Pulsar Mu

User Manual

www.pulsar.audio

Table of contents

Introduction

This manual describes the features and operation of the Pulsar Mu effect processor. To be sure you understand how to use your plugin and understand all its subtleties, please read it completely.

The information contained in this manual is believed to be correct at the time of publication. However, if an error has unfortunately crept into its contents, please let us know.

IMPORTANT: The prolonged use of amplified instruments, speakers or headphones may cause permanent hearing loss. Ensure you monitor your exposure level, and take regular breaks. In case of tinnitus or suspected hearing loss, please consult an ENT specialist.

Welcome

Our experience

Thank you for choosing Pulsar Audio quality!

With more than 15 years' experience in plugin development for the biggest names in the industry, we decided to create Pulsar Audio to push the quality requirements of our products even further.

For each product, our quest for excellence requires us never to rest on our technical achievements, and to expand our knowledge ever further.

Sound and science

With solid expertise in audio signal processing, but also in electronics, sound techniques and music practice, we take great care in modeling all the small details and imperfections of analog equipment that make the difference between a « mathematical » exact sounding algorithm and a rich, living and musical processing, and we produce this famous « 3rd dimension » sound so much sought after.

In addition, our close collaboration with music production professionals requires us to be rigorous in order to produce professional quality tools.

Our user interfaces

The user interface of a plugin is the link between the creative drive and the technical implementation; it must therefore be clear, intuitive, and as pleasant as possible to use. We take great care to create the most beautiful and fluid interfaces possible, with an emphasis on intuitiveness.

The search for the right equipment

Rarely do you find two analog machines that sound exactly the same. It is therefore important, when developing an emulation, to carefully choose the hardware units to be used as models. We only use units in perfect condition and measure them with the best recording equipment.

A final word

We hope you will enjoy this plugin as much as we enjoyed creating it. Be sure to visit our website www.pulsar.audio and find out about updates, new products, tips and other resources. There, you will also be able to contact us to ask for help or simply to tell us about your experience!

The Pulsar Team

« Variable-bias » compressors

The term « variable-bias » refers to a dynamic compression technology that appeared in the 1950s (for example, the mythical Fairchild 670, originally designed for the mastering vinyl process, but very quickly adopted in the studio), which consists in varying the polarization of a tube to control the gain. The compression ratio is variable, and also depends on these parameters.

The result is compression with a slower attack than a FET or VCA compressor, but an extremely musical behavior, making it an exceptional bus compressor, both for a drum bus and for an entire mix. This type of compressor is also present in the vast majority of mastering studios.

In addition, its design based on transformers and tubes generates a slight harmonic distortion that brings subtle presence and warmth, and this famous « glue » effect that makes all the sounds sit musically in the mix.

The Pulsar Mu will offer you:

- The sound of a legendary machine, present in studios all over the world
- A musical bus compression
- Unparalleled warmth and presence
- The advantages of « organic » analog sound as well as the advantages of digital sound (presets, parameter automation, etc.)

Quick start

Installation

Pulsar Mu is available as a plugin in VST2, VST3, AU and AAX formats for use with all major DAW software such as Live, Cubase, Logic, Pro Tools, etc.

Installation from the supplied installer is automatic. The installer takes care of copying the different plugins as well as presets, manual, etc. into the appropriate locations.

Note: If you are using the VST2 format in Windows, you will be asked by the installer to specify the installation folders for the 32-bit and 64-bit VST2 plugins respectively. The paths that seem most appropriate for your computer will be recommended by default, but we advise you to check them before completing the installation. If the plugin is not installed in the same folder as your other possible plugins, your DAW software may not detect it.

Activation

All our plugins are protected by PACE's iLok system. For correct operation, we recommend you ensure that you have the latest version of the « iLok License Manager » software, available for free download at www.ilok.com .

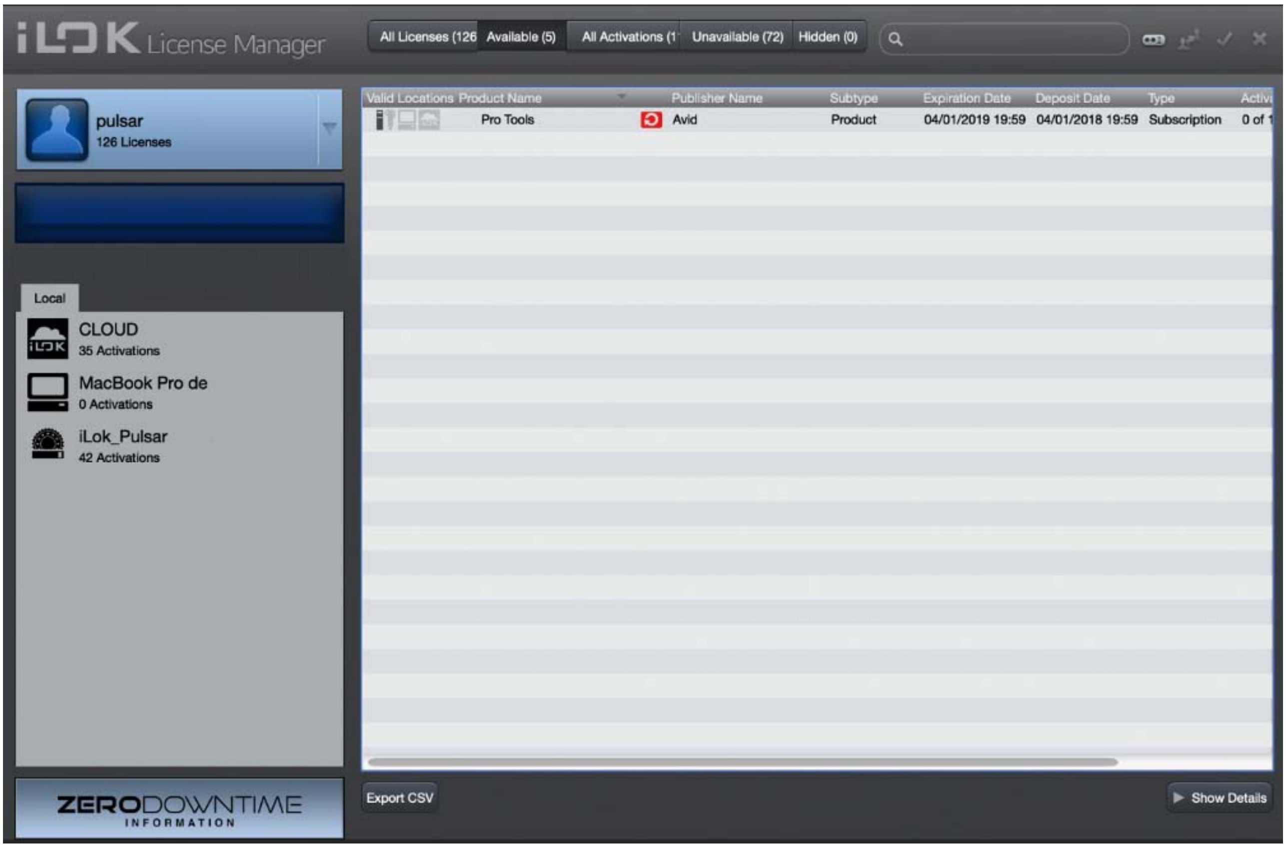
You can choose between three activation methods:

- Activation on a hardware USB dongle such as iLok 2 or iLok 3, which will enable you to use your plugin on several machines (you can order a dongle online at www.ilok.com or buy it from your music retailer)
- iLok Cloud activation which will enable you to use your plugin on several machines but requires a permanent internet connection
- Machine activation, which does not require a dongle or a permanent internet connection, but only activates your plugin on one machine

Important: If you choose the iLok Cloud system, you have to open a Cloud session on your computer by going to the « File > Open Cloud Session » menu of your iLok License Manager. If you choose an iLok 2 or 3 dongle, you have to connect it to your computer before any operation.

When you purchase your software, you will receive:

- Either a license deposited directly onto your iLok account. Just go to the « Available » tab and drag it to the destination of your choice (here CLOUD for a cloud license, or iLok_Pulsar for an iLok 2 or 3 dongle)
- Or an activation code. Simply paste it into the « Licenses > Redeem Activation Code » menu to receive the license on your account, and drop it off at the destination of your choice (CLOUD or iLok 2 or 3 dongle)



iLok License Manager's "available" license tab

First Steps

Load the Mu on the track of your choice in your DAW. The base preset is a good starting point. You can now:

- Adjust the « Dual Input » gain in order to obtain the Gain Reduction suggested in the preset, for example -3 dB for the base preset. The amount of Gain Reduction is displayed in the visualization rack at the top.
- Adjust the amount of compressed sound with the « Mix » knob.
- Adjust the release attack times using the « Attack » and « Release » knobs

You can also review the available factory presets to find inspiration quickly, without getting into technical considerations.

The user interface

The user interface

The user interface consists of 2 separate panels:

- The toolbar, common to all Pulsar Audio plug-ins (top)
- The control panel, specific to the plug-in. This panel consists of 3 racks: the visualization rack, the compressor control rack and the advanced detection functions rack

Use of parameter controls

The parameter control knobs have several modes of use:

- The normal editing mode (use a classic mouse drag, or the mouse wheel)
- The fine editing mode (hold the Ctrl or Cmd key while dragging or while using the mouse wheel, or drag with the right mouse button)
- The « reset to default » action (double-click, or click while holding the Alt key)
- The « menu » action (right-click, or click while holding the Ctrl key)
- Only for some controls, the alternate edition mode (hold Shift while dragging), which can have various functions, for example to temporarily link two parameters

Parameter locking

It is possible to lock certain parameters, so that they are not changed when loading a preset. For example, one possible use of this feature is to set the input and output gains of a compressor to achieve the desired amount of gain reduction, lock these parameters, and then scroll throught the list of factory presets to find the most appropriate tone.

Locking the Dual Input knob

To lock a control, right-click it with the mouse, or click while holding down the Ctrl key on the keyboard. If the control can be locked, a menu will appear offering to lock it. When a parameter is locked, a small padlock icon appears next to the control.

Control surface and multi-channel parameter edition

If you are using a control surface such as AVID S1/S6, Mackie HUI, so on. to control your plug-in, and in the case of a plug-in whose L/R or M/S channels can be controlled independently:

- When the "link" option for the controls of the two channels is deactivated, the reading/writing of automations and the control of parameters by the surface function "normally" (each control is controlled independently, as expected).
- When the link option is activated, only channel A parameters are used (which corresponds to the Left or Mid channel, depending on the stereo mode selected). By controlling channel A parameters from the control surface, or reading automations from channel A, the link automatically synchronizes channel B parameters. **Automations on channel B parameters are ignored, as are changes to channel B parameters from control surface** . Channel B automations are not written.
- **Caution:** in Pro Tools, automations are written to both channels A and B, and even with the link option enabled, playing an automation on A does not synchronize channel B (it is assumed that channel B's automations have been recorded at the time of writing, and that channel B's automation tracks are being replayed independently of channel A's).

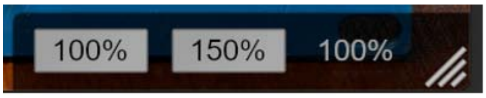
Using the GUI resize control

Located at the bottom right of the interface of all Pulsar Audio plugins, this control allows you to resize the plugin's interface to your liking. It comes in the form of three lines, like a classic resizing handle:

Resizing handle

Note that in some DAWs, this resizing can be problematic, depending on how the DAW developer has designed its windowing.

It is also possible, by clicking in the corner, to open a small popup window with buttons offering a choice of fixed size resizing (100% - 150%):



Resizing window

The Toolbar

Located at the top of the plugin interface, it contains all the functions relating to parameters, presets, communication with Pulsar Audio, etc.

The toolbar

Undo / Redo

The two arrow buttons on the left of the toolbar have the function Undo and Redo, i.e. respectively the cancellation and restoration of the last action. All parameter changes and more generally the state of the plugin are stored in a history. You can click on « Undo » at any time to return to the previous state (or to the nth previous state) and on « Redo » to return to the current state.

Note: a right-click on one of these buttons gives access to the list of stored operations.

Undo / redo buttons

Preset Selection

The preset selection area, located in the center of the bar, allows you to:

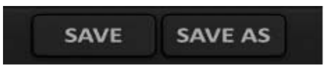
- Read the name of the current preset. If an asterisk appears after the preset name, it means that the state of the plugin no longer matches the saved preset
- Select a preset from the list of available presets, arranged in sub-banks
- Delete the current preset (« Delete Preset » option)
- Rename or move a preset to another sub-bank (« Move / Rename Preset » option)
- Set the current preset as the one that will be loaded by default when creating a new instance of the plugin (« Set This Preset As Default » option)
- Open the presets directory. This can be handy for making backups of your preset files and restoring them. Note that renaming and reorganizing presets must be done from the plugin menu, not by using your system's file explorer.
- Restore factory presets. This will also overwrite any changes you have made to your factory presets
- Quickly navigate between the presets to find inspiration, using the left and right arrows

The preset selection area

Save / Save As

The Save button saves the current preset.

The Save As button saves the current state of the plugin under a new preset name.



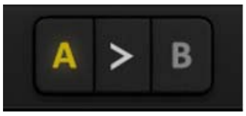
Save and Save As buttons

A / B

This section allows you to compare 2 different states of the plugin, or 2 different presets. Slots A and B, accessible through these 2 buttons, represent 2 completely independent states.

For example, when state A is active, you can load a preset and/or make settings from the interface, then click on button B; then load another preset and/or make other settings; buttons A and B now allow you to quickly switch between the two states and easily compare the 2 presets or sets of settings.

It is also possible to copy the state A to B or vice versa using the > or < buttons located between A and B.



A, B and Copy buttons

Menu Button

The button located on the far right of the bar encompasses various options.

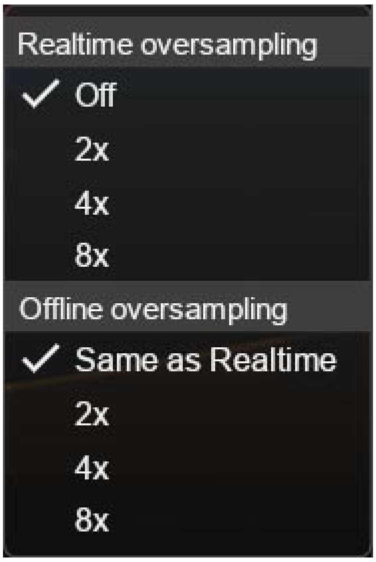

The Menu button

Oversampling settings

The first menu item is used to set the oversampling. Oversampling allows the sound to be processed at a higher sampling rate within the plugin, in return for higher latency and CPU consumption. Oversampling is disabled by default, as all Pulsar Audio products use advanced technologies that allow in most cases to process the sound without oversampling, with no compromise on quality. This makes oversampling useful mainly when you saturate a lot.

The maximum available oversampling rate is not the same in all Pulsar Audio plug-ins and depends on a trade-off between the need for oversampling and the CPU consumption induced by oversampling in this plug-in.

Please note that Pulsar Audio products use very high-quality linear phase upsampling and downsampling filters. This means that the x2 oversampling will generally be of higher quality than the x2 setting in a competitor's product, but will also be more CPU intensive.



Oversampling options

The "Offline oversampling" option allows you to choose an oversampling setting for final rendering (and other non-real-time processing) independent of the setting applied in real time. This enables to reduce the CPU consumption during the use of the plugin, while having the best quality during the final rendering.

Disable Static Noise option

All analog equipment introduces a hiss, mainly caused by thermal noise in the electronic components, the amplitude of which differs from one model to another.

In some Pulsar plugins, we assumed that modeling this noise was appropriate, although at a lower level than in real life (often around -90 dBFS), because it contributes slightly to the character of the original device.

In some cases (if the output of the plugin is strongly amplified), this noise can become audible and undesirable, so it is possible to deactivate it using the "Disable Static Noise" option.

Other options

Other functions accessible through this menu are:

- Enabling / disabling the help balloons
- Access to the website
- Access to social media
- Access to communication with technical support
- Link to this user manual

The control panel

The Mu's control panel

The Mu's control panel is inspired by a famous hardware machine, with some additions to expand the sound palette.

This panel consists of 3 racks:

- The visualization rack
- The compressor control rack
- The rack of advanced detection functions

The visualization rack

This rack allows you to view important levels such as the GR level (gain reduction).

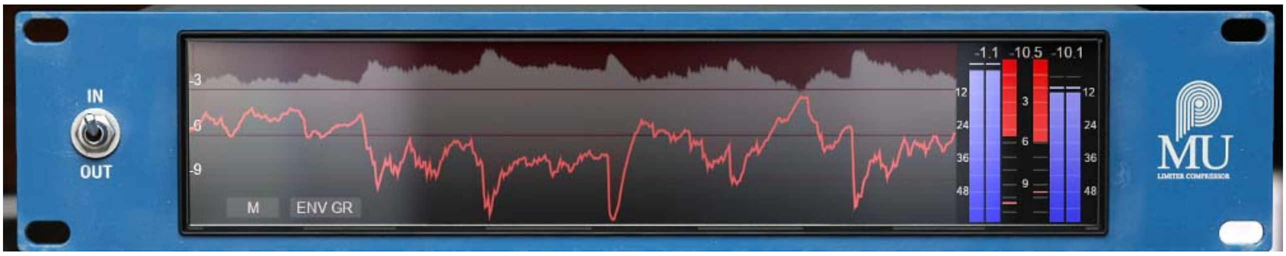
By clicking on the banner, you can access 2 different types of visualization: Classic VU meters, or a more modern visualization of the GR over time.

In « classic » mode, the 2 displays indicate the gain reduction in dB.

« Classic » mode

In « modern » mode you have access to more information:

- Gain reduction over time (screen with scrolling)
- 3 stereo bar graphs on the right side indicating the input level (in dBFS), gain reduction (in dB) and output level (in dBFS)



« Modern » mode

Note: The recommended gain reduction for this compressor is between 2 and 6 dB in Comp mode, and 2 and 4 dB in Limit mode.

Control rack

This rack contains the main controls (switches and knobs) useful for setting up the actual dynamic compression.

Bypass

The switch on the left of the display panel is a bypass, it allows you to activate and deactivate the effect.

Bypass switch

Dual Input

This knob is used to adjust the amount of signal at the compressor’s input. The higher this input level is, the more audible the color of the tubes will be.

Dual Input knob

Output

This knob is used to adjust the output level after compression (to compensate for the input gain, and the gain reduction due to compression).

Output knob

Threshold

This knob sets the compression trigger threshold. The lower it is, the more the machine will compress the signal.

Threshold knob

Note: the compression ratio also depends on this setting. The lower the Threshold, the higher the ratio.

Attack

This is the knob for adjusting the compressor attack time. The shorter the attack time, the more the compressor will be triggered on percussive elements. This time is between about 10 milliseconds (Fast) and 100 milliseconds (Slow).

Attack knob

Note: the actual attack time also depends on the Release setting.

Release

This knob is used to adjust the compressor release time. The shorter the time, the more audible the pumping effect will be. This time is between about 70 milliseconds (Fast) and 1,8 seconds (Slow).

This parameter can be set to any of the 1-2-3-4-5 positions, or to any value in between using the fine edit mode (by holding down the Ctrl or Cmd key).

Release knob

Note: The actual release time also depends on the Attack setting.

COMP / LIMIT switch

This switch is used to adjust the compressor ratio (« quantity » of compression):

- COMP mode: ratio of about 1.5:1 – light compressor
- LIMIT mode: ratio of about 4:1 – limiter

The effective ratio can grow to much higher levels in case of large gain reduction.

COMP / LIMIT switch

It should be noted that due to the electronic structure of this compressor, the ratio is not fixed; it depends on the position of this switch, but also on the input gain and threshold levels.

Link Controls

This switch links the settings of the two channels. If enabled (upwards) each setting that is adjusted on channel 1 is also adjusted on channel 2 (and vice versa).

Link Controls switch

Link Sidechain

This switch is used to connect the sidechains (detection circuits) of the 2 channels. If it is activated (upwards), the 2 detection channels are mixed, and the gain reduction is the same for both channels. This enables, for example, the stereo coherence of a master bus to be kept in case of high compression.

Link Sidechain switch

L-R / M-S

This switch selects the stereo management mode: traditional Left/Right stereo, or Mid/Side stereo. In the latter case, the left channel is the Mid (Center), and the right channel is the Side (Sides).

L-R / M-S switch

Mix

This knob is used to crossfade between the compressed signal (MAX) and the original signal (MIN).

Mix knob

The rack of advanced detection functions

Advanced detection functions

This control panel allows you to adjust various settings for the detection signal, which is the (non-audible) signal used to calculate the gain reduction.

Source (INT / EXT)

This switch is used to select the detection source. In « INT » mode, the compressor operates in the usual way, i.e. using the audio input as a detection source. If « EXT » is selected, an external audio signal will be used as the detection source. It will then be essential to assign a sidechain channel (external detection) in your DAW – Please refer to your DAW's manual for more details on its routing.

Look (AHEAD / BEHIND)

This knob allows you to add a positive or negative delay in the detection channel. The more the knob is turned towards « AHEAD », the more the detection will be done in advance compared to the compressed signal, thus allowing very fast transients to be compressed. The more we go to « BEHIND », the more we let the transients pass, so for example they can be limited with a limiter or a maximizer.

The switch above the knob allows to deactivate the whole « AHEAD » part. This may be desirable when zero-latency operation is required. Indeed, in look-ahead mode (i.e. when the knob is set to values between -5 ms and 0 ms), the pplugin introduces exactly 5 ms of latency. This latency is automatically compensated by your DAW, so you won't have to worry about it in most cases. However, it may be necessary to remove it in tracking or live situations.

Thus, when this switch is turned off, the left part of the « Look » knob will be disabled. The right part, corresponding to the « BEHIND » function, remains usable because it does not need latency to work.

Sidechain EQ

This section allows you to filter the detection signal using a simple parametric equalizer:

- The high-pass filter (knob on the left) will allow you, for example, to not trigger the compressor on a bass drum
- The high-pitched band centered at 3kHz (knob on the right) will reduce compression on cymbals to maintain a brilliant mix (or on the contrary will increase this compression)
- The bell filter (the two middle knobs) will allow you to selectively trigger the compressor more or less on a part of the spectrum, between 50 Hz and 4 kHz (a snare drum or a voice for example)

Listen

These buttons allow to listen to:

- The compressor output (OUTPUT)
- The detection signal (SIDECHAIN), useful to refine the EQ settings
- The L or M channel (depending on stereo mode)
- The R or S channel (depending on stereo mode)

Minimum Configuration

This plugin is compatible with all major sequencers on the market (Cubase, Nuendo, Pro Tools, Logic Pro, FL Studio, Ableton Live, Bitwig, Digital Performer, Studio One, Reaper, Adobe Audition...)

Available formats:

- VST 2.4 (Windows: 32/64-bit, Mac: 64-bit)
- VST 3 (Windows: 32/64-bit, Mac: 64-bit)
- AAX (Windows: 64-bit, Mac: 64-bit)
- Audio Unit (Mac: 64-bit).

Windows

- CPU: Intel Core i3 / i5 / i7 / Xeon
- Memory: 4 GB RAM / 1 GB free disk space
- Operating system: Windows 7 and higher
- Screen resolution: minimum 1024×768 / recommended 1280×1024 or 1600×1024

MacOS

- CPU: Intel Core i3 / i5 / i7 / Xeon / Apple Silicon (M1, M2, etc.)
- Memory: 4 GB RAM / 1 GB free disk space
- Operating system: 10.9 and higher
- Screen resolution: minimum 1024×768 / recommended 1280×1024 or 1600×1024

License agreement

This license agreement concerns and describes your rights and the conditions under which you may use your Pulsar Audio software. We recommend that you read the entire agreement. By accepting the present agreement or by using Pulsar Audio software, you accept all these conditions.

This license agreement applies to all Pulsar Audio software, plugins and programs that you may use during the evaluation period and/or thereafter subject to the acquisition of a license, for any version, update, or supplement.

License

The software is not sold to you: you are granted a license to use it. You are allowed to install and use the software on as many machines as you wish. You may not rent, lend, or license this software. You may not alter, decompile, disassemble, or reverse engineer this software.

Updates

This license gives you the right to all minor updates (e. g. 1.1 to 1.2), but excludes major versions (e. g. 1.x to 2.x).

License transfer

You may transfer all your rights to use the Software to another person provided that you transfer this Agreement and the Software to that other person; and that the recipient accepts the terms and conditions of this Agreement and any other provisions pursuant to which you have acquired a valid license to use this Software.

Activation

Pulsar Audio will not be held responsible for any failure to activate PACE's iLok protection system / license.

Trial

Pulsar Audio offers a 14-day trial license, starting at the time of transfer of the license to an iLok key. After expiration, the plugin can no longer be used, and in the event that no permanent license is acquired, it must be deleted.

Third-Party Software

VST is a registered trademark of Steinberg Media Technologies GmbH. AAX is a registered trademark of Avid Technology, Inc. Audio Units is a registered trademark of Apple Computer, Inc.

Disclaimer

Neither Pulsar Audio nor anyone else who has been involved in the creation, production, or delivery of this product shall be liable for any direct, indirect, consequential, or incidental damages arising out of the use or inability to use this product (including, without limitation, damages for loss of business profits, business interruption, loss of business information and the like) even if Pulsar Audio has previously been advised of the possibility of such damages. Some jurisdictions do not allow limitations on the duration of an implied warranty or limitation of incidental or consequential damages, in which case the above limitations or exclusions may not apply to you. This warranty gives you specific legal rights, and you may also have other rights that vary from jurisdiction to jurisdiction.