



AMPLIFIED INSTRUMENT PROCESSOR

user manual AIP 1.1

KORNEFF
the other side of sound



Amplified Instrument Processor

The Amplified Instrument Processor is a complete suite of tools designed to help you sculpt the ultimate guitar, bass or keyboard tone. But while it was designed initially to work with amplified instruments, its power and versatility make it an excellent processor for virtually any audio task. The AIP can work wonders on vocals, acoustic instruments, pianos, group buses and even across the whole mix. Heck, you can even master a finished recording with it. It's that good, and it's that versatile.

The AIP combines a seriously powerful four band EQ, a VCA based compressor, specialized filter networks, a psycho-acoustic stereo widener, a frequency dependent compressor, and a proprietary signal processing unit, a real time analyzer.. oh god... it's basically a killer audio channel that has everything except delay, reverb and an assistant to get you coffee.

Using his years of experience shaping rock records behind the console, Dan Korneff developed a processor that contains the entire mixing chain for his signature guitar and bass sounds. Not only does it speed up his work flow, but the result gives him the extra flexibility and features of the digital world without compromising analog vibe.

AIP 1.1 adds a user requested feature: the FDR now has an expanded range and any frequency from 20Hz to 20kHz can be selected. And there's now an input meter on the FDR as well. Further, 1.1 fixes a number of bugs. Finally, we've added System/Machine authorization to the iLOK security that protects our plugins. You can now license your plugin via iLOK dongle, and iLOK cloud account, and via System/Machine authorization.

Behold, the AIP!

Minimum System Requirements

Quad Core i5 Processor
4GB of RAM
Mac OS X 10.14 or Later
Windows 7 & Above

Supported Plugin Formats

Intel Mac 64bit: VST3, AU, AAX
Windows 64bit: VST3, AAX



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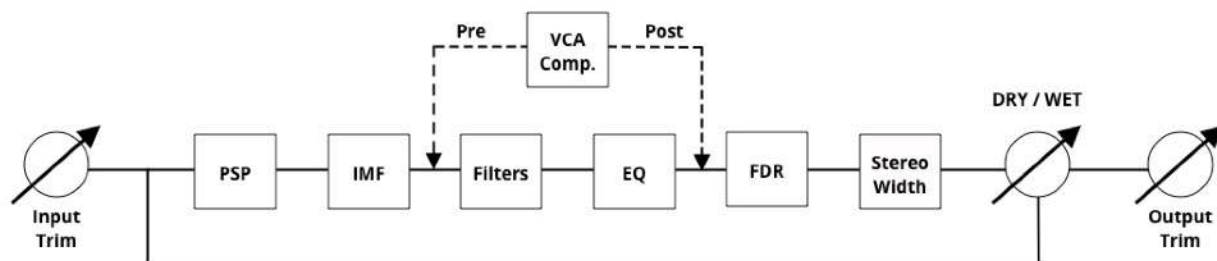
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OVERVIEW

There is a lot going on with the Amplified Instrument Processor (we are going to call it the AIP from here on out), and it has a lot of controls and functionality.

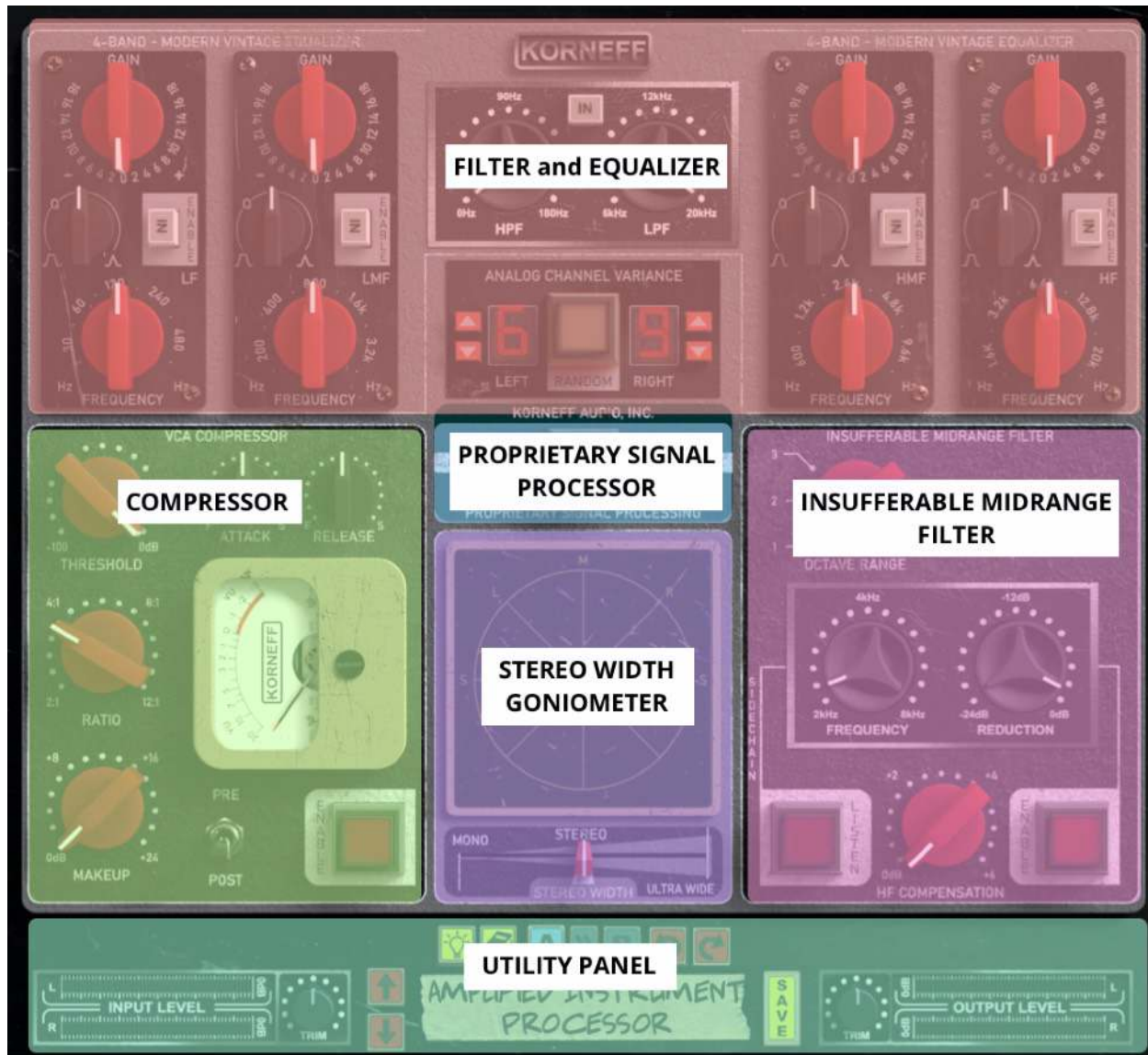
Many of the AIP processing sections have controls on both the front and back panels of the plugin. Generally, the front panel controls can do 90% of what you need to do, and the back panel provides additional tweakability to either get exactly the sound you want or add a bit of vintage analog flavor and unpredictability to things.

SIMPLIFIED BLOCK DIAGRAM



The signal flow of the AIP is similar to that of an audio console channel strip. The biggest difference is that all of the AIP's processing can be used in parallel via the **DRY / WET** control. Please note that the position of the compressor is switchable—it can be before or after the equalization section.

FRONT PANEL

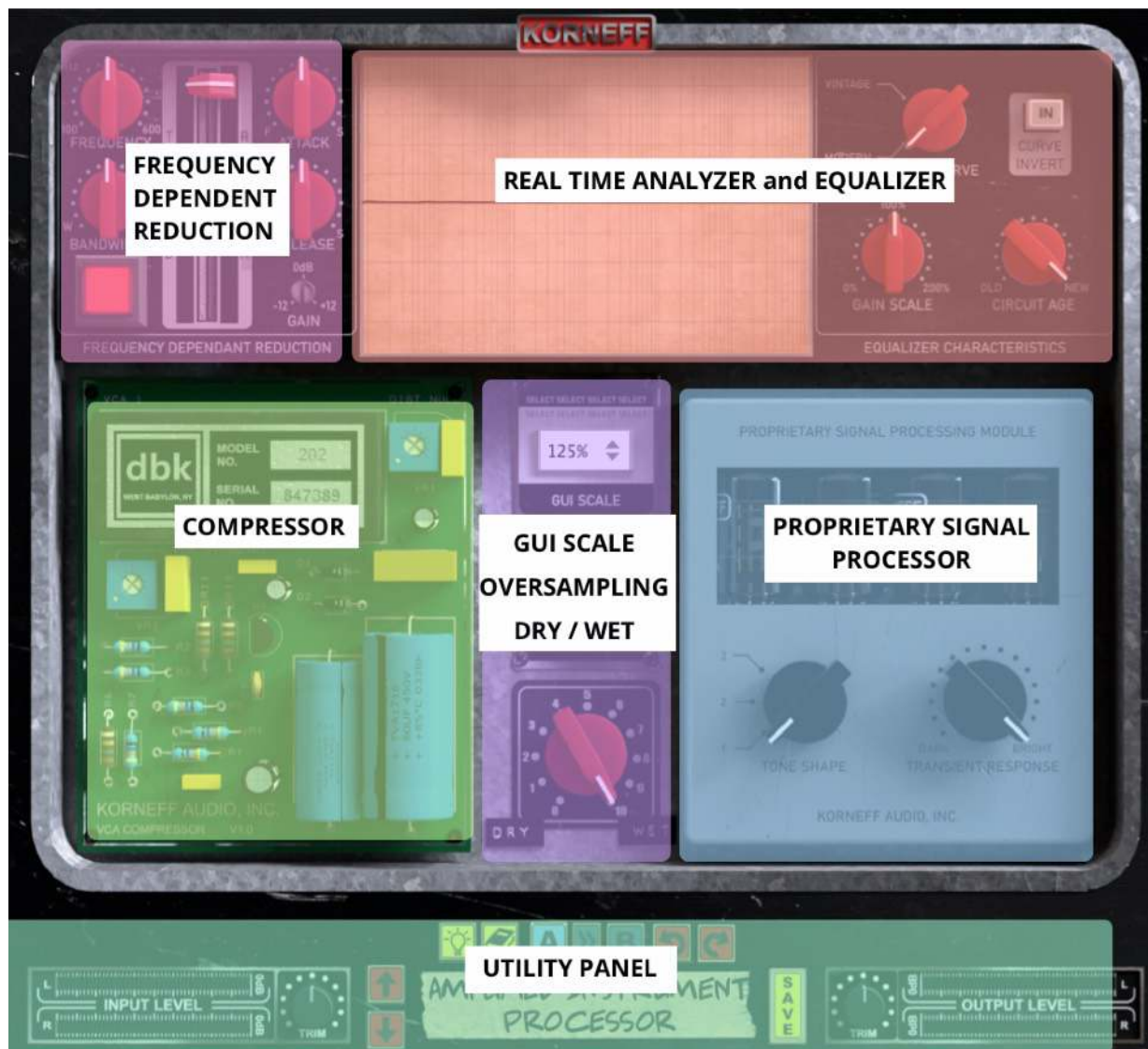


The front panel is divided into an eq section, a compressor section, the Insufferable Midrange Filter (**IMF**) section, the Korneff Proprietary Signal Processing (**PSP**) section, and a stereo widening section that includes a goniometer. There are **ENABLE** buttons all over the place to insert the various sections in and out of the signal flow.

At the bottom of the AIP is our **UTILITY PANEL**, which allows you to trim the input and output levels, access the preset library, AB compare settings, and bring tool tips and this manual if you need information.

Click on the **KORNEFF NAMEPLATE** at the top to go to the back panel of the plugin.

BACK PANEL



The "other side" of the plugin is dominated by a **REAL TIME ANALYZER** at the top. This RTA lets you see the frequency response of whatever audio is going through the plugin, and it also gives you visual feedback on how your eq settings are affecting the overall response.

Below the RTA are additional tweaks for the compressor section and the Proprietary Signal Processing. There's also the **FREQUENCY DEPENDENT REDUCTION (FDR)** unit, which is a compressor that is tunable to work on specific frequencies. You can also control the size of the **GUI** and switch in **OVERSAMPLING** to reduce aliasing and digital artifacts. There's also a **DRY / WET** control that lets you use the AIP as a parallel processor.

Click on the **KORNEFF NAMEPLATE** at the top to go to the front panel of the plugin.

PROCESSING SECTIONS

EQUALIZATION SECTION

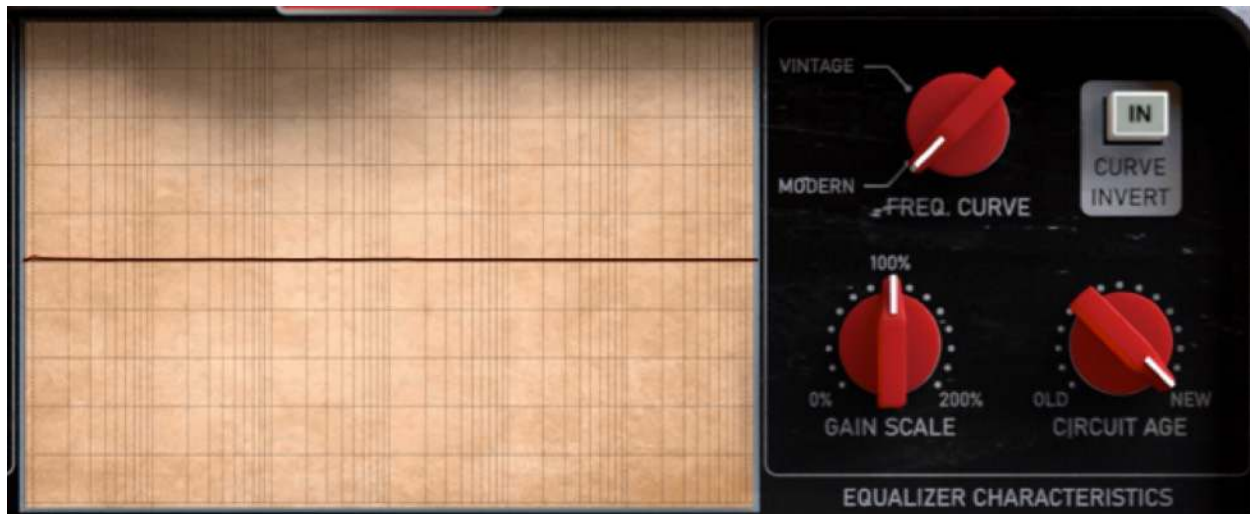


FILTER and EQUALIZER front panel

HPF/LPF : These are first order high pass and low pass filters with gentle 6db/octave curves. Use them to get rid of low end rumble and high end chuff. The **IN** button takes them in and out of the signal path.

Four Band Fully Parametric Equalizer: This is a ridiculously powerful and wonderful sounding eq. Based on the response curves of old German Klangfilm eq's, the AIP adds full continuous control of frequency, gain and Q (bandwidth). It covers from 20hz to 20kHz in four separate bands that have reasonable overlap and are independently switchable.

Back in the good old Klangfilm days, equipment was handmade and the tolerances on components was pretty loose. Hence, there was variation in the frequency response of individual channels in equipment. In modern gear, the right channel and the left are very likely identical in sound. In the old days, not so much. The AIP's **ANALOG CHANNEL VARIANCE** has 16 different response curves that you can set between the two channels to recreate a vintage difference in channel tone. If you want the channels to sound the same, set them to the same number. Press the **RANDOM** button to generate almost infinite channel mismatch. The effect is subtle, but it is definitely there. We'll have some hints on how to use this effect later in the manual. And no, we won't tell you what the 16 different response curves are. And there are also some more EQ controls on the Other Side...



REAL TIME ANALYZER and EQUALIZER back panel

Oh! There's that nifty **REAL TIME ANALYZER** that shows the frequency response of your recording and the curves of your eq's. Please note that this wasn't a feature on Klangfilm eq's made in 1957, but we thought you would like it.

You can further futz with the eq by using the **FREQ CURVE** control to switch the eq's response curves from **MODERN** to **VINTAGE**. The VINTAGE setting simulates a passive eq curve, which adds a slight dip at each of the bandwidth cutoff frequencies. You can see all this in action on the RTA.

GAIN SCALE is a multiplier for the front panel eq gains. At 100% , each gain knob has the has +/- 18dB of gain. Set it to 200% if you need something insane, like 36dB of boost! Set it lower than 100% and the gain control for each frequency will become more precise to use but with less overall gain, similar to how a mastering eq might feel. Dan loves the **CIRCUIT AGE** control, which simulates capacitors and other components in the eq circuit wearing out. Turn it towards **OLD** and you'll introduce ripples in the response of the eq that bring it to life in an old analog way—you can see this on the RTA. Click **INVERT** to reverse the curve of the eq.

VCA COMPRESSION SECTION



VCA COMPRESSOR front panel



VCA COMPRESSOR back panel

VCA Compressor: The VCA compressor is built around an emulation of the venerable DBX 202 VCA integrated circuit. Sound wise, this compressor is similar to a cross between a DBX 160 and an SSL channel compressor, but smoother and perhaps cleaner. If you want huge amounts of punch and color, go patch in our Pawn Shop Comp or Talkback Limiter. But if you want to beautifully smooth things out while still controlling articulation and details, the AIP's VCA compressor is your new best buddy.

It has a full range of controls.

THRESHOLD: 0dB to -100dB

RATIO: 2:1 to 12:1

ATTACK: 3ms to 200ms

RELEASE: 10ms to 4 seconds

MAKEUP: 0dB to +24dB gain

Use **THRESHOLD** to set the level at which the compressor kicks in. Use **RATIO** to adjust how much the signal is compressed, and use **ATTACK** and **RELEASE** to adjust articulation and detail. **ENABLE** puts the compressor into the signal chain, and **PRE POST** places it before the eq section or after it. And there are more compressor controls on the back!

There are two additional controls for the compressor on the back panel. **DISTORTION NULL** lets you adjust the amount of grit and saturation you get when you hit the compressor hard. Set it high to get more of a gnarly, vintage 70's sound. The **VCA SYMMETRY** introduces inconsistencies between the compression curves of the right and left channels of the compressor. This is analogous to the the **ANALOG CHANNEL VARIANCE** function of the equalizer.

PSP - PROPRIETARY SIGNAL PROCESSING SECTION



**THIS IS THE VERY IMPORTANT
ENABLE SWITCH FOR THE
PROPRIETARY SIGNAL PROCESSOR!**

It will find you in your dreams...

**PROPRIETARY SIGNAL PROCESSOR
front panel**



**PROPRIETARY SIGNAL PROCESSOR
back panel**

PSP: This section adds a sweet secret sauce to the input section of the plugin. It packs a lot of functionality into its very minimal controls—you'll be amazed at what the PSP can do for your tracks.

Pop the PSP in and out by pressing the big red button on the front panel. Hopefully you'll be able to find it...

On the back, the **TONE SHAPE** lets you select between three different shaping options that simulate the effects of various tubes, transformers, transistors and whatnot. All three sound different and all are very useful.



- 1** - Tube-ish. Warms things up and rounds out the top a bit. Lovely on bass!
- 2** - Tape-ish. This setting gives you the sound of pushing the signal a bit into analog tape. Extended bottom end with a high end sparkle. A bit like a loudness button if you're into Fletcher-Munson.
- 3** - Solid State-ish: Like a vintage 70's channel strip off an API, Electrodyne or Quad Eight console, this setting has a forward push to the midrange.

Transient Response is an emulation of Slew Rate. What is that, you ask? Basically, it is how quickly power is available to reproduce a fast incoming transient. A high slew rate means that there is a lot of power immediately available and the transient is reproduced cleanly and linearly; a low slew rate means that power isn't available quickly and the transient is somewhat rounded off. Another way to think of this is as compression that effects the high end of signal—from about 1kHz up on the AIP. When engineers refer to a piece of equipment as being "fast" or "slow" they are probably referring to slew rate. Set it to **BRIGHT** for fast slew and a nice, sharp attack. Set it to **DARK** to round off the transients a bit and get the sound of a channel with a slower slew rate, like an old tube console or a Neve 1073 preamp. Dan tends to use BRIGHT for his main guitars and DARK for clean and less dirty sounds. Luke thinks of it as way of pushing things forward or back in the mix - BRIGHT brings things forward, DARK pushes them back.



IMF - INSUFFERABLE MIDRANGE FILTER SECTION



INSUFFERABLE MIDRANGE FILTER
front panel

The IMF doesn't have back panel controls.
But we're sure Dan will figure out something.
You can never have too many things to adjust!

INSUFFERABLE MIDRANGE FILTER
back panel

Electric guitars have a lot of energy in the presence region—from 2kHz to 8kHz. It tends to impart a "whistling" sort of sound to a recording. One guitar on a song, maybe not so bad. But after a few overdubs... AY CARAMBA! So much 4k it hurts! The **IMF** is a notch filter specifically designed to clean up the whistle on electric guitar tracks, but it works well to tame cymbals, keyboards and vocals as well. Click in the **LISTEN** button to hear exactly where you're tuning the filter. Set the center frequency of the notch filter by adjusting **FREQUENCY** to the spot where the whistle is most annoying, and then turn **REDUCTION** counterclockwise to nip it out. Use the **OCTAVE RANGE** control to also remove the whistle frequency's upper harmonic by setting it to 2. Remove both the upper harmonic and the subharmonic by setting it to 3. You'll have to play with this control to get the result you want.

The **HF COMPENSATION** is a gentle shelf boost starting at the center frequency of the notch to restore some life to the high end. **ENABLE** inserts the IMF into the signal path. Be sure to pop out the LISTEN button when you're done unless you want things to sound like a broken iPhone. The IMF only has controls on the front panel, thank the Lord above.

FDR - FREQUENCY DEPENDENT REDUCTION SECTION



FREQUENCY DEPENDENT REDUCTION back panel only

On the back side only, you have access to a nifty compressor that can be tuned to reduce the gain of a small range of frequencies. Think of it as a cut only parametric eq with a threshold control on it. It works great to keep chugging guitar parts from taking over your mix, or taming resonant notes on a bass, etc. The **FREQUENCY** control selects the center frequency, from 20Hz up to 20kHz. **BANDWIDTH** lets you select how wide an area is affected. Pull the **THRESHOLD** slider down to kick the compressor in—a red LED bar will give you a visual indicator of how much gain reduction is happening. To the left of a gain reduction, a yellow LED bar will show input level. **ATTACK** and **RELEASE** give you control over the attack and release times of the compressor, and you can adjust the output using the **GAIN** trim. Beneath the **BANDWIDTH** knob is the **ENGAGE** button. Use the RTA to adjust your settings— you'll see a red negative peak curve that moves in accordance with the amount of gain reduction happening.

STEREO WIDTH and GONIOMETER



STEREO WIDTH and GONIOMETER
front panel



PICTURE OF RAMPANT HUGE GORILLA

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At the end of the signal chain is a psycho-acoustic processor that works like a charm to increase the apparent width of your signal. It has been fine tuned to work especially well with guitars, but it works great for anything. The **STEREO WIDTH** slider controls the amount of effect. Push to to the right to increase the width to **ULTRA WIDE**. Pull the slider to the left to make the output **MONO**.

The **GONIOMETER** give you a visual indication of the width of your signal. If the meter shows a line running up and down the center, surprise! Things are mono! The wider and all over the place the dots are, the wider and all over the place the signal is.

GUI SCALE, OVERSAMPLING and DRY / WET



GUI SCALE



OVERSAMPLING



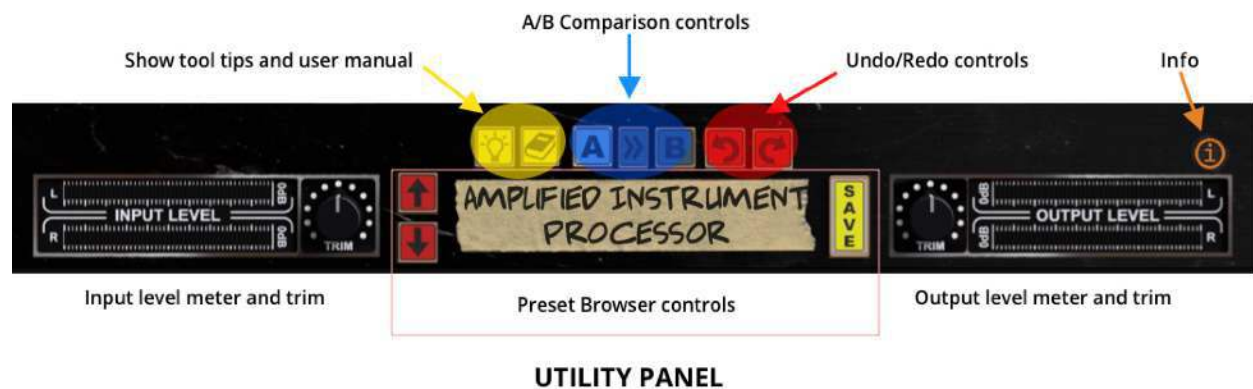
DRY / WET

The **GUI SCALE** allows you to change the size of the AIP's interface on your screen. This can make it easier to see things and impressive to people that are easily impressed.

OVERSAMPLING increases the sample rate of the plugin 4 times over the current sample rate of your project. Oversampling minimizes deleterious digital artifacts that can lessen the sound quality of signals. However, engaging OVERSAMPLING will increase the load on your computer's processors, which can cause stuttering and other issues with your audio. If you are working at sample rates above 48kHz, you probably don't need to use oversampling because artifacts will most likely already be minimized. At rates below 48kHz, clicking in oversampling can yield an improvement in sound quality.

DRY / WET: The default position of this control is 100% wet, meaning all the audio coming out of the AIP has been processed by the AIP. As you lower this control towards DRY, counterclockwise, signal from the input of the plugin—before any processing—will be mixed in with the processed output. This allows you to parallel process signals and fine tune how much you "hear" of the AIP on your audio. Generally, we use the AIP with this set to 100% wet, but there are times, such as when you're getting down to the final stages of a mix, that you might decide that you want to back off the processing of the AIP. The DRY / WET control allows you to do this without changing any of the setting of the eq, compressor, etc. It can be very handy, so keep this control in mind. It can save your ass on occasion.

UTILITY PANEL



TOOL TIPS/USER MANUAL - The **LIGHTBULB** brings up the tool tips, and the **BOOK** opens the user manual, which you are currently reading.

A/B COMPARISON - The comparison tools are very cool. **A** and **B** are two independent snapshots of the AIP's settings. Click **A** and make a few adjustments to the AIP's controls. Click **B** and make some different adjustments. Click back on **A** and you'll see all the adjustments you made for **A**. Click back on **B** and you'll see all the adjustments you made on **B**. The **>>** key copies whatever setting you have in **A** over to **B**.

UNDO/REDO - **UNDO**, to the left, reverses all the damage mankind has done to the planet. **REDO**, to the right, restores all the damage mankind has done to the planet, starting with the latest control you tweaked on the AIP. There are unlimited levels of Undo, so feel free to go back to the Cretaceous Period.

PRESET BROWSER - Click on the piece of tape with the Sharpie text on it at the bottom of the unit to access the preset browser and load and save presets. The red arrows to the right of the tape allow you to step through and audition presets. The yellow **SAVE** button saves the current setting of the AIP to the preset library.

INFO (**i** within a circle) - Click this to see release information about the plugin and to access online resources.



USAGE IDEAS

The AIP is the most versatile and flexible plugin we've yet devised at Korneff Audio. You can use it on any individual channel and have just about every tool you need to get kick ass sounds. You can also use it on grouped channels, across the mix, on reverb returns—wherever. Don't let the name limit your creativity. Go AIP!

GUITARS - The AIP was specifically designed for use with big ass, heavy electric guitars. Some thoughts by the developer of the plugin, Dan Korneff, on using the AIP:

"My Philosophy when working with instruments is to remove the stuff I don't like first and then sweeten the sound. I always reach for my filters first and remove any excess rumble or fizz. Then I'll add a character piece of gear to help shape the tone in an overall sense.

The PSP (Proprietary Signal Processing) contains three tone shapes that I use frequently on my tracks. Once I find an overall tone that I'm really happy with, I'll reach for the EQ to help the track fit into the mix.

I'll usually add a little compression to help accentuate detail. Depending on the source content, that compressor maybe pre or post equalizer. At this point, I reach for the FDR (Frequency Dependent Reduction) module to tame down the low end build up that happens on heavier guitar parts. My final process in the signal chain is the Stereo Widener, which I use to spread those guitars out to the sides."

BASS - The AIP is an excellent processor for bass. Switch the COMPRESSOR to PRE so that when you fiddle around with eq settings you don't have to constantly reset the compressor threshold. Start with the RATIO at about 3:1, the ATTACK at about 80ms, the RELEASE at around 300ms. Watch the meter—you want its movement to seem similar to the bass part: if the part is slow, the meter should react slowly; if the part is fast, the meter should be moving quickly. Use the eq with wide (counterclockwise) Q settings for general low end shaping. You can get rid of thumpy low end garbage by popping in the HIGH PASS filter and getting rid of unmusical low stuff.

The FDR is especially useful for keeping neck and body resonances from making certain notes pop out loudly. Start with the BANDWIDTH at about 2 o'clock, pull down the THRESHOLD slider and adjust the FREQUENCY down into the 60 - 200 Hz area, which is where most of the fundamentals on the bass live.

KEYBOARDS - The secret weapon on the AIP for keyboards is the PSP, which will allow you to easily add an analog vibe to an otherwise sterile sounding keyboard patch. For additional mojo, slap the VCA COMPRESSOR to post, drop the threshold down low and then go around to the back and adjust the DISTORTION NULL to add some crunch. For keyboard



pad parts, max out the STEREO WIDENER to really get amp up the space and movement of the part.

ACOUSTIC INSTRUMENTS - The AIP combines an equalizer that has an especially sweet top end with a compressor that is very smooth. This makes the AIP an excellent plugin for treating groups of instruments, such as a string ensemble or even an orchestra (or orchestral samples). Set the farthest right band of the eq to a wide Q setting and then turn the FREQUENCY up above 17kHz or so. Flip around to the back, set the GAIN SCALE to about 50%, then flip back to the front and use the GAIN control to gently lift up the high frequencies to give the ensemble air and upper harmonic extension. This trick works beautifully on acoustic guitars as well. The ANALOG CHANNEL VARIANCE is fun to play with here. Experiment with changes the curves of the two channels to accentuate the differences in the instruments. Variance is also fun to play with whenever you're running doubled tracks through the AIP. Things matching exactly is boring. Little differences = cool ear candy. And don't forget that S T E R E O W I D E N E R !

GROUP BUSES - the AIP can be used across a group of similar tracks to bind them together and simplify your overall processing workload.

As an example, group a bunch of background vocal tracks and strap the AIP across that bus. Use the VCA Compressor to "glue" the vocals together. Set the RATIO to 2:1, ATTACK to 20ms, RELEASE to over 700ms and bring down the THRESHOLD such that the meter is almost always moving a bit. At those settings, the AIP's VCA compressor will work in a manner similar to an Opto unit, albeit without the tubes. The IMF can be used to nip out sibilance. Set the OCTAVE RANGE to 2 and poke around in the 4kHz region to find the offending frequencies. Add a bit of sheen with the HF COMPENSATION knob. You can control the obtrusiveness of the backing vocals by switching in the PSP and setting it to 1, which will pull the vocals back a bit in the mix, or 3, which will push them forward and make them more "in your face." Pulling back a bit on the TRANSIENT RESPONSE knob will also tend to tuck the vocals back. Lastly, use the STEREO WIDTH to either get the vocals to "wrap" around the lead vocal (push the slider to the right) or pull the slider to the left toward mono to group them spatially tighter and render them less distracting.

MASTERING or ACROSS the MIX BUS - A combination of a powerful eq, a gentle compressor and stereo widening make the AIP very capable as an overall mix processor and even as a mastering suite. Set the VCA COMPRESSOR to PRE and adjust the ATTACK to 10ms and the RELEASE to 200ms. Set the RATIO for something between 4:1 and 12:1. The higher you set ratio, the more noticeable the compression. Pull down the THRESHOLD so that the meter is showing maybe 4dB of gain reduction. Use the eq to sculpt the overall response—think wide BANDWIDTH. Use the GAIN SCALE on the back panel— set it to 25% as a starter—to reduce the throw of the front panel eq GAIN controls so that your movements are more precise. Finally, use the STEREO WIDENER to expand and accentuate the space of the finished mix. Many stereo widening schemes widen the soundstage at the



expense of weakening the center. You'll find that the AIP's Stereo Widener maintains a strong center presence even at extreme settings.

Ok, there it is— the War and Peace of plugin user manuals!

Thanks again for supporting Korneff Audio. We hope you love your AIP.

Make great records! Make great music.