



P42 CLIMAX

User Guide

Version 2.0

Forward



The year was 1958. A young high school student by the name of Ed Wolfrum was faced with a personal dilemma that, as it turns out, would lead to a pivotal time in the history of recorded music. He was exceptionally passionate about recording bands, but he was simply unable to afford microphones. This predicament led to the design of a device that allowed him to record instruments directly, without the need for expensive microphones and with the ability to capture a purity of sound that had been previously unheard of. The world's first DI box was born.

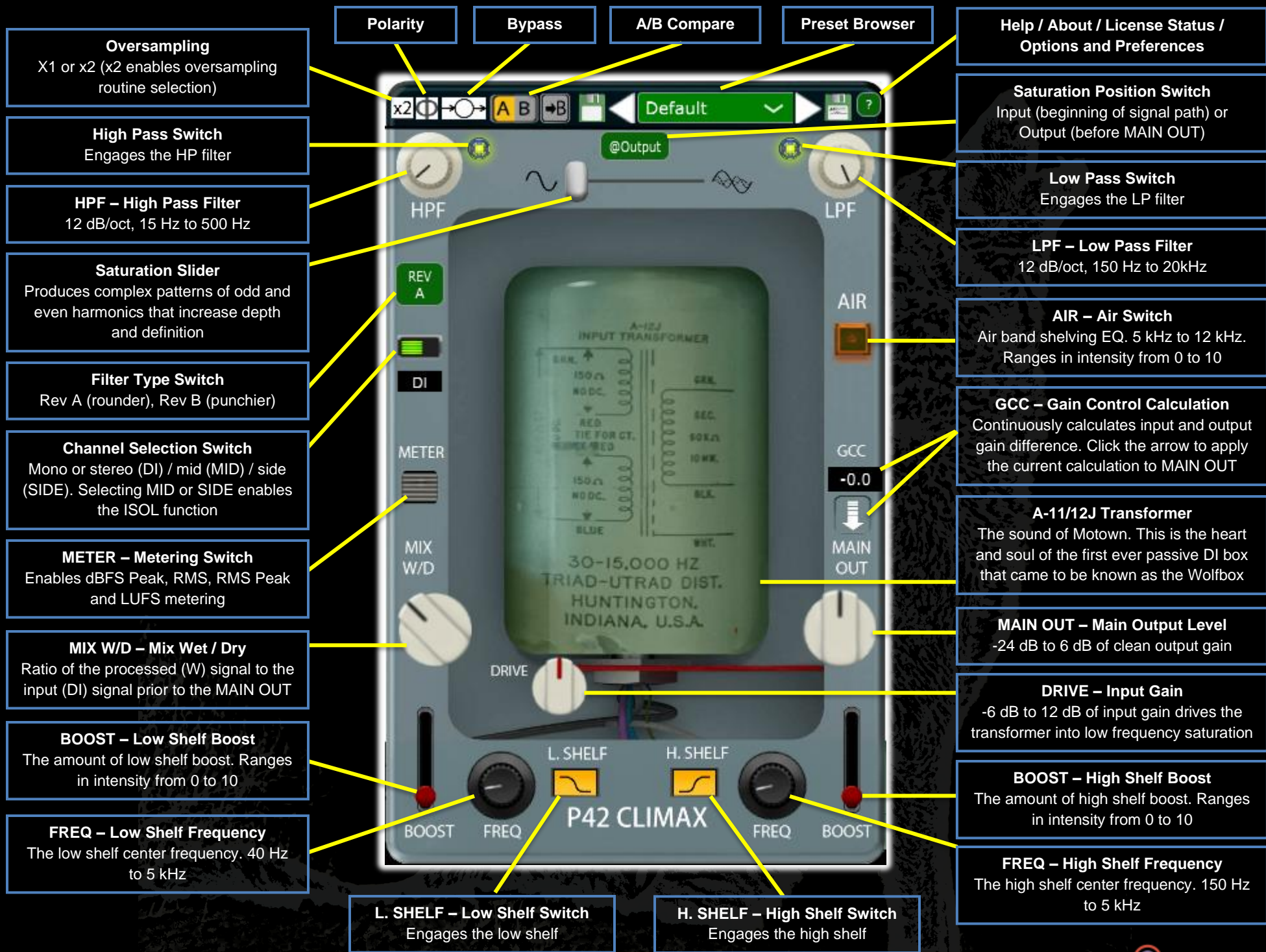
He soon went on to work for Detroit studios like Motown Records and United Sound where his direct boxes were used to capture some of the greatest moments in musical history, performed by artists who were to become legends. This little device was critical in giving Motown its big, distinct, fat bottom, full range sound. At the center of this device, which went on to become affectionately known as the Wolfbox, was the Triad A-11/12J transformer.

Fast-forward to 2020. Pulsar Modular, the audio plugin division of Pulsar Novation, begins work on recreating this elusive piece of history. Originally, this endeavor did not prove to be fruitful and it was put on pause. If the sound is not fully indistinguishable from the target, it is not considered to be a success. Close enough is not good enough. Several months later, a reinvigorated development team once again took on the challenge with a new sense of determination. Once again, countless adjustments were made and seemingly endless variations were carefully evaluated and rejected until finally and almost unexpectedly, there it was. Immeasurable hours and tireless efforts had finally culminated in the unmistakable tone of the Wolfbox DI.

But alas, as with many tales of exploration, this was not to be the end. This is where we find the P42 story just beginning. Pulsar, after all, does not aim to recreate hardware. Instead, it aims to exceed it in every way. Recreation of the Wolfbox was only the foundation. All the different trials and experiments had resulted in a unique set of tools and techniques that would be integrated back into this creation to elevate and shape it into what you are hearing and experiencing today.

P42 is not a DI emulation. It is not a saturator. It is not a collection of filters. It is not a static set of curves and harmonic charts. Don't think of it as any of these things individually. Think of it as a palette of shapes, colors, depth, density and imaging. Let your ears guide you and experience your sound in a way you have not heard before.

In honor of the inspiration that launched this journey, please have a deeper look into the history of the Wolfbox as told by Ed Wolfrum himself. Check out the Wolfbox Story at <http://www.audiographicservices.com/wolfbox.html>.



Oversampling
X1 or x2 (x2 enables oversampling routine selection)

High Pass Switch
Engages the HP filter

HPF – High Pass Filter
12 dB/oct, 15 Hz to 500 Hz

Saturation Slider
Produces complex patterns of odd and even harmonics that increase depth and definition

Filter Type Switch
Rev A (rounder), Rev B (punchier)

Channel Selection Switch
Mono or stereo (DI) / mid (MID) / side (SIDE). Selecting MID or SIDE enables the ISOL function

METER – Metering Switch
Enables dBFS Peak, RMS, RMS Peak and LUFS metering

MIX W/D – Mix Wet / Dry
Ratio of the processed (W) signal to the input (DI) signal prior to the MAIN OUT

BOOST – Low Shelf Boost
The amount of low shelf boost. Ranges in intensity from 0 to 10

FREQ – Low Shelf Frequency
The low shelf center frequency. 40 Hz to 5 kHz

Polarity

Bypass

A/B Compare

Preset Browser

Help / About / License Status / Options and Preferences

Saturation Position Switch
Input (beginning of signal path) or Output (before MAIN OUT)

Low Pass Switch
Engages the LP filter

LPF – Low Pass Filter
12 dB/oct, 150 Hz to 20kHz

AIR – Air Switch
Air band shelving EQ. 5 kHz to 12 kHz. Ranges in intensity from 0 to 10

GCC – Gain Control Calculation
Continuously calculates input and output gain difference. Click the arrow to apply the current calculation to MAIN OUT

A-11/12J Transformer
The sound of Motown. This is the heart and soul of the first ever passive DI box that came to be known as the Wolfbox

MAIN OUT – Main Output Level
-24 dB to 6 dB of clean output gain

DRIVE – Input Gain
-6 dB to 12 dB of input gain drives the transformer into low frequency saturation

BOOST – High Shelf Boost
The amount of high shelf boost. Ranges in intensity from 0 to 10

FREQ – High Shelf Frequency
The high shelf center frequency. 150 Hz to 5 kHz

L. SHELF – Low Shelf Switch
Engages the low shelf

H. SHELF – High Shelf Switch
Engages the high shelf



Oversampling options allow P42 to operate at the host sample rate or at double the host sample rate. White indicates x1 and yellow indicates x2. When operating at double the host sample rate, selection of an oversampling filtering routine is enabled.



Intel. mode (intelligent) scans the full frequency spectrum and attenuates any aliasing signals. The amount of processing applied by this advanced filtering is highly dependent on the signal and the degree to which P42 is being pushed. Aliasing is proportional to saturation intensity as well as level and density of higher frequency content. As a result, processing applied can end up being minimal or if it is heavily used, it is highly selective and as such, this mode will generally be more transparent than the Vint. mode.



Vint. mode (vintage) applies smooth filters to upper frequencies to maintain a classic rolled-off characteristic and allows any aliasing signals to remain unfiltered. This provides the ability to creatively combine a smooth, vintage top end with modern inharmonic distortion. This is most effective when oversampling at a 44.1 kHz or 48 kHz host sample rate.

Tip: Don't assume that an oversampled signal is better. Try both and use your ears. At x1 when working at a sample rate of 44.1 Hz or 48 Hz, there may be some aliasing present, however this very often creates a pleasing color. If you can't hear a difference between x1 and x2, stick with x1 and save the CPU cycles.

Note: Please have a read over the Tips, Tricks and Techniques section of this user guide for helpful information on effectively using oversampling.



Polarity inverts the audio signal. Yellow indicates it is inverted.

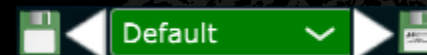


Bypass allows the unaffected audio signal to pass through without being processed. Red indicates audio is being bypassed.



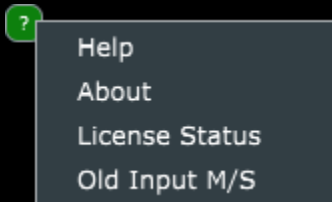
A/B allows for temporary storage of different settings for quick comparison. The arrow button allows for copying the active side to the inactive side.

Tip: When comparing settings, clicking the A/B button will perform the toggle. This is a single button so it is not necessary to move the mouse pointer to alternate back and forth. This makes it easy to compare without knowing which one is selected. We recommend doing this with your eyes closed for maximum focus.



The preset browser allows for modification of the currently selected preset using the save icon to the left of the browser (direct save) or for creation of new presets using the save icon to the right of the browser (save as...). A red asterisk* will appear beside the left save icon indicating the loaded preset has been modified and can be overwritten.

Tip: Modified factory presets will be preserved when updating the software as long as the install presets option is not selected. This is the default installer setting.



Help – Open this user guide.

About – Check the version number or demo expiration date.

Authorize – Manage your license.

Old Input M/S – Enables legacy M/S DRIVE and @Input saturation behavior. Prior to 1.9.2, these were applied to both MID and SIDE signals when one of the two modes was selected. As of 1.9.2 and higher, these are applied to the signal of the selected mode only. This is normally only needed for legacy session compatibility, but feel free to experiment with it!



The signature high pass filter features a 12 dB/oct slope and a frequency range of 15 Hz to 500 Hz. Use in tandem with the low shelf to influence the focus, weight and warmth of the low end of any signal.

The filter can optionally be turned off using the green light adjacent to the knob.

Note: When using a sample rate of 384 Hz or higher, the HPF becomes unstable when set below 25 Hz. P42 will automatically change the HPF to 25 Hz if these conditions are met. Please keep this in mind and plan accordingly.

Tip: Multiple stages of filtering will result in the accumulation of phase shift (a characteristic of all analog filters) and will cumulatively increase gain reduction per octave. The option to turn the filter off can be useful in mastering or mixing situations where external filters are being used and multiple stages of filtering is not desired.

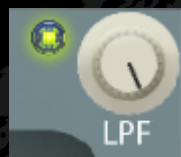


If the saturation position switch is set to Output, the saturation circuit is positioned immediately before MAIN OUT. If it is set to Input, the saturation circuit is positioned at the beginning of the signal path. Output lends itself to smoother, mellower tonal shaping while Input lends itself to aggressive, rich tonal shaping.



The saturation slider influences the amount of saturation applied to the signal. It gradually builds into a complex matrix of harmonics that is generated not only based on the signal, but also by a self-generated saturation chart.

Both odd and even harmonics are generated, dominated mainly by odd order harmonics. The filters, the shelves and the drive control all play important roles in conducting this harmonic symphony. The result is a striking sense of weight, thickness, depth, clarity and presence that is normally reserved for vintage and modern classic hardware.



The signature low pass filter features a 12 dB/oct slope and a frequency range of 150 Hz to 20 kHz. Use in tandem with the high shelf to shave sharp, harsh transients and clear space for harmonic content without introducing dullness or loss of transient energy.

The filter can optionally be turned off using the green light adjacent to the knob.

Note: Please review the note on use of multiple stages of filtering in the corresponding high pass filter control description.



The filter type switch allows you to choose between two filter types. Rev A promotes warmer, rounder treatment of transient energy while Rev B promotes punchier, more forward treatment of transient energy.



Switches processing between mono / stereo (DI), mid (MID) or side (SIDE). The mid and side options are only available when the plugin detects a stereo signal.

When MID is selected, the center is processed and the sides pass through untreated/dry, bypassing the P42 circuitry. When SIDE is selected, the sides are processed and the center passes through dry.

The ISOL switch allows either the mid or side signals to be heard in isolation. When not selected, the full stereo image is always heard and in the case of mid or side processing, includes the processed and dry signals as described above.

Tip: This methodology allows for working within the context of the full mix.

To process both mid and side signals separately, one of two approaches can be taken, both of which are equally as effective. The first option is to use two instances of P42 in series, one set to MID and the other set to SIDE, both with the ISOL switch off. The second option is to duplicate your track (or route your signal accordingly) and use parallel instances of P42, one set to MID and the other set to SIDE, both with the ISOL switch on.

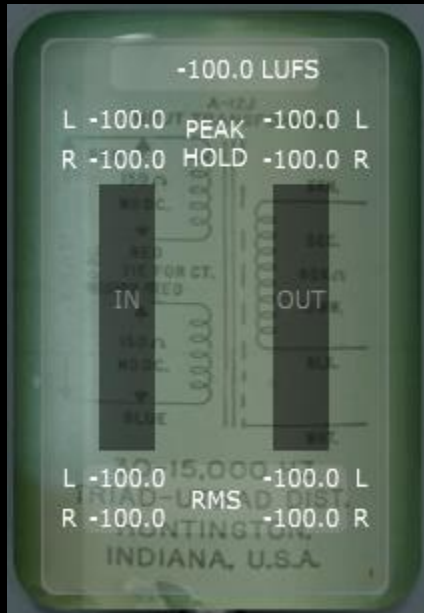


The AIR switch engages a carefully tuned high shelf designed to gently unmask high frequency detail and clarity.

While adjusting the level and frequency, focus on the imaging and positioning of the top end of the signal. When adjusted appropriately, it will have a significant effect on the height and depth of upper frequencies without introducing a hint of harshness, brittleness or sibilance.



The METER switch engages a projected metering heads up display that is superimposed on top of the transformer display.



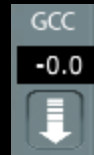
The metering HUD (heads up display) show several pieces of information including LUFS (Loudness Units relative to Full Scale) measurements, dBFS (Decibels relative to Full Scale) PEAK HOLD measurements, and switchable RMS / PEAK RMS (Root-Mean-Square) bar meters and measurements.

The PEAK HOLD measurements hold the highest measured value. They will be reset if any measurements in the HUD are clicked or if the

DAW is stopped/started.

The switchable RMS / PEAK RMS bar meters and measurements can be switched by either clicking on the word RMS or the words PEAK RMS, depending on what is currently displayed. These measurements are updated automatically but can also be reset if any measurements in the HUD are clicked or if the DAW is stopped/started.

Tip: As an alternative to, or in conjunction with the Gain Control Calculation functionality, gain can be evaluated and calibrated visually by relying on the metering and audibly by relying on what you are hearing.



The Gain Control Calculation feature continuously measures the input level and compares it to the output level. The calculated compensation value is shown in the display box below the GCC label.

The compensated level can be applied to MAIN OUT at any time by clicking the arrow below the value display.

Tip: GCC is especially useful when used in tandem with A/B comparison. After applying GCC, copy current settings from A to B, then switch to B and adjust parameters as desired. Now you can apply GCC to B, then switch back and forth while being level matched for eyes-closed comparisons that will not be influenced by loudness differences.



The Wet / Dry mix knob allows for blending a desired amount of dry signal in with the processed wet signal. The output of this stage feeds into the main output.

Note: Fully counterclockwise is 100% wet.



The main output control features -24 dB to 6 dB of clean gain.

Tip: The wide range of attenuation to -24 dB helps to gain stage the signal when the drive is pushed to increase the audibility of the harmonics.



The drive control pushes additional input gain into the transformer which increases the even order harmonics. Use this in tandem with the saturation slider to dial in an ideal harmonic balance that flatters your material.



The advanced filter topology of this low shelf filter features a musical boost that ranges from 40 Hz to 5 kHz. The boosted portion of the signal is followed by a frequency dip that helps to frame the area of focus.



The high shelf filter shares a similar advanced filter topology to that of the low shelf filter and features an ultra-smooth boost that ranges from 150 Hz to 5 kHz. The boosted portion of the signal is preceded by a frequency dip that helps to frame the

area of focus.



Tips, Tricks and Techniques

.: Influencing focus and weight in the low end .:

To either focus or give weight to the low end of a signal, try using a combination of saturation along with the HPF and L. SHELF.

Set the HPF by ear so that it is backed off just a little bit from the point where you hear it affecting the low end of the signal. Set the L. SHELF to be double the frequency of the HPF. Using the BOOST slider, increase the level until you hear the low end gain focus and solidity. Increase or decrease the saturation slider to balance out the upper end of the signal and increase or decrease the sense of depth and thickness. From this point, it is a matter of fine tuning what you are after. Increase the L. SHELF frequency to increase the weight and body of the signal while adjusting the BOOST and saturation according to what your ears are telling you.

A Low Shelf close to double the HPF frequency will help to give focus. Increasing the Low Shelf frequency from there will enhance the body. [KE]

.: Layer instances of P42 to combine complimentary tonal coloration .:

Consider an example of a drum bus where you might want to tease out additional weighty lows and clear highs.

1st instance: Engage the Shelving and get the sound you desire from the bottom end (Low Shelf). In tandem, work with the HPF (so let's say for example you settled for 38 Hz). On the High shelf, get the 1st color you want. Probably get the sound of that snare in the mix to "puck" instead of "poke" sounding. So maybe High shelf at 700 Hz with a light touch of boost. Leave the LPF at max 20 kHz.

2nd instance (inserted after the first one): Disable the Low Shelving EQ (Already you nailed it in the 1st instance). HPF all the way down to 10 Hz (already you nailed it in the 1st instance). Get the second color you want by engaging the High Shelf filter and start searching with the Freq. knob, while playing the music for a color you want to accentuate. Ahh.. say you got it at 1.4 kHz. Now work the Boost till you get it just right. Now turn on the AIR and work the boost level. If you feel it is not doing its job. then Turn it off and initiate a 3rd instance!

3rd instance: Turn off the Low Shelf EQ. HPF at minimum and LPF at maximum. Now turn on the AIR and work the boost level. Optional: engage the High Shelf EQ and scan for other interesting muffled high frequencies, around 3 to 4 kHz. Now work the LPF to smooth things out at the higher end.

Revisit each instance and do the little tweaks to get everything working together. This could mean adjusting the LPF on each instance. [ZS]

.: Conditioning tracks for a killer mix (aka P42 makes every plugin that comes after it sound better) :.

Set up your console for a mix with just this at the top slot and start prepping your tracks for the mix (general hi/low pass and some sweetening). Don't be shy with the saturation or the shelves and filters. Once it's sounding pretty good, mix on top of that. This makes it easy to get a huge sound without having to go to other CPU heavy plugins.

This can add weight/girth/openness/depth in a way that is very solid feeling with a comically low CPU hit.

P42 capable of way more than many initially think. Personally, I went from completely skeptical to "this gets used in every session" in no time flat. [LA]

You can also use P42 Climax on each channel of your drum kit (the tape studer is a great starting point) and then push the kick insert separately to taste. The beautiful boom from beyond and the cumulative vividness of the single channel workflow is most valuable.

When your DAW supports a channel-link function you can build groups and drive those with just one instance of this group. On Cubase it's also possible with selected channels and Alt-Shift plus tweaking a knob in one instance for all selected.

Prefect for driving different groups in your mix.

I usually put the plugin on the group stem, tweak it to taste and then move it on the first group's channel and copy and paste from there for all other channels of that group, then adjust the drive (linked) to fit. That would be the old-school tape machine approach. Then you only have to do smaller tweaks for some individual channels afterwards. [MK]

.: Achieving a wider stereo image with dual mono :.

In DAWs that allow dual mono like Apple Logic Pro and Avid Pro Tools, inserting P42 Climax as dual mono on a stereo track gives a wider stereo image due to the analog behavior of P42. This is not always going to be better because it depends on what you want to achieve. It is simply different, so as always, use your ears to decide based on your material which methodology works best for you. [ZS]

.: Oversample for cumulative benefit :.

There is a lot of information (and opinion) about this topic, but here is a short introduction to it. The Nyquist frequency is a term used in digital audio to describe the upper limit frequency that can be represented when working at a particular sample rate. For example, at 44.1 kHz, the Nyquist frequency is 22050 Hz. Many types of plugins generate additional signals, most notably harmonic distortion, that can very easily exist at frequencies beyond Nyquist. Since they can't be mathematically represented beyond the allowable sample limit however, they 'fold back' and are rendered in the audible spectrum as inharmonic distortion.

Taken on it's own, inharmonic distortion is not necessarily a negative thing. In fact, many people use it to creative effect. It is, after all, just another form of distortion. If it sounds good to your ears, don't allow any rules or mathematical theory persuade you otherwise. This is especially true when using a single plugin or a small number of plugins.

However, build up of any type of signal can sometimes have a cumulatively negative effect on audio. In the case of inharmonic distortion, this cumulative effect may influence the sound stage by shrinking the image and decreasing depth. This occurs because the inharmonic content will blend with an existing element, influencing the frequency content and ultimately affecting both top to bottom positioning and front to back depth perception.

When working with a project at a 44.1 kHz or 48 kHz sample rate with P42 on every channel in a mix (and let's face it, you will be using it on every channel of every mix), engaging oversampling will generally make a significant difference. When oversampling is engaged, the potential inharmonic cumulative buildup will be eliminated and the potential negative influence mentioned above is no longer present. Combined with qualities P42 brings to the individual elements, the result will be a clear sense of front to back depth and top to bottom positioning.

When employing this tip, mix into P42 using x2 from the start because it will help to inform you when considering all decisions that are to follow. The CPU will increase but P42 is a lean, mean, CPU friendly machine even at x2. The CPU usage will depend on your host and hardware, but in tests, P42 is typically hovering around 0.4% CPU at x1 and only around 0.9% at x2. It is all about dimension. [KE, ZS, LA]

.: Be strategic about choosing the most effective oversampling filter routine :.

We have to remember that while mixing you apply EQ differently based on what you hear. It is never advisable to change oversampling options after making final mixing or mastering decisions. Ask yourself why you would make all critical decisions at one setting or sample rate and then switch to another at the end of a mix or when rendering without spending additional critical time evaluating how that might affect your results.

We also have to keep in mind that when it comes to oversampling, a 96 kHz signal will have much less aliases and as such the upsample/downsample routine will sound differently then that applied on a 48 kHz signal that has accumulation of high frequencies due to aliases. This increased aliasing will happen mostly when you are applying hefty saturation.

Oversampling can have a dramatic effect on space, clarity and depth. For example, in Intel. mode, since it is suppressing the aliases across the signal, the bass frequencies will have a bigger space to play in and as we know, lowering the high frequencies is same to the ear as raising the bass freq. This is why x2 gives more dimension to the bass.

Which is better? In mastering, It depends on the song- try both modes and consult your ears.

In Mixing, on individual tracks- if the track has mostly low/mid frequencies then I would insert P42 without oversampling. For high frequency sounds I would use x2 in P42 and place a p42 in x2 with Intel. mode on the 2-bus and I will mix into it (not to be placed as an afterthought).

If you want to apply P42 at the end of a mixing session (as an afterthought), then probably Vintage mode will conserve what you were already hearing when mixing the whole song.

Again, you need to try and see which works best for you and this can be done only during actually using it in several mix or mastering sessions. [ZS]



Managing Presets

Basics

If the option to install presets is selected during installation, updates will overwrite the original presets but custom named presets will remain untouched. Be sure to save your own presets with different names using the save as option (to the right of the preset browser), or alternatively, ensure the preset installation option is not selected when updating the software.

Below the Default preset is the -FLAT preset. This is an ideal starting point for fully shaping your own character from scratch. It initializes P42 with the least amount of signal processing that is possible (no saturation, no shelving).

Fine Tuning Mode

You need to press and hold the modifier key (in Mac: “control, option or command”, in Windows: CTRL or ALT) while moving the knobs.

Special Edition Presets

Special thanks to Synthient Sound for providing his custom presets along with descriptions and suggested uses.

.: **Bass-Prog Rock Ricky** - bright/dirty bass sound for Geddy Lee and Chris Squire type tones (Ricky aka Rickenbacker users) . Can be used as a DI or in front of bass amp sim. Should sound good with Fender Jazz bass as well.

.: **Bass-Slap Funk** - slightly scooped mid, cleaner slap-n-pop bass sound. For DI or in front of bass amp sim.

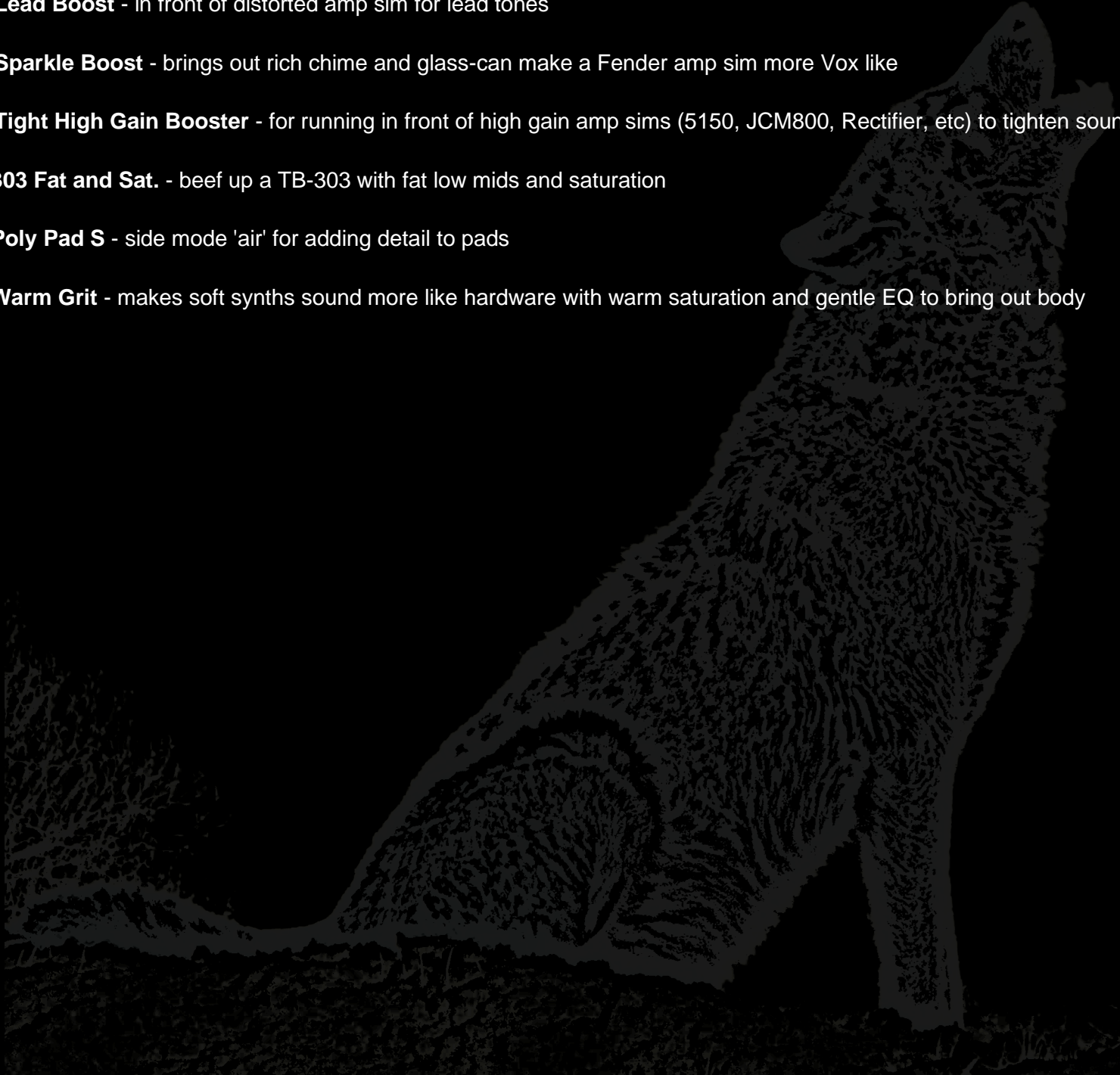
.: **Drum Machine-Colorizer** - soft saturated with a tuned fat low end and crispy highs. Great with an 808.

.: **E.Piano-Vintage Grit** - softly overdriven which lightly compresses and brings out detail in the upper mids and highs.

.: **Guitar-Direct Amp Sim** - DI to sound roughly like a mid-60s Fender Deluxe Reverb

.: **Guitar-Dirty Boost** - great for boosting into an a non-master volume amp sim (like a Vox AC-30 or Marshall Plexi)

- .: **Guitar-Fat Blues** - DI or in front of cleaner amp sim for fat blues tones
- .: **Guitar-Lead Boost** - in front of distorted amp sim for lead tones
- .: **Guitar-Sparkle Boost** - brings out rich chime and glass-can make a Fender amp sim more Vox like
- .: **Guitar-Tight High Gain Booster** - for running in front of high gain amp sims (5150, JCM800, Rectifier, etc) to tighten sound for metal
- .: **Synth-303 Fat and Sat.** - beef up a TB-303 with fat low mids and saturation
- .: **Synth-Poly Pad S** - side mode 'air' for adding detail to pads
- .: **Synth-Warm Grit** - makes soft synths sound more like hardware with warm saturation and gentle EQ to bring out body



Uninstalling P42 Climax

FOR WINDOWS

- In 'C:\Program Files\Common Files\VST3', locate the 'P42 Climax.vst3' file and delete it.
- In 'C:\Users\Public\Documents\Pulsar Modular', locate the 'P42 Climax' folder and delete it. This folder contains the user guide and presets. If no other folders exist under 'Pulsar Modular', this can be deleted as well.

FOR MAC OS X

- In '/Library/Audio/Plug-Ins/Components', locate the 'P42 Climax.component' file and delete it.
- In '/Library/Audio/Plug-Ins/VST3', locate the 'P42 Climax.vst3' file and delete it.
- In '/Users/Shared/Pulsar Modular', locate the 'P42 Climax' folder and delete it. This folder contains the user guide presets. If no other folders exist under 'Pulsar Modular', this can be deleted as well.

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