Allrounder

User Manual - Version 1.0

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1.1 Device Overview

Allrounder is an audio plugin that has four different algorithms for 'rounding off' the data of your signal. The plugin uses four different techniques to augment the sample resolution of the audio signal. The order of all four has been optimised for the best possible outcomes. A saturator and various filters types, that can placed pre or post the algorithms, are accessible in the plugin to help bring out more of the four algorithms.

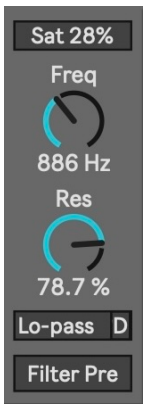
1.2 System Requirements

- Ableton Live 11 or 12.
- Max for Live (included with Live Suite).

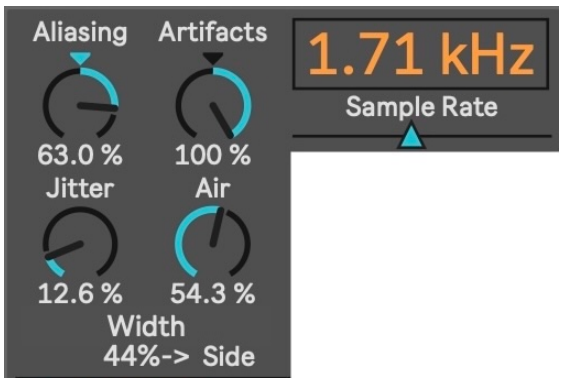
1.2 Installation

- Double-click the installation file or drag it directly onto an open instance of Ableton Live.
- A dialogue box will appear with a notice and ask if you'd like to continue.
- Click Yes
- The device will now be installed under: Packs > Allrounder
- Drag the device onto a Audio track or a MIDI track that has an synth or other sound generating device in Live to begin using it.

2.1 Device Structure



- A Saturator and Variable Filters, which can be placed pre or post the resolution algorithms

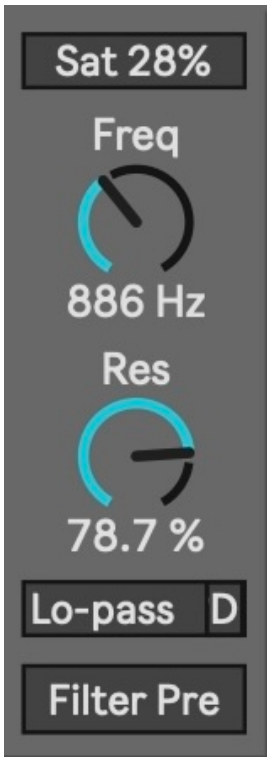


- Algorithm 1 : Sample rate resolution

- Algorithm 2/3 : Bitwise AND / XOR

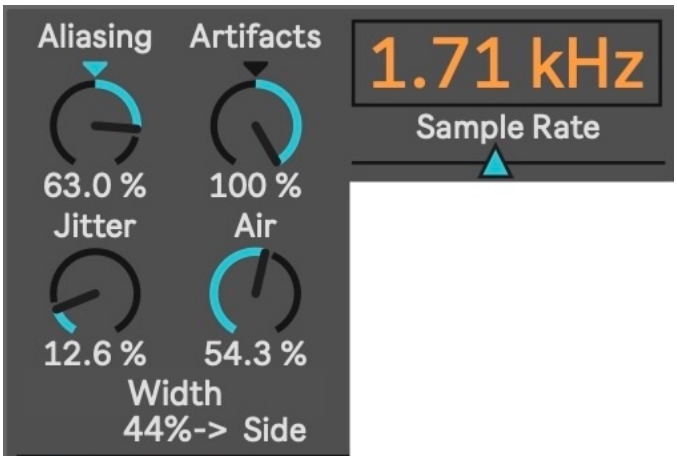
- Algorithm 4 : Custom rounding algorithm and Output control

3.1 Saturator and Variable Filter



- The saturator starts the signal chain and affects everything, sonically, in the plugin. Changing this setting will also affect the Round algorithm (explained later in this manual) if engaged.
- Freq - Controls the frequency of the filter.
- Res - Controls the resonance of the selected filter.
- D - C Button. Dirty or Clean are the two filter types that can be accessed. In the Clean (C) configuration the filters are typical biquad filters: low-pass, notch, hi-pass. All with a smooth filter response. In the Dirty (D) configuration the filters have extreme resonance characteristics and emphasise or add a high shelf to the high frequency content in particular setting. Be warned that at low frequencies and high resonances that feedback can occur. There is a clipper and limiter that has been place on the plugin to ensure the track will not blow up and you can get distorted sounds from the filter if desired.
- Filter Pre / Post - The filter can be set Pre (after the saturator but before the algorithms) or Post (after everything before the output)

4.1 Algorithm I - Bit Crusher



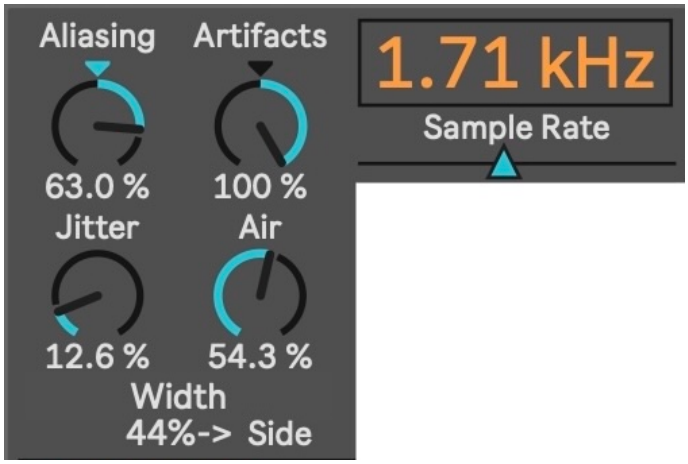
This is a high quality and very flexible bit crusher. Using a few techniques to bring out and control the sound of bit crushing with the following controls. Please note that a lot of the following parameters are influenced by the cutoff frequency that you set.

- kHz (Sample rate) - This controls the sample rate or cutoff frequency of the bit crusher. Depending on the sample rate of your DAW, the plugin will correspond to it. So if your DAW is set to 48 kHz the plugin will max out at 48, not 44.1. You can use the triangle slider or click on the number box to transition the sample rate from its the maximum resolution to 1000 times less, 44.1kHz to 44.1Hz or 98kHz to 98Hz. Since the slider does not have a fixed frequency range, it works with 100 incremental steps between the minimum and maximum sample rate. Please remember when inputting data into this that it is a number between 0 and 100 and not the actual sample rate. Typing the number 50 into this at the sample rate of 44100 will set the slider to 2.00 kHz so the slider is logarithmic and not linear.

4.1 Algorithm I - Bit Crusher (continued)

Aliasing & Artifacts - These are 2 filters that create the tonal colour of the bit crusher. They run serial, one after the other, Aliasing first and Artifacts second.

- Aliasing - This filters out frequencies above the set cutoff point when set at 0%.This changes when the cutoff point is changed. When set to 100% no filtering is applied to the signal and all aliasing is created. Set at 0% the filter will match the cutoff frequency and filter out aliasing frequencies above the cutoff. Between -100% and 0% the filter can be set from 100 Hz to the cutoff point. This filter frequency will change when the cutoff frequency changes so 50% with a cutoff frequency of 2.5 kHz will be 1.25 kHz.
- Artifacts - This filters out frequencies above the set cutoff point when set at 0%. This changes when the cutoff point is changed. When set to 100% no filtering is applied to the signal. Set at 50% the filter will be set at half the 'distance' from the cutoff point and nyquist.
- Jitter - This controls the intensity of random deviations of the bit crusher filter frequency. This leads to smearing or lessening of the bit crusher effect.
- Air - A variable frequency high shelf filter based on the cutoff frequency. 0% is half the cutoff and 100% is 1.5 the cutoff frequency.
- Width - Is a MS controller which focus more on the mono or stereo elements of the signal. Clicking on the Width text resets to zero which is Stereo. Going left or right moves to mono or the side information.

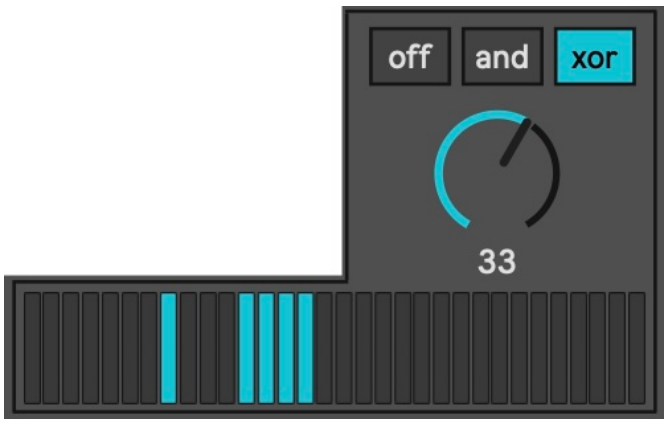


5.1 Algorithm II - Bitwise AND /OXR

These two algorithms are bitwise AND / OR operations that perform a bitwise intersection on the incoming float-point signal as raw 32-bit data. The algorithm is engaged when both the audio and the plugin recognise the same bit state as an AND or XOR. A simple understanding of AND and XOR operations is that AND recognises the same while XOR recognises the differences. AND creates a one when two signals are both non zero. XOR is when two signals are analysed and if one of the two signals is not zero, it will output a one. But not when both signals are non zero. So this is used in the float conversion process on particular parts of the float point signal (seen visually). This is after the saturator but before algo I and II.

- Bit - This controls the bit resolution of the AND bitwise operation.
- Depth - Controls the finer variation of the bit setting.an only work when the drive effect is being engaged.

- XOR – This dial is a custom setting of bitwise operation resolution. The dial at 0 is off and then operates between bits 15 and 6 (1-55).



6.1 Algorithm III - Round

The round algorithm is a simple rounding of the bits. It is placed first in the effects signal and colours all the other processes. When both gain dials are set to -6 dB, the plugin will have the same gain staging as the pre plugin input.

- Round - Is a custom bit depth rounder. This is before algo I but after algo II. If the saturator is engaged the round algorithm changes based on where the saturation is set. This is done to try and keep some consistency in the rounding algorithm when saturation is present.
- Dry Gain - Controls the gain of the unprocessed signal.
- Bit Gain - Controls the gain of the effected signal.
- Ø - Flips the phase of the affected signal. This is useful when blending the Dry and Bit gains. Flipping the phase can cancel out different frequencies and create other colour.