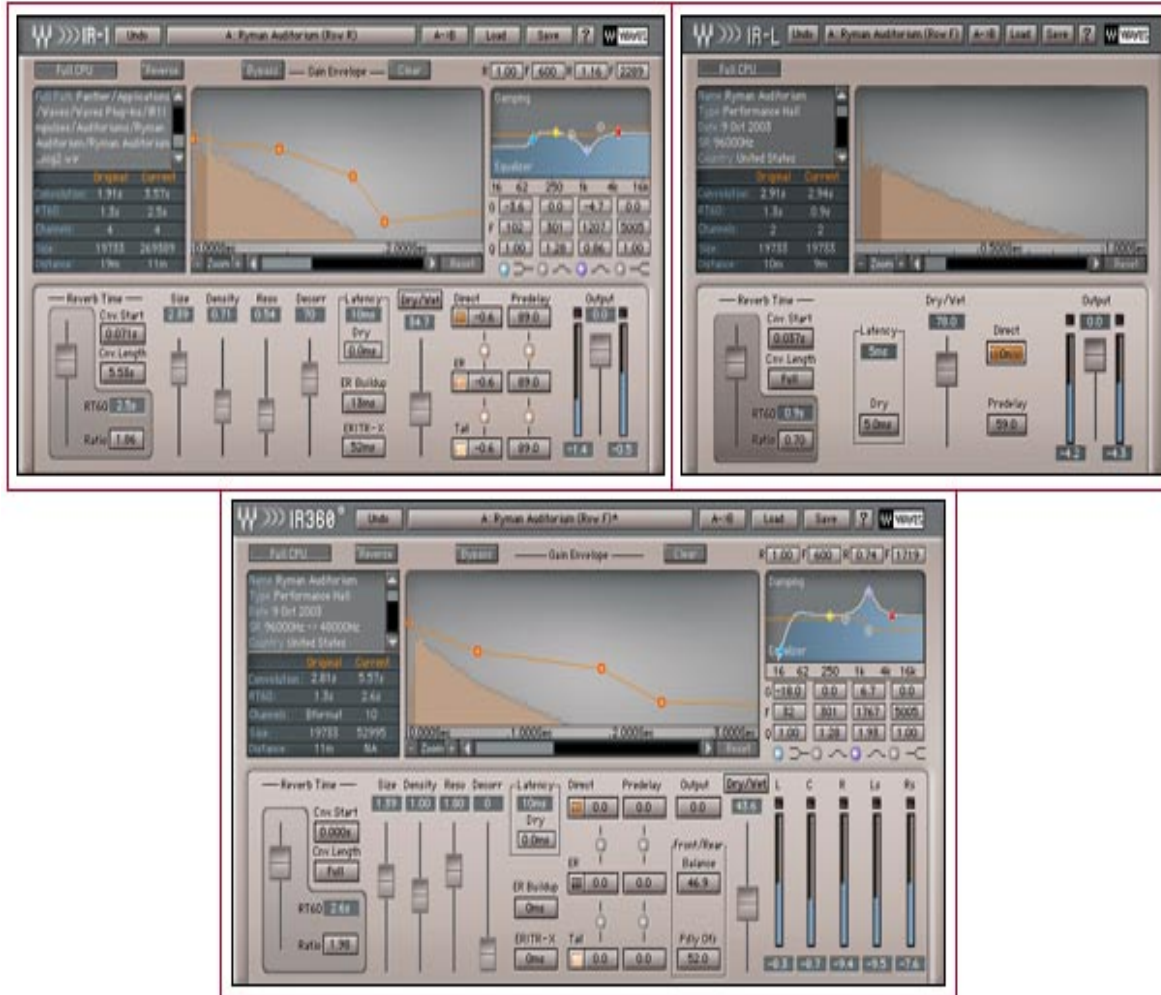


# The IR-1, IR-L and IR-360<sup>0</sup> Parametric Convolution Reverbs



## Waves Parametric Convolution Reverb

### IR-1, IR-L and IR-360

### User's Guide

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## Introduction

The Waves **IR-1**, **IR-L** and **IR-360** plug-ins create extremely high quality replicas of acoustical spaces and synthetic reverbs. They also allow the user to alter many aspects of the sound of these reverbs.

For the purpose of this Manual, when discussing attributes shared by the Waves **IR-1**, **IR-L** and **IR-360** Parametric Convolution Reverb plug-ins, we will simply refer to them all as the **IR-series**.

## Overview

The Waves **IR-series** are convolution-based reverb processors. They give the user total control of a reverb's properties. *Impulse Response Convolution* is the process of forcing a certain acoustic signature on any incoming audio signal. An Impulse Response (IR) is created by feeding a very short pulse (Impulse) into a linear system and measuring the resultant output (Response).

### ***Synthetic Digital Reverberation***

Historically, most digital reverberation processors have used a synthetic reverberation engine for high efficiency. This would usually consist of two reverberation generators:

- *The Early Reflections Generator.* This generator is based on a short Finite Impulse Response (FIR) filter with less than 100 reflections. A short stage of convolution is performed with this FIR.
- *The Reverberation Tail Generator.* This is based on a recursive Infinite Impulse Response (IIR) filter. The IIR comprises a feedback loop with a decay factor and inner dispersion filters. It makes the reverb more complex, less resonant, and more rich and natural sounding.

The tail generator is efficient. It can generate a one-second reverberation time or a five-second reverberation time using the same amount of processing power. Such reverbs usually allow a great deal of control over the reverberation properties. These controls have become classics. They include Reverb Time (RT), Predelay, Size, High and Low Frequency Damping, and Diffusion.

Typically, the goal for using a reverb is to emulate the acoustics of natural spaces such as rooms or halls, or to add "wet" reflections to "dry" sources for creative and aesthetic reasons.

Examples of these reverb processors include the Waves **TrueVerb** and **Renaissance Reverb**.

## ***Sampled-Acoustics Convolution Reverberation***

In math, the term *convolution* means an integral which expresses the amount of overlap of one function  $x$  as it is shifted over another function  $y$ . In the literal sense, it means to “roll together” or to “fold together”. A convolution reverb blends together an input signal with another type of audio signal called an *impulse response*.

Sampled-acoustics convolution processors are a different type of digital reverb than the digital reverbs that preceded them. They are much more calculation-intensive, but offer the ability to faithfully *convolve* any given impulse response with any input signal.

The Impulse Response (IR) of a certain reverb hardware setup or the acoustics of a real space can be captured and imported into the sampled acoustics reverb. It will use the Impulse Response as a filter consisting of thousands of reflections. The longer the IR, the more intensive the calculation, and thus the more processing power that will be required.

The result is that running a dry signal through a typical sampled acoustics reverb will sound much like running it through the original device with the same settings, or like recording the signal in the sampled space with a microphone setup similar to the one used for capturing the acoustics. These reverbs usually offer very little control of the *reverberation properties* and are essentially designed to faithfully super-impose the *sampled acoustics*.

## ***The Perfect Hybrid***

The Waves **IR**-series plug-ins are designed to bridge the gap between *synthetic digital reverberation* and *sampled acoustics convolution reverberation*. They provide a convolution-based reverb processor with total control of the reverb parameters. The **IR**-series uses the *impulse response* as raw material. This allows the user to process the impulse response with controls that are similar to those offered in classic digital reverbs.

The process is power intensive. However, the manipulations performed on the impulse response are calculated offline so that they don't consume all of the CPU's power. Then they are applied by way of straightforward convolution.

The result is that you have the natural and high-quality sound of sampled space acoustics as well as the controls to shape the sound to your needs. These manipulations go beyond those offered in regular digital reverbs.

## ***The Impulse Responses***

Another major part of the **IR**-series is the Impulse Response (IR) Library. The library offers a set of high-definition IR's. Waves paid special attention to this part of the product because it is the basic reverb sound of the **IR**-series reverbs. We established a *special recording setup and method* specifically for IR recording. This method provides exceptional signal-to-noise ratio, very low harmonic distortion, and a wide frequency bandwidth (20Hz-32kHz). Thus, it offers optimal sonic depth and dimension.

This method for taking acoustical measurements was developed by Waves in collaboration with Professor Angelo Farina of the University of Parma in Italy. The basis of the acoustic capture method was documented and presented in the AES (Audio Engineering Society) paper *Recording Concert Hall Acoustics for Posterity*. To see this article, please check out the following link:

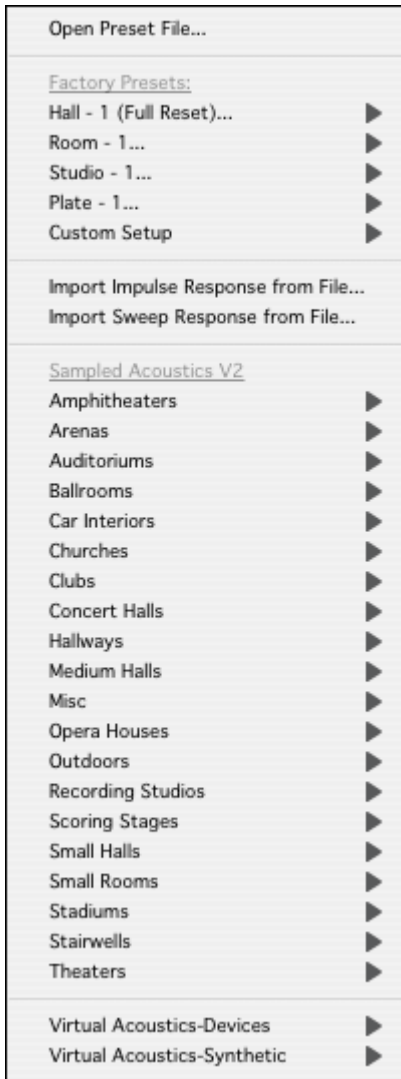
[http://www.waves.com/objects/pdf/general/article\\_farina05.pdf](http://www.waves.com/objects/pdf/general/article_farina05.pdf)

## **IR Library**

The library that is included with the purchased **IR**-series covers all of the reverberation needs of the most demanding studio professionals. It consists of 4 main parts:

- ***The Basic IR set*** is installed with the **IR**-series. It is available in the downloadable demo as well as in the full product.
- ***Sampled Acoustics V2*** contains a set of world-class Concert Halls, Opera Houses, Cathedrals, Studios and more. These are natural spaces captured in the USA, Italy, Japan, the UK, Israel, Australia, and more. Our original Sampled Acoustics library has been reprocessed with enhanced quality, and new micing techniques were added. In accordance with our vision of continuously expanding the IR library, new venues have also been added to the library.
- ***Virtual Acoustics - Devices*** contains Sampled IR's from classic reference reverb hardware.
- ***Virtual Acoustics – Synthetic*** contains synthetically generated or pre-processed IR's.

Accessing these IRs is easy. Select the desired preset from the plug-in's "Load" button, which will present the menu as follows:



**Factory Presets** is the first section. It is available with the installation of the IR-series, even for demo purposes. The IR data for these presets resides in the folder "IR1Impulses V2" in a directory called Basic.

The Path is:  
Waves Plug-Ins dir/IR1Impulses V2/Basic.

The *Custom Setup* group contains presets that only have settings. These presets can be loaded onto any IR.

## Sampled Acoustics V2

The presets in this section are based on samples of real places. The **contents** of the Sampled Acoustics V2 folder on the DVD have to be copied into the **IR1Impulses V2** folder inside the Waves Plug-Ins folder. You can add or delete folders from this directory. Each time upon loading, the Plug-In will scan the directory and dynamically build its preset menu according to the folder hierarchy.

## Virtual Acoustics - Devices and Synthetic

The presets in the Virtual Acoustics-Devices section are samples of a reference hardware Reverb Unit. The Virtual Acoustics-Synthetic presets are synthetically generated and processed IR's. The **contents** of the Virtual Acoustics folder on the DVD have to be copied into the **IR1Impulses** folder inside the Waves Plug-Ins folder. You can copy the Devices and/or Synthetic folders into the **IR1Impulses** folder, and the Plug-In will scan them upon loading.

## Import Impulse Response from File...

The **IR-series** allows you to import your own IR's. You can sample them yourself, download them from the Internet, or find them elsewhere.

## Import Sweep Response from File...

The **IR-1** and **IR-360** allow you to import the Sweep Response that you record in an acoustical space or hardware device. The plug-in will then process that Sweep Response into a new IR.

Here are step by step instructions to make your own IR:

- Play your sweep file supplied on the CD and also downloadable from the Waves website (a 15 second sweep followed by 10 seconds of silence) through speakers in the room to be captured.
- Record the result with 1 or more microphones to 1 or more tracks.
- Create a new folder (We'll name it "My Sweeps") inside your "IR1Impulses V2". folder, where the plug-in searches for new IRs.
- Put your recorded sweep files inside your new "My Sweeps" folder.
- Load the **IR-1** or **IR-360**; Press Import Sweep Response From File...
- Guide the browser to the "My Sweeps" folder.
- Load the .wav files one by one as you are prompted by the plug-in.
- After some calculations, an Impulse Response .wir file will be automatically saved next to your sweep files in the "My Sweeps" folder.
- To save this new IR as a preset, press Save, then select Save to New File.
- Navigate to the "My Sweeps" folder.
- Reload the **IR-1** or **IR-360**; The new preset will appear in the load menu.
- Congratulations! You have created a new IR! You may now optionally delete the recorded sweep files from the "My Sweeps" folder.



## Tips for Sampling an Acoustic Space

When sampling an acoustic space, a typical setup includes a multi-track recorder, an amplifier and speaker to play the sweep signal into the space, and one or more microphones (and mic preamplifier(s), if needed) to capture the sound.

Ideally, the speaker, amplifier, microphone(s), and microphone preamplifier(s) used should be as transparent as possible, having flat frequency responses. A microphone of any polar pattern will work, although we recommend cardioid or omni-directional microphones for optimal results.

In most venues, the speaker that plays the sweep tone should be placed where the performers would be; on stage in a concert hall, for example. The microphone should be placed somewhere in audience area; naturally, a microphone placed closer to the stage will capture more direct sound and less reverberation than a microphone placed in the last row. Keep in mind that the presence of a full audience, as well as musicians or musical equipment on the stage, will absorb sound and dramatically change the reverberation quality of the acoustic space.

### ***MEASURING VINTAGE PLATES, SPRING REVERBS OR ANY REVERB UNIT***

The measuring procedure is quite similar to a normal voice processing (via the plate or the spring unit). The difference is that we do not use an aux send. Instead we connect the output of our sweep playback device directly to the plate or spring reverb device.

#### **SET UP:**

Connect the output of your source (Pro-Tools HD card, etc...) directly to the input of the measured unit. As for the output of the measured unit, connect it directly to the recording device (HD card, etc...)

Output source (HD card, etc...) → Input of the plate.

Output of the Plate → Input of the recording device (HD card, etc...).

#### **SESSION:**

Create two different channels. The first channel will be the sine sweep channel. We use a simple sweep signal, which is a pure tone whose frequency is changed continuously from low to high frequency. The second channel should record the wet signal, which we receive from the plate reverb. If there is more than one setting on the unit please measure them all. For instance, reverb settings (1-5), mono-to-mono, mono to stereo, and stereo to stereo.

## **MEASURING HALLS, ROOMS**

The measuring procedure of a venue is slightly different than that of measuring devices. Measuring venues requires a larger setup, which includes a playback device, a microphone and a loudspeaker. We can use the same session we were using for the device recordings, only instead, we connect the output of our playback device to a speaker, and a microphone should be assigned to the input of our recording device.

### **SET-UP:**

Connect the output of your source (HD card etc...) directly to or through the mixing console to the loudspeaker. For the input, connect the microphone cable directly to or through the microphone pre amp to the recording device.

Output source (HD card etc...) → Input of the speaker.

Microphone cable → Input of the recording device (HD card etc...).

You can use a microphone of any kind, with any polar patterns. We recommend a cardioid or omni-directional microphone. After deciding which microphone to use, set up the microphone with the speaker located in front of the microphones, to achieve the best results.

### **MONO SESSION:**

To make a Mono IR, create two different channels. The first channel will be the sine sweep channel and should be assigned to the loudspeaker. The second channel should record the microphone signal.

### **MONO TO STEREO SESSION:**

Simultaneously record one speaker (Center) with the Left and Right mics.

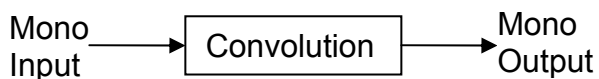
### **FULL STEREO SESSION:**

To create a 4 convolution IR, Record your Left speaker with the Left mic (LL) and Right mic (LR). Then record the Right speaker with the Left mic (RL) and the Right mic (RR).

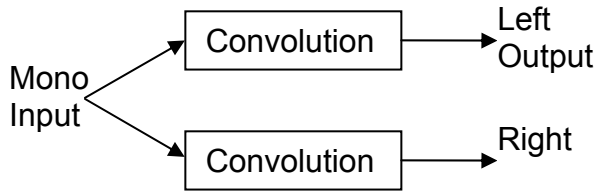
## **IR-SERIES PLUG-IN CHANNEL COMPONENTS**

The IR-series is available in several channel configurations as follows:

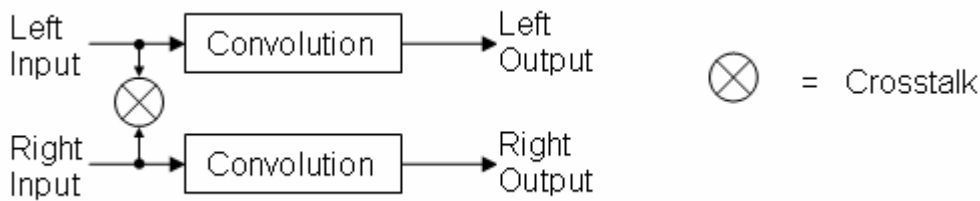
*Mono to Mono:* 1 convolution



*Mono to Stereo*: 2 convolutions: 1 left, 1 right.

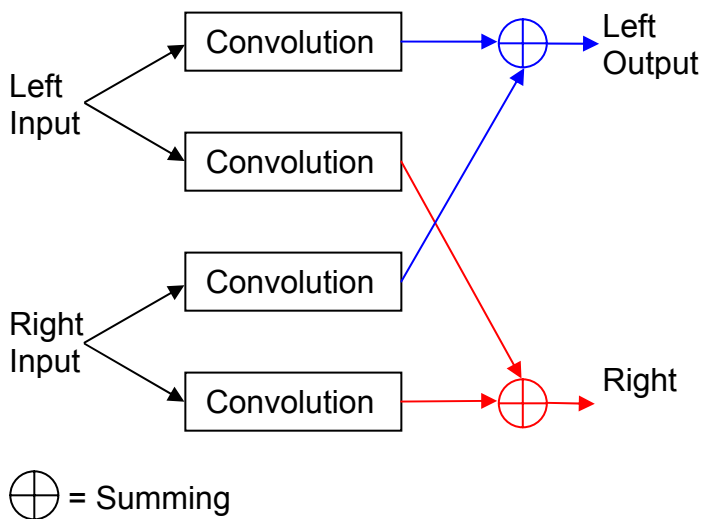


*Efficient Stereo*: 2 convolutions: Left in to Left out and Right In to Right Out. This component does multi-mono processing.



- With *Efficient Stereo*, the added control for input channel *Crosstalk* can keep the channels separate, or sum the wet path input. This component is very practical for stereo-to-stereo reverberation. It provides beautiful reverberation with all the depth and dimension that you might need on an auxiliary stereo reverb.

*Full Stereo to Stereo*: 4 convolutions, Left Channel to Stereo, and Right Channel to Stereo. This component presents a more purist approach to Stereo Reverberation than *Efficient Stereo*, but will use twice the CPU.



## About the IR-360:

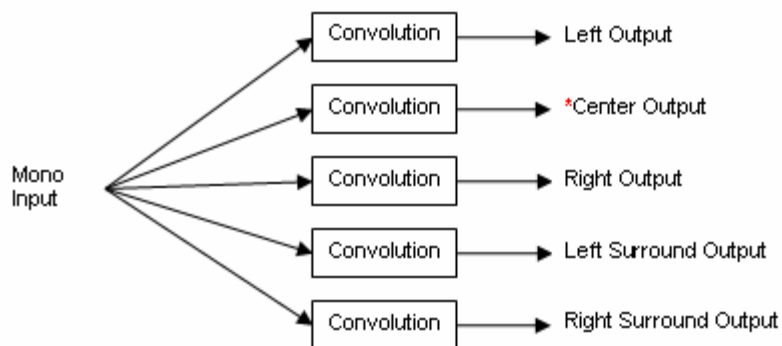
The **IR-360** is the surround sound version of the **IR-1**. The **IR-360** is HTDM for Pro Tools TDM Users. **IR-360** owners will also receive an **IR-1** included in the purchase.

Supported channel configurations of the **IR-360**:

<u>Channel Configuration</u>	<u># of Convolutions</u>
Mono to 5 Channel Discrete Surround processor	4 or 5
Efficient Stereo to 5 Channel Discrete Surround processor	4 or 5
Mono to 5 Channel Sound Field	3
Efficient Stereo to 5 Channel Sound Field	3
Full Stereo to 5 Channel Sound Field	6
Sound Field Surround to Surround	6
Efficient Discrete Surround to Surround	4 or 5

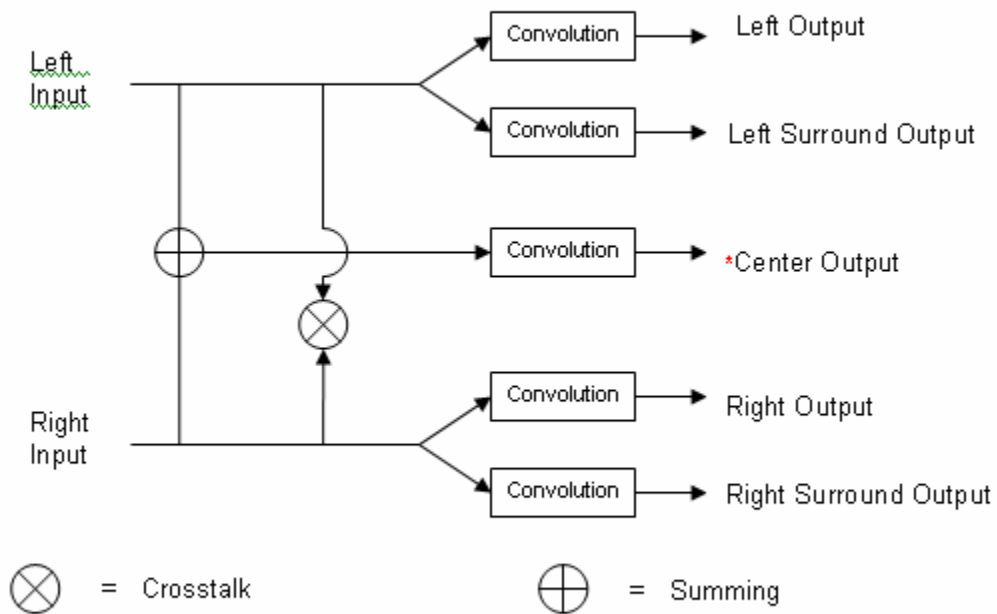
*Diagrams of these channel configurations are shown below:*

### **Mono to 5 Channel Discrete Surround Processor: 4 or 5 Convolutions**



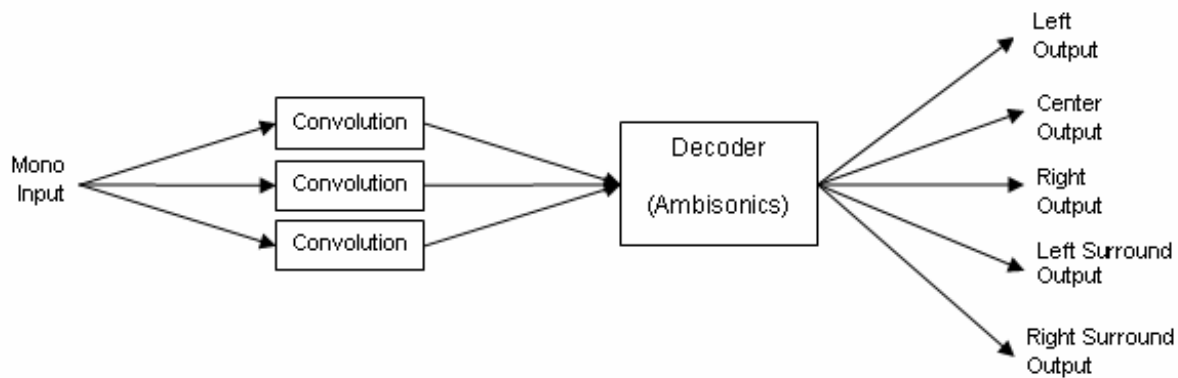
\*Available through importing only – The supplied discrete IRs have 4 channels (4 convolutions).

## Efficient Stereo to 5 Channel Discrete Surround Processor: 4 or 5 Convolutions

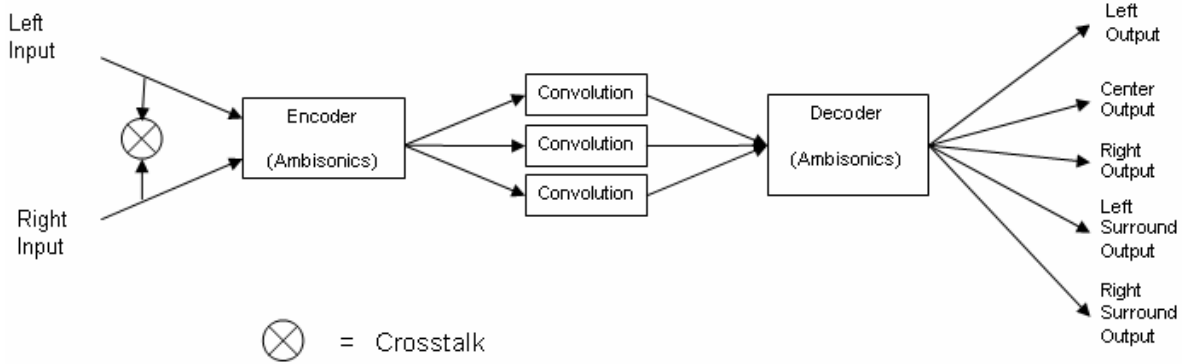


\*Available through importing only – The supplied discrete IRs have 4 channels (4 convolutions).

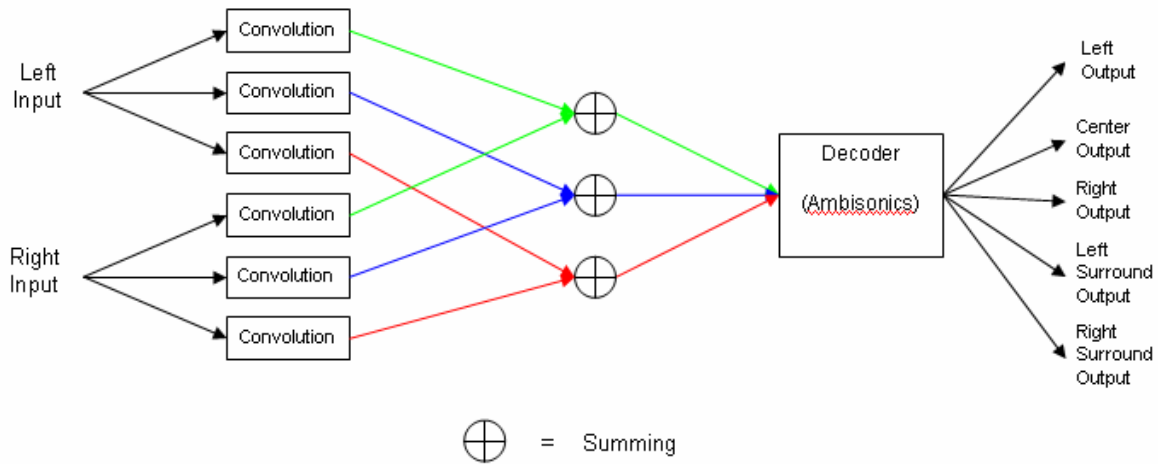
## Mono to 5 Channel Sound Field: 3 Convolutions



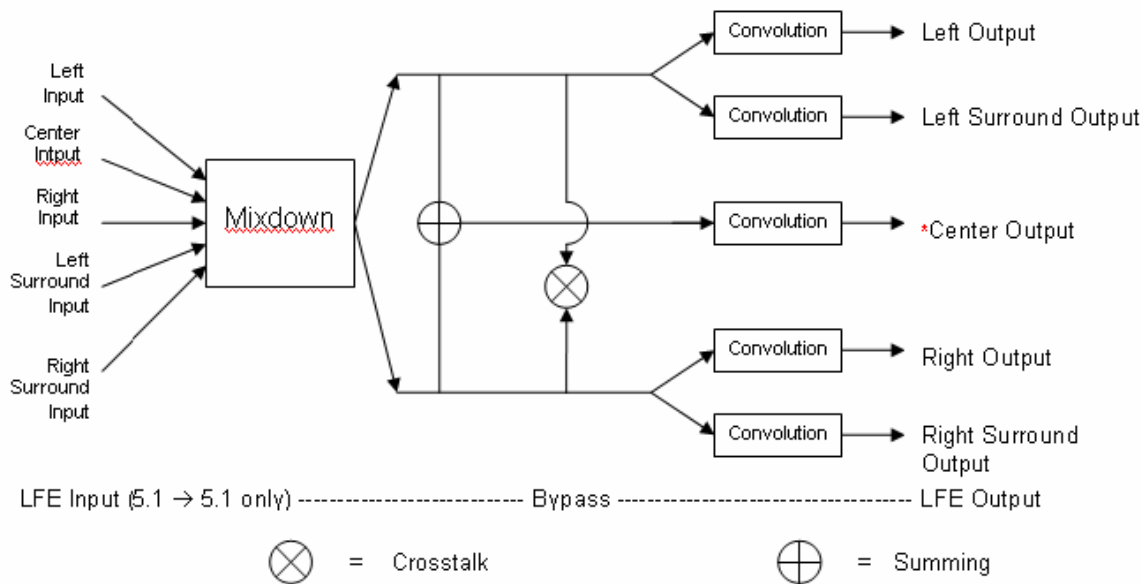
### Efficient Stereo to 5 Channel Sound Field: 3 Convolutions



### Full Stereo to 5 Channel Sound Field: 6 Convolutions

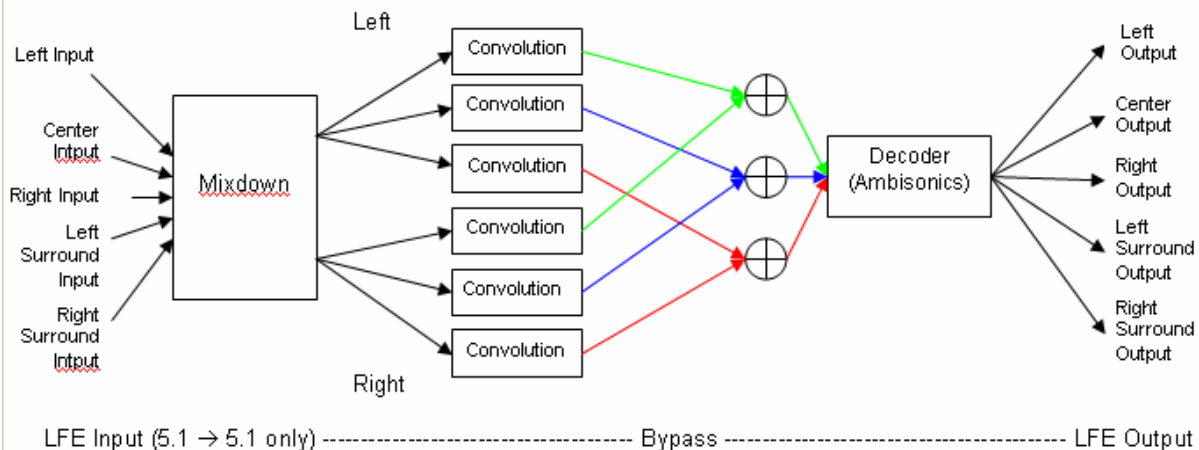


## Efficient Discrete Surround to Surround: 4 or 5 Convolutions



\*Available through importing only – The supplied discrete IRs have 4 channels (4 convolutions).

## Sound Field Surround to Surround: 6 Convolutions



## IR-SERIES CONTROLS AND DISPLAYS

The **IR-series** has a large main window with controls similar to those of a standard digital reverb. Most of the values are displayed as a ratio of the nominal value of the original IR. The **IR-360** has the most adjustable parameters, followed by the **IR-1** and then the **IR-L**.

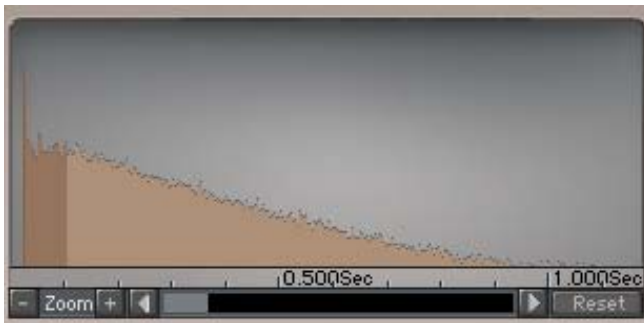
### About the IR-L

The **IR-L** is essentially an **IR-1** with fewer adjustable parameters. It has the same high sound quality as the **IR-1**. The **IR-L** can be upgraded at any time to an **IR-1** or **IR-360**.

### IR-L CONTROLS AND DISPLAYS

*(IR-1 and IR-360 users can skip to the following section labeled "IR-1 and IR-360 Controls and Displays" on page 17)*

### THE IR GRAPH



The IR graph shows dB level on the vertical axis and time on the horizontal axis. The range of the IR display is from 0dB at the top to -90dB at the bottom.

The scroll bar on the bottom allows you to zoom in and out horizontally as well as scroll back and forth. Note that while IRs are limited to six seconds in length, the window itself has a maximum length of ten seconds.



## Reverb Time



The Reverb Time section includes these 4 parameters:

**CONVOLUTION START:** Range: 0 TO 1 second; Default: 0 seconds

The Convolution Start determines the starting point of the convolution process. It enables you to remove unwanted pre-delay at the beginning of an IR.

**CONVOLUTION LENGTH:** Range: 0 seconds to 6 seconds; Default: Full

This control sets the length of the process filter. By default it is set to Full, which means that the **IR-L** will convolve the input with the whole Impulse Response. However, if the IR is longer than 6 seconds then it will be cut to 6 seconds. This will be indicated in the IR Properties display. You can also set the Convolution length manually to be shorter than the RT60 Reverb time which will save some CPU. This will make the Impulse Response stop at some point down the slope, creating a “Gated” reverb sound. The default Full setting will only use the amount of CPU it needs for the loaded IR, so there is nothing to gain by manually setting it to 2 seconds for a 2 second IR.

**RATIO:** Range: .25 to 1.00; Default: 1.00

The Reverb Time Fader controls the *Reverb Time Ratio*. The resulting reverb time will be the original RT60 multiplied by the ratio.

**RT60:** **RT60 (Reverb Time, 60dB)** is defined as the time it takes the sound pressure level of a sound to drop by 60dB in gain once the source has been shut off. It is actually a slope factor. RT60 is a term that is commonly used in building construction and architecture to describe a room’s acoustic properties.

This field shows the reverb’s current RT60. It is the original RT60 multiplied by the Reverb Time Ratio. To determine what the original RT60 is, set the Ratio to 1.00. The original RT60 is calculated by analysis of the loaded IR.

The RT60 display is active *only* when RT60 may be computed on the IR. There is a minimum IR length (about 100 ms) for computing RT60 as well as for activating the reverb time ratio.

**DRY/WET CONTROL:** Range: 0 to 100% wet; Default: 100% wet



This control sets the balance between the “Dry” signal and the convolved signal. A 50% Wet setting is the “natural” balance point.

The **IR-series** lets you work either with the recorded Direct signal, or with the Dry signal (that is, the unprocessed input to IR-1). When the Direct signal is turned off, and Dry/Wet is set to 50%, the Dry signal will have the same level and relative delay as the recorded Direct signal, so the Dry will replace the Direct. This is useful if you do not want any coloration of the Dry signal. When working with the Dry signal, it is generally recommended to turn *off* the Direct.

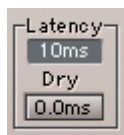


**DIRECT:** Range: ON/OFF; Default: Off

This switches the direct signal on or off.

**PRE-DELAY:** Range: 0-500ms; Default: 0

This controls the amount of time that the entire IR is delayed.



**LATENCY:**

The **IR-series** does not declare any inherent latency. The reason for this is that the Dry path goes to the output without any actual latency. The process, however, does have some latency: 11.6 ms at 44.1 to 48kHz and 5.3 ms at 88.2 to 96kHz.

Since reverbs are commonly used in an aux send configuration and since most host applications cannot compensate for the delay for plug-in latency on auxiliary sends/returns, the Wet path latency is shown *within* the plug-in and regarded as a nominal predelay.

**DRY:** Range: 0-30ms; Default: 0ms

When working as an insert, the Dry latency control can be used to align the dry signal to the wet signal or even to advance it ahead.

**CPU MODE:** Range: Full CPU or Low CPU. Default: Full CPU



This control allows you to select between Full CPU (Central Processing Unit) or Low CPU. Low CPU presents an option to slightly compromise the resolution of the convolution process while saving up to 45% of the CPU cycles (depending on the IR length and Sample rate). The average savings is about 20%.

### Offline Calculation indicator

Manipulating the impulse response to reflect new controls settings requires recalculation of the whole impulse response. This calculation process can take up to several seconds. While the impulse response calculation is taking place, the Reverb IR graph displays “Calculating...” and the audio is muted. After the calculations are performed, the graph updates and the audio un-mutes.

**OUTPUT GAIN:** Range: 0dB to –50dB; Default: 0dB



This controls the overall output level of the **IR-1**.

**CROSSTALK** : Range: 0.0 to 100.0; Default: 0.0

*This control is only available in the Efficient Stereo component.*

It controls how much Crosstalk is allowed between the **IR-series'** input channels. When the Crosstalk is 0.0 the process is multi-mono, convolving left to left and right to right.

When crosstalk is 100.0, the plug-in will convolve a summed stereo input. The dry signal is not affected, so the reverb will have elements from the left channel in the reverb of the right channel and vice versa.

## IR-1 AND IR-360 CONTROLS AND DISPLAYS

(IR-L users can skip this section and resume reading at **Toolbar Wavesystem Controls** on page 30)

### THE IR GRAPH



The IR graph shows dB level on the vertical axis and time on the horizontal axis. The range of the IR display is from 0dB at the top to -90dB at the bottom. The display incorporates an envelope curve (in orange). The scaling of the envelope curve is different: its flat bypass point is two-thirds of the graph's height and its range is from +30dB to -60dB.

The scroll bar on the bottom allows you to zoom in and out horizontally as well as scroll back and forth. There's a Reset button on the right of the scroll bar that sets the window to its maximum length. Note that while IRs are limited to six seconds in length, the window itself has a maximum length of ten seconds.

Above the IR window are buttons for bypassing and clearing the envelope curve. Clearing the envelope curve resets it to its flat point of 0dB (two-thirds of the graph's height). To the left and above the IR window is the Reverse button which flips the IR front-to-back.

## Reverb Time



The Reverb Time section includes these 4 parameters:

**CONVOLUTION START:** Range: 0 TO 1 second; Default: 0 seconds

The Convolution Start control determines the starting point of the convolution process. It enables you to remove unwanted pre-delay at the beginning of an IR.

**CONVOLUTION LENGTH:** Range: 0 seconds to 6 seconds; Default: Full

This control sets the length of the process filter. By default it is set to Full, which means that the **IR-series** will convolve the input with the whole Impulse Response. However, if the IR is longer than 6 seconds then it will be cut to 6 seconds. This will be indicated in the IR Properties display. You can also set the Convolution length manually to be shorter than the RT60 Reverb time in order to save some CPU. This makes the Impulse Response stop at some point down the slope, creating a “gated” reverb sound. This gating effect can be avoided by using the Envelope to make the IR fade out earlier. The default Full setting will only use the amount of CPU it needs for the loaded IR, so there’s nothing to gain by manually setting it to 2 seconds for a 2 second IR.

**RATIO:** Range: .25 to 4.00; Default: 1.00

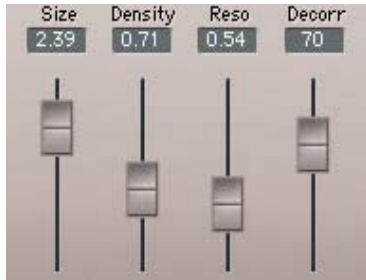
The Reverb Time Fader controls the *Reverb Time Ratio*. The resulting reverb time will be the original RT60 multiplied by the ratio.

**RT60: (Reverb Time, 60dB)** is defined as the time it takes the sound pressure level of a sound to drop by 60dB in gain once the source has been shut off. It is actually a slope factor. RT60 is a term that is commonly used in building construction and architecture to describe a room’s acoustic properties.

This field shows the reverb’s current RT60. It is the original RT60 multiplied by the Reverb Time Ratio. To determine what the original RT60 is, set the Ratio to 1.00. The original RT60 is calculated by analysis of the loaded IR.

The RT60 display is active *only* when RT60 may be computed on the IR. There is a minimum IR length (about 100 ms) for computing RT60 as well as for activating the reverb time ratio.

### Reverb Control Parameters



**SIZE:** Range: .25 to 4.00; Default: 1.00

The *Size* is a ratio of the original IR, and controls the dimensions and volume of the actual room. This defines the traveling time of sound between reflections. The *Size* value will be indicated in the IR Properties display.

**REVERB DENSITY:** Range: 0.25 to 4.00; Default: 1.00

The *Density* control is a ratio of the density in the original IR. Reducing density will usually result in a more grainy sound; increasing density will usually create a thicker and smoother sound. The outcome will vary according to the nature of the reflections in the original reverb.

**REVERB RESONANCE:** Range: .25 to 4.00; Default: 1.00

*Resonance* extends or reduces any existing resonances in the original IR.

**DECORRELATION CONTROL:** Range: 0 to 100%. Default = 0%

The *Decorr* control increases spaciousness by increasing the Decorrelation between channels.

You can use this control to make a stereo output out of a mono-recorded IR. To achieve this, load a single-channel IR into a stereo or mono → stereo plug-in component. Then, cancel the "import" dialog when the plug-in asks for the second channel.

**DRY/WET CONTROL:** Range: 0 to 100% Wet; Default: 100% Wet



This sets the balance between the “Dry” signal and the convolved signal. 50% Wet setting is the “natural” balance point.

The **IR-series** lets you work either with the recorded Direct signal, or with the Dry signal (that is, the unprocessed input to IR-1). When the Direct signal is turned off, and Dry/Wet is set to 50%, the Dry signal will have the same level and relative delay as the recorded Direct signal, so the Dry signal will replace the Direct signal. This is useful if you do not want any coloration of the Dry signal. When working with the Dry signal, it is generally recommended to turn *off* the Direct.

### **Wet Properties**



The Wet Properties section has separate on/off, gain and delay controls.

The IR-1 engine analyzes the impulse response loaded into it, and separates from it the 'Direct', 'Early Reflections', and 'Reverb Tail' portions. Each part has its own Gain and Pre-delay value. All of these parts make up the original loaded IR. So, in order to use the original IR in its pure state, all parts should be *On* and the Dry/Wet control should be at 100% Wet. To use a specific IR as a reverb, it is recommended to turn off the Direct signal and use the Wet/Dry control to reach the desired balance between source and reverb. In aux send configurations, the IR-1 is best used with the *Direct Off* and *100% Wet* (the default status).

## DIRECT SIGNAL

*Please Note: The Direct signal is not the dry signal!* The dry signal is the input signal to the plug-in.

The *Direct* signal is the direct arrival of the sound source to the listener, traveling the shortest straight-line path between them. The Direct signal is the signal that was directly emitted from the speaker and recorded by the microphone. It is the input signal convolved with the first reflection in the IR. The direct signal differs from the *Dry* signal (that is, the unprocessed input to the IR-1) since it includes the effects of air damping in the original room and the transfer function of speakers and microphones used. For purist simulation of the sampled acoustics, you should turn the Direct *On*, and set the Wet control to *fully wet*.

Since most of the time you will be using it as a reverb, the default setting of the **IR**-series has the Direct signal *Off*. The idea is that in a reverb, it is more likely to use the Dry signal rather than the Direct, especially when working in send mode. The Direct is *not* the Reverb and therefore it is *Off* by default.

Note that IRs that are sampled from Reverb devices at 100% wet *do not have a direct signal*. The **IR**-series is programmed to auto detect if a direct signal is available or not. If the imported IR has no direct signal, then the direct signal On/Off control will be disabled.

In some cases, when importing an IR that doesn't have a direct signal, the **IR-1** may mistake the first Early Reflection as a direct signal. In this event, the direct On/Off will be enabled and should be turned *On*. Otherwise, you may lose the first Early Reflection of your IR.

If you reduce the "Direct Gain" to "Off", it will only attenuate that parameter of the IR. However, if you switch the Direct to "Off" by using its "On/Off" button, the Direct will be completely cut from the beginning of the IR and the whole IR will be shortened and re-aligned so that the wet signal is aligned with the latency (the IR will start with the true direct arrival time). This enables the user to delete any unnatural pre-delay caused by the pre-roll of the impulse response.

## EARLY REFLECTIONS

The early reflections start at the first reflection after the Direct, and extend several tens of milliseconds. They maintain their discrete echo sound. The **IR-1** separates them by analysis of the IR and you can still move the ER/Tail separation point using the ER/TR-X control. Early reflections are mostly responsible for a natural-sounding room and the perception of localization and distance.



Sometimes shutting off the ERs or reducing their level vs. the tail level will leave a lot of reverb, but will sound less distant so that the signal sounds close and wet.

You can also apply different Predelay values to the ERs and Tail. Note that the division is implemented by crossfade, so sliding the ER Predelay only slightly will produce some comb filtering in the part that overlaps with the tail reflections. Turning the ERs *off* will result in the reverb going straight to the Tail part, which will give less of the perception of a room sound. This will sound less true-to-life, but that may be just what you are looking for.

## **REVERB TAIL**

The reverb *Tail* is everything after the early reflections.

The Tail is the most diffused part of the reverb. The reflections are totally dispersed, and the sound is like a smooth smearing of the input. It is the longest and most significant part of the reverb and its decay. The Reverb Tail has a *spectral damping signature* that is unique to each IR. The Tail can be used without Early reflections and you can also apply negative Predelay to have it closer in time to the source.

**DIRECT:** Range: ON/OFF; Default: Off

This switches the direct signal on or off.

**DIRECT GAIN:** Range: 0.0dB to Off; Default: 0.0dB

Direct Gain controls the level of the direct signal.

**DIRECT PREDELAY:** Range: 0ms to 500ms; Default: 0ms

This controls the amount of time that the direct signal is delayed. Direct delay is bound by the ER delay and cannot exceed it. When you apply separate gains or pre-delays to different parts of the IR, automatic cross-fades are used by the plug-in when needed.

**ER:** Range: On/Off; Default: On

This switches the Early Reflections On or Off.

**ER GAIN:** Range: 0.0dB to Off; Default: 0.0dB

ER Gain controls the level of the early reflections.

**ER PREDELAY:** Range: 0ms to 500ms; Default: 0ms

This controls the amount that the early reflections are delayed.

**TAIL:** Range: On/Off; Default: On  
Tail switches the reverb tail on or off.

**Tail Gain:** Range: 0.0dB To Off; Default 0.0dB

Tail Gain controls the level of the reverb tail.

**TAIL PREDELAY:** Range: -100ms to 500ms; Default: 0.0ms

This controls the amount of time that the reverb tail is delayed.

The Tail Predelay can be used to make the reverb Predelay longer but it is always in addition to the inherent process latency and to any Predelay embedded into the original IR file.

In non-reverse mode you can take the tail backwards in time to overlap the ER. Shifting the tail to negative Predelay values does not increase latency. Also, link mode must be *off* to shift the ER tail backwards.

**LINK:** Range: On/Off; Default: On

All elements have link controls so that all gains can move together as if in multi-selection. When an offset is made, re-linking will preserve the offset and limit the range to the extreme denominator.

#### **LATENCY:**



The IR-1 Plug-In does not declare any inherent latency. The reason for this is that the Dry path goes to the output without any actual latency. The process, however, does have some latency: 11.6 ms at 44.1 to 48kHz and 5.3 ms at 88.2 to 96kHz.

Since reverbs are commonly used in an aux send configuration and since not all host applications cannot compensate for the delay for plug-in latency on auxiliary sends/returns, we show the Wet path latency *within* the plug-in and regard it as a nominal predelay.

When working as an insert, the Dry Latency control can be used to align the dry signal to the wet signal or even advance it ahead up to 30ms.

**CPU MODE:** Range: Full CPU or Low CPU; Default: Full CPU

Full CPU

This control allows you to select between Full CPU and Low CPU. Low CPU presents an option to slightly compromise the resolution of the convolution process and saves up to 45% of the CPU cycles depending on the IR length and Sample rate. The Average savings is about 20%.

**REVERSE:** Range: ON/OFF; Default: Off.

Reverse

This will simply reverse the convolution filter to create true reverse reverbs. All Predelays become negative, but continue to display and function normally. In Reverse Mode, you can adjust the Convolution control to align the reversed IR with the dry signal. This is useful for avoiding a slow ramp up of the IR resulting from the reversed slow delay of the original IR.

## REVERB ENVELOPE



This control allows you to change the nature of the IR to achieve non-linear responses, such as faster or slower decay, gated response or special creative gain manipulations.

The control has a gain line representing Time on the X-axis and Gain on the Y-axis. The line can be edited to change over time. *Double clicking* adds and deletes points and dragging them to a time and gain coordinate will create a curve that will alter the gain of the original IR. The lines between points are always decaying exponentially and are indicated as a straight line because the vertical scale is in dB.

Double Clicking an existing point will delete it. This is useful because some hosts will not work well with the Delete button and will either delete items in the host or just ignore and not pass the event to the IR-1 Plug-In.

If you bypass the Reverb Envelope, you can make several adjustments without having to wait for each change to calculate. When you enable the Reverb Envelope, it will calculate all of the changes at once.

## Offline Calculation Indicator

Manipulating the impulse response to reflect the controls settings requires recalculation of the whole impulse response. This calculation process can take up to several seconds. While the impulse response calculation is taking place, the reverb envelope graph displays “Calculating...” and the audio is muted. After the calculations are performed, the graphs are updated and the audio un-mutes.



**ENVELOPE BYPASS:** Range: ON/OFF. Default: Off.

This control bypasses all Envelope manipulations and allows you to audition the IR without Gain Envelope manipulations. This does not bypass any of the IR manipulations such as size or density; only the Gain Envelope is bypassed.

## CLEAR ENVELOPE

This control deletes all envelope points and resets the envelope to its default unity gain.

## REVERB DAMPING



This section controls the relative reverb time of the high and low frequencies. The Reverb Damping controls allow you to modify the Low and Hi Frequency decay times of the IR relative to its overall Reverb Time.

The controls are grouped together in a line, immediately above the Frequency Response graph. From left to right you'll see the low frequency ratio and frequency value (Hz), then the high frequency ratio, and frequency value (Hz). This arrangement allows for easy modification of both Ratios at the same time, by shift-clicking them and dragging to change values.

*Please Note:* The Ratios are relative to the spectral properties of the original IR. For example, an IR (remember: it is the measured response of a real space) may have a long low-frequency decay time, but the low-frequency ratio would indicate the number “1”. This is different from synthetic digital reverbs which use these types of controls to define the frequency response of a room's reverberation. With a synthetic reverb, the low frequency ratio would have to be greater than “1”.

**LOW FREQUENCY DAMPING RATIO:** Range: 0.10 to 2.00; Default: 1.00

This control defines the ratio by which the frequencies below the Low Damping Frequency will decay in relation to IR's general Reverb Time. Values of less than 1.00 are shorter than the IR's Reverb Time, and values greater than 1.00 are longer than the IR's Reverb Time.

**LOW DAMPING FREQUENCY:** Range: 16Hz to 1600Hz; Default: 600Hz

This defines the frequency below which the Reverb time is controlled by the Low Damping Ratio Control.

**HI FREQUENCY DAMPING RATIO:** Range: 0.10 to 2.00; Default: 1.00

This defines the ratio by which the frequencies above the Hi Damp Frequency will decay in relation to the IR's Reverb Time. Values of less than 1.00 are shorter than the IR's Reverb Time, and values greater than 1.00 are longer.

**HI DAMPING FREQUENCY:** Range: 1000Hz to 21000Hz; Default: 2500Hz

This defines the frequency above which the Reverb time is controlled by the Hi Damping Ratio Control.

## **MORE ABOUT REVERB DAMPING**

**Damping** is the frequency response of a room's reverberation. This information is captured in the Impulse Response. The total decay time of a room might be 2 seconds, but high frequencies might have a decay time of 1 second. In this case, the high frequencies would have one-half the total time. This can be expressed as a Ratio. For this example, the HF ratio would be 0.5 (that is, 2 seconds multiplied by 0.5 = 1 second). A room with hard, reflective walls, such as a bathroom with tiles, can have a HF ratio greater than 1, giving longer decay times (and a very bright sound).

**HF damping** is the reduction of reverberation time at high frequencies due to absorption of high frequencies by the air and absorption in the walls. Generally speaking, this high frequency reduction in reverberation time is necessary to avoid a very splashy or sibilant sound. The degree of HF damping actually changes in real rooms according to furnishings, audience size, air temperature and humidity.

- Typical HF damping frequencies may be between 4 and 7 kHz.
- Very dry air increases HF damping. Stone walls, such as in many churches, give less HF damping than do wooden or furnished rooms.

**LF damping** is the reduction of reverberation time at low frequencies, due to low frequency transmission in the walls. Most rooms with solid walls have little low frequency damping. In general, the reason for this is that a wall with more LF transmission has a greater level of “noise nuisance” caused by bass sounds leaking out of the room and outside bass sounds leaking into the room. If noise nuisance isn’t a problem, (and it won’t be in a virtual room!) then LF damping increases the clarity of sounds, giving a clearer bass, especially on speech and orchestral music.

- Many rooms actually have a longer reverberation time in the bass than at middle frequencies. If you want some extra warmth, set the LF damping at slightly greater than 1.00x.

### Reverb EQ



The **IR-1** has a 4-band Parametric EQ section with a Low Shelf filter, 2 Bell-Shaped filters and a High Shelf filter. All 4 bands have a variable Q that controls bell bandwidth as well as overshoot resonance for the shelves.

The EQ section operates on the *Wet* signal *only*.

#### LOW SHELF FILTER

- **Gain:** Range: -24dB to +12dB; Default: 0
- **Frequency:** Range: 16Hz to 1600Hz; Default: 100Hz
- **Q:** Range: 0.71 to 1.41 Default: 1

#### LOW BELL-SHAPED FILTER

- **Gain:** Range: -24dB to +12dB; Default: 0
- **Frequency:** Range: 16Hz to 1600Hz; Default: 301 Hz
- **Q:** Range : 0.26 to 6.50; Default: 1.28

#### HIGH BELL-SHAPED FILTER

- **Gain:** -24dB to +12dB; Default: 0
- **Frequency:** Range: 1000Hz to 21000kHz; Default: 1200 Hz
- **Q:** Range : 0.26 to 6.50; Default: 1.64

## HIGH SHELF FILTER

- **Gain:** Range –24dB to +12dB; Default: 0
- **Frequency** Range: 1000Hz to 21000Hz; Default: 5005 Hz
- **Q:** Range: 0.71 to 1.41; Default: 1

**OUTPUT GAIN:** 0dB to –50dB; Default: 0dB



The Output Gain controls the overall output level of the **IR-1**.



**ER BUILDUP:** Range: 0 to 150ms; Default: 0ms

The Early Reflections Buildup allows control of the buildup slope of the early reflections. It determines a transient (abrupt) or smooth attack for the early reflections. The control is effective only when the Direct control is set to Off.

**ER/TR-X CROSSOVER:** Range  $\pm 100$ ms; Default: 0ms

This control allows you to modify the time reference point that splits the Tail Reflections (TR) from the Early Reflections (ER).

When using the IR library, this control will get its nominal value from the IR properties. When importing a user IR, the **IR-series** will attempt to define the crossover point automatically.

When you load an IR, the plug-in attempts to locate the Direct signal, Early Reflections (ER), and Tail Reflections (TR) automatically, so that they can be manipulated separately.

In natural IRs, there is no distinct delineation between ER and TR; the definition of where the best ER/TR-X crossover point is subjective. The user is allowed to modify the automatic decision of the plug-in by using the ER/TR-X control, in milliseconds, relative to the automatic crossover point.

In IRs where the direct signal is not found, the Direct Gain and Predelay become disabled.

If the direct signal has already been removed by the user before importing an IR, or if the source of the IR is not acoustically natural, and the direct path is not a part of it, then the plug-in may detect the first early-reflection as direct.

In that case, the user should make sure that the Direct On/Off button is On. In some cases, if the IR is too short or if its behavior is not acoustically natural, the plug-in may fail to detect the Direct, ER, or Tail as separate parts.

This is typical of small rooms or spaces with a reflective floor. The direct signal and the ER cannot be separated, theoretically, since the first ER arrives before the direct path response terminates. They appear as mixed in the IR. In such cases the plug-in presents the best separation, possible, but the direct would appear to include some reflections.

**CROSSTALK** : Range: 0.0 to 100.0; Default 0.0

*This control is only available in the "Efficient" components.*

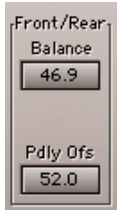
This controls how much Crosstalk is allowed between the **IR-series'** input channels. When the Crosstalk is 0.0 the process is multi-mono, convolving left to left and right to right.

When Crosstalk is at 100.0, the plug-in will convolve a summed stereo input. The dry signal is not affected, so the reverb will have elements from the left channel in the reverb of the right channel and vice versa.



## IR-360 Controls:

*These Controls are exclusive to the IR-360:*



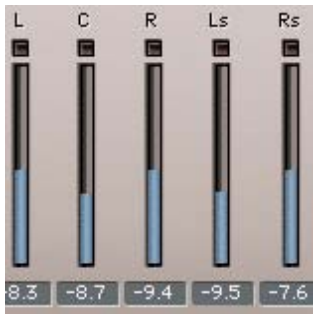
**FRONT/REAR BALANCE:** Range: Front to Rear; Default 50.0

The Front/Rear Balance control determines how much of the reverb is going to the front vs. the rear speakers. For instance, if the control is set to 65, then 65% of the Reverb will be put in the front speakers and 35% will be put in the rear speakers.

**PRE-DELAY OFFSET:** Range: 0 to 80ms; Default: 0

This control adds a time delay to the reverb on the rear speakers. It can create an effect similar to that of a digital delay plug-in.

**OUTPUT METERS VIEW:** Range: -30 to 0 dB



These meters show how much total signal (combined wet and dry) is going into each surround channel.

## Toolbar WaveSystem Controls

There is a view of the Toolbar WaveSystem Controls on the following page:



## Loading Impulse Responses

The **IR-1** load menu is the same load menu used in all WaveSystem plug-ins. Use “Open preset file...” to load a Waves setup file \*.xps. Use “Import Impulse Response from WAV...” to import IR’s either from standard .wav files or from Waves Impulse Response (.wir) files.

Under “Factory Presets” you can select “IR-1 Full Reset...” to load the default IR, “Hall - 1”, by choosing “Impulse Response and Reset”. This loads the IR and resets all plug-in settings to the factory defaults. You can also choose to load just the default IR by choosing “Impulse Response”. This leaves all other parameters unchanged.

Full Reset sets the plug-in to a flat setting that allows for auditioning the original Impulse Response, with two exceptions:

- The Direct On/Off switch will be set to “Off”. To audition pure convolution, you will need to set it to *On*, unless the control is disabled.
- The Wet/Dry slider is untouched by the Full Reset factory preset. To audition pure convolution, make sure it is set to *100% Wet*.

Note that for most studio work, you should use the IR with auto-removal of its acoustic direct portion (set Direct to *Off*) and with the dry signal replacing the acoustic direct.

The next three factory presets: Room – 1, Studio – 1, and Plate – 1 are from the Waves basic IR library. This IR Library is available with any install of the **IR-1** plug-in.

Most Factory Presets offer two load options:

- Impulse Response and Reset
- Impulse Response

The first option will load the IR and reset all parameters to their default state. The latter will load the selected preset's IR into the current setup, automatically applying the current parameters to the newly loaded IR.

Again, you can audition the *pure convolution* using the original impulse response by loading the "IR-1 Full reset" factory-preset (or the "Impulse Response and Reset" option in each IR sub-menu), and setting Wet/Dry to *100% wet* and Direct On/Off to *On*.

The Impulse Responses supplied with the **IR-series** include the product's basic IR set and two IR Libraries on CD's. This gives you almost One Gigabyte of Impulse Responses with 32-bit 96kHz quality.

The IR library is installed into the Waves Plug-Ins folder in a sub-folder called **IR1Impulses**. The Impulse files are stored in a proprietary Waves format and use the extension **.wir** (Waves Impulse Response). The **.wir** files cannot be opened in normal Audio Editors, but they can be imported into the IR-1.

### ***Saving***

The Save operation will always save all data: Control Settings, IR properties and a pointer to the IR file. This is true whether saving to a new file or to a user preset.

The impulse response data is stored in a **.wir** file, separate from the IR-1 **.xps** preset file. The **.xps** preset file contains the file name of the relevant **.wir** file. A complete preset is valid only if the related **.WIR** file is found. In the event that the **.wir** file specified in the preset is not found, a dialog box saying: "Please locate: Filename" opens. You can browse and navigate to the file, point to a different file, or cancel.

Canceling returns to the plug-in without loading a setup and will show a message "IR was not loaded". Pointing to a different file will load that file in place of the one saved with the preset, and a note on the IR Graph "Different IR loaded to preset" will be displayed.

### ***Importing Impulse Responses***

You can import IRs in .wav and .wir formats (supported bit-depths: 16, 24 and 32-floating point, with sample rates between 44.1kHz and 192kHz). The plug-in will automatically sample-rate convert the IR to the sample rate you are working in (session sample rate is 96kHz maximum).

Note: an imported .wav is *not* saved. If you want to move your session to another system, you will need to include in your backup any IRs and .wav(s) you need for the session.

When importing IRs the **IR-series** will normalize loud peaks to  $-12\text{dB}$ . If the imported IR's peak is lower than  $-12$ , then no normalization will take place, and the IR will be imported as is. The  $-12\text{dB}$  norm is applied in order to allow some headroom for the expected peak growth and clipping that will occur when convolving a hot signal with a hot IR. This may still prove to be too high, especially when importing IRs that do not have a Direct signal. In this case, it is recommended to scale the IR level using the Direct/ER/Tail gains. A recommended approach is to try to reach the point where the *perceived loudness* of the *fully wet* signal is similar to that of the *fully dry* signal.

Depending upon the channel component of the plug-in, **IR-series** will expect a certain number of channels to be imported. If you select an IR with fewer channels than expected, for example loading a stereo file to the IR-series Full Stereo (4 convolutions), the result will be another dialog box saying, "IR requires 2 more channels. Please select a file." You can create some wild sound effects by loading in completely different impulse files.

If the user chooses to cancel rather than open more files, the **IR-series** will use what it has and multiply it for the missing channels. The IR-1's graph screen will show the following message: "IR was loaded using fewer files".

### **IR Info and Properties**



	Original	Current
Convolution:	2.81s	5.57s
RT60:	1.3s	2.6s
Channels:	Bformat	10
Size:	19733	52995
Distance:	11m	NA

IR Properties are stored in the **IR-1** presets and are loaded along with the IR. Some of the properties are fixed and are shown in the top left pane (IR info) of the display. Properties that can be modified using the Plug-In's controls are displayed beneath the fixed properties section (IR properties).

The IR info pane shows:

- Name – Name of the location.
- Type – Location type i.e. Concert Hall, Theatre, Opera House, etc.
- Date – Date the location was measured.
- SR – Sample rate of the IR and the sample rate of session you are working in.
- Country – Country of the sampled location.
- City – City of the sampled location.
- Emitter type – Playback loudspeaker used in the measurement of the location.
- Receiver Type – Microphone used in the measurement of the location.
- Source Location – Where the emitter was placed in the location (i.e. Stage.)
- Mic Location – Where the microphone was recording i.e. 5<sup>th</sup> row, seat 24.
- Dimensions – The dimensions of the location in meters.
- Full Path – Shows the full path to the loaded IR file of the current setup.

The IR properties pane shows Original and Current (manipulated) values for:

- Convolution – length of the convolution filter for the current IR. The output length may differ from the input length due to one of the following reasons:
  1. Convolution user control gating (i.e. lowering the Convolution control shortens the length of the IR, thus “gating” it and saving CPU resources)
  2. Sample rate conversion upon import
  3. The imported IR is longer than 6 seconds
  4. IR manipulations (reverb-time, resonances, size, etc.).
- RT60 – the decay time value in seconds according to the RT60 standard
- Channels – the number of convolution used vs. the number channels of convolution provided in the IR file
- Size – the size of the location in cubic meters
- Distance – the distance of the source from the receiver in meters

For imported IRs, the IR properties (Distance, Size, and RT60) will remain blank if the information is not available, and will display "NA" when the value is not applicable.

## **NOTES ON “FLAT POINTS” OR UNITY GAIN SETTINGS FOR CONTROLS**

- For Reverb controls having a ratio scale of 0.25 to 4 (Reverb Time, Density, Resonance, Size) the flat point is 1.
- For Reverb controls having a percentage scale of 0% to 100% (Decorrelation, Crosstalk, Dry/Wet), the flat point is 0%.
- Predelay reverb controls (Direct Predelay, ER Predelay, Tail Predelay, ER/TR-X, Dry latency) have a flat point of 0.
- Flat point for Direct/ER/Tail gains and switches is 0dB with all switches “On”.
- For Full-CPU/Low-CPU, the flat point is Full-CPU.
- In the Gain-Envelope graph, the flat point is obtained by using the Gain Envelope
- Bypass control or by using the Gain Envelope Clear control.
- In the Damping Filters section, set the ratios to 1 for the flat point.
- In the EQ filters section, the flat point is obtained by setting all bands to Off, or by setting all the gains to 0dB.
- You can set the flat point of any control by option-clicking (Mac) or alt-clicking (PC) on that control. The only exception is the Wet/Dry control. Opt or alt-clicking the Wet/Dry control sets it to 100% (but the flat point is 0%).

## SYSTEM REQUIREMENTS

Minimal system requirements for running at least 1 instance of the Full Stereo in real time at 48kHz:

Native:

MAC G4 867, 512Mb RAM.

PC Intel - P4, Celeron 1.4 Gig and P3-700, or AMD Athlon XP, 512Mb RAM.

## NOTES CPU ABOUT OPTIMIZATION

Here are a few points to use in order to increase performance, sorted by priority:

**BUFFER SIZE:** This doesn't compromise any aspect of audio quality. Thus it is the first place the user should try to optimize system performance.

A buffer of 512 is optimal. Lower buffer sizes will increase the peak CPU consumption. Higher Buffer sizes may reduce the average CPU consumption, but the peaks will be similar.

Beware that not all hosts have friendly default buffer settings. In fact, ASIO host users use small buffers for lower latency, as do people who use virtual instruments.

### **CPU MODE SWITCH**

The CPU Mode switch at the top left allows switching to low CPU mode. This can cut peak consumption by a third. This mode gives excellent results, and the quality difference will be negligible for most applications. It is not the purist path, but it is a fine way to convolve the original sample with good accuracy.

### **COMPONENTS**

*Efficient Stereo* consumes the same amount of resources as the Mono to Stereo, and about half of what the Full Stereo consumes. While it presents a compromise on the purist natural reverb, the overall sound quality isn't compromised in any way, and it is often a more practical choice. You may like Efficient Stereo as much as the Full Stereo in most cases.

### **CONVOLUTION LENGTH**

You can set the convolution control to a shorter time. While the compromise is obvious (shortening the Reverb Tail decay time or gating it at a certain level where it would naturally continue to decay), please note that the noise floor in IR-1 is around  $-92\text{dB}$  to  $-106\text{dB}$ , and there is no compromise in sound quality.

The original convolution length shows in the IR properties display. Sometimes shortening by a bit presents negligible compromise on sound and meaningful savings in CPU, but the savings is in steps.

In other words, while not every millisecond will free some CPU, every 100ms should. You can get great results by using a shortened convolution length and using the envelope to smooth the slope so that it doesn't sound gated. On continuous material it

sounds like the rich reverb that it is, and it is a “mix-friendly” optimization.

It is important to distinguish between Reverb Time RT60 and Convolution length. The ear is more sensitive to the slope at the higher gains, so while shortening RT60 will sound like a smaller reverb, shortening the Convolution length will simply sound less linear at the lower IR gains.

### ***IR-1 HTDM***

There are two different Digidesign stream managers (HTDM Enablers), and the one that many HTDM Instruments install and use is 128 rather than 512. For better performance with IR-1 the 512 Stream Manager is required.



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