Dirty Tricks Reference Manual

INDEX

Introduction

The “Dirty Tricks” suite is a bundle of 12 Max for Live devices: use them to distort, modulate, decimate, degrade and eventually destroy the sound at your will!
There are more than 200 presets (both individual, for each device, and audio racks) which will help you to start experimenting with the bundle: have fun!

Brickwall

A band-pass/band-reject brickwall filter; the slope can go up to 192 dB of attenuation per octave (in other words it can be extremely selective)

C. Freq: the center frequency of the filter.

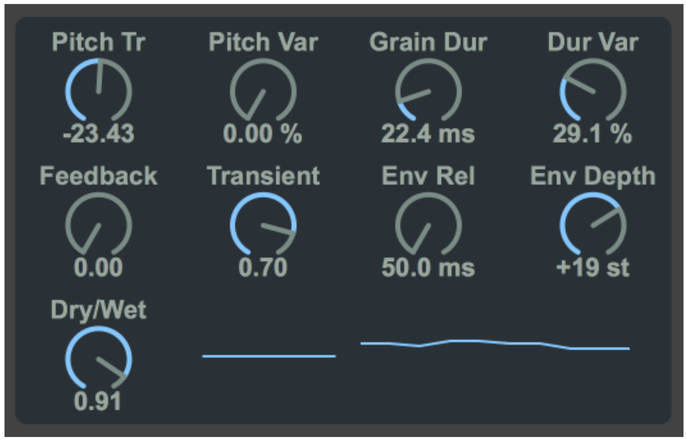
Width: filter bandwidth, in normalized values from 0 to 1.

Bell/Hole switch: switches from band-pass to band-reject.

Slope: filter attenuation per octave, from 12 to 192 dB.

ReGain: the signal energy can be severely reduced by the filter, especially with high slope values; the ReGain parameter can be used to compensate the energy loss.

Growler



A real-time pitch shifter with feedback, transient detector and envelope follower: ideal to create “monster” voices and other scary effects.

Pitch Tr: transposition in semitones.

Pitch Var: the signal is divided in grains (see Grain Dur below) and this parameter sets a percentage of pitch variation for each grain.

Grain Dur: duration of the sound grains.

Dur Var: variation of the duration of the sound grains.

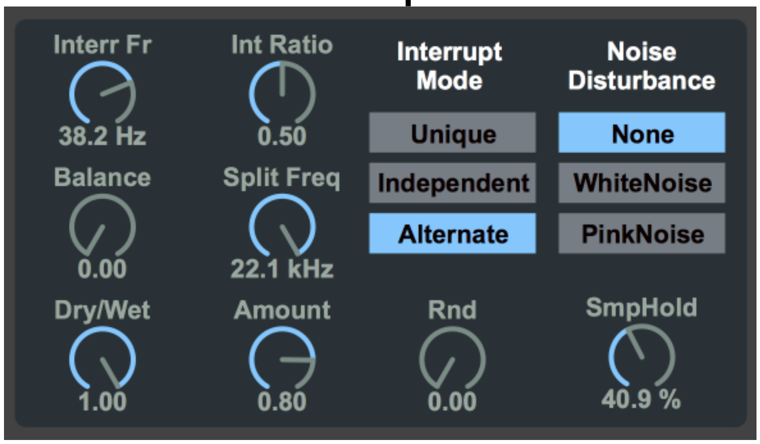
Feedback: amount of output signal sent back into the effect for further transposition.

Transient: detects transients. 0 means “no detection”. With positive values the transients are sent unprocessed to the output (they are not pitch shifted), with negative values the transients are suppressed.

Env Rel: release time of the envelope follower. The envelope follower is used to alter the pitch.

Env Depth: amount of pitch alteration generated by the envelope follower.

Interruptor



As the name implies, this device creates interruptions in the audio stream. It can also statistically sample and hold the signal.

Interr Fr: this parameter sets how many interruption you have per second.

Int Ratio: this is the ratio between the interruption and the sound. If set to 0.5 the sound has the same duration of the interruption, if set to 0.1 the sound has 1/10 of the duration and the interruption 9/10 and so on.

Balance: when this parameter is 0, only the signal below the split frequency is interrupted, when it is 1, only the signal above the split frequency is interrupted. Other values cause different mixes of interruption above and below the split frequency.

Split Freq: you can interrupt the signal above or below the split frequency (see the Spec Balance above).

Unique/Independent/Alternate switch: Unique = both channels are interrupted at the same time; Independent = the two channels are interrupted independently; Alternate = the cannels are interrupted alternatively.

None/WhiteNoise/PinkNoise switch: None = during the interruption the sound is silenced; WhiteNoise = the sound is replaced with white noise; PinkNoise = the sound is replaced with pink noise.

Amount: this is the depth of the interruption. If it is 1 the sound is completely interrupted (silenced). If it is 0.5 the sound amplitude is halved during the interruption and so on.

Rnd: sets the randomness of the interruptions. When it is 0% they are regular, when 100% they are completely random.

SmpHold: sets the percentage of samples that are held (not updated). When it’s high this parameter completely destroys the sound.

Klamper

Clamps the signal with a few special features.

Normalize: the klamped signal can be normalized, i.e. raised to the level specified by the Norm Level parameter to the right.

Norm Level: when the Normalize switch is on, this is the level for the klamped signal.

Balance: when this parameter is 0, only the signal below the split frequency is klamped, when it is 1, only the signal above the split frequency is klamped. Other values cause different mixes of klamping above and below the split frequency.

Split Freq: you can klamp the signal above or below the split frequency (see the Spec Balance above).

Klamp Lev: when the signal amplitude is above this level, it is “klamped”; i.e. if the Klamp Level is 0.5, all signal values above 0.5, or below -0.5 are transformed according to the Mode parameter (see below).

Mode: you can choose among 6 modes of “klamping” for the signal exceeding the Klamp Level.

Klamp: the signal is clipped.
Fold: the signal is folded back into the limits.
Wrap: the signal is wrapped in the opposite direction.
Annihil: the signal is annihilated (reduced to 0).
Sine: the signal is replaced with a sinusoidal oscillation.
Square: the signal is replaced with a square wave oscillation.

Osc Freq: frequency parameter for the “Sine” and “Square” modes above.

Mod

With Mod you can modulate any parameter of any device (native or Max for Live), any plug-in and any track in the Live environment. You can use an LFO (choosing among different waveforms) or a step sequencer to control the parameter.

Click the button on the upper left of the device (labelled Map), then click the parameter you want to control: the name of the button changes from Map to the name of the controlled parameter. To release the parameter (i.e. to stop controlling it) you can click on the red “R” button.

LFO Mode Buttons (Free/Sync) : in Free Mode the LFO rate is expressed in Hertz, in Sync Mode it is expressed in Note Values (1/2, 1/4, 1/8 and so on).

LFO Waveform Menu
Use this menu to choose the LFO waveform. The items are:
no (LFO is not active)
sine waveform
square waveform
triangle waveform
upsaw (upward ramp) waveform
downsaw (downward ramp) waveform
samp&hold waveform (random steps)
random (interpolated) waveform
step: selects the Step Sequencer

Freq (or Rate): with this parameter, you can set the LFO frequency in Hz (when it is in Free Mode) or in Note Values (when it is in Sync Mode)

Width: this parameter sets the amplitude of the LFO modulation.

Control: sets the center of the LFO oscillation in percentage

Seq Mode Buttons (Cycle/Step): this switch sets how the Step Sequencer uses the Freq/Rate parameter:
- In Cycle Mode the Freq/Rate parameter represents the time to generate all the steps
- In Step Mode the Freq/Rate parameter represents the time to generate a single step

Direction Menu: this menu is only available in Step Mode (see above) and sets the direction of the Step Sequencer.
The items are:
up, the steps are executed cyclically from the first to the last one
down, the steps are executed cyclically from the last to the first one
updown, the steps are executed alternatively forward and backward
random, the steps are executed randomly
urn, the steps are executed randomly without repetition, i.e. a step is executed again only after all the steps have been executed

Step Edit Arrows: the Left and Right arrows shift the steps to the left and to the right respectively
The Up and Down arrows increase or decrease the steps values.

Step Edit Buttons:
Random: fills the Step Sequencer with random values
Scramble: scrambles the steps without altering the values
Reset: zeroes all the step values

Nsteps: sets the number of steps for the Step Sequencer

Ovrdrv

Ovrdrv is a distortion device: it goes from very subtle to total destruction. It can operate on the signal below (Saturation) or above (Excitation) a given frequency.

Distort/Skream switch: there are two modes of distortion; “Distort” is warmer (sort of!), “Skream” is heavier

Depth: the amount of saturation/excitation. Caution! It can be very loudly

Sat/Exc: When this parameter is 0, only the signal below the split frequency is distorted (see below), when it is 1, only the signal above the split frequency is distorted. Other values cause different mixes of distortion above and below the split frequency.

Split Freq: this is the parameter used to separate the two distortion bands (Saturation and Excitation).

Philtre

The Philtre device has two configurations:
Dub: dual low-pass high-pass state variable filter with resonance and overlap factor.
Morph: a triple low-pass/band-pass/high-pass state variable filter with resonance and continuous morphing between the filter types.

You can switch between the two configurations using the switch button on the center of the device

Dub Parameters:
Filter: this parameter sets the cut off frequency and the filter type. From -100 to 0 it is a low pass filter with cutoff frequency gradually increasing from 0 Hz to 15000 Hz, from 0 to 100 it becomes an high pass filter with cutoff frequency gradually increasing from 0 Hz to 15000 Hz.

Overlap: the overlap factor influences the Filter behavior. When it is 0 the Filter parameter has the effect described above. Increasing the overlap factor makes the two filter types (low pass and high pass) to gradually overlap (i.e. the high pass effect begins before the low pass section has completed its path), and when the overlap is at its maximum, the filter behaves like a band pass (because the low-pass and the high-pass sections have nearly the same cut off frequency)

Morph Parameters:
Frequency: from 10 to 11.000 Hz.

Type: the filter type can be morphed gradually from low-pass (0) to band-pass (0.5) to high-pass (1).

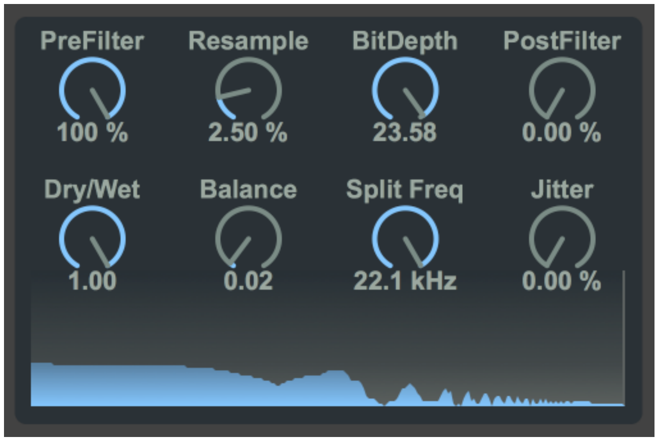
Other Parameters:
Reson: this is the amount of resonance the filter has at its cut off frequency.

Slope: this parameter gradually changes the filter order from 12 dB to 24 dB.

EnvDepth: this parameter sets the amount of variation the internal envelope follower has on the filter cutoff frequency.

EnvSpeed: this parameter sets the speed of the envelope follower, i.e. how fast it follows the input signal amplitude.

Reduktor



Decimates and reduces the bit depth of your signal. It is dual band, i.e. the reduction can be applied only to a portion of the signal.

PreFilter: filters the input signal with a brickwall low-pass to try to avoid foldover, when it is 0% the signal is not filtered, when it is 100% it is filtered at the nyquist frequency of the resample factor.

Resample: this parameter is a percent of the actual sampling rate: i.e. if the SR is 44100 Hz, a Resample of 50% would give a 22050 Hz sampling rate, 10% would give 4410 Hz and so on.

Bit Depth: changes the bit depth of the signal.

PostFilter: filters the output signal with a brickwall low-pass to try to avoid aliasing, when it is 0% the signal is not filtered, when it is 100% it is filtered at the nyquist frequency of the resample factor.

Spec Balance: when this parameter is 0, only the signal below the split frequency is decimated, when it is 1, only the signal above the split frequency is decimated. Other values cause different mixes of decimation above and below the split frequency.

Split Freq: you can decimate the signal above or below the split frequency (see the Spec Balance below).

Jitter: sample rate reduction instability factor.

Ringer

A dual band ring modulator with distortion and delay with feedback.

RingPitch: Frequency (in MIDI note values) of the modulator.

Shape: the shape of the modulator; 0 = sine, 1 = saturated sine.

Feedbk: the amount of modulated signal sent back into the modulator.

Balance: When this parameter is 0, only the signal below the split frequency is modulated (see below), when it is 1, only the signal above the split frequency is modulated. Other values cause different mixes of modulation above and below the split frequency.

Split Freq: this is the parameter used to separate the two modulation bands.

Free/Sync switch: switches between delay calculation in milliseconds (Free) and note values (Sync)

Delay/Rate dial: delay time

Shifter

A dual band frequency shifter with feedback delay

Freq1: frequency shifting for the lower band (see below)

+/- button: frequency shifting sign (positive/negative)

Freq2: frequency shifting for the upper band (see below)

+/- button: frequency shifting sign (positive/negative)

Split Freq: this is the parameter used to separate the two bands (lower and upper)

Feedbk: the amount of modulated signal sent back into the shifter.

Stereo Flip button: when “on” the left and right channel shift is of the opposite sign (i.e. +20 Hz on the left and -20 Hz on the right)

Free/Sync switch: switches between delay calculation in milliseconds (Free) and note values (Sync)

Delay/Rate dial: delay time

Swarmer

A “superchorus”, excellent to create super fat synth sounds (and more). It is composed of a chain of modulated delay lines.

Tank: this is the feedback parameter, sets the amplitude of the signal sent back to the circuit.

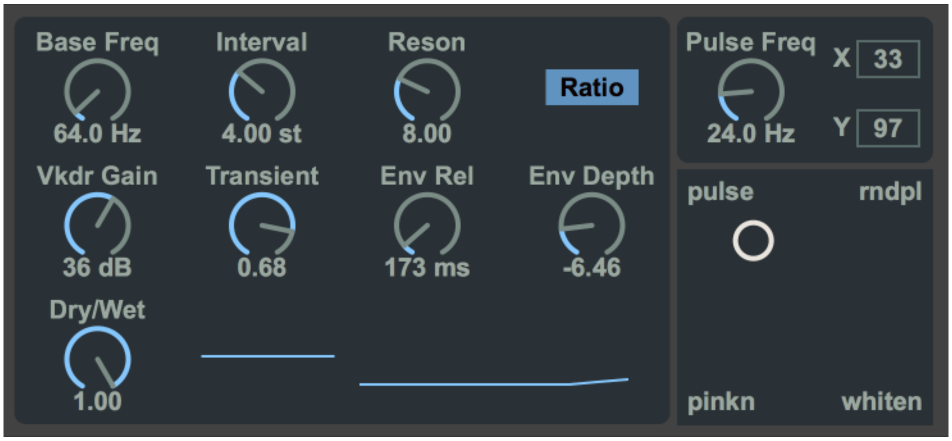
Tank Filter: a filter for the feedback signal.

LoPass/HiPass switch: changes the Tank Filter mode

Activity: how fast the delay lines are modulated.

Depth: sets the amount of delay in the delay lines.

Vkdr



This is a vocoder effect. Not to be confused with the "phase vocoder", the vocoder is a filter effect which analyses a signal and recreate it using a broadband source (for instance a white noise) and a bank of filters.

Ratio/Harm switch: (on the upper right of the first panel) when in "Ratio" mode the filter bank filters are equally spaced in semitones, when in "Harm" mode they are equally spaced in hertz

Base Freq: the frequency of the lowest filter of the filter bank

Interval: (appears in Ratio mode, see above) the distance in semitones between adjacent filters in the filter bank

Hstretch: (appears in Harm mode, see above) when this parameter is 1 the interval between adjacent filters is equal to the lowest filter frequency (i.e. they are arranged in a harmonic series), values below or above 1 shrink or stretch the filter spacing

Reson: the resonance (Q) factor of the filters

Vkdr Gain: gain factor for the processed sound

Transient: detects transients. 0 means “no detection”. With positive values the transients are sent unprocessed to the output, with negative values the transients are suppressed.

Env Rel: release time of the envelope follower. The envelope follower is used to alter the Pulse Frequency (see below).

Env Depth: amount of pitch alteration generated by the envelope follower.

Pulse Freq: the frequency of the pulse generator and of the random noise generator (see below)

X and **Y:** the coordinates of the X/Y slider (see below)

X/Y slider: (lower right panel)
The signal used to recreate the analysed sound can be a pulse generator, a random noise generator, a white noise or a pink noise. The four signals can be mixed with the 2D X/Y slider.

Credits

The Philtre device contains a modified version of the saturating state variable filter by Yofiel:
http://www.yofiel.com/software/cycling-74-patches/svf-saturating-filter
Many thanks to Ernest Leonardo Meyer!