Auria User Guide

WAVEMACHINE
LABS

Auria User Guide v1.150
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Part One - The Basics

1

GETTING STARTED

Welcome

Welcome to Auria, a full-featured DAW designed from the ground up exclusively for the iPad. This User Guide is designed to introduce the various concepts of Auria, explain the available features, and serve as a reference for each of the tools, settings, and processes found inside the app.

Installation

Installing Auria is very easy, as the app will install automatically after purchasing through Apple's App Store. And if the app ever needs to be reinstalled (on a new iPad, for example), simply use the Purchased tab of the App Store from the iPad, find Auria in the list, and tap to immediately reinstall.

3rd party effect plug-ins can be purchased in-App via the Auria Store, and will immediately be available for use. Any purchases made through the in-app store can be automatically reinstalled (on a new/restored iPad) by using the Restore Purchases button inside the Auria Store.

Screen Orientation

Auria is designed to primarily be used in landscape orientation, i.e. held so that the screen is wider than it is tall. Portrait orientation (taller than it is wide) is available only in the Mix Window, where full-length 100 mm faders can be shown (with the tradeoff of less channels being in view); the Edit Window won't be available when in Portrait Mode, however, as it is fully optimized just for landscape mode.

Gestures

As a DAW designed for the iPad, Auria cannot rely on common input methods found in other computer-based DAW's. There are no keyboard shortcuts or right-mouse buttons available, so every input must be made with a touch "gesture". This section will go over some of the basic gestures used in Auria, which with a little bit of practice should become very natural in use.

There are several common gestures found in iOS apps that are utilized in Auria:

1. Tap
2. Pan (or Flick)
3. Pinch
4. Tap and hold
5. Double-tap
6. Double-tap and hold

Tap – The simplest gesture, performed by quickly tapping (and letting go) one finger a single time. Used for switches (like Mute or Solo), selecting entire regions in the Edit Window, closing pop-up windows, etc.

Pan – Also called a Swipe or Flick. A simple sweeping gesture made with one finger, either horizontally or vertically, which shifts the field of view or moves an object. In the Mix window a horizontal pan shifts between visible sections of the mixer. In the Edit window panning horizontally shifts the view along the timeline, while vertical panning shifts between visible tracks.

Pinch – A two finger gesture, where both fingers are used at once. Only used in the Edit window for zooming, either horizontally (time) or vertically (track heights). Note: Panning and pinching can be combined in the Edit window, allowing the user to both adjust zoom level and position in the timeline simultaneously.

Tap and hold – Also referred to as a long hold. A basic single tap that is held down. Used primarily in the Edit window for moving regions and adding automation control points.

Double-tap – Two quick taps in a row.

Double-tap and hold – Also called a long double-tap. Two quick taps in a row without letting go after the second tap. Used in the Edit window to highlight a section: double-tap the desired selection start and then swipe horizontally to highlight.

Differences Between iPad Models

At the time of writing, Apple has produced four primary iPad models, referred to in this manual as First Generation, Second Generation, Third Generation, and Fourth Generation. As each model has different technical specifications (such as CPU, RAM, disk size and speed, etc), Auria has been tuned differently for specific models.

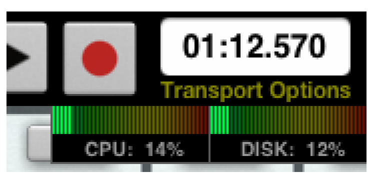
	1st Generation iPad	2nd Generation iPad	3rd/4th Gen iPad
Track Count	24	48	48
Subgroups	4	8	8
Aux Delay Compensation	No	Yes	Yes
Sample Rates Supported	44.1/48 kHz	44.1/48/96 kHz	44.1/48/96 kHz

System Optimization

While Auria has been designed to maximize the potential of the iPad as a recording/mixing platform, it is possible to over-tax the iPad's system resources just like on a desktop-based DAW. When Auria detects either low CPU or RAM resources remaining it will pop-up a warning message. The best rule-of-thumb, especially when working with large projects, is to close all other background apps when using Auria.

To close running background apps, double-tap the iPad's Home button (or use a Four-Finger Up swipe) to reveal the Multitasking Bar, and close any other running apps by touching and holding the first app until an X appears, and then clicking any open apps to close them.

Included in Auria's Mix window is an optional CPU/Performance meter (enabled from the main Settings window), a helpful indicator to how hard the iPad is currently working and how many resources remain. The meter can be cycled through the following displays by simply tapping the meter:



The Performance Meter

- **CPU & DISK** – Displays the current CPU and Disk sub-system performance, in percentages. Lower numbers indicate lower usages.
- **MAX CPU & MAX DISK** – Same as previous except the largest values are held. Double-tap to clear held values.
- **BATTERY & FREE SPACE** – Remaining Battery and available storage Space, in percentages.
- **FREE MEMORY** – Amount of unused RAM available, in megabytes.

If Auria detects either the CPU or Memory usage approaching a critical state a warning pop-up message will be displayed. In the case of either message immediate action must be taken to prevent the iPad from becoming unstable and potentially closing Auria.

CPU Overload - Please use a higher record buffer size, or reduce the number of active plug-ins in your project.

Low Memory - Your iPad is running low on memory. It is recommended that you close any unnecessary background applications, or reduce the number of plug-ins loaded immediately.

One easy step to decrease both CPU and Memory usage is to Freeze audio tracks which contain effects (like the ChannelStrip or insert effects). When a track is frozen Auria will automatically render the active effects on that track to a new audio file, bypass those effects, and free up system resources. See the Mixing chapter for more information on freezing tracks.

If the CPU Overload warning appears during recording try setting the Record Latency to a higher value (found in the Settings menu). This will make Auria more stable under high track counts during record with the tradeoff of higher monitoring latency - using an external USB audio interface which supports hardware monitoring will avoid this latency by monitoring at the interface's input instead.

3rd-party Plug-ins

Auria supports the purchase of 3rd party effect plug-ins through the built-in Auria Store. After purchase these plug-ins instantly become available and ready for use in projects. Developers such as PSPaudioware, FabFilter, Overloud, and others have converted their plug-ins to the iOS platform and offer them through in-app purchase.

Windows and OSX format plug-ins are not directly compatible with Auria, as iOS is a unique operating system with a different architecture than traditional Macs or PC's. In order for a particular plug-in to work in Auria it must be compiled by the plug-in developer specifically for iOS, and then distributed through the Apple App Store (via in-app purchase).

These plug-ins must normally be purchased through Auria's in-app store because Apple does not allow iOS users to install their own software manually. All software distributed on iOS devices must go through Apple's own systems, there is currently no way to install outside software in iOS.

However in Auria 1.13 an additional system was introduced where certain stand-alone audio apps available in the App Store (such as Sugar Bytes' Turnado and WOW2 Filterbank) have been updated and, if the full iOS version of said app has been purchased, an Auria-specific version will be unlocked in plug-in form automatically – without requiring an additional purchase.

WaveMachine Labs has an SDK available for plug-in developers who are interested in converting their plug-ins to the iOS platform. These plug-ins can then be offered through Auria's in-app store. For more information please contact WaveMachine Labs directly.

New in iOS version 7 Apple introduced a feature called Inter-App Audio, where separate standalone iOS apps can be used with Auria's Inserts and Aux Channels. For more information on using Inter-App Audio within Auria please refer to the Inter-App Audio and AudioBus chapter.

Saving in Auria

Auria is not a traditional Mac or Windows program but a fully iOS-based app, and as such treats saving projects like other iOS apps. Unlike a Mac or Windows program Auria does not have a normal Save button, as iOS apps are designed to always save automatically in the background, eliminating the need to worry if a document has been saved or not. Any time there is a change in a project, any change whatsoever, Auria has automatically saved the project.

There is an extensive file management system included in Auria which includes the ability to copy, rename, backup, and delete projects, and also manage the individual audio files which make up the actual projects. Please refer to the File Management chapter for more information.

Differences between Auria and Auria LE

In addition to the full version of Auria, there is a scaled-down version available called Auria LE. The primary differences between the two versions include:

	Auria LE	**Auria**
Tracks - Playback	24	48
Tracks – Recording	8	24
Subgroups	2	8
Sample Rates	44.1kHz	44.1/48/96kHz
AAF Project Import/Export	No	Yes
WIST and AuriaLink	No	Yes
Output Matrix	No	Yes
Track Freeze	No	Yes
Normalize, DC Offset, Reverse, Silence	No	Yes
Double-Precision 64-bit Mix Engine	No	Yes
Aux Channel Plug-in Delay Compensation	No	Yes
Audio Scrubbing	No	Yes
Time Stretching	No	Yes
Automatic Tempo Matching on Import	Yes	Yes
Convolution Reverb	Optional Purchase	Included

Brickwall Limiter	No	Yes
PSP Master Meter	No	Yes
MIDI Sync/Remote/Plugin Control	No	Yes
Price	$24.99 USD	$49.99 USD

Lynda.com Video Tutorials

Online software training site, lynda.com, has recently added Auria to its library of tutorial videos. This is a subscription service, so while the full 2.5 hours of Auria coursework is only available to lynda.com members, there are still several public videos available in order to check out the full course.

The full series goes fairly in-depth and includes topics like automation, crossfades, Audiobus, and importing video. The full list of tutorials is available on the lynda.com site.

lynda.com's Course Description - Auria is the first major digital audio workstation designed specifically for the Apple iPad, and in this course, author and professional musician Garrick Chow demonstrates how to use its recording, editing, and mixing tools to create great-sounding music. First, Garrick reviews the hardware you'll need to start capturing audio, from microphones to cables and input devices. He then demonstrates how to record anything from a single audio track to a complete multitrack capture of a live band performance. Once the recordings are done, he shows you how to edit them by adding splits and trims, as well as how to apply effects and use automation in creating a final mix. Lastly, Garrick reviews the options for exporting your project from Auria in several formats to share it with the world.

2

NEW USER TUTORIAL

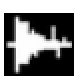
In this section we will create a simple project and learn some of the essential features necessary before using Auria to record, edit, and mix your own creations. As we want to keep this as easy to follow as possible please make sure that no external audio interfaces are connected (except headphones). Auria's USB recording capability is an exciting feature but it will be simpler to start off using the built-in hardware only.

The first time you open Auria it will open a demo project called, "The Approach". For tutorial purposes let's start with a brand new project:

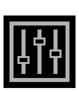
1. Tap Menu on the top menu bar
2. Tap New Project
3. The New Project dialog will open. Change the Tracks selector to 0 tracks, and leave everything else at the default values.
4. Tap Return on the keypad

Auria will open a brand new project (called New Project 2), and display the project's Mix window. This will be a completely empty project, without any tracks, regions, or effects, so the only channels you will see are the Subgroups (either 4 or 8, depending on your iPad model) and Master channel.

First let's navigate to the Edit window:



Tap the Edit window icon in the upper-left hand corner and Auria will switch displays. Make sure you are in Landscape Orientation (i.e. hold the iPad so it's wider than it is tall) as the Edit window cannot be accessed in Portrait mode.

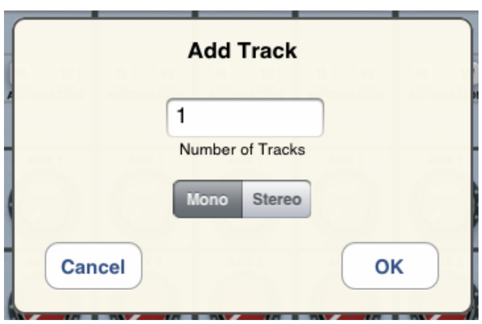


Switch back to the Mix window by tapping the adjacent Mix window icon.

An empty project – note only Subgroups and Master exist at first

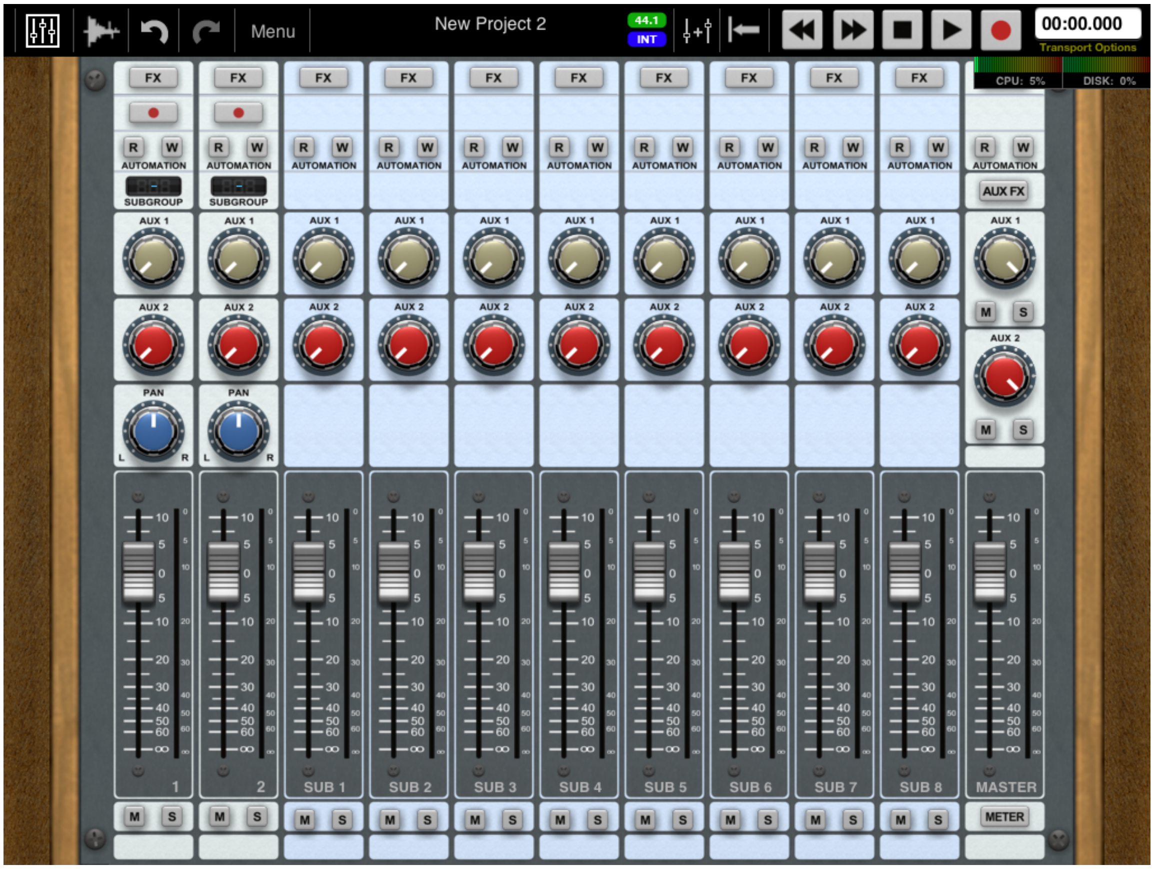
Recording

Before we can record anything we'll need to create some tracks. Tap the word Menu at the top of the screen to display the Main Menu, and tap Add Track. A dialog box will open asking for the number of new tracks to create, and whether they should be Mono or Stereo.



Add Track dialog

Change the options to 2 Tracks, Mono and tap OK. We now have two new channels displayed in our mixer, simply labeled as 1 and 2.

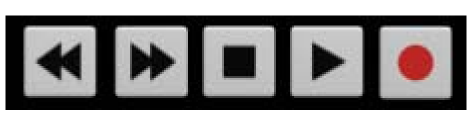


Mix window with two tracks on the far left

Let's record some audio on that first track. For the purposes of this tutorial we will use the built-in iPad microphone for recording, so make sure there are no audio interfaces connected (again headphones are fine).

Arm track 1 for recording by tapping the red button at the top of the channel (and just below the word FX) and the button should start flashing, indicating that track is armed.

Since we have no external audio devices connected Auria will default to the iPad's built-in microphone. A simple snap of your fingers near the mic (found on the same side as the headphone jack, near the middle) should make track 1's meter display some signal. Once you see signal on the meter we're ready to record!



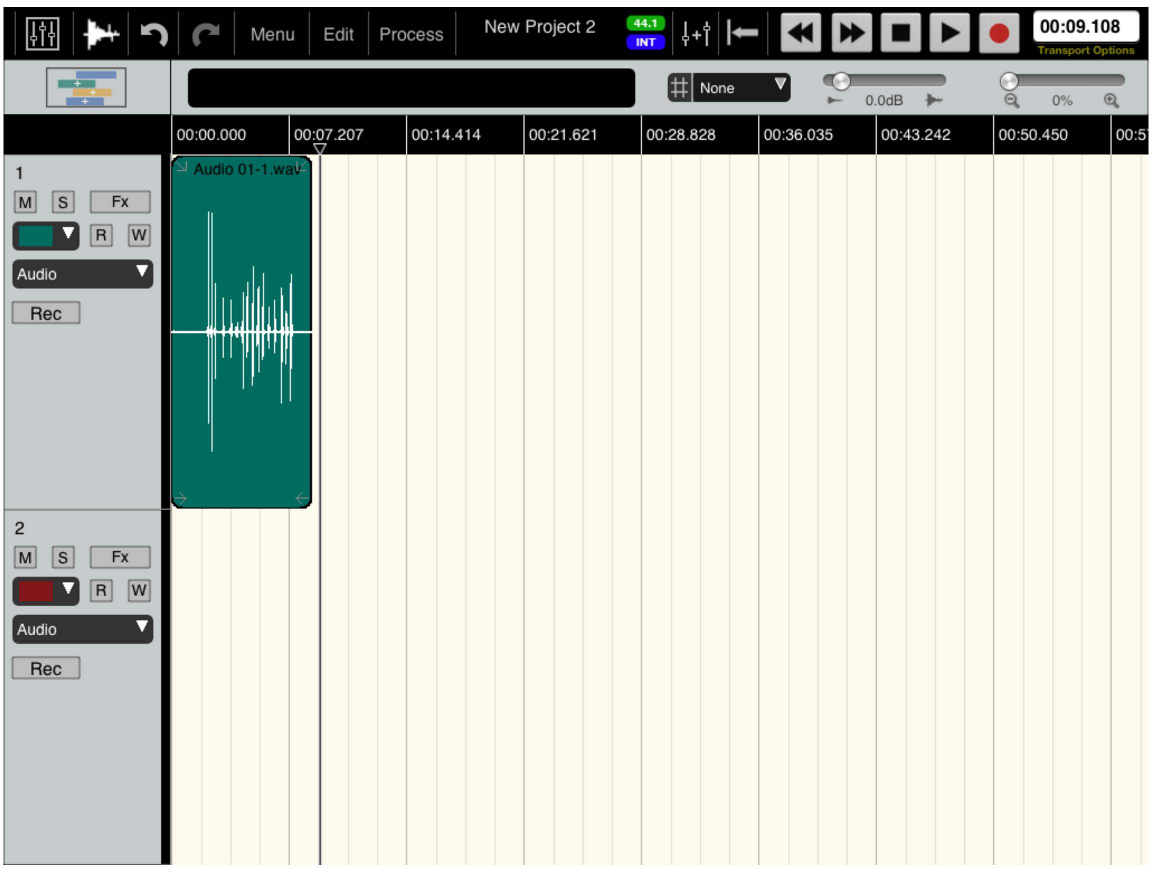
The Tranport – Record is on the far right

Press the large Record button on the transport (upper-right hand corner) to indicate we're recording, and then press the Play button to begin the actual recording. Now, make some noise! Talk into the built-in mic, snap your fingers, tap out a groovy pattern on your table, but just make sure it's loud enough to make the meter move. Once you've recorded several seconds worth of audio press Stop on the transport (a brief window will flash across the screen while Auria saves and updates the project), and lastly tap the flashing red button to disarm track 1 (it should respond by no longer flashing). Congratulations, you've just finished your first tracking session in Auria!

Let's run the project back and take a listen to this masterpiece. Double-tap the Rewind button in the transport to automatically jump to the start of the project, and then press Play to have a listen. Make sure your iPad's volume control is up/un-muted in case you don't hear anything, but don't worry too much if it's a little quiet as that will work fine for the purpose of this tutorial.

Editing

Let's take a look at your first effort in the Edit window. Switch to the Edit window and you should see a single region has been added on track 1; let's take a closer look at that region. To do this we'll need to learn how to zoom using a pinch gesture.



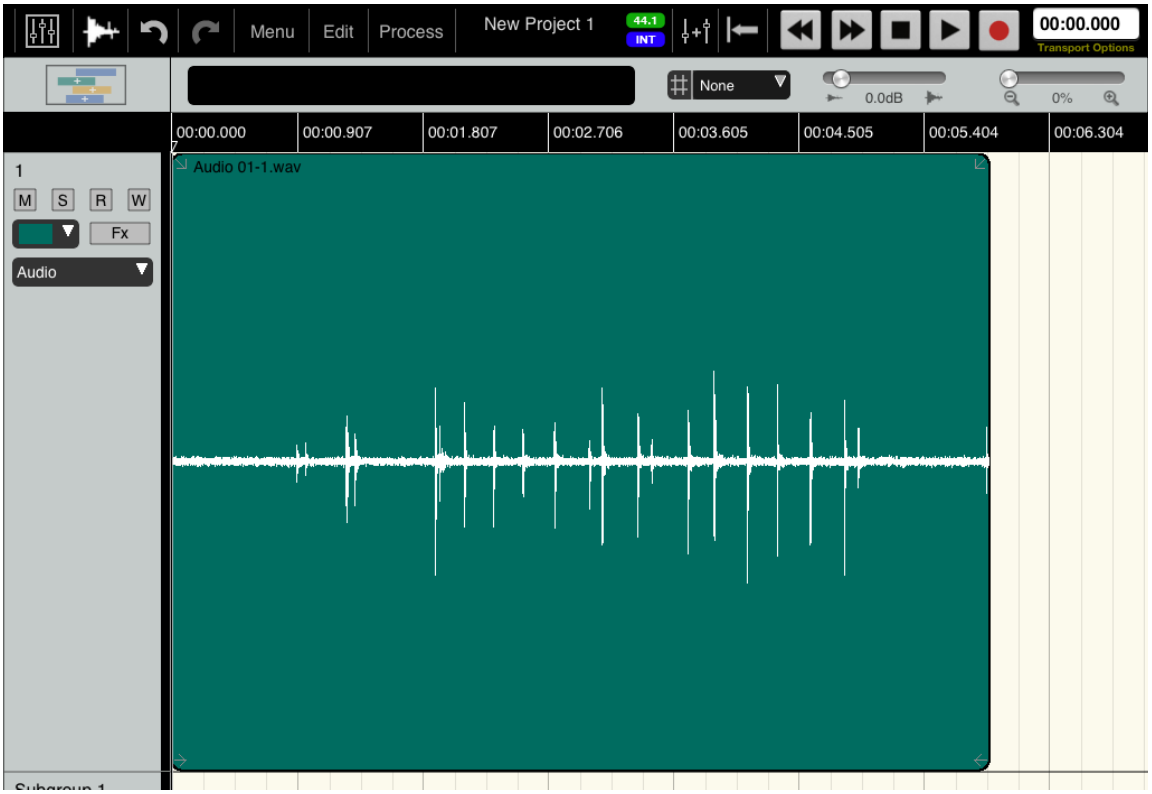
Our new region is at the very top – time to take a closer look

Touch the screen with your thumb and index finger, one above the other and close together, and slide the two fingers away from each other vertically. This will zoom vertically and make each track's height larger. Don't worry if you ended up zooming in to the empty track: simply use one finger to scroll up to track 1 by swiping downward (this may seem counter-intuitive at first, but imagine you were sliding a piece of paper across the table) until you reach the top and can see Track 1 with your region.

Vertical pinch (zoom) adjusts track heights

We'll zoom in time-wise (i.e. horizontally) by using another pinch, but this time place your thumb and index finger near each other side-by-side, and slide them away from each other horizontally; this will zoom in along the timeline. Again don't worry if you zoomed into an empty area, you can swipe horizontally to move earlier along the timeline and find your region (if you zoomed in after the region, swipe to the right in order to move left – again imagine you're sliding a piece of paper.) Where you place your fingers when pinch zooming has a direct relationship on the zoom itself, you are in essence stretching the display between your fingers.

Horizontal pinch (zoom) adjusts the timeline



Our region zoomed in nice and tight – good table tapping!

You should now be viewing your newly recorded region up close. Let's take another listen so double-tap the Rewind button and press Play. Ah ha! Yes, that was the Cursor moving across the screen, which shows you exactly where you are in the project. The cursor can also be moved more precisely by tapping/swiping in the Timeline ruler itself, try it out and move the cursor to a few different places; you should see the Counter display (upper-right corner) updating right along with the cursor itself. The cursor is useful both in signifying where to start playback but also in editing: regions can be split at the cursor, and objects can be snapped to the cursor as well.

Let's try out some non-linear editing with Auria and do a little slicing and dicing. Auria offers several methods that should be familiar to other DAW users such as moving regions, trimming, splitting, separating, deleting, and even utilizing crossfades. We'll begin with trimming, which in Auria is referred to as editing the region's Trim Handles.

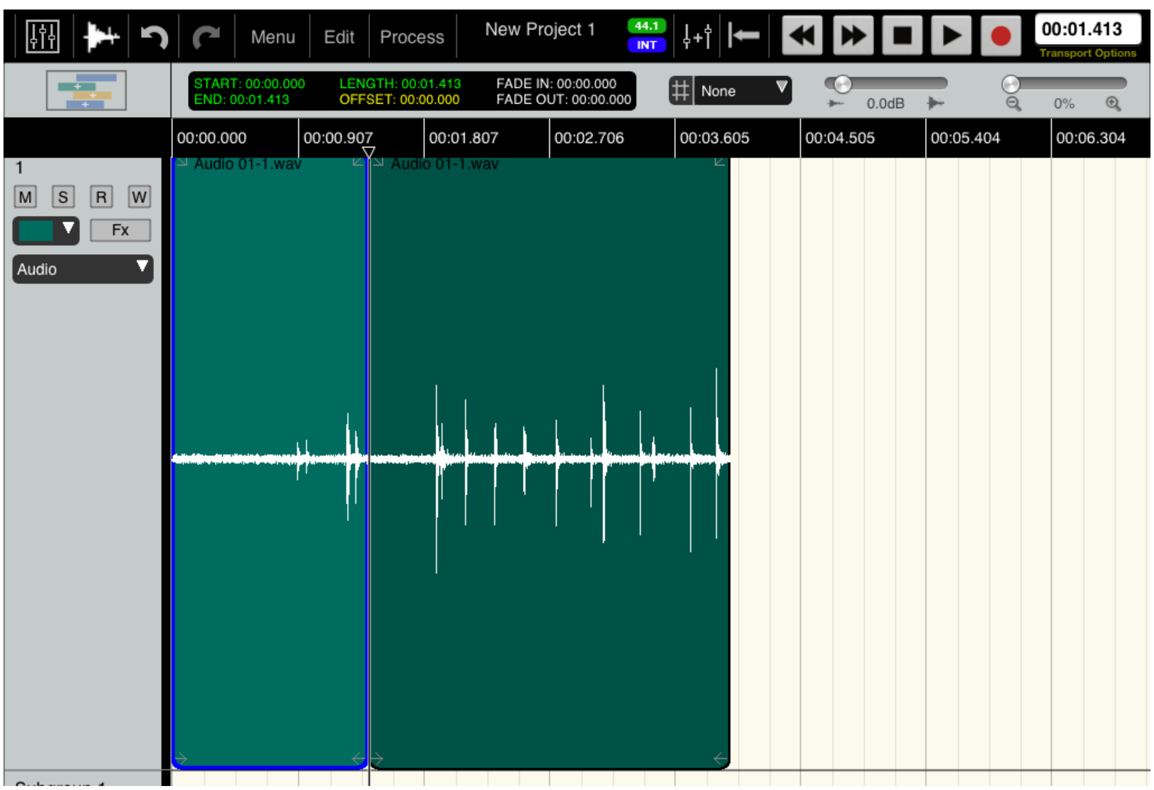
Look at your region's two bottom corners and you should see two arrows, each one pointing inwards. [→ ←] These are the Trim Handles and either can be moved to re-size the region; we'll start by shortening the end. Tap and hold the arrow (←) in the bottom-right of your region, the whole region should turn blue around its edges and a much larger arrow should appear. Keep your finger pressed to the screen and start swiping to the left, the region's end handle should begin trimming.

Note: If you find it difficult to touch the trim arrow itself (and accidentally end up selecting the track below it instead), keep in mind that you can tap anywhere near the corner and not just the arrow itself; give yourself some room and aim further inside the region's corner.

At this point if you make a mistake and accidentally do something you'd rather forget just tap the undo button in the upper-left corner to go back one step (or more!)

Now try out the same technique by trimming the beginning of the region by dragging its handle to the right. When you're done you will have shortened your recording, a very useful technique to eliminate background audio left over from the beginning and end of a take.

Now that we've shortened this region, let's split it in two. Tap the Timeline ruler somewhere in the middle of the region to place the Cursor, and then tap the region itself once (again it will turn blue) to select it. Tap the Edit menu up in the menu bar and find the Split command, and give it a tap to split your region in two.



The region split in two

Let's add some time between our two clips by moving the second one later in the timeline. Moving a region is very simple in Auria: tap and hold anywhere in the middle of the region (just stay clear of the corners!) and the region's edges will turn red – this indicates you are actively moving the region. Keep touching the region so it stays red and swipe your finger to the right - the region should move right along with you. Slide it over until there's a noticeable gap between the two regions of at least a second or two.

After moving the second region

So far we've trimmed the beginning and end of the region, split it in two, and moved the second region. Now let's just get rid of that second region altogether. Tap the second region again to make sure it's still selected (blue), open the Edit menu, and this time select Delete Region. This does exactly as the name suggests and removes the second region entirely.

However, suppose later on you decide you want to bring back the section you've already removed. You could just tap Undo to go back one step, but as Auria is a non-linear editor we never permanently destroyed anything and can recover the deleted data at any time – even after closing and reopening the project. Just tap and hold the remaining region's right (i.e. ending) trim handle (the ← in the bottom-right of the region) until the region turns blue and the big arrow appears, and swipe to the right, revealing the "deleted" audio as you swipe. Go ahead and drag the trim handle all the way to the right, restoring your entire region as you go.

The last important part of editing to cover initially is highlighting, as this allows you to work with just a smaller section of a larger region. We'll now highlight a section of audio in the middle of the region. This is done by double-tapping and holding inside the region at the place you want to start highlighting. Then, with your finger still pressed to the screen, swipe to the right until you've highlighted a few seconds of audio before lastly letting go. You should see a light-gray highlighted section inside your region.

A highlighted section

If you want to adjust the beginning or the end of the highlighted section merely tap near either edge and swipe, the highlighting will follow your finger.

With something highlighted go back up to the Edit menu and tap Separate. Auria will create a brand new region from your highlighting, giving you three in total (one before and one after the highlighted section).

Our newly separated region

Congratulations! You've now recorded something into Auria and learned some of the simpler editing functions. The last thing we'll cover in this tutorial is getting around in the Mix window.

Mixing

Switch back to the Mix window by clicking on its icon in the upper-left corner. Try moving the track 1 fader by touching the fader itself and swiping up and down; the fader will follow along with your finger. Since the iPad supports multi-touch input, use both hands and adjust several subgroup faders at once. The same thing works with buttons (and knobs) so try muting a few channels at once. Can't do that with a mouse, can you? Once you've messed up our perfectly zero-ed mixer simply double-tap any fader or knob to reset it to 0, and make sure our channel is not muted.

Tap the FX button to open the PSP ChannelStrip. Here we will find the included Expander/Gate, EQ, and Compressor sections. Each section has its own Enable/Disable switch, and as you should see the EQ is the only one currently active, though no bands are in use.

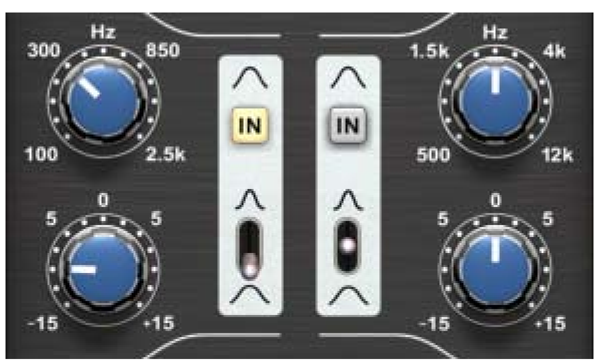
PSP ChannelStrip

The very first thing we need to do is take the ChannelStrip out of bypass by tapping the Bypass button in the upper-left corner of the window (ChannelStrips are bypassed by default in new projects as a means to save on processing power), once the Bypass light is off the ChannelStrip is active.

Let's try to EQ the mid-range a little bit. Find the two fully parametric EQs just below the High and Low pass filters.

Two bands of sweepable parametric EQ, with adjustable Q

Engage the left-side parametric section by clicking the IN button (causing it to light up). Now find a frequency and try sweeping the Boost/Cut (gain) knob. By default Auria uses vertically-linear knob control, which simply means that to adjust a control tap and hold the knob and swipe up or down to move the knob. Try the 3-way Q toggle (tap the switch, hold, and slide up and down) to adjust the bandwidth.



EQ set to cut about 10dB at 300 Hz with a wide Q

Rewind back to the beginning of the project (double-tap the Rewind button), press Play, and take a listen to your EQ. Feel free to tweak the frequency, gain, and Q controls to get a better sense of the interface.

Now that we've explored the ChannelStrip head back to the mixer by clicking the X in the upper-right corner to close the ChannelStrip, revealing the mixer behind. We'll try out the Aux section next. You will notice every audio track and subgroup channel has two Aux Sends, labeled 1 and 2. We are going to setup a simple reverb on Aux 1.

Find the Aux Effects section above the Master fader. The returns will already be turned up, so we just need to tap the AUX FX button to open the Aux Effects section.

From there click the top insert point (just below the AUX label) to open the list of available effects and tap ClassicVerb to insert a simple reverb on Aux 1. The reverb's interface will open automatically.

The ClassicVerb, a very low CPU usage reverb

Since we're using this on an aux send let's start by turning the MIX knob to 100% (wet). Adjust the TIME knob however you'd like, and once you have a good starting point tap the X and close the ClassicVerb dialog, then tap the X and close the Aux Effects dialog as well.

Find the AUX 1 Send knob on track 1, turn it up (at least half-way), and press Play in the transport. You should now hear your recorded masterpiece with a little verb, but if not try turning the send up all the way.

Well done! You've successfully used Auria in all three phases: Recording, Editing, and Mixing. From here feel free to explore the rest of the User Guide for more in-depth information on the various features, tools, effects, and settings you'll find inside the app.

Part Two – Working in Auria

3

RECORDING

Audio Hardware Setup

The first step before recording audio in Auria is to take a look at the hardware setup. Auria supports two main categories of audio hardware, the iPad's built-in internal I/O (Speaker, Mic, and 3.5mm headphone mini-jack), and external audio interfaces connected through the 30-pin/Lightning MFi port (either directly or through the Camera Connection Kit).

Internal I/O

The iPad's built-in audio hardware is available to Auria, and includes the following devices/connectors:

- Speaker
- Microphone
- 3.5mm mini-jack – Includes both mono input plus stereo out.

When no external audio interface is connected, Auria will display INT in the top Menu bar, just below the current sample rate. Devices like IK Multimedia's iRig series, which connect through the 3.5mm mini-jack, are considered to be internal (INT) devices in Auria because they still use the iPad's own I/O.

Note: The built-in microphone and speaker do work with Auria, and work well for simple testing purposes or "scratch track" recording; using an external audio interface (through MFi or USB) is the best option for high-quality audio recording.

If Auria is not recording any incoming audio with iOS 7 installed

iOS 7 can block all audio inputs (from both the microphone and from interfaces) to an app unless it is specifically allowed.

- When first opening an app under iOS7 a pop-up asks, "Auria would like to access the microphone."
- Tap "Allow" to enable all audio input to Auria
- If "Don't Allow" was accidentally selected then all audio input to Auria is blocked

- To fix this open iPad Settings, go to General, then Privacy, then Microphone. Find Auria in the list and enable access by tapping the slider so it is green.

USB/MFi

In addition to using the iPad's internal audio hardware, Auria also supports external audio interfaces through the 30-pin/Lightning connector (or MFi). There are two types of devices which can be connected through this port:

- MFi - These interfaces connect directly to the 30-pin/Lightning MFi connector. One popular example is the Apogee Jam.
- USB Class 2 Compliant interfaces - These require an Apple Camera Connection Kit (replaced by the Lightning to USB Camera Adapter) connected between the iPad and interface. Examples include the RME Fireface UCX, Presonus AudioBox 1818VSL, and the Focusrite Scarlett series interfaces.

For the latest list of tested USB interfaces please refer to the Auria Interface Compatibility page on www.auriaapp.com. Please note that some USB interfaces require a powered USB hub between the Camera Connection Kit and the interface. The list of compatible USB Class 2 devices is changing rapidly, so please check both with the audio interface manufacturer and the Auria website's Support section for the most up-to-date information.

When Auria detects a compatible audio interface connected through either through MFi or the Camera Connection Kit the "INT" indicator on the top Menu Bar will change to "USB", and the list of available inputs will update.

USB Soft Start

During testing some USB audio interfaces were found to make brief but noticeable noise during initialization. USB Soft Start mode will initialize USB interfaces more slowly before recording, preventing this noise. This mode is on by default.

Project Setup

Before recording a brand new project in Auria, there are a few project options which affect the upcoming recording.

New Project Dialog

- **Project Name** – Enter the name of the project
- **Sample Rate** – Select the desired sample rate (this cannot be changed later)
- **Tracks** – Select the number of pre-made mono tracks (stereo tracks can be added later)

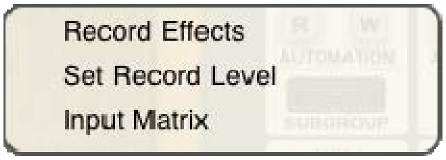
Note: Not all sample rates available on all iPad models, see Getting Started chapter for more information.

Track Setup

Recording audio on a track requires some initial configuration of elements like input routing and level adjustment.

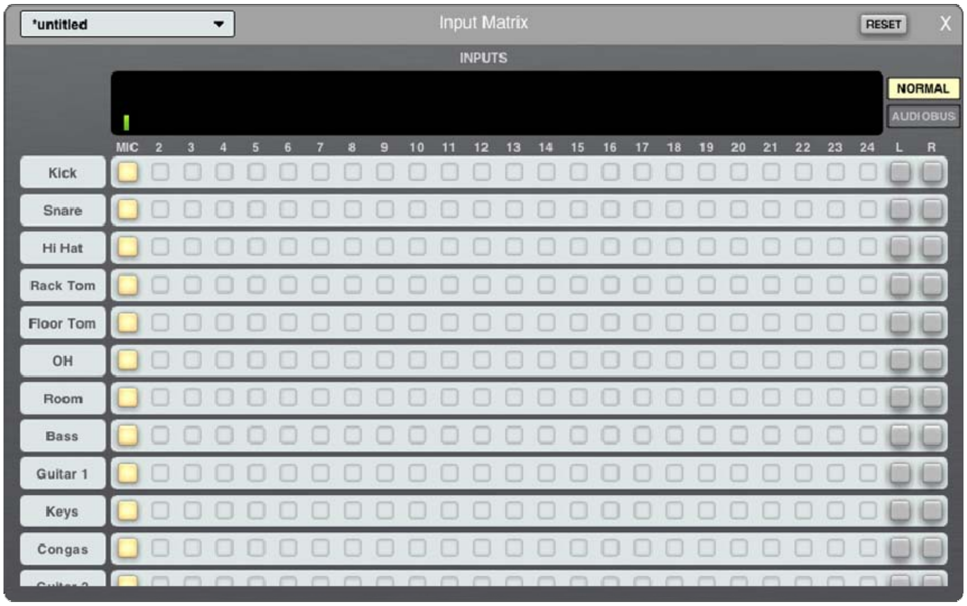
Input routing will depend on the particular audio hardware being used; using the built-in iPad audio system is very simple, using a multi-channel USB interface can mean a large number of inputs to navigate. Fortunately Auria includes a single view to make the task as painless as possible: the Input Matrix.

Input Matrix



Record Options Pop-Up

To open the Input Matrix, either tap the main Menu and select it, or tap and hold the red record enable button found at the top of the track and select it from the pop-up list.



The Input Matrix

There are two axes in the matrix: the physical hardware inputs on the horizontal axis (including Left and Right master bus, used for bouncing audio), and Auria's track list on the vertical axis (slide up and down to move between all 48 tracks).

To route a particular hardware input to a specific track simply find the intersection of input and track and tap the corresponding radio button. By default Auria will route Input 1 to Track 1, Input 2 to Track 2, and so on. To reset Auria to this default tap the Reset button in the upper-right corner.

Above the matrix itself is a handy meter bridge which displays the signal level currently present on each input. Seeing signal across all the inputs at once is useful during complicated setups for tracing signals during a line check.

Different input routing schemes can be saved as presets using the menu in the top-left corner, making complicated setups much easier to switch between. Tap the drop-down menu to save the current setup as a preset, or load an already existing scheme from the list.

Audiobus users can also route Audiobus ports to specific tracks for recording. To do this first switch from the Normal inputs (i.e. hardware) to Audiobus by tapping the corresponding button in the upper-right corner of the panel. For more information please see the Audiobus chapter.

Output Matrix

Auria also includes a panel for routing Auria's outputs to specific channels on a connected USB interface.

Available outputs include:

- Subgroups
- Aux 1 & 2
- Master (LR bus)

To open the Output Matrix select (tap) it from the Main menu.

Output Matrix

The Output Matrix is very similar in layout to the Input Matrix, and includes the same preset system for storing common routing setups. There are also some additional buttons found only on the output panel:

- Mono – Sums that particular subgroup or aux channel to mono. Useful when routing a mono instrument (like bass guitar) to a subgroup for monitoring purposes, as it then won't take up two separate outputs on the audio interface.
- Int – Short for internal. When lit (on) routes the selected output to the Master (LR) bus in addition to any other hardware routing assignments. On by default, turning off is useful in instances where a particular output shouldn't be heard in the main outs, like when using aux sends for a separate headphone mix.

Record Level

Auria includes a track-specific input gain adjustment for controlling record levels, separate from the audio interface's own gain controls. To adjust the recording level on a particular track tap and hold the red record enable button until a pop-up menu appears, and then tap Record Level.

Record Level pop-up

- Tap and drag the gain knob to boost or cut the recording signal level.

As a general rule-of-thumb, digital-based recording systems, like Auria, have a very large amount of dynamic range available. This, coupled with a good low-noise analog signal chain before the analog-digital converter, means that concern over the signal-to-noise ratio in traditional analog recording is a much less significant factor in the digital realm. To put it more plainly, since there isn't any concern over things like tape noise or channel crosstalk, it's best to allow for a healthy amount of headroom when recording. One common recommendation is to aim for peak levels around -18 dBFS (or even lower), though this is up to the end user.

One final note is that this is a software-based level control, and as such it is only scaling the audio signal present at the physical input. Please take care to monitor signal level at the audio interface, as any clipping that occurs there will not be fixed by lowering Auria's record level; it will simply record a quieter version of distortion.

Recording with Effects

An additional option found in Auria's Recording Options pop-up (accessed by tap-and-holding the red record enable button on the top of each track) is whether to record with effects or not.

Normally Auria records the dry signal only, and any effects present on the record track will be monitored only (like the ClassicVerb) and not end up in the actual recording. However, some engineers may prefer to record the channel effects (possibly to save on system resources later during mixing). To enable this function simply tap the Record Effects option in the Record Options pop-up (it will become checked).

Note: The Convolution Reverb is not available in this mode as it is too CPU-intensive to record stably.

Record Monitoring

One important consideration in any digital recording system is the issue of latency. In reference terms latency refers to the amount of time which passes between a stimulus and the response, and in the digital audio world many components have some degree of inherent latency.

Auria includes a number of options that will affect that latency, based on a particular recording setup. The following options are found under the Settings window and are concerned with monitoring during recording.

Record Monitor – Turns on or off monitoring of recording tracks through software. When turned on Auria will send signal present on record-enabled tracks to the main out, allowing monitoring to be done through software. Some external USB interfaces support monitoring at the inputs (i.e. hardware monitoring), in those cases turn off Record Monitoring to prevent hearing doubled audio.

Record Latency – Dropdown box that selects the size of the audio recording buffer. The higher the number, the more stable the system is. The lower the number, the less latency is present when record monitoring through software. If using hardware monitoring through an external interface then it is recommended to set this to the maximum latency for the most stable recording environment.

Disable Effects While Recording – Defaults to No, which means any effects present in the project will be heard during recording. If set to Yes, all effects will be bypassed when Record is pressed. Included as a means of reducing CPU and memory usage during recording.

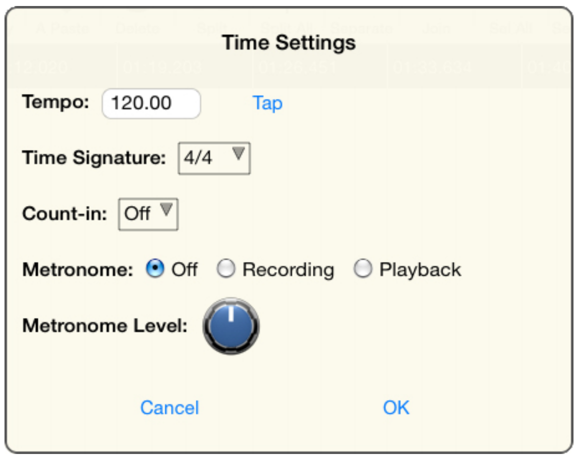
Record Latency Adjustment – Enter a time value, in samples, to shift recorded audio earlier during recording. Auria automatically attempts to detect the hardware latency of the attached audio interface (both internal or USB interfaces), and then compensates for it so that recorded tracks line up correctly. If a particular interface either doesn't report its latency, or the reported latency is incorrect, use this setting to manually compensate and have Auria shift recorded tracks by the amount entered. To determine an interface's internal latency a Loop-Back test should be performed (see below).

Loop-Back Test – To test an audio interface's latency, record (or import) a single percussive audio impulse to a track. Then, connect a cable from the audio interface's audio output back into its input, and re-record that impulse on a new track. Finally, measure the difference (in samples) between the two impulses using the Edit window. Switch to the Sample Time Format (tap Counter), zoom in, and highlight between the two impulses;

the Info box at the top of the editor will display the selection length in samples. Enter this value back in the Record Latency Adjustment box.

Metronome

The metronome can be used to provide a click track during recording, and can be accessed from the Time Settings dialog (double-tap the box displaying 120.00 BPM 4/4 in the upper-left of the Edit window).



The Time Settings window

- **Tempo** – Sets the project tempo
- **Time Signature** – Selects the project's time signature
- **Count-in bars** – Selects how many bars of the metronome to play before recording
- **Metronome** – Sets when the metronome should play
- **Metronome Level** – Adjusts metronome volume

The metronome can also be quickly toggled on/off by tapping the Counter (in the upper-right corner) and checking/unchecking the Metronome option.

Auto-Punch

Normally Auria will start recording wherever the cursor happens to be when the Record and Play buttons are tapped in the transport, and it will stop when the Stop button is tapped. Sometimes, though, a more specific section needs to be recorded (such as overdubbing an individual word on a vocal), and in these cases Auria can be set to automatically record only during a specific section. This is done in Auto-Punch mode.

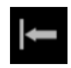
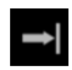
When recording with Auto-Punch, rewind to spot before the locators and tap Record and Play as usual in the transport, Auria will start playing but wait to start recording until reaching the Locator In point and then automatically jump into record. When the Locator Out point is reached Auria will automatically jump out of record and back to playback.

There are two components to using Auto-Punch: setting locators for the in and out points, and then enabling Auto-Punch mode under Transport Options.

Locators

Locators can be set in one of two ways, either using the Locator In/Out button on the top Menu Bar while the project is playing, or by highlighting across the Edit window Timeline.

To set the locator in and out points during playback:

1. While the project is playing tap the Locator In button on the top Menu Bar 
2. A confirmation window will flash on the screen
3. Continue playing until reaching the end of the desired Auto-Punch
4. Tap the Locator Out button (in the same spot as the In button) 
5. Another confirmation window will flash indicating the Out point was set

To set the locator in and out points by highlighting in the Editor:

1. From the Editor window, double-tap in the Timeline Ruler at the desired Locator In point
2. While continuing to hold, swipe to the right
3. When highlighted section reaches desired Locator Out point release finger from screen
4. Locator points can be adjusted by simply tapping and swiping the highlighted beginning and end points

One the Locator In and Out points are set, simply tap the Counter to open Transport Options, and tap Auto-Punch to enable it.

Bouncing

Bouncing is an important tool in a DAW which allows summing multiple audio tracks or sub-groups, each with their own effects and automation, and recording the results to a new track. This is most often needed when running into Auria's track limit, or to simplify a complex project by combining related tracks (i.e. bouncing down the 16 tracks of roto-toms to a single stereo track). Since the digital domain allows for bit-perfect copies to be made without any generation loss, bouncing can be done safely without worry of added noise.

To perform a bounce:

1. Setup the mixer so that only the desired contents of the bounce are audible, i.e. mute tracks or effects that aren't part of the bounce source.
2. Create a new track (either mono or stereo, depending on the source material)
3. Open the Input Matrix
4. Assign the new track's input to L and R (for stereo), or for mono content just L or R
5. Record enable the new track
6. Rewind to the beginning (or set the specific beginning of the bounce with the Edit Window cursor)
7. Tap Record and then Play in the transport
8. Be sure to listen to the bounce as it happens as a safety check
9. Tap Stop when playback reaches the end of the desired bounce source

4

MIXING

The Mix window in action

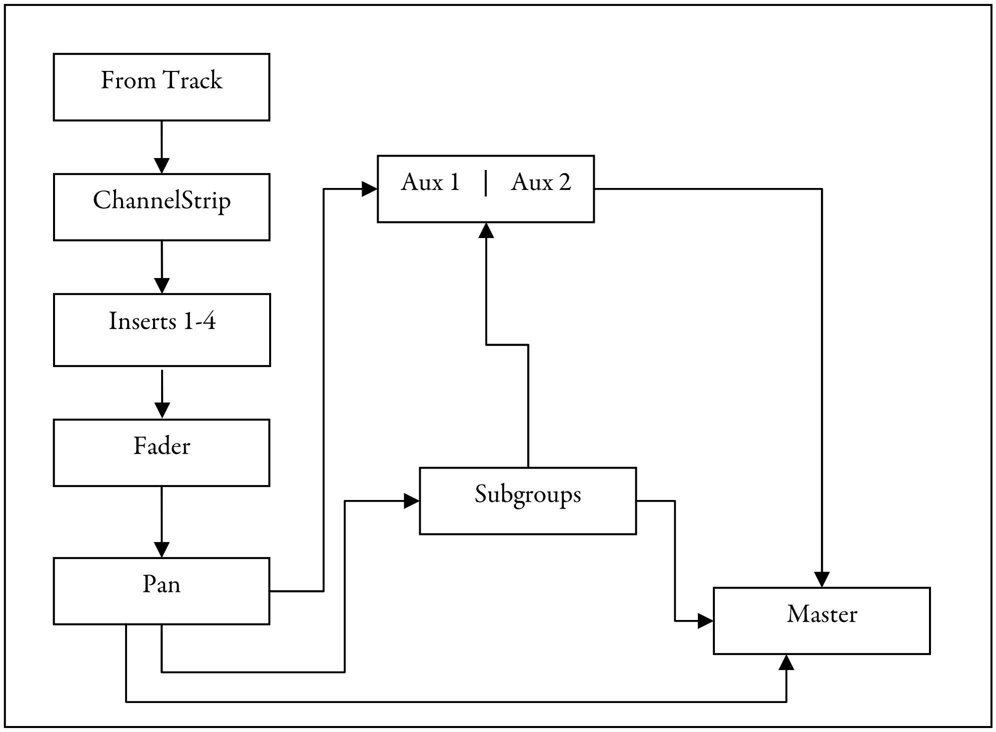
Auria has within it a powerful mixing system, capable of playing back 48 tracks of audio (on an iPad 2 or newer) at once, combined with top-notch effects processing (including EQ, compressors, gates, reverbs, chorus, delay), all designed to allow final mixes that can rival those done on desktop-based systems.

Just like a traditional mixing console, Auria's mixing interface is split up into several different sections:

- Channels
- Subgroups
- Aux Sends/Returns

- Master Stereo Channel

The basic signal flow between mixer sections looks something like this:



Auria Mixer Signal Flow

As the above diagram shows, signal can be routed from the individual channels through the aux sends and subgroups, with all signals ending up summed at the Master channel (which then feeds the monitor output: headphones, built-in speaker, external USB, etc).

In the following sections this guide will examine each of these components.

Channels

Auria can have up to 48 individual channels (but only up to 24 on iPad 1), in any combination of mono/stereo channels; these channels appear in the mixer and occupy the left-hand section. These channels are directly tied to their corresponding tracks in the Edit window, and in the context of Auria can be treated as essentially the same thing. In other words, track 1 will appear on channel 1, track 2 on channel 2, and so on; if channel 15 on the mixer is record enabled the resulting recording will appear on track 15.

Each mixer channel includes:

- PSP ChannelStrip, containing standard processing such as EQ and compression.
- 4 Insert Points

- Track Freeze
- Saturation
- Polarity switch
- Mute and Solo
- Pan knob
- Fader
- Track/Channel name

Some of the above controls are available right on the mixer itself, like the fader, pan knob, and mute/solo buttons. The rest are only visible when the channel's ChannelStrip window is open. Whenever there is a duplicate control that is found both on the mixer and in the ChannelStrip, Auria mirrors the control. If the channel's Mix window fader is visible along with its ChannelStrip, moving one fader will move the other automatically.

To open a channel's ChannelStrip, tap the FX button found at the top of the mixer (also found in the Edit window on the left-hand track pane).

The ChannelStrip view is split into two main sections, the three processing modules on the left (Expander-Gate, EQ, and Compressor), and the right-hand Fader section. For more information on the ChannelStrip's processing modules, please see the ChannelStrip chapter.

Note: Both the pan knob and the aux sends can be accessed from the ChannelStrip by switching from the Inserts panel over to the Pan/Aux view; tap the PAN/AUX tab in the upper-right corner.

Inserts

Each of Auria's channels contains four insert slots for additional processing. In terms of signal flow these insert points are post-ChannelStrip (meaning an EQ used in the ChannelStrip will affect a chorus placed on an insert). The insert slots are available on the right-hand side of the ChannelStrip.

Tapping an insert slot opens a scrollable window which shows all the available effects. These will include the standard effects included with Auria: PSP StereoDelay and StereoChorus, ClassicVerb, Convolution Reverb, and ReTune. See the Insert Effects chapter for specific information on each.

Any optional plug-in purchases will appear here as well. For more information on these in-app purchasable effect plug-ins please see the Optional Plug-ins chapter.

Note: Each plug-in includes both mono and stereo versions, and depending on the track type one or both might be available. This way it is possible to insert a stereo effect onto a mono track as the channel will then output in stereo (while the pan knob will still be in mono operation).

Track Freeze

No matter how powerful a computer system is, be it either desktop, laptop, or tablet, it will always be possible to need more processing power than the system can provide at once. In these situations Auria provides a track freeze option which can greatly reduce CPU and RAM usage, freeing up much needed resources during heavy mix sessions.

Track freezing works by automatically bouncing a particular track "in place", which means creating a new audio recording and substituting it into the same exact spot as the original. This new recording will include all of the ChannelStrip settings plus the 4 insert effect slots, so the frozen track will sound exactly the same as the pre-frozen version. Finally, Auria will automatically disable all of the aforementioned effects, freeing up their associated CPU and memory usage.

Once a particular track has been fully tweaked and is sitting well in the mix it is generally a good idea to go ahead and preemptively freeze it, even if the CPU meter shows plenty of headroom remaining. As the following section will show it is extremely easy to both freeze and un-freeze a track during mixing.

An unfrozen track's ChannelStrip, note the many enabled effects, including inserts

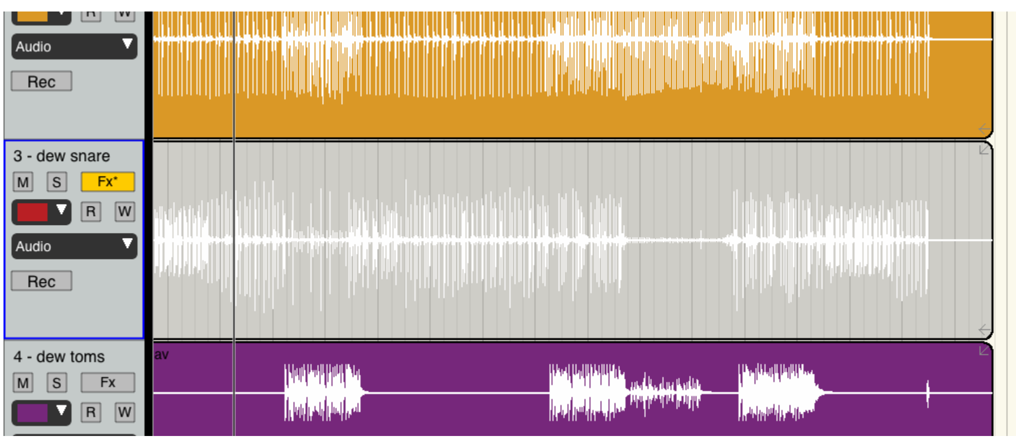
To freeze a track and free up its CPU usage:

1. Open the track's ChannelStrip window by tapping the corresponding FX button, either on the Mixer or in the Edit window
2. Tap the Freeze button, found to the right of the fader
3. A progress bar will pop-up while the track is automatically bounced in-place
4. Once the progress bar closes the track is frozen

The same track after freezing

A frozen track's ChannelStrip window will become grayed out, with a large snowflake superimposed over the entire window.

When a frozen track is viewed from the Edit window, its regions will also be grayed out, and a small snowflake will be displayed next to the FX button (the snowflake will be visible in the Mix window as well).



A frozen track as seen from the Edit window

Note: When a track has been frozen both the track's effects and regions will be locked from editing. To edit a frozen region or tweak a frozen effect the entire track must be first un-frozen. The track's fader, pan knob, aux sends, and mute/solo buttons will still be available even when frozen.

To un-freeze a track:

1. Tap open the ChannelStrip view
2. Tap the Freeze button
3. The track will immediately un-freeze and be freely editable again

Saturation

This algorithm emulates the sound of analog-style saturation by creating additional harmonics.

Polarity

This switch (Ø) inverts the channel's polarity. As an example, when recording both a top and bottom mic on a snare the bottom mic will often need its polarity flipped so it is in phase with the top mic.

Mute and Solo

Each channel has its own Mute (M) and Solo (S) buttons:

- Mute – Mutes, or cuts, the signal on the channel, removing it from being heard in the overall mix.
- Solo – Isolates the selected channel so only it is heard, used for auditioning a specific channel that is part of a larger overall mix. Multiple channels can be soloed at once. When one or more channels are soloed, a flashing red Solo message will appear below the Project Name on the top bar; tapping this flashing message will cancel all solos. This is an After Fade Listen solo, or AFL; the channel's fader position, panning, ChannelStrip, and insert effects will all be heard.
- Solo Safe Mode – Found in the Settings Menu, this global parameter determines what happens to the aux returns when a channel is soloed. When Enabled (the default), soloing a channel also solos both aux returns.

Pan

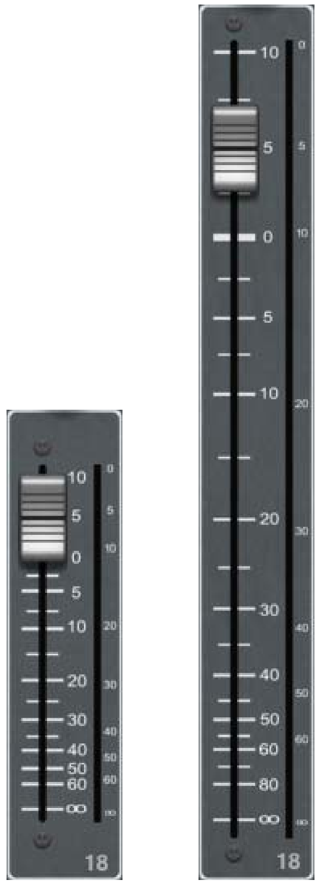
The pan pot controls the channel audio's position in the stereo field. Its behavior will depend on whether the channel contains either mono or stereo audio:

- Mono – Pans the mono audio across the stereo spectrum, from full left-center-full right. Includes selectable pan laws in the Settings menu.
- Stereo – Adjusts the relative balance of the stereo audio's left channel versus the right channel.

Fader

The channel fader determines how much signal is sent to the channel's destination (Master Channel or a Subgroup). In other words, it controls how loud the channel is in the overall mix.

The iPad screen orientation will affect the overall length, or "throw", of each fader. In the normal landscape orientation (where the width is greater than the height) each fader is about 45 mm in length. Turning the iPad to portrait mode (height greater than width) increases the fader throw to 100 mm, making smaller adjustments possible and more closely emulating large-format consoles, but decreasing the number of channels visible at once.



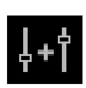
Comparison of fader throw length: Landscape mode on left, portrait on right. Images to scale, not exact size

Fader Grouping

Individual faders can be grouped together, so that adjusting one will equally adjust the rest of the group. When two or more faders are grouped their respective mute and solo buttons are grouped as well.

To create a fader group:

1. Tap the Group icon on the top bar
2. Tap each fader to add it to the group



3. Lastly tap the Group icon again when done adding faders
4. Each grouped fader will display a number signifying which group it belongs to

To remove a fader from a group:

1. Tap the Group icon
2. Tap the desired fader
3. The fader's group number will disappear
4. Tap the Group icon again when done removing faders

A couple of import caveats:

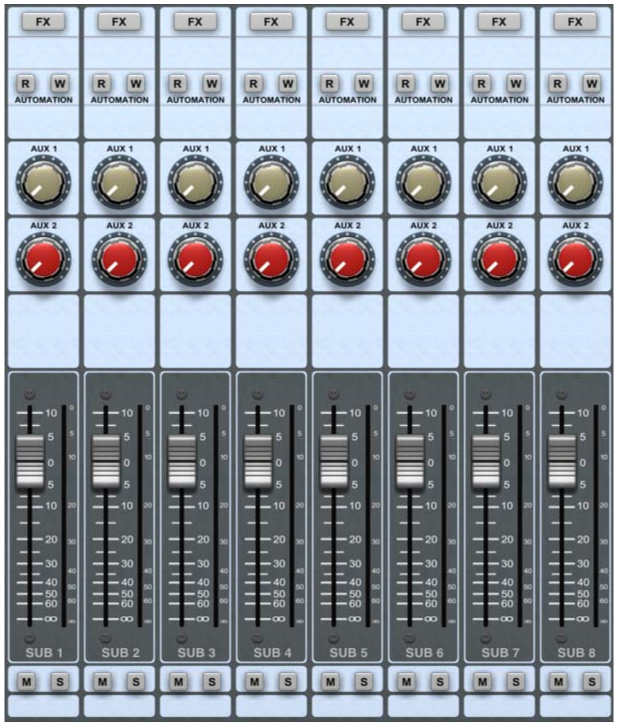
- All existing groups can be removed at once with the Clear All Groups command, found in the top Menu.
- Only channel faders can be group, not subgroups or the master channel.
- A fader can belong to only one group at a time.
- Existing groups can have faders removed (see above) but not added. To do this first ungroup the faders, then re-group them with the additional fader.

Track/Channel Names

Mixer channels can be named by simply double-tapping the blank scribble pad at the very bottom of the channel, just below the Mute and Solo buttons (this also applies to subgroups).

Subgroups

Auria's mixer contains 8 separate subgroups (only 4 on iPad 1) which can be used for advanced routing and bus processing. It works be re-routing the channel's output to a specific subgroup, instead of directly to the master channel, allowing selected channels to be summed and processed together first before being bussed to the master. For example, each drum channel can be assigned to a subgroup, and then by enabling a compressor on that subgroup the entire kit can be compressed together (which sounds different than setting up separate compressors on each individual channel).



Subgroup section of the mixer

To assign a channel to a subgroup:

1. Tap the SUBGROUP display towards the top of the channel
2. A drop-down list will appear displaying all the subgroups
3. Tap the corresponding subgroup from the list

Each subgroup includes its own PSP MasterStrip, which includes bus-centric EQ, compression, limiting, and saturation (see the PSP MasterStrip chapter for more information), in addition to the other standard controls: Aux sends, mute, solo, etc.

Subgroup MasterStrip

Notes: A channel can only be routed to one subgroup at a time, Auria does not support channels with multiple destinations (like assigning track 10 to subgroups 2 and 3). However, duplicating the original track and assigning each to different subgroups can accomplish the same thing.

Subgroups can be renamed just like tracks. Double-tap the blank scribble pad at the very bottom of the subgroup's fader, just below the Mute and Solo buttons, to open the Rename window.

Auxiliaries

Auria has two stereo auxiliaries, labeled as Aux 1 and Aux 2, with each available on every channel and subgroup. By default these are post-pan sends, which means both the fader and pan pot will directly affect the signal sent to the auxiliary sections; either aux channel can also be switched to pre-fader mode in the Settings menu.

To insert an aux effect:

1. Tap the Aux FX button found on the Master channel (Mix window)
2. Choose either Aux 1 or 2 and tap the corresponding insert slot
3. A drop-down list of available effects will appear
4. Tap the desired effect from the list
5. The effect will be added to the selected aux and the effect's GUI (window) will open

The amount of overall signal from each auxiliary can be controlled with the two Aux Return knobs, also found on the Master channel.

Aux Delay Compensation

Some effect plug-ins may cause some additional latency, depending on the nature of the plug-in. Auria includes on option in the Settings menu called Aux Delay Compensation which can automatically correct for this latency. When enabled Auria will calculate the amount of latency on the aux channel and automatically compensate for this in the mix engine.

This option is off by default, as using it requires large amounts of system resources. Since by nature most Aux-style plug-ins tend to not require being perfectly time-aligned (like a reverb, chorus, or delay) including their latency generally will not degrade the overall mix. However, in a case where such latency is unacceptable, this option can simply be enabled.

Master Channel

The Master Channel is the final part of Auria's mixing section, as everything ends up here. It has its own MasterStrip, plus a brick-wall limiter, extra-large meter, and a Mono switch for checking the mix in mono.

The master channel's primary MasterStrip modules (EQ, BussPressor, and Limiter) operate exactly as the subgroup's version, so please refer to the [corresponding chapter](#) for more information on those modules.

There are some additional controls found only on the master channel, however.

Brick-Wall Limiter

In addition to the MasterStrip's limiter module, the Auria Master Channel also includes a distinctly brick-wall style limiter. Tap the Limiter tab to display the following controls:

- Input – Determines the amount of gain feeding the input of the brick-wall limiter
- Release – Adjusts the release time, in milliseconds
- Check – When enabled switches monitoring to just the signal being attenuated, useful as a way to hear what is being lost during limiting
- Soft – Toggles between a hard (the default) and soft knee
- Lim – Switches the brick-wall limiter on or off

Mono Button

This button switches the master channel from normal stereo operation to mono, this is most useful for checking a mix in mono for indications of phase issues.

Meter

Tapping this button will display a more detailed master channel meter. The meter is designed to display both peak and RMS at once: peak in the middle, RMS on the outside, with a peak-hold display showing the loudest peak so far. All meters display dBFS (full scale) values.

Any digital overs will result in the OVER light displaying a warning.

5

EDITING

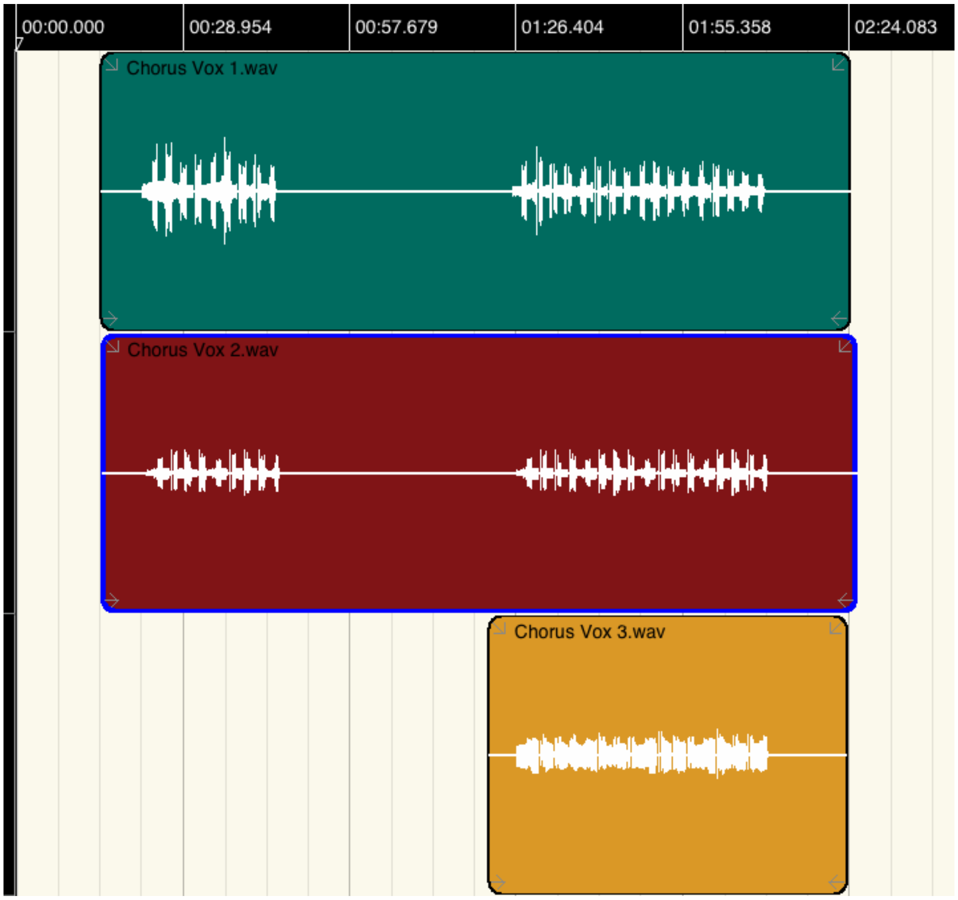
Auria contains a powerful non-destructive editing system that can handle just about any required editing task. This chapter will discuss in-depth how to accomplish various edits within Auria.

Before beginning, one important difference to understand between Auria and other desktop-based DAWs is the nature of user input. Standard desktop/laptop DAWs use a combination of long-standing input devices, such as the mouse, keyboard, trackball, etc, to tell the program what to do. In a typical DAW the use of these devices has evolved over time to incorporate a whole mountain of various keyboard shortcuts, multiple mouse buttons, mouse + keyboard combinations, macros, and even dedicated control surfaces to mimic actual recording consoles. Whole vast sections of DAW manuals are dedicated to Keyboard Shortcuts, or documenting thirteen different ways to move a region from one spot to another.

Auria is different. As an iOS application, Auria is 100% controlled by touch; to move a region from one place to another the user need only reach out with a finger and move it. There is no Appendix in this manual dedicated to Keyboard Shortcuts. There is only one input device: you.

Selecting Regions

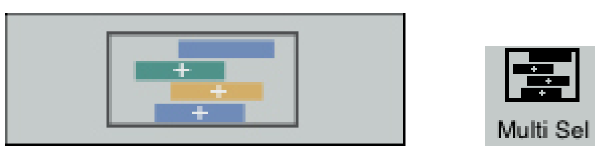
Before performing any edits to a particular region (or regions), that region must first be selected. To select a single region simply tap it once to select the entire region; the edge of the selected region will turn blue to indicate it has been selected.



A blue edge indicates the current selection

To de-select a region simply tap the selection again, or tap once in an empty section of the Edit Window; the blue edge will disappear indicating the region is no longer selected.

In addition to selecting a single region or track, Auria allows the user to select multiple objects at once with the Multi-Select Tool. This is very useful when an entire section needs to be edited, for example moving all the drum tracks at once, or copying all of the background vocals to a new chorus.

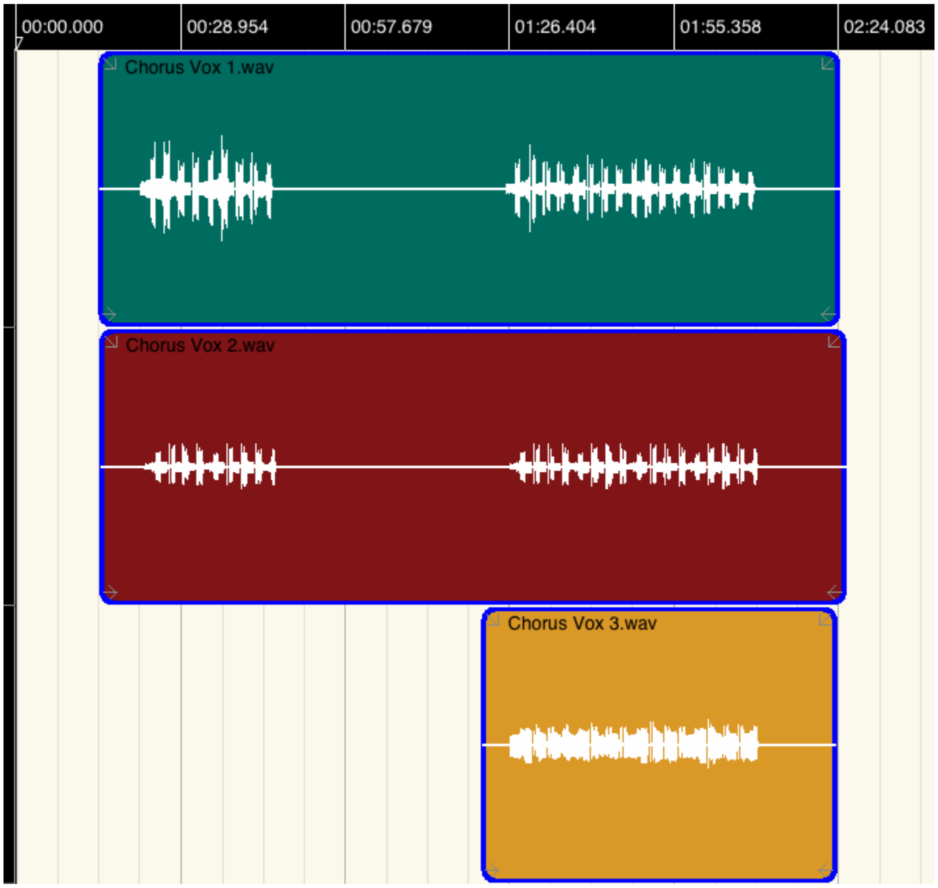


The Multi-Select Tool: Tap it once to select multiple items

To begin selecting multiple objects:

1. Tap the Multi-Select tool once - it will begin flashing to indicate multi-select mode is on
2. Tap each object once to add it to the selection; each will become blue in turn

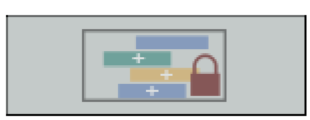
3. When all the desired selections have been made tap the Multi-Select Tool again to finish; any subsequent edits/processes performed will affect all selected objects



Multiple selections indicated in blue

De-selecting multiple regions works similarly as when working with one: tap once in an empty section to de-select all the objects, or tap individual elements to de-select one by one.

To lock the current selections double-tap the Multi-Select Tool, this will prevent changing the current group of selections while performing any moves or edits. This is useful in preventing accidental de-selections when working with multiple objects; simply double-tap the Multi-Select Tool again to unlock the selections.

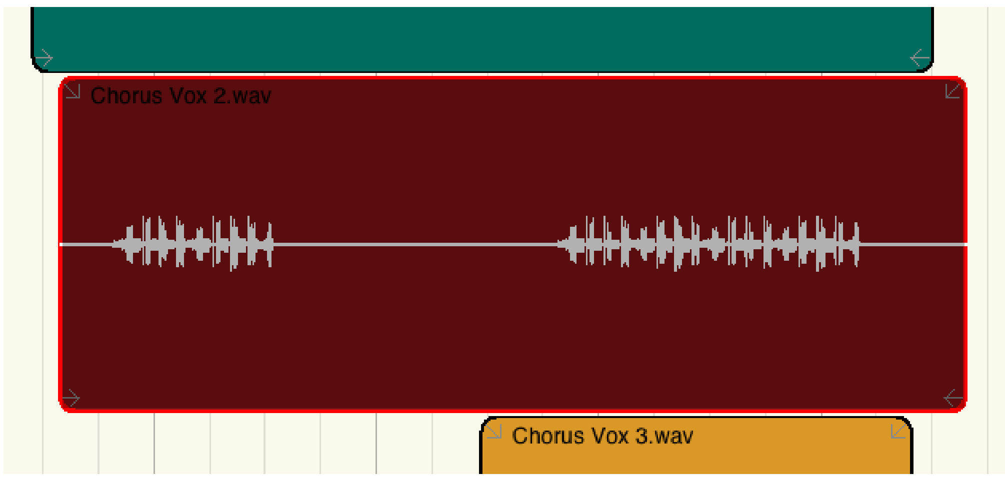


Group Lock Mode – Prevents accidentally de-selecting objects during editing

Moving Regions

One of the most common edits used in digital audio is to simply move a region from one place to another. Auria regions can be moved either earlier or later in the timeline (horizontally) or between different tracks (vertically), each done with a simple gesture.

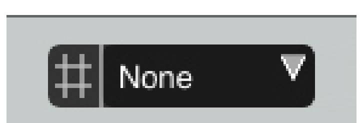
To move a region horizontally tap and hold the region itself: after a moment the region's edge will turn red signaling that it is ready to be moved. Then simply swipe your finger left or right and the region will move accordingly; when the region is in the new desired position simply let go.



A red outline indicates the region is being moved

Moving a region between tracks is done in essentially the same manner; again tap and hold to indicate which region you want to move, wait for it to turn red, and then drag it up or down (of course there must be another track either above or below the original for this to work), letting go when the region is on the correct track.

When moving regions between tracks Auria will "magnetically" snap the region in the same place along the timeline, preventing the region from sliding left or right. This snap can be over-ridden by simply moving the region sideways enough to "break" the magnetic effect.



The Snap Menu

By choosing a value in the Snap menu Auria will snap all region movement to the selected grid. This is useful when performing edits to regions which correspond to Bars and Beats, as Auria can automatically move regions by whole bars, beats, or even beat subdivisions. Another useful snap option is Events which automatically snaps regions so that they start/stop at the same time as another region.

Deleting Regions

Deleting an unwanted region is also quite straight-forward:

1. Select the region to be deleted by tapping (the region will be outlined in blue)
2. Tap open the Edit Menu
3. Tap Delete Region to remove the region

As Auria is a non-destructive editor the actual audio recording will not be deleted from the iPad and can be immediately brought back with a tap on Undo.

Cut-Copy-Paste-Duplicate

Just like a word processor, Auria supports standard cut, copy, and paste functions. These options can be used to manipulate selections just like a text editor. Each of the following commands are found in the Edit Menu, and require making a selection before being available:

Cut – Copies the selection to the clipboard and deletes the source from the project (but not from the iPad itself).

Copy – Copies the selection to the clipboard and leaves the source in place.

Paste – Pastes the contents of the clipboard to the selected track. First select the destination track by tapping the left-hand track pane (i.e. where the track number is displayed), then use the cursor to indicate where the item should be pasted along the timeline.

Duplicate – Makes an exact copy of the selection and automatically places it at the end of the original. Useful when working with regions that are edited to complete measures, like a drum loop, that should be repeated a number of times. For a quicker version see Automatic Region Duplication.

AudioCopy/AudioPaste™

Auria supports the AudioCopy and AudioPaste iOS standard developed by Sonoma Wire Works. These functions allow Auria to move audio between other audio apps intuitively and easily.

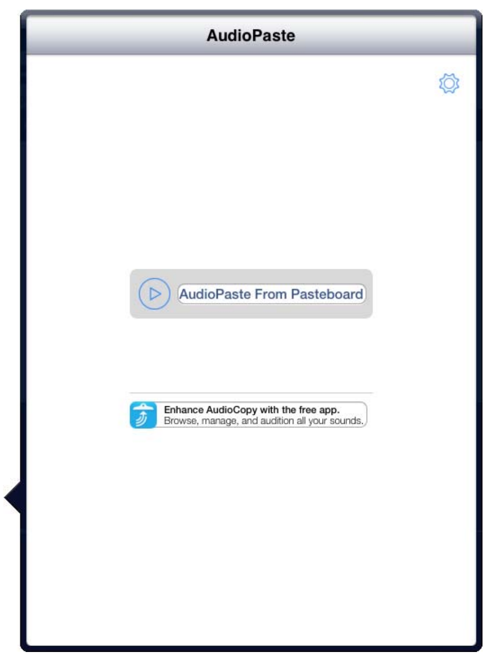
AudioCopy – Used to move audio from Auria into another compatible app.

AudioCopy Dialog

To copy audio out of Auria:

1. Select the desired region, then tap open the Edit Menu and select AudioCopy
2. The AudioCopy dialog will open
3. Tap Copy Audio
4. Wait for the selection to copy
5. Close Auria, open the destination app and choose AudioPaste to finish the copy

AudioPaste – Used to move audio from an outside app into Auria.



AudioPaste Dialog

To paste audio into Auria:

1. Start in the outside app and copy the desired audio using AudioCopy

2. Back in Auria tap the Edit Menu and select AudioPaste
3. The AudioPaste dialog will open
4. Set parameters as desired (tap gear icon to open):

    Paste to – Assign destination track and start time in Auria

    Loops - Set number of repetitions of pasted item (used when pasting in loops)
5. Tap Paste to add the copied audio to the current project

Note: The AudioCopy and AudioPaste functions are 16-bit only, so when AudioCopying from an Auria project that is a higher bit depth (i.e. 24 or 32) the audio will be automatically converted to 16-bit during the copy.

Locking Regions

To prevent un-intended edits to a particular region (or regions), Auria includes a Lock Region function; any locked region cannot be moved or edited.

To lock a region:

1. Select the region (with a single tap)
2. Tap open the Edit Menu and select Lock Region
3. A small pair of lock symbols will appear in the lower corners of the region to indicate its locked status

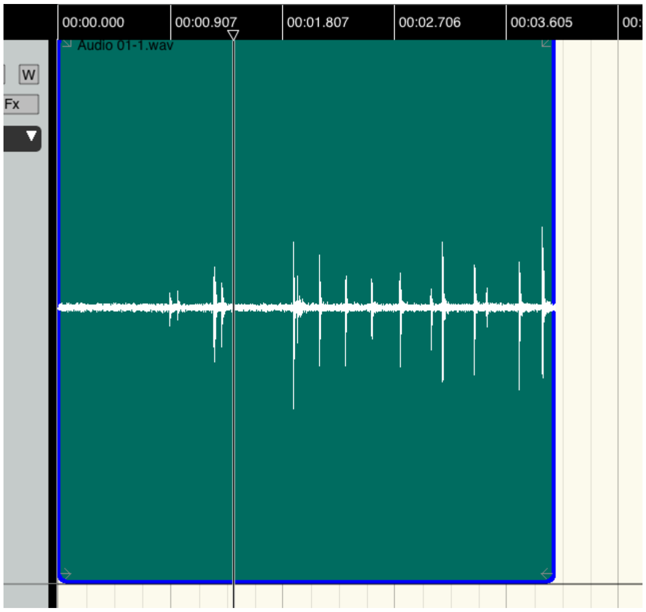
The Unlock Region menu item can be used to enable editing once more.

Splitting Regions

Any region can be split in-half using the corresponding Edit Menu command, Split, using the Cursor to designate where the split should occur.

To Split a region:

1. Select the corresponding region (single tap it to turn it blue)
2. Position the cursor at the desired split point. Fine movements of the cursor can be accomplished by zooming in horizontally and then sliding a finger across the Timeline Ruler

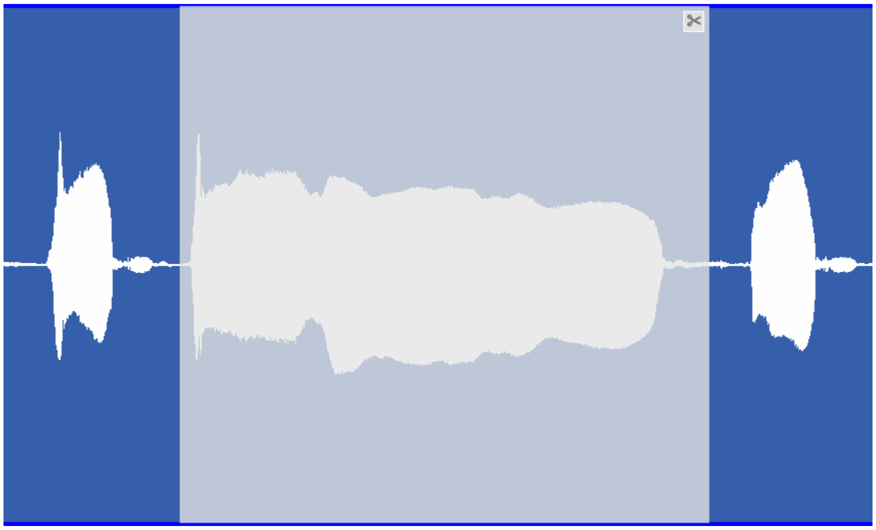


A region ready for splitting

3. Once the cursor is in the desired split point tap open the Edit Menu
4. Tap Split to cut the selected region in two.

Highlighting

Sometimes only a section of a particular audio region needs to be changed, such as altering the gain of a particularly loud vocal plosive or a too-quiet snare hit. In these cases the specific section can be highlighted in Auria to allow more fine-tooth editing.



Highlighted portion of an audio region

To highlight a section of audio:

1. Double-tap at the beginning of the intended selection, keeping the finger held on the screen (i.e. a double-tap and hold)
2. Swipe finger horizontally across the entire desired area
3. Release finger

Highlighting can easily be fine-tuned by tapping and swiping at either the start or end of the selection and adjusting the end points.

Separate

A highlighted section of audio can be separated from a larger region into its own new region. When there is a highlighted section of audio:

- Tap the Edit Menu and select Separate, or
- Tap the scissors icon in the top-right corner of the highlighted area.

Scrubbing

Similar to rocking the reels on a tape machine, audio can be scrubbed from the Edit window in order to find specific spots along the Timeline. While many times desired edit points can be found visually by examining a waveform, there is no substitute for hearing a slowed-down section to really discover what is happening.

Scrubbing works just like sliding the Timeline cursor, except instead of using one finger use two fingers at once to slide back-and-forth along the Timeline Ruler.

Note: Zoom level determines how fast or slow the audio can be scrubbed. For slower speeds zoom further into the Timeline; to move quickly between sections zoom out.

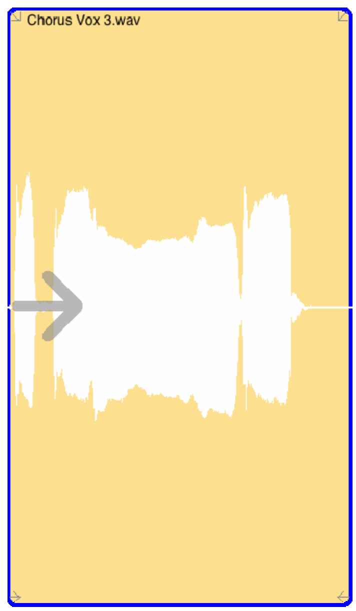
Trim Handles

One of the most common edits is changing the beginning or end of a particular region, such as rimming off the background noise before a vocal take starts, or removing the unwanted audio at the end of a recording. Auria includes a very quick way to perform these edits non-destructively: trim handles.

Every region in Auria has a trim handle at both its beginning and end, represented by an inward-facing arrow. They are in the bottom corners of the region.

To use a trim handle:

1. Tap and hold near the arrow icon in the corresponding bottom corner (left for beginning, right for end)
2. The region's outline will turn blue and a large arrow will appear on the side of the region



Trimming the start of a region

3. Swipe in the direction of the desired edit, the region boundary will move along with the swipe
4. Release the corner of the region to finish

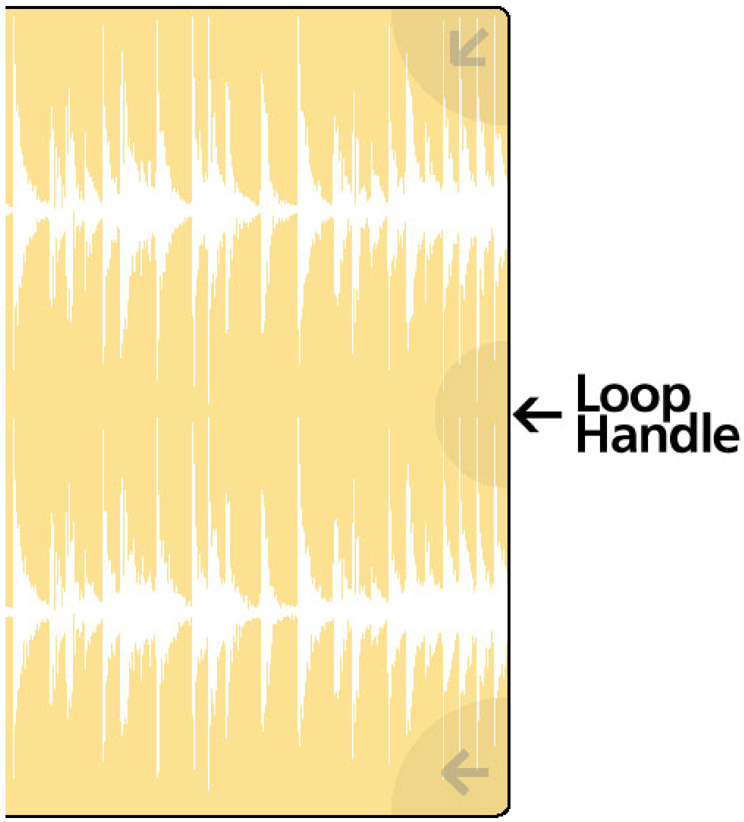
Note: Handles are only visible when zoomed in adequately far enough to clearly display the handles, so if the handles aren't visible try zooming in farther.

Trim handle editing is non-destructive, which means the underlying audio recording isn't affected when adjusting the beginning or end of a region. At any time the changed trim handle can be slid back to where it came from, revealing the original audio along the way.

Note: If you find it difficult to touch the trim arrow itself (and accidentally end up selecting the track below it instead), keep in mind that you can tap anywhere near the corner and not just the arrow itself; give yourself some room and aim further inside the region's corner.

Automatic Region Duplication

When working with audio loops, being able to quickly and easily repeat certain loops can greatly speed up workflow, so Auria has a dedicated handle that can be used to automatically duplicate (repeat) a particular region simply through dragging.



Drag the handle to the right to duplicate the region

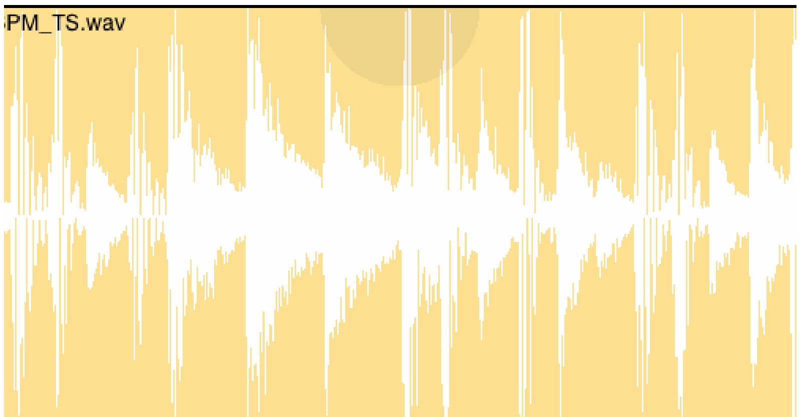
To automatically duplicate a region:

1. Tap and hold the handle along the middle of the right edge
2. A large right-facing arrow will appear, indicating the region is ready to be duplicated
3. Drag the handle to the right. To prevent unintended duplication there is an initial "safety buffer" that the handle must be dragged beyond, just keep dragging to the right
4. A second copy of the region will automatically appear at the end of the original region
5. Continue dragging to the right to create additional copies

Note: Automatic duplication works even with edited regions, no need to bounce or condense a region before duplicating.

Gain Handle

Individual region's can have their overall gain adjusted non-destructively by using the Gain Handle:

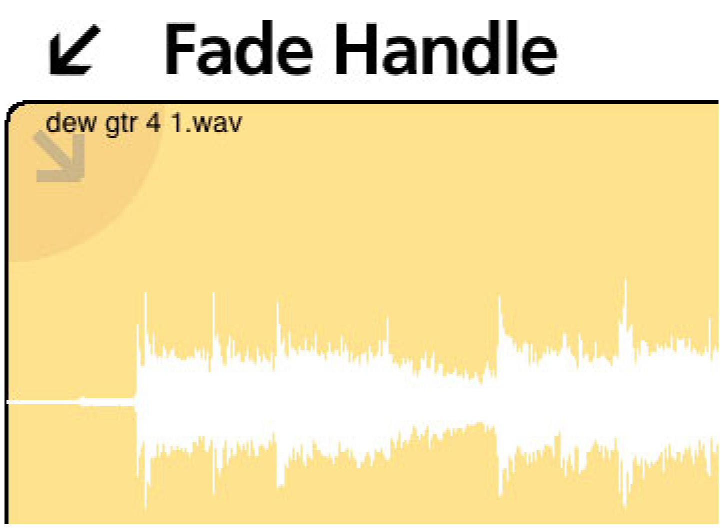


Gain Handle shown at the top-middle of the region.

To adjust a region's gain simply tap and hold the handle at the top and center of the region, then slide up or down to raise or lower the gain of the region. A horizontal line will appear to show the relative amount of gain change, as well as precisely displayed in the Region Info Box.

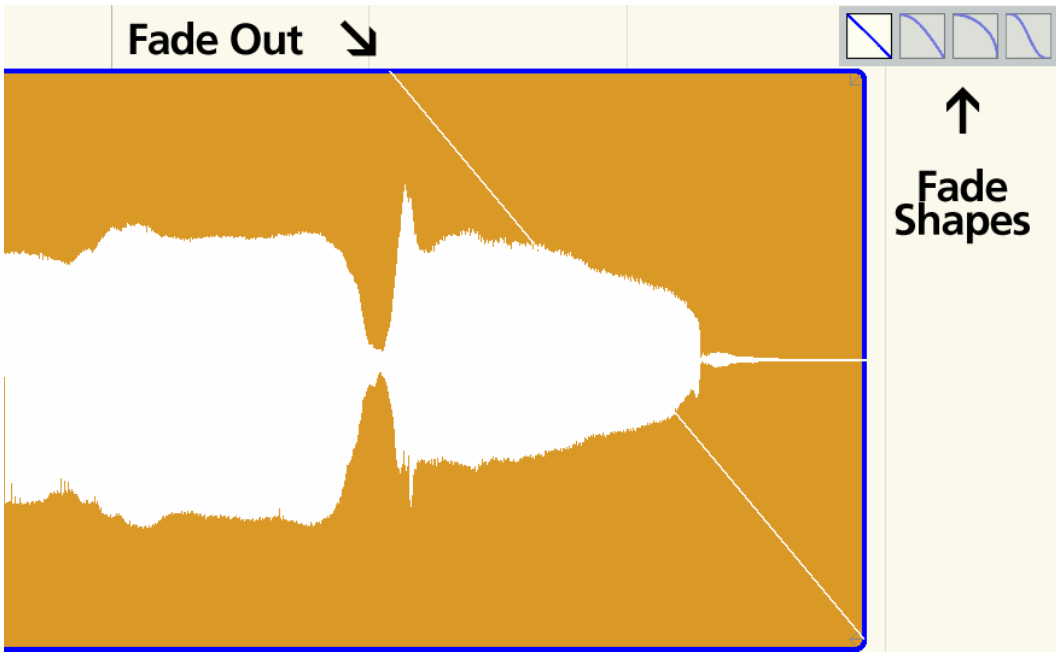
Fades

Auria's regions can include both non-destructive fade ins and fade outs. These are manipulated in a similar fashion to the Trim Handles, through two separate Fade Handles in the top corners of a region.



Fades have two different parameters, fade length and fade shape:

- Fade Length – Sets the duration of the fade in time
- Fade Shape – Includes 4 selectable fade types, each with a unique sound



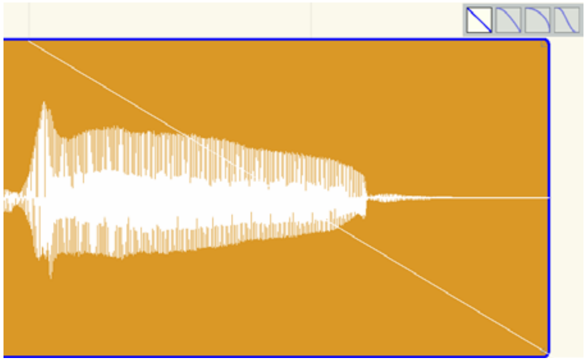
Fade Length is adjusted in the same manner as trimming, through the use of fade handles in the upper corners.

To create a new fade:

1. Tap and hold the arrow icon in the corresponding upper corner (left for fade in, right for fade out)
2. The region's outline will turn blue and a diagonal fade line will appear
3. Swipe in the direction of the desired fade, the fade will be drawn along with the swipe
4. Release the corner of the region to finish the fade

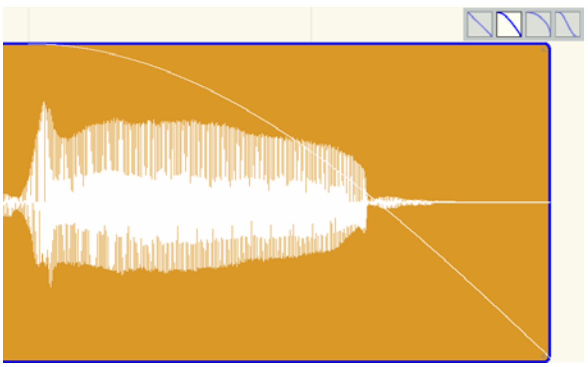
Note: If you find it difficult to touch the fade arrow itself, keep in mind that you can tap anywhere near the corner and not just the arrow itself; give yourself some room and aim further inside the region's corner.

Once the fade itself has been drawn a new fade shape can be selected from the pop-up box. Four types of fades are available:



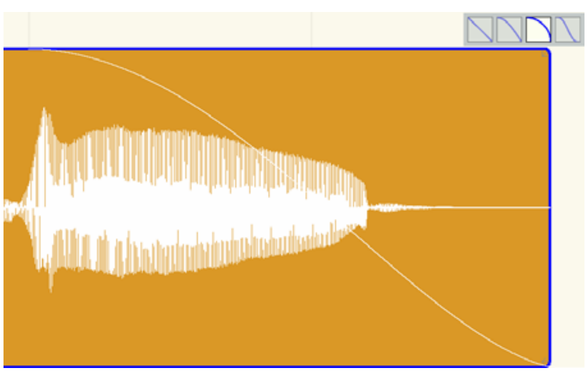
Fade 1 – Linear (default)

Linear – Default type, gain changes at a constant rate.



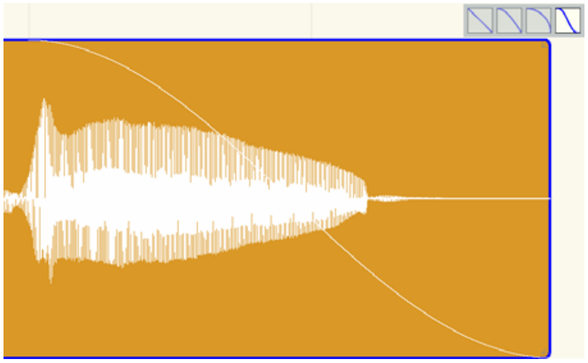
Fade 2 – Exponential (Slow)

Exponential (Slow) – Non-linear fade which changes gain logarithmically, which results in a more musical fade than linear. Creates a smooth, gradual fade.



Fade 3 – Exponential (Fast)

Exponential (Fast) – Non-linear fade which changes gain logarithmically, though a faster, more abrupt fade than Slow.



Fade 4 – “S” Curve

“S” Curve – Non-linear fade which changes gain more slowly at the beginning and end of the curve, which results in a Slow-Fast-Slow shape.

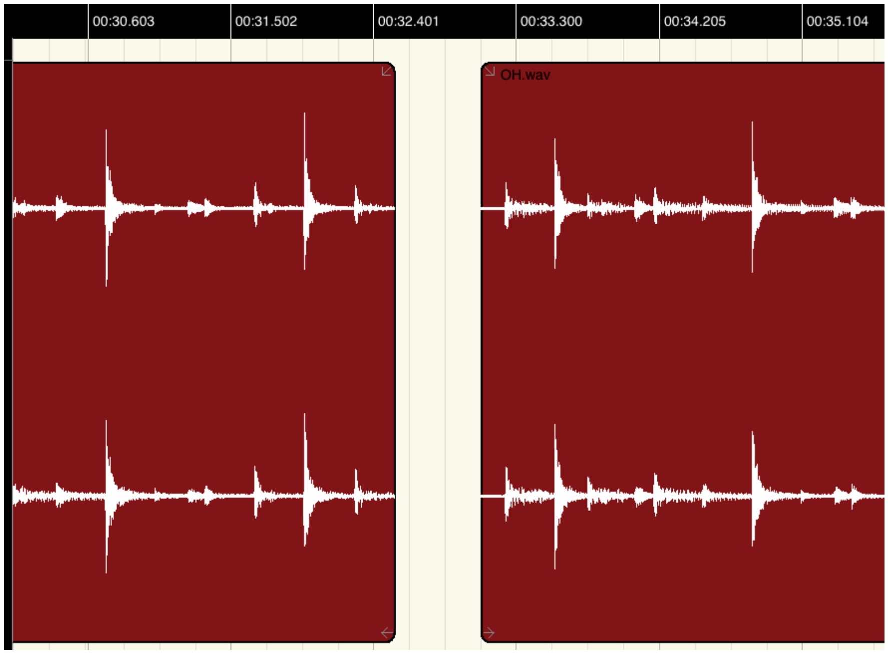
Crossfades

Auria normally only plays one region at a time per track. If two (or more) regions are present in the same place on the timeline and reside on the same track, Auria only plays the top-most region, ignoring those regions which may be underneath. Sometimes, though, two regions need to play at once during a transition, for example when editing together a single performance made up of multiple takes. To accomplish this Auria includes a Crossfade function.

Crossfades are performed by arranging two separate regions so that they overlap, and then applying the crossfade across the overlapping section. In the following example two separate drum overhead takes will have a crossfade applied to create a seamless edit.

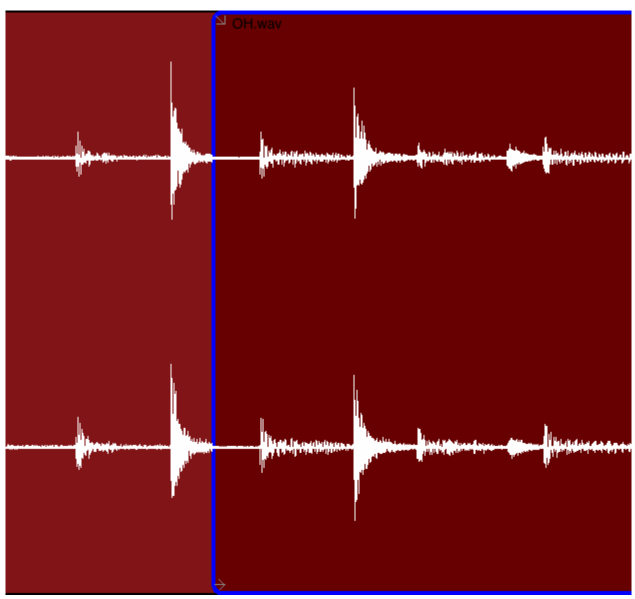
To create a crossfade between these two regions:

1. Place both regions on the same track and trim the beginning and end points, if needed.



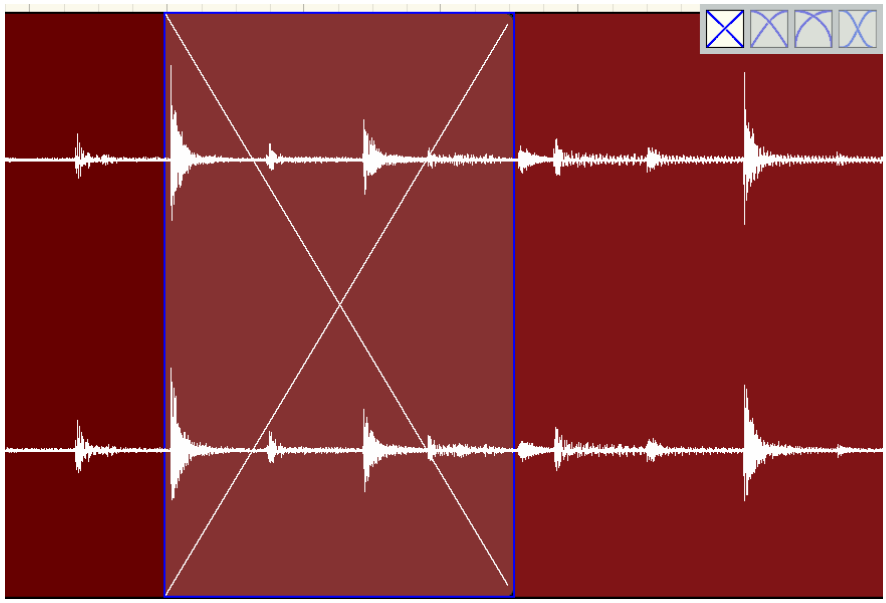
Two regions before a crossfade. Both have had their beginning/end trimmed as needed.

2. Move one region so it overlaps the other region. The timing of the two regions should sound correct, both in context with the overall project and in regards to each other.



Two regions being lined up so that the downbeats of each are aligned

3. Select both regions (using the Multi-Select tool), tap open the Process menu, and select Crossfade.



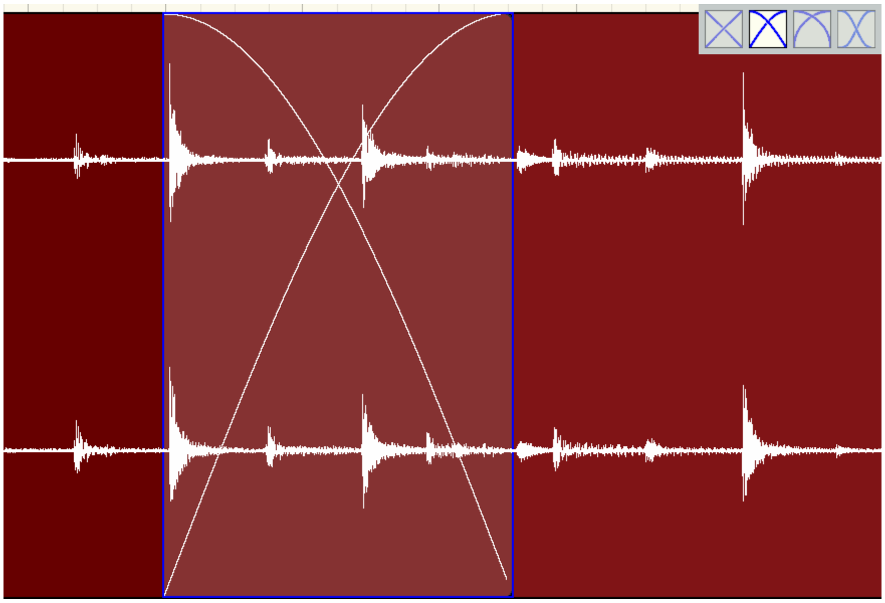
Equal Gain (Linear) Crossfade

4. A crossfade is created, corresponding to the overlapping section of the two regions. The size of the crossfade (i.e. the start and end points) can be edited just like trim handles.

The default crossfade shape type is Equal Gain (linear), as shown above. While this shape works well for certain kinds of audio other shapes are included, four in total.

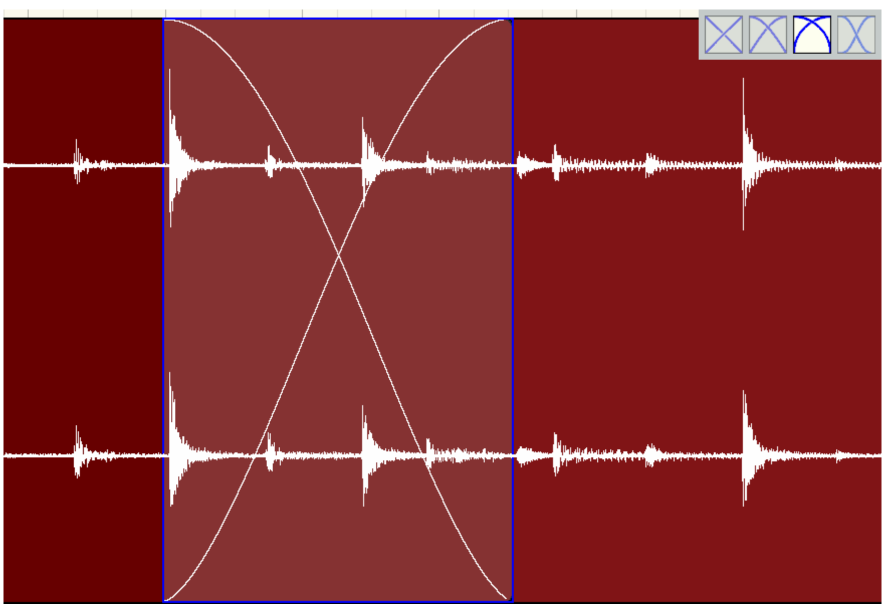
Equal Gain (shown above) – The two regions change gain at a constant rate so that the sum gain is equal throughout the crossfade. Best when working with very similar pieces of audio (i.e. correlated), like editing together two takes of the same guitar part.

Equal Power – Gain changes so that the sum power is equal through the crossfade. Best for dissimilar pieces of audio (de-correlated), like a transition between two very different music beds .



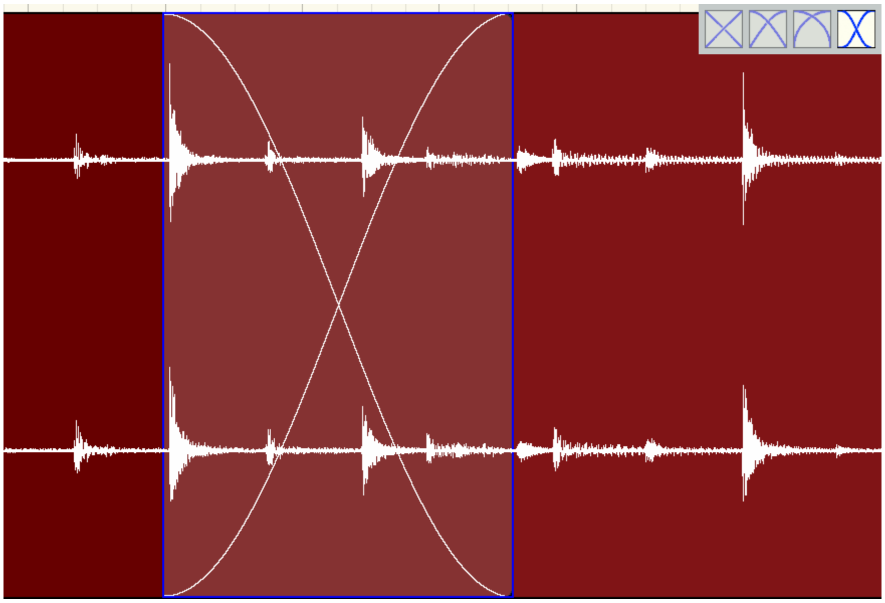
Equal Power

Exponential – In-between Equal Gain and Equal Power, this fade works well when the two regions are somewhat similar, but the previous fades are both too noticeable.



Exponential

"S" Curve – Similar to the first curve, but the fade tapers more slowly than the perfectly linear #1.



Equal Gain (S-Curve)

Destructive Processing

In addition to Auria's non-destructive editing, there are a suite of audio tools which will destructively change the actual audio content itself (though Undo will still work!)

The following commands are found in the Process Menu, and will be applied to the current selection:

- **Gain** – Changes the gain of the region or highlighted selection by the desired amount.

- **Normalize** – Normalizes the region or highlighted selection to the desired level. Choose between Peak and RMS modes, as well as how clipping will be treated. The 'Release' parameter is only active if 'Limit' is selected. Limiting is processed by the built-in Brick Wall Limiter.

- **DC Offset** – Removes DC offset from the region or highlighted selection.
- **Reverse** – Reverses the region or highlighted selection.
- **Silence** – Converts the region or highlighted selection to silence.
- **Crossfade** – Creates a crossfade between the selected regions (two overlapping regions must be selected using the Multi-Select Tool). Selecting a crossfade region will display the crossfade curve options in the top-right corner. See Crossfade section earlier in this chapter.
- **Reset Fades** – Reset the fades of the selected region.
- **Condense Regions** - Trims away unused audio in a project to save space.

Time Stretch

One very powerful feature found in Auria is the ability to time stretch audio, without affecting the frequency (pitch) of the recording. This allows for speeding up or slowing down individual regions, such as voice-overs and sound effects, as well as stretching loops of various tempos to fit the project tempo. The specific time stretching algorithm comes from the well-known Dirac 3 Pro signal processing suite.

Time Stretch a Specific Region

Any region can be time stretched by simply dragging its handle, using two fingers instead of the usual one, until the region reaches the new desired length. For example, if there is a narration track that starts at the beginning of the project that is currently 33 seconds long, but it needs to be 30 seconds, simply drag the region's right handle (using two fingers) to the left until it reaches the 30 second mark on the Time Ruler, and let go.

If working with a loop the easiest method is to set the Time Format to Bars:Beats, move the loop so it starts at the desired measure (bar), and then drag out the right handle (using two fingers) until the loop is the correct number of bars in length.

Automatic Tempo Matching with Import Audio

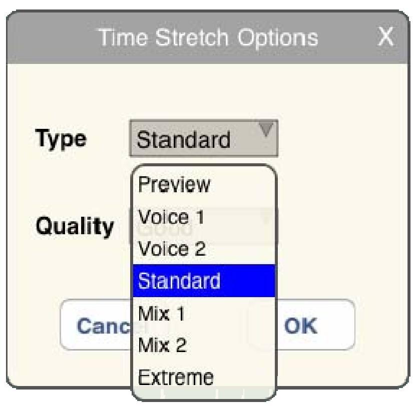
When importing audio into a project, Auria will detect if the selected clip includes tempo information (in either Acid or Apple Loop formats), and can automatically time stretch the loop to match the current project's tempo.

Time Stretch Options

Whenever the Time Stretch feature is used, a pop-up window will be displayed with specific settings for that region/file.

Time Stretch Options Dialog

The following section regarding the time stretching options of Type and Quality are taken from the Dirac 3 Pro documentation, which is rather technical in nature. Don't worry, there are additional guidelines at the end of each section.



Time Stretch Type

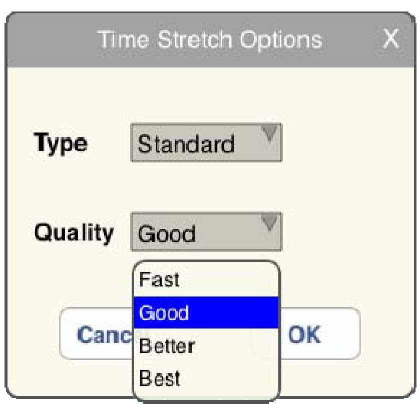
"Dirac uses a novel algorithm that can be scaled to provide good time domain localization or good frequency localization, or both. This ability is controlled by a parameter that Auria labels as Type. As a rule of thumb, settings found higher in the list provide good time localization (good for voice and single instrument recordings), while settings lower in the list are good for entire mixes. These Mix settings take slightly more time to process but are not considerably slower."

"The following Type settings are available:

- Preview - This automatically selects the best time/frequency tradeoff for preview performance. It is the fastest setting but might not provide the best results in all cases.
- Voice 1 - Selects full time localization. Good setting for single instruments and voice.
- Voice 2 - Time/frequency localization with emphasis on time localization. If a setting of Voice 1 produces echoes this might be a better choice.
- Standard - This sets the time/frequency localization halfway between time and frequency domains. It is the best setting for all general purpose signals.
- Mix 1 - Higher frequency localization and less time localization. Might be a better choice for mixed music than the previous settings.
- Mix 2 - Highest frequency localization. This might not be an ideal choice if you're dealing with signals that have very sharp attack transients but it might be useful for sensitive material such as classical.
- Extreme - Special mode for very large stretch ratios (2x to 4x). Good for transcription purposes."

Note: Generally speaking, the Standard setting works well on all types of audio (including drums). Voice 1 and Voice 2 are best on voice and single instruments. Mix 1 and Mix 2 are best on full mixes containing multiple instruments. Preview is the fastest method, for when simply testing time stretching on a particular part. And Extreme is only for those cases when stretching something more than twice as fast/slow, such as transcribing very fast musical performances.

A quick rule of thumb would be to try the Standard setting first, and then try the other settings if the Standard results aren't good enough.



Time Stretch Quality

"The second parameter is used to set the processing quality. Settings higher in the list provide excellent performance at a slightly lower algorithm quality, while settings lower in the list render the results in more time but at a significantly higher resolution."

"The following four quality settings are available:

- Fast - This quality mode offers preview quality, which is usually good enough for a preview to see the effects of the parameter settings.
- Good - A better quality mode than Fast. It is recommended as the default quality setting for non-preview processing.
- Better - Very good quality mode but takes more CPU.
- Best - The highest quality mode. Note that this setting can be very slow."

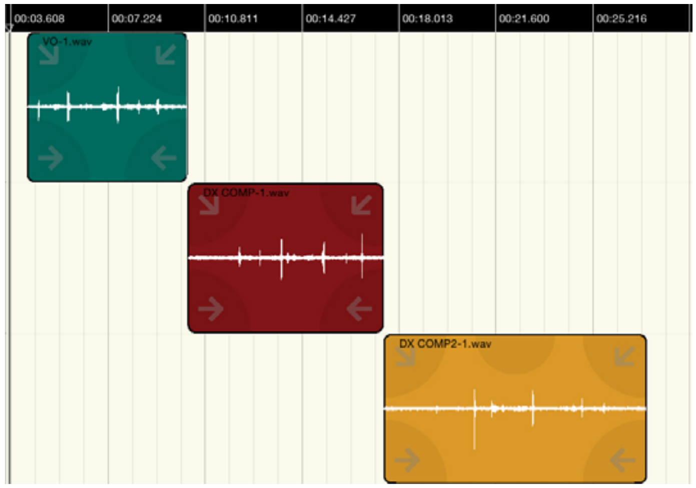
Note: The main idea here is of the tradeoff between quality and processing speed, i.e. how long it takes to render the stretched audio. The Good setting is a great compromise between speed/quality, as the Better and Best settings result in small improvements in sound quality but with much longer rendering times. In those instances where quality is of the foremost concern? Use the better settings; just be prepared to wait a while.

Ripple Edit Mode

Sometimes an edit should affect more than just the individual region being worked on; it should alter the entire timeline of the project. Imagine editing the dialog for a movie where the director wants to remove a specific line being spoken by an actor. Simply selecting the unwanted speech and deleting it ends up leaving a gap, though, an undesired hole in the dialogue; what's needed is a quick way to both delete the region AND shift everything that occurs afterwards ahead in time, eliminating the gap . This is where Ripple Mode editing comes in.

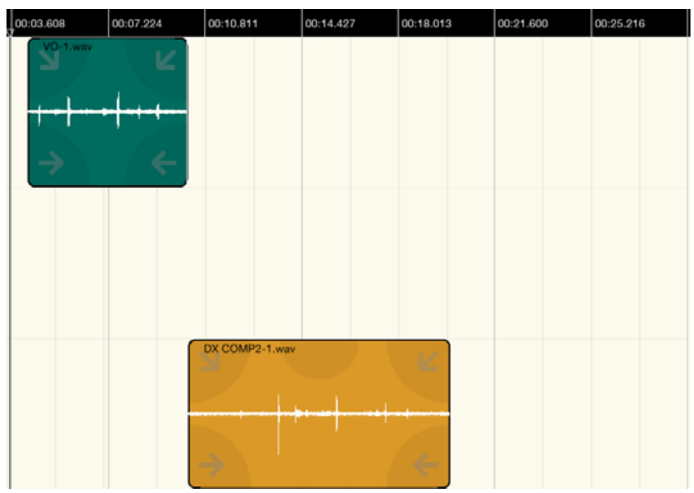
To enable this mode tap open the Edit menu, look towards the bottom of the menu, and then tap RIPPLE EDIT MODE to switch it on (a check mark will appear). Now edits made to regions will ripple "downstream" to later events and shift them in time. Removing a region will cause later regions to move earlier in time to close the gap, while adding a region (via recording, pasting, etc) will cause the later regions to make room for the new audio by shifting later in time.

Here is an example of deleting a region with Ripple Mode turned on. This project has three tracks, each with one region, with one right after the other.



Three regions spread across three tracks

Normally, if that middle (red) region was deleted the third (orange) region would stay in place and a hole would be left in place of the removed region. But, with Ripple editing enabled, that last region would automatically slide over to the left exactly the same amount in time as the length of the deleted region.



The "downstream" region has moved to the left, closing the gap caused by the edit

Ripple editing has two different modes of operation, these modes are selectable under the GENERAL tab in the SETTINGS MENU:

- **Active Track Only** – Only regions found on the same track being edited will shift, all other tracks will ignore the edit.
- **Global** – Every region on every track that occurs after the edit will be shifted in time.

6

AUTOMATION

Auria includes a powerful system which enables automation of every mixer and plug-in parameter. Automation can be created in two different ways: via recording from the Mix window or graphically from the Edit window.

Automation in the Mix Window

At the top of every channel and plug-in window are buttons labeled R and W, or Read and Write. To enable automation on a particular channel/plug-in simply press the corresponding button:

- **Read Mode** - Plays back existing automation
- **Write Mode** - Records new automation

Note: enabling Write mode will automatically place the channel in Read mode also.

To record an automation pass, first make sure the channel is in Write mode (the W button should be lit yellow). Press Play in the transport to begin playback of the project, and then simply move a control as desired: perform a fade out with the channel fader, sweep the pan knob, adjust an aux send, etc. The W button will turn red when Auria senses a control being touched, and then turn back to yellow when that control is "let go". Auria uses Touch Mode style automation recording, which means that automation data is only recorded while actively touching a control.

Lastly, rewind back to the start of the automation pass, and press Play. The automated control should play back the recorded moves.

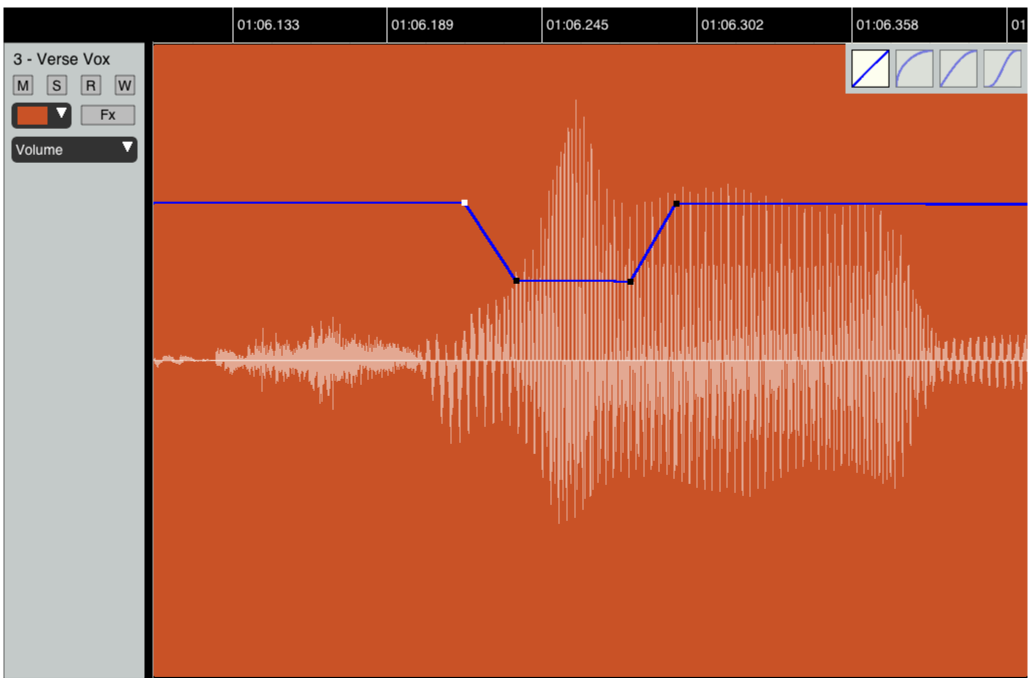
To edit automation data from the Mix window, simply overdub new automation over existing data as needed. For more fine control switch to the Edit window, which is detailed in the next section.

Automation in the Edit Window

Automation can be created and edited from the Edit window as well, and in fact it is generally easier to perform edits to automation via the Edit window.

To view/work with a particular control parameter, first select the desired control from the track's drop-down menu. Auria will then overlay the automation data in that track pane on top of the audio regions. The track height may need to be increased (by using an expanding vertical pinch) in order to see the drop down menu.

This data is made up of two elements: Control Points, and the lines which connect those points. Each control point represents two different parameters: Time and Value. The time is simply where in the timeline the point exists, and the value is the relative "height" of the point inside the track pane.



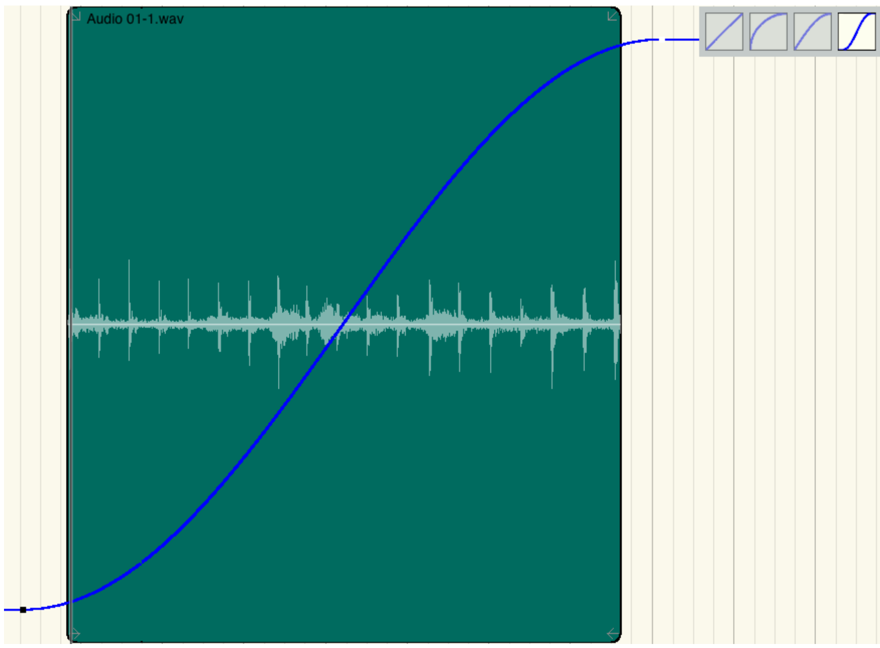
Example of volume automation, with one Control Point selected and Curves window visible

A simple fade can be created by adding two control points, one each at the beginning and end of the desired fade. Auria will automatically connect these two points with a linear line segment.

- To add a new control point, tap and hold inside the track's pane, and a new control point will be added in that specific location.
- To move an existing control point, tap and hold on the control point itself, and drag the point to a new location.

- To delete an existing control point, tap on the specific control point to select it (it will turn white as an indication). Then choose Delete Control Point from the Edit Menu to delete to point.

By default Auria will connect control points with a linear (i.e. straight) line segment, however this shape can be changed. When a control point is selected (with a simple tap) the curve-type window will open, and tapping the desired curve shape will change the line segment to the left of the current control point.



The 4 Curve Shapes displayed

Moving Automation

There are several ways to move existing automation around in a project.

Cut/Copy/Paste

Entire sections of automation can be moved via Cut/Copy/Paste. To Cut or Copy a section of automation:

1. Select the correct automation type from the track's drop-down box so that it is visible on the track
2. Highlight across the desired automation (double-tap and swipe)
3. Tap either Cut or Copy as needed
4. Place the Cursor in the new desired location along the Timeline
5. If needed select the alternate track the automation is being moved to
6. Tap Paste to place the Cut/Copied automation in its new location

Note: One necessary caveat is that only directly compatible automation data can be moved between different tracks, for example plug-in automation can only be moved to a different track when the destination track already has that same plug-in inserted. Also, due to the proprietary differences between regular audio tracks, Subgroups, and the Master channel, channel automation (e.g. volume, Aux sends, mutes, etc) cannot be moved between the three different types of channels; volume automation from the Master channel cannot be copied to a Subgroup, a track's Aux send automation cannot be pasted to a Subgroup, and so on.

Linking Automation to Regions

This option, found in the Settings Menu on the Editor tab, links automation to its corresponding region (i.e. the region the automation overlaps). With this enabled, any time a region is moved (or deleted, cut, copied, duplicated, etc) any automation data will move along with the region and stay tied to it.

One example would be the case of a drum loop. A filter plug-in could be inserted and automated to sweep its filter frequency in time with the loop, and then, with linking enabled, when that loop is duplicated across the project the corresponding filter automation will duplicate along right along with it.

7

FILE MANAGEMENT

Auria saves all data inside its own directory on the iPad, something Apple refers to as "sandboxing", which means that no other iPad app can access it. This is done to separate all iOS apps from each other as a safety precaution.

When starting a New Project, Auria creates a special directory (with the same name as the project) for storing all the project-related files. Auria provides several methods to add/remove/modify these projects as well as manage the file structure of the Auria documents folder.

Project Bundles

Auria stores all its projects in packaged Project Bundles, an iOS/OSX convention that simplifies more complex file hierarchy. Essentially a packaged bundle is a directory which contains all relevant files to that project, but this directory appears to iOS and OSX as a single file (Windows users using iTunes File Sharing will see them as normal directories with a ".project" extension).

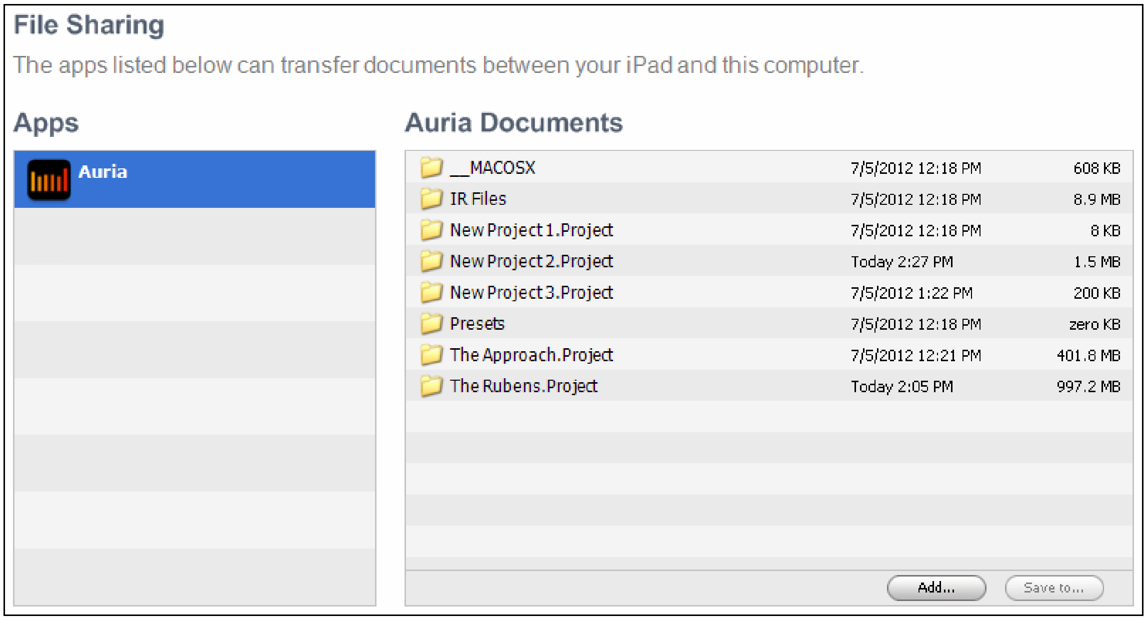
All necessary parts of a project (the project file, all audio files, and region overviews) are stored inside its bundle, making project transfers back and forth from the iPad very simple: only the project bundle itself needs to be moved (or the project directory if viewed from Windows).

When loading a saved project, Auria can automatically scan the project folder for orphaned audio files (i.e. regions no longer being used in the project) and delete them from the project bundle. This automatic cleanup process can save critical space on the iPad's internal storage. This function can be enabled through the Settings window.

Automatic Project Backup

Auria maintains automated backups in the unlikely case of a corrupted project. Auria makes three backups of the project in a round-robin system, and will automatically load the most recent working version whenever a project fails to load successfully.

iTunes File Sharing



The iTunes File Sharing window, shown on Windows

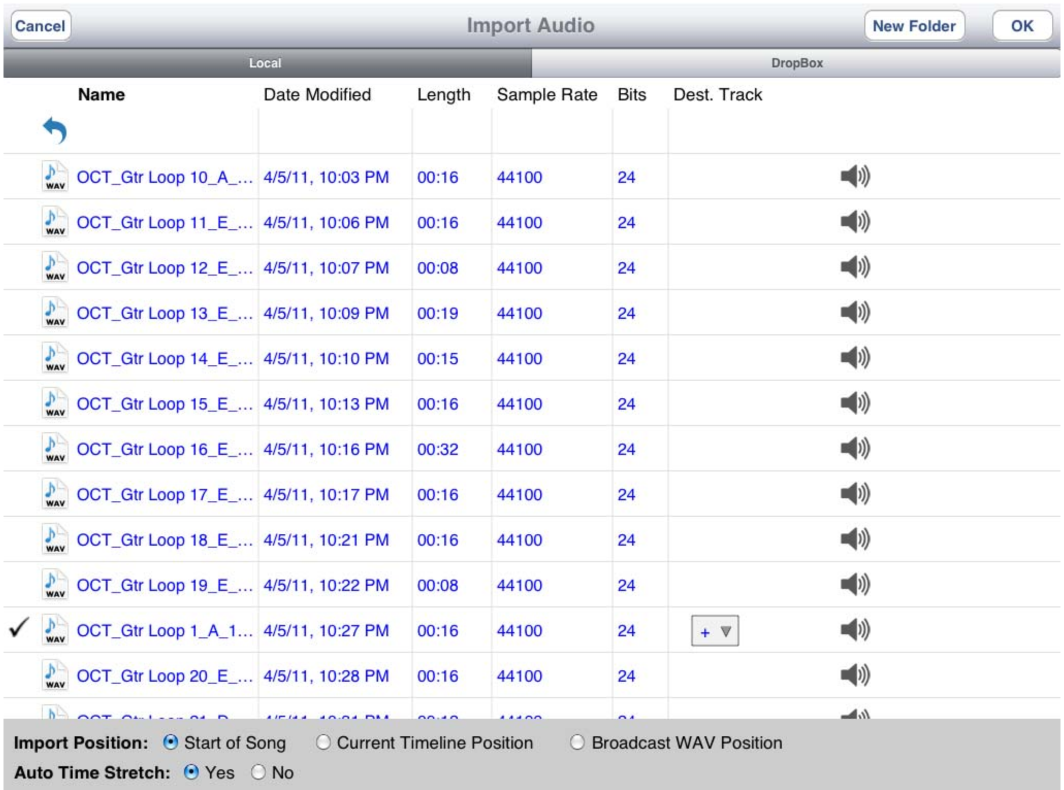
All files that are used within Auria can be maintained by connecting the iPad to a Mac or Windows system and using the iTunes File Sharing system, found in the main iTunes application. Auria will appear in the File Sharing section of the Apps tab in iTunes, and audio files, projects bundles, AAF projects and other files supported by Auria can be dragged in and out of the "Auria Documents" window.

Note: All plain folders must be zipped before dragging them into Auria's Documents directory (Auria will automatically decompress them), due to a limitation in iOS.

Import Audio

In addition to recording audio directly into Auria, separate audio files can be brought into projects directly. Loops, samples, or even entire other multitrack recordings can be brought in through importing.

To start importing audio tap the main Menu, then Import Audio.



The Import Audio dialog

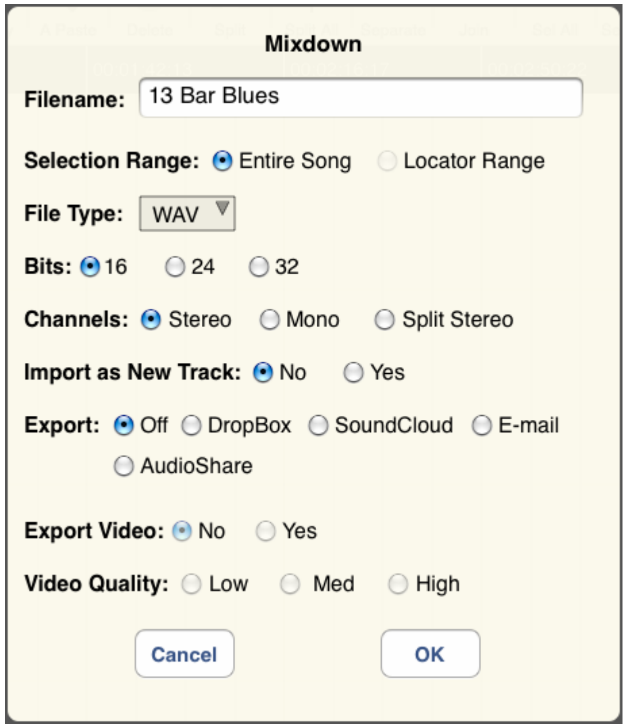
All audio files, including those that have been loaded into Auria via iTunes File Sharing, will be listed here. Tap a track to select it for import. The destination track number will increase sequentially as you select tracks but can be manually modified as well. Audio files can be previewed by clicking the speaker icon. To import audio via DropBox tap the appropriate option on the Local/DropBox bar.

The insertion point can be selected at the bottom, to either line up at the start of the song (project), the current position of the cursor in the timeline, or the time-stamp of a Broadcast WAV file.

When importing either Acidized loops or Apple Loops (which contain tempo information) there is an option to automatically time stretch the loop to match the project tempo. When turned on the time stretch dialog will appear so that time stretch settings can be applied. See the Time Stretch section for more information.

Mixdown

Found under the main Menu, this creates a final mixdown of the project.



Filename - Enter a name for the mixdown.
Selection Range - Choose whether the mixdown is of the entire project or just a section between the locators.
File Type – Select the mix file format, including standard .wav, AIFF, and M4A. Also includes Stems option which will automatically bounce every track, including effects and automation, to individual WAV files.
Bits – Choose the file's final bit depth, 16-bit, 24-bit, or 32-bit (float).
Channels – Choose either a stereo (interleaved) file, summed mono file, or split stereo files (i.e. two mono files for left and right channels).
Import as New Track - Indicate whether the resulting mixdown should also be imported as a new track.
Export – Select an external service the mixdown should be exported, including DropBox, SoundCloud, an email recipient, or AudioShare.

For information on the optional Video export settings please see the Video chapter.

AAF Import/Export

One critical component in any professional DAW is compatibility with other systems, since many projects need to be passed between multiple users running multiple systems. For example, a project started on Auria may need to be moved to a traditional desktop system later (or vice-versa), and being able to maintain things like edits and automation data is extremely important. For these situations the Advanced Authoring Format, or AAF, was developed. Outside DAW session data can be shared using Auria's AAF import/export options (projects can be transferred to and from an iPad using iTunes File Sharing or DropBox, instructions are detailed below).

Multiple DAW's support the AAF format, including Pro Tools, Nuendo, Logic, Digital Performer, Samplitude, and others. Each DAW does differ, however, on what elements of AAF it supports. Some support interleaved (i.e. stereo) audio files, while others do not. Some support non-destructive fades and some render all fades destructively on export. And some are designed with video systems in mind and align every region to frame boundaries, while others support sample-based edit lists.

We have attempted to make Auria as compatible as we can with every AAF-compatible DAW on the market, but because of some drastic differences in their AAF support there aren't well-defined guidelines. In the following sections we've included special instructions on specific DAW compatibility where needed.

Exporting an AAF Project from Auria – The only parameter is whether to export stereo audio files as Normal (interleaved) or Split (non-interleaved). If the destination DAW does not support interleaved files (like Logic or Pro Tools) select Split Stereo in the Settings Menu before exporting the project.

The AAF project can be exported either locally or to DropBox. Local projects will be saved in a new folder which can then be transferred off the iPad using iTunes File Sharing. DropBox projects will be uploaded to the connected DropBox account.

Importing an AAF Project into Auria – There are no import settings in Auria as it has been designed to work with all the major DAW's automatically. Frame Boundaries, interleaved audio, and non-destructive fades are all supported when importing into Auria.

To import an outside AAF project from a computer to the iPad, the recommended method is to first zip the entire AAF into one folder which contains both the .aaf file and the individual audio files, and then use iTunes File Transfer to copy that Zip file onto the iPad. Auria will automatically unzip the AAF folder and then display it under the Import AAF file option in the Main Menu.

Note: After importing an AAF project into Auria it is recommended to delete the original AAF folder in order to save space. During the import process Auria will automatically create a new project bundle which includes all the needed audio files, making the original AAF folder no longer needed.

DropBox

Auria allows file management via DropBox for numerous functions. When loading a file in Auria from DropBox (Load Project, Import Audio, Import AAF file) simply tap the DropBox header on the top-right corner of the screen. Save Project to DropBox and Export AAF to DropBox have their own menu items. Additionally, there is an option to Export to DropBox in the Mixdown dialog.

When uploading a project to DropBox Auria will only copy those audio files in use in the project.

SoundCloud

The Mixdown dialog includes an option to Export to SoundCloud. When selected a "Share to SoundCloud" dialog will open requesting SoundCloud credentials in order to transfer the bounced file.

Snapshots

Snapshots of a particular project can be saved and loaded from the Main menu. A snapshot contains all the current settings from a project, minus the actual audio files. In other words, a snapshot contains all of the mixer, editor, and automation information from a project.

This can be useful when alternate versions of a particular project are needed, as it won't use up as much storage space as when using the Save Copy of Project option (which copies all of the project's audio files, too). This way multiple mix versions can be stored and recalled from within a single project.

Auria will automatically save Snapshots of the current project every 10 minutes in a round-robin fashion, adding an additional layer of backups that can easily be recalled.

8

AURIALINK & WIST

AuriaLink

With AuriaLink, two separate iPads running Auria can connect and sync via Bluetooth, enabling two separate projects (one per iPad) in sync, doubling the total number of tracks available. The two projects will play in sync, plus the Edit windows will be intelligently linked so that the transports, scrolling, zooming, and resizing tracks will occur on both devices simultaneously.

Setting up AuriaLink is nearly identical to WIST:

1. On the first device tap open the Settings window
2. Under Device Linking select AuriaLink
3. Tap Connect
4. The iPad will attempt to connect to any nearby iOS devices through Bluetooth.
5. A list of available devices will open, and then tap the desired slave device
6. A confirmation window will open on the second (slave) device
7. Confirm the AuraLink request on the slave device
8. The two devices are now connected via AuriaLink

Transport locked out in slave mode

Once the two devices are connect via AuriaLink, start and stop messages will be automatically sent so that the slave device automatically chases the master; as such the Transport will be unavailable on the slave. Fast-forwarding and rewinding are synced as well, and as the Edit windows are linked scrubbing the Timeline Cursor on the master device will scrub the cursor on the slave, too.

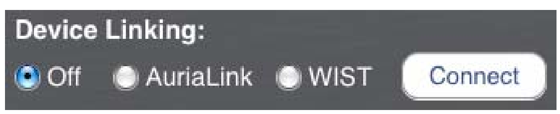
Note: Since synchronization depends on a Bluetooth connection, there will be some unavoidable latency between the two devices. This latency is quite small (usually under 20 ms), so for sync with an outside sequencer or virtual synth will probably not be noticeable. However, with AuriaLink, and especially when recording across two iPads at once, the Bluetooth inconsistency will mean the slave iPad may be ~10 ms behind the start of the Master; this latency total will vary every time a new connection is made. The good news is that this delay should remain constant throughout recording (or at least as constant as two separate audio interfaces without a common clock), so a manual adjustment in region start times should remove the latency from the recording. Altogether the syncing should be about as tight as it was locking two 2” inch recorders together back in the day.

WIST

(Wireless Sync-Start Technology) Created by Korg as a means to connect two iOS devices near each other through Bluetooth, and synchronize compatible apps on the different devices. WIST allows Auria and a separate 3rd-party compatible app to play together in sync.

WIST works by designating one device as the master and the other as the slave, and the master then sends Start/Stop messages to the slave device so the two apps play in sync together. This makes it possible to sync Auria with another iPad running a WIST-compatible sequencer, virtual instrument, or drum machine.

Enabling WIST in Auria is done through the Settings window:



1. On the first device, running Auria, tap open the Settings window (this device will become the slave)
2. Under Device Linking select WIST
3. Tap Connect
4. The iPad will attempt to connect to any nearby iOS devices through Bluetooth.
5. A list of available devices will open, and then tap the second device
6. A confirmation window will open on the second (master) device
7. Confirm the WIST request on the slave device
8. The two devices are now connected via WIST

Once the two devices are connected, pressing play in the master will cause the slave to start playing in sync.

9

VIDEO PLAYBACK

Available as an optional add-on purchase in the Auria Store, Auria can load a video and play it back in sync with a project, and then export a new version of the video which includes the project audio. The video preview window will stay locked with Auria's timeline, and the video preview will even "scrub" in-time with the timeline cursor. The video's main stereo stream can even be imported to its own audio track for further manipulation.

Edit Window with Video Preview. Double-tap preview window to go full-screen

Loading Video

To load a video into Auria:

1. Copy the video into Auria's Documents folder via iTunes File Sharing,
2. Tap the main Menu and select Load Video

3. Select the movie from the File Browser (DropBox is also available when importing video)
4. In the Video Import window, adjust the following as needed:
    - **SMPTE Start** – Used to only import part of a long video file. Start time designates what spot on the video to begin importing from
    - **SMPTE End** – Designates what point, after the SMPTE Start time, to import to
    - **Import Audio Track** – If Yes, Auria will copy the video file's stereo audio stream onto a new audio track
5. Tap OK to finish the import process.

Video Playback

Once the video has been imported, a Video Preview window will appear in the project. This preview will stay in sync with the project itself, and even scrub along with the Timeline Cursor in the Edit window. The video preview is locked to the project timeline with sub-frame accuracy, so audio edits can be performed with better than frame accuracy.

- Double-tap the video preview window to toggle to full-screen
- Video Preview can be toggled on/off from the top Menu

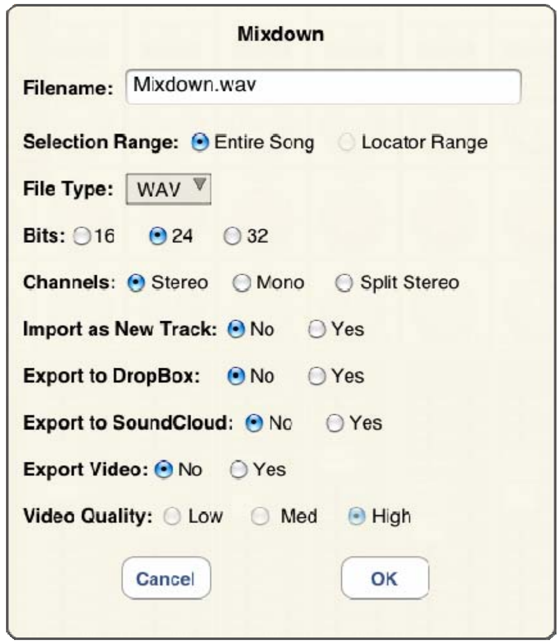
Video section of the Settings window

There are some video-specific options found in the Settings window which affect how Auria interacts with the video file:

- **SMPTE Frame Rate** – Auto-detected when importing the video, but can be changed (or set in a video-less project) to a new value. Supports most common frame rates (plus drop and non-drop) used in film and video.
- **Video Offset** - Determines where in the imported video should correspond to 00:00 in the project. Different than the SMPTE Start/End options when importing, as those determine a section of video to import, while the Offset determines if there should be any pause before starting the video (to allow for count-offs, etc).

Video Export

Auria includes an option to export a video in the Mixdown dialog. When exporting the video the project audio will be used as the video's audio stream. Auria exports all videos in QuickTime mp4 format.



Mixdown window with video export options

To export a video tap open the Mixdown window (under Menu) and select the video options:

- **Export Video** – If Yes, Auria will create a video file which includes the project's audio as its audio content
- **Video Quality** – Determines rendering settings. High is the best looking/largest file/slowest rendering, while Low is the smallest file/quickest rendering/worst looking, with Med in between. Use Low/Med for draft versions and High for the final export.

Auria will export both the video mp4 file and the audio-only version simultaneously.

10

INTER-APP AUDIO AND AUDIOBUS

Inter-App Audio (IAA) Overview

Introduced by Apple in iOS 7, Inter-App Audio enables music apps to share their audio with each other in real time. This allows Auria to record audio from compatible 3rd -party instrument apps, or to use compatible 3rd-party effect apps on both effect inserts and aux channels as if they were standard plug-ins. Any installed apps which support Inter-App Audio will automatically be detected and displayed, in blue lettering, as available plug-ins in Auria's various inserts and aux channels.

The Inter-App Audio protocol (abbreviated as IAA) must be implemented by both the host and synth/effect apps in order to work, so not every music app in the App Store will support IAA inside Auria. Please contact your favorite music iOS makers to find out if they currently support (or plan to support) Inter-App Audio.

Inter-App Audio Requirements:

- Auria 1.13 (or higher)
- iOS 7 (or higher)
- Other 3rd -party apps which support IAA

Differences between native Auria plug-ins and IAA plug-ins:

While Inter-App Audio is designed to function like plug-ins inside Auria, there are some key differences between using Auria's own plug-ins versus IAA:

- IAA plug-in parameters not saved (i.e. knobs and other settings) in projects
- No automation
- Only one instance of a particular plug-in can be active at a time (though multiple unique IAA plug-ins will run at once)
- IAA apps run full screen and do not show any of Auria's display in the background
- Compatible apps have their own IAA bar which is used for switching between apps. It may also have transport controls to start/stop Auria without switching windows

Another important distinction between using Auria's own native plug-ins versus Inter-App Audio is the fact that IAA is connecting two (or more) separate audio apps together. This means that each app must use the same audio settings, like sample rate and buffer (frame) size. iOS 7 will handle configuring this automatically, but it means that any IAA app must be compatible with Auria's current project settings; a 96kHz Auria project requires the other Inter-App Audio app to support 96kHz also.

Note: If the IAA app doesn't support Auria's playback buffer size of 4096 samples then temporarily running Auria in record mode, by arming any audio track, will switch Auria to a smaller buffer size which should then be compatible with any 3rd-party app. As using any IAA instrument automatically puts Auria into record mode this is only an issue when using Inter-App Audio for effect processing.

Hopefully Apple will continue to update the Inter-App Audio spec and add support for features such as saving parameters and automation, but in the meantime IAA is still a very useful feature in the iOS world.

Using IAA with Instruments

As IAA can be used to route audio between apps, probably the most common use will be for recording 3rd-party instrument apps into Auria projects. Doing this is fairly straight-forward within Auria, as the following example will outline.

Blue-colored names in the insert list represent available Inter-App Audio compatible apps

Inserting and Recording an IAA Instrument in Auria

1. On a blank audio track (either mono or stereo as needed) open the Insert section by tapping the FX button
2. In the list of available plug-ins any IAA-compatible apps will appear in blue, so select the desired instrument
3. The audio track will automatically switch to a record-enabled state (required for monitoring)
4. The iPad will switch over to the instrument's display

At this point an instrument will be inserted on one of Auria's audio tracks and be ready for recording. In this example a synth app called Magellan is shown, but other IAA instruments work similarly.

Inter-App Audio compatible apps will display a special bar when run via IAA

5. Tap Record followed by Play on the IAA transport bar to start recording in Auria
6. Play the instrument app
7. Tap the Play/Pause button again on the IAA transport to stop recording
8. If needed simply rewind and re-record, and if not...
9. ...Tap the Auria icon in the transport bar to switch back to Auria
10. Disarm the track and remove the instrument from the insert when finished recording. Close the background app by double-tapping home and sliding the instrument up off the screen

Inter-App Audio instruments can be synced via MIDI Clock just like any other virtual MIDI app, and will appear in Auria's MIDI Settings tab (once they are inserted on an audio track). This way IAA-compatible instruments which utilize sequencing (like drum machines) can be synced with Auria's tempo and recorded in-time with existing projects. Please refer to the MIDI Chapter for more information on how to set this up.

Using IAA with Effects

The other main use of IAA is for processing existing audio in Auria projects, utilizing 3rd-party effects such as reverbs, EQs, delays, etc. As Inter-App Audio is designed to allow using outside apps just like they were plug-ins inside Auria, IAA-compatible effects can be used in several places:

- Track inserts
- Subgroup inserts
- Master insert
- Aux effects channels

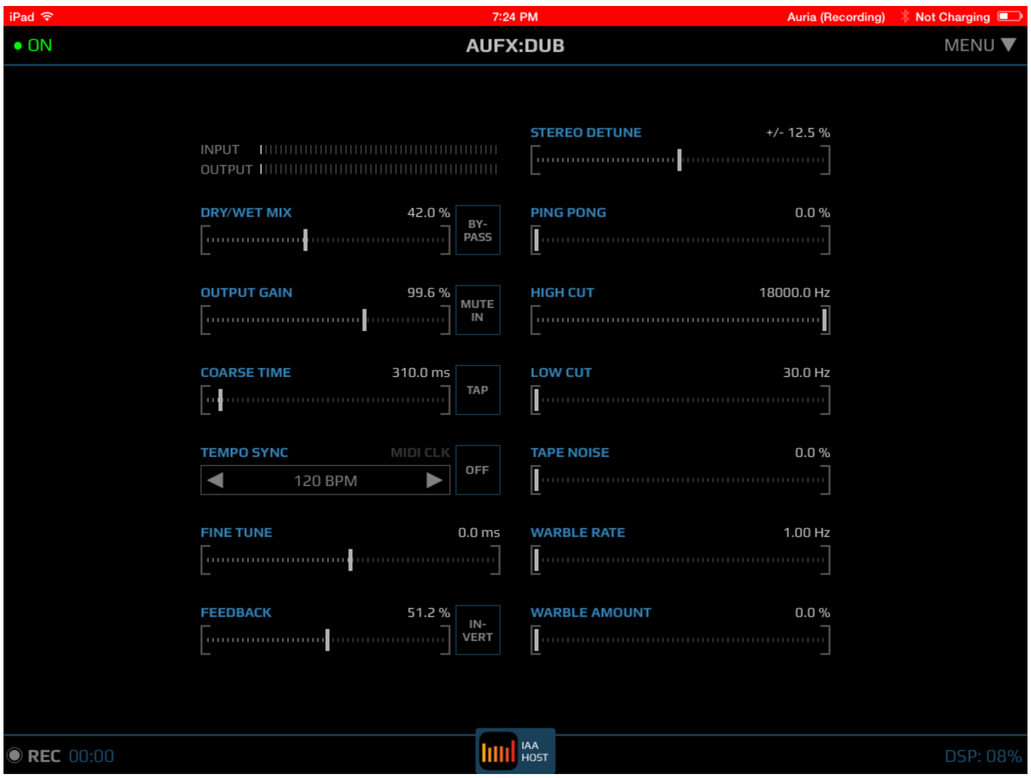
Inserting an IAA Effect on a Track/Subgroup/Master

Using Inter-App Audio a compatible 3rd-party app can be inserted on an audio track, subgroup, or master section, and used like it was one of Auria's own plug-ins (with certain limitations, see above section).

Opening the Insert list on an audio track, with available IAA plug-ins colored blue

To use Inter-App Audio on an insert:

1. On any active audio track, subgroup, or master channel, open the Insert section by tapping the FX button
2. In the list of available plug-ins any IAA-compatible apps will appear in blue, so select the desired effect
3. The display will switch over to the effect's display
4. Tap play in the IAA's transport control to start Aura's playback, if it has one. If the effect does not have an IAA transport control it will be necessary to return to Auria to start playback and then switch back to the effect. Tap the Auria icon in the IAA bar to toggle back to Auria, and the e button in Auria to re-open the IAA plug-in
5. Listen and adjust the effect as needed while playing back the project in Auria



AUFX: Dub from Kymatica running inside Auria via Inter-App Audio. Note the IAA bar at the bottom, used for toggling back to Auria's display

At this point the effect could be left active to run in real-time, and performing a mixdown would include this Inter-App Audio effect as it sounds in the mix. However, just like with regular plug-ins, leaving the IAA running will use valuable system resources, like ram and CPU. Additionally, as only one instance of an IAA plug-in can be active at a time, if two or more instances of this effect are needed then the first instance must be processed in-place, freeing it for use in a new instance.

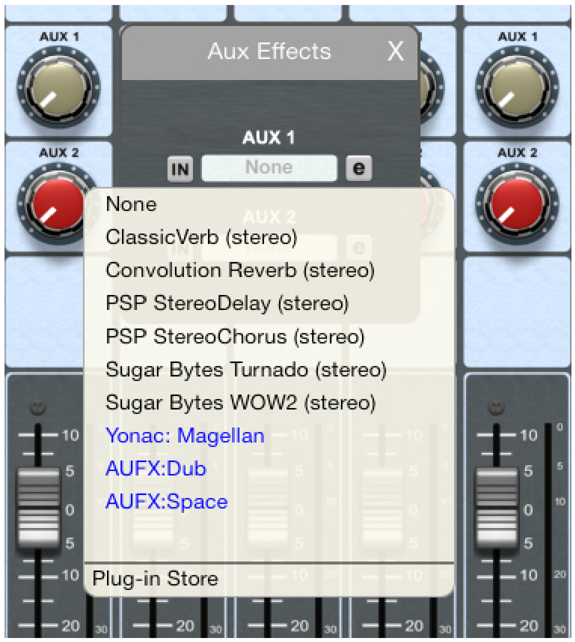
Luckily, Auria's CPU-saving systems work with IAA, just like they do with native plug-ins:

- Track Freeze

- Bounce to a new track using Mixdown : first simply solo either the particular track or subgroup, then use Mixdown with the Import as New Track option enabled. After bouncing the current IAA can be removed or bypassed

Using an IAA Effect as an Aux Effect

Inter-App Audio can also be used through Auria's AUX 1 and AUX 2 channels. This way a single 3rd-party effect, such as a reverb, can be applied to multiple tracks (or subgroups) at once.



Adding an IAA app as an aux effect

To use Inter-App Audio on an aux send:

1. Tap the mixer's AUX FX button to open the Aux Effects window
2. Tap the effect slot for either AUX 1 or AUX 2
3. In the list of available plug-ins any IAA-compatible apps will appear in blue, so select a desired effect
4. The display will switch over to the effect's display
5. Adjust the effect's initial parameters as needed
6. Switch back to Auria by pressing the Auria icon in the effect's IAA bar
7. Turn up at least one corresponding aux send knob on an active track or subgroup
8. Tap Auria's play button in the transport
9. Toggle back to the effect by tapping the e button found in the Aux Effects window which corresponds with the active slot
10. Listen and adjust the effect while playing back the project in Auria, toggling between apps as needed to adjust the various parameters and aux sends as needed

Note: Aux effects are designed to be run in real-time, so in most cases 3rd-party effects should be left active. However if, under special circumstances, it becomes necessary to bounce an aux channel then there is a way to accomplish this:

1. The aux send utilizing this particular effect must be set to Pre Fader mode. This setting can be found in the Mixer tab under Settings
2. Mute every track on the mixer, even those being sent to the aux channel in question – Pre Fade aux mode will keep signal flowing to the aux
3. Mute the other aux return
4. At this point when playing back the project only the necessary aux return should be heard
5. Either Mixdown to a new track, or…
6. …Create a new stereo track, use the Input Matrix to route the L/R signal to that track, and record the aux effect onto that track
7. Un-mute all the other parts and remove the effect from the AUX FX section
8. Adjust the level of the new effect track as needed

AudioBus Support in Auria

Added in version 1.06, Auria supports both Audiobus input and output modes, meaning Auria can both record other Audiobus compatible apps directly onto audio tracks, or Auria can route its own outputs through Audiobus to other apps for recording/processing.

For more information on Audiobus itself, please visit http://audiob.us/

Recording from Audiobus

Auria supports recording an outside Audiobus app directly to an audio track. To do this, first setup Audiobus with the desired apps using the following steps:

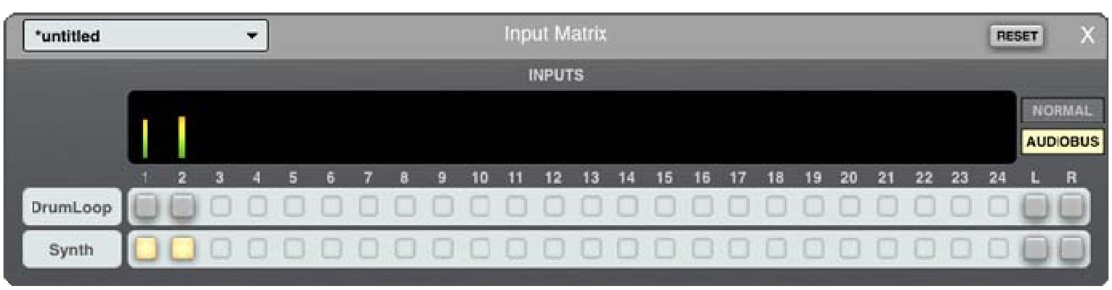
1. Open Audiobus
2. Insert Auria in Audiobus's Output slot
3. Insert the desired source app (such as a synth) in the Input slot

The Audiobus app setup for recording into Auria, using Magellan in the input slot

Auria will automatically create a brand new track (or tracks) that correspond with whatever is in the Input slot and record-enable them so in most cases Auria will now be ready to record.

In advanced setups, Audiobus may also be routed manually to existing tracks:

1. Open the Input Matrix panel, found under the Main menu
2. Tap Audiobus (below the Normal button) on the right-hand side to display Audiobus's ports instead of the normal hardware audio inputs
3. Using the radio buttons, tap the available Audiobus buttons which correspond with the desired destination track (be careful not to select the L and R bus buttons on the right-side of the panel)
4. Record-enable the destination track and proceed to record as normal



Auria's Input Matrix, with Audiobus selected and routed to track 2

Audiobus Live Monitoring

Normally the audio input from Audiobus gets routed into Auria's mixer, through its plug-ins, then out to the speaker/headphones. This kind of routing provides powerful flexibility, because Auria's effects can be added in real time to the Audiobus input. However this also adds an additional layer of latency, enough to be noticeable when recording live into Auria.

To eliminate that additional latency when recording, Auria has a mode called "Audiobus Live Monitoring" (found in the Settings Menu). If enabled, Auria will let Audiobus handle the input monitoring and bypass Auria's mixer, lowering the amount of total latency when recording Audiobus.

Note: Be sure to mute any record-enabled Audiobus tracks in Auria when recording with this mode, as otherwise two separate Audiobus streams will be heard.

Routing Auria's Outputs to Audiobus

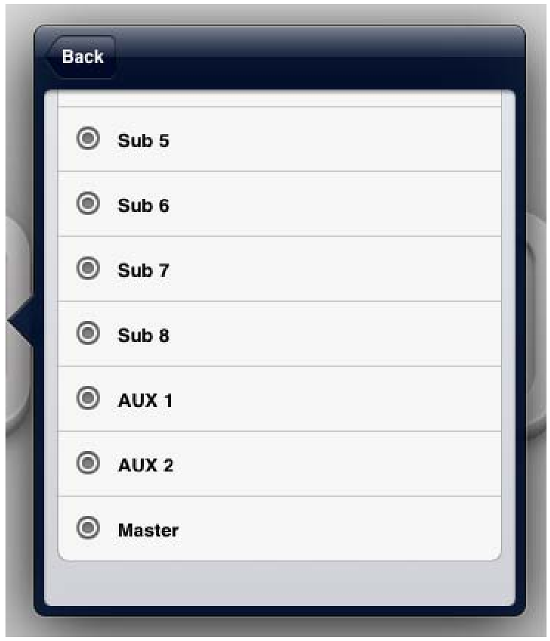
Auria can also route its outputs directly to Audiobus, for additional processing or recording to another app. This routing is accomplished from within the Audiobus app itself.

When inserting Auria in the Input (left-hand) slot Audiobus should display a blue arrow, indicating that Auria has multiple choices of outputs available.

Tap the blue arrow to display all of Auria's available outputs

After tapping the blue arrow, all of Auria's available outputs will be listed. These include:

- Subgroups
- Aux 1 & 2
- Master (Main L/R Bus)



Tap the desired Auria output to route it to Audiobus

Once a specific output has been selected, any Audiobus app inserted in the Output slot will receive that audio stream for recording or additional processing purposes.

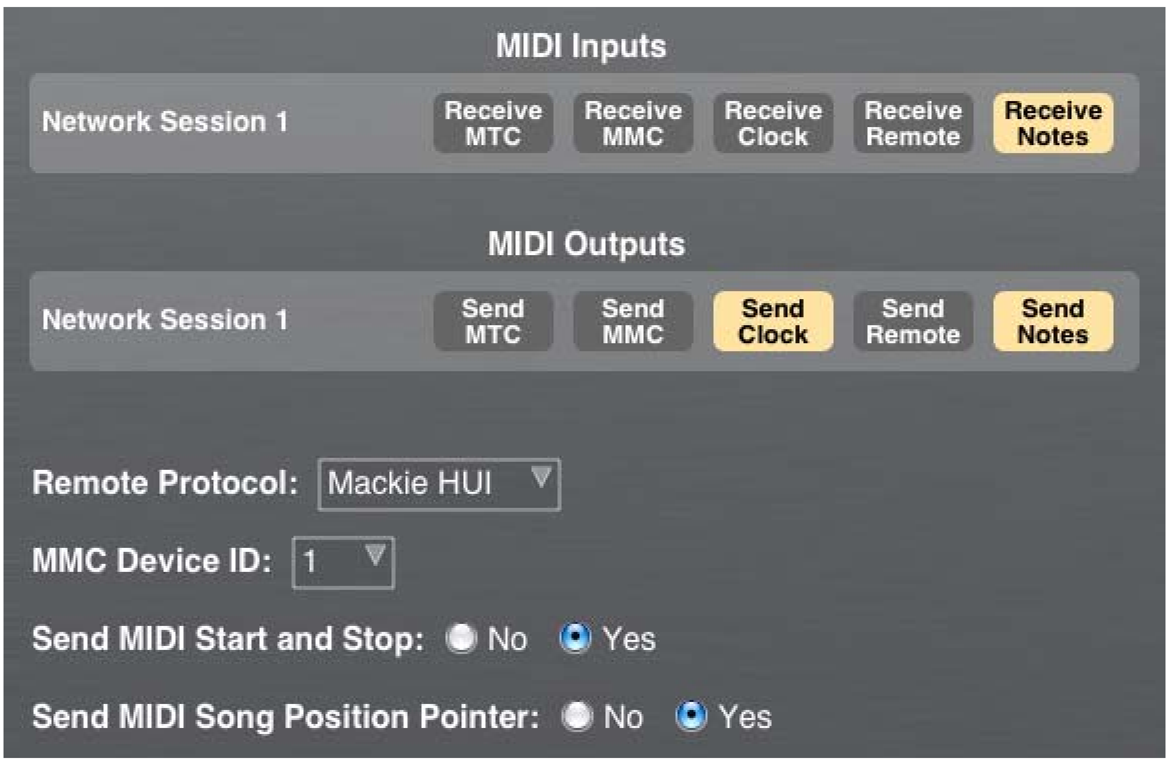
Audiobus Performance

Audiobus sets a fixed hardware buffer size of either 256 or 512 samples when any audio app uses it. Auria normally uses a much higher buffer size during playback, so reducing the buffer size increases the CPU usage considerably. When using Audiobus with Auria it is recommended to set Audiobus's Hardware Buffer Size to the larger 512 samples setting, and to create a new project, with minimal effects, for recording Audiobus apps into that project. If recording into existing projects then freezing tracks or bypassing effects can help reduce the CPU and memory usage and allow Audiobus to run more effectively. Running other audio apps decreases the amount of CPU and memory available to Auria, so only necessary apps should be running alongside Auria and Audiobus.

11

MIDI SYNC & REMOTE

Auria can send and receive many kinds of MIDI control information, including sync, transport, and even mixing parameters. These can be used to lock Auria together with another sequencer, control plug-ins from outside controllers, and even control Auria's entire mixer from a control surface. All of the MIDI parameters can be found under the Settings section.



MIDI Settings Window

Supported MIDI Formats

Auria can send and receive many types of MIDI information, including:

- **MTC** - MIDI Time Code is a time-based synchronization protocol where a Master device transmits SMPTE-based time information to slave devices. Most often used by primarily time-centric systems like video systems, multi-track tape, or other DAWs. To slave another device (DAW, sequencer, tape machine) to Auria enable sending MTC. The specific frame rate can be selected from the Settings window. To make

Auria chase an external device enable receiving MTC and enable External Sync under the Transport Options.

- **MMC** - MIDI Machine Control transmits Transport messages between devices, such as Play, Stop, Rewind, Fast Forward, and Record.
- **Clock** - MIDI Clock is a tempo-based synchronization protocol where a Master device transmits continuous MIDI ticks to slave devices. Most often used by MIDI sequencers, drum machines, or synth's with tempo-based parameters. Auria can send MIDI Clock so outside sequencers can sync to Auria, this generally requires also using MIDI Start and Stop and Song Position Pointer (SPP).
- **Remote** – MIDI Remote Control allows outside control of Auria's mixing parameters, including faders, pan, aux controls, even plugin control, allowing it to be controlled by an external control surface. This requires 2-way communication between devices so both an In and Out must be selected.
- **Notes** - For external MIDI control of Auria's plugins; this allows controlling individual plugin parameters inside Auria from an outside source (controller, sequencer, etc). The individual plugin must specifically support MIDI control for this to work.

MIDI Devices

MIDI under iOS can be transmitted through several types of connections, both physical and virtual.

Virtual MIDI

iOS apps can transmit MIDI back and forth directly between each other using Virtual MIDI. Whenever a Virtual MIDI compatible app is running alongside Auria its name will appear in Auria's list of detected connections.

Physical Interface

Hardware MIDI interfaces can be connected directly to the iPad using the MFi connector, and will appear in Auria as available MIDI ports. These are used to connect external physical controllers like keyboards and control surfaces.

Network Session

iOS devices can also connect to the outside world using wireless network connections with a desktop/laptop computer. Current versions of Mac OSX have this functionality built-in under the "Audio MIDI Setup" utility, just make sure that the iPad and Mac are on the same network. Once a session has been created the iPad and computer can transmit MIDI back and forth over the air. For more information read Apple's Knowledge Base article on Sharing MIDI Information over a Network.

Windows users will need to install a 3rd-party utility to add network MIDI support such as rtpMIDI. This site also includes a helpful tutorial section for setting up network MIDI on PC.

MIDI Sync

Auria can be synchronized to outside systems by using some form of MIDI sync. These outside systems include both apps running simultaneously on the iPad as well as devices running on entirely different hardware.

Some examples would include:

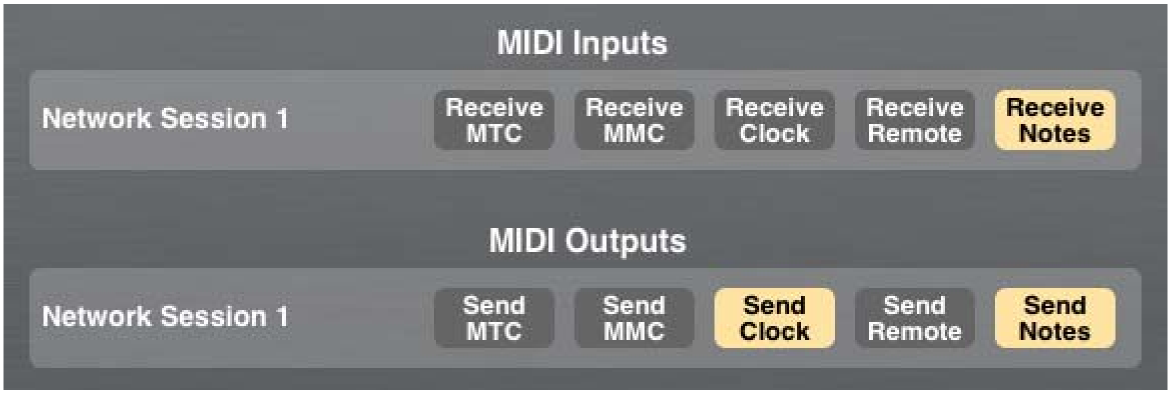
- An iOS app with its own built-in sequencer, like a drum machine or workstation-style synthesizer
- A desktop DAW such as Pro Tools

When discussing the synchronization of two systems there are two important distinctions: which device is the master and which is slave? Simply put one system needs to be the reference system (the Master), which runs normally, and the other (the Slave) chases that reference. If the master speeds up or slows down the slave needs to adjust accordingly. In the days of tape the general rule of thumb is that the slowest machine needed to be the master, because it was easier for a quick system to follow a slow one. In those cases the tape machine was generally the master and the other systems (mixing console automation, hardware sequencers, etc) chased tape. With most systems today being non-linear in fashion (like Auria) this is less important, but it isn't a bad idea to keep that rule in mind when deciding which device should do what.

Auria is capable of both being a master device (the master reference which sends out the sync information) and a slave device (chasing another system), so it is very powerful when needing to integrate with others.

Auria as Master

Auria is capable of sending both MIDI Clock as well as MIDI Time Code (MTC), both of which can be used for synchronization. Generally speaking MTC, because it uses SMPTE divided into sub-frames, is a finer-grain reference and is usually preferred when tight sync is needed. MIDI Clock is tempo-based, so at lower BPM has a higher amount of drift. In practical terms most devices out in the world only support one or the other method so use whichever format is compatible.



To setup Auria as a master:

1. Tap open the Settings Menu
2. Tap the MIDI tab
3. Find the appropriate MIDI Output port that corresponds to the slave(i.e. the app name or Network Session)
4. Choose and tap either Send MTC or Send Clock for that port, depending on what the other device requires
5. Enable other MIDI parameters such as SPP and Start/Stop as needed by the slave device

Auria will now send either MTC or MIDI Clock whenever playback is started.

Auria as Slave

Auria can chase external MTC (but not MIDI Clock), so it can sync to other DAW's like Pro Tools.

To setup Auria as a slave:

1. Tap open the Settings Menu
2. Tap the MIDI tab
3. Find the appropriate MIDI Input Port that corresponds to the external master device
4. Tap Receive MTC for that port
5. Tap open Transport Options
6. Enable External Sync

When External Sync is enabled a clock symbol will appear on the transport's Play button, denoting the current sync state:

- Solid green clock means Auria is locked to external timecode
- Flashing green clock means Auria is chasing but not yet locked
- Blue clock means no timecode is currently detected

Note on using chase sync: When Auria is set to chase an outside source, Auria will vari-speed audio playback to match the other clock source, if needed. One caveat is that when slaved to external sync only the main audio outputs (1 & 2) are active.

Plug-in Control

Some of Auria's plug-ins support MIDI control, such as the optional Fabfilter effects. This means that those plug-in's parameters can be controlled by an outside controller or sequencer, allowing more flexibility when automating changes or dialing in specific settings. For example a filter's cutoff frequency could be linked to a controller keyboard knob, allowing the frequency control to be "played" from the keyboard.

To enable plug-in MIDI control:

1. Tap open the Settings Menu
2. Tap the MIDI tab
3. Find the appropriate MIDI Input Port that corresponds to the controller
4. Tap Receive Notes for that port

For more information on using MIDI control with Fabfilter plug-ins consult that plug-ins individual section later in the User Guide under the MIDI Learn heading.

Remote Control

Auria supports two different remote control protocols, Mackie's HUI (Human User Interface) and MCU (Mackie Control Universal), for controlling Auria from an outside control surface. Auria supports many aspects of the HUI and MCU specs, allowing integration with options like:

- Channel faders and pan pots
- Aux sends
- Metering
- Time/Counter display
- Transport control
- Markers
- Plug-in parameters
- Track names

To enable remote control:

1. Tap open the Settings Menu
2. Tap the MIDI tab

3. Find the appropriate MIDI Input and Output Ports that corresponds to the control surface
4. Tap both Send and Receive Remote for that port (remote control requires 2-way communication)
5. Select the appropriate Remote Protocol, either HUI or MCU, depending on the connected device

Note: When using the Mackie MCU Pro configure it for "Logic Mode" for compatibility with Auria.

Other MIDI Settings

The MIDI Settings window contains some additional MIDI parameters:

Remote Protocol – Selects the MIDI remote control protocol Auria should emulate, which enables controlling Auria via external control surface. Both Mackie's HUI and MCU protocols are supported.

MMC Device ID – Assigns a unique ID MIDI Machine Control device ID number to Auria. Every MMC device in a chain requires a unique ID assignment.

Send MIDI Start and Stop – Enables transmitting MIDI Start and Stop messages when tapping Play and Stop on the transport. Used to remotely start and stop external MIDI devices, and is generally required when transmitting MIDI Clock.

Send MIDI Song Position Pointer – Transmits the current project's song position in MIDI.

Part Three – Reference

12

MENU BAR

The Top Menu Bar

1. Mixer Window
2. Edit Window
3. Undo
4. Redo
5. Main Menu
6. Edit Menu (only available in Edit Window)
7. Process Menu (only available in Edit Window)
8. Project Name
9. Sample Rate/USB Interface Indicator
10. Grouping
11. Locator In/Out Point
12. Transport
13. Counter

Main Menu

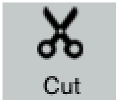
- **New Project** – Creates a new project. Enter a name for the project, as well as select the desired sample rate and number of initial audio tracks. All audio tracks created initially will be mono—to create stereo tracks, add them after the project has been created.
- **Load Project** – Loads a previously saved project. Locally stored projects will appear in the list. Projects stored in a DropBox account can be accessed by touching the DropBox menu header.
- **Load Recent** – Loads one of the 6 most recent projects.
- **Save Copy of Project** – Saves a copy of the current project locally under a different name.
- **Load Snapshot (Under Snapshots)** – Loads a previously saved mixer snapshot.

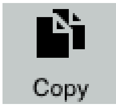
- **Save Snapshot (Under Shapshots)** – Saves the current mixer, automation, and editor settings (essentially everything in the project except the audio files). Good for trying alternate mixes of a particular song without resorting to duplicating the whole project and its audio files.
- **Save Project to DropBox** – Saves a copy of the project to DropBox.
- **Rename Project** – Changes the name of the current project.
- **Import Audio** – Imports audio files (WAV, AIFF, M4A). Locally stored files will appear in the list. Files stored on a DropBox account can be accessed by touching the DropBox menu header.
- **Add Track** – Creates the specified number of new audio tracks (Mono/Stereo).
- **Mixdown** – Creates a final mixdown of the project. For more information see the Mixdown section.
- **Import AAF file (Under AAF File)** – Imports an AAF file. Locally stored AAF files will appear in the list. AAF files stored on a DropBox account can be accessed by touching the DropBox menu header.
- **Export AAF file (Under AAF File)** – Exports project as an AAF file and saves it locally.
- **Export AAF to DropBox (Under AAF File)** – Exports project as an AAF file and saves it to DropBox.
- **Reset Mixer** – Resets all parameters on the mixer.
- **Clear All Groups** – Clears all grouped channels.
- **Clear All Automation** – Clears all automation.
- **Input Matrix** – Opens the Input Matrix. This is used to route physical inputs (or Audiobus ports) to their destination tracks for recording. Selecting L R will record the main stereo bus onto that track, i.e. perform a bounce.
- **Output Matrix** – Opens the Output Matrix. This panel is used to route subgroups, aux sends, and the master output to assignable physical outputs on an attached multi-channel audio interface.
- **Settings** – Opens the settings page. See Settings chapter.
- **User Guide** – Opens this guide on the iPad.
- **Auria Forum** – Link to Auria's official discussion board.
- **Auria Store** – Access the optional In-App Store, with purchasable plug-ins, convolution IR files, and demo projects.

Edit Menu

Notes: If any of the following commands are also found on the Icon Toolbar their corresponding icon is shown below. Also, this menu only appears when viewing the Edit Window, it is not visible from the Mix Window.

- **Undo** – Undo the most recent action.
- **Redo** – Redo the previous action.

- 

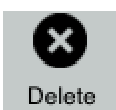
Cut – Cuts the region or highlighted selection.
- 

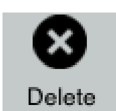
Copy – Copies the region or highlighted selection.
- 

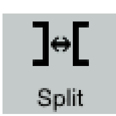
Paste – Pastes the region or highlighted selection most recently cut or copied to the cursor.
- 

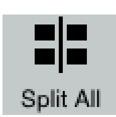
AudioCopy™ – AudioCopies™ the region or highlighted selection. Note: AudioCopy/Paste works at 16-bit only. Any higher bit-depth audio will be converted to 16-bit during the Copy/Paste. For more information see the section on AudioCopy/AudioPaste.
- 

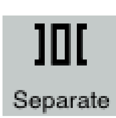
AudioPaste™ – AudioPastes™ the region or highlighted selection most recently AudioCopied™ at the cursor. Note: AudioCopy/Paste works at 16-bit only. Any higher bit-depth audio will be converted to 16-bit during the Copy/Paste. For more information see the section on AudioCopy/AudioPaste.
- **Export to AudioShare** – Copies the selected region(s) to the optional AudioShare app, helpful when needing to more audio into an outside app, including those that only support the General Pasteboard.
- **Import from AudioShare** – Pastes audio from the AudioShare app into the current Auria project.
- 

Delete Track – Deletes the selected track.
- 

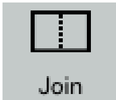
Delete Region – Deletes the selected region.
- 

Delete Control Point – Deletes the selected control point.
- 

Split – Splits the selected region at the cursor.
- 

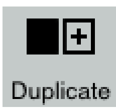
Split All – Splits every region in the project at the cursor, and selects all regions to the right of the cursor. Useful to insert time in an existing project.
- 

Separate – Separates the highlighted section of a region.

- 

Join – Joins two or more selected regions together, creating one new region.
- 

Select All – Selects all regions in the edit window.
- **Loop to Selection** – When a region is selected tap this command to automatically set the Locators to match the region's beginning and end times; it will also enable Loop playback.
- 

Select Highlighted Regions - Selects any regions currently highlighted (similar to using a "lasso" tool).
- 

Duplicate – Duplicates the selected region. For a quicker version see Automatic Region Duplication.
- 

Lock/Unlock Region – Toggles between locking/unlocking the selected region in place on the timeline.
- 

Mute/Unmute Region – Mutes/unmutes the selected region(s).
- **Ripple Edit Mode** – Toggles between normal editing and Ripple editing, where performing an edit automatically moves affected regions on the timeline. For more information see the Ripple Mode section of the Editing chapter.
- **Link Locators and Highlight** – When enabled setting the Locator points will also automatically highlight the same section, and vice-versa. Turned off by default.
- **Show Sub/Master Tracks** – Toggles On/Off display of the subgroups and Master channel in the Edit window. Primarily toggled on when needing to edit automation data in those particular channels.
- **Show Icons** – Toggles the Icon Toolbar on and off. For more information see the section on the Icon Toolbar.

Process Menu

Please note that the following options (except crossfades) are destructive in nature, and will alter the actual recording itself. This menu is only visible from the Edit Window.

- **Gain** – Changes the gain of the region or highlighted selection by the desired amount.

- **Normalize** – Normalizes the region or highlighted selection to the desired level. Choose between Peak and RMS modes, as well as how clipping will be treated. The ‘Release’ parameter is only active if ‘Limit’ is selected. Limiting is processed by the built-in Brick Wall Limiter.
- **DC Offset** – Removes DC offset from the region or highlighted selection.
- **Reverse** – Reverses the region or highlighted selection.
- **Silence** – Converts the region or highlighted selection to silence.
- 

Crossfade – Creates a crossfade between the selected regions (two overlapping regions must be selected using the Multi-Select Tool). Selecting a crossfade region will display the crossfade curve options in the top-right corner. See larger section on Crossfades for more information.
- **Reset Fades** – Reset the fades of the selected region.
- **Condense Regions** – This destructive process will scan all the used audio regions in the current project and automatically delete any unused audio from the project, specifically regions with edited trim handles. Useful for regaining storage space in a project with many edited regions. Can also be used on specific regions by first selecting those regions before choosing this option; if no regions are selected then the entire project is scanned.

Project Name – Displays name of the current project. Double-tapping opens Rename Project dialog. Also global Solo light appears here, lighting up whenever any channel is soloed. Tap the flashing Solo indicator to turn off all solos.

Sample Rate & USB Interface Indicator – Displays the sample rate of the current project and indicates whether the internal microphone is in use, or an MFi/USB interface is connected.

Grouping - Assigns channel grouping. When tapped, channels can be grouped or un-grouped by touching their respective faders. Tap control again when done adding channels to the group.

Locator In/Out Point – Sets locator in/out points. Used in loop mode or Auto-Punch. Enable looping or Auto-Punch by tapping the Counter display in the top-right corner and selecting Loop or Auto-Punch.

Transport

- **Rewind** - Double tap to rewind the cursor to 0:00
- **Fast Forward** - Double tap to move the cursor to the end of the last region

- **Stop -** Double tap to rewind the cursor to 0:00 (or Start Locator, if it is set)
- **Play** – Tap to start playback. When External Sync is enabled a clock symbol will appear on the play button: a solid green clock means Auria is locked to external timecode, a flashing green clock means Auria is chasing but not yet locked, and a blue clock means no timecode is currently detected
- **Record** - Activates record mode; press Play to begin recording

Counter

Displays the current position of the cursor. Touch for additional transport options.

Transport Options

- **Time Format** – Toggles between Min:Sec, Samples, Bars:Beats, and SMPTE
- **Set Marker 1-4** – Places a marker at current cursor position.
- **External Sync** – When enabled Auria will chase external MTC sync.
- **Go to Marker 1-4** – Moves cursor to marker position.
- **Clear Locators** – Clears the locator points.
- **Clear Markers** – Clears all markers.
- **Lock Locators** – Locks Locate points, preventing them from being moved.
- **Lock Markers** – Locks existing markers, preventing them from being moved.
- **Loop** – Enables a playback loop between the locator points.
- **Auto Punch** – Records only between set locator points.
- **Auto Scroll** – When enabled, the edit window will automatically scroll to keep the cursor in view during playback.
- **Metronome** – Turns the metronome on or off. Metronome settings can be found in the Time Settings window.
- **Count-In** – Turns metronome count-in on or off.
- **Auto Return** – When enabled, stopping playback (by tapping Stop) results in the cursor returning to the same spot as when Play was pressed. Off by default.

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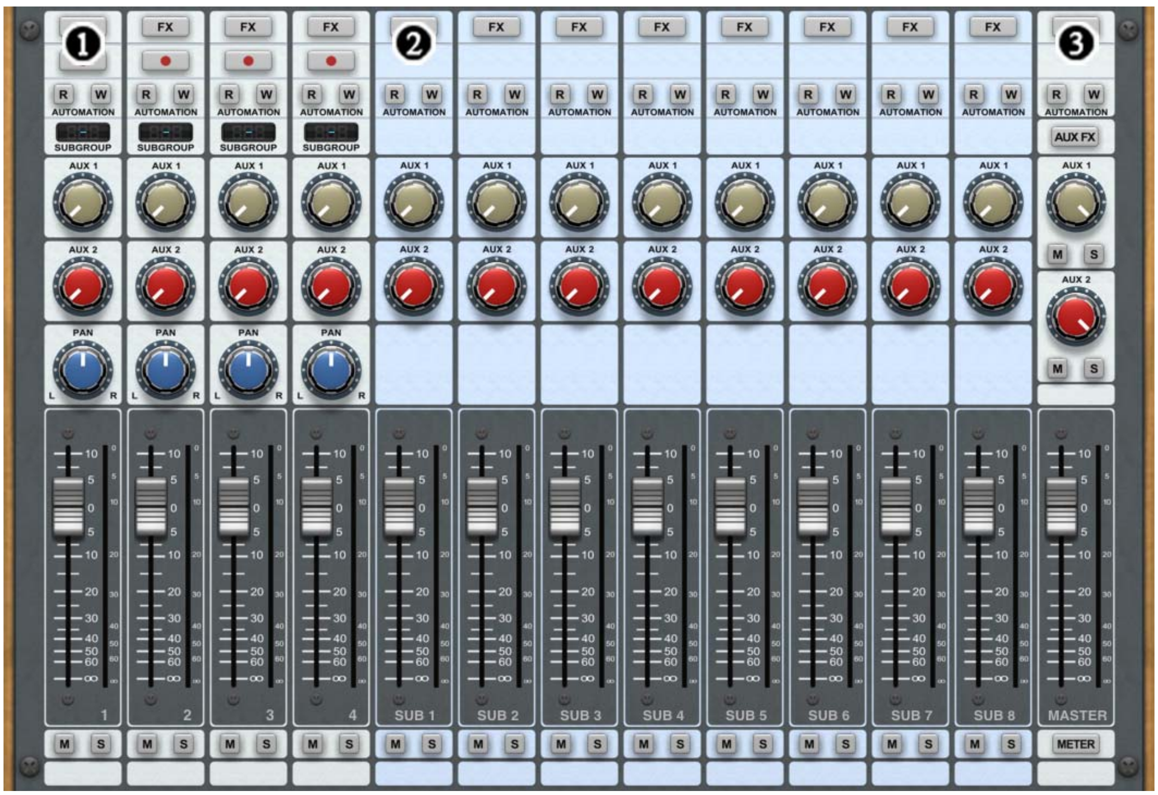
MIX WINDOW



The Mix Window

The mixer window consists of:

1. Channels
2. Subgroups
3. Master Channel

Swiping left or right with one finger will scroll to bring off-screen areas of the mixer into view. The mixer window can be used in either landscape or portrait orientations—turning the iPad vertically into portrait mode will elongate the mixer, enabling 100mm faders.

Channels

1. FX – Opens the PSP ChannelStrip and Effects Inserts, as well as the Freeze Track, Saturation (SAT), and Polarity Inversion (Ø) buttons.

2. Record Enable – Arms the track for recording. Press and hold to open Record Options menu:

- Record Effects – when enabled, active effects are recorded. When disabled, effects are monitored only.
- Set Record Level – press to adjust input level.
- Input Matrix – press to open the Input Matrix. Here, active audio inputs can be routed to all tracks available for recording.

3. R/W – Read/Write automation: 'Read' mode enables playback of previously written automation; 'Write' mode arms the track to record new automation of any modifiable parameter.

4. Subgroup – Assigns track output to either the main L/R mix or one of eight stereo subgroups.

5. AUX 1/AUX 2 – Adjusts the level of audio being sent to the AUX FX.

6. Pan – Adjusts the stereo spread of a mono channel or the balance of a stereo channel. Pan Law can be selected in Menu > Settings.

7. Fader – Adjusts the level of channel output being sent to the Master Channel.

8. Volume Meter – Displays the channel output level. Meter Type (Peak/RMS), as well as Pre-Fader/Post-Fader metering, can be selected in Menu > Settings. A (∞) symbol above the meter indicates that it is a stereo track.

9. M/S – Mute/Solo. Soloing a track will also solo the AUX returns on the Master Channel.

10. Track Name – Double-tap to modify the track name.

Subgroups

1. FX – Opens the PSP MasterStrip and Effects Inserts, as well as the Saturation (SAT) and Polarity Inversion (Ø) buttons.

2. Automation - Read/Write automation: 'Read' mode enables playback of previously written automation; 'Write' mode arms the track to record new automation of any modifiable parameter.

3. AUX 1/AUX 2 – Adjusts the level of audio being sent to the AUX FX.

4. Fader – Adjusts the level of channel output being sent to the Master Channel.

5. Volume Meter – Displays the channel output level. Meter Type (Peak/RMS), as well as the order in which the signal is metered (Pre-Fader/Post-Fader) can be selected in Menu > Settings.

6. M/S – Mute/Solo. Soloing a track will also solo the AUX returns on the Master Channel.

7. Track Name – Double tap to modify the track name.

Note: First generation iPads will only have 4 Subgroups available.

Master Channel

1. FX – Opens the PSP MasterStrip and Effects Inserts, as well as the Brick Wall Limiter.

2. R/W – Read/Write automation: 'Read' mode enables playback of previously written automation; 'Write' mode arms the track to record new automation of any modifiable parameter.

3. AUX FX – Opens the AUX 1/AUX 2 Effects.

4. AUX 1/AUX 2 – Adjusts the overall level of all audio being sent through the AUX 1/AUX 2 FX to the Master Channel.

5. AUX M/S –Mute/Solo for the AUX returns.

6. Fader – Adjusts the signal level of the Master Channel.

7. Volume Meter – Displays the signal level. Meter Type (Peak/RMS) can be selected in Menu > Settings.

8. Meter – Opens the PSP MasterMeter.

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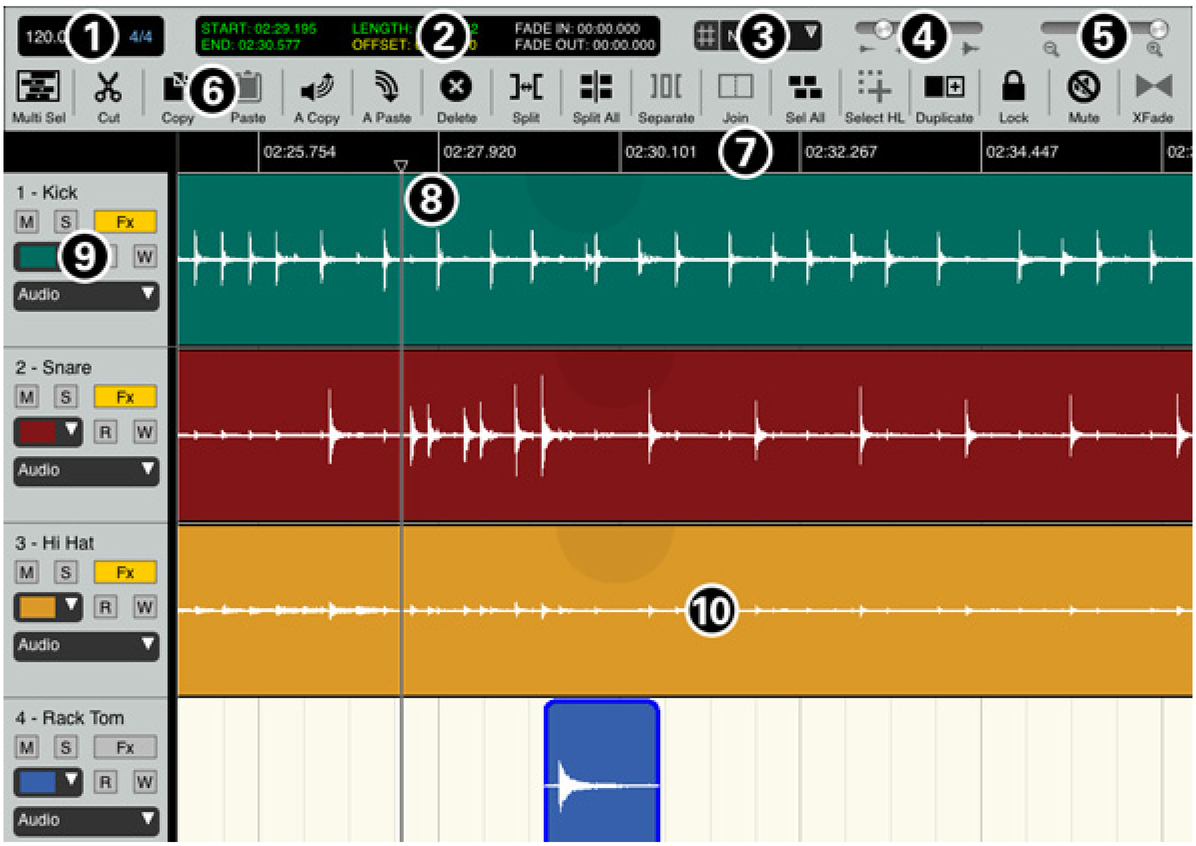
EDIT WINDOW



The Edit Window

The Edit Window consists of:

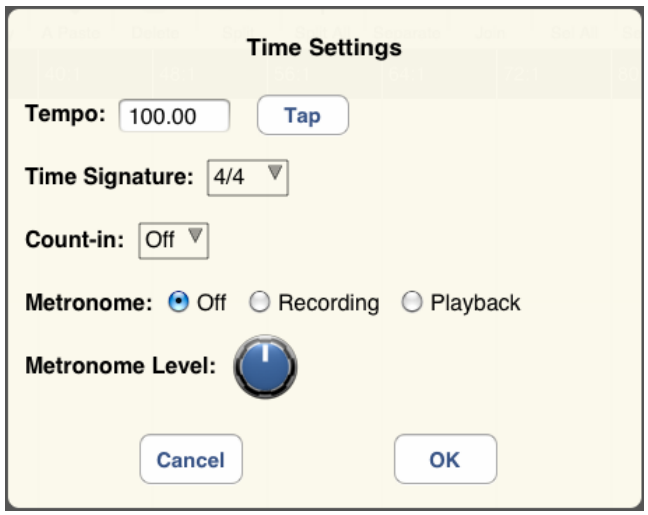
1. Time and Tempo Settings
2. Region Info Box
3. Snap
4. Waveform Display Gain
5. Zoom
6. Icon Toolbar
7. Timeline Ruler
8. Cursor
9. Track Display
10. Regions

Swiping left or right with one finger will scroll to bring off-screen areas into view, swiping up or down will scroll between tracks. Pinching with two fingers will zoom either vertically or horizontally, though only one direction at a time.

The Edit Window can only be used in the landscape orientation.

Time and Tempo Settings

Double-tap to open the Time and Tempo Settings dialog, used to set tempo, time signature, and adjust metronome settings:



Time and Tempo Settings Dialog

Tempo – Sets the project tempo (BPM). Tempo can either be entered through typing the specific numeric value (including fractional tempos, i.e. 120.25 BPM), or by using the Tap button to enter the tempo through tapping.

Time Signature – Determines the time signature of the project.

Count-in – Assigns the number of measures played during count-in. Can also be toggled On/Off in the Transport Options menu.

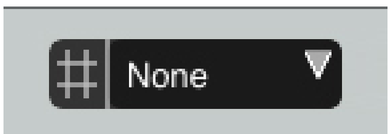
Metronome – Toggles between no metronome (Off), metronome during recording (Recording), and metronome during both recording and playback (Playback). Can also be toggled On/Off in the Transport Options menu.

Metronome Level – Determines the volume of the metronome.

Region Info Box

Displays information pertaining to either the selected region or the highlighted selection, using the same time units as the current Time Format setting. When actively editing a region it will also show both how far the region has been moved (as Shift), or how much time stretching is being applied (as a percentage of the original). Regions undergoing gain adjustments will also display the current amount of gain change.

Snap Menu



Determines the unit of time to which regions will snap, based on the selected Time Format.

All Available Time Formats when Snapping:

- None
- Events (i.e. other regions)
- Cursor
- Markers
- Locators
- Highlight

Time Format: Min:Sec

- .0001 Second
- .001 Second
- .01 Second
- .1 Second
- 1 Second

Time Format: Samples

- 1 Sample
- 10 Samples
- 100 Samples

- 1000 Samples
- 10000 Samples

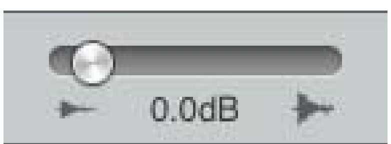
Time Format: Bars:Beats

- Bars
- Beats
- 1/4 Beat
- 1/8 Beat
- 1/16 Beat

Time Format: SMPTE

- Frame
- 1/2 Frame
- 1/4 Frame
- 1/8 Frame
- 1/16 Frame

Waveform Display Gain



Increase or decrease the displayed waveform gain. Double-tapping the control resets the display gain to 0 dB. Moving the waveform gain slider fully to the left will disable waveform drawing, allowing much faster re-draw rates for slower iPads and larger projects and speeding up scrolling in the Edit window.

Zoom

Zoom between sample level (100%) and length of the longest region (0%).

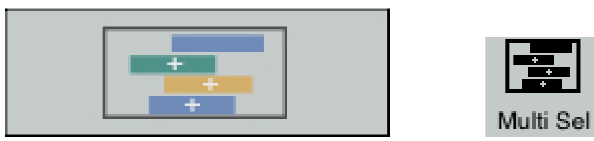
Icon Toolbar

The Icon Toolbar

A list of the most commonly-used editing commands, simply tap an icon to perform that command. Details on each of these specific functions will be found in an [earlier section](). This bar can be toggled On/Off from the Edit Menu by un-checking Show Icons.

Note: When a particular function is not currently possible it will appear as a lighter shade of gray instead of black. In the above example of the Icon Toolbar the XFade option is currently gray, meaning that function cannot be performed - in this case a crossfade requires a very specific situation where two overlapping regions are selected, so only in that instance will the icon turn black.

Multi-Select Tool

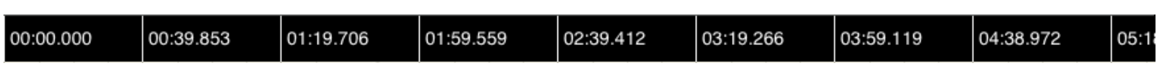


Multi-Select Tool. Larger version is shown when Icon Toolbar is disabled, smaller version on the right when the Icon Toolbar is turned on.

Allows the selection of multiple regions or tracks at once. Tap once to begin selecting (the tool will begin to flash), then touch each desired region or track; tap the tool again when done selecting objects.

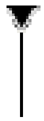
Group Lock Mode – Double-tap the Multi-Select Tool to “lock” the current multi-selection to prevent accidental de-selection when moving or editing the objects; double-tap again to unlock the selection.

Timeline Ruler



Displays the project time in relation to the regions. Touch anywhere along the timeline to move the cursor to that spot. When locate points are set (by touching the Locator In/Out Point button in the Menu Bar), they will appear on the timeline, and can be dragged to change the size of the loop/Auto-Punch. To change the time format of the ruler select an option in Transport Options.

Cursor



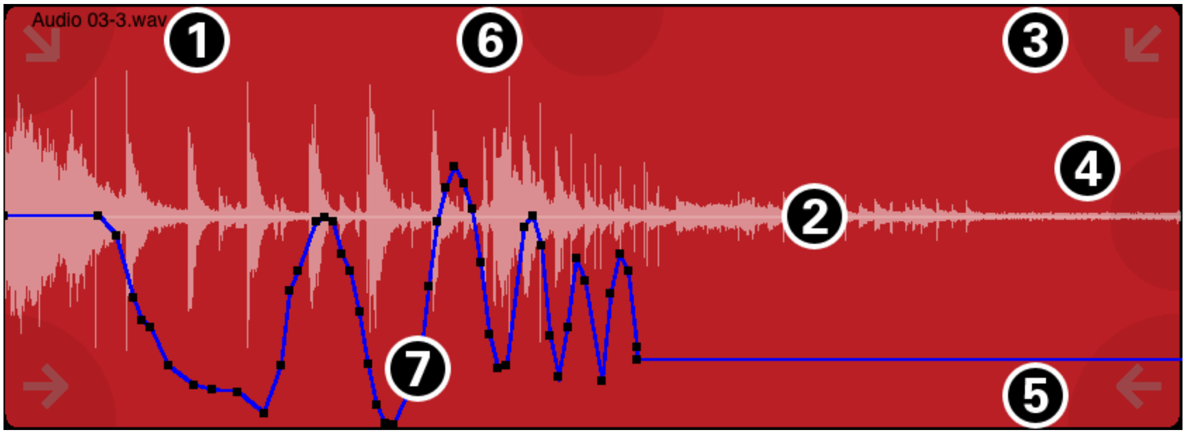
Indicates the current time in the project.
Touch and hold to drag along the timeline.

Tracks

1. **Track Name**
2. **M** – Mute
3. **S** – Solo
4. **R** – Read; enables playback of previously written automation.
5. **W** – Write; arms the track to record new automation of any modifiable parameter.
6. **Color** – Changes the color of regions displayed within the track.
7. **FX** - Opens the PSP ChannelStrip and Effects, as well as the Freeze Track, Saturation (SAT), and Polarity Inversion (Ø) buttons.
8. **Waveform/Automation** – Selects which automation parameter is displayed. 'Audio' displays the waveform only.
9. **Meter** – Displays the channel output level. Meter Type (Peak/RMS), as well as the order in which the signal is metered (Pre-Fader/Post-Fader) can be selected in Menu > Settings.
10. **Rec** – Record Enable button.

To move a track, touch and hold the track name. It will pop up, indicating it is ready to be moved, and can be dragged to its desired position.

Regions



1. **Region Name** – Corresponds to region's filename.
2. **Waveform**
3. **Fade Handles** – The arrows in the upper left and right are fade controls. Tap, hold, and swipe to adjust. See the Fades section for more information.
4. **Region Duplication Handle** – The arrow on the far-right is the automatic duplication handle. See the Automatic Region Duplication section for more information.
5. **Trim Handles** – The arrows in the lower left and right are the trim handles. Tap, hold, and swipe to adjust. See the Handles section for more information.
6. **Gain Handle** – The arrow at the very top in the middle is the gain handle. See the Gain Handle section for more information.
7. **Automation** – Displays the automation for the parameter that is currently selected in the Automation Display Menu. Tap points to select or move them. Add a point by touching anywhere on the line. Points can be deleted using the Edit menu. See Automation Chapter for more information.

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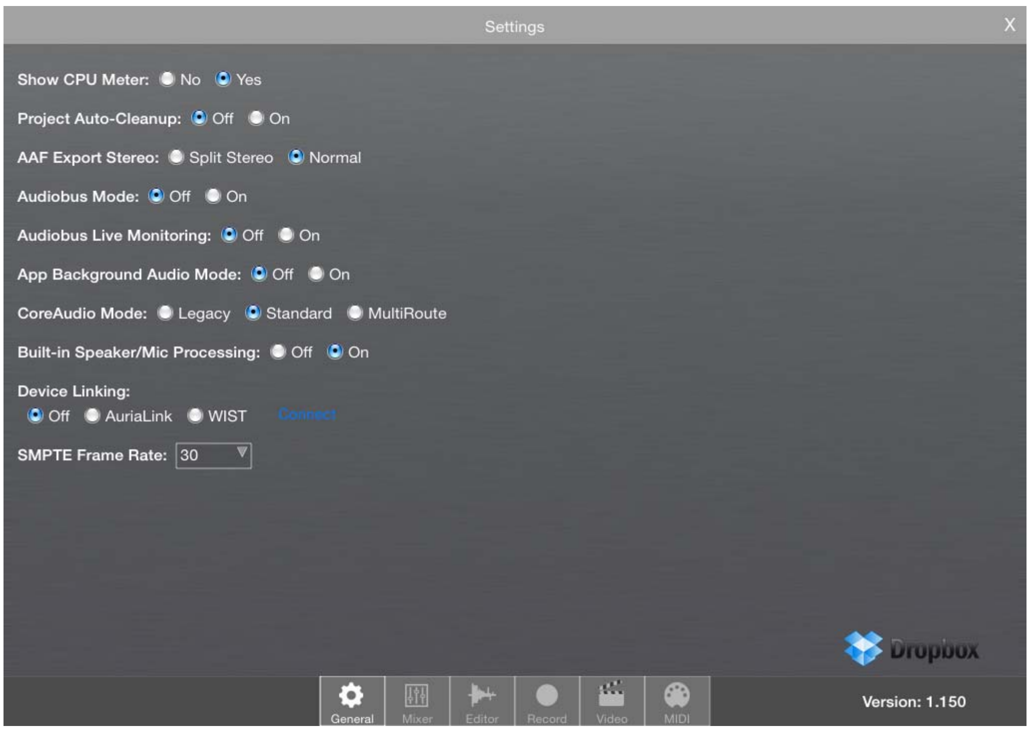
SETTINGS MENU



Auria's General Settings Menu

Auria's Settings Menu contains all of the app's "under the hood" options, and is grouped into multiple tabs; simply tap between the tabs to change which settings are currently viewed.

Note: When contacting WaveMachine Labs for help the Technical Support Department may ask for the currently installed version of Auria, this number will be displayed in the lower-right handle corner of the Settings Menu.

General Settings

Show CPU Meter - Displays the CPU/DISK meter in the top-right of the Mix window. Tap meter to cycle between current and max values, the BATT/SPACE meter, which shows remaining battery charge and storage space, and MEM, which displays the amount of free memory available.

Project Auto Cleanup – Turns on automatic Project Cleaning, where Auria scans current projects and deletes unused audio regions from disk. This is a destructive process so the default value is "Off"; before cleaning be sure to backup critical projects.

AAF Export Stereo – When exporting AAF projects determines if stereo tracks should be exported as Normal Stereo files (interleaved) or Split Stereo (two separate Left/Right mono files).

Audiobus Mode – Enables a compatibility mode for use with AudioBus, recommended for projects using Audiobus.

Audiobus Live Monitoring – Toggles monitoring of Audiobus's direct output, i.e. before it reaches the Auria mixer. This is similar to Hardware Monitoring, but substituting Audiobus for an external audio interface. Note: If enabled be sure to mute any Audiobus tracks set to Record Enable, otherwise a second, slightly delayed audio stream from Audiobus will be heard.

App Background Audio Mode – When enabled, Auria will continue to stream audio when the app is closed.

CoreAudio Mode – Switches between 3 different iOS audio systems:

- MultiRoute – The latest system in iOS, supports sending different audio mixes to a USB device and headphone jack simultaneously. Can be incompatible with other background audio apps and interfere with certain USB devices.
- Standard – The default setting and the most compatible, this mode lacks the ability to route different audio streams to headphones and USB devices.
- Legacy - Deprecated system from iOS 5. Reliable, but may be eliminated in a future iOS version. This is the only mode available for iPad's running iOS 5.

After switching CoreAudio mode both Auria and any USB interface must be rebooted; for USB devices this means disconnecting their power source for 30 seconds and reconnecting.

Note: This setting may take some trial and error to find the best mode for a particular setup, as different audio apps and interfaces can each behave differently in regards to CoreAudio. Standard mode should work well for the majority of users.

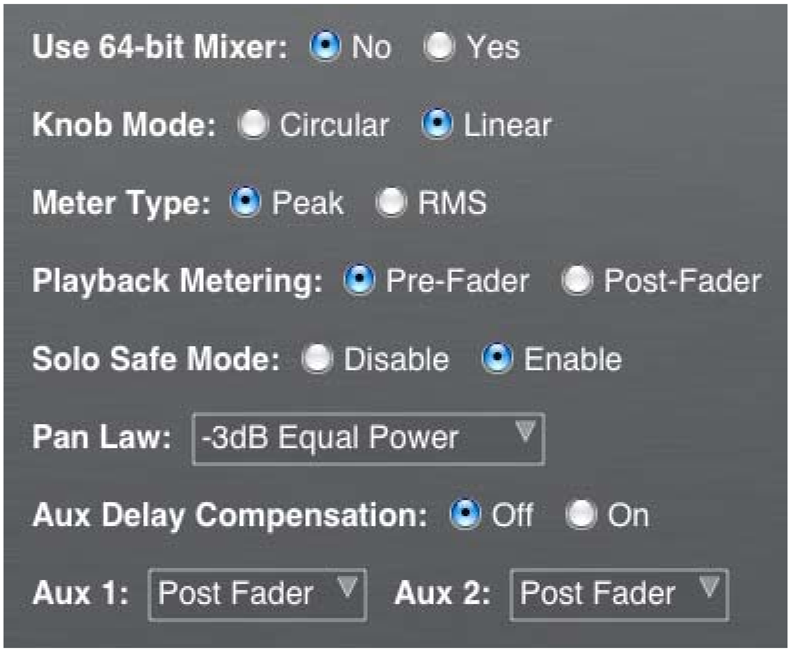
Built-in Speaker/Mic Processing – Toggles on and off the iPad's built-in audio processing for its on-board microphone and speaker, which adds some EQ and dynamic processing. Before iOS 7.1 this was always on, it is now optional. This has no affect on external audio interfaces.

Device Linking - Enables linking with additional iPads via either WIST or AuriaLink. Tap Connect to begin setup. For more information see the AuriaLink and WIST chapter.

SMPTE Frame Rate – Sets the project's frame rate, influences the Frames time display in the Counter, Timeline, and Snapping features. Also used when sending MIDI Time Code.

DropBox – Click to link a DropBox account. Used with importing/exporting items via DropBox.

Mixer Settings



Mixer Settings

Use 64-bit Mixer: Enables Auria's 64-bit mix engine. When disabled, Auria uses a more CPU friendly 32-bit mix engine at the expense of potentially more quantization error (but still quite low).

Knob Mode – Toggles between Linear and Circular modes for all knobs.

Meter Type – Toggles between Peak and RMS modes for all meters.

Playback Metering (Pre-Fader/Post-Fader) – Switches meters between Pre or Post Fader.

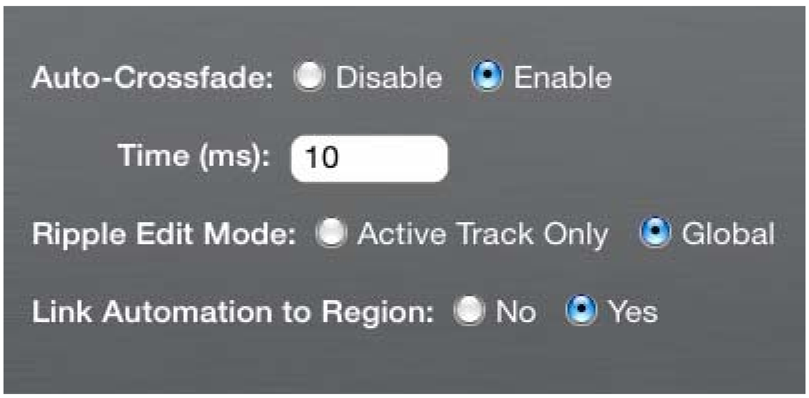
Solo Safe Mode – When enabled, soloing a track will also solo the AUX returns.

Pan Law – Switch between various equal power (-3, -4.5, and -6 dB) or linear modes.

Aux Delay Compensation – When enabled, any latency induced by plug-in effects on the AUX Sends will be automatically compensated for (but uses more CPU).

Aux 1 & 2 – Assigns either pre or post fader operation to aux sends. Default is post-fader.

Editor Settings



Editor Settings

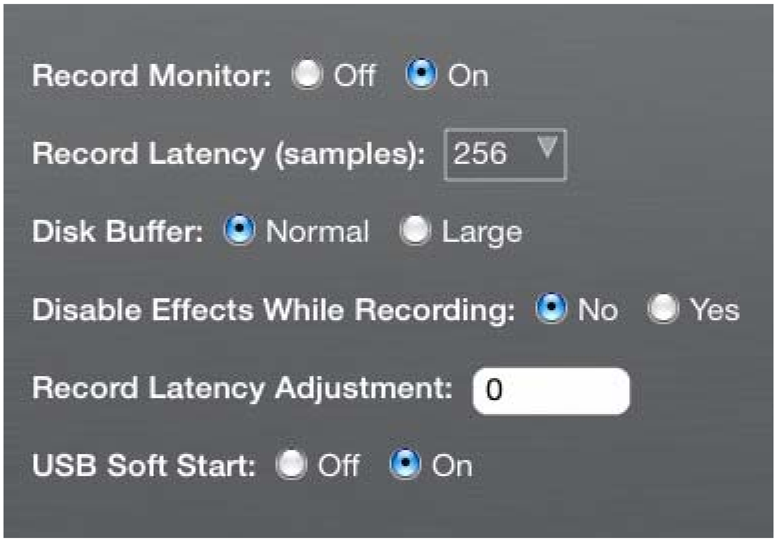
Auto-Crossfade - When enabled Auria will automatically fade regions in and out, unless two regions are snapped back-to-back. Use the Time (ms) box to change the default length of the Auto-Crossfade.

Ripple Edit Mode – When Ripple Mode Editing is enabled, this setting determines which tracks will be rippled:

- Active Track Only – Only the specific track being edited will ripple
- Global – All project tracks will be affected by the edit

Link Automation to Region – When enabled automation is linked to the region it corresponds to, so that moving that region (by using Cut/Copy/Paste, Delete, and Duplicate) will move the automation as well.

Record Settings



Record Settings

Record Monitor - Enables software monitoring when recording via USB. Useful when hardware monitoring is not available on an interface.

Record Latency (128-4096 samples) – Determines the recording latency of Auria. Higher buffer sizes will increase recording stability, but also increase latency when employing software monitoring.

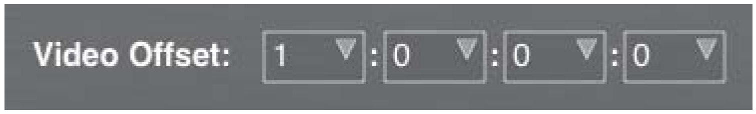
Disk Buffer – Sets the size of the I/O disk buffer. Default is "Normal", but try "Large" mode if encountering disk overload messages when using high channel-count USB interfaces.

Disable Effects During Recording – When "Yes", all effects will be disabled during recording.

Record Latency Adjustment – For audio devices which don't correctly report their recording latency, manually enter latency (in samples). To determine actual latency try running a loopback "ping" test.

USB Soft Start –Initializes USB audio interfaces more slowly before recording to prevent possible noise. This mode is on by default.

Video Settings

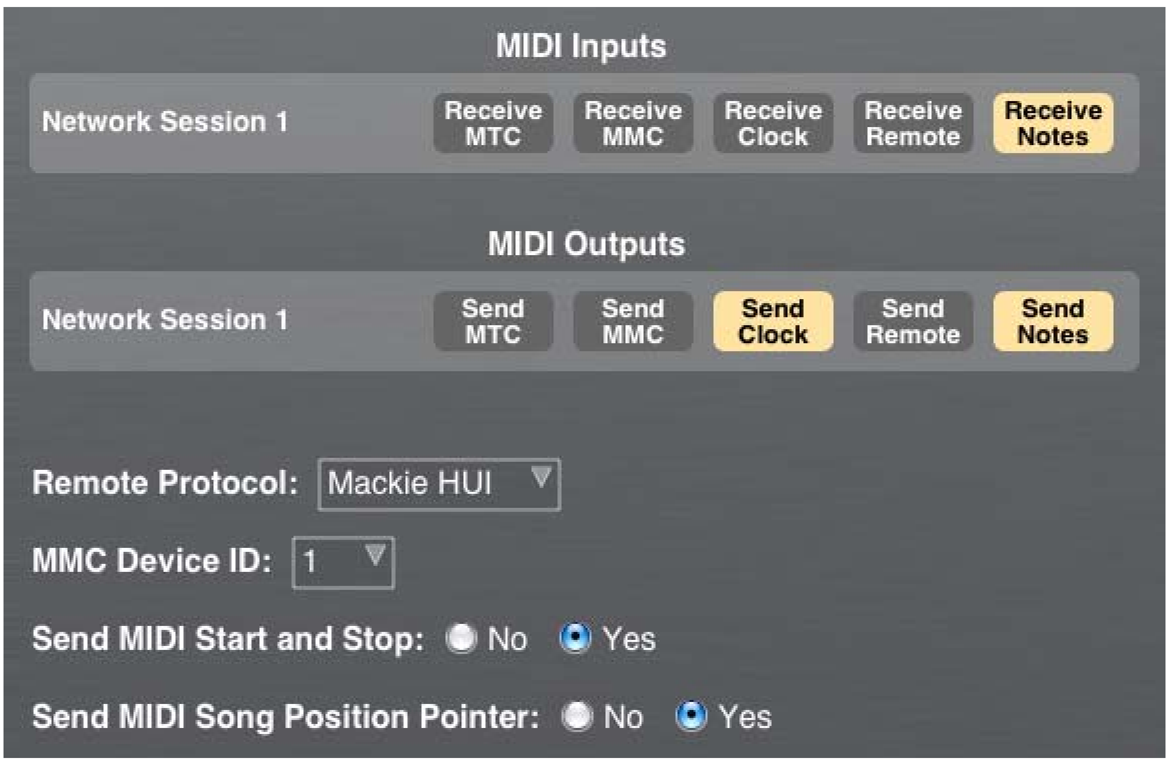


Video Settings

Video Offset – In Hours:Minutes:Seconds:Frames. Used with optional Video Preview feature. Determines what point in the video's timeline should coincide with 0:00 in Auria.

This tab will only be present when the optional Video Import add-on has been purchased.

MIDI Settings



MIDI Settings

For more information regarding the use of MIDI please refer to the MIDI Chapter.

MIDI Inputs – Selects which types of the MIDI information to receive from a particular connection. Any detected MIDI connection will be listed here, including Virtual MIDI (labeled as Network Session), other running apps, and connected MIDI interfaces.

MIDI Outputs – Selects which types of MIDI information to transmit out a particular connection. Any detected MIDI connection will be listed here, including Virtual MIDI (labeled as Network Session), other running apps, and connected MIDI interfaces.

Remote Protocol – Selects the MIDI remote control protocol Auria should emulate, which enables controlling Auria via external control surface. Both Mackie's HUI and MCU protocols are supported.

MMC Device ID – Assigns a unique ID MIDI Machine Control device ID number to Auria. Every MMC device in a chain requires a unique ID assignment.

Send MIDI Start and Stop – Enables transmitting MIDI Start and Stop messages when tapping Play and Stop on the transport. Used to remotely start and stop external MIDI devices, and is usually required when transmitting MIDI Clock.

Send MIDI Song Position Pointer – Transmits the current project's song position in MIDI.

16

PSP CHANNELSTRIP

PSP ChannelStrip

The PSP ChannelStrip combines the kind of channel processors you're likely to run across on a high-end mixing desk. It provides an expander/gate, equalizer, and compressor. All of these modules and their controls are optimized for mono and stereo channel processing.

Expander

Threshold

- Adjusts the threshold of the expansion or gating. If you need to set a very low threshold, press the -24dB button to lower the scale of the knob by 24dB.

Ratio

- Sets the expansion ratio. You can select between 1:1 and five expansion settings, as well as switch to a dedicated GATE mode.

Gain Reduction Meter

- This meter displays the immediate attenuation provided by the expander or gate.

ATK (attack)

- Sets the attack (gate open) time. The attack phase in FAST mode is covered by an internal pre-delay, which makes this setting click free. This is the best setting for transient content like drums and percussion. Use medium and SLOW settings whenever you need a smoother fade-in on open.

RELease

- Sets the expansion or gating release (gate close) time.

Low Pass Filter

- Sets the cut off frequency of the low pass filter for the side chain (control path). Use the IN button to engage the filter.

High Pass Filter

- Sets the cut off frequency of the low pass filter for the side chain (control path). Use the IN button to engage this filter.

MONitor

- This button allows you to listen to the side chain signal (control signal) including any activated filters.

RANGE

- Sets the maximum attenuation of the expander or gate.

EXP button

- Engages the expander module when lit.

EQualizer

High Pass Filter

- Sets the cut off frequency of the high pass filter.
- Use the IN button to engage this filter.

Low Pass Filter

- Sets the cut off frequency of the low pass filter.
- Use the IN button to engage this filter.

Low Middle Filter

- Use the frequency knob to adjust the middle frequency of the bell type low mid filter. Use the gain knob to set the amount of gain for this filter. The gain knob is not scaled in precise dB as the actual gain value varies depending on the Q setting. Use the switch to control the Q factor of the filter. Use the IN button to engage this filter.

High Middle Filter

- Use the frequency knob to adjust the middle frequency of the bell type high mid filter. Use the gain knob to set the amount of gain for this filter. The gain knob is not scaled in precise dB as the actual gain value varies depending on the Q setting. Use the switch to control the Q factor of the filter. Use the IN button to engage this filter.

Low Shelf Filter

- Use the knob to set the corner or middle frequency of the low filter. Use the gain knob to set the amount of gain for this filter. The gain knob is not scaled in precise dB as the actual gain value varies depending on the steepness/type setting. Use the switch to control the steepness of the shelf filter or to switch to a bell mode. Use the IN button to engage this filter.

High Shelf Filter

- Use the knob to set the corner or middle frequency of the high filter. Use the gain knob to set the amount of gain for this filter. The gain knob is not scaled in precise dB as the actual gain value varies depending on the steepness/type setting. Use the switch to control the steepness of the shelf filter or to switch to a bell mode. Use the IN button to engage this filter.

EQ->CMP switch

- This switch determines which module will be first in the signal chain, the equalizer or compressor module.

EQ

- This button toggles the EQ module in or out of the ChannelStrip.

Output

- Adjusts the output gain of the EQ module.

Compressor

THRESHold

- Sets the threshold of the compressor.

SOFT button

- Use this button to engage the soft knee mode.

MkUp (makeup)

- Use this button to engage the automatic make-up.

Gain Reduction Meter

- This five LED meter shows the average compression level.

RATIO

- Sets the compressor ratio. LIM puts the compressor in limiter mode.

High Pass Side Chain Filter

- Sets the cut off frequency of the side chain (control path) filter. Use the IN button to engage this filter.

ATTACK

- Sets the attack time of the compressor.

RELEASE

- Sets the release time of the compressor

RMS button

- Engages the RMS processing mode, which offers a slightly more “polite” compressor response.

CMP

- When lit, this button engages the compressor module.

OUTPUT

- Sets the output gain of the compressor module.

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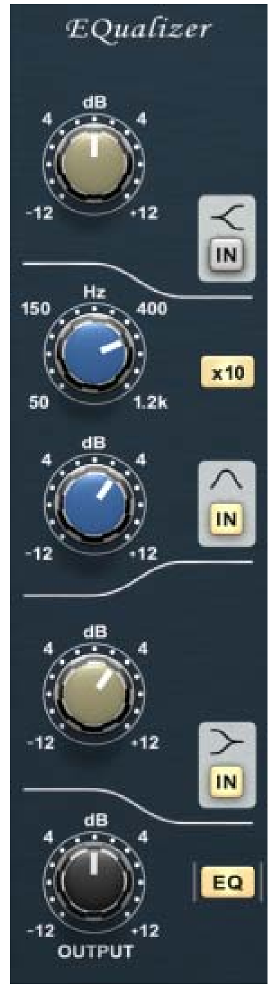
PSP MASTERSTRIP

PSP MasterStrip combines group and master channel strip processors into the same channel strip style processor. It offers an equalizer, bus compressor, and limiter. All of these modules and their controls are optimized for stereo group and master channel processing. The BussPressor has the same algorithm as the Mac/PC plug-in version.

PSP MasterStrip is available on subgroups and the master channel

EQualizer



High Shelf Filter

- The IN button toggles the high shelf filter on or off. The gain knob adjusts the gain of the filter.

Middle Filter

- The IN button toggles the middle filter on or off. The frequency knob adjusts the EQ frequency between 500Hz and 12kHz. The gain knob adjusts the gain of the filter.

Low Shelf Filter

- The IN button toggles the low shelf filter on or off. The gain knob adjusts the gain of the filter.

EQ

- This button toggles the EQ module in or out of the MasterStrip.

Output

- Sets the output gain of the equalizer module.

BussPressor

Gain Reduction Meter

- This meter displays the gain reduction level of the compressor.

THRESHold

- Sets the threshold of the compressor.

RATIO

- Sets the compressor ratio. In general, a ratio of 10:1 is considered a limiter.

MAKE-UP

- Sets the amount of manual make-up gain.

ATTACK

- Sets the attack time of the compressor.

RELEASE

- Sets the release time of the compressor.

AUTO

- Engages auto-release mode. In this mode the RELEASE knob sets the basic release time while the auto algorithm calculates a multi-stage release stage based upon a set release time value. The default setting of the RELEASE knob for the AUTO mode is in its center position.

High Pass Side Chain Filter

- Sets the cutoff frequency of the side chain (control) filter.

MIX

- Adjusts the ratio of dry and processed signal. This is useful for “parallel” or “New York” compression, which blends both the compressed and uncompressed signals together.

EQ->CMP switch

- Use this switch to place the EQ module before the compressor (up) or the compressor before the EQ module (down).

CMP

- Engages the compressor module when lit.

OUTPUT

- Sets the output gain of the compressor module.

Limiter

High Pass Side Chain Filter

- Sets the cutoff frequency of the side chain (control) filter. Press the IN button to engage or disengage this filter.

Gain Reduction LEDs

- These five LEDs serve as a rough meter of the depth of gain reduction of the limiter module.

INPUT

- Sets the input gain of the limiter.

SoFT button

- Engages soft knee limiting mode, which is a wider-range and faster limiting mode.

CEILING

- Sets the maximum level of the limiting curve. This works similar to threshold level controls in compressors, except compressor thresholds are at the beginning of the compression process.

ATTACK

- Sets the attack time of the limiter.

RELEASE

- Sets the release time of the limiter.

OPTO

- Engages the opto mode for the limiter. In this mode the release characteristics of the limiter are similar to analog limiters with an opto cell.

LIM

- Engages the limiter module when lit.

OUTPUT

- Sets the output gain of the limiter.

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INSERT EFFECTS

Auria comes with several different effect modules built-in (separate from the Channel and Master strips).

PSP StereoChorus

PSP StereoChorus is a high quality stereo chorusing processor. PSP StereoChorus offers you full control of its processing, allowing for a wide variety of modulation effects from a subtle thickening effect to wild stereo flanging. PSP StereoChorus also comes with a handful of extremely useful factory presets that cover the variety of modulation effects you can get from this very powerful processor.

- **FILTERS (high pass)**: Use this to set the high pass filter for the processed signal. The range is 20Hz to 2kHz. A setting of 20Hz bypasses the filter.
- **FILTERS (low pass)**: Use this to set the low pass filter for the processed signal. The range is 200Hz to 20kHz. A Setting of 20kHz bypasses the filter.

- **Spread**: Values to the right of M(iddle) are normal stereophony. Values to the left of M reverse the stereophony, thereby reversing the channels of the chorus effect. The value of M provides a monophonic wet signal.
- **TimeVar**: This varies the length of the chorus line in each channel. Values to the left shorten that channel's delay compared to the right channel, and vice versa.
- **PhaseVar**: This parameter sets the phase variance for the modulation. Higher values widen the stereo image.
- **Freq**: Use this knob to set the LFO frequency of the modulation between .1Hz and 10Hz. Use higher values for more extreme modulation.
- **Depth**: Controls the depth of the modulation effect. Higher values result in a more pronounced effect.
- **FeedBack**: Determines how much of the modulated signal is fed back into the processor. Higher values are useful for deep flanging effects (especially with short delay time).
- **Dry**: Sets the level of the unprocessed signal. Use this in tandem with the Wet knob to dial in the right amount of effect.
- **Wet**: Sets the level of the processed signal. Use this in tandem with the Dry knob to dial the right amount of effect.
- **Mode**: This button lets you chose either a sine wave or a triangle wave for the modulation.
- **Time**: Determines the delay time in milliseconds. Drag your finger to increase or decrease the delay time. The more delay, the more "out of time" the delay effect.
- **In**: Tap this button to enable or disable the processor.

PSP StereoDelay

PSP StereoDelay is a high quality stereo delay and echo processor. PSP StereoDelay can be used for a wide variety of delay effects from a simple slap back and sustain through ping-pong delays and unusual spacious echoes. PSP StereoDelay also comes with a handful of extremely useful factory presets that cover a wide range of this plug-in's settings.

- **FILTERS (high pass)**: Use this to set the high pass filter for the processed signal. The range is 20Hz to 2kHz. A setting of 20Hz bypasses the filter.
- **FILTERS (low pass)**: Use this to set the low pass filter for the processed signal. The range is 200Hz to 20kHz. A Setting of 20kHz bypasses the filter.
- **Ping-Pong**: Sets the amount of the ping-pong effect. There is no ping-pong delay present in the C(enter) position. Moving the control to the left from C sets the plug-in's left delay shorter then the right one. Moving the control to the right from C sets the right delay shorter then the left one. For a standard, balanced ping-pong effect set this control to 3R or 3L.
- **Saturate**: Sets the amount of tape-like saturation on the delayed signal. Experiment with various drive and filter settings to mimic analog tape echo effects.
- **Spread**: Controls the stereo spread of a fed back signal. Values to the left of M(iddle) reverse the stereophony, settings close to M narrow the delay with every echo. Setting this knob to S+ provides normal stereo repeats of the echo with constant panning.
- **FeedBack**: Determines how much of the delayed signal is fed back into the processor. The higher the value the longer the echo will be. Settings over 8 provide a feedback greater then 0dB that will result in the repeated echoes increasing in volume. Please be conscious of this volume effect and keep it under control!
- **Dry Pan**: Sets the panning or balance of the dry signal.
- **Dry Level**: Sets the level of the unprocessed signal. Use this in tandem with the Wet knob to dial in the right amount of effect.
- **Wet Pan**: Sets the panning or balance of the wet signal.
- **Wet Level**: Sets the level of the processed (delayed) signal. Use this in tandem with the Dry knob to dial the right amount of effect.
- **Tap**: Tap this button to set the delay time.

- **Time**: Determines the delay time in milliseconds. Drag your finger to increase or decrease the delay time.
- **In**: Tap this button to enable or disable the processor.

ClassicVerb

A simple, CPU-friendly reverb.

- **Time** – Sets total reverb length, in seconds
- **Filter** – Low-pass filter on reverb output
- **Mix** – Sets amount of wet (reverb) to dry (original signal). For Aux sends typically set to 100% (wet)
- **Output** – Sets output gain of reverb

Convolution Reverb

A reverb designed to recreate the acoustics of specific places/devices. Works by first loading particular Impulse Response, taken from the specific place/device being emulated.

- **Impulse Response Selection** – Loads an individual response (IR) file

- **Size** – Adjusts the total length of the reverb. When 100% the entire IR will be used
- **Offset** – Adjusts the start point of the IR
- **Delay** – Sets the amount of pre-delay
- **Low CPU Mode** – When lit (On), runs in a more CPU efficient mode, though it will lack reverb detail
- **Mix** – Sets amount of wet (reverb) to dry (original signal). For Aux sends typically set to 100% (wet)
- **Output** – Sets output gain of reverb

Side-Chain

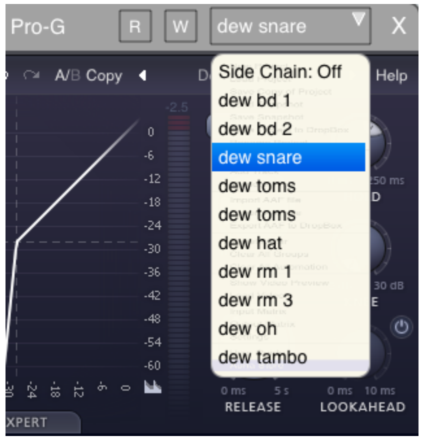
Certain plug-in effects available for Auria support Side-Chaining, sometimes also referred to as using a key input. This routes the audio from a different, separate track into the specific plug-in for use as an outside trigger element. So track 1 can be routed through a side-chain into a plug-in running on track 2.

One common example would be when working with kick drum and bass guitar the two parts often compete for the same low-frequency range and don't "sit" together easily. So, a compressor is inserted on the bass guitar track, and the kick drum track's signal is then fed into that compressor via a side-chain input. Then, whenever the kick drum is hit, the compressor is triggered on the bass guitar, automatically ducking (lowering the volume of) the bass and allowing the kick to come through more easily.

FabFilter Pro-G Gate, a side-chain compatible plug-in

To use Side-Chain on a particular plug-in:

1. First, the plug-in must support side-chaining. If it does then a special drop-down menu will appear in the top-right corner of the plug-in window.
2. Tap the Side-Chain Source drop-down menu to see a list of every track in the project.
3. Tap the appropriate track name to route that track into the plug-in's side-chain input.



Selecting the Side-Chain Source

Note: Routing a track to a side-chain does not affect the source track at all, it simply duplicates the signal and splits it off to the plug-in.

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OPTIONAL PLUG-INS

This chapter details the optional effect plug-ins available for purchase through the in-app Auria Store.

WaveMachine Labs Drumagog 5

Drumagog 5 Real-time Drum Replacer

What is Drumagog?

Drumagog is a software plug-in which replaces acoustic drum tracks with your choice of other samples. But Drumagog also offers the "secret weapons" top mix engineers use to give hit records polish and power. It is this exclusive combination of replacement and enhancement tools that has made Drumagog the industry standard for more than a decade.

File Browser

Use the File Browser to select a particular GOG file used for replacement. Tapping the speaker icon will play a preview of the selected GOG.

Samples Window

The Samples Window displays all the individual samples that make up the currently loaded GOG. Individual samples can be Soloed, Muted, or Played (auditioned) by first selecting a particular sample and then tapping the corresponding control.

Options Panel

- Dynamic Multisamples – Toggles dynamic sample playback. When Off Drumagog will ignore all dynamic groups and simply play a single sample from the pool for every hit.
- Random Multisamples – Toggles random sample playback. When On Drumagog will randomly select a sample from the corresponding dynamic group, never playing the same sample twice in a row. When turned off Drumagog will only play the first sample in each dynamic group.
- Articulations – Turns On/Off articulation samples, like rim shots, sidesticks, etc.
- Use Left/Right Hand – Toggles the use of Left vs. Right hand samples. When enabled Drumagog will automatically alternate between samples designated Left and Right hand when detecting quick fills or rolls.
- Dynamic Tracking – Toggles Drumagog's dynamics detection. When enabled Drumagog will automatically match the detected amplitude of each drum hit and scales the playback volume of the new sample to match. When disabled Drumagog will not adjust the amplitude of triggered samples.
- Stealth Mode – Toggles stealth mode. In stealth mode Drumagog will let audio below the sensitivity threshold pass through, even with the blend control set to 100% wet. Useful with "loose"-style micing, when a particular drum track has "bleed" that is wanted. One example would be when using a single mic for both snare and hi-hat.
- Auto Align 2.0 - Designed in collaboration with Fraunhofer, this powerful algorithm not only detects the initial transient of a drum hit, it analyzes the full waveform (i.e. the entire drum hit) and finds the best alignment which results in the most in-phase output throughout the entire sound. Auto-Align will even automatically invert the polarity of the replacement sample when needed! When switched Off Drumagog uses a simpler transient-based alignment algorithm,

- Live Mode – When enabled Drumagog runs in an ultra-low latency mode (3-4 ms total) suitable for live use; the trade-off is that in live mode Drumagog is much less accurate, especially to tracking dynamics. When mixing this mode should almost always be disabled for more accurate replacement.

Visual Triggering

Drumagog uses a visual triggering interface that streamlines the adjustment of the related controls. A scrolling, real-time waveform display is shown with the controls superimposed on top. This provides a visual indication of exactly how the various controls interact with the incoming audio. For example, the sensitivity control is represented as a horizontal line that can be moved up or down, and as the incoming audio scrolls past it's easy to see which audio impulses will trigger Drumagog and which will not. All the audio above the line will cause a trigger, and the audio below it will be ignored. The incoming audio that scored a "hit" is displayed as a white dot, making it easy to see a history of Drumagog's triggered hits.

- Sensitivity - Adjusting Drumagog using the Visual Triggering window is quite intuitive. As seen above, the "Sensitivity" control is represented by a horizontal line; any peak above this line will trigger Drumagog. If there is background noise or bleed from other instruments present in the audio track just raise the horizontal line above the noise floor. Simply adjust the line above the noise floor and below the softest drum hits.
- Resolution - The "Resolution" control is represented by a vertical line; moving this line right or left adjusts how long Drumagog must wait before triggering again. If your track doesn't contain any hits closer than 1 second apart, just set the resolution to 1000ms. This way Drumagog will ignore all audio for 1000ms after each hit, even if it's above the "Sensitivity" threshold. The default setting is "Auto", the quickest response with no additional wait time.
- Transient Detail - This slider changes the amount of detail Drumagog detects when triggering. If the slider is set to the right, Drumagog will pick up all the tiny nuances of a drum hit, including "ghost notes", etc. However, if the slider is set too far to the right it might pick up too much detail and falsely trigger. Moving the slider to the left reduces the amount of detail, useful for noisy tracks. The default setting in the middle is good for most tracks, but some adjustment may be necessary.
- Triggering Circles - Drumagog displays a white circle above each peak that triggered. This is useful in determining if Drumagog is triggering on the desired hits, or if miss-hits are occurring.
- Input - The input control adjusts the level of audio entering Drumagog's triggering engine. The input control can also be used to force Drumagog to choose different multi-samples, because Drumagog uses the volume of the incoming audio to determine which sample to use. Input is adjustable from -96.3 to +4dB.
- Output - This controls Drumagog's audio output level. Output is adjustable from -96.3 to +4dB.

NOTE: The height of the drum circle represents Drumagog's actual output volume level. If dynamic tracking is turned off, the circles will always be drawn at the same height. If the calculated output level is higher than 0dB the circles will be drawn above the waveform. This provides an accurate picture of Drumagog's output volume in relation to the original track.

Mixer Panel

- Blend - Drumagog's "Blend" control provides control of the mix between the original audio and the replaced samples. By using this feature, it's easy to find the "sweet spot" by combining the sound of your original drum with the new replaced sound from Drumagog. A setting of 100% (full-right) outputs only the replaced audio, while a setting of 0% (full-left) outputs the original, unchanged audio only.
- Dynamic Tracking – This slider adjusts the amount of dynamic tracking Drumagog uses. At 100% (the default) Drumagog will recreate exactly the same amplitude for the replaced drum sound as is found in the original. At 0% the replaced sound is played at a constant volume regardless of the original track's dynamics. Normally a value of 100% is used. Lowering "Dynamic Tracking" to a value less than 100% is an effective means to control overly dynamic tracks without the artifacts imparted by traditional compression. It also works well "upstream" from a compressor, when the compressed effect is desired, but not at the level required to achieve the desired dynamic control.
- Articulation Slider - This control is used in conjunction with a sample set that contains articulation multi-samples. Articulation multi-samples are groups of samples used to represent different stick striking positions or hi-hat pedal positions; the Articulation control selects which group should be played from that layer. For example, if a snare drum sample contains Articulation multi-samples, this controls where the drummer plays the drum. Moving the control to "Rimshot" selects a rimshot hit, while "Side stick" selects the cross-stick hit on the rim. Like all other controls, this slider can be automated, making it possible to switch between different articulations during a single performance.
- Room Sound Mixer – These three faders adjust the amount of "room" mics to mix in with each GOG file. Drumagog 5 comes with GOG files that contain both an "O/H" slot (stereo Over-Heads) and a "St Room" slot (Stereo Room). Simply slide the fader higher on the mixer to add these new ambient mics to Drumagog's playback. Some GOG files can also have a "Direct" fader; this controls the amount of direct or "close-mic" in a GOG file. Other GOG files may have custom specialty faders to mix in elements such as distortion, compression, snare bottom mics, etc.
- Trigger Filter - Many drum tracks are easy to replace, but on occasion there may be excessive bleed from other instruments which cause problems when triggering. Drumagog contains a Trigger Filter for use in these cases. This filter enables you to fine-tune the frequency range of interest, making the desired drum easier for Drumagog to replace. This technique greatly reduces any chances for mis-triggering when bleed is present on the source track. Drumagog's trigger filter has four modes: Hi-Pass, Low-Pass, Band-Pass and Notch. Using the filter it's even possible in some cases to trigger individual drums from a mixed stereo track.

NOTE: the filter doesn't actually change the audio of the samples being output, it strictly filters the incoming audio, making it easier to trigger.

PSP Echo

PSP Echo

PSP Echo is a high quality echo processor. PSP Echo's powerful and unusual features combined with its smooth operation makes it ideal for all kinds of creative uses from simple slap back and sustain effects through ping-pong delays and spacious echoes. Use the delay sliders to add special tape echo effects for even more unique effects. The tape wow control and built in ducker further extend the creative potential of PSP Echo. Internally, the Echo is like a combination of four mono tape delays—two for the initial ping-pong pre-delay and two for the main stereo echo. PSP Echo includes a set of extremely useful factory presets that cover a wide range of this plug-in's settings.

Top section controls

- Wow Freq: Sets the tape wow frequency.
- Wow Depth: Sets the depth of wow effect.
- Input: Sets the input level of entire effect.
- Tape Speed: Controls the speed of all built in tape delays. The reference speed is 15'.

- Ping-Pong: Sets the amount of the ping-pong effect. There is no ping-pong delay present in the C(enter) position. Moving the control to the left from C sets the plug-in's left delay shorter then the right one. Moving the control to the right from C sets the right delay shorter then the left one. For a standard, balanced ping-pong effect set this control to 3R or 3L.

Ducker

- Ducker button: Tap on the round Ducker button to open the Ducker view.
- Ducker LED: Indicates the state of the ducker. Green indicates the ducker is in the opened state or opening and red indicates the ducker is closed or closing. If the ducker is disengaged the LED will not illuminate.
- Ducker In/Out: Use this button to engage (In) or disengage the ducker.
- Ducker Threshold: This knob controls the threshold of the ducker.
- Ducker Range: Sets the amount of attenuation when the signal on the input is above a threshold.
- Ducker Open: Controls the opening time of the ducker when the input signal goes below threshold.
- Ducker Close: Controls the closing time of the ducker when the input signal goes above threshold.

Center Panel

- ms/bpm switch: Sets the display mode to milliseconds or beats per minute. In both settings the reference is 15' speed and a quarter note on sliders.
- Time/Tempo display: Sets the overall delay (echo) time or tempo referenced to 15' speed and a quarter note. Tap on a selected digit and move up or down to change values.
- Panic: Tap and hold for 0.5s on the time/tempo display to reset all delays' signals.
- Note Sliders: Use sliders to set a musical note values used for a delay. The reference note is a quarter note. You can glide to shorter or longer values on silence or on signal to get uncommon - tape like - effects.
- Echo LEDs: These LEDs blink green when a wet signal on the corresponding echo channel occurs. They will blink red when a tape is noticeably saturated.
- PSP Echo tap: Tap your finger anywhere in the white rectangle below the words “tap tempo” to tap out the tempo you wish for PSP Echo

Left and Right channel settings

- FB-Pan: Sets the feed back panorama for this channel’s echo effect. This allows for various cross-feedback and echo-narrowing effects.
- FeedBack: Controls the amount of feedback of this channel.
- FILTERS (high pass): Use this to set the high pass filter for the processed signal. The range is 20Hz to 2kHz. A setting of 20Hz bypasses the filter.
- FILTERS (low pass): Use this to set the low pass filter for the processed signal. The range is 200Hz to 20kHz. A Setting of 20kHz bypasses the filter.

- Drive: Sets the amount of tape-like saturation on the delayed signal. Experiment with various drive and filter settings to mimic analog tape echo effects.
- Level: Controls the gain of the echo effect's output.

Output section

- Link switch: Turns channel linking on or off. You can switch the link on and off during setting up PSP Echo without losing independent settings of channels however if you want to retain your independent settings please remember to turn it to the Off position before closing Auria. When a project is stored and PSP Echo is in Linked mode all independent settings of channels will be lost.
- Dry Spread: Controls the stereo spread of a dry signal. Values to the left of M(iddle) reverse the stereophony, settings close to M makes the signal narrow. Setting it to S+ provides a normal stereo dry signal on the output.
- Dry Balance: Sets the balance between dry left and right channels.
- Dry Level: Sets the dry signal gain to the output.
- Wet Level: Sets the wet (echo) signal gain to the output.
- Wet Balance: Sets the balance between wet left and right channels.
- Wet Spread: Controls the stereo spread of a wet signal. Values to the left of M(iddle) reverse the stereophony, settings close to M makes the signal narrow. Setting it to S+ provides a normal stereo wet signal on the output.
- In-Bypass switch: This switch engages or disengages entire Echo effect.

PSP MicroWarmer

PSP MicroWarmer

PSP MicroWarmer is a high-quality digital simulation of an analog-style single band limiter/tape emulator. This plug-in combines rich, warm analog-style processing with a straightforward user interface. We paid careful attention to PSP MicroWarmer's overload characteristics, so that this processor is fully capable of generating saturation effects typical of analog tape recorders. PSP MicroWarmer also incorporates VU metering together with accurate overload indicators, thereby assuring professional results.

Displays

- VU Meters: PSP MicroWarmer’s analog-style meters indicate VU levels.
- Pre/G.R./Post: The Pre/G.R./Post switch determines the point in the processing chain at which the meters measure the audio signal. “Pre” mode shows the signal level after equalization. “G.R.” (the default state) shows the signal gain reduction. “Post” mode shows the signal level after all processing and the Output knob.
- VU Ref.: This switch sets the VU reference level to -10dBFS or -12dBFS according to your needs.
- Parameter Display: The Parameter Display shows the value of the knob currently being operated.

Knobs

- Drive: The Drive knob sets the input level for the limiter. It can range from -24dB to +24dB. It is active when the red switch is in the 'On' position. The default value is 0dB.
- Speed: The Speed knob sets the compressor's combined attack and release times. The name refers to tape speed. A setting of 0 refers to a very slow tape speed resulting in very fast limiter/compress time setting, while a setting of 100 refers to the highest available tape speed which results in a smooth and very slow limiter/compressor timing. The default value is 50%.
- Release Multiplier: The Release knob is a multiplier control that sets the release time relative to the Speed setting. The default value is "x1".
- Low Adjust: The Low Adjust knob sets the low shelving pre-limiter gain. The default value is 0dB.
- High Adjust: The High Adjust knob sets the high shelving pre-limiter gain. The default value is 0dB.
- Knee: The Knee knob sets the knee range of the limiter. The 0% setting indicates that the knee is "bent" at 0dB ("hard knee"), which is suitable for limiting. Mid range settings can be used to create analog tape-style effects. The 100% setting provides a wide-range soft knee for deep and fast compression. The default value is 50%.
- Output: The Output knob sets the final output signal level. This is the last operation in the signal chain. The default value is 0dB.

Switches

- Link: Sets PSP MicroWarmer's channel linking mode. When set to On both channels work with the same amount of gain reduction. This results in a proper stereo field even with extreme processing but increases improper distortion effects when Speed is set to 50%. The Link Off mode provides more natural saturation behavior but doesn't preserve the proper stereo field on deep processing.
- On / Off: The On/Off switch turns the processor on and off. When the processor is off, all processing routines are bypassed except for the VU metering.

PSP oldTimer

PSP oldTimer is a vintage-style compressor designed for track and program compression and limiting. Our goal in developing this plug-in is to provide a simple compressor that offers an exceptionally musical sound while requiring a minimum of tweaking. This plug-in is not based on any specific hardware, rather it is inspired by vintage circuits and is designed to emulate some favorite characteristics of such compressors.

Controls

PSP oldTimer's controls are quite simple and intuitive. The plug-in gives you the ability to set integration Time, Compression depth and Output level. It also offers a Valve/Clear/Off switch to engage or disengage compression and Ratio rotary switch to change its compression curve.

- Valve/Clear/Off - Use this switch to engage (Valve) or disengage (Off) compression. If the switch is set to its middle position (Clear) the internal tube rounding is disengaged, resulting in more transparent processing.
- Valve level - Use this screw pot to set up a valve processing reference level. Choose between seven positions from "-" to "+".
- Ratio - PSP oldTimer features "over-easy" transition characteristics for 1.2:1 and 1.5:1 ratio, and old school peak-through-soft-knee characteristics for higher ratios. The specific amount of compression possible at the maximum compression depth depends on the ratio and ranges from about 8dB at a 1.2:1 ratio setting to about 30dB for a 10:1 ratio setting.
- Attack ratio - Attack ratio screw pot adjusts the attack time in reference to the overall Time knob. In other words it controls the ratio between the attack and release. A middle point setting and the default value for earlier versions of the PSP oldTimer refers to 1:10 attack to release time ratio.

- Time - Time sets up the combined attack and release time. Timing characteristics are tuned internally to insure a smooth and musical sound and the usual attack to release ratio is 1:10 whenever the Attack ratio screw pot is set to its default value. Since attack and release are heavily program dependent, you can manually adjust them to meet the specific program requirements using the Time knob. Time knob settings of 0-3 result in a fast attack/release, perfect for drums limiting. Time settings of 4-7 are good general values, offering times typical for opto or valve compressors. Time values around 8-10 set long values for leveling.
- Compression - Compression controls the amount of gain reduction by adjusting the threshold point. The greater the compression value the lower the threshold point, resulting in more compression. Even if the Compression is set to 0 the compressor may still influence the sound.
- Output - The Output knob sets the compression make-up gain. PSP oldTimer offers up to 30dB of gain in 0.5dB steps.
- Gain Reduction Meter - This meter gives you a readout of how much gain reduction is being performed by PSP oldTimer. In general, you'll want to keep the compression values shown on the meter between 4-8dB for the most transparent compression/limiting. When the Time knob is set between 8-10 a deeper compression of about 12-15 dB can be used for program leveling.

PSP PianoVerb 2

PSP PianoVerb2 is a creative resonant reverb plug-in. It creates its unique sound with twelve resonant filters that mimic the behavior of piano strings. The ability to transpose, tune and detune the set of strings allow you to set up the PSP PianoVerb2 to deliver a wide range of reverberations ranging from traditional wide spread reverb to more unusual resonances. With the addition of optional modulation, A-B settings of time and damping, and decay freezing this little plug-in delivers amazing options for creative use.

Whenever you want to add some natural resonance to a weak piano track, vitalize your leading guitar with a bit of sustain or simply add a touch of a nice reverb to selected track, the PSP PianoVerb2 can do the job.

Controls

- Time - Sets the reverberation decay time for settings A or B respectively. It ranges from 0 % for a very short reverb to 100 % for an almost frozen reverberation.
- Damping - Sets the damping factor for high frequencies. It ranges from 0 % for bright reverberation to 100 % for very dark reverberation.
- Transpose - Allows changes in transposition for a string system. It is possible to change transposition in the range from –24 to + 24 semitones. Transposing to a lower octave gives a less resonant and a more natural reverberation while transposing to a higher octave gives a highly resonant sound.
- Tune - The string system is tuned to A 440Hz by default, however, it is possible to manually tune to another reference frequency using this control. The string system can be tuned within a range of +/- 100 cents.

- Detune - Using this knob, the system can be detuned independently of the Tune control. Setting it to 0% results in tuned sound while turning it to 100% gives completely detuned results.
- Spread - sets the stereo width of the reverb.
- HPF - sets the low frequency cut off frequency for settings A or B.
- Select button - this button allows you to select Time and Damp settings A or B.
- Dry - controls the amount of a dry signal on the output. Set to 5 to achieve 0dB gain.
- Wet - sets the amount of a wet - reverb signal on the output. Set to 5 to achieve 0dB gain.

PSP SpringVerb 2

PSP SpringVerb2 is an emulation of a hardware spring reverberator. It recreates several sound features typical for a spring reverb like a convincing "boing" on transients or repeatable yet resonating character with a musical and adjustable presence. A selection of configurations from two to four springs total is provided and ability to set a stereo spread to suit various mix set ups - from a usual guitar reverb to an interesting option as a send reverb in the mix.

Two channel setup for A and B settings, together with a range of presets, allows a fast and easy operation. Whenever you want to add sustain to your guitar or add a sound of a mechanical reverberator to your mix the PSP SpringVerb2 can do the right job.

Controls

- Select button - this button allows you to select Time and Damp settings A or B.
- Type selector - sets up the spring configuration. 2 and 3 spring configurations and a dual 2 spring settings (one 2 spring tank for a left and one for a right channel) are available.
- Feed switch - when engaged an audio signal feeds the resonant structure of the SpringVerb2
- Time A/B - sets the length of the resonant reverberation for settings A or B respectively.
- HPF A/B - sets the low frequency cut off frequency for settings A or B.
- Presence A/B - sets the high mid frequency boost with a slight attenuation of low frequencies for settings A or B. It changes a tonal character of a reverb from dark to bright.
- Damp A/B - sets the high frequency damping for settings A or B.

- TrimB - sets the relative gain for a setting B.
- Diffusion - sets the amount of diffusion of the reverberation tail.
- Spread - sets the stereo width of the reverb.
- Dry - controls the amount of a dry signal on the output. Set to 5 to achieve 0dB gain.
- Wet - sets the amount of a wet - reverb signal on the output. Set to 5 to achieve 0dB gain.

FabFilter Micro

FabFilter Micro – a lightweight filter and envelope follower

FabFilter Micro is the ultimate lightweight filter plug-in, making the classic FabFilter sound affordable for everyone. With just one filter and an envelope follower to modulate its frequency, it can be used for simple filtering tasks, sound coloring and creative filtering effects.

The key feature of FabFilter Micro is its unique resonating, screaming and saturating filter that we first created for the FabFilter One synthesizer. It features both LP and HP filter shapes and an adjustable envelope follower to modulate the filter frequency according to the incoming audio signal. Furthermore, independent input and output gain controls enable you to saturate the filter more or less depending on your distortion needs!

Filter controls

- Frequency - The Frequency knob sets the cut-off frequency of the filter over the entire audio range.

- Peak - The Peak knob adjusts the resonance of the filter. A little resonance will cause the filter to create warmer and more characteristic tones. At maximum resonance, the filter will self-oscillate at the center frequency.
- Response - The Response buttons select between low-pass and high-pass filter shapes. FabFilter Micro's filter always has a 12 dB/octave steepness.
- EF Level - At the far left, the Level knob sets the amount of filter frequency modulation with the built-in envelope follower. You can set both negative and positive modulation. In the center position, no modulation will take place. The little reset button at the top of the level knob lets you easily turn off modulation altogether.
- EF Speed - The Speed knob adjusts how quickly the envelope follower reacts to changes in the input signal. When turned all the way to the left, the envelope follower reacts quickly and aggressively to changes. When turned all the way to the right, the response to changes is very slow and smooth. The default position in the center provides a good overall behavior.

Tip: the EF light above the envelope follower controls shows you how much filter frequency modulation is currently taking place.

Output options

At the right-hand side of the bottom bar in the interface, FabFilter Micro contains independent input and output gain parameters.

- The Input Gain parameter sets the gain that is applied before the signal enters the filter. You can use this to amplify the signal so it saturates the filter more or less, adjusting the amount of distortion.
- The Output Gain parameter sets the gain that is applied after the filter. If you have amplified the incoming signal with the Input Gain parameter, you can use the Output Gain to attenuate it again to obtain a reasonable output level.

MIDI Learn

Controlling FabFilter Micro's parameters directly with MIDI is very easy using the MIDI Learn feature. With MIDI Learn, you can associate any MIDI controller with any parameter.

Click the MIDI Learn button in the bottom bar to enter MIDI Learn mode. The interface dims and the parameters that can be controlled are highlighted. Each parameter has a small text balloon that displays the associated controller number. Now do the following to associate a controller number with a parameter:

1. Touch the control of the desired parameter in the interface that you wish to control. A red square will mark the chosen parameter.
2. Adjust the slider or knob on your MIDI keyboard or MIDI controller that you want to associate with that parameter.

That's it! The parameter will now be controlled with the MIDI controller. You can now go back to step 1 to associate a different parameter. Note that there is no warning when you associate a different knob with a controller number that is already used. It will just be replaced.

To exit MIDI Learn mode, click the MIDI Learn button again, or click Close at the top of the interface.

Click the small menu drop-down button next to the MIDI Learn button to access the MIDI Learn menu:

- Disable/Enable MIDI - This globally turns MIDI control of parameters on or off: useful in hosts that automatically send all MIDI events on a track to all effect plug-ins associated with that track as well.
- Clear - This submenu shows all parameter associations and lets you delete individual associations or clear all associations in one step.
- Revert - Reverts to the last saved MIDI mapping (or the state when the plug-in was started).
- Save - Saves the current MIDI mapping so Revert will go back to this state. The current mapping is automatically saved when closing the plug-in.

FabFilter Pro-C

There is one signal-processing tool that is almost impossible to do without in any form of audio recording or post-production: compression. Compression is available in a wide variety of different formats, flavors, designs and degrees of quirkiness.

Knee

The Knee switch chooses between a custom soft knee that is different for each compressor style, or a simple hard knee. You can view the resulting transfer function in the transfer function display.

Style

FabFilter Pro-C comes with 3 different styles of compression: Clean, an allround, low distortion, feedforward, program dependent, soft knee style; Classic: a vintage, feedback, very program dependent style; and Opto: a relatively slow, very soft knee, more linear opto style.

Input

The Input knob controls the amount of input gain that is applied before the input signal enters the compressor. If Expert mode is active, it is also possible to pan the input signal.

Threshold

The Threshold knob sets the threshold at which compression begins. Lower thresholds give heavier compression.

Ratio

The Ratio knob sets the amount of compression. At a ratio of 10:1, just one dB of output signal above the threshold remains for every 10 dB of input signal above the threshold. You can click on the small dots around the Ratio knob to jump to certain fixed ratio amounts. If you move the knob completely to the left (1:1), no compression will take place. If you move it completely to the right (infinity), everything above the threshold will be completely compressed away, making Pro-C act as a limiter.

Attack

The Attack knob sets the time after which gain reduction sets in. For transient-rich program material like drums, fast attack times are needed to minimize overshoot. For other program material, too short attack times may dull the sound a bit. FabFilter Pro-C is capable of very fast attack times and they are program dependent.

Release

The Release knob sets the time that the compressor takes to recover from gain reduction. The various compression characteristics of Pro-C use different release models, and in most cases, the release time is very program dependent.

Auto Release

The Auto Release button enables a smart auto release feature. When enabled, the compressor adjusts the release time depending on the current amount of gain reduction, so this actually introduces an additional form of program dependency. When Auto Release is used, the Release knob changes into the Auto Release Speed knob that adjusts the overall effect of the auto release feature on the release time.

Program dependency

The different compressor styles in FabFilter Pro-C all have their own kind of program dependency. This means that the compressor reacts differently to different kinds of input (program material). For example, Pro-C will recover very fast from transients (fast changes/peaks), but will react quite a bit slower after longer periods of gain reduction. Both the attack and release times are program dependent.

The Classic style is by far the most program dependent style. Even at the fastest release time setting, the actual release time can increase up to a few seconds! Also the Clean style uses a form of program dependency to sound smooth on various types of program material. The Opto style implements only little program dependency.

Output

When all audio is processed the way you want, it is sent to the output. The output section of the interface contains controls for adjusting main volume. The output of the compressor and the dry (unprocessed) signal have their own output knob (and panning rings when Expert mode is activated).

This allows for parallel compression. Parallel compression refers to mixing the dry signal with a compressed copy of itself. The dynamics in the dry signal are preserved while the compressed signal adds body and character to the overall sound. The advantage of this is that the sound is reinforced where it needs it, but without the risk of crushing any peak transients.

For example, you can smash the living crap out of the snare drum to get whatever effect you are looking for while at the same time just lightly blending that snare into the mix. A very powerful function so we suggest you experiment with this.

The Auto Gain knob will help you to restore volume after compression. This is also known as "make-up gain" because it compensates for the gain reduction introduced by the compressor.

Warning: Auto Gain is not a guaranteed way of getting the "right" result. Rather, it can be a useful tool while you are tweaking the threshold and ratio parameters. Especially when using Expert mode functions like filtering, panning and side chain levels, it is better to switch Auto Gain off and adjust the Output level manually.

Displays

FabFilter Pro-C comes with some very cool ways of looking at all the dynamic information.

On the left, you'll find the basic transfer function display which shows the input/output relationship. The horizontal axis corresponds to the input signal level, and the vertical axis is the output level (measured in decibels). The red area in the display shows the current signal level. Furthermore the threshold, ratio and knee are visible so you have a clear view on those compressor settings. See Compressor basics for more information.

In the center is a very useful other window: the animated level display showing the actual input, output and gain-reduction levels in a 2D animated display. Very useful if you want to know exactly what's happening. The input level is shown in grey, the gain reduction as a red line, and the output level is yellow. Use the three small colored knobs underneath the display to adjust the opacity of each of the three curves to your liking.

To the right, there are three very accurate peak level meters, which are always displaying the current input level, gain reduction level, and output level as well. In addition, the output meter will freeze the peak level if it has clipped (shown in red). Click on the red clipping indication to reset it.

Notes

- The animated level display will slide under the knee display when Expert mode is enabled.

- You can change the scale of the level display and the peak level meters with the Meter Scale drop-down button, ranging from 48 dB (the default) to 8 dB for very precise mastering purposes. The transfer function display always keeps the 48 dB range.
- With the small Display Enabled button, you can turn off all animated displays in case you find them distracting. The peak level meters will always be visible, though.
- To save all display settings so they will be re-used in future plug-in instances, choose Options > Save As Default from the plug-in's preset menu. The display settings will also be saved in songs.
- If the output meter indicates clipping, this does not imply distortion in Pro-C: it can handle levels above 0 dB easily. Rather, this indicates that the output signal might clip in another part of the audio chain, for example your sound card or host software.

Undo and Redo

The Undo and Redo buttons at the top of the plug-in interface enable you to easily undo changes you made to the plug-in.

- The Undo button at the left undoes the last change. Every change to the plug-in, such as dragging a knob, or selecting a new preset, creates a new state in the undo history. The Undo button steps back through the history to restore the previous states of the plug-in.
- The Redo button cancels the last Undo command. It steps forward through the history until you are back at the most recent plug-in state.

If the plug-in parameters are changed without using the plug-in interface, for example with MIDI or automation, no new undo states are recorded.

The Undo and Redo buttons will disable themselves if there is nothing to undo or redo.

A/B

With the A/B feature in FabFilter Pro-C, you can easily switch between two different states of the plug-in.

- The A/B button switches from A to B and back. Before switching, the current state of the plug-in is saved, so if you click this button twice, you are back at the first state. The button highlights the currently selected state (A or B).
- The Copy button copies the active state to the inactive state. This marks the current state of the plug-in and allows you to go back to it easily with the A/B button. After clicking Copy, the button disables itself to show that both states are equal, so there is nothing to copy anymore.

Presets

- To load a preset, click the preset button. The presets menu will appear with all available presets. Click a menu item to load that preset. The currently selected preset is highlighted with check marks.

- To explore the presets one by one, click on the little arrow buttons to the left and right of the main preset button. This will load the previous or next preset in the menu.

The preset button shows the name of the current preset. If you have changed the preset by adjusting one or more parameters, the name is dimmed to indicate that this is not the original preset anymore.

- To save the current setting as a preset, click the preset button, and then click Save As. A standard Save dialog will appear. Type a name for the new preset and click Save to finish.
- In the Save dialog, you can also rename and delete existing presets and create a new folder to store presets in. New folders will show up as new categories in the preset menu.

MIDI Learn

Controlling FabFilter Pro-C's parameters directly with MIDI is very easy using the MIDI Learn feature. With MIDI Learn, you can associate any MIDI controller with any parameter.

Click the MIDI Learn button in the bottom bar to enter MIDI Learn mode. The interface dims and the parameters that can be controlled are highlighted. Each parameter has a small text balloon that displays the associated controller number. Now do the following to associate a controller number with a parameter:

3. Touch the control of the desired parameter in the interface that you wish to control. A red square will mark the chosen parameter.
4. Adjust the slider or knob on your MIDI keyboard or MIDI controller that you want to associate with that parameter.

That's it! The parameter will now be controlled with the MIDI controller. You can now go back to step 1 to associate a different parameter. Note that there is no warning when you associate a different knob with a controller number that is already used. It will just be replaced.

To exit MIDI Learn mode, click the MIDI Learn button again, or click Close at the top of the interface.

Click the small menu drop-down button next to the MIDI Learn button to access the MIDI Learn menu:

- Disable/Enable MIDI - This globally turns MIDI control of parameters on or off: useful in hosts that automatically send all MIDI events on a track to all effect plug-ins associated with that track as well.
- Clear - This submenu shows all parameter associations and lets you delete individual associations or clear all associations in one step.

- Revert - Reverts to the last saved MIDI mapping (or the state when the plug-in was started).
- Save - Saves the current MIDI mapping so Revert will go back to this state. The current mapping is automatically saved when closing the plug-in.

FabFilter Pro-DS

Every mix or mastering engineer often has to deal with over-sibilant vocals. Even when using the best microphones, pre-amps and converters, 's' and 't' sounds can easily get over-accentuated during post-processing like compression, saturation or limiting. For all these cases, FabFilter Pro-DS comes to the rescue!

With its highly intelligent 'Single Vocal' detection algorithm, it accurately and transparently attenuates sibilance in vocal recordings. Plus when using Pro-DS in 'Allround' mode, it's a great tool for high-frequency limiting of any material, like drums or full mixes. Try it out yourself!

Overview

The interface of FabFilter Pro-DS is designed to be easy to use while providing all necessary information and controls. It consists of the following elements:

Real-time level display

In the top section of the plug-in, the real-time moving level display and the level meters show you at a glance what's happening to your audio. The unaffected output level is shown in transparent grey, while affected parts (where gain reduction is applied) are shown in light green. See Metering.

- Level metering

At the right of the interface, the output level meter, gain reduction meter and their read-outs provide an immediate overview of the current output and gain reduction levels. See Metering.

- Basic controls

Using the large yellow Threshold and Range knobs, and the high-pass and low-pass filter sliders, you can set up basic de-essing functionality. These controls determine the amount of gain reduction and the frequency range on which the de-esser will trigger. To the right of the large knobs, you can toggle between the highly intelligent Single Vocal and classic Allround modes. See Basic controls.

- Advanced controls

Centered in the interface, under the Mode buttons, you can toggle between Wide Band and Split Band processing, and between using the internal or external input for the side-chain. See Advanced controls.

- Stereo linking and lookahead

Next to the level meters, at the right of the interface, you can control the amount of stereo linking (with optional Mid-only or Side-only processing), and lookahead time. See Advanced controls.

- Oversampling

The Oversampling setting sets the amount of internal oversampling, which reduces possible aliasing for fast/aggressive deessing at the cost of additional CPU usage. See Oversampling.

- Input and output options

On the far right of the bottom bar, you can bypass the entire plug-in and adjust the initial input and final output levels. See Input and output options.

- Presets, undo, A/B, help

With the preset buttons, you can easily browse through the factory presets or save your own settings so you can re-use them in other songs. The Undo, Redo, A/B and Copy buttons at the top of the plug-in interface enable you to undo your changes and switch between different states of the plug-in. Finally, the Help menu provides access to help and version information.

Basic controls

The large Threshold and Range knobs, the trigger frequency sliders and the Mode button are the most important controls in Pro-DS. They greatly affect how Pro-DS reacts and sounds.

Threshold

The Threshold knob sets the threshold of the side-chain level above which the de-esser will trigger and apply gain reduction. The circular side-chain level meter around the Threshold knob shows the level of the filtered and possibly stereo-linked signal that is used for detection. This feedback makes it a lot easier to choose a proper Threshold setting.

Using the round Audition Triggering button, at the left top of the Threshold button, you can hear on which parts of the audio Pro-DS is triggering and how much de-essing is taking place. This helps you choose an appropriate Threshold level as well, making sure Pro-DS is catching all necessary peaks without triggering on anything else.

Range

The Range knob simply scales the detected gain reduction so that it stays within a desired range, enabling you to easily change the desired amount of de-essing.

Single Vocal vs. Allround mode

When you want to de-ess a single vocal track, it's best to set the mode to Single Vocal. This enables a highly intelligent detection algorithm, which splits sibilance from non-sibilance.
In Allround mode, triggering only depends on the frequency range, specified by the HP and LP filtering sliders, in combination with the Threshold setting of course. This is intended for processing entire mixes, for example.
Tips

- In Single Vocal mode, you can lower the Treshold all the way to -INF dB. This reduces the dynamic range of the gain reduction: all sibilance will then be reduced by (roughly) the same amount (specified by Range).
- To brighten up your sound during mastering, you can use Pro-DS as a high-frequency limiter. Choose the Allround mode and
- Split Band processing, and limit the transients of the high frequency range. Then, bring up that frequency range again using a high shelving EQ filter (from FabFilter Pro-Q for example). Try it yourself!

Processing

Using the processing buttons, you can choose between Full Band processing and Split Band processing. When Full Band processing is enabled, Pro-DS will lower the overall gain of the audio when sibilance is detected. When Split Band is chosen, only high frequencies will be attenuated. The split frequency is determined automatically according to the chosen high-pass sidechain filtering setting.

When working with single vocals, the Wide Band option often gives you great results already, but when de-essing full mixes, or more complex audio, it's often best to use the Split Band mode and leave the lower frequencies untouched. In some cases however, especially with single vocal material, Wide Band sounds more natural. Note also that split-band processing introduces some additional latency (see Lookahead below).

Split-band processing (in combination with the Allround mode) is also often used during mastering, where the de-esser functions as a high frequency limiter. The basic trick is to compress transients in the high frequencies of the music, and bring up that same frequency range again afterwards using a high shelving filter (using FabFilter Pro-Q for example). This can really brighten up the music, and glue the overall sound, giving it a nice edge!

Stereo linking and mid-only / side-only processing

Using the single Stereo Link knob and its Stereo Link Mode button (only available in the stereo version of the plug-in of course), you can both set a variable stereo linking for the trigger input signal, and choose between normal stereo or mid-only/side-only processing.

The first half of the knob's range sets stereo linking from 0% (fully unlinked, channels operate independently) up to 100% (fully linked, resulting in the same gain reduction for both channels).

When turning the knob even further, you will eventually process only the mid-signal (mono content of the processed audio), or only the side-signal (stereo content of the processed audio). Using the small Stereo Link Mode button at the right bottom of the Stereo Link knob, you can toggle between the two.

Mid- or side-only processing can be very useful, for example while mastering. A lead vocal is often placed in the center of the stereo image, so only de-essing the mid-signal will leave all stereo content untouched, ensuring the most transparent end result possible. Alternatively, when de-essing backing vocals in a stereo mix (often panned left or right), you could choose side-only deessing, ensuring that Pro-DS leaves all mono content untouced, like the lead vocal.

To better understand the working of these settings, enable the Audition Sidechain button right under the side-chain filtering sliders. You can now directly hear the effect of stereo linking and mid-only or side-only processing!

Lookahead

With the Lookahead knob, Pro-DS can be set to start de-essing up to 15 ms before the trigger audio level actually exceeds the threshold. This is an excellent way to catch transients and/or the start of sibilance.

Often, when de-essing vocals, a lookahead value of about 10 ms is optimal to ensure you're catching all of the sibilance. Leaving a bit of the initial transient untouched might help to preserve a 'natural' s-sound, so take your time to experiment and choose an appropriate value for the audio you're processing.

You can enable or disable look-ahead with the Lookahead Enabled button, just at the right-top of the Lookahead knob. When lookahead is disabled, processing is set to Wide Band and oversampling is off, Pro-DS works without any latency. When look-ahead is enabled, the latency will be 15 ms, plus a small additional latency if oversampling or split-band processing is used.

Tips

- When choosing a look-ahead value, it might be handy to enabled the Audition Triggering mode, using the round button at the left top of the Threshold knob. This way, you can hear right away whether you're catching all of the detected sibilance.

The real-time level display

The large real-time waveform level display at the top part of the interface shows you the incoming audio signal (after the input gain has been applied), while highlighting the parts that are actually affected by the de-esser. The unaffected parts are always dimmed and semi-transparent.

The highlighted parts show the reduced gain in light green, and the resulting gain in darker green, giving you an immediate insight in how much processing is going on.

The detection input meter

The circular side-chain level meter around the Threshold knob shows the level of the filtered and possibly stereo-linked signal that is used for detection, helping you to choose a proper Threshold setting.

The spectrum display

Built into the sidechain filtering section, a real-time spectrum analyzer will light up as soon as audio is running through the plug-in. It highlights the strong frequencies in the side-chain input signal, helping you to narrow the triggering frequency range using the high-pass and low-pass sliders. Note that the analyzer shows you the spectrum after it has been filtered as indicated by the sliders.

The level meters

At the right of the interface, the level meters show output and gain change. Above the meter is a dB read-out, showing the maximum level that has been detected. You can reset it by simply clicking on it.
When the output exceeds 0 dBFS, the meters will indicate clipping. To reset this, just click on the meter area or on the read-out. Note: the actual audio is not clipped within Pro-DS. The clipping indication is primarily useful for hosts that always keep all audio within the 0 dBFS range.

Oversampling

The de-essing algorithm often needs to make very quick changes to the audio when attenuating sibilance. These sudden changes can introduce a little aliasing, which causes distortion and generally reduces the quality of the audio signal. Oversampling is a way to reduce that aliasing by running the internal process at a sample rate that is two or four times higher than the host's sample rate. When should you use oversampling? You need it more when the de-esser is triggered often, and when using higher Range settings.

Of course, in return for a reduction of possible aliasing/distortion, the plug-in will use more CPU power when using oversampling. In addition, oversampling introduces a small latency, in addition to any split-band and look-ahead latency (see Advanced controls).

Input and output options

At the right-hand side of the bottom bar in the interface, FabFilter Pro-DS contains the bypass, input level and output level parameters.

Bypass

You can bypass the entire plug-in with the Global Bypass toggle button to the left of the input level button. While most hosts already provide the ability to bypass plug-ins, our internal global bypass feature is guaranteed to correctly compensate the latency of the plug-in and it also applies soft bypassing to avoid clicks. While the plug-in is bypassed, the display dims and a red light glows in the bypass button itself.

Input and output levels and panning

With the input and output level/pan knobs, you can fully adjust the stereo audio level both before and after of the de-essing process.

Undo and Redo

The Undo and Redo buttons at the top of the plug-in interface enable you to easily undo changes you made to the plug-in.

- The Undo button at the left undoes the last change. Every change to the plug-in, such as dragging a knob, or selecting a new preset, creates a new state in the undo history. The Undo button steps back through the history to restore the previous states of the plug-in.
- The Redo button cancels the last Undo command. It steps forward through the history until you are back at the most recent plug-in state.

If the plug-in parameters are changed without using the plug-in interface, for example with MIDI or automation, no new undo states are recorded.

The Undo and Redo buttons will disable themselves if there is nothing to undo or redo.

A/B

With the A/B feature in FabFilter Pro-DS, you can easily switch between two different states of the plug-in.

- The A/B button switches from A to B and back. Before switching, the current state of the plug-in is saved, so if you click this button twice, you are back at the first state. The button highlights the currently selected state (A or B).
- The Copy button copies the active state to the inactive state. This marks the current state of the plug-in and allows you to go back to it easily with the A/B button. After clicking Copy, the button disables itself to show that both states are equal, so there is nothing to copy anymore.

Presets

- To load a preset, click the preset button. The presets menu will appear with all available presets. Click a menu item to load that preset. The currently selected preset is highlighted with check marks.
- To explore the presets one by one, click on the little arrow buttons to the left and right of the main preset button. This will load the previous or next preset in the menu.

The preset button shows the name of the current preset. If you have changed the preset by adjusting one or more parameters, the name is dimmed to indicate that this is not the original preset anymore.

- To save the current setting as a preset, click the preset button, and then click Save As. A standard Save dialog will appear. Type a name for the new preset and click Save to finish.
- In the Save dialog, you can also rename and delete existing presets and create a new folder to store presets in. New folders will show up as new categories in the preset menu.

MIDI Learn

Controlling FabFilter Pro-DS's parameters directly with MIDI is very easy using the MIDI Learn feature. With MIDI Learn, you can associate any MIDI controller with any parameter.

Click the MIDI Learn button in the bottom bar to enter MIDI Learn mode. The interface dims and the parameters that can be controlled are highlighted. Each parameter has a small text balloon that displays the associated controller number. Now do the following to associate a controller number with a parameter:

5. Touch the control of the desired parameter in the interface that you wish to control. A red square will mark the chosen parameter.
6. Adjust the slider or knob on your MIDI keyboard or MIDI controller that you want to associate with that parameter.

That's it! The parameter will now be controlled with the MIDI controller. You can now go back to step 1 to associate a different parameter. Note that there is no warning when you associate a different knob with a controller number that is already used. It will just be replaced.

To exit MIDI Learn mode, click the MIDI Learn button again, or click Close at the top of the interface.

Click the small menu drop-down button next to the MIDI Learn button to access the MIDI Learn menu:

- Disable/Enable MIDI - This globally turns MIDI control of parameters on or off: useful in hosts that automatically send all MIDI events on a track to all effect plug-ins associated with that track as well.
- Clear - This submenu shows all parameter associations and lets you delete individual associations or clear all associations in one step.
- Revert - Reverts to the last saved MIDI mapping (or the state when the plug-in was started).
- Save - Saves the current MIDI mapping so Revert will go back to this state. The current mapping is automatically saved when closing the plug-in.

FabFilter Pro-G

A gate/expander is one of those workhorse studio tools that you probably use in every mix. Whether you need to suppress noise on your vocal tracks, reduce bleed on your drum recordings, gate a guitar before distortion or enhance the dynamics on your drum or master bus, FabFilter Pro-G will do the job in style!

Dynamic controls

The section in the top-left corner of the FabFilter Pro-G user interface is where the conventional gate/expansion settings can be found: threshold, ratio and range. These determine the dynamic behavior and the amount of expansion.

Threshold

The Threshold knob sets the threshold at which the gate/expander will open. With a lower threshold, the gate/expander will open earlier. Finding a good setting always depends on your audio, and the real-time display and metering will help you set the right level.

Ratio

The Ratio knob sets the amount of expansion when the signal level drops below the threshold. At a ratio of 4:1, every dB under the threshold results in an target reduction of -4 dB. If you set the knob to 1:1, no expansion will take place at all. If you set the ratio to values larger than about 5:1, Pro-G will starting acting more and more as a gate.

Range

With the Range knob (often also called floor), you can specify the maximum expansion range of Pro-G. For example, when you set it to 20 dB, Pro-G will reduce the signal level with a maximum of 20 dB. This way, you can choose to suppress unwanted background noise only a bit, or expand only a specific area of the dynamic range.

Time controls, style and knee

The section at the right of the interface contains the controls that affect the speed and feel of the current chosen gate/expander style.

Style

With the Style selection button, you can choose between various gate/expander styles, all tailored and fine-tuned carefully to meet specific needs or offer a certain character:

- Classic - This style brings you the flavor of gating and expansion as often found in vintage, high end mixer channel strips. It can be quite aggressive, but also subtle when needed. It's a great all-round style for mixing purposes and works especially well on drums.
- Clean - Designed to be as clean as possible and minimize flutter and distortion, the Clean style is great for transparent gating and expanding.
- Vocal - Since a gate/expander is very often used on vocals, we have developed a special vocal gating algorithm. It retains the natural feel of the vocal, opening the gate gently when the singer breathes in, and releasing gently, yet fast enough to reduce unwanted noise or bleed.
- Guitar - Another common application of a gate/expander is on electric guitar before distortion, to reduce or minimize rumble. With this style, especially when used in the lower ratio range (2:1 to 5:1), Pro-G gently follows the natural decay of the guitar sound, ensuring that even after distortion, the result still sounds very natural and lively.
- Upward - As a special treat, an upward expansion algorithm is also included. When you choose this style, separate Threshold and Ratio parameters are used with custom, smaller ranges. In Upward mode, the expander will amplify signals above the threshold instead of reducing them below threshold. When used moderately and with care, you can achieve very natural and transparent sounding expansion effects.
- Ducking - Finally, Pro-G also features a dedicated ducking mode, as found in many classic gates. A typical application of ducking is to automatically lower the level of a musical background track when a voice-over starts, and to automatically bring the level up again when the voice-over stops (in movies and on radio broadcasts). It is similar to compression with a side-chain, but it can sound quite different, because it's

All styles have built-in, carefully tuned hysteresis where needed. This will cause the gate to close at a slightly lower threshold level than the threshold at which it will open (as set by the Threshold knob), avoiding flutter when the incoming audio signal hovers around the threshold.

Attack

The Attack knob sets the speed with which the expander/gate will open when the signal level exceeds the threshold. For transient-rich program material like drums, fast attack times are needed to preserve punch. FabFilter Pro-G is capable of very fast attack times and they are program dependent.

Release

The Release knob sets the time that the expander/gate takes to close and reach maximum gain reduction. Just like the attack, the behavior is very program dependent, depending heavily on the audio you're processing.

Hold

The Hold knob sets the minimum time that the gate/expander will remain fully opened after the sound level has exceeded the threshold.

Knee

By choosing a custom soft knee setting, the gate/expander will react more gradually when sound drops below the threshold. You can of course clearly see the effect of the Knee in the transfer curve shown within the level display.

Lookahead

With the Lookahead knob (often also called pre-open), the gate/expander can be set to open up to 10 ms before the audio level actually exceeds the threshold. This is an excellent way to preserve transients, while still avoiding ultra-fast attack times that might cause distortion or aliasing.

You can enable or disable look-ahead with the Lookahead Enabled button, just at the right-top of the Lookahead knob. When lookahead is disabled and oversampling is off, Pro-G works without any latency. When enabled, the latency will be 10 ms, plus a small additional latency if oversampling is enabled.

Tips

- When using Upward expansion, it's best to avoid using high ratios in combination with a low threshold, because this can lead to quite extreme amplification.

The real-time level display

The large display in the center of the interface shows the output level (light blue) versus the input level (dark blue), with a fixed 60 dB scale. This immediately displays when the gate/expander is open or closed, and how it is reacting to the incoming audio signal.

You can enable or disable the real-time display using the toggle button in the right-bottom corner, right under the meter scale, in case you find it distracting while listening.

The transfer curve

Within the level display, the transfer curve is shown. It visualizes the effect of the Threshold, Ratio, Range and Knee settings. Together with the thin dashed threshold lines, this makes it easy to set up the gate/expander.

The level meter

Directly next to the level display, the level meter shows the output level versus the input level as well, which makes it easy to interpret the current gating/expansion process. Above the meter is a dB read-out, showing the maximum level that had been detected. You can reset it by simply clicking on it. When the output exceeds 0 dBFS, the meters will indicate clipping. To reset this, just click on the meter area or on the read-out.

Note: the actual audio is not clipped within Pro-G. The clipping indication is primarily useful for hosts that always keep all audio within the 0 dBFS range.

Undo and Redo

The Undo and Redo buttons at the top of the plug-in interface enable you to easily undo changes you made to the plug-in.

- The Undo button at the left undoes the last change. Every change to the plug-in, such as dragging a knob, or selecting a new preset, creates a new state in the undo history. The Undo button steps back through the history to restore the previous states of the plug-in.
- The Redo button cancels the last Undo command. It steps forward through the history until you are back at the most recent plug-in state.

If the plug-in parameters are changed without using the plug-in interface, for example with MIDI or automation, no new undo states are recorded.

The Undo and Redo buttons will disable themselves if there is nothing to undo or redo.

A/B

With the A/B feature in FabFilter Pro-G, you can easily switch between two different states of the plug-in.

- The A/B button switches from A to B and back. Before switching, the current state of the plug-in is saved, so if you click this button twice, you are back at the first state. The button highlights the currently selected state (A or B).
- The Copy button copies the active state to the inactive state. This marks the current state of the plug-in and allows you to go back to it easily with the A/B button. After clicking Copy, the button disables itself to show that both states are equal, so there is nothing to copy anymore.

Presets

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The preset button shows the name of the current preset. If you have changed the preset by adjusting one or more parameters, the name is dimmed to indicate that this is not the original preset anymore.

- To save the current setting as a preset, click the preset button, and then click Save As. A standard Save dialog will appear. Type a name for the new preset and click Save to finish.
- In the Save dialog, you can also rename and delete existing presets and create a new folder to store presets in. New folders will show up as new categories in the preset menu.

MIDI Learn

Controlling FabFilter Pro-G's parameters directly with MIDI is very easy using the MIDI Learn feature. With MIDI Learn, you can associate any MIDI controller with any parameter.

Click the MIDI Learn button in the bottom bar to enter MIDI Learn mode. The interface dims and the parameters that can be controlled are highlighted. Each parameter has a small text balloon that displays the associated controller number. Now do the following to associate a controller number with a parameter:

7. Touch the control of the desired parameter in the interface that you wish to control. A red square will mark the chosen parameter.
8. Adjust the slider or knob on your MIDI keyboard or MIDI controller that you want to associate with that parameter.

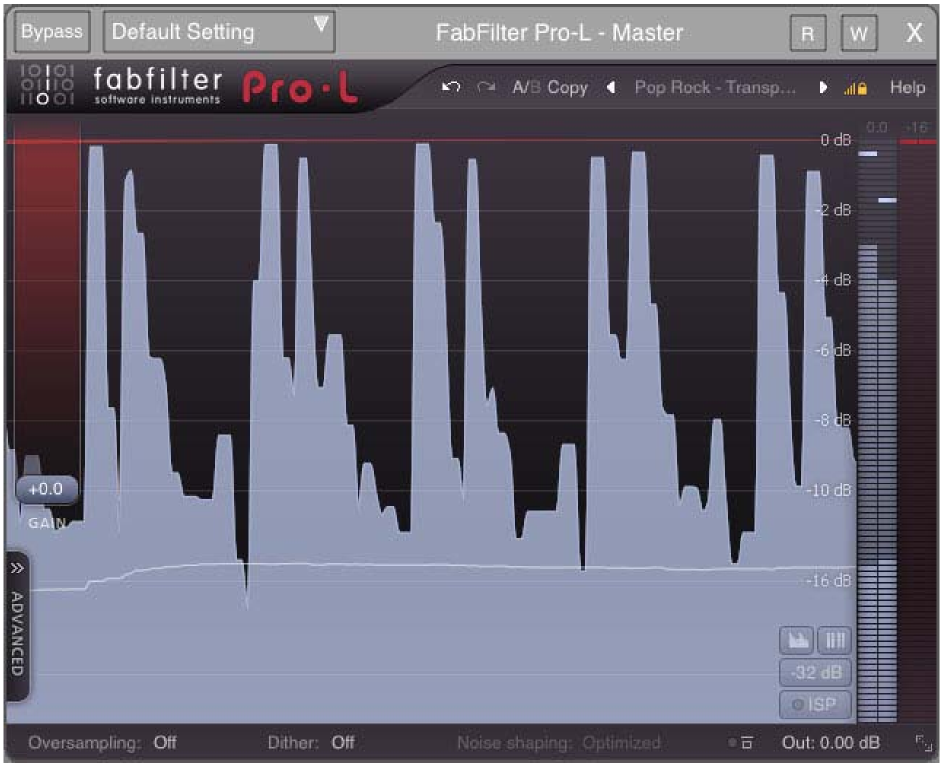
That's it! The parameter will now be controlled with the MIDI controller. You can now go back to step 1 to associate a different parameter. Note that there is no warning when you associate a different knob with a controller number that is already used. It will just be replaced.

To exit MIDI Learn mode, click the MIDI Learn button again, or click Close at the top of the interface.

Click the small menu drop-down button next to the MIDI Learn button to access the MIDI Learn menu:

- Disable/Enable MIDI - This globally turns MIDI control of parameters on or off: useful in hosts that automatically send all MIDI events on a track to all effect plug-ins associated with that track as well.
- Clear - This submenu shows all parameter associations and lets you delete individual associations or clear all associations in one step.
- Revert - Reverts to the last saved MIDI mapping (or the state when the plug-in was started).
- Save - Saves the current MIDI mapping so Revert will go back to this state. The current mapping is automatically saved when closing the plug-in.

FabFilter Pro-L



The limiter is an indispensable tool in modern mastering and mixing, and there are a lot of different flavors available on the market today. Some limiters try to be as transparent and 'safe' as possible, while others are designed to just go loud. One limiter may work best on rock music, while another performs best on electronic dance music.

FabFilter Pro-L is a professional brickwall limiter, that combines all these flavors and qualities in one plug-in, making it suitable for any type of music/audio. It can be as transparent as needed, and can go as loud as you want it to go. Equipped with excellent metering, oversampling and dithering, Pro-L is everything you need in a limiter.

Key features include four different limiting styles, all with their own characteristics, adjustable look-ahead, attack and release settings, separate channel link settings for transients and release, dithering with three different noise shape settings, up to four times oversampling, accurate and clear metering with K-System support and a unique real-time limiting display.

Recommended workflow

To get the best results with FabFilter Pro-L, we recommend the following steps:

Step 1: Choose a good starting point preset

Of course, the simplest way to start working with Pro-L, is to just open it with its Default Setting preset and move the Gain slider up until you reach the desired level. We have chosen the default preset carefully, to work well on almost any audio.

Instead of just using the default settings however, you can also try one of the excellent factory presets that come with FabFilter Pro-L. They are divided per musical genre and have descriptive names, so you can easily choose the preset that works best for your purpose and audio material.

To make it easy to try and compare different presets on your audio, while leaving the current amount of limiting, the specified output level and other output settings unaffected, make sure the Lock Output button is enabled. It's placed directly right to the preset controls, at the top of the interface. When enabled, the current values of the Gain, Output Level, Oversampling, Noise Shaping and Dithering parameters will be preserved while loading presets.

After a while, you'll probably have a favorite preset and favorite settings that you use most of the time. In that case, it's a good idea to override the Default Preset so that the next time you open FabFilter Pro-L, you're ready to go right away. To save all current settings as default, simply choose Options > Save As Default from the preset menu.

Step 2: Refine settings if needed

If you like to, you can of course adjust or refine the settings that influence the sound and flavour of limiting. Just open up the Advanced panel to get access to settings like Style, Lookahead, Attack, Release and Channel Link.

To learn how to interpret these settings and adjust them with sense, see Advanced settings.

Step 3: Set up Oversampling, Dithering and Output level with care

The final step in setting up the limiter is choosing the correct output settings. Whether you use dithering and noise shaping depends on your preferences and requirements. See Dithering and noise shaping for more information.

"So what's an appropriate Output Level setting? Should I choose -0.1 dB, -0.2 dB or -0.3 dB?"

Actually, there is not a single correct value for the Output Level parameter. It mainly depends on the presence and strength of inter-sample peaks in the outgoing audio signal. In short, our advice is simple. Turn on ISP metering, which exposes these inter-sample peaks. Then adjust the level so that the level meter doesn't exceed the maximum of 0.0 dBFS. This makes it very likely that a subsequent D/A conversion (or any other conversion) will handle your audio without introducing distortion.

Notes

- Not everyone cares about inter-sample clipping: see Output options for the full discussion.
- Inter-sample peaks are introduced when the limiter needs to react very quickly to the incoming signal. By increasing the Lookahead setting, you allow the limiter more time to respond, and therefore reduce distortion and inter-sample peaks.
- Another way to reduce inter-sample peaks is by using oversampling. In most cases, choosing 4x oversampling in combination with a minimum lookahead time of 0.1 ms, keeps inter-sample peaks within the range of 0.1 dB.

Metering

Accurate metering is extremely important in a limiter plug-in. To give you a perfect view of what's happening to your audio, FabFilter Pro-L offers very accurate output and gain change meters, including a textual representation of maximum peak levels, as well as a large real-time level display, showing levels and limiting over time. Using the Meter Scale button, you can choose a scale that fits your need. You can choose between three normal scales and three K-Metering scales.

Normal metering scales

FabFilter Pro-L has three normal metering scales. All three general scales have linear precision in the upper part of the metering, offering the best precision where limiting mostly happens:

- 16 dB: Showing the top 5 dB of input, output and gain reduction meters in the linear upper part of the metering, this scale offers a precise view of limiting in the top ranges.
- 32 dB: With a bit less, but still enough detail, combined with a fairly large overall range, this scale offers the best of both worlds; a good insight in the applied limiting and a proper impression of overall levels.
- 48 dB: Covering a wide range of 48 dB, using this scales gives you the best a general overview input and output levels.

Like the K-System meters described below, the normal output meter shows the RMS level and the peak level at the same time, but with a longer RMS integration time of 2000 ms. Above the meters, the maximum peak output level and gain reduction is displayed. Click on the level text to reset it.

K-System metering scales

The K-System, introduced by mastering engineer Bob Katz in 1999, is a protocol for setting mix and monitor calibrations in a studio environment. It is an attempt to standardize leveling practices throughout the audio industry. It uses three separate standards known as K-20, K-14, and K-12. With each step (from K-20 to K-12), the available dynamic range decreases as the average level increases. The top label of the meter scale indicates the maximum head-room (either 20dB, 14dB or 12dB), and just as with normal metering this matches the the full-scale digital. Your monitor gain should be calibrated carefully, so that the level at the 0 dB label of the meter matches 83 dBC.

The K-System meters show both peak and RMS level at the same time. The top red zone of the meters is the loud or fortissimo zone. In music recording, the RMS level should only reach the red zone in the loudest passages, climaxes or occasional peak moments. If you find yourself using the red zone all the time, you might want to check whether your monitor gain is properly calibrated.

- K-12: This scale is intended to be used exclusively for broadcast material, be it radio or television. With this system, -12dBFS = 0VU = 85dB SPL. The limited headroom of 12dB explains its exclusiveness to heavily compressed broadcast material.
- K-14: This should be the standard for the majority of commercial recordings created for home listening. Pop music and home theatre mixes are examples of material that would fall under K-14, where -14dBFS = 0VU = 85dB SPL. The available headroom is 14dB. The K-14 scale is probably the most widely used of the three standards.
- K-20: Offering the widest available dynamic range of the three systems, this scale should be used primarily for large theatrical mixes, dynamic music mixes, and Classical style mixes. Any material with a wide dynamic range should be reserved for the K-20 standard. In this system, -20dBFS = 0VU = 85db SPL. As you might guess, 20dB of headroom is available.

To read more about the K-System and how to use it properly (including monitor gain calibration), read Bob Katz' article: An Integrated Approach to Metering, Monitoring, and Levelling Practices .

The real-time level display

The large real-time level display shows input level (grey), output level (light blue), gain reduction (red) and RMS level (white line) at the same time. It gives you a very good insight in the amount of limiting going on, and the overal peak and average levels.

In case you don't want to be distracted by the display, you can simply disable it using the Show Display button, right under the meter scale select button, left of the level meters. The Show Meters button next to it disables the level meters so you can turn off all visual feedback.

ISP (Inter-Sample Peak detection)

Digital audio processing, and especially ultra-fast limiting or hard clipping in the digital domain, can introduce harmonic frequencies that can't be expressed properly with the sample rate you're using. Still, a D/A converter needs to interpret that signal and translate it to an analog wave form. At some points, especially at sharp transients as a result or limiting/clipping, the resulting wave form that is constructed out of the samples, can have peaks that are higher than the peaks in your original digital signal. The quality of your D/A converter will determine how these peaks are handled, and how they affect the sound.

Of course, it's always best to minimize inter-sample peaks, and at the same time ensure that any peaks in your audio, both normal and inter-sample, stay within the 0 dBFS range. Then you can be sure that D/A conversions do not introduce unwanted distortion.

Using the ISP button, you can enable inter-sample peak detection. Pro-L's output level meter will then show the inter-sample peaks in the resulting audio, so you can correct the output level for them. The level readout above the output meter will show exactly how much more headroom is needed. After adjusting the output level, click on the level text to reset it.

Normally, using Pro-L's oversampling (preferably 4x) in combination with a minimum lookahead time of 0.1 ms (which is still very fast), brings down inter-sample peaks to a range of only about 0.1 dB. However, the ISP option will show you exactly what's going on, so you don't have to guess.

Tips:

- After working with Pro-L for a while, you probably have your own favorite metering settings. You can simply save all current parameter and interface settings of the plug-in as the default startup settings, by choosing Options > Save As Default from the plug-in's preset menu.
- You can also switch to the Compact interface layout to hide the real-time display and see larger output meters. See Compact view.

Advanced settings

Most of the time, choosing the right preset from FabFilter Pro-L's preset menu works very well. But in some cases, you might want to adjust and fine-tune the limiting behavior. To access Pro-L's advanced settings, click the Advanced button right under the Gain slider at the left of the interface, which makes the advanced panel slide out. Clicking the button again will hide the panel.

Style

FabFilter Pro-L comes with four advanced and highly program-dependent limiting algorithms, which all have their own distinct character. One is not better than the other; they are all great, and they all go very loud if you want them to.

It is best to select the appropriate algorithm depending on what kind of effect you want to achieve: some are designed to be as transparent as possible, while others may add a nice punch or flavor. Here's a short description of all algorithms:

- Transparent: As the name says, this algorithm is designed to stay true to the original sound and feel as much as possible, avoiding pumping effects and coloring. It works great on most program material, but especially

- Punchy: Of all algorithms, this one is the most apparent. When pushed, it becomes quite punchy and will introduce a bit of pumping, which can add some nice flavor. Since it's fairly 'safe', minimizing distortion most of the time, it works miracles on single tracks, like vocals, bass or guitar and can give a nice edge to beat-oriented music. It performs best with a normal Lookahead (> 1.0 ms), an Attack around 250 ms and Release around 500 ms.
- Dynamic: By enhancing transients before actually applying limiting, this algorithm excels in preserving the original punch and clarity of your audio. It's probably the best algorithm for rock music, but it can work surprisingly well on other types of audio as well. The best starting point is to choose a very short Lookahead time (< 0.5 ms) and set the Attack knob and Release knob half way, and adjust them from there. Note however that this algoritm uses more CPU power than the other algorithms, and also introduces more latency.
- Allround: This algorithm is designed to be very 'safe', minimizing distortion while still going as loud as possible. It can work well on almost any program material! Try starting with a fast Lookahead (around 0.5 ms), an Attack around 250 ms and Release around 500 ms.

Lookahead

The Lookahead knob sets the look-ahead time for the initial 'transient' stage. This allows the limiter to examine the incoming audio in advance and predict the amount of gain reduction needed to meet the requested output level. If the look-ahead time is very short, the limiter doesn't have much time to move to the desired level: this will generally have the effect of preserving transients better and increasing the apparent loudness, but at the expense of possible distortion. Longer look-ahead times are safer, but less loud.

Very short look-ahead times (less than 0.1 ms) will approximate 'hard clipping', introducing distortion and aliasing. This causes inter-sample peaks which can cause further distortion later on. To reduce aliasing and inter-sample peaks, we advise to use oversampling. Also, turn on ISP metering to visualize the inter-sample peaks generated. This enables you to adjust the output level accordingly.

Attack and Release

Apart from the fast 'transient' stage, the limiter has a slower 'release' stage that responds to the overall dynamics of the incoming audio. The Attack and Release knobs control how quickly and heavily the release stage sets in. Shorter attack times will allow the release stage to set in sooner; longer release times will cause it to have more effect.

In general, short attack times and long release times are safer and cleaner, but they can also cause pumping and reduce clarity. On the other hand, long attack times and short release times can increase apparent loudness and presence, but at the expense of possible distortion.

Channel linking

When limiting a stereo signal, it is generally desirable to process both channels in the same way to avoid changing the stereo image inadvertently. However, when a short peak occurs in one channel, removing it is often almost inaudible. In this case, it is better to remove it only in the channel where it occurs.

You can control this behavior completely with the two channel linking knobs. The Transient knob controls the amount of channel linking for the 'transient' stage that mainly operates on short peaks. It often works well to choose less than 100% here. The Release knob controls the channel linking for the 'release' stage, where it is best to start with 100% which will completely link the channels. However, you can of course experiment with different settings depending on the level of limiting and the character of the incoming audio signal.

Tips:

- When you first start to work with FabFilter Pro-L, try to use the factory presets. These are great starting points, smartly divided in music categories, and with descriptive names. These might just do the trick already, without even needing to open the Advanced Panel.
- If you have found a preset that you really like, which works well on most music you usually process, you can easily save it as the default preset! Just choose Options > Save As Default from the plug-ins preset menu, and the next time, Pro-L with startup with your favorite settings.

Oversampling

The limiting algorithm often needs to make very quick changes to the audio in order to remove peaks while preserving transparency and apparent volume. These sudden changes can introduce aliasing, which causes distortion and generally reduces the quality of the audio signal. Oversampling is a way to reduce that aliasing by running the internal limiting process at a sample rate that is two or four times higher than the host's sample rate.

"When do I need to turn on oversampling?"

You need it more when the limiting process operates faster (using short lookahead times), and when limiting more heavily, both leading to a higher level of aliasing. The aliasing will cause inter-sample peaks and these can cause distortion later on, for example during D/A conversion. There are only two small drawbacks to oversampling: it increases CPU usage, and it can introduce a very slight pre-ring due to the phase-linear filtering that is needed. Generally this effect is so small that it's inaudible, but it's good to be aware of this and not blindly assume that oversampling is always better.

"Why can my output level exceed the specified Output Level setting when oversampling is enabled?"

When using oversampling, limiting is applied to the upsampled audio (two or four times the normal sample rate), ensuring that no sample value in the upsampled result will exceed the specified Output Level. However, even though most aliasing is filtered out during the final downsampling stage, still some inter-sample peaks may exist. Because of these peaks, the downsampling process which reconstructs the audio in the original sample rate, can generate waveforms with a slightly higher level than the specified Output Level. The amount of that overshoot highly depends on the speed and amount of limiting. In most cases, using a minimum lookahead time of 0.1 ms keeps the overshoot within the range of 0.1 dB.

Using ISP metering and oversampling

As we just explained, oversampling might already reveal the presence of inter-sample peaks in the resulting audio. But to fully expose them, you should use ISP metering. This also clearly visualizes the benificial effects of oversampling. Using 4x oversampling will dramatically reduce the inter-sample peaks, which in turn allows you to increase the desired output level without inter-sample clipping. You can also clearly see that if you use a slightly longer look-ahead time, there will be less inter-sample peaks, both with and without oversampling.

Notes

When you use very short look-ahead times of less than 0.1 ms (clipping), it may become difficult to keep inter-sample peaks within a workable range. Again, in most cases, choosing 4x oversampling in combination with a minimum lookahead time of 0.1 ms, keeps inter-sample peaks within a small range of 0.1 dB.

Dithering and noise shaping

In modern music production, most people are used to working with 24-bit audio to preserve as much resolution and precision as possible. Plug-ins usually work with 32-bit or 64-bit floating-point sample values. However, most audio finally ends up on a normal CD that only uses 16 bits of resolution. This means that at some point, the bit depth has to be reduced to suit the final medium.

The simplest way to reduce the bit depth of an audio signal, is to just truncate the least significant bits of every sample. However, this causes quantization distortion (in the form of unwanted harmonics) in the resulting audio. The best way to avoid this distortion, is by adding a tiny bit of white noise to the audio signal before truncating any bits. This eliminates the nasty quantization distortion at the cost of a slightly higher noise level in the final audio. This is called dithering.

To make the effect of applying dithering noise less audible in the final audio (in other words: to improve the signal-to-noise ratio), we can filter the noise introduced by the dithering process. That way, we don't end up with plain white noise (having a flat spectrum), but with noise that is less audible at frequencies where the human ear is most sensitive.

Dithering in FabFilter Pro-L

In the bottom bar of FabFilter Pro-L's interface, you'll find the dithering settings. With two simple controls, you can specify your prefered dithering and noise shaping settings:

- The Dithering parameter specifies the desired bit depth of the resulting audio. You can choose to dither/quantize to 24, 22, 20, 18 and 16 bits. Of course, if you don't want any dithering, quantization (or noise shaping) to occur, just choose 'off'.
- The Noise Shaping setting lets you choose between various noise shaping algorithms:
    - The Basic setting lowers the overall noise floor a few dB, at the cost of increasing noise levels for frequencies above 6 kHz.
    - With the Optimized setting, the effect is more extreme; you'll get an even lower overall noise level, but noise frequencies above 10 kHz are boosted more extremely.
    - Weighted noise shaping will transform the noise spectrum according to the ear's sensitivity to certain frequencies at low listening levels. Theoretically, this results in the lowest audible noise. This noise shaping setting is designed to be used at 44.1 kHz. It still works at other sample rates, but the frequency spectrum of the resulting noise isn't optimal anymore.

Myths and facts

Theoretically, dithering the best way to retain as much resolution as possible when quantizing your audio. However, in the real world, dithering often has little to no audible effect. Here are a few things to keep in mind:

- Most of today's music is mastered at quite loud (if not ridiculously loud) average levels, leaving very little dynamics in the final result. This already masks the small level of distortion due to quantization, so dithering probably won't have any audible effect.
- A lot of audio recordings already have a relatively high noise floor, due to the use of microphones, amplifiers, analog outboard, mixing consoles etc. In that case, dithering will have no beneficial effect at all; it will just increase the existing noise floor!
- Dithering should only be used as the final stage of audio processing/mastering. With any further processing, like gain changes, applying effects, or converting to yet another bit depth, the effects of dithering will be lost. If your host offers a post-gain effect insert slot on the master channel, use this slot for FabFilter Pro-L when dithering is enabled.
- Dithering more than once doesn't make any sense. It will just increase the overall noise level in your audio.

So when should you use Pro-L's dithering? The rule of thumb would be: when you use FabFilter Pro-L in the final stage of mastering, handling audio with a very low noise floor of itself, and the end result is still fairly dynamic. But the most important advice of all is.... use your ears!

Notes

- The white noise used for dithering in FabFilter Pro-L is the industry-standard TPDF noise, 2-bit peak-to-peak.

Output options

At the right-hand side of the bottom bar in the interface, FabFilter Pro-L contains the output level and bypass parameters.

The Output Level knob plays an important role: it sets the desired maximum output level for the limiter. It seems logical to set it to 0.00 dBFS: you want the output to be as loud as possible, right? Not so fast.

Due to its ultra-fast behavior, the limiting process can generate inter-sample peaks: while none of the outgoing sample values are higher than 0 dBFS, the analog wave form that will be constructed out of the samples by the D/A converter can actually exceed this by several dB. Actually, any conversion that reinterprets the wave form can expose inter-sample peaks. This will lead to unpredictable clipping and therefore possibly audible distortion.

"How bad is clipping caused by inter-sample peaks?"

This is very hard to tell: it depends on many things, such as the quality of the D/A converter and the character of the music. Many professionally mastered albums contain inter-sample clipping and this doesn't have to be a problem. The main effect is that the music may suffer from slight distortion when played by low-quality D/A converters.

"So what do you recommend?"

We recommend to turn on ISP metering to visualize the inter-sample peaks so you are at least aware of them. If you want to ensure that there won't be any inter-sample clipping during subsequent conversions, you need to adjust the output level to keep inter-sample peaks under 0 dBFS.

You may find that the inter-sample peaks are very high and that you need to reduce the output level too much to avoid clipping. In this case, there are two ways to reduce the generated inter-sample peaks: turning on oversampling, and slightly increasing the Lookahead setting in the Advanced panel. In most cases, choosing 4x oversampling in combination with a minimum lookahead time of 0.1 ms, keeps inter-sample peaks within the range of 0.1 dB, so the corresponding output level setting would be -0.10 dB.

You can bypass the entire plug-in with the Global Bypass toggle button to the left of the output level button. While most hosts already provide the ability to bypass plug-ins, our internal global bypass feature is guaranteed to correctly compensate the latency of the plug-in and it also applies soft bypassing to avoid clicks. While the plug-in is bypassed, the display dims and a red light glows in the bypass button itself.

Notes

- When loading presets, enable the Lock Output option next to the presets button to preserve the current gain and output settings. All factory presets were saved with a default output level of 0.0 dB.
- You can directly adjust the output gain by clicking and dragging the output button vertically, so there is no need to click it first to view the output knobs. You can also double-click the output button to directly enter a value using the keyboard.
- The resize button next to the Out setting lets you switch the interface to Compact view.

Undo and Redo

The Undo and Redo buttons at the top of the plug-in interface enable you to easily undo changes you made to the plug-in.

- The Undo button at the left undoes the last change. Every change to the plug-in, such as dragging a knob, or selecting a new preset, creates a new state in the undo history. The Undo button steps back through the history to restore the previous states of the plug-in.
- The Redo button cancels the last Undo command. It steps forward through the history until you are back at the most recent plug-in state.

If the plug-in parameters are changed without using the plug-in interface, for example with MIDI or automation, no new undo states are recorded.

The Undo and Redo buttons will disable themselves if there is nothing to undo or redo.

A/B

With the A/B feature in FabFilter Pro-L, you can easily switch between two different states of the plug-in.

- The A/B button switches from A to B and back. Before switching, the current state of the plug-in is saved, so if you click this button twice, you are back at the first state. The button highlights the currently selected state (A or B).
- The Copy button copies the active state to the inactive state. This marks the current state of the plug-in and allows you to go back to it easily with the A/B button. After clicking Copy, the button disables itself to show that both states are equal, so there is nothing to copy anymore.

Presets

- To load a preset, click the preset button. The presets menu will appear with all available presets. Click a menu item to load that preset. The currently selected preset is highlighted with check marks.

- To explore the presets one by one, click on the little arrow buttons to the left and right of the main preset button. This will load the previous or next preset in the menu.

The preset button shows the name of the current preset. If you have changed the preset by adjusting one or more parameters, the name is dimmed to indicate that this is not the original preset anymore.

- To save the current setting as a preset, click the preset button, and then click Save As. A standard Save dialog will appear. Type a name for the new preset and click Save to finish.
- In the Save dialog, you can also rename and delete existing presets and create a new folder to store presets in. New folders will show up as new categories in the preset menu.

MIDI Learn

Controlling FabFilter Pro-L's parameters directly with MIDI is very easy using the MIDI Learn feature. With MIDI Learn, you can associate any MIDI controller with any parameter.

Click the MIDI Learn button in the bottom bar to enter MIDI Learn mode. The interface dims and the parameters that can be controlled are highlighted. Each parameter has a small text balloon that displays the associated controller number. Now do the following to associate a controller number with a parameter:

9. Touch the control of the desired parameter in the interface that you wish to control. A red square will mark the chosen parameter.
10. Adjust the slider or knob on your MIDI keyboard or MIDI controller that you want to associate with that parameter.

That's it! The parameter will now be controlled with the MIDI controller. You can now go back to step 1 to associate a different parameter. Note that there is no warning when you associate a different knob with a controller number that is already used. It will just be replaced.

To exit MIDI Learn mode, click the MIDI Learn button again, or click Close at the top of the interface.

Click the small menu drop-down button next to the MIDI Learn button to access the MIDI Learn menu:

- Disable/Enable MIDI - This globally turns MIDI control of parameters on or off: useful in hosts that automatically send all MIDI events on a track to all effect plug-ins associated with that track as well.
- Clear - This submenu shows all parameter associations and lets you delete individual associations or clear all associations in one step.

- Revert - Reverts to the last saved MIDI mapping (or the state when the plug-in was started).
- Save - Saves the current MIDI mapping so Revert will go back to this state. The current mapping is automatically saved when closing the plug-in.

FabFilter Pro-MB

FabFilter Pro-MB

About FabFilter Pro-MB

Multiband compression and expansion are very powerful and useful tools, but many people find the concept quite difficult to grasp, or don't know how and when to use it. With FabFilter Pro-MB, multiband dynamics processing becomes intuitive yet powerful at the same time.

The traditional approach to multiband processing is to simply split the whole incoming signal into bands using a set of crossovers. In most cases however, this is overly complicated as you're often only interested in working with a particular frequency range. Instead, we have chosen to approach this from the user's perspective: think bands, not crossovers. You're working on some audio and want to adjust a certain frequency range... so just create a band at that frequency range and start working! The interface clearly reflects that the rest of the spectrum stays untouched.

Next, we have also pushed the limits to achieve the best possible sound. We've gone through a lot of research and developed our own unique Dynamic Phase processing mode. It has virtually the same frequency response as traditional multiband processing, but features zero latency operation, no pre-ringing effects, and only introduces phase changes when actually changing the gain. This mode really makes Pro-MB stand apart! Of course, we have also included an excellent Linear Phase mode and a traditional Minimum Phase mode.

As you've come to expect from FabFilter, Pro-MB has a highly intelligent and intuitive interface, making it easy to create, organize and adjust bands that are freely placed in the frequency spectrum. With its well-thought-out display, designed to achieve an optimal workflow, FabFilter Pro-MB is an absolute time-saver and a joy to use.

Key features

- Up to six processing bands, freely placed anywhere in the spectrum
- Unique Dynamic Phase processing mode plus excellent Linear Phase and traditional Minimum Phase modes
- Any form of dynamics processing, from highly transparent compression, limiting and expansion to pumping upward compression and punchy gating
- Fully customizable per band: lookahead (up to 20 ms), variable knee, variable stereo linking with mid-only or side-only processing, band/free side-chain triggering (external or internal), variable slopes between 6 dB/oct and 48 dB/oct (in Dynamic and Linear Phase mode)
- Intelligent, highly program- and frequency-dependent attack and release curves
- Unique interactive multiband display, designed for an optimal workflow
- Global dry/wet mix from 0% to 200%
- High-quality audio processing algorithms with 64-bit internal processing where needed
- Up to four times oversampling
- Accurate and smooth real-time frequency analyzer with pre- and post-processing options and 'freeze' feature
- Precise output metering

Overview

The interface of FabFilter Pro-MB is designed to be easy to use while providing all necessary information and controls. It consists of the following elements:

- Interactive multiband display - Using the interactive display, you can easily create, organize and adjust processing bands. See Display and workflow.
- Band controls - The band controls let you adjust the dynamics, level and triggering settings of one more selected bands. See Basic band controls.
- Level metering - At the right of the interface, the output level meter and level read-outs provide an immediate overview of the current output level. See Input and output options.
- MIDI Learn - MIDI Learn lets you easily associate any MIDI controller with any plug-in parameter. See MIDI Learn.
- Processing mode - The Processing mode setting specifies how the incoming signal is split into bands before processing them. See Processing mode.
- Oversampling - The Oversampling setting sets the amount of internal oversampling, which reduces possible aliasing for fast/aggressive dynamics processing and improves high-end frequency response for the Minimum Phase and Dynamic Phase processing modes, at the cost of additional CPU usage. See Oversampling.

- Analyzer settings - Using the Analyzer settings, you can enable and customize the built-in spectrum analyzer that lets you visually judge the effect of processing on the incoming signal. See Spectrum analyzer.
- Input and output options - At the far right of the bottom bar, you can bypass the entire plug-in and adjust the initial input and final output levels. See Input and output options.
- Presets, undo, A/B, help - With the preset buttons, you can easily browse through the factory presets or save your own settings so you can re-use them in other songs. The Undo, Redo, A/B and Copy buttons at the top of the plug-in interface enable you to undo your changes and switch between different states of the plug-in. Finally, the Help menu provides access to help and version information. See Loading presets and Undo, redo, A/B switch.

Display and workflow

The large display provides an overview all bands and lets you easily create new bands and edit them. Each band visualizes its potential dynamic range, while the thick yellow curve shows the overall dynamic frequency response at the present moment.

Unlike traditional multiband tools, the unique workflow in FabFilter Pro-MB does not require you to divide the entire spectrum in bands. You just create one or more bands at the frequency range that you actually want to work on, and leave the rest of the spectrum unprocessed. We take care of the rest!

Creating bands

- To create the first band, just click anywhere in the display.
- To create more bands, hover over an empty area or over an existing band in the display and click the + button at the top.
- Alternatively, you can drag the yellow overall curve or double-click in an empty area.
- After creating a band, you can immediately start dragging to adjust it.

Selecting bands

- Click anywhere inside a band to select it.
- Click and drag on the display background to select adjacent bands by dragging a rectangle around them.
- Click once in an empty area to deselect all bands.

Adjusting and editing bands

Click the peak dot in the center of a band and drag vertically to adjust the gain, or horizontally to adjust the center frequency. All selected bands are adjusted in parallel. As you move a band across the spectrum, the other bands are squashed and then moved as well to make space.

If you move a band close to another band, it will snap to it, and they will start sharing a single crossover. Likewise, you can also snap a band to the display edge. If you want, you can use this to divide the entire spectrum in bands for a more traditional approach to multiband processing.

When tapping a snapped crossover line, two buttons appear. The Split button creates a new band between the two snapped bands. The Unsnap button unsnaps the two bands, creating a small empty area in between. This lets you move them independently again.

When you select a band, additional controls appear in the display:

- The solo/mute buttons enable you to mute a band or listen to it exclusively. Hold down the solo or mute button to solo or mute a band only temporarily, as long as the button is pressed.
- The crossover lines let you drag each crossover independently. In contrast, dragging the peak dot horizontally changes the center frequency, adjusting both crossovers in parallel.
- The slope buttons set the low and high crossover slope for the band. Each crossover can have an independently variable slope value from 6 dB/oct to 48 dB/oct. To change multiple slope values at the same time, click and drag a rectangle around the slope buttons to select more than one.
- The range line lets you easily adjust the Range parameter for a band by dragging the line up and down.

In the top-right corner of the display, there is a drop-down button to choose the display range: +/- 3 dB, 6 dB, 12 dB or 30 dB. When you are dragging a curve outside the current range of the display, the range will expand automatically as needed.

Tips

- You can turn off the automatic adjustment of the display range by clicking Auto-Adjust Display Range in the Help menu.
- Note that two display scales are drawn: the yellow scale is adjusted by the Display Range drop-down button and corresponds to the band curves, range and the yellow overall curve. The gray scale is used by the spectrum analyzer and output level meter.
- When you double-tap the peak dot, you start editing the gain of the band; double-click on the center frequency in the parameter value display to edit the center frequency instead.

Basic band controls

Once one or more bands are selected in the multiband display, controls for the selected bands will appear at the bottom of the display. The band controls will be positioned below the currently selected bands. Note that the arrow at the top of the container has a glow that matches the color of the band it's controlling right now. A subtle yellow glow indicates that you are controlling multiple bands simultaneously.

- The Threshold knob sets the threshold level for compression or expansion. Whether the band triggers on signals above or below the threshold depends on both the Range parameter and the current dynamics mode (Compress or Expand): see Dynamics Mode below. The circular side-chain level meter around the Threshold knob shows the level of the filtered and possibly stereo-linked signal that is used for detection. This feedback makes it a lot easier to choose a proper setting.
- The Range knob limits the maximum amount of applied gain change. In addition, the Range knob chooses between downward and upward compression or expansion: see Dynamics Mode below.
- The Dynamics Mode buttons select between compression and expansion. In combination with the Range knob, four different types of dynamic processing are possible: see Dynamics Mode below.
- The Attack knob sets the speed with which gain reduction sets in. Fast attack times are needed when you want to react on transients as fast as possible, for example to achieve limiting (Compress mode) or gating (Expand mode). The Attack knob shows a percentage value from 0% to 100%, because actual attack times are very program dependent, and even depend on the placement of the band in the frequency spectrum.
- The Release knob sets the speed at which the compressor/expander recovers from gain reduction. Higher release values will result in more subtle leveling. Like Attack, the Release knob shows a percentage value from 0% to 100%, because actual release times are very program dependent, and even depend on the placement of the band in the frequency spectrum.
- The Ratio slider adjusts the amount of compression or expansion that is applied, scaling the dynamic effect of the band on the input signal. For example, when applying compression with a ratio setting of 4:1, three of every four dB above the threshold will be attenuated. In comparison, the Range knob limits the final amount of compression or expansion rather than scaling it.
- The Knee slider sets the type of knee to use for the compressor/expander. A soft knee setting causes it to react more gradually around the threshold, somewhat smoothing the dynamic effect.
- The Lookahead slider sets the compressor/expander to start reacting up to 20 ms before gain change is actually detected. This is an excellent way to preserve transients, while still avoiding ultra-fast attack times that might cause distortion or aliasing. You can globally enable or disable lookahead with the Lookahead Enabled button in the bottom bar.
- The Output Level knob adjusts the final level of the band: this is equal to the gain value controlled by the peak dot for a band in the display. The Output Pan ring around the level knob adjusts the panning between mid and side levels of the band, which is very useful in many multiband processing situations. For example, you can easily make the low-end of your signal more mono, or increase high-end stereo width.
- The bypass button at the left top lets you easily bypass the currently selected bands. While a band is bypassed, it is dimmed in the display and the bypass button itself glows red. You can temporarily bypass a band by holding down the mouse on the bypass button.
- The delete button at the right top removes the currently selected bands. If you have accidentally deleted some bands, you can easily restore them using the Undo button at the top of the plug-in interface.

- The band preset button, just left of the delete button, opens a drop-down menu that lets you save and load specific settings for a band. You can simply overwrite the Default preset to change the default settings for new bands.
- The Expert button at the right enables or disables the expert controls for all bands.

Dynamics Mode

FabFilter Pro-MB can apply any kind of dynamics processing per band, using the Dynamics Mode buttons in combination with the Range knob. When the Dynamics Mode is set to Compress, use either a negative or positive range to apply downward (normal) or upward compression. The same applies to Expand mode. Here are diagrams to visualize the four different combinations:

- Downward compression - Using Compress mode in combination with a negative Range will result in normal, downward compression. The dynamic range of the signal is reduced by attenuating peaks that exceed the specified threshold level.
- Upward compression - Using Compress mode in combination with a positive gain does the opposite of normal compression: instead of reducing peaks above the threshold, it adds gain as soon as the level drops below the threshold. So this reduces the dynamic range from the noise floor up instead of from the peaks down. Upward compression can be very useful to add loudness and body, while leaving the transients untouched. Also, when used with extreme range, ratio and release values, you can achieve creative pumping effects.
- Downward Expansion - When using Expand mode in combination with a negative range, the signal will be attenuated as soon as it drops below the threshold, increasing the perceived dynamics of the signal around the threshold. This is the most common type of expansion and with higher ratio and range values, it's often called gating.
- Upward Expansion - Expand mode in combination with a positive range will again do the opposite of normal expansion: instead of attenuating the signal when it drops below the threshold, it will add gain as soon as the signal exceeds the threshold, emphasizing the peaks in the audio. So this increases the dynamic range from the threshold up instead of from the threshold down. Upward expansion is a great way to enhance transients. For example, you can easily increase the impact of a snare in a drum loop using upward expansion.

Tips

- If multiple bands are selected, the band controls will adjust all selected bands simultaneously.
- If lookahead is disabled, oversampling is turned off, and the processing mode is set to Dynamic Phase or Minimum Phase, FabFilter Pro-MB works without any latency. When lookahead is enabled, the latency will be 20 ms, plus possible additional latency for Linear Phase processing and oversampling.

Expert band controls

Next to the normal band controls, FabFilter Pro-MB contains additional expert controls, enabled by the Expert button at the right of the floating band controls. These consist of advanced yet highly useful side-chain triggering and stereo-linking options.

- The Band/Free buttons select the frequency range of the trigger signal: either the band input signal itself (which is the default setting, used when Expert mode is off), or a freely chosen range anywhere in the spectrum. Enabling Free mode will reveal side-chain filtering controls just above the band controls. It uses intelligent and adaptive filtering, which makes triggering on narrow frequency ranges a lot easier.
- The In/Ext buttons choose between the internal, normal plug-in input, or the external side chain input. For more information on connecting the external side chain in various hosts, see External side chaining.
- The Audition button lets you listen to the filtered and stereo-linked signal that will be used to trigger dynamics processing for this band. You can turn Audition mode on or off with a single click, but you can also click-and-hold the button to temporarily audition the trigger signal.

Stereo linking and mid-only/side-only processing
The Stereo Link slider sets the amount of stereo linking for the trigger input signal, and also selects between normal stereo processing or mid-only/side-only processing.

The first half of the slider range sets stereo linking from 0% (fully unlinked, channels operate independently) up to 100% (fully linked, resulting in the same gain reduction for both channels).

By dragging the slider further, the band will eventually process only the mid-signal (mono content of the processed audio), or only the side-signal (stereo content of the processed audio). Using the small Stereo Link Mode button at the right bottom of the Stereo Link slider, you can toggle between the two.

Mid- or side-only processing can be very useful, especially during mastering. For example, bass or lead vocals are often placed in the center of the stereo image, so only processing the mid-signal will leave all stereo content untouched, ensuring the most transparent end result possible.

To better understand the working of these settings, enable the Audition button. You can now directly hear the effect of stereo linking and mid-only or side-only processing!

Tips

- When triggering on a very specific frequency, choosing a very narrow range in Free mode often works better than the default Band mode.
- The Stereo Link slider is of course only enabled in the stereo version of FabFilter Pro-MB.

Processing mode

The Processing Mode button in the bottom bar selects the algorithm that is used to split the incoming audio into bands to be able to process them separately.

While developing FabFilter Pro-MB, we've gone through a lot of research and developed our own unique Dynamic Phase processing mode. It has virtually the same frequency response as traditional multiband processing, but features zero latency operation, no preringing effects, and only introduces phase changes when actually changing the gain. This mode really makes Pro-MB stand apart! Of course, we have also included an excellent Linear Phase mode and a traditional Minimum Phase mode.

Linear Phase
In Linear Phase mode, the resulting spectrum after splitting the signal into bands and summing them together again, is always guaranteed to have a flat phase response.

FabFilter Pro-MB's linear phase implementation guarantees an excellent frequency response, even at the lowest frequencies! Also, note that changing crossover frequencies in Linear Phase mode sounds just as smooth as when using the other modes, no zipper effects whatsoever. This might sound trivial, but it's actually quite unique in linear-phase processing.

Linear phase processing will give much more transparent results than the traditional Minimum Phase method, at the expense of quite a bit of extra latency and possible pre-ringing artifacts (which are an inevitable side-effect of any kind of linear phase processing).

Minimum Phase
Minimum Phase mode uses a traditional way of splitting the signal into bands using filters. This doesn't introduce extra latency, but it will introduce static phase changes at the crossover frequencies instead. Especially when using higher slopes, these phase effects will become very apparent and often unpleasant, which makes this method virtually unusable for mastering purposes. However, with gentle slope settings, you could use the phase effects for creative purposes.

Dynamic Phase
Finally, the unique Dynamic Phase mode gives you the best of both worlds: it results in a flat/linear phase response when no gain processing is applied, but doesn't introduce latency or pre-ringing artifacts (like linear-phase processing) or static phase distortion (like minimum phase processing). Minor phase effects will only be introduced when the gain of a band is actually changed. This is possible because the input signal isn't actually split into bands, but is treated with intelligent dynamic filtering, offering almost the exact same frequency response as with linear phase and minimum phase processing.

Dynamic Phase is by far the most transparent mode, suitable for both mastering and mixing, so we've chosen this as the default processing mode for new plug-in instances.

Tips

- You can customize the default processing mode setting (and other default plug-in settings) used for new plug-in instances via the preset menu, by choosing Options > Save As Default.

Oversampling

The dynamics algorithms often need to make very quick changes to the audio when compressing or exampling. These sudden changes can introduce a small amount of aliasing, which causes distortion and generally reduces the quality of the audio signal.

Oversampling is a way to reduce that aliasing by running the internal process at a sample rate that is two or four times higher than the host's sample rate. Additionally, this will also improve the frequency/phase response for high frequencies (near the Nyquist frequency at half the sample rate) in the Dynamic Phase and Minimum Phase processing modes.

When should I use oversampling?
You need it more when the compression or expansion is more aggressive and apparent. Usually, this is the case when using lower Attack and Release and/or higher Ratio and Range settings.

Of course, in return for a reduction of possible aliasing/distortion, the plug-in will use more CPU power when using oversampling. In addition, oversampling introduces a small latency, in addition to any processing mode or lookahead latency.

Tips

- Do you want to use zero-latency processing? Disable oversampling and lookahead in the bottom bar, and use either Dynamic Phase or Minimum Phase processing.

Spectrum analyzer

To help you judge the effect of the different dynamics processing bands on the incoming audio signal, FabFilter Pro-MB includes a powerful real-time frequency analyzer.

The spectrum analyzer is controlled by the Analyzer menu in the bottom bar.

- Off turns the spectrum analyzer off so it doesn't consume any CPU power.

- Pre turns the spectrum analyzer on and connects it to the incoming audio signal before it is modified.
- Post turns the spectrum analyzer on and connects it to the outgoing audio signal, after all the dynamics processing bands have been applied.
- Pre+Post turns the spectrum analyzer on, showing both the spectrum of the incoming audio signal and the outgoing audio signal at the same time.
- The Resolution submenu selects how precise the spectrum analyzer works. Higher resolution settings allow more precision in the low-frequency area, but because more incoming samples are needed to calculate a single spectrum, dynamic changes are less clearly visible. The Low value corresponds to a resolution of 1024 points, Medium to 2048, High to 4096, and Maximum to 8192 points.
- The Speed submenu selects the release speed of the spectrum. A fast release shows dynamic changes more clearly, while a slow release gives you more time to examine the spectrum before it disappears. Higher resolution settings generally work better with slower release speeds.
- The Tilt setting tilts the measured spectrum around 1 kHz with a specified slope, expressed in dB per octave. The default setting of 4.5 dB/oct results in a natural looking spectrum, resembling best how loudness is perceived by the human ear.
- Using the Freeze button next to the analyzer menu ( ), the spectrum will stop falling and show the maximum over time.

The spectrum analyzer uses the gray gain scale at the righthand side of the multiband display, ranging from -100 to 0 dB.

Notes

- By default, the spectrum analyzer is on and set to pre+post. If you would like different default settings, set the desired mode for the analyzer in the bottom bar, and then choose Save As Default from the Analyzer meny or overwrite the Default Setting preset. Now all parameter values you just saved, including the analyzer mode, are used as the default setup for any new Pro-MB instances.
- When skipping through presets, the current analyzer settings are not changed, but they are saved in songs.
- Hold down the Freeze button to freeze temporarily until you release the mouse button again.

Input and output options

At the righthand side of the interface, FabFilter Pro-MB offers a high-resolution output level meter. At the top of the meter, the maximum output level is displayed together with a clipping indicator. Simply click the level reading to reset it. Note that the level meter uses the gray display scale, ranging from -100 to 0 dB.

At the right bottom of the interface, you'll find the bypass, mix, input level and output level controls.

- The Global Bypass toggle button to the left of the Mix button bypasses the entire plugin. While most hosts already provide the ability to bypass plug-ins, our internal global bypass feature is guaranteed to correctly

- The Mix knob enables you to mix between the dry and processed signals, scaling the overall dynamic and static gain changes for all bands. Because the Mix knob ranges from 0% to 200%, you can also choose to increase overall gain processing instead of fading it out!
- The Input Level/Pan knob adjusts the level and L/R panning of the input signal before any processing is applied. You can use this as an alternative to changing the threshold of all bands.
- The Output Level/Pan knob adjusts the level and L/R panning of the final output signal. This lets you compensate globally for any gain added or removed by dynamics processing.

Tips

- You can directly adjust the input or output gain by clicking and dragging the button vertically, so there is no need to click it first to display the knobs.
- You can reset the level meter peak read-outs by clicking them.

MIDI Learn

Controlling FabFilter Pro-MB's parameters directly with MIDI is very easy using the MIDI Learn feature. With MIDI Learn, you can associate any MIDI controller with any parameter.

Click the MIDI Learn button in the bottom bar to enter MIDI Learn mode. The interface dims and the parameters that can be controlled are highlighted. Each parameter has a small text balloon that displays the associated controller number. Now do the following to associate a controller number with a parameter:

1. Touch the control of the desired parameter in the interface that you wish to control. A red square will mark the chosen parameter.
2. Adjust the slider or knob on your MIDI keyboard or MIDI controller that you want to associate with that parameter.

That's it! The parameter will now be controlled with the MIDI controller. You can now go back to step 1 to associate a different parameter. Note that there is no warning when you associate a different knob with a controller number that is already used. It will just be replaced.

To exit MIDI Learn mode, click the MIDI Learn button again, or click Close at the top of the interface.

Click the small menu drop-down button next to the MIDI Learn button to access the MIDI Learn menu:

- Enable MIDI - This globally turns MIDI control of parameters on or off: useful in hosts that automatically send all MIDI events on a track to all effect plug-ins associated with that track as well.

- Clear - This submenu shows all parameter associations and lets you delete individual associations or clear all associations in one step.
- Revert - Reverts to the last saved MIDI mapping (or the state when the plug-in was started).
- Save - Saves the current MIDI mapping so Revert will go back to this state. The current mapping is automatically saved when closing the plug-in.

Undo, redo, A/B switch

The Undo and Redo buttons at the top of the FabFilter Pro-MB interface enable you to easily undo changes you made to the plug-in.

With the A/B feature, you can quickly switch between two different states of the plug-in.

- The Undo button at the left will undo the last change. Every change to the plug-in (such as dragging a knob or selecting a new preset) creates a new state in the undo history. The Undo button steps back through the history to restore the previous states of the plug-in.
- The Redo button cancels the last undo command. It steps forward through the history until you are back at the most recent plug-in state.
- The A/B button switches from A to B and back. Before switching, the current state of the plug-in is saved, so if you click this button twice, you are back at the first state. The button highlights the currently selected state (A or B).
- The Copy button copies the active state to the inactive state. This marks the current state of the plug-in and allows you to go back to it easily with the A/B button. After clicking Copy, the button disables itself to show that both states are equal, so there is nothing to copy anymore.

Notes

- If the plug-in parameters are changed without using the plug-in interface, for example with MIDI or automation, no new undo states are recorded.
- The Undo and Redo buttons will disable themselves if there is nothing to undo or redo.

Loading presets

FabFilter Pro-MB comes with a selection of excellent factory presets that provide many useful multiband setups for different scenarios and types of audio.

- To load a preset, click the preset button. The presets menu will appear with all available presets. Click a menu item to load that preset. The currently selected preset is highlighted with check marks.
- To explore the presets one by one, click on the little arrow buttons to the left and right of the main preset button. This will load the previous or next preset in the menu.

The preset button shows the name of the current preset. If you have changed the preset by adjusting one or more parameters, the name is dimmed to indicate that this is not the original preset anymore.

Tips

- The Default Setting preset is loaded automatically when FabFilter Pro-MB is started. To change the default settings, simply overwrite this preset by clicking Options > Save As Default in the presets menu.
- To open a preset outside the presets folder, click Options > Open Other Preset. This might be useful if someone sends you a preset by e-mail, for example.
- If somehow the factory presets are lost or not installed properly, click Options > Restore Factory Presets in the preset menu to restore them.

MIDI Program Change and Bank Select
Loading a presets can also be done via MIDI, using Bank Select and Program Change messages. Click Options > Enable MIDI Program Changes in the preset menu to enable or disable this feature. When enabled, the corresponding bank/program numbers are shown in front of the preset name (for example: (2/65) My Preset). This means that you can load that preset by first sending a Bank Select message to select bank 2 and then sending a Program Change message to select program 65.

Important: All the presets in your preset folder are numbered automatically, starting with bank 0 and program 0. This way, you are able to access any of the presets via MIDI. However, this also means that when you add new presets to the menu, bank/program numbers of other presets might change. Be aware of this when recording program changes in a session!

Saving presets

You can easily extend the included presets with new settings to build your own library of presets for FabFilter Pro-MB that you can reuse in various projects. This is also a good way to copy settings across multiple instances of FabFilter Pro-MB in a session.

To save the current setting as a preset, click the preset button, and then click Save As. A standard Save dialog will appear. Type a name for the new preset and click Save to finish.

In the Save dialog, you can also rename and delete existing presets and create a new folder to store presets in. New folders will show up as new categories in the preset menu.

FabFilter Pro-Q

FabFilter Pro-Q

The equalizer is by far the most used tool in audio recording, mixing and mastering. That's why there are probably around 1001 EQ plug-ins available. But the strange thing is: most of them don't meet top quality standards by far! Most just lack sound quality and the ones that do sound okay often have poorly designed interfaces that obstruct your workflow and creativity.

Now here's where we come in. FabFilter Pro-Q gives you the best of both worlds: the highest possible sound quality, and an innovative interface that is designed to help you get 'that' sound quickly and easily.

Key features include up to 24 separate EQ bands (bell, low cut, low shelf, high shelf and high cut), state-of-the-art filter algorithms with precise analog modeling and unlimited internal headroom, zero-latency or linear-phase processing, mid/side support, customizable stereo placement for every band (stereo, left/mid or right/side), 6 dB, 12 dB, and 30 dB display ranges and the easiest yet most powerful EQ interface ever.

Interactive EQ display

The large display shows all EQ bands and lets you easily create new bands and edit them. The thick yellow curve shows the overall frequency response of the equalizer.

- To add a new EQ band, simply click on the yellow overall curve and drag it up or down. Alternatively, double-click or Ctrl-click (Command-click on Mac OS X) on the display background. The shape of newly created curves is determined automatically depending on where you click.
- Click the dot on an EQ band to select it.
- Hold down Ctrl (Command on Mac OS X) and click another dot to select multiple bands. Hold down Shift and click a dot to select a consecutive range of bands.
- Click and drag on the display background to select adjacent bands by dragging a rectangle around them.
- Click once on the display background to deselect all bands.

Once you have one or more EQ bands selected, the display highlights the shapes of the selected bands. The easiest way to adjust them is simply by dragging them around:

- Click and drag a selected dot to adjust the frequency and gain of all selected bands. If you have multiple bands selected, the gain of all selected bands will be scaled relative to each other.
- In the top-right corner of the display, there is a drop-down button to choose the display range: +/- 3 dB, 6 dB, 12 dB or 30 dB. When you are dragging a curve outside the current range of the display, the range will expand automatically as needed.

Tips

- It is possible to turn off the automatic adjustment of the display range by clicking Auto-Adjust Display Range in the Help menu.
- Even though frequencies above 20 kHz are generally inaudible, the display extends to 30 kHz so you can put filters above this limit. The left part of the filter, extending into the audible frequency spectrum, still affects the sound. This gives you even more possibilities to shape the frequency response of the equalizer just the way you need it.

Editing EQ bands

The selection controls below the interactive EQ display show the current settings of the selected EQ bands and enable you to adjust them precisely.

From left to right, the following parameters are available:

- The L/stereo/R buttons control which channels are affected by the selected bands.
- The shape parameter selects the filter shape of the selected bands:
    1. Bell, the traditional parametric EQ shape and probably the most versatile of them all

2. Low Shelf, to boost or attenuate low frequencies
3. Low Cut, to cut all sound below the filter frequency
4. High Shelf, to boost or attenuate high frequencies
5. High Cut, to cut all sound above the filter frequency
6. Notch, to cut a small section of the spectrum

When Low Cut or High Cut is selected, a slope parameter appears below the shape parameter that sets the steepness of the filter from 6 dB/octave to 48 dB/octave.

- The frequency knob sets the frequency of the selected band between 5 Hz and 30 kHz. If multiple bands are selected, this knob is disabled because it would otherwise set the frequency of all bands to the same value.
- The gain knob sets the gain in dB of the selected bands between -30 and +30 dB. If the Low Cut or High Cut filter shape is selected, this parameter is not used.
- The Q knob sets the bandwidth of the selected bands, widening or narrowing them. Because there are different interpretations of Q values in various EQ plug-ins and scientific papers, we have chosen the value 1 to correspond to the average bandwidth. For the shelf filters, the internal Q values are chosen such that they result in a good range of shelf shapes. Keep this in mind when trying to reproduce the filter shapes of another EQ plug-in in Pro-Q: the interpretation of the Q values might not be the same.
- The 6 dB/octave variant of the Low Cut and High Cut filters does not have an adjustable Q setting.
- The bypass button lets you easily bypass the selected EQ bands. While an EQ band is bypassed, it is dimmed in the display and a red light glows in the bypass button.
- The delete button removes the selected EQ bands. If you have accidentally deleted some bands, you can easily restore them using the Undo button at the top of the plug-in interface.

Finally, the previous/next buttons at the far right let you easily advance the selection to the adjacent band in the display. To the left of these buttons, the band number of the currently selected EQ band is shown to help you to identify this band in the host when automating EQ parameters.

Tips

When multiple bands are selected, adjusting the gain or Q knobs will set the gain and Q for all selected bands to the same value. In contrast, when you drag multiple bands in the display, this modifies the frequency, gain and Q settings in parallel without setting them all to the same value.

Solo

When you tap near an EQ dot, a parameter value display pops up showing the current parameter values for the corresponding EQ band.

Click and hold the solo button (with the headphones icon) to enter solo mode for the current EQ band. The other EQ bands will dim, just like the yellow overall curve. Simply drag the solo button to change the frequency of the band.

In solo mode, you don't hear the effect of the EQ band itself, but instead you will hear the part of the frequency spectrum that is being affected by that band. Of course, the frequency range depends on the frequency and Q settings, and is visualized in the display as well.

When using solo mode with Low Cut or High Cut bands, you will hear the frequencies that are being cut away instead of the frequencies that pass, which helps you to determine whether you are cutting the right frequencies.

Generally, solo mode aims to expose the parts of the incoming audio that matter to the current EQ band, but that you can't hear just by listening to the regular EQ sound.

Tips

You can turn the parameter value display on and off by clicking Show EQ Parameter Display in the Help menu.

Stereo options

One of FabFilter Pro-Q's best features is that it's very easy to equalize both stereo channels in a different way. This is a great way to surgically remove unwanted sound artifacts, or even to add stereo effects.

To make this even more powerful, Pro-Q offers both Left/Right and Mid/Side channel modes. In the default Left/Right mode, each EQ band works either on both stereo channels, or on the left or right channel only. This is controlled by the stereo options at the left-hand side of the selection controls:

- Click the L or R button to let the selected bands affect only the left or right channel.
- Click the stereo button (in the middle) to let the selected bands affect both stereo channels.
- Click the split button underneath the buttons to duplicate the selected band, making two identical copies, one operating only on the left channel and one operating on the right channel. This makes it very easy to slightly adjust one of the channels.

As soon as one or more of the EQ bands are operating on a single channel, the EQ display switches to per-channel mode, where it shows two overall frequency response curves: a white one for the left channel, and a red one for the right channel.

Mid/side mode

The Channel Mode parameter in the bottom bar switches between Left/Right and Mid/Side operation. In Mid/Side mode, the incoming stereo signal is converted into Mid (mono) and Side parts, which you can then easily filter independently. This is often an even better way to fix artifacts or modify stereo information because it represents the stereo signal in a more natural way.

In Mid/Side mode, everything works as described above, except that the stereo options above change to M/stereo/S buttons. In addition, the display shows the two overall frequency response curves in white (Mid) and light blue (Side) so you know at a glance in which mode Pro-Q is currently operating.

Techniques

Independent channel equalization is very useful when dealing with stereo audio containing unbalanced frequency content over the stereo field. Let's say you want to combine a stereo drum recording with a stereo acoustic guitar recording. The drum recording contains more low-mid frequencies in the left channel (for example a low tom-tom), and more high frequencies in the right channel (like cymbals or a hi-hat). The guitar sound, recorded with a mic capturing the sound-board/hole panned left and one capturing the fretboard/neck panned right, might have similar frequencies as the drum recording, making it hard to combine them in a balanced way. By using independent left/right channel EQ-ing, it is possible to balance these elements so that they do not fight each other. Instead of EQ-ing the whole stereo track of the drums and guitars one can simply EQ where it is necessary to get the two elements to complement each other.

Mid/Side EQ is perhaps most commonly used to bring some stereo elements further up within a recording, either by cutting certain frequencies in the mid channel or by boosting the wanted frequency range in the side channel. It is great for adding a bit of depth to typical hard panned rock/heavy guitar recordings where you boost the "bite" frequency range of the guitars (around 2-4kHz) with a quite narrow eq. Combine this with cutting some of the "mud" away from the side channels will give the illusion of huge guitars that still sit well within a mix.

Independent Mid/Side equalization is also often used during mastering. For example, raising high frequencies in the Side channel can freshen up the sound, while a low-cut filter in the Mid channel can work very well to clear up the low end.

Consider using linear-phase processing when filtering both stereo channels (either in Left/Right or Mid/Side mode) differently to avoid introducing unwanted phase changes.

Mono operation

FabFilter Pro-Q can also work as a mono equalizer plug-in, but in this case the stereo options and the Channel Mode parameter are not available, of course. When loading 'stereo' presets (containing EQ bands that work on e.g. the left or right channel) in the mono version of Pro-Q, all EQ bands are treated as if they work on the mono channel. You should be aware that this can sometimes yield unexpected results. For example, if a stereo preset contains two bands working on the left and right channels respectively, at the same frequency, with gain=+10 dB, this will result in a +20 dB peak in the mono version. Therefore it is best not to use any presets that use per-channel processing in the mono version of Pro-Q.

Linear-phase processing

When filtering audio, traditional analog and digital filters always introduce phase problems. What happens is that the phase of different frequencies in the signal is changed in different ways. This can subtly change the sound (not necessarily in a bad way though). It does affect transients and it can make the sound less transparent.

Moreover, problems arise when you mix a filtered and phase-altered signal with another similar signal that has not been filtered, or that has been filtered in a different way. In this case, it is very likely that the different phase components of both signals won't match up properly and will cancel each other to some extent.

This situation can for example occur when mastering. It is quite common to apply an equalizer only to a part of the song, using crossfades at the beginning and end of the affected region. Because the phase information in the original and filtered parts is different, the fades won't work as intended.

Linear-phase equalization provides an answer to these problems. Linear-phase filters change the phase of the incoming signal in the same way for all frequencies. This ensures that no unwanted phase cancellation will take place, preserving transients and the transparency of your music.

However, linear-phase filters also have some disadvantages. First of all they introduce latency: the entire signal is delayed when passing through the plug-in. Longer latency means greater resolution when making low-frequency changes to the signal, but unfortunately this also can lead to the creation of 'pre-echoes' that can make low frequency transients (e.g. a kick drum) lose their edge. Choosing the latency correctly is a compromise depending on the program material and your personal preference.

To give you the best of both worlds, FabFilter Pro-Q provides both zero-latency and various linear-phase processing modes. Apart from the differences discussed above, the EQ has the same frequency response in both modes.

To change the processing mode, use the Processing setting in the bottom bar of the interface.

Zero latency mode is the default. While it introduces phase changes, it is CPU-efficient and doesn't result in any latency, so it is the best mode for e.g. live usage. Also, it's quite possible you might like the coloration introduced by the phase changes when mixing, for example.

- Linear Phase - Low Latency provides linear-phase processing with a minimal latency. Use only with low Q settings, or when only changing the mid-high part of the spectrum. With a sample rate of 44.1 kHz, it results in a total latency of 3072 samples (about 70 ms).
- Linear Phase - Medium Latency is a good compromise between low-frequency resolution and latency and we recommend using this in general for linear-phase processing. The total latency is 6144 samples at a sample rate of 44.1 kHz (about 139 ms).
- Linear Phase - High Latency gives very good low-frequency resolution. If you need to use high Q settings when changing the low end of the spectrum, use this mode. The total latency is 12288 samples at a sample rate of 44.1 kHz (about 279 ms).
- Linear Phase - Maximum Latency results in even better low-frequency resolution at the expense of latency and possible pre-echo problems. The total latency here is 24576 samples at a sample rate of 44.1 kHz (about 557 ms).

To conclude, Pro-Q lets you freely choose between zero-latency and linear-phase processing as you go. If you use high Q settings combined with low-frequency filtering, you need to use a higher latency; if you only work on the mid-high frequencies, you can get by with a lower latency.

Notes

- When working with different sample rates, the latency in samples of the various linear-phase modes can change to give you approximately the same low-frequency resolution (and the same latency in ms).
- Due to Pro-Q's advanced filter design, the CPU usage is very low, even when using up to 24 EQ bands, and it doesn't change much with the different linear-phase processing modes. On the other hand, the latency might be higher than in other plug-ins.

Spectrum analyzer

To help you judge the effect of the combined EQ bands on the incoming audio signal, FabFilter Pro-Q includes a powerful real-time frequency analyzer.

The spectrum analyzer is controlled by the Analyzer menu in the bottom bar.

- Off turns the spectrum analyzer off so it doesn't consume any CPU power.
- Pre-EQ turns the spectrum analyzer on and connects it to the incoming audio signal before it is modified.

- Post-EQ turns the spectrum analyzer on and connects it to the outgoing audio signal, after all the EQ bands have been applied. Note that the global bypass parameter is ignored by the Post-EQ mode: if global bypass is active, the analyzer will still show the signal after having been filtered by all EQ bands.
- Pre+Post turns the spectrum analyzer on, showing both the spectrum of the incoming audio signal and the outgoing audio signal at the same time.
- The Resolution submenu selects how precisely the spectrum analyzer works. Higher resolution settings allow more precision in the low-frequency area, but because more incoming samples are needed to calculate a single spectrum, dynamic changes are less clearly visible. The Low value corresponds to a resolution of 1024 points, Medium to 2048, High to 4096, and Maximum to 8192 points.
- The Speed submenu selects the release speed of the spectrum. A fast release shows dynamic changes more clearly, while a slow release gives you more time to examine the spectrum before it disappears. Higher resolution settings generally work better with slower release speeds.

If the spectrum analyzer is enabled, a separate gain scale is shown at the left-hand side of the EQ display, ranging from -72 to 0 dB.

Notes

- When skipping through presets, the current analyzer settings are not changed, but they are saved in songs.
- By default, the spectrum analyzer is off. If you would like it to be on by default, set the desired mode for the analyzer in the bottom bar, and then overwrite the Default Setting preset. Now all parameter values you just saved, including the analyzer mode, are used as the default setup for any new Pro-Q instances.

Output options

At the right-hand side of the bottom bar in the interface, FabFilter Pro-Q contains a set of global output level and bypass parameters.

- The Output Gain parameter lets you adjust the output level between minus infinity and +36 dB. You can use this to correct any overall level change that the EQ bands might introduce. Note that FabFilter Pro-Q features unlimited internal headroom so it won't clip internally at any level. You only need to be concerned about any clipping that might occur after the signal has left Pro-Q.
- The Output Pan parameter lets you change the relative levels of the left and right audio channels. When Mid/Side mode is active, it adjusts the relative levels of the mid and side channels instead.
- You can bypass the entire plug-in with the Global Bypass toggle button to the left of the output level button.
- While most hosts already provide the ability to bypass plug-ins, our internal global bypass feature is guaranteed to work correctly in linear-phase mode (compensating for the latency of the plug-in) and also

- The output level meter at the far right of the bottom bar shows the current output level, together with a clipping indicator that lights up red if the output signal has exceeded 0 dB. Click on the meter to reset it. You can hide/show the meter by clicking Show Output Level Meter in the Help menu.

Note that FabFilter Pro-Q has unlimited internal headroom and will never clip itself: the clipping indicator merely warns against possible clipping during further processing of the output signal.

Tips
- You can directly adjust the output gain by clicking and dragging the output button vertically, so there is no need to click it first to view the output knobs.

You can hide/show the small output level meter by clicking Show Output Level Meter in the Help menu.

MIDI Learn

Controlling FabFilter Pro-Q's parameters directly with MIDI is very easy using the MIDI Learn feature. With MIDI Learn, you can associate any MIDI controller with any parameter.

Click the MIDI Learn button in the bottom bar to enter MIDI Learn mode. The interface dims and the parameters that can be controlled are highlighted. Each parameter has a small text balloon that displays the associated controller number. Now do the following to associate a controller number with a parameter:

11. Touch the control of the desired parameter in the interface that you wish to control. A red square will mark the chosen parameter.
12. Adjust the slider or knob on your MIDI keyboard or MIDI controller that you want to associate with that parameter.

That's it! The parameter will now be controlled with the MIDI controller. You can now go back to step 1 to associate a different parameter. Note that there is no warning when you associate a different knob with a controller number that is already used. It will just be replaced.

To exit MIDI Learn mode, click the MIDI Learn button again, or click Close at the top of the interface.

Click the small menu drop-down button next to the MIDI Learn button to access the MIDI Learn menu:

- Disable/Enable MIDI - This globally turns MIDI control of parameters on or off: useful in hosts that automatically send all MIDI events on a track to all effect plug-ins associated with that track as well.
- Clear - This submenu shows all parameter associations and lets you delete individual associations or clear all associations in one step.
- Revert - Reverts to the last saved MIDI mapping (or the state when the plug-in was started).
- Save - Saves the current MIDI mapping so Revert will go back to this state. The current mapping is automatically saved when closing the plug-in.

FabFilter Saturn

The distortion effect has played a huge role in music history. By driving and distorting a guitar amplifier, rock-and-roll was born in the 1960s! Since that day, distortion has been used in many forms, and not only to get that crunchy electric guitar sound. Today, distortion is used to color sounds in various ways while mixing, by driving vacuum tubes, saturating tape and even by reducing bit rate.

FabFilter Saturn offers various flavors of distortion, and combines it with multi-band audio processing and virtually endless modulation possibilities. From subtle, clean and warm tube or tape saturation to the wildest multi-band guitar amp effects: FabFilter Saturn delivers!

Overview

FabFilter Saturn's interface is divided into multiple sections:

- Presets, undo, A/B, help - The Undo, Redo, A/B and Copy buttons at the top of the plug-in interface enable you to undo your changes and switch between different states of the plug-in. With the preset buttons, you can easily browse through the library of factory presets or save your own settings so you can re-use them in other songs. The Help button provides access to the help file and other information and options.

- Interactive multi-band display - Via the interactive display, you can directly create and select frequency bands. At the same time, it's a real-time frequency analyzer, making it easy to decide where to set the cross-over frequencies.
- Band controls - Using the band controls, you can adjust the settings of the selected bands. For each band, you can separately adjust the distortion type, drive, feedback settings, dynamics, tone, level and mix settings.
- Modulation button - The modulation button shows or hides the entire modulation section at the bottom of the interface. FabFilter Saturn offers virtually unlimited modulation possibilities, but all this power might be a bit intimidating. That's why the modulation section is hidden by default, and you can look 'under the hood' when you want to tweak a preset or design your own.
- Source selection bar - The source selection bar shows all modulation sources at a glance and lets you easily scroll around and create new sources. FabFilter Saturn offers XLFO, Envelope Generator (EG), Envelope Follower (EF), MIDI and XY Controller sources. See also Modulation.
- Modulation slots and sources - The bottom section contains the modulation sources. The modulation section in Saturn is fully modular — but without the cables! We found a simple way to show you everything that is modulating, and what is modulated by what. Above each modulation source, the modulation slots show exactly what targets are modulated by this source and let you adjust the amount of modulation. You can very easily set up modulation connections with drag-and-drop. All in all, we think we made sound design easier and more fun!
- Resize - The resize button in the lower-right corner lets you choose between normal and wide interface layouts. The wide layout eliminates scrolling in the top part of the interface and provides more space for the modulation sources at the bottom of the interface. Most hosts support dynamic resizing of the interface; otherwise just close and re-open the interface window.

What-you-use-is-what-you-see

Often an impressive feature list results in an impressively difficult-to-use interface full of controls for parameters you might never even use. For almost every plug-in developer one of the greatest challenges when making a complex full-featured plug-in is to design an interface that is easy to use. And we think we did it! FabFilter's interface concept: What-you-use-is-what-you-see.

The idea is simple yet powerful. At all times, the interface only contains the modulation sources and slots that you are actually using. You can even hide the complete modulation section if you only want to browse presets, and don't feel the need to look 'under the hood'! This results in an intuitive user interface that experienced producers and novices alike will embrace.

And if you need power, it's at your fingertips. Do you want another XLFO? Just add one! Do you want an envelope generator? Just add one and start modulating things! Of course there is a limit to the number of sources you can create, but in practice it feels like you can create as many sources as you will ever need.

Interactive multiband display

FabFilter Saturn's top section consists of a large interactive multi-band display, which makes it very easy to create and select frequency bands and adjust their level and drive settings. At the same time, it's also a real-time frequency analyzer.

- Simply click anywhere in a band to select it.
- Click and drag on the display background to select adjacent bands by dragging a rectangle around them.

If only one band is active, it will be selected by default and the band controls are automatically linked to that single band. Once you have multiple frequency bands available, the display highlights the level buttons of the selected bands and the band controls are linked to these bands. In the screen shot above, the middle band is selected. In addition, the band controls will slide underneath the selected band.

- Click and drag a selected level button vertically to adjust the level setting of all selected bands. If you have multiple bands selected, the level of all selected bands will be adjusted with the same relative change.
- To change a crossover frequency, click and drag the vertical crossover split. Alternatively, click and drag a level button horizontally to change the crossover frequencies at both edges of the band.
- Double-click a level button or a crossover split to type a level value or frequency directly.

Tips

- About everything in the display can be modulated, even the cross-over splits. Simply drag and drop a modulation source to it!

Solo and mute

If you hover the mouse over the top of the display and there is more than one band, small solo and mute buttons in the left-top corner of each band will appear. The solo button lets you listen to a single band, while the mute button does the opposite and mutes the band, letting you hear all other bands. Of course, you can solo or mute multiple bands at the same time, just like it works with tracks in your DAW.

- Hold down the solo or mute button to solo or mute a band only temporarily, as long as the mouse button is pressed.

Solo and mute changes can be automated and are saved with the other parameters, so you can also use them for creative effects.

Band controls

FabFilter Saturn contains one set of controls to adjust the currently selected frequency bands. When only one band is used, it will be selected by default and the controls are automatically linked to that single band. Once you have multiple frequency bands available, the display highlights the level buttons of the selected bands and they are linked to all of them.

From left to right, the following parameters are available:

- The Enabled button lets you easily bypass the selected frequency bands. When bypassed, the dry input signal of the band is passed to the output straight away. Note that you can also solo and mute bands with the solo/mute buttons in the display.
- The Section Preset button lets you quickly save or restore the parameters for the selected bands. See Section presets.
- The Mix sets the combination of unprocessed (dry) signal and the processed/distorted signal for the band.
- The Feedback Amount knob sets the level of feedback for the band, which feds the processed audio back into the input of the band. The Feedback Frequency knob sets the ringing frequency of the feedback loop. You can simply compare this to the distance of a microphone which picks up the signal of an amplifier that outputs its own signal: the closer the mic gets to the speaker, the higher the ringing frequency.
- The Dynamics knob can be used to either gate or compress the band signal. Turning the knob to the right will add heavily pumping compression, while turning the knob to the left will introduce great all-purpose gating/expansion.
- The Style button selects the type of distortion applied to the signal. You can choose between:
  - Tube emulations: from clean, high quality tubes to juicy or even broken tube sound.
  - Tape emulation: subtle, warm or extreme tape saturation.
  - Amplifier emulation: from smooth and crunchy amplifiers to screaming power amps.
  - Smudge: This creative distortion algorithm smudges and stretches the audio in weird and unexpected ways. The Drive knob sets the amount of smudging/stretching.
  - Rectify: A crunchy combination of rectified sound, DC offset removal and soft clipping.
  - Destroy: A destructive combination of bit-crushing, sample rate reduction and clipping.
- The Drive knob is obviously one of the most important parameters, setting how much the clipping stage is driven with input signal. While increasing the drive, the output level will be adjusted automatically, to ensure that the overall sound level doesn't get out of control. With the pan ring, you can change the balance of the drive amount between the stereo channels (see also Mid/Side processing).
- The Tone controls set the bass, mid, treble and presence of the processed band signal, allowing you to tweak the harmonics generated by the distortion algorithm.
- The Level knob sets the output level of the selected bands between minus infinity and +36 dB. With the pan ring, you can change the relative levels of the left and right audio channels (or mid/side when Mid/Side processing is active).
- The Remove button deletes the currently selected bands.

Tips

- If there is more than one band, you can also easily adjust the level or drive settings of any band via the interactive multiband display.

- All continuous band parameters can be modulated, of course!
- When adjusting band settings with a continuous range (for example Drive, Level, Dynamics) for multiple selected bands at once, the relative differences between the bands will be preserved. When adjusting discrete parameters (HQ, Distortion Style), the parameter for all bands will be set to the same new value. If you double-click a knob to enter a value, this value will be applied to all bands directly as well.

Modulation

The real fun with FabFilter Saturn starts with the incredible modulation options. Almost any parameter can be modulated. These are called modulation targets. They can be modulated by any of the available modulation sources: XY controllers, XLFOs, envelope generators, envelope followers and MIDI sources. The modulation signal always flows via a modulation slot that allows you to vary the extent of modulation.

Use the Modulation button at the top to show or hide the entire modulation section, which consists of the following elements:

- Source selection bar - The source selection bar shows a schematic overview of all modulation sources at all times. Simply click on a source button here to scroll the source into view. The highlighted section of the bar shows the currently visible part, and it can be dragged to scroll the sources as well. The top segment of each source button lights up according to the modulation signal it is currently sending.
- Modulation slots - As said before: every modulation source uses a modulation slot to send its signal to the target. Saturn always groups all modulation slots above the source that they're connected to. Each slot displays the destination, graphically shows the amount, and you can quickly turn it on or off, or reverse its output.
- Modulation sources - The modulation sources are organized in a horizontally scrolling strip below the source selection bar. There are 5 different kinds of sources available: The XLFO can generate almost any waveform you can imagine and can be synchronized to the host tempo. The Envelope generator is of the usual ADSR kind and triggered by audio or MIDI. The Envelope follower will follow the loudness of the incoming audio or side-chain signal. The MIDI source transforms any incoming MIDI data into a modulation signal. Finally, the XY controller lets you modulate two targets using horizontal and vertical mouse movements.

To add a modulation source, click the + button in the source selection bar.

To delete a modulation source, click the remove button in the top right corner in the source interface. When a source is deleted, modulation slots that use that source will also be deleted automatically.

Drag-and-drop modulation slots

One of the best features of FabFilter Saturn is undoubtedly the ability to set up modulation connections with drag-and-drop. There is no need to search through long drop-down menus containing dozens of sources and targets or to find your way in cluttered and obscure matrix views. This simple method of making modulation connections makes sound designing become fun, easy and, above all: fast. So how does it work?

First, grab the source drag button that you would like to use as a modulation signal, for example XLFO 1. The moment you click on the source drag button, the interface dims and all modulation targets are highlighted.

The moment you start dragging, you will see a line from the source drag button to the icon that you are dragging. The cursor will snap to any available modulation target.

Now drop the icon on the highlighted knob of the parameter that you would like to modulate, for example the Drive knob of Band 2. That's all there is to it!

If you wish, you can also add a slot manually using the small plus button above each modulation source. You can also modulate slot level knobs, which makes incredibly complex modulation setups possible. To sort the slots click the + button in the source selection bar and select Sort Slots from the menu that pops up.

Once a slot has been added, you can edit it:

- Use the Level slider to adjust the amount of modulation. Like with knobs, hold down Shift for fine-tuning; hold down Alt to adjust all slot levels for the same source; Ctrl-click (Windows) or Command-click (OS X) to reset the level to the default value.
- To the left of the Level slider, you can invert the modulation signal with the +/- button.
- When you hover over the slider on the left an on/off button appears. Use this to temporarily disable the slot. On the right a menu is accessible that gives direct access to all available modulation targets.
- To delete the slot, click the Remove button to the right of the Level slider.

Our what-you-use-is-what-you-see interface makes complex programming very easy. Saturn uses dynamic slot highlighting to visualize all the sources that modulate a specific target. When a parameter is modulated a small modulation indicator "M" appears.

Click the M modulator indicator to highlight all slots that modulate this target. In the source selection bar the sources that modulate the target are also highlighted.

This feature makes programming so much more fun because it's easy to see what is happening inside a patch. To return to the normal interface click anywhere on the interface background or click the modulation indicator again.

When a modulation indicator appears in a band in the interactive multiband display or envelope generator, this means one or more parameters of that band or EG are modulated. When you click that indicator it will highlight all slots that modulate a target of the band or envelope generator.

XLFO

The XLFO is like a classic LFO but it can do so much more! It can also be used as a 16 step sequencer with an individual glide parameter for every step. The display shows the waveform that is used by the XLFO. Steps can be freely added or deleted to shape the funkiest of waveforms.

To add an XLFO as a modulation source, click the + button in the source selection bar and click New XLFO.

At the left of the XLFO interface, you find the global parameters that affect the entire waveform:

Frequency

The frequency knob sets the time it takes for 1 cycle of the waveform to complete. This knob is a modulation target, so you could let one XLFO modulate the frequency of another XLFO. The XLFO can be synchronized to the tempo of the plug-in host or set to run free. With the options at the top-right corner of the frequency button you can choose the different settings:

- Free running mode will allow values from 0.02 to 500 Hz, so the minimum cycle length is 0.002 seconds.
- When using any of the synchronized cycle lengths (16 to 1/64, measured in bars) the frequency knob changes into the Offset knob. It now acts like a cycle length multiplier and therefore is capable of changing the cycle length relative to the host tempo, from half to two times the host tempo. Click the dots around the knob to jump to certain predefined offsets for dotted and triplet synchronization. Note: the Offset parameter is not a modulation target, but you can modulate the Phase offset instead.

Balance

The outer ring of the frequency knob adjusts the time balance of the first and last halves of the step sequence. For example, when turned to the left, the first half of the wave form is generated more quickly than the last half.

Snap

This function makes it possible to use the XLFO as an arpeggiator. When you enable Snap, a small piano keyboard appears, the range of the XLFO turns into 2 octaves, and steps "snap" to notes on the piano keyboard. Now when you modulate a frequency parameter, turn the slot level to maximum, and the total amount of modulation will exactly correspond to 2 octaves.

Glide

The global Glide knob acts like an overall glide offset. The amount of glide determines the point within a step at which the XLFO starts to interpolate to the value of the next step. The global Glide value is added to the glide value for individual steps to arrive at the final glide value for each step. The final glide value is limited between 0 (no interpolation) and 1 (full interpolation). Because the global Glide value can range from -1 to 1 it can completely overrule the individual step glide values at the extreme settings. It is also a modulation target which allows for very cool effects.

Phase offset

In the step editor you can see a triangular shape. The vertical line of the shape indicates the beginning of each cycle. You can move this triangular shape, and thus change the beginning of a XLFO cycle. This phase offset is a modulation target, so when the XLFO frequency is set to 0, you can use another modulator to cycle through the different steps.

At the top right of the global settings, the Presets button provides access to the XLFO section presets. The Remove button deletes the XLFO source. By default, the XLFO starts with two steps that make a sine wave. You can customize this by overwriting the predefined Default section preset.

Editing Steps

You can shape the waveform of the XLFO in almost any way you want by editing the individual steps.

- Drag a step up or down to change the value for the step.
- Click a step to select it.
- Click next to a step to deselect all steps.
- Click the + button at the end of all steps to add a new step. The new step is added to the right of the selected step, or at the end of all steps.
- Click the - button at the end of all steps to remove the selected steps. If no steps are selected, the last step is removed.

If one or more steps are selected, the XLFO expands to show the step interface where the parameters for the selected steps can be edited:

- Random - The Random button enables random values for this step. If enabled, the XLFO will use a new random value for the step each time it encounters it. The display also changes to show that the value is chosen at random (see step 3 in the screen shot above).
- Value - The Value knob adjusts the value of step. This is the same as dragging the step up and down, except that with multiple selected steps, the value of all steps is set to the same value. In contrast, when you drag multiple selected steps, the relative distance is kept the same.

- Curve - The Curve button selects the curve that is used to interpolate to the next step when the final glide value is higher than 0: Linear, Sqr, Sqrt and Sine.
- Glide - The Glide knob sets the per-step glide value. This is combined with the global glide value to determine at which point the XLFO starts to interpolate towards the next step.

To start exploring the many sound shaping possibilities start with an XLFO that modulates a Drive knob or Crossover Frequency to make the sound change over time. You'll be amazed by the many possibilities. Have a look at the presets to see the XLFO in many different setups to get an idea of what it can do for you and start creating your own sequences!

Envelope generator

The envelope generator (EG) generates a traditional ADSR envelope. The envelope being the way in which the level changes with time and is controlled by the Attack, Decay, Sustain and Release parameters. Its function is to modulate a parameter over time, based the amplitude of the input signal.

To add an envelope generator as a modulation source, click the + button in the source selection bar and click New Envelope Generator.

The following EG parameters are:

Trigger

The EG can be triggered by the main input signal. Depending on the type and amplitude of the incoming signal you need to adjust the threshold for optimal functioning. Look at the top segment of the source button for the EG to see when it is in the triggered (Attack-Decay-Sustain) state.

Delay

The time it takes for the attack to start after the key is.

Attack

The Attack portion of the envelope is the time taken for the amplitude to reach maximum value. For percussive effects, the attack time should be as short as possible.

Decay

After the sound has reached its maximum level, it starts to decay until it reaches the Sustain level at a time set by the Decay Time setting.

Sustain

This is the level reached after the decay time. The EG will hold this level as long as a key is pressed. Note that this parameter specifies a volume level rather than a time period.

Hold

Once the key is released, the value will remain at the sustain level for a time set by the hold parameter.

Release

After the hold time the sound resumes its decay, this time at a new rate determined by the Release setting.

Tips

- At the top right of the EG interface, the Presets button provides access to the EG section presets. The Remove button deletes the envelope generator. You can customize the default EG settings (used when creating a new EG) by overwriting the predefined Default section preset.

Envelope follower

The envelope follower modulation source outputs an envelope signal based on the plug-in input or side-chain audio level. You can set the Attack and Release parameters to 'smooth out the bumps'.

To add an envelope follower as a modulation source, click the + button in the source selection bar and click New Envelope Follower.

The two buttons at the top of the EF source interface select which signal is used to trigger on: the main input signal or the signal from the side-chain input.

At the top right of the source interface, the Presets button provides access to the EF section presets. The Remove button deletes the envelope follower. You can customize the default EF settings (used when creating a new EF) by overwriting the predefined Default section preset.

XY Controller

The XY Controller makes for more tweaking fun. It's a classic, and we didn't dare to leave it out! It can control two parameters with one mouse movement. When browsing presets don't forget to listen to the sound mangling possibilities provided by these controllers.

To add an XY controller as a modulation source, click the + button in the source selection bar and click New XY Controller.

Because the XY controller has two "outputs", it also has two source drag buttons labeled X and Y. The slots for the XY controller are grouped in two rows, with the X-slots at the top. For example, in the screen shot above, the X axis controls the output panning, while the Y axis controls the level.

The Remove button deletes the XY Controller.

Mid/Side processing

The Channel Mode button in the bottom bar selects between normal Left/Right processing and Mid/Side processing.

Mid/side is a representation of stereo sound as the sum and difference of the two channels. The concept has its origin in stereo microphone techniques using two microphones (more about that here on Wikipedia) but also gives us many options to change a stereo audio signal.

When you set the Channel Mode to Mid/Side, the incoming stereo signal (normal left and right audio) will be converted to mid (mono information; audio that's equally present in both channels) and side (pure stereo information; audio only present in either the left or right channel). The mid information is processed by the internal left channel, and the side information by the right channel. Of course, after the internal audio processing, the mid/side signal is converted back to stereo again.

Any conventional stereo signal can be converted to Mid/Side stereo, and back again, with no loss of information. Using the Mid and Side channels to treat them differently gives many creative opportunities, providing tremendous amount of control over the stereo spread.

You can use mid/side processing in FabFilter Saturn to achieve various useful and creative effects. For example, with the Pan ring around the Drive knob, you can drive the stereo image of a sound while keeping the mono/mid signal intact. Anything is possible!

Input and output controls

Besides the MIDI learn and Channel Mode buttons, the bottom bar controls various input/output options and settings.

Auto Mute Feedback

The Auto Mute Feedback option reduces the feedback used in the distortion bands if there is no incoming audio signal. Depending on the amount of feedback and type of distortion processing, the plug-in can start to self-oscillate endlessly. The auto-mute feature enables you to use extreme feedback settings while ensuring that the continuous ringing will stop when you stop playback in your host.

HQ

The HQ (high quality) mode controls oversampling of the internal distortion algorithms. Adding distortion to a signal introduces digital aliasing effects, especially when applying a lot of drive. Enabling HQ mode will drastically reduce aliasing artifacts by oversampling the internal saturation section 8 times, at the cost of using more CPU power and introducing a very small latency. Note: The HQ setting is not changed when you skip presets, but you can save its initial setting by overwriting the default preset.

Audition

The audition switch (recognizable by its headphones icon) lets you listen to either the normal output signal (default setting), the input signal (bypassing the entire plug-in) or the side chain signal. When setting up side chaining in your host this is very useful to confirm that the correct side chain signal is routed to the plug-in.

Input

The input button shows the current input gain and lets you adjust it from -36 dB to +36 dB. To change the gain, simply drag the button up and down. For precise adjustments or to change the panning, click the input button once to open a pop-up window with the actual input/pan knobs. Click the button again to let the pop-up window disappear. The input and pan knobs are also modulation targets.

Output

The output button shows the current output gain, also adjustable from -36 dB to +36 dB. It works the same as the input button and is also a modulation target. Note that you can overdrive the filters by increasing the input gain and reducing the output gain at the same time.

Mix

You can use the mix button to mix some of the original (dry, unprocessed) input signal back into the output signal, reducing the amount of filtered (wet) signal. Like the input and output buttons, this is also a modulation target.

Undo and Redo

The Undo and Redo buttons at the top of the plug-in interface enable you to easily undo changes you made to the plug-in.

- The Undo button at the left undoes the last change. Every change to the plug-in, such as dragging a knob, or selecting a new preset, creates a new state in the undo history. The Undo button steps back through the history to restore the previous states of the plug-in.
- The Redo button cancels the last Undo command. It steps forward through the history until you are back at the most recent plug-in state.

If the plug-in parameters are changed without using the plug-in interface, for example with MIDI or automation, no new undo states are recorded.

The Undo and Redo buttons will disable themselves if there is nothing to undo or redo.

A/B

With the A/B feature in FabFilter Saturn, you can easily switch between two different states of the plug-in.

- The A/B button switches from A to B and back. Before switching, the current state of the plug-in is saved, so if you click this button twice, you are back at the first state. The button highlights the currently selected state (A or B).
- The Copy button copies the active state to the inactive state. This marks the current state of the plug-in and allows you to go back to it easily with the A/B button. After clicking Copy, the button disables itself to show that both states are equal, so there is nothing to copy anymore.

Presets

- To load a preset, click the preset button. The presets menu will appear with all available presets. Click a menu item to load that preset. The currently selected preset is highlighted with check marks.
- To explore the presets one by one, click on the little arrow buttons to the left and right of the main preset button. This will load the previous or next preset in the menu.

The preset button shows the name of the current preset. If you have changed the preset by adjusting one or more parameters, the name is dimmed to indicate that this is not the original preset anymore.

- To save the current setting as a preset, click the preset button, and then click Save As. A standard Save dialog will appear. Type a name for the new preset and click Save to finish.
- In the Save dialog, you can also rename and delete existing presets and create a new folder to store presets in. New folders will show up as new categories in the preset menu.

MIDI Learn

Controlling FabFilter Saturn's parameters directly with MIDI is very easy using the MIDI Learn feature. With MIDI Learn, you can associate any MIDI controller with any parameter.

Click the MIDI Learn button in the bottom bar to enter MIDI Learn mode. The interface dims and the parameters that can be controlled are highlighted. Each parameter has a small text balloon that displays the associated controller number. Now do the following to associate a controller number with a parameter:

13. Touch the control of the desired parameter in the interface that you wish to control. A red square will mark the chosen parameter.
14. Adjust the slider or knob on your MIDI keyboard or MIDI controller that you want to associate with that parameter.

That's it! The parameter will now be controlled with the MIDI controller. You can now go back to step 1 to associate a different parameter. Note that there is no warning when you associate a different knob with a controller number that is already used. It will just be replaced.

To exit MIDI Learn mode, click the MIDI Learn button again, or click Close at the top of the interface.

Click the small menu drop-down button next to the MIDI Learn button to access the MIDI Learn menu:

- Disable/Enable MIDI - This globally turns MIDI control of parameters on or off: useful in hosts that automatically send all MIDI events on a track to all effect plug-ins associated with that track as well.
- Clear - This submenu shows all parameter associations and lets you delete individual associations or clear all associations in one step.
- Revert - Reverts to the last saved MIDI mapping (or the state when the plug-in was started).
- Save - Saves the current MIDI mapping so Revert will go back to this state. The current mapping is automatically saved when closing the plug-in.

FabFilter Timeless2

Welcome to the wobbly world of one of the most versatile delay plug-ins: Fabfilter Timeless 2. At its heart there are two independent, programmable delay lines. The addition of a high quality filter section and incredible new modulation features will get you time-warped where no man has gone before.

All these controls provide an almost unbelievable array of sound manipulation possibilities, ranging from simple repeat echo to genuinely original sounds that you wouldn't expect from a delay plug-in.

Overview

FabFilter Timeless 2's interface is divided into multiple sections:

Presets, undo, A/B, help

The Undo, Redo, A/B and Copy buttons at the top of the plug-in interface enable you to undo your changes and switch between different states of the plug-in. With the preset buttons, you can easily browse through the vast library of factory presets or save your own settings so you can re-use them in other songs. The Help button provides access to the help file and other information and options.

Feedback and delays

This is where the magic begins. The delay time is controlled by a big knob and can be synchronized to the host tempo. Both delay lines have there own feedback and cross feedback knobs which determine the amount of repeats. See Delay lines.

Filter section

And in the filter section, the magic goes on! Our state-of-the-art multimode filters let you morph the delayed sounds, adding filtering effects ranging from gentle sweeps up to self-oscillating madness. See Filters.

Dry/wet level

Here you control the audio output gain. The dry (unprocessed) signal and the output of the delay lines have their own output volume knobs. (The input gain control is located at the far left of the plug-in.) See Input/Output stage.

Modulation button

The modulation button shows or hides the entire modulation section at the bottom of the interface. FabFilter Timeless 2 offers virtually unlimited modulation possibilities, but all this power might be a bit intimidating. That's why the modulation section is hidden by default, and you can look 'under the hood' when you want to tweak a preset or design your own.

Source selection bar

The source selection bar shows all modulation sources at a glance and lets you easily scroll around and create new sources. FabFilter Timeless 2 offers XLFO, Envelope Generator (EG), Envelope Follower (EF), MIDI and XY Controller sources. See also Modulation.

Modulation slots and sources

The bottom section contains the modulation sources. The modulation section in Timeless 2 is fully modular — but without the cables! We found a simple way to show you everything that is modulating, and what is modulated by what. Above each modulation source, the modulation slots show exactly what targets are modulated by this source and let you adjust the amount of modulation. You can very easily set up modulation connections with drag-and-drop. All in all, we think we made sound design easier and more fun!

Resize

The resize button in the lower-right corner lets you choose between normal and wide interface layouts. The wide layout eliminates scrolling in the top part of the interface and provides more space for the modulation sources at the bottom of the interface. Most hosts support dynamic resizing of the interface; otherwise just close and re-open the interface window.

What-you-use-is-what-you-see

Often an impressive feature list results in an impressively difficult-to-use interface full of controls for parameters you might never even use. For almost every plug-in developer one of the greatest challenges when making a complex full-featured plug-in is to design an interface that is easy to use. And we think we did it! FabFilter introduces a revolutionary new interface concept: What-you- use-is-what-you-see.

The idea is simple yet powerful. At all times, the interface only contains the modulation sources and slots that you are actually using. This results in an intuitive user interface that experienced producers and novices alike will embrace. You can easily create more modulation sources. Do you want another XLFO? Just add one! Do you want an envelope generator? Just add one and start modulating things! Of course there is a limit to the number of sources you can create, but in practice it feels like you can create as many sources as you will ever need.

To help you understand even the most complex presets, modulation slots are grouped with each source. Each component, knob or controller that is being modulated is marked with a little M button. Simply click the M to highlight the modulation source and slots responsible for the modulation. See also Modulation.

Another interface innovation are the filter buttons in the filter section. You can control the main filter parameters simply by dragging on the filter buttons, which makes for an uncluttered interface that is easy to overview.

Delay lines

The delay lines are the center of FabFilter Timeless 2. Of course, they cause a delay in the transmission of a signal passing through.

There is a wide range of effects possible with a digital delay: repeat echo, slap-back delay, chorus, vibrato, and resonant 'tunnel' echo.

There are two delay lines: one receiving input from the left channel, and the other from the right channel (except in Mid/Sidemode).

You control each delay line with the following parameters:

Delay time

Well, guess what... this sets the delay time! To be more precise: the time of the delay given to a signal passing through.

The delay time can be locked/synchronized to the tempo of your sequencer host. When this is activated using the curved switch the knob controls the sub-multiples of this tempo (we call this the Delay Offset instead of the

Delay Time). The small dots that appear around the knob make it easier to get precise and quick access to certain fractions that are related to your sequencer tempo.

When the delay time is not locked to your sequencer tempo it is possible to 'tap' the tempo of the delay by clicking on the number-display above or below the knob. The display will turn into an illuminated TAP button. The next time you click here the time between the clicks is calculated and used as delay time. Just tap it a few times to get some values you want to work with.

In case you want to use the exact same delay time for both delay lines, enable the Delay Link switch between the delay lines. This makes it easier to set up both delay lines with the same settings.

Delay pan

Pans the output of each delay line to the left or right channel.

Feedback

You can vary the feedback to produce more than one repeat from a single sound. All the feedback control does is to send some of the delayed output (after passing through the filters) back to the input so it gets delayed again; the more feedback, the more repeats. There are separate knobs for the left and right filter output for both delay lines.

When a signal coming out of a delay line is routed back into the other delay line this is called "cross-feedback" hence the names on the interface. Cross-feedback is used to mix different delay times and creates beautiful stereo effects. The amount of total feedback determines the number of audible repeats. Higher levels will have more repeats and above a certain level feedback will cause higher volumes at every cycle and thus create sonic mayhem! Be careful with your ears and speakers, and don't use too high feedback levels.

There is a convenient lock icon that makes it possible to set up feedback settings for both delay lines.

Feedback invert switch

Very interesting effects can be achieved when inverting the phase of one of the feedback signals. The effect of this is most noticeable on effects that use a very short delay time. By inverting the phase of the signal fed back to the input, it allows different harmonics to be accentuated by the filtering process, and so gives a choice of two types of tonal coloration, one usually sounding thinner than the other. On longer delay times it might alter the stereo perception of the sound.

Delay style

There are two different ways the digital delay can behave:

1. Tape which behaves like a classic tape delay. When the delay time is changed in positive direction i.e. the delay time gets shorter, you will hear a increase in pitch of the delayed signal. Conversely when the delay time is made longer you will hear a decrease in pitch of the delayed signal. This is the way analog delays sound and makes 'playing' the delay so much fun.
2. Stretch makes this plug-in simply unique. It means that no matter whether the delay time gets shorter or longer, the pitch will remain constant using granular techniques. This is NOT possible with an analog delay and we thought this to be a highly creative addition. Listen to some of the presets using this algorithm and you will hear what sonic possibilities this option has to offer.

Freeze

The Freeze button lets you freeze the sound that's currently in the delay lines. As soon as you activate freeze, the input to the delay line is cut off, so no new sounds will be stored. The delay lines will keep playing the current sound, which you can now filter continuously. Also, you can of course change the delay time which will also transform the sound in the buffers. This can really warp the sound and change it into something completely different! The Freeze option is not stored in presets because it really needs to be turned on and off dynamically.

The settings of all delay parameters can be stored as a section preset.

Tips

- By setting a delay time of between 30 and 100 ms and adding a little gentle modulation with no feedback, you get the classic chorus effect.
- At very short delay times (5 to 50 ms), increasing feedback will give a resonant cardboard tube or tunnel echo sound, the pitch of the resonance being set by the delay time. This effect is useful in creating new sounds or modifying existing ones beyond recognition; used with a synth, it can create the illusion of ring modulation or phase sync.
- Short delays of between 30 and 100 ms are used to create slap-back echo effects, which are quite effective on vocals and guitar.
- Delay times in excess of 100 ms will give you the familiar tape echo type of sound, and this is a valuable effect for warming up vocals and guitar.

Filters

FabFilter Timeless 2 comes with two high-quality filters, each with no less than eleven different sound characteristics. These multimode filters are based on our award winning filters first developed for FabFilter One. You can use them individually or combine filter characteristics to create your own sounds in any way imaginable. Both filters are stereo filters.

The filter buttons let you easily adjust the main filter parameters, simply by clicking and dragging on the button. As soon as you move the mouse cursor over a filter button, value displays will pop up to show the current values of the associated parameters.

- Click and then drag horizontally or vertically to change the filter frequency and peak parameters.
- To view all filter parameters, click the one of the filter buttons once. This will expand the filter section to show the complete filter interface. Click the filter button again to hide the interface. While the filter section is expanded, you can scroll the top section of Timeless' interface with the left and right scroll buttons at the far ends of the interface.
- The filter buttons have an on/off switch in the left top corner, to quickly enable or disable the filter.

We strongly suggest for you to try all these movements yourself, and you'll find it's a great aid in quickly setting up the filters in Timeless the way you like. The most important parameters are always available, and if you need access to all parameters, they are just a mouse click away.

Tip: You can turn off the parameter value displays for the filter buttons with the Show Component Displays option in the Help menu.

Filter routing

Above the filter buttons, the filter routing can be set. There are three different ways of configuring the filters in the audio signal path:

1. Serial will put both left and right channel of the delay lines first thru filter 1 and than thru filter 2.
2. Parallel: The output of delay line 1 into both filter 1 and 2, and the output of delay line 2 into both filter 1 and 2.
3. Per channel: delay lines and filters are working in 2 groups. Delay line 1 uses filter 1 and delay line 2 uses filter 2.

Filter parameters

By clicking on one of the filter buttons in the filter section, the filter section expands to show all filter parameters and the interactive filter display.

You control each filter in Timeless 2 with the following parameters:

Frequency

The filter frequency is adjustable over the entire audio range. The Frequency controls the center or cut-off frequency of the active filter and can be controlled in real time, either manually or via external devices.

Pan

The Pan ring around the Frequency knob lets you filter the left and right channels differently. It works as a stereo balance setting for the center frequency of the filter. For example, when you turn the Pan knob to the left, the left channel will be filtered with a lower center frequency, and the right channel will be filtered with a higher center frequency. You can use this to create various stereo filtering effects, especially in combination with modulation.

Peak

The Peak knob adjusts the resonance of the active filter. A little resonance will cause the filter to create warmer and more characteristic tones. At maximum resonance, the filter will self-oscillator with most filter characteristics. (The Auto Mute Self- Osc option in the bottom bar will help to keep this manageable. See Input/output options.)

Characteristic

FabFilter Timeless offers the possibility to choose between three different filter characteristics:

1. FabFilter One, the original filter characteristic taken from our award-winning FabFilter One synthesizer
2. Smooth, like the cream in your coffee
3. Raw, a filter with lots of overdrive and exhibits a character of its own
4. Hard, moderately distorting filter, with a nice clean whistle
5. Hollow, juicy moderate distortion with fairly much low-end self-oscillation
6. Extreme, for more wild sonic ideas
7. Gentle, a more smooth and clean general purpose characteristic
8. Tube, with a warmer sound and nice overdrive, great for synth sounds
9. Metal, with a rough, sharper sound and distortion
10. Easy Going, a softer version of the Tube filter
11. Clean, linear behavior with no clipping or distortion at all

Response

The response of each filter can be set to either Low Pass, High Pass, or Band Pass. In Low Pass mode, the filter will pass through frequencies lower than the center frequency. In High Pass mode, frequencies higher than the center frequency will be passed through. In Band Pass mode, only the frequencies around the cut-off frequency will be passed through.

Slope

The slope switch sets the steepness of the filter, which controls how aggressively the frequencies around the center frequency are filtered. You can choose between 12 dB/octave, 24 dB/octave or 48 dB/octave settings. For example, if the response is set to Low Pass, more high frequencies will remain at 12 dB/octave than at 48 dB/octave.

Enabled

The filters can each be switched on or off with the small buttons left of the characteristics drop-down menu in the filter section. While a filter is bypassed, it will look disabled, but the controls can still be used to adjust the filter.

The settings of the filter parameters can be stored as a section preset.

Interactive filter display

The interactive filter display gives an overview of the filter parameters and makes it very easy to adjust multiple filter parameters simultaneously. The vertical lines in the background represent a logarithmic scale that corresponds to the actual filter frequencies.

To open the filter display, click on one of the filter buttons.

- Drag a filter dot to adjust the Frequency and Peak parameters for that filter.
- Drag the link dot between filter 1 and 2 to adjust both filters simultaneously.

Tip: Of course, all changes made in the filter display can be automated!

Modulation

The real fun with Timeless 2 starts with the incredible modulation options. Almost any parameter can be modulated. These are called modulation targets. They can be modulated by any of the available modulation sources: XY controllers, XLFOs, envelope generators, envelope followers and MIDI sources. The modulation signal always flows via a modulation slot that allows you to vary the extent of modulation.

Use the Modulation button at the top to show or hide the entire modulation section, which consists of the following elements:

Source selection bar

The source selection bar shows a schematic overview of all modulation sources at all times. Simply click on a source button here to scroll the source into view. The highlighted section of the bar shows the currently visible part, and it can be dragged to scroll the sources as well. The top segment of each source button lights up according to the modulation signal it is currently sending.

Modulation slots

As said before: every modulation source uses a modulation slot to send its signal to the target. Timeless 2 always groups all modulation slots above the source that they're connected to. Each slot displays the destination, graphically shows the amount, and you can quickly turn it on or off, or reverse its output.

Modulation sources

The modulation sources are organized in a horizontally scrolling strip below the source selection bar. There are 5 different kinds of sources available: The XLFO can generate almost any waveform you can imagine and can be synchronized to the host tempo. The Envelope generator is of the usual ADSR kind and triggered by audio or MIDI. The Envelope follower will follow the loudness of the incoming audio or side-chain signal. The MIDI source transforms any incoming MIDI data into a modulation signal. Finally, the XY controller lets you modulate two targets using horizontal and vertical mouse movements.

To add a modulation source, click the + button in the source selection bar.

To delete a modulation source, click the remove button in the top right corner in the source interface. When a source is deleted, modulation slots that use that source will also be deleted automatically.

Drag-and-drop modulation slots

One of the best features of FabFilter Timeless 2 is undoubtedly the ability to set up modulation connections with drag-and-drop. There is no need to search through long drop-down menus containing dozens of sources and targets or to find your way in cluttered and obscure matrix views. This simple method of making modulation connections makes sound designing become fun, easy and, above all: fast. So how does it work?

Grab a source... ... drag it to a target... ... and drop it.

First, grab the source drag button that you would like to use as a modulation signal, for example XLFO 1. The moment you click on the source drag button, the interface dims and all modulation targets are highlighted.

The moment you start dragging, you will see a line from the source drag button to the icon that you are dragging. The cursor will snap to any available modulation target.

Now drop the icon on the highlighted knob of the parameter that you would like to modulate, for example the Delay 2 Time knob. That's all there is to it!

If you wish, you can also add a slot manually using the small plus button above each modulation source. You can also modulate slot level knobs, which makes incredibly complex modulation setups possible. To sort the slots click the + button in the source selection bar and select Sort Slots from the menu that pops up.

Once a slot has been added, you can edit it:

- Use the Level slider to adjust the amount of modulation. Like with knobs, hold down Shift for fine-tuning; hold down Alt to adjust all slot levels for the same source; Ctrl-click (Windows) or Command-click (OS X) to reset the level to the default value.
- To the left of the Level slider, you can invert the modulation signal with the +/- button.
- To delete the slot, click the Remove button to the right of the Level slider.

Our what-you-use-is-what-you-see interface makes complex programming very easy. Timeless 2 uses dynamic slot highlighting to visualize all the sources that modulate a specific target. When a parameter is modulated a small modulation indicator "M" appears.

Click the M modulator indicator to highlight all slots that modulate this target. In the source selection bar the sources that modulate the target are also highlighted.

This feature makes programming so much more fun because it's easy to see what is happening inside a patch. To return to the normal interface click anywhere on the interface background or click the Modulation Indicator again.

When a modulation indicator appears next to a filter button or envelope generator, this means one or more parameters are modulated. When you click that indicator it will highlight all slots that modulate a target of the component or envelope generator.

XLFO

The XLFO is like a classic LFO but it can do so much more! It can also be used as a 16 step sequencer with an individual glide parameter for every step. The display shows the waveform that is used by the XLFO. Steps can be freely added or deleted to shape the funkiest of waveforms. But there is more... This XLFO can also be used as arpeggiator! The values can be equally be distributed over 2 octaves, so when connecting it to any pitch parameter, it will function like an arpeggiator. We couldn't make it more funky!

To add an XLFO as a modulation source, click the + button in the source selection bar and click New XLFO.

At the left of the XLFO interface, you find the global parameters that affect the entire waveform:

Frequency

The frequency knob sets the time it takes for 1 cycle of the waveform to complete. This knob is a modulation target, so you could let one XLFO modulate the frequency of another XLFO. The XLFO can be synchronized to the tempo of the plug-in host or set to run free. With the options at the top-right corner of the frequency button you can choose the different settings:

- Free running mode will allow values from 0.0 to 500 Hz, so the minimum cycle length is 0.002 seconds.
- When using any of the synchronized cycle lengths (16 to 1/64, measured in bars) the frequency knob changes into the Offset knob. It now acts like a cycle length multiplier and therefore is capable of changing the cycle length relative to the host tempo, from half to two times the host tempo. Click the dots around the knob to jump to certain predefined offsets for dotted and triplet synchronization. Note: the Offset parameter is not a modulation target, but you can modulate the Phase offset instead.

Balance

The outer ring of the frequency knob adjusts the time balance of the first and last halves of the step sequence. For example, when turned to the left, the first half of the wave form is generated more quickly than the last half.

Snap

This function makes it possible to use the XLFO as an arpeggiator. When you enable Snap, a small piano keyboard appears, the range of the XLFO turns into 2 octaves, and steps "snap" to notes on the piano keyboard. Now when you modulate the filter frequency, turn the slot level to maximum, and the total amount of modulation will exactly correspond to 2 octaves. With filter frequency parameters, you will hear individual notes if used with high filter peak settings.

Glide

The global Glide knob acts like an overall glide offset. The amount of glide determines the point within a step at which the XLFO starts to interpolate to the value of the next step. The global Glide value is added to the glide value for individual steps to arrive at the final glide value for each step. The final glide value is limited between 0 (no interpolation) and 1 (full interpolation). Because the global Glide value can range from -1 to 1 it can completely overrule the individual step glide values at the extreme settings. It is also a modulation target which allows for very cool effects.

Phase offset

In the step editor you can see a triangular shape. The vertical line of the shape indicates the beginning of each cycle. You can move this triangular shape, and thus change the beginning of a XLFO cycle. This phase offset is a modulation target, so when the XLFO frequency is set to 0, you can use another modulator to cycle through the different steps.

Tip: Like with knobs, you can Ctrl/Command-click on the phase offset slider to reset it.

At the top right of the global settings, the Presets button provides access to the XLFO section presets. The Remove button deletes the XLFO source. By default, the XLFO starts with two steps that make a sine wave. You can customize this by overwriting the predefined Default section preset.

Editing Steps

You can shape the waveform of the XLFO in almost any way you want by editing the individual steps.

- Drag a step up or down to change the value for the step.
- Click a step to select it.
- Click next to a step to deselect all steps.
- Click the + button at the end of all steps to add a new step. The new step is added to the right of the selected step, or at the end of all steps.
- Click the - button at the end of all steps to remove the selected steps. If no steps are selected, the last step is removed.

If one or more steps are selected, the XLFO expands to show the step interface where the parameters for the selected steps can be edited:

Random

The Random button enables random values for this step. If enabled, the XLFO will use a new random value for the step each time it encounters it. The display also changes to show that the value is chosen at

Value

The Value knob adjusts the value of step. This is the same as dragging the step up and down, except that with multiple selected steps, the value of all steps is set to the same value. In contrast, when you drag multiple selected steps, the relative distance is kept the same.

Curve

The Curve button selects the curve that is used to interpolate to the next step when the final glide value is higher than 0: Linear, Sqr, Sqrt and Sine.

Glide

The Glide knob sets the per-step glide value. This is combined with the global glide value to determine at which point the XLFO starts to interpolate towards the next step.

To start exploring the many sound shaping possibilities start with a XLFO that modulates a Delay Time or Filter Frequency knob to make the sound change over time. You'll be amazed by the many possibilities. Have a look at the presets to see the XLFO in many different setups to get an idea of what it can do for you and start creating your own sequences to funkify your life!

Envelope generator

The envelope generator (EG) generates a traditional ADSR envelope. The envelope being the way in which the level changes with time and is controlled by the Attack, Decay, Sustain and Release parameters. Its function is to modulate a parameter over time, based the amplitude of the input signal.

To add an envelope generator as a modulation source, click the + button in the source selection bar and click New Envelope Generator.

The following EG parameters are:

Trigger

The EG can be triggered by the main input signal. Depending on the type and amplitude of the incoming signal you need to adjust the threshold for optimal functioning. Look at the top segment of the source button for the EG to see when it is in the triggered (Attack-Decay-Sustain) state.

Delay

The time it takes for the attack to start after the key is.

Attack

The Attack portion of the envelope is the time taken for the amplitude to reach maximum value. For percussive effects, the attack time should be as short as possible.

Decay

After the sound has reached its maximum level, it starts to decay until it reaches the Sustain level at a time set by the Decay Time setting.

Sustain

This is the level reached after the decay time. The EG will hold this level as long as a key is pressed. Note that this parameter specifies a volume level rather than a time period.

Hold

Once the key is released, the value will remain at the sustain level for a time set by the hold parameter.

Release

After the hold time the sound resumes its decay, this time at a new rate determined by the Release setting.

Tips

- At the top right of the EG interface, the Presets button provides access to the EG section presets. The Remove button deletes the envelope generator. You can customize the default EG settings (used when creating a new EG) by overwriting the predefined Default section preset.

Envelope follower

The envelope follower modulation source outputs an envelope signal based on the plug-in input or side-chain audio level. You can set the Attack and Release parameters to 'smooth out the bumps'.

To add an envelope follower as a modulation source, click the + button in the source selection bar and click New Envelope Follower.

The two buttons at the top of the EF source interface select which signal is used to trigger on: the main input signal or the signal from the side-chain input.

At the top right of the source interface, the Presets button provides access to the EF section presets. The Remove button deletes the envelope follower. You can customize the default EF settings (used when creating a new EF) by overwriting the predefined Default section preset.

XY Controller

The XY Controller makes for more tweaking fun. It's a classic, and we didn't dare to leave it out! It can control two parameters with one mouse movement. When browsing presets don't forget to listen to the sound mangling possibilities provided by these controllers.

To add an XY controller as a modulation source, click the + button in the source selection bar and click New XY Controller.

Because the XY controller has two "outputs", it also has two source drag buttons labeled X and Y. The slots for the XY controller are grouped in two rows, with the X-slots at the top. For example, in the screen shot above, the X axis controls the output panning, while the Y axis controls the level.

The Remove button deletes the XY Controller.

Input/output options

FabFilter Timeless 2 provides a rich variety of input and output options. In the main section interface, you will find basic volume controls:

Input Level

Located on the left of the main interface section is the main input volume control. This will set up the amount of signal going into the delay lines. Remember that overdriving the filter gives an more harmonically rich sound so feel free to experiment with higher levels.

Dry/Wet Level

On the right of the main interface section, the Dry Level and Wet Level knobs control the amount of dry (unprocessed) signal and the amount of delayed and filtered signal that is coming out of the plug-in. Panning settings are also included.

Dry Enabled

Located above the Dry Level knob, the Dry Enabled button is an easy way to stop dry signal coming through. Since a delay is often used as a send effect (inserted on a bus) you wouldn't want dry signal coming through in that case. This is a valuable feature when you are browsing presets which were not specifically designed for this kind of usage. The Dry Enabled setting cannot be saved in a preset and therefore will not be altered when browsing presets. However, you can save the current setting as the start-up default by clicking Options > Save As Default in the presets menu (useful for example if you always happen to use Timeless 2 as a send effect).

In the bar at the bottom of the plug-in, there are additional options:

Channel Mode

The Channel Mode option lets you choose between Left/Right and Mid/Side operation of Timeless 2. In Mid/Side mode, the incoming stereo signal is converted into Mid (mono) and Side parts, which are then routed to the separate delay lines. This enables you to delay the stereo part differently than the mono part of the signal, ensuring interesting and totally original stereo effects! At the end of the plug-in, after the Dry/Wet Level knobs, the signals are converted back to a normal stereo signal.

Note that Mid/Side mode also allows you to easily adjust the balance between Mid and Side using the panning rings around the Dry Level and Wet Level knobs.

Auto Mute Self-Oscillation

The Auto Mute Self-Osc option reduces the resonance of the filters if there is no incoming audio signal. Depending on the filter characteristic you can push the filters into self-oscillation with increasing peak values. The auto-mute feature will make higher peak settings possible while the filters will not be howling continuously when you stop playback in your host.

Audition

The audition switch (recognizable by its headphones icon) lets you listen to either the normal output signal (default setting), the input signal (bypassing the entire plug-in) or the side chain signal. When setting up side chaining in your host this is very useful to confirm that the correct side chain signal is routed to the plug-in.

Undo and Redo

The Undo and Redo buttons at the top of the plug-in interface enable you to easily undo changes you made to the plug-in.

- The Undo button at the left undoes the last change. Every change to the plug-in, such as dragging a knob, or selecting a new preset, creates a new state in the undo history. The Undo button steps back through the history to restore the previous states of the plug-in.
- The Redo button cancels the last Undo command. It steps forward through the history until you are back at the most recent plug-in state.

If the plug-in parameters are changed without using the plug-in interface, for example with MIDI or automation, no new undo states are recorded.

The Undo and Redo buttons will disable themselves if there is nothing to undo or redo.

A/B

With the A/B feature in FabFilter Timeless2, you can easily switch between two different states of the plug-in.

- The A/B button switches from A to B and back. Before switching, the current state of the plug-in is saved, so if you click this button twice, you are back at the first state. The button highlights the currently selected state (A or B).
- The Copy button copies the active state to the inactive state. This marks the current state of the plug-in and allows you to go back to it easily with the A/B button. After clicking Copy, the button disables itself to show that both states are equal, so there is nothing to copy anymore.

Presets

- To load a preset, click the preset button. The presets menu will appear with all available presets. Click a menu item to load that preset. The currently selected preset is highlighted with check marks.
- To explore the presets one by one, click on the little arrow buttons to the left and right of the main preset button. This will load the previous or next preset in the menu.

The preset button shows the name of the current preset. If you have changed the preset by adjusting one or more parameters, the name is dimmed to indicate that this is not the original preset anymore.

- To save the current setting as a preset, click the preset button, and then click Save As. A standard Save dialog will appear. Type a name for the new preset and click Save to finish.
- In the Save dialog, you can also rename and delete existing presets and create a new folder to store presets in. New folders will show up as new categories in the preset menu.

MIDI Learn

Controlling FabFilter Timeless2's parameters directly with MIDI is very easy using the MIDI Learn feature. With MIDI Learn, you can associate any MIDI controller with any parameter.

Click the MIDI Learn button in the bottom bar to enter MIDI Learn mode. The interface dims and the parameters that can be controlled are highlighted. Each parameter has a small text balloon that displays the associated controller number. Now do the following to associate a controller number with a parameter:

4. Touch the control of the desired parameter in the interface that you wish to control. A red square will mark the chosen parameter.
5. Adjust the slider or knob on your MIDI keyboard or MIDI controller that you want to associate with that parameter.

That's it! The parameter will now be controlled with the MIDI controller. You can now go back to step 1 to associate a different parameter. Note that there is no warning when you associate a different knob with a controller number that is already used. It will just be replaced.

To exit MIDI Learn mode, click the MIDI Learn button again, or click Close at the top of the interface.

Click the small menu drop-down button next to the MIDI Learn button to access the MIDI Learn menu:

- Disable/Enable MIDI - This globally turns MIDI control of parameters on or off: useful in hosts that automatically send all MIDI events on a track to all effect plug-ins associated with that track as well.
- Clear - This submenu shows all parameter associations and lets you delete individual associations or clear all associations in one step.
- Revert - Reverts to the last saved MIDI mapping (or the state when the plug-in was started).
- Save - Saves the current MIDI mapping so Revert will go back to this state. The current mapping is automatically saved when closing the plug-in.

FabFilter Volcano2

FabFilter Volcano has proven to be one of the few plug-ins that offer convincing high quality digital filtering with unique analog character. With tons of features, modulation options and a unique user interface, Volcano 2 is the absolute top of its class!

Volcano 2 is more than just an update of Volcano 1. The original idea of high quality filters being modulated with several sources is still present but Volcano 2 has been redesigned from the ground up. Now it is not only capable of high quality filtering effects, but it can even be used for phasing, flanging, chorus and many other cool effects!

Volcano's interface is divided into horizontal sections:

Presets, undo and A/B

The Undo, Redo, A/B and Copy buttons at the top of the plug-in interface enable you to undo your changes and switch between different states of the plug-in. With the preset buttons, you can easily browse through the factory presets or save your own settings so you can re-use them in other songs.

Interactive filter display

This is the graphic representation of the filter settings which allows you to drag all filters individually or simultaneously and control the filter settings with key combinations. See Interactive filter display.

Filter controls

At the bottom of the display, the filter controls provide full control over all filter parameters. The filters offer a choice of lowpass, high-pass and band-pass modes and come with 11 algorithms that offer different filter characteristics. Each filter has a special delay option with short delay times up to 30 ms, creating all kinds of stereo effects. At the left of the filter controls, the routing button selects the routing of the four filters.

Modulation button

The modulation button shows or hides the entire modulation section at the bottom of the interface. FabFilter Volcano 2 offers virtually unlimited modulation possibilities, but all this power might be a bit intimidating. That's why the modulation section is hidden by default, and you can look 'under the hood' when you want to tweak a preset or design your own.

Source selection bar

The source selection bar shows all modulation sources at a glance and lets you easily scroll around and create new sources. FabFilter Volcano 2 offers XLFO, Envelope Generator (EG), Envelope Follower (EF), MIDI and XY Controller sources. See also Modulation.

Modulation slots and sources

The bottom section contains the modulation sources. The modulation section in Volcano 2 is fully modular — but without the cables! We found a simple way to show you everything that is modulating, and what is modulated by what. Above each modulation source, the modulation slots show exactly what targets are modulated by this source and let you adjust the amount of modulation. You can very easily set up modulation connections with drag-and-drop. All in all, we think we made sound design easier and more fun!

Input/output/mix

The bottom bar contains the various output options and mix buttons. See Output controls.

Resize

The resize button in the lower-right corner lets you choose between normal and wide interface layouts. The wide layout provides more space for the modulation sources at the bottom of the interface. Most hosts support dynamic resizing of the interface; otherwise just close and re-open the interface window.

What-you-use-is-what-you-see

Most of the times an impressive feature list results in an impressively difficult-to-use interface, full of controls for parameters you might never even use.

For almost every plug-in developer one of the greatest challenges when making such a complex full-featured plug-in is to design an interface that is easy-to-use. And we think we did it! FabFilter introduces a revolutionary new interface concept: What-you-use-iswhat- you-see.

The idea is simple yet powerful. Do you want another filter? Just add one! Do you want an envelope follower? Just add one and start modulating things! At all times, the interface only contains the filters, modulation sources and slots that you are actually using. Of course there is a limit to the number of sources you can create, but in practice it feels like you can create as many sources as you will ever need.

To help you understand even the most complex presets, modulation slots are grouped with each source. Each component, knob or controller that is being modulated is marked with a little M button. Simply click the M to highlight the modulation source and slots responsible for the modulation. See also Modulation.

Filters

Volcano 2 comes with four multi-mode stereo filters, which can be routed in almost every possible way, even per-channel and midside.

Every filter can be switched between low-pass, high-pass, and band-pass responses with 12, 24 and 48 dB/octave slopes and a staggering amount of eleven different high-quality filter characteristics that define the unique sound and overdrive of the filter. They range from smooth with moderate overdrive to raw, self-oscillating and over-the-top! All characteristics have been tuned very carefully, using our state-of-the-art FabFilter filter technology.

One set of filter controls is shared among all filters, which shows the parameters of the active filter.

- To activate a filter, click one of the numbered filter buttons. The button for the currently active filter lights up and is connected visually with the filter controls section.
- To bypass the active filter, click its filter button again. The first click activates the filter; once it is active, the filter button enables and disables the filter. When a filter is bypassed, the filter controls will look disabled, but can still be used to adjust the filter.
- To add a filter, click the + button next to the last filter button. The newly added filter will copy most of its settings from the currently active filter.
- To remove a filter, activate it and then click the Remove button at the top right of the filter controls section. If there is only one filter left, you cannot remove it.

- Click the Presets button to save the current filter and routing settings, or to access previously saved section presets. Note that this affects all filters and the routing, not just the active filter. See also Section presets.

In the filters section, we applied our new what-you-use-is-what-you-see concept: you only get filter buttons for filters that you are actually using at the moment. Even the Routing button only contains the routing options available for the current number of filters!

When using L/R or M/S configuration, the amount of filters will always be 2 or 4. So when adding or deleting filters just one click will add or delete 2 filters automatically.

You can use the filters individually or combine filter characteristics to create your own sounds in any way imaginable. To experience the full potential of Volcano 2's filters, try it on some signals with rich harmonics (synth sounds, distorted guitar or complete mixes are good sources for filtering).

Filter parameters

Volcano 2 contains up to four independent multi-mode filters. You control the parameters of the active filter with the following settings:

Frequency

The filter frequency is adjustable over the entire audio range. The Frequency controls the center or cut-off frequency of the active filter and can be controlled in real time, either manually or via external devices.

Pan

The Pan ring around the Frequency knob lets you filter the left and right channels differently. It works as a stereo balance setting for the center frequency of the filter. For example, when you turn the Pan knob to the left, the left channel will be filtered with a lower center frequency, and the right channel will be filtered with a higher center frequency. You can use this to create various stereo filtering effects, especially in combination with modulation.

Peak

The Peak knob adjusts the resonance of the active filter. A little resonance will cause the filter to create warmer and more characteristic tones. At maximum resonance, the filter will self-oscillator with most filter characteristics. (The Auto Mute Self- Osc option in the bottom bar will help to keep this manageable. See Output controls.)

Characteristic

FabFilter Volcano 2 lets you choose between 11 different filter characteristics:

1. FabFilter One, the original filter characteristic taken from our award-winning FabFilter One synthesizer
2. Smooth, like the cream in your coffee

3. Raw, a filter with lots of overdrive and exhibits a character of its own. Great for distortion guitar sounds
4. Hard, moderately distorting filter, with a nice clean whistle
5. Hollow, juicy moderate distortion with fairly much low-end self-oscillation
6. Extreme, for more wild sonic ideas
7. Gentle, a more smooth and clean general purpose characteristic
8. Tube, with a warmer sound and nice overdrive, great for synth sounds
9. Metal, with a rough, sharper sound and distortion
10. Easy Going, a softer version of the Tube filter
11. Clean, linear behavior with no clipping distortion at all

Response

The response of each filter can be set to either Low Pass, High Pass, or Band Pass. In Low Pass mode, the filter will pass through frequencies lower than the center frequency. In High Pass mode, frequencies higher than the center frequency will be passed through. In Band Pass mode, only the frequencies around the cut-off frequency will be passed through.

Slope

The slope switch sets the steepness of the filter, which controls how aggressively the frequencies around the center frequency are filtered. You can choose between 12 dB/octave, 24 dB/octave or 48 dB/octave settings. For example, if the response is set to Low Pass, more high frequencies will be passed above the cutoff frequency using at 12 dB/octave than at 48 dB/octave. But let your ears decide! Just listen to the sound as you move the filter around and see if you like it...

Delay

Each filter has an delay function that will, well yes, delay the sound passing through. This feature will add some more wobblyness to your sounds! For example the creation of comb-filter effects (chorus/flanging). These effects occur when 2 or more signals are added together while the delay time is changing of at least 1 of the signals. You can set it up by using an XLFO sending a simple sine wave to modulate the delay time of a filter. Use slow XLFO rates to create the classic flanging effect. In this case the Stereo configuration works best.

Or how about some crazy stereo effects using the "Haas" effect. Mr. Haas found out that time differences are very important for your stereo perception. So when the left channel is delayed we will hear it coming from the right speaker.

To set this up start with 2 filters and select the L/R configuration with the routing button. The schematic diagram will show us the filter 1 is used for the left channel and filter 2 is used for the right channel. If you raise the delay of filter 1 it will sound like the sound is panned to the right purely based on delay times, not volume. Now reset the delay time (both filters) to 0 and take a new XLFO to modulate the delay time of both filters and inverse 1 of the Modulation Slot. Now you will hear the stereo image move from left to right. Sweet innit! From here you can add more filters with different settings and pan them differently and it will create some truly blissful stereo effects.

The delay offers so many extra sound design opportunities that we strongly suggest that you take some time to experiment with it.

Tip: When you make changes to filter parameters that are modulation targets (cut-off frequency, peak, pan and delay), the modulation slots that use that target are automatically shown. You can return to view all connections by using the Show All Slots button that appears on the left in the modulation slots bar.

Interactive filter display

The interactive filter display gives an overview of the filter parameters and makes it very easy to adjust multiple filter parameters simultaneously. The vertical lines in the background represent a logarithmic scale that corresponds to the actual filter frequencies.

- Drag a filter dot to adjust the Frequency and Peak parameters for that filter. The active filter will have a light colored ring around its filter dot in the display (filter 2 in the screen shot above).
- Drag the link dot (between filter 1 and 3 in the screen shot above) to adjust all linked filters simultaneously. To link filters, click on the link buttons that appear when the mouse is just above the filter buttons. For example, you can set them up as 4 resonant band-pass filters and sweep the cutoffs simultaneously. This configuration will give you access to all manner of 'vocal' sounds, as well as even more dramatic formant-based timbres.

Tip: Of course, all changes made in the filter display can be automated!

Routing

Volcano 2 lets you route the four available filters in almost any way you can think of. There are 3 basic routing configurations: Stereo, L/R or M/S. By clicking the main routing button, the routing possibilities are presented for each configuration depending on the amount of filters used.

The three buttons at the right of the routing button switch between the main routing configurations:

- Stereo - In this default mode, both the left and the right channel pass through all filters. You can use the Pan rings to introduce differences in cutoff frequencies for each output channel and thus create beautiful stereo effects.
- Left/Right - In this case the left and right channel will go through different filters. This allows for more extreme settings and stereo effects. If there are two filters, the left channel will go only through filter 1, and

- Mid/Side - This is the icing on the cake, never seen before in a filter plug-in. Now the signal is split into a Mid and a Side channel using the sum and difference of the left and right channels. Each signal is sent through different filters (as in L/R mode) after which it is reconstructed into the normal left and right channels. The Mid/Side concept has its origin in stereo microphone techniques using two microphones but also gives us many options to process a stereo audio signal.

Modulation

The real fun with Volcano 2 starts with the incredible modulation options. Almost any parameter can be modulated. These are called modulation targets. They can be modulated by any of the available modulation sources: XY controllers, XLFOs, envelope generators, envelope followers and MIDI sources. The modulation signal always flows via a modulation slot that allows you to vary the extent of modulation.

Use the Modulation button at the top to show or hide the entire modulation section, which consists of the following elements:

Source selection bar

The source selection bar shows a schematic overview of all modulation sources at all times. Simply click on a source button here to scroll the source into view. The highlighted section of the bar shows the currently visible part, and it can be dragged to scroll the sources as well. The top segment of each source button lights up according to the modulation signal it is currently sending.

Modulation slots

As said before: every modulation source uses a modulation slot to send its signal to the target. Volcano 2 always groups all modulation slots above the source that they're connected to. Each slot displays the destination, graphically shows the amount, and you can quickly turn it on or off, or reverse its output.

Modulation sources

The modulation sources are organized in a horizontally scrolling strip below the source selection bar. There are 5 different kinds of sources available: The XLFO can generate almost any waveform you can imagine and can be synchronized to the host tempo. The Envelope generator is of the usual ADSR kind and triggered by audio or MIDI. The Envelope follower will follow the loudness of the incoming audio or side-chain signal. The MIDI source transforms any incoming MIDI data into a modulation signal. Finally, the XY controller lets you modulate two targets using horizontal and vertical mouse movements.

- To add a modulation source, click the + button in the source selection bar.
- To delete a modulation source, click the remove button in the top right corner in the source interface. When a source is deleted, modulation slots that use that source will also be deleted automatically.

Drag-and-drop modulation slots

One of the best features of FabFilter Volcano 2 is undoubtedly the ability to set up modulation connections with drag-and-drop.

There is no need to search through long drop-down menus containing dozens of sources and targets or to find your way in cluttered and obscure matrix views. This simple method of making modulation connections makes sound designing become fun, easy and, above all: fast. So how does it work?

Grab a source... ... drag it to a target... ... and drop it.

First, grab the source drag button that you would like to use as a modulation signal, for example XLFO 3. The moment you click on the source drag button, the interface dims and all modulation targets are highlighted.

The moment you start dragging, you will see a line from the source drag button to the icon that you are dragging. The cursor will snap to any available modulation target.

Now drop the icon on the highlighted knob of the parameter that you would like to modulate, for example the Filter 1 Cut-off knob. That's all there is to it!

If you wish, you can also add a slot manually using the small plus button above each modulation source. You can also modulate slot level knobs, which makes incredibly complex modulation setups possible. To sort the slots click the + button in the source selection bar and select Sort Slots from the menu that pops up.

Once a slot has been added, you can edit it:

- Use the Level slider to adjust the amount of modulation.
- To the left of the Level slider, you can invert the modulation signal with the +/- button.
- When you hover over the slider on the left an on/off button appears. Use this to temporarily disable the slot. On the right a menu is accessible that gives direct access to all available modulation targets.
- To delete the slot, click the Remove button to the right of the Level slider.

Our what-you-use-is-what-you-see interface makes complex programming very easy. Volcano 2 uses dynamic slot highlighting to visualize all the sources that modulate a specific target. When a parameter is modulated a small modulation indicator "M" appears.

Click the M modulator indicator to highlight all slots that modulate this target. In the source selection bar the sources that modulate the target are also highlighted.

This feature makes programming so much more fun because it's easy to see what is happening inside a patch. To return to the normal interface click anywhere on the interface background or click the Modulation Indicator again.

When a modulation indicator appears next to a filter button or envelope generator, this means one or more parameters are modulated. When you click that indicator it will highlight all slots that modulate a target of the component or envelope generator.

XLFO

The XLFO is like a classic LFO but it can do so much more! It can also be used as a 16 step sequencer with an individual glide parameter for every step. The display shows the waveform that is used by the XLFO. Steps can be freely added or deleted to shape the funkiest of waveforms. But there is more… This XLFO can also be used as arpeggiator! The values can be equally be distributed over 2 octaves, so when connecting it to any pitch parameter, it will function like an arpeggiator. We couldn't make it more funky!

To add an XLFO as a modulation source, click the + button in the source selection bar and click New XLFO.

At the left of the XLFO interface, you find the global parameters that affect the entire waveform:

Frequency

The frequency knob sets the time it takes for 1 cycle of the waveform to complete. This knob is a modulation target, so you could let one XLFO modulate the frequency of another XLFO. The XLFO can be synchronized to the tempo of the plug-in host or set to run free. With the options at the top-right corner of the frequency button you can choose the different settings:

- Free running mode will allow values from 0.0 to 500 Hz, so the minimum cycle length is 0.002 seconds.
- When using any of the synchronized cycle lengths (16 to 1/64, measured in bars) the frequency knob changes into the Offset knob. It now acts like a cycle length multiplier and therefore is capable of changing the cycle length relative to the host tempo, from half to two times the host tempo. Click the dots around the knob to jump to certain predefined offsets for dotted and triplet synchronization.

Balance

The outer ring of the frequency knob adjusts the time balance of the first and last halves of the step sequence. For example, when turned to the left, the first half of the wave form is generated more quickly than the last half.

Snap

This function makes it possible to use the XLFO as an arpeggiator. When you enable Snap, a small piano keyboard appears, the range of the XLFO turns into 2 octaves, and steps "snap" to notes on the piano keyboard. Now when you modulate the filter frequency, turn the slot level to maximum, and the total amount of modulation will exactly correspond to 2 octaves. With filter frequency parameters, you will hear individual notes if used with high filter peak settings.

Glide

The global Glide knob acts like an overall glide offset. The amount of glide determines the point within a step at which the XLFO starts to interpolate to the value of the next step. The global Glide value is added to the glide value for individual steps to arrive at the final glide value for each step. The final glide value is limited between 0 (no interpolation) and 1 (full interpolation). Because the global Glide value can range from -1 to 1 it can completely overrule the individual step glide values at the extreme settings. It is also a modulation target which allows for very cool effects.

Phase offset

In the step editor you can see a triangular shape. The vertical line of the shape indicates the beginning of each cycle. You can move this triangular shape, and thus change the beginning of a XLFO cycle. This phase offset is a modulation target, so when the XLFO frequency is set to 0, you can use another modulator to cycle through the different steps.

At the top right of the global settings, the Presets button provides access to the XLFO section presets. The Remove button deletes the XLFO source. By default, the XLFO starts with two steps that make a sine wave. You can customize this by overwriting the predefined Default section preset.

Editing Steps

You can shape the waveform of the XLFO in almost any way you want by editing the individual steps.

- Drag a step up or down to change the value for the step.
- Click a step to select it.
- Click next to a step to deselect all steps.
- Click the + button at the end of all steps to add a new step. The new step is added to the right of the selected step, or at the end of all steps.
- Click the - button at the end of all steps to remove the selected steps. If no steps are selected, the last step is removed.

If one or more steps are selected, the XLFO expands to show the step interface where the parameters for the selected steps can be edited:

- Random - The Random button enables random values for this step. If enabled, the XLFO will use a new random value for the step each time it encounters it. The display also changes to show that the value is chosen at random.

- Value - The Value knob adjusts the value of step. This is the same as dragging the step up and down, except that with multiple selected steps, the value of all steps is set to the same value. In contrast, when you drag multiple selected steps, the relative distance is kept the same.
- Curve - The Curve button selects the curve that is used to interpolate to the next step when the final glide value is higher than 0: Linear, Sqr, Sqrt and Sine.
- Glide - The Glide knob sets the per-step glide value. This is combined with the global glide value to determine at which point the XLFO starts to interpolate towards the next step.

To start exploring the many sound shaping possibilities start with a XLFO that modulates a Delay Time or Filter Frequency knob to make the sound change over time. You'll be amazed by the many possibilities. Have a look at the presets to see the XLFO in many different setups to get an idea of what it can do for you and start creating your own sequences to funkify your life!

Envelope generator

The envelope generator (EG) generates a traditional ADSR envelope. The envelope being the way in which the level changes with time and is controlled by the Attack, Decay, Sustain and Release parameters. Its function is to modulate a parameter over time, based the amplitude of the input signal.

To add an envelope generator as a modulation source, click the + button in the source selection bar and click New Envelope Generator.

The following EG parameters are:

Trigger

The EG can be triggered by the main input signal. Depending on the type and amplitude of the incoming signal you need to adjust the threshold for optimal functioning. Look at the top segment of the source button for the EG to see when it is in the triggered (Attack-Decay-Sustain) state.

Delay

The time it takes for the attack to start after the key is.

Attack

The Attack portion of the envelope is the time taken for the amplitude to reach maximum value. For percussive effects, the attack time should be as short as possible.

Decay

After the sound has reached its maximum level, it starts to decay until it reaches the Sustain level at a time set by the Decay Time setting.

Sustain

This is the level reached after the decay time. The EG will hold this level as long as a key is pressed. Note that this parameter specifies a volume level rather than a time period.

Hold

Once the key is released, the value will remain at the sustain level for a time set by the hold parameter.

Release

After the hold time the sound resumes its decay, this time at a new rate determined by the Release setting.

Tips

- At the top right of the EG interface, the Presets button provides access to the EG section presets. The Remove button deletes the envelope generator. You can customize the default EG settings (used when creating a new EG) by overwriting the predefined Default section preset.

Envelope follower

The envelope follower modulation source outputs an envelope signal based on the plug-in input or side-chain audio level. You can set the Attack and Release parameters to 'smooth out the bumps'.

To add an envelope follower as a modulation source, click the + button in the source selection bar and click New Envelope Follower.

The two buttons at the top of the EF source interface select which signal is used to trigger on: the main input signal or the signal from the side-chain input.

At the top right of the source interface, the Presets button provides access to the EF section presets. The Remove button deletes the envelope follower. You can customize the default EF settings (used when creating a new EF) by overwriting the predefined Default section preset.

XY Controller

The XY Controller makes for more tweaking fun. It's a classic, and we didn't dare to leave it out! It can control two parameters with one mouse movement. When browsing presets don't forget to listen to the sound mangling possibilities provided by these controllers.

To add an XY controller as a modulation source, click the + button in the source selection bar and click New XY Controller.

Because the XY controller has two "outputs", it also has two source drag buttons labeled X and Y. The slots for the XY controller are grouped in two rows, with the X-slots at the top. For example, in the screen shot above, the X axis controls the output panning, while the Y axis controls the level.

The Remove button deletes the XY Controller.

Output controls

The bottom bar controls various options and settings for the output signal of Volcano 2.

Auto Mute Self-Osc

The Auto Mute Self-Osc option reduces the resonance of the filters if there is no incoming audio signal. Depending on the filter characteristic you can push the filter into self-oscillation with increasing peak values. The auto-mute feature will make higher peak settings possible while the filters will not be howling continuously when you stop playback in your host.

Audition

The audition switch (recognizable by its headphones icon) lets you listen to either the normal output signal (default setting), the input signal (bypassing the entire plug-in) or the side chain signal. When setting up side chaining in your host this is very useful to confirm that the correct side chain signal is routed to the plug-in.

Input

The input button shows the current input gain and lets you adjust it from -36 dB to +36 dB. To change the gain, simply drag the button up and down. For precise adjustments or to change the panning, click the input button once to open a pop-up window with the actual input/pan knobs. Click the button again to let the pop-up window disappear. The input and pan knobs are also modulation targets.

Output

The output button shows the current output gain, also adjustable from -36 dB to +36 dB. It works the same as the input button and is also a modulation target. Note that you can overdrive the filters by increasing the input gain and reducing the output gain at the same time.

Mix

You can use the mix button to mix some of the original (dry, unprocessed) input signal back into the output signal, reducing the amount of filtered (wet) signal. Like the input and output buttons, this is also a modulation target.

Undo and Redo

The Undo and Redo buttons at the top of the plug-in interface enable you to easily undo changes you made to the plug-in.

- The Undo button at the left undoes the last change. Every change to the plug-in, such as dragging a knob, or selecting a new preset, creates a new state in the undo history. The Undo button steps back through the history to restore the previous states of the plug-in.
- The Redo button cancels the last Undo command. It steps forward through the history until you are back at the most recent plug-in state.

If the plug-in parameters are changed without using the plug-in interface, for example with MIDI or automation, no new undo states are recorded.

The Undo and Redo buttons will disable themselves if there is nothing to undo or redo.

A/B

With the A/B feature in FabFilter Volcano2, you can easily switch between two different states of the plug-in.

- The A/B button switches from A to B and back. Before switching, the current state of the plug-in is saved, so if you click this button twice, you are back at the first state. The button highlights the currently selected state (A or B).
- The Copy button copies the active state to the inactive state. This marks the current state of the plug-in and allows you to go back to it easily with the A/B button. After clicking Copy, the button disables itself to show that both states are equal, so there is nothing to copy anymore.

Presets

- To load a preset, click the preset button. The presets menu will appear with all available presets. Click a menu item to load that preset. The currently selected preset is highlighted with check marks.
- To explore the presets one by one, click on the little arrow buttons to the left and right of the main preset button. This will load the previous or next preset in the menu.

The preset button shows the name of the current preset. If you have changed the preset by adjusting one or more parameters, the name is dimmed to indicate that this is not the original preset anymore.

- To save the current setting as a preset, click the preset button, and then click Save As. A standard Save dialog will appear. Type a name for the new preset and click Save to finish.
- In the Save dialog, you can also rename and delete existing presets and create a new folder to store presets in. New folders will show up as new categories in the preset menu.

MIDI Learn

Controlling FabFilter Volcano2's parameters directly with MIDI is very easy using the MIDI Learn feature. With MIDI Learn, you can associate any MIDI controller with any parameter.

Click the MIDI Learn button in the bottom bar to enter MIDI Learn mode. The interface dims and the parameters that can be controlled are highlighted. Each parameter has a small text balloon that displays the associated controller number. Now do the following to associate a controller number with a parameter:

1. Touch the control of the desired parameter in the interface that you wish to control. A red square will mark the chosen parameter.
2. Adjust the slider or knob on your MIDI keyboard or MIDI controller that you want to associate with that parameter.

That's it! The parameter will now be controlled with the MIDI controller. You can now go back to step 1 to associate a different parameter. Note that there is no warning when you associate a different knob with a controller number that is already used. It will just be replaced.

To exit MIDI Learn mode, click the MIDI Learn button again, or click Close at the top of the interface.

Click the small menu drop-down button next to the MIDI Learn button to access the MIDI Learn menu:

- Disable/Enable MIDI - This globally turns MIDI control of parameters on or off: useful in hosts that automatically send all MIDI events on a track to all effect plug-ins associated with that track as well.
- Clear - This submenu shows all parameter associations and lets you delete individual associations or clear all associations in one step.
- Revert - Reverts to the last saved MIDI mapping (or the state when the plug-in was started).
- Save - Saves the current MIDI mapping so Revert will go back to this state. The current mapping is automatically saved when closing the plug-in.

ReTune

Mu Technologies' ReTune processor performs real-time monophonic pitch correction and harmonization. Use it to fix a vocal part with inconsistent pitch, or create additional harmony parts from just a single track.

Impact - Adjusts the tuning intensity.

Tune Tolerance - Adjusts the threshold at which tuning is enacted, in percentage of a semitone. This is the distance away from the pitch center the audio must be before it is affected.

Speed - Adjusts the amount of time, in milliseconds, that will elapse before the audio is affected after it has crossed the Tune Tolerance threshold.

Shift - Transposes the signal by the designated number of semitones. When 'Q' is enabled, the amount of transposition is quantized to discrete semitone steps.

Gender - performs a formant shift to adjust between male and female timbres.

Humanize – Adds random variance to the pitch-shifted audio, as a percentage.

Dry (M/S) - Mute, solo, and adjust the level of the original audio.

Wet (M/S) - Mute, solo, and adjust the level of the affected audio.

Range

- Lower - Sets the lower frequency boundary (in Hz) above which audio will be affected.
- Size - Adjust the window size (in semitones) within which audio is affected.

Invert - Inverts the affected audio signal.

Key - Determines the root note of the scale to which audio is tuned.

Pitch - Adjust the tuning reference (in Hz) to which the audio is tuned.

Scale - Determines the musical scale to which the audio is tuned.

OverLoud THM

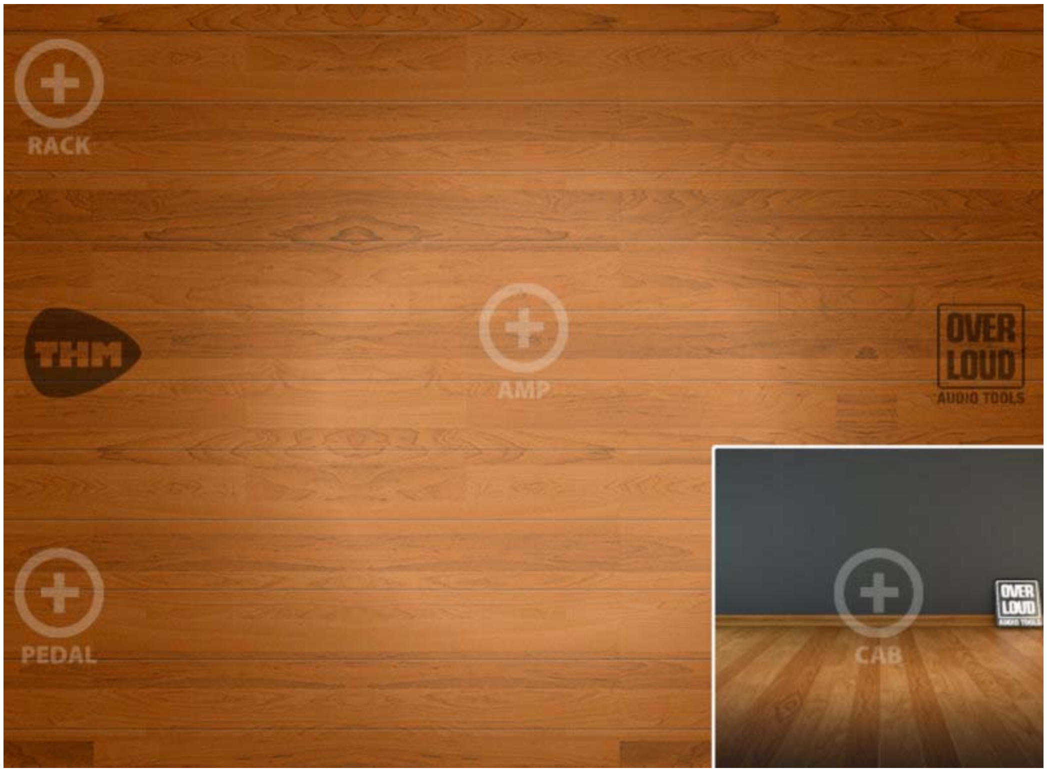
THM brings a complete guitar rig on your iPad. Build your dream effect chain choosing the modules from a gorgeous collection of the best vintage and contemporary instruments.

Start by playing the factory presets and then go on with your own adjustments.

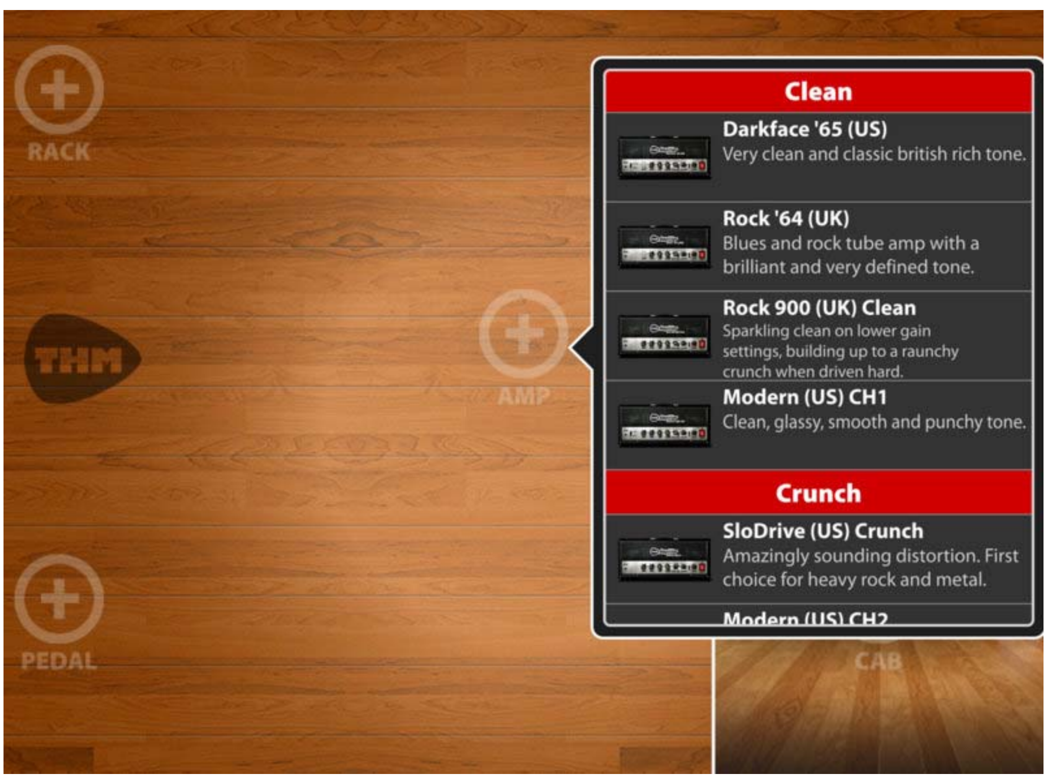
Everything in THM is extremely intuitive and easy to do, and works just like you would in reality, with no compromise on the sound quality. The audio engine has been optimized for the iPad allowing you to play more instances of THM in total mobile freedom.

MAIN INTERFACE

The main interface of THM has 4 areas: PEDAL, AMP, CAB, and RACK for pedal effects, amplifier, cabinet and rack modules respectively.



You can add new modules by touching the '+' button of the corresponding area.



For example touch '+ AMP' to add an amplifier and choose one item from the list that will pop up. The amplifier will take place into the current THM setup.

You can proceed in the same way to add all modules that you need to build your own THM setup.

Here is an example of how THM might look after building a complete guitar rig:

Pedal and Rack effects get arranged in scrollable horizontal lists. You can then have several modules linked together. If you want to move one module one step left or right, or either if you want to remove it, you can make the module enter the Edit Mode by keeping a long touch on it (choose an area of the module which is free from knobs, switches or the like).

When a module is in Edit Mode it shows some additional buttons around its shape (which buttons are present depends on the position of the module across the module list and on the fact that there are or not other modules next to it).

The red button with a white cross in the top left corner is to remove the module. The two grey buttons with white arrows move the modules one step left or right respectively.

The blue button with a white '+' on the bottom left corner inserts a new module just before the edited module.

To make a module exit Edit Mode, just touch the THM interface somewhere out of the module list.

CABINET

The cabinet area of THM includes two modules: the cabinet itself and the microphone. As for any other module, you can change cabinet and microphone models by double-tapping either of them.

Cabinet parameters

Touching the cabinet you can see the cabinet parameter bar.

These four buttons respectively control the following cabinet properties (from left to right):

- ReSPiRe (Real Sound Pressure Response)
- HPF (high pass filter)
- LPF (low pass filter)
- PHASE (reverse phase)

Microphone alignment

Touching the microphone you can change its alignment: in-axis, off-axis and far.

ABOUT THM

You can see some information about THM by clicking the THM logo.

THM PRESETS

THM integrates a complete preset manager which greatly simplifies your daily work allowing you to start scrolling the factory presets and then to go on by saving your own presets.

Presets are organized into banks. Each bank contains 128 presets numbered from 1 to 128.

Banks and presets have names. You can use those names to organize your presets into banks following some kind of logic. For example you can have banks for Jazz, Rock, or Acoustic presets.

Before going into the detailed description of the Preset panel of THM, let's take a look at the other controls on the bottom bar.

The first button, Preset, opens the Preset panel that we will see later in detail.

THM refers to the current preset (the last preset loaded) with the display which comes next. The display shows the bank number, the preset number and the preset name with the following syntax: [bank#] : [preset#] [preset name]. So if you see "1:4 Guitar Hero", then you are playing the preset 4 from bank 1, named "Guitar Hero".

Next you see the Save button. This button saves the changes you have done to the current preset. When you load a preset the button is black. As soon as you make a change, it turns red to warn that there are unsaved changes. When you touch Save, the changes get saved and the button goes back to black.

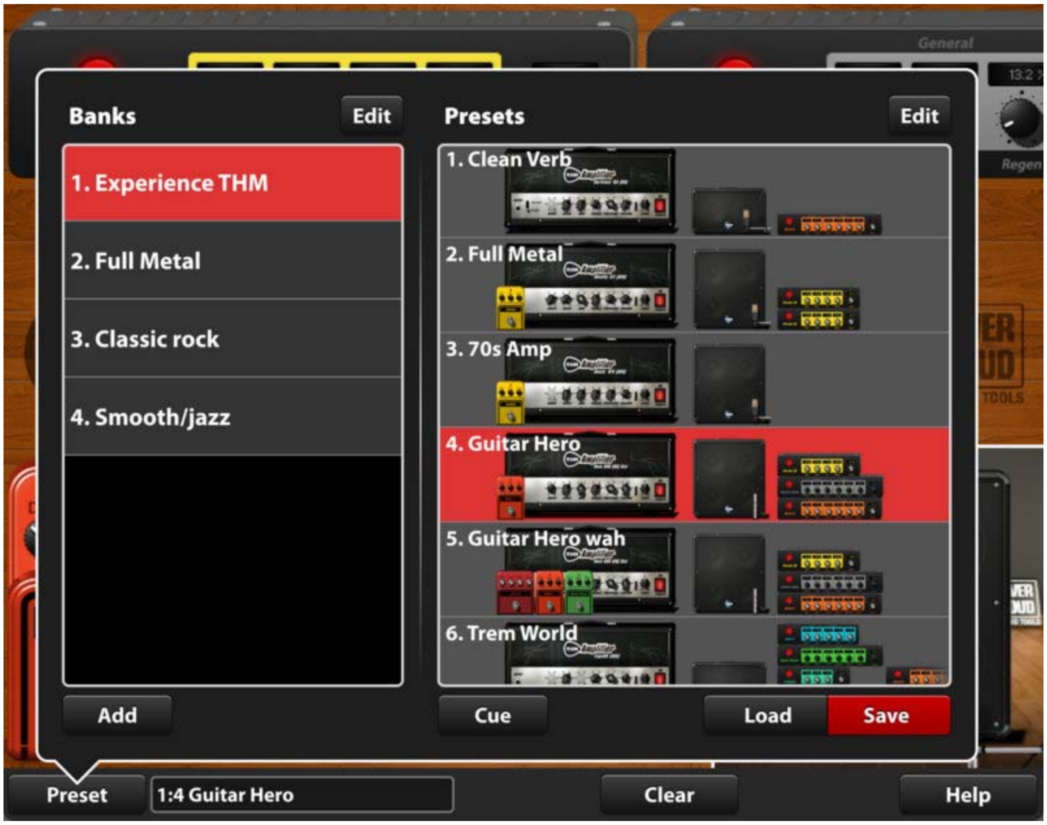
Then there is the Clear button. This button is a shortcut to empty the THM setup without having to manually remove all modules. It's useful when you want to start a brand new setup.

Lastly there is the Help button which shows a short reference guide to the main features of THM.

PRESET PANEL

When you touch the Preset button on the bottom bar, the Preset panel appears.

Here you can manage the factory presets and your own presets.



The Preset panel has two main lists, the Banks list and the Presets list. When you select a bank on the Banks list, the presets of that bank appear on the Presets list.

In this case, you can see that the preset 4 of bank 1 is selected. This is the current preset (look at the display on the bottom bar which still reads "1: 4 Guitar Hero").

Save a preset

Now you can push Save to save this preset, or you can select another preset and overwrite it (please note that the Save button text changes to Overwrite to warn you that you will lose the overwritten preset).

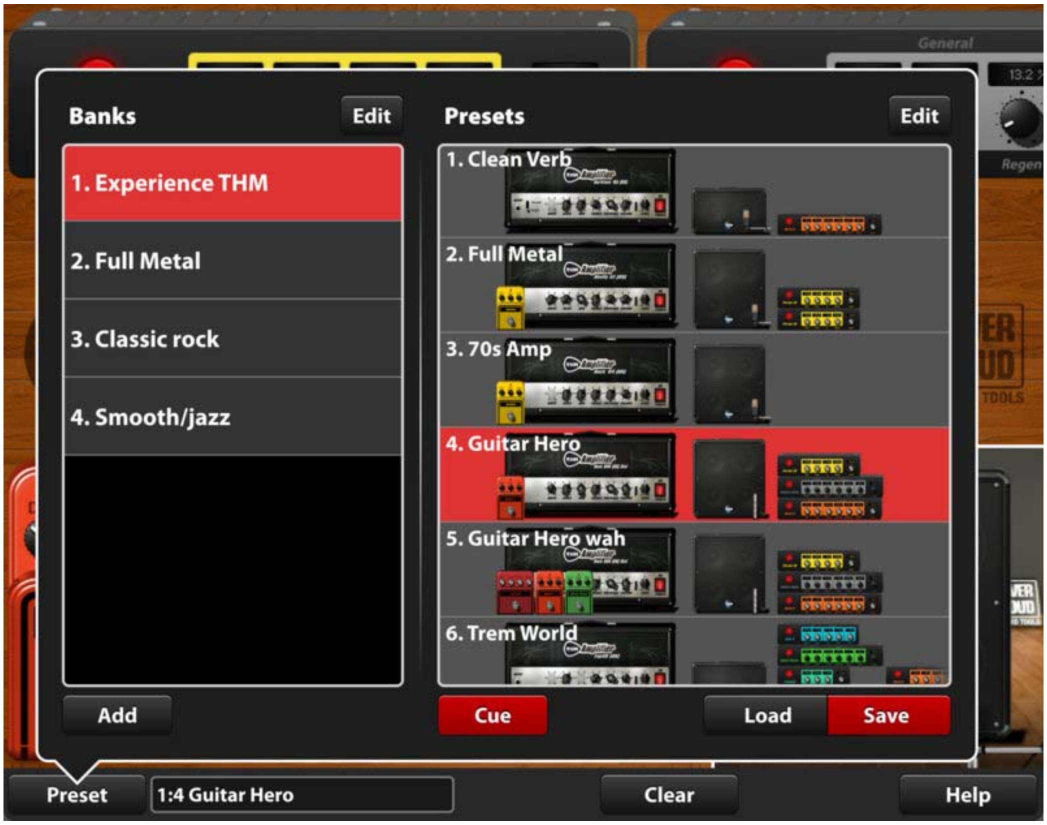
If you want to create a new bank to store a new collection of presets, just touch the Add button below the Banks list and a new bank will be added. It will initially contain 128 empty presets.

Load a preset

If you already know which preset to load, then just select it and touch Load. The preset will be loaded and the Preset panel will close, so you will get back to the main interface with the new setup of THM ready to play. You can also double-touch a preset as a shortcut to load it.

Cue presets

If you are not sure about the preset to load, you can take advantage of the Cue function.



Touch the Cue button. It will turn red to show that the Cue function is enabled. Then you can touch several presets from the Presets list and they will be instantly loaded without closing the Preset panel. That's very useful when you have a track playing on the background or if you are playing your guitar live. Once the cued preset satisfies you, just push Load and the preset will be finally loaded.

Edit banks

Items of Banks list can be edited. Touch the Edit button on the top of the Banks list to have access to four additional buttons.

The first button, Empty, empties the bank (erasing all contained presets). Note that if the selected bank already only contains empty presets, then the button text reads "Del" and it will delete the bank.

Next, there are the Up and Down buttons which can be used to move the selected bank one position up or down. This way you can sort your banks as needed.

Then the Name button allows you to rename the selected bank.

Edit presets

Items of Presets list can be edited. Touch the Edit button on the top of the Presets list to have access to four additional buttons.

The first button, Empty, empties the preset (erasing all modules).

Next, there are the Up and Down buttons which can be used to move the selected preset one position up or down. This way you can sort your presets as needed.

Then the Name button allows you to rename the selected preset.

THM MODULES

Here following are the complete lists of THM modules with the corresponding names of the emulated, modeled or captured real instruments*.

Amplifiers

- Darkface '65 (US): Fender® Twin Amp '65
- Rock '64 (UK): Marshall® JTM45
- SloDrive (US): Soldano® X88R Crunch
- Rock 900 (UK) Clean: Marshall® JCM900 Clean
- Rock 900 (UK) Dist: Marshall® JCM900 Dist
- Modern (US) CH1: MesaBoogie® Dual Rectifier Clean

- Modern (US) CH2: MesaBoogie® Dual Rectifier Crunch
- Modern (US) CH3: MesaBoogie® Dual Rectifier Lead
- Top30 (UK): Vox® AC30
- Heavy 51 (US): Peavey® 5150

Cabinets

- 1x12 Clst (UK): Marshall® 1974CX
- 2x12 OB Darkface '65 (US): Fender® Twin '65
- 2x12 OB Top 30 (UK): Vox® AC30 TopBoost Brian May
- 4x10 OB Tweed '59 (US): Fender® Super Reverb
- 4x12 Green (UK): Marshall® JCM800
- 4x12 Vintage (UK): Marshall® 1960
- 4x12 Heavy 51 (US): Peavey® 5150
- 4x12 Modern (US): Mesa® Rectifier Standard

Microphones

- Austria Gold 414: AKG® 414
- Austria 451: AKG® C-451
- TechJapan 4033: Audio Technica® AT 4033
- GermanFet 87: Neumann® U87
- RadioElectro 16: Electro-Voice® RE20
- Square 421: Shenneiser® MD 421
- American 57: Shure® SM57
- American 58: Shure® SM58

Pedals

- TubeNine: Ibanez® Tube Screamer
- BeeDeeToo: Boss® BD-2
- Distort+: MXR Distortion+
- MetalTone: Boss® MT-2 Metal Zone
- CHR-5: Boss® CE-5
- Wave Flanger: Overloud® Flanger
- Chorus Phaser: Overloud® Chorus Phaser

- DDelay: Boss® DD-3
- Noise Gate: Overloud® Noise Gate
- Tremolo: Boss® TR-2
- RSS Compressor: ROSS® Compressor
- Auto Wah: Mu-tron® III
- Env Filter: Boss® FT-2

Rack effects

- Reverb: Overloud® BREVERB Plate
- Pattern Delay: TC Electronic® TC 2290
- CHR-5: Boss® CE-5
- Digital Phaser: Overloud® Digital Phaser
- Wave Flanger: Overloud® Wave Flanger
- Tremolo: Boss® TR-2
- Auto Wah: Mu-tron® III
- Param EQ: Overloud® Full Parametric EQ Band

Positive Grid JamUp

Positive Grid brings the industry-acclaimed tone quality into the Auria system, the JamUp Essential Package provides in total 22 authentic models, it's the one stop to cover the must have tones, ready for serious recording.

The collection of amps and effects includes 6 historic amp models with matched convolution cab simulator, spring and digital reverb, tangy modulations, tape and digital delays, classic and fuzzy distortions, filters, compressors, noise gate and other essentials. Each effect parameter can be accessed with a simple touch. There's no digging through menus, no learning process and no hassles. A fully configurable signal path allows players to try out different pre and post arrangements, and it provides a great and flexible way to create your own tone.

In the signal path area, you can turn on/off each amp and effect, and change the order of the signal path. Single tap the small icon on the signal path and you can select the control panel of the amp/effect to adjust each parameter.

For more information visit http://www.positivegrid.com/auria-jamup/

FXpansion DCAM EnvShaper

DCAM EnvShaper takes a different approach to dynamics processing than a standard compressor. It allows you to adjust the attack and sustain portions of transients in order to change the dynamic shape of a signal. It is particularly useful when dealing with complex or already-mixed audio material such as a drum mix bus, but is also extremely effective on single instrument tracks.

Side Chain section

- HP Freq - The HP Freq control allows you to apply a variable high pass filter on the key signal that is used for the amplitude detection circuit. This control is useful when there is too much low-end in the input signal, which can result in the peak detector reacting too heavily.

Dynamics section

- Attack - The Attack control adjusts the intensity of the attack phase of detected peaks in the audio signal. Increase the control to intensify attack transients, and decrease it to soften transients.
- Sustain - The Sustain control adjusts the intensity of release portions of detected peaks in the audio signal, which increases or decreases the apparent sustain of sounds in the signal. Increase the control for more sustain, and decrease it for less sustain. This control is useful for adjusting the perceived level of ambience in a signal. Negative settings can produce damping effects for drum sounds.
- Signal Bias - The Signal Bias control adjusts the sensitivity and release characteristics of the EnvShaper. At low settings (towards the Fast setting) it is more sensitive to short transients while at higher settings (towards the Slow setting) it is more sensitive to longer transients (towards the Slow setting).

Master section

- In Gain - The In Gain control adjusts the level of the input signal, from -inf dB to +18 dB.
- Out Gain - The Out Gain control adjusts the level of the final output signal, from -inf dB to +6 dB.
- Mix - The Mix control allows you to blend the final output mix between the input signal (0%) and output signal (100%). This is useful for performing parallel dynamics processing.

FXpansion DCAM ChanComp

DCAM ChanComp is based on a classic limiting amplifier design commonly used as a channel compressor. It features very fast attack response and is usually intended to be applied to individual tracks.

In Gain

Increase the In Gain control to make the sound more compressed (higher signals engage the compression circuit more).

Out Gain

Adjust the Out Gain to reduce the final level as required.

Ratio

The Ratio specifies the gain reduction applied by the compressor.

Five Ratio settings are available: 4, 8, 12, 20 and Inf (Infinite). The numbered settings correspond to ratios of 4:1, 8:1, 12:1 and 20:1. The numbers represent the change in gain after compression.

For example, assuming that the threshold level has been breached, then a Ratio of 4:1 would mean that for every 4 dB of increased signal level coming into the compressor, the output level rises by 1dB.

The Infinite setting is an emulation of the ‘all buttons’ ratio mode on a classic limiting amplifier design. It affects the compression characteristics in various ways, affecting the attack and causing limiting and distortion effects, resulting in rather brutal, heavy sounds.

Attack

The Attack control adjusts the speed at which the program (input signal) gain is reduced when a peak is detected. DCAM ChanComp is designed for fast compression with Attack times from 0.02ms to 1.2ms.

Release

The Release control sets the speed at which the gain level returns to normal after a transient has passed.

The Release time ranges from 50ms to 1.2 seconds.

Bias

The Bias control continuously varies between different capacitor values which were used on various hardware revisions of the hardware on which DCAM ChanComp is based.

Settings between -25% and +25% result in subtle sonic variations in the compression characteristics. More extreme settings are useful for driving the compression circuit hard.

Mix

The Mix control allows you to blend the final output mix between the input signal (0%) and output signal (100%).

This is useful for parallel compression, allowing you to achieve a 'huge' compressed sound while keeping the original signal's transients in the mix.

G.R.

By default, the meter displays the output level from the compressor.

Activate the G.R. button to enable metering of the amount of gain reduction.

FXpansion DCAM BusComp

DCAM BusComp is based on a classic bus compressor design from the centre section of a well-known British large-format mixing console. It is usually intended to be inserted on subgroups like drum mixes or the entire master output to add 'glue' and power to a mix of tracks. However, it is also versatile enough to work very well as a channel compressor.

Sidechain section

HP Freq

The HP Freq control allows you to apply a variable high pass filter on the key signal that is used for the compressor's amplitude detection circuit. This control is useful when there is too much low-end in the signal fed into the peak detection circuit, which can result in the compressor reacting too heavily.

Listen

Activating the Listen button allows you to monitor the signal used for the compressor's detection circuit, according to the current settings of the HP Freq and External controls.

External (External sidechain)

Activate the External button in order to use the sidechain signal as the source for the peak detection circuit, allowing you to control the dynamics of the input signal with another signal entirely.

This function involves routing the desired external sidechain signal to the sidechain input of the DCAM BusComp plugin.

Envelope section

Attack

The Attack control adjusts the speed at which the program (input signal) gain is reduced when a peak is detected.

Six attack times are available: 0.1 ms, 0.3 ms, 1 ms, 3 ms, 10 ms, 30 ms.

Release

The Release control sets the speed at which the gain level returns to normal after a transient has passed.

The following Release settings are available: 0.1 ms, 0.3 ms, 0.6 ms, 1.2 ms and Auto.

Compressor section

Threshold

The Threshold represents the input level at which the compressor starts to react - any signals over the Threshold level engage the compressor circuit.

Makeup

The Makeup control increases the output gain after the compressor circuit has applied gain reduction to the input signal.

Ratio

The Ratio specifies the gain reduction applied by the compressor. 3 Ratio settings are available: 2:1, 4:1 and 10:1. The numbers represent the change in gain after compression.

For example, assuming that the threshold level has been breached, then a Ratio of 4:1 would mean that for every 4 dB of in- creased signal level coming into the compressor, the output level rises by 1dB.

Master section

In Gain

The In Gain control adjusts the level of the input signal, from -inf dB to +18 dB.

Out Gain

The Out Gain control adjusts the level of the final output signal, from -inf dB to +6 dB.

Mix

The Mix control allows you to blend the final output mix between the input signal (0%) and output signal (100%).

This is useful for parallel compression, allowing you to achieve a ‘huge’ compressed sound while keeping the original signal's transients in the mix.

Saturate

Activating the Saturate button enables DCAM BusComp's saturation circuit. The saturation behaviour is dependent on the level of the input signal. Note that this function is not a peak clipper - the signal can still exceed 0dB depending on the input level and compression settings.

G.R.

By default, the meter displays the output level from the compressor. Activate the G.R. button to switch to metering the amount of gain reduction.

Sugar Bytes Turnado

Turnado is a new multi-effect tool for real-time beat and audio manipulation. With 24 new effect algorithms and a completely new, one-knob approach to working with Effects.

- Turn the Knob ...The effect is on.
- Turn the Knob further ...The effect parameters are being modified.
- Turn the Knob back ...The effect is off.

HOST INTEGRATION

Press the "FX" button on a track to open the “ChannelStrip” window and load Turnado as an insert effect. (Make sure it's not bypassed.)

Automation

Press the "FX" button and scroll down to Turnado.

Select the parameter you want to control via automation and draw your curve.

- The first 8 parameters are controlling the 8 main knobs.
- Then comes the Dictator control, the dry/wet and the randomize control.
- Then for all 8 engines the parameters will follow.

MIDI Learn

Long press a control and click "Learn" to assign incoming MIDI CC's and Clear to delete the assignment.

Turnado sends the values of the 8 Main Knobs as CC's 1-8.

Zoom

Double click an empty area in order to zoom in or out.

MAIN PAGE

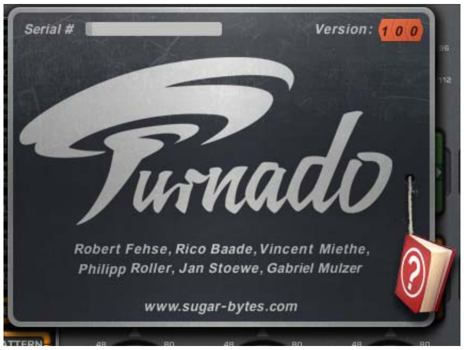
The Main Page of Turnado is divided in three sections.

Overview

Here you can chose a global preset, sequentially using the arrow tabs, or from a drop-down menu accessed by clicking on the Preset Name Display. Click on the grey buttons to access the Settings or Dictator window. Click on the Turnado or Sugar Bytes logo to open the About Screen.

- Below the preset menu you find the Global DRY WET control to set the Overall Dry/Wet Mix of Turnado.
- To reverse or to recover changes use the Undo/Redo Buttons.
- Above is a randomize button, which randomizes all effects.

About Screen



The About Screen displays the version number and names of contributors. Your serial number is shown in the top left, along with its validation status.

Just click on the little book for quick access to this manual.

Settings

Default settings can be made to adjust the way Turnado operates in certain situations. Every adjustment can be made Per Preset or applied generally.

The available settings are:

- Dynamic Displays: The displays of the parameters show Real-time values.
- CC Recall Lock: MIDI Learn Settings will be kept and not be changed when selecting a different preset.
- Dynamic Signal Flow: When checked, the last activated effect lies in front of the previously activated effect in the signal chain. Otherwise the signal flow follows the sequence of the respective effect slots.
- Ignore Program Change: Incoming program changes will not change effects.
- Reset After Load: The Main Knobs will be set to zero when a preset is loaded.
- FX Off at Knob Full On: The effect will be deactivated at zero and full rotation of the Main Knob
- Activation Threshold: Determines the position at which the Main Knob triggers the effect.
- Activate MIDI Out: Sends the position of the 8 main knobs as CC data. The knobs are mapped to CC 1-8.
- Turn Off Knobs on Host Stop: When stopping your host, all effects are turned off in Turnado.
- Keep Bypass State: The Bypass State will be changed or not when loading another preset.

Effect Browser

There are 24 effects in the effect browser. You can drag and drop them into any of the 8 available Effect Slots. The effects groups are labelled with colours for easy differentiation.

Be aware that some effects sample the audio signal and then interchange it for the original signal. For example, Looper or Slice Arranger turned on before audio was played can lead to unexpected silence.

Effect Slots

Each of the 8 Effect Slots has a Preset Menu for the chosen effect type. To adjust parameters and performance control of the current effect there is a separate Edit Page, which can be opened using the Edit button.

Main Knob

Each Effect Slot has a corresponding Main Knob. This is used to turn the effect on or off and adjust the assigned Effect Parameters. The standard MIDI allocation is CC1 to CC8 and the 0 to 127 scale corresponds to the MIDI specification. Each Main Knob bears a number corresponding to the Effect Slot number, when illuminated this number indicates that the effect is active.

Use the little lamp beside the Main Knob to bypass the effect.

EDIT PAGE

The Main Controller, in the centre, is simply the Edit Page representation of the corresponding Main Knob on the Main Page. The Main Controller shows the position and activity of the Main Knob. Coloured circuits help to illustrate possible routing options between Effect Parameters, Modulators and the Main Controller.

EDIT PAGE SECTIONS

Main Controller Overview

In the top left is a set of Mini Controllers, one for each of the eight effects. It is intended to give an overview of the Main Knob positions and quick access to the other Effect Slot's Edit Pages. By clicking on one of the eight controllers you can jump to the corresponding Edit Page. Moreover, the eight Main Knobs can be controlled here. This enables you to quickly test different combinations of effects, by turning each effect on or off without leaving the Edit Page. The controller of an active effect is illuminated in green and the currently edited effect is illuminated red.

Effect Name/Menu

Right of the Main Controller Overview is the Effect Name. This drop-down menu enables you to choose other effects for a particular slot right from the Edit Page.

Effect Preset

Here you can chose a preset, sequentially using the arrow tabs, or from a drop-down menu accessed by clicking on the Preset Name Display. With the Save function, you can also save your own presets and modifications.

Key Sync

The Key Sync enables you to quantise Turnado effects to the beat. When set to "Off" then the effect will activate as soon as you turn it on. If it is set to "¼" bar, then the effect will activate at the beginning of the next beat regardless of when you turn the effect on.

Close

Click the close button in the top right corner to quit the Edit Page and return to the Main Page.

EFFECTS PARAMETERS

The Effect Parameters of the chosen effect can be adjusted using the five red controllers in the top half of the Edit Page. The red display to the right of each Effect Parameter shows the current value, while the green display, when present, is a drop-down options menu for that Effects Parameter.

Dry/Wet

All effects have the same Dry/Wet Effect Parameter. The five different modes give you greater control over how Turnado works within the mix.

- Equal: The cross-fade is shaped according to the "Equal Power Law", which leads to some signal attenuation at a 50/50 mix.
- X-Fade: Source audio and effect signal are being mixed using a linear transformation.
- Dry: The original signal is being mixed into the effect signal.
- Wet: The effect signal is being mixed into the original signal.
- Wet Only: Only the effect signal is audible and the Dry/Wet Effect Parameter becomes a volume controller. This setting is particularly useful when using Turnado as a send effect.

Gain

The Gain fader lets you adjust the volume of the effected signal.

Amount Controller

The Amount Controller, located underneath each parameter controller, determines the influence of the Main Controller on that parameter. Double clicking on any controller will return it to centre, in this position nothing happens. If the Amount Controller is turned anti-clockwise, then the Main Controller has a subtractive relationship to the Effect Parameter. When the

Amount Controller is turned clockwise, then the Main Controller has an additive relationship to the Effect Parameter.

In the middle of the Amount Controller is a waveform illustrating the transformation curve, which the modulated parameter will follow, in response to movement of the Main Controller. Underneath the Amount Controller is a drop-down menu for choosing transformation curves. These curves offer a range of transformation patterns allowing modulation to occur at a later point in time, stop intermittently, follow steps, or follow logarithmic and exponential curves.

On the right hand side of the Amount Controllers are the Modulator Allocation Switches. When the Modulator Allocation Switches are in centre position, the Modulators have no influence on the Effect Parameters. In position "+" the modulator has an additive relationship to the Effect Parameter. In the "–" position the modulator has a subtractive relationship to the Effect Parameter. The intensity of the Modulators is adjusted with the Amount controller in the respective Modulator.

The parameter range, and therefore expected intensity of the modulation, is shown in the middle of the Effect Parameter controller itself. The white band illustrates the defined range of the modulation. The red indicator arm shows the real-time value, as well as the influence of the Modulators and the Main Controller.

Main Controller

The Main Controller in the centre of the page is simply the Edit Page representation of the corresponding Main Knob on the Main Page. The Main Controller shows the position, and activity, of the Main Knob. It is useful for testing the current effect settings, as well as keeping track of the current Main Controller position when using a MIDI control device.

Modulators

Turnado has three independent Modulators. There are two LFO's, which can be used as Step Sequencers or Envelope Generators. Between the LFO's is an Envelope Follower, which generates an envelope from the incoming audio signal. The Modulators can be assigned to any Effect Parameter and can have an additive or subtractive influence on the parameter value. You do not have to choose between the Modulators, all three Modulators can be used on each Parameter at the same time.

LFO Function Panels

To the left and right of the Main Controller are the LFO Function Panels for the two LFO's. Here, LFO curves or the Step Sequencer can be chosen and edited by clicking/dragging values directly on the panel itself.

Waveforms

In this menu the LFO Waveforms are selected. The Step Sequencer is at the bottom of this list and can be edited directly on the LFO Function Panel.

One-shot/Loop/Host Sync

Here you can choose whether the LFO/Step Sequencer triggers only once or loops continuously. When Host Sync is selected the LFO will run synced to the host clock.

Time Factor

Here you can choose between the three Time Factors; Sync (LFO Rate synchronized to the beat), Hz (LFO Rate in Hz) and Triplet Sync (LFO Rate in synchronized to the beat in triplet and dotted patterns).

Quantise

This setting defines the number of steps in the LFO Waveform and Step Sequencer data. When Quantise is “Off” the LFO Waveform and Step Sequencer will allow the full data range to be used. Changing the Quantise to “2” will reduce the data range to two values, maximum and minimum. Increasing the Quantise value further increases the number of data points between maximum and minimum, from “3” to “12”.

Main Parameters of the Modulators

Each of the Modulators has three Main Modulator Parameters. The Main Controller can influence the Main Modulator Parameters in the same way as Effect Parameters. There are separate Amount Controllers for each of the Main Modulator Parameters, allowing you to define the range of modulation and select different transformation curves.

LFO

The LFO (Low Frequency Oscillator) generates modulation with a continuously repeated waveform. In addition, the Turnado LFO offers a Step Sequencer, which is also capable of triggering a sequence only once, enabling you to generate many different types of envelopes. The Main Controller also activates the LFO.

Rate

The LFO Rate determines the frequency of the LFO, or the speed of the Step Sequencer pattern. Note that with the 1.5 update you have a rate multiplier in the waveform view, which allows you to multiply the selected rate up to 8 times. This allows ultra slow movements.

Phase

Here, the starting point within the LFO Waveform and the Step Sequencer is chosen.

Amount

Determines the intensity of the modulation being sent. Whether the modulation affects the end Parameter in an additive or subtractive way, is determined with the "+" and "-" buttons of the respective Effect Parameter.

Envelope Follower

The Envelope Follower generates an envelope from the source audio signal. This Modulator is very dependent on the dynamics of the incoming signal. Whilst strong beats create obvious modulations, signals without significant dynamics will create more subtle modulation.

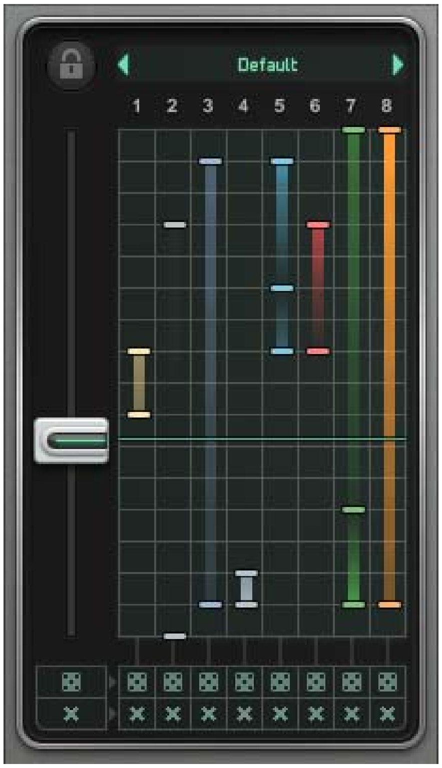
Attack

Determines the lead-in time of the Envelope Curve. The shorter the lead-in time the faster the reaction to the audio dynamics. While a longer lead-in time gives a slower reaction it can also massively reduce the effect of the Envelope Follower.

Release

Determines the decay time of the envelope. Short decay times enable the Envelope Follower to react quickly to the audio signal whereas long decay times create a more sustained modulation.

DICTATOR MODE



The powerful Dictator mode enables you to create an automation sequence for the 8 Main Knobs and assign the entire sequence to one fader.
The Dictator window shows eight vertical tracks, which correspond to the 8 Effect Slots. On the left is a fader that enables you to move through the sequence of Main Knob automations.
To create automation for one of the Main Knobs, simply click on the vertical track corresponding to the effect you want. A coloured bar will appear with a shadow, this is an Automation Point. The intensity of the shadow's colour indicates the range of the automation.
Load the "Allinarow" Preset from the Preset Menu at the top of the Dictator window. Moving the fader up and down while watching the Main Knobs will give you a good indication of how the Dictator works.

Some of the key functions are:

- Two Automation Points are necessary to create an automation
- Press to create more Automation Points
- The default range of an Automation Point is 50%
- Long press to delete Automation Points
- The Dice Buttons allow for random generation of automations for all or individual tracks
- The X Buttons delete all or single tracks
- While active the X Buttons inhibit creation of Automation Points
- The Preset Menu enables you to load and save Dictator Presets

THE EFFECTS - Delays

Pattern Delay

The Pattern Delay has 8 delay lines and offers a number of pre-defined Patterns, each with different timing and pitch. The first Effect Parameter is Delay Time, which can be synchronised or unsynchronised in relation to the tempo. The second Effect Parameter selects the Pattern. Every pattern is individual so you should test the various settings to get a feel for them. Changing the Pattern option to "Fade" enables cross-fading between patterns. The Amount Parameter determines the intensity of the pitch component of the Pattern Delay, while the Decay Parameter determines the volume relationship of the 8 delays. Alternate delay lines can be turned off using the Option Settings 2nd and 3rd, which respectively enable only every 2nd or 3rd delay line to be heard.

Reverse Delay

Reverse Delay layers a played-backward delay signal over the source audio. The Time Left and Time Right Parameters control the delay time as well as the length of the played-backwards material. The reversed signal is blended in and out to avoid clipping. The Fade Parameter determines the length of the cross-fade, while the Feedback controls the intensity and duration of the delay tails.

Pitch Delay

Pitch Delay is a classic delay with an integrated filter and the additional option of modifying the pitch of the delay tails. The Time Left, Time Right and Feedback Parameters are standard delay controls. Along with synchronisation settings you can also set up when the incoming signal is routed into the delay.

- "¼ ..." Allows the incoming signal to pass only once for the length of a quarter note.
- "¼--- ..." Allows the incoming signal to pass once for the length of a quarter note and then pauses for 3 quarter notes.
- "¼- ..." Allows the incoming signal to alternate between passing and pausing for the length of a quarter note.
- Option settings for eighth note and sixteenth note denominations follow the same pattern.

The Pitch Parameter adds a positive or negative pitch change to the delayed signal. Furthermore you can add the filter to create special new Dub-style delay effects. There are several filter settings available so you can decide whether the Pitch and Filter work separately, together, or inversely. For every Filter Type there is a low and high (Q) resonance setting. The first menu entry "Pitch" has no filter and only modifies the pitch. A "+" in front of the menu entry gives an additive cutoff modification, when turned up the Pitch Controller also increases the Filter Frequency. A "-" in front of the menu entry will result in the Filter Frequency decreasing when turning up the pitch. If the menu entry has no prefix then only the filter is active. With the 1.5 update there are new modes with a constant filter-frequency, the knob will adjust the input level of the signal.

THE EFFECTS - Modulation Effects

Flanger

This is a classic Modulation Effect using very short delay times to create Flange and Chorus effects. The Delay Left and Delay Right Parameters enable you to set independent delay times for each of the respective channels. Using the Flanger/Chorus settings you can select different delay time ranges. In Flanger Mode the range is 1-20 milliseconds and in Chorus Mode the range is 25-50 milliseconds.

The Feedback Parameter returns the signal back to the delay input to enhance the effect. The "Inverse" Option of the Feedback Parameter inverts the phase of the feedback by 180 degrees and creates a softer, more diffused sound. With a negative setting the Filter Parameter works as a Low-pass filter and with a positive setting as a High-pass filter. The menu on the Filter Parameter offers three different resonance values to give different tonal qualities to the filter.

Phaser

As well as generating classic phase effects using the LFO's, you can also directly change the Phaser value. Assigning the Phase Parameter to the Main Controller and applying different transformation curves can create some interesting effects. The intensity is modified with the Feedback and Depth Parameters. Feedback can be inverted for a fluffier sound here too and the Width Parameter shifts the phase of the left and right channel creating stereo movement.

Tonalizer

The Tonalizer is a special delay that uses short, tonal delay times and high feedback to create tuned delay tails. Note Right and Note Left Parameters define the pitch of the left and right channel, while the Option Menus enable you to choose between various tonal intervals, from semi-tones through to octaves. High settings of the Feedback Parameter will widen and intensify the tonal effect. When turned clockwise of centre position the Hold Parameter freezes the wet signal, creating a continuous tone. In this position no incoming audio is being processed and only the frozen, Tonalized delay will be heard.

In the Hold menu you'll find the root note option which let's you define a root-note. The notes defined by the first 2 parameter are then on top of this one. For the first 2 parameter there are now scales available. This makes sure that the delay-tuning fits into the selected scale. This gives you the possibilities to adjust the sound related to your song tuning.

THE EFFECTS - Reverbs

Reverb

The Reverb is a first class Echo Effect. The Size Parameter controls the room size, while Reverb Time defines the length of the reverb tail. The Reflectivity Parameter determines the intensity of reflections from the virtual reverb space. The Input Parameter determines the volume of the incoming signal being sent to the effect. Special effects can be created through dynamic activation of the Input Parameter. For example you can set up an LFO to control the Input Parameter so the Reverb activates in a rhythmical way. Taking this idea further you can assign a Step Sequencer to the Input Parameter to trigger the Reverb effect in more complex patterns. There are some filter modes under the reflectivity control. You can add a highpass filter with fixed cutoff or BP, Comp or HP where the cutoff is controlled by the Reflectivity knob.

Freezeverb

The Freezeverb Effect is a special Echo Effect that freezes the Echo signal. The Size Parameter defines the virtual room size and the Damp Parameter dampens the reflection of the virtual reverb space. Width Parameter creates a broader stereo picture by slightly offsetting the left and right signals. The Freeze Parameter has two positions, off and on. When turned anti-clockwise of centre the Freeze is inactive, when turned clockwise of centre it is active. Remarkable effects can be created by modulating the Freeze Parameter with the Envelope Follower. Be aware that when the Freeze Parameter is active no audio is being sent to the reverb. Freeze should be off when you start the audio otherwise there will be silence when you play the source signal.

THE EFFECTS - Transformation Effects

Ringmodulator

The Ringmodulator is an effect where an oscillator modulates the amplitude of the audio signal. The VCA Parameter transforms the Ringmodulator defining how the internal oscillator is being amplified or the incoming signal multiplied. The AMT Parameter determines the harmonisation between the source signal and the oscillator. A major chord for example, has 4 ring modulators instead of one. Use the Waveform Parameter to select from four available waveforms: Sine, Triangle, Pulse or Sawtooth.

Vocodizer

The Vocodizer is more an instrument than an effect as it can create independent melodies, rhythms and sounds. The Sound Parameter is the spectral-dynamic reaction to the incoming signal. There are four base waveforms: Sawtooth, Triangle, Pulse and Sine. In the sub-menu these are also available in Unison Mode with the suffix "2" for a more powerful and harmonically richer sound. The Note Parameter determines the base note. The submenu defines the note range (one or two low or high octaves). The Parameter Spread creates a chord and determines the number of voices. In the sub menu, the chord type is chosen. The parameter Arp creates an arpeggio out of the selected chord. In the sub menu the arp style and pattern is chosen. The "Trig"-Entries make the Arp play the next note when the Arp Control gets above 50%. Therefore the envelope follower should be used, but also using the stepsequencer is a good choice. When the menu entry contains a „2", duophonic arp melodies will be created.

THE EFFECTS - Amplifier

Levelizer

The Levelizer is a basic effect for modifying volume, panorama, bit depth and sample rate. Although these parameters are relatively simple, with the use of Modulators many classic effects like Compression, Autopan, Tremolo or Gating can be achieved. The Volume Parameter can double the amplitude of the audio signal or reduce it to silence. The Pan Parameter moves the signal into the left or right channel. The Crush Parameter reduces the bit depth and has three options: "Normal" reduces the bit depth to create a classic low-bit sound. "Hi" mode reduces the signal in a non-linear way, resulting in loud signals being bit-reduced more than quiet ones. "Low" mode works inversely, so quiet signal parts are being bit-reduced more than loud ones.

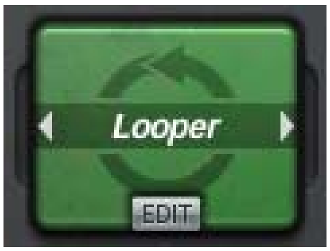
There are also three Options for the Sample Rate Parameter. "Hard" creates classic sample rate reduction until complete destruction. "Dynamic" reduces the sample rate dynamically according to the amplitude of the incoming signal. The louder a signal the greater the reduction of the sample rate will be. In "Absurd" mode the sample rate will drop towards zero, creating even harsher overtones than in "Hard" mode.

Guitar Amp

The Guitar Amp is an Amplifier/Distortion emulator with integrated multi-band EQ. It allows for individual amplification or distortion of the three EQ Bands and therefore offers greater control of the amplifier overtones. The Drive Parameter controls the total amplification. Stereo and Mono mode are available. The Low, Mid and High Parameters define the frequency bands which are to be amplified. Dynamic effects can be created through modulation of the Effect Parameters using a fast Envelope Follower. Direct assignment to the Main Controller also creates some interesting sound effects. With the 1.5 update there are now some options available to limit the output of the amp. The 3 limiting options will make sure the signal stays in 0db range.

THE EFFECTS - Loop Effects

Looper



Besides the classic looping, this effect can add swing to your looped signal. The Size Parameter determines the length and repetition rate of the loop. You can choose between the three synchronisation styles in the Option Menu. “Sync”, “Sync TP” and “Free” options dictate the synchronisation style. “Sync” quantises the loop length into ½ to 1/128-bar steps. Triplet or Dotted Note steps can be selected using “Sync TP”. “Free” mode offers loop lengths from 1 to 500 milliseconds. SyncX mode ensures that Size changes will only be activated at sync time. The Swing Parameter defines the relative lengths of alternate loops, lengthening and shortening them in order to create swing while maintaining synchronisation. The Trigger Parameter samples new audio material into the Looper when turned clockwise from centre. Assigning this parameter to an LFO or Step Sequencer can automate the sequential sampling of source audio producing excellent beat and rhythm variations. The AMP Parameter determines the volume and serves to gate loops or blend them in and out.

Pitch Looper

The Pitch Looper gives you the ability to add pre-defined pitch sequences to the looped slice. The Size Parameter defines the loop length and the Pattern Parameter selects the pitch sequence. In the Options Menu you can select “Glide” to bend between pitches in a similar way to Portamento. The Trigger Parameter functions in the same way as the in the Looper. The Decay Parameter sets the number of loop repetitions, and for silencing individual slices it also has a few options:

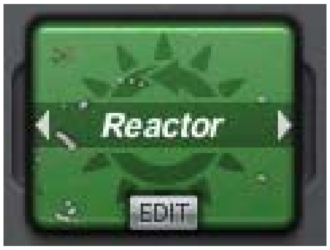
- 2nd - Every second slice is being played
- Triple 1 - Two out of three slices are being played
- Triple 2 - Two out of three slices are being played (Variation)
- Swing - A swing is being generated

Pan Looper

In addition to the alteration of the stereo picture of the slices, the Pan Looper can also manipulate the pitch. The Size and Amp Parameters define the loop length and volume respectively. The Pan Parameter changes the panorama of the loop.

The Pitch Parameter defines the pitch, while the Amp Parameter offers additional options to create swing and rhythmical elements.

Reactor



The Reactor effect is a Transient Looper and is therefore a self activating loop tool. The Active Parameter adjusts the responsiveness of the loop trigger. The Holdtime Parameter defines the length of the sampling period. Parameters two and three are switchable. Selecting "Freeze" from the Option Menu the Parameters Speed and Pitch are operable. With the Option "Reverb", Parameters Mix and Intensity are available. The Intensity Parameter is a special Reverb Hold Level that produces some great effects when combined with Modulators. With the 1.5 update there is a new sync option available called SyncX. In this mode changes of the size will be done as if they had run from the beginning. This means for example if you start with 1/8 size and then change to 1/4, another 1/8 slice will be inserted in order to ensure correct alignment for the 1/4 slice.

Slice Arranger

The Slice Arranger does exactly that, slicing the incoming signal into new patterns. The audio signal is recorded in real-time, then divided into slices and arranged according to the Pattern and Fill Parameters. With the Pattern Parameter there are a number of pre-defined patterns. You can choose from eight pattern types in the Options Menu with 50 patterns for each type. The two Fill Parameters incorporate Rolls and Microloops into the selected Pattern. Furthermore, each pattern allocates different slices to the Fills.

Here are the Fill Options:

- Repeat: The final phase of the slice is being looped
- Up: Upwards-pitch
- Down: Downwards-pitch
- Hard: Dramatic repeat
- Pong: Forwards and backwards repeat
- Soft: Simple more subtle repeat

The Decay Parameter controls the length of the slice envelopes as well as highlighting the end of the sliced audio with help of an amplitude envelope. With the Options "Beat" and "2 Beats" the slice arranging can be applied to the audio over one or two beats.

THE EFFECTS - DJ Tools

Granulizer

The Granulizer is a Grain Effect with control over tempo and pitch. The Amount Parameter has two Options: "Amount" mode sets the tempo to be changed in relation to the original audio signal. Full off is the original tempo and full on is stop. In "Position" mode the Amount Parameter navigates you through the recorded audio signal. The Grainsize Parameter defines the length of each individual grain in milliseconds. There are also several Options for cross-fading and the tempo of the individual grains is determined using the Distance Parameter. To avoid gaps in the audio signal, the Grainsize value should be larger than the Distance value. To change the play-direction of the grains, the Distance Parameter has several Options, "Forward", "Backward" and "Ping Pong". The Pitch Parameter determines the pitch of the grains. If no time-stretching is active and the pitch is being moved upwards, then a minimal delay is being mixed into the audio signal which may not be audible.

Stutter

The Stutter Effect continually adds an Amplitude Envelope to the audio signal. The Size Parameter determines the repetition rate of the envelope. As with all the other time based parameters there are also synchronisation options. SyncX mode ensures that size changes will only be activated at sync time. The SyncPPQ mode will sync the stuttering to your host clock. The Decay Parameter determines the absolute end of the envelope and offers in the options "rhythmic", "shuffled" and "swing" variations. The Shape Parameter determines the waveform of the envelope. This parameter blends continuously through the following waveforms: Downwards Sawtooth, Sine, Pulse and Upwards Sawtooth. The Pan Parameter enables control of the stereo position of the effect signal and can be combined with Modulators to create dynamic Panning effects.

Vinylizer

The Vinylizer Effect simulates the stopping and scratching of vinyl discs. The Size Parameter defines the time interval of the stops and scratches. If the value is set to OFF it will stop only once. With the Slow Down Parameter the speed of the stopping is determined and there are three options. "Stop" stops the audio signal with the rate defined by the Slow Down parameter. "Scratch" plays the audio signal alternately forward and backwards while slowing down and speeding up the audio signal. This happens in relation to the selected interval with acceleration rate controlled by the Slow Down Parameter. The Downslope and Upslope Parameters define the shape of breaking and acceleration. In full anti-clockwise position the slope becomes logarithmic this transforms into an exponential curve as you rotate through to a fully clockwise position. Setting these two parameters differently gives you the quick-stopping and slow-starting vinyl record sound.

THE EFFECTS - Filters

Filter

With the Filter Effect a quality, Stereo, Multi-mode-filter is available. The Cutoff Parameter controls the cutoff frequency and offers the modes “Highpass” (only allows frequencies above the cutoff to get through), “Bandpass” (only allows the frequencies around the cutoff to get through), “Lowpass” (only allows frequencies below the cutoff to get through), and “Comb” (a special delay which works with filter frequencies as delay times). To synchronise the left and right channel there is a sub-menu on the Cutoff R Parameter with the option to “Link”. The resonance for the left and right channel is set with the Reso L and Reso R Parameters. Be careful with resonance values, high settings can produce harmful volumes for you and your equipment.

Filter Pattern

The Filter Pattern Effect adds pre-defined Filter Patterns to the audio signal in which filter settings run through a sequence. The Pattern Parameter enables you to choose from 25 different patterns, each with 10 variations. With the 1.5 update there is a new variation called 'Beardyman', this variation will offer more simple filter rides with only one filter type. The Resonance Parameter controls the Q Factor. When in centre position the Sweepspeed Parameter will set the filter sweep to complete once. Turned anti-clockwise the speed of the sweep is reduced so it will only partially complete. Turned clockwise the sweep will accelerate and complete more than one cycle. In this situation the option setting dictates what happens to the sweep, either being repeated “Repeat”, passed forward and backwards alternately “Ping Pong” or synced to the host clock “Synced”. With the 1.5 update there is a new mode here called Synced X. In this mode the sweepspeed is always aligned to the original clock. (see Looper for detailed description)

The Sweeprange Parameter defines the range of the frequency sweep. Be careful with resonance values, high settings can produce harmful volumes for you and your equipment.

Vowel Filter

The Vowel Filter offers a powerful Talkbox sound, making the audio signal sound as if it were being spoken. The Vowel A and Vowel B Parameters identify the vowels between which the Mix Parameter cross-fades. The Vowel A Parameter can have a filter assigned to it from the sub-menu, “Highpass”, “Lowpass”, “Bandpass” and “Comb”. Keeping the Resonance Parameter settings high will ensure you get a rich vowel sound.

Spectralizer

The Spectralizer is a filter-bank with 32 delays. Every delay has its own delay time and filter frequency. The delay time and density are determined by the Delay Time Parameter. Available modes are “Tonal”, “Sync” and “Free”. The Frequency Parameter adjusts the cutoff frequency with which the delays work, the available sub menu offers different relations of the delaylines and their frequencies. The Resonance Parameter controls the resonance of all the filters. You can select 2 Pan modes here which will bring some space into the sound. The Bands Parameter determines the number of delays used. Be aware that this effect is quite complex and results depend greatly on the source audio. With drum parts for example, a high resonance value with full delay size produces a very bizarre effect. Test the various presets to get an overview of the variety of sounds that you can create with this effect. With the 1.5 update there are 2 pan modes (under the resonance parameter) available which offers stereo distribution of the bands.

FAQ

Silence - Take note that some effects sample the audio signal and then replace it with the sampled signal. For example if a Looper or Slice Arranger controller is turned on, or the Freeze in Freezeverb active before audio is played, you will hear nothing.

Contact

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Sugar Bytes WOW Filterbank

What is WOW2

WOW2 is a state of the art filter, offering 21 high quality filter types. Each filter type can be operated in vowel mode, giving you the best vocal sounds available. WOW2 comes with the craziest modulation system you will ever find. Modulators can modulate and randomize each other, and the wobble generator is on board, for rhythmic control voltage without headache. Enjoy the juiciest, punchiest filters with close to analog quality, only available in WOW2.

Host Integration

Press the "FX" button on a track to open the “ChannelStrip” window and load WOW2 as an insert effect. (Make sure it's not bypassed.)

Automation

Press the "FX" button and scroll down to WOW2.

Select the parameter you want to control via automation and draw your curve.

Zoom

Double click an empty area in order to zoom in or out.

WOW2 Structure

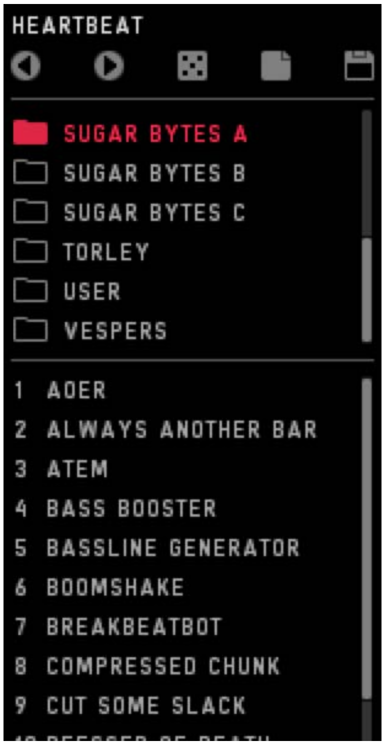
About Screen

Click on the WOW2 logo to open the About Screen.

Here you find your WOW2 version number, if it's needed to download new updates.

Click on the « ? » button to automatically open the WOW2 manual.

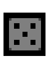
The Presets

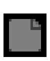


In the left half of the screen you will find the preset browser.

The top line shows the current preset name and offers different file operations.

 Step through the presets, forward or backwards.

 Load a random preset.

 Load a clean preset to start from.

 Opens a save dialog to save your preset.

Filter Section

The red part of WOW2 includes the actual filter section.

It features Dry/Wet, Level, Cutoff, Resonance, Vowel Mode and Distortion.
You can choose from 7 distortion types, 21 filter types and use the Vowel Mode here. All these Parameters can be modulated. Long press a parameter, to assign modulation and MIDI Learn to it.

Additionally, all modulation amounts are available on the bottom of the modulation section, in order to have these controls automated and MIDI-learned. This means, WOW2 can assign a modulation from source to target, but also from target to source, which is most convenient.

Classic Controls

Cutoff

The biggest knob is the Cutoff control. Basically, it determines the cutoff frequency, which is the frequency where the filter works. This frequency is boosted by the resonance and determines where the Lowpass cuts high frequencies off, or where the Bandpass lets the signal through.

In Vowel Mode, the Cutoff Control morphs from Vowel A to Vowel B.

Resonance

The resonance is actually the feedback level of the filter circuit. Turning this control up will expose the cutoff frequency. In some filter types the resonance can introduce « self-oscillation » , which turns the filter into a sine oscillator which oscillates at cutoff frequency, turning WOW2 into a dub siren, or whatever you make of it.

If you run in unwanted self-oscillation, use the Envelope Follower to push the Reso up to where you want it, while silence will hold the Reso below self-oscillation.

Distortion

The Distortion Parameter always shows the name of the selected Distortion type. If the Distortion Parameter is set to zero, the distortion is bypassed. In the upper half of the distortion menu you find the Signal Flow control. Put the Distortion in front or behind the filter for a vast range of sounds.

7 Distortion types are available:

1. Parabolic. Tube-like overdrive, creating a rich harmonic spectrum, four times oversampled for finest harmonics without aliasing.
2. Hyperbolic. Tube-like double-drive for rather angry distortions. Four times oversampled for crystal clear harmonics without aliasing.
3. Diabolic. Diode-like distortion, for the apocalyptic punch. Four times oversampled for high definition apocalypse without aliasing.
4. One Bit. Turns everything into a pulse wave. Four times oversampled.

5. Sine. The audio signal drives a sine function. Creates sounds from roasted ham to spider invasion. Four times oversampled.
6. Crush. The bit crusher you would have asked for.
7. Digitize. The sample rate reducer you would ask for.

Level

The Level control can double up the signal volume, so when the Level control is at 50%, you have 1:1 Level from input to output.

Use the Envelope Follower to turn down the Level according to the input signal, to create compressor effects. The modulation system offers all kinds of Level modifications, like stuttering, tremolos, compressors.

Dry/Wet

Here you mix between input and output signal. Especially the Vowel Mode often requires not a full wet mix for best performance. The modulation system and be used to blend in the filter in every possible way.

Filter Types

Select the filter type here. We included Moog and MS filter models, as well as some Sugar Bytes creations like the « 030 Lowpass ». 030 and MS models are oversampled for best performance at the whole frequency range. The (sat) filter models include a saturator, so turning up the resonance will not turn down the input level. These are two existing philosophies about handling the Resonance, so we decided for both ways.

Available are the following filter types:

Highpass

Cuts off low frequencies according to the cutoff frequency.

- 2Pole: 12db Highpass from a SVF (State Variable Filter)
- 2Pole(Sat): 12db Highpass SVF including saturation. See text above.
- 4Pole: 24db Highpass SVF
- Diode MS: 12db 1pole Highpass, based on the MS diode ladder filter.

Bandpass

Cuts off low and high frequencies and passes thru the cutoff frequency. Best filter for talkbox effects using the Vovel Mode.

- 2Pole: 12db Bandpass SVF
- 2Pole(SAT): 12db Bandpass with saturation (see text above)
- 4Pole: 24db Bandpass SVF
- Diode MS: 12db Bandpass, based on the MS diode ladder filter.
- Ladder MG: 24db Band/Lowpass filter, based on the Moog Ladder Filter.

Lowpass

Cuts off high frequencies according to the cutoff frequency.

- 2Pole: 12db Lowpass SVF
- 2Pole(SAT): 12db Lowpass SVF with saturation (see text above)
- 4Pole: 24db Lowpass SVF
- 8Pole: 48db Lowpass SVF
- Ladder MG: 24db Lowpass, based on the Moog Transistor Ladder filter
- Diode MS: 12db Lowpass, based on the MS Diode Ladder filter
- 030: 18db Lowpass, based on the Roland TB-303 filter.

Special

This category contains filter types beyond the HP/BP/LP classics.

- Mid Boost: 24db SVF Lowpass/Bandpass combination

- Peak: Creates a peak at the cutoff frequency without losing original audio.
- Notch: Creates a hole at the cutoff frequency, keeping original audio.
- Band Reject: Erases frequencies around the Cutoff Frequency and creates two peaks with high resonance, very good in VOWEL MODE.
- Mid Clear: 12db Highpass/Lowpass combination which eliminates the mid frequencies between them, while they run away from each other as you turn up the cutoff.
- Comb: A feedback delayline with delaytimes according to the cutoff frequency. Should be used with high resonance. This filtertype is working very good with the VOWEL MODE and can produce intensive flanger- and chorus-effects in classic mode.

Vowel Mode

Put the filter into Vowel mode, where the Cutoff knob is used to fade between two vowel frequencies. Bandpass and Comb Filter modes are recommended for achieving the best vowel sounds. A high Resonance is usually needed to create the « Formant » necessary for the vocal sound.

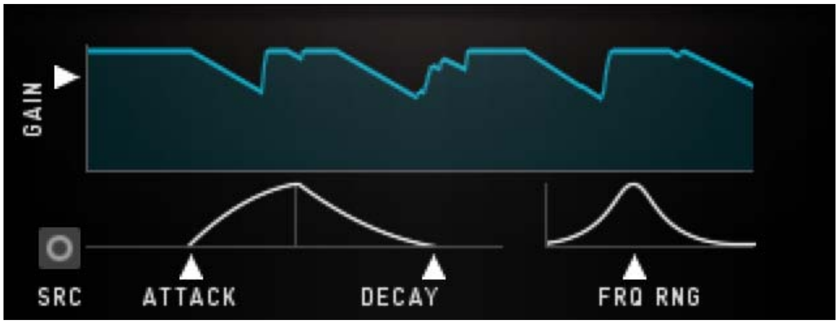
[i:]	as in feet
[e]	as in men
[æ]	as it bat
[y]	as in tu (French)
[ə]	as in the
[ɑ]	as in father
[ɔ]	as in awe
[o]	as in copy
[u]	as in boot

Long press a phonetic symbol to modulate the Vowel Mode. You can see the phonetic symbols switching. Click on the phonetic symbol to see the basic value.

Modulation Section

The modulation section includes 4 modulation engines that can modulate each other as well.

Envelope Follower



Follows the amplitude of the incoming audio signal to produce a control curve.

Gain

Set the level of the controller signal here.

Attack

Smoothes the rising part of the controller signal.

Decay

Smoothes the falling part of the controller signal.

Freq Range

A Bandpass filter in the sidechain circuit allows frequency selective envelope following. That way you can grab a kick, a snare or other signals to produce a controller signal. Turn the knob down to bypass the filter and feed the whole spectrum into the Envelope Follower.

Source

The Envelope Follower can be provided with the input signal or with the output signal, to create the envelope from. Since the Filters might create high levels, the Envelope Follower can be used to modulate the Level control.

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The Circle symbol indicates that the WOW audio output is used for the Envelope Follower.

The Line symbol indicates that the WOW audio output is used to create the envelope.

LFO

An oscillator generates a control curve with different waveforms. The LFO has three sync modes.

1. Sync, Audio Trig:
2. Free, Audio Trig: The LFO runs at divisions of your host bpm and can be retriggered by the incoming audio signal.
3. Songposition: The songposition is used to generate the LFO waveform. Using that method, the LFO wave is absolutely reproducible in every position of your song. For example, on beat 132 the waveform will be in the same position, no matter if you started the playback at beat 12 or beat 36.

Wave

The LFO waveform. Sine, Saw, Square, Triangle and random are available.

Rate

The speed of the LFO curve. Always in sync with your song.

Audio Trig

The LFO can be retriggered by the audio signal.

Trig Sens

Sensitivity of the LFO retrigger.

Step Sequencer

A sequencer with 16 steps generates a control curve.

Tempo

Speed of the Sequencer. It´s always in sync with your song.

Direction

The reading-direction. Forward, Backward, Pingpong or Random.

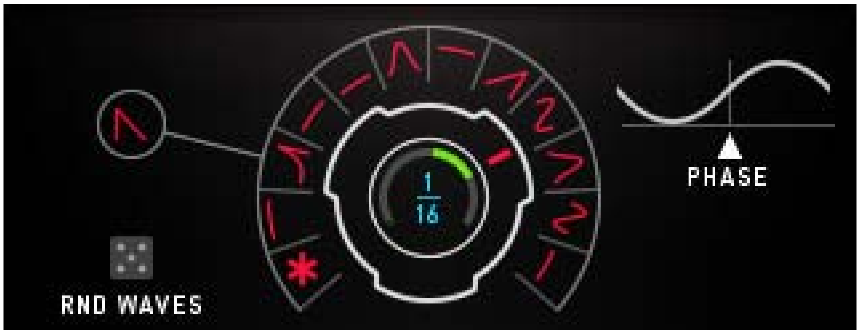
Glide

Glides the sequence steps to a continuous curve.

Random

Selects a random control curve.

Wobbler



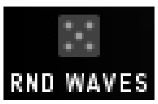
The Wobble Knob lets you choose from 12 LFO Speeds and 16 Waveforms.

 Special about the Wobble LFO are the fixed values.
The snowflake indicated Freeze mode. Like a sample and hold module, the Freeze will maintain the last active value as long as it is selected.

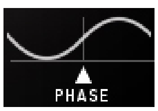
If the Wobble Knob turns from a sine wave to the snowflake, the last value of the sine wave will be held until another wave form is selected.


Furthermore, there are 5 fixed valued available. These values are 0%, 25%, 50%, 75% and 100% and are displayed by square with different sizes. These values make it possible to use the Wobble Generator like a step sequencer.


The central wave form control sets all wave forms at once.



The Random Button sets a random wave form situation. Can be modulated, in order to auto-randomize at certain events.



Here you can set the starting phase of the wobble LFO. This determines if the modulation will go upwards or downwards after you hit a note.

Modulation Assignment

In order to assign modulators to their targets, you have two choices: Long press the target parameter to choose from the available modulators, or select the source-modulator and find the target parameters on the bottoms of the modulators area.

The Mod-Amount knobs control the modulation intensity from -100% to +100%. The modulation can be set positive or negative, so it will add or subtract with the value given by the parameter. The Reset Button resets all modulations to zero. Assigned modulations can be identified by the moving ring around the parameter.

Double click a control to set it to zero.

These are the Modulation Assignments below the Modulators. Each Modulator has its own set of 6 assignment screens. The Amount Controls can be MIDI learned and automated:

Main Filter Assignment

LFO Assignment

Env Follower Assignment

Step Seq Assignment

Wobbler Assignment

Vowel Assignment

MIDI Learn

Long press a control to open the MIDI Learn function. Using MIDI Learn, you can control a WOW2 parameter with an external MIDI controller. Just assign a MIDI device to WOW2 and use it to send MIDI CC´s to WOW2.

Questions?

WOW2 makes no sound

Are the level meters showing an Audio Signal?

If Audio In works, but Audio Out is silent, your filter setting might be filtering out all audible frequencies (HP too high?, LP too Low?).

Level could be modulated to zero. Long press Level to check out the modulation assignments.

Have I found a bug?

Please write to support@sugar-bytes.de

Impressum

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