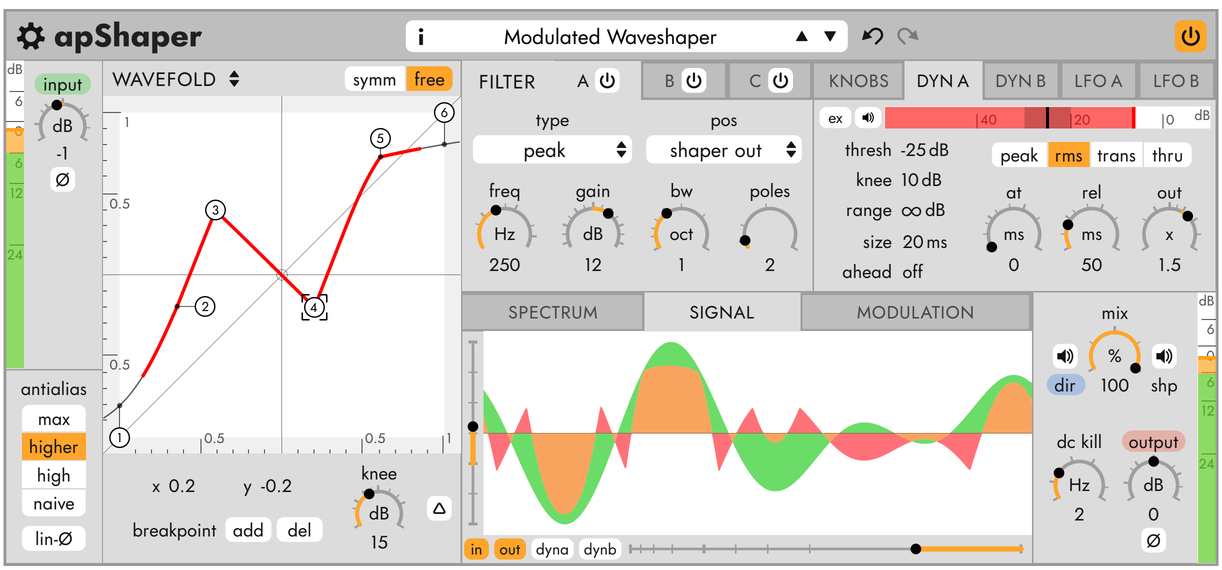
apulSoft apShaper v1.2.5 Manual

harmonic distortion audio plugin
(VST/VST3/AU/AAX)



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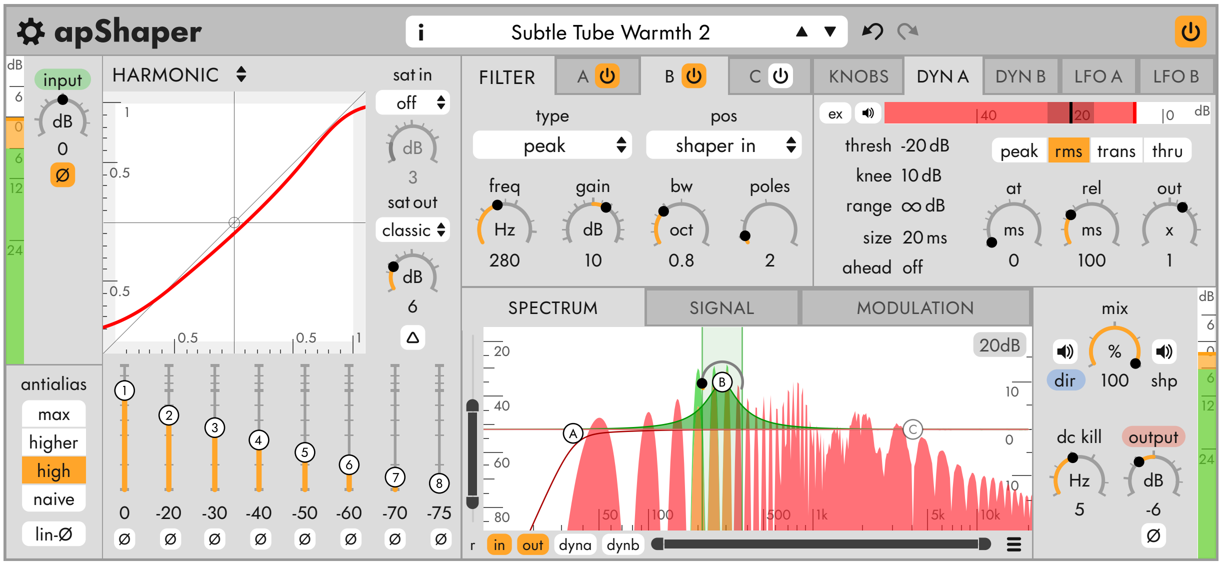
VST plugin technology by Steinberg.
AU plugin Technology by Apple.
AAX plugin Technology by Avid.
Manual written with LaTeX on May 5, 2025

Introduction



apulSoft apShaper is an audio insert effect plugin to add harmonic content to audio signals.

It combines a waveshaper with three filter bands and various modulation options - two LFOs, two dynamics detectors, and three user-configurable knobs that can be automated (as meta parameters).

apShaper comes with five shaper engines ranging from adding harmonic distortion to freely configurable polygonal wave folding.

The waveshaper in apShaper uses sophisticated algorithms to reduce aliasing. It approximates continous-time (idealized analog) waveshaping using integration. This reduces digital artifacts more than traditional oversampling could, especially if the transfer curve is not smooth (wave folding/bit crushing). The process only adds 7 samples of latency to the signal which means apShaper can be used for real-time performance.

The three filter bands offer high-quality filtering without high-frequency squeezing. The variable-slope low-pass filter type allows smooth slopes not possible with traditional filtering - ideal to gently shape signals with large amounts of distortion applied. Multiple tone control types modeled after classic circuits enable classic bass/mid/treble tone shaping.

Contents

1 End User License Agreement

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2 System Requirements

- macOS
  - macOS 10.11 or newer on an Intel or Apple Silicon CPU (64-bit only).
  - A host application compatible with VST, VST3, AU or AAX plugins.
  - Pro Tools (AAX): Version 11 or newer. Pro Tools on Apple Silicon is natively supported.
- Windows
  - Windows 10 or newer. Both 32-bit and 64-bit versions of Windows are supported.
  - A CPU with SSE2 instruction support.
  - A host application compatible with VST, VST3 or AAX plugins running in 32-bit or 64-bit mode.
  - Pro Tools (AAX): Version 10.3.5 or newer.
  - An application to view pdf files to read this manual.

3 Installation

- macOS
  - Quit all plugin host applications.
  - Double-click **apshaper-mac-(..).pkg**.
  - Follow the OSX installation procedure.
  - Open a host and create an instance of apShaper in a plugin slot.
  - The apShaper window shows a welcome screen with the options to run the plugin in demo mode or to buy or enter license information.
- Windows
  - Quit all plugin host applications.
  - Double-click the **apshaper-win-(..).msi** to start the installation. On newer versions of Windows, it may be necessary to confirm the launch because of user access management.
  - Follow the installation procedure. During the install, you have the option to set the path to the apShaper data folder. That is where settings, presets and the manual will be installed.
  - The installer provides the option to select destination folders for VST2 plugins running as 32-bit and 64-bit binaries.
  - Open a host and create an instance of apShaper in a plugin slot.
  - The apShaper window shows a welcome screen with the options to run the plugin in demo mode or to buy or enter license information.

4 Interface Overview

The apShaper user interface consists of a global top bar and a large bottom section.

Most of the user interface controls have **tooltips** that get displayed if the mouse hovers over them for some time (with the tooltips preferences setting enabled). In order to keep the manual brief, the tooltip information is not repeated in the manual. If the function of a control is unclear, use the tooltips.

Many controls on the user interface can be dragged with the mouse to change values.

- If the Shift key is held down the values snap to predefined markers at round values during dragging.
- Holding Ctrl switches dragging to be scaled by 1/20 for fine adjustments.
- Double-clicking most of the value-based controls opens a popup editor to enter a new value with the keyboard.

4.1 Title Bar

Click the **gear button** on the left or the plugin title to open the preferences/information dialog.

On the top right corner, the **enable processing** button allows to bypass audio processing. Modern hosts supporting AU/AAX or VST3 plugin formats hook this up to their plugin bypassing system.

The **license** section to the left of the enable button changes depending on your license status. In demo the →full version button is displayed. It brings up a dialog with options to buy an apShaper license online, to enter the purchased license information or to keep running in demo mode.

4.2 Preset Selection/Menu/Undo/Redo

The white box shows the name of the current preset. Click to open the preset menu and double-click to edit the name. The "**i**" button on the left opens the preset description popup/editor.
On the right side are up/down buttons to cycle through presets. Depending on the "cycle all presets" setting, these go through all presets or stay inside the current folder.

The preset menu (opened by clicking the preset name) lists all available presets with folders as they are on disk. Below the presets the following entries are available:

save current preset.. The current state of the plugin is added to the preset menu. In the prompt that pops up, the preset name can be edited and folder paths can be added which automatically creates folders on disk if necessary. Presets in the menu can be overwritten by using the same name/path.

import preset.. Load a preset from a .shprpreset file anywhere on the local filesystem.

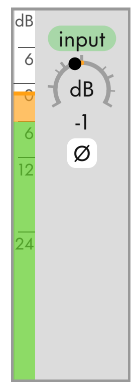
export current preset.. Store the current state of the plugin as a .shprpreset to any location on your local filesystem.

manage preset folder in finder/explorer.. This opens the filesystem folder that contains the presets shown in the menu. Normal file operations can be used to restructure this folder which will be reflected by the presets menu.

To the right of the white preset box are the **Undo** and the **Redo** buttons. apShaper supports unlimited undo.

Note: Saving and importing of presets is only available in the full version of apShaper.

5 Input

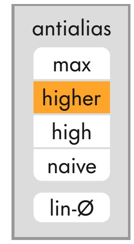


The apShaper input section consists of an input **gain** control, an input **meter** and a **phase** switch button.

The input gain knob can be used to adjust the input signal level. This is done after dynamics detection, but before any other processing. For best results, the meter should reach the yellow range (-6dB - 0dB), but not enter the red range (>0dB).
As long as the signal stays inside the red part of the meter, calculations are executed without clipping, but unintended things might happen. 0dB means a signal value of 1.0. The transfer functions of the shaper engines are designed for signals in the -1↔1 range.

The **phase** toggle button below the gain knob inverts the signal on input. This makes a big difference if both the incoming signal and the transfer function are asymmetrical. Most natural signals have some degree of asymmetry due to containing even overtones.

6 Antialiasing



apShaper has multiple levels of aliasing suppression. They differ by the amount of digital artifact reduction, CPU usage and oversampling ratio. Oversampling leads to a falloff of the highest frequencies very close to half the samplerate.
Oversampling can be minimal phase (for the lowest latency) or linear phase (for a more linear phase response and an untouched dry channel).
Note: User filters will still change the overall phase response.

quality	oversampling	integration order	∼CPU usage	aliasing suppression
max	8x	2nd	12x	awesome
higher	4x	2nd	6x	great
high	2x	2nd	3x	good
naive	none	none	1x	none at all

The **naive** quality setting is only recommended in case aliasing is desired or for comparison with the better modes.

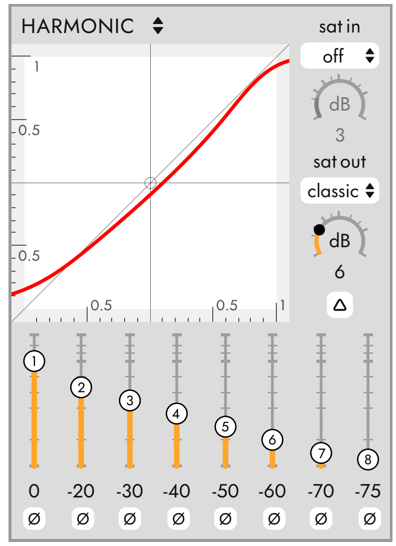
7 Shaper Engines

Clicking on the shaper engine title shows a popup menu with the available types. apShaper provides five different shaper engines and only one can be active at any time. To combine them, create multiple instances of apShaper.

The effect the algorithms have on the audio signal is visualized by their **transfer graphs** that are prominently features on each editor. The values of the incoming signal are represented by the x/horizontal position and the resulting output values by the y/vertical position. The hinted diagonal line from bottom left to top right shows the transfer curve that would do nothing. The more the curve moves away from this line, the more change is applied.

While audio is being processed by apShaper, a section of the transfer curve is drawn in red. This shows the signal range of the processed audio. Ideally the whole width of the display is used occasionally without overshooting too much. The input gain knob provides range adjustment.

7.1 Harmonic

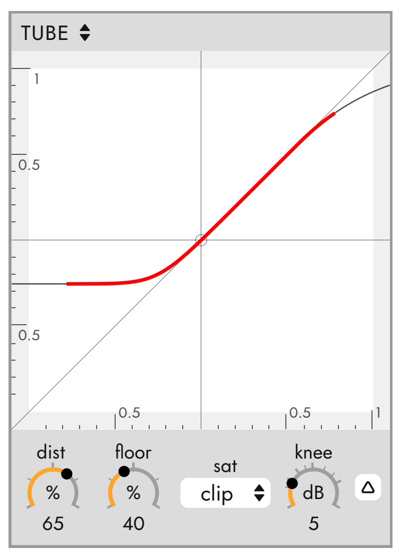


The **harmonic** shaper engine adds low order harmonics to the signal. It uses summed-up Chebyshev polynomials. The intensity of the first eight harmonics can be adjusted using dB sliders and the phase of each harmonic can be switched. For a sinus wave at full gain, the result is an exact sum of the sines with the set intensities. Once multiple input frequencies are present, intermodulation distortion is produced as well. The output is then a sum of all the multiples of the single frequencies as well as all difference between all frequencies present.

The harmonic shaper engine has two saturation stages **sat in** and **sat out**. Higher-order Chebyshev polynomials have very steep slopes outside the -1↔1 range. The input clipper limits the incoming signal so the full curve range can be used.
If multiple harmonics are added with high levels, the signal can get very loud. The output saturation allows to limit the maximum range of the output. Both stages produce extra harmonics to the ones set up with the sliders.

The **delta** mode (∆ button) subtracts the linear portion from the shaper output. For the harmonic engine this means the linear parts of all odd harmonics are removed from the output - completely reverting slider 1. This enables checking just the added frequencies from higher order parts.
With the right gear, the harmonic distortion of analog devices can be measured and these measurements can be entered in the shaper engine to approximate their harmonic effect.

7.2 Tube



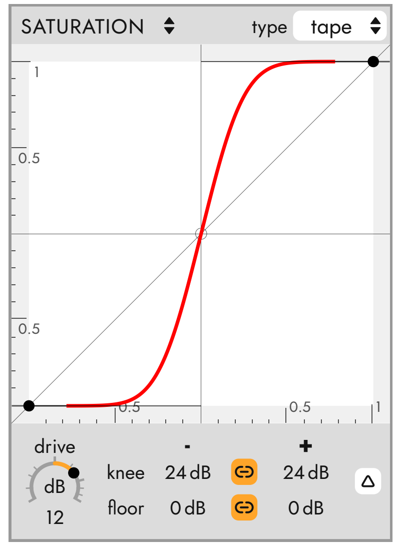
The **Tube** shaper engine is based on an idealised tube transfer function. It produces asymmetrical waveforms. It can go from gently adding harmonics all the way to a half-wave rectifier at extreme settings.

An output saturation circuit with softknee can be enabled to limit high output values. Using this, the effect goes from tube-like added even harmonic at low levels to pure odd harmonics at high levels.

As the tube-like transfer curve is asymmetrical, the tube engine leads to DC offset. Use the built-in DC kill filter at the output section to remove that from the output signal to avoid clicks.

The **delta** mode (∆ button) subtracts the shaper input from the output to get just the nonlinear part. It produces the difference to the diagonal guideline.

7.3 Saturation



Saturation gradually reduces the signal level on the positive and negative sides using various formulas.
The **drive** parameter boosts the signal before applying the saturation. The **knee** ranges determine the levels where reduction starts. The **floor** values set the maximum value the output can reach. Both knee and floor have settings for both the negative/left and positive/right side of the transfer function. The **link** buttons in the center couple positive and negative sides.
The **delta** mode (Δ button) subtracts the shaper input with drive applied from the output to get just the nonlinear parts. For the saturation engine these are the harmonic distortion products created by the signal going into the knee region.

As long as the settings for positive and negative sides are the same, the saturation engine will produce odd harmonics only. When unlinked, asymmetrical functions become possible that produce even harmonics as well and can lead to DC offset.
Using ∞ knee, zero floor and zero drive leads to the pure algorithms over the full signal range.

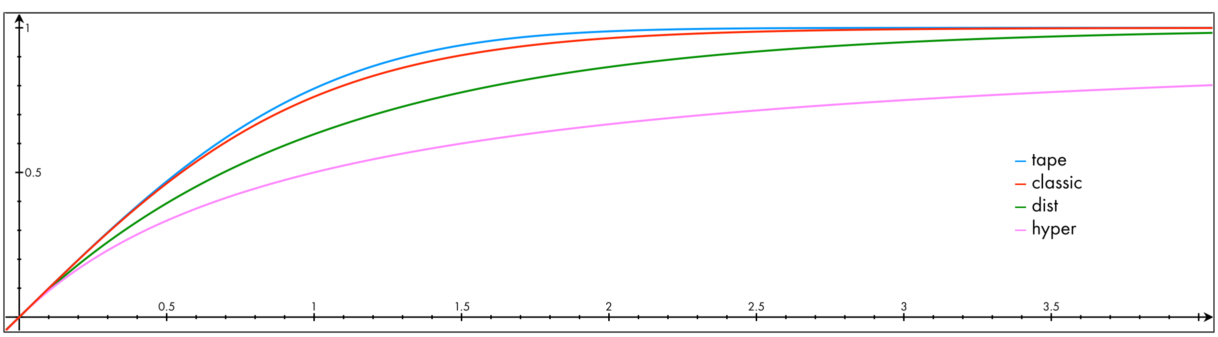
Using the same parameters, later saturation functions will produce more and higher overtones.

tape Uses the error function erf $\left(x \cdot \frac{\sqrt{\pi}}{2}\right)$ to calculate the curved part of the transfer function. This is similar to how analog tape saturation behaves.

classic Based on the tangens hyperbolicus tanh (x). This is the classic function used to calculate circuit saturation.

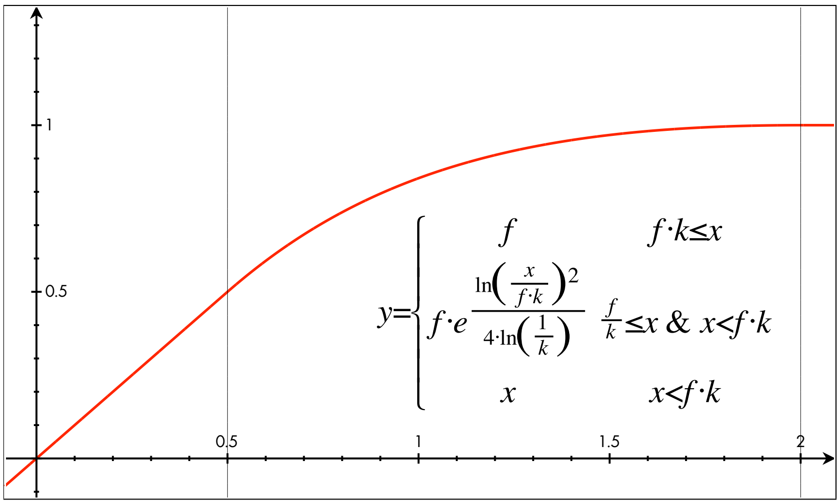
dist Distortion simulation based on the transfer function $1 - e^{-x}$.

hyper Hyperbola-based distortion using the function $\frac{x}{x+1}$.



7.3.1 Clip Function

The **clip** function works differently than the other four algorithms.
It start with a straight section, then has a logarithmic knee and ends in a constant value.
While the other four functions approach 1 at infinity, the clip function stops growing at floor + knee. If knee is set to 6dB, then everything between -6dB and +6dB is smoothly condensed into -6dB to 0dB, while higher values are clipped to 0dB.



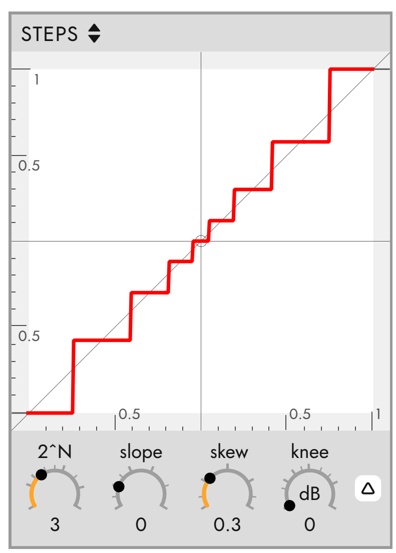
Clip curve for f(floor) = 1(0db) and k(knee) = 2(6.02dB)

The clip function has a maximum knee value of 48dB. It cannot go to infinity as that would result in silence (and numerical instability).

7.4 Steps

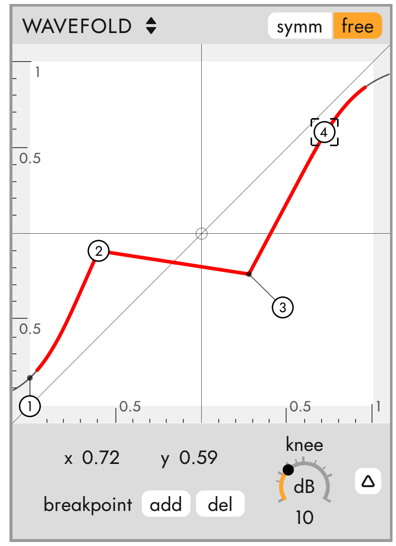
The steps engine provides bitcrushing. The number of bits can be adjusted continuously using the **2^N** knob with N being the number of bits.

The **slope** knob defines the steepness between the steps. If the slope is zero, the result is classic bitcrushing. Other values make the effect more or less pronounced as they add/subtract the original signal to/from the result.
The **skew** parameter adjusts where the steps are close/wide. The lower the value the more concentrated the steps are near zero. 0.5 leads to all steps having the same size.
The **knee** parameter adjusts the classic saturation that is applied after the steps function. Saturation is easily reached when low 2^N and high slope values are used.



The **delta** mode (△ button) subtracts the shaper input from the output to get just the nonlinear part. It produces the difference to the diagonal guideline.

7.5 Wavefold



The wavefold engine provides polygonal wavefolding using user-definable breakpoints. It can be run in symmetrical/**symm** and asymmetrical/**free** modes. The symmetrical mode produces odd harmonics only. When it is enabled, only the right half of the graphical display is active for editing, the left side is just a mirror image.
The asymmetrical mode often leads to DC offset which has to be filtered using the DC kill filter of the output stage.

The wavefold engine supports up to 8 breakpoints. Add new points using the **add** button or by double-clicking the desired location on the transfer graph. Delete them by double-clicking the numbered handles or using the **del** button once a breakpoint has been selected.

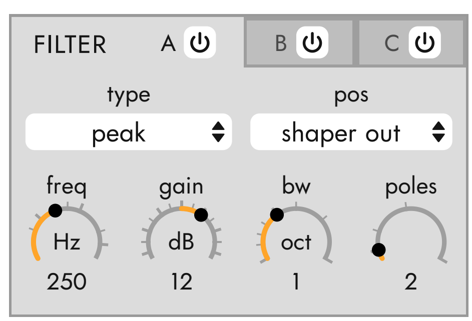
To select a breakpoint, click the numbered handle. The x/y values show up under the graph and can be dragged or edited using the keyboard after a double-click.

The **knee** knob controls the softness of clipping applied. The clipping prevents the transfer function from leaving the -1↔1 range.

Due to the clipping or if modulation rules control breakpoint positions, the breakpoints can move away from the control handles. In that case, a black line leads from the handle to a little black circle indicating the position currently used for processing. To adjust the point, still use the main handles.

The **delta** mode (△ button) subtracts the shaper input from the output to get just the nonlinear part. It produces the difference to the diagonal guideline.

8 Filter Section



apShaper has three filter bands. They are independent, but all have the same configuration options. The three filter bands can be individually turned on and off by toggling the **enable** buttons next to the **A, B & C** labels.
Filter settings depend on filter type and can be tweaked using the four knobs or directly on top of the frequency spectrum view. For more information, check the **spectrum display chapter**.
Filter **type** and **position** can be selected using popup buttons.

8.1 Filter Types

low pass Cuts frequencies above a cutoff value and lets low frequencies through. The gain value adds resonance to the lowpass which boosts/attenuates frequencies around the cutoff frequency.

high pass Cuts frequencies below a cutoff value and lets high frequencies through. The gain value adds resonance to the high pass which boosts/attenuates frequencies around the cutoff frequency.

peak Boosts/attenuates the frequency band around a center frequency. It has an adjustable bandwidth in octaves that is defined by half the gain-value. Higher pole versions lead to a flat gain plateau around the center frequency.

band stop Cuts out a band of frequencies completely. Bandwidth defines the -3dB points. This type has no gain parameter.

band pass High and low frequencies outside the bandwidth area are removed from the signal and gain is applied. The bandwidth spans between the two -3dB points.

low shelf Low shelf filters boost or cut low frequencies by the amount set up by the gain parameter. There is a transition area which width depends on the number of poles used. The frequency parameter defines the middle of the transition region where half the gain is applied.

high shelf High shelf filters boost or cut high frequencies by the amount set up by the gain parameter. There is a transition area which width depends on the number of poles used. The frequency parameter defines the middle of the transition region where half the gain is applied.

tilt shelf A tilt shelf filter boosts or reduced the frequencies above the center frequency by a set amount while applying the opposite change below the center frequency. Higher the poles count lead to steeper transition regions.

low pass variable slope The **low pass vs** cuts frequencies above the cutoff value like the regular low pass and additionally allows to adjust the frequency/octave slope to any value, breaking the usual limitations of pole/zero-based filters. As an added bonus it can do negative slopes, boosting the frequencies above the cutoff linearly.
The slope value parameter can be modulated by other modules and using negative values allow to create a unique variable-slope boost filter. The gain value adds resonance which boosts/attenuates frequencies around the cutoff frequency. **Note:** A slope of 6.0206 dB/octave is what traditional filters produce for each pole added (= filter order).

tone control: vintage 1 A tone control filter with bass/mid/treble bands derived from a famous guitar amp passive tone stack. All bands influence each other.

tone control: vintage 2 A tone control filter with bass/mid/treble bands derived from another famous guitar amp passive tone stack. All bands influence each other.

tone control: vintage bass A tone control filter with bass/mid/treble bands derived from a famous bass amp passive tone stack. All bands influence each other.

tone control: modern A more modern approach to tone control using multiple shelving sub filters. Setting all bands to the same value results in no frequency change at all.

8.2 Filter Positions

input At the plugin input after the input gain adjustment, affecting everything going into apShaper.

shaper in Before the sound enters the selected shaper engine. Harmonics generation is changed by the filter. The direct/dry signal stays untouched.

shaper out After the shaper engine. The distorted/shaped signal get filtered, but the direct/dry path is not changed.

direct The filter works on the direct (and no longer dry) signal only and doesn't affect the shaper engine in any way.

output Just before the DC killer working on the summed direct/shaped audio.

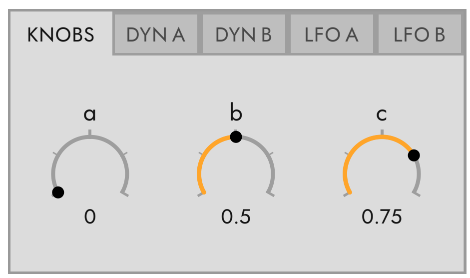
dynamics a in On the side-chain going to the dynamics detection A. The filter does not change sound, but changes the modulation output of the dynamics module A.

dynamics b in On the side-chain going to the dynamics detection B. The filter does not change sound, but changes the modulation output of the dynamics module B.

in+ out- The filter is applied to the shaper input as is and a gain-mirrored copy is applied to the shaper output. They cancel each other out completely on the direct signal, and partially on the shaped signal. What remains is the change in generated harmonics, allowing to emphasize frequency regions where more or less harmonics are generated.
Note: This is only possible for some types of filters: peak, shelfing and modern tone control.

9 User Knobs

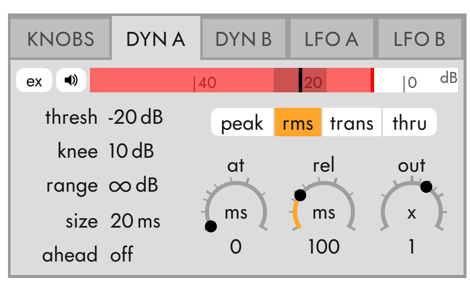
The knobs section provides three user-definable knobs. The knob values can control various parameters using modulation rules of the modulation system. One knob can change multiple values at the same time using multiple values. Initially, they are called **a**, **b** and **c**. The names can be changed using double-clicks on the labels.



The knobs can be controlled using host automation which can provide an extra layer of control.

10 Dynamics A & B

The two dynamics sections (dyn A & dyn B) convert an audio signal into a modulation signal based on dynamics. They can be used as an envelope follower, a peak hold detector, a transient detector, or feed through a modulation signal on the sidechain unprocessed.



The meter on top shows the dB level of the incoming signal.
Layered onto that is the threshold value showing the knee area as a gray bar. A black frame displays the dynamic range setting (if not 0 or infinite).
The threshold, the knee boundaries and the range can be adjusted by dragging with the mouse.

On the left of the signal bar, there are two buttons related to the dynamics section input. **ex** enables external/sidechain input. For this to work, a sidechain needs to be set up in the host application, otherwise the input will not change or be silent if **ex** is enabled.
The **pre-listen** button next to it allows feeding the dynamics section input to the plugin output. This allows listening to the sidechain (if enabled) and/or checking the effect of the "dynamics in" filters

Three detection modes can be selected:

peak The peak hold mode produces a dynamics signal that reacts quickly to fast increases. The maximum sample values are held for the time set up by the **size** parameter.

rms Rms stands for "Root Mean Square". For the duration of the **size** parameter, squared samples values are summed up and divided by the size. **rms** mode is good for slower material and is a way to measure the energy contained in the signal.

trans The trans mode uses a special algorithm to detect transients in the audio. Transients are steep rises of energy in the signal usually at the beginning of musical notes. In this mode the **size** parameter defines the scale of transients to capture. The transient mode only captures transients above the threshold.

thru The thru mode lets the input pass through to the output without any processing. Only the "out" knob is applied as a multiplier. This mode allows to use the sidechain input of the plugin to be used as a modulation source in modular hosts.

Multiple parameters can be adjusted to the left and using knobs at the bottom:

thresh The threshold value in dB. This is also shown as a line on top of the input level meter. It determines the level above which dynamics processing starts.

knee The soft-knee control defines the area over which the dynamics processing gradually sets in in dB. 10 dB means dynamics processing begins 5 dB below the threshold and the full amp setting is reached 5 dB over the threshold. This value is shown as a grey area on top of the threshold meter/slider combo view.

range The maximum dynamic range. Setting this to anything below infinite causes the signal to become constant once it surpasses threshold + range. This can be used to limit the maximum modulation.

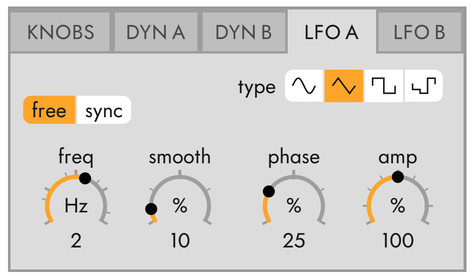
size The size parameter works differently depending on the dynamics module mode (set on the module title bar). In **peak** mode, it is the duration for which maximum/peak samples are held. In **rms** and **trans** modes it is the size of the processing window, the duration of audio that is considered for each output value.

ahead Stands for Lookahead. This is the time the dynamics module "looks into the future" during the processing so modulation can occur before transients have already happened. If lookahead is set above zero, the dynamics module will introduce latency in the signal chain. Hosts with plugin delay compensation can make up for that by sending audio earlier, but not all hosts support changing latencies while transport is running. To make sure, things are running in sync, restart the host transport.

attack/release The attack/release times of the built-in envelope follower. These set the time the output signal takes to rise/fall by 10 dB and can be used to smoothen the output signal.

out The value above the threshold is taken and multiplied by the out value.

11 LFO A/B



apShaper features two independent LFOs (Low frequency oscillator) to feed into the modulation section. Using modulation rules they can be used to shift many of the plugin parameters over time. Using multiple rules, one LFO can modulate multiple parameters at the same time.

In **free** mode, the LFO generates its own tempo based on a Hertz value and in **sync** mode the LFO can sync to the host beat tempo. In that case, the frequency knob adjusts the ratio to the host beat. **Note:** not all hosts supply beat information.

The LFOs support these waveform types:

sine A sine curve.

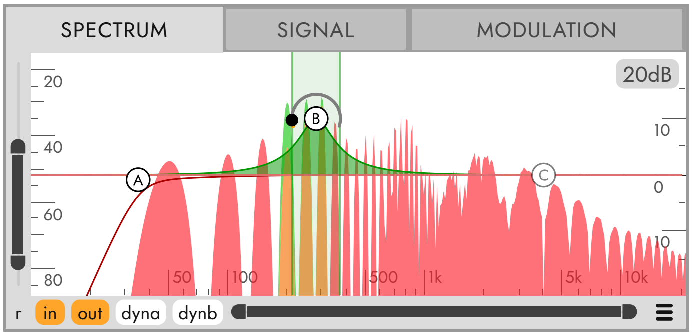
triangle A triangle wave.

square A square wave switching between the minimum and maximum values.

random hold At the beginning of each oscillator cycle, a random value is generated and then held for the cycle duration.

The **freq** knob adjusts the LFO frequency or its ratio to the host beat tempo.
The **smooth** button defines the amount of smoothing applied to the LFO output. The smoothing is always relative to the oscillator frequency so the waveform shape does not change if the frequency changes.
The **phase** setting offsets the waveform by a % value. A 50% phase setting means the waveform starts at its middle. Multiple LFOs with the same settings usually run in sync and using phase fixed ratios can be obtained. **Note:** If the frequency knob is adjusted, LFOs might loose sync and only restarting host playback will make then sync again.
The **amp** knob sets the LFO output amplitude.

12 Spectrum Display



12.1 Frequency Spectrum

The spectrum view display the frequencies contained in the input and/or output signals. It uses FFTs to calculate the graphs. The spectrum view range for frequencies and gains can be adjusted using sliders at the bottom and on the left.

At the bottom, some buttons allow controlling the spectrum view.

r Reset the graph viewport area to the values defined in your **Init** preset.

in enables the input analyzer right at the plugin input before any processing is applied by apShaper.

out enables the output analyzer which is places just before the plugin output after all the processing has happened including output gain and output clipping.

dyna enables the analyzer at the input of the dynamics section A. This includes all filters used on the dynamics A input. If dynamics A uses external input, the sidechain audio fed through dynamics in filters is displayed.

dynb enables the analyzer at the input of the dynamics section B. This includes all filters used on the dynamics B input. If dynamics B uses external input, the sidechain audio fed through dynamics in filters is displayed.

menu The bottom right menu button opens a dialog with additional spectrum settings:

> **res** This setting defines the number of samples used as FFT block size. This is directly proportional to the diplayed frequency resolution. The higher the resolution, the more detailed the frequency graph gets. However at the same time the larger blocks mean the display reacts slower to changes and the transients get smoothed more. Higher resolutions also require more processing power.
>
> **tilt** This setting optionally applies a tilt to the graphical output. At the **0 dB**/Oct setting the display is mathematically accurate and white noise gets displayed as a horizontal line. Music tends to fall off towards higher frequencies and the **3 dB** and **4.5 dB** settings allow to

compensate for that. Common sound then appears more like a horizontally balanced graph. **Note:** Tilt is only applied to the analyzer graphs, not the filter frequency curves!

decay The speed at which the frequency graph moves downward. **frz** means all freqs stay at their maxima.

12.2 Direct filter editing

On top of the frequency curves, filter response curves for the filter bands **A, B** & **C** are drawn. Various handles allow direct filter manipulation. The filter curve colors reflect their position on the signal path:

green plugin input.

dark geen shaper input.

dark red shaper output.

blue direct path (bypassing shaper).

red output.

blue green dynamics A input.

violet dynamics B input.

green & inverted gain red in+ out- position emphasis filters.

In the top right corner of the display, the dB range of the filter curves is shown and can be adjusted using a popup menu. The filter curve gains are not necessarily scaled the same amount as the spectrum frequencies.

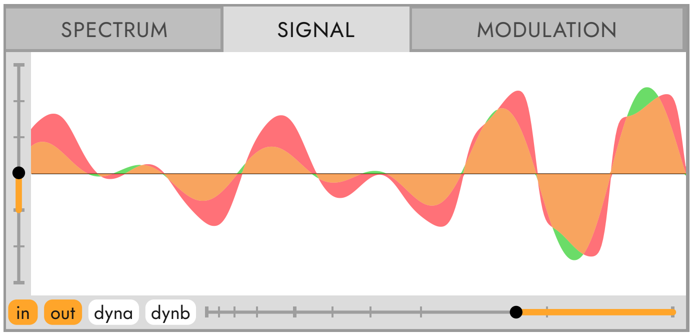
When a filter handle is clicked on the spectrum, it gets selected. Additional handles appear and the filter details view on top of the spectrum switches to the pane of the selected band.

Filter bandwidth can be adjusted by dragging the borders of the colored bandwidth area.

Filter types can be changed by right-clicking the filter handle.

Filters can be disabled/enabled by double-clicking the filter handle.

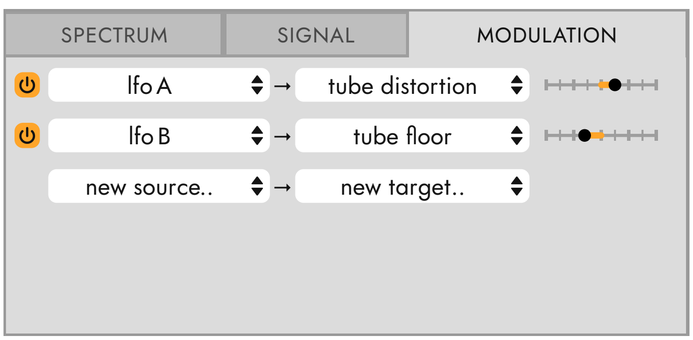
13 Signal Display



The signal display shows the waveform present at the plugin input and/or output as well as the signals generated by the dynamics sections. At the bottom, toggle buttons define which signals are drawn. Input is drawn green, output red and the overlap orange. The dynamics A signal is drawn as a blue line on top and the dynamics B signal as a violet line.
The large horizontal slider sets the time duration displayed. For short durations, an autocorrelation algorithm is used to make the display stable (in sync with the waveform).

On the left side a slider allows vertical zooming of the waveform. Input and output are centered at the middle while the dynamics curves always displays zero at the bottom.

14 Modulation



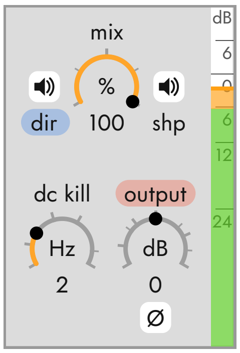
The modulation system allows to create up to 16 modulation rules. Each rule connects one modulation source (f.i. LFO) to one target (f.i. saturation drive). Multiple rules can use the same sources/targets. Each rule has an independent gain slider to control the amount of modulation.

To create a new modulation rule, use the **new source..** or **new target..** popup boxes to assign a source/target. These are always displayed on the lowest rule position. Once a source or target has been assigned, additional controls appear to disable/enable the rule and to control its gain.

The available targets change based on the shaper engine and filter band types used.

To remove a rule, use the lowest entry on the source or target selection boxes.

15 Output



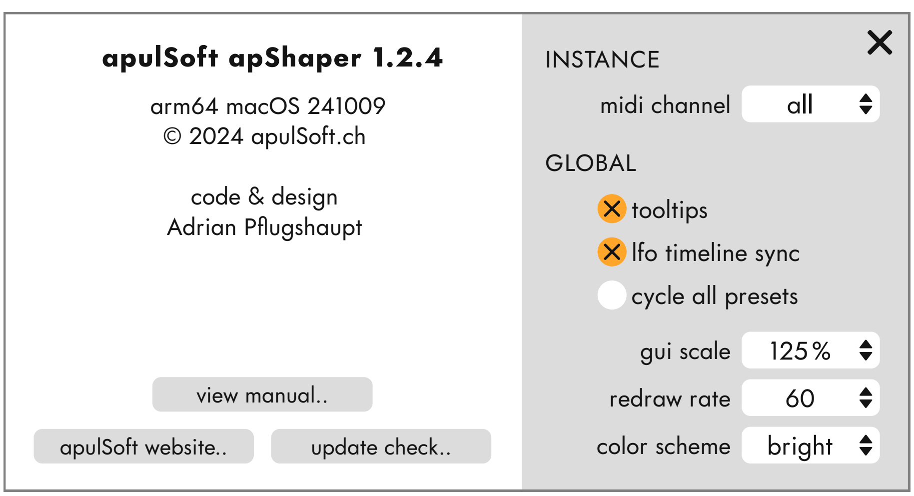
The output section or apShaper provides direct/shaped signal mixing, a dc kill filter, output gain/phase and the output level meter.

The **mix** knob sets the amount of shaped signal in the output. It works like a regular dry/wet mixer, but additionally the direct signal can be filtered by the filter bands. The two **speaker icon** buttons temporarily switch the output to direct-only and shaped-only to check the signals while creating a setup.

The **dc kill** knob controls the -6dB point of a 12dB/oct high-pass filter. It removes frequencies below the hearing range from the output signal which includes dc offset. Check **section 18** to get an explanation why waveshaping can create dc offset. If the knob is turned all the way to the left/minimum, the dc kill filter is removed from the signal chain. This is not recommended, but can at times be useful to check what the shaper is doing.

The output gain amplifies the output signal as the last step of signal processing. Afterwards the level is displayed in the output meter. Adjust the gain control to make sure the output level is reasonable. The output phase switch only influences things that happen to the audio signal once it has left apShaper. If the next processing steps are sensitive to asymmetrical signals, changing the phase/inverting the signal could make a big difference as apShaper can produce highly asymmetrical signals.

16 Plugin Settings & Information Dialog



This dialog is opened by clicking the gear icon or the plugin name in the top left corner of the main apShaper interface.
The left side of the dialog shows some basic information about the plugin and has four buttons at the bottom.

apulSoft homepage.. This opens the systems default browser and points it at https://www.apulsoft.ch.

view manual.. The apShaper manual is opened in the default pdf viewer application.

update check.. This opens a special page on the apulSoft homepage and sends version information. The homepage checks the version against the latest release and provides links to downloads if newer versions are available.

The **instance** settings at the top-right corner are only valid for the current instance of the plugin. **midi channel** allows to set the midi channel if multiple instances should be individually controlled using the same midi stream.

Global settings apply to all instances of apShaper. All instances using the same plugin format in the same host update immediately, other instances once the plugin is reloaded.

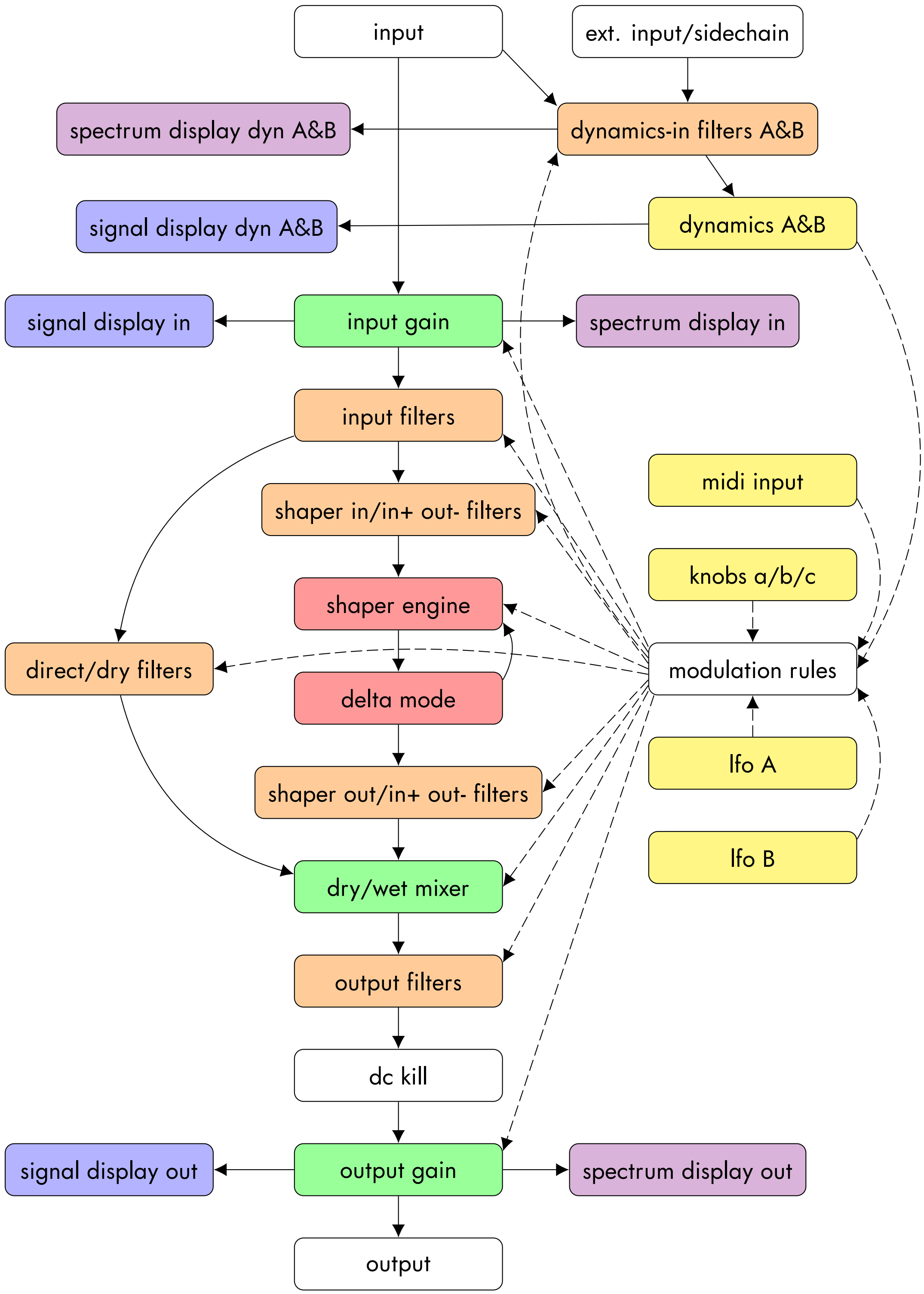
tooltips If this is activated, orange rectangles with little hint texts pop up if the mouse hovers in place over a control for a second.

gui scale Choose how large the plugin interface should be drawn in %. The right-side popup features a few presets and it is also possible to just enter any value between 25 and 500. Some hosts might only display the plugin correctly at the new size once the plugin window is closed and reopened. In extreme cases, the host might need to be restarted.

redraw rate The number of interface redraws per second. A slow computer might not be able to reach high rates. High refresh rates only work well with small host audio buffer sizes.

color scheme Switch between multiple color schemes for the interface. The menu shows all installed schemes. New ones can be added by the user by renaming and editing the existing scheme files. They are are located in a folder called ColorSchemes next to the apShaper presets folder. Use the **manage in finder/explorer..** entry of the presets menu to navigate to the presets folder and go one level up. The color schemes use an xml based format and can be edited in any text editor. More information can be found inside the **bright.xml** file.

17 Plugin Signal Flow



18 Waveshaping and DC offset

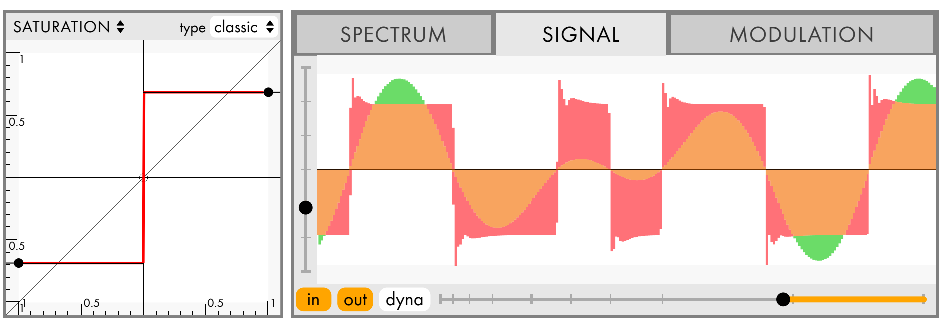
A signal without DC offset has a sum of zero for large sample sections over time. The positive and negative samples are perfectly balanced. If not, the signal has DC offset. This is undesirable as it lowers the maximum possible sample values, leads to artefacts in later stages and reduced loudspeaker performance if it is not filtered out before.

Applying non-linear waveshaping to a signal easily leads to DC offset as it applies different amounts of amplification to different sample values. This is most of the time even true for symmetrical transfer curves because even a signal without DC offset most of the time has an asymmetrical waveform which then produces different amounts of positive and negative result signal. Asymmetrical transfer functions easily lead to DC offset for all input signals.

That is why apShaper includes a dc kill filter in the output section. It is a 2-pole high-pass filter (12 dB/oct steepness) that is applied before the final output gain. apulSoft recommends to always keep this filter running. When DC offset is produced, inaudible very low frequencies are often produced with it. That's why the DC kill filter can be tuned up to 25 Hz. The higher the frequency, the faster DC offset gets removed from the signal. Modulated transfer functions might require high DC kill frequencies in order not to produce any offset.

If the built-in DC removal does not meet the requirements, it can be turned off and one of the filter bands A,B or C can be used in output position with the high-pass type to filter out low frequencies including DC offset.

19 Gibbs Phenomenon



If a transfer function includes sharp corners (like high-drive saturation or waveshaping), very high frequencies up to half the sampling rate are added to the signal. To calculate the output signal correctly in such cases, apShaper uses integration in combination with oversampling to reduce aliasing as much as possible. When high frequencies are accurately added to the resulting corner-y signal, the so-called gibbs-phenomenon occurs. It shows up as high-frequency sinus-like spikes near corners of the signal that can overshoot the regular signal range. This can lead to clipping in later processes.

If this is a problem, the output gain of apShaper can be lowered to make the spikes stay inside the

-1↔1 signal range. Alternatively the signal input can be lowered, drive can be lowered or one of the filter bands can be used to filter some high frequency content away to lower these signal spikes. Higher and max quality produce less overshoot than high quality. Linear phase oversampling leads to half as much overshoot as the overshoot becomes symmetrical. Naive quality does not produce any overshoot, but this comes at the cost of heavy aliasing.

20 Midi Input & Program Changes

Please consult the manual of your DAW to find out how to send midi data to an audio insert plugin. **Note:** Not all host applications support this feature.

Using modulation rules, midi pitch bend information as well as various midi controllers can be used to change effect parameters.

Sending midi program change messages switches to different presets. The sent program change number is used to select a new preset based on the order of the current preset folder.
Midi messages are filtered based on the instance **midi channel** setting.

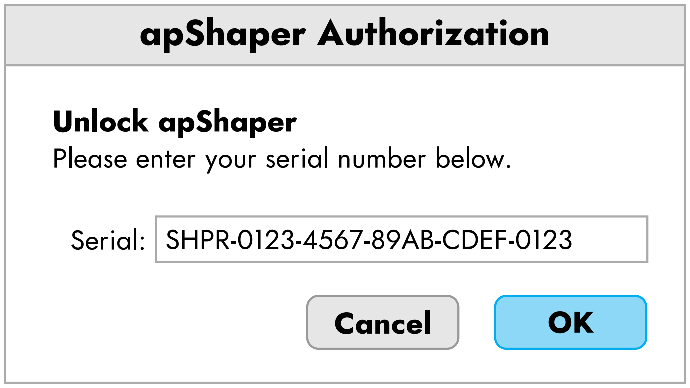
21 Init and Factory Presets

apShaper installs a number of example presets by default. All factory presets include descriptions that are accessible by clicking the **i** button on the left side of the current preset name.

21.1 Init Preset

A special preset is the **Init** preset. The first time the plugin is opened it is auto-generated from the plugin's default values. Every time a new instance of apShaper is created, the **Init** preset is loaded. This allows you to set up your personal default values by overwriting this preset once the plugin is in the desired default state.

22 Unlocking the Full Version of apShaper



Once you bought an apShaper license via 2Checkout from the apulSoft homepage, there are two ways to unlock the plugin.

- When you open the apShaper plugin interface in demo mode, a welcome screen appears with a **enter serial..** button. Click this button open the serial entry dialog.
- If the plugin is running in demo mode, the →full version button is displayed on the top right which brings up a license dialog where the **enter serial..** button can be used to open the serial entry dialog.

Enter the serial exactly as received and click **OK** to unlock the full version.

In case the serial is not accepted, check the following things:

- The serial needs to be an apulSoft apShaper serial consisting of **SHPR** followed by 5 sections of 4 hexadecimal digits (0-9, A-F).
- If copy/paste was used, try typing manually as copy/paste sometimes copies more than was intended (white spaces, tab stops, etc).

23 Frequently Asked Questions (FAQ)

- **I lost my serial. How can I retrieve it?**
  Just head to https://www.apulsoft.ch/contact and get in touch. Please add enough information to locate your order in the database and you will receive your serial as soon as possible.

- **What to do if the window size does not match the interface size after adjusting the GUI scale?**
  Depending on how the host application handles resizing of plugins triggered by the plugin, changing the GUI scale might not immediately work correctly. Any change to the GUI scale is stored in a global preferences file that is read whenever a new instance of apShaper is created. If this problem occurs, first try to just close and reopen the plugin window/editor. If that does not help, set the desired scale on the settings pane and then restart your host application. As long as GUI scale is not changed again, window and content should match.

24 Changelog

- Version 1.0.1
  - Improved quality and lowered cpu usage of filters.
  - Improved ctrl-drag fine mode on supported controls.
  - Improved mouse-wheel handling which now includes a ctrl fine mode.
  - BUGFIX: Tube engine issues at high distortion values and with low frequency input signals
- Version 1.0.2
  - Improved preset compatibility between different versions of the plugin.
  - BUGFIX: window size issues on Vst3/Windows/HiDPI.
- Version 1.0.3
  - Improved waveshaping quality. New "max"-quality uses third order integration to reduce digital artifacts.
  - Improved the way the LFO module resets.
  - Increased LFO output amp range.
  - Support for Notarization on modern versions of OS X.
- Version 1.1.0
  - New vintage and modern tone control filter types.
  - BUGFIX: Crash in FL Studio when bypassing apShaper in the fx rack.
- Version 1.1.1
  - Added options to choose the clipping/saturation algorithm for the harmonic and tube engines.
  - Compatibility with Apple Silicon Macs.
- Version 1.1.2
  - Improved timing accuracy of host-automated parameters.
  - BUGFIX: installer signing issues on older macOS versions.
- Version 1.1.3
  - Input and output meters now have units, markers and peak hold.
  - Increased the number of user modulation knobs to three.
  - The preset cycle buttons can optionally cycle through the entire preset structure.
  - LFO module: option to resync on playback time jumps (including cycle playback)
  - Provide better info to hosts about changing parameter names.

- Compatiblity improvements with macOS Big Sur.
- BUGFIX: Various tooltips and user-defined colors.
- BUGFIX: Stepper parameters not visually reacting to preset switches.

• Version 1.1.4

- The dry signal is now phase-synced to the wet antialiased signal to improve dry-wet mixing.
- Improved oversampling quality.
- New linear phase oversampling mode.
- Reworked max quality setting. It now uses 8x oversampling and is superior to higher quality in all cases.
- Renamed 'ultra' antialiasing quality name to 'higher'.
- New filter position option: before the dynamics module input.
- BUGFIX: Spectrum display ranges recall from Init.shprpreset not working.
- BUGFIX: Sharpness of display rulers on HiDPI screens.
- BUGFIX: Fixed LFO time-line sync.
- BUGFIX: Bypass behaving strangely in Cubase.

• Version 1.1.5

- New filter position: 'in+ out-' for emphasis filters.
- BUGFIX: gui data was updated during offline bounces.
- BUGFIX: Work around a plugin loading issue on macOS 10.11 and 10.12.

• Version 1.2.0

- AAX: Compatibility with Apple Silicon native.
- User-ID no longer required/displayed on the GUI.
- Side-chain option for dynamics input.
- Option to pre-listen to the dynamics input.
- Frequency analyzer display for dynamics input.
- Improved DC removal filter.
- Less noise when switching presets.
- Improved spectrum graph drawing.
- Better display of numerical values.
- Accept both ',' and '.' as decimal separators for keyboard entry.

• Version 1.2.1

- Added a second dynamics detector.
- BUGFIX: dynamics spectrum and signal displays sometimes not updating correctly.

  - BUGFIX: unable to switch presets when not processing audio.
  - BUGFIX: sidechain connections not working after project reloads in some vst3 hosts.
  - BUGFIX: incorrect mono to stereo processing when sidechain is used.

- Version 1.2.2
  - HOTFIX: auval failure in mono rendering test.

- Version 1.2.3
  - New saturation function: clip with adjustable knee.
  - New delta option: get the difference between the linear signal and the shaper signal.
  - Fixed Init.preset mechanism in the AAX version.
  - Added support for automation menu keyboard shortcuts in Pro Tools.
  - BUGFIX: Preset menu layout issues on Windows HiDPI screens.
  - BUGFIX: type-switches were delayed in the saturation engine.
  - BUGFIX: incorrect curve drawing in the saturation engine.

- Version 1.2.4
  - New thru-mode for dynamics sections to use sidechain as a modulation signal.
  - Reworked GUI, new dark color scheme.
  - .msi installer for Windows.
  - Dynamics graphs now also show negative values.
  - BUGFIX: Tone control filter band default values.
  - BUGFIX: Wrong bottommost pixel for some gui scales in some hosts.

- Version 1.2.5
  - Improved Windows GUI performance.
  - BUGFIX: Occasional crash when creating a new .vst3 instance in Reaper.