u-he

Uhbik

user guide

Uhbik 2.0 is safe if used as directed

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Table of Contents

Introduction

Installation

Go to the Uhbik webpage, grab the appropriate installer for your computer system, double-click on the downloaded file and follow instructions.

Online Resources

For u-he news, downloads and support, go to the u-he website
For a lively discussion about u-he products, go to the u-he forum at KVR
For friendship and informal news, go to the u-he facebook page
For video tutorials and much more, go to our youtube channel
For 3rd party presets, go to our patch library

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Shared Features

The elements described in this chapter are common to all ten Uhbik plugins.

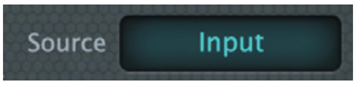
GUI Elements

Knobs

Values are adjusted via click-and-drag. For finer steps, hold down a SHIFT key. Note that some of the knobs are bipolar—zero is in the center so you can set negative values. Double-click to set the default. For MIDI remote control see the Configuration chapter.

Mouse wheel control: You don't always have to click and hold to adjust values: Try hovering over a knob or selector and rolling the mouse wheel. For extra-fine adjustment, hold down a SHIFT key.

Selectors



Left-click to select from a list of options, or hover over the selector and roll your mouse wheel.

Note: The grey modulation target selectors at the bottom of the GUI behave differently: right-click opens a list of options, while left-click is used for drag and drop assignment.

Parameter Locking

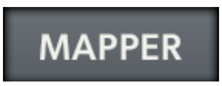
To prevent a knob or selector from changing its value when you switch presets, use the parameter lock function: right-click on the knob/selector and select *Lock* from the context menu. A small padlock icon will appear over the knob or selector.

Control Bar

MIDI activity indicator: The small rectangle in the left side panel flashes whenever the plug-in receives a MIDI message.

ON/OFF button: Disable this to bypass the effect. A rapid cross-fade ensures click-free switching.

Regular view where most Uhbik parameters appear.



Expanded view of the mapper offering more comfortable editing than in MAIN.

The preset browser.

KEYC (Key Control): Experimental feature: Activate this button, click on the desired control or navigate to it using cursor keys. Enter a value then confirm with Return or Enter. For negative values, type a minus before confirming. Increment / decrement values via plus (+) and minus (-) keys. Tip: +/- also works for on/off switches.

Hold Shift for fine steps, option (Mac) / ctrl (Win) for steps of 10. Backspace resets to the default value and zeroes selected Mapper points.

Data Display

In the center of the control bar is a display area which performs several functions:

7.1 surround

This area usually displays the name of the current preset. While editing a parameter, it shows that parameter's name and value. Hovering with your mouse over a control reveals its current value.

As a quick alternative to using the powerful preset browser you can load presets directly from here: Click the preset name in the center to open a list of all presets in the current folder, then select one. Use the arrows (❮ or ❯) to step through presets.

You can also drag & drop a preset file from an external folder onto the data display—that preset will be loaded but not saved.

Surround Processing

The Uhbiks can handle up to 8 channels surround, including e.g. quadraphony, 5.1 and 7.1.

Along the bottom of the data display are 8 pairs of small "VU meters" for the input levels (upper) and output levels (lower) of each active channel. The idea was to make Uhbik work in surround without cluttering up the GUI or requiring multiple versions of each plug-in.

Right-click on the **data display** for the following options...

Auto/Surround .... This default mode tests the number of output channels, then automatically switches to 1.0, 2.0, 3.0, 3.1, 5.0, 5.1, 7.0 or 7.1. The assumed channel order is: left, right, center, LFE, left surround, right surround, left rear, right rear. This order may deviate from your host's meters, but you can always check channel activity in Uhbik's own little VU-meters below the data display.

Multichannel................. All channels are spread evenly: the first channel is the furthest left, the last channel is the furthest right. For example if 4 channels are connected, the assumed channel order is left rear, left, right, right rear.

Stereo 1+2 etc............. Only the selected pair is processed, while all others are passed on "dry". Use these modes if you want to process a specific pair of channels only.

Mono................................ Only processes the first channel (typically 'left') and sends the result to all active channels. Unlike *mono-surround* mode (see below), all outputs are processed equally.

Stereo-Surround..........The input signal of the first two channels is copied (as a pair) to all other available channels. With the 'all channels' option active, channels 1 and 2 are copied to channels 3+4, 5+6, and 7+8 (depending on availability). With the 'dry C + lfe' option (see below), channels 3 and 4 are skipped i.e. the unprocessed input signal is routed directly to outputs 3 and 4. With the 'dry lfe' option this only applies to channel 4.

Mono-Surround...........Like stereo-surround, but with mono input. Unlike *mono* mode, the output channels are processed differently.

The next two options let you exclude certain channels (LFE and Center) from being processed:

Bypass LFE ....The LFE channel will not be processed

Bypass Center.............The center channel will not be processed

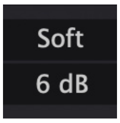
When applied to multichannel tracks, the channel **Offset** parameter in the Flanger, Phaser, Grainshift and Tremolo works across all active channels. For instance you can create dramatic effects that rotate around your head.

Note: the Delay's Pan controls are not just stereophonic—they will pan across all active channels!

To the right of the data display we have...

UNDO / **REDO**: The pair of curved arrows let you step backwards or forwards through your edit 'history' (maximum 10 steps). Note: UNDO even works if you happen to select a new preset by mistake before saving your edits!

Save: Opens a dialog box where you can specify a preset name and information text before clicking on 'Apply'. Saved presets will appear either in the User folder or in the currently selected folder, depending on the global preference Save Presets To.



Clipper: Algorithm (Bypass, Hard, Medium or Soft) and Threshold (-6, 0, 6 or 12 dB) level for **protective** output 'soft clipping'. The default is Medium, 6 dB.

Out: Adjusts the final gain so you can balance the processed and bypassed levels, for instance. The range is +/- 12dB. The shield image in the center flashes whenever the clipper is triggered.

u-he Badge: Opens a menu containing links to this document, to our website, to our user support forum and to our social network pages. Below the badge you can see the plug-in revision number.

Configuration: Click the cogwheel button in the top-right corner to access three pages for setting up MIDI remote control or adjusting global preferences such as UI size and brightness. For details, see the Configuration chapter.

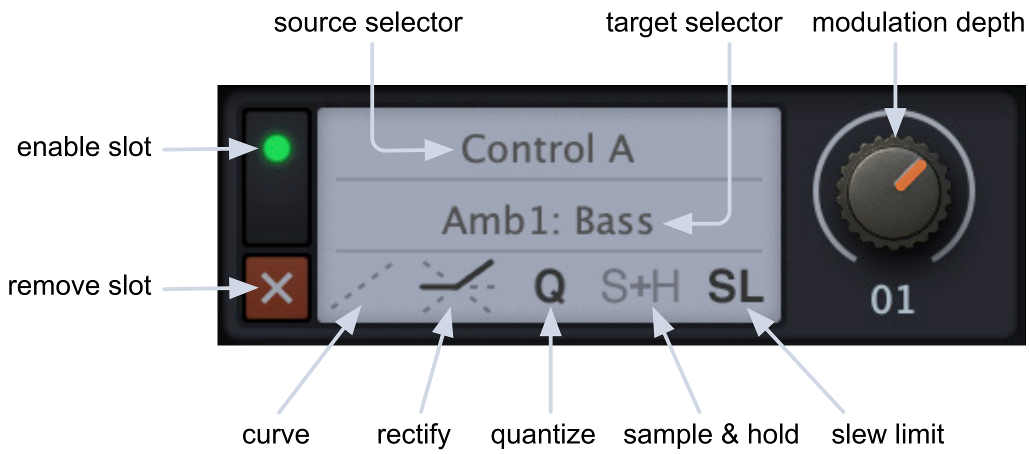
GUI Size

Right-click anywhere in the background (e.g. in the 'wooden' side panels) to select a size between 50% and 300% (options larger than your monitor are disabled).

This setting is temporary—for a more permanent GUI size option see the Preferences.

Modulation Matrix

The 8-slot modulation matrix is used for creating motion by connecting **sources** to **targets**…



Assignments can be made very comfortably via drag & drop. Try this: In any Uhbik, click on the data display and select *init*. In the LFO panel, click on the **pip** (source picker) to the left of the Triangle icon and drop it on the Out knob in the control bar. In the matrix, the source in slot 01 is set to *LFO Tri* and its assigned target is *Output DB*.

Source Bar

The bar above the matrix contains source pickers for MIDI as well as Mod Noise. See Drag & Drop.

Source Menu

Left-click on the source selector to open a menu with the following options:

MIDI**	Mod Wheel, Pitch Wheel	MIDI keyboard lefthand performance controls
	Control A, B, C, D	MIDI CC general purpose controls (see Preferences)
	Gate, Key Fol, Velocity	MIDI Note data (Key Fol = key follow)
	Aftertouch	MIDI channel pressure or poly pressure
Internal	Mod Noise	Modulation noise: Add ‘flutter’ to any modulation target!
	LFO Tri, Sqr, Rnd, Sine	The four LFO waveforms: Triangle, Square, Random, Sine
	Envelope	The Envelope signal
	Env. Rising / Env. Moving	Gate signals derived from the envelope – view them in the scope!
	Mod Mapper	The Mapper signal.

** For how to route MIDI into effect plug-ins, please refer to the documentation of your host app.

Target Menu

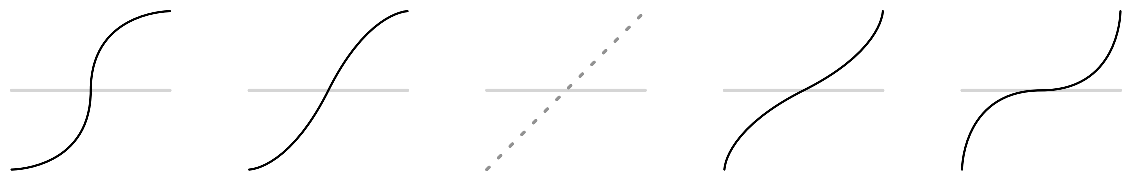
Right-click (!) on the Target Selector to open a menu of all targets. An alternative to the drag & drop method available via via left-click.

Slot Modifiers

Below the target selector is a row of buttons for modifying the shape of the modulation signal...

Curve

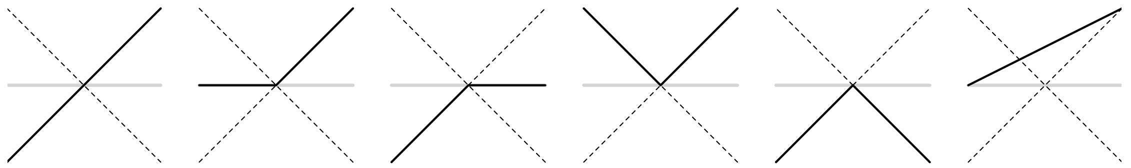
These options let you 'map' the source onto an s-curve. A bipolar ramp, for instance from a rising sawtooth LFO or from the pitch bend control, would be transformed into one of these curves:



Note that positive unipolar modulation sources will only use the upper half of the curve.

Rectify

Half-wave or full-wave rectification in positive and negative versions, as well as *unipolarize*. The symbols show how a bipolar ramp wave (like *none* here) would appear after rectification.



Half Wave + / -..........................ignores negative values
Half Wave - ....ignores positive values
Full Wave + ...turns negative values positive
Full Wave -..................................turns positive values negative
Unipolarize.................................shifts the entire signal to positive-only

Quantize (Q)

After applying a curve, the modulation signal can be forced to adopt discrete values. The 'steps of' options make the modulation more or less steppy, with *steps of 12* transforming bipolar sources into maximum 5 values (unipolar = maximum 3) including zero.

Caveat: As the quantize options have been standardized across most u-he plugins, you will find some of them more useful than others. Try them in Uhbik Grainshift first!

Integer / Steps Of nn ....multiple integers, from 1 to 12
Overtone Series ....harmonic overtones
Minor / Major Scale...............natural minor, regular major
Minor / Major Chord..............chord tones (root, third fifth)
Minor / Major Series ....third intervals (repeats after two octaves)
Fifths And Octaves ....7 and 12 semitone intervals only

Note: Quantization is applied **after** modulation depth, so a lower depth simply means fewer steps. For instance, if you quantize an LFO with the *steps of 12* option, there will be no modulation at all unless the Depth is set to at least 25.

Sample & Hold (SH)

Whenever the signal selected here crosses zero in the positive direction, the modulation source will be sampled and held i.e. stepped.

Slew Limit (SL)

Lets you soften transients in the modulation source. To develop a feel for this feature, modulate something drastically from a square or random LFO, then try out the various SL options.

Envelope (ENV)

Similar to the *function generators* seen in modern modular synthesizers, Uhbik's envelope is a very flexible modulation source. The selected Mode affects the appearance of the panel...

Mode

EF ....The **Envelope Follower** mode smoothes the selected Source, with Attack and Decay independently controlling the slope of the leading and trailing flanks.

AD ....In the **Attack / Decay** mode, the Attack starts whenever the Source signal exceeds the Threshold, and is followed immediately by the Decay. The envelope can be retriggered at any time (see OS below).

AR ....The **Attack / Release** mode is like AD except that the envelope stays at maximum until the source drops below the Threshold, which starts the **Release** stage.

CYC..............In **Cycle** mode the envelope loops automatically as long as the Threshold is exceeded.

OS ....The **One Shot** mode is the same as AD (see above) except that it will only retrigger after the Decay has finished.

Source

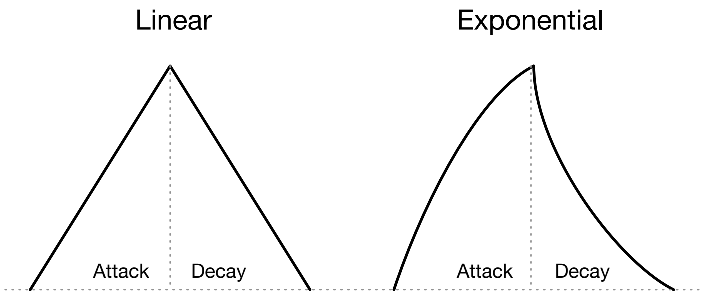
The envelope can be triggered by various audio / modulation signals:

Audio	Input, external sidechain, output
Modulation	MIDI Gate, LFO waves, mod mapper, modulation wheel or MIDI controls A / B.

To make use of the *Ext. Sidechain* option you will need to route **extra audio** channels into Uhbik. For details please refer to the documentation of your host application.

LIN / EXP

Sets the basic shape of the two envelope stages to linear (LIN) or exponential (EXP).



Gain (EF only)

A level control for the envelope follower (see Mode above), with a range of -24 to +24 dB.

Threshold

The level at which the input signal will trigger the envelope. The 'traffic light' indicator between the Threshold and Attack knobs show retriggering.

Tip: To monitor the resulting envelope, activate the SCOPE instead of the MAPPER, drag & drop the pip next to the ENV label into the scope or onto one of the four fields along the bottom of that area.

Attack

The slope of the leading flank. Effectively the time it takes to rise from zero to maximum.

Decay / Release

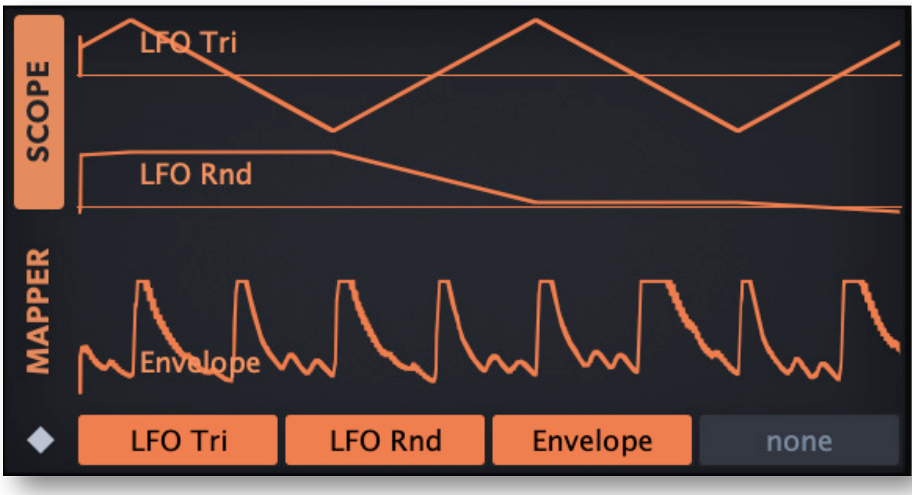
The slope of the trailing flank. Effectively adjusts the time it takes for the envelope to fall from maximum to zero. Labelled 'Release' in **AR** mode (see above).

Scope

The area below the envelope can display the audio output or up to four modulation waveforms.

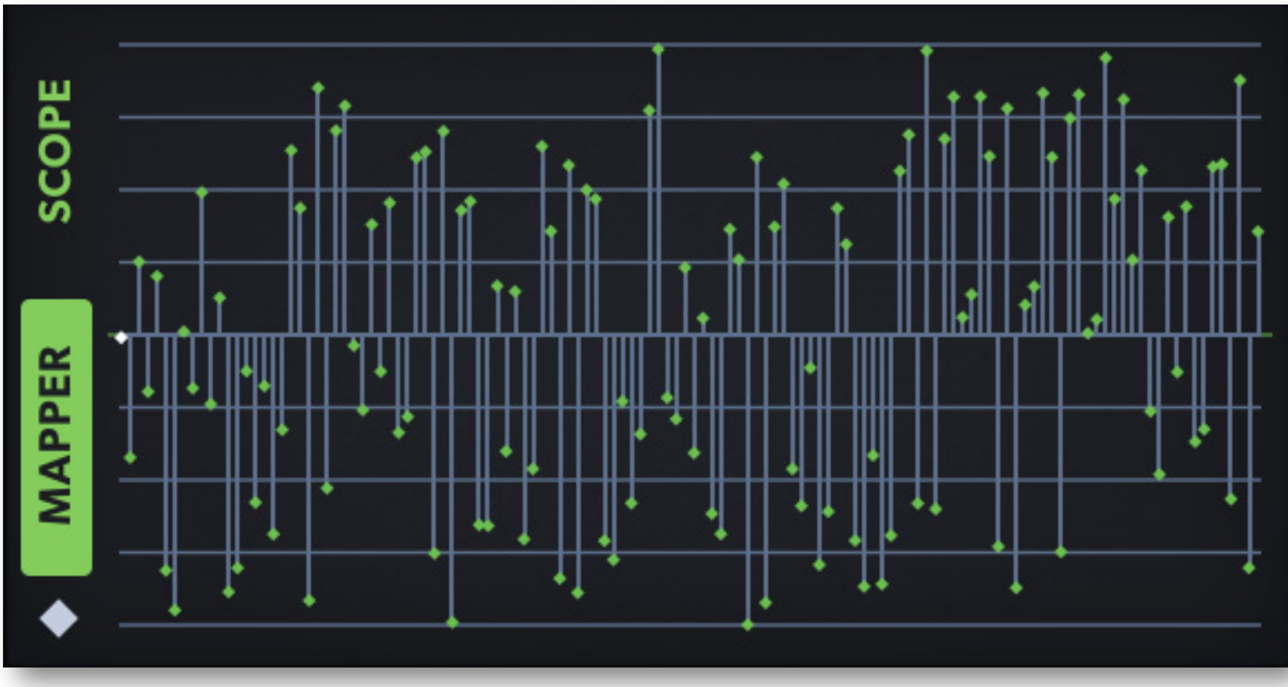
If you view a single source, the audio output appears as a mono sum:

If you select more than one modulation source, only those signals will appear:



Mapper

The Mapper is a modulation source comprising from 2 to 128 values from -100.00 to +100.00. How the values are interpreted depends on the Mode (see the next page).



the simple view of the Mapper in MAIN, set to 128 random steps here

The expanded view (click on the MAPPER button in the control bar) includes Mode and Source options as well as a manual reset button:

the expanded view of the Mapper, set to 48 steps here

Mode

The Mode switch specifies how the values in the Map are interpreted:

Key..............................the 128 MIDI notes, ignoring any selected Source

Map Smooth..........the specified Source (see below) mapped onto the values then interpolated

Map Quantize.........the specified Source mapped onto the values, not interpolated i.e. steppy!

Increment................New MIDI notes or zero-crossings in the Source will step through the table

Source

Selects a modulation source (MIDI, LFO, envelope) to be mapped. The MIDI control sources will only work if Uhbik is receiving MIDI (please refer to the documentation of your host application).

Manual Reset

Momentary button that causes the step index to jump back to the first value.

Steps

Specifies the number of values in the map. Also available in the menu (Step Count).

Drawing Tools

As the tools are standard across all u-he synths with modulation mappers (ACE, Bazille, Zebra2), you will find certain options and functions more useful than others.

If the Draw mode is set to *Freehand*:

Draw in the window by clicking and dragging. Click on a point and drag up or down to move it vertically. For fine tuning, hold a *shift* key beforehand.

To select parts of the map, hold down a SHIFT key while dragging. Note: The functions listed on the next page are applied to the current selection, if one exists.

To deselect, either click outside the current selection or right-click and choose *Deselect* from the *Selection* sub-menu (this entry will only appear if something is selected).

Mapper context menu

Right-clicking in the edit window opens a menu of editing tools:

Copy / Paste	Copies the map or selection to the clipboard, or replaces the map / selection with a previously copied one.
Shapes	Draws a ramp, triangle, sine, cosine, root or quadric shape (spectralize is a special function that interprets the data as harmonics then replaces it with the corresponding waveform). If you create a shape with fewer than 128 steps, the pattern will be repeated to fill all 128 steps.
Draw	Basic drawing mode: ***Freehand*** (painting), *Line* (straight lines), *Level* (multi-point steps) or *Halfsine* (curve segments).
Cmd/Alt-Draw	Alternative drawing mode—hold **cmd** (Mac) / **alt** (Win) before drawing. The options: *Erase* (zero), *Scale* (multiply), *Shift* (2D move) or *Warp* (2D bend).
Selection	Functions: *invert*, *shift left*, *shift right*, *every 2nd / 3rd / 4th*. If nothing is selected, only the 'Every' options will appear in the submenu.
Reverse	Flips the selection horizontally
Invert	Flips the selection vertically
Randomize	Adds random offsets to the selection
Soften	Interpolates between values
Normalize	Expands vertically to minimum / maximum
Make Unipolar	Shifts values so they are all above the line, rescaling if necessary
Straighten	Draws a straight line between the first and last point of the selection
Zero	Sets all values in the map to zero i.e. the vertical center
Subdivisions	The number of guides (from 2 to max. 12) above as well as below zero
Soft Snap	While drawing, values will only snap to guides if close enough
Hard Snap	While drawing, values will always snap to the closest guide
Step Count	Same as Steps, but also available in the small mapper view in MAIN.

LFO

A multi-waveform, general purpose low frequency oscillator you can use for any cyclic modulation, for Sample & Hold triggering, or as a source for the Mapper:

Polarity [+]

Shifts the LFO wave 'upwards' so that it only outputs positive values.

Waveforms

On the left are drag & drop sources for Sine, Triangle/Saw, Square and Random waveforms.

Time

Units for the basic speed of the LFO. The list of options includes three absolute times (a tenth, one or ten seconds) plus several synchronized divisions from 1/64 (a 'hemidemisemiquaver') to 8 bars, including various dotted and triplet times.

Rate

Progressively halves or doubles the *Time* with each integer step. The range of -5 to +5 is therefore much wider than you might think!

Phase

The horizontal position of the LFO waveform. This parameter is particularly useful when the LFO is restarted via MIDI note (see FREE / GATE below).

Symmetry

This bipolar knob skews the LFO waveform as follows:

Triangle ....From falling **saw** through pure triangle to rising saw
Square................0% pulse with through perfect square to 100% pulse width
Random............From stepped through trapezoid to triangular (no plateaus, whatever the rate)
Sine.....................Unused—Symmetry has no effect on the sine wave

FREE / GATE

LFO Restart option: If Uhbik is receiving MIDI notes (check that!), setting this switch to GATE will cause the LFO to restart at the current Phase (see above) each time a note is played.

Swing

Swing effect across 2 cycles: Drag the LFO Sine pip onto the Scope and watch how the LFO shape changes as you turn Swing up!

Flow LFO

The three cyclic effects include another LFO hardwired to the main 'character' parameter. Only available in Uhbik **Flanger**, **Phaser** and **Tremolo**:

Time Base, TIMES

The column of option buttons and the TIMES knob work together to control the rate of the effect.

Multiply..................Synced to host tempo, measured in quarter notes. For example, setting TIMES to 16 means that the LFO wave will last for 16 quarter notes. Higher TIMES result in slower motion.

Divide ....Synced to host tempo, measured in fractions of quarter notes. Setting TIMES to 16 means that the cycle will last for only one 16th (a semiquaver). Higher TIMES result in faster motion.

Seconds ....16 TIMES means 16 seconds (higher TIMES are slower).

Hertz ....16 TIMES means 16 cycles per second (higher TIMES are faster).

Manual ....The waveform is static unless you shift the Phase (see below). In Manual mode, TIMES adjusts how many Flow LFO cycles are included in the full range of the Phase knob. Example: If TIMES is set to 4 and you move Phase from 0 to maximum, you will have scanned through 4 complete cycles.

Phase

Shifts the Flow LFO to the right so that the effect can rise and fall precisely where you want within the music. The Phase setting is therefore especially useful in Time Base modes that depend upon the song tempo.

Offset

Shifts the Flow LFO phase between audio channels, in addition to the Phase setting. The simplest case would be shifting a stereo signal in opposite directions. LFO offset is a fairly common feature in conventional stereo effects, but please remember that Uhbik's channel offset also works in e.g. 5.1 surround, in which case the phase is shifted outwards and to the rear.

Wave, Symmetry, Scale

Wave continually adjusts the basic Flow LFO waveform from triangle to sine.

Symmetry skews the waveform horizontally so that the second half of the cycle is either longer (negative values) or shorter (positive values) than the first half.

Scale skews the waveform such that the lower half is either longer and more subtle (negative values) or shorter and more pronounced (positive values) than the upper half.

Together, these three parameters allow for precise control over the shape of the "flow." For example, with certain settings, you can focus most of the effect on the offbeat.

Note: *Wave, Symmetry and Scale* have rather different functions in Uhbik Tremolo, which offers multiple waveforms and chopped patterns.

Drag & Drop

This method provides an easy way to assign matrix slots and also allows you to view modulation signals—such as envelopes, LFOs, or even the output of a matrix slot—in the scope.

Drag & drop modulation sources are the diamond-shaped **pips** next to...

- the ENV label: Source 'Envelope'
- the arrows below the ENV label: 'Env Rising' (single arrow) and 'Env Moving' (double arrow)
- the MAPPER label
- the LFO waveform icons
- the labels in the modulation source bar

...as well as the matrix target selectors.

Drag & drop **targets** are:

- most of the knobs, excluding rotary switches
- matrix Depth controls
- matrix source selectors
- the Scope (the main area as well as the 4 slots)

Ambience

While most of the other effects in the Uhbik family are geared towards synthetic, radical effects, **Uhbik Ambience** is the master of understatement. Good ambient reverb should be inconspicuous, but when it's not there, something important is obviously missing.

The plan was not to create the most natural ambient reverb possible, but rather to create the most pleasant sounding one. It should not be as "in-your-face" as convolution reverbs tend to be, the reverb needs to blend well with the dry signal to create a coherent audio image. To achieve this we combined two concepts that are seldom seen together: **early reflections** and **plate reverb**.

Early reflections are rapid-fire echos that appear after only a few milliseconds and determine our immediate perception of room size and structure. The plate reverb (late reflections) appears more or less soon afterwards, with a chaotic character that smears the original sound. It refines our perception of the surroundings, including the texture of the walls.

Input Gain

Uhbik Ambience has several gain controls required as **modulation targets** for complex effects, but which can be left at maximum for standard reverb. The **Input Gain** knob at the top left is one such control: it adjusts how much signal is sent into the reverb. Unlike the MIX knob, Input Gain does not affect the level of the dry signal.

Tip: Modulating Input Gain with the envelope can deliver interesting results, but as this does not affect the length of the reverb tail it will not give you a typical 'gated reverb'. For such effects you could experiment with modulating the LATE Gain or the MIX instead.

Pre-Delay

Global delay before the onset of early reflections. Pre-delay can make the dry signal appear closer and is also useful for slapback style delay effects. See OLD/EXT...

LF / HF Range

A pair of filters which can be used to narrow the bandwidth of the signal before it is passed on to the reverb. **LF** adjusts the low frequency and **HF** adjusts the high frequency. By default, LF is at minimum and HF at maximum.

Q

The **Q** (quality) knobs adjust the ‘steepness’ of each filter.

Early Size

Early reflections are irregular echos, the number, arrangement and levels of which depend on the selected algorithm (see MODEL below). Early Size affects the time-span of these echos and therefore the perceived dimensions of the immediate surroundings, with a range of about 1 millisecond (shoebox) to over 60 ms (about 40 meters between the enclosing walls).

Spread

Adjusts a channel-independent shift of up to 20 ms between individual echos. This is irregular i.e. each channel has a different reflection pattern. Using a lot of spread can result in rather extreme spaces which still sound transparent. Works differently in the Natural MODEL – see below.

Gain / Pan

Level and position controls for the EARLY reflections.

MODEL

Uhbik Ambience includes four basic reverb models. Although these are based on very different algorithms, they all share a common set of controls. The choice of model depends on the source material and its function within the music…

The **Open** option is probably your best choice for subtle ambience, while **Direct** is often more suitable for up-front reverb. The **Small** model was designed for rooms with prominent early reflections and relatively short reverb tails.

Selecting the **Natural** model reveals a Distance control (see the next page). Note that the Spread parameter (see above) has a different function in this mode: The channel pairs are decorrelated to ensure a stable stereo field, and Spread moves these pairs further apart. The result: low Spread creates local resonances for a warmer sound, while higher Spread will give you a flatter response.

Modulation

The structure and relative lengths of delays within the network were chosen to produce the most natural-sounding reverb. However, percussive sounds – the true test of quality reverb – can easily sound too cold and metallic. Uhbik Ambience’s Modulation parameter adds slowly changing offsets to all delay times, which effectively adds warmth to the reverb effect. Note, however, that stronger Modulation can lead to unwanted flanger-like motion.

Density

The level of the forward channels, and therefore the diffusion. For long decays you would normally set a high density, and for short decays a low density (otherwise the reverb can get too metallic).

Distance (Natural model only)

Effectively front-back localization. Imagine that you are standing with your back to one wall while somebody is singing from the opposite wall, the maximum Distance. The room's reverb behaviour remains constant, but the reflections reach your ears differently depending on your position.

Bass

A wide-range low shelf filter within the reverb, effectively the decay time from very short to about twice the normal length. In natural environments, low frequencies generally remain audible for longer than high frequencies. However, to avoid clashing between tracks you might like to shorten them by turning down the Bass.

Treble

Similarly, the Treble knob sets the decay time for high frequencies...

Frequency

... and the associated Frequency knob sets the cutoff position of the treble filter (high shelf).

DECAY

The reverb tail is generated by means of a complex network of short delays. Some of these pass their signal forward, while others pass it back to an “earlier” position within the network. DECAY controls the levels of these feedback channels, and consequently the length of the reverb tail.

Gain / Pan

Level and position controls for the reverb tail (LATE).

REVERB

The REVERB knob crossfades between the initial (EARLY) reflections and the reverb tail (LATE). Each has its own Gain and Pan control so you can set up and even modulate 'strange spaces'.

MIX

The MIX knob controls the overall reverb level, crossfading between 100% dry and 100% wet.

Compressor

Most compressors offer only the basic parameters *Threshold*, *Ratio*, *Attack* and *Release,* which are often enough to get the job done. In the role of a generic compressor, Uhbik Compressor also does a fine job bringing level fluctuations under control or delivering compact, dense, breathing compression. But there's more – **Uhbik Compressor** actually comprises two compression units in series, one of which is dedicated to the unique INFLATION process…

INPUT GAIN

Adjusts the signal level before any processing.

INFLATION

All the other controls affect regular downward compression; familiar territory to most readers. Loud parts are made quieter, and the output volume is turned up to compensate.

Inflation is practically the opposite of downward compression: it amplifies the quieter parts until a certain threshold is reached. Often referred to as "upward compression", only very few devices actually do this, and in practice their use is limited. As very quiet signals are too easily amplified in an unpleasant way, upward compression is hard to handle, typically suffering from unnatural envelope timing and troublesome level tracking. This is the main reason we still consider our implementation "work in progress". We welcome user feedback to improve this feature.

Level Indicators

The left bar graph (labeled **INFL**) indicates the amount of inflation, while the right bar graph (labeled **GR**) displays gain reduction. The scale between them is measured in decibels (dB).

THRESHOLD

Sets the level at which gain reduction starts to happen. At low values the compressor already responds to quiet signals, while at high values it will only respond to the loudest of signals.

SOFT KNEE

The Threshold isn't a specific point along a response curve. A good compressor should be quite forgiving when the parameters are not perfectly set up. Finding the optimum threshold can be tricky, and that's where Soft Knee comes to the rescue, creating smooth transitions between 'uncompressed' and 'fully compressed'. Soft Knee effectively turns a precise threshold into an 18dB window within which the compression smoothly increases between 1:1 and the set RATIO...

RATIO

How much compression is applied when the input signal level exceeds the threshold. At 1:1 there is no compression at all, while the maximum 20:1 is almost brickwall limiting.

Tip: A ratio of **4:1** is a good starting point for general-purpose instrument compression. At this ratio, +4dB above the threshold results in only +1dB gain at the output.

ATTACK

As an instantaneous jump above the threshold would lead to unwanted distortion, compression is usually applied within a 'time window', smoothing out any abrupt changes. Fast Attack will still catch sudden transient peaks, while slow Attack can give you a naturally smooth sound or a more synthetic, 'artistic' contour.

FEED-BACK

Most modern compressors (including Uhbik Compressor), use **feed-forward** detection by default. This process looks at the input signal and adjusts the output gain accordingly. Implementing feed-forward in hardware is difficult, however, so classic compressors tend to use **feed-back** detection instead, a method similar to the thermostat in home heating systems: Gain circuitry at the input reacts to a control signal derived from the output. Circuits applying this principle are easy to build, and feed-back detection also has the advantage of naturally compensating for less-than-perfect component tolerances and nonlinearities.

The two detection types sound different: Feed-forward is direct and 'perfect', while feed-back is more forgiving, more 'musical' at the price of a more limited range: Compression ratios well above 3:1 are impossible. To achieve *brickwall limiting* in a feed-back compressor would require infinite gain in the control loop. To deliver its maximum ratio, Uhbik Compressor applies a LOT of gain!

Which of the two modes works best depends on the audio material. FEED-BACK can easily slam the signal down much too strongly. If in doubt, let your ears decide.

RELEASE

The time it takes to recover to its idle state once the signal level falls below the threshold. Like fast ATTACK, fast RELEASE allows more rapid fluctuation while risking distortion or modulation artifacts. The latter can become problematic if the release is shorter than the fundamental wavelength of the input signal, so it is best to set RELEASE no shorter than absolutely necessary. On the other hand, if RELEASE is set too long the entire process becomes ineffective, as the compressor never has a chance to recover.

Adaptive

The **Adaptive** knob introduces semi-automatic release. The higher the value, the more the release will be prolonged by a factor derived from the momentary signal energy: Steady signals will have longer release times, while fluctuating signals will retain most of their dynamics.

DETECTION HPF

Inserts a high-pass filter into the detection path so that frequencies below 75Hz are excluded from detection. Use this option to boost material that is both rich in transients as well as relatively bass heavy—for instance drums, bus-mixes or complete mixdowns.

AUTO MAKEUP

As downward compression attenuates signals above the threshold, the output volume would normally be turned up to compensate. AUTO MAKEUP does this automatically by calculating how much signal loss would occur at the current THRESHOLD and RATIO settings.

Note that the makeup gain actually applied is only half the calculated signal loss, as with moderate ratio settings this usually sounds more 'correct' to our ears. On the other hand you might have to turn the output up a little if RATIO is set very high.

Wet Gain

A makeup control for the compressed signal only, letting you balance the levels of the dry and processed signals before the final MIX. A more useful modulation target than MIX...

MIX

The overall amount of compression, a crossfade between 100% dry and 100% wet.

Delay

Uhbik Delay is a multi-tap delay resembling a classic tape echo unit with several record and play-back heads. The delay time is a function of tape speed: the slower the tape, the longer the delay. Like classic echo units, Uhbik Delay includes a FEEDBACK control which governs how much of the output is fed back into the input.

Real tape echo has several technical imperfections and limitations. Magnetic tape already has a limited frequency and dynamic range; tape wears out over time; transport mechanisms become erratic etc.. Previously considered serious drawbacks, such irregularities (colouration, flutter) have become quite popular in this era of digital perfection!

Nevertheless, digital delays have some significant advantages, for instance precise and repeatable timing as well as freely adjustable **taps** (the digital equivalent of playback heads).

Uhbik Delay combines all these concepts in a single package, adding song tempo synchronization and much more. As Uhbik Delay has no physical playback heads, the delay times can be as short as you like, and the tap positions can even cross over...

1/16

Uhbik Delay has five adjustable taps. The position of each tap can be set from 0 to 16 semiquavers (a.k.a. 16th notes) via the lefthand row of knobs labelled **1/16**.

Here's a handy conversion table for converting rhythmic values into knob values. Of course you can add two or more of them together e.g. 1/2 plus a 1/4 triplet is 8.00 + 2.67 = 10.67.

1/32	0.50
1/16 triplet	0.67
1/32 dotted	0.75
1/16	1.00
1/8 triplet	1.33
1/16 dotted	1.50
1/8	2.00
1/4 triplet	2.67

1/8 dotted	3.00
1/4	4.00
1/2 triplet	5.33
1/4 dotted	6.00
1/2	8.00
1/1 triplet	10.67
1/2 dotted	12.00
1/1	16.00

PAN

The panorama position of each tap. See Multichannel Operation a couple of pages down.

VOL

The volume of each tap. Note that VOL does not affect the amount of feedback for each tap.

FEEDBACK

Controls overall feedback intensity—effectively how long it takes for echos to fade to silence.

1—2—3—4—5

These buttons **activate feedback** for each tap, allowing complex patterns to be set up very easily. Note: If none of the selectors are active, an 'invisible' tap at 16/16 is used for feedback.

The tape echo paradigm still holds true here: The recorded signal is read by multiple playback heads at several points along the 'tape', and feedback signals are re-recorded.

If several taps are active, the delay can build up instead of decaying. However, this isn't a serious problem as non-linear processes (limiting and distortion) enforce a ceiling. Also, activating feedback for more taps automatically reduces the feedback levels of all taps, which also helps to limit feedback. The best approach is usually to TURN IT UP and see what happens!

Low Cut, High Cut

Additional sound-shaping within the feedback loop: Two shelving filters that progressively colour the echoes as they fade—typical of tape echo machines, and fully adjustable in Uhbik Delay.

Soft Clip

Distortion within the feedback loop causes repeat echos to lose fidelity. Another typical behaviour of tape echo machines which is adjustable in Uhbik Delay.

SPEED

Scales the delays of all taps. Not only useful for altering the overall timing, SPEED is also the easiest modulation target for pitch effects.

Mod Type

SPEED can be modulated directly: *LFO* creates a smooth, animated effect similar to chorus, while *Flutter* introduces less regular echoes, making it the better choice for authentic tape echo.

Rate, Depth

Flutter/LFO speed and intensity.

GAIN

Separate **Input** and **Wet** (Output) levels. These two are particularly interesting candidates for modulation in the matrix.

MIX

Crossfade between 100% dry and 100% delay.

Multichannel Operation

Like all other Uhbik models, the Delay can also function in a surround environment, with one notable enhancement: No longer simple stereo position controls, the **Pan** knobs scan through all possible positions. For instance in 5.1 they 'pan' through channels in the order surround-left, left, center, right, surround-right. For 7.0 and 7.1, add rear-left and rear-right to that list.

Note that none of the delay signal is sent to the LFE channel (the '.1' sub channel). Also, if effects set to stereo are played on surround systems, they might need to be adjusted so that the delays don't only appear in one side or only in the rear speakers, for instance.

Tips

Groove delay

As mentioned above, a tap can still contribute to the feedback even if its volume is set to zero—remember that the **VOL** levels only affect the final output from each tap.

Imagine an echo that repeats every quaver (a.k.a. quarter note) although precise quavers are not part of the echo signal: Set one of the taps to 8.00 with the volume at 0.00, and active Feedback. Set two other taps to values close to (but not precisely) 8.00 and 4.00, and adjust their volumes.

Ping-pong delay using the **Haas effect**

Set up Tap1 as an inaudible (VOL = 0.00) feedback delay, length = 4.00. Set taps 2 and 3 both to 2.00 and pan them fully L-R. Similarly, set taps 4 and 5 to 4.00 and pan them fully L-R. Listen to the results with taps 2 to 5 all set to maximum volume with feedback at around 50.00. Now spread the lengths of each pair slightly, in opposite directions like this:

Delay2 = 1.90
Delay3 = 2.10
Delay4 = 4.10
Delay5 = 3.90

Although the left and right panned echoes in each pair (2 & 3, 4 & 5) are almost simultaneous, one appears to come from the left and the other from the right. This is the Haas effect at work, a subtle but interesting stereo effect that would be impossible using only two taps.

EQualizer

Background

You will find equalizers (EQ) in any professional studio. Apart from volume control and summing, equalization is the most common type of signal processing—so it is hardly surprising that EQ is a highly controversial topic in the audio world!

For some, the most important factor is the number of frequency bands, for others it is the ease of use. EQ characteristics are often described in highly subjective terms such as ‘transparency’ or ‘warmth’. Heated debates are commonplace: to quote a literary friend, "the altercation is so bitter because so little is at stake."

Certain EQ properties can be judged objectively. For instance, many digital EQs suffer from overly steep HF filters, which either leads to irritating artifacts or to a lack of high end. Similar applies to the bass end of the spectrum, where certain algorithms require mathematical precision.

Also, the compulsion to minimize CPU usage at all costs can tempt developers to cut corners and accept lower-quality results, which is especially bad news for an EQ.

Uhbik EQ goes for maximum flexibility with a minimum of controls, and lowest CPU-usage without compromising audio quality. It combines freely tunable frequency bands with presets for all other options, and Q-factors which automatically adjust to suit the current gain.

BOTTOM

A low-shelf filter specially designed for bass frequencies. It is not fully tunable, but is more precise than conventional models. Click on a button or use the rotary switch to select a frequency between 40 Hz and 350 Hz, then adjust the gain from -24 to +24 dB using the lower knob.

BAND 1 | BAND 2

The 6 switches select the filter type.

Upper row

Off......The band is deactivated, it will not affect the sound.

Low Shelf......Classic low-shelf filter. The gain knob controls the amplitude of frequencies below the value set by the frequency knob (see below).

Wide Bell......The gain knob controls the amplitude of frequencies around the value set by the frequency knob. Very low Q-factor.

Lower row

Flex Bell......The gain knob controls the amplitude of frequencies around the value set by the frequency knob. The Q-factor automatically increases with greater gain values (negative or positive) so that the perceived volume remains fairly constant.

Narrow Bell......The gain knob controls the amplitude of frequencies around the value set by the frequency knob. High Q-factor!

High Shelf......Classic high-shelf filter. The gain knob controls the amplitude of frequencies above the value set by the frequency knob.

Frequency (unlabeled)

Uhbik EQ provides two **semi**-parametric filters covering a wide frequency range—from the low mids up to frequencies beyond the upper limit of human hearing (>20 kHz). Unlike **fully** parametric EQ devices, the Q-factors are managed internally, ensuring optimal bandwidth without the need for manual adjustment.

Gain (unlabeled)

Attenuates or amplifies the signal between -24 and +24 dB.

Low Cut, High Cut

In addition to three classic EQ filter bands, Uhbik EQ has two quasi **brick-wall** filters, with a choice of fixed frequencies below/above which very little of the signal will pass. These are useful for rigorously eliminating e.g. rumble and hiss, leaving the other bands free for other tasks.

MASTER GAIN / MID GAIN / BELL GAIN

The large dial near the center changes its function according to the selected **Gain Type**, and the label changes accordingly:

Gain Type

Output........................Regular output gain, +/- 24 dB. MASTER GAIN.

Wide Mids ....The gain of an extra, very wide mid range frequency band spanning almost the entire audible spectrum. Only very low and very high frequencies remain unaffected. MID GAIN.

Center Bell ....The gain of an extra bell-shaped band (similar to the *flex bell* option of the two semi-parametric bands) whose frequency lies exactly between the band 1 and band 2 frequencies. Of course the width also depends on the band 1 and 2 settings. BELL GAIN.

Question…

Why are the frequencies of the two main bands adjustable, while their Q-factors are not?

Answer: tunable frequency bands can be 'modulated' at will via automation. This feature transforms Uhbik EQ into a highly flexible creative tool. Early plans to add more bands and more control were dropped in favour of wider, more promising horizons. After all, many reputable hardware simulations only let you switch between a few fixed frequencies.

Flanger

Uhbik Flanger offers two distinct types, the original **through-zero** effect (tape flanging) as well as the much more common **delay-based** flanging effect...

Flanger History

The original flange effect was made during the mid 1960s by mixing the output of two tape machines playing back the same material. Either tape could be slowed down slightly by applying pressure to the flange of the reel. When the “earlier” of the two tapes is then slowed down until it becomes the “later”, this results in a dramatic effect called *through zero flanging*. In extreme cases i.e. using noise or full-range signals, flanging sounds like the *whoosh* of a jet as it passes overhead. There’s a good example in the middle of the song *Itchycoo Park* by The Small Faces (1967). This type of flanging is seldom realized in software, one notable exception being u-he Satin.

Guitar 'stomp-box' flangers apply a different method: A very short delay is fed back into itself, and this delay time is modulated, creating intense comb filtering which becomes more resonant as the feedback is turned up. Although less dramatic than through zero flanging, this method has the advantage of being almost equally effective on low frequency material.

Virtual Tape Machines

Uhbik Flanger simulates two tape machines (called A and B here) per audio channel, with playback and recording heads that can even occupy the exact same position (which is of course impossible in the real world). This means that feedback from tape A can be recorded to both tapes A and B while one of them is “catching up” with the other.

The Uhbik Flanger applies the principle that only one tape machine needs to be slowed down or sped up for through-zero effects: it is only necessary that both audio signals can *somehow* meet at zero. If the speeds of both virtual tapes were affected (as they usually are in real tape flanging), it would be impossible to maintain precise timing between tracks in a multitrack environment.

The Flow LFO slows down and speeds up tape B only so that the delay between the two machines, and therefore the comb filter effect, is smooth and continuous.

Operation, Delay, DEPTH

The **DEPTH** knob controls the amount of Flow LFO applied to Tape B's playback position – effectively how much the LFO affects the speed of Tape B. The range is 0 to 20 ms.

The **Delay** knob applies a constant delay of up to 10 ms. Although useful for chorus and flanging effects with feedback (see below), too much Delay will diminish the tape-flange character.

The **Operation** switch lets you reduce CPU usage by setting Eco (economy) mode. The loss of audio fidelity is seldom noticeable.

Auto Mix

Controls the amount of Flow LFO modulating MIX (see below). Use this to accentuate the bottom or top of the wave, or to prevent cancellation when MIX is in the -50% position.

Feedback

This parameter simulates the typical stomp box flangers, allowing negative as well as positive values. Note that extreme feedback can lead to self-oscillation, just like in the hardware.

Bass Sanctuary

Inserts a highpass crossover filter (with *Low*, *Mid* or *High* frequency) between tapes A and B so that the bass frequencies remain unprocessed and therefore stable.

Drive

This adds second harmonic distortion to the signal resulting in a brighter, more pronounced effect. High values can introduce a significant amount of distortion, which can be useful for boosting the presence of solo instruments.

MIX

The large **MIX** knob controls the relative volumes of the virtual tape machines. At 12 o'clock (**A**), the output is dry, so you should hear no effect. Moving the knob to the left or right mixes in tape B (the delayed signal) while fading out tape A (the dry signal). The **–** and **+** symbols signify 50% i.e. an equal mixture of both. If MIX is at either minimum or maximum you will only hear tape B…

Negative MIX inverts the tape B signal so the delayed signal is effectively subtracted instead of added, resulting in a more pronounced jet effect.

Grainshift

Pitch Shifting History

The ancestor of today's pitch-shifting and time-stretching devices was developed in Germany by the conductor Hermann Scherchen (1891-1966), the first Apparatus for Independent Control of Pitch and Tempo of Audio Recordings (original: Apparat zur unabhängigen Kontrolle von Tonhöhe und Tempo von Tonaufnahmen). Little information is available, but development is likely to have been during the 1930s, shortly after the invention of magnetic tape recorders.

Scherchen's device had four playback heads attached to a small rotating drum which was mounted between the original playback head and the capstan of a conventional tape recorder. When the drum is rotated, the heads take turns to read the tape, but unlike fixed playback heads this can happen at a variable rate. As long as the drum is not rotating, the pitch remains unaffected. If the drum is rotated in the opposite direction to the tape, very short (smoothly) overlapping samples of the recording are played back while gaps between them are effectively skipped, resulting in a higher pitch. The opposite effect happens when the drum is moving in the same direction as the tape - the pitch is lowered because multiple samples of the same audio material are sent to the output. The effect turns into time-stretching when the tape itself is slowed down or sped up to compensate for the pitch change.

This same principle was applied in commercial devices e.g. the *Eltro Information Rate Changer* used for HAL's death scene in *2001 - A Space Odyssey*. There's a short article about this device on Wendy Carlos' website (an Internet search for "wendy eltro" will take you there).

Although the contraption seems bizarre, the basic principle is still applied in modern granular pitch-shifters, which first appeared as hardware in the 1980s and as software in the 1990s.

Pitch-shifters effectively cut the audio material into small snippets which are then played back, overlapping, at a variable rate. These snippets are **grains**, and their duration is the **grain size**. In the old tape-based devices, the radius of the drum determines grain size while the length of tape in direct contact with the drum determines the overlap between grains. Uhbik Grainshift is not bound by the physical limitations of rotating drums, so you can make the 'drum' impossibly small!

GRAIN SIZE

The maximum GRAIN SIZE in Uhbik Grainshift is about 2 seconds. Unlike conventional pitch-shifters, this parameter affects the duration of grains as they appear at the output, while the input grain size is adjusted automatically.

OPERATION

Uhbik Grainshift can be switched into a fundamentally different mode called **Phase Voc** (phase vocoder), with three quality levels replacing the grain size control:

Phase Vocoder: GRAIN SIZE is replaced by QUALITY options

In Phase Vocoder mode, pitch shifting is achieved by time-stretching/compressing the spectrum of the input signal. The signal is split into its component sine waves via Fourier analysis, and the phase and position of these waves are adjusted via SEMI and SCALE. Note: In Phase Voc mode the grains cannot be played in reverse—negative SCALE is the same as positive.

The effect is highly dependent upon the audio material. While granular often sounds rather rough, phase vocoding can sound mushy due to the loss of transients. However, when applied to vocals and pads, the effect can be very impressive.

Like all FFT-based (Fast Fourier Transformation) effects there is a noticeable latency between the input and output signals. In Uhbik Grainshift this has been left uncompensated for, as the granular algorithm itself would otherwise have required extra delays. If you want to use Phase Voc mode for rhythmically critical material despite the loss of transients, you should try shifting the audio track about 2000 samples forward i.e. earlier.

GRAIN RESET

Although sometimes subtle, a GRAIN RESET can significantly improve the timing of effects.

The two buttons:

Auto resets grains after a moment of silence to suit the audio input material.
Manual is a momentary button that instantly resets grains.

ITERATION

Feeds the output back into the input, often resulting in multiple detuning effects e.g. echos with constantly rising or falling pitch.

SEMI, SCALE

The pitch is controlled by two parameters: SEMI adjusts the grain pitch in semitones (+/- 12) while SCALE multiplies the playback rate within a range 4 octaves. A scale value of 1 retains the original pitch, values between zero and 1 lower the pitch. At SCALE zero, only a single sample is played back, but as grains still overlap, this results in an effect reminiscent of low-pass filtered sample-rate reduction.

The SCALE knob is bipolar: negative values play grains backwards, and this can deliver some very interesting reverse effects!

SEMI and SCALE can be used at the same time: For instance, play grains backwards (scale = -1) while transposing them up an octave (semitone = +12).

OFFSET

This parameter scales the effect differently *per available channel*. For instance, if you set SCALE to zero and OFFSET to 100, the grains in the right channel will play back normally while those in the left channel will be played in reverse.

MIX

The relative volumes of the original (dry) and effect (wet) signals.

Phaser

Phaser History

Closely related to tape flanging is the classic **phasing** effect, which may originally have been an attempt to simulate tape flanging using electronic circuitry. While flanging is delay-based, phasing is achieved using a *frequency-dependent* phase shift.

Either method results in a 'comb filter' effect with multiple peaks and troughs when mixed with the untreated signal, but there is a difference: In flanging any modulation e.g. from an LFO affects the distance between the comb "teeth", whereas in phasing this remains fairly constant.

Frequency-dependent phase shifting is the domain of *allpass filters* which, although these don't affect the timbre of the audio material passing through them, will affect its phase. A rich, deep phasing effect requires several allpass filter stages arranged in series, and the more stages a phaser has, the more teeth will appear in the comb. Two allpass stages are required per tooth.

Most phasers include a feedback channel for added resonance, but something special is going on here: As the signal is phase-shifted each time it is fed back through the filters, frequencies are created that were not present in the original signal – which is why phasers can sound metallic.

Operation

This selector specifies the number of allpass filters: Deep 14, 28 or 42, or Classic 4, 6, 12. Most phasers have fewer than 10 filters, and even Uhbik Phaser's lowest setting (Classic 4) can sound lush. 28 filter stages should be plenty for a complex wash of sound, but the **Deepest 42** setting makes Uhbik Phaser one of the most dramatic-sounding phasers available.

Note that modulation effects tend to become deeper the more allpass filters you use: Compensate by adjusting the TIMES and DEPTH parameters to taste.

DEPTH

Controls the amount of Flow LFO modulation. Note: As the modulation requires some headroom to work fully, its depth is reduced as Spectrum (see below) approaches minimum or maximum.

MIX

Relative volumes of the dry and wet signals (negative values invert the wet signal). Note that the phasing effect is normally most pronounced when MIX is set to +/- 50%. With more extreme Feedback (see below), however, you should try other MIX values, even removing the dry signal entirely by setting MIX to +/- 100%.

Bass Sanctuary

Like in Uhbik Flanger, this is a highpass crossover filter (with *low*, *mid* or *high* frequency) so that the bass remains unprocessed i.e. is not tossed around by the effect.

Feedback

Feedback not only widens the cancellation areas (gaps between the teeth of the comb) and makes the peaks more resonant. Negative values invert the phase of the feedback signal.

Like in Uhbik Flanger, high feedback can lead to self-oscillation (squealing). For the sake of stability, Uhbik Phaser was carefully calibrated so that self-oscillation is always short-lived.

Spectrum

Moves the comb around the frequency spectrum. **Like cutoff** in conventional filters, it specifies the center position before any modulation.

Runciter

Runciter, the characterful distortion filter, always stood out in the Uhbik lineup by not bearing the *Uhbik* name. But now, with all version 2 Uhbiks sporting fuller, more descriptive titles, Runciter has officially joined the fold as **Uhbik Runciter**—finally cozying up with the rest of the family!

About Filters

Filters are powerful tools for shaping sound—one reason they've remained so popular over the years. While they are technically related to the parametric EQs found on mixing consoles, filters can be used much more aggressively, especially when high feedback levels come into play.

Several classic filter designs exist, each with distinct characteristics. The two most common are '*cascade*' and '*state variable*', both of which allow multiple modes—lowpass, bandpass, and highpass—as well as feedback and resonance. Runciter's filter is a 4-pole (24 dB/octave) state variable capable of delivering all three modes simultaneously.

Distortion as Feature

One of the most intriguing qualities of analogue filters is their overdrive or distortion behaviour. Depending on the circuit, certain components can be pushed past their intended limits, creating a uniquely musical distortion. This led to filters designed with post-distortion in mind—such as the iconic wah pedal. Runciter embraces this concept and more: It not only emulates the character of overdriven analogue components, but includes an extra 'fuzz' stage for more sonic flexibility.

MODULATION

The section top left is for direct CUTOFF modulation, as indicated by lines on the panel. Please note that the resolution here is finer/faster than in the modulation matrix.

Source | Depth

Modulation source selector and bipolar amount.

CONTOUR

Shape

Selects one of the follow envelope types:

Fast .... A very short attack time paired with a slow release time. This option is particularly good for percussive filter effects.

Ride .... Medium attack / decay times. Most suitable for smoothly ‘riding’ the input.

Slow .... Reacts slowly to rising volume, but falls more rapidly during the quiet parts. Most suitable for adding motion to long, mostly static tones.

Transient .... The input signal is analyzed for positive transients i.e. sudden jumps in the input signal, and these are used to trigger the envelope (with a short attack and an exponential decay). Best for percussive material such as drums or plucked strings.

MIDI 1 .... Similar to transient, but triggered by MIDI notes. The envelope starts at a value corresponding to the note velocity. Adjust dynamic response via **Sense**.

For details about routing MIDI data into effect plug-ins (e.g. Runciter), please refer to the documentation of your host application.

MIDI 2 .... Like *midi 1* except that the envelope does not start at the velocity value of the MIDI note, but approaches it at the specified **Rate** (see below), then holds that value until the next Note On. Smoother than *midi 1*, and particularly useful for rhythmic note sequences where velocity is easier to edit than automation. Experiment!

Depth | Sense | Rate

Depth controls the modulation amount. **Sense** determines the threshold of the analysis, just like a compressor's Threshold knob. **Rate** affects the modulation speed by smoothing the envelope.

Envelope movement is displayed in a bar above the Sense knob. The best strategy is often to set a medium **Rate**, find the **Sense** value that gives you the most movement in the indicator, then tweak CONTOUR **Depth** and filter **CUTOFF** to taste...

FILTER

FILTER TYPE

Classic................the original Runciter, 24 dB / octave
2-pole SV...........State Variable, 12 dB / octave
3-pole SK...........Sallen-Key, 18 dB / octave
4-pole CC ....OTA Cascade, 24 dB / octave

Fuzz | Colour

Fuzz adds strong distortion, and its tone is controlled via **Colour**. Of course the distortion sound is highly dependent upon the original audio being processed.

The Fuzz 'circuit' is positioned post- filter in the signal chain. If the FILTER TYPE *Classic* is selected there are actually two fuzz units—one between the two stages and one at the end.

Note: Runciter's fuzz is a level-dependent offset which forces the signal against a ‘brick wall’ within the non-linear filter circuitry. This method creates even-numbered harmonics similar to tube distortion. It is usually considered pleasant, but often means losing some bass.

Input Drive | Output

The **Input Drive** parameter adjusts the input gain by a generous +/- 48 dB, and has a significant impact on the amount of distortion the filter circuit delivers. Input Drive is pre-Fuzz.

The **Output** knob compensates for significant changes in signal level caused by Input Drive in combination with the various FILTER TYPEs. Filter Output is post-Fuzz.

Note: The lower the input level, the more prominent the resonance becomes. Conversely, higher input levels reduce the relative amount of resonance, and distortion takes center stage.

CUTOFF | RESONANCE | MIX

CUTOFF controls filter frequency within a range of 20 Hz to 20 kHz, scaled logarithmically in octaves from 0 to 10 (1 unit = 1 octave). **RESONANCE** controls filter feedback from very mild to almost self-oscillation, while **MIX** crossfades the output from 100% dry to 100% filtered.

Cutoff Pan

CUTOFF offset for each channel pair. Negative Cutoff Pan values mean that the *left*, *left surround* and *left rear* channels will be brighter while *right*, *right surround* and *right rear* will be darker. Neither the *center* nor the *LFE* channels are affected.

Lowpass | Bandpass | Highpass

These knobs mix the output of all three filter types. You can e.g. emphasize the cutoff frequency of the lowpass (without using resonance) by adding a little bandpass. If the lowpass and highpass are set to the same value, the result is a so-called *peak filter*. The height of the peak can be controlled via the RESONANCE knob.

Shifter

From serene barber-pole waves to full frequency freak-out..

Frequency Shifter History

The origins of the Frequency Shifter go back to the early days of radio technology. Frequency shifting is related to ring modulation (RM): two signals are multiplied together, resulting in two **sidebands**. One of these is the sum of all frequencies in both signals, the other is the difference. Unlike RM, frequency shifters output a single sideband shifted up or down by a constant value. That's why frequency shifters are sometimes called 'Single Sideband Filters'.

Like RM, strong frequency shifting gives the signal a metallic character because all frequencies are shifted by a constant (e.g. 100Hz) instead of a factor (e.g. 2 times). For instance, when shifted 100Hz up, 440Hz becomes 550Hz, while its octave (880Hz) becomes 980Hz—which is NOT an octave above 550Hz: Harmonic relationships are therefore destroyed by frequency shifting.

Frequency shifters are not only only suitable for special effects e.g. horror-movie voices. Used in moderation, frequency shifting is similar to chorus or phasing, but without the need for an LFO. While the pleasant beating of mildly detuned oscillators can become irritatingly fast when you play further up the keyboard, frequency shifting keeps this movement constant. Uhbik-S can synchronize beating to the song tempo.

There are similarities with phasing, as both cause a comb effect to move around the audio spectrum. However, frequency shifters move constantly downwards or upwards ("barber pole"). Phase cancellations that disappear out of the top reappear at the bottom and vice versa. Like a phaser, resonances can be accentuated via feedback.

The Uhbik Shifter was designed so that all negative side-effects are either minimized or eliminated. Most other sideband filters have a poor frequency response – perhaps because the lowpass filter cutoff is set too low, just to be on the safe side. Also, a quality sideband filter needs either a high latency (e.g. for Hilbert transformation) or clever routines to prevent otherwise inaudible sidebands from folding back into the audio range.

SHIFT

The Shift knob controls the amount of frequency shift relative to the value set by the *Range* switch (see below).

Offset (SHIFT)

An frequency shift offset per audio channel. For instance, the shift can be continuously rising in one channel while continuously falling in the other.

Range

There are four absolute frequency ranges here: 1 Hz, 10 Hz, 200 Hz and 4 kHz. Note that the latter two are more suitable for extreme effects than for subtlety.

The 1/1 and 1/16 options are factors relative to the current song tempo. An example: If the tempo is 120 bpm, +100% of 1/1 is equivalent to 0.5 Hz i.e. the effect repeats every 2 seconds. Don't be put off by a little arithmetic here, it will soon become second nature.

PHASE

This parameter is a unusual for a frequency shifter. It adjusts the phase of the frequency-shifted signal within a range of 0° to 360° i.e. one complete cycle. Although the frequency shift has effectively disengaged the processed signal from the dry signal i.e. they are out of phase anyway, it does open a few interesting doors. Firstly, it lets you manually adjust phase when the shift is zero. Secondly, it lets you position the cyclic effect to suit your music.

Offset (Phase)

A per channel offset for the PHASE.

Auto, Manual Reset

These buttons reset the phase to 0° i.e. to be in sync with the dry signal. The latching **Auto** button automatically resets the phase whenever the input signal level drops below a certain (very low) threshold. The momentary **Manual Reset** button instantly resets the phase.

FEEDBACK

This works like phaser feedback—it increases cancellation as well as resonance. Please note that extreme feedback can lead to self-oscillation.

MIX

Sets the relative volumes of the dry and wet signals. The maximum value is suitable for special effects such as robot voices, while the central position (50.00) is best for phasing.

Tremolo

Tremolo is most commonly defined as a regular, repetitive change in volume, achieved through whatever method the instrument allows. In electronic music, tremolo typically refers to amplitude modulation using an LFO. **Uhbik Tremolo** incorporates many different elements, and in this case, the Flow LFO plays a much larger role...

In parallel with amplitude modulation, the signal can be moved across multiple audio channels (e.g. for quadrophonic panning), and the tone can be modulated using the lowpass filter.

The perceived position can be shifted using a short delay between stereo channels, making use of the **Haas Effect**... How we hear position is affected by slight differences in the time it takes for the sound to reach each ear. Up to about 40ms, the longer the delay between the left and right channels, the more extreme the panning effect will be.

The Flow LFO can modulate three parameters at the same time: the stereo / surround position (applying the *Haas effect*), filter attenuation (tone), and gain (normal tremolo). See the next page.

While the most useful rates for flanging and phasing are quite slow (one cycle often extending over several bars), tremolo is more musically useful if the Flow LFO is set faster e.g. several cycles per second. Also, soft LFO shapes tend to be less interesting than hard ones such as pulse waves or choppy patterns.

To make this work in Uhbik Tremolo, it has a much more complex Flow LFO than the other Uhbiks. You can even set different patterns for each channel (Pattern Y mode).

HAAS DELAY

The maximum delay for each channel. The Haas effect only appears if the LFO phases are shifted apart via Flow LFO **Offset**. When summed to mono, the Haas effect becomes a kind of chorus.

FILTER ATTENUATION

The Filter Attenuation knob controls how much LFO affects the cutoff of a lowpass filter. As well as making a sound quieter, reducing high frequencies will also give it a less defined position.

GAIN

How strongly the Flow LFO modulates volume. The **Gain Law** selector determines the range and response of modulation: either linear, or exponential at -12dB, -30dB or -96dB.

GENERATOR

The row of buttons below the Flow LFO selects one of six options:

1x | 2x | 3x | 4x

1x means the Flow LFO behaves exactly the same as it does in the other Uhbik effects. For the other numbered options it completes 2, 3 or 4 cycles in the same time as it would otherwise complete only one. What makes these extra modes interesting is that the Symmetry parameter still works as if there were only a single cycle. Consecutive LFO cycles are squashed and stretched, resulting in a kind of tremolo "swing" - which can be perfect for music with a complex groove.

Px | Py

The two **pattern** modes are for creating complex rhythms or dramatic gate effects. The patterns are user-definable, so these modes have a special Flow LFO editor:

Uhbik Tremolo with the GENERATOR set to Px

Pattern Grid

Up to 11 patterns with up to 16 steps can be defined. The description here might seem a little confusing at first. Simply reading this chapter will not be enough, you will have to try it out!

The **Patterns** replace Uhbik's standard Flow LFO shapes. For 16 semiquavers, set the TIMES to 4 quarters (Multiply). For typical gating effects, 2 quarters is often enough.

The Wave control is replaced by **Fade Y**, which sets a **pattern position** on or between the patterns. The value of this control corresponds with the numbers on the left i.e. 0 to 100, while clicking on a point will display a "Gate number" (pattern index from 1 to 11).

Setting it to e.g. 5.00 will apply patterns 1 and 2 with equal strength. Note that intermediate values are particularly useful for adding accents, like in a drum-machine.

The Symmetry control is disabled (amplitude scaling is redundant for on/off values), and Scale has been replaced by a **Smooth** control, which you should find very useful for avoiding clicks!

STEPS

Pattern length from 4 to maximum 16. Note that the steps are always spread across the total length of the LFO, so when you reduce this number, the steps will be played back more slowly. Example: if you reduce the number of steps from 16 to 12 and want the shorter pattern to play back at the same rate, you should also reduce TIMES from 4 quarters down to only 3.

Px vs Py

In **Pattern X** mode, the Offset X parameter separates the phase position for each channel (stereo, quad etc.), just as it does in the normal Wave modes. In **Pattern Y** mode, the Offset Y parameter separates the pattern i.e. it lets you use different patterns in different channels—which also works in surround. Note that in Pattern X mode, Offset X splits the phase position indicator while in Pattern Y mode, it splits the pattern position indicator.

Two Tips for Uhbik Tremolo

Try Uhbik Tremolo with a complex drum loop, and experiment with all the controls using the various GENERATOR modes. In Px or Py mode, activate a few points in the grid. You can click+drag around the grid to 'paint' or erase multiple points. Adjust Offset X and all modulation depths to accentuate or soften transients.

While adjusting the LFO rate, Uhbik Tremolo can sometimes get out of step. This can be remedied by stopping and restarting playback in the host application (which resynchronizes the LFO). This also works in the other Uhbik modulation effects, but is particularly important for Uhbik Tremolo as the rhythmic content of the processed audio can be so much stronger.

Why No MIX Knob?

A dry/wet mix control wouldn't make sense in Uhbik Tremolo because adding the dry signal to the Haas delay would result in an obvious chorus which is likely to overpower the tremolo. However, you can isolate the dry signal by setting HAAS DELAY, FILTER ATTENUATION and GAIN all to zero.

Combining all three effect modules in Uhbik Tremolo can deliver some interesting spatial effects: “A bit of everything” is often more effective than a pure tremolo or rhythmic pan here. As the volume can be kept fairly constant, it is particularly easy to mix into a song.

Preset Browser

“You will only recognize the beauty of a preset if you run the ‘right’ audio through it”

Uhbik includes a powerful preset browser. Click on the PRESETS button, then select the DIRECTORY tab if necessary. Folders will appear on the left, presets are in the center, and any details about the currently loaded preset will appear on the right.

After loading a preset (click on its name) you can step through all the others using the cursor keys.

If no presets appear in the central area, click on ‘Local’ or one of its subfolders. If you don’t see a ‘PRESET INFO’ label on the right, click on the [≡] button (top right) and select *Show Preset Info*.

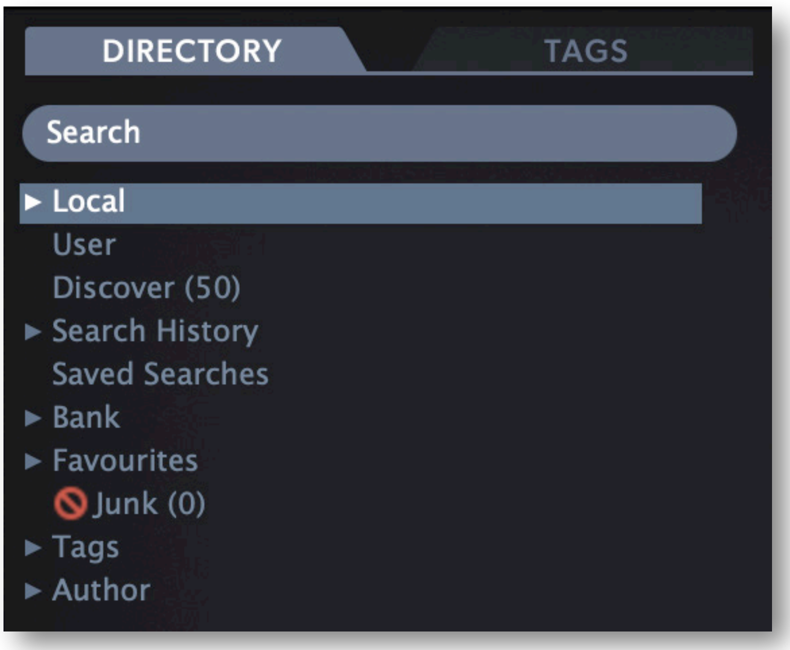
That’s all you really need to know to get started. For those who wish to dig deeper, however, the browser offers many features, including a powerful search engine. For details, read the rest of this chapter and/or watch our video at https://youtu.be/9A3nPN_Nn4M.

The default preset

When a new instance of Uhbik starts it checks whether the ‘Local’ root directory contains a preset called simply ‘default’, which is then loaded instead of the standard one. If you would like to change the default preset, make sure that the root of the *Local* folder is selected, and [SAVE] the one you want under the name ‘default’. Note: Even if it exists, ‘default’ won’t appear in the browser.

If a fresh instance of Uhbik is not loading your new default preset, it probably landed in the ‘User’ folder instead of ‘Local’ – see the preference Save Presets To.

Directory Panel



Local

Uhbik's factory presets are sorted into subfolders within 'Local'. To find the factory presets on your hard disk, right-click on 'Local' and select *reveal in Finder* (Mac) / *open in Explorer* (Windows).

Note: It is best not to touch the contents of 'Local', but to put your own creations as well as any third party soundsets in the 'User' folder (see below).

MIDI Programs

Up to 128 presets in this folder are loaded into memory when the first instance of Uhbik is launched. Once loaded, they can be selected using MIDI "Program Change" messages.

Note: Some hosts automatically route all incoming MIDI data to effect plug-ins, while others require manual setup. Please refer to your host application or DAW documentation for instructions.

Since MIDI Program presets are accessed in alphabetical order, it's best to rename them by adding a numerical prefix—for example, "000 rest-of-name" through "127 rest-of-name". Unlike regular presets, MIDI Programs cannot be added, removed, or renamed on the fly: Any changes will only take effect after the host application has been restarted..

MIDI Programs can include up to 127 sub-folders, each containing up to 128 presets. These can be selected using a MIDI 'Bank Select' message (CC#0) followed by a Program Change message. The main 'MIDI Programs' folder is bank 0, and sub-folders are assigned bank numbers in alphabetical order, starting from bank 1.

When Uhbik receives a MIDI Program Change message, it displays the bank and program numbers to the left of the preset name (e.g., "0:0" for the first preset in the first bank). Note that some hosts label the first bank or preset as "1" instead of the correct "0".

To avoid potential confusion, make sure there are no *junked* presets in the MIDI Programs folder. All files there are addressed—even hidden ones!

User

This is the best address for your own creations as well as presets from other sources. You can select the 'User' *folder* immediately before saving or set a global preference to always save to this folder (or one of its sub-folders) – see the preference Save Presets To.

Tip: To locate the 'User' folder on your computer, right-click on it and choose *Open in Finder* (Mac) or *Open in Explorer* (Windows).

Smart Folders

The other top-level folders don't actually contain files, but the results of querying a database. The content is therefore dynamic; it will change whenever the underlying data changes. **Caution**: When you delete files from smart folders, the referenced originals will be moved to the trash!

Discover

A random selection of presets. You can refresh the list by right-clicking the folder and choosing *Rebuild* from the bottom of the menu.

Search History

Open this folder to view results from your last 10 searches. To save a search permanently, right-click a search entry and choose *Save Search*... — it will be moved to the 'Saved Searches' folder (see below). To clear the history, right-click the 'Search History' folder and select *Clear*.

Saved Searches

The ‘Saved Searches’ folder contains a list of results saved from ‘Search History’ via right click (see above). To remove individual entries, right-click and select delete. Tip: Entries dragged from ‘Saved Searches’ and dropped onto real folders within ‘Local’ or ‘User’ will create a folder containing real copies of all presets!

Favourites

8 smart folders, one for each Favourite colour (1-8). See *Presets Context Menu* on the next page. Only one Favourite colour/number can be set per preset. Presets dropped onto one of the ‘Favourites’ folders will be marked as such.

The Favourite status can be removed from all presets of one particular colour / index by right-clicking on the ‘Favourite’ folder and selecting *Remove All Favourite (n) Marks*.

Junk

A smart folder pointing to all junked presets. See *Presets Context Menu* on the next page. Presets dropped here will disappear from the rest of the browser unless made visible (see *Show Junk* in the Presets context menu).

Tags

Smart folders for each *Category*, *Application* and *Character* tag. Presets dropped onto these folders will inherit the corresponding tag. Presets dropped onto the ‘Untagged’ smart folder will have all *Category*, *Application* and *Character* tags removed.

Author

Smart folders for each preset author. Tip: Instead of signing your creations individually you could sign just one of them then drag & drop others onto your new author folder. As the process cannot be undone, please use this feature with caution!

Directory Context Menu

Right-clicking on any folder within ‘Local’ or ‘User’ will open a menu:

Refresh
Create New…
Rename…
Open in Finder

Move to Trash

On Open Expand to >
Hidden Folders >

Refresh: Update the browser content. Necessary after you have moved, added, removed or renamed any folders or presets using Explorer / Finder.

Create New… Insert an empty subdirectory.

Rename… Edit a folder’s name.

Reveal in Finder / Open in Explorer: Open a system window showing the clicked folder. After adding, removing or renaming preset files or folders there, please remember to *refresh*.

Move to Trash: Moves the selected folder to the system trash. If you right click on the Junk folder, this entry will be replaced by ‘Remove All Junk Marks’. If you right click on a Bank smart folder, it will be replaced by ‘Remove Presets from Bank’ (see Smart Folders above).

On Open Expand to: These options determine how deeply the browser will open subdirectories when the GUI is opened again or the refresh function is used.

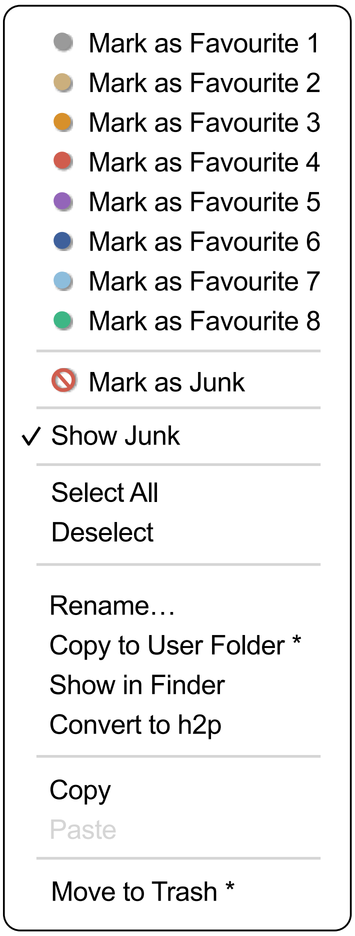
Hidden Folders: Select the smart folders that you do not wish to see in the directory.

Presets Panel

The central, unlabelled area of the browser is where you click to load presets...

Presets Context Menu

Right-click to open a menu containing functions which can be applied to individual presets.



Mark as Favourite: Choose one of eight *Favourite* marks. The selected entry will be replaced with unmark as favourite.

Mark as Junk: Instead of deleting any unloved presets, you can mark them as 'junk' so that they disappear from the browser...

Show Junk: Activate this option to display junked files (see above) instead, but mark them with a STOP symbol.

Select All, Deselect: See Multiple Selection on the next page.

Rename: Change the name of a preset.

Duplicate / Copy to User Folder: The function here depends on the status of the preference Save Presets To and whether the source presets are in the *Local* or *User* folder. Selected presets are copied with an appended index (like the 'Auto Versioning' preference).

Show in Finder / Open in Explorer: Opens a system window for the selected preset. Remember to *Refresh* the directory after adding, removing or renaming any preset files there!

Convert to native / h2p / h2p extended: Converts the selected preset(s) to the format previously selected via right-click on the SAVE button.

Copy / Paste: Clipboard functions. Individual or multiple presets can be copied then pasted elsewhere, even between the Uhbik browser and system windows (Finder, Explorer).

Move to Trash / Recycle Bin: This function moves all selected preset(s) to the system trash, so please be careful. Note that it also works on files in any smart folder, meaning the originals will be moved to the trash.

Restore

While in the browser you can audition as many presets as you like without losing track of the one that was previously loaded: Clicking *Restore* will always get you back to where you started.

Multiple Selection, Drag & Drop

A block of adjacent presets can be selected via SHIFT+click, and individual presets can be added to the selection via *command+click* (Mac) / *control+click* (Win). Presets can be moved to a different folder via drag & drop. Use SHIFT etc. to highlight the files you want to move, drag them from the files area and drop them onto the target folder.

To clear the selection either click on a preset that is not currently selected, or choose *Deselect* from the context menu.

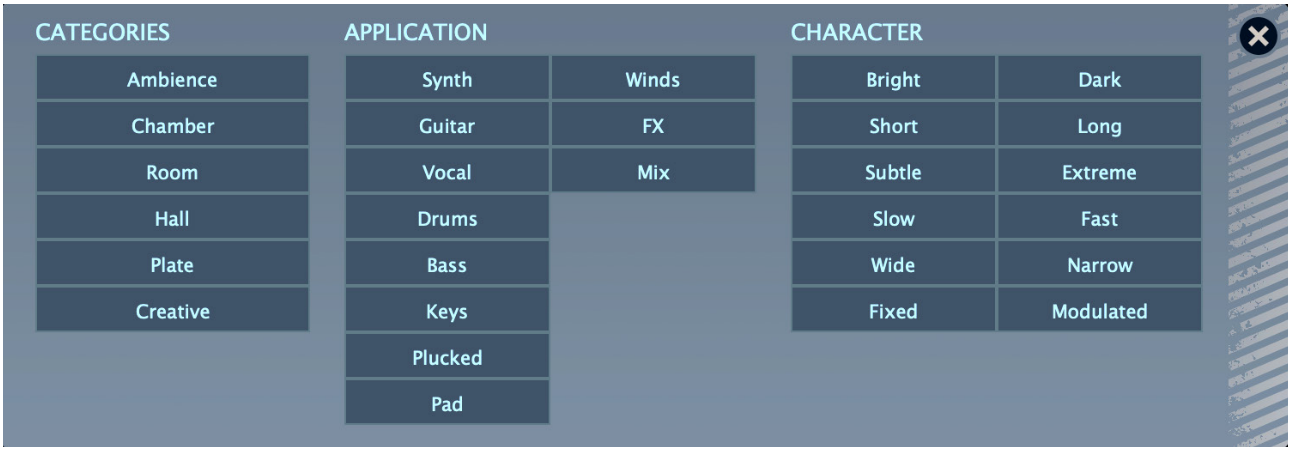
Preset Tagging

“Tags” are metadata attributes you can add to presets so that they can be found more easily.

IMPORTANT: Clicking on [SAVE] isn’t required, as tags are updated immediately. One obvious advantage is that presets don’t need to be saved every time you edit tags. The main disadvantage is that you should only edit tags after saving your preset: If you edited tags while in the process of creating a new version of something, you would also be changing the tags in the original preset!

The Tagging Window

Right-click on the [**Save**] button and select *Tag this Patch*:



The Uhbik Ambience tagging window

In Uhbik, the *Category* tags describe a preset according to the type of effect, *Application* tags describe typical usage, while *Character* tags are pairs of (more or less) opposite attributes from which you can select just one.

Tagging via PRESET INFO

In the PRESET INFO panel, right-click on CATEGORIES, APPLICATION or CHARACTER and select or unselect tags from the menu. Note: This method only works for individual presets.

The function *Create Search from Tags* looks for presets with an **identical** set of tags.

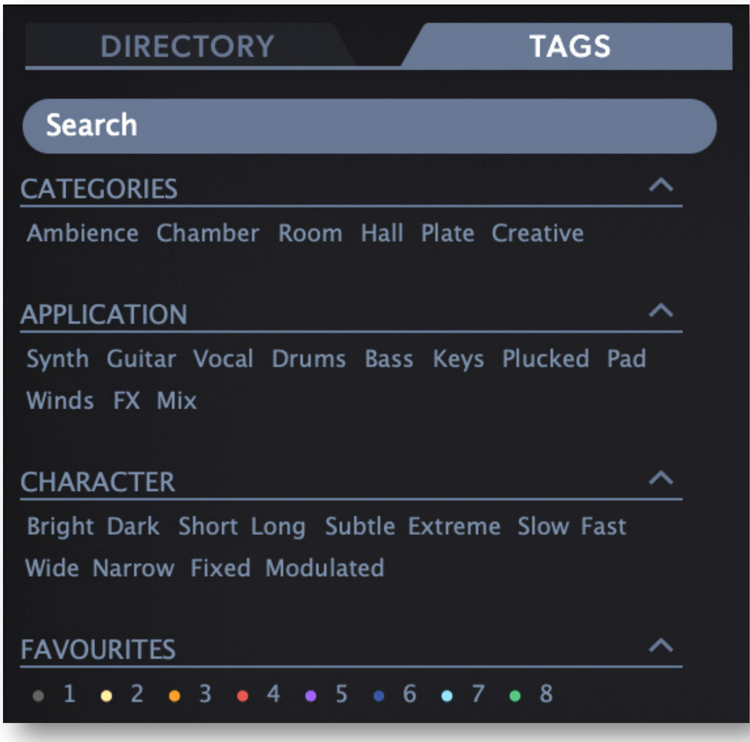
Tagging via the Tags smart folder

You can tag presets by drag & drop onto one of the *Tags* smart folders. To remove all tags, drag presets onto the ‘Tags/Untagged/‘ smart folder.

Search Functions

Search By Tags

In the preset browser, click on the [TAGS] tab. The buttons let you set up search criteria according to existing tags with just a few mouse clicks. Here's the TAGS panel for Uhbik Ambience:



There are four sets of buttons: The first three correspond to the tags in the tagging window (see the previous page), while the bottom row lets you find any presets tagged as *Favourites*. Clicking on the [^] icon to the right of each label hides the options for that set of tags.

A practical example for Uhbik Delay

Click on the [DIRECTORY] tab, right-click on the 'Search History' folder and select *Clear*. Double-click on 'Local' to restrict the scope to that folder (presets in the 'User' folder will not appear in the results). The selected path appears below the Search field. To exit, click on the [^] symbol.

Click on the [TAGS] tab and select the [Digital] and [Comb] categories. Presets tagged with either will appear in the presets panel. Click on the DIRECTORY tab again: "#Delay:Digital #Filter:Comb" appears in the Search field as well as in 'Search History'. Adventurous souls can try editing the contents of the Search field now – the results will be updated accordingly.

Note: Unlike selecting multiple Categories tags, which expands the scope of the search, selecting Application, Character or Favourites tags restricts the scope for fewer, better hits.

Search by Text

The Search field lets you find presets according to a string of text. Here's an easy example: If you remember that the preset you're looking for has the word "space" in either its name or the description, simply enter "space" into the Search field and hit Return.

The search routine normally looks into the preset name, the author, the DESCRIPTION and USAGE (see the PRESET INFO panel). Searches are not case-sensitive, and quotes are not required unless you need to include spaces between multiple words.

To limit the search to a particular path, double click on that folder. The path will appear below the Search field. The [^] button to the left moves the search path up one level, while the [X] button to the right resets the search path to the 'Local' or 'User' root directory.

Try it: Click on Search, enter three or four characters then hit Return. For instance, “sta” will find all files containing the text string “sta” (e.g. “instant” or “custard”). Entering "star wars" (including the quotes!) would find e.g. “Battlestar Warship”, if such a preset existed.

Scope

You can limit the scope of the search to just the preset name or specific parts of PRESET INFO by using *name* (preset name), *author*, *desc* (*description*) or *use* (*usage*) followed by a colon. For instance, “author:the” finds all presets by sound designers whose author names contain “the”. Similarly, “desc:space” will find all presets with the word “space” in the description.

Logic

These logical operators can be used between text strings, but not between tags:

AND requires that presets contain both words. It can be written explicitly or simply left out. For example, “star AND wars” or “star wars” will find presets that contain both “star” and “wars”.

OR means that presets can contain just one of the words… or both. For example, “star OR wars” will find presets that contain “star” as well as presets that contain “wars”.

NOT excludes presets containing the specified word. To find all presets that do contain “star” but don’t contain “wars”, enter “star NOT wars”.

Including Tags

In the current version of the browser, text items must appear before any tags. For technical reasons, tags appear in the form #type:category (the *type* is invisible in the TAGS panel).

Tags can be entered into the Search field if preceded by a ‘#’. For example, “name:int #Delay” will find all presets with "int" as part of the name that are also tagged with the [Delay] category.

Please note that *Search by Text* remains work in progress. We hope to remove any remaining inconsistencies and improve the functionality in future versions!

Configuration

The cogwheel gives you access to the global configuration pages. Here, you can adjust the window size and brightness, as well as control Uhbik parameters remotely via MIDI CC. Click and select MIDI Learn [L], MIDI Table [≡] or Preferences [tools]…

MIDI Remote Control

MIDI assignments are truly global. They apply not only to all instances in the current project, but to ALL instances in ALL your projects. For details about how to route MIDI into effect plugins, please refer to the documentation of your host application!

MIDI Learn

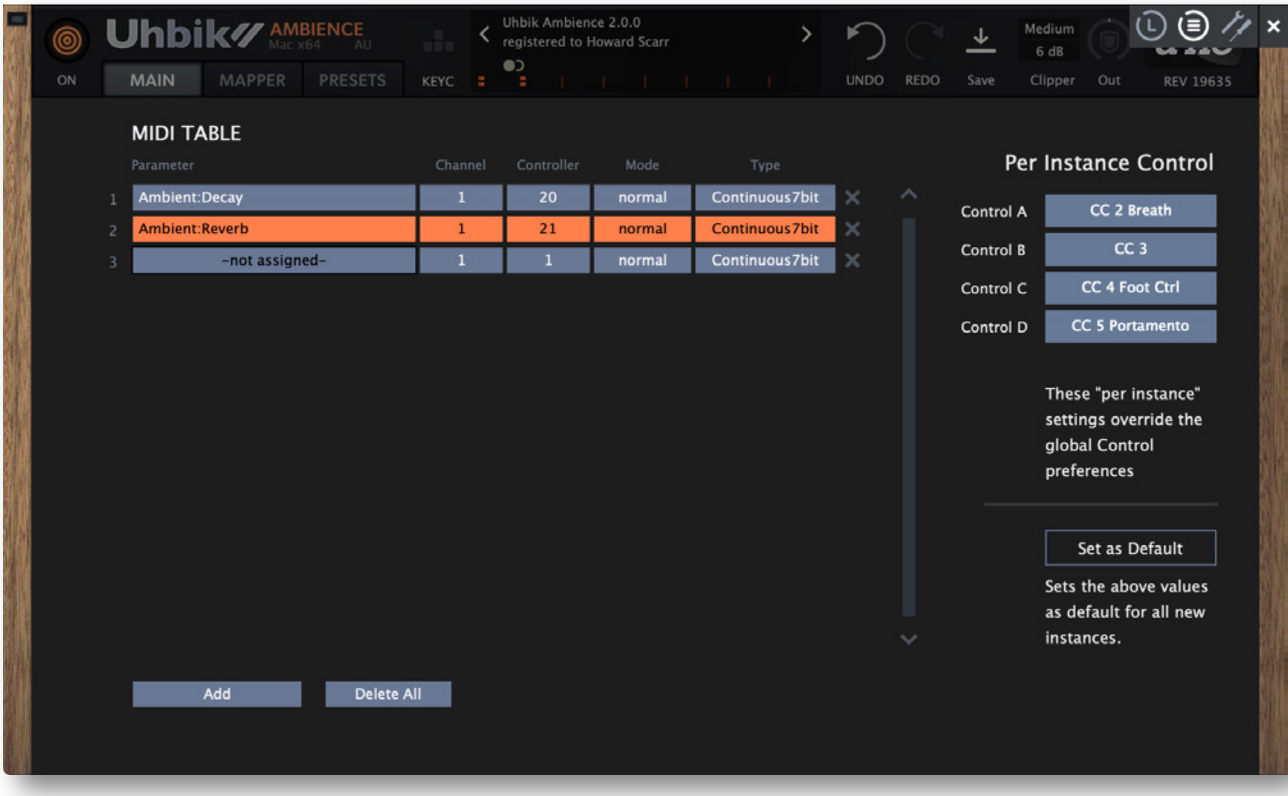
This page lets you assign MIDI CC ('control change') to Uhbik parameters. The CC data can be generated by hardware knobs / sliders or tracks in the host application. For details about how to route MIDI data into effect plug-ins, please refer to the documentation of your host application. For Uhbik Ambience it will look something like this:

The window shows all MIDI-learnable elements as selectable outlines. Those that are already assigned will appear filled (like DECAY in the above image), and the currently active control i.e. the one ready to be MIDI-learned is highlighted (like the MIX control here).

If your setup allows, try this: While in the MIDI Learn page, click on the REVERB knob then send Uhbik some MIDI CC to make the assignment. Test that the knob can now be remote-controlled. Finally, double-click on REVERB to remove the assignment again.

MIDI Table

The MIDI Table page lets you review and edit the MIDI assignments created using MIDI Learn (see the previous page), or add more assignments. If a few assignments have already been made, it will look something like this:



Parameter

This field shows the assigned target – click to select a different one. At the bottom of the list is an experimental feature you should try: Select *Last Clicked Control*, enter the number of an unused controller your hardware can send and exit the configuration pages. The most recently clicked knob or switch will now respond to that CC. The *Fine* option is similar, but with a reduced range.

Channel / Controller

These fields specify the MIDI channel (1 to 16) and CC number (0-127).:

Mode

Specifies the range and/or resolution:

normal..........................full range, continuous
integer ....full range, whole numbers only
fine.................................0.01 steps between the two integers closest to the current value

Type

Specifies the kind of hardware used. If in doubt, set *Continuous7bit* here.

Encoder127 ....'relative mode' endless rotary controls that repeatedly send the CC value 1 when turned in the positive direction, or 127 (interpreted as -1) when turned in the negative direction

Encoder64 ....'relative mode' endless rotary controls that repeatedly send the CC value 65 when turned up, or 63 when turned down

Continuous7bit...........7-bit MIDI CC (normal resolution, absolute values, common)

Continuous14bit ....14-bit MIDI CC (high resolution, absolute values, less common)

Adding more assignments

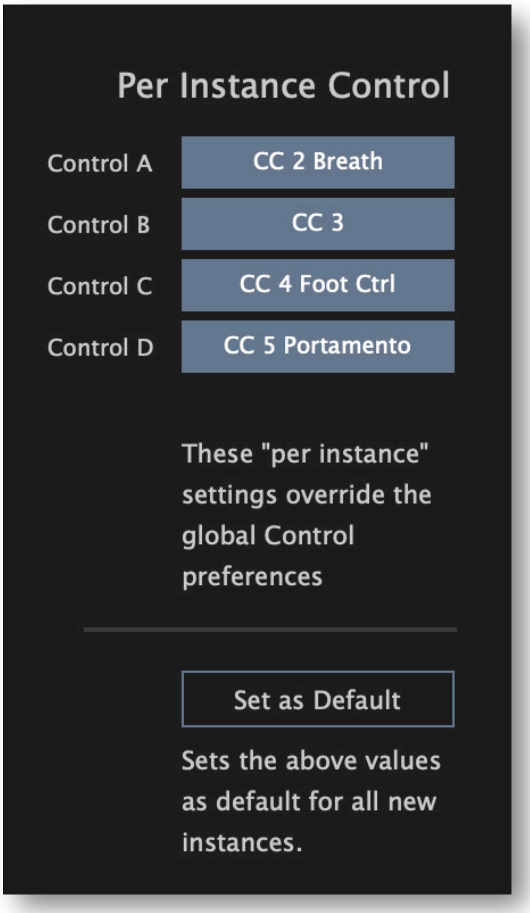
You can either MIDI-learn them as described above, or click on the [**Add**] button at the bottom of the window then select a Parameter, Channel etc. from the lists.

Removing assignments

Individual assignments can be removed by clicking on the small [**x**] to the right of each line. To remove all assignments at once, click on the [**Delete All**] button at the bottom of the window.

Per Instance Control

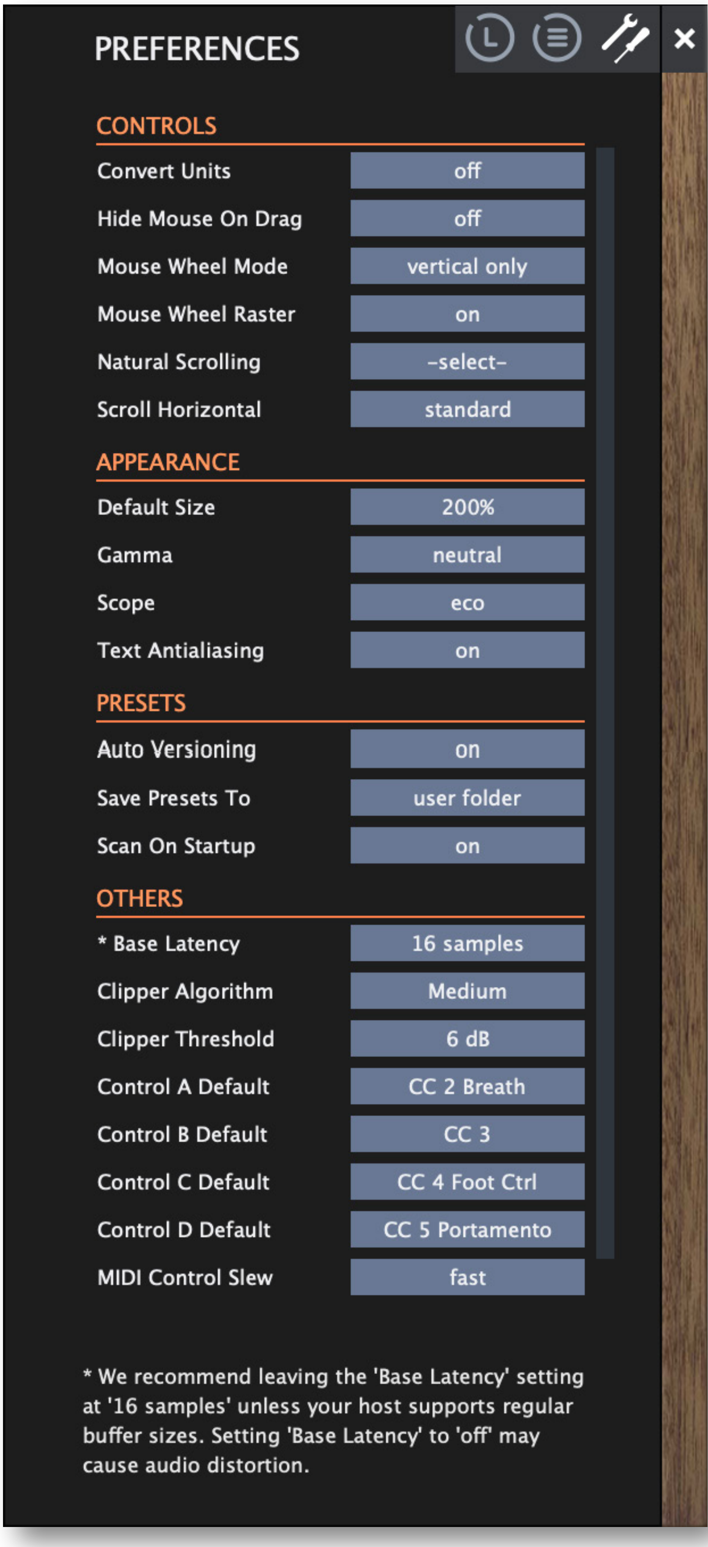
Local versions of the Control A/B preferences (see below).



The [**Set as Default**] button copies the Control A / B / C / D settings to the global preferences...

Preferences

Click on the [Tools| button to open the 'Preferences' page, where you can set several global defaults to suit your mouse and monitor. The image here shows all options (you will have to scroll down to see the very bottom of the panel) in the Apple Mac versions:



CONTROLS

Convert Units

How values appear in the data display:

off ....parameter value only (usually 0 to 100)
display both ....Decibels / Hertz as well as the parameter value
converted only..............Decibels / Herz only

Note: Certain parameters e.g. envelope Gain always appear with units, whatever is selected here.

Hide Mouse On Drag

On: The mouse pointer will disappear while you are adjusting a parameter, and reappear in the same position afterwards.

Mouse Wheel Mode

Mainly for browser navigation using a 2D mouse wheel or touchpad. Work in progress!

auto-detect ....tries to determine whether your input device has a horizontal mouse wheel, then respond accordingly
vertical only ....responds to vertical motion only—the default for standard mouse
vertical & horizontal ......responds to motion in either direction

Mouse Wheel Raster

If your mouse wheel is rastered (you can feel it clicking slightly as you roll the wheel), set this to *on* so that each click will increment / decrement values in sensible steps.

Natural Scrolling

Mac: Inverts the mouse wheel if 'Natural scrolling' is activated in System Settings / Mouse.

Windows: Inverts touchpad scrolling direction if 1) the OS includes a Natural Scrolling setting and 2) this is activated and set to '*Downwards motion scrolls up*'. Does not affect the mouse wheel.

Scroll Horizontal

Direction of scrolling within the presets panel.

APPEARANCE

Default Size

Sets the default GUI size for each new instance. You can temporarily change the GUI size without entering the Preferences – simply right-click in the background.

Gamma

Determines GUI brightness.

Scope

Sets a default effect for the oscilloscope (Scope).

Text Antialiasing

Switches the smoothing of labels and values on / off. Note: Only in certain special cases will switching it off improve readability.

PRESETS

Auto-Versioning

If '*on*', an index is automatically appended to the name and incremented each time it is saved. Saving 'Space' 3 times in a row would thus give you 3 files: 'Space', 'Space 2' and 'Space 3'.

Save Presets To

Choose the *user folder* option if you always want saved presets to land in the 'User' folder.

Scan On Startup

Determines whether the preset library should be scanned and the database recreated when the first instance of Uhbik is started, e.g. every time you reopen a project.

OTHERS

Base Latency

If you are sure that your audio system—hardware as well as software—uses buffers that are all multiples of 16 samples in size (refer to the respective documentation), you can safely disable this. Otherwise leave it set to the default '16 samples' to prevent crackles.

Note that the new Base Latency only takes effect when the host allows, e.g. on playback or after the sample rate is switched. Reloading Uhbik will always update Base Latency.

ABOUT BUFFERS

Internally, Uhbik processes audio in chunks of n x 16 samples. The 'block processing' method reduces the CPU load and memory usage of all our plug-ins.

For example. if the number of samples to be processed is 41, Uhbik will process the first 32 and keep the remaining 9 in a buffer (16 samples is enough). Those 9 samples are then processed at the start of the next call... and so on.

The extra buffer is only necessary if the host application or audio driver processes 'unusual' audio buffer sizes. Many hosts process buffers of 64, 128, 256 or 512 samples (all multiples of 16), in which case you could try switching off Base Latency so that Uhbik can work latency-free.

Clipper Algorithm / Threshold

Default algorithm and threshold for the protective output clipper. See Control Bar.

Control A / B / C / D Default

Default MIDI CC numbers for the user-definable sources..

MIDI Control Slew

The strength of parameter smoothing for all performance controls—pitch bend, modulation wheel, Control A/B/C/D and Pressure. With MIDI Control Slew 'off', Uhbik is more responsive to modulation wheel data (for instance), but the result of rapid modulation can sound rather grainy. The default 'Fast' setting is a good compromise between speed and smoothness.

The 'Slow' option is adaptive: Whenever incoming control data jumps immediately between values that are further apart, no slew is applied.