

# PRESSWERK

## DYNAMICS PROCESSOR



## USER GUIDE

version 1.1

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# Introduction

If you have any questions, feel free to post a message in our lively [KVR user forum](#)!

## Installation

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Go to the [Presswerk](#) page, download the latest installer for MacOSX or Windows, and unzip the compressed file. Open the *Presswerk10* folder and start the installer app.

The only demo restriction is a mild crackling sound at irregular intervals after about two minutes of use. The demo is otherwise fully functional. The crackles disappear after you have entered a serial number. For more information, please refer to the *ReadMe* file included with the installer.

By default, Presswerk uses the following directories:

Windows presets	...\\VstPlugins\\u-he\\Presswerk.data\\Presets\\Presswerk\\
Windows preferences	...\\VstPlugins\\u-he\\Presswerk.data\\Support\\ (*.txt files)
Mac presets	MacHD/Library/Audio/Presets/u-he/Presswerk/
Mac presets (User)	~/Library/Audio/Presets/u-he/Presswerk/
Mac preferences	~/Library/Application Support/u-he/com.u-he.Presswerk... (*.x files)
Mac resources	MacHD/Library/Application Support/u-he/Presswerk/

To uninstall, delete the plugin files then *Presswerk.data* (PC) or the two Presswerk folders (Mac).

## Resources

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### u-he online

- For downloads, news articles and support, go to the [u-he website](#)
- For heated debates about u-he products, go to the [u-he forum](#)
- For friendship and informal news updates, go to our [facebook page](#)
- For u-he presets (payware and free), go to [u-he patch library](#)
- For video tutorials and more, go to our [youtube channel](#)

### the u-he team

- Urs Heckmann (framework code, strategy, discipline)
- Sascha Eversmeier (presswerk code, user guide expertise)
- Sebastian Greger (graphics, GUI design)
- Rob Clifton-Harvey (support, databases)
- Jan Storm (practically anything)
- Howard Scarr (user guide)
- Micheal Pettit (videos, marketing)
- Viktor Weimer (support, the voice)
- Oddvar Mannlig (everything else!)

with special thanks to Brian “you can call me Brian” Rzycki for maintaining [PatchLib](#)

### terms of use

Please read the document **license.txt** included with the Presswerk installer.

## The Presswerk Story

Presswerk is a powerful but easy-to-use dynamics processor which can deliver excellent results in any computer-based audio workplace (OSX or Windows). With its rich feature set, Presswerk is suitable for a wide range of sound-shaping tasks – as you will soon discover.

Presswerk grew out of an in-house codebase we have been using for various kinds of dynamics signal processing. Whenever we need something 'compressorish', we can load existing modules and immediately try them out.

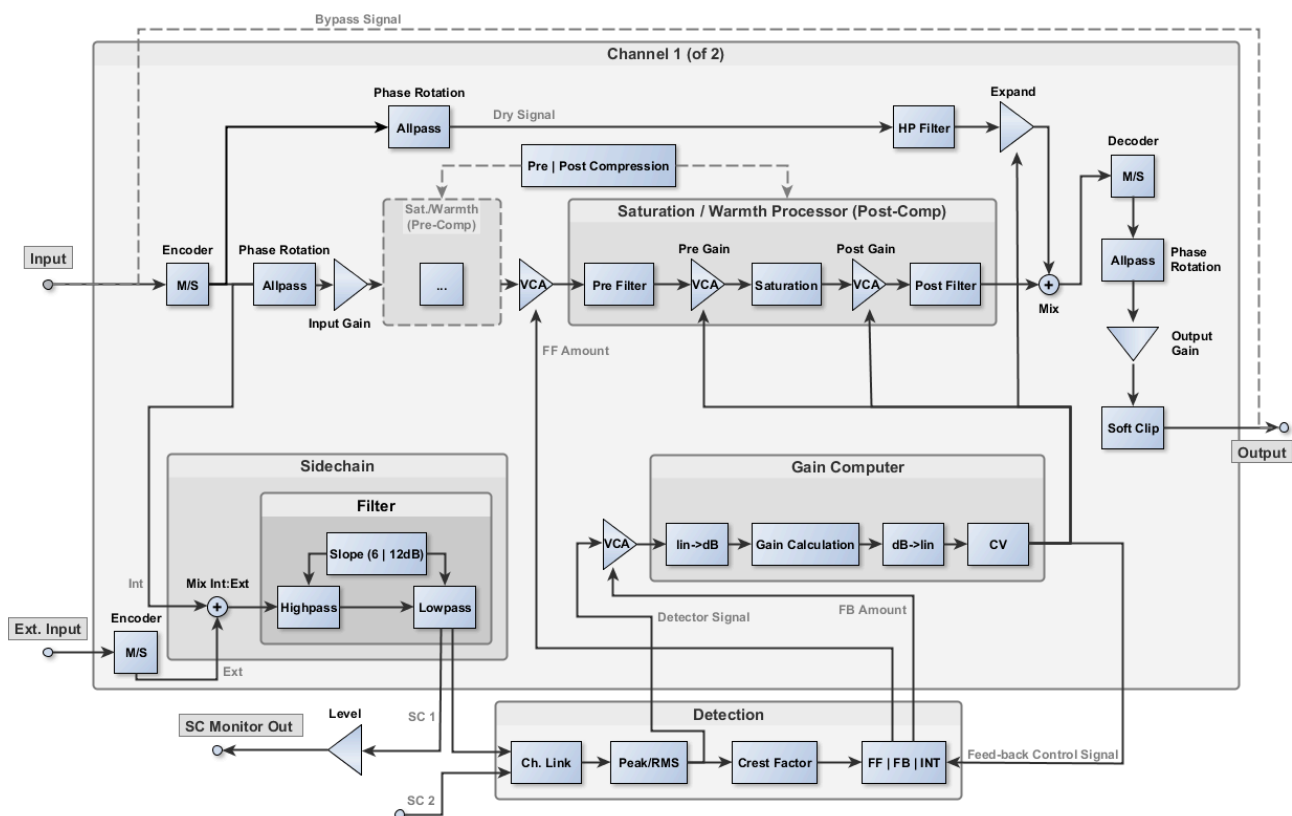
A few years ago, out of sheer curiosity, we stuck some of those modules together to form a simple compressor with an M/S matrix and saturation stage. Then we tested several hardware compressors, paying attention to every microscopic detail. It soon became clear that we had no alternative: we would start developing a full-blown compressor plug-in!

The reactions of testers to an early alpha version were encouraging, but it wasn't quite 'there' yet. We decided to redesign parts of the core architecture from scratch – after studying the classic hardware more thoroughly. We wanted Presswerk to sound even more 'analogue', we were determined to give it as much musical soul as possible...

We hope you agree that the result was worth the extra effort!

– Sascha Eversmeier

Here is a simplified signal flow diagram:



There is a much [larger version](#) of this image on the last page of this document.



## A Peek at the Special Views

While browsing through the presets you must have noticed those ‘special views’: Simpler panels with fewer controls (some of which are ‘macros’ combining several parameters). It’s almost like having a bundle of specialist plugins, except that you don’t need to swap them out.



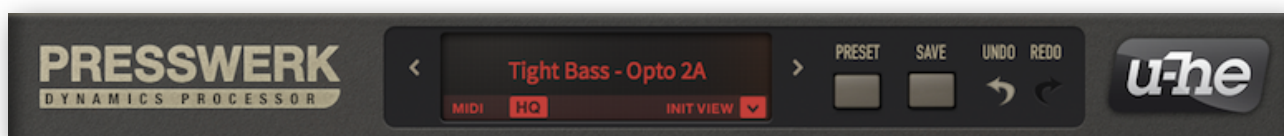
This is just a peek. For descriptions of each one, please refer to the [Special Views](#) chapter.

## UI Size

The size of Presswerk’s window is adjustable in 10% steps from 70% to 200%. Simply right-click in the background and select from the list.

Note, however, that this is temporary. You can change the default size in the global [Preferences](#).

## Control Bar



### Data Display

This normally shows the name of the selected preset. While a parameter is being edited, however, it shows the value of that parameter. After a few seconds, it reverts to showing the preset name.

You don't always have to open the browser to select presets: clicking on the arrows steps forwards and backwards through them. Click directly on the data display to open a drop-down list of all presets in the current directory. Of course Presswerk also has a regular preset browser (see below).

At the bottom left of the data display is a normally dark **MIDI** activity indicator, which flashes whenever Presswerk receives MIDI messages. For information about how to route MIDI into effect plug-ins such as Presswerk, please refer to the documentation that came with your host application.

Next to this is an **HQ** button. Although usually barely audible, aliasing is possible, especially when the envelope settings are relatively short. Oversampling helps: At 44.1kHz or 48kHz, activating HQ sets the internal sample rate to 4 times the original (but only two times at 88kHz or 96kHz). HQ also minimizes aliasing generated in the saturation stage, and the improvement in audio quality is greater than from gain-riding the stages. We recommend switching HQ on when rendering your project offline. Note that HQ status, although global, is remembered for each instance.

The dropdown symbol labelled **INIT VIEW** opens a menu containing templates for the various alternative views, which are described in detail in the [VIEWS](#) chapter (starts on page 20).

### Preset / Save

PRESET opens the browser (see the next page). SAVE opens a dialog box in which you can enter information before finally storing via the 'Apply' button. **IMPORTANT:** SAVE uses the currently selected folder. If necessary, click on PRESET and select the required folder in the lefthand panel!

### Undo / Redo



The curved arrows let you step backwards and forwards through your editing 'history'. Note that *Undo* and *Redo* even works if you select a new preset before saving your edits!

### u-he badge



Click on 'u-he' for quick access to our website, to this user guide, to our KVR support forum, or to our social networking channels (currently Twitter, Facebook, YouTube).

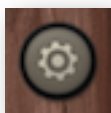
### Bypass



The button at the top left needs little explanation – activating it simply connects the audio input directly to the output, so the signal is not processed. The button flashes as a constant reminder that this particular instance of Presswerk is currently bypassed.

Note: Even while bypassed, Presswerk continues to run so you can still check how your adjustments *would* affect the compression.

## Configuration



The cogwheel icon at the top right opens Presswerk's configuration pages, which offer MIDI remote control assignments as well as several GUI preferences. For details, go to the [Configuration](#) chapter.

## Presets

To browse through the presets, click on the PRESET button to the right of the data display. A translucent window opens in the middle of the GUI, looking something like this:



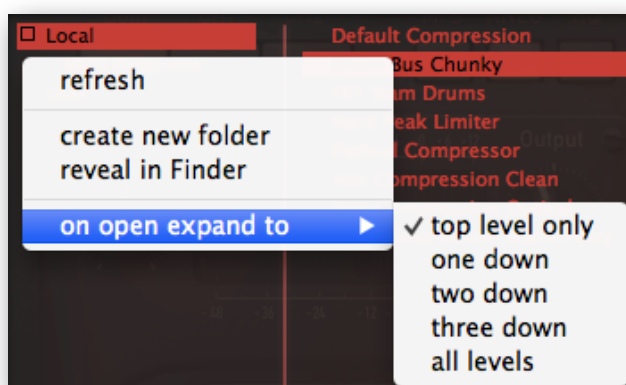
Folders are listed on the left, presets in the centre, extra information to the right. After clicking on a preset you can use the up/down cursor keys on your computer to step through the others.

Note: You can still make adjustments to Presswerk parameters while the preset browser is open. To reveal all controls, simply slide the browser window up or down.

Tip: If there's a preset called **default** in the local root, it will be loaded automatically whenever a fresh instance of Presswerk is started. Try this: Load your favourite preset, open the root folder (Local) and save it under the name 'default'. Start a new instance of Presswerk and check...

## refresh / create / reveal / on open

Right-click in the lefthand (folders) pane to see this context menu:



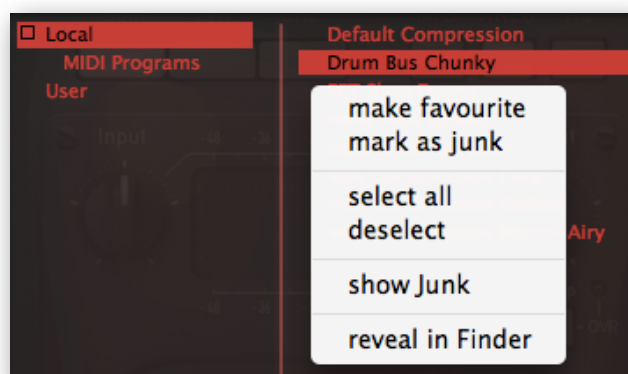
As Presswerk's browser is functionally limited, the context menu has a *Reveal in Finder / Explorer* option which opens a regular file window at the correct path.

Whenever you need to update the list e.g. after having moved or renamed presets, select *refresh*. Clicking on any folder should also refresh the list.

*On open expand to* lets you select the maximum depth of open subdirectories whenever the GUI is opened (or the list is refreshed). Try the various options to find out which one you like best. Note that the current preset will always be visible.

## favourite or junk?

Right-clicking in the righthand (presets) pane opens a different context menu:



Right-click on a preset to classify it as *Favourite* or *Junk*. Favourite files are marked with a star, junked files disappear from the list. Select *show junk* from the same context menu whenever you want to see the junked presets (which appear with a bright ‘stop’ symbol).

If you right-click on a file that is already selected, or if multiple files have been selected via SHIFT etc., the top two options are replaced by *make selected favourite* and *mark selected as junk*.

## MIDI Programs

*Local* also contains a special directory called *MIDI Programs*, which is initially empty. The point of having this folder is that those presets can be selected via *MIDI Program Change* messages. For information about how to route MIDI into effect plug-ins, please refer to the documentation of your host application / DAW.

If you place a bunch of presets (up to 128) in the *MIDI Programs* folder, they will **all** be loaded into memory (for performance reasons) when the first instance of Presswerk is loaded. Note: this means that changes to the content of *MIDI Programs* will only be recognized after the host application is restarted. MIDI Programs cannot be added, removed or renamed on the fly, however often you try to refresh!

MIDI Programs are accessed in alphabetical order, so it’s a good idea to put a number at the beginning of each name e.g. ‘000 preset-name’ to ‘127 preset-name’ or similar.

## Knobs and Buttons

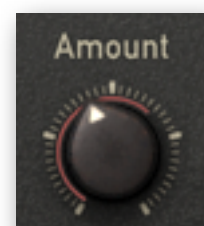
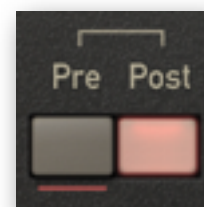
All knobs are adjusted by clicking and dragging vertically. To fine tune, hold down SHIFT before moving the knob.

Buttons bracketed together with a line above the labels are so-called ‘radio buttons’: activating one switches the other(s) off. All the rest are simple on/off.

Buttons that flash when activated (Bypass, sidechain monitor channel selectors) do this as a constant reminder to switch them off again!

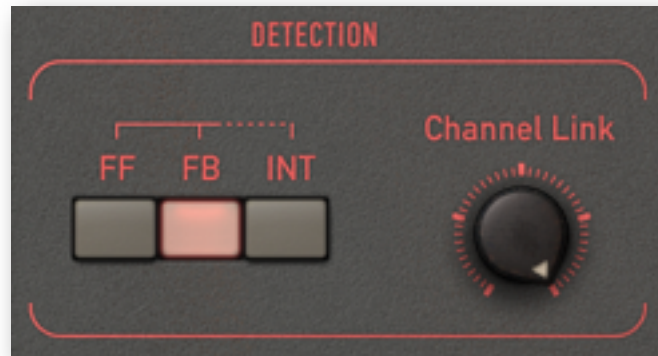
If you have the GUI skin **Original Plus** (see [Preferences](#)) selected, the values of parameters in “the other channel” are indicated by red arcs (knobs) or underlines (buttons). In the image to the right, the ‘Amount’ in this channel is set lower than it is in the other channel. See [Edit](#) mode a few pages down.

To reset a knob to its default value, either double-click on the knob or hold down Alt (Mac) or Ctrl (Windows) then single-click.



# Global Parameters

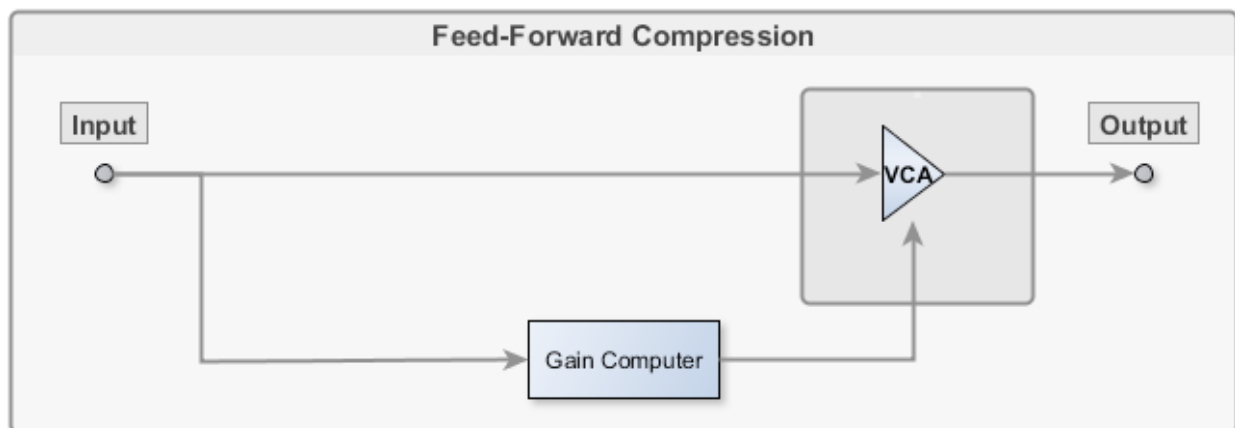
## Detection



Most dynamics compressor-limiters have a fixed topology, so the origin of the signal feeding the detector circuit and gain-computing device (VCA, voltage divider etc.) is also fixed. In modern units this is usually the *input*, and the method is called *feed-forward* (FF) detection...

### Feed Forward (FF)

Feed-forward detection tracks the input very precisely. The resulting sound could be described as 'predictable' and 'direct', as the effect strictly follows threshold and ratio settings etc..

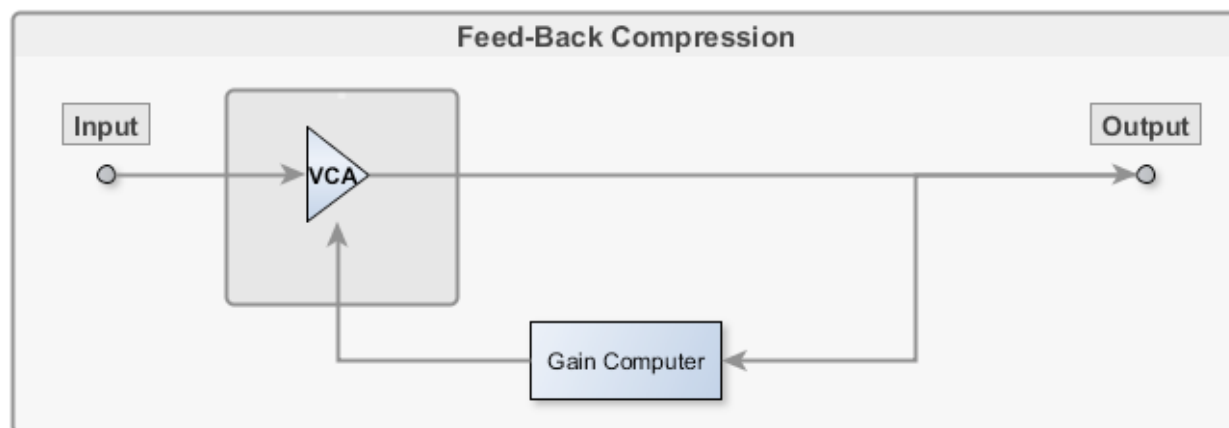


Feed forward detection isn't particularly forgiving – it requires careful adjustment, especially when applied to complex audio material.



## Feed Back (FB)

In contrast, the *feed-back* (FB) design found in classic hardware is less obtrusive, more 'musical'. The FB principle, greatly simplified for audio circuits, is standard in control engineering devices e.g. domestic heating and aircraft flight automation. The principle specifies: Look at the output, and as soon as there's 'a bit too much', adjust the input to compensate. So with FB detection, everything tends to adjust itself automatically, and the system requires little attention.



Feedback detection does have its drawbacks, however. Not only are the threshold and ratio settings fairly arbitrary and program-dependent (i.e. governed by characteristics of the material being processed), but the compression ratio is also limited to values between 2:1 and 3:1. To limit the input signal, the feedback loop would need masses of extra gain. In fact this should be infinite, and we all know that infinity can be "problematic"!

Some designs offer a huge amount of gain anyway, which is released as soon as saturation happens. For Presswerk, we chose not to allow more than about 2.5:1 ratio in FB mode, and ensure a more linear response within the headroom range.

## Interactive (INT)

Can we have the best of both worlds? When the interactive detection (INT) mode is active, the signal is continuously analyzed, and the two internal gain multipliers are blended accordingly. This is achieved using the *crest factor* (peak-to-RMS ratio) of the signal. A high crest factor means that the signal has lots of transients but little steady-state energy.

With transient-rich material, detection in INT mode leans more heavily toward FF, while smoother material will keep it more in FB mode. The result is almost as unobtrusive as FB gain-riding, but INT mode allows for higher ratios (up to about 5:1).

## Channel Link

By default, the two channels are linked together 100%. The detector derives its control signal from the maximum of the two audio signals (the loudest channel 'wins'), ensuring that the spatial integrity required for most stereo-source processing is retained. Linked setups usually work best with coherent material at moderate to high compression ratios.

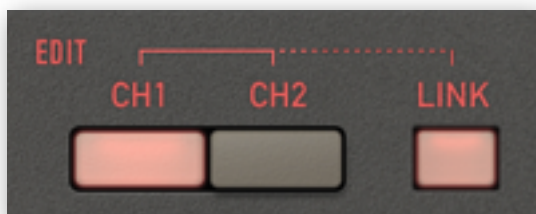
The opposite effect is *brickwall limiting*, where maximum loudness and overshoot protection is required and where linking the channels would result in giving up valuable headroom. When 0.0% linked (i.e. unlinked), sudden gain reduction can lead to a momentary 'tilt' of the audio to one side.

Tip: If your audio material is difficult, it's worth spending some time and effort on finding the best intermediate settings.

## Operation Modes

### EDIT

Presswerk can operate on two channels independently. However, the standard mode is *Link*: the stereo or dual-channel signal is then controlled by a single set of parameters consisting of all sections on the user interface except those with red borders.



*Link* is the best choice for most jobs, as it ensures spatial coherence with a minimum of tweaks. If in-depth adjustments are necessary, however, you can disable LINK and activate CH1 and CH2 to adjust the individual channels independently. Remember that *Channel Link* (see the previous page) ultimately determines how independent the two channels really are.

Note that the values of parameters in “the other channel” are indicated by red arcs (for the knobs) or underlines (for the buttons).

### M/S



Traditionally, M/S was the standard dynamic treatment applied before cutting acetates. As groove size is limited, strong signal peaks are usually placed dead-center.

M/S (mid-side) mode gives you more opportunity for detailed stereo signal processing. Channel 1 acts on the mid (L+R), while channel 2 is the side-chain (L-R).

Note that dynamics processing in M/S mode is independent of the selected EDIT mode. Even with *Link*, the channels will still operate independently unless you set *Channel Link* to maximum.

M/S encoding happens after the input controls (like on certain popular hardware compressors) so the stereo remains intact. If you really need to change the M/S balance, use *Threshold* or *Output*.

### DPR



DPR (Dual Phase Rotator) emulates a trick often used by radio broadcasters to tame vocal transients. Originally designed to increase headroom by reducing asymmetries in the human voice, it can also be used as a general-purpose ‘phase smearing’ option.

Note: Digital compressors tend to leave the phases and frequency response intact, whereas components built into analogue equipment have capacities and inductivities that can significantly smear the phase of a signal.

How DPR works: The phase of low mids (around 200 Hz) in the input stage is shifted, as are bass frequencies in the output stage. The former can e.g. keep kick drums under control (achieving a more snappy, slightly coloured attack) while the latter compensates for the drop in low frequency impact caused by momentary gain reduction.

However, there’s no free lunch. If you have been wondering why insanely hot-mastered music sounds harsh and wimpy on the radio, here’s your answer: Phase rotation exaggerates symmetrical distortion, making the sound worse. It can only work its magic when applied to intact material.

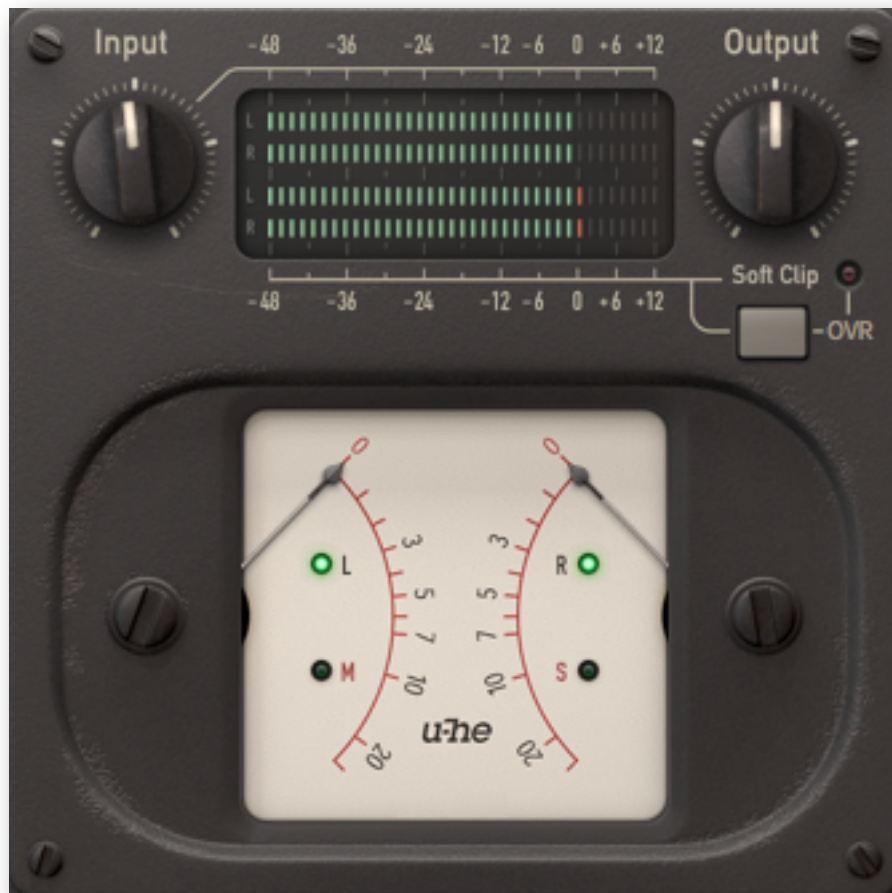
While the taming of transients via phase rotation is usually inaudible, it can change our perception of low frequencies. The effect manifests itself as more ‘oomph’ (for lack of a better word). The altered attack can also make the sound more focused and fresh, as well as a little brighter.

In the end, your decision about whether to enable DPR will depend as much on your personal preference as on the job at hand. Please keep in mind, however, that rotating phases can cause problems if Presswerk is running in parallel with another track carrying the same original signal!

# Channel Parameters

## Levels and Meters

In the centre of the window is an analogue-style VU meter which measures the gain reduction (in decibels) for each of the two channels. Gain reduction results from the compression process itself as well as from saturation ('warmth').



Each channel has its own **Input** and **Output** stage. The levels are indicated by four 'LED strip' type peak meters – the green bars in the above image. The upper two bars show input levels, while the lower two show output levels.

The **Input knob** adjusts the gain of the incoming signal to a level suitable for compression. It can also fine-tune compression response as, once the available headroom is exhausted, the detector amplifies *non-linearly*. Detector saturation, an interesting effect in its own right, prevents any further gain reduction.

The **Output** knob adjusts the output gain so you can match the processed signal level to the unprocessed (bypassed) level.

The Output control also drives the optional **Soft Clip** stage, ensuring that the level stays below 0dBFS and that the transition between 'clean' and 'clipped' peaks is relatively smooth. If the signal is hot, the indicator will light up, meaning there is some soft clipping. With **Soft Clip** disabled, however, the same indicator lighting up means digital clipping – which you should generally avoid.

## Compression Curve

On the left is a panel used for setting up the compression curve. The graphic display shows the resulting input (x) versus output (y) curve. The detector signal is updated in realtime, so it is fairly easy to set up a suitable threshold and ratio.



The compression parameters are:

- **Threshold:** the level (in dB) at which all processes start to kick in.
- **Ratio:** the gain reduction above the threshold. For instance, '4' here means 4:1. For instance, if the incoming audio in this case is 8dB above the threshold, the output will be only 2dB above the threshold.
- **Soft Knee:** increases the range within which compression is *gradually* applied. The maximum setting here means the smoothest transition from 1:1 up to the specified ratio, while the minimum setting makes the transition 'razor sharp'. The more complex the material and the more transparent you want your compression to sound, the softer the knee should be. Low 'hard knee' settings are more suitable for limiting or special effects e.g. 'squashed' drums.
- **Non Lin:** in the central position, gain reduction above the threshold follows a straight line. Positive values bend the curve downwards, while negative values bend it upwards. The *NonLin* control helps simulate the behaviour of specific compressors or gain-reduction circuits.
- **Auto Makeup:** compensates for the drop in output level caused by gain reduction. It determines the theoretical compensation at a given threshold / ratio setting – for instance -20dB and 4:1 gives 5dB – and applies half of that value (2.5dB) to achieve the same *perceived* loudness.

Such precise ratios only apply in feed forward (FF) mode. In FB and INT modes, the ratio is program-dependent – the input-output curve displayed in FB and INT modes is only an approximation. The same applies to the visible effects of *Soft Knee* and *NonLin* settings.

## Envelope

Audio-range compression is a time-variant process, it uses a response-time envelope. Otherwise we would simply be doing non-linear amplification... better known as 'distortion'! The envelope determines how fast the device reacts to audio input, and how it recovers.



Of course manually finding the best settings is important, but we tried to make Presswerk sound great with any settings by including algorithms that automatically 'ride' the audio and adjust various parameters. We didn't take the idea too far, though – you can still control the following:

- **Attack:** the time it takes for gain reduction to take 'full' effect (typically 90%). Short attack times make compression kick in almost instantaneously, but to keep the sound natural, longer times are allowed. Do you notice the smack of drums (or vocal plosives) when you increase the attack just a little? With longer attack times, we recommend enabling *DPR*, as this usually delivers the most natural results.
- **Release:** the time it takes to recover from compression, to about 63%. This is often a critical setting when the audio material is complex. If *Release* is set too short, bass-heavy material will distort. If set too long you will hear the famous/infamous *pumping* effect.
- **Adapt:** as finding the optimal release time can be tedious, allowing automatic adjustment is often your best choice. This saves time while delivering pretty good results. Higher *Adapt* values mean a wider 'window' around the nominal release time. Short, percussive signals will lead to shorter release times, while smooth passages will make recovery much slower.
- **RMS Window:** RMS (Root Mean Square) is a common mathematical method for approximating how our ears perceive loudness. *RMS Window* adjusts the time allowed for sensing the incoming audio: At minimum, the detector has a 'true peak' response, meaning the absolute level will always be used to trigger the compression. The longer the window, the more the 'steady' portion of the signal will be used.

For most material, values between 1 and 5 ms should be fine. Extremely short settings or instantaneous peak response can be used for limiting or for more 'creative' purposes. *RMS Window* greater than 10 ms can be used to emulate the lag of opto-electronic compression units, and can help treat difficult material such as vocals or piano as naturally as possible.



## Sidechain

The *Sidechain* panel governs what the detector ‘sees’. Often, sidechain refers to an external input on a compressor device, where the term ‘keying’ is used to describe what’s happening: the external signal takes control of the compressor’s action. Typical uses are ‘ducking’ the music whenever a DJ’s voice comes in, or having a drum pattern add rhythm to synth pads.

On Presswerk, both origins – the plug-in’s input as well as an external signal – can feed the detector, and their levels (balance ratio) can be determined freely.



**Important:** Routing a second signal (‘sidechain’) into Presswerk requires that the host application actively supports sidechaining – please consult the documentation that came with your DAW.

- **Int:Ext:** this bipolar knob adjusts the balance between internal (Int) and external (Ext) audio. As *Int* means the regular audio input, Int:Ext is set to minimum by default so that only the internal signal is used for driving compression.
- **Delay:** bidirectional delay time. Negative values mean traditional ‘lookahead’ sidechaining (with fewer transients), while positive values mean that the sidechain is behind (for more transients).
- **Low Cut:** gently cutting out the lower frequencies from the detection signal can add punch to rhythmic material, it can make mixes louder and less prone to unwanted triggers.
- **12dB:** the default slope of both filters is 6dB per octave, and activating the *12dB* button doubles this value. Tip: Shallow slopes are often better for wide frequency transitions and natural compression. Steep slopes isolate signals more ‘surgically’ and require more careful adjustment.
- **High Cut:** cutting mid to high-frequencies from the detection signal prevents spurious clicks and other transients from triggering the compression.

A combination of low and high cut can narrow the sidechain down to a certain frequency range e.g. in a de-esser, compression is only applied in the upper mids / treble.

## Sidechain Monitor

The controls in the *sidechain monitor* are used for auditioning the sidechain signal:



Sidechain monitoring is useful in the following scenarios, for instance:

- The bass drum is used to slightly duck a sub-group of rhythm guitars, making them 'breathe' a bit. Finding the optimum balance between the contribution of the internal and external signal to the compression effect is easier if you can audition them separately.
- The snare drum track contains too much bleed from the kick and cymbals to be compressed cleanly, so the filters need to be adjusted with special care.
- In M/S mode, precise adjustments need to be made independently for each channel.

Soloing a channel makes the signal appear as a mono sum in both outputs. The volume can be adjusted using the *Level* knob. Like on most mixing desks, however, the solo buttons are additive i.e. switching one on doesn't automatically switch the other off. When both buttons are active, the monitoring is practically 'solo in place', retaining the stereo position.

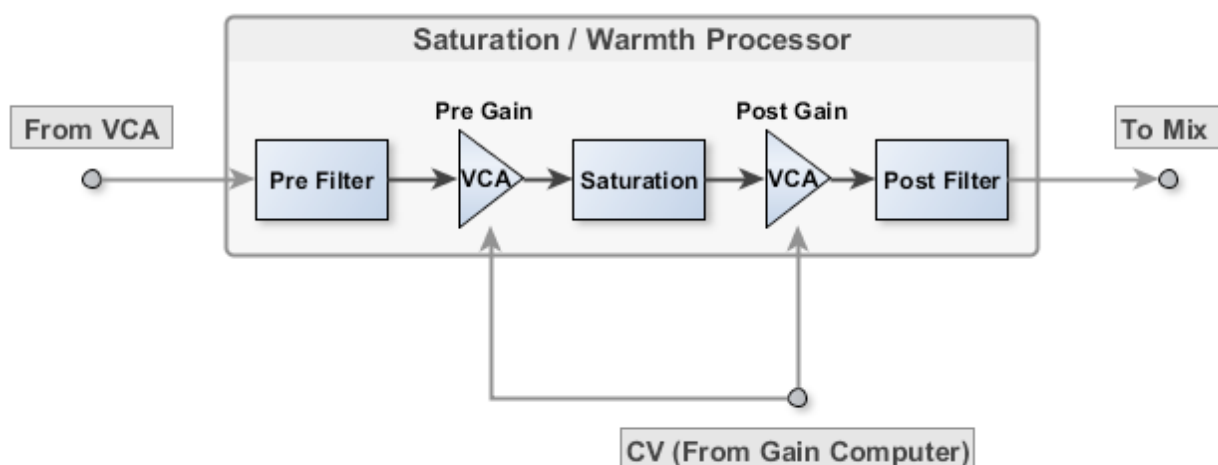
Active buttons flash as a constant reminder that you are only listening to the sidechain.

## Saturation

We are leaving the realm of pure, transparent compression, and are now entering some serious 'colouration' territory here...



The saturation stage consists of a pre filter, a non-linear amplifier with distortion, and a post filter:

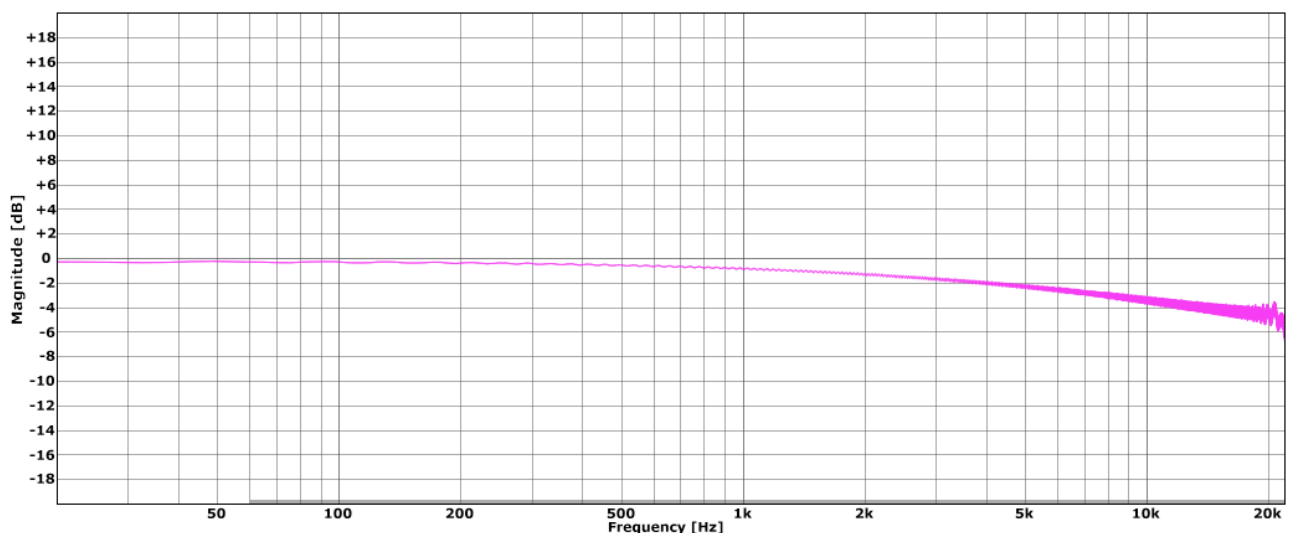


In moderation, saturation can help tame harsh transients or left-overs from the compression process. It can also warm things up. We have control over the two most important factors: *amplitude* and *frequency*. Amplitude comes into play as non-linear amplification, either as a static gain factor or as a user-definable amount of dynamic movement. Frequency comes into play as the ability to saturate (distort) certain parts of the audio spectrum more than others.

- **LED:** a small lamp to the left of the label indicates the amount of saturation.
- **Pre / Post:** selects whether saturation happens before (pre) or after (post) compression.
- **Amount:** a master gain control for the saturation stage. Turning *Amount* up overdrives the non-linear amplifier, adding odd-numbered harmonics. This sounds similar to an overdriven 'class AB' tube stage or an amp that uses FETs as its input transistors.
- **Dynamics:** if set to minimum, amplification remains constant. When turned all the way up, the *Amount* is multiplied by the inverse of the compressor's gain reduction – so extra saturation is only generated if compression is actually happening.
- **Warmth:** Shifts the energy towards the treble end of the spectrum – but this does not mean a brighter sound. On the contrary, *Warmth* causes lower frequencies to be removed from the saturation, so the treble will be saturated more strongly.

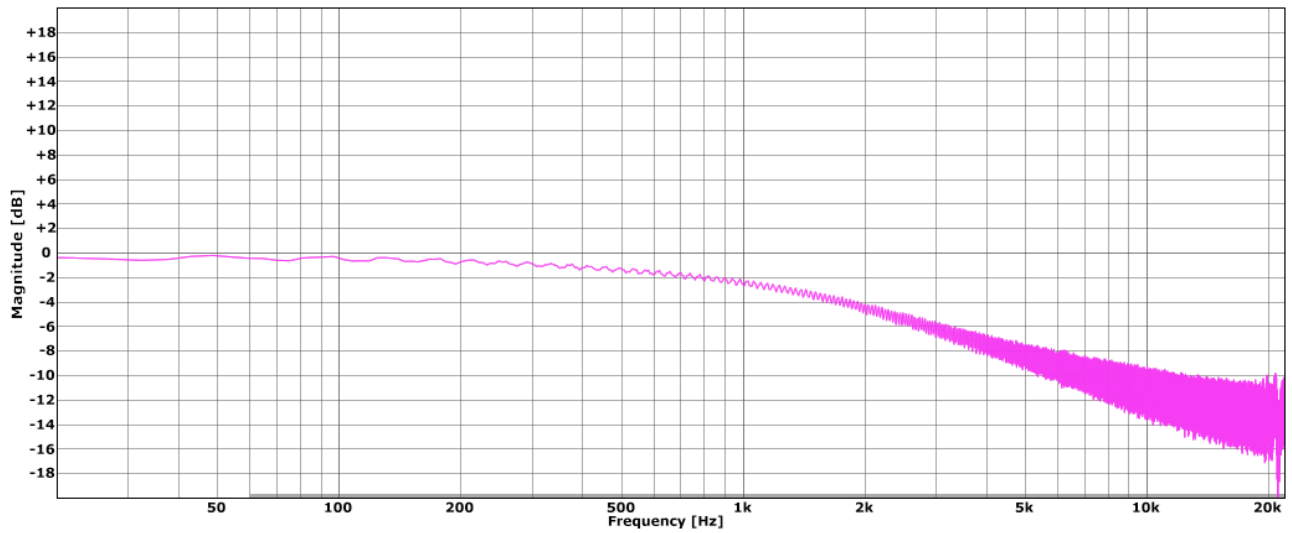
Warmth is a kind of **tilt filter**: After saturation, the inverse filter curve is applied so that the frequency response remains flat when *Amount* is at minimum. With Warmth turned up, the higher the *Amount* (i.e. gain / distortion), the more the treble is squashed while low frequencies remain intact. The result is like tape overdrive in that it removes harshness in a similar way.

Note: Adjusting saturation via *Amount* and *Dynamics* makes the compression effect more lively, more organic. It's actually quite easy to mimic the distortion behaviour of the FET-based voltage dividers found in popular studio gear of the late 1960s.

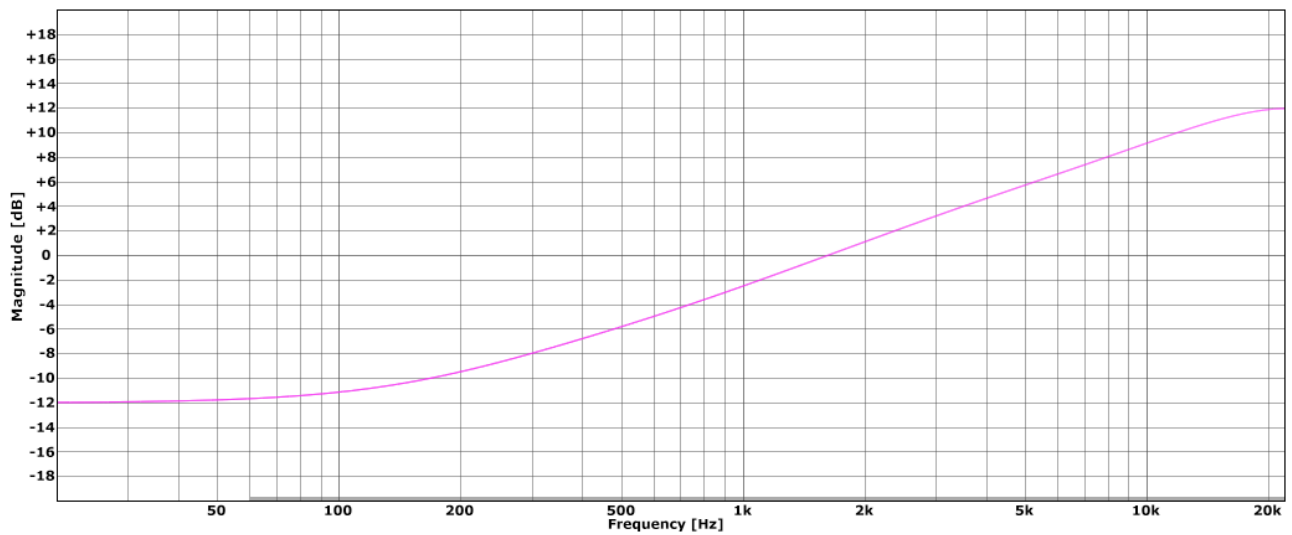


*Saturation amount +5dB combined frequency response*

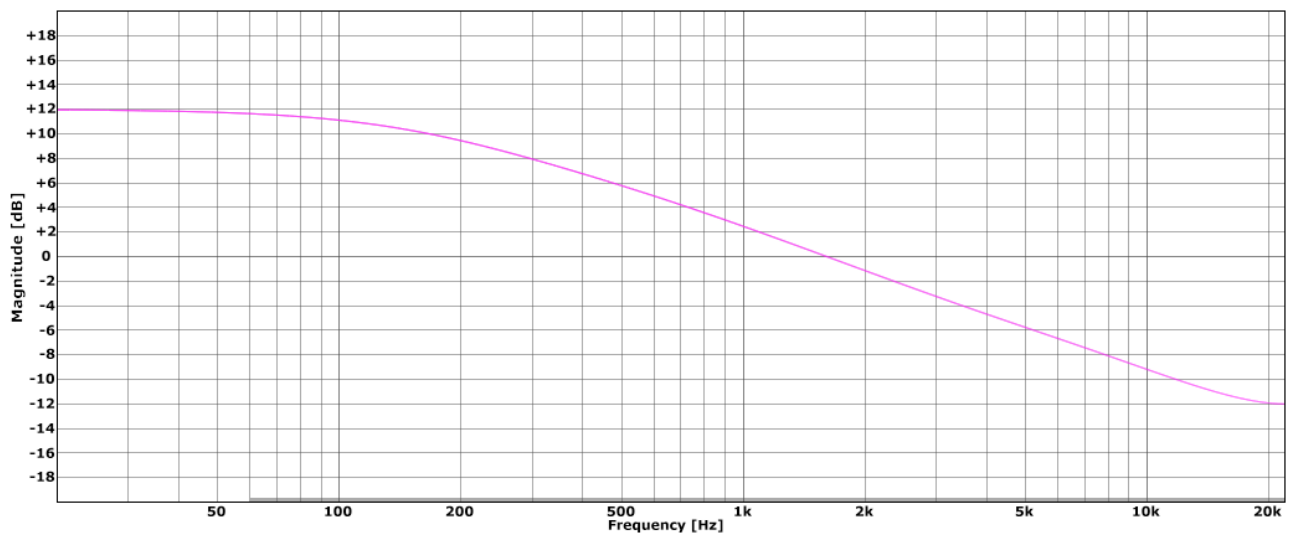
More plots on the next page...



*Saturation amount +15dB combined frequency response*



*Saturation pre-filter warmth +12dB*



*Saturation post-filter warmth +12dB*

## Mix

Presswerk is capable of parallel compression internally. Although it is generally used as an insert effect, some of the untreated (dry) signal can be mixed in with the compressed signal...



At first glance, mixing dry signal back in seems counterproductive. However, this is quite a popular trick as it keeps the dynamics fairly intact and makes the overall effect more transparent. Especially with 'difficult' audio material, parallel compression is often the key to achieving the most natural-sounding result. As long as the untreated signal keeps everything pretty much in shape, tracks can be compressed quite a lot.

Note: Some engineers compare parallel compression to upward compression – instead of attenuating everything above the threshold, everything below it is boosted.

The MIX controls are:

- **Dry:Wet:** cross-fade between untreated (dry) and compressed (wet) signal.
- **Level:** a makeup control for the wet signal only.
- **HP:** a high-pass filter applied to the dry signal. This adds a 'freshener' to the concept of parallel compression, and is great for signals that tend to lose clarity when heavily compressed. Try this on vocals: Turn HP up quite high and use Dry:Wet to mix a bit of the high-pass filtered dry signal back in. The voice should now sound more 'airy', as if it were processed with an 'enhancer'.
- **Expand:** when this is switched on, the dry signal is multiplied with the inverse gain reduction, creating downward expansion and, at higher ratios, gating effects. In conjunction with the HP filter, Expand can make the sound a lot more lively.



# Special Views

Presswerk is a typical u-he product in that it sounds great and is super-flexible. But now and again you might prefer to have fewer ‘distractions’: Simpler panels with fewer controls, faster workflow, signature sounds... that’s what Presswerk’s special views are all about.

It’s like having a bundle of different compressor plug-ins, except that you never need to swap them out. You can check out the various compressor types by browsing the presets, or you can initialize one of the special views and quickly dial in the settings to suit your track. And should you ever need more detailed control, simply exit to the **Main View** by clicking on the ‘house’ icon.

IMPORTANT: While the Main View easily adopts all Special View settings when you exit, the other way round i.e. switching from Main to Special wouldn’t work: Many of the parameter settings from the Main View would get lost in the process. That’s why we decided to implement the special views as *initialization options*... which makes switching back and forth conveniently impossible!

## Easy Compressor

As the name suggests, this general-purpose option is probably the best choice whenever you want to quickly even out the dynamics of an instrumental track, resulting in a more compact signal...



While THRESHOLD and RATIO are also available in this view, a RESPONSE knob replaces the more common pair of Attack and Release knobs, which can be difficult to balance.

A fast Response means that the plugin not only catches up quickly with transients, but also recovers quickly. Conversely, a slow Response setting allows for more time to pass before anything happens, and recovery is also relatively slow. The Easy Compressor comes with a fairly short detection window and a good portion of ‘adaptive release’, which minimizes unwanted amplitude ripple and/or intermodulation artifacts when the Response is set very fast.

An optional SOFT CLIP gently limits strong output signals, thus avoiding digital clipping.

Although the Easy Compressor is suitable for any kind of audio material, we found it particularly good on electric guitars and basses!

## Vocal Compressor

The ideal compressor for even, powerful vocal dynamics should have as few controls as possible and plenty of automatic behaviour. Adjusting a voice manually is a delicate task, it is often a game of hit-and-miss. So, like in the Easy Compressor, we simplified attack and release by combining them in a single 'musical' RESPONSE control aimed chiefly towards vocal compression.



The compression threshold is internally fixed so the INPUT knob is used to drive the compression process directly as well as for easily setting up operation levels. Unlike the Easy Compressor, the compression RATIO can be adjusted manually.

We also added an ENHANCER control. This blends in a highpass filtered version of the dry input signal with the processed part, so that even highly-compressed vocals will stay clear and 'airy'.

The whole compressor can be switched into a DE-ESS mode at the push of a button: The detection circuit then reacts mainly to *sibilant frequencies* ('s' sounds, upper mids / treble).

## Drum Compressor

A good compressor for drums always needs two characteristics: First, it must be lively and responsive. Second, it must be able to give you lots of coloration and fatness as fast as you want it!



Set at a fixed threshold, the Drum Compressor view's INPUT knob governs the overall signal level as well as the point where compression starts to kick in.

Use the RATIO control to dial in 'how much'. Notice how the audio is slammed way down, with the gain reduction meters bouncing all the way but without causing any break up.

The RESPONSE and SATURATION can be adjusted to seriously thicken up your drums, making them 'smack' and 'bite'.

Using the SIDECHAIN filter control to take the low end out of the detection creates some extra oomph for e.g. kicks and overhead or room microphones.

An optional SOFT CLIP gently limits strong output signals, thus avoiding digital clipping.

For some of that New York flavour, try bringing a bit of the direct signal back in by backing off the MIX value: The transients should stay mostly intact while all that lovely ambience, rumble and noise are greatly accentuated.



## Bus Compressor

The Bus Compressor offers a similar feature set to the Drum Compressor. Possible applications also include drums, but this View was tailored to suit complex bus or group material – to deliver that oft sought-after 'glue'!



The Bus Compressor sounds less aggressive, more neutral than the Drum Compressor. This view gives you regular access to the ATTACK and RELEASE parameters, which are often necessary for more complex signals.

An optional SOFT CLIP gently limits strong output signals, thus avoiding digital clipping.

The ADAPT button switches adaptive release on. When active, a modicum of automatic behaviour helps you get difficult material sitting right, while still leaving you enough manual control over the envelope.

The RATIO is selectable in three steps, which covers all practical settings for this model and gets you 'on track' very quickly.

## M/S Program Compressor

While all the other special views offer a single set of parameters for both channels, the M/S view provides two sets of controls. It is aimed at independent manipulation of mid- and side-signal information of a stereo source.



In the hands of a skilled audio engineer, M/S processing is a powerful tool. It requires a certain degree of background knowledge and practice. With Presswerk's M/S Program Compressor view, however, it all becomes so much easier...

There are no Threshold or Ratio controls here. Use the **Input** knobs to set up a basic level, then tweak the **Compression** (which adjusts both traditional parameters at once).

Once again, **Response** combines Attack and Release in a single control, making adjustments to the envelope a breeze.

The **Sidechain** filter control actually processes the two channels independently, although you only see a combined set of knobs.

The rectangular TOUCH LINK pads between the upper and lower row of knobs are for controlling both channels at the same time. The knobs above and below are ganged together, although any differences between the two values are retained. To move them in opposite directions, hold [Cmd] (Mac OSX) or [Ctrl] (Windows) before clicking on the pad.



## Limiter

Setting up Presswerk as a sum / master brickwall limiter is perhaps not such an easy job. Such a process is very delicate in terms of transient control, audible distortion as well as envelope smoothness. But with the dedicated **Limiter** view, we reckon we managed it. Use this one just like any brickwall limiter unit: Kick it, make it loud!



Three modes give you access to distinct flavours:

'Smooth' uses enough look-ahead (1.25 ms) to catch up with fast transients, and an overall smooth release envelope behaviour. Distortion is minimum with this mode.

'Punchy' is quite the opposite with a very short (0.35 ms) look-ahead, longer attack (0.2ms) and DPR switched on. It lets some transients slip through the limiting process (to be fed to the succeeding clipping stage) and recovers much faster than Smooth mode. Bass frequencies are pronounced.

'Balanced' is a healthy compromise between the two, and is probably the best mode to start with for most sources. No DPR.

Tip: Use THRESHOLD to set the point where you want limiting to start. Notice how the output volume rises by the same amount. Also notice how the gain-reduction meters drop accordingly: The meter action is a function of limiting, but also shows any additional SATURATION.

Watch out that you don't overdo brickwall limiting. Although it creates minimum artifacts, this view can easily push things into 'insanely loud' territory!

# Configuration



The cogwheel icon at the top of the righthand side-panel is your gateway to the global configuration pages, which let you adjust the window size and brightness as well as most of Presswerk's parameters via MIDI remote control. A row of selectors appears for MIDI Learn [L], MIDI Table [≡], Preferences [tools] and Close [X].

## MIDI Remote Control

This page is for assigning MIDI CC ('continuous controller' or 'control change') to individual parameters in Presswerk. The CC data may be generated by hardware knobs / sliders, or come from automation tracks in the host application. For how to route MIDI data into effect plug-ins, please refer to the documentation that came with your host application.

## MIDI Learn

To open the **MIDI Learn** page (if not open already), click on the [L] button. The Main view should look something like this:



*the Configuration / MIDI Learn page of the Main view*

While MIDI Learn is active, the window has a dark translucent overlay. All MIDI-learnable controls appear as outlines, and those that are already connected to a MIDI CC are highlighted (like two of the knobs in the above image). To MIDI-learn another control, click within its outline and move a knob or slider on your hardware controller.

Switch to the MIDI Table (see below) to check that the new connection has appeared in the list, then click on the large **[X]** at the top right to exit the configuration pages. If you don't want the new connection to be permanent, remove it from the MIDI Table first.

## MIDI Table

This page not only lets you define MIDI CC assignments 'manually', but also lets you specify a few details about the type of hardware control Presswerk should expect:

	Parameter	Channel	Controller	Mode	Type	
1	Threshold 2	1	2	normal	Continuous7bit	X
2	Threshold 1	1	2	normal	Continuous7bit	X
3	Ratio 2	1	11	normal	Continuous7bit	X
4	Ratio 1	1	11	normal	Continuous7bit	X
5	-not assigned-	1	1	normal	Continuous7bit	X

The **Parameter** field shows the assigned parameter. Click to select a different one.

The **Channel** and **Controller** fields specify the MIDI channel (1 to 16) and CC number (0-127).

The **Mode** setting specifies the range and/or resolution of values...

- Normal: full range, continuous
- Integer: full range, whole numbers only
- Fine: 0.01 steps between the two integers closest to the current value

... and **Type** specifies the type of hardware (by far the most common is Continuous 7-bit)

- Encoder127: unipolar encoder
- Encoder64: bipolar encoder
- Continuous7bit: 7-bit MIDI CC (normal resolution, common)
- Continuous14bit: 14-bit MIDI CC (high resolution, rare)

## Adding more assignments

Either MIDI-learn them as described above, or click on the **Add** button at the bottom of the window then select the Parameter, Channel etc. from the options 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.



## Preferences

Clicking on the ‘tools’ icon open the Preferences window, in which you can set several global defaults to suit your computer mouse and monitor:

How you set the **MouseWheel Raster** parameter depends on whether your mouse wheel is rastered i.e. whether you can feel it clicking slightly as you roll the wheel. If so, set YES so that each little click will increments / decrement in sensible value steps.

**Curve Display** affects the appearance of the compression curve: *eco*, *fast* or *glow* (in order of CPU usage).

**Default Size** sets the GUI size for each new instance of Presswerk. Note that you can temporarily change the GUI size without entering the *Preferences* page – simply right-click in the background.

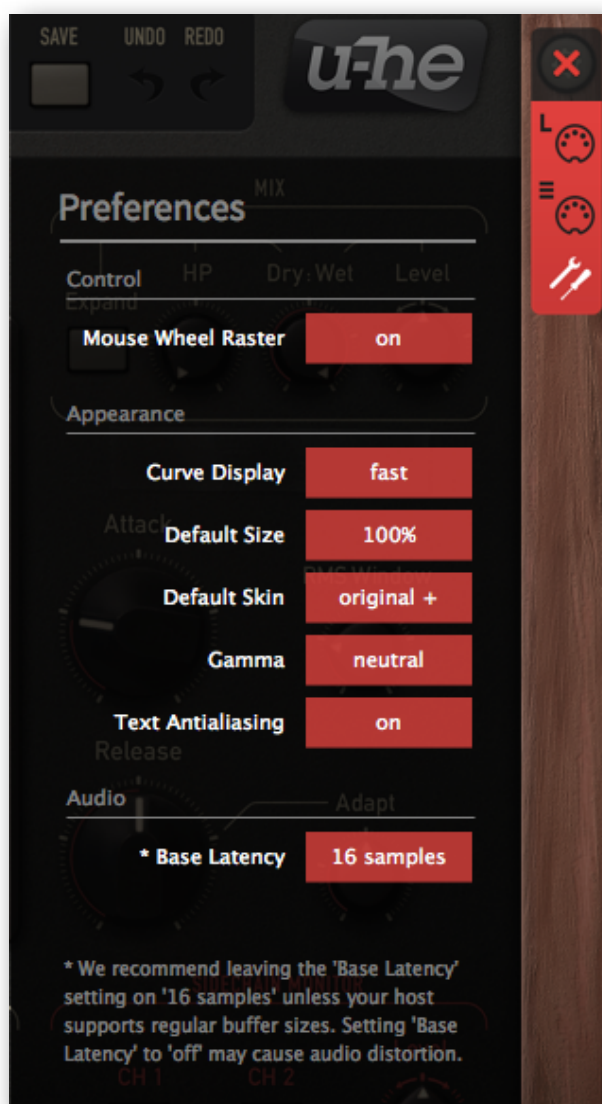
**Default Skin** gives you two options in this version. With *original* + selected, the values of parameters in the ‘other’ channel are indicated by red arcs (for knobs) or underlines (for buttons).

**Gamma** determines GUI brightness.

**Text Antialiasing** switches the smoothing of labels and values on or off. Normally left on, in certain cases switching it off will improve readability.

**Base Latency:** If you are certain that your audio system – hardware as well as software – uses buffers that are a multiple of 16 samples in size (please refer to the respective documentation), you can safely disable Presswerk’s base latency. Otherwise leave it set to the default ‘16 samples’ to prevent crackles.

Note that a new Base Latency setting will only take effect when the host allows e.g. on playback or after switching the sample rate. Reloading Presswerk will always work.



### MORE ABOUT BUFFERS

Internally, Presswerk processes audio in chunks of  $n \times 16$  samples. This so-called ‘block processing’ method significantly reduces the CPU load and memory usage of all our plug-ins.

If the number of samples to be processed is say 41, Presswerk processes the first 32 and keeps the remaining 9 in a small 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 either the host or audio driver processes ‘unusual’ buffer sizes. In the many host applications that process buffers of e.g. 64, 128, 256 or 512 samples (all multiples of 16), try switching it off so that Presswerk can process latency-free.

# Specifications

## Main compression parameters

- Threshold: -60dB – 0dB
- Ratio: 1:1 – 20:1
- Attack Time: 0.1ms – 150ms
- Release Time: 15ms – 2500ms (with 'Adapt' set to 0%)
- Detector integration time (RMS Window): 0.1ms – 30ms
- Look-ahead delay: 0 – 5ms
- Sidechain delay: 0 – 5ms

## Sidechain filter

- High-pass corner frequency: 10Hz – 10kHz
- Low-pass corner frequency: 100Hz – 20kHz
- Filter slope (selectable): 6dB or 12dB per octave

## Saturation stage

- Saturation amount: -40...+40dB (gain factor)
- 'Warmth' corner frequencies: 500Hz and 5kHz, 'Tilt'-type filter (pre- and de-emphasis)

## Sample rates

44.1kHz – 192kHz (HQ mode off), 176.4kHz – 192kHz (HQ mode on)

## DPR frequencies

70Hz (input stage), 450Hz (output stage)

## Latency

17 samples (typical, using Base Latency = 16 samples + look-ahead delay)



## Signal Flow

