



# P455 MDN Sidecar

## User Guide

Version 1.5

## Marc Daniel Nelson: Biography



Marc Daniel Nelson is a Grammy and French Academy Award nominated mixing engineer, music producer and creative director. He has been mixing, producing and managing creative content for over 23 years.

His music credits include Fleetwood Mac, Jason Mraz, Joni Mitchell, Colbie Caillat, Eric Burdon, Ben Harper, Need To Breathe, Robert Duvall, Ozomotli, John Fogerty, Reik and more. As Protégé for both legendary producer / engineer Bill Schnee and Ken Caillat, Marc has carried the torch for impeccable quality sound and production.

His film credits include Solo, Blade Runner, The Vietnam War, Mulan, The Expanse, Wild Horses, Point Break, No Manches Frida, Fractured, Amanda, Father Figures, Ya, Ty, Vin, Vona and more.

His creative management credits include executive producing the 13 episode PBS television series, creating and executive producing the national video campaign for Guitar Center. Creative directing for Alcon Sleeping Giant, ArtistMax, Produce Like A Pro, Inside Blackbird and Warner Chappell.

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A stylized, handwritten signature of Marc Daniel Nelson in a dark, cursive font.

MARC DANIEL NELSON

## Foreword

My passion for the sounds behind the music started well over 23 years ago when I embarked on my first internship at a professional studio. Over the years, I became obsessed with the magic and personalities of the sonics – why some albums sounded more 3D and dream-like, and why some sounded more intimate and organic.

Albums like Fleetwood Mac's *Rumours* and Stevie Wonder's *Songs in the Key of Life* were two major albums that spoke to me with immense personality and character. It wasn't just the amazing music but also the sonic landscape that moved me, even at a very young age.

It wasn't until I learned the true meaning of critical listening under Bill Schnee and Doug Sax that I understood it wasn't about what you use, but how you use things to get the sounds you want. Gear has always been a huge part of the record making process for me, and my passions have always been tied to trying out all sorts of combinations of sounds while mixing – what a line amp can add or hurt in a mix, what a converter that captures your analog chain really does to the signature of that sound, and why compression can flat out make or kill your mix.

I spent many years testing out summing mixers, mastering EQs and compressors to find the best combination of clarity, size, and power. Being a mixer that works on many different genres in music, I was looking for a special chain that could work with any type of music – from orchestral scores to hip hop.

In 2018, I received a call from a friend wanting to sell me his Sidecar of the first 8 buckets of the Sausalito Record Plant's beautiful API console. This console recorded not only *Rumours* and *Songs in the Key of Life*, but also a massive list of historical albums from artists like Bob Marley, The Grateful Dead, Huey Lewis And The News, Rick James, Prince, and many many more. I found that using this as a 4 stereo stem summing mixer instantly brought me the familiar sound I grew up loving – massive headroom, a forward midrange, and just the right bit of saturation when pushing into it. Pairing this Sidecar with my AT-101

Fairchild and a stereo API 5500 equalizer clicked with how I envision music in my head. This EQ gives a very punchy sound, and the AT-101 gives a velvety chocolate fatness.

This brings me to the Analog-to-Digital converter that “captures” my mix after it leaves the analog domain. The one thing I’ve never really seen considered while designing a virtual version of analog hardware is the sound of the box capturing the analog signal and returning it into the digital world of the DAW. Each converter has its own sonic character, and that imprints a huge final effect to the chain. I chose my two favorite mastering grade Analog-to-Digital converters as they both have such different sounding identities.



So, what is the P455 exactly? After almost a year of development, Pulsar Modular and I designed what I’d consider the closest stereo channel plugin to emulate the characteristics of console summing, as well as being able to recreate the effect of pushing into a console. Pairing that with my favorite tube line amplifier, master stereo EQ, and light tasteful compression. The frosting on the cake, and what I would call the magic of the P455, are the Analog-to-Digital conversion character settings – Willie and Atticus. The Willie setting is very punchy, open, and forward sounding. The Atticus setting is wide and magical sounding, with notes that seem to extend, and the stereo field gives a sense of holographic imagery.

With all the choices of the P455, you have the flexibility and power to achieve the best pairing for whatever music comes your way to faithfully achieve the space, depth, and clarity that you get out of the box. I can’t think of another piece of software that has been able to do what this can do.



Marc Daniel Nelson

# Preface

P455 began with a simple idea – an idea fueled by collaboration, a shared passion for sonic excellence, and the vision of bringing a complete audio workflow to your fingertips.

My acquaintance with Marc Daniel Nelson traces back to his use of P42 Climax and his invaluable beta testing contributions to P11 Abyss. What began as a conversation about emulating his distinctive 2-bus stereo EQ expanded into a grander vision – the emulation of his entire audio workflow. Marc's openness about his signature sound, showcased in educational videos, laid the foundation for a project that aimed not only to replicate but to enhance.

In the spirit of Marc's streamlined yet potent external audio workflow, simplicity is power. You, the user, wield control effortlessly – choosing the ADC, opting for the tube line amp treatment, or deciding between the 2-channel tightness or the 8-channel expansiveness on the Sidecar mixer. The magic lies in your choices, while the meticulously selected high-quality equipment Marc curated takes the lead in the heavy lifting.

In alignment with this philosophy, the compressor was conceived. Our vision was clear: the compressor should handle the intricacies, allowing the user to listen and request more or less with ease. The workflow emulation already weaves a tapestry of color and saturation, leaving the compressor to provide a subtle touch of cohesion when desired.

To realize this vision, we engineered a bespoke 2-bus compressor, finely tuned for compatibility within this unique workflow. Auto attack and auto release features gracefully adapt, and user-friendly dials grant you the power to request more or less grip, a tightly stretched or leisurely release. The ratio options are delicately calibrated, yet they reveal a magical compression prowess when pushed assertively with a low threshold. It's a testament to the belief that sometimes, in the realm of audio, less truly is more.

As we delved into the intricacies of Marc's equipment choices and the compressor's operation, it became evident that we were not merely replicating hardware – we were rebuilding, aiming for a sonic experience better, stronger, and faster, echoing the ethos of the TV classic Six Million Dollar Man.



The development process may have seemed a fool's errand to anyone observing it from the outside. The ultimate success of this project was built upon countless failed attempts. Much like with past Pulsar Modular efforts, we decided that if we could not meet or exceed all of our demanding requirements, we would not release the software. You are here reading this foreword, so the results of those promises to ourselves are no mystery. Marc's unwavering commitment, keen insights, and insistence on preserving the soul of his signature sound steered the project through countless iterations. It felt like catching shadows at times, yet with every failure, success loomed nearer.

We invite you to experience the culmination of dedication, collaboration, and the relentless pursuit of audio perfection.

Ziad Sidawi  
Audio Equipment Designer & CEO  
Pulsar Novation LTD










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

  Polarity inverts the audio signal.

*Tip: You might think that flipping polarity simply results in an inverted copy, but keep in mind that the push and pull of speaker cones is also inverted. This may result in additional punch, body and impact. Try it on the main bus!*

   The Hammer is a program and frequency range dependent circuit that selectively pushes and saturates different audio regions using unique weightings, without increasing the peak level.



Intensity is not universally applied, with lows and the lower end of the high midrange being more predominantly affected than other regions. It is an additional boost of power, just as the name suggests.

  Stage Focus affects the location of elements within the soundstage.

When it is enabled (default), the results are true to Marc's analog chain. This option prioritizes separation, space and spread between elements. When it is disabled, elements still benefit from the characteristic enhancements that

P455 imparts, but element interactions are tighter, more up front and more focused.

*Tip: On a full mix, having Stage Focus enabled can impart a more relaxed, vintage vibe, whereas having it disabled can impart a modern vibe. This doesn't have to be an all or nothing proposition. Consider having Stage Focus enabled on buses such as drums or backing vocals to help them sit back in the soundstage, and having it disabled for lead lines and vocals to enhance the forward and focused qualities of these elements.*

  Analog Variance is a characteristic of all analog equipment. Random fluctuations can occur due to various factors such as electrical interference, thermal noise, component imperfections and environmental conditions. This can introduce small variations or distortions into the signal, causing it to deviate slightly from its original form.


When the default STEP option is engaged, P455 produces 1:1 results that match the grit and aggressiveness of Marc's analog chain. When the SMOOTH option is enabled, variations are continuous and fluid, which leads to less distortion and saturation.

*Tip: If you prefer the SMOOTH option to be your default, simply open the Default preset, switch to the SMOOTH icon and save the preset.*



**OS OS** OS enables or disables EQ oversampling. This option is available at sample rates of 88.2 kHz or higher when the EQ is enabled. EQ oversampling is always active at sample rates of 48 kHz and lower.

**GR 0.0 dB** The Gain Reduction Meter provides a precise numerical representation of the RMS dB gain reduction applied to the audio signal by the 2-bus compressor. This meter provides continuous real-time updates with a predetermined hold time.

 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.*

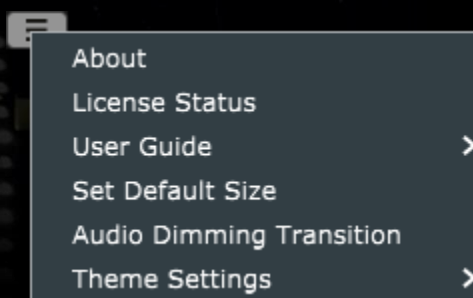
**A B ↔ B** A/B allows for temporary storage of different settings for quick comparison. The arrow button allows for copying the active side to the inactive side.

*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.*

**MOD1 MOD2** Analog-to-Digital Converter MOD options. These options become available when 8 CH Summing is selected and the LINE AMP circuit is ON. MOD1 represents the most accurate recreation of the hardware while MOD2 gives a different flavor that extends the usability of the emulation.

**UI NEW** Choose the look of brand-new factory fresh piece or a vintage barn find treasure. Either way, sound will not be altered.



Options Menu

About – Check the version number or demo expiration date.

License Status – Use this option to authorize or deauthorize your software. The internet is required for this action.

User Guide – Open the user guide in PDF format.

Set Default Size – Saves the size of the current P455 GUI as the default size used when inserting new P455 instances.

Audio Dimming Transition – When checked (by default), the audio will dip momentarily by -3 dB when switching between ANALOG, LINE AMP, 2 CH / 8 CH and MOD1 / MOD2 options. This will reset your ears before hearing a different emulation. If you don't want this behavior, you can uncheck this option and save to your Default preset.

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

- New: The New theme is always used.
- Old: The Old theme is always used.
- Preset: The choice of New or Old will be stored and recalled with each preset. When this option is used, a theme selection button is shown beside the Options Menu button.

# Metering / Gain Staging

## RMS INPUT / OUTPUT Meter

Measures average audio signal levels. RMS metering helps to visualize levels in a similar way to how they are heard.

## GLOBAL IN / OUT

Bypass control (synchronized with the toolbar bypass).

## INPUT

Raises or lowers the raw input signal prior to hitting the input transformer.

## RAW / RAW + IN

Input corresponds to the white ballistic needle. Click to change the measurement point (SHIFT-Click changes both L and R).  
If RAW is selected, it is measured at the beginning of the signal chain.  
If RAW + IN is selected, it is measured after the INPUT knob.



## OUTPUT

Output corresponds to the red ballistic needle.

## PEAK dBFS

Peak dBFS numeric readout with brief hold.

## Channel Summing

8-channel mixer or 2-channel stereo bus.

## BIAS

Calibrates the INPUT transformer and LINE AMP to run hot or run cool.



The P455 VU meters provide a nuanced representation of perceived loudness by presenting the average input and output levels calculated over a brief period (RMS). Peak dBFS values are shown in the lower quadrants of the meters.

The white needle indicates the input signal, while the red needle indicates the output signal. Clicking on the RAW + IN label adjusts the position in the signal path where input is measured. Holding the SHIFT key while clicking changes both simultaneously.

**RAW + IN** The input needle reads the raw signal entering the plugin.

**RAW + IN** The input needle reads the sum of the raw signal + the INPUT knob level adjustment.

*Tip: RMS is useful for understanding the average loudness of a track. You may want to consider the difference between the*

*peak and RMS values. For example, if there is a large gap between RMS and peak values, you can assume that this audio track is fairly dynamic, while if RMS and peak values are close together, you can assume the track is heavily compressed or has minimal dynamics.*



GLOBAL IN/OUT is a bypass control, allowing the unaffected audio signal to pass through without being processed.



INPUT controls the amount of signal pushed into or pulled back from the input transformer. Pushing (clockwise) results in a more saturated and circuit-compressed signal.

Pulling (counterclockwise) results in less saturation and less circuit compression.

INPUT affects the input transformer non-linearly. As it is pushed, varied harmonics are generated and its behavior fluctuates as it reacts to different signal levels. This can result, for example, in a low-end bump or a tightening of the mids.

Hold the shift key while adjusting INPUT using the left mouse button to have OUTPUT L and OUTPUT R automatically compensate. Do the same but use the right mouse button for fine control.



P455 offers the sound and feel of analog summing without the need to insert plugin instances on each track.

Channel Summing can be set to either 8 CH for an emulation of an 8-channel mixing board (4 stereo stems) or 2 CH FOR a 2-channel bus (single stereo stem).

This feature emulates a signal processing workflow whereby stereo audio signals are routed from a DAW to an 8-channel analog mixer, where they are summed and take on distinctive characteristics of the hardware. The 8 CH option gives a wider, more relaxed, spacious feel. The 2 CH option gives tighter, more aggressive color.

*Tip: There are no rules that dictate 2 CH must be on one type of source (e.g. mix bus) and 8 CH must be on others (e.g. drum bus). Rather, consider what a track simply needs. 2 CH is more center focused, tighter and grittier with articulate, punchy mids. 8 CH is wider and deeper with a relaxed center low end and a wider all-encompassing stereo image. Use these characteristics to your advantage throughout your session, whether they are single stereo tracks, sub-groups or buses.*

*Tip: Real-time emulation of an 8-channel mixing board requires complex non-linear, cumulative and mutually interactive calculations. As such, for optimal results, consider integrating P455 into the mix from the very beginning, since its distinctive*

*characteristics play a foundational role in shaping the overall sound. If working with stems, a stereo mix or if adding P455 during the late stages of a mix, audition different emulations to determine what works best in that context.*



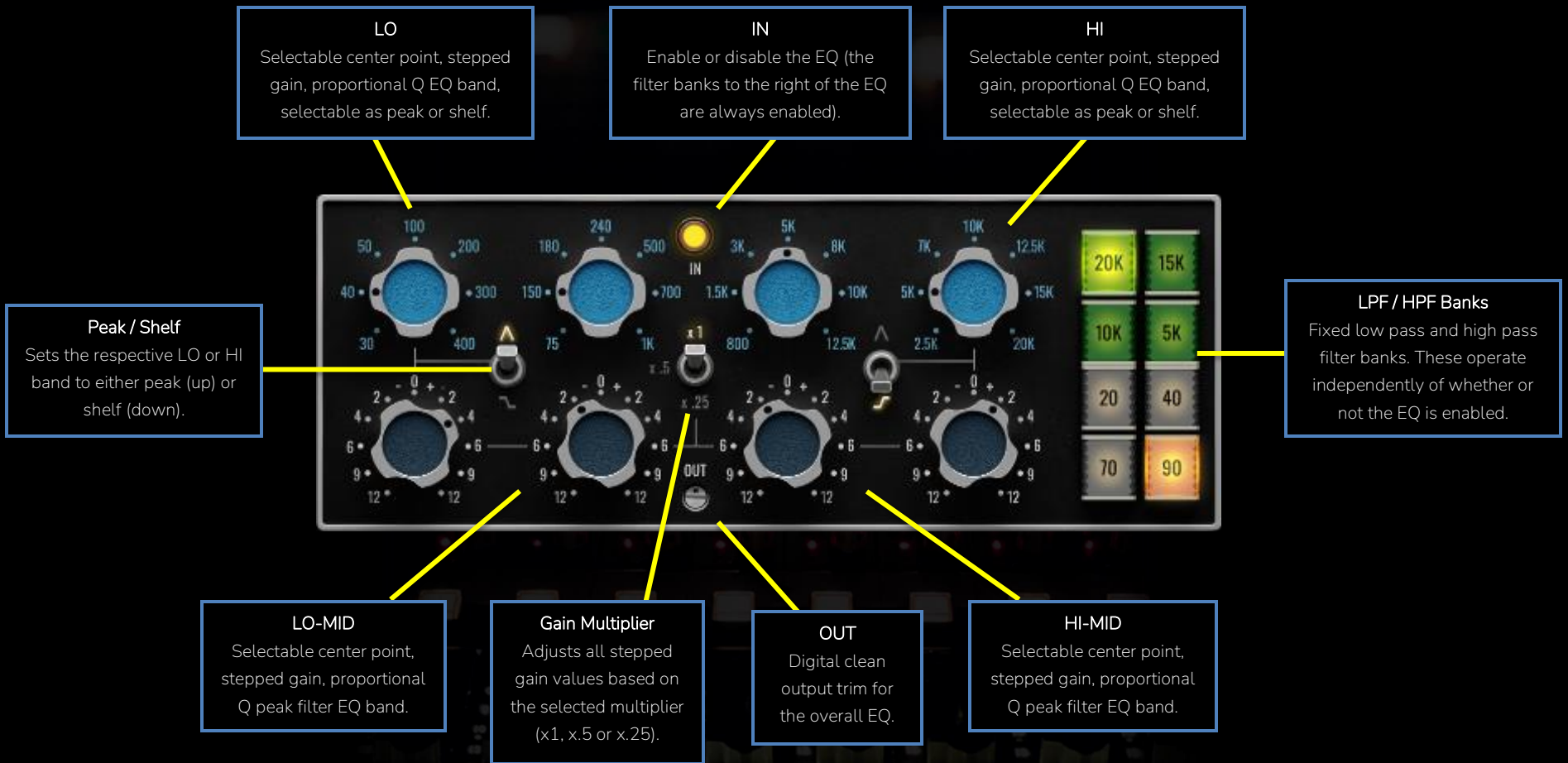
BIAS calibrates the INPUT transformer and LINE AMP circuits, enabling them to 'run hot' with less available headroom or 'run cool' with more available headroom.

Turning BIAS clockwise will calibrate the circuit for a tighter, forward, more aggressive tone. Turning BIAS counterclockwise will calibrate the circuit for a more relaxed, deeper tone with more clarity and space.

*Tip: Review both the BIAS and INPUT sections of this document while considering the complementary relationships that can be created between them. Experiment with your ideas on a mix bus without touching other controls to get familiar with all the colors that can be achieved just with these two parameters.*



# EQ and Filters





To fully realize Marc's workflow, the P455 EQ initially drew inspiration from a renowned stereo bus EQ known for delivering punch. The P455 EQ improves upon the original with no noise, more balanced delivery, reworked and highly improved high frequency response and better sounding low end – all while maintaining its highly sought-after punch



The IN indicator LED switch can be pressed to enable or disable the EQ. The EQ imparts hardware-like passthrough characteristics to the audio even if all bands are in neutral gain positions.

*Note: When the EQ is enabled, the signal is oversampled within the EQ module only. As a result, enabling the EQ introduces latency and delay compensation is required. When the EQ is disabled, P455 operates at 0 latency. See the OS section of this document for more information about oversampling when operating at sample rates of 88.2 kHz and higher.*



The P455 EQ offers four frequency bands logically arranged from low to high: LO, LO-MID, HI-MID and HI.

Each band features selectable frequency points and stepped gain with proportional Q, whereby the Q factor is adjusted

dynamically based on the amount of boost or cut. As the boost or cut is increased, Q narrows proportionally.

Every frequency point underwent meticulous tuning by ear at each gain step and multiplier setting, expressing craftsmanship in every detail.

To temporarily bypass EQ bands, press and hold CTRL+ALT (Windows) or CMD+OPTION (macOS) while the mouse is over a control. Additional bands can be bypassed by continuing to hold the key combination and hovering the mouse over them.



The leftmost and rightmost bands can be set as either peaking or shelving filters by using the corresponding peak/shelf switch. The up position selects peak, while the down position selects shelf.



The Gain Multiplier switch changes the stepped gain dB amounts according to the selected position.

Calculations are made by multiplying the selected option (x1, x.5 or x.25) by the value corresponding to the tick on the gain knob.

Tip: While you might get similar gain value if you used 2 dB gain with an x1 multiplier verses a 4 dB with a x.5 multiplier, please note that the frequency point and the shape of the Q is a bit different. As such, use your ears to audition which works best on the audio you are working with.

Note: When gain is adjusted or a gain knob is clicked, the popup numeric display shows the calculated gain, which is the selected gain value multiplied by the multiplier switch value.



The EQ output trim pot provides clean digital boost or cut. It can be used to adjust the overall EQ gain to compensate for EQ band adjustments.



The LPF (low pass filter) and HPF (high pass filter) banks come after the EQ and are not part of the EQ circuit. Turning off the EQ will not affect these filter banks.

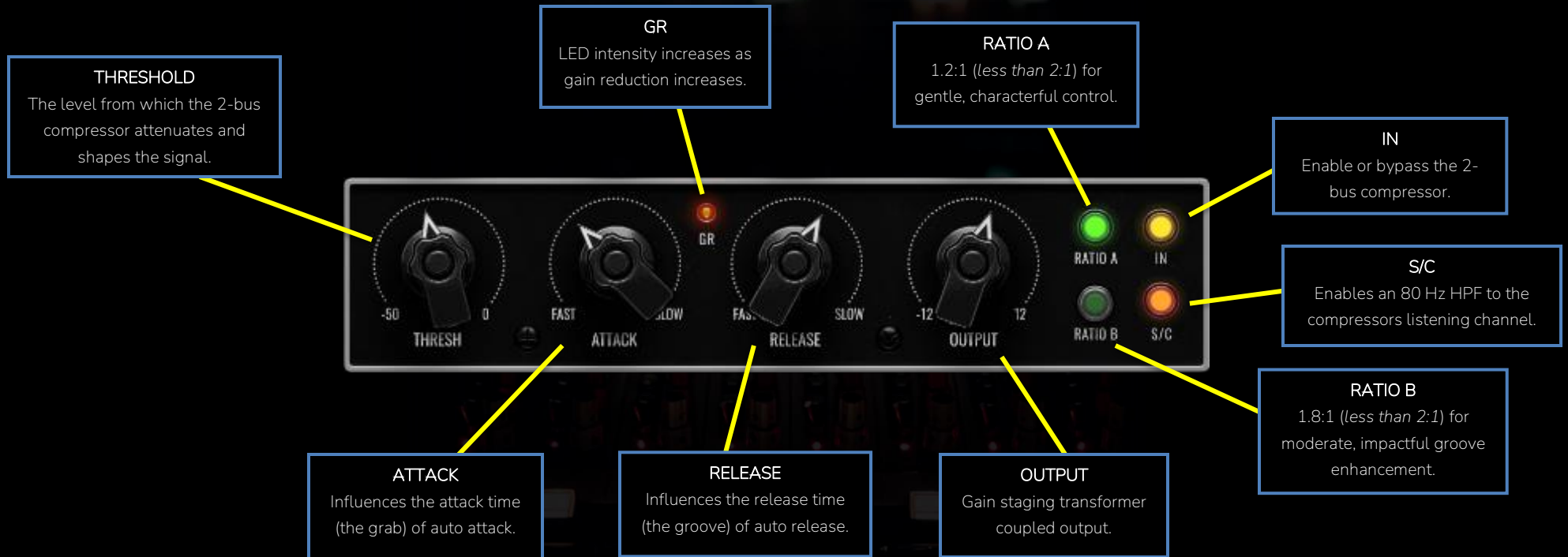
The low pass filters are exceptionally smooth and the high pass filters are particularly punchy due to their very unique designs. These are not your garden-variety filters.

Clicking on a frequency point button toggles it on or off. One frequency point per bank can be on.

Note: As an extension of being independent from the EQ, the LPF and HPF banks are not enabled or disabled with the EQ IN indicator LED switch. The selected option is always enabled.

Tip: Pushing a band or shelf EQ into these filters produces beautiful, playdough-like shapes. As a simple example, try enabling the 20 Hz HPF along with a 30 Hz low shelf bump of 2 dB on a kick drum for simultaneous tightening and embeefening; or try a 5 kHz LPF with a 4 dB boost from the high shelf at 2.5 kHz on a drum group. This will contain the drums, focus it and make it super punchy with less high frequencies.

# Custom 2-Bus Compressor





The P455 custom 2-bus compressor embodies a wholly original design, independent of any specific device or class of devices. It was conceived and brought to life as a seamlessly integrated companion to all other characteristics of Marc's signature sound and workflow.

The intent is to provide effortless, beautiful, flattering sound that allows for making artistic and not technical decisions. No need to fret over details. Engage the 2-bus compressor, pull the threshold down and let it automatically work the groove. Choose the most flattering ratio and if needed, adjust the automatic behavior to be a little faster or slower. That's it, just listen and enjoy compression that is one with your sound.



THRESH (for threshold) controls the level from which the 2-bus compressor will affect the dynamic range and feel of the audio signal.

The detector circuit is program dependent and as such, the range scale represents a general reference point (as opposed to being a precise value measurement such as RMS or peak dB). While adjusting THRESH, listen, feel and observe the intensity of the GR LED and check the toolbar GR meter if necessary (but trust what you hear despite what you may see).

The 2-bus compressor's listen channel is always active and fed directly with the raw input signal. The compressor's internal sidechain is made up of a series of custom ear-tuned emphasis and de-emphasis curves. Parameter changes within P455 (outside the compressor module) will not affect the behavior of the compressor (since it is listening to the raw input signal). This design choice allows for shaping the audio, character, image and color, without affecting the gain reduction on the signal.

*Tip: Notice how the compression shape and action very musically follows and accentuates the groove, regardless of where ATTACK and RELEASE are set. Take the time to really feel how the 2-bus compressor is enhancing the mood, texture and emotion of the audio. Fine tune ATTACK and RELEASE while focusing on these sensations. Turning THRESH very low temporarily to exaggerate the gain reduction and compression action may help to quickly dial in the feel you want.*



The attack time adapts automatically according to changes in the input signal. By doing so, the 2-bus compressor optimally processes the rapid onset of transients, ensuring balanced and controlled output while respecting the natural dynamics of the source.



The ATTACK knob grants the ability to influence the reaction speed from FAST to SLOW. This allows for additional precision in shaping the dynamics if the automatic attack is not perfectly adapted to the intricacies of your unique audio.

*Tip: Set ATTACK anywhere where it flatters your audio of course, but auto-attack is very good at what it does, so if the default position isn't already perfect, somewhere between 10 and 2 o'clock generally nudges it into the ideal position.*



The GR LED illuminates when gain reduction is occurring. It is synchronized with the attack and release timings as well as the intensity of the compression.



The release time responds automatically to the varying characteristics of the input signal. This allows the 2-bus compressor to adapt to rhythmic flow as it transitions from measure to measure and from section to section throughout a composition.

In scenarios with quick transients, the release time is shortened, allowing the 2-bus compressor to recover swiftly and avoid unnecessary pumping or distortion. Conversely, in situations with sustained sounds, the release

time extends, providing a smooth and transparent reduction in gain.

The RELEASE knob serves as a manual adjustment control to fine-tune the automatic release behavior of the 2-bus compressor from FAST to SLOW. This dynamic feature accommodates the unique characteristics of different audio sources, providing the flexibility needed to achieve nuanced and tailored compression.

*Tip: To encourage more movement, create some space after the current recovery and the onset of the next transient by adjusting the knob toward FAST. To smooth out and stabilize the dynamics, adjust the current recovery time to slightly overlap the onset of the next transient by adjusting the knob toward SLOW. The default position for RELEASE is often perfect, but if not, a position between 10 and 2 o'clock is typically ideal.*



The OUTPUT knob adjusts the output level after compression. This makeup gain amplification stage features a Class A transformer that provides clean, balanced and characterful analog boost and attenuation.

Hold the shift key while adjusting OUTPUT using the left mouse button to have OUTPUT L and OUTPUT R

automatically compensate. Do the same but use the right mouse button for fine control.

*Tip: Creative use of internal gain staging abounds with options to use deep compression threshold with level compensated makeup gain; or to lower the INPUT knob (harmonics) and raise the 2-bus compressor OUTPUT (heft); or attenuate with the EQ OUT trim pot and compensate with the 2-bus compressor OUTPUT. These are just a few examples so be sure to experiment!*



The 2-bus compressor features very gentle ratio options, both of which are less than 2:1. RATIO A sets the ratio to 1.2:1 and RATIO B sets the ratio to 1.8:1. Be assured that these seemingly polite ratio options leave nothing to be desired when it comes to strong character. The P455 2-bus compressor vigorously hugs audio signals and noticeably increases rhythmic impact.

*Tip: While the 2-bus compressor was designed with a focus on enhancing cohesion in mind, in practice, it has revealed itself to deliver exceptional results when using extreme settings. Try using RATIO B with THRESH dialed very aggressively to give 10+ dB of GR and adjust OUTPUT to make up for the gain reduction. Even at these settings, the 2-bus compressor maintains a deep and natural feel.*



IN enables and disables the 2-bus compressor.



S/C enables or disables the internal 80 Hz sidechain high pass filter. By enabling this, the 2-bus compressor becomes less sensitive to low-frequency content that can excessively trigger compression.

*Tip: This can be particularly useful in scenarios where you want to preserve the natural low-end of a mix or when dealing with instruments with strong low-frequency components, such as kick drums or bass guitars.*

# The Dog Pound / Output Stage





Analog-to-Digital Conversion (ADC) is a fundamental part of modern audio workflows that span in-the-box and out-of-the-box hybrid workflows. The flexibility to choose the right ADC for the right situation is a critical part of Marc's workflow and signature sound.

The quality of the ADC directly impacts the overall sonic integrity of the sound source. A high-quality ADC captures not only the analog signal with precision, it also imparts its own nuanced signature on the signal.



P455 offers the choice between two accurate recreations of Marc's favorite ADC options. Meet Willie and Atticus. Willie is the Norwich Terrier on the left. Atticus is the Weimaraner on the right (and can often be spied lounging on the couch behind

Marc in many of his videos).

Willie is tight, punchy, strong, open and forward.

Atticus is laid back, velvety, wide and magical with an ability to extend notes and create holographic imagery.



The LINE AMP circuit is a painstakingly realized recreation of Marc's most iconic mojo boxes. Comprised of an enviable selection of tubes, transformers and inductors, these dual devices are a key part of the signature sound Marc has crafted over years of experience and experimentation.

The results of engaging the LINE AMP are immediate and breathtaking. The audio undergoes an expansive and dimensional transformation, extending not only along the conventional vertical plane but also expanding its presence horizontally and spatially.

The expansion and extension of the audio signal in various directions can be attributed to the perfect marriage of tube harmonics and transformer saturation.

This is what everyone is talking about when they say 'soundstage'.

*Note: When the LINE AMP is OFF, it continues to be modeled, but with tubes off (in other words, the device is not true bypass, so signal is still affected by virtue of passing through it).*



OUTPUT L and OUTPUT R provide clean digital gain at the end of the P455 signal path.

Hold the shift key while adjusting OUTPUT L and OUTPUT R using the left mouse button to have INPUT automatically compensate. Do the same but use the right mouse button for fine control.



The link button links OUTPUT L and OUTPUT R together so adjusting one control results in simultaneous adjustment of the other control.



## Tips, Tricks and Techniques

### **.: Understanding INPUT and BIAS .:**

*The INPUT knob pushes signal into or pulls signal back from the internal circuits at the onset of the signal path, prior to the input transformer. Increasing INPUT (clockwise) results in a subtly saturated and tightened signal with a reduced sense of depth. This is very useful for encouraging defined positioning when you want to place something toward the front or the back. Conversely, decreasing INPUT (counterclockwise) leads to a more open, breathy, dynamic sound, which is very useful on key elements and the overall mix.*

*The BIAS knob operates in somewhat of a similar manner in that it also encourages tightening or relaxing of the signal, but it does it in a different way and in a different position in the signal path for different and complementary characteristics when compared to INPUT. BIAS calibrates operating characteristics of the INPUT transformer and the LINE AMP circuitry, allowing them to operate anywhere from hot and aggressive to cool and relaxed.*

*So, how can we use these circuits to our advantage? There are many possibilities, but we will introduce one here for you to kickstart your imagination. Start by pushing INPUT to achieve a desired low-end bump and saturation. If the bump and saturation is dialed in perfectly but the overall feel is too tight and aggressive, turn the BIAS knob counterclockwise to bring the signal back down to the original level (but now it maintains the bump and saturation). Now try the same approach but go in the opposite direction for both knobs. Pull INPUT back to reduce unneeded harmonic content and push BIAS to compensate. Compare and think about everywhere you can go just using these two parameters.*

### **.: P455 as stem channel strips .:**

*P455 is highly effective when used as a channel strip for stereo stems (aka group buses) since it provides cohesion within a group and as an extension of being self-complementary (hence cohesion), it encourages complementary and mutually flattering character between groups.*

Take a typical setup of stems for a contemporary pop or rock song for example. You may have groups such as drums, bass, guitars, keys, main vocals and backing vocals. Use an instance of P455 on each group and focus first on just the summing, Analog-to-Digital conversion and line amp settings that work best for each group (make sure to listen in the context of the whole mix because it is the inter-relationships you are after here).

As you are auditioning options, listen for thickness, depth, positioning and imaging within the group and judge how it works in relation to all the other groups. Think about the characteristics of each component and how that will help to present the production as you envision it. Willie is punchy and attacky. Where will that help? Atticus is more saturated and deeper. Where could that make something sit better? The 8 CH summing option is wide and relaxed. Is there a group that could benefit from that? The 2 CH option is tighter and more focused. Is there a group that needs this?

All settled and eager to do more? Look over the 'Understanding INPUT and BIAS' TT&T entry, then consider some subtle finalization with EQ and compression settings if anything more is needed or desired.

## **.: P455 on individual tracks :.**

P455 is focused on uncompromising quality, which means there is a slight CPU tax to be paid. If you have a modern CPU, or are not averse to freezing the odd track, P455 has proven to be a wonderful track level channel strip for use just on key tracks or on every track if you wish.

P455 does not suffer from excessive buildup of character when used for multiple stages. The variable colors/BIAS really help to create separation between tracks without getting muddy or harsh in any way. Even when all are set exactly the same, there are no issues with buildup of frequencies over many channels.

Soundstage width, height and depth is definitely enhanced with this plugin, making placement within the mix easier. There is more space around everything, hence more space to play. Surprisingly, even though it has oodles of character, it is not overbearing in any way. More or less BIAS results in a closer/farther kind of thing (similar to P440 but more range). I really love how BIAS is almost a compressor in its own right which in turn gives the P455 2-bus compressor more range/aggression because the dynamic range is already tamed quite a bit if you crank BIAS.

*Originally I thought this would work only on group/master duties but aside from a few issues that required a parametric EQ to fix specific problems this just got out of the way and told me to have fun! And it IS fun. It gives immediate and inspiring results. Add to that a willingness to work with just about everything I have without any issues whatsoever and I think we have a new standard in what can be expected from an algorithmic plugin.*

*When working with individual elements like a pair of guitars, the first thing I recommend is to play with the gain setting. 1 dB, .5 dB and .25 dB all sound very different to each other and are at least as distinct as Willie and Atticus are from each other. It is well worth finding which one works best for the particular guitar. Then you just pile more gain if the values are smaller but (and I cannot stress this enough) the setting's sound shifts the spectrum in interesting ways. The more complex the source the more noticeable it is.*

*Additionally, you can push the low shelf into the HPF and the bottom end is nice and tight. This thing (like all PM EQs) cuts beautifully and once you tighten the bottom and cut the gunk the top shelf does the "sparkle without hardness" thing and viola, crunchy, beefy guitars. [LA]*

## **.: Compression without a compressor :.**

*Each P455 emulation, even with BIAS and INPUT knobs at zero and with the EQ and 2-bus compressor off, will regulate micro-dynamics by shaping and breathing into your signal in a non-linear way. Each emulation interacts with your signal frequencies differently and delivers a different behavior.*

*Some of these interactions can be described as: how each emulation grabs the transients or releases them; how they translate the low end; how they emphasize the upper-low end frequencies or go for deep sub-low frequencies; how they make the vocals sit in the mix, pushed on top, floating or working side by side with the other elements.*

*Even without turning on the 2-bus compressor, each emulation delivers non-linear compression behavior, element positioning in the stereo image and movement.*

*Now if you bring into the picture the INPUT and BIAS knobs, you will be able to expand even further on the above-described behavior.*

## **.: Understanding the VU meter, gain staging and the INPUT knob .:**

*P455 measures audio levels using dBFS, where -18 dBFS corresponds to 0 dBVU on most typical hardware analog VU meters.*

*The ballistic needles in the P455 VU meter display the RMS level. RMS is a measurement of the average level of a signal over time. The needles do not display peak levels. The peak display readouts in the lower corners of the VU meter show the exact dBFS peak of the input and output signals.*

*You will want to find the sweet spot for your music by adjusting the INPUT knob to gain stage internally. To automatically compensate for the output level, hold the shift key while adjusting INPUT. To achieve more harmonic saturation, push the INPUT knob harder (clockwise). To achieve more depth and headroom, dial the INPUT knob back (counterclockwise).*

*Important to note:*

- The input transformer will saturate and act as a limiter as the signal approaches 0 dBFS.*
- The output volume faders are clean digital gain, so boosting or attenuating these won't alter the signal harmonics..*

# Managing Presets

## Basics

If the option to install presets is not deselected 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\P455 MDN Sidecar\Presets'`

### For macOS

`'/Users/Shared/Pulsar Modular/P455 MDN Sidecar/Presets'`



# General

## Mouse and Key Modifiers

### Fine adjustment of knobs, sliders and other controls

Press and hold control (^) on macOS or CTRL on Windows, then left 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.

### Temporarily bypass one or more EQ bands

Press and hold command + option (⌘ + ⌘) on macOS or CTRL + ALT on Windows while moving the mouse cursor over the chosen EQ bands to bypass them. Release the modifier keys to re-engage the bypassed bands.

### Enable parameters for automation (Pro Tools only)

Press control + command + option (^ + ⌘ + ⌘) on macOS or CTRL + ALT + START (⊞) on Windows.

## Uninstalling P455 MDN Sidecar

### For Windows

- VST3: In 'C:\Program Files\Common Files\VST3', locate the 'P455 MDN Sidecar.vst3' file and delete it.
- AAX: In 'C:\Program Files\Common Files\Avid\Audio\Plug-Ins', locate the 'P455 MDN Sidecar.aaxplugin' folder and delete it.
- Shared: In 'C:\Users\Public\Documents\Pulsar Modular', locate the 'P455 MDN Sidecar' 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: In '/Library/Audio/Plug-Ins/Components', locate the 'P455 MDN Sidecar.component' file and delete it.
- VST3: In '/Library/Audio/Plug-Ins/VST3', locate the 'P455 MDN Sidecar.vst3' file and delete it.
- AAX: In '/Library/Application Support/Avid/Audio/Plug-Ins', locate the 'P455 MDN Sidecar.aaxplugin' folder and delete it.
- Shared: In '/Users/Shared/Pulsar Modular', locate the 'P455 MDN Sidecar' 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).

# Signal Flow



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DSP Engine Design: Ziad Sidawi  
Plugin Development: Pulsar Modular Team  
GUI Development: Max Ponomaryov / azzimov GUI design – [www.behance.net/azzimov](http://www.behance.net/azzimov)  
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