STLTONES

IGNITE AMPS

NadIR

AUDIO PLUG-IN

USER MANUAL

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Introduction

NadIR is a zero latency, dual Impulse Response (IR) convolver, designed to be used as a cabinet simulator for guitar and bass (pre)amplifiers (AAX/AU/VST/VST3 or even hardware).

It has been designed to perform pristine quality convolution in real time, while being light on the CPU and easy to use, providing advanced built-in filters and delay controls to let guitarists and bassists shape their tone with ease, without the need to be professional audio engineers.

NadIR is meant to be used as a cabinet simulator for live playing and jamming, tracking or mixing inside hosts capable of AAX, AU, VST or VST3 Plug-Ins support.

Minimum System requirements

Windows:
Windows 7/8/10 (32/64 bit)
Intel Pentium 4 or AMD Athlon XP

Mac:
OSX 10.9
Intel processor with SSE2 instructions support

Installation

To install the NadIR Plug-In, just run the supplied installer and follow the instructions.

For Windows AAX/VST/VST3 format, we provide separate x86 (32 bit) and x64 (64 bit) binaries, so make sure to choose the right one according to your operating system and plug-in host specifications.
Keep in mind that x64 binaries will not run on 32 bit environments, while x86 binaries will most likely run on 64 bit environments, although we do not reccomend it for performance and stability reasons.
We strongly advice the Windows user against putting both x86 and x64 versions in the host AAX/VST/VST3 folder(s), as it may cause one of the versions to not be recognized as a plug-in.

Mac plug-ins (AAX/AU/VST/VST3) are compiled in Universal Binary format for Intel processors, containing both 32 bit and 64 bit code in the same bundle, which means that the user doesn't need to care about choosing x86 or x64 version, as the system will handle that automatically.

After that, you should (re)start your favourite AAX/AU/VST/VST3 host, making sure it re-scans your Plug-Ins folder(s) to recognize the NadIR as a new “Effect” Plug-In (please note that some hosts may not re-scan the plug-in folder automatically at every start-up, so you may need to do it manually. Refer to your host’s manual for instructions).
If everything is right, you should now see the NadIR entry into the “Effects” Plug-Ins list of your host.

About Impulse Responses and Convolution

NadIR is an “impulse response convolver”, but what does it mean? Let's start with the “impulse response” term. From Wikipedia:

> *In signal processing, the **impulse response (IR)**, or impulse response function (IRF), of a dynamic system is its output when presented with a brief input signal, called an impulse. More generally, an impulse response refers to the reaction of any dynamic system in response to some external change. In both cases, the impulse response describes the reaction of the system as a function of time*

The use of impulse responses in digital signal processing, has spread enormously in recent years, expecially for the implementation of reverberation processors, but this is not the only field where they can be used.

Impulse responses contain a lot of useful and very detailed informations about the system from which they've been captured, one of which is its **frequency response**. This makes them perfect for the simulation of systems like equalizers and, mostly, guitar or bass **cabinets**.

If you've ever looked at the frequency response graph of a loudspeaker, you've surely noticed how complex it is, with hundreds of sharp peaks and notches, which are basically impossible to replicate accurately with standard digital filters.
Properly capturing the impulse response of a cabinet and processing it through a math operation called "**convolution**", gives extremely accurate results in terms of frequency response fidelity.

The process of "capturing" the impulse response of a system, usually involves these simple steps:

- A generic test signal (which can be a Dirac, a sinesweep, or noise) is sent as input through the system of interest.
- The output of the system is recorded through a soundcard or a transducer (like a microphone)
- A math operation called "**deconvolution**" is performed using both the input test signal and the recorded system response, generating the impulse response of the system.

In audio signal processing, the impulse response generated through the deconvolution process, is usually stored into an audio file (in Wave, Raw or AIFF format, for example) which can have various lengths, depending on the system from which it has been captured. They can be up to 10 seconds (or more) long, when capturing big reverberation spaces or under 100 millisecond long when capturing loudspeaker cabinets or equalizers. This makes those files portable and conveniently small in terms of byte size, allowing users to have huge libraries with thousand impulse responses of various systems.

NadIR is designed to load these audio files and perform convolution of these impulse responses with its input signal, to recreate the frequency response of the captured system with great accuracy, in real time and at low CPU cost.
Being optimized to be used as a cabinet simulator and not as a general purpose convolution processor, it supports audio files long up to about 0.5 seconds, which may seem too restrictive, but is way more than enough to get impressive accuracy for cabinet simulation.

Some impulse response files and presets will be installed along with NadIR, so you'll have everything you need to create the most awesome tone you've ever heard.

Main Features

- Zero Latency
- Low CPU usage
- Three routing modes: Mono, Dual Mono and Stereo
- Selectable quality control for max IR length (up to 0.5 seconds)
- Automatic high-quality resampling for IRs with different sampling rates
- High-quality analog shaped filters
- Selectable delay for phase interactions between loaded IRs
- Continuous morphing control between loaded IRs
- Global input level and single IR level controls
- Fully automatable controls

NadIR Processing Diagrams

Mono Routing

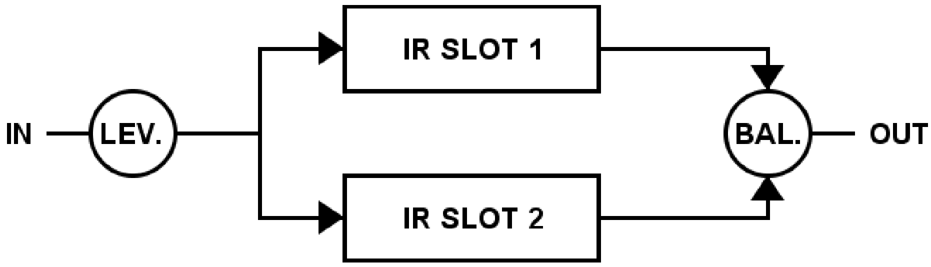

Fig. 1 - Mono processing diagram

Dual Mono Routing

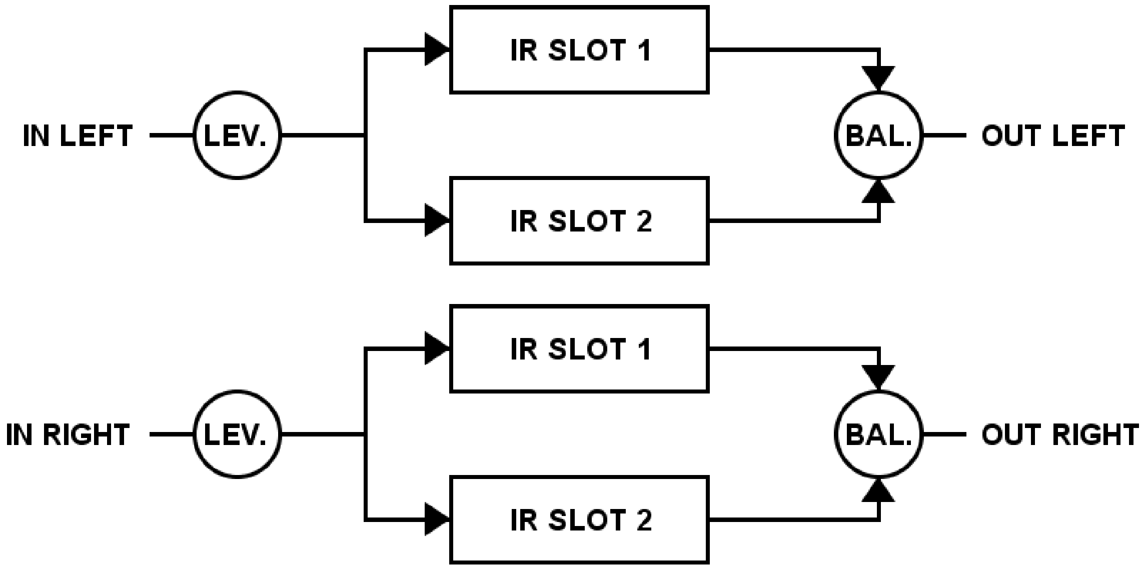

Fig. 2 - Dual Mono processing diagram

Stereo Routing

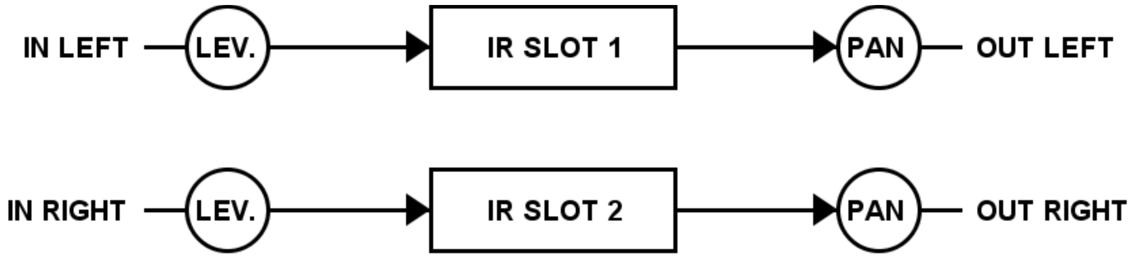

Fig. 3 - Stereo processing diagram

Graphic User Interface

Fig. 4 – Graphic User Interface

As you can see from the GUI screenshot (fig.4), we've decided to make NadIR interface really simple, in order to make the user experience easier.
The GUI is composed by a **header**, containing the Presets Management System plus other convenient functions, the **main view** with all the main NadIR features and a **footer**, containing the global controls of the plug-in.

Main View Controls

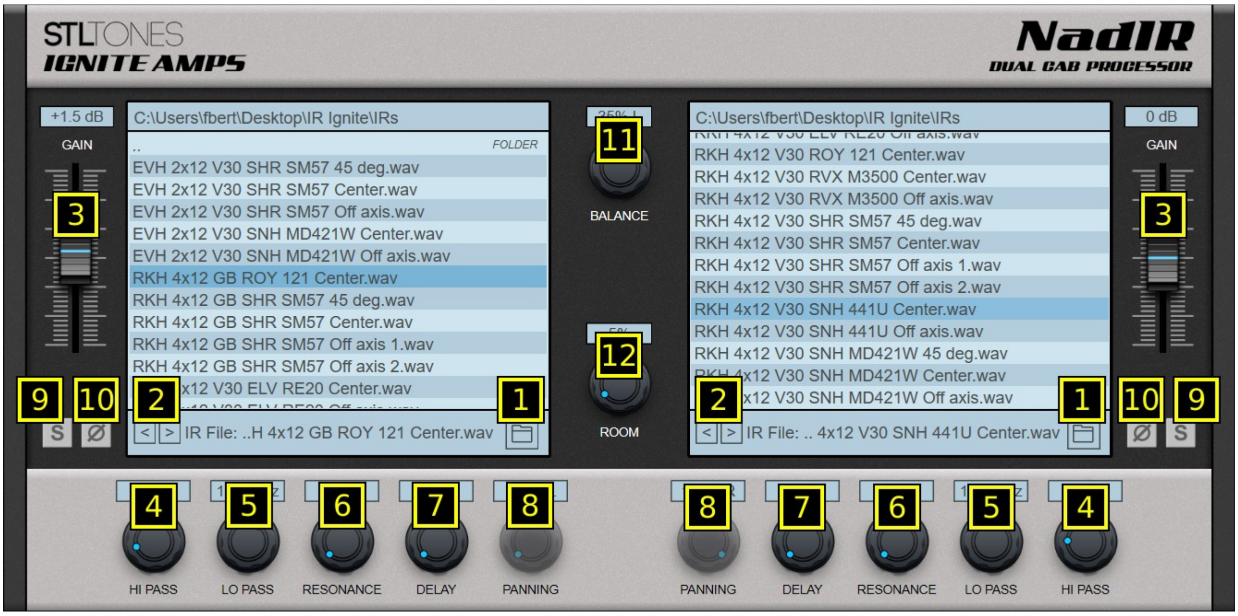

Fig. 5 – Main View

In the main view of NadIR (fig. 5) you'll find controls to load and manage the cabinet IRs loaded into NadIR.

[1] **Folder**: lets you load an IR file from your computer. After clicking on the folder icon, a file explorer will appear and you'll be able to locate and select any Wave, Raw or AIFF file which represents your IR. Once loaded, the name of the selected IR folder will be displayed next to the folder icon, the folder path will appear at the top and a list of all the available IRs and folders available at that path will be displayed into the IR Browser

[2] **Prev/Next**: lets you load the previous or next IR file in alphabetical order, with respect to the one currently loaded, cycling in the same working directory.

[3] **Gain**: lets you control the output volume of the IR slot. It can be useful to achieve the perfect balance between IRs when using both slots. It ranges from -18 dB to +18dB, default is 0dB.

[4] **Hi Pass**: lets you select the frequency of the high pass filter. It ranges from 10Hz to 400Hz and it can be really useful to control the low end response of the IR, preventing excessive resonance.

[5] **Lo Pass**: lets you select the frequency of the low pass filter. It ranges from 6 KHz to 22 KHz and it can be useful to control the high end response of the IR, preventing harshness, especially when the plug-in is used in conjunction with high-gain (pre)amplifiers.
NadIR uses advanced filtering algorithms to avoid the typical frequency “warping” affecting most digital filters, guaranteeing perfectly analog-like sound and perfectly clean transient response.

[6] **Resonance**: simulates the power amp + speaker interaction in tube amplifiers. It provides a boost at guitar cabinet speaker resonant frequencies.
Use this control at **high values** if you're using IRs captured using **solid state power amplifiers**, at low values as a tone shaping tool.

[7] **Delay**: lets you delay the convolution output by a short amount of time, from zero (default, no delay) to 20 milliseconds. Using short delay values can be useful to emulate phase interactions happening when using multiple microphones at different distances, to record guitar or bass tracks. Increasing the delay can be seen as moving the microphone away from the speaker. Considering that the sound travels at circa 340 m/s, 0.01 ms delay means

a distance of 0.34 cm from the source, 0.1 ms is equal to a distance of 3.4 cm and so on. Longer delay values can be used in conjunction with Stereo routing and wide panning, to emulate stereo guitars from a mono source.

[8] **Panning**: lets you decide the output panning of the convolution process. This control is enabled **only** when using NadIR in **Stereo** routing mode.

[9] **Solo**: sets the selected convolver in "solo" mode. By switching it on, you'll only hear the sound from the selected convolver. Please note that the output volume in "solo" mode is affected by both the Gain [3] control and the Balance [11] control

[10] **Phase**: lets you flip the phase of the loaded IR.

NadIR will automatically detect negative phase IRs upon loading and invert their phase to avoid cancellation when combining IRs, so use this control only if you want to achieve phase cancellation on purpose.

[11] **Balance**: when using two IRs at the same time, it lets you control their balance, effectively morphing the output sound between the two cabinet slots.
This control is enabled **only** when using NadIR in **Mono** or **Dual Mono** routing mode.
When NadIR is in **Stereo** routing mode, this control is disabled, as the left IR is convolved with the left channel and the right IR is convolved with the right channel exclusively.

[12] **Room**: lets you add a room reverb tail to the output. This control must not be mistaken with a general purpose reverb effect as it will only act on both cabs separately in mono mode, avoiding stereo diffusion.
The purpose of this control is to add back the room sound that may get lost when IRs are cut too short.

Header controls

Fig. 6 – Header

In the the NadIR header section (fig.6), you'll find controls for the Ignite Amps proprietary Preset Management System and other useful features:

[1] **Bank**: lets you change the name of the current bank. A bank is a group of presets which can be imported or exported to file, in order to save or recall settings and eventually share them with other the NadIR users, or just move them from one DAW to another.
Clicking on this control, will enable text editing, so just type in the new name and hit Enter to update.

[2] **Load**: lets you load a previously saved bank (using the Save [3] control) from a file. Clicking on this button will open your OS file manager to select the desired bank file. Once the file is chosen, all the presets contained in the bank will be available on the Preset Selector [4] and the first one will be automatically loaded.
Please note that when loading a new bank, all the previous bank settings will be discarded, unless you saved them on a file.

[3] **Save**: lets you save the current bank on file. Clicking on this button will open your OS file manager to select the path and the name of the file in which the bank will be stored. Once the file is chosen, all the presets contained in the current bank will be saved on the selected storage device and made available for future loading via the Load [2] control.
This is the only control that persists bank data on disk. Any other function of the Preset

Management System will act on the plug-in memory, so no changes will be saved on file unless you use this control explicitly.

[4] **Preset Selector**: lets you switch between presets contained in the current bank. Clicking on this control will open a popup menu showing all the available presets. Selecting a preset will immediately update the plug-in settings to the ones stored into it.
Additionally, when the mouse cursor is over this control, a button labeled “**E**” (as “Edit”) will appear on the right side: by clicking on it you can edit the name of the current preset through a dialog box.
Once a preset is loaded, as soon as you edit one of the plug-ins settings, an **asterisk** (“*”) will appear next to the preset name, in order to remind you that the settings for that preset are changed. You can revert the settings back using the Revert [7] function or permanently update them using the Store [8] function.

[5] **Add Preset**: lets you add a new preset to the current bank. Clicking on this button will create and load a new preset with a default name (“Preset <N>”), using the current plug-in settings. You can change the preset name by clicking 2 times on the Preset Selector [4].

[6] **Remove Preset**: lets you remove a preset from the current bank. Clicking on this button will erase the current preset and load the settings of the previous one on the list (or the next one, in case the removed preset was the first of the bank).
There is no undo function, so use this control carefully.

[7] **Revert**: lets you revert the selected preset settings to the original state. Clicking on this button will discard all the current plug-in settings and reload the last saved ones. This control is enabled only when a preset has been changed from its saved state.

[8] **Store**: lets you store the selected preset settings as its original state. Clicking on this button will save all the current plug-in settings and mark them as the last saved state, meaning that every successive use of the Revert [7] function, will recall these settings. This control is enabled only when a preset has been changed from its original state and will be disabled as soon as you click it (you'll also notice the asterisk next to the preset name disappear).

[9] **Copy**: lets you copy the current preset settings on the plug-in's clipboard. You can then use the Paste [10] function to reload them. The cool thing about this control, is that the plug-in's clipboard is shared among different the NadIR instances, so you can conveniently copy and paste settings from one to another, without having to explicitly save and load the bank.

Please note that as soon as all the instances of the NadIR are removed from the project, the clipboard data will be lost.

[10] **Paste**: lets you load the preset settings available on the plug-in's clipboard. You can then use the Copy [9] function to store them. The cool thing about this control, is that the plug-in's clipboard is shared among different the NadIR instances, so you can conveniently copy and paste settings from one to another, without having to explicitly save and load the bank.
Please note that as soon as all the instances of the NadIR are removed from the project, the clipboard data will be lost.

[11] **About**: clicking on this button will show up all the the NadIR additional information. Just click anywhere in the plug-in graphic interface to make it disappear.

Footer Controls

Fig. 7 – Footer

In the NadIR footer section (fig.7), you'll find controls to manage the plug-in to suit your system and mixing environment at best:

[1] **Input level**: it is a simple way to adjust the amount of signal going through the convolution engine.

[2] **Oversampling**: *not available in NadIR (not necessary).*

[3] **Routing**: controls the signal routing of the plug-in. Clicking on the text box will show up a drop down menu with three different options: Mono, Dual Mono and Stereo. To understand the difference between these options, we suggest you to take a look at fig. 1, fig. 2 and fig. 3, respectively.

[4] **Output**: lets you change the overall output level of the plug-in. Unlike the Volume controls located in the front panel, this control is completely linear and doesn't affect the dynamic behaviour of the plug-in in any way.

Tips for “digital” guitarists and bassists

- Always use the high impedance (Hi-Z) input of your sound-card (when featured). This will ensure less noise and signal loss. Most real (pre)amplifiers and stomp boxes, have an input impedance of 1MegaOhm, so it would be a good idea to get a sound-card with 1MegaOhm input impedance to use Ignite Amps simulators at their best.

- Make always sure to have the highest input signal before the AD conversion, while still avoiding clipping.

- Amp sims, stomp box and cabinet simulators are not noisy, they do not add noise. In fact, they're a lot less noisy than real hardware. If you have noise issues, check your guitar electronic circuit, cables and sound-card settings.

- In almost all cases, amp sims and stomp box simulators don't introduce noticeable latency. NadIR doesn't introduce any latency. If you're experiencing latency issues, check your sound-card settings (specifically reduce the “Input Buffer Size”).

- NadIR is a cabinet simulator, so it needs a preamp (like our Emissary, NRR-1, The Anvil, SHB-1 plug-ins, or even a real hardware preamp) and, optionally, a power amplifier simulator (like our TPA-1 plug-in), to sound like a real mic'd guitar/bass rig.

Acknowledgements

Ignite Amps wants to thank all the musicians interested in Ignite Amps projects who have shown great enthusiasm toward us, always pushing us to improve our work, helping us beta test and find bugs, everyone who has provided precious suggestions, kick-ass audio clips or videos, or have donated money for our research and development in the DSP field. Without these people, this plug-in would have never been created.

Thanks to You too, for downloading and trying the NadIR plug-in and for reading the f***ing manual! :-)

Sincerely
The Ignite Amps Team

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Ignite Amps uses:

- **Symbiosis** to provide Audio Unit support
- **Takuya Ooura**'s **OouraFFT** library, to perform fast Fourier transform
- **Aleksey Vanev**'s **r8brain** library, to perform high quality resampling