

**NEWFANGLED AUDIO**

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Elevate proudly uses the Eigen C++ library, which is available at <http://eigen.tuxfamily.org>.

Elevate also proudly uses the PFFFT library, which is available at <https://bitbucket.org/jpommier/pffft>.

Newfangled Audio

[www.newfangledaudio.com](http://www.newfangledaudio.com)

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## Chapter 1

# Newfangled Audio Elevate



Newfangled Audio's Elevate plug-in is a professional audio grade mastering limiter and audio maximizer designed to transparently increase the loudness of your audio signal while maintaining or improving the dynamic perception and tonal balance of your mix. The Elevate limiter uses several new technologies to achieve this goal, but the central idea is to use an auditory filter bank to separate the audio into frequency bands similar to the frequency curve in the human ear, it then uses adaptive algorithms to recombine these frequency bands to give you maximum flexibility while avoiding artifacts like distortion, pumping, or shifts in the stereo image or tonal balance. In addition to the adaptive

algorithms the MULTIBAND parameter pages allow you to tweak this mixture, giving you control over the tonal balance and transients even after the final limiting stage. Finally, the UI has been designed to give you fast and easy control over every aspect of the final master, as well as giving you clear and accurate metering to help you make good decisions quickly.

## **1.1 A Word About Loudness**

Audio maximizers are designed to increase and balance the loudness profile of your audio while maintaining a high quality sound. This is an important audio processing step, especially when the quality of sound reproduction systems and listening environments can vary. However, there is such a thing as too loud. When audio is consistently too loud it can lead to listening fatigue for the user and a sub par experience. While an audio maximizer can be an essential tool which will help you get a good audio mix across a range of listening systems it's wise to pay attention to the overall mix levels and aim for the best sound quality not necessarily just the loudest audio when determining the setting of Elevate's parameters.

## Chapter 2

# Installation and Instantiation

Newfangled Audio plug-ins are distributed by Eventide and like other plug-ins Eventide distributes, Elevate uses PACE's iLok.com licensing system, with or without an iLok hardware dongle. Each license provides two activations which can reside on either your computer or on an iLok license dongle. Once you've purchased your plug-in, you'll need to register it on Eventide's website, activate your license, and install the plug-in on to your computer.

### 2.1 Registering Your Plug-in

When you purchase Elevate, you'll receive a Serial Number and License Key. The Serial Number will be two letters followed by 6 numbers. If you have an individual Elevate license, the Serial Number will start with NL (i.e. NL-#####). The License Key will be 3 sets of 4 characters, a letter or a number, each; like XXXX-XXXX-XXXX.

Once you've received these codes, you can register your plug-in on the Eventide website. To do so, please log in to <http://www.eventideaudio.com>, navigate to My Account in the top right corner, and select Register a New Product. Then, fill out the form by selecting Native Plug-in (VST, AU, AAX) in the Product Category field, select Elevate in the Product list, and enter your Serial Number, License Key, and iLok.com account name. If you don't yet have an iLok.com account, you can create one for free at <http://www.ilok.com>. Once you've done so, press Register.

Once you've entered this information and pressed the Register button, Eventide will send the applicable plug-in license to your iLok.com account, which you will need to activate to your computer or iLok dongle.

## 2.2 Activating Your License

To activate and manage your plug-in licenses you'll need to install PACE's iLok License Manager software which you can download from <http://www.ilok.com>. If you don't have this software installed, please download and install it now.

Once you have installed and launched iLok License Manager you should be able to log in to your account by clicking the large Sign In button in the upper left hand corner of the application. Once you have, you should be able to see available licenses by choosing the Available tab at the top of the iLok License Manager application. If you have successfully registered your plugin, your Elevate Native license will be available in this list. Please activate this license by dragging it to either your computer or iLok dongle listed on the left. When you do so, you will be asked to confirm the activation, and you will be able to see it by clicking on the location you have chosen. At this point your license is activated.

## 2.3 Installing Your Plug-In

You should have been given a link to the Elevate plug-in installer when you purchased your plug-in, but if you haven't, you can find downloads for all of our plug-ins at <http://www.newfangledaudio.com/downloads>. Please download and launch the correct installer for your system.

Once you've launched the plug-in installer, it will take you through several pages of options. We have tried to choose defaults for these options which will best serve the majority of users, but it is worth a minute to make sure you understand these options before clicking through to the next page. A common issue with Windows VST plug-ins is choosing the correct VST directory, which can be different on each system. Please pay special attention to this setting. Once you have followed through the installer, your plug-ins and presets should be in your chosen locations, and you can hit finish to end the installer application.

At this point, you should be ready to use your Elevate Plug-In.

## 2.4 Moving or Removing an Activation

If at any point, you decide to move your plug-in activation, you can do so in iLok license manager. To move an activation between an iLok dongle and your computer, simply plug in the iLok, locate the license in its current location, and drag it to its new location. To deactivate a license, find it in its location, right click on it, and choose deactivate.

Remember that each Elevate Plug-In License comes with two activations, which can be used on either a computer or iLok dongle, meaning you can use Elevate in two locations at the same time.



## 2.5 Removing or Uninstalling your Plug-In

We're sure you'll love your Newfangled Audio plug-ins, but if you ever want to remove them from your machine it's as easy as removing the following files.

For Mac:

- Macintosh HD/Library/Audio/Plug-Ins/Components/<Plug-In Name>.component
- Macintosh HD/Library/Audio/Plug-Ins/VST/Newfangled Audio/<Plug-In Name>.vst
- Macintosh HD/Library/Audio/Plug-Ins/VST3/Newfangled Audio/<Plug-In Name>.vst3
- Macintosh HD/Library/Application Support/Avid/Audio/Plug-Ins/Newfangled Audio/<Plug-In Name>.aaxplugin
- /Music/Newfangled Audio/<Plug-In Name>/

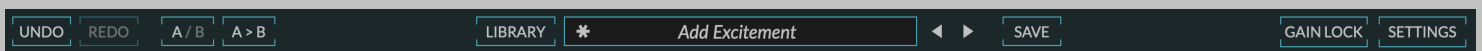
For Windows:

- c:\Program Files\Common\Steinberg\VST2\Newfangled Audio\<Plug-In Name>.dll
- c:\Program Files\Common\VST3\Newfangled Audio\<Plug-In Name>.vst3
- c:\Program Files\Common\Avid\Audio\Plug-Ins\Newfangled Audio\<Plug-In Name>.aaxplugin
- My Documents\Newfangled Audio\<Plug-In Name>\

## Chapter 3

# Navigation Bar

At the very top of the plug-in is a navigation bar which allows you to undo and redo changes, access the preset librarian, settings, and manage several other global features of the plug-in.



### 3.1 UNDO/REDO

On the far left is an UNDO and REDO button. After changing any plug-in parameter in the plug-in interface the UNDO button will activate. Pressing it at this point will undo this action. At this point the REDO button will become active, and pressing it will redo this action. Multiple levels of undo are available and you can move back through several changes to reach an earlier state.

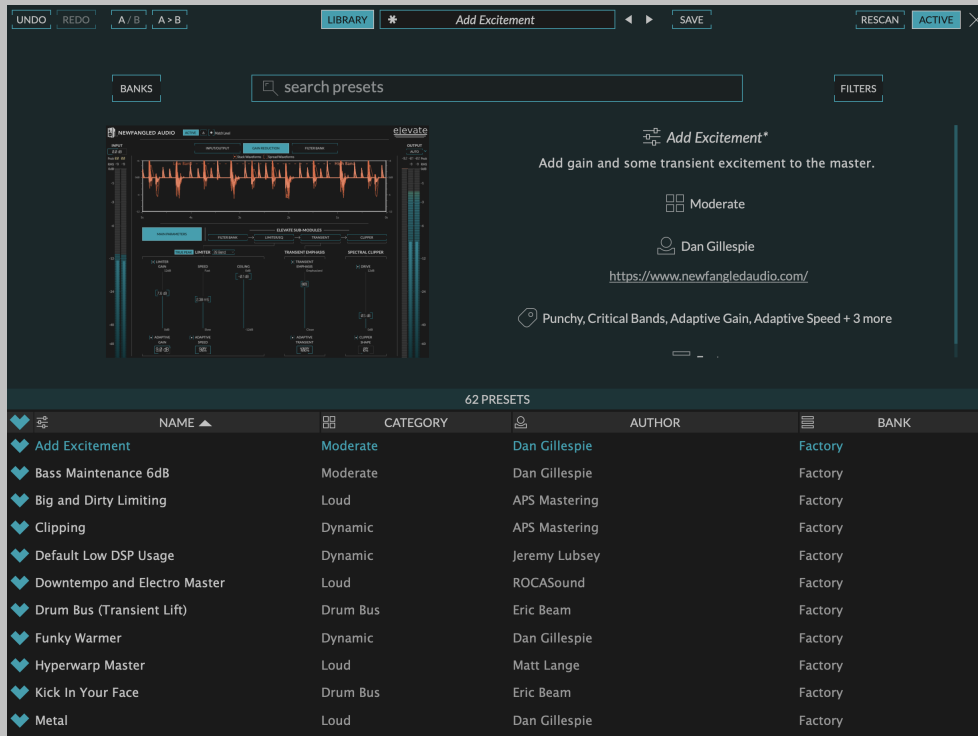
### 3.2 A/B COMPARE

Next you'll find buttons labeled A/B and A>B. These two buttons allow you to easily toggle between two states of the plug-in and compare the sounds. By default the plug-in is in the A state and you'll notice that the "A" is highlighted. In this state pressing the A>B button will copy the A state to the B state. Toggling the A/B button will switch to the B state, highlighting the "B" and switching the A>B button to B>A (allowing you to copy the B state to the A state).

Any changes you make in the plug-in interface will update the highlighted state. In this way, you can copy the A state of the plug-in to the B state, make changes to the A state, and use the A/B button to compare these two states.

## 3.3 LIBRARY

The Library button launches the Preset Librarian which can be used to explore the plug-ins presets based on any combination of category, author, tags, or favorites.



The Preset Librarian is split into four main sections.

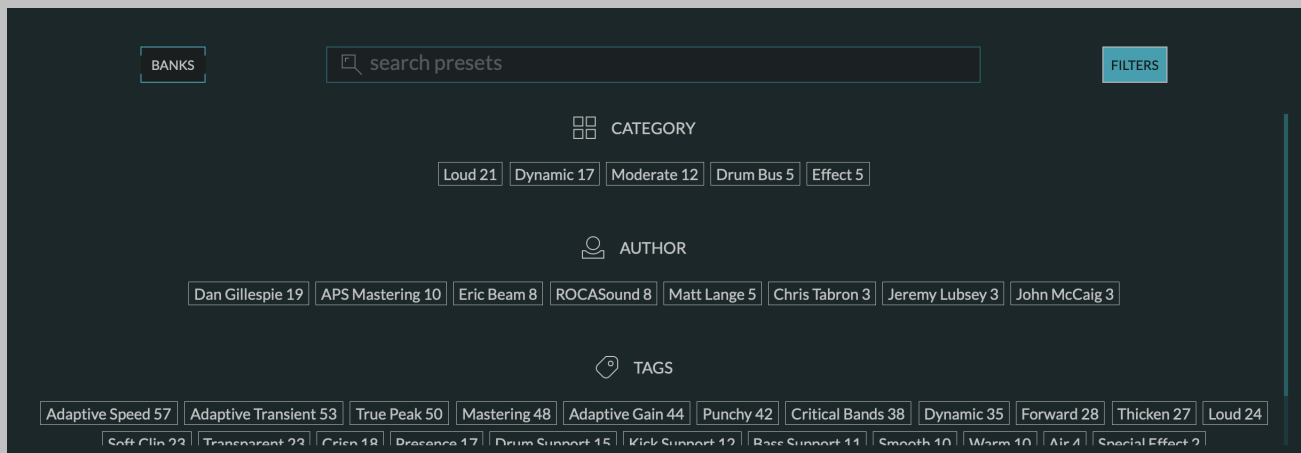
### 3.3.1 SEARCH and FILTERS

The top of the Preset Librarian shows the BANKS, SEARCH BAR, and FILTERS.

Typing a phrase into the search bar will allow you to search all the presets by a given phrase. You can further restrict this search by selecting a BANK or FILTERS.

Pressing the BANKS button opens a screen which allows you to restrict your search to the presets in either the FACTORY or USER bank. Or additional preset banks if you have any installed.

Pressing the FILTERS button opens a screen which allows you to restrict your search to presets matching a specific CATEGORY, AUTHOR, or one of several TAGS. Additionally, you can restrict your search to FAVORITES if desired.



### 3.3.2 PRESET LIST

The bottom of the Preset Librarian shows the list of presets which match the current criteria. It displays the FAVORITE status using a heart icon, the preset NAME, the preset CATEGORY, the preset AUTHOR, and the preset BANK. You can sort the list by any of these. The up and down arrows on your keyboard will allow you to toggle through the presets.

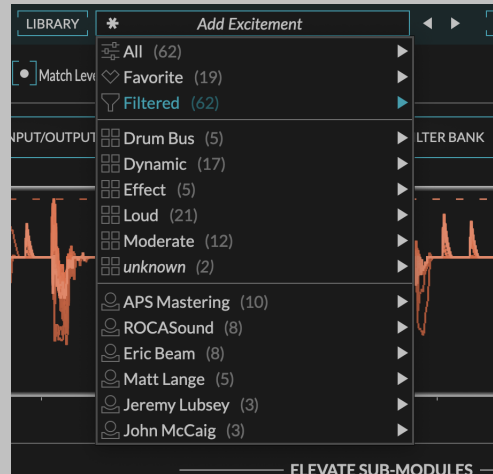
### 3.3.3 PRESET INFO

The center right of the Preset Librarian displays information about the currently loaded preset. This area shows the preset name, a short description if available, the category, the preset author - including a link to their work, the preset's tags, and its bank.

### 3.3.4 PLUGIN UI

The center left of the Preset Librarian displays a miniature version of the plug-in UI. This plug-in UI is not editable, but it is live and will show you the current settings of the plug-in, as well as any meters. We've found it incredibly useful to see this display as we've toggled through presets to get a sense of what each preset does.

## 3.4 PRESET SELECTOR



In the center of the navigation bar is a dropdown preset selector. Clicking on this preset selector will show you the plug-ins presets in several views.

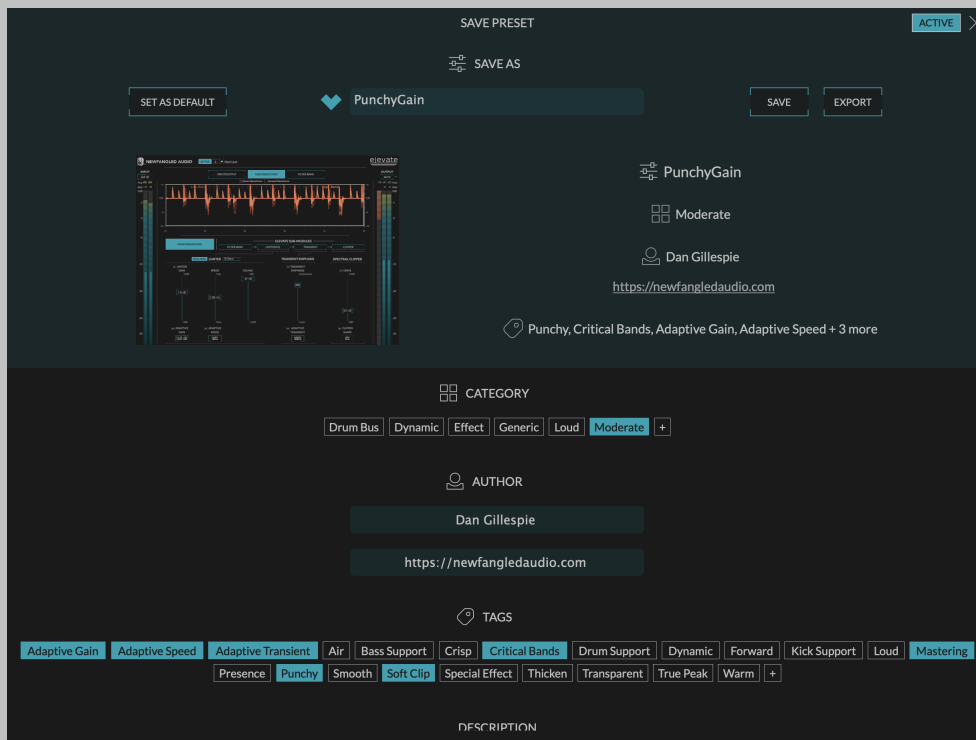
1. All will show you all the presets in alphabetical order.
2. Favorite will show you all your favorite presets in alphabetical order.
3. Filtered will show you all the presets which correspond to the current filter in alphabetical order. This filter is set up in the preset librarian.
4. Underneath these the presets are displayed according to their Category, as assigned when a preset is saved.

Once you load a preset from this dropdown list the Left and Right arrow buttons allow you to toggle through the selected subgroup in order. For more advanced preset selection click the LIBRARY button to enter the preset librarian.

Additionally, if you like the sound of the current preset you can click the heart icon to set the loaded preset as a favorite. If a loaded preset is modified the heart icon will become an asterisk and the preset name is italicized, indicating that the current state no longer matches that of the saved preset.

## 3.5 SAVE

The SAVE button launches the preset save screen where you can save a preset and assign categories, tags, and descriptions, as well as set the default state of the plug-in.



### 3.5.1 PRESET NAME

This field sets the preset name. There can only be one preset with a given name.

### 3.5.2 SET AS DEFAULT

This button will set the current state of the plug-in as a plug-in's default. This means when you load the plug-in it will default to these settings.

### 3.5.3 HEART ICON

This will mark this preset as a favorite when you save it.

### 3.5.4 SAVE BUTTON

This button saves the preset

### 3.5.5 EXPORT BUTTON

In order to be browsed by the plug-in the presets are saved in a particular folder. However, if you want to save the preset elsewhere on your computer you can use the EXPORT button to export it as a file.

### **3.5.6 PLUGIN UI and DESCRIPTION**

This section shows the current settings of the plug-in, and how it will appear in the preset librarian.

### **3.5.7 CATEGORY**

Set the category for the preset you are saving. A preset can have only one category. If the preset doesn't fit into any of the existing categories you can use the (+) button to create a new one.

The existing categories are based on those that are found in the existing presets. If you create a new one it will appear in this list with an "X" on it. If you decide not to use the newly created CATEGORY you can remove it by clicking on the "X". However, once the preset is saved this category will be permanent and available to all newly saved presets, unless you delete all presets which use it.

### **3.5.8 AUTHOR**

The AUTHOR section allows you to add your name and a url if you maintain a presence on the web. This will be automatically populated with the information from the last saved preset, so you will only likely need to enter it once.

### **3.5.9 TAGS**

You can add any number of TAGS to a saved preset. These TAGS will allow your preset to be found in the preset librarian. It's best to add the tags that apply to the preset, and don't add the ones that don't. You can create a new tag using the (+) button.

The existing tags are based on those that are found in the existing presets. If you create a new one it will appear in this list with an "X" on it. If you decide not to use the newly created TAG you can remove it by clicking on the "X". However, once the preset is saved this tag will be permanent and available to all newly saved presets, unless you delete all presets which use it.

### **3.5.10 DESCRIPTION**

If desired you can add a short description to the preset. It's best to use this field to add information about what a preset is for, or any hints for a user about how to use it.

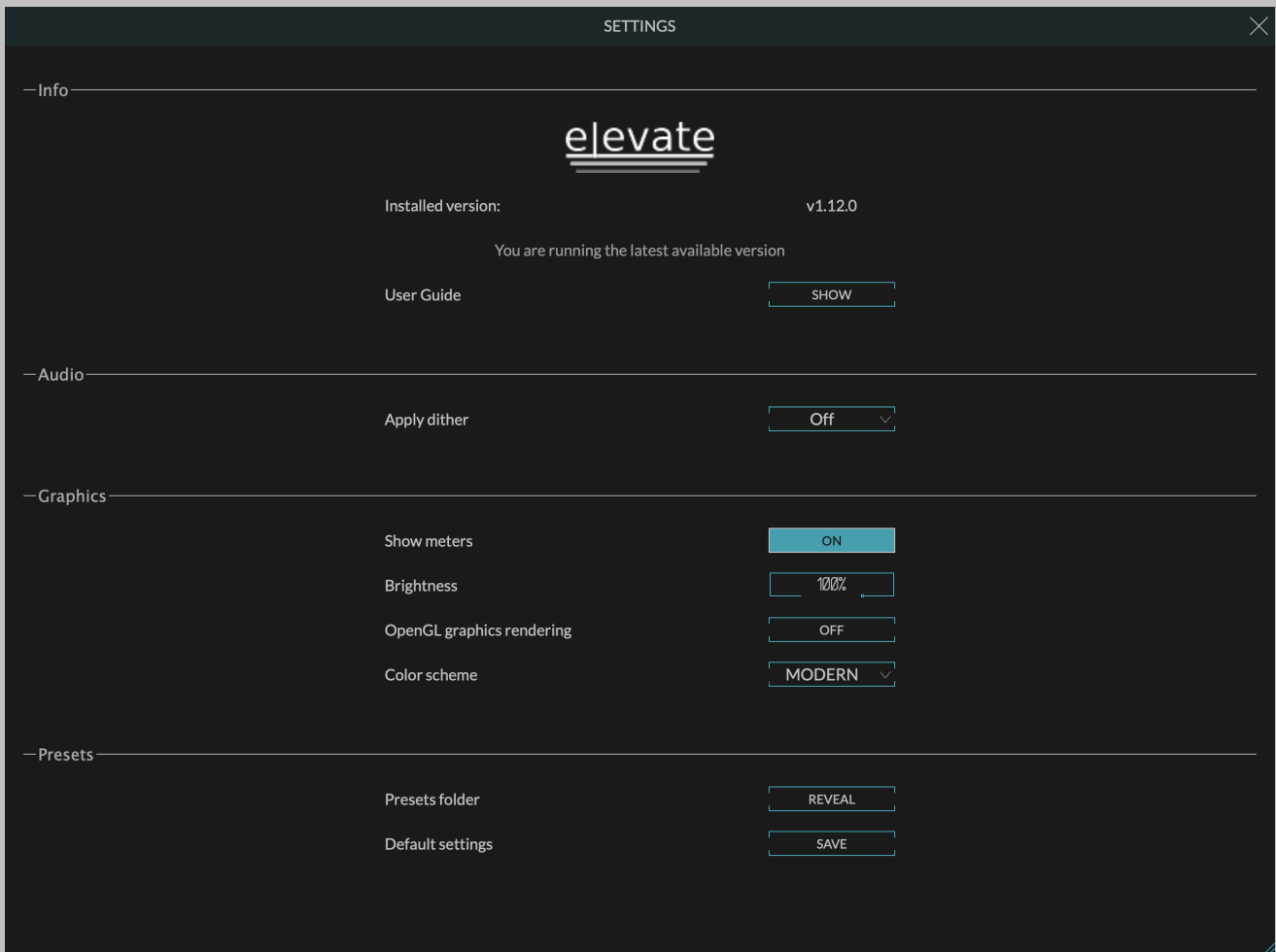
## **3.6 GAIN LOCK**

The purpose of Elevate is to add gain to your audio, however, the presets give you slightly different takes on this. For this reason, just flipping

through presets can be jarring if they're all stored with different amounts of gain. The GAIN LOCK button solves this problem by locking all the gain related parameters so the total level settings in the plug-in won't change and you will get fair comparisons. Specifically, this button will cause the GAIN, DRIVE, CEILING, INPUT LEVEL, and OUTPUT LEVEL values to not be updated based on the preset settings so they will remain the same as you are currently listening to. The one exception is that the ratio of the GAIN to DRIVE setting will be updated based on the preset values but the total amount of GAIN + DRIVE will not be, so the sound will change, but the requested gain will not.

## 3.7 SETTINGS

The SETTINGS button pops down a settings page which contains additional options.





### **3.7.1 INSTALLED VERSION**

This displays the version number that is currently installed. If you are running the latest version it will tell you. If there is an update an UPDATE button will appear. Clicking this button will bring you to the downloads page where you can get the latest version.

### **3.7.2 USER GUIDE**

Pressing the SHOW button will launch this user guide.

### **3.7.3 APPLY DITHER**

This settings allows you to dither down to 24, 20, 16, 12, or 8 bits.

### **3.7.4 SHOW METERS**

The SHOW METERS button will show or hide the additional meters in the plug-in. Specifically, turning it off will hide the glow in the RADAR control, the envelope graph behind the LEVEL DETECTOR section, and the curve graphic behind the CURVE section.

### **3.7.5 BRIGHTNESS**

The BRIGHTNESS control will control the brightness of the glow in the RADAR control, the envelope graph behind the LEVEL DETECTOR section, and the curve graphic behind the CURVE section.

### **3.7.6 OPENGL GRAPHICS RENDERING**

The USE OPENGL button will enable OpenGL rendering of the UI. For this setting to take effect you must close and reopen the UI. If you find that the plug-in UI renders better with OpenGL on or off on your computer you can save this as the default setting and new instantiations will always load using this setting.

### **3.7.7 COLOR SCHEME**

The COLOR SCHEME dropdown will allow you to choose one of several different color schemes for your viewing pleasure.

### **3.7.8 PRESETS FOLDER**

Clicking the REVEAL button will bring you to the presets folder. This is only necessary if you want to share your presets with someone else, or access these files for another reason.

### **3.7.9 DEFAULT SETTINGS**

Pressing the SAVE button will save all the current settings as the default for the plug-in. This is useful if you'd like a different starting point than the one we've provided.

## **3.8 Resize**

The bottom right corner of the UI will allow you to resize the plug-in. To change the default plug-in size drag the plug-in to the desired size and save over the default preset.

## Chapter 4

# Parameters and Metering

Elevate is designed to be incredibly easy to use, but also incredibly powerful. The MAIN PARAMETERS page exposes ten core parameters which give you control over an adaptive limiter, transient engine, and final clipping stage. If you want to tweak further, Elevate also exposes band parameters for the filterbank and each band in the 4 Elevate sub modules. In addition to this, Elevate shows you several different kinds of metering to keep an eye on what's happening. Here are brief descriptions of these pages, along with some of the other features of the plug-in.

### 4.1 Controls

Elevate has several control types, including vertical and horizontal sliders, rotary knobs, drop-down menus, and toggle and radio buttons, all of which have tool-tips which will give you more information about their function if you hover your mouse above them.

In general, all sliders and knobs can be double clicked to type in a specific value, option-clicked to return them to their default value, or command-clicked or control-clicked to enter a vernier mode for fine tuning. In addition, some controls support special shift-click functionality, such as using shift-swipe to draw an EQ curve, or shift clicking the gain sliders to set output level compensation. Where available, these shift-click functions are mentioned in the tool-tips for the individual controls.

Where appropriate, most controls have a related on/off button which will deactivate this control, or even the entire section. Turning sections off when you're not using them can reduce the amount of processing power used by Elevate.

## 4.2 Global Parameters

### 4.2.1 ACTIVE



The ACTIVE button activates or bypasses all the processing in the plug-in. If the nearby Match Level button is on, the inactive state will apply the equivalent clean gain that you are asking for in the ACTIVE state. If you have enough headroom, this will allow you to compare the output of the Elevate plug-in at the same theoretical level to more easily hear what effects the processing is having on your sound. Realize that if you don't have enough headroom on your master fader this will cause the output of your DAW to clip.

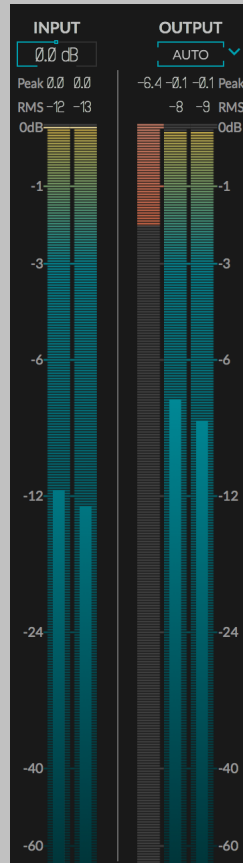
### 4.2.2 Δ

The Δ symbol engages Delta Listen Mode. When you toggle this button we subtract the plug-in output from the input to show only what is changing in the processor. Think of this as what the process is removing. Careful use of this can be a powerful tool to hear what the plug-in is doing to your audio.

### 4.2.3 MATCH LEVEL

As mentioned above, MATCH LEVEL will boost the dry signal when the plug-in is inactive in order to allow you to compare the limited signal to the non-limited signal with the same amount of gain. This can give you a fair comparison between the clean and effected audio and help you understand what artifacts the algorithm may be imparting. When MATCH LEVEL is on and the plug-in is inactive the output level will likely be much higher than 0 dBfs, so make sure you have appropriate headroom in your playback system to avoid clipping.

#### 4.2.4 INPUT AND OUTPUT LEVEL and BAR METERS



Above the INPUT meter is a knob for the MASTER INPUT LEVEL, and above the OUTPUT meter is an AUTO button for setting the MASTER OUTPUT LEVEL. By default AUTO is off and adding gain will result in a louder sound. However, if you turn the AUTO output level button on, the MASTER OUTPUT LEVEL will be reduced by the total amount of GAIN and DRIVE added to the signal. This will allow you to easily compare the limited sound to the bypassed sound and determine that you are still getting transparent limiting. This is a very handy feature, but make sure to turn AUTO off before bouncing your final track, or else you will be reducing the peak gain, rather than increasing the RMS gain.

If you'd prefer not to use the AUTO button to set the MASTER OUTPUT LEVEL and would like to set it manually instead, clicking the down arrow next to the AUTO button will allow you to select a manual mode which will display a knob for the MASTER OUTPUT LEVEL. The MASTER INPUT LEVEL is just before the INPUT meter and the MASTER OUTPUT LEVEL is just before the OUTPUT meter, these controls should be used as TRIM controls only when you are targeting an input or output standard other than 0 dBfs, or when the input signal is too low or too high as indicated by the INPUT

or INPUT/OUTPUT meter.

When the LIMITER section is in TRUE PEAK mode the OUTPUT meter will show the TRUE PEAK output rather than the sample based output.

### **4.3 PARAMETER PAGES**

Most of Elevate's most important controls are on the MAIN PARAMETERS page. Many times you will be able to get the results you want without ever leaving this page. However, there are also four ELEVATE SUB-MODULE pages which are arranged to show the flow of audio through Elevate's sub-modules. We will discuss the parameters on each of these pages here.

### **4.4 MAIN PARAMETERS PAGE**

The MAIN PARAMETERS page contains the primary parameters for making the input signal louder while avoiding artifacts. It contains 5 parameters for the LIMITER, 2 parameters for the TRANSIENT EMPHASIS section, and 2 for the SPECTRAL CLIPPER.



#### 4.4.1 LIMITER SECTION

On the left hand side is the LIMITER section. You will see sliders for the standard limiter parameters GAIN and SPEED (sometimes called threshold and release) as well as a slider for CEILING. Below these parameters you will see knobs for ADAPTIVE GAIN and ADAPTIVE SPEED, these control the adaptive algorithms that reduce audible artifacts and make Elevate so powerful. Elevate works by using an auditory filter bank to split the signal into the critical bands of human hearing, then using these adaptive algorithms to know when and where to best apply limiting. If you deactivate the ADAPTIVE parameters and the PER BAND parameters on the EQ and TRANSIENT pages, the LIMITER will act as a standard single band limiter, but that would be boring.

##### TRUE PEAK

By turning on the TRUE PEAK button Elevate will enable TRUE PEAK limiting, otherwise known as inter-sample peak limiting. When TRUE PEAK

is on Elevate will limit the incoming audio based not on the incoming samples themselves, but based on the peak values measured between samples. Many users believe that inter-sample peaks that go above 0 dB can create audible distortion when the audio is rendered in listeners playback systems. Additionally, TRUE PEAK limiting is required for several broadcast standards, and may be important for signals that are converted to lossy formats like mp3 and mp4.

Elevate's TRUE PEAK mode is designed to match all standards while maintaining maximum clarity. You should not be able to detect a significant difference between in audio between TRUE PEAK on and off, however, it does alter the processing, so you may detect differences, especially at extreme settings. Additionally, turning on TRUE PEAK will raise the CPU usage of Elevate slightly.

Finally, when Elevate's TRUE PEAK mode is engaged Elevate's OUTPUT METER becomes a TRUE PEAK output meter and instead of displaying the maximum sample values it will display the maximum inter-sample peaks.

## **NUMBER OF FREQUENCY BANDS**

The first stage of signal processing is the FILTER BANK and next to the LIMITER section label is a drop down that allows you to select the number of frequency bands being used by the adaptive algorithms in both the LIMITER and TRANSIENT sections. This can be viewed as a resolution parameter for the core Elevate algorithms, as is set to the maximum be default. However, this can use a lot of processing power, and the number of bands can often be reduced and still provide good results. If you find that you're running out of processing power, just select a smaller value.

More fine tuning of these frequency bands can be done on the FILTER BANK sub-module page.

## **GAIN**

The GAIN is the amount of gain you wish the ELEVATE limiter to apply. In a nod to the compression that happens in mastering limiters this control is often called threshold, but we felt that GAIN was more appropriate given the purpose of this control. The ELEVATE limiter uses an optimally soft knee to allow very fast release times while producing minimal distortion. To the left of GAIN is a button which will allow you to deactivate this part of the algorithm.

## **ADAPTIVE GAIN**

The ADAPTIVE GAIN parameter allows the algorithm to adapt the gain applied to each frequency band individually, allowing more gain reduction to be applied to the loudest bands if the signal is near the limit. Note that similar to a multi-band limiter this has the potential to effect



the tonal balance of your mix if used too heavily. The setting is the number of dB by which each band can stray from the others. To the left of ADAPTIVE GAIN there is a button which will allow you to deactivate this part of the algorithm.

## **SPEED**

The SPEED parameter is the combined attack and release time of the LIMITER. Moving this slider up speeds up the dynamics of the LIMITER and allows higher output levels, but also introduces the possibility of distortion. The perception of distortion and therefore the best setting is highly program dependent so pick the best setting for your mix.

## **ADAPTIVE SPEED**

The ADAPTIVE DYNAMICS parameter adapts the attack, release, and look-ahead parameters for each filter band and greatly reduces "pumping and breathing" artifacts as well as perceived distortion.

## **CEILING**

The CEILING slider allows you to set the maximum output level of the plug-in. This affects the behavior of both the LIMITER and SPECTRAL CLIPPER.

## **4.4.2 TRANSIENT EMPHASIS SECTION**

Look-ahead style mastering limiters have a tendency to squash transients and reduce the dynamic impact of music. Elevate alleviates this problem by re-emphasizing these transients and driving them into a clipper stage. The MAIN PARAMETERS page shows TRANSIENT EMPHASIS and ADAPTIVE TRANSIENT controls.

## **TRANSIENT EMPHASIS**

The TRANSIENT EMPHASIS slider allows you to compensate for the squashing effect by allowing fast transients through the limiter or even emphasizing them for a more explosive sound. To the left of TRANSIENT EMPHASIS is a button which allows you to deactivate this part of the algorithm.

## **ADAPTIVE TRANSIENT**

The ADAPTIVE TRANSIENT parameter allows the TRANSIENT EMPHASIS engine to be adapted to each filter band individually. Settings in between 0% and 100% make sure that adjacent bands are similarly emphasized, but that distant bands are not. This allows transients from kick drums, snares, and cymbals to maintain their impact while not effecting the entire mix or each other. TRANSIENT EMPHASIS and ADAPTIVE TRANSIENT

are powerful parameters, so use with care. To the left of ADAPTIVE TRANSIENT is a button which allows you to deactivate this part of the algorithm.

#### **4.4.3 SPECTRAL CLIPPER SECTION**

The SPECTRAL CLIPPER sits at the end of the signal chain and allows you to clip transients, or even drive the entire signal into it. Its innovative design allows it to overdrive a signal without effecting the tonal balance or allowing the signal to get "tubby", which makes it ideal for overdriving an entire mix. Clipping short transients is often the ideal way to increase level without introducing pumping, and using the DRIVE parameter will allow you to add an extra oomph to more aggressive styles of music. These parameters are duplicated on the CLIPPER page along with a graphical indication of their behavior.

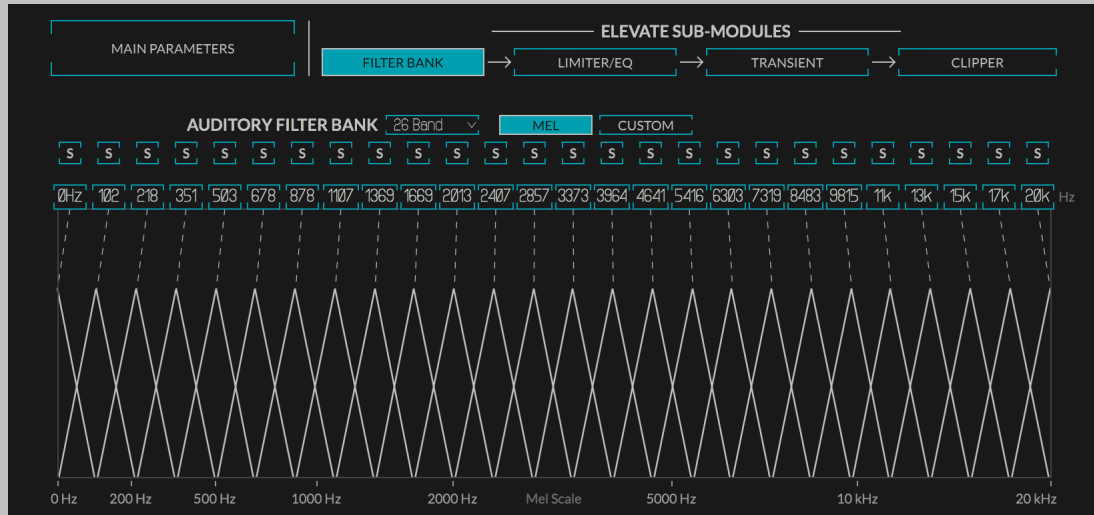
##### **DRIVE**

The DRIVE parameter allows you to increase the LIMITER output into SPECTRAL CLIPPER to increase loudness and distortion. It does this according to a gain curve determined by the SHAPE parameter.

##### **SHAPE**

The SHAPE parameter changes the shape of the gain curve from soft to hard. This clipper uses an ideal set of shapes which allow for maximum boost with minimal added harmonic content. When DRIVE is 0 dB no gain is being provided and the soft and hard SHAPEs are the same.

## 4.5 FILTER BANK SUB-MODULE PAGE



The Elevate plug-in uses a triangular filter bank to split the input signal into a number of frequency bands that resemble the critical bands in the human ear. The FILTER BANK parameter page allows you to control the filters that split the input signal into these frequency bands. This filter bank outputs a number of audio streams which the adaptive algorithm uses to set the gain reduction across bands. In general, the higher the number of bands, the more fine grained control the adaptive algorithms will have, and the more gain you can add before hearing artifacts. However, a larger numbers of bands does increase the algorithms processing requirements significantly, and increasing the number of bands may not always produce a noticeable increase in quality.

The number of bands selected in the auditory FILTER BANK will change the number and frequency of bands in the graphic EQ and TRANSIENT emphasis pages. Selecting the 1 BAND option will also deactivate the ADAPTIVE parameters, as these work across the bands. By selecting 1 BAND you are turning the limiter section into a standard look-ahead limiter, with transient emphasis.

### 4.5.1 AUDITORY FILTER BANK

This drop-down allows you to select the number of bands you are splitting the signal into. Setting the number of bands using this drop-down will automatically space them on the mel scale between the minimum value and the maximum value.

### **4.5.2 MEL/CUSTOM**

This set of interactive radio buttons allow you to distribute the filter center frequencies using the MEL scale or a CUSTOM scale of your choosing. The MEL scale is an auditory scale based on the perceptual distance between pitches, selecting MEL will equally distribute the chosen number of filters between the current minimum frequency and current maximum on the MEL scale. CUSTOM is selected by either choosing the radio button, or by moving any of the filter center frequencies away from their MEL based values. By customizing the filter bank to your program material you can use the EQ and TRANSIENT pages to hone in on particular frequency ranges that are important in your program material. For instance you might choose one filter to focus on the resonant frequency of a kick drum, and another for the bass guitar and use the EQ and TRANSIENT pages to increase punch in just the bass drum without effecting the relative loudness of the bass guitar.

### **4.5.3 SOLO BUTTONS**

There is a set of toggle buttons which allow you to solo the output of one band, or for a number of bands by shift-clicking or ctrl-clicking. These are the same solo buttons that appear on the EQ and TRANSIENT pages.

### **4.5.4 INSERT FILTER**

The (+) button beneath the solo for the selected band will insert a new filter with a center frequency between the band chosen and the one to it's right. This can be especially helpful if you find you need more resolution in an area when fine tuning the filter center frequencies. If you are already using the maximum number of filters, the INSERT FILTER buttons will be greyed out.

### **4.5.5 REMOVE FILTER**

The (-) button beneath the solo for the selected band will remove the current band without effecting the center frequencies of the surrounding bands. This can be helpful if you find one or more bands to be redundant while fine tuning center frequencies.

### **4.5.6 FILTER CENTER FREQUENCIES**

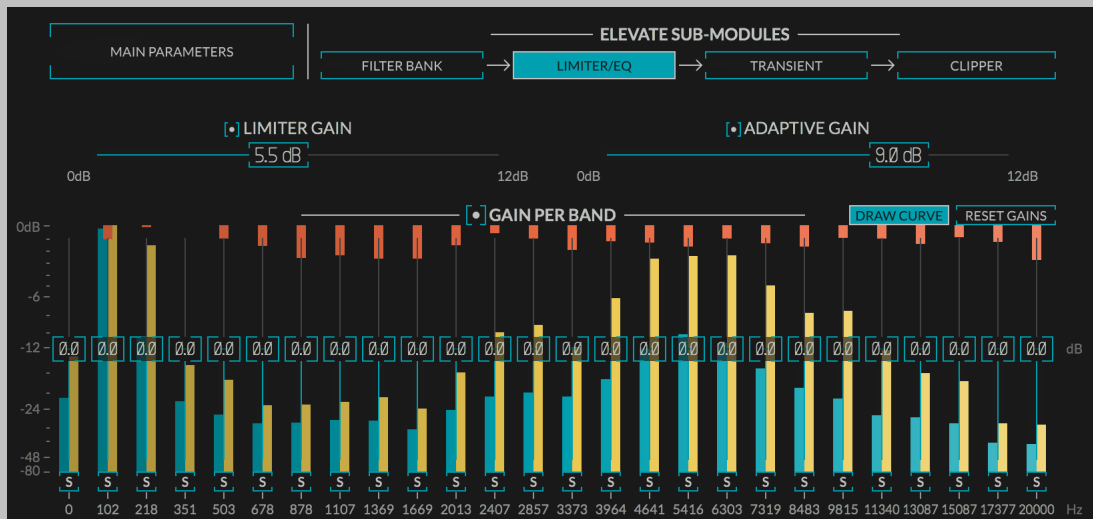
There are also a set of text box sliders that allow you to set the CENTER FREQUENCY of each band. By default these are set to match the critical bands of the human ear, however these bands can be fine tuned to more closely match important frequency ranges in your program material as well. Doing so will allow you to use the band specific gain and transient sliders in the other sub-modules to fine tune your result.

## 4.5.7 FILTER BANK DISPLAY

The FILTER BANK DISPLAY shows you the relative distance and shape of each filter using the MEL scale. This will help you visualize how the filters are arranged across the auditory spectrum. The triangular shape of these filters help adjacent band have some amount of similarity, leading to a smoother overall sound.

## 4.6 LIMITER/EQ SUB-MODULE PAGE

The LIMITER/EQ sub-module page allows you to fine tune the GAIN settings for each frequency band and see the relative gain reduction being applied. The GAIN and ADAPTIVE GAIN parameters are the same as on the MAIN PARAMETERS page, but each filter band now also breaks out:



### 4.6.1 GAIN PER BAND

A GAIN slider which allows you to adjust the amount of additional gain or cut to be applied to the band, similar to the Equivocate graphic equalizer. You can set these slider individually by clicking and dragging, or draw a curve by shift-clicking and dragging across the parameter field. These gains can also be deactivated by clicking on toggle button to the left of the GAIN PER BAND title.

### 4.6.2 RESET GAINS

Clicking the RESET GAINS button will reset all gains to 0dB.

### 4.6.3 BAND SOLO

A SOLO button which allows you to hear the output of this or a combination of bands. This can allow you to zero in on the effects of the parameters from a specific band to either find problem spots, or better understand what the algorithm is doing. When a SOLO is active the entire algorithm is still running across all bands, but you are simply changing what you can see and hear.

### 4.6.4 DRAW CURVE

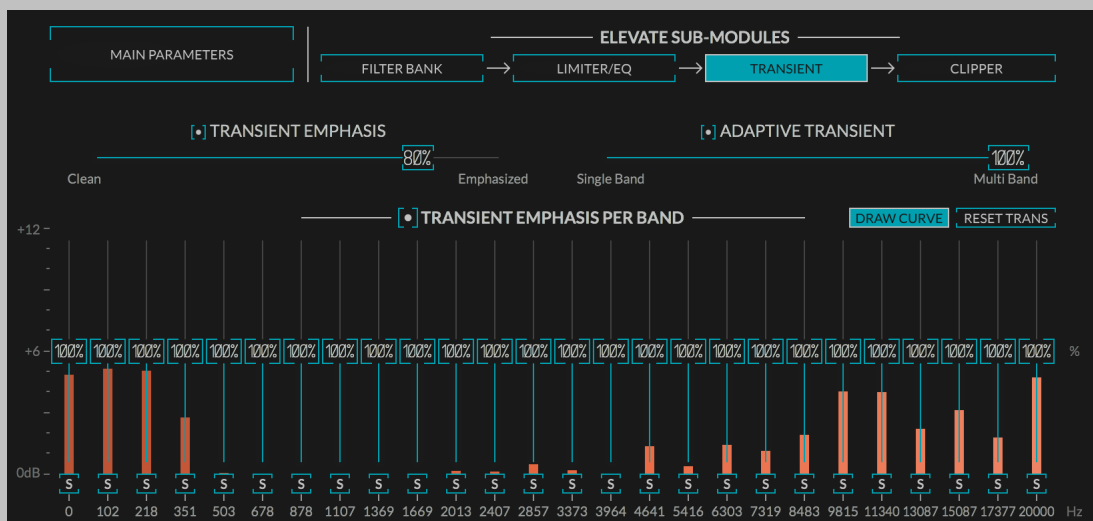
Clicking the DRAW CURVE button will allow you to draw a single gain curve across all the GAIN sliders. Turning this off will allow you to fine tune each GAIN slider individually. Whichever mode you're in you can use the opposite behavior by pressing the shift key.

### 4.6.5 METERING

Drawn behind the sliders for each band is an input, output, and gain reduction meter for each band. These are the same meters shown in the FILTER BANK meter page in the top section of the GUI. Note that this meter is slowed down from the gain reduction actually applied to the signal to make it easier to view. This metering is a good way to see the effects of the ADAPTIVE GAIN control.

## 4.7 TRANSIENT SUB-MODULE PAGE

The TRANSIENT sub-module page allows you to fine tune the TRANSIENT EMPHASIS for each band. The TRANSIENT EMPHASIS and ADAPTIVE TRANSIENT parameters are the same as are on the MASTER PARAMETERS page, but each filter band now also breaks out:



### **4.7.1 TRANSIENT EMPHASIS PER BAND**

A TRANSIENT EMPHASIS slider which goes from 0% to 200%. This allows you to reduce or increase the amount of TRANSIENT EMPHASIS being applied to each band. This can be helpful for bringing out just the transients in particular frequency ranges, like kick or snare drums. The effects of these additional transient emphasis controls can be turning off using the toggle button next to the TRANSIENT EMPHASIS PER BAND title.

### **4.7.2 RESET TRANS**

Clicking the RESET TRANS button will reset all transient emphasis sliders to 100%.

### **4.7.3 BAND SOLO**

A SOLO button which allows you to hear this output of this or a combination of bands. This is the same SOLO button available on the MULTIBAND GAIN parameter page and its state will transition across pages.

### **4.7.4 DRAW CURVE**

Clicking the DRAW CURVE button will allow you to draw a single gain curve across all the TRANSIENT sliders. Turning this off will allow you to fine tune each TRANSIENT slider individually. Whichever mode you're in you can use the opposite behavior by pressing the shift key.

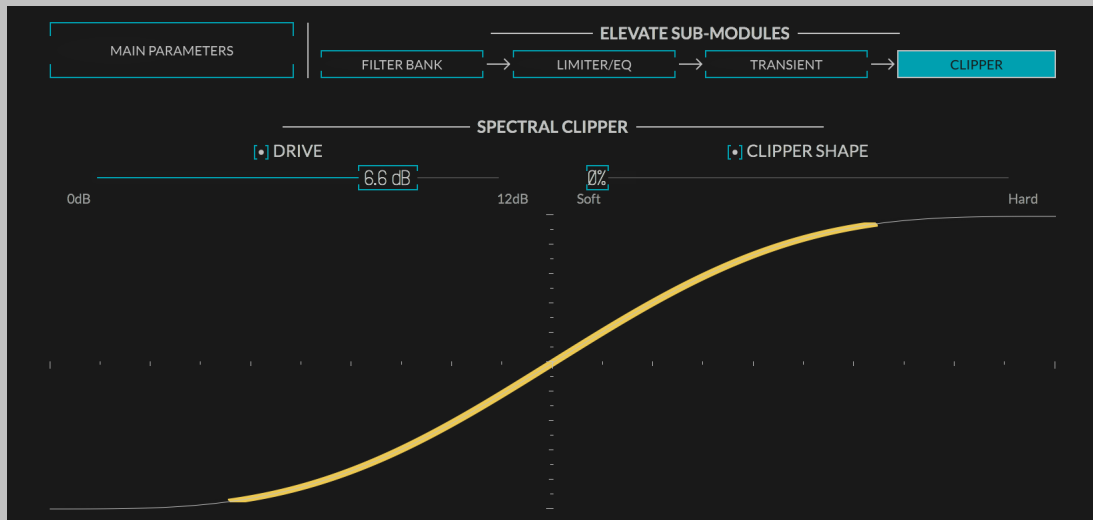
### **4.7.5 METERING**

And a TRANSIENT EMPHASIS meter which shows the amount of TRANSIENT EMPHASIS being applied to each band. These meters will show you the effect of the ADAPTIVE TRANSIENT slider well. Similar to the FILTER BANK meters these are slowed down to give you a better sense of the spectral ramifications of the transient emphasis section. These meters are a good way to see the effects of the ADAPTIVE TRANSIENT control.

## **4.8 CLIPPER SUB-MODULE PAGE**

The final stage of the Elevate plug-in is the SPECTRAL CLIPPER. This clipper is primarily designed for clipping the fast transients allowed through the transient emphasis section, but also allows you to add up to an additional 12 dB of distortion based gain. The SPECTRAL CLIPPER uses an optimal soft clipping curve to add the minimal amount of harmonics for a particular gain setting, as well as a special frequency domain algorithm to avoid the low-passed "tubby" sound that can be produced

when over-driving full bandwidth audio. The SPECTRAL CLIPPER parameter page allows you to control the DRIVE and CLIPPER SHAPE of the SPECTRAL CLIPPER using duplicates of the parameters found on the MAIN PARAMETERS page. An animated meter below indicates the shape of the clipper by mapping it's output on the Y axis to it's input on the X axis. A meter also shows what the level in to and out of the SPECTRAL CLIPPER is at any given moment.

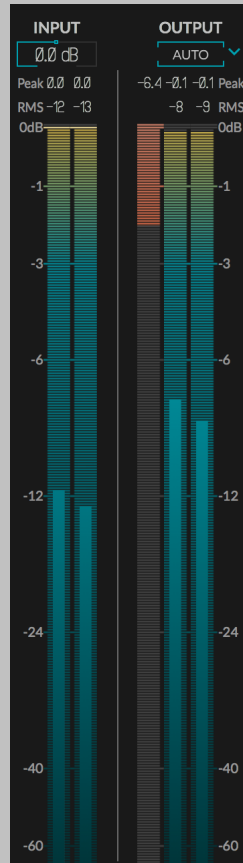


## 4.9 METERS

In addition to the plug-ins parameters there are a selection of meters available.



### 4.9.1 INPUT and OUTPUT BAR METERS

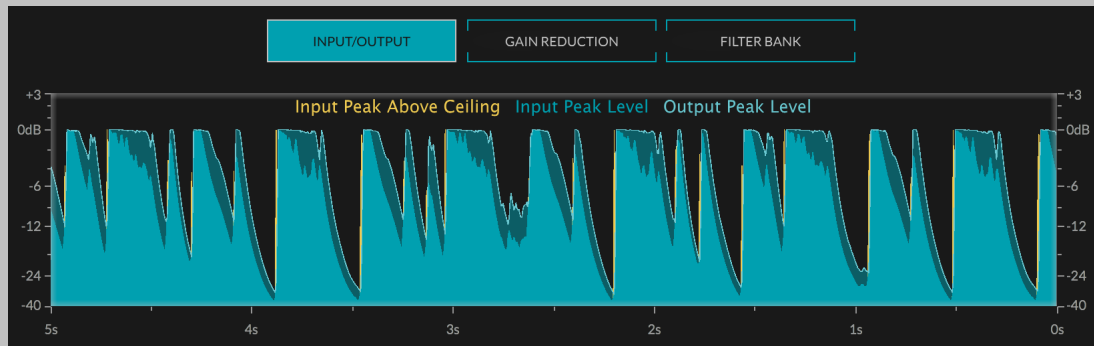


The left and right hand sides of the plug-in have matching vertical bar meters which display the INPUT (on the left) and OUTPUT (on the right) levels in Peak (horizontal ticks), RMS (solid bar and numerical display), and Peak Hold (numerical display) formats. Additionally, the right hand side includes a global gain reduction meter, which shows the maximum gain reduction being applied across the bands at any given moment. These meters are always present in the display and clicking the Peak Hold section, or bypassing the plug-in will clear the held peak value.

### 4.9.2 CENTER SECTION METER

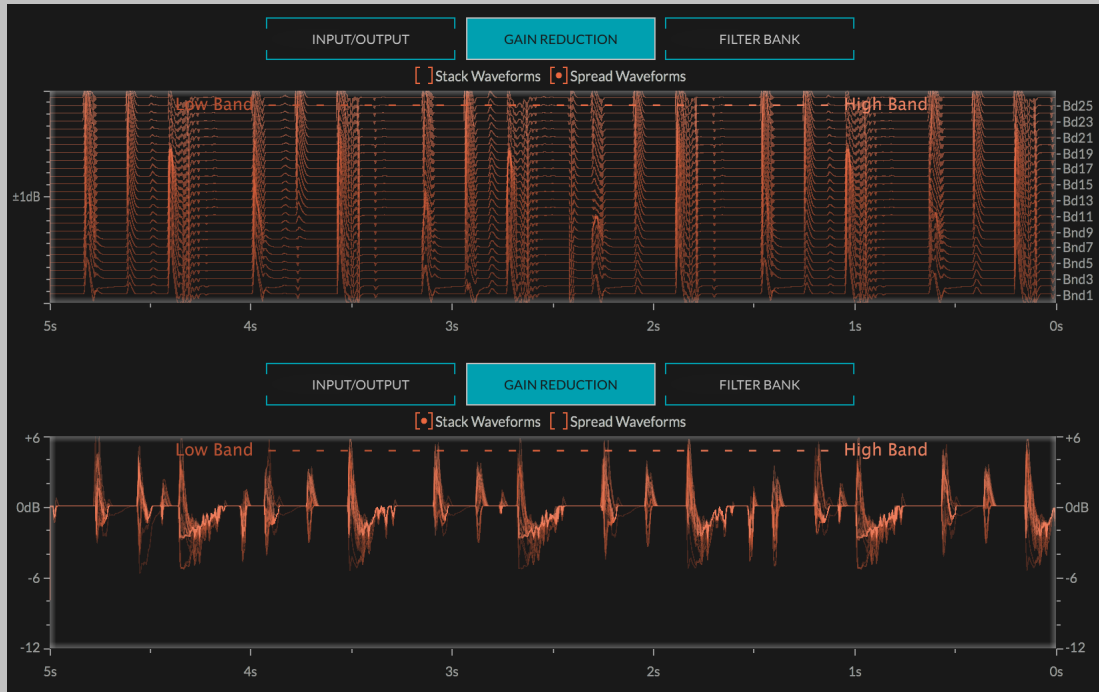
Additionally, when OpenGL rendering is on, the top center section allows you to see one of:

### 4.9.3 SCROLLING INPUT/OUTPUT METER



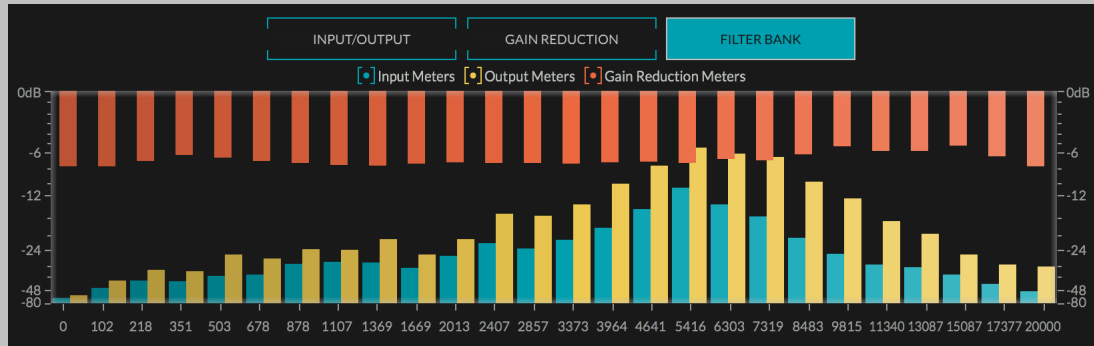
The scrolling INPUT/OUTPUT meter displays the input peak value overlaid with the output peak value over time. This allows you to visualize the INPUT level over time as the top of the solid color, the OUTPUT level over time as the top of the overlaid color, and the amount of GAIN being applied by the processing as the width between these. This meter also gives an indication of when the INPUT level exceeds the output when the input pokes out over the output. For instance, in the NEW-FANGLED color scheme, if the red input curve is poking out above the green curve it's an indication that you may want to use the MASTER INPUT LEVEL control to reduce the input level and make this up with the GAIN control.

## 4.9.4 SCROLLING GAIN REDUCTION METER



The scrolling GAIN REDUCTION meter displays a combination of GAIN REDUCTION and TRANSIENT EMPHASIS for each active band. It has two views. In "Stack Waveforms" view each individual GAIN meter is stacked on top of each other with the same dB scale. Lower frequencies are displayed in a darker color while higher frequencies are displayed in a lighter color. This meter give you a good view of the dynamics of each gain signal over time (they can be quite fast), and allows you to see how the parameters are effecting these gain signals. In "Spread Waveforms" view, each individual GAIN meter is has it's own waveform, with the low frequencies being represented on the bottom and the high frequencies being represented on top. This is good at showing relative gain and gain reduction for each band over time, and shows how the adaptive algorithms are working well. This is probably the best meter to show you what Elevate is doing over both time and frequency.

## 4.9.5 FILTER BANK METER



Finally, the FILTER BANK meter is a bar graph showing the input level, output level, and gain reduction for each individual band. These meters are slowed down to show you an average amount of level and gain reduction over time to get a better sense of the tonal effects of the processing being applied. While the GAIN REDUCTION meter gives a good indication of the dynamics of the Elevate algorithm, the FILTER BANK meter gives a good indication of the frequency distribution in the input and the output.

## Chapter 5

# Conclusion

We hope you enjoy Newfangled Audio's Elevate plug-in. If you have any questions, comments, or concerns please write us at [support@newfangledaudio.com](mailto:support@newfangledaudio.com).

## Chapter 6

# About Newfangled Audio

The Oxford English Dictionary defines Newfangled as "objectionably new".

Music technology can sometimes be a backward looking pursuit. This is understandable, the purpose of music technology should be to help musicians make great music. There have been many great pieces of gear in the past and we should seek to keep these pieces and make them available to people who want to use them, and the time and dedication required to master a musical instrument means that changes in their design are often evolutionary rather than revolutionary.

However, the great pieces of gear from yesteryear are more often than not those that contained new ideas in their time. The reverence we have for these pieces can sometimes turn into fetishism, and mindless re-creation of classic gear can fail to inspire musicians and artists to take new risks. It's important to make sure artists and engineers have access to good tools that inspire them and don't stand in their way, but these tools should never be used as a security blanket to stand in the way of an artist or engineers ears and taste.

Newfangled Audio seeks to only make gear that incorporates new ideas. We want to make gear that is great, but only using ideas that others are not. We realize that deviation from the norm might sometimes be objectionable. The Oxford English Dictionary defines Newfangled as "objectionably new".