

The **Hardware Interface** is always present at the top of the rack. It is primarily used to connect Reason software to your computer's sound card. You cannot add a new Hardware Interface device, and you cannot delete it. In that, it is a "Primary" device which operates differently than the others in the rack. However, it is necessary for Reason to function. It is broken up into 4 sections: Audio Input / Output, More Audio, the Big Meter, and Advanced MIDI. Each of these 4 Panels can be opened or closed, however, one panel must always be open (whichever one you choose). Though you can fold the device closed if you wish.

At a Glance:

- Top Panel & 4 Sub-Panels
- Up to 64 Audio Outs / 64 Ins
- Sampling Input with Monitor
- Big Meter to view 2 channels (VU / PPM / Peak)
- 4 Banks / Up to 64 Channels of MIDI input for external control of Reason devices
- Device Type: I/O to sound card
- Patch Capability: No
- Cost: N/A
- Manufacturer: Propellerhead
- Web: www.propellerheads.se
- Tutorials: 60

Pros:

- Provides a lot of Audio / MIDI interface options and control. You probably will never need more than 64 tracks.
- Big Meter is a welcome addition, as is the Sampling options.

Cons:

- There are no cons to the device itself, but depending on your hardware and setup, it may be frustrating to get working correctly
- All hardware inputs have some degree of latency. Any additional connection in a chain produces some degree of latency.

Device Front

panels available from the Hardware Interface: **Audio I/O** (Audio Input / Output), **More Audio**, **Big Meter**, and **Advanced MIDI**. Click the button for each panel to show it in the rack. Note that at least one button must be selected at all times / at least one panel must be displayed at all times. When selected, a green line above each button will light up (signaling it is enabled). The **Active Audio Driver** is a non-editable field that lists the computer's audio driver (set up in the **Edit > Preferences > Audio** tab). **Input Focus** (MIDI & Audio) & **Play Focus** (MIDI Sync) deal with how Reason receives MIDI & MIDI Sync from an external source when multiple songs are open. If there is no MIDI Sync (Edit > Preferences > Advanced tab), the song on top will always have focus. If there is MIDI Sync, when both buttons are enabled for a specific song, this song will always have focus, regardless of whether it's on top or not. If one song has Input Focus enabled and another has Play Focus enabled, then the first song will receive incoming MIDI & Audio, while the other will only receive MIDI Sync. When ReWiring the song, the **Rewire** LED lights up. When MIDI Sync is sent to the song, the **MIDI Sync In** LED will light up.

Foldout Arrow: Use this to expand the Hardware Interface (and all selected panels) or fold it up into the Top Panel only.

Audio I/O: This section is subdivided into 3 areas: 1. **Sampling Input** with left / right input channels to receive audio for sampling. The **Monitor** button allows you to monitor (listen) to the sampling input in isolation. When the **Auto** button is selected, the Monitor only activates when audio is being received (sampled) via the inputs. **Level** sets the amplitude for the Monitor signal, not the sampled audio! 2. **Audio Input** has 16 Channels of Audio Input that can be sent into Reason. 3. **Audio Output** has 16 Channels of Audio Output that can be sent out of Reason. Each channel has a **Meter** that goes from Green to yellow to Red. Red indicates the audio is clipping. If it clips constantly, try lowering the level somewhere prior in the audio chain. Below each channel is a Selection Button used to focus on a specific channel pair that display metering in the Big Meter. When on, the button turns red.

More Audio: This panel expands on the Audio I/O panel by providing more Audio Inputs and Audio Outputs. The top section provides Audio Input channels 17-64, while the bottom provides Audio Output channels 17-64. All channels have corresponding Meters and select buttons, so that you can view the metering in the Big Meter panel. See the Audio I / O section for an explanation.



The Big Meter: This panel lets you meter any audio In / Out or Sampling Input. This panel is explained in greater detail on the next page.

How to Read the Meters: If you've never worked with meters before, it helps to understand how they work and what they are telling you. Without getting bogged down in too much theory, here's the bare bones of what you need to know.

All 3 meters explain the same thing, but in different ways. The VU (Volume Unit) meter is averaging a period of time, and provides you with average volume level for what you're monitoring. The PEAK meter, is providing you with an instantaneous Peak level for what you're monitoring. It may seem from reading this that the VU meter isn't accurate for measuring clipping, but this is not entirely true. While PEAK is a better indication of the actual highest point of your audio level, the VU meter is closer to what the human ear perceives as loudness. Therefore, it has value when metering your audio.

For example, you may have a sound without a loud peak, but with a very loud average volume (for example, long sustained pad sounds). This can lead to audio that seems too loud when you hear it, but still doesn't clip. Or you may have a sound that has very high peaks, but seems too soft to your ears (vocals and transients can easily fall into this category). These two meters can help guide you to ensure you're on the right path.

Patch Creation suggestions: Generally speaking, I adjust most of the amplitude levels during patch creation; as I go, since most changes you make to a patch will affect amplitude in some way. So I'm forever adjusting and tweaking the levels as I create the patch. However, I tend to work from the inside out. For example, I'll adjust patch settings inside the Voice and Global sections of Thor first, making sure those settings are correct. Then I may make an adjustment to the main level of the Thor patch via the big Volume knob after the patch settings are in place. In addition, I usually (but not always) work by following the signal flow. For instance, I'll first adjust the drive setting of Filter 1 before adjusting the Gain setting of the Amp section inside Thor. This helps to get the levels correct, and also helps to understand the signal flow of the patch you're creating.

Advanced MIDI Device: Used to select a **MIDI Bus** and **MIDI Channel** for your MIDI Input. If you want to control Reason devices via external sequencer (as opposed to the Reason Sequencer). To use this, you will have to do 3 things: 1. Go into **Edit > Preferences > Advanced**, and select an input for one, some, or all MIDI Busses (A - D). This is where you select the External Sequencer. 2. Select a Bus (A - D buttons) on the Advanced MIDI Device panel. 3. Select a Reason Rack device to control from the **Channel** drop-down. Each bus has 16 channels (64 Channels total). You can control up to 64 Reason devices with external sequencer(s). When a channel is in use, the LED to the left of the channel label lights. The label above each Channel shows the name of the device being controlled. To remove a device, click the drop-down and select "Disconnect."

Device Front (The Big Meter)

VU Offset: VU Offset allows you to change the VU scale in 2 dBFS steps, from -20 dBFS to 0 dBFS. Usually, and practically, this does not ever need to be changed. I would leave it just as it is. If you do change it, use Ctrl + click on the knob to get it "reset" to its default. Note also that this is merely a display preference, and you will only see the changes if the meter is in VU or VU+PEAK modes.

Big Meter Display: The Big Meter display shows the meter itself using LED lights. Green is well within limits, yellow is still within limits, but cautionary. And Red indicates clipping is occurring. In this mode, VU legend is listed along the top and PEAK along the bottom. However, the LED displays the Peak as a single set of LED lights at the farthest right of the meter, while the VU is indicated by the LED lights extending out from the left. As shown here, the VU is set to -4 dBFS, and PEAK at -8 dBFS.

Clip Indicator: The Clip Indicator will light up in Red if your Audio clips. Once the audio clips, it will remain red until you use the Reset Button.

Reset Button: This button will "reset" the peak or clipping. Note that once the audio clips, it will not reset back unless you press this button.

Mode Selection: Lets you select different Metering Options, which are: VU, PPM, PEAK, VU+PEAK, and PPM+PEAK. Personally, I find the VU+PEAK mode is the best one to use when constructing sounds or mixing. This way you have access to both modes visually at the same time.

Peak Hold: In PPM and PEAK mode, the Peak Hold determines how long the maximum peak is displayed in the Big Meter display area. Personally, I would leave this at 5 seconds. Otherwise, the peak indicator remains indefinitely. 5 seconds allows for the Peak to "reset" itself.



Channel Select Rotary: This rotary selects the appropriate Audio Channel from the Hardware Interface, so that it shows up in the metering on the Big Meter Display. Alternately, you can select the appropriate Channel by clicking the Channel Select Button (small square below the corresponding Audio Input / Output Channel pair).

VU Meter: Indicates the Audio VU (in this example, -4 dBFS)

Peak Meter: Indicates the Audio PEAK (in this example, -8 dBFS).

Note about Peak & VU values: Usually, you want your audio to PEAK around -6 dBFS or -8 to -4 VU dBFS. This is merely a practical guideline as different sounds will require different settings. Your ears should be the ultimate arbiter of the sound level you are seeking.

Tips & Tricks

Setting up the Big Meter to Monitor your Sound:

1. First, click on the **Audio I/O** and **Big Meter** buttons in the Hardware Interface. This opens both, and expands the Big Meter below the Audio I/O.
2. Next, click the little **square** below Audio Output 1/2 (or whichever audio output you want to monitor). Alternately, use the Big Meter **Channel** rotary to turn the Big Meter's focus on the proper output.
3. Using the Big Meter **Mode** button, adjust until you have **VU+PEAK** mode selected. Now you can monitor your audio using both meters. VU is listed in light blue along the top of the Big Meter scale, and PEAK is listed in orange along the bottom. When viewing the audio output LED lights on the Big Meter, note that the VU LEDs are always to the left of the PEAK indicators. And the Peak indicators are displayed as two LED lights to the farthest right. It takes a little getting used to, but it's a valuable way to monitor your sound and ensure there is no clipping.
4. Ensure the **Peak Hold** is set to **Five Sec** (to indicate that the PEAK setting readjusts after every 5 seconds).

A Word about Amplitude: While I know everyone has their own opinion on what the "correct" level of a sound should be, unless I'm designing something solely for myself, and solely for a specific reason, I generally open up the Big Meter in Reason's Hardware Interface, set it to VU+PEAK mode, and ensure the levels of your audio do not exceed approximately -4 dBFS (VU) or -6 dBFS (Peak). This is dependent on a lot of factors of course, and the meters are only an aid to help you. They are not meant to replace your ears. If the audio sounds right, then it IS right. But one thing is for sure: you don't want the meters going in the red. This means your signal is clipping, and should be avoided. Better to be too cautious than too aggressive in your sounds.

Note that clipping occurs when you see the Meter go into the red. With Reason's default settings, this is a value of 0 dBFS (VU) or 0 dBFS (PEAK). Ensure that your audio signal is always consistently below these values, no matter what performance parameters are used in the device, and no matter what combination of these performance parameters are used (more on this below).

It's also important to understand that the amplitude of your sound revolves around several interrelated aspects: The Level of the sound (sometimes multiple level stages along the signal route), how many Oscillators are being used, the Polyphony setting, The Decay / Sustain setting of the Amp Envelope, the Gain setting of the Amp section, and the Filter settings, to name just a few. All these settings factor into the loudness of your sounds, and whether or not they are clipping. In fact, everything you do with your sound in Reason will affect the amplitude one way or another. That's just the nature of the beast.

Testing Amplitude During Performance: I categorize the Polyphony, Envelopes, Velocity, Aftertouch, and Key Scaling settings in this group of volume control. I generally know in advance if I'm going to be creating a patch that is monophonic versus Polyphonic. And if the patch is monophonic, I set the volume levels while playing a single note at maximum velocity. If the patch is polyphonic, I'll adjust amplitude settings based on playing a 3- or 4-key chord at maximum velocity. It's also important to play the patch up at high registers

and down at lower registers to see if the volume is consistent across the entire keyboard. Often times, a synth that has a proper volume in the C2 and C3 range, can be overly bright at higher registers (which can increase the volume), or can overload the bottom end amplification-wise. Without getting too deep into this conversation, just keep in mind that the number of voices (polyphony) affects amplitude, and should be tested accordingly.

The envelopes you set up for the Oscillators are also important. The sound can start off low, and then build up over time to peak much higher than your maximum. So if you're creating a long drawn-out pad sound which develops over time, be sure to listen and monitor the sound played over the entire duration of the envelope and check that big meter to make sure you don't go too high.

Volume Consistency: There's also one other reason to pay close attention to the volume level of your sounds: Consistency. If you're designing sounds for a ReFill, for example, you don't want one sound to be extremely loud and another to be extremely soft. You want to provide some degree of consistency across your patches, and indeed your ReFill.

For me, a good consistency is when the patches or sounds are all somewhat close to each other in volume, both average volume and without causing any clipping. Of course, this also depends on the type of instrument you're creating, the frequency range, the rise and fall of the Amp Envelope, etc. But you still want them all to be somewhat consistent relative to each other.

Note: A good test to see how accurate your ears are is to turn off the big meter entirely, and hide all the metering (put a piece of tape over the metering displays on your screen if you have to). Design a few different sounds or listen to your tracks without the aid of any metering. Then go back through the sounds and see how close you came to getting them all set to consistent levels.

In summary, it's a good idea to think of volume and amplitude in your patches as a system of pulleys and levers that need to equalize at the outcome. If you pull one lever up, you generally need to pull another level somewhere in the audio chain down. Kind of like balancing a budget. If you spend more in one area, you have to save in another. Thinking in these terms will help you come out with a better patch in the end, which is consistent with your other patches. When you produce a ReFill, getting the levels even across the board is very important, and it's something that will make your ReFills better in the long run. So take a minute to explore how volume is affected when you change things around in your patch, and learn to compensate when you raise that drive on that filter.

The Last Stop: The Hardware Interface is the last stop before the audio is output to your sound card. And it has no level or amplitude adjustment of its own. If you see your audio clipping here via the output meters, you'll need to reduce the volume before the audio goes into this device. Personally, I always try to reduce the audio levels on the source instrument, as all Instruments have level controls (the External MIDI Instrument being the only exception - you'll have to reduce the level being input if you record your audio to an Audio Track which can be controlled on your Audio Interface). You can also control volume at the Mixer stage (after the Instrument, but before the Hardware Interface). Since the Hardware Interface is the last stop in the signal chain before output, clipping here is the most important clip meter to look at and correct.

Device Back

Sampling Input: If sampling in Mono, use the Left Sampling Input only. If sampling in Stereo, use both Left / Right Sampling Inputs. If you are sampling an external instrument, like a Guitar or external synth, ensure it is set up through your audio interface and use the available "From Audio Input" channel sent into the Sampling Input. If you are sampling an Instrument within Reason, you have a few options. To use the final output of all instruments, use the Master Section left / right Master Out into the left / right Sampling Inputs. To use a specific device, send that device's main left / right output into the left / right Sampling Inputs.

Active / Inactive Channel Indicator: On the front of the device, just below the channel and above the "Select Channel Button" is a tiny LED. When red, it means the channel has a cable connection, but it is unavailable (to make a channel available you must set it up in the "Edit > Preferences > Audio" tab). When yellow, it means the channel is available, but there are no cables connected. When green, it means the Channel is available and in use; an audio cable is connected. The rear panel also shows this same indicator just above the top row (left) channel jacks, and just below the bottom row (right) channel jacks.

Panel Buttons: The same panel buttons that you find on the front of the device are found on the back. Audio I/O, More Audio, Big Meter, and Adv. MIDI (Advanced MIDI). This way you can close or open each of these panels from the back of the device.

From Audio Input / To Audio Output: These inputs mimic the front panel. You can connect up to 64 Stereo pair of jacks that can be used for the Audio Input from your audio interface. Technically, these jacks will pull audio from the audio that is going through your computer's soundcard. The "To Audio Outputs" also mimic the front panel. And this is where you set the output from Reason to your soundcard and out of your computer. You have up to 64 audio outputs that can be sourced. Technically, the output isn't sent out of your computer, unless you set it up that way. In most cases, you'll probably be converting the Reason output to a wav file that is stored somewhere on your computer.

Big Meter / Advanced Panels: These two panels have no connections, however, there is an explanation of how the "Channel LED Indicators" operate on the Audio Input / Output and More Audio panels above. A nice use of some empty space.



Default Connections: The image here shows the basic default connections for the Hardware Interface and Master Section. The Master section left / right Output is sent to the first available "To Audio Output" left / right channels (channels 1 & 2 out). In addition, the first "From Audio Input" left / right channels (channels 1 & 2 in) is sent to the left / right Sampling Input. Even if the Audio Inputs are unavailable (as indicated here with the red LED indicators). Sampling Input, on the other hand, is always available, as it requires no special audio setup within Reason to sample. This Sampling setup is the one you would normally use if you are sampling an external instrument sent into Reason via your audio interface.

To quickly set up your system to sample the entire song with all effects intact, move the 1 / 2 "To Audio Output" into the left / right Sampling Input (both on the Audio I/O panel). Ensure Monitoring is turned on at the front of the Audio I/O panel, so that you hear what's being sampled. Move it back to the default connections once you're finished.

Alternately, if you want to sample frequently from your entire song, create a Spider Audio Splitter and split the signal going from the Master Section's left / right Master Out. Send one split into the 1 / 2 "To Audio Output" and the other split into the Sampling Input. Now, you can have the same signal that's output in the default configuration, and another sent to the Sampling Input. In this scenario, you won't need to Monitor the signal going into the Sampling input, so you can turn that off.

