



P42 CLIMAX

User Guide

Version 3.0

Foreword



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.

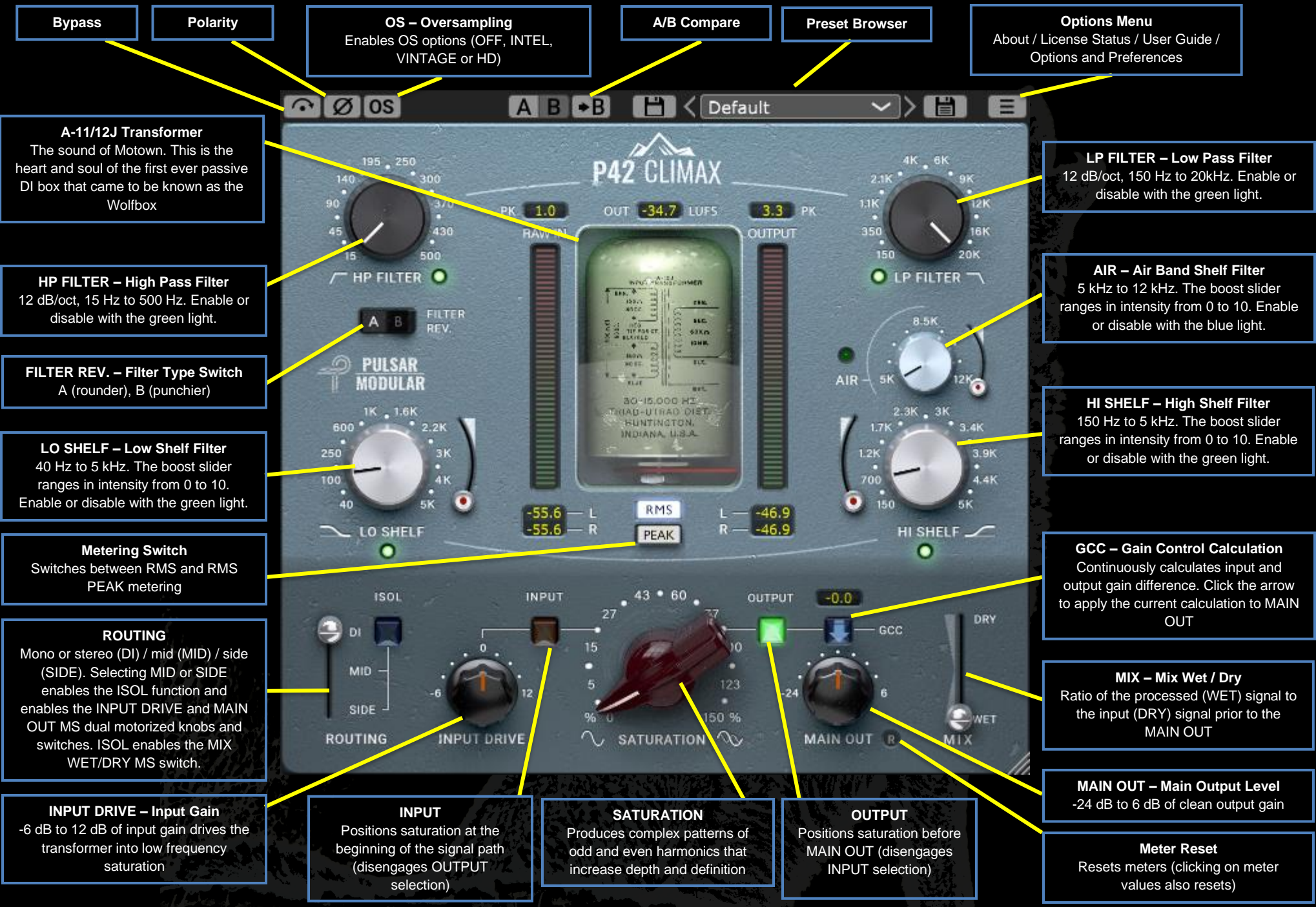
He soon went on to work for Detroit studios like Motown Records and United Sound where his direct boxes were used to capture some of the greatest moments in musical history, performed by artists who were to become legends. This little device was critical in giving Motown its big, distinct, fat bottom, 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.

The development process has truly been a labor of love for all involved. We sincerely hope you enjoy the results as much as we do. Pulsar Modular – **The sound is unbelievable.**



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

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

 Oversampling options allow P42 to optionally operate at a multiple of the host sample rate. Grey indicates OS off and yellow indicates OS on. With OS off, P42 operates at the host sample rate (x1).

When oversampling is on, different options are made available. See the descriptions of INTEL mode, VINTAGE mode and HD mode below.

Tip: Don't assume that an oversampled signal is better. Try each option and use your ears. At x1 or to a lesser degree at x2, 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.

INTEL INTEL (intelligent) mode operates at double the host sample rate (x2). It 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 VINTAGE mode.

VINTAGE VINTAGE mode operates at double the host sample rate (x2). It 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.

HD HD mode operates at an internal sample rate of 384 kHz. It utilizes the same full frequency scan filtering strategy as INTEL mode. The high sample rate and filtering mechanism make this a pristinely high-quality option at a surprisingly efficient CPU load. This mode is equally suitable for mastering duties or for key tracks when mixing.

To achieve HD oversampling, P42 applies the following logic:


- 44.1 and 48 kHz oversamples at x8
- 88.2 and 96 kHz oversamples at x4
- 192 kHz oversamples at x2, thereby enabling INTEL and VINTAGE options
- 384 kHz disables oversampling options

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

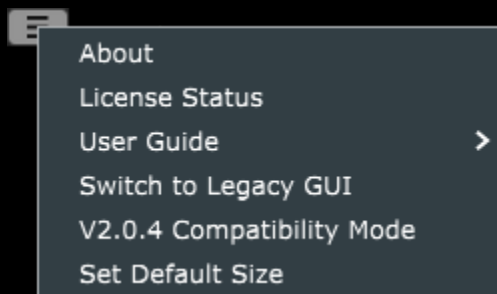
Note: If you switch between the available OS modes, buffer sizes between 0 (OS off) and 61 samples (HD with x8 OS) are generated. This can result in audible clicks that are harmless to your system.

A B ↔ 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 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 if the install presets option is not selected. This is the default update version installer setting.



Options Menu

About – Check the version number or demo expiration date.

License Status – Manage your license.

User Guide – Open the EN or DE user guide.

Switch to Legacy GUI – Switches the selected GUI option to the legacy version (see the 'Alternative GUI: Legacy' section of this document).

V2.0.4 Compatibility Mode – Indicates the current instance was originally saved using a legacy version of the software. This is normally automatically managed and no user interaction is required.

Set Default Size – This option can be selected to apply the size of the current P42 instance as the default size for all instances of P42.



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.

Note: When using a sample rate of 192 kHz or higher, the HPF becomes unstable when positioned below a set limit. P42 will automatically change the HPF lower limit to 25 Hz if the sample rate is 192 kHz and 30 Hz if the sample rate is 384 kHz or higher. 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.



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



The metering shows several pieces of information including LUFS (Loudness Units relative to Full Scale) measurements, dBFS PK (decibels relative to full scale peak hold) measurements, and switchable RMS / PEAK RMS (Root-Mean-Square) bar meters and measurements.

The PK (peak hold) measurements hold the highest measured value. They will be reset if any measurements are clicked, if the DAW is stopped/started or if the Meter Reset button adjacent to the MAIN OUT label is clicked.

The switchable RMS / PEAK RMS bar meters and measurements can be switched by clicking the RMS / PEAK radio buttons. These measurements are updated automatically but can also be reset as described above.

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 advanced filter topology of this low shelf filter features a musical boost that ranges from 40 Hz to 5 kHz. The boost slider ranges in intensity from 0 to 10.

The boosted portion of the signal is followed by a frequency dip that helps to frame the area of focus.

Enable or disable with the green light.



The AIR Band Shelf Filter is a carefully tuned high shelf designed to gently unmask high frequency detail and clarity. The boost slider ranges in intensity from 0 to 10.

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.

Enable or disable with the blue light.



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 boost slider ranges in intensity from 0 to 10.

The boosted portion of the signal is preceded by a frequency dip that helps to frame the area of focus.

Enable or disable with the green light.



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. Selection of MID or SIDE also enables the MS motorized dual function knobs and switches. See the sections for INPUT DRIVE and MAIN OUT for details.

The ISOL switch (not available if DI is selected) 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: There are several options available for processing mid/side signals using P42 including in series, in parallel and using dual mono with a mid/side encoder/decoder. Please have a look at the Tips, Tricks and Techniques section for descriptions of setting up and using different workflows.

Note: Always keep in mind that an instance of P42 is dedicated to a single ROUTING channel selection. Separate control over DI, MID or SIDE processing requires separate instances. These options are exclusive and are never combined in a single plugin instance.



The INPUT DRIVE knob 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.



When MID or SIDE is selected in the ROUTING options, the INPUT DRIVE MS dual function motorized knob and switch become enabled. By default, the switch is on as indicated by the blue light.

When the switch is on, the INPUT DRIVE affects the MID or SIDE signal only. When the switch is off, the INPUT DRIVE affects the stereo signal. Since the knob is motorized dual function, it recalls both settings and applies them simultaneously. For compatibility with versions prior to 3.0, the MS switch should be turned off.

Note: The dual function does not apply across MID settings and SIDE settings, it is only within one of those selections. This means that if you are in MID, the knob will remember if you are changing the mid only (MS light blue) or the full stereo signal (MS light off).

Note: Please have a read over the Tips, Tricks and Techniques section of this user guide for helpful information on use of the dual function motorized knob functionality.



If the saturation position option 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 knob 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 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 above 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 main output knob 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.



When MID or SIDE is selected in the ROUTING options, the MAIN OUT MS dual function motorized knob and switch become enabled. By default, the switch is on as indicated by the blue light.

When the switch is on, MAIN OUT affects the MID or SIDE signal only. When the switch is off, MAIN OUT affects the stereo signal. Since the knob is motorized dual function, it recalls both settings and applies them simultaneously. For compatibility with versions prior to 3.0, the MS switch should be turned off.

Note: The dual function does not apply across MID settings and SIDE settings, it is only within one of those selections. This means that if you are in MID, the knob will remember if you are changing the mid only (MS light blue) or the full stereo signal (MS light off).

Note: Please have a read over the Tips, Tricks and Techniques section of this user guide for helpful information on use of the dual function motorized knob functionality.



The Meter Reset button resets current measurements when clicked. Measurements can also be reset by clicking on any of the metering display values.



The MIX WET/DRY slider 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.



When MID or SIDE is selected in the ROUTING options and the ISOL switch is selected, the MIX WET/DRY slider MS switch becomes enabled. By default, the switch is off as indicated by the switch being unlit.

When the switch is off, the slider affects the stereo signal, blending the unaffected stereo signal with the affected and unaffected MID or SIDE signal. When the switch is on, the slider affects only the MID or SIDE signal in isolation, meaning the processed signal can be compared with the unprocessed signal without hearing the influence of the full stereo field.

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 HP FILTER 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 LO SHELF to be double the frequency of the HP FILTER. Using the LO SHELF 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 LO 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 HP FILTER frequency will help to give focus. Increasing the Low Shelf frequency from there will enhance the body. [KE]

.: Layer instances of P42 to combine complementary 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 HP FILTER (so let's say for example you settled for 38 Hz). On the HI SHELF, get the 1st color you want. Probably get the timbre of that snare in the mix to "puck" instead of "pock" sounding. So maybe high shelf at 700 Hz with a light touch of boost. Leave the LP FILTER at max 20 kHz.

2nd instance (inserted after the first one): Disable the Low Shelving EQ (Already you nailed it in the 1st instance). HP FILTER all the way down to 15 Hz (already you nailed it in the 1st instance). Get the second color you want by engaging the HI SHELF filter and start searching with the frequency 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 Band Shelf Filter 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 LO SHELF EQ. HP FILTER at minimum and LP FILTER at maximum. Now turn on the AIR and work the boost level. Optional: engage the HI SHELF EQ and scan for other interesting muffled high frequencies, around 3 to 4 kHz. Now work the LP FILTER 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 LP FILTER 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 A812" preset is a great starting point). You can tune each instrument of your kit according to your taste. The beautiful boom from beyond and the cumulative vividness of the single channel workflow is most valuable.

If your DAW offers a channel link function, you can use it to define individual instrument groups in your session. This feature is available in Cubase, for example. Each of these linked groups gives you their very special sound character. To do this, simply open a single P42 instance in a group that you want to customize. If you edit any setting here, it will be automatically transferred to all linked instances. If no shortcut function is available by default, you can, for example, select several channels at the same time in Cubase and work with the key combination ALT + Shift + left mouse button in a P42 instance. To achieve the desired effect for the selected instances, all P42 instances must be at the same horizontal insert level. This method is perfect for giving different instrument groups individual drive and creating more air and separation in the mix or for specifically highlighting or concealing groups in their mix.

For a simple and fast implementation, load a P42 instance in the insert of a group channel (stem) and define its desired sound there. If you are satisfied with the result, drag and drop this P42 instance away from the group channel and place it on the first instrument channel of the group in question. From there, copy this instance (in Cubase with ALT + left mouse button) to the second instrument channel, etc., until all individual channels are populated. Now you can further optimize as in the previously described link mode or fine-tune individual channels as required. This roughly corresponds to the old-school method with multi-channel tape machines. In any case, this approach saves them a lot of work and you will quickly find the right basic sound for a complete group. Precisely because the change of presets in link mode is not transferred to the other instances, this approach is a priceless time saver.

This way of working allows for easily and quickly mix grouped elements into the foreground in a mix, to emphasize them or to push them discreetly and effortlessly into the background. This gives your mix significantly more separation, depth and liveliness. [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 its 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, buildup 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 OS on (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 than 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 OS on (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]

:: M/S Dual Mono Workflow ::

This technique requires a DAW that supports dual mono (such as Logic Pro) and use of a MS encoder/decoder plugin (such as Voxengo MSED).

Placed prior to the single instance of P42, the first MS encoder/decoder plugin will encode the LR stereo signal into the MS signal, outputting the mid signal on the left and the side signal on the right. The next plugin in this series is P42. When switching the DAW to the L channel, P42 will be processing the mid signal and when switching the DAW to the R channel, P42 will be processing the side channel. While processing either the mid or side signals, all features of P42 are available, including the WET/DRY slider. The exception to this is the ROUTING options because each instance of P42 is processing a mono signal when dual mono is used. The final plugin required in this series is a second instance of the MS encoder/decoder, which will decode the MS signal back to the LR stereo signal. [ZS]

.: M/S In Series Workflow .:

This is perhaps the most straight forward method to use P42 to apply M/S techniques. This technique requires use of two P42 plugins in series, one of which processes the mid signal and the other of which processes the side signal. This workflow lets you work while listening to the full stereo field so you are working in context.

The first plugin in the series should have the ROUTING option set to MID and the second plugin should have the ROUTING set to SIDE. While working in this manner, it is sometimes desirable to temporarily engage ISOL to isolate just the channel you are working on to focus on subtle details of that channel for a short time. The selected channel becomes mono and the opposite channel is not heard when ISOL is engaged. For example, if you are in SIDE and you choose ISOL, you only hear the SIDE in mono and not the MID so you lose the stereo field context but you can hear the effect just on the side signal individually. When in ISOL, a MS switch appears beside the WET/DRY MIX slider. This switch, when lit blue, indicates that the MIX slider affects only the isolated signal. In effect, this lets you bypass just the processed signal to compare to the original signal. 100% DRY is the original signal while 100% WET is the processed signal.

This option allows P42 to be used in M/S without the need for a separate M/S encoder/decoder utility plugin. [ZS]

.: M/S In Parallel using ISOL Workflow .:

This workflow requires you to duplicate the stereo track or use the routing capabilities of your DAW to achieve a similar duplication of your stereo signal if it supports it. Each stereo track requires an instance of P42.

P42 on one stereo track should have the ROUTING option set to MID and the second plugin on the other stereo track should have the ROUTING set to SIDE. Both instances should have ISOL engaged. The P42 instance set to MID will be outputting two channels, however the mid signal is output both on the left channel and the right channel. The P42 instance set to SIDE will be outputting two channels, however the side signal is output on the left channel as mid+side and the right channel is output as mid-side. The two stereo signals can now be blended as desired.

This option allows P42 to be used in M/S without the need for a separate M/S encoder/decoder utility plugin. [ZS, KE]

.: Advanced M/S processing using the dual function knobs and switches :.

The dual function INPUT DRIVE and MAIN OUT knobs and MS switches work very well when working in M/S in series. It is important to remember that these features are only enabled when ROUTING is set to MID or SIDE. The dual function gives you more routing and coloring options than before.

As an example, assuming you are in MID, your first choice is INPUT DRIVE. Remember that although you are working on the MID, you still hear the full stereo field in context. Ask yourself if you want to push and saturate the full stereo signal or only the MID? Say you determine you want to push the whole signal 6 dB and the MID channel 3 dB. Here is where you can use the dual function of the INPUT DRIVE. To accomplish this, ensure the MS switch is lit blue, indicating that you are processing only the channel. Set the INPUT DRIVE to 3 dB. Now disable the MS switch so it is grey, indicating you are processing the full stereo field. Set the INPUT DRIVE to 6 dB. That's it! Click the MS button a few times and enjoy watching the slick motorized fader!

You do not need to use all these features together, but now you have the capability and choice. Note that all other controls (other than MAIN OUT which also has its own motorized knob) apply only to the channel of the ROUTING option you have selected. [ZS, JB]

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).

Backing Up Presets

Presets can be backed up and restored using your operating system file manager. Simply perform a copy/paste of either individual preset files or the full presets folder to a backup location of your choosing. The presets folder can be found in the following locations:

FOR WINDOWS

'C:\Users\Public\Documents\Pulsar Modular\P42 Climax\Presets'

FOR MAC OS X

'/Users/Shared/Pulsar Modular/P42 Climax/Presets'

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 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 (Aciiiid :))
- .: **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

General

Fine Tuning Mode

Press and hold the modifier key (in macOS: “control, option or command”, in Windows: CTRL) while moving the knobs.

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 and presets. If no other folders exist under 'Pulsar Modular', this can be deleted as well.

Alternative GUI: Legacy

Bypass

Polarity

Oversampling
Enables oversampling routine selection

A/B Compare

Preset Browser

About / License Status / User Guide / Options and Preferences

Low Pass Switch
Engages the LP filter

High Pass Switch
Engages the HP filter

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

REV A

DI

METER

MIX W/D

DRIVE

L. SHELF

H. SHELF

P42 CLIMAX

FREQ

BOOST

MAIN OUT

GCC

AIR

HPF

LPF

@Output

Developers / Legacy GUI Design: Pulsar Modular Team
Modern GUI Design: Max Ponomaryov / azzimov GUI design – www.behance.net/azzimov
User Guide (EN): Kevin Eagles
User Guide (DE): Matthias Klein
Testers: Chris Allen Allan Klinbail
 Leo Alvarez Vince Riccio
 Kevin Eagles Synthient Sound
 Jason Fernandez Hilton Stroud
 Nil Hartman Wheeliemix
 Matthias Klein John Marshall
 Conan Manchester Nicolas Lefèvre
 Jory Berger Andi Vax

Please kindly report any errors or omissions in this user guide to psupport@pulsarmodular.com.

P/N: 15013, Rev. 3.0

Pulsar Modular is a registered trademark of Pulsar Novation Ltd.

P42 Climax is a plugin name owned by Pulsar Novation Ltd.

AAX and Pro Tools are trademarks of Avid Technology. Names and logos are used with permission.

Audio Units is a trademark of Apple, Inc.

VST is a trademark of Steinberg Media Technologies GmbH.

All other trademarks contained herein are the property of their respective owners.

Pulsar Novation Ltd.

Demircikara District, 1419 Street, Ocean City Block B, Floor 4

Muratpaşa, ANTALYA 07100 +90-530-111-4907

www.pulsarmodular.com