IGNITE AMPS

engineering for the moshpit

NadIR

AUDIO PLUG-IN

USER MANUAL

Summary

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 (VST/AU 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 VST or AU Plug-Ins support.

Minimum System requirements

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

Mac:
OSX 10.5
Intel processor with SSE2 instructions support

Installation

To install the NadIR Plug-In, just follow the instructions below, according to the platform and plug-in format you want to use.

Windows VST:
Copy the file **NadIR.dll** into your VST Plug-Ins folder.
(for example C:\Program Files\Steinberg\VSTPlugins)

Mac OSX VST:
Copy the bundle **NadIR.vst** into the path: /Library/Audio/Plug-Ins/VST/

Mac OSX AU:
Copy the bundle **NadIR.component** into the path: /Library/Audio/Plug-Ins/Components/

For Windows VST format, we provide separate x86 (32 bit) and x64 (64 bit) binaries, ***so make sure to choose the right one according to your operative 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 recommend such usage for performance and stability reasons.
We strongly advice Windows users against putting both x86 and x64 versions in the host VST folder(s), as it may cause one of the versions not to be recognized as a plug-in.

Mac OSX plug-ins (VST/AU) 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 VST/AU host, making sure it re-scans your Plug-Ins folder(s) to recognize 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 that manually. Refer to your host’s manual for instructions).
If everything is right, you should now see the NadIR entry into the “Effect” Plug-Ins list of your host.

About Impulse Responses and Convolution

NadIR is an “impulse response convolver”, but what does it mean? Lets start from 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 “capture” of the impulse response of a system, usually follows 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)
- Through a math operation called “**deconvolution**”, performed using both the input test signal and the recorded system response, the impulse response of the system is generated.

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 a low CPU cost.
Since it is optimized to be used as a cabinet simulator and not as a general purpose convolution processor, it supports audio files long up to 0.185 seconds, which may seem too restrictive, but is way more than enough to get impressive accuracy for cabinet simulation.

A lot of free and commercial cabinet impulse response libraries are available, so just find the one that works better for your taste and music style, load its impulse responses into NadIR and 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 0.185 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

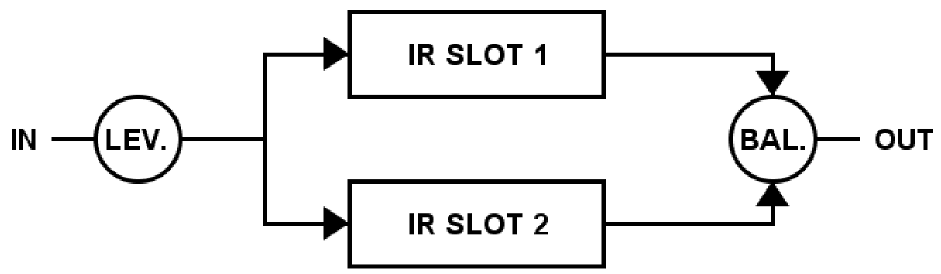

Fig. 1 - NadIR Mono processing diagram

Dual Mono Routing

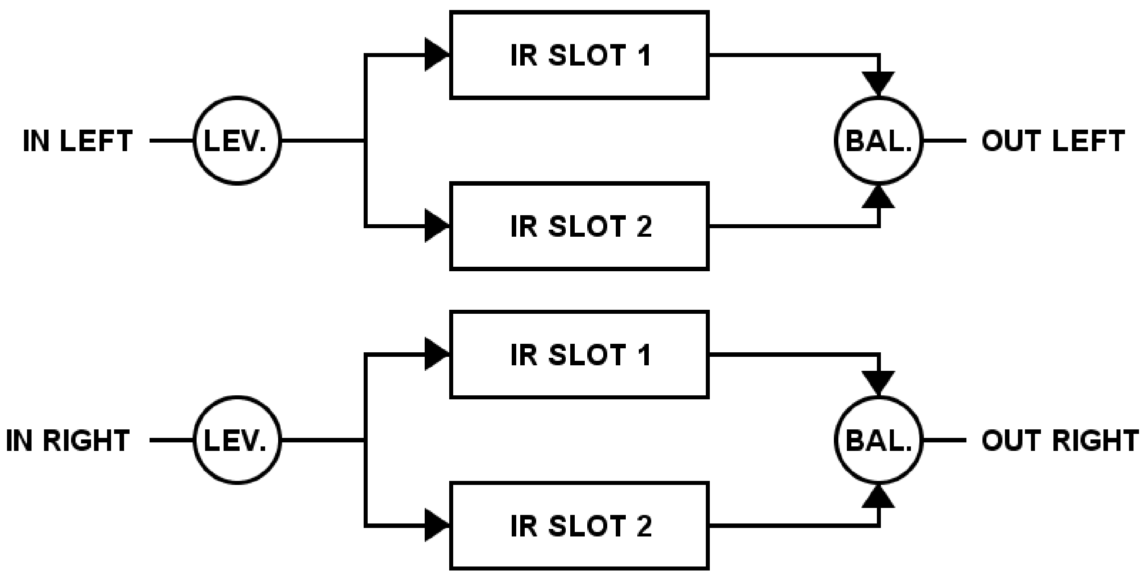

Fig. 2 - NadIR Dual Mono processing diagram

Stereo Routing

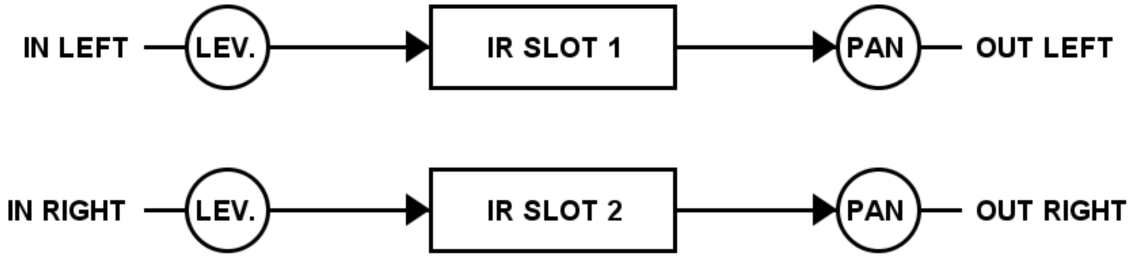

Fig. 3 - NadIR Stereo processing diagram

Graphic User Interface

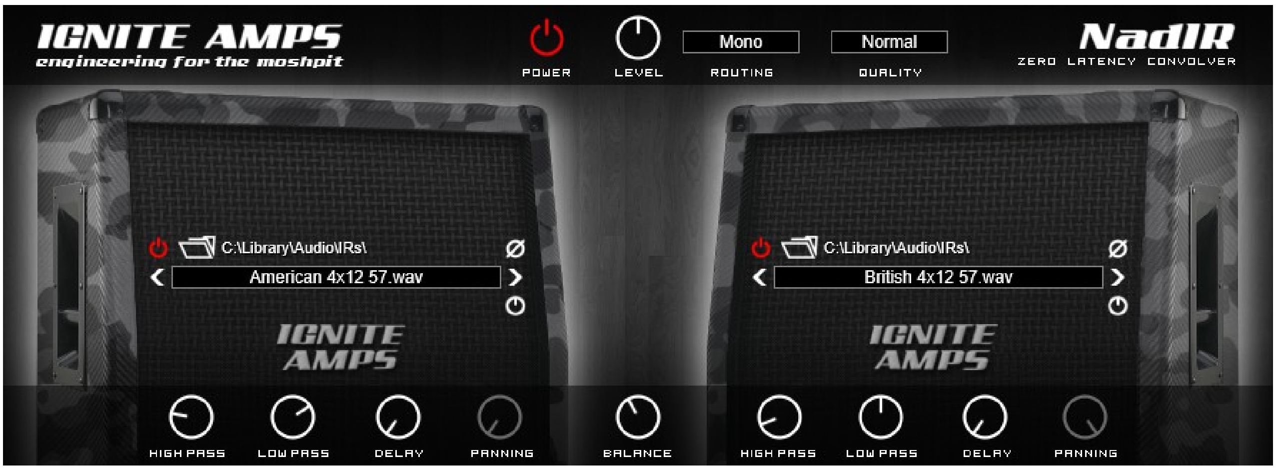

Fig. 4 – NadIR Graphic Interface

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 divided into three main sections:

- a header with global controls
- a central area to load cabinet IRs and modify their status
- a footer with controls for shaping the single cabinet IR sound and their balance

Controls

Header Controls

In the header panel of NadIR you'll find all the global controls to change the plug-in status, input level, processing mode and quality. From left to right:

> **Power**: controls the status of the plug-in. When enabled (red icon), NadIR will convolve the input signal with the user selected IRs. When disabled (white icon), the sound will be left untouched

Level: controls the global input level of the plug-in. It ranges from +18 dB to silence. Default is 0 dB or unity gain.

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.

Quality: controls the resolution of the selected IRs. The quality and the accuracy of an IR depends on the amount of samples it contains. Longer IRs provide a more detailed frequency response of the system from which they've been captured, at the cost of an increase of math operations (and so, an increase of CPU usage).
If the IR is not long enough, a considerable amount of details can get lost, expecially in the low frequencies, which are extremely important when trying to simulate a cabinet + microphone system.
Supposing you're recording at a sampling rate of 44100Hz, a minimum of 1024 samples is needed to achieve decent accuracy for cabinet simulation. Using shorter IRs can easily lead to a noticeable decrease of low-end resonance tightness and unnatural increase of very high frequencies, perceived as digital noise.
On the other hand, using longer IRs will result in a more accurate representation of the cabinet + microphone system, with a tighter low-end response, increased fidelity in the speaker resonant frequency and accurate high frequency content, without unnatural "fizz".
Using very long IRs (over 2048 samples at 44100Hz) may be important when searching for some additional reverberation caused by early reflections occurring in the room where the cabinet IR has been captured.
It is really important to note that the effective length of an IR must be calculated excluding all the zeros located on the "tail" of the IR itself, since they don't contribute to the convolution result. If an IR is composed by 1024 samples, but the last 256 samples are all zeros, its effective length is 768 samples (1024 – 256 = 768).

NadIR features four quality options:

Low: 0.023 second long IRs (or 1024 samples at 44100Hz)
Normal: 0.046 second long IRs (or 2048 samples at 44100Hz)
High: 0.092 second long IRs (or 4096 samples at 44100Hz)
Extreme: 0.185 second long IRs (or 8192 samples at 44100Hz)

Any IR loaded into NadIR will be automatically cut down according to the selected quality. Additionally, all the consecutive zeros located at the end of the IR tail will be ignored to avoid wasting CPU cycles.
NadIR is designed to be used as an IR based cabinet simulator, not as a general purpose convolver, so don't try to load reverb IRs (which are usually much longer than the maximum supported length) into it.

Cabinet Controls

In the central area of the interface, you'll find controls to load and manage the cabinet IRs loaded into NadIR. From left to right, top down:

Power: controls the status of the IR slot. When active (red icon) the incoming signal will be convolved with the loaded IR. When switched off (white icon), the slot will output silence.

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 path to the selected IR folder will be displayed next to the folder icon, so you'll always know in which directory you're currently working.

Phase: lets you flip the phase of the loaded IR. This can be very useful when using both IR slots, to achieve or avoid phase cancellation, if needed.

Left arrow: lets you load the previous (supported) IR file in alphabetical order, with respect to the one currently loaded, cycling in the same working directory.

IR Name label: it shows you the name of the currently loaded IR file. If you click on the label, a drop down menu will appear with a list of all the (supported) IRs located in the same working directory. You can conveniently select a different IR from the drop down list itself.

Right arrow: lets you load the next (supported) IR file in alphabetical order, with respect to the one currently loaded, cycling in the same working directory.

Volume: 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 -12 dB to +12dB, default is 0dB.

Footer Controls

In the footer of the GUI, you can find additional controls to shape the sound of the single IR slot to a deeper detail. From left to right:

High Pass: lets you select the frequency of the high pass filter. It ranges from 20Hz to 200Hz and it can be really useful to control the low end response of the IR, preventing muddiness and excessive resonance.

Low Pass: lets you select the frequency of the low pass filter. It ranges from 4000Hz to 22050Hz and it can be useful to control the high end response of the IR, preventing “fizziness”, especially when the plug-in is used in conjunction with high-gain (pre)amplifiers.
NadIR uses Ignite Amps' advanced filtering algorithms to avoid the typical frequency “warping” affecting most digital filters, guaranteeing perfectly analog-like sound and perfectly clean transient response.

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.

Pan: lets you decide the output panning of the convolution process. This control is enabled only when using NadIR in Stereo routing mode.

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.
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.

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 hardwares. 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 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.

Acknowledgments

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 Crew

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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