

# ZIP 1.0 Manual

by unfilteredaudio

The interface features a top bar with the ZIP logo, navigation icons, a 'Default' preset selector, settings and lock icons, and the 'unfilteredaudio' brand name. The main area is divided into a spectrum analyzer on the left and a grid of controls on the right. The spectrum analyzer shows a blue waveform with a white diagonal line, a y-axis from 0 dB to -70 dB, and a pink vertical bar on the right. The control grid includes: Filter (OFF, Audition), Threshold (0.0 dB, Auto), Ratio (1, Expand), Knee (0 dB), Mix (100%, Bypass), Lookahead (0 ms), Attack (6.00 ms), Release (180 ms), Color (0.0%), Gain (0 dB), Input (two blue dots), Amplitude (left-right arrow), Peak, Classic (left-right arrow), Contrast (left-right arrow), and AGC. At the bottom are 'MODULATION' and 'MUTE ALL' buttons.



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## *Introduction*

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Zip is Unfiltered Audio's creative take on dynamics processing. It goes beyond simple amplitude analysis, offering up methods for analyzing a signal's brightness, tonality, and more.

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## **Features**

- Analyze a signal's Amplitude, Quietness, Brightness, Darkness, Noisiness, or Tonalness.
- Choose between compression and expansion.
- A detailed visualizer shows all aspects of the algorithm.
- A multi-mode Color control adds detail and finish to a track.
- Multiple Attack/Release envelope shapes for quickly capturing a compression "feel".
- A wide-ranging Ratio knob can go from surgical precision to full-on squash.
- Multi-mode sideband filtering.
- Automatic gain compensation.
- Click-free soft bypass.

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## Getting Started

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### Installing Zip

Simply run the provided installer. On Windows, you may need to select your VST directory if the installer doesn't pick your preferred directory by default.

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### Browsing Presets

Zip comes with a wide array of presets to get you started. The preset manager at the top of Zip's interface is used to load, save, or browse presets. If you want to get a good feel for the power of Zip, spend some time with these presets before browsing the manual.

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## Standard Unfiltered Audio Features

### CONTROLS

- All knobs and vertical sliders are controlled in a smooth up-and-down motion.
- Use the Command key (on OS X) or Control key (on Windows or OS X) to fine-tune controls.
- Double-click or Alt-click on a control to return it to a default setting.

### PRESETS

- All presets are saved with a .uap file extension. These presets are compatible across all platforms and plug-in formats. They use a simple, non-encrypted XML format for easy editing.
- Use the "Lock" menu to prevent specific parameters from changing while browsing presets.

### INTERFACE

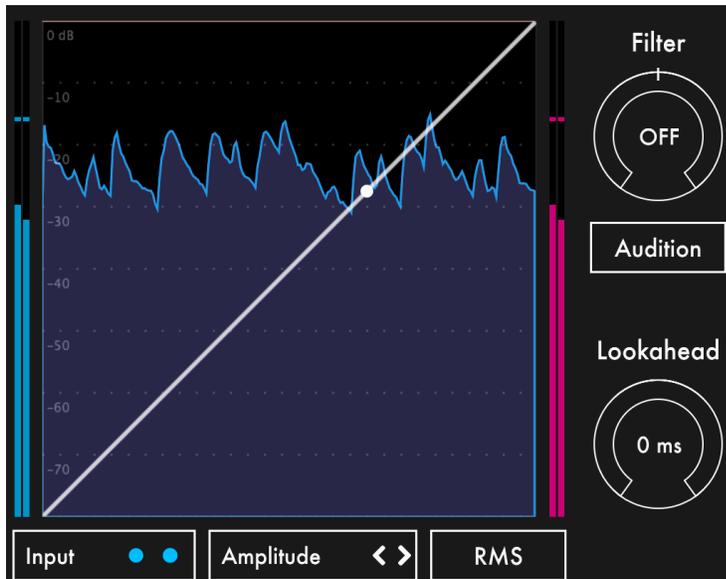
- Click the Gear icon on the preset manager to bring up interface options.
- Dark and Light skins provide a comfortable view for all studio situations.
- Resizable interface for easy viewing on all monitors.

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## Controls

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### Analysis



This section affects how the incoming signal is analyzed.

**Input Selector:** This menu will select whether the main input or external sidechain is used for analysis. The blue circles represent the amplitude of each channel of the signal.

**Analysis Mode:** Selects Zip's active analysis target. This will show up in blue on the analysis visualizer. Each mode permits creative mixing strategies. For instance, analyzing on Brightness will cause the amplitude to duck when the signal becomes brighter.

### AMPLITUDE MODES

**Amplitude:** This is the most traditional dynamics mode. The loudness of the signal is used here.

**Quietness:** Anti-amplitude. The quieter a signal is, the "louder" it will be to the analyzer. Silence will appear as full loudness.

**Brightness:** This measures the signal's frequency response. A signal with more energy in higher frequencies will appear "louder" to the analyzer.

**Darkness:** Anti-brightness. A signal with more energy in lower frequencies will appear "louder" to the analyzer. Silence will appear as full loudness.

**Noisiness:** This also measures the signal's frequency response. A signal with rapidly changing frequencies will appear "louder" to the analyzer.

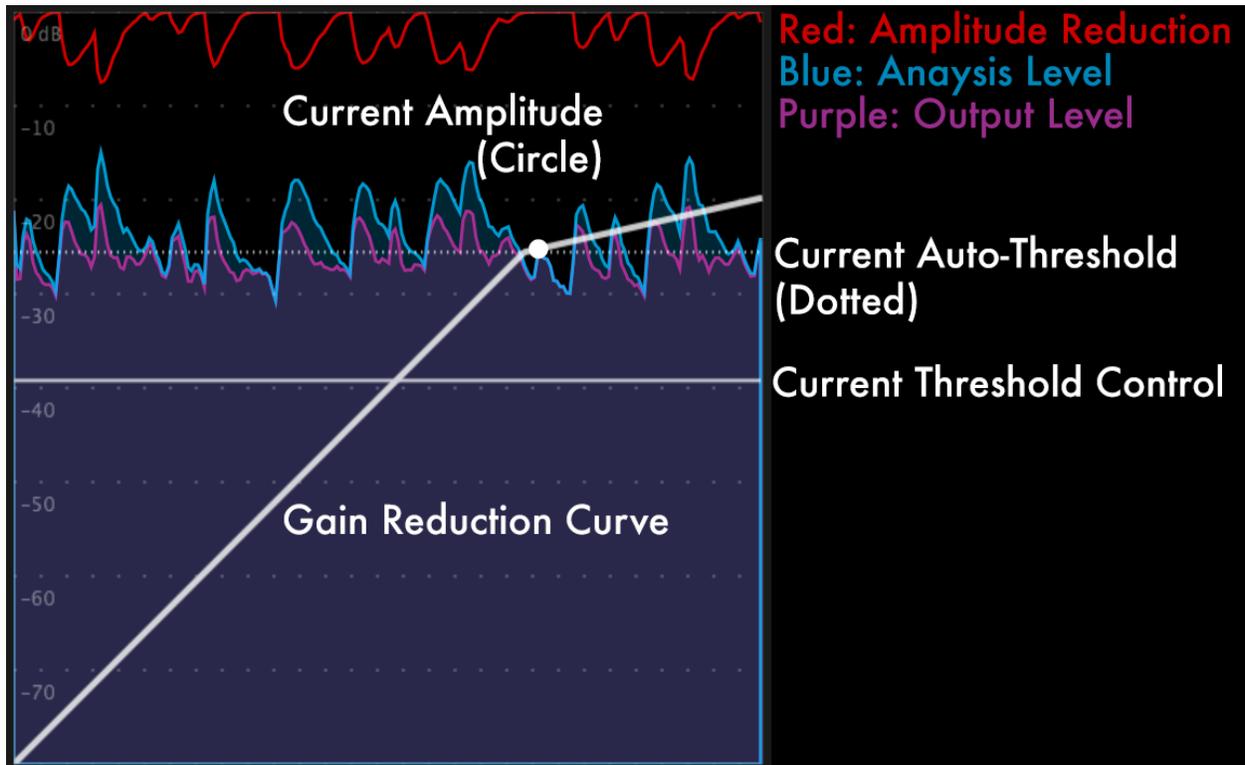
**Tonalness:** Anti-noisiness. A signal with stable frequencies will appear "louder" to the analyzer.

**RMS/Peak:** Changes the analysis mode between RMS (Root Mean Square) and Peak on all meters in the plugin. This will change how the compressor responds. Root Mean Square is a bit slower and smoother, while Peak is very rapid.

**Filter:** Controls the cutoff of the sidechain filter, allowing the analyzer to focus on a specific frequency range. Values below 12 o'clock apply low-pass filtering, while values over 12 o'clock apply high-pass filtering.

**Audition:** When active, this mode will send the filtered sidechain directly to the plugin's output, allowing you to hear exactly what is being analyzed. It is useful to listen to how the filter is working, especially if you're using a sidechain input.

**Lookahead:** This adds a delay to the output signal so that analysis will occur slightly ahead of where it is applied. You can adjust the lookahead value to improve transient response. Higher lookahead values improve the plug-in's reaction to sudden transients. However, this will add latency to the plug-in. Keep this at 0 ms if using Zip for real-time processing.



**Display Zoom:** This control is visible when your mouse cursor is over the display. Here, you can change what amplitude region is displayed in the visualizer. The full 80 dB range is the default.

## Dynamics

This section contains the dynamics processor. This stage can act as a compressor or as an expander.

**Threshold:** Sets the decibel level at which the dynamics processor will activate.

**Auto-Threshold:** When active, this mode tries to keep the threshold value in between the average amplitude over the last few seconds. Also when active, the threshold knob "nudges" the calculated value up or down.

**Ratio:** Sets how much gain reduction will be applied when the amplitude crosses the Threshold. Higher values mean more reduction. In Expand mode, this sets how much reduction occurs when the amplitude is below the Threshold.

**Expand:** When active, the dynamics processor becomes a downward expander. Downward expansion mode reverses the procession behavior, so amplitudes over the Threshold are unaffected but amplitudes below will be further reduced, somewhat like a noise gate.

**Knee:** This determines how sharp the transition is between an unaffected signal and gain reduction.

**Attack:** This determines how quickly the gain reduction curve will reach its target after the analysis signal exceeds the Threshold.

**Release:** This determines how quickly the gain reduction curve will go back to zero after the analysis signal drops below the Threshold.

**Envelope Style:** Each mode changes the shape and behavior of the Attack and Release curves.

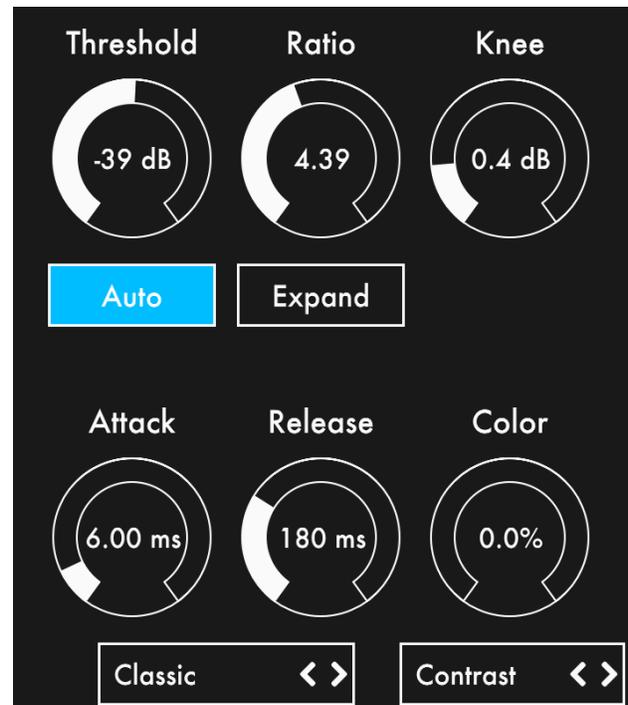
### ENVELOPE STYLES

**Classic:** Standard envelope behavior.

**Goopy:** Sluggish envelope behavior. Adds a vintage, muddy feel.

**Quick:** Faster envelope response. Useful for detailed percussion processing.

**Extreme:** Very fast envelope response. Useful for dealing with precise transients.



**Color:** Determines the intensity of the selected last-stage Color effect.

**Color Mode:** This menu selects the active last-stage Color effect.

## COLOR MODES

**Contrast:** Bright, phase-modulated distortion. At low values, this can bring a signal out in a mix. At high values, it can act as an aggressive distortion.

**Saturation:** Simple, warm soft saturation. Adds a pleasant grit to loud signals.

**Low-Pass:** A 2-pole low-pass filter.

**High-Pass:** A 2-pole high-pass filter.

**Auto LP:** A 2-pole low-pass filter that is modulated by the analysis curve.

**Auto HP:** A 2-pole high-pass filter that is modulated by the analysis curve.

**Bitcrush:** A digital degradation effect.

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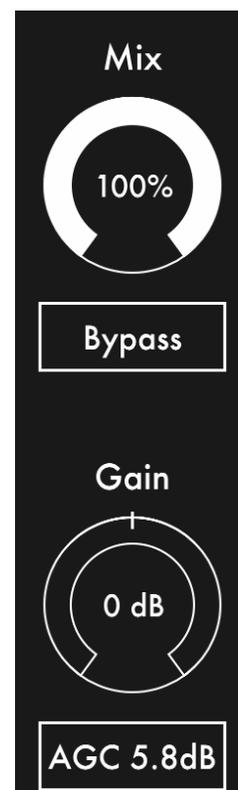
## Output

**Mix:** Controls the balance between the wet, processed output and the dry, unprocessed input. Reduce the Mix amount to create parallel compression.

**Bypass:** Quickly bypass Zip without changing latency or producing pops. This can be modulated rapidly for unusual effects.

**Gain:** Sets the level of the output signal. At 0.0 dB, no gain is applied. This can be used to add “makeup” gain in order to boost the final signal after compression is applied. Please be careful with this control if you do not have the output limiter turned on (in the options menu).

**AGC:** Automatic Gain Compensation. This applies a fixed amount of gain that is determined by Threshold, Ratio, and Analysis Mode settings. It is always positive.



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# Modulation System

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## Introduction

Zip utilizes Unfiltered Audio's flagship modulation system, with which you can quickly patch anything from simple parameter changes to complex and even self-evolving systems. Starting off is very easy due to the direct cable-routing user interface.

The first step is to reveal the modulation manager by clicking the "Modulation" button in the bottom corner of the interface. You'll see the plugin interface get taller, revealing modulation sources, outputs, and inputs on each modulatable plugin parameter.

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## Patching

### Input and Output Ports

Outputs and inputs are both denoted by a simple circle. To avoid confusion, just remember that outputs are always located on the top left of each modulator. Everything else is an input.

Output ports have a small circular scaling knob below them which can be used to alter its calculated output value. This knob is set to 1.0 by default (fully clockwise), but can be turned down to 0.0 (center) to zero out its values entirely, or -1.0 (fully counter clockwise) to invert them.

### Creating Connections

To create your first modulation "patch" simply click and drag on one of the outputs. You'll see a cable appear next to your mouse, the end of which you can drag to any input on the interface. When you get close enough you'll see the cable "lock" into place, showing you that the path is valid. Once you release the mouse, the connection completes and you'll see the modulation start to animate in real-time.

You can create a second connection from the same output source by clicking on it again. Likewise, try clicking and dragging from an input to an output, which works in the same fashion.

### Moving or Removing Connections

In order to move a connection, click on one of the thicker ends of the cable, near the input or output port. You'll see the cable lock onto your mouse, allowing you to move it to a different input/output source. If you release the mouse while the cable is not "locked" onto any port, it will be removed entirely, which is how connections are deleted.

## Inter-Modulation

As you might have noticed, all of the knobs on each modulator have input ports of their own, capable of receiving modulation just like parameters on the main section of the interface. Any output port can be connected to these inputs, meaning that a modulator can even modulate its own parameters! Experimenting with complex inter-modulation chaining can lead to dynamically evolving behavior, and even emergent systems.

## Automation

Our modulation system provides 16 automation “slots” that can be assigned to one or more modulation parameters. Every knob and button on a modulator can be assigned to one of these automation slots by right clicking on it and following the “Add X to Automation Slot...” menu. Once assigned to a slot, this element will mirror the automated value sent in through your DAW. You’ll also see a small number appear next to the element, denoting the automation slot number to which it is assigned.

## Adding Modulators

You can add a new modulator by clicking the “+” button in the modulation header. At this time a maximum of 6 modulators can be used at once, although that is a somewhat arbitrary maximum that we plan to increase in the future.

## Removing and Duplicating Modulators

Modulators can be removed or duplicated by right clicking on them and then selecting the appropriate option.

## Mute Modulation

The “Mute All” button in the modulation header allows you to quickly mute and un-mute all modulation sources.

## Saving and Loading

All modulation routing, values, automation, and general state is saved with the preset and/or with the DAW’s session. Presets are a great way to quickly share interesting modulation patches with other users.

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## Modulator Types

You can change a modulator’s type by clicking the dropdown menu next to its symbol. There are currently six types to choose from, each with its own characteristics and use case. Common parameters such as frequency will remain at their selected value, making the process of auditioning different modulator types both quick and easy.

## LFOs: Sine/Saw/Tri/Square

The LFO family of modulators all use common waveforms which are useful for classic “envelope” style parameter modulation.

The Amplitude knob is common to all of them, corresponding to modulation depth. This can be either Bipolar or Unipolar which is determined by the state of the +/- button in the modulator’s bottom left corner. Bipolar modulation fluctuates between negative and positive, whereas unipolar modulation is always positive when the amplitude is above 0% and always negative when it is below 0%.

The Frequency knob corresponds to modulation speed. It is set in Hz (cycles per second) when in non-tempo-synced mode. Tempo Sync can be activated by clicking the “Hz” button on the left side of the modulator, which will turn into a musical note symbol indicating that it has been initiated. In tempo sync mode, modulation speed is synchronized to divisions of the DAWs global tempo setting- for instance setting the frequency to  $\frac{1}{4}$  will correspond to quarter-note modulation cycles.

The Phase knob determines the value of the LFO at its starting point. The starting point occurs when the DAW’s transport (playback) resets. The LFO can be manually reset with the trigger button on its left side, or via the reset modulation input below that. An input signal transitioning above zero here will trigger the reset.

The Saw/Tri modulator has an additional Shape knob which determines the slope of the triangular wave. At the default value of 0.5, you get a normal triangle wave. At 0.0, you get a downward sloping triangle wave and at 1.0 you get an upward sloping triangle.

The Square modulator has a Width knob that changes the pulse width of the square wave. At the default value of 0.5, you get a normal half-up, half-down square wave. As you approach 0.0, the wave will be almost entirely down, and conversely as you approach 1.0, the wave will be almost entirely up.

## Input Follower

The input follower modulator allows you to easily “sidechain” parameters to the incoming audio stream. The Smooth knob determines the speed of the rms meter used to track the incoming audio. Lower values correspond to a more instantaneous reading, while higher values smooth everything out.

## Macro Control

The macro modulator can act as a manual control knob capable of outputting values to multiple parameters. This allows control over many parameters with a single gesture. Alternatively, you can patch many modulation sources into the amplitude knob’s input port, turning the macro control into a summing modulation bus.

## Sample and Hold Noise

The sample and hold noise modulator chooses random values at the chosen frequency rate. The Slew knob controls the amount of interpolation time whenever a new value is set. At a slew

of 0.0 values change instantly, whereas at 1.0 they interpolate over the entire cycle period. Tempo Sync and Uni/Bipolar can be set on this modulator just like LFOs (see above).

## Step Sequencer

The step sequencer is a modulator that cycles through up to 16 manually chosen values.

**T:** “Trigger” input. This includes both a manual button and a modulation input. If the button is clicked or the modulation input transitions to a positive signal, the sequencer’s active stage will increase by one.

**R:** Reset input. This includes both a manual button and a modulation input. If the button is clicked or the modulation input transitions to a positive signal, the sequencer’s active stage will reset immediately to the stage set by the START control.

**AMP:** Sets the maximum amplitude of the modulation outputs.

**FREQ:** Sets the rate at which the sequencer automatically increments. With tempo sync off, this can be set to 0 Hz, making the sequencer only increment based on the state of the T input.

**LENGTH:** Sets the number of active steps in the sequence.

**START:** Sets the first stage in the sequence. This stage is selected when playback restarts or the Reset input is triggered.

**SLEW:** Determines how quickly the output values change. At 0.0, the modulation output is stepped. As SLEW increases, new values are smoothly selected, adding a “slide” effect between values.

## ROLI Lightpad

This modulator connects to the excellent ROLI Lightpad (<https://roli.com/products/blocks>) for use as a performance controller. Please note that because this is part of the modulation system, this modulator is not intended for writing automation. If you want to use the Lightpad for automation, you can use the ROLI Blocks Dashboard for loading the interface of your choice and sending MIDI CC.

To get started, connect the Lightpad to your computer using either its USB-C cable or Bluetooth. For instructions on setting up a Bluetooth connection, see the Lightpad’s manual.

Once a Lightpad is connected to your computer, it should appear in the drop-down menu on this modulator. Select it and click “Connect”. The interface should turn white. You are now in Location + Pressure Mode. To change the active mode, click the large control button on the side of the Lightpad.

### LOCATION + PRESSURE MODE

In this mode, the modulator will track the position of one finger on the Lightpad along with its pressure. There are separate outputs for X, Y, Z (Pressure), and On (Touch Gate: on when a finger is touching the Lightpad, otherwise off).

### SLIDERS

In this mode, the modulator will output the value of four vertical sliders. Due to the resolution of the Lightpad’s screen, it may appear that there are only 15 possible values for the slider. However, the information that is read is higher resolution.

## **TOGGLES**

In this mode, the modulator will output the value of four toggle switches.

## **QUAD PRESSURE**

In this mode, the modulator will output four separate pressure values (one for each quadrant of the Lightpad).

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## Options

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The following unique options are available in Zip:

**Enable Display:** This toggles the analysis display on the left side of Zip's interface. If you have multiple open interfaces and want to reduce the workload on your graphics hardware, it can be useful to turn this off.

**Average Stereo Analysis:** This determines how Zip handles stereo amplitude analysis. When enabled, a stereo signal is averaged to mono. When disabled, the loudest sample from each channel will be used.

**Limit Output:** When enabled, a soft limiter will be used to keep the output within a safe range.

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## *Recipes and Ideas*

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### **Parallel Compression**

- Set the Mix control to a value below 100% Wet.
  - Turn up the Color control to provide a very different type of compression sound.
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### **Brightness Filter**

- Set the Analysis Mode to Brightness.
  - Set the Color Mode to Auto LP.
  - Change the depth of the Color control. The filter's cutoff will be related to how much high-frequency content is present.
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## *Credits and Thanks*

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Zip is written by Joshua Dickinson and Michael Hetrick.

We'd like to thank:

Benn Cooper

Andres Cabrera for teaching us the ins-and-outs of dynamics processing.

Olivier Gillet of Mutable Instruments for his many contributions to DSP education.

Lance Putnam for his flexible DSP library, Gamma.

Plugin Alliance for helping us do this full-time.

Our families, for putting up with the constant stream of weird sounds coming out of our offices (and random stress over esoteric bugs).

<http://www.unfilteredaudio.com>