

A diver in a deep-sea environment, illuminated by a bright lantern. The diver is wearing a full-body suit and a large, round helmet. The background is dark blue with some coral visible on the left and right sides. The text "P11 Abyss" is in the top right corner, "User Guide" is in the center, and "Version 2.0" is at the bottom center.

P11 Abyss

User Guide

Version 2.0



Core Action and Behavior

These controls influence the fundamental characteristic behavior and shape of the compression. Simple yet infinitely flexible options for reproducing the characteristic behavior of any compression circuit topology are presented here.

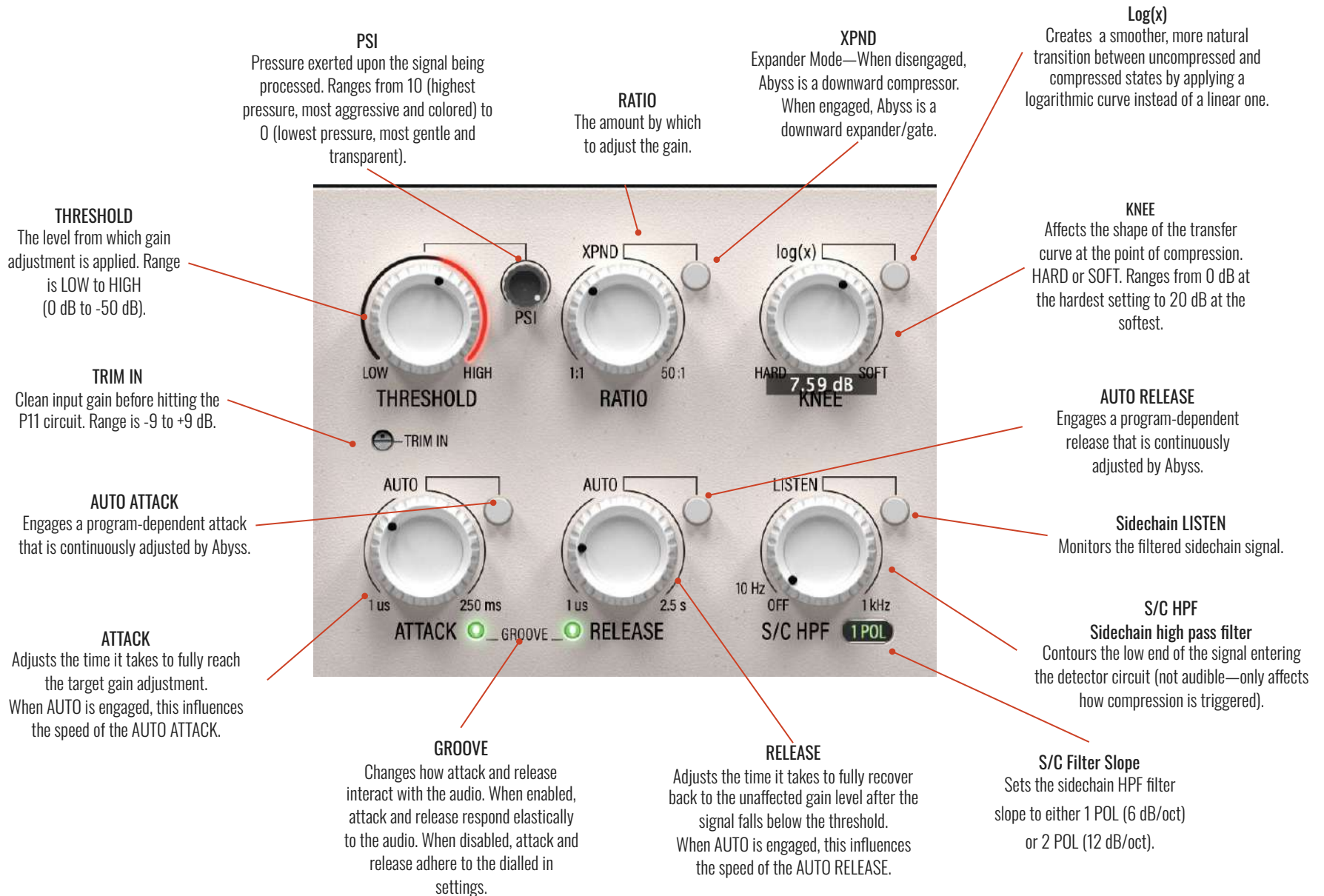
The Central Scrutinizer

Presents a wealth of metering and other visual feedback to alert you of the potential consequences of everyday activities. Extended functionality is also available here to help prevent you from doing wrong things. It's a way of life.

Character, Depth and Color

These controls provide options to extensively alter the internal characteristics of the processed audio. These are not layered additions to the sound, but rather they profoundly and interactively manipulate the core character.

Core Action and Behavior





Sets the level from which gain adjustment is applied. When the S/C (side chain) signal crosses the threshold level, compression on the input signal is applied by a factor determined by the **RATIO** knob. Please note that the S/C signal is what matters to the **THRESHOLD** and not the input signal.

The threshold ranges from 0 dB (HIGH) to -50 dB (LOW).

The PSI (Pressure) Knob shapes the very nature of how Abyss responds to audio. It ranges from 10 (highest pressure) to 0 (lowest pressure).



At high pressure, Abyss reacts with force—strong, assured, firm, and ferocious. As pressure eases, it reveals its more sensitive side—calm, delicate, transparent, and refined. Abyss thrives confidently at either extreme, as well as across the full spectrum of ambient pressures—from the surface to the deepest trench.

PSI is central to Abyss's ability to emulate the behavior of various hardware compressors. It subtly adjusts attack and release characteristics with each 0.1 increment, offering a vast range of dynamic textures.

And yet, despite its depth, PSI is incredibly intuitive: click the knob, close your eyes, make adjustments, and feel. Listen to how it changes not just the track, but its relationship to everything else in the mix.

Tip: Always remember to explore combinations of MOD, PSI, SOUL and O2 when fine-tuning your settings to really get to know what these controls impart on their own and in combination.

Tip: Hear how PSI fundamentally influences the compression in our No-nonsense Audio Workshop at https://www.youtube.com/watch?v=EEA_KC6pXpo.



RATIO determines the factor by which gain is adjusted once the S/C signal crosses the threshold line. The gain reduction (gain in XPND mode) is applied to the input signal.

The gain value is calculated based on the settings of your **THRESHOLD**, **KNEE**, **RATIO**, **ATTACK** and **RELEASE** parameters in relation to the side chain signal.



The **KNEE** is the slope (smoothness of transition) of the ratio. In the **SOFT** setting, the ratio is gradually increased to reach the set ratio value, while in **HARD** setting, the ratio value is applied immediately.

Log(x) enables logarithmic interpolation of the compressor's knee, creating a smoother, more natural transition between uncompressed and compressed states by applying a logarithmic curve instead of a linear one.

Tip: In practice, softening the knee reduces the noticeable transition from uncompressed to compressed signal. As such, softening the knee can be less aggressive and sound more natural by allowing affected transients to be attenuated more gradually. In simple terms, this can mean having a fast attack while still allowing natural transient energy and punch to come through.



The TRIM IN trim pot provides clean input gain that allows for level adjustments to the incoming signal prior to any other processing. This is useful for cleanly raising or lowering input signals that are too low or too high for normal operation. Hold the shift key when adjusting the TRIM IN or TRIM OUT trim pot to compensate equally with the opposite trim pot.

Tip: Try using this to gain stage into a preset to achieve the desired gain reduction without having to change the THRESHOLD.



ATTACK controls the time it takes to fully reach the target gain adjustment based on the threshold and ratio settings. If AUTO is engaged, the attack time is continuously and dynamically determined by Abyss. The speed of the program dependent attack time can be influenced by adjusting the ATTACK knob. The values range from -10 to 10 when AUTO is engaged.

Tip: On group and 2-bus or mastering duties, try using higher values than you are accustomed to because of the nature of Abyss being an RMS and not a Peak compressor. If you feel the mix is choked, don't be afraid to use an attack of 100 to 150 ms!

Tip: While attack is always thought of in terms of fast or slow, keep in mind that it is always relative to the input signal. What is fast for a bass guitar is slow for a snare drum, so always keep context in mind.

Tip: Setting fast or slow attack is all about how the transients should be handled (or not handled). A faster attack will affect the transient directly, controlling or manipulating it in some way. For example, this can help to even out an unbalanced performance when combined with a medium to slow release, or it can help a flabby kick to become punchy and defined when combined with a fast release. A slower attack will allow the transient to pass unaffected, emphasizing the initial articulation and definition of a well-captured source. Nothing is automatically good or bad; everything doesn't need more punch or more control; listen within the context of all the audio being processed and decide what it needs.

Note: Please have a read over the Tips, Tricks and Techniques section of this user guide for ways to use AUTO ATTACK and AUTO RELEASE.

ATTACK GROOVE RELEASE

GROOVE affects how the attack and release curves interact with the audio.

With GROOVE disabled, the compressor adheres to the exact values dialed in for attack and release. With GROOVE enabled, the ratio and real-time gain reduction come into play, affecting how the compressor interprets the attack and release values.

Attack: When enabled, the attack behaves in an almost elastic manner, exhibiting an inflating motion while simultaneously being tightly constricted. In contrast, when disabled, the attack rigidly shapes the transient.

Release: When enabled, the audio pushes against the curve during release, resulting in slight deformation with a tethered, swinging motion. Conversely, when disabled, the curve and timing are strictly adhered to without any deformation.

Tip: Experiment with different combinations of Attack and Release Groove settings. Different situations may call for different approaches!

Tip: Learn how to hear and feel the Groove circuit in our No- nonsense Audio Workshop at https://www.youtube.com/watch?v=EEA_KC6pXpo.



RELEASE controls the time it takes to fully recover back to the unaffected gain level once the signal falls below the threshold. If AUTO is engaged, the release time is continuously and dynamically determined by Abyss. The speed of the program dependent release time can be influenced by adjusting the RELEASE knob.

The values range from -10 to 10 when AUTO is engaged.

Tip: On a full mix, increasing the Release will deliver a wider stereo image.

Tip: A faster release tends to enhance the rhythm of a signal because the controlled gain movement helps to push and pull an element forward and backward. A slower release tends to increase the density of a signal and solidify the position it occupies because it reduces the dynamic range over a more stable period of time. Attack and release characteristics are not an all-or-nothing proposition.

Think about how multiple compressors can be used together to affect an element.

Note: read over the Tips, Tricks, and Techniques section of this user guide for ways to use AUTO ATTACK and AUTO RELEASE.



S/C HPF applies a high-pass filter to the detector circuit. This filter does not directly affect the sound of the processed signal.

The slope of the HPF can be adjusted using the 1 POL / 2 POL switch to the right of the S/C HPF label. The 1 POL option gives a 6

dB/oct filter slope, while the 2-pole option gives a 12 dB/oct filter slope.

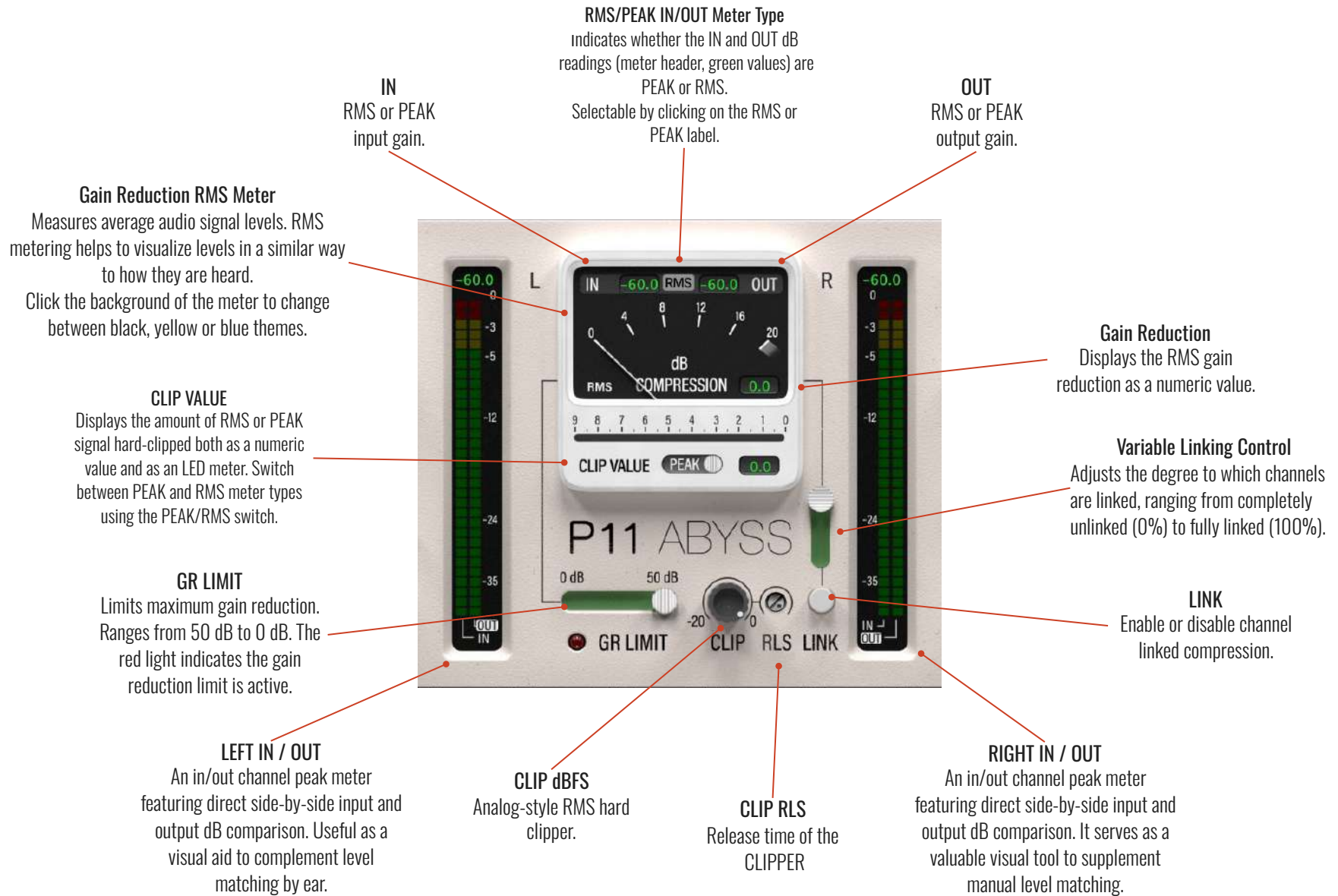
It is not uncommon for an element to contain more energy in the lower range of frequencies it spans, despite sounding balanced to the human ear. In some cases, this energy will exceed the threshold far more than upper frequencies within the same element, and this can result in an erratic dynamic response. This filter allows for attenuation of lower frequencies to encourage stable dynamic processing.

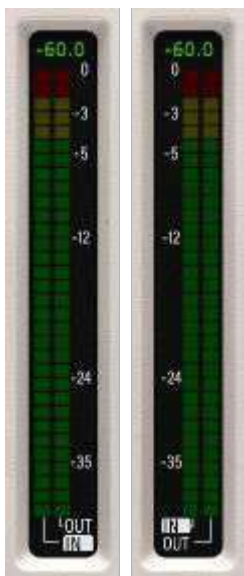
Use the LISTEN button to hear the signal fed into the detector.

Pressing CTRL+ALT (Windows) or CMD+OPTION (macOS) on the keyboard while the mouse is positioned over this control will temporarily disable it until the keys are released.

Tip: Also have a look at the sidechain input EQ section (TO SC setting of the EQ Target Switch). Used together with the S/C HPF filter, the inaudible signal that influences the detector circuit can be rebalanced and reshaped in virtually any way you can imagine. This affords very fine and flexible control over how the detector circuit is reacting to your audio beyond what the audible signal triggers.

The Central Scrutinizer





The LED meters shows peak and RMS input and output levels simultaneously, with RMS represented as the 'solid' continuously updated LED block in the main part of the meter, and with peak represented by the 'held' LED light, which will always appear above the RMS block.

The value shown at the top of the meter is the L or R channel-specific peak or RMS value (as opposed to the LR averaged value shown in the VU meter header). Whether this displays a peak or RMS value is determined by and synchronized with the VU meterheader RMS/PEAK selection.

At the bottom of the meter, either the IN label or OUT label is highlighted. If IN is selected, the value shown at the top of the meter is synchronized with the IN value shown in the VU meter header, and conversely, if OUT is selected, the value is synchronized with the OUT value shown in the VU meter header. Click either label to switch to the alternate option.

These meters provide a visual calibration aid that can be used in tandem with what you are hearing when gain matching input and output levels.

These can be very helpful to visually support what you are hearing but rely first and foremost on your ears.



None shall pass. The CLIP knob sets the dBFS at which the analog clipper hard clips the signal with beautifully transparent, pristine quality clipping. As with real-world imperfect analog clipper circuits, overshoots can sometimes occur randomly within a +0.3 dB tolerance. The clipper can be pushed in ways that a traditional peak level clipper cannot be pushed. As such it delivers transparent, high- energy output that is bigger than life.

Use the RLS screw to manipulate the sound from relaxed release time of 10 ms to hard 0.1 ms setting.

Tip: Loud but controller sound shaping can be done by enabling the Clipper and setting its release time, then engaging the limiter (LMTR) and again shape it with the release slider.

Tip: To best set the clipper value, ensure the RMS/PEAK IN/OUT meter header in the GR window is set to RMS since this measurement type aligns with the CLIP operation.

Note: The clipping routine can be positioned in various locations in the signal path by using the configuration option on the top toolbar.



The dB COMPRESSION gain reduction meter main window shows the RMS (Root-Mean-Square) gain reduction applied to the signal as indicated by the ballistic needle.

To the right of the needle is a display box that shows a numeric

Abyss is capable of GR that goes well beyond -20 dB. When this occurs, the needle will remain in the dark area below the 20 dB

marker but the display box will show an accurate gain reduction value.

Along the top of the gain reduction window is the RMS/PEAK meter header. The current metering type is shown in the middle of the header and this can be switched by clicking on the RMS or PEAK label. The IN and OUT values on either side of the RMS/PEAK label show the LR averaged peak or RMS value as determined by the current metering type selection. The discrete non-averaged values for the L and R channels are shown at the top of the vertical LED meters to each side of the VU meter area.

The CLIP VALUE display shows the RMS or PEAK amount of signal that is clipped pre or post compression, depending on where the clipper is positioned in the signal path. The position can be configured using the top toolbar. The clip value is represented both in the display box to the right of the PEAK/RMS switch and in the horizontal meter above the CLIP VALUE label.

The metering type can be switched between PEAK and RMS using the PEAK/RMS switch beside the CLIP VALUE label.

Clicking on the background of the meter changes the meter theme between black, yellow and blue options. Combine this with light or dark mode and tailor Abyss to what inspires you visually.

Note: Changing either the header meter type or the clipper PEAK/RMS option changes the metering option only, not the way the detectors are listening. The detectors are always based on RMS.

Note: The ballistic needle and the numeric representation thereof are always RMS regardless of the header meter type selection.



Limits the maximum gain reduction to the specified value.

Without this set, the compressor or expander/gate will apply gain reduction as determined by the ratio and

threshold without restrictions. With this set, the gain reduction will not exceed this value.

The red LED lights up when GR LIMIT is active.

Tip: Learn how CLIP can be used to create uniform compression for wildly dynamic sources by taming the sidechain signal in our No- nonsense Audio Workshop at https://www.youtube.com/watch?v=_yC53UrOmD8.



When the LINK button is turned on, the compression process is based on the average signal of both the left and right channels combined. The degree to which the channels are linked is determined by the Variable Linking Control. When set at 100% linked, the same amount of compression is applied to both channels whenever a signal crosses the threshold on either the left or right channel. This fully linked compression leads to uniform dynamic movement and enhances the tightness and solidity of the stereo image.

On the other hand, when the LINK button is turned off or set to 0% linked, each channel is compressed independently, as if using two separate compressors. Compressing each channel individually results in variable dynamic movement between the channels, as the compressor responds uniquely to each signal. This can lead to a widening or opening effect in the stereo image.

Tip: When LINK is disengaged or set to a low percentage, pay attention to transient heavy percussive elements, particularly dynamic elements or heavily accentuated elements that are panned out because they can trigger 'wandering' of the center image. The SIDECHAIN HPF and the SC EQ controls can be used to influence and smooth out the detector circuit to help with any extreme movement. Additionally, carefully considered gain reduction will help when finding the right compromise between opening the soundstage and maintaining a solid center image.

Character, Depth and Color

MODE Selector

These are tied to the EQ Target Switch. Once enabled, these remain active even when the other EQ target is selected. Choices are shelf and bell boost/attenuate. Enable and disable an EQ using the button below the EQ label.

EQ Target Indicator

Indicates an EQ Target is active. The green light on the left indicates the TO IN target is enabled. The yellow light on the right indicates the TO SC target is enabled.

EQ Target Switch

Selects the current target of EQ section controls (MODE, Frequency and Gain) to adjust. Choices are TO IN (input, prior to compression) and TO SC (sidechain detector).

02

A unique signature Pulsar Modular designed engine that breathes life into everything, making it simply sound better. Compress gently... then inject 02, smash it... then inject 02, do whatever you want... yes, now you get the picture, then inject 02.

02 Position Switch

Positions the 02 knob before the MIX knob (PRE) or after the MIX knob (POST).

TRIM OUT

Clean output gain for adjusting signal after leaving the Abyss. Range is -9 dB to 9 dB.

MIX

Ratio of the processed (WET) signal to the input (DRY) signal prior to MAIN OUT.

MAIN OUT

-24 dB to 24 dB of clean gain.

Motorized EQ Frequency Selector

See MODE Selector for the relationship to the EQ Target Switch. Ranges from 15 Hz to 20 kHz.

Motorized EQ Gain Slider

See MODE Selector for the relationship to the EQ Target Switch.

Amplifier Type Selector

Off- no transformer; Class-A amplifier for clean signal reproduction with sharper transients; Class-A/B amplifier for thicker, more characterful amplification.

T. IN

Tied to the Input Amplifier Type Selector 0 dB to 12 dB of clean or transformer-coupled gain.

T. OUT

Tied to the Output Amplifier Type Selector ± 24 dB of clean or transformer-coupled gain or attenuation.

MIX Position Switch

Positions the MIX knob before MAIN OUT (PRE) or after MAIN OUT (POST).





Two multi- functional EQ bands can be used to simultaneously alter the signal sent to the detector circuit (TO SC) and to alter the signal sent through the processing chain prior to

dynamics adjustment (TO IN).

The MODE Selector can be used to set the current filter to shelf or bell boost/attenuate. Each EQ can be enabled or disabled by clicking the button below the EQ label.

The shelf filters feature a broad, fixed Q. When boosting, the bell filter features a very sweet and smooth symmetrical fixed Q curve. When attenuating, the bell filter features a proportional Q that narrows as the signal is further attenuated. Each motorized EQ is carefully designed to maintain musicality as a shaping tool while adjusting the sidechain or the input signal.

The EQ Target Switch can be used to select either TO SC to show the EQ pair that is routed to the detector circuit, or TO IN to show the EQ pair that is routed through the processing chain.

Each pair remains active when not visible.

The lights above the switch indicate which EQ Target is active. The green light on the left indicates the TO IN target is enabled. The yellow light on the right indicates the TO SC target is enabled.

Note: Please have a read over the Tips, Tricks and Techniques section of this user guide for ways to use these sidechain and input EQ features.

Tip: Learn how the sidechain EQ can control movement and create texture in our No-nonsense Audio Workshop at https://www.youtube.com/watch?v=YDxlwkc_KXw



The MAIN OUT knob features -24 dB to 24 dB of clean gain.

The WET DRY knob allows for blending a desired amount of dry signal in with the processed wet signal.

The WET DRY position is configurable using the PRE/POST switch found to the left of the MAIN OUT label. When set to PRE, WET DRY output feeds into the MAIN OUT knob, allowing for final volume adjustment of the overall processed signal. When set to POST, MAIN OUT feeds into the WET DRY knob allowing for MIX adjustment after the final level of the processed signal has been set.

Note: An additional option is available for having both a PRE and POST positioned gain control. Abyss features a MAIN OUT gain adjustment knob as well as a T. OUT gain knob. The T. OUT knob affects the processed signal only and is positioned before the WET DRY knob in the internal signal chain, whereas MAIN OUT is positioned after the WET DRY knob when set to PRE and will affect the fully combined signal.



O_2 is a signature Pulsar Modular circuit algorithm that works to subtly and beautifully enhance any audio signal that passes through it. Born from trial and experimentation, it defies traditional description as it is not akin to any other

hardware circuit or software algorithm that existed prior to the advent of Abyss.

As O_2 is increased, the signal opens and breathes in a way that is reminiscent of introducing a long, deep, airy breath into the audio, expanding it beyond the confines of your speakers.

The O_2 position is configurable using the PRE/POST switch found to the left of the O_2 label. When set to PRE, O_2 output feeds into the mix knob, so it is applied only to the wet signal. When set to POST O_2 is positioned after the MIX knob, so it affects the wet/dry blended signal. The O_2 circuit is always positioned relative to the MIX knob positioning. Pressing CTRL+ALT (Windows) or CMD+OPTION (macOS) on the keyboard while the mouse is positioned over this control will temporarily disable it until the keys are released.

Tip: O_2 has a very interesting relationship with SOUL. Try starting with O_2 OFF, finding a SOUL setting that works for the material at hand and then introducing complementary drive using O_2 . Once you find a SOUL setting that gives the source a constructive nonlinear shape (across all 3 axis: front to back, side to side and top to bottom), you can then drive O_2 for a little harmonic fattening and flattening.

Note: When the O_2 circuit was created, it was placed after the MIX knob because in this position, usage results in a homogenous sound. This is the default whenever a new instance of Abyss is loaded. To have Abyss start with the position set to PRE with each new instance, the Default preset can be modified by selecting Default from the Preset Browser, changing the switch from POST to PRE and saving over the preset using the save icon with the red asterisk*.

Tip: Learn how KNEE, PSI, SOUL, T. IN, T. OUT and O_2 can be used to accent snap, smack or swoosh in our No-nonsense Audio Workshop at https://www.youtube.com/watch?v=YDxlwkc_KXw



The T. IN and T. OUT gain knobs provide either pristine digitally clean or mojo infused transformer coupled amplifiers for adjusting gain pre and post dynamic processing.

Note: For brevity, throughout this document, the TRANSFORMER IN and TRANSFORMER OUT knobs will commonly be referred to as T. IN and T. OUT.

Hold the shift key and left click the Amplifier Type Selector to turn the transformer circuit OFF. Left click cycles clockwise/forward, while right mouse click cycles backward/anticlockwise.

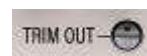
Hold the shift key when adjusting the T. IN or T. OUT knob using left click to trigger the MAIN OUT knob to compensate equally with opposite gain.

The Amplifier Type Selector features the following options:

- **OFF:** The gain is clean and will rival any high-end hardware mastering compressor.
- **Class A:** A transformer coupled class A amplifier featuring a clean and accurate reproduction of an analog signal.
- **Class A/B:** A transformer coupled class A/B amplifier featuring varying degrees of adaptive analog distortion.

Tip: Choose OFF for the most pristine gain adjustments, Class A for sharper transients and Class A/B for additional color and thickness.

Note: T. IN does not push additional level into the compressor. Please have a read over the Tips, Tricks and Techniques section of this user guide for details and ways to use the T. IN and T. OUT gain.



The TRIM OUT trim pot provides clean output gain that allows for level adjustment to the output signal after all other processing. This is useful for cleanly raising or lowering the output signal to ensure equal gain staging without the need for an external plugin. This is particularly useful when driving into the internal limiter (LMTR), which is positioned after MAIN OUT. Hold the shift key when adjusting the TRIM IN or TRIM OUT trim pot to compensate equally with the opposite trim pot.



Bypass allows the unaffected audio signal to pass through without being processed.



Delta solo allows you to hear the difference (or delta) between the wet and dry signal. This allows you to hear just what the plugin is adding to or removing from the unprocessed dry signal.



Dry polarity inverts the unaffected dry audio signal.



Wet polarity inverts the input signal, so all internal processing is applied to the inverted signal.



The external sidechain button enables use of an external source as the signal feeding the compression detector circuit. Consult the documentation for your DAW for external routing options and instructions.



The clipper selection button changes the position of the clipper relative to other controls in the signal path.



Left-click the button to move forward to the next option.



Right-click the button to move backward to the previous option.



Hold the shift key and left-click to switch to CLIP OFF from any other position.

CLP>OUT: Signal path is T. IN >> COMPRESSOR >> CLIP >> T. OUT >> MIX >> MAIN OUT. This is the default clipper position. Use this option to even out the signal after compression but prior to the output signal path to achieve firm control while allowing imparted character to remain unrestrained.

CLP>MAIN: Signal path is T. IN >> COMPRESSOR >> T. OUT >> MIX >> CLIP >> MAIN OUT. Use this option to get 'larger than life' sound that results from pushing a signal into a clipper that is simultaneously holding it back.

CLP>S/C: Clipper is placed before the sidechain. Signal path is T. IN >> COMPRESSOR >> T. OUT >> MIX >> MAIN OUT. Use this if you want to keep the raw signal unaffected by clipping but you still want an evenly controlled signal hitting the threshold.

CLIP OFF: No clipper. This allows the signal to go beyond 0 dBFS without being subject to clipping.

RAW>CLP: Signal path is CLIP >> T. IN >> COMPRESSOR >> T. OUT >> MIX >> MAIN OUT. Use this option if you want an audibly clipped signal that is also very controlled when it hits the threshold.

Note: As will be mentioned elsewhere in this document, for brevity, the TRANSFORMER IN and TRANSFORMER OUT knobs will commonly be referred to as T. IN and T. OUT.

OS OS Oversampling options allow P11 Abyss to optionally operate at a multiple of the host sample rate.

When oversampling is off, Abyss operates with zero latency at the host sample rate (x1).

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

INT INT (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 Abyss is being pushed.

VIN VIN (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 INT 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, Abyss applies the following logic:

- 44.1 and 48 kHz oversamples at x8
- 88.2 and 96 kHz oversamples at x4
- 176.4 and 192 kHz oversamples at x2, disabling the HD mode and enabling the INT and VIN modes
- 384 kHz disables oversampling options

TX LO
TX LM
TX MID
TX HM
TX HI The transformer selection option affects the infrasonic frequencies (below 20 Hz). Different cutoff frequencies up to 20 Hz are available. Setting it to LO results in more bottom end, setting to HI results in tighter bass. Use your ears to decide which works best for the material at hand. The default is LO.

Left click cycles forward, right click cycles backward.

MOD MOD When engaged gives you an alternative circuit based on Class A/B pre-amplification stage with compression behavior tuned by ear by the Pulsar Modular chief designer, Ziad Sidawi.

A B →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.

Note: When saving a preset, only the active parameters in the selected A/B slot are saved; the opposite slot is not saved.

Tip: Please review and enjoy the Presets Guide section of this document. This section outlines detailed descriptions of some of the included presets and gives guidance for how the preset designer recommends working with the preset. Where descriptions are not included, adjust the threshold while considering likely gain reduction targets as a starting point and tweak other parameters from there. There is a good chance the preset designer did not have 0.5 dB of GR in mind with a preset called 'Crush'!

SOUL 100 As is the case with all quality hardware, the SOUL is the signature sound of audio passing through the device without any parameters of the device being engaged.

The SOUL slider ranges from 100% where the full device signature is experienced, through to 0% where the signal nears a pristine digital state while still retaining the essence of the device character. In between these extremes, all manner of analog behavior that is possible but not easily achieved in the hardware world is revealed.

Tip: Really get to know MOD, PSI, SOUL and O2. These controls are at the heart of how Abyss is able to emulate the most beloved of hardware devices, ranging from the secret weapons of mixing legends to the most exquisitely designed processors utilized by the world's topmost mastering engineers.

Tip: Listen to how SOUL controls the richness of the device characteristics in our No-nonsense Audio Workshop at https://www.youtube.com/watch?v=EEA_KC6pXpo.

LMTR 0.0 dB 10.0 ms GR 0.0 dB Abyss features an analog style brickwall limiter that is positioned after the MAIN OUT in the signal path.

The limiter does not allow the signal to exceed the configured dBFS value. Take note, however, that since this is an analog-style limiter, overshoots may occur. Pushing the signal into the limiter results in very light but thick coloration.

The limiter release allows you to shape the sound from soft (10 ms) to hard (1 ms).



The routing switch switches processing between mono or stereo (DI), mid (MID) or side (SIDE).

Abyss does not do M/S channel processing within the same instance of the plugin. You will need to insert two instances, one for MID and one for SIDE if you decide to process both channels.

When working in MID or SIDE, Abyss outputs the combined channels (one being processed while the other is not) so as to hear the processing in the context of a stereo mix. If you want to isolate the channel, press the S button.



The S button (not available if DI is selected) allows either the mid or side signals to be heard in isolation. The S button will blink while engaged to remind you that it is soloing the channel.



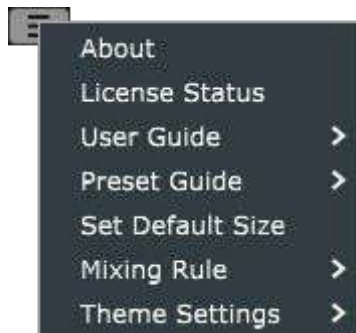
Switches the plugin faceplate from light mode to dark mode. Light mode is enabled by default.



Browse, load and save presets using the Preset Browser. Save over the current preset by clicking the left save icon or create a new preset with the right save icon. A red asterisk* will show up next to the left save icon to indicate the preset has been changed from its original parameters.

Note: Modified factory presets will be overwritten when updating the software unless the install presets option is deselected. User-created presets with different names than the provided preset names will not be replaced or deleted.

Note: The Light / Dark Mode and the background of the VU meter can be customized per instance and even saved as part of a preset. This becomes especially helpful when ALL your compressor instances are Abyss! Big blue GUI with a yellow meter for the master keeps it readable at a glance; smaller blue GUI with black VU for the drums, blue VU for the bass, white skin and blue VU for the vocals, etc.



Options Menu

About – Check the version number or demo expiration date.

License Status – Manage your license.

User Guide – Open the user guide.

Preset Guide – Open the preset guide. The preset guide provides a wealth of information on how to effectively use the signature presets. Signature presets are easily identified by the designer's initials, which appear at the end of the preset name. Get into the mind of the designer, find out the intention behind a preset, get recommendations on how to adapt it to your material and learn something useful along the way.

Set Default Size – Sets the default size for new plugin instances to the size of the current instance. This is a global setting. Existing instances will not be affected.

Mixing Rule – Sets the WET / DRY MIX behavior to one of the following options:

- **Linear:** This is the traditional or standard mixing rule commonly used by many plugins. The dry level is equal to full gain minus the wet level.
- **Balanced:** With MIX at 50%, the dry and wet signals are both at full gain, resulting in a level increase compared to Linear. As MIX turned counterclockwise towards WET, the

dry level is decreased. As MIX is turned clockwise towards DRY, the wet level is decreased.

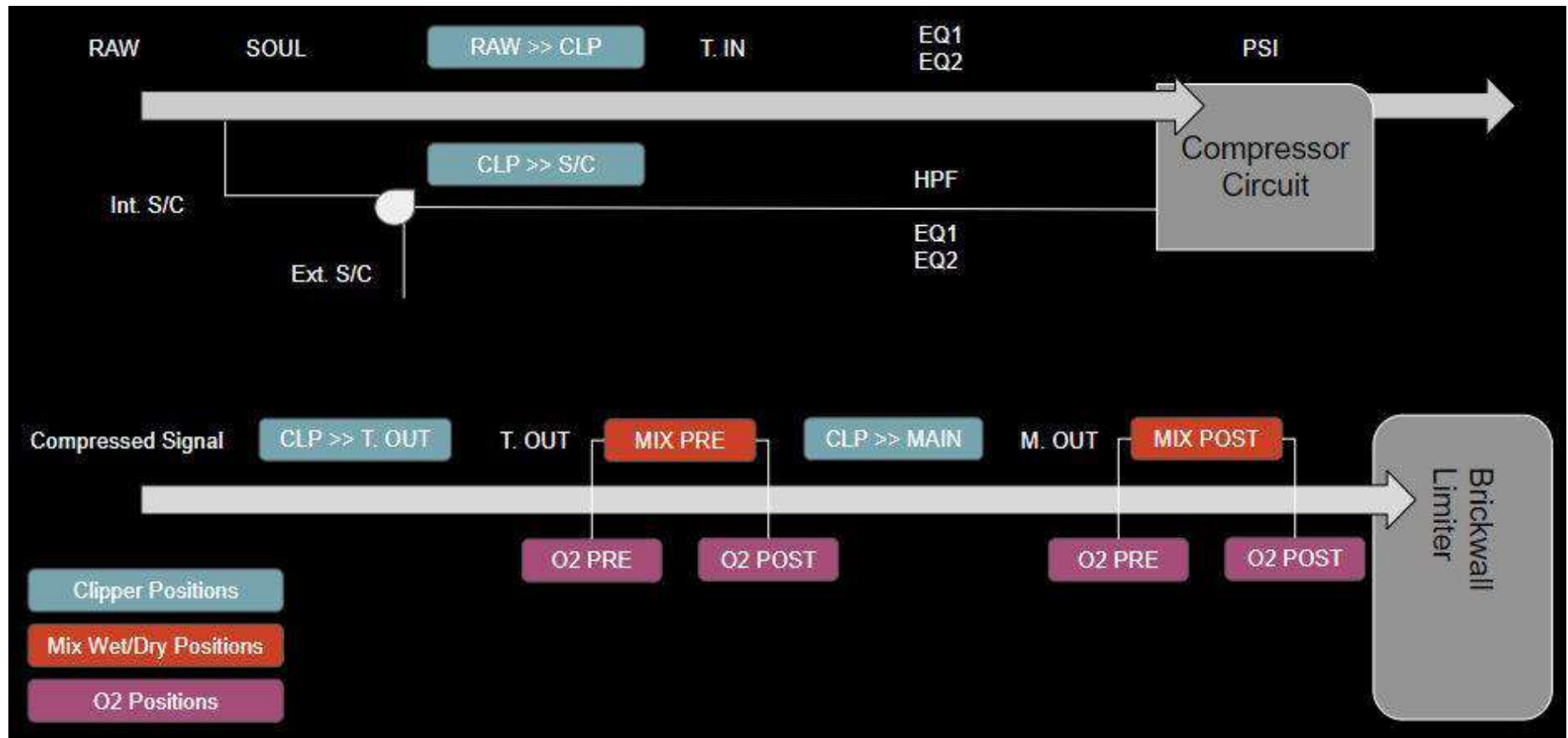
- **Sin3dB:** Uses a 3 dB equal power sine law, meaning that with MIX at 50%, the signal will have a 3 dB increase in gain. The relation of dry to wet signal is similar to that of the Linear option, however levels are adjusted using a sine wave shaped slope.
- **Sin6dB:** Uses a 6 dB equal power sine law. With MIX at 50%, this is close in level to the Linear option. Like Sin3dB, levels are adjusted using a sine wave shaped slope, however this is more subtle and natural than the Sin3dB option, with a behavior a little more similar to the Linear option.

Tip: Try Sin3dB or Sin6dB for more creative comb filtering effects and Linear or Balanced for more traditional mix behavior.

Theme Settings – Switches the faceplate and controls to a desired visual theme based on the following choices:

- **Bright:** The Bright theme is always used.
- **Dark:** The Dark (blue) theme is always used.
- **Grey:** The Grey theme is always used.
- **Preset:** The choice of Bright, Dark or Grey will be stored and recalled with each preset. When this option is used, a theme selection button is shown beside the Options Menu button.

Signal Flow Diagram



Tips & Tricks

- **When to use AUTO ATTACK and RELEASE? It depends on your audio**

Pun intended. Using AUTO ATTACK or AUTO RELEASE enables program dependent behavior in Abyss and this can be useful in several ways.

Think of AUTO ATTACK and AUTO RELEASE as virtual plainsmen. They continuously adjust the planes and rudder automatically to keep the ship as one with the surrounding depths, reacting to every current, every pulse, every tide in a sophisticated and coordinated unity with the depths.

If AUTO ATTACK is engaged and AUTO RELEASE is disengaged, Abyss digs deeply into the transient decay to create separation between the attack and sustain portion of the signal resulting in increased definition.

If both AUTO ATTACK and AUTO RELEASE are engaged, Abyss moves and breathes very naturally along with the source while simultaneously being able to keep a signal very stable and locked in place relative to other elements (depending on ratio and threshold settings). Try it with MOD engaged for very deep, clean, natural control. [KE]

- **How to reach deep into an uneven signal using the internal sidechain EQ**

When you want to compress a track that contains multiple instruments that are at uneven levels, the sidechain EQs (TO SC) can play a crucial role.

For example, on a full drum track you may want to compress a kick more than a snare, but your snare may be louder, so lowering the threshold to reach the kick as you desire might overly affect the snare. Use a sidechain EQ to push the level of the kick up in the internal sidechain so the threshold is hit more evenly or even with more emphasis on the kick than on the snare. This way, your compressor will now act on your kick and compress it as if it were deep inside the mix without being compromised due to the level of the snare.

For even more control over the compression action of an uneven signal, position the clipper before the internal sidechain (CLP►S/C) and knock down peaks before they hit the compression routine. [ZS]

- **The ins and outs of IN and OUT (transformers, that is)**

If you want your signal to have more presence, push T. IN. Although it is tempting to think so, this does not 'push' into the compressor, rather it increases the breadth of harmonic content in the signal that the compressor works on. If you are looking for the sensation that additional signal is being pushed into the compressor circuit, simply lower the threshold to get the same effect.

If you are looking for that 'larger than life' sensation that a pushed transformer brings, push T. OUT. This brings up the volume and harmonic content, working directly on the post compressed signal.

Use the TRANSFORMER Amplifier Type Selector for different colors. Think of OFF as clean, Type A as more transient and Type A/B as richer.

Experiment to find a balance between the T. IN and T. OUT along with MAIN OUT to find the sound you are looking for within the RMS you are targeting. Take for example, a scenario where you are compressing aggressively to around 10 dB of GR. If you want to hear more of the 'larger than life' sound, increase T. OUT and decrease T. IN. If you want to hear more snappiness in the transients, raise T. IN and lower T. OUT or use CLP►OUT. If you want both snappiness and 'larger than life', increase both T. IN and T. OUT while lowering MAIN OUT or use CLP►OUT. We got both kinds... we got Country AND Western.

For bonus points (aka more RMS level), set the clipper to be positioned before MAIN OUT (CLP►OUT) and clip the signal and/or push the whole thing into the brick-wall limiter. [ZS]

- **Motion in the ocean and propensity for density**

Dial in generous amount of gain reduction with super-fast attack (like, 3ms tops), auto release and low ratios (1.5:1 max), and this will work on practically anything as a marvelous densifier AND movement enhancer. This works equally well when both mixing and mastering.

Once you've dialed something groovy, please experiment with lowering the PSI drastically while simultaneously softening the knee and bringing in some O2: enjoy the magic, but don't stop there! Unlink channels, use some 1pole HPF SC, set the SOUL and clipper to taste... and let it bloom.

You're welcome :) [NH]

- **Keep it clean folks, this is a family show**

Abyss is full of long, hairy, silky mojo but it can also get pretty close to clean and pristine without ever venturing into the realm of stogy digital sterility. Try this on for size...

Engage the MOD circuitry to lower the noise floor but introduce additional harmonics (flip back and forth later to see what you like most). Adjust PSI to be somewhere between 0 to 2 (ish). Pay a lot of attention to what PSI is doing because it very literally changes attack and release character with every single tick.

Reduce the SOUL slider to peel back layers of emulated hardware circuit complexity. Adjust your makeup gain using MAIN OUT for clean gain.

After getting the threshold and PSI to where you roughly want them, consider using the EQ Target Switch set to TO IN to slightly push areas where you want a little more density or if you want to fine tune the way the compressor is being triggered, switch the target to TO SC and push the frequencies that will balance out the detector the way you want it to react.

All the above operations used like this work to minimize or avoid imparting additional Abyss mojo (but this does not mean lack of character - character is still there in spades).

Experiment with ⁰2 as well because while this is not clean per se, it enhances things in such a way that it imparts openness so it works really nice when wanting to enhance but maintain a pristine signal. [KE]

- **EQ your way to balance, selective density, framing, sculpting, fame and fortune**

Try these easy and intuitive sidechain and input signal EQ techniques to level up your dynamic control mastery (and look good doing it). You could discover that all the character and shaping capabilities you need are right here in Abyss.

For very smooth and balanced detector action and compression, fine tune the detector using the internal sidechain filters. Start by turning the sidechain listen function on by clicking the LISTEN button to the right of the S/C HPF knob. You are now hearing what the detector circuit hears. Set the EQ Target Switch to TO SC. Enable an EQ, set the MODE Selector to the bell boost option and increase frequencies that you want to hit the detector harder with.

Enable the other EQ, choose the bell attenuation option and move the slider to pull down frequencies that you want to hit the detector softer with. Mix and match based on what is needed. Use one, use both, set both to boost, set both to attenuate. Whatever is needed. Keep in mind that it doesn't need to sound good, it needs to sound balanced and even. Trust your ears first and foremost, but you can also keep an eye on the ballistic needle for how the detector is reacting. Don't forget to disengage the LISTEN button when you are done!

To increase or decrease frequency specific density, set the EQ Target Switch to TO IN so you are working with the input signal prior to compression. Imagine a scenario where you have an element that is a little muddy and needs more energy and presence (think electric guitar, main or backing vocals, keys, and so on). Set EQ1 to bell attenuation and EQ2 to bell boost using the MODE Selector. Set the attenuation frequency somewhere in the range of 250 Hz to 500 Hz and set the boost frequency to somewhere in the range of 1 kHz to 5 kHz (depends on finding what is troublesome or flattering to your specific instrument). Increase each slider to taste and listen for how the attenuation cleans up the signal while the boost increases presence and energy. An easy way to decide upon the best frequencies is to drastically overdo the amount of boost or attenuation and listen for where the instrument fits into the mix in the most effortless but over-exaggerated way and then back off the intensity until you are just left with clean, full, present and flattering sound. This is just the start. By the way, you can do the exact opposite for instruments that are thin and/or harsh.

The sidechain EQ can be used to fashion a frame of sorts around elements to either get them to punch through or to get them to be very finely controlled. For example, you can punch something up by cutting it into the SC. For a rounder, punchier kick + bass marriage, try cutting 100 Hz into the SC to let the compression frame around it. For another example, if you want to deal with those pesky fight or flight frequencies (2 - 4 kHz) dynamically, boost them into the SC so that the compressor becomes hyper vigilante of them. You can do both of these tricks in the same instance with two bands.

Also note that since the bell boost has a broad symmetrical fixed Q curve and the bell attenuation has a narrower proportional Q, placing them on the same frequency point creates a sculpting effect where a broad boost is complemented by a narrower cut at the same center point. Use this tip within a tip to combine different techniques as described above! [KE, RR, SDC]

Managing Presets

Basics

If the option to install presets is not de-selected during installation, the installer will overwrite the factory presets. User created presets will remain unaltered. To safeguard any modifications made to factory presets and preserve them during an update, make sure to deselect the install presets option when running the installer. Also, remember to save your own presets with different names using the 'save as' option located to the right of the preset browser.

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\P11 Abyss\Presets'

For macOS

'/Users/Shared/Pulsar Modular/P11 Abyss/Presets'

Modifier keys

Temporary bypass the parameter

CTRL+ALT (Windows) or CMD+OPTION (macOS) +Mouseover:

- S/C HPF
- O2
- CLP

Gain Compensate

SHIFT

- T. IN counters MAIN OUT.
- T.OUT counters MAIN OUT.
- TRIM IN counters TRIM OUT.
- TRIM OUT counters TRIM IN.

Fine adjustment of knobs, sliders and other controls

Hold control (^) on macOS or CTRL on Windows, then click and drag. Alternatively, right-click and drag without a key modifier.

Return controls to their default state

Press option (⌘) on macOS or ALT on Windows and left-click. Alternatively, double-click without a key modifier.

Enable parameters for automation (Pro Tools only)

Control + command + option (^ + ⌘ + ⌘) on macOS or CTRL + ALT + START () on Windows.

Uninstalling P11 ABYSS

For Windows

- VST3: 'C:\Program Files\Common Files\VST3\Pulsar Modular', locate the 'P11 Abyss.vst3' file and delete it.
- AAX: 'C:\Program Files\Common Files\Avid\Audio\Plug-Ins\Pulsar Modular', locate the 'P11 Abyss.aaxplugin' folder and delete it.
- Shared: 'C:\Users\Public\Documents\Pulsar Modular', locate the 'P11 Abyss' 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 macOS

- AU: '/Library/Audio/Plug-Ins/Components', locate the 'P11 Abyss.component' file and delete it.
- VST3: '/Library/Audio/Plug-Ins/VST3/Pulsar Modular', locate the 'P11 Abyss.vst3' file and delete it.
- AAX: '/Library/Application Support/Avid/Audio/Plug-Ins/Pulsar Modular', locate the 'PP11 Abyss.aaxplugin' folder and delete it.
- Shared: '/Users/Shared/Pulsar Modular', locate the 'P11 Abyss' 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.

Restrictions

The USER may not reverse engineer, disassemble, re-sample, create Impulse Response profiles or re-record, decompile, modify, alter in whole or in part PULSAR NOVATION LTD. audio plugins for the intent of renting, leasing, distributing, repackaging (whether for profit or not).

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P/N: 14961, Rev. 1.0

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