POWAIR

User Manual

SoundRadix

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PLUG-IN OVERVIEW

POWAIR is an innovative, multi-stage dynamics processor that allows a user-customized blend of the gain-riding, peak-limiting, and tone-shaping aspects of compression.

We use compression for many different purposes. In some cases, we're solving problems, like erratic dynamics in a performance. Other times, it is an artistic tool, used to color an instrument and change its overall timbre. Compression can reshape transients, or alter the amount of ambience in a recording. As loudness standards for broadcast and streaming music services evolve, so do the roles of volume leveling and peak limiting.

We often seek a combination of these compression functions, and as a result, "stack" plug-ins on a single track or bus. Rather than having to bounce around between a number of plug-in windows, compensating for the interactions between these effects, using POWAIR brings all of these processor types into one clean, simple GUI. Without flipping through pages or tabs, multiple compression types can be addressed while being analyzed with a single set of powerful metering tools.

What makes POWAIR really special, however, goes beyond merely aggregating processors into a single toolkit. It's all about giving you flavor when you want it and unprecedented transparency when you don't. The core compression stage doesn't rely on a traditional envelope follower to calculate the signal and gain reduction. This allows it to adapt to the signal in a way that produces a purer sound, truer to its unprocessed state. POWAIR also utilizes proprietary controls which can alleviate common compression pitfalls and trade-offs.

Adaptive Compression

A couple of years ago at the AES show, multi Grammy award winner Frank Filipetti pulled us aside and said (more or less in these words), "Listen guys, I'm working on a vocal track and I get the compressor to work all smooth and nice on one section, and then when the loud part comes in, the compressor over-compresses and kills my sound. Now, I could automate the threshold, but what I really want is a compressor that will keep the sound that I had in the soft part, in the loud part as well. Can you do it?"

After a few drinks and sleepless nights we're happy to say "Yes we can!"

In an industry first, POWAIR features Adaptive Compression to maintain consistent compression effects across different signal levels, adding intensity and glue, while keeping the natural dynamics of the recording.

Punch

The unique Punch feature enables full control over transients' levels during the compressor's attack period. This makes shaping transients' lengths and smoothing transients' levels completely independent functions.

Auto-Leveler

There are several tools on the market that can ride a virtual fader to preset target level, and others that can normalize an offline signal to a standardized LKFS loudness level. Realtime, automatic level-riding based on an ITU-R BS.1770 target level, aided by a customizable speed and range, make POWAIR's auto-leveler a one-of-a-kind tool.

SYSTEM REQUIREMENTS

MAC:

- Intel Core CPU
- 2 GB RAM
- OS X 10.7 or higher

WINDOWS:

- Intel Core CPU
- 2 GB RAM
- Graphics card supporting OpenGL 2.1 or higher
- Windows 7 or higher

PLUG-IN FORMATS:

- AAX
- Audio Unit
- VST
- VST3

AUTHORIZATION:

To use POWAIR, you'll need a free iLok account and the iLok License Manager application. To create an iLok account and download the iLok License Manager, please visit https://www.ilok.com/.
An iLok USB Device is not required to use POWAIR.

1. Log-in to your User Area at https://www.soundradix.com/users/
2. Enter your license redeem code in the "New License Activation" box, then click "Redeem."
3. Enter your iLok Account User ID and email address, then click "Redeem."
4. POWAIR will now appear in your product downloads and the license will become available in your iLok account.

Once your POWAIR license has been added to your iLok account, it can be used to authorize the plug-in one of two ways:

iLok USB Device:
Can be purchased on-line or at retailers where professional audio equipment is sold.

Pros:

- If you work on multiple machines, it allows you to easily migrate licenses from one machine to another.
- For freelancers who travel to different studios, you can have an assistant engineer install the plug-in before you arrive, but you don't have to provide them with your log-in credentials to authorize the software. You just bring the key with you.
- When upgrading machines, there is no need to deactivate licenses.

Cons:

- Requires the "dongle" to be connected in order to use the software, which requires an available USB port.
- An iLok USB Device can be lost or stolen.

Host Drive Authorization:

Pros:
- Free (Doesn't require the purchase of an iLok USB Device).
- Opens up a USB slot, or in the case of many modern laptops which don't have USB slots, prevents the need for USB adapters or hubs.
- Perfect solution for those using a single computer.

Cons
- Migrating a license to an alternate machine requires returning the license to your account's "license cloud" through the iLok License Manager.
- When traveling to different studios, an additional step is required upon your arrival.

It is important to note that regardless of which method is used, a single license can only be used on one machine at a time. In order to allow you to use POWAIR on multiple computers simultaneously, we provide three licenses with every purchase. If you wish to use your Sound Radix plug-ins on more than three workstations, additional licenses must be purchased. Also worth noting, if a POWAIR license is activated on a Host Drive, it can later be transferred to an iLok USB Device associated with the same iLok account, at no charge.

INSTALLATION:

Download and install the POWAIR installer. Run the application and follow its on-screen instructions. Please note that you may need administrator permissions and password in order to install POWAIR. When installation is complete, quit the installer.

Prior to the first DAW launch after installation, your POWAIR license can be activated to your Host Drive or iLok USB Device through the iLok License Manager. If this has not been done, you will be prompted to activate the plug-in during your DAW's startup sequence. You will need to enter your iLok Account credentials, so please have them handy when using this authorization method.

GUI MAP

The plug-in's GUI breaks into a number of areas with singular functions which are described directly below. Controls with more complicated functionality are broken out onto subsequent pages, as indicated.

1. **Input Peak Level:** Displays the level of incoming sample peaks prior to processing.

2. **Output True-Peak Level:** Displays the post-processing output level in dBTP (True-Peak) as specified by ITU-R BS.1770-4. While traditional peak meters display the value of the loudest digital sample, they do not account for "inter-sample" peaks. When DA converters and oversampling processors reconstruct an audio waveform from a sequence of digital samples, new peaks are created between existing sample positions. The levels of these peaks, which can exceed the value of the loudest audio sample, are True-Peaks.

3. **Input LKFS:** Shows the K-weighted loudness level of the incoming, pre-processing signal. Measurement is based on the ITU-R BS.1770 "perceived loudness" algorithm. You can determine the speed of the reading by using the LU Integration Time Slider in the Input and LU Integration Time Drawer (See Page 14).

4. **Output LKFS:** Shows the short-term K-weighted loudness level of the outgoing, post-processing signal.

5. **Master Output Gain:** Provides a final gain adjustment after all other processing, The level displayed indicates the amount of boost or cut that is being applied to the signal. All output meters display the level following this control, indicating the product of its effect.

6. **Output Meter:** Meters the post-processing output level using the Sound Radix "Spectral Level Meter." This meter displays the intensity of signal across different frequency bands using co-centric, color-coded bar-graphs. The lowest frequencies are represented by a wide, red bar, and the highest frequencies by a narrow violet one. The frequencies in between follow the "RYGBV" color order.

7. **Wet/Dry Ratio:** Allows parallel processing by combining input signal with audio that has been processed by the **Compressor**. If the **Leveler** is active, its output will serve as the "Dry" signal, instead of the un-processed input signal

   - 100%: Only the fully compressed signal will be heard, post the **Leveler** and **Compressor**.
   - 0%: Only the raw input or post-Leveler signal will be heard. The **Compressor** is effectively inactive.

8. **Master Bypass:** Pressing the Sound Radix logo disables all plug-in functionality, including the **Master Output Gain** control. Applying this function "grays out" the plug-in interface.

9. **A-B Settings:** This function provides an easy way to change several parameters, but then compare the results to the previous settings. Before making a change, press "B" to create your alternate setting. After it is dialed in, press "A" again to hear the original. The ">" button will toggle back and forth between the two versions.

10. **Preset Browser:** Here you can load factory presets, and also save or load your own. Unlike presets saved by using your DAW's preset menu, presets saved here will appear in the preset list of every DAW, not the just one that created the preset. Use the arrow buttons to "flip through" the preset list.

11. **Undo/Redo:** "Undo" will reverse the last parameter change that you have made. "Redo" will reinstate that change.

A. **Auto-Leveler**: See Page 7

B. **Compressor**: See Page 9

C. **Stereo Link and Side-Chain Filters**: See Page 12

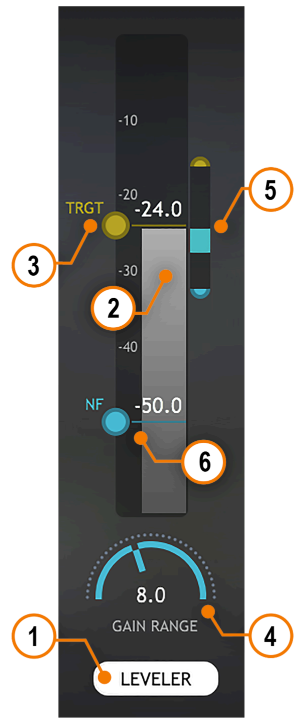
D. **Input and LU Integration Time Drawer**: See Page 14

E. **Settings Drop-Down Menu**: See Page 15

SoundRadix

A. AUTO-LEVELER

The POWAIR Auto-Leveler is designed to provide smooth and transparent level-riding to match a user-defined, targeted LKFS level. This makes it perfect for automatically conforming speech to EBU and ATSC specifications, but it is very useful for musical applications, as well:



1. **Leveler Enable:** Activates or bypasses the Leveler component of POWAIR.

2. **Input Level Meter:** Displays the level of incoming signal, prior to changes imparted by the Leveler.

3. **Target Volume:** Sets the target level in LKFS to which the Leveler will match incoming signals. The speed at which an incoming signal's LKFS will be read and adapted is set in the Input and LU Integration Drawer (See Page 14).

4. **Gain Range:** The Leveler behaves much like "riding a fader," pulling signal that is averaging below the Target Volume upward, and signal above that level downward. Gain Range determines how dramatic the "fader moves" are allowed to be.

   If the average incoming level deviates from the Target Volume by an amount that is smaller than the Gain Range, the signal will be pulled towards the target by an amount equal to the level of deviation, so that it will match the Target Volume precisely.

   If the average incoming level deviates from the Target Volume by an amount that is greater than the Gain Range, the signal will be pulled towards the target by an amount that is equal to the Gain Range, in order to minimize the extremity of the level-riding.

   This means that, when set to the maximum value of "20 dB," signals as great as 20 dBLU above the target or as low as 20 dBLU below the target can be normalized, creating a full 40 dBLU range that can be matched strictly to the target. By contrast, when the Gain Range is set to the minimum value of "0 dB," the Leveler is effectively disabled.

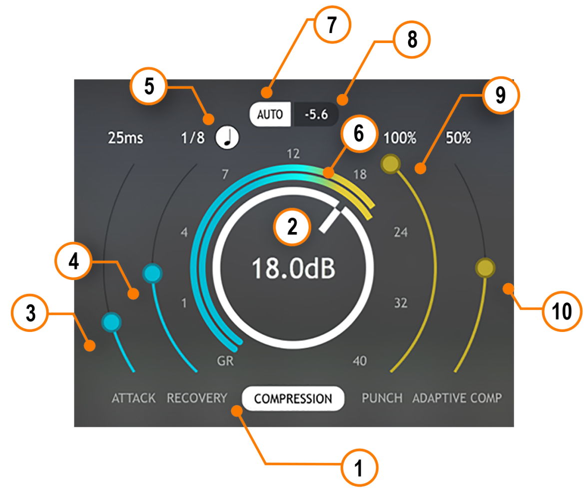
5. **Gain Meter:** Displays the activity of the Leveler. The appearance of the teal lower bar graph indicates that signal has exceeded the Target Volume, and displays the level-change applied to that signal in order to match the Target Volume. The appearance of the yellow, upper bar graph means the opposite. In this case, signal has fallen below the Target Volume, and the meter displays the level-change applied to bring that signal up to the Target Volume. Pulling the teal and yellow "handles" at either end adjusts the Gain Range.

   The overall post-Leveler LKFS level can be read in the Output LKFS indicator above the Output Meter (See Page 6), but note that this meter's reading will also be affected by the Compressor (See Page 9) and Master Output Gain (See Page 6), if those components are engaged.

6. **Noise Floor:** Because the **Gain Range** is equal above and below the **Target Volume**, in order to pull the loudest signals down the necessary amount, the lower portion of the gain range might extend into the noise floor. This would cause the **Leveler** to pull noise upward towards the **Target Volume**. Setting the **Noise Floor** value sets a cutoff, so that signal below the selected value will not be boosted.

B. COMPRESSOR

The Compressor follows the Auto-Leveler in terms of signal flow. It features a fixed-threshold, soft-knee design with the following controls:



1. **Compressor Enable:** Activates or de-activates the Compressor component of POWAIR.
2. **Compression Amount:** Adjusts the level of the signal driving the input of the Compressor as well as its detector "circuit." The dB level displayed reflects the amount of gain applied to the input signal. To increase gain reduction, increase this control and more compression will be applied.
3. **Attack:** Sets the amount of time that it will take for the Compressor to reach the maximum compression based on source signal and selected parameters.
4. **Recovery:** Sets the "release time," or the amount of time that it will take the signal to return to unity gain after gain reduction has been applied.
5. **Sync to Tempo:** Changes the values of the Recovery slider from their default milliseconds scale to divisions of musical meter. The tempo of these note values is automatically synchronized to the active tempo setting of the host DAW.
6. **Gain Reduction Meter(s):** Indicate the amount of gain reduction being applied to the input signal. When used on a mono track, a single meter is shown, but when processing stereo tracks, the meter is split into two channels. If the Stereo Link control (See Page 12) is set to L/R, the top meter shows gain reduction for the left signal, the bottom displays the right. In M/S mode, the top meter becomes the "Mid" meter, and the bottom becomes the "Side."
7. **Auto Makeup Gain:** Using machine learning, the plug-in continuously analyzes the perceived volume of the post-Compressor signal, and compares it to the apparent loudness of the input signal. Auto Makeup Gain will adaptively attenuate the processed signal's level to psychoacoustically match that of the unprocessed signal.

   Because we all suffer from the "louder sounds better" syndrome, it is easy to believe that our compression settings are making a positive change as long as they are making things louder. In many cases, once we've manually adjusted the makeup gain, while A-B'ing with the unprocessed signal, we find that the compression is doing significantly less than we perceived, or even producing negative effects.

   When adjusting the Attack, Recovery, Punch or Adaptive Compression, the level will be compensated, allowing the effects to be evaluated objectively, without loudness increases or decreases biasing your perception.

8. **Manual Makeup Gain:** If the Auto Makeup Gain button is deselected, a Manual Makeup Gain value can be entered by clicking once and typing a value. Clicking and dragging upward or downward will also smoothly scroll through, and change values. This is particularly useful when using the Wet/Dry control (See Page 6) for parallel compression with extreme compression settings.

9. **Punch:** Punch is a proprietary design that allows unique transient-shaping abilities. Because of POWAIR's extremely fast detector "circuit," coupled with a brief look-ahead time, the plug-in can analyze transients and compress them with maximum gain reduction nearly instantly. This allows non-transient information, like room ambience, or the body resonance of an acoustic instrument to be pronounced, while the transients are backed off.

   - **0% Punch:**
     In this state, the Attack time value becomes essentially irrelevant, as all transients are compressed with maximum gain reduction, relative to the incoming signal level and the Compression Amount control, instantly. This effect can be sculpted using the Recovery and Compression Amount controls.

   Increasing the Punch value allows transients to be passed, while still providing control over their level. Best of all, this attack level remains independent of the time-based Attack control. On traditional compressors, setting a fast attack time in order to control transients can be effective, but it has the added byproduct of altering the shape of the transient in order to produce the desired result.

   - **100% Punch:**
     When the Punch control is maxed out, the Compressor behaves like a traditional compressor, with its Attack time and attack level effectively tied together. Decreasing Attack time will cause the more transients to be compressed, but they will also be more susceptible to coloration due to the Compressor's effects.

     Note: Setting the Attack time to "0 ms" with the Punch controls set to any value, is essentially the same as setting the Punch control to 0%.

   Lowering the Punch control to a value between 0% and 100% allows you to attenuate transients, regardless of the Attack time setting. Even transients which are faster than the set Attack time can be reduced in volume, but because they are not subject to the Attack control, their original shape can be maintained, even as they play at a lower volume. The overall shape can still be customized using the Attack and Recovery controls, and the overall amount that they are reduced is still tied to the Compression Amount control.

   - 50% Punch:
     If the Punch control is set to 50%, transients will play at roughly half the volume that they would if the Punch value were set to 100%.

10. **Adaptive Compression:** Adaptive compression is another POWAIR original. The ideal application would be a very dynamic performance, like a vocal or acoustic instrument in a ballad. Using compression to thicken up the instrument would be tricky, because for one, compression would also alter the dynamics of the performance. On top of that, the character of the compression would be inconsistent with quieter sections being under-compressed and louder sections being over-compressed.

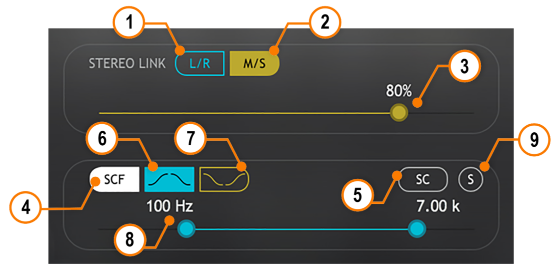
Adaptive compression solves this problem by maintaining a consistent compressed sound *and* preserving the natural dynamics of the performance. It's like flavoring the piano or singer's voice with compression, before they even perform. This is accomplished by establishing the desired compression amount and then adjusting the **Adaptive Compression** slider to determine how dynamic the product of the **Compressor** is allowed to be.

- 0% Adaptive Compression:
  At this setting, the **Compressor** and the dynamics of the performance will interact as they do in a normal compressor. Louder sounds will be compressed more, and quieter ones will be compressed less. The **Compressor** will also have the typical effect of taming dynamics.

- 100% Adaptive Compression:
  The natural dynamics of the performance will be fully preserved, and compression will remain at a consistent level. The target compression amount will be based off of the **Target Volume** setting in the **Leveler** section of POWAIR (See Page 7). Keep in mind that the **Leveler** will alter the dynamics of a performance before it even hits the **Compressor**, so if you truly want to maintain the original dynamics, disable the **Leveler** or minimize the **Gain Range** control.

C-1. STEREO LINK

The Stereo Link section allows the left and right channels of stereo tracks to either be processed separately or equally. The built-in mid/side decoder allows the identical, centered information from the stereo signal to be processed separately from the unique stereophonic information in the two channels. The stereo link controls have no effect on mono tracks.



1. **L/R:** Sets Stereo Linking to left/right stereo processing mode.
2. **M/S:** Switches Stereo Linking to mid/side (sum/difference) processing mode.
3. **Stereo Linking:** Adjusts amount of correlation that the Compressor's processing will have across the two channels, either left and right, or mid and sides, based on the above selection.
    - At a setting of 100%, both channels will be equally compressed by the maximum gain reduction.
    - At 0%, the two channels will be compressed completely independently.

C-2. SIDE-CHAIN FILTER

4. **Side-Chain Filter:** By default, the post-Leveler input signal feeds the Compressor's detector. When operating in this manner, the SCF button will engage filters which can process that signal prior to feeding it to the detector in order to skew the focus of the compression action.

    For example, because low frequencies have more energy at the same apparent loudness as high frequencies, low frequencies which are being masked by higher-order harmonics can cause the Compressor to become overactive. On bass, try filtering out the frequencies below 100 Hz, for a smoother overall response. With vocals try zeroing in on the 100-400 Hz vowel range, to smooth out the notes without excessively pumping breaths and sibilance.

5. **Side Chain:** Actives the external side-chain and feeds its signal to the Compressor's detector "circuit." Each DAW routes plug-in side-chains differently, so if you are unfamiliar with your particular DAW's implementation of this type of functionality, please see your software's owner manual.

    When using the Side Chain to feed the Compressor's detector, the SCF button applies its filters to the incoming, side-chained signal.

6. **Band-Pass Filtering:** Creates a band pass filter across the signal feeding the Compressor's detector, by shelving out highs and lows, above and below the selected frequencies. This mode would be used to execute the above examples.

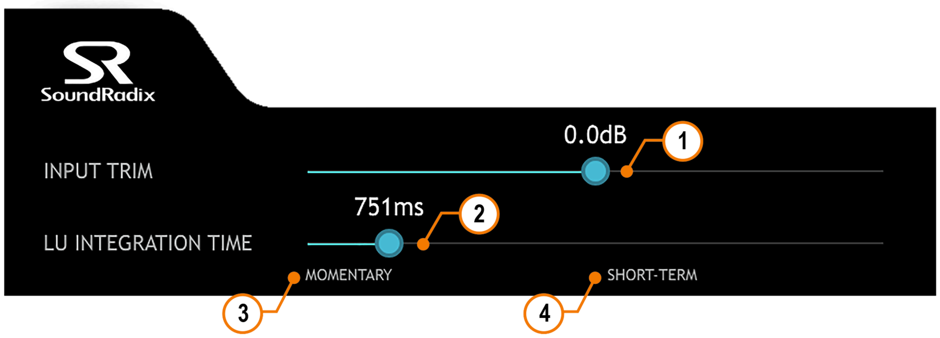
7. **Band-Reject Filtering:** Excludes the range between the two selected frequencies, and instead focuses the **Compressor**'s detector on the frequencies outside (above and below) that range.

8. **Frequency Selectors:** Slide each dot to the left or right to set the **Side-Chain Filter** range. Holding the "Shift" key while moving one of the sliders will automatically move the second slider as well, while maintaining the bandwidth between them. Clicking on a numeric value allows manual entry.

9. **Side-Chain Filter Solo:** When **SCF** is enabled, the **Solo** button will isolate the filtered signal which is being fed to the Compressor's detector, letting you hear what the filters are doing. If **SCF** is disabled, the **Solo** button allows the filters to be used as a post-compression equalizer, without affecting the **Compressor**'s operation.

D. INPUT AND LU INTEGRATION TIME DRAWER

Press the white arrow at the bottom of the plug-in GUI in order to open or close this additional settings drawer:



1. **Input Trim:** Adjusts the gain of signal arriving at the input of the plug-in. This adjusts the level prior to the **Leveler**.

2. **LU Integration Time:** Sets the speed of the the **Leveler**'s automatic gain-riding action.

   The ITU-R BS.1770 specification is designed to measure the loudness for the entire duration of a program, however, in order to achieve the release country's broadcast loudness standard, it is important to track the level in the short term while making level adjustments. The ITU recommendation doesn't specify any standardized short-term measurement durations, only a means for measuring in the short term. The **LU Integration Time** slider follows this formula, but allows you to freely slide to any desired speed.

   In order to produce consistency between meters and normalization tools from different manufacturers, the EBU drafted the TECH 3341 document, which serves as a supplement to EBU R 128 broadcast loudness standard, which is based off of ITU-R BS.1770. This document presents two measurement and response speeds which can be easily selected in POWAIR by clicking on the labels below the **LU Integration Time** slider.

3. **Momentary:** Sets **LU Integration Time** to the standardized 400 ms Response Time, known as "M".

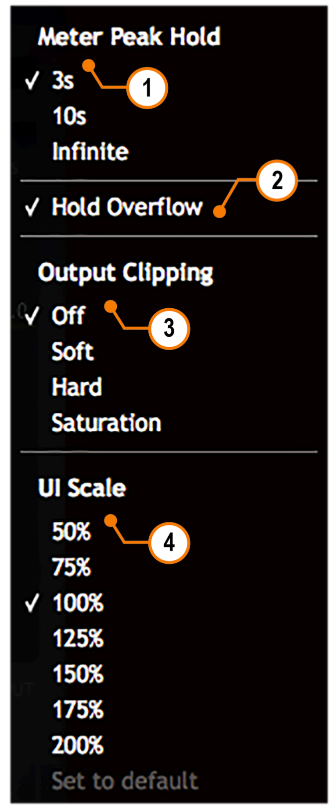
4. **Short-Term:** Sets **LU Integration Time** to the standardized 3 Second Response Time, known as "S".

E. SETTINGS DROP-DOWN MENU

The Version 1.1 Update brings a new drop-down menu with additional features:

1. **Meter Peak Hold:** Sets the duration that the white peak-level bar in the Spectral Level Meter (See Page 6) holds its maximum value.

2. **Hold Overflow:** When output signals exceed 0 dBTP, the Output True-Peak Meter (See Page 5) display box turns red. If Hold Overflow is enabled, this "overflow" indicator will remain lit until manually clicked. If disabled, it will refresh based on the Meter Peak Hold time.



3. **Output Clipping:** Clips off samples which exceed 0 dBFS and determines how the remaining waveforms will be shaped. This clipping effect takes place after the Compressor's Makeup Gain (See Page 9), and before the Master Output Gain (See Page 6). Though sample peaks will be clipped by engaging this function, True-Peaks can still exceed 0 dBTP, so keep an eye on the Output True-Peak Level indicator, and adjust the Master Output Gain to safely avoid True-Peak overloads.
    - **Off:** Disables clipping and allows signal to exceed 0 dBFS when passing through the plug-in's output.
    - **Soft:** Enables clipping, and slightly rounds the edges of squared-off waveforms to produce a transparent-sounding effect.
    - **Hard:** Clips off peaks with hard edges, producing an aggressive character.
    - **Saturation:** Adds colored, harmonic distortion to clipped output signals. This mode is particularly prone to producing True-Peaks which exceed 0 dBTP, so be sure to adjust the Master Output Gain accordingly.

4. **UI Scale:** Allows the plug-in interface to grow, in order to be seen across the room, or shrink to eat up less screen space. The "Set to default" command will cause all instances of the plug-in to open at the selected scale.

TIPS AND PRESETS

In order to help you understand the controls of POWAIR, we've put together a handful of presets that take advantage of the plug-in's unique features, so that you can see them in action. We've even given you some tips on how to tweak these presets, so that you can hear their effects. Try these out on their intended applications, and you should feel much more comfortable with all of the exciting features that POWAIR has to offer.

THICKER CHUGGING GUITAR

In this case, we've disengaged the **Leveler** to the allow the natural expressiveness of the performance come through. The slow **Attack** setting with light **Punch** accentuates the "chunk" while the compressor works on taming the body resonance that follows. The overall product should be thicker, but with a punchy character that will cut through the mix.

Because we're pushing a lot of gain into the **Compressor**, and allowing the transients to pass through uninhibited, if **Auto Makeup Gain** were disabled, there would be a considerable gain increase at the **Compressor**'s output. **Auto Makeup Gain** is maintaining an equal perceived loudness before and after compression. Unlike the **Leveler**, however, which continually tracks and corrects level changes, **Auto Makeup Gain** is a static adjustment which only updates in response to changes in control values.

Because of this, with the **Leveler** disabled, the dynamics of the guitarist's performance are maintained. With that in mind, this preset utilizes POWAIR's **Adaptive Compression**, so that even when the guitarist backs off on certain notes and chords, the same thickening effect is applied to the signal. For this to work correctly, find the loudest part of their performance and adjust the **Target Volume** of the **Leveler** so that it sits right where the **Input Level Meter** is averaging. Though the **Leveler** is not active, remember that this control set the target gain reduction for **Adaptive Compression** as well.

To achieve a more even, less dynamic sound, switch it up by enabling the Leveler, and sliding back the **Adaptive Compression** control to 40%. The **SCF** is dialed in, but bypassed. For an even thicker sound, un-bypass the **SCF** and now the the compression will focus its effects on the pick attack and "thunking" sounds.

VO COMPLIANCE

The VO Compliance presets are designed to level out VO's in order to maintain compliance with EBU, ATSC, or different streaming service's target loudness standards. These are designed for use with POWAIR as the last plug-in on the source track, prior to feeding the master. This way, the Leveler can account for gain changes produced by any other EQ, de-essing, or exciting which precede its input on the track. Don't forget that if you have additional processing on your master bus, this can affect the overall loudness, so an additional LKFS meter should be added to the end of the master bus's plug-in chain.

In all of these presets, the Leveler is set to a 10 dB Gain Range with a 1 second LU Integration Time which should smooth things out without obvious pumping sounds. We recommend that you adjust the Input Trim so that the Input Level Meter is hovering around the Target Volume line for the best results.

In all of the "VO" presets, the Compressor is bypassed, as the purpose of the preset is "fader-riding." However, an appropriate compression setting for voiceover is dialed in, so if you wish to use it, just enable the Compressor. Notice how the SCF is focusing the Compressor's effects on a range that will help the VO sit on top of a music bed. Also note that Auto Makeup Gain has been disabled, as that function uses its own algorithm to match perceived loudness, and cannot guarantee adherence to the desired LKFS standard. Each VO preset has a Manual Gain value entered which we found to maintain an average level at the Target Volume on our test materials. You should adjust this value as necessary if you alter the Compressor's settings or find the Output LKFS to be straying from the target.

BALLAD VOCAL

With its fast Attack and slower Recovery, this preset allows the Compressor to bolster the vocal as it starts to trail off. Meanwhile, the Punch value of 60% allows the singer's consonants to cut through, despite the fast Attack time. If you want less consonants, back off the Punch control. The Leveler is disabled and Adaptive Compression is set to 50%. This allows the same thickening effect that you hear during louder passages to be applied at quieter parts. Likewise, the Adaptive Compression will prevent the Compressor from over-compressing to a degree, but because it's only at 50%, the Compressor is still able to tame really loud moments in the performance.

If it feels like the Compressor is overactive in louder sections, first, double-check your Target Volume control. In this case, find a section of average volume, not the loudest or quietest part, and make sure that the Input Level Meter is hovering around the Target Volume. If not, adjust the Target Volume control. At this point, if the Compressor is still crushing the vocal at the loudest parts, try increasing the Adaptive Compression control.

Also note that the SCF is enabled and soloed, so that not only will the Compressor be focused on a range of 80 Hz and up, but the high-pass filter will be audible. If the singer has a higher voice, you might be able to increase that value.

CHICAGO DRUMS

This preset is designed to be placed across the drum bus, giving you a crushed, roomy sound by utilizing very fast **Attack** and **Recovery** times. If you have a favorite tape emulator plug-in, place that on the insert position right before POWAIR with a subtle setting, and you'll love the synergy of the two plug-ins working together. We recommend that you set the **Input Trim** to around -20 dB LKFS.

There is a lot of room to play with this one:

The preset **Wet/Dry** ratio is 50%, giving the effect a more mainstream sound, but crank it all the way to 100% for maximum effects. For a less punchy, more industrial sound, try bypassing the **SCF** and embrace the chaos. If things are getting too gritty, try pushing up the **Recovery** control, but keep it under 350 ms or you'll lose the intended sound. Conversely, for a little extra mayhem, slide back the release and see what happens. If you want to hear more room, slide back the **Stereo Link** control towards 20% or lower, and you'll hear some nice wide reverberations.

If you're concerned about keeping your kick and snare punching through the mix, or maintaining small speaker (cellphone) compatibility, ease the **Punch** control upward. Do it slowly, though, because a little goes a long way!

DIRTY BEATS

This preset is having some fun with the **Auto-Leveler**, by using a fast tracking speed and a wide **Gain Range**. It is designed for use with hip hop and EDM drum loops, and by activating or bypassing the different stages of POWAIR, different sounds can be achieved for different parts of a song. With the **Compressor** and **Leveler** both bypassed, you'll still hear the sound of the **Output Clipping** in **Saturation** mode, so you can build from fully bypassed (Press the **Sound Radix** icon and the interface will "grey out"). Un-bypassing will add the **Output Clipping**, then you can build to the **Compressor**, whose **SCF** should be tuned to focus on the snare drum in the beat.

Slamming the **Leveler** across everything will be too abrupt, so dial the **Gain Range** back to 0 dB, enable the **Leveler** and ramp the **Gain Range** back up to 20 dB for the full effect. For the wildest sound, bypass the **Compressor** so that it can no longer keep the **Leveler** under control. The combination of the unbridled **Leveler** and the **Saturation**-mode **Output Clipping** is pretty sweet. Keep an eye on the **Output True-Peak Level** meter and adjust the **Master Output Gain** if necessary.

ACKNOWLEDGEMENTS

POWAIR uses the following libraries:

- JUCE by ROLI Ltd. - https://juce.com
- KISS FFT by Mark Borgerding - https://sourceforge.net/projects/kissfft
- Protocol Buffers by Google - https://developers.google.com/protocol-buffers/
- VST Plug-In Technology by Steinberg Media Technologies GmbH - https://www.steinberg.net/

To Frank, your next cigar is on us.

Sound Radix

www.soundradix.com

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