

# **preFIX**

**MANUAL**

revision 1.0

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# 1 Introduction

## 1.1. License

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## 1.2. Installation

Requirements:

- Win32 compatible system with SSE2 (or higher) instruction set support
- Tested and known to work in many VST compatible hosts

Put the DLL file contained in this archive in the VST plug-in folder of your host.

## 1.3. Overarching topics

**Warning:** Lower your listening volume while operating the plug-in to avoid hearing damage or damage of speakers or any other equipment.

**Note:** Each section contains a separate on/off switch on the bottom which must be powered on to run each module.

Usage tips:

- Use your host's switch to toggle the plug-in on/off for A/B comparisons
- Use <ctrl> + mouse left click on a knob or switch to restore default position
- Use <shift> + mouse left click on a knob to fine adjust values
- Use this plug-in as an insert effect in any (stereo) channel of your VST host

## 1.4. Credits

Thanks to *Patrick Barca* for insisting on this plug-in idea and for contributing his outstanding UI artwork again!

Special thanks to Dax Liniere and *susiwong* for helping on some crucial concept design choices.

Thanks to *trogluddite* for sharing his goniometer code.

Thanks to *Andrew J* on sharing all his insights about oversampling.

Special thanks to *Dozius* for all that community support and inspiration.

And many thanks to all the beta testers of course.

# 2 Overview

## 2.1. *preFIX* at a glance

*preFIX* – getting those alignments done.

*preFIX* is a pre-mixing and audio alignment tool which typically takes place upfront the mixing process. It provides a clever tool set to clean-up, fix and align audio tracks (typically taken from recordings) concerning overall frequency correction, phase alignment, spatial stereo field corrections and routing. It contains a complete gate/expander solution with a dedicated and comprehensive sidechain filtering path as well.

### Highlights

- smoothest audio frequency filtering
- comprehensive gate and expander audio treatments
- detailed phase corrections
- easy routing and stereo imaging changes

### Plug-in specification

- Win32 / VST compatible
- state-of-the-art digital signal processing
- performance-critical parts are written in assembler
- completely SSE optimized

## Features

- **Main audio path filter**
  - Baxandall style shelving filter with pristine audio quality
  - Smooth Butterworth high- and lowpass filter with switchable characteristics (12 and 24dB per octave)
  - Both are oversampled and match their analog model curve behavior
- **Sidechain path EQ and filter**
  - Whole spectrum “tilt style” balancing filter with adjustable center frequency
  - Smooth Butterworth high- and lowpass filter with switchable characteristics (12 and 24dB per octave)
  - A dedicated parametric peaking EQ
  - Internal/external sidechain switchable
  - Sidechain listening option
- **Gate/Expander**
  - Adjustable threshold between -80 and +6dBFS
  - Freely adjustable knee from hard- to soft-knee behavior up to 1:2 downward expander mode
  - Range limiting option (floor)
  - Channel link switchable
  - Envelope follower section with attack, hold and release control
  - Attack timing features console style peak timings as well as two rms modes
  - Additional gate pre-open timing option
- **Phase alignment**
  - Analog style step-less signal phase corrections
  - Detailed options for adopting phase response curve regarding polarity, frequency center and width
  - Additional digital signal delay option
  - Phase control switchable to channel 1, 2 or both
- **Advanced control**
  - Detailed output channel routing with six different modes
  - Stereo image rotation option visually supported by a goniometer
  - Output level control with special mono mode to mix in channel 1 and 2

## 2.2. Getting started

### Overall signal path flow



The overall main audio path routing **is not** strictly from left to right through the interface but goes according the following order:

1. EQ
2. Gate
3. Phase alignments
4. Output routing and stereo imaging
5. Output level control

The routing for the SC signal path takes its input directly from the plug-ins inputs 1+2 (internal sidechain) or 3+4 (external sidechain) which are routed then through the SC EQ section and directly into the gate/expander.

If the sidechain listen function is activated then the gates audio output signal is dropped and replaced by the SC EQ output. All processing afterwards in the main audio path (phase alignment, routing etc) still applies!

**Note:** Each section contains a separate on/off switch on the bottom which must be powered on to run each module.

# 3 Advanced

## 3.1. The modules explained



### Main audio path filters

There is one EQ/filter section fixed in the main audio path (the left most one).

That one is for the main audio path only and is a modern Baxandall EQ adoption but with freely adjustable frequency and gain settings. It also offers smoothest Butterworth high- and lowpass filters which can be switched between 12 and 24dB per octave characteristics.

This whole section is oversampled for best quality and precise curve match even in the highest register. Its an incredibly good sounding unit and you can't harm any incoming audio by dialing in these (shelving) filters. All frequency readouts are in Hz.



## SC path filters

Another EQ/filter is permanently located in the SC path (titled SC).

This one only works on the gate/expander sidechain path and offers a 12/24dB HP, a peak/notch filter, the “tilt style” whole spectrum balancing filter (with additional possibility to adjust the center frequency) and a 12/24dB LP.

This EQ lane can be listened to with the little speaker symbol switch. External sidechain is also fully supported by the EXT/INT switch. Note that for external sidechaining your plug-in host must route a proper audio sidechain signal to input 3 and 4 of this plug-in.



## The gate/expander

Adjusting the transfer curve with the 3 sliders:

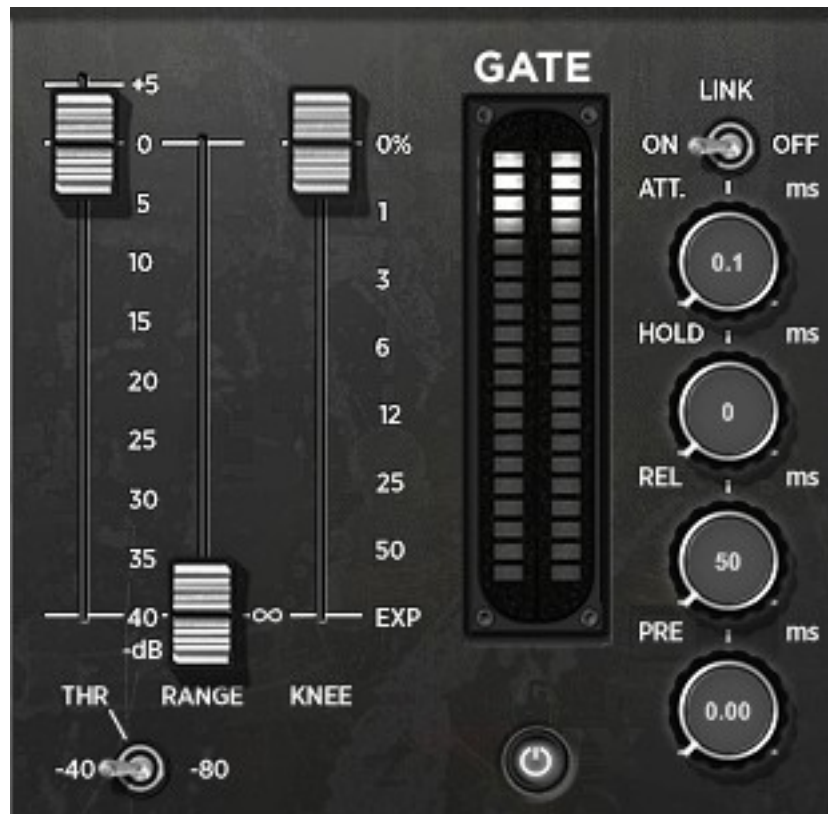
- TRH sets the threshold level at which gating/expanding should begin.
- RANGE limits the range of the gating/expanding (also known as "floor"). In bottom position this function is off (no range limiting).
- KNEE sets the curve characteristic from instant/hard (0%) down to a smoother knee curve. In bottom position the curve represents a 1:2 downward expander curve with smooth soft knee.

Adjusting the timing:

- ATT: switches between five fast peak signal detection timings (0.1 – 1.5ms) into two different RMS detection methods (5 and 15ms). The peak timing behavior relates to some console channel gates. The RMS timings can be useful to easily avoid fluttering of the gate on a micro dynamics level (e.g. when gating toms in the release tail) or to better catch a snare in a whole drum mix for example.
- HOLD: Sets the hold time for the envelope.
- REL: Sets the release time for the envelope.
- PRE: Adds a gate pre-open control so that the gate opens earlier than the audio event actually occurs.

Additional options:

- LINK: Links both channels in the sidechain path.
- 40/80: Sets the threshold level range from 0...-40dB down to -40...-80dB



## The scope section



In the scope section there is a Goniometer included which represents the stereo field imaging. The display can be adjusted with the SIZE and GLUE knobs **which do not affect the audio itself!**

There is a selector knob OUT which allows to select from six different output routing modes. Afterwards in the audio path, the ROT knob rotates the audio signal in the stereo field.

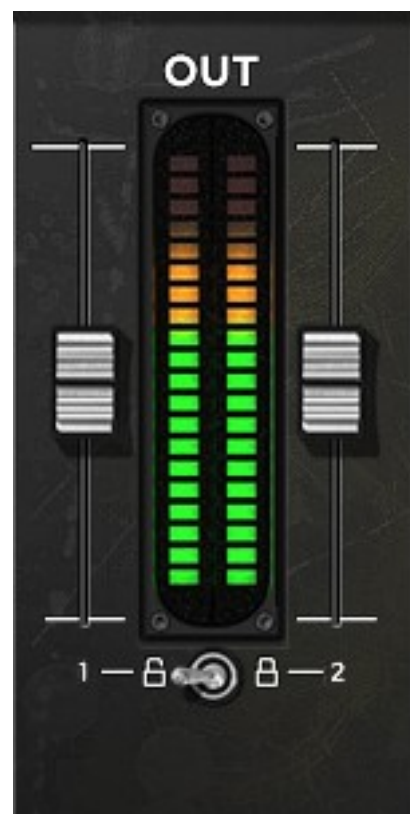
The six routing modes explained:

stereo	Ch1 input routes to Ch1 output and Ch2 input routes to Ch2 output
mono	The mono content of Ch1 and Ch2 are routed to both, Ch1 and Ch2
mono*	Same as mono but the mono conversion is applied <b>after output level faders</b>
swap	Same as stereo but with swapped channels
1->1&2	Routes Ch1 to both, Ch1 and Ch2
2->1&2	Routes Ch2 to both, Ch1 and Ch2

## Output level control

To the right side of the panel two faders are provided for output volume control per channel which is displayed then in the final volume meter. Both sliders can be linked.

Note: In **mono\*** mode both sliders are active on each channel upfront converting the output to mono.



## Phase alignments



The phase alignment tool basically matches an analog phase shifter with some additional digital options.

This section provides:

- A channel switch: Sets the phase operation to Ch1 or Ch2 only or both.
- DLY: Introduces a digital signal delay up to 4ms.
- PHASE (the knob on the right side): Alters the phase transition around its center.
- 0/INV: The polarity switch.
- 90/180: Sets the width of the phase alteration.
- LO/HI: Sets the center of the phase transition.

## **3.2. Tips and tricks**

### **Improving the stereo field perception**

Beside the stereo field rotation option in the scope section the phase alignment is the ticket. Just apply this alignments to just channel 1 (or 2) with the switch on the top. Dial in some subtle phase alterations and check stereo field behavior visually with the scope. Check for mono compatibility with the mono routing option. Works great with vocal groups or rhythm guitar sections!

### **Focusing the spectrum and tonality of all tracks**

Applying high- and low-cuts to each and every track is a well known rule of thumb before starting mixing. Though, this should be considered with care especially for acoustic recordings. Some audio lowend content might actually be the frequency body of an instrument and the HF typically contains the overtone structure. So, choose all cuts with care. preFIX provides two filters dedicated for this task in the first EQ section.

### **Dealing with gate release tail distortions**

Might be a difficult task if the audio content is sensible to distortion artifacts, e.g. as with release tails of toms. Utilize the HOLD timing option of the gate in this case or tweak the sidechain EQ to better focus the detector for the gate. Soft knee is also a good option.

### **Dealing with fast gate attack timings and distortions**

If fast attack timings are needed but attack transition becomes a problem try to utilize the pre-open feature of the gate to move the gates opening upfront the audio event itself.

### **If the track needs some “balls”**

Then the EQ is the ticket: Use highpass and shelving filter altogether to boost the LF department while cutting the very lows the same time.

### **If the track needs some “air” or “shimmer”**

Just dial in the pristine sounding Baxandall HF shelving filter – you can't go wrong.

### **Align stacked recordings**

The phase section is the right place to look: Just apply this alignments to just channel 1 (or 2) with the switch on the top of the section.

### **Getting those tight (electronic) drum tracks**

Use the gate, Luke! Use it on the kick, the snare, the hi-hats, on everything. Use it on single tracks and the drum sub-groups. Set each tracks focus with the main EQ filter section on top. Always check for mono compatibility.

### **Mixing a stereo track into mono**

Check out the mono\* option for this task which gives you the opportunity to adjust each channels mixing level **before** everything is converted to a mono signal.

### **Workflow: separate track alignment from mixing**

Start first to check all tracks and if mixing is feasible with them. Some might not match the quality and needs to be replaced. Some just need some basic alignments. Do all creative decisions and treatments later during the mixing process. Mixing is always context dependent.

### **Alignment does not mean restoration!**

If you prepare a track and something like “where is my damn 64dB/octave brickwall filter?” comes to your mind then this might be because you are not upfront any mixing process but are working on audio restoration instead. Or the sources just might be crap. Always remember: garbage in, garbage out.

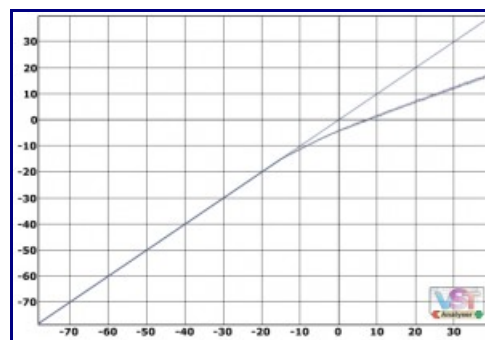
## **3.3. Closing comments**

preFix was developed together with quite a bunch of recording engineers. The specific technical design and feature combination found in this unique audio plug-in might not be self-explanatory in the very first place but I can just encourage to explore all the possibilities to discover how things are working nicely hand in hand. Some hints and applications are already given in the previous chapter.

# 4 Addendum

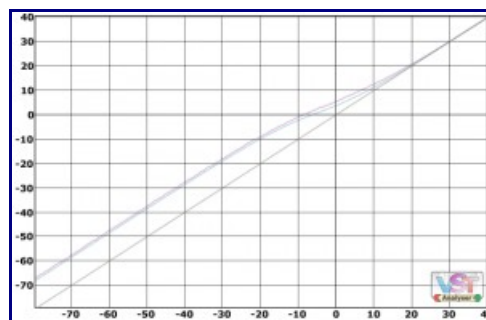
## 4.1. Compressor, gate and expander

Some might get confused sometimes when compressor, expander and gate are discussed and especially when concepts like “upward”, “downward”, “parallel” or such-like are thrown in. Fortunately, things can easily be explained just by looking at the according transfer curves and as an added sugar some more sophisticated insights can be obtained as well.



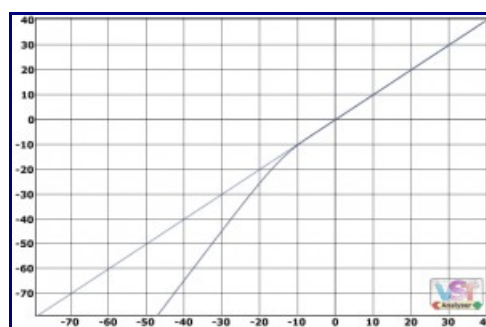
*typical downward compression curve*

The “ordinary” compression transfer curve which is used in most audio dynamic compressors is the downward compression curve somewhat similar to the image above where a 1:1 curve is shown for reference as well (and so in all other diagrams here). It basically works in a way that a signal level above a certain threshold – in this case roughly around minus 10dB - leads to an attenuation which is linear as well (ok, almost – and just in case of a hard-knee compressor) but with a lower amplification ratio. The so-called upward compression works the other way around: A signal above the threshold level remains unaltered level wise but those below are getting amplified to a certain degree now. This is shown in the diagram below where a threshold is set somewhere around +20dB:



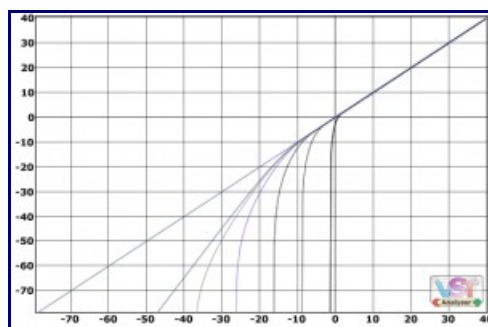
*upward compression curves*

Since the upward compression transfer curve looks quite similar to a \*parallel\* downward compressed one (which is an uncompressed dry signal mixed back into downward compressed one) some might guess that this is completely the same but actually it's not. The difference lies in the fact that envelope curve based dynamic treatments are only partially applied to the parallel compressed signal and the other part remains completely unmanaged. This is different in the upward compressor: the whole signal will be shaped by the envelope curve and therefore the transient response is actually different.



*a downward expander curve*

Opposed to a compressor an expander features basically just different transfer curves as shown in the diagram above where a typical 1:2 downward expansion curve is shown: In this case, below a threshold at around -10dB downward expansion occurs which even lowers the already quieter signals. Upward expansion (not shown there) is just the other way around and preserves the signals below a threshold and amplifies the already louder ones. The downward expansion is often referred to when it comes to the so-called Gate/Expander device. This can easily be understood just by looking at the according transfer curves again which in this case shows that the classic gate curve is just a special case of the downward expander where the ratio becomes infinite:



*different gate transfer curves*

Additionally, other concepts like soft-knee curves (already shown above) or processing range limiting (like a range control in a compressor or a floor setting in a gate) can be implemented just by proper transfer curve designs as well. This leads to the question, if one could implement just one single device for all that three purposes – compressing, expanding and gating – by simply utilizing the very same basic engine and just swapping the respective transfer curves. Well, in theory this idea might be attractive and sound but practise has shown that this does not lead to optimal results. All the other relevant aspects such as time or frequency dependent behaviour or even non-linearity is not that interchangeable between such devices and their specific application domains in general.

#### **4.2. Updates and further information**

Refer to my Blog at <http://varietyofsound.wordpress.com> for some additional information and updates on this plug-in or leave a note there if any issues did occur.

There are already some readings available concerning gate, expander and stuff.

Peace,  
Herbert