Dent 2.0 Manual

by unfiltered**audio**

Introduction

Dent is a waveform workstation, a distortion that blends analog and digital traditions. It mixes many analog-style distortion techniques (soft clipping, wavefolding, rectification, bias) with digital destruction (hard clipping, bitcrushing, waveset processing). When combined with our cutting-edge modular modulation system, an unbelievable palette of tonal processing opens up.

Features

- Multimode bitcrusher: 6 unique modes, including CRISPR, Bent, XOR, AND, and standard bitcrushing. All five modes are continuous and non-stepped, giving you the widest possible range of sounds.
- Multimode clipper: 4 clipping flavors, from digital (hard clipping) to analog (soft clipping) and futuristic (wavefolding).
- Probabilistic Waveset processing: Turn Dent's distortion features on and off at zero-crossings to create pulsating grit, interference patterns, and cable dropouts.
- Variable rectification: Continuously morph between full- and half-wave rectification, along with all of the timbres in between!
- Resonant Multimode Filter: This filter packs some serious tone, from liquid cutoffs to screaming resonance.
- Automatic Gain Compensation.
- 2x/4x Oversampling.

Getting Started

Installing Dent

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

Browsing Presets

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

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.

Controls

Top Row

Dent's Top Row contains controls that use analog-style techniques for distorting a signal.

In/Out Gain: Controls the amount of gain applied to the incoming and outgoing audio signals. Both gains only affect the Wet signal. Dent has separate input and output gain controls due to the number of Feedback controls present. In Gain, for example, may be useful for saturating the signal before it is filtered. Out gain, meanwhile, may be useful for amplifying quiet details or adding second-stage distortion to the filtered signal.

Bias: This control adds DC bias to the entire waveform, ignoring whether the waveform is currently positive or negative.

Shape: This control affects the amplitude curve of the signal. At 12 o'clock, the signal is unaffected. The shaper acts by effectively rounding the audio signal toward specific amplitude values. When the Shape control is turned clockwise, it will morph the signal to a square-like waveform. Counter-clockwise will produce needle-like waveforms.

Split: This control takes the positive and negative components of the signal and separates them from each other. In technical terms, this adds a symmetrical amount of DC bias to both components.

Upper/Lower X: These controls modify the amplitude and polarity of the positive (upper) or negative (lower) components of the signal.
At 12 o'clock, these controls mute the associated component of the signal. This means that these controls can be used to rectify a signal.

Trim: This control affects the amplitude of the final output. This occurs *after* the Out Gain control and last-stage clipper. This control will not add any gain. If you add a lot of gain to saturate your signal, this control will help bring the final level down for mix compliance.

AGC: Automatic Gain Compensation. When enabled, a smoothing algorithm will match the amplitude of Indent's wet output to its dry input.

Clip Mode: Chooses how the signal is treated when its amplitude exceeds a threshold.
1. <u>Hard Clip</u>: If the signal's amplitude exceeds +/- 1.0, it is simply clipped to that boundary. This is an aggressive, digital-sounding form of distortion.
2. <u>Soft Clip</u>: The signal is saturated using a parabolic equation. This will actually affect the sound of the entire signal in a subtle way. This clipping is more pleasant and analog-sounding.
3. <u>Wavefold</u>: Signals that exceed the threshold are "folded" around the threshold. There is more sophisticated shaping at play here, but essentially this can change the perceived frequency of the sound. A great way to hear this is to use a simple sine wave as an input, then increase the gain on the input and outputs.

4. Overfold: A combination of wavefolding and overdrive. Try this mode out on bass frequency sawtooth waves!
5. Harsh Fold: A much more brutal, digital-sounding implementation of wavefolding.
6. Distortion: A more traditional distortion algorithm. Excellent for guitars.
7. Tube: A partially rectified saturator that greatly attenuates negative components of signals.
8. Tape: A modified soft clipper with a parabolic equation that more closely matches the saturation behavior of tape.
9. Gloubi: "Gloubi-boulga" waveshaper.
10. Cubic: The signal is multiplied by itself three times.

There are three clip mode selectors. The first clipper appears after the In Gain process. The second occurs after the middle distortion controls (before the bitcrusher). The final clipper appears after the Out Gain process (before the Trim and AGC).

Complex: When this mode is enabled, a different clipping mode can be selected for each section. With Complex mode disabled, the first clipping mode is applied to all clippers (the same as Dent 1.0 behavior).

Bottom Row

The bottom section of Dent features three more effects that can be used to drastically alter your signal.

BITCRUSHER

A Bitcrusher is an effect that manipulates the resolution of a signal. It manipulates the signal in two dimensions: temporal resolution and amplitude resolution. The "sampling rate" of a signal is the temporal resolution, or the speed at which a signal is acquired. In Dent, this means how often Dent checks for a new input and generates a new output. The "bit depth" of a signal is the amplitude resolution, or how much precision the computer uses to represent the value of a signal.

SR: Reduces the sampling rate of Dent's DSP. Note that this doesn't simply reduce the sampling rate of the input. Changing this knob will affect the quality of the Filter section in a dramatic way. The only part that is unaffected is the DC Blocking filter, as that needs to run at the full sampling rate to do its job correctly!

Bits: Reduces the resolution of the signal. The method that it uses to do this will depend on the selected mode.

Bitcrush Mode: Chooses the algorithm under control of the Bits knob.
1. Bitcrush: Standard bitcrushing algorithm.
2. Bitwise AND: Turns the signal into a 16-bit integer and uses an AND comparison against another 16-bit integer set by the Bits knob. Extremely aggressive. This mode is recommended for synthesizers.
3. Bitwise XOR: Similar to the AND mode, but uses XOR logic. Try this on a sine wave.
4. Bent: Simulates bit destruction in a circuit bent toy. This algorithm was created by Olivier Gillet of Mutable Instruments.
5. CRISPR: A milder, micro-bitcrush. Gets deep into your signal and makes it crispy.

6.   <u>Dust</u>: An unusual masking-based crush that thins out your signal.

FILTER

Dent contains a highly-resonant multimode filter. This filter can be used to soften artifacts produced by the distortion and bitcrushing processes, or to add screaming resonance to the signal path.

Cutoff: Sets the cutoff frequency of the multimode filter.

Rez: Sets the resonance of the multimode filter.

Mode: Sets the active filter mode. LP (Low-Pass) removes higher frequency components, while HP (High-Pass) removes lower frequency components. BP (Band-Pass) only permits a band of frequencies near the cutoff, while BR (Band-Reject) removes a narrow band of frequencies near the cutoff.

Post-Dist: When active, the filter will appear after the top-row distortion and bitcrusher algorithms.

WAVESET PROCESSOR (PROBABILITY)

Waveset Distortion is a concept invented by a composer named Trevor Wishart. A "waveset" is defined as three zero-crossings of a signal. In practice, a waveset is usually one cycle of a waveform (For example: one cycle of a sine wave crosses 0.0 three times).
There are many ways to distort wavesets, but most of them are offline processes (For a great list of methods, check out *Microsound* by Curtis Roads, or look at Composer's Desktop Project).

Dent implements a form of Waveset Distortion in real-time. Dent's Waveset Processor exists in the "PROBABILITY" section. This section determines the probability that a waveset will pass through Dent's top-row distortion algorithm.

With "Seq" mode off, there is only one Probability control. This control shows a percent chance that each waveset will be processed. This occurs randomly.

With "Seq" mode on, there are two "Probability" controls. This is a bit of a misnomer, as the process becomes deterministic. Now, X out of Y wavesets will be processed, where X is the first knob and Y is the second. Furthermore, all X wavesets are processed in a row. As an example, if the first knob is 8 and the second knob is 12, the first 8 wavesets will be distorted, while the remaining 4 are left clean.

"Drop" mode determines what happens to unprocessed wavesets. With "Drop" mode off, the waveset ignores the top-row distortion controls. With "Drop" mode on, the waveset is replaced with silence.

Finally, one very critical thing to realize about Waveset Distortion is that it is heavily frequency dependent. Wavesets will occur more rapidly at higher frequencies.

MIXER

Mix: Controls the balance between the incoming dry signal and the outgoing wet (processed) signal.

DC Block: When active, components below 20 Hz will be filtered out of your signal. This is the last part of the signal chain, and is useful for filtering out DC bias generated by the top-row distortion controls.

Modulation System

Introduction

Dent 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 show 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.

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.

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 primitive wave types which are useful for classic "envelope" style parameter modulation.

The Amplitude knob is common to all of them, corresponding to the depth of modulation. 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 always 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 the speed of modulation. 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 ¼ will correspond to quarter-note modulation cycles.

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 which 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. In this way you can control many knobs 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).

Options

Dent 2.0 introduces an expanded options menu. The following unique options are available in Dent:

Smooth Wavesets: Dent 1.0 used interpolation to remove DC offset in Probability mode. This provides click-free behavior at the expense of waveset accuracy. Turn this mode off to hear the true sound of waveset processing.

Oversampling 0x/2x/4x: Oversampling runs Dent's DSP algorithm multiple times to remove foldover aliasing from occurring as a result of the many clipping algorithms. Higher levels of oversampling provide improved anti-aliasing at the cost of much higher CPU usage.

Recipes and Ideas

Rectification

- Half-Wave: Set the "Lower X" control to 12 o'clock (0.0).
- Full-Wave: Set the "Lower X" control to full counter-clockwise (-1.0).
- For true rectification, turn off DC filtering.

Auto-Wah

- Add an Input Follower modulator to the modulation section.
- Turn down the cutoff frequency of the LP filter.
- Attach the Input Follower to the cutoff control. Increase modulation depth and filter resonance to taste.

Credits

Dent is written by Joshua Dickinson and Michael Hetrick.

http://www.unfilteredaudio.com