (46) State update equations

$$X_1[n] = \frac{Z}{P_1} \cdot V_x - X_1[n-1]$$

↑ need to calculate.

$$X_2[n] = \frac{Z}{R_2} V_{R2} - X_2[n-1] \quad \text{more complicated.}$$

Let's find V_{R2} as function of V_x

$$\text{use } V_x = V_{P_1} + V_{R_2} + V_{R5}$$

Write all terms with V_{R2} , use currents equal

$$\frac{V_{P_1}}{P_1} = \frac{V_{R2}}{R_2} - X_2 \quad V_{P_1} = \frac{P_1 V_{R2}}{R_2} = P_1 X_2$$

$$\frac{V_{R5}}{R_5} = \frac{V_{R2}}{R_2} - X_2 \quad V_{R5} = \frac{R_5}{R_2} V_{R2} - X_2 R_5$$

now plug in

$$V_x = \frac{P_1 V_{R2}}{R_2} P_1 X_2 + V_{R2} + \frac{R_5}{R_2} V_{R2} - X_2 R_5$$

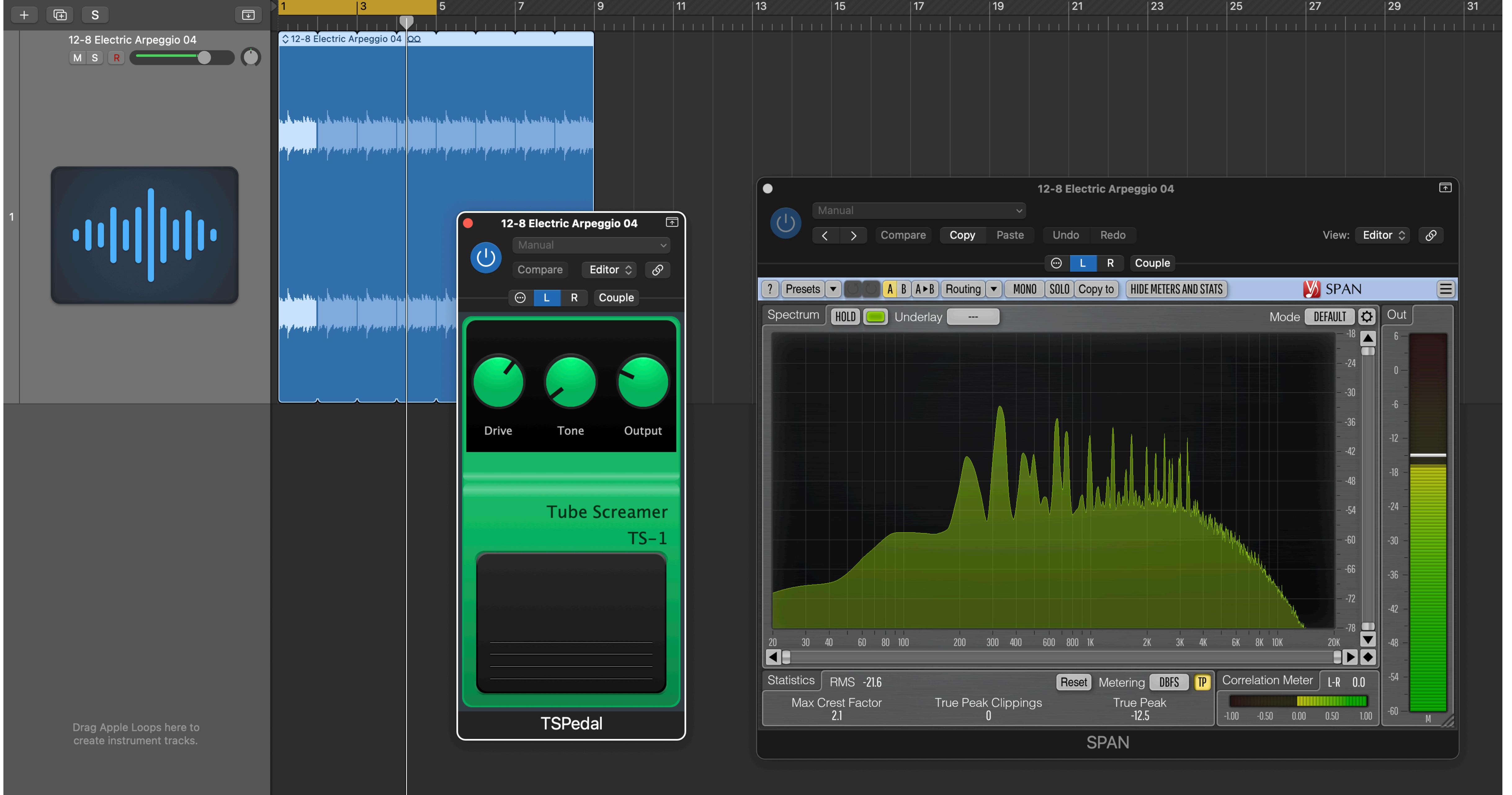
Solve for V_{R2}

$$V_x + P_1 X_2 + R_5 X_2 = V_{R2} \left(\frac{P_1}{R_2} + 1 + \frac{R_5}{R_2} \right)$$

$$V_{R2} = \frac{V_x}{G_R} + \left(\frac{P_1 + R_5}{G_R} \right) X_2$$

~~X₂~~ Plug into original state equation

$$X_2[n] = \frac{Z}{R_2} \left(\frac{V_x}{G_R} + \left(\frac{P_1 + R_5}{G_R} \right) X_2[n-1] \right) - X_2[n-1]$$



hack audio

Alternatives?

SOUL

The Future of Audio Coding



The SOUL project is creating a new language and infrastructure for writing and deploying audio code. It aims to unlock improvements in latency, performance, portability and ease-of-development that aren't possible with the current mainstream techniques that are being used.

[Try SOUL in your browser](#)

[Watch the Keynote](#)

<https://soul.dev/>

```

405 mesh_square(1) =
406     si.bus(4) <: par(i,4,*(-1)), (si.bus(4) > ((N)) >: si.bus(4)) :> si.bus(4);
407
408 // rectangular NxN square waveguide mesh:
409 mesh_square(N) = si.bus(4*N) : (route_inpt
410 ~ (prune_feedback(N/2)))

```

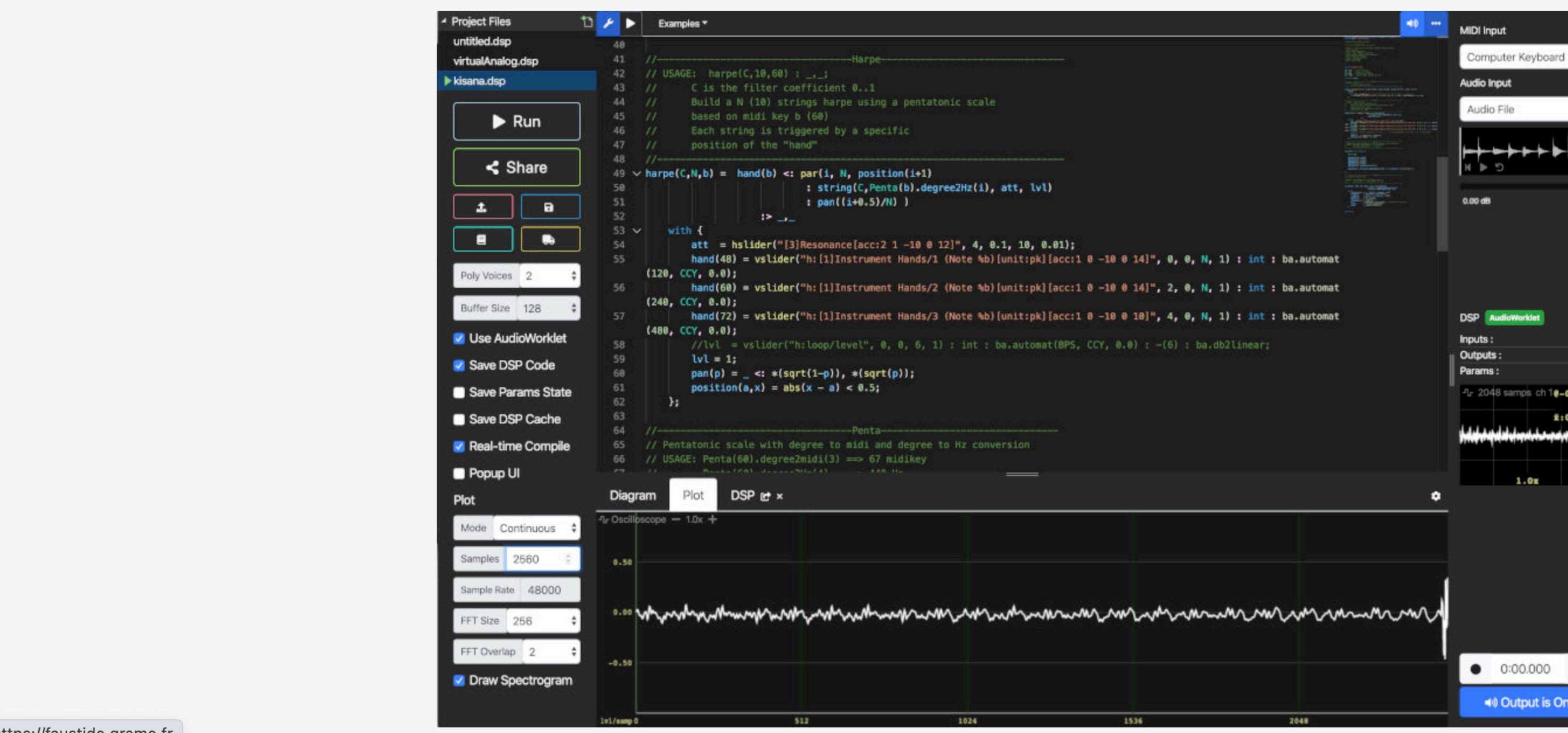
Faust
 Functional Programming Language for Real Time Signal Processing
[GitHub](#) [Quick Start](#) [Try It Online!](#)

What is Faust?

Faust (Functional Audio Stream) is a functional programming language for sound synthesis and audio processing with a strong focus on the design of synthesizers, musical instruments, audio effects, etc. Faust targets high-performance signal processing applications and audio plug-ins for a variety of platforms and standards.

The core component of Faust is its compiler. It allows to "translate" any Faust digital signal processing (DSP) specification to a wide range of non-domain specific languages such as C++, C, LLVM bit code, WebAssembly, Rust, etc. In this regard, Faust can be seen as an alternative to C++ but is much simpler and intuitive to learn.

Thanks to a wrapping system called "architectures," codes generated by Faust can be easily compiled into a wide variety of objects ranging from audio plug-ins to standalone applications or smartphone and web apps, etc.



<https://faustide.grame.fr>

It's time to **simplify** the way we write audio software

Elementary is a modern platform for writing high performance audio software that helps you build quickly and ship confidently.

[Get Started](#)[Try in your Browser](#)

FEATURES

A fundamentally different way to write audio software

Elementary brings the functional, reactive programming model to a world dominated by imperative, object oriented code. That means faster development, simpler code, and unimpaired creativity.



Declarative

Elementary makes it simple to create



Functional

Build highly reusable components that



Accessible

One language, one paradigm, no hidden

<https://www.elementary.audio/>

Web Audio API

https://developer.mozilla.org/en-US/docs/Web/API/Web_Audio_API



SOURCES

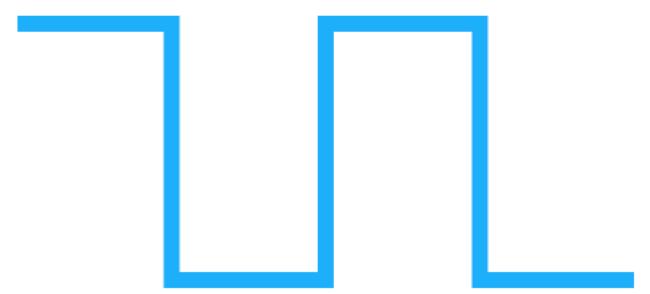
AMPLITUDE

FILTER

MODULATION



Square Oscillator



Amplitude 100.0%

Width 0.0%

LFO Amount 0.0%

Envelope Amount 0.0%

Saw Oscillator



Amplitude 0.0%

Detune Fine 0.00 ct

Detune Coarse 0.00 st

Pitch

Pitch Offset 0.00 st

LFO Amount 0.00 st

Envelope Amount 0.00 st

Glide 0.00 ms

Noise

Amplitude 0.0%





OSCILLATORS



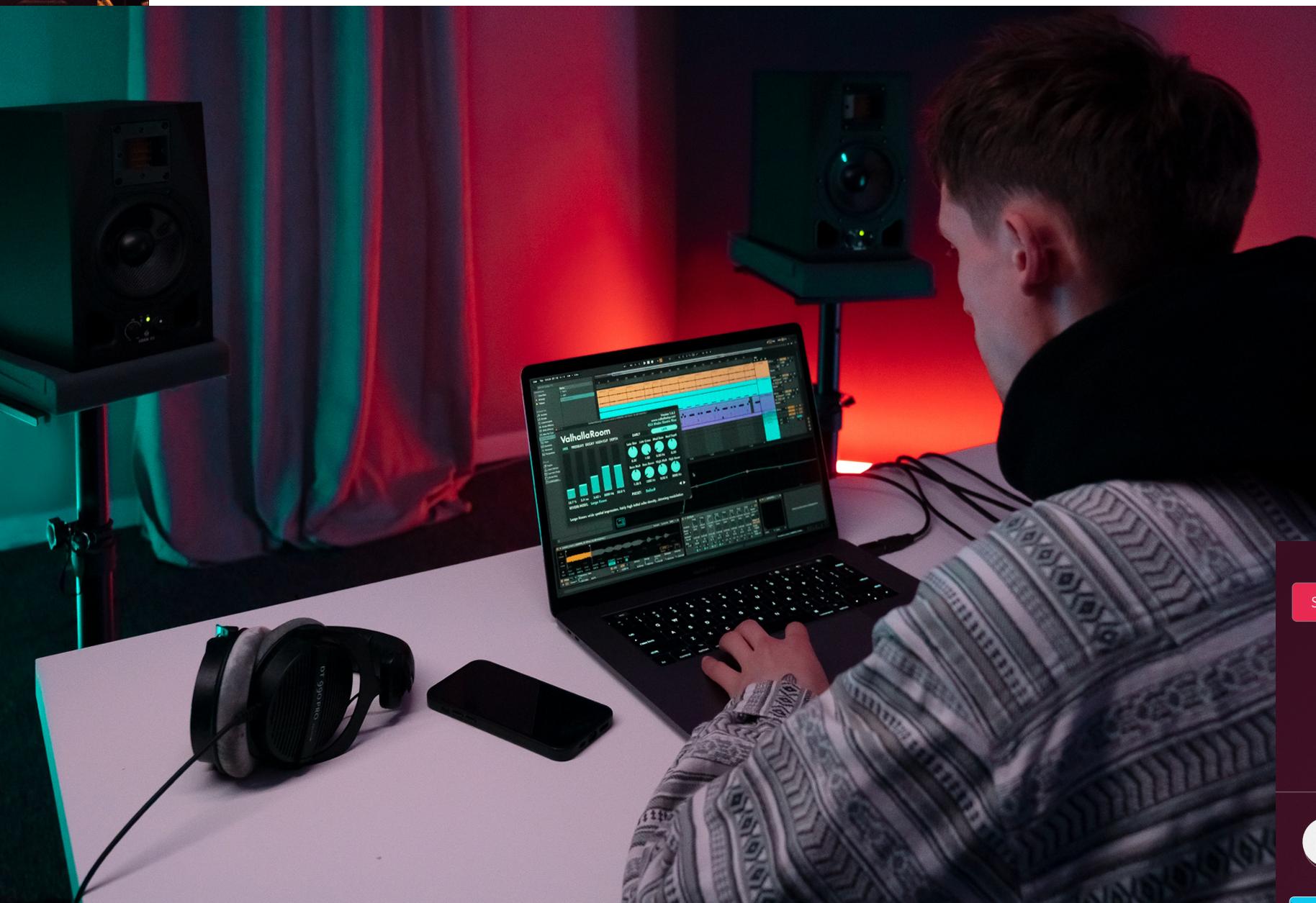
```
oscillator.type = 'square';  
oscillator.frequency.value = 1113;
```

<https://musiclab.chromeexperiments.com/>

Web Assembly



<https://virtualcz.io/>

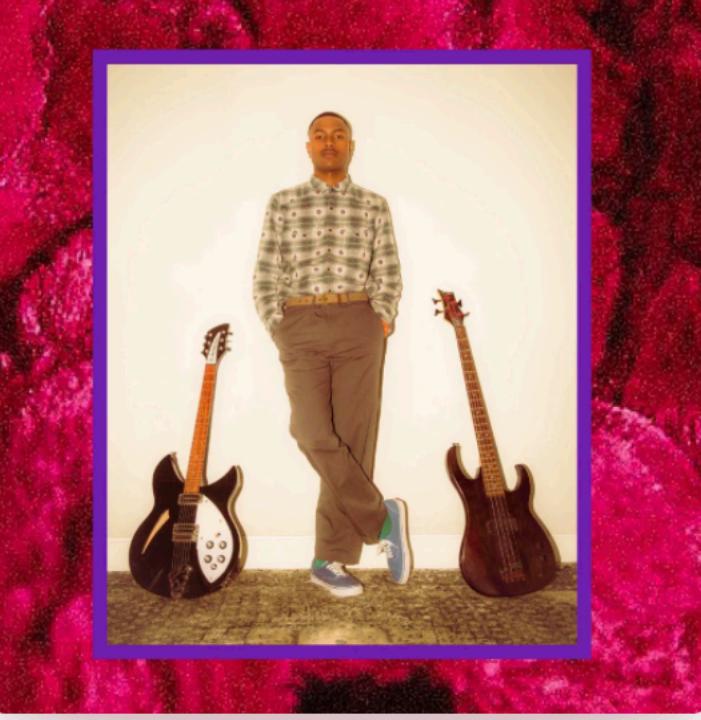


Mobile music producers

Henny Tha Buziness



Steve Lacy



Steve Lacy's Demo - EP E
Steve Lacy
R&B/SOUL · 2017

Preview ...

1	Looks	1:31	...
2	Ryd	2:20	...
★	3 Dark Red	2:53	...
4	Thangs	1:50	...
5	Haterlovin <small>E</small>	1:56	...
6	Some	3:00	...

<https://music.apple.com/us/album/steve-lacys-demo-ep/1207820151>

Resources



<https://www.youtube.com/c/TheAudioProgrammer>

<https://www.theaudioprogrammer.com/community/discord>

<https://audio.dev/>

<https://www.hackaudio.com/>

<https://www.programmingformusicians.com/>

Thank you