Auria 2.0 User Guide

For Auria/Auria Pro



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Auria User Guide v2.18
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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.

Two Versions of Auria

Auria is currently sold in two main editions, Auria ($24.99 USD) and Auria Pro ($49.99 USD), and this User Guide will discuss those features found in both of these versions. For standard Auria users (i.e. non-Pro), those chapters which discuss specific Pro-only features (such as MIDI Sequencing or Audio Warping) are marked as Pro Only in their title. When a particular chapter deals with features found in both editions this Guide will attempt to specify whenever Pro-only features are discussed. If there is ever any question regarding which version includes a specific feature please contact WaveMachine Labs.

For a list of the feature differences between the two editions, please refer to the chart found later in this chapter.

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:

Tap
Pan (or Flick)
Pinch
Tap and hold
Double-tap
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.

System Requirements

Since Auria's initial release back in 2012, many new iPad versions and iOS updates have occurred. With the release of Auria and Auria Pro 2.0 the system requirements have been updated to reflect the latest technology available.

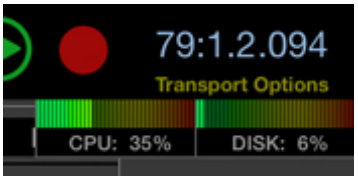
- iPad 4 or later recommended.
- iOS 6.1 or later required.
- Auria will record audio from the iPad mic, but a compatible USB Class 2 audio interface and Camera Connection Kit is recommended.
- Auria Pro supports CoreMIDI compatible MIDI interfaces.

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.

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 Buffer Size 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 Auria iOS platform and offer them through in-app purchase.

Windows and OSX format plug-ins are not 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 Auria, 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.

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.

Audio Unit (AUv3) support has been added to Auria starting with v2.06. For more information please refer to the Audio Unit section.

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.

Auria includes an extensive file management system 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 2.0 and Auria Pro 2.0

In addition to the original version of Auria, there is a more powerful version available called Auria Pro, which adds a number of additional features. The primary differences between the two versions include:

	Auria	**Auria Pro**
MIDI Sequencing	No	Yes
Piano Roll Editor	No	Yes
Lyra, a multi-format sample player	No	Yes
FabFilter One and Twin2 Synths	No	Yes
Real-time Quantize Engine	No	Yes
Groove Template Quantize Engine	No	Yes
Tempo and Time Signature Track	No	Yes
Real-time Audio Warping	No	Yes
Audio Quantize	No	Yes
Transient Detection and Slicing	No	Yes
Flexible bus routing w/ multiple outs	No	Yes
Bounce Track in Place	No	Yes
Direct Outs	No	Yes
Auxiliary Sends	2	6
Tracks - Playback	Unlimited	Unlimited
Project Templates	Yes	Yes
Offline Time Stretching	Yes	Yes
Sample Rates	44.1/48/96kHz	44.1/48/96kHz
Track Freeze	Yes	Yes
AAF Project Import/Export	Yes	Yes
MIDI Sync	Yes	Yes
Convolution Reverb	Yes	Yes
Vintage-inspired ChannelStrip by PSP	Yes	Yes
PSP MasterStrip	Yes	Yes
PSP Stereo Chorus	Yes	Yes
PSP Stereo Delay	Yes	Yes
Price	$24.99 USD	$49.99 USD

Existing original Auria customers can upgrade to Pro for $24.99 USD through Auria's in-app store. Original Auria users must install the free update to Auria version 2.0 for this option to appear.

Transfer Purchases from Auria

The recommended procedure for any Auria and Auria LE users who want to add the features of Auria Pro is to use the in-app store to upgrade. This is the simplest and least expensive way to own Auria Pro without any limitations. Any owners of Auria who choose to purchase the full version of Auria Pro instead of upgrading through the app will find that at first their Auria in-app purchases are not automatically available, due to the fact that Apple has no built-in mechanism to transfer purchases between separate apps.

To get around this limitation, there is a Transfer Purchases from Auria option in the SETTINGS menu which will transfer any Auria plug-in purchases over to the Pro version. The process can take a few moments and will cause both apps to open and close several times.

Note: For this process to work both Auria and Auria Pro must be installed on the iPad and both must be running version 2.00 or later. Make sure all purchased plug-ins are currently available in Auria before beginning this process.

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.

Part Two – Working in Auria

2

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 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

iOS 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 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 lightning connector (or MFi). There are two types of devices which can be connected through this port:

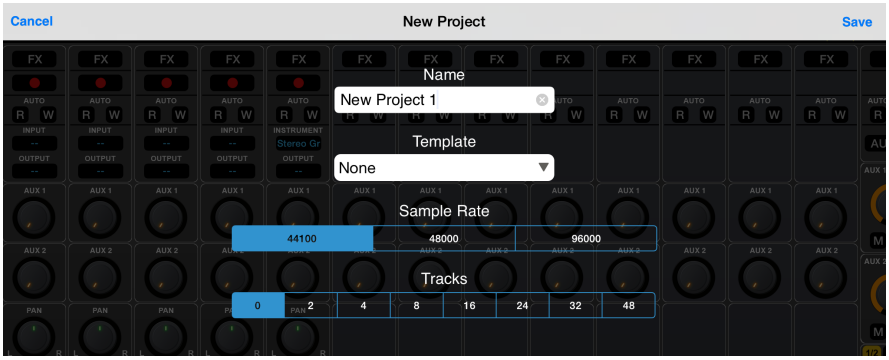
- MFi - These interfaces connect directly to the lightning MFi connector. One popular example is the Apogee Jam.
- USB Class 2 Compliant interfaces - These require an Apple Camera Connection Kit (replaced by the Lightning to USB Camera Adapter) connected between the iPad and interface. Examples include the RME Fireface UCX, Presonus AudioBox 1818VSL, and the Focusrite Scarlett series interfaces. When purchasing this optional adapter, we strongly recommend a genuine Apple adapter, as third party "clones" often cause issues with audio 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.

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
- **Template** – Select a project already set up with a specific track configuration
- **Sample Rate** – Select the desired sample rate (this cannot be changed later)
- **Tracks** – Select the number of pre-made mono audio tracks (stereo and MIDI tracks can be added later)

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

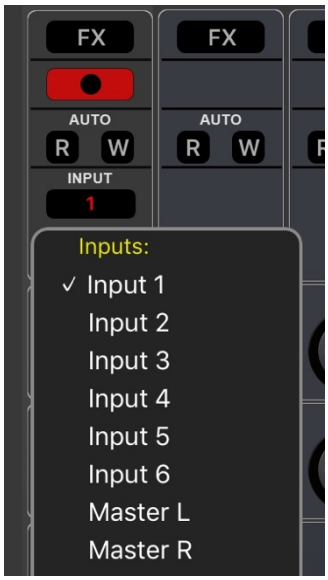
Track Setup

Recording audio or MIDI on a track requires some initial configuration of elements such as input routing and audio level adjustment.

Input routing will depend on the particular audio or MIDI hardware being used; using the built-in iPad audio system is very simple, while using a multi-channel USB interface can mean a large number of inputs to navigate. Fortunately Auria includes a simple way to assign track inputs through the INPUT box.

Input Assignment

To quickly assign a specific input to a track, first make sure it is Record Enabled, and then tap the INPUT box to bring up a list of available inputs.

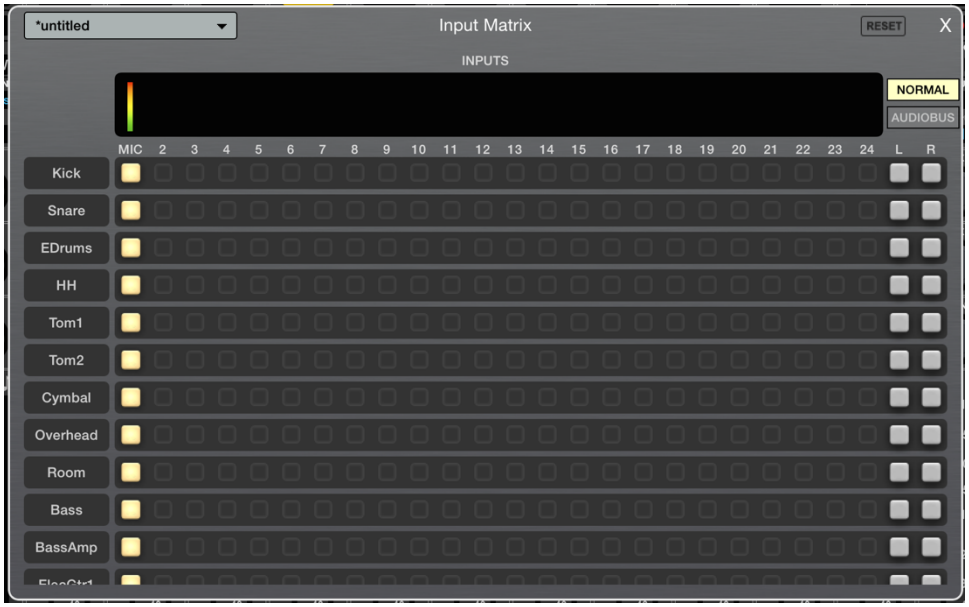


Track Input Selector, showing a 6-input USB interface currently connected.

Input Matrix

In addition to the simpler individual track input assignment, detailed above, Auria also includes a powerful pair of windows that make more complicated routing assignments straight forward, the Input and Output Matrices. This view is especially helpful when recording using a large number of inputs at once.

To open the Input Matrix 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 available 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.

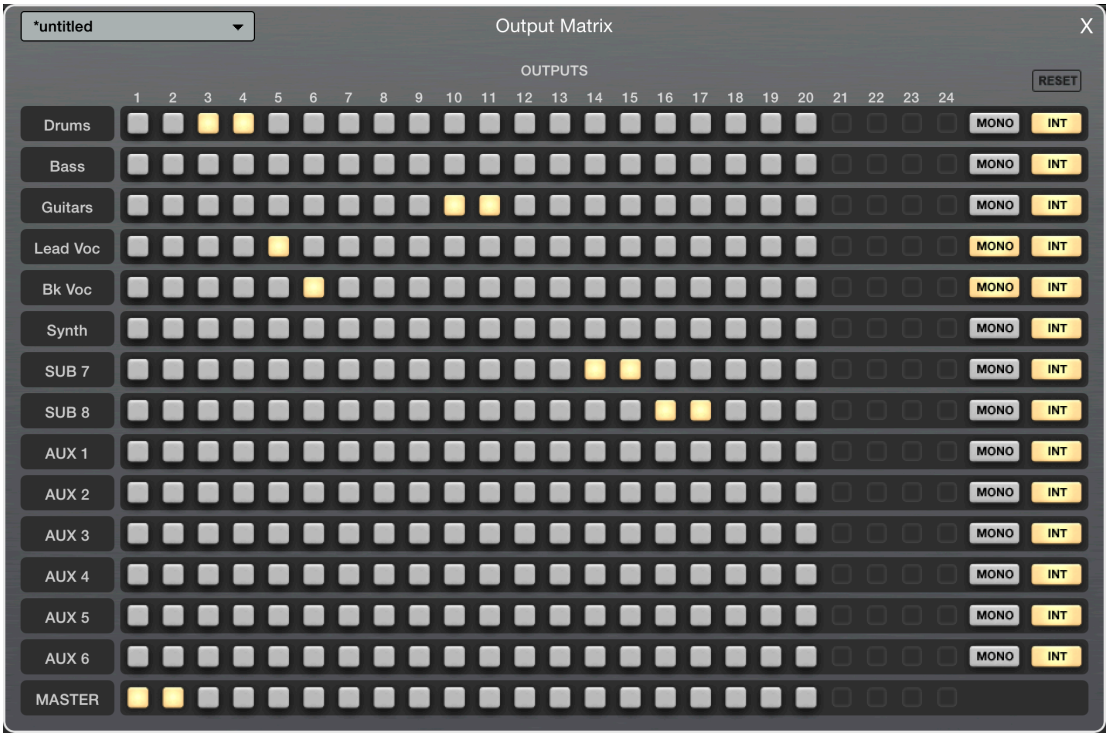
Output Matrix

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

Available outputs include:

- Subgroups
- Aux 1 – 6 (Pro Only, standard Auria has 2 auxes total)
- Master (LR bus)
- Direct Outs (Pro Only)

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



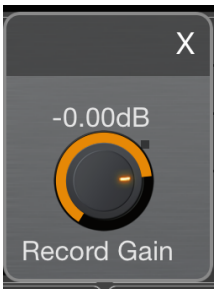
Output Matrix from Auria Pro

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 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

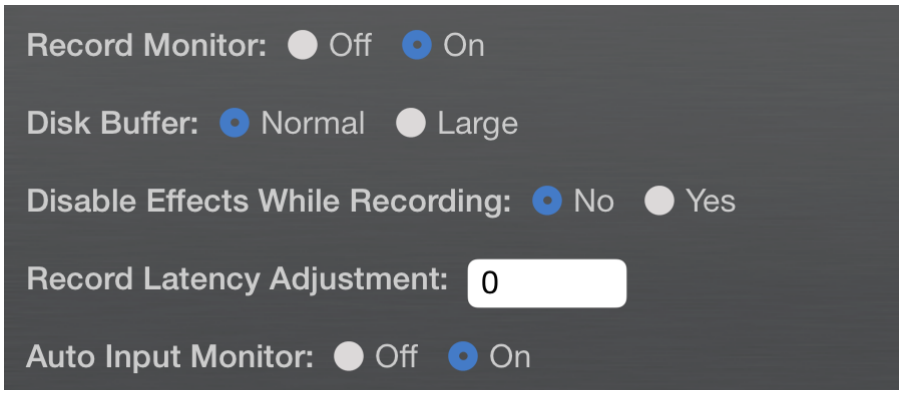
An additional option found in Auria's Recording Options pop-up (accessed by tap-and-holding the red record enable button on the top of each track) is whether to record with effects or not.

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).

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 throughout the Settings windows and are concerned with monitoring during recording.



Record Settings Tab

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.

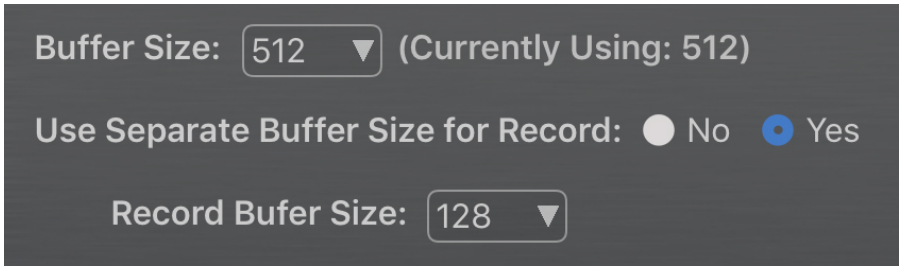
Disable Effects While Recording – Defaults to No, which means any effects present in the project will be heard during recording. If set to Yes, all effects will be bypassed when Record is pressed. Included as a means of reducing CPU and memory usage during recording.

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.

Auto Input Monitor – When enabled Auria will attempt to automatically handle input monitoring when recording.



General Settings Tab

Buffer Size – Drop-down box that selects the size of the audio buffer. The higher the number, the less CPU the system uses. 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 to lower the CPU usage. If another audio app was started before Auria its audio buffer setting has priority, and iOS may set a different buffer size; in this case the actual buffer size will be displayed in parenthesis.

Use Separate Buffer Size for Record – If set to NO then the same buffer sizes are used for both recording and playing back. If set to YES then a separate selectable recording buffer size is available.

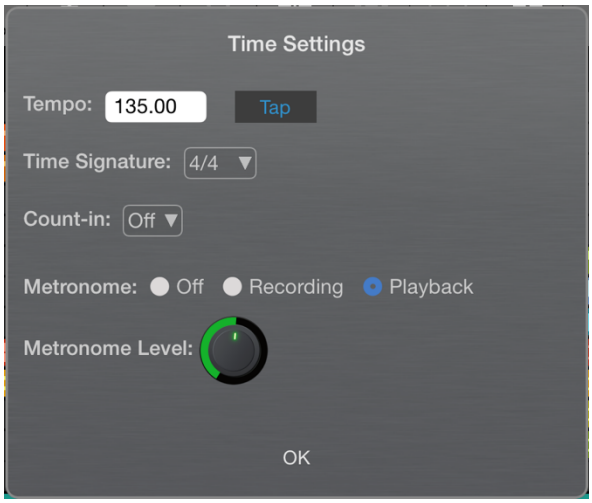
Record Buffer Size – Drop-down box that selects the size of the audio buffer when recording. This allows for using a low latency buffer size for recording and a larger and less CPU-intensive buffer for playback.

Note: Using MIDI tracks in Auria Pro requires a buffer size of 512 samples or less, as anything higher makes iOS's CoreMIDI system unusable. However, if Auria Pro detects no MIDI tracks in a project it will allow larger values (up to 4096 samples).

There is a tradeoff between buffer sizes and CPU usage, as the smaller the audio buffer the higher the CPU usage. Use the separate playback and recording buffers to enjoy low latency when recording (small buffer size) but lower CPU usage when mixing (larger buffer size).

Metronome

The metronome can be used to provide a click track during recording, and can be accessed from the Time Settings dialog (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
- **Use Tempo/Time Signature Track** – Determines whether the project's tempo track should be followed, the default setting is enabled. When enabled the other tempo and time signature settings will be unavailable. Pro Only

The metronome can also be quickly toggled on/off by tapping the Counter (in the upper-right corner) and checking/un-checking 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 of the Mixer while the project is playing, or by highlighting across the Edit window Timeline.

To set the locator in and out points from the Mixer 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 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 needing 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 Selector
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

Bounce Track in Place – Auria Pro Only

In addition to performing a traditional bounce as illustrated above, Auria Pro also allow both audio and MIDI tracks to be individually "bounced in place". This feature will perform a bounce without creating a new track. applying any effects currently set in the ChannelStrip or insert effects, as well as automatically converting MIDI tracks into audio tracks; this newly rendered audio will replace the existing audio already on the track.

To use this feature first select a specific track. Once a track is selected, tap the BOUNCE TRACK IN PLACE item from the PROCESS menu. A progress bar will appear while the new audio data is rendered.

Note: This is a destructive action. Any effects currently inserted on the track will be applied to the audio and then removed from the insert list. This is unlike the behavior of track freezing, which preserves the insert list but disables any effects on the list.

In order for Bounce Track in Place to work with MIDI, the selected track must be using either one of Auria Pro's included instruments or a connected Inter-App Audio synth. An outside keyboard connected via MIDI can only be recorded onto audio tracks via a physical iPad input or interface.

3

MIXING



The Mix window in action

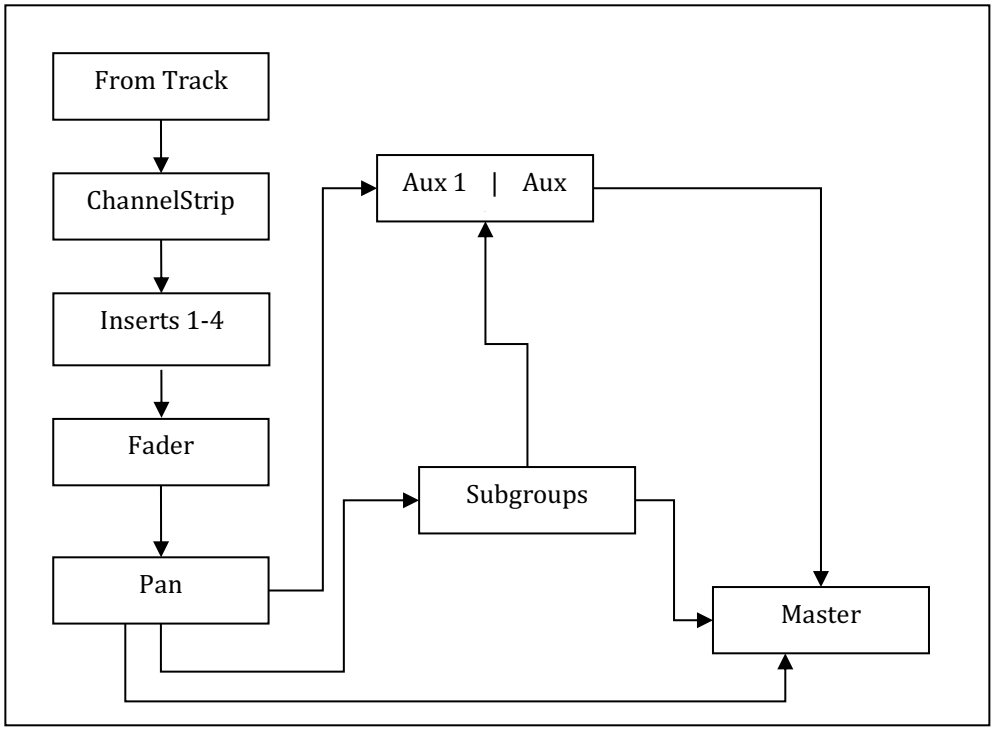
Auria Pro has within it a powerful mixing system, capable of playing back unlimited audio and MIDI tracks at once, combined with top-notch effects processing (including EQ, compressors, gates, reverbs, chorus, delay), all designed to allow final mixes that can rival those done on desktop-based systems.

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 Pro can have an unlimited number of individual channels, in any combination of mono/stereo and MIDI; 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
- Bus assignments (Pro Only)
- Direct Out (Pro Only)

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, and Convolution Reverb. 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:

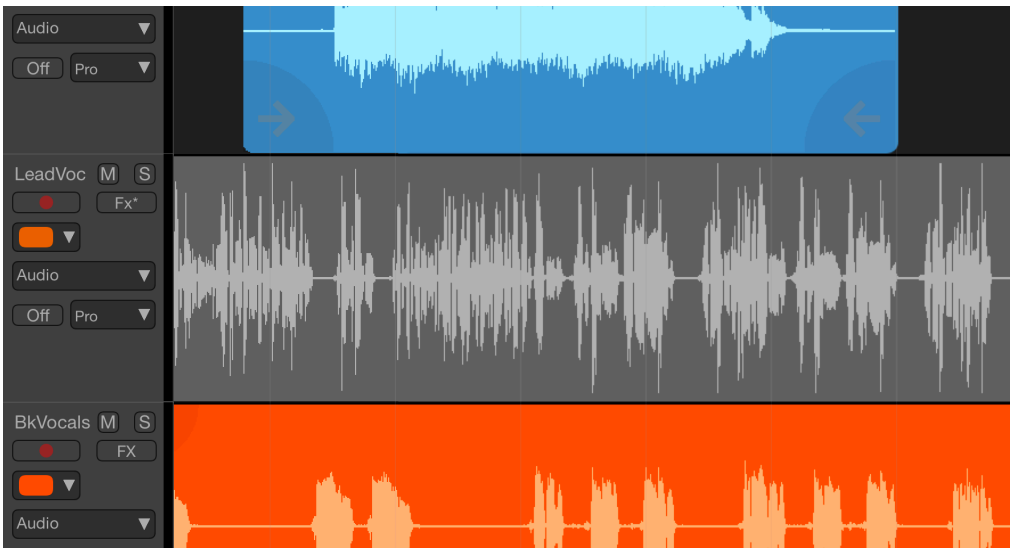
1. Open the track's ChannelStrip window by tapping the corresponding FX button, either on the Mixer or in the Edit window
2. Tap the Freeze button, found to the right of the fader 
3. A progress bar will pop-up while the track is automatically bounced in-place
4. Once the progress bar closes the track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



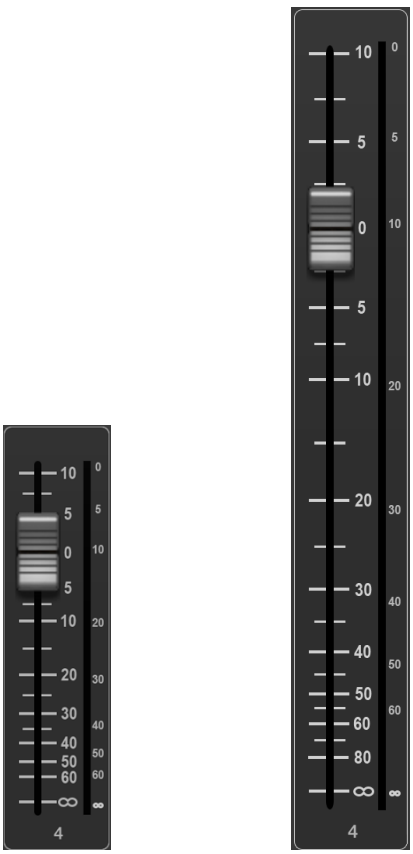
The pan pot controls the channel audio's position in the stereo field. Its behavior will depend on whether the channel contains either mono or stereo audio:

- 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).

Buses (Pro Only)

New to Auria Pro are up to 32 assignable buses (the default number of buses is eight), which can be used as output sends from individual channels to create even more flexible mixing setups than even the subgroups can offer.

Any channel output in the Mixer can be assigned to one or more buses, which in turn can be assigned to another channel's input. The really powerful part of this is that a single channel can have multiple bus output destinations at once, unlike subgroups, allowing much more flexible routing options than ever before.

Another addition is that the Aux Returns are now considered buses, so in Auria Pro any aux channel can be returned to an individual stereo channel in the Mixer, where as before all aux returns were "straight wired" directly to the Master Channel. With this change aux returns can be processed just like any other channel, so use the PSP ChannelStrip to EQ the reverb return, or compress the vocal delay, etc etc.

Assigning Buses

Buses can be assigned to both channel inputs and outputs through the Bus Selector menu. From the Mixer window, tap either INPUT or OUTPUT to open the appropriate menu.



The Bus Output Selector Menu, showing available output destinations. Buses appear after Subgroups.

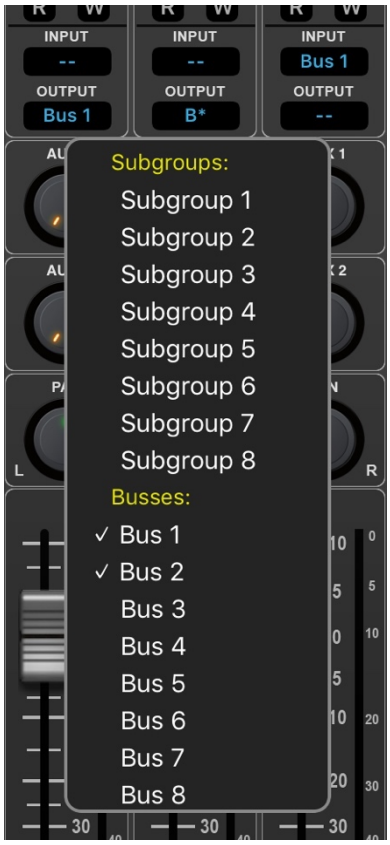
Input:

- Track – This is the default option, and simply plays back the audio content already on this track.
- Buses – Select a specific bus source.
- Auxes – Select an Aux Channel, used to perform additional processing/mixing on aux returns.

Output:

- Master L/R – This is the default option, which sends the channel's post-fader output directly to the Master Channel.
- Subgroups – Sends the channel to the selected Subgroup.
- Buses – Sends the channel to the selected buses; tap each bus destination to enable that route. Can be used to send a single track to multiple bus destinations.
- Direct Out – Sends the channel's pre-fader output directly to a specific hardware output.

Note: When a track is record-enabled a different menu of inputs is displayed, listing the available physical inputs on the iPad and optional USB audio interface.



Multiple buses outputs selected at one, note the B displayed under OUTPUT*

Note: When more than one output bus is selected, the OUTPUT label will read B*, meaning two or more bus outs are currently active on that channel.

Managing Buses

Auria Pro defaults to eight total buses but can support up to 32 at once. To configure Auria Pro's buses, tap open the MANAGE BUSES window from inside the Mixer Settings window.



Manage Buses

- Add Bus – Tap the + sign to create an additional bus, either using the default name or creating a custom-named bus.
- Delete Bus – Select the bus to be deleted, and then tap the – sign.

Subgroups

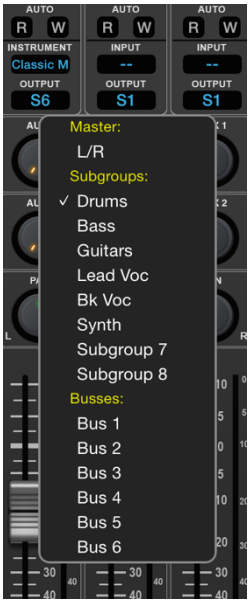
Auria's mixer contains eight separate subgroups which can be used for advanced routing and bus processing. It works by re-routing the channel's output to a specific subgroup, instead of directly to the master channel, allowing selected channels to be summed and processed together first before being bussed 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 subgroup 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. Auria Pro's new bus system does support multiple bus destinations, however.

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.

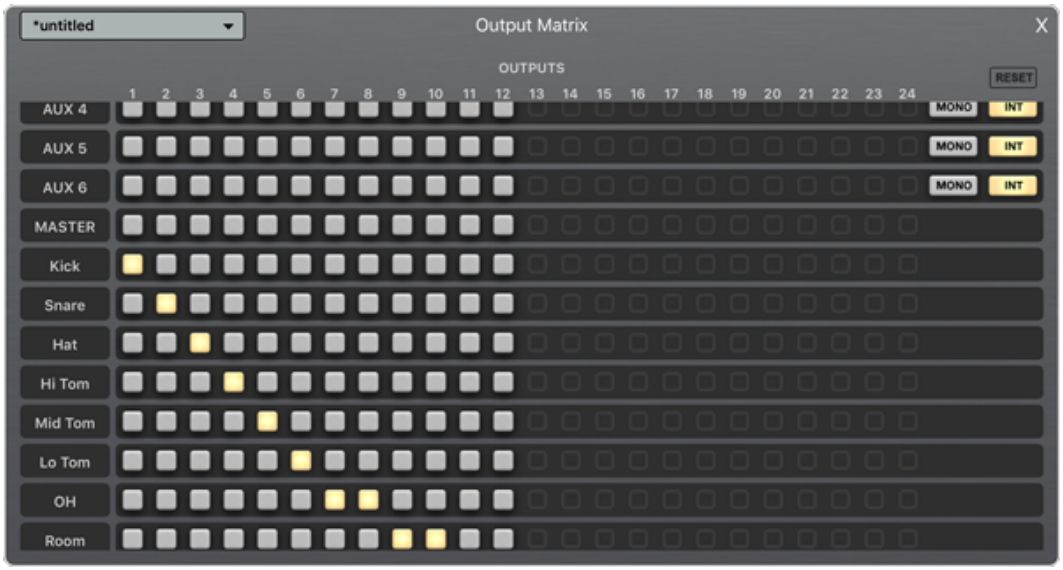
Direct Outputs (Pro Only)

Starting with Auria Pro v2.01, individual tracks can have their pre-fader output routed directly to an externally connected audio hardware's output. This is useful whenever an individual track needs any form of external processing, such as a hardware compressor or EQ. It can also be used to allow mixing tracks for Auria Pro in an external mixer, i.e. mixing "outside the box".

Enabling a track's direct output can be done either from the individual track's Output Selector or via the Output Matrix. Both mono and stereo tracks can use direct outs, however a stereo track requires two hardware output channels versus a mono track's single output.



Assigning a stereo track's direct outs to hardware channels 5 and 6.



Assigning multiple tracks direct outputs to a multi-channel USB audio interface via the Output Matrix

Note: All direct outputs are pre-fader, so the signal level being sent out is not affected by the channel fader; the ChannelStrip and any insert effects will be applied, however.

Auxiliaries



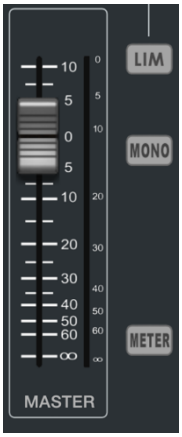
Auria Pro has six stereo auxiliaries, while standard Auria has two, with sends available on every channel and subgroup. By default these are post-pan sends, which means both the fader and pan pot will directly affect the signal sent to the auxiliary sections; either aux channel can also be switched to 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 visible Aux 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 Aux Return knobs, also found on the Master channel. Below the Aux Returns in Auria Pro are the buttons to toggle between the three banks of Aux channels.

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



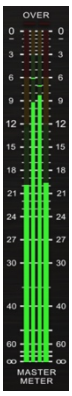
In addition to the MasterStrip's limiter module, the Auria Master Channel also includes a distinctly brick-wall style limiter. Tap the Limiter tab to display the following controls:

- 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

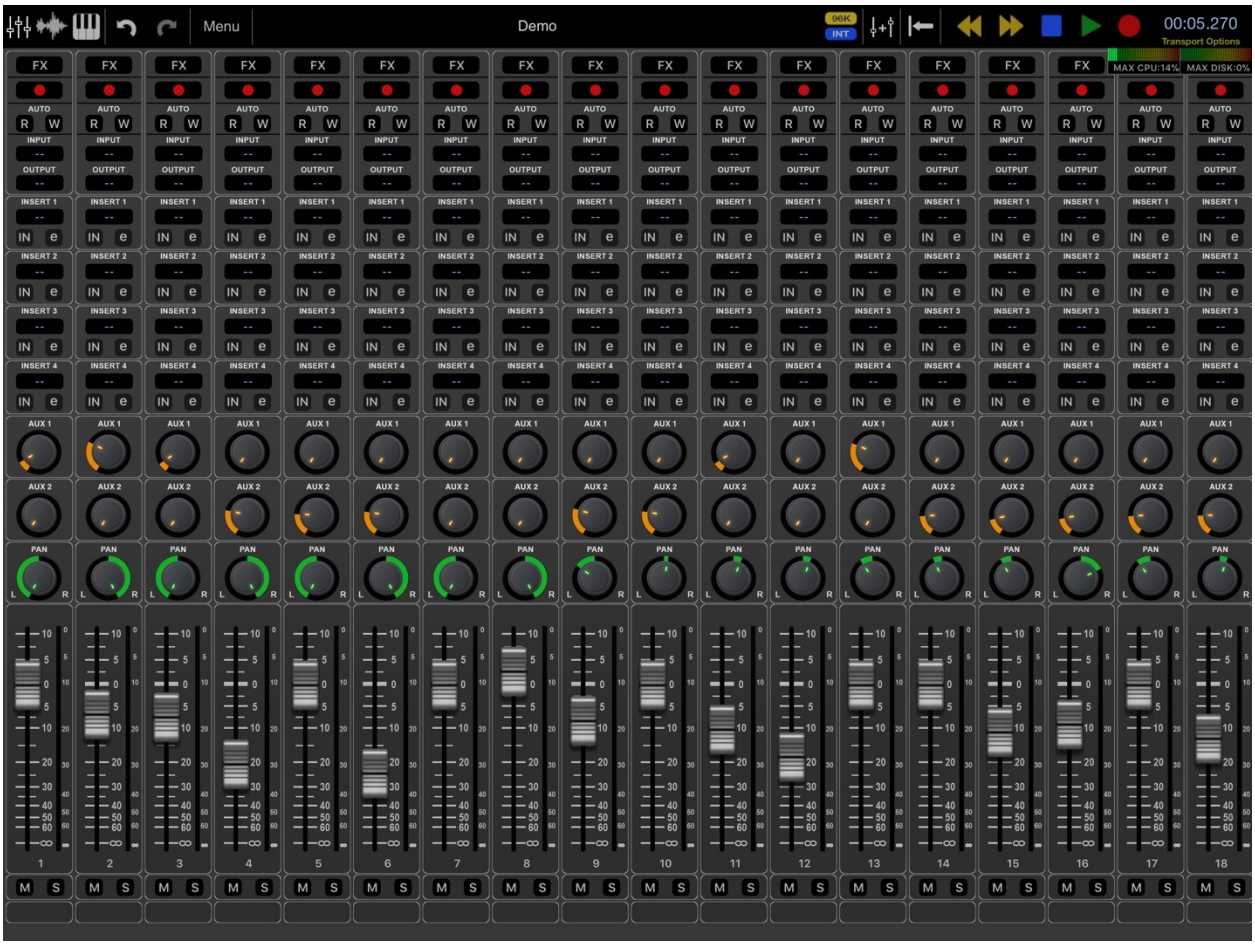


Tapping this button will display a more detailed master channel meter. The meter is designed to display both peak and RMS at once: peak in the middle, RMS on the outside, with a peak-hold display showing the loudest peak so far. All meters display dBFS (full scale) values.

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

iPad Pro Screen Optimization (iPad Pro models only)

For users of Auria and Auria Pro exclusively working on iPad Pro models (2016 and later) the settings page contains a screen optimization option to take advantage of the large screen space. In this mode, Auria's Mixer page will display the first 4 inserts for each channel strip to assist with a faster workflow when mixing. The editor page is also adjusted to allow for more visible tracks and icons. This setting is not available on non-Pro iPad models.



Auria Pro's Mixer shown with the iPad Pro Screen Optimization setting turned on

To toggle the iPad Pro Screen Optimization settings, navigate to the Auria Settings (Menu> Settings). The on/off switch will be found towards the bottom of the General section. The Screen Optimization can be turned off if you would like to retain the original Auria Pro graphics in a larger size on the iPad Pro screen.

4

EDITING

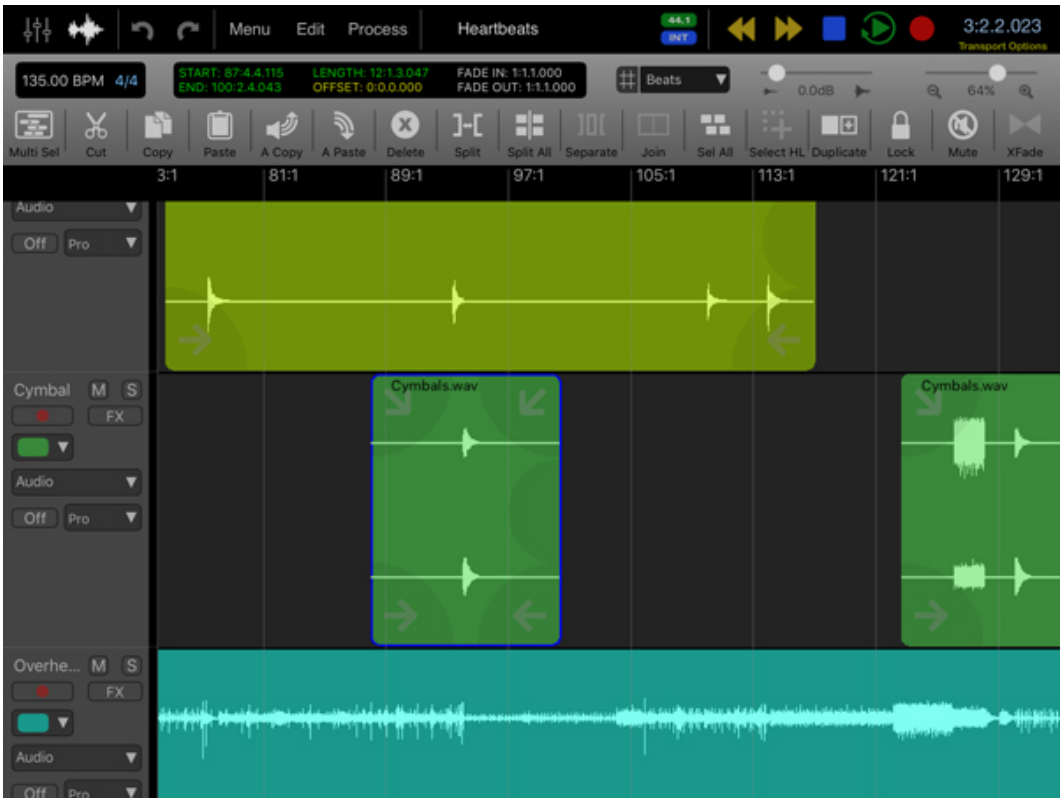
Auria contains a powerful non-destructive editing system that can handle just about any required editing task. This chapter will discuss in-depth how to accomplish various edits within Auria.

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 region selection

To de-select a region simply 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

Tap each object once to add it to the selection; each will become blue in turn

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

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)

Tap open the Edit Menu

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 both the Edit Menu and the Icon Bar, 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

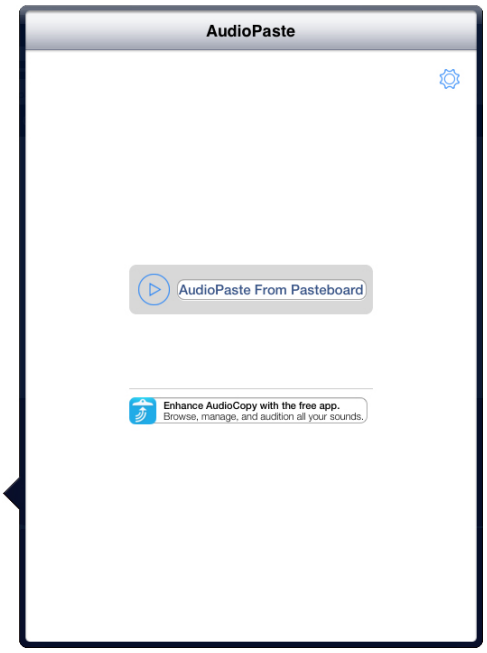
The AudioCopy dialog will open

Tap Copy Audio

Wait for the selection to copy

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

Back in Auria tap the Edit Menu and select AudioPaste

The AudioPaste dialog will open

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)

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)

Tap open the Edit Menu and select Lock Region

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.

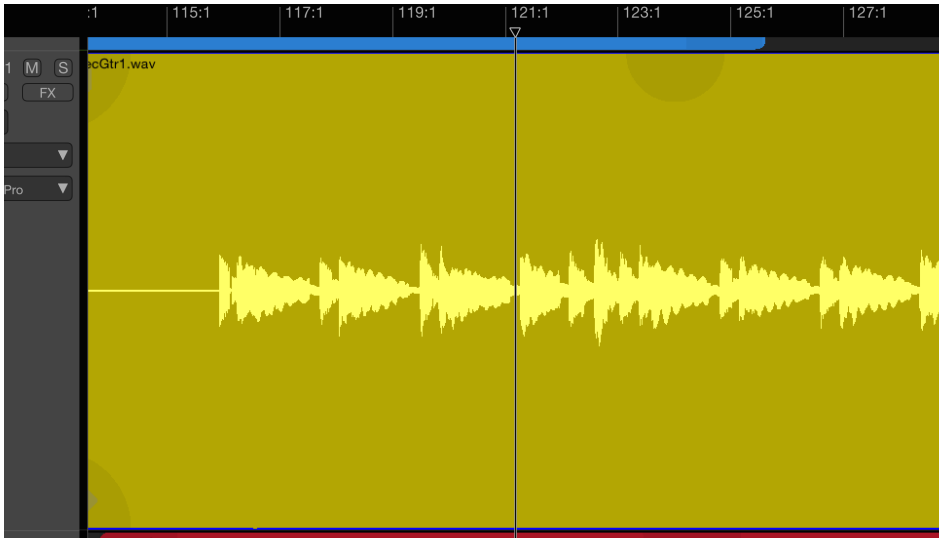
Splitting Regions

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)

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

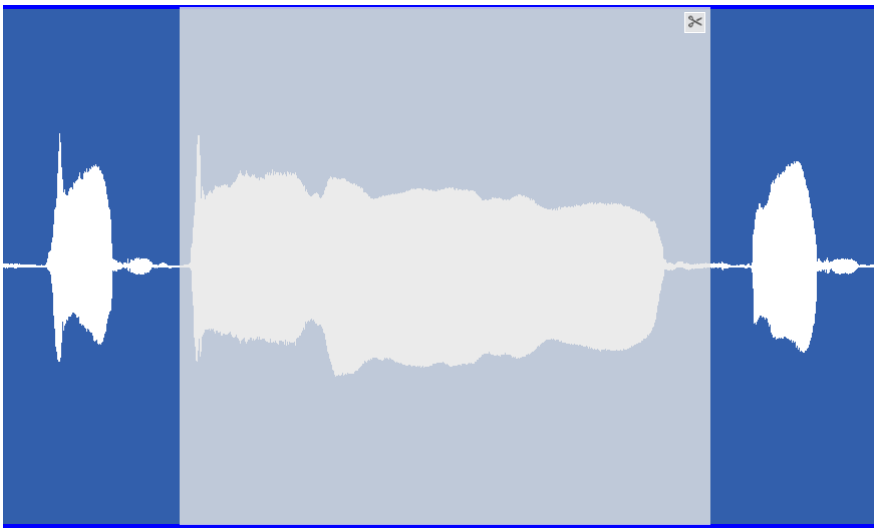


A region ready for splitting

Once the cursor is in the desired split point tap open the Edit Menu
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)

Swipe finger horizontally across the entire desired area

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.

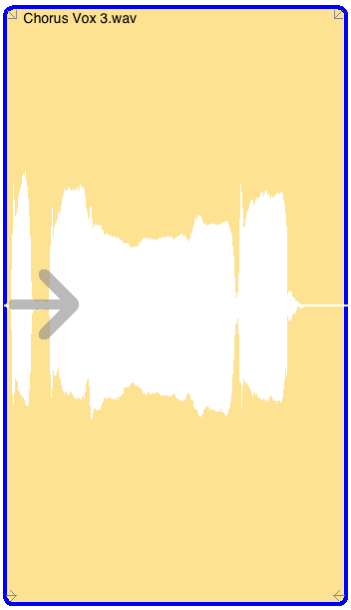
Every region in Auria has a trim handle at both its beginning and end, represented by an inward-facing arrow. They are in the bottom corners of the region.



To use a trim handle:

1. Tap and hold near the arrow icon in the corresponding bottom corner (left for beginning, right for end)

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

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

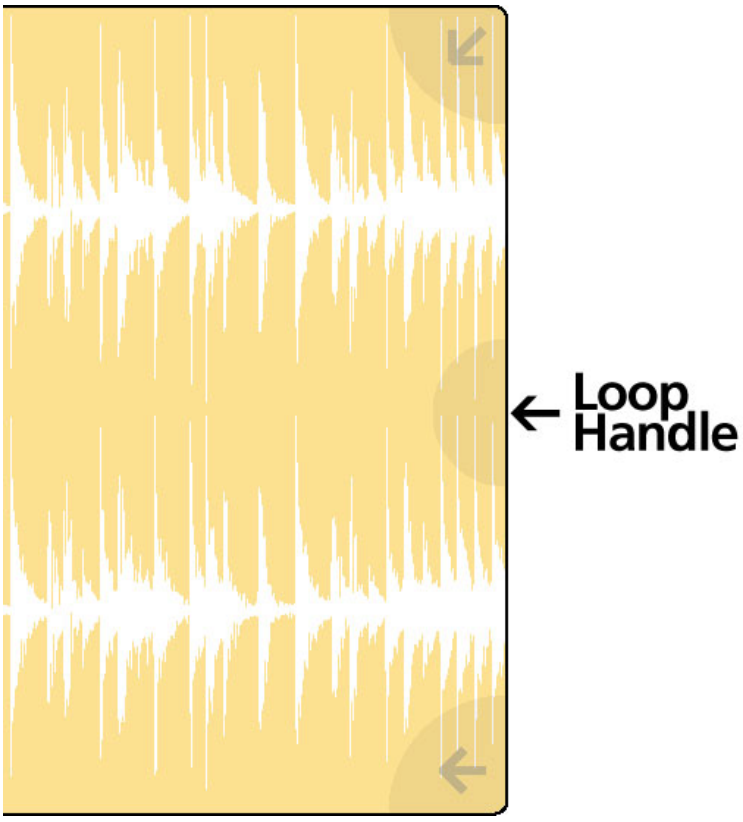
Note: Handles are only visible when zoomed in adequately far enough to clearly display the handles, so if the handles aren't visible try zooming in farther.

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

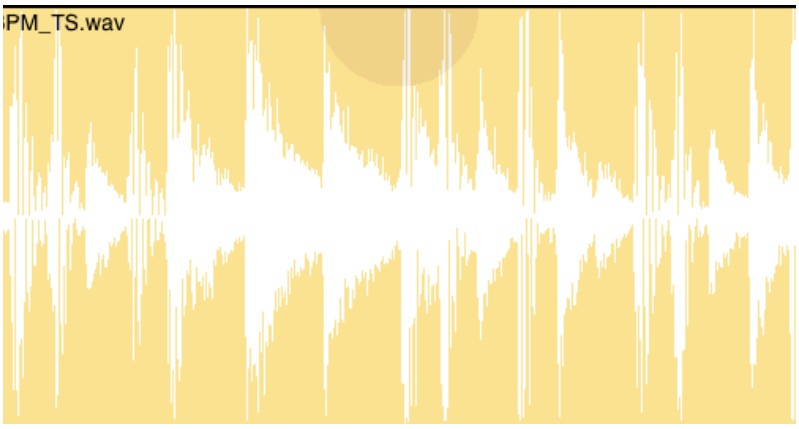
To automatically duplicate a region:

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:

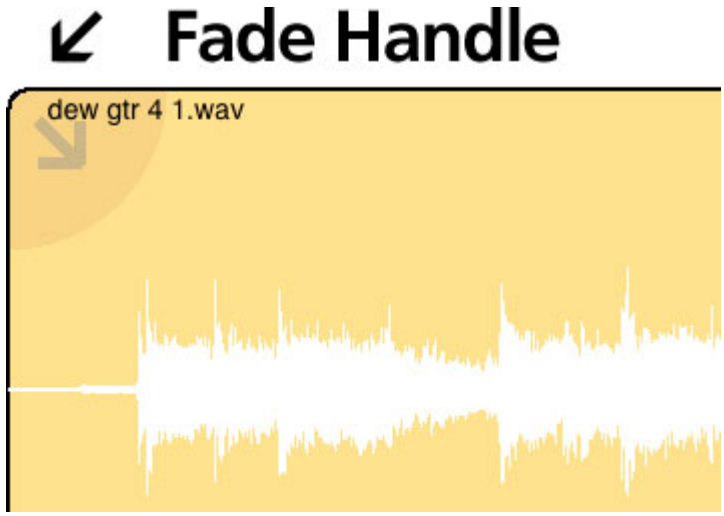


Gain Handle shown at the top-middle of the region.

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.

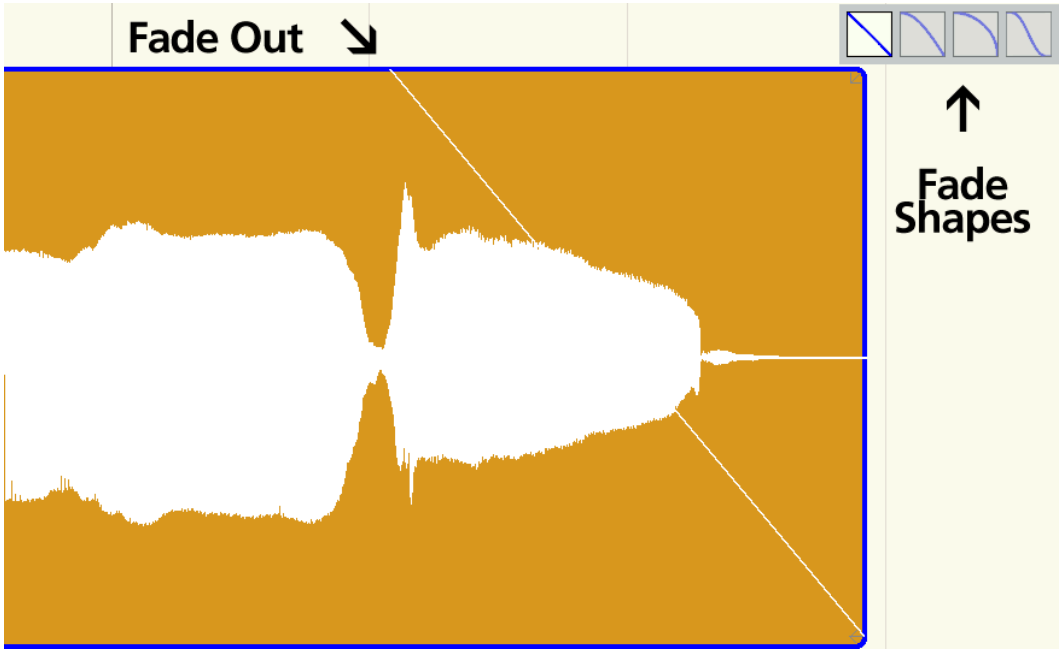
Fades

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.

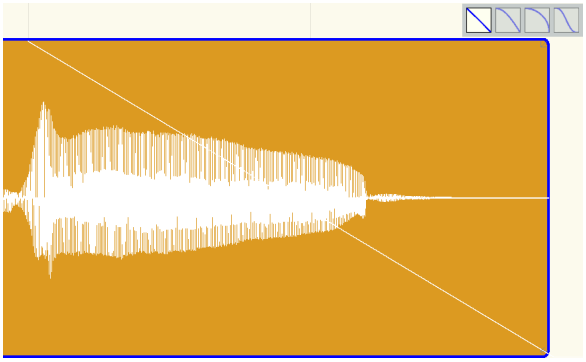
To create a new fade:

1. Tap and hold the arrow icon in the corresponding upper corner (left for fade in, right for fade out)

The region's outline will turn blue and a diagonal fade line will appear
Swipe in the direction of the desired fade, the fade will be drawn along with the swipe
Release the corner of the region to finish the fade

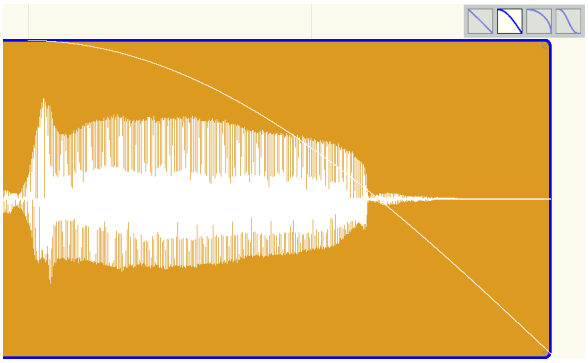
Note: If you find it difficult to touch the fade arrow itself, keep in mind that you can tap anywhere near the corner and not just the arrow itself; give yourself some room and aim further inside the region's corner.

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:



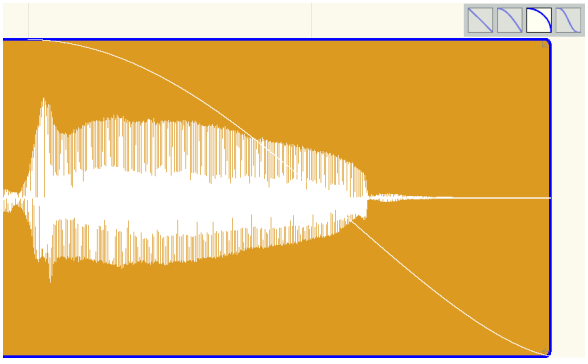
Fade 1 – Linear (default)

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



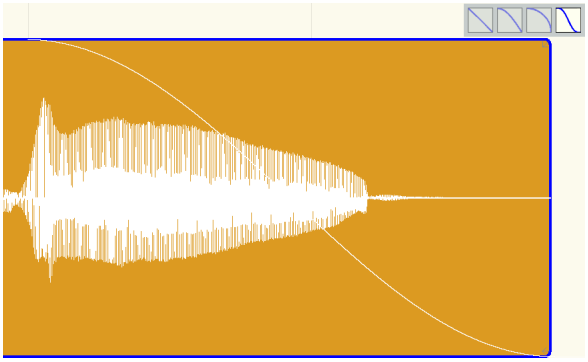
Fade 2 – Exponential (Slow)

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.

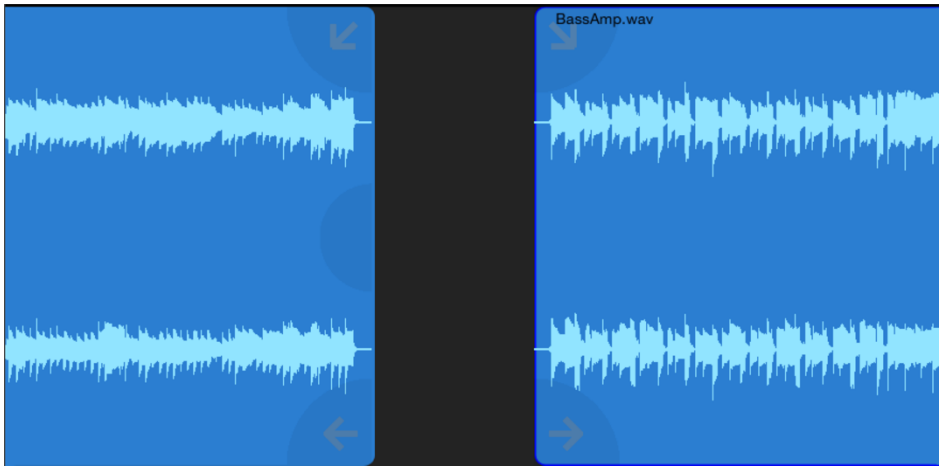
Crossfades

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.

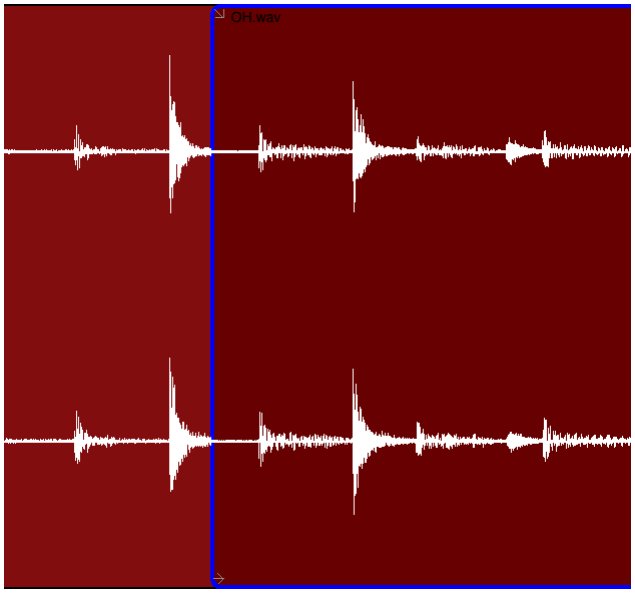
To create a crossfade between these two regions:

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



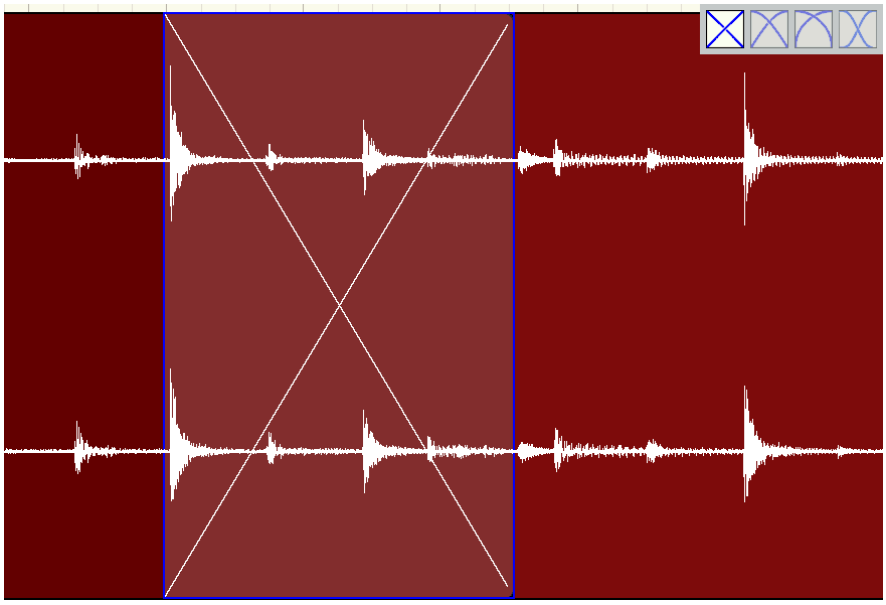
Two regions before a crossfade. Both have had their beginning/end trimmed as needed.

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

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



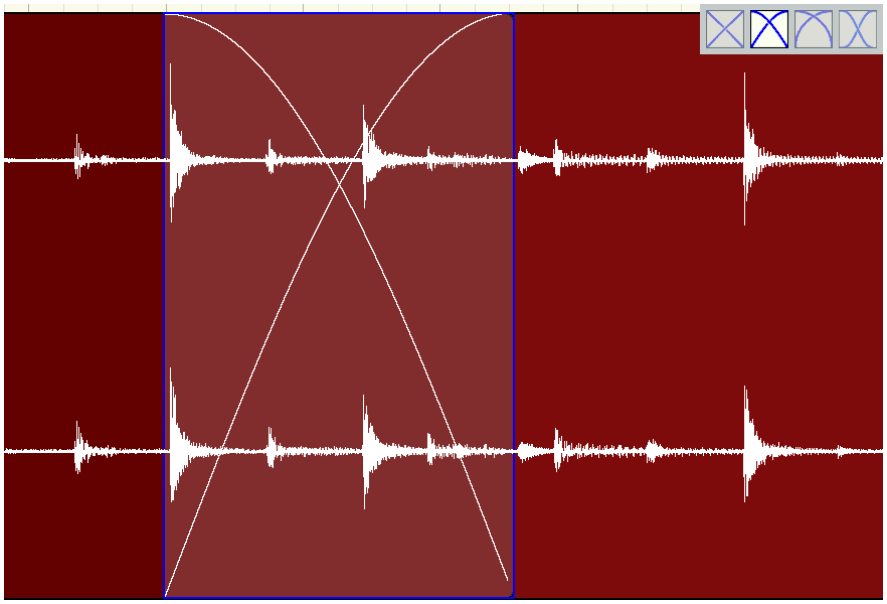
Equal Gain (Linear) Crossfade

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.

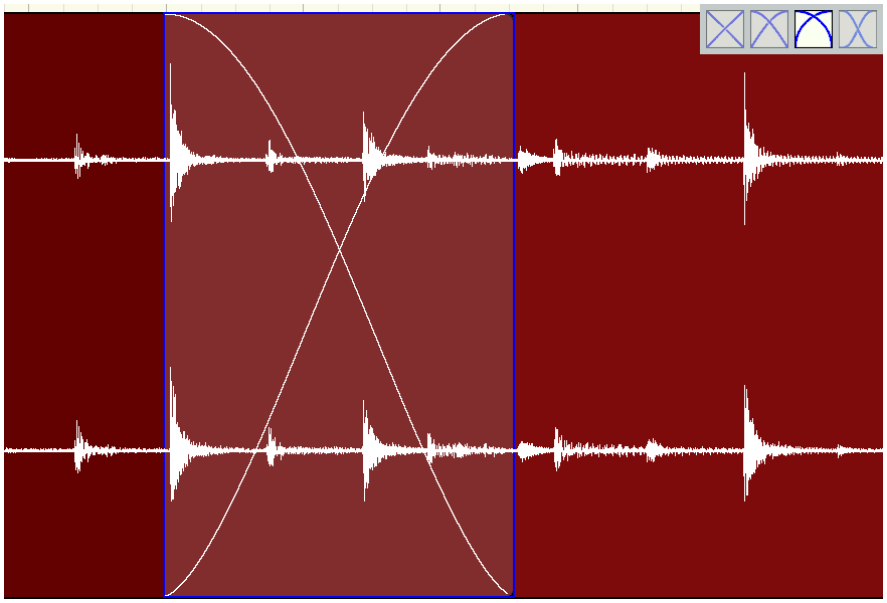
Equal Gain (shown above) – The two regions change gain at a constant rate so that the sum gain is equal throughout the crossfade. Best when working with very similar pieces of audio (i.e. correlated), like editing together two takes of the same guitar part.

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 .



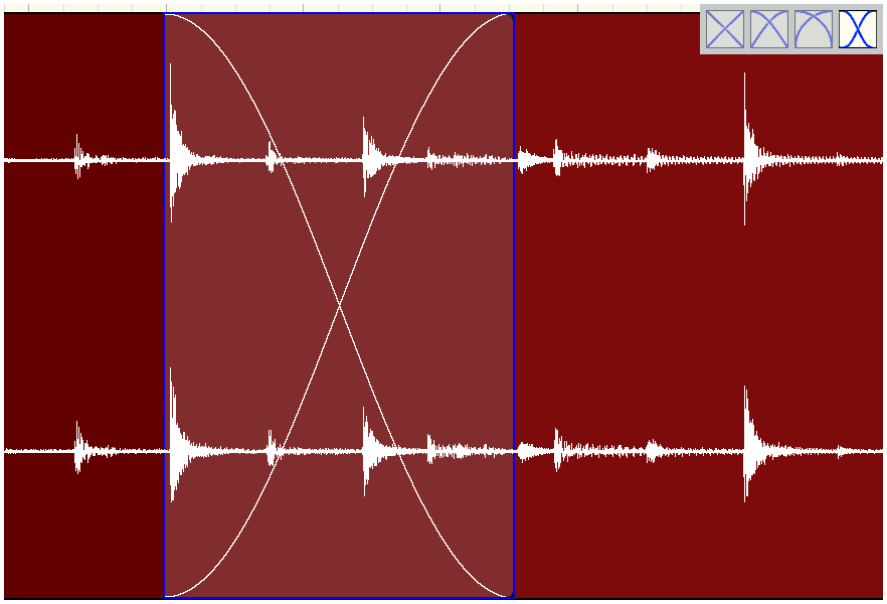
Equal Power

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.



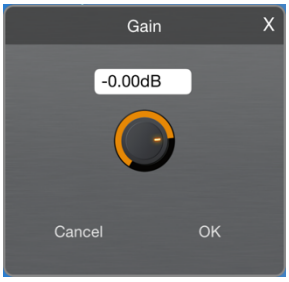
Equal Gain (S-Curve)

Destructive Processing

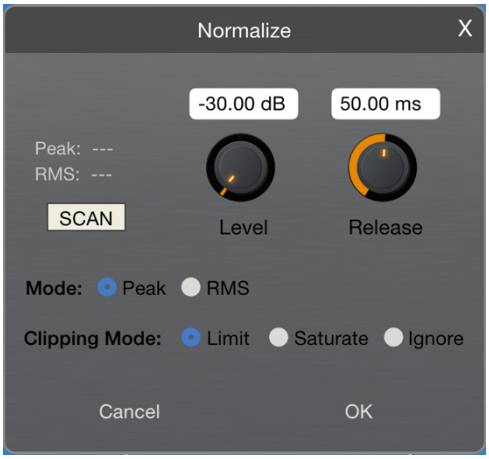
In addition to Auria's non-destructive editing, there are a suite of audio tools which will destructively change the actual audio content itself (though Undo will still work!)

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.
- **Pitch** – Utilizes élastique Pro's pitch-shifting algorithm to either raise or lower the selected audio's pitch. See the Process Menu section for more information.
- **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.
- **Bounce Track in Place** – Mixes-down selected track and applies any current non-destructive effects or processes, making them destructive. The newly processed track will take the place of the original. Can be used to bounce instruments on MIDI tracks. Pro only.
- **Use as Groove Template** - Copies the selected events (like Warp Markers or MIDI notes) to the Groove Clipboard, making it available when utilizing Groove Quantizing. Pro only.

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 algorithms come from both the well-known ZTX 3 Pro signal processing suite as well as élastique's algorithms

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 ZTX 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

"ZTX 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."
- ElastPro, ElastEff, & ElastSolo – these three élastique modes are described in full in this section.

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.

5

MIDI SEQUENCING – PRO ONLY

Auria Pro adds a full MIDI sequencer and built-in instruments to Auria's existing audio recording and editing capabilities. There is a new MIDI-type track available which will record MIDI from a variety of sources, a Piano Roll editor specifically designed to edit MIDI parts quickly and easily, a collection of helpful musical functions which can be applied to existing MIDI selections, and the capability to play back these MIDI tracks using both the included MIDI instruments as well as 3rd-party iOS instruments.

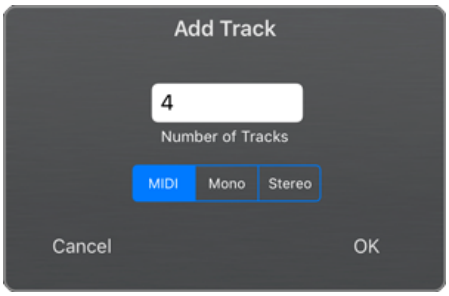
This chapter assumes the user is already familiar with Auria's Mixer and Transport controls, so please refer to these earlier sections to first become familiar with basic Auria operations.

The MIDI track

The best place to start is to look at the new type of track available in Auria Pro, the MIDI track. Certain elements, like the volume fader, mute and solo buttons, aux sends, etc, function exactly as their audio track equivalents, so this section will concentrate on the MIDI-specific differences.

Creating MIDI tracks

New MIDI tracks are created in the same fashion as audio tracks, simple add them using the ADD TRACK function inside the Main Menu. Enter the number of desired tracks and select the MIDI track type.



Adding new MIDI tracks

Once a new, empty track has been created it is ready for recording. MIDI data can be recorded from a number of different sources, such as an external MIDI controller or keyboard, a 3rd-party instrument, or from one of the built-in instruments like Lyra, WaveMachine Labs' sample player;

Lyra is the default assigned instrument for any new MIDI tracks. For more in-depth information on Lyra see the chapter on Auria Pro's included MIDI instruments.

Recording MIDI from Lyra tutorial

To record onto a MIDI track using Lyra's on-screen keyboard, start from the Mixer window and choose an instrument from Lyra's list of available samples. On the newly created MIDI track tap the INSTRUMENT box to bring up the list of currently installed samples (and there are additional free and paid samples available in Auria Pro's in-app store), and tap an available instrument from the list under LYRA SAMPLER. Tap the regular red Arm Track button to enable MIDI recording on this track.

Lyra's on-screen keyboard is part of its graphic interface, which can be opened quickly by simply double-tapping the INSTRUMENT box.

The playable on-screen keyboard is at the bottom of the Lyra interface window.



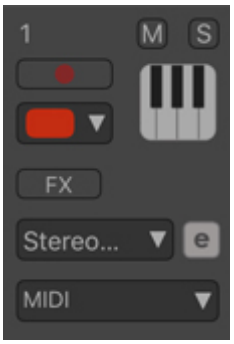
Lyra's graphical interface and on-screen keyboard

As long as the track is record-enabled, touching the keys should play the selected instrument. Tap the RECORD BUTTON in the upper-right corner TRANSPORT followed by the PLAY button to start

recording. Play a few notes in during recording, and then tap the STOP button when done. Rewind to the beginning and hit play to listen back to the new recording.

MIDI tracks in the Edit Window

Auria Pro's Editor Window will show both existing audio and MIDI tracks in the track panes. In addition to the standard Auria controls, such as MUTE and SOLO, there are a few new MIDI-specific selections available on MIDI tracks.



MIDI track

- The musical keyboard button switches the view to the Piano Roll Editor
- The FX button opens the Real-Time MIDI CONTROL parameters (and ChannelStrip)
- The INSTRUMENT drop-down selects the desired MIDI instrument and patch/sound (default selection is STEREO GRAND piano)
- The e button next to the INSTRUMENT drop-down opens the selected instrument's graphic interface

Note: If not all of these controls are visible in the Editor then the track heights need to be increased by zooming vertically.

Also note: While Auria Pro allows audio buffer sizes up to 4096 samples when using audio tracks, the use of MIDI tracks requires a buffer size of 512 samples or less. If an existing audio-only project is set to a buffer size of 4096 but later has a MIDI track added to it its buffer size will automatically be changed to 512.

Real-time MIDI Control

Once a MIDI track contains some MIDI data it can be played back using a number of different changeable parameters which control the track's overall sound. This is accessed by tapping a MIDI track's FX button from either the Mixer or Editor windows – just like opening the ChannelStrip on an audio track.



From here many different MIDI parameters can be adjusted

The available parameters are grouped together into different sections, which will be described below. For more detailed information on each individual control please refer to the real-time MIDI parameters section in Part Three – Reference of this guide.

- **INSTRUMENT** – Select an available MIDI instrument and patch/sample
- **PORT** – Select a specific MIDI channel and make input/output port assignments
- **PARAMETERS** – Includes controls for functions like transposing notes, adjusting velocity levels, and adding randomness; each of these functions can be adjusted and heard in real-time
- **QUANTIZE** – Adjusts the rhythmic timing of the track's playback to conform with a selected grid, and can be used with either a strict grid or with more complex groove templates, all in real-time
- **LEVELS** – Adjusts the track's MIDI volume and MIDI panning

Note: The MIDI Volume and Pan controls work differently than the normal volume fader and pan knob found in the Mixer Window (as well as on the right side of the ChannelStrip). The MIDI versions of the controls affect the playback of the selected instrument, while the mixer version simply controls the track's audio volume and panning. One example would be when creating a custom patch in one of FabFilter's included synths it is possible to overload the output of the synth (through "creative" use of the filters and envelopes) so that it sounds distorted. Lowering the track's volume (audio) fader from the Mixer window will simply make the already-distorted signal

softer; instead, lower the MIDI volume of the track until the synth is no longer overloading and then raise the track's audio volume fader to compensate.

Groove Quantizing

Groove quantizing works similarly to normal quantizing, except it utilizes a MIDI-note pattern instead of the strict beat grid. This can be used to give a MIDI track more complex rhythmic feel than by simply adding swing, and Auria Pro comes with several built-in groove templates by Numerical Sound, the makers of DNA Groove Templates. For specific information on the individual parameters in this section please refer to the real-time MIDI Parameters section.

The built-in groove templates also have different variations available for each groove. First select a specific template from the drop-down list.

Groove templates list:

- 16th Feel – Straight ahead feel
- Medium Shuffle – Syncopated feel
- Triplet Shuffle – For adding syncopation to triplet-based music (i.e. 6/8 and similar time signatures)
- Triplet – Straight ahead feel for triplet-based music

Variations list:

- Main – Primary groove
- Ahead – Slightly ahead of the beat
- Pushing – More pronouncedly pushing ahead of the beat
- Behind – Slightly behind the beat
- Laid Back – More pronouncedly laying back behind the beat

Capturing custom grooves

In addition to the built-in grooves templates and the optional add-on DNA Groove packs available in the In-App Store, custom grooves can be captured from existing Auria Pro projects and used on separate MIDI parts.

1. Select an existing MIDI region to use as the template
2. Tap USE AS GROOVE TEMPLATE from the Process menu to copy the MIDI data to the clipboard
3. Open the Groove Quantizing panel on any desired MIDI track
4. In the Groove Selector drop-down tap CLIPBOARD
5. Assign a RESOLUTION setting for this groove, typically equal to the smallest duration note found in the original MIDI selection
6. Tap SAVE to name and remember this custom groove template for later use

Piano Roll Editor

While overall MIDI playback can be adjusted using Auria Pro's various real-time parameters, editing individual MIDI parts is a basic element of any MIDI sequencer. Auria Pro has a dedicated MIDI editor which provides precise control over adding, deleting, and moving MIDI notes and controller data. This editor is based on the now-ubiquitous piano roll-style editor, which has its roots in player pianos going all the way back to the late 19th Century.

Opening the Piano Roll

Double-tap an existing MIDI region to automatically open it inside the Piano Roll editor. Or, tap the piano button found on the MIDI track (zoom in vertically if at first this button is hidden from view):



The Piano Roll icon

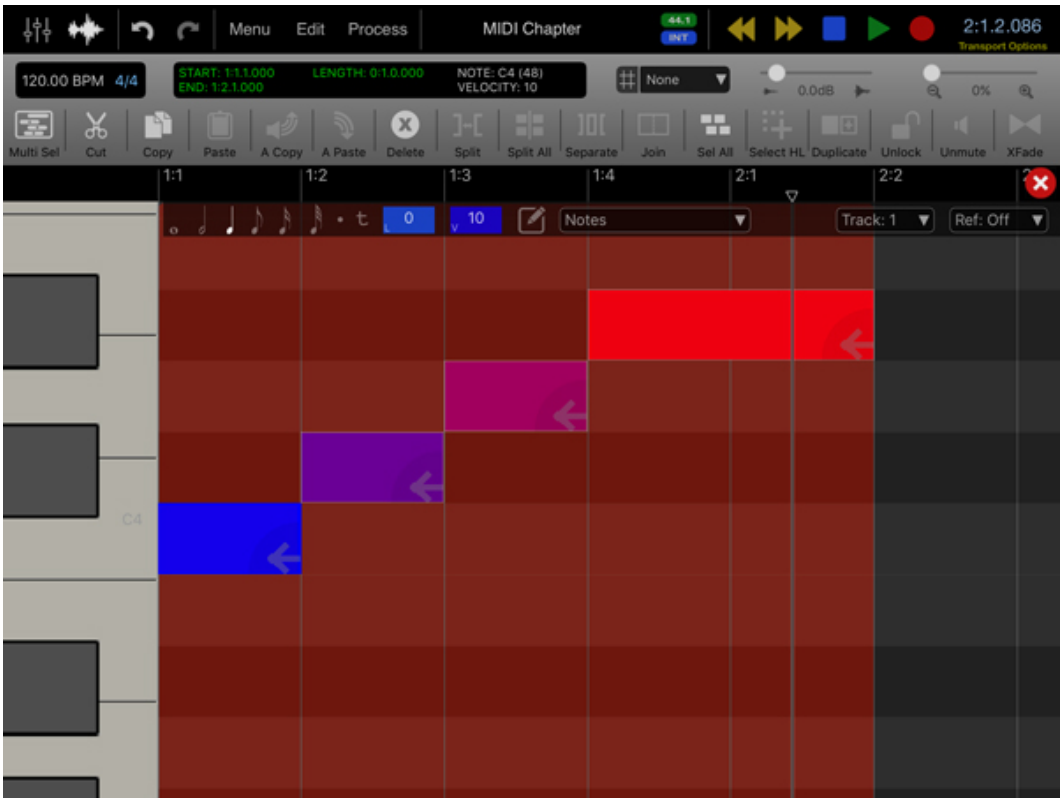
Description of the Piano Roll

The Piano Roll shares some similarities with the main Editor view in that there is a time ruler running along the top of the Piano Roll, displaying time in musical bars and beats; events happening earlier in time appear to the left on the window, later events are further to the right, and the current playhead time is shown with the vertical cursor. Each MIDI note is represented by a small, horizontal rectangle drawn on the window.

Unlike the main Edit Window, however, there is a sideways piano keyboard running along the left side of the Piano Roll, indicating that musical pitch is represented vertically, with notes going up in pitch from bottom to top.

When in the Piano Roll editor the default gestures for panning (both Up/Down and Left/Right) and zooming the view work differently then elsewhere in Auria's Editor. By default the Piano Roll cannot be panned around using a single finger – as this will cause new notes to be accidentally added; use two fingers (like a normal pinch zoom) to both adjust zoom level and pan up/down or left/right by sliding the two fingers together in the desired direction.

Please note that the currently selected Piano Roll editing mode affects the way pan and zoom gestures work; refer to the detailed sections below for the differences between Draw mode and Standard mode.



The Piano Roll with four notes visible

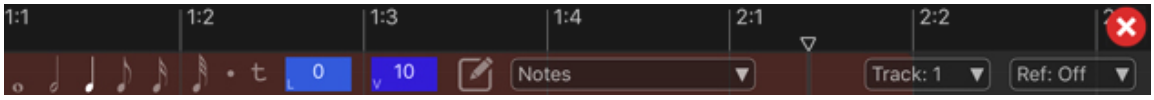
MIDI information is made up of a stream of multiple messages that, when combined together, tell a MIDI instrument when and how to playback musical events; MIDI is actually made up of several different types of messages:

- Note On – The time when a specific note is supposed to start playing; also includes the pitch and velocity of the struck key
- Note Off – The time when a currently playing note should stop playing; also includes the note's pitch
- Other "non-note" messages, like pitch bend and other controller data

The left edge of the rectangle in the Piano Roll represents the start of a note (Note On), the right edge represents the end of the note (Note Off), so therefore the length of the rectangle is the note's duration. The note's pitch is represented by the vertical orientation of the rectangle; the rectangle will line up with the corresponding piano key on the left. Finally, the color of the rectangle represents the note's MIDI velocity, running the spectrum between blue (small velocity value) and red (large velocity value); except for the rarest of cases, MIDI velocity determines a note's loudness – the bigger the velocity value, the louder the note.

Note Bar

Just below the Timeline Ruler are a series of various buttons and sliders used to modify MIDI notes and their data.



Located below the Timeline Ruler, the Note Bar controls allow both quick and precise changes to MIDI data

These controls can be divided into several different sections:

- **Note duration selectors** – tap a note's picture to select that corresponding duration; Whole, Half, Quarter, dotted, triplet, etc
- **Note length slider** – For when more varied control is needed over note length than the left-hand preset values allow, touch and hold this box and slide up or down to adjust note length
- **Velocity slider** – touch and hold this box to change velocity, sliding up or down to adjust values. If multiple notes are selected then this will adjust all of their velocities relative to each other, i.e. adding 10 to each note's velocity value
- **Draw mode** – This pencil icon toggles between separate editing modes, detailed below, which allows for fast note entry and editing. Draw mode is ON by default
- **View selector** – Drop-down to select between viewing MIDI notes, pitch bend, aftertouch, controllers
- **Track selector** – Drop-down box for switching between different MIDI tracks by tapping corresponding track number
- **Reference track** – Drop-down box to select a separate MIDI track to superimpose underneath the current track, to use as a reference when editing MIDI

Draw vs. Standard modes

The Piano Roll offers two distinct modes for editing MIDI notes, the default Draw mode and Standard mode. Standard mode works essentially the same way as Auria's main Editor window, allowing notes to be manipulated using the same gestures as audio regions. Draw mode uses different gestures to quickly add, delete, and change MIDI notes. Most of the standard edit functions, both those found along the Icon bar and those inside the Edit menu, will work when using the Piano Roll in both editing modes.



Draw mode (default) uses specific gestures to make adding and deleting notes quick and easy:

- **Add note** – Tap once in an empty section of the Piano Roll to instantly add a new note, which will use the Duration and Velocity settings selected above in the Note Bar. Or…
- **Add custom duration note** – Tap once in an empty section of the Piano Roll, hold, and slide to the right to add a new note of custom duration; the note off will correspond to where the finger is lifted from the editor screen.

- **Select note** – Tap and quickly let go to select a note. To select multiple notes use highlighting (see below).
- **De-select** – When a note (or notes) are already selected, single tap in empty space to de-select it.
- **Delete note** – Long tap an existing note to delete it; it will delete once the finger is lifted.
- **Move note** – Tap and slide notes in any direction (up/down or left/right).
- **Duration** – Slide a note's right edge to change its length. Zoom further in to make it easier to grab just the right edge and not move the entire note. Durations can also be adjusted using the preset Note Duration buttons or the Note Length slider.
- **Highlight** – Tap and slide a single finger diagonally to draw a highlighted box and select multiple notes at once. Every note contained within the highlighted box will become selected.

Note: When in Draw mode the standard gestures for panning and zooming the view work differently. The Piano Roll view in Draw mode cannot be panned using a single finger – as this will cause new notes to be accidentally added; use two fingers (like a normal pinch zoom) to both adjust zoom level and pan up/down or left/right by sliding the two fingers together in the desired direction.

Repeat Draw mode

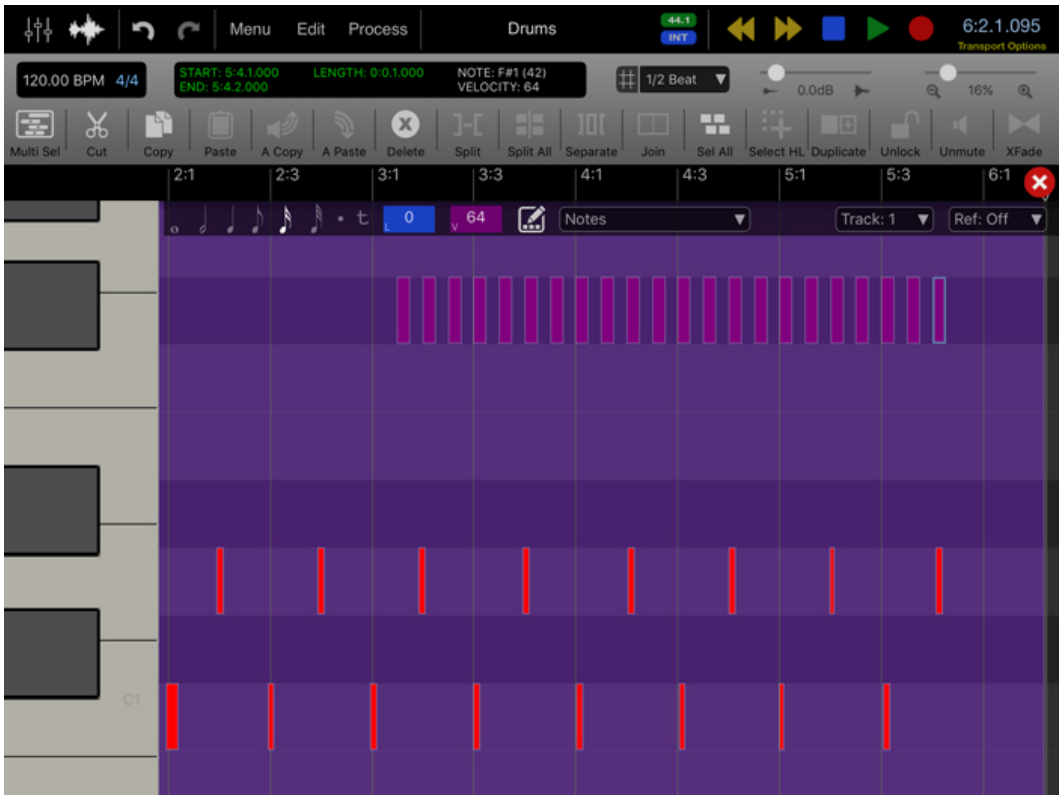


Draw mode also has a variation called Repeat Draw mode, where multiple notes can be created by simply sliding a finger across the piano roll. Both a note duration value and a snap value must be selected for this mode to function, as the created notes will follow both of those selections. Double-tap the pencil icon to toggle between normal Draw and Repeat Draw modes (while a single-tap toggles between Draw mode and Standard mode).

- **Add** – Slide a finger horizontally across the piano roll to add repeating notes using the note and snap values currently selected.
- **Delete** – Long tap an existing note to delete it.
- **Move** – Tap and slide existing note in any direction.

In **Standard editing mode**, MIDI notes are edited in much the same way as audio regions:

- **Move** - Tap and hold a note to move it either up and down (i.e. change its pitch) or left and right (in time).
- **Select** - Tap a note once to select it. This also works with Auria's normal Multi-Select icon.
- **Duration** - Touch and slide the end handle along the right edge of the note, either to the left or right, to adjust the duration of the note. Durations can also be adjusted using the preset Note Duration buttons or the Note Length slider.
- **Add** - Tap and hold in an empty part of the Piano Roll to create a new note, which will use the Duration and Velocity settings selected above in the Note Bar.
- **Delete** – Select a note (so its edges turn blue) and then tap the DELETE icon.



Creating a repeating pattern of notes with Repeat Draw Mode enabled. Note both a grid and note value are selected.

Controller editing

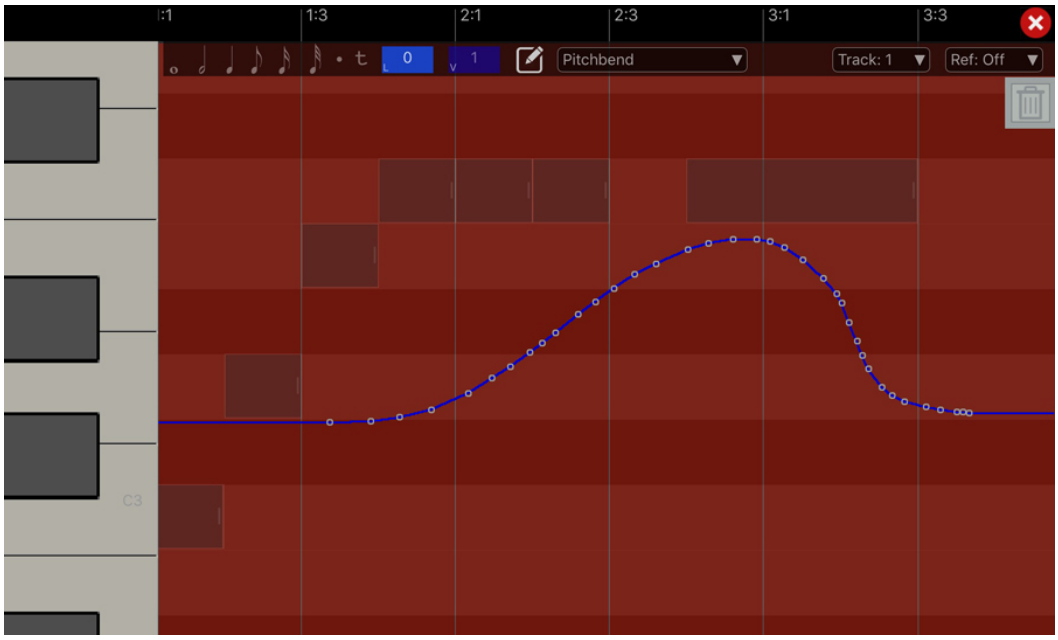
MIDI controller and other non-note data can be edited in the Piano Roll using the familiar control point system used to edit mixer and plug-in automation. MIDI notes are displayed by default, but the View selector drop-down found in the Note Bar can be used to switch between other types of data: MIDI controllers, velocity, pitch bend, aftertouch, etc.



View selector found along the Note bar

Once a different type of data is selected, like pitch bend, any existing data of this type will be displayed using Auria's control point system (the same one used in Auria's automation system). If in Standard mode, single control points are added, moved, and deleted using the same system used with automation data. However, switch to Draw mode (tap the pencil icon) for a quicker method of working with multiple control point data.

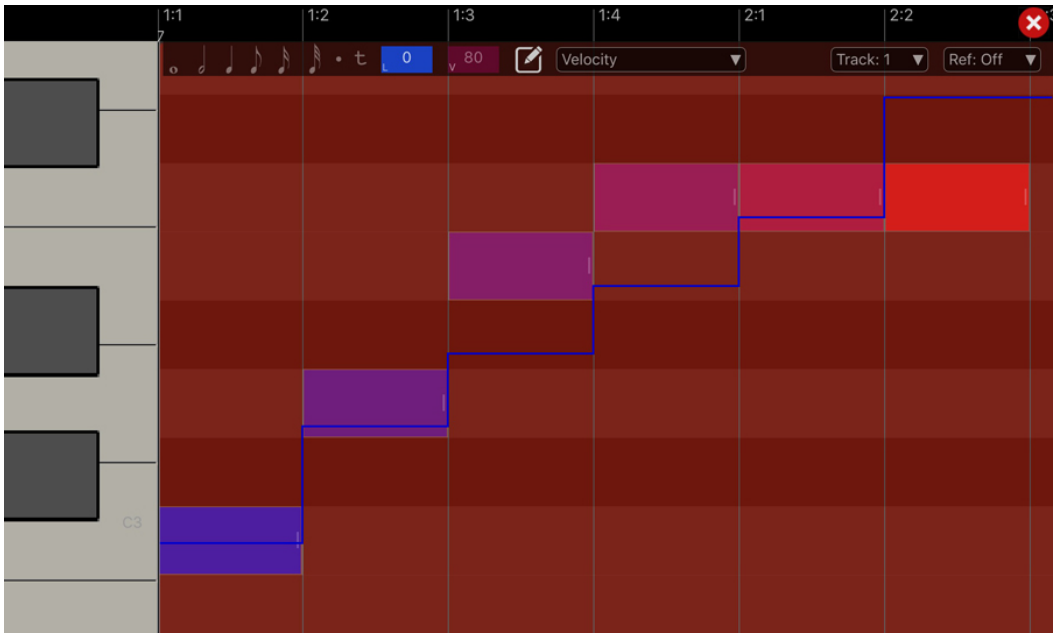
In Draw Mode, simply use a finger to draw new curves instantly on the screen, and Auria Pro will automatically interpolate the gestures into controller data. Existing control points can be edited by simply re-drawing them on the screen with a finger.



Pitch bend data drawn in using Draw Mode

To delete control points, tap the garbage can icon in the top-right corner of the editor (the icon will darken) to toggle Delete mode; then touch any control points to instantly delete them.

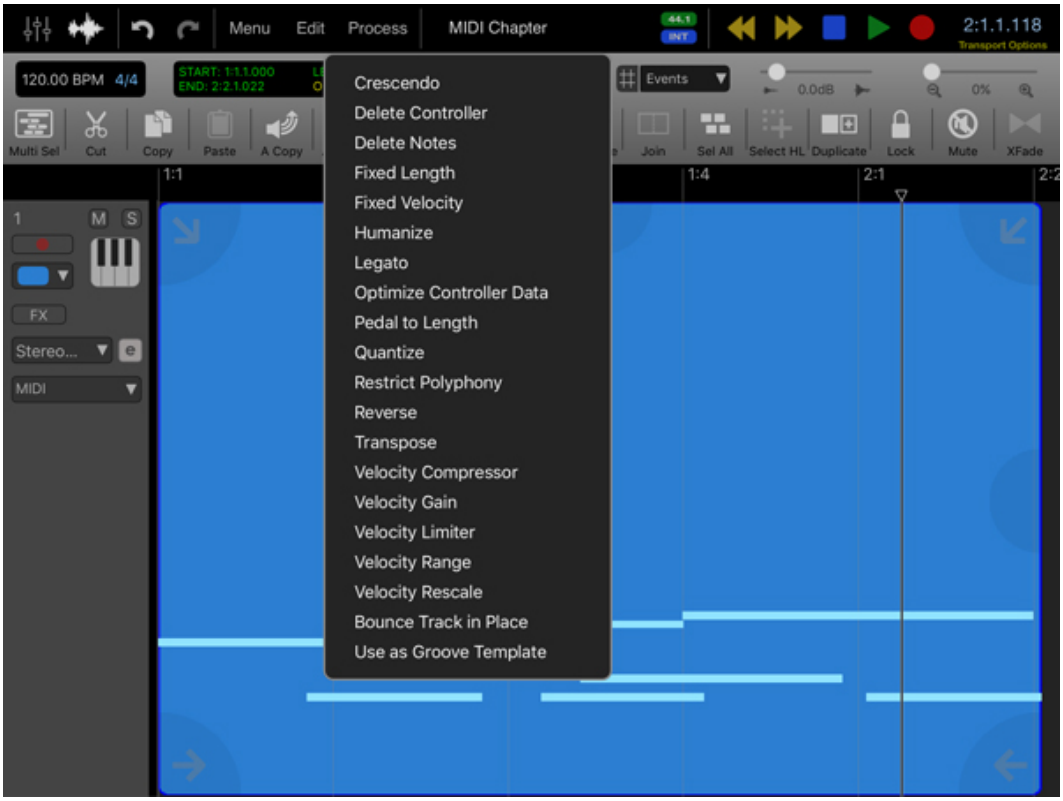
Note: Changing note velocities using this view works in much the same way as other controller data, however as velocity is a strictly note-based parameter each control point will snap to an existing note.



Velocity data in the controller view, note that each control point lines up with a note

MIDI Process Menu

Whenever some MIDI data is selected, either in the Piano Roll editor or a MIDI region in the main Edit window, the top Process menu will switch from the usual audio functions over to MIDI-specific processes.



The complete MIDI Process menu

Each of these processes are destructive, meaning the currently selected MIDI data will be permanently changed by using one of these functions. Some of these functions are available in a non-destructive manner via the MIDI CONTROL window.

For specific information on every available MIDI function in this list please see the corresponding Process Menu Reference in Part Three of this guide.

MIDI Global Keyboard

Auria features a global floating keyboard that can be used to play internal Auria Pro MIDI instruments as well as 3rd party virtual instruments.



Global MIDI Keyboard button

To open the Global MIDI keyboard, tap on the keyboard icon on the upper left corner of the Auria Pro screen to the right of the Edit window button.

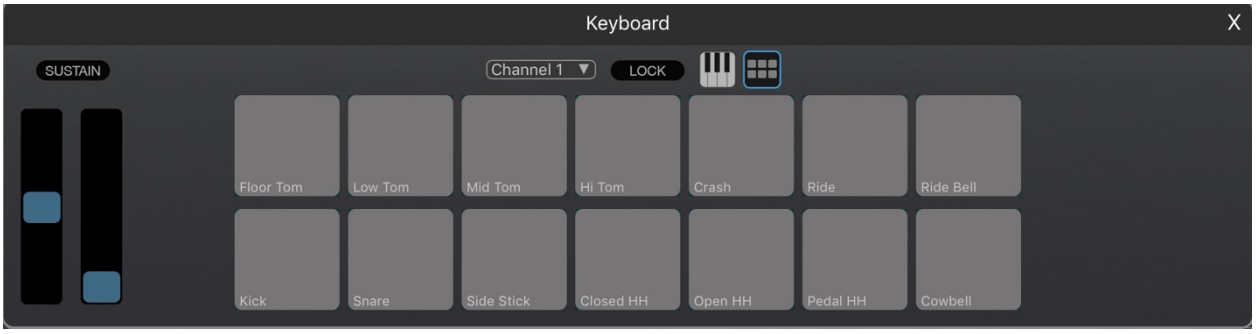


The Global MIDI Keyboard shown floating on top of a 3rd party AUv3 plugin inside Auria Pro

The Global MIDI keyboard has a simple set of controls and buttons:

- **Pitch and Mod -** The two sliders found on the far left of the Global Keyboard represent the standard Pitch and Mod wheels found on most keyboards. By default the Modulation slider will transmit on MIDI CC 1
- **Sustain-** Press and hold to sustain the current notes being played

- **Transmit Channel-** By default the Global MIDI keyboard will transmit on MIDI Channel 1. Tap to open the dropdown window to select a different MIDI channel. Only one instance of the Global MIDI Keyboard can be used at a time.
- **Lock-** The keys on the Global MIDI Keyboard can be resized and shifted left and right to change octaves. Use the standard pinch-to-zoom to change the size of the keys and swipe left and right to change the octave. Tap to highlight the Lock button to keep the keys from resizing or shifting.
- **Keys/Pads toggle-** The Global MIDI Keyboard will open showing piano keys. You can press the Drum Pad toggle to switch to the view to playable drum pads.



Global MIDI Keyboard shown with playable drum pad view

The drum pads are currently fixed to the following MIDI note numbers:

- Floor Tom- 41
- Low Tom- 45
- Mid Tom- 48
- Hi Tom- 50
- Crash- 49
- Ride- 51
- Ride bell- 53
- Kick- 35
- Snare- 38
- Side Stick- 37
- Closed HH- 42
- Open HH- 46
- Pedal HH- 44
- Cowbell- 56

6

MIDI INSTRUMENTS – PRO ONLY

WaveMachine Labs' Lyra

Lyra is a sample-based instrument included with every copy of Auria Pro. Lyra can load large and complex sample files; through its disk streaming engine even 4GB piano samples can be loaded and played with ease on a simple iPad.



Sample Formats

The Lyra instrument included with Auria Pro plays sampled instruments. There are various formats of samples but they all have one thing in common, they are collections of samples combined with articulation and playback information allowing the sampler to treat the samples as the basis of instruments. A piano sampler instrument is normally, for example, a collection of wave files sampled (recorded) by playing a piano one key at a time and recording each key. The instrument then maps each of the samples to a specific MIDI key allowing the sampler to re-create the original instrument sound as a digital signal. Some sampler instruments contain a few samples and are simple, while more complex instruments might map to many thousands of samples, supporting hundreds of detailed key and velocity splits to retain the characteristics of the original instrument.

Lyra supports three different sample formats:

- SF2 (SoundFont)
- SFZ 1.0
- EXS

Most sample instruments have specific parameters already assigned to them, such as filter and envelope settings, which are inherent to the sound of the individual instrument. As noted in the later section detailing Lyra's various parameter controls, these knobs actually behave as global modifiers, allowing the user to quickly customize any loaded patch.

File Management

Lyra stores its instruments inside the SAMPLER INSTRUMENTS folder, found inside Auria Pro's primary DOCUMENTS folder. All of Lyra's default samples plus the instruments available for download from the in-app store are saved inside categorized folders.

Note: For Users running iOS 11+, you can now use the Apple Files App to directly access the Auria Documents folder using your iPad. See Apple Files App section.
On iOS 10 and below a Mac or PC file utility is required to directly access Lyra's sample library, including Apple's own iTunes application or the excellent 3rd party utility iFunBox.

Custom loaded samples can either be saved into these same categories, or new folders could be created by the user, for example a My Imported Samples folder; just make sure that the folder is directly inside the existing SAMPLER INSTRUMENTS folder. These category folders show up when selecting Lyra samples from Auria Pro's INSTRUMENT selector menu found on every MIDI track:



The Instrument Menu, showing available categories of instruments and patches

Some sampled instruments will be contained all within a single file, like SF2 samples. Others will include a small instrument file plus a folder of samples; both SFZ and EXS formats can work this way. In both cases simply include every sample-related file next to each other inside a Category folder, this way Lyra can easily find all of the necessary files.

Note: Lyra only recognizes two levels of Category folders, if too many folders are nested together then they will not all be displayed in the Instrument list. And make sure these folders appear directly inside the SAMPLER INSTRUMENTS folder.

Lyra Parameters

Note: To the user of normal synthesizers, some of the parameters on the interface will seem confusing, because they have both a positive and negative range, unlike common counterparts on regular synthesizers such as envelope attack time. This is because with Lyra, the currently loaded sample instrument also has its own default parameter values, and the values from the graphic interface can be added to these underlying sampler instrument values to form a final computed value.



Lyra's panels of adjustable parameters

Sample Selector: This menu shows all the currently installed Lyra samples. Either tap a particular sample to load it, or use the Up and Down arrow to quickly step through every sample.

Master Panel

Volume: Controls the volume. This is the absolute volume level and is multiplied with internal volume levels after all modulation to the internal volume sum has been computed, thus the minimum value will always produce silence regardless of the values of other modulation levels, such as gain and envelopes.

Key Tracking: Controls how much the MIDI key affects the volume. This is a range in decibels. The default setting is zero, so it has no effect by default. You can use it to make notes louder or softer depending on which side of the center key the note is. For example, if you set the parameter to 12dB and then press notes above the center key, the notes will get louder and louder the farther from the center key the note is.

Velocity Tracking: Controls the degree to which the MIDI note velocity affects volume. By default this is set to have no effect since most samples use the velocity internally, but you can still adjust it for added effect. For example, you could set it so that pressing notes harder makes them play softer, simple by setting this parameter to a negative value.

Voice Mode: Controls the voice allocation mode, either Normal, Legato, or Mono.

Polyphony: This is the maximum number of voices. For example, if this value is set to 3 and you hold a basic 3 note major chord down and then press a fourth note, one of the voices in the chord

will be released, and a new voice will start playing. The way that notes get zapped when the maximum polyphony is reached is to quickly fade them out. The note selected to be zapped is the one that has been playing the longest.

Release Mode: The release mode controls what to do when a note-off arrives at the envelope. Normally the envelope will release from the current level and fade to zero, but you can override this behavior, including choosing to completely ignore the note-off event and let the envelope continue as normal.

X/Y Controller: Control any two of Lyra's controls at once with a single finger, simply assign both the X and Y axis drop-downs to the desired parameters and then slide a finger across the X/Y Controller pad to modify both parameters at once.

Filter Panel

The filter parameters act as global modifiers to the currently loaded sample's own filter usage.

Filter Type: Selects the kind of filter to use. Choose between multiple variations of both lowpass and highpass filters. If you do not wish to use the filter and are happy with the sample sound as is, then use the OFF option to save CPU cycles for other plugins and algorithms Auria Pro. The filter can be an expensive overhead if it is not used, especially in the case where you have many voices playing at the same time in many tracks.

Cutoff Frequency: The cutoff frequency adjusts which frequencies are passed or rejected by the filter. For a lowpass filter, the frequencies below the cutoff frequency are passed, while those above the cutoff frequency are rejected. For a highpass filter, it is the other way around, where frequencies above the cutoff frequency are passed, and frequencies below the cutoff frequency are rejected.

Cutoff Key Tracking : Controls how much the MIDI key affects the cutoff frequency.

Velocity Key Tracking: Controls how much the note velocity affects the cutoff frequency.

Resonance Q: The resonance (abbreviated in most synths as "Q"), is a setting that in simple terms controls the feedback of the filter. Setting a high Q level, and a low cutoff frequency for a lowpass filter, is a common filter effect.

Resonance Key Tracking : Controls how much the MIDI key affects the amount of resonance.

Resonance Key Tracking: Controls how much the note velocity affects the amount of resonance.

Tuning Panel

Coarse: Coarse tuning is in semi-tones and controls the coarse pitch adjustment.

Fine: Controls pitch adjustment in cents. A cent is one hundredth of a semitone.

Random Tuning: Allows random tuning of each note. Each note that is played will add random tuning to the computed pitch. This is handy for some percussion and FX instruments.

Portamento Time: This is the time for the pitch glide. When you hit a new key, the pitch can glide from the most recent pitch of the previous note to the new pitch of the new note over this time.

Portamento Tune: Added to the pitch to glide from when a new note is initialized and it will do a pitch glide.

Portamento Mode: Determines how to handle the Pitch glide.

- Off - Ignore all pitch gliding.
- Normal - Just glide to the note pitch from the previous note pitch.
- Reset - If there are no notes playing, then do not do a pitch glide for the first note on, only any subsequent notes. This can be used to make runs of notes for some kinds of sampled instruments where the real instrument actually works this way, like some wind instruments for example.
- Tuned - Ignore the previous notes pitch and just glide to each notes pitch from the offset determined by the Portamento Tune parameter. For example, set the Tune to -12, and then set the Time to 0.2 seconds and you will hear each note "jumping" up to the pitch of the note from 12 semitones down.

Envelope Panel

Each envelope parameter uses time for units, and serves as a global modifier to all of the currently loaded sample's own preset envelope settings.

Delay: Envelope delay is the amount of time to wait before the envelope attack starts; this will delay the start of the note.

Attack: Controls the attack time. This occurs just after the envelope delay (if any) has elapsed.

Hold: After the envelope has risen from the initial value to the max value, then it can be made to hold for some time at the max value before entering the decay phase.

Decay: The decay phase drops the envelope level from the max value to the level of the sustain parameter.

Sustain: If the player is still holding down the key of the note by the time that the delay period has elapsed then the sustain phase is entered. This is a constant phase and remains in effect until the player releases the key.

Release: The release portion of the envelope occurs when the player releases a note.

Distortion Panel

On/Off switch: Used to turn the distortion effect on or off.

Drive: For most distortion functions, this controls the strength of the distortion. This parameter is an adder, so it combines with any settings that might be in the currently loaded sample.

Damp: This is used to control the smoothness of some distortion functions. Normally it is implemented like a simple one pole lowpass filter to reduce the harsh sound that some distortion functions create. This parameter is an adder, so it combines with any settings that might be in the currently loaded sample.

Amp/Level: This is used to scale the loudness of the affected samples post-distortion. After applying the distortion function, the signal might be too soft or too loud. This parameter is an adder, so it combines with any settings that might be in the currently loaded sample.

Keyboard Panel

Pitch Bend Range Coarse: Use this parameter to set the pitch bend range, such that when you move the Pitch wheel all the way up or down it will increase or decrease the pitch by a higher or lower amount. The default value (RPN standard) is two semitones up or down.

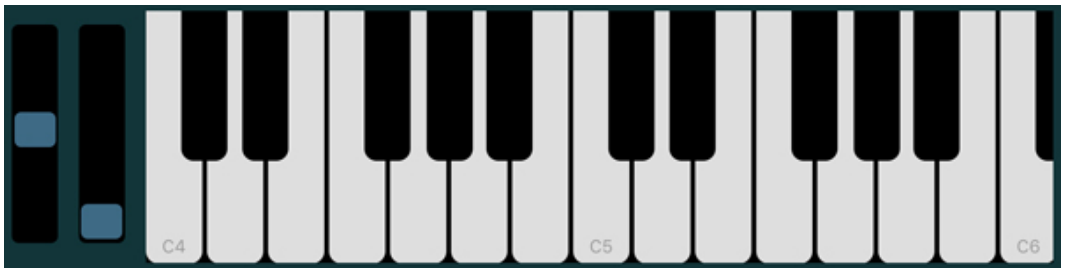
Pitch Bend Range Fine: Use this parameter to tweak the pitch bend range by additional cents.

Sustain: Tap to send a momentary sustain pedal message.

Lock: Locks the current keyboard range and zoom level from changes.

Keyboard/Drum pad toggle: Switches between the keyboard and drum pad layouts.

Keyboard and Drum Pads



Lyra has two different playable layouts, a standard keyboard and a collection of drum pads.

Lyra's keyboard is touch-location sensitive, where tapping higher up the key results in a smaller velocity value (softer) while tapping lower on the key produces a larger velocity (louder). The keys are also zoomable, so using a two-finger zoom can adjust the key size as desired. This size can then be locked (along with the current display key range) with the LOCK button (see above).



Lyra's drum pads

Lyra's drum pads are configured for a standard GM drum mapping (which all of the included sampled drum kits follow), and support WaveMachine Lab's VelAUcity™ technology.

Note: The display has two wheels on the left, in order they are the Pitch Bend Wheel and a Mod Wheel.

VelAUcity™

Utilizing the iPad's internal microphone, VelAUcity detects the intensity with which the drum pads are tapped and translates that into MIDI velocity, in essence making Lyra's drum pad's pressure sensitive. Tap a crescendo onto a pad and Lyra will respond accordingly. Tap the VelAUcity button to the upper-right of the drum pads to enable this mode.

Best performance notes:

- Use headphones to eliminate false readings caused by the iPad speaker
- The quieter the ambient noise level the better velAUcity's performance
- For best performance use a buffer size of 128 samples
- Use of any USB or MFi audio interfaces will disable VelAUcity, as connecting a USB or MFi audio interface disables the iPad's internal microphone

Important compatibility note: The iPad Air 2 does not support the use of VelAUcity. Apple instituted a change in this model which introduced an extreme amount of latency with the internal microphone that renders VelAUcity useless. Hopefully Apple will fix this issue down the road and allow all iPads to work with VelAUcity.

FabFilter One



The FabFilter One interface is divided into several sections, somewhat like the front panel of a traditional analogue synthesizer:

- Oscillator: The oscillator is the heart of the synthesizer, producing the raw tones that are modified by the other components. It also contains portamento, additional noise settings and settings that control the amount of frequency modulation by the modulation generator and envelope generator.
- Filter: The filter alters the tones from the oscillator to make them warmer and less harsh. Like a true analogue filter, it can easily be driven to self-oscillation to create completely distorted sounds, or a warm lead with a little raw edge. The filter section also contains settings that control the amount of frequency modulation by the modulation generator and envelope generator.

- Envelope generator: The final touch and feel of the sound is defined by the envelope generator. Tweak the various settings here to create anything between short percussive hits to long reverberating effects.
- Modulation generator: The modulation generator generates a low-frequency square or triangle wave that can be used to modulate both the frequency of the oscillator and the cut-off frequency of the filter. This is where the fun starts!
- Additional features: This section contains a few useful extras. You can choose between monophonic and polyphonic (10-voice) mode, set the type of pulse width modulation, set the amount of filter keyboard tracking and set the amount of keyboard velocity modulation for the volume and the filter cut-off frequency.
- Presets: With the preset button, you can easily browse through the factory presets or save your own sounds so you can re-use them in other songs.

Oscillator

At the heart of the synthesizer, the oscillator produces the raw tones that are later modified by other components, such as the filter and the envelope generator.

You control the oscillator with the following settings:

- Wave Form: Selects the wave form that the oscillator produces: triangle (a soft, warm tone), sawtooth (a sharp bright tone that is ideal for filtering), square (a more metallic sound, adjustable with the pulse width setting), and white noise (which is great for percussive sounds and effects).
- PW/PWM: Adjusts the pulse width of the square wave form, from an even square to an infinitely thin pulse (which is inaudible). In combination with the Pulse Width Modulation switch in the Patching Features section, this knob also adjusts the amount of pulse width modulation.
- Scale: Transposes the tones that are produced by the oscillator by full octaves. The oscillator produces rich bass tones at the 32' position and high whistles at 4'. The 16' and 8' settings are good for lead sounds.
- Pitch: Slightly detunes the oscillator, which can be necessary if you want to play along with a record, or with live instruments.
- Portamento: Adjusts the amount of portamento or glide, which makes the oscillator slide from one note to another. A little portamento is great for smooth keyboard solos, or when playing large bass intervals. The portamento feature in FabFilter One is unique because it also works in polyphonic mode, so you can play sliding chords!
- Noise: Adds extra noise to the oscillator signal. Use the Noise Type switch at the bottom to choose between white and pink noise. A small amount of noise makes instruments sound more natural, and you can create thundering sound effects with lots of pink noise. (Try the Pink Noise preset in the Basic submenu for a good starting point.)

- Filter: The 12 dB/octave low-pass filter is one of the most important components in the synthesizer. Especially when using the modulation features, it is the key to creating great sounds.

The Frequency knob adjusts the cut-off frequency. Basically, all sounds below the cut-off frequency will go through the filter, but sounds with higher frequencies are damped. At position 10, the cut-off frequency is far above 20 kHz and therefore the filter will allow all frequencies to go through unchanged. At position 0, the cut-off frequency is just a few Hz and the filter will block almost all sound from the oscillator.

The wave forms produced by the oscillator contain many harmonics: high frequencies that are multiples of the fundamental tone. Therefore, they sound bright and a little harsh. By turning the Frequency knob to the left, you reduce the number of harmonics to change the character of the wave forms and create warmer sounds.

The Peak knob adjusts the filter resonance. A little resonance will cause the filter to create warmer and more characteristic tones. Around position 8, the filter starts to self-oscillate at the cut-off frequency which creates terrific effects in combination with the sound from the oscillator. At these high peak values, the filter produces deep, ringing bass tones if you set the Frequency knob around position 3 or 4.

We've carefully tuned the filter to avoid the whistles and other digital artifacts that are common with other digital filters when you change the cut-off frequency at maximum resonance. Our filter sounds crazy but never over the top!

You can modulate filter cut-off frequency with the modulation generator (the MG knob), the envelope generator (the EG knob), and with keyboard velocity. Read the modulation basics section to learn more about the modulation features of the FabFilter One.

Envelope generator

The envelope generator changes the volume of the sound dynamically when you press and release keys. It shapes your sounds, adjusting the timbre and release. It's also frequently used to modulate the filter for dynamically changing sounds.

- Hold: Sets the extra time that a key appears to be held down after you have released it. After releasing a key, the envelope generator pretends it is still down for the duration of the hold time. This enables you to press new keys while the previous note is still playing.
- Attack: Sets the attack time, which controls how long it takes for the volume to increase when you press a key. The volume always increases from zero to the maximum value. Set this to 0 for

staccato instruments and drums. Use larger values for more natural sounds, such as flutes and pads.

- Decay: Sets the decay time, which controls how long it takes for the volume to decrease after the maximum value has been reached at the end of the attack stage. The volume always decreases from the maximum value to the sustain level. The effect of the decay setting depends on the sustain level. With a low sustain level and short attack and decay times, the envelope generator produces a sharp tick whenever you press a key. With the sustain level at its maximum, the volume does not decrease at all, so the decay setting has no effect.
- Sustain : Sets the sustain level: the volume that is produced when both the attack and decay stages are over, but the key is still held down. If you want a sharp attack (especially when you also use the envelope generator to modulate the filter), you have to use a low sustain level. If you set the sustain level to 0, the sound will be inaudible after the attack and decay stages, which is often useful for drums and percussion.
- Release: Sets the release time, which controls how long it takes for the volume to decrease from the current level to zero when you release a key and the hold time has elapsed. When playing the synthesizer live, it's often nice to have a little longer release time because this gives you time to play a new note while the current note still fades away.

Tips

- When you work with short attack, decay, and release times, it's often easier to adjust the knobs using the rotational drag mode. To do this, drag the knob arrow around, and move your finger further away from the knob for more precise adjustments.
- If you use a short decay time, a low sustain level, and a long release time and a key is released when the envelope generator is not yet in the sustain stage, it enters the release stage immediately which can cause the sound to last much longer. Although this can be a nice effect, you can avoid it by choosing a larger hold time.

Knobs

It is easy to control FabFilter One's parameters with the large round knobs. They will light up when you move your finger around to indicate that you can adjust them. The moment you touch a knob with your finger, a parameter value display will pop up, which shows the name and the current value of the parameter.

All knobs support four ways of control:

1. Vertical mode: Tap on the center area of a knob and drag up or down to rotate it. The knob reacts to the speed with which you are dragging, so if you move your finger slowly, you make precise adjustments.
2. Rotate mode: Touch the arrow of the knob and drag it around. By moving your finger further away from the knob while dragging it, you can make precise adjustments.

Modulation basics

So far, we've discussed how to use the oscillator, the filter, and the envelope generator to create basic sounds. To make things more interesting, you can also let the synthesizer adjust the oscillator frequency and the filter cut-off frequency dynamically while playing a sound. This is called modulation.

You can modulate the oscillator frequency and filter cut-off frequency with the modulation generator and the envelope generator. The filter cut-off frequency can also be modulated by velocity.

The modulation generator generates a low-frequency square or triangle wave. Use this to adjust the oscillator frequency and the filter cut-off frequency periodically. The amount of modulation is controlled with the MG knobs. The modulation generator contains the following settings:

- Wave form : Adjusts the balance between rising and falling ramps in the wave form. To reset the balance, double tap on the knob
- Sync mode: Chooses between arbitrary frequency settings and synchronizing with the tempo and song position of the plug-in host.
- Frequency / Sync type: Sets the speed of the wave form that is produced by the modulation generator. When synchronized to the host's tempo and position (the closed lock symbol), you can choose a 'frequency' in bars, between 16 bars and 1/64 bar. Otherwise, you can set an arbitrary frequency. When set to 0, very slow ramps are produced that can for example be used with white noise to imitate ocean waves rolling in.

You can also switch between square waves and triangle waves. Triangle waves are good to slowly change the filter cut-off frequency or to simulate a car siren. The square wave form is very useful to repeatedly switch between two different tones. With the frequency modulation MG knob, you can tune the two tones to be exactly one octave apart, for example.

The envelope generator enables you to modulate the sound dynamically while you press and release keys. The amount of modulation is controlled with the EG knobs. Modulating the filter cut-off frequency with the envelope generator is essential for creating percussive attack effects and simulating real instruments.

When the envelope generator is in its sustain stage, it has no effect as a modulator. In the attack or decay stage, it increases the oscillator or cut-off frequency; in the release stage, it decreases the frequency. It depends on the EG knobs how much the frequency is changed.

Tips

- Set the modulation generator to a moderate frequency and turn the frequency modulation MG knob up slightly to create a tremolo effect. The modulation wheel on your keyboard will also have this effect.
- Set the modulation generator to a low frequency and turn up filter modulation MG knob to make sounds livelier by changing the timbre dynamically.
- In the envelope generator, set short attack and decay times and use a low sustain level. In the filter, set the cut-off frequency to about 5 and then turn up the filter modulation EG knob while pressing different keys. Also try increasing the peak value for more pronounced attack effects.
- In the filter, set the cut-off frequency to 3 and the peak to 10. In the envelope generator, set attack to 0, decay and release to 1, and sustain to 8. Now turn up the filter modulation EG knob to about 6. This will create a solid bass drum (it's best with white noise). Experiment with different sustain levels and cut-off frequency values.

Modulation: I can't stop

We just can't get enough of those modulation tricks, so we've added a few extra gadgets.

The Pulse Width Modulation switch is only effective if the oscillator wave form is set to square, so be sure to do that first. It changes the meaning of the PW/PWM knob in the oscillator section. You've probably noticed that it sounds funny to turn that knob quickly, so why not modulate it? We've come up with the following options:

- Off: This is the default setting. In this case, the PW/PWM knob just sets the pulse width of the square wave form.
- MG: With this setting, the PW/PWM knob adjusts the amount of pulse width modulation by the modulation generator. If you set the modulation generator to triangle waves, this creates a quite nasty sound. With square wave modulation, the oscillator periodically switches between two different pulse width values.
- EG Normal : With this setting, the PW/PWM knob sets the amount of pulse width modulation by the envelope generator. Because the envelope generator increases the pulse width setting during the attack phase, the sound actually gets thinner during attack, and thicker during release. This may or may not be what you're after.
- EG Inverted : In this case, the PW/PWM knob sets the amount of pulse width modulation by the inverted envelope generator signal. This creates thicker sounds during attack/decay, and thinner sound during release, which sounds more natural. Our favorite choice.

When modulating the pulse width, it's often better not to turn the PW/PWM knob all the way to the right, because that will mute the square wave at some point. Just leave it at 8 or 9 (unless you like that, of course).

The Frequency Mod EG switch at the bottom of the oscillator section and Filter Mod EG switch at the bottom of the filter section invert the envelope generator signal before it is used to modulate the oscillator frequency or the filter cut-off frequency. You can sometimes use this to create unusual effects.

The Keyboard Velocity settings change the amount with which the keyboard velocity modulates the volume and the filter cut-off frequency. If the volume knob is set to 0, a tone will always sound at maximum volume.

MIDI Learn

Controlling FabFilter One'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.

Tap 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, tap the MIDI Learn button again, or tap Close at the top of the interface.

Tap 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 and redo

The Undo and Redo buttons at the top of the FabFilter One interface enable you to easily undo changes you made to 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.

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.

Creating Sounds

The FabFilter One comes with a solid library of great sounds, but the real fun starts with creating your own presets. However, just randomly turning some knobs won't help you to get closer to the sounds you have in mind.

The topics in this chapter will help you start creating your own sounds, explaining the best settings step by step. They're divided into the following categories:

- Bass sounds
- Lead sounds
- Drum sounds
- Sound effects

Creating Bass Sounds

The Oscillator: For bass sounds, you can use any of the first three wave forms. Using triangle waves will result in deep, sine-like sounds, great for R&B and soul. Sawtooth waves will get you a more solid, defined sound, good for funk, techno and dance sounds as well as for emulating original bass guitar sounds. If you're trying to create an electro-like bass sound, the square wave form is just what you need.

The Envelope Generator: A good start for bass sounds as well as lead sounds is setting the attack, decay and release fairly low, the sustain value quite high and the hold time off, resulting in a 'snappy' envelope. When your sound needs to have a sharper attack, lower the sustain value.

The Cut-off Filter: Using the cut-off filter really is essential when creating bass sounds. To get warm and deep sounds, start with the frequency and peak values around position 5. For techno and dance sounds, turn the peak value up to get those freaky filter effects.

Modulation: One thing that definitely works great with bass sounds, is modulating the cut-off frequency using the envelope generator (the EG knob in the filter section). Slightly modulating it with the modulation generator (the MG knob) works quite well for moody, deep sounds.

The Finishing Touch: Adding a bit of portamento often works great for funky and analog-like sounds. Also adding just a little bit of pink or white noise can give good results in combination with the cut-off filter. It can be very worthwhile to experiment with the filter modulation EG value and with keyboard velocity modulation.

Creating Lead Sounds

One of the things that FabFilter One excels at is producing great lead synth sounds of any type. You can create anything from the smoothest seventies analog sounds to the hardest techno sounds.

The Oscillator : Especially the sawtooth and the square wave forms are suitable for lead synthesizer sounds. With the sawtooth wave form as a starting point, you can create smooth and soft lead sounds as well as more techno-oriented sounds. The square waveform is great for electro stuff.

The Envelope Generator : Most lead sounds use a fairly short attack, a short decay, a high sustain and a noticable release. This will give your sound just that little bit of attack at the start and a good full sound while holding down a key.

The Cut-off Filter: Adjust the frequency and peak values to get the timbre that you're after. Modulating the cut-off filter is not essential when creating lead sounds. However, it can give your sounds that extra rawness and body they need, especially when modulating the cut-off frequency.

Modulation: Use the modulation generator to achieve great vibrato or tremolo effects by modulating the oscillator's frequency or the filter's cut-off frequency. It also works really well to use a bit of cut-off frequency modulation by the EG. When using the oscillator's square wave form, you can also modulate the pulse width using the patching features. Modulating it with the EG or with the MG can both get you awesome results.

The Finishing Touch: One thing that will definitely give wonderful results with almost all your lead sounds, is FabFilter One's great and smooth portamento. Even using it just a little bit really brings your sounds alive. Experiment with it! To get those awesome raw seventies synth sounds, push

your filter to the limit, using high peak values and combining different types of cut-off frequency modulation. Try changing the cut-off frequency while playing, to really adapt it to your play.

Creating Drum Sounds

FabFilter One can create a whole range of different percussive sounds, like bass drums, snares, toms and hihats. The keys to success are extreme peak values and modulating the cut-off frequency of the filter.

The Oscillator: The wave form you need really depends on the type of drum sound you would like to create. For instance, tom sounds or snare sounds could use any of the first three waveforms (triangle, sawtooth or square).

To get a good hihat or brush sound, the noise type would do the trick.

Creating bass drums works differently. Because great bass drum sounds heavily depend on the self-oscillation of the filter, you can experiment using any of the four wave forms. But the noise wave form usually gives the most solid results.

The Envelope Generator : Most drum sounds, like bass drums and snares, are short and have a really strong attack. To get this effect, you can set the attack (almost) to 0, use fairly short decay and low sustain values and use a short release. Especially with bass drum sounds, the sustain level can have a huge effect on the sound. Just experiment with this value in combination with the filter modulation to get the results you're after.

The Cut-off Filter: The cut-off filter really is the key component in the process of creating drum sounds, especially for bass drums. To create the fattest kick sounds, use the self-oscillation that occurs with the highest peak values, in combination with extreme cut-off frequency modulation by the envelope generator (turn up the EG knob in the filter section).

Modulation : As mentioned before, modulating the cut-off frequency with the EG is what you want to do here. For a good bass drum, start with maximum EG filter modulation and fine-tune if needed, adjusting the sustain value as well. For other percussive sounds, most of the time just a bit of EG modulation is enough.

The Finishing Touch : A great trick for giving snare drums that little extra kick, is adding some white or pink noise to the signal (using the oscillator's noise knob and noise type switch). Also inverting the EG cut-off frequency modulation can give great effects. Try to experiment with the envelope generator, expecially when creating bass drum sounds. For example, using higher release values and slightly higher sustain values will give you great results.

Creating Sound Effects

For creating sound effects you can really exploit every option that FabFilter One offers you. Especially the modulation features can help you create the coolest sounds. Because sound effects can be very diverse, ranging from a storm sound to a raindrop, there is not one ideal initial setting that will help you on the way. To be able to create a sound effect that you have in mind from scratch, you should be comfortable with the various options that FabFilter One provides. To give you an idea how different effect sounds are created, we will discuss two basic factory preset sounds.

In the explanations, the knob settings leading to the sound will be discussed. Knobs that are not mentioned are assumed to be set to their default value. You can quickly set all knobs to the default values by loading the Sawtooth preset from the Basic category.

Rain Drop

The first thing that comes to mind when you think of a rain drop, is that it sounds quite warm and 'round', somewhat like the sound of a sine wave. This indicates that a triangle is the best wave form to use. A rain drop sound is quite high, so a 4' scale setting would be good to start with.

1. Set the oscillator wave form to triangle.
2. Set the oscillator scale to 4'.

The second important thing that characterizes the sound is the fast rising frequency, starting low and ending very high. To obtain this effect, we need frequency modulation by the inverted envelope generator signal (the EG knob in the oscillator's frequency modulation section and the invert switch below it) in combination with a short envelope. A rain drop sound starts right away, so an attack value close to 0 should be used. To get the correct frequency modulation effect, short decay and release values are required.

1. Set the EG knob in the frequency modulation section around value 9.
2. Set the invert switch below the EG know to 'inverted' (at the right-hand position).
3. Set the attack almost to 0, the decay to 2, the sustain to 6 and the release to 2.5.

We now have that characteristic fast rising sound. However, it still needs some tweaking. To give the sound a little bit more body, we use the cut-off filter and some cut-off frequency modulation by the envelope generator (the EG knob in the filter section).

1. Set the cut-off frequency to 4.
2. Set the EG knob in the filter section to 10.

Storm

Many effect sounds use the oscillator's noise wave form. A storm sound is of course one of the best examples of such a noise-based sound. The oscillator produces white noise. This type of noise is a quite high, hissing sound. FabFilter One also offers the possibility to add white or pink noise to its signal (using the noise knob and the noise type switch below it). Pink noise has a more dark and rumbling feel. White noise mixed with a bit of pink noise would be just fine for our storm sound.

1. Set the oscillator's wave form to noise.
2. Set the noise knob to 3 and the noise type to pink.

A storm sound is characterized by slowly swelling and fading winds. This swelling effect can easily be achieved using the modulation generator at a low frequency to modulate the filter's cut-off frequency (the MG knob in the filter modulation section).

1. Set the cut-off frequency to 6.5 and the peak to 5.5.
2. Set the modulation generator's wave form type to triangle and frequency to 0 and wave form to its center position (a perfect triangle).
3. Set the MG knob in the filter section to 4.
4. To get proper fade in and fade out effects for the storm sound, we use high attack and release values. Set the attack to 5 and the release to 6.

Loading presets

The included presets give a great overview of what you can do with FabFilter One. You can either use the presets as they are, or tweak them further to create your own unique sounds.

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

To explore the presets one by one, tap 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.

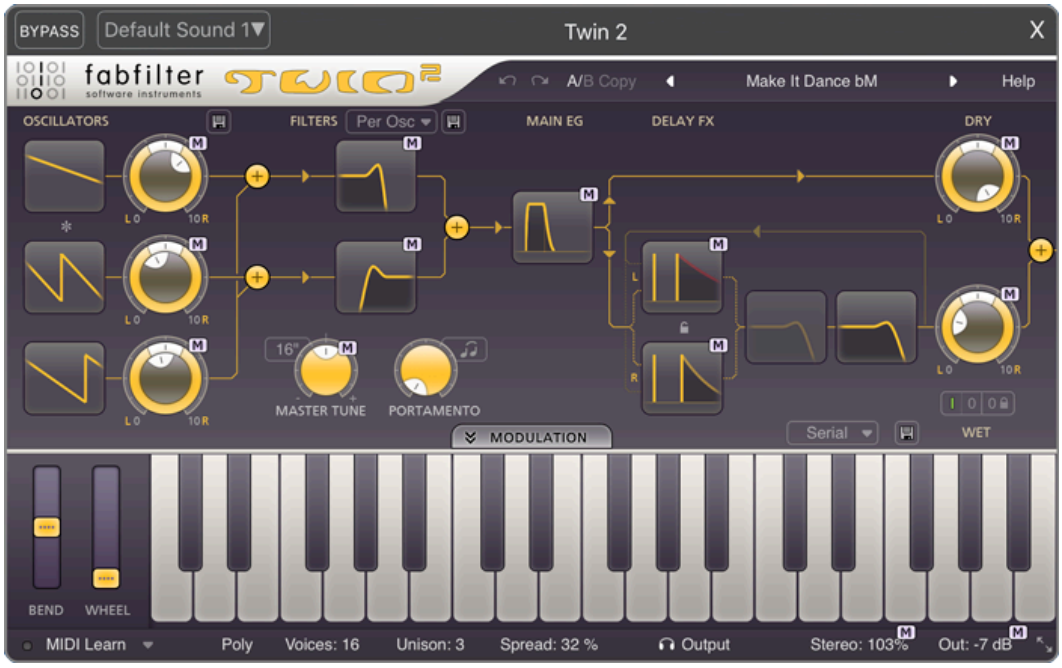
The preset button shows the name of the current preset. If you have changed the preset by adjusting one or more parameters, the name is dimmed to indicate that this is not the original preset anymore.

Saving presets

You can easily extend the included presets with new settings to build your own library of presets for FabFilter One that you can reuse in various projects. This is also a good way to copy settings across multiple instances of FabFilter One in a session.

To save the current setting as a preset, tap the preset button, and then tap Save As. A standard Save dialog will appear. Type a name for the new preset and tap Save to finish.

FabFilter Twin2



FabFilter Twin 2's interface is divided into multiple 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 vast library of factory presets or save your own settings so you can re-use them in other songs.

Synth section

The top section is the actual synth, the sound-producing heart of the instrument. Here you'll find the oscillators, the filters, the main envelope generator and the delays logically displayed, together with general settings like portamento, master tune and volume controls. Each component of the synthesizer is represented by a component button that shows its state and lets you edit it directly.

Modulation button

The modulation button shows or hides the entire modulation section at the bottom of the interface. FabFilter Twin 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 Twin 2 offers XLFO, Envelope Generator (EG), Envelope Follower (EF), MIDI and XY Controller sources.

Modulation slots and sources

The bottom section contains the modulation overview. Twin 2 is a fully modular synthesizer 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!

MIDI Learn, monitoring and output mix

The bottom bar contains the MIDI Learn feature which sets up any MIDI controller to control any plug-in parameter. Next, we find various polyphony and unison settings which give you the possibility to create super-fat sounds. At the far right are stereo spread and output level settings.

Resize

The resize button in the lower-right corner lets you choose between normal and wide interface layouts. The wide layout is designed such that the component display at the top half of the interface never needs to be scrolled. **Presets**
With the preset buttons, you can easily browse through the factory presets or save your own sounds so you can re-use them in other songs.

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 tap the M to highlight the modulation source and slots responsible for the modulation.

Another interface innovation are the component buttons. These make it possible to do more in depth editing of parameters you don't need to see all the time. Only when you want, the interface can show more parameters of the oscillators, filters or delays.

Tip: the resize button in the lower-right corner of the interface allows you to make the interface wider, so you can see more components and modulation sources at the same time.

Knobs

It is easy to control FabFilter Twin 2's parameters with the large round knobs. They will light up when you move your finger around to indicate that you can adjust them. The moment you touch a knob with your finger, a parameter value display will pop up, which shows the name and the current value of the parameter.

All knobs support four ways of control:

- Vertical mode: Tap on the center area of a knob and drag up or down to rotate it. The knob reacts to the speed with which you are dragging, so if you move your finger slowly, you make precise adjustments.
- Rotate mode: Touch the arrow of the knob and drag it around. By moving your finger further away from the knob while dragging it, you can make precise adjustments.

Component buttons

All components that make up the synthesizer in FabFilter Twin are displayed as large component buttons in the top section of the interface: three oscillators and following the signal path, two filters, two delays, and two more filters in the delay section.

You can edit the most important parameters of a component directly by taping and dragging on the component button. As soon as you tap and hold your finger on the component button, value displays will pop up to show the current values of those parameters.

Drag horizontally and vertically to change the parameters. There are two sets of parameters.

To view all parameters of a component, such as a filter or oscillator, tap the component button once. This will expand the complete interface for all related components. Tap the component button again to hide the interface. While a component interface is expanded, you can scroll the top section of Twin's interface with the left and right scroll buttons at the far ends of the interface.

Each of the buttons has an on/off switch in the left top corner, to quickly enable or disable the component.
We strongly suggest for you to try all these movements yourself, and you'll find it's a great aid in quickly setting up Twin the way you like. The most important parameters are always available, and if you need access to all parameters, they are just a finger tap away.

Tips

- Tapping the Main EG button does not open a dedicated EG section, but rather highlights the Main EG source in the modulation section.
- You can turn off the parameter value displays for the component buttons with the Show Component Displays option in the Help menu.
- The resize button in the lower-right corner of the interface allows you to make the interface wider, so you can see more components at the same time.

Oscillators

FabFilter Twin 2 comes with three full featured oscillators. Combined with our award winning filters this gives you an exceptional broad palette of sounds. The oscillators are very analog sounding and completely aliasing-free (for the digifoobs!). For every oscillator the following parameters are available:

- Enabled: This will turn an oscillator on or off, so you can easily check the contribution of an oscillator to the overall sound, or turn it off to reduce CPU usage.
- Scale: Each oscillator can be transposed up to 3 octaves above the master tune.
- Waveform : The Waveform parameter selects the waveform that the oscillator produces:

1. Triangle (a soft, warm tone)
2. Sawtooth (a sharp bright tone that is ideal for filtering)
3. Square (a more metallic sound, adjustable with the pulse width setting)
4. Sine (a simple sine wave, the most basic waveform without overtones)
5. White noise (has a flat frequency response, meaning all frequencies at equal power)
6. Pink noise (is like white noise but with less power in the higher frequencies)

- Phase Sync: Next to the waveform setting, the Phase Sync button (with a MIDI icon) toggles oscillator phase sync on and off. When oscillator phase sync is on, the oscillator resets its phase to halfway the waveform on each note-on event. With short, percussive sounds, this works really well to create a sharp and aggressive attack with every note. However, normally you should turn it off because it sometimes introduces unwanted taps.
- Detune: The Detune knob sets the detune for each oscillator. The detune knob has a range of +/- one octave, and is extra sensitive around the 0 position. It is a modulation target as well.
- Pulse Width: The Pulse Width knob adjusts the pulse width of the square waveform, from an even square to an infinitely thin pulse (which is actually inaudible). The pulse width is a modulation target and is only available when a square wave is selected.
- Sync: The Sync knob adjusts the level of hard sync for its oscillator. When set to 1, the oscillator works normally without hard sync. When set to a higher value, the oscillator will 'squeeze' more phases into one actual audible phase, which drastically changes the harmonics in the sound with a metallic-sounding effect. Sync is a modulation target.
- Level/Pan: The volume and pan of each oscillator can be adjusted using the combined Level/Pan knobs. The inner knob set the overall level and outer ring sets the panning. Both are modulation targets.
- Ring modulation: You can ring modulate the first oscillator with the second oscillator using the small * button between the component buttons for oscillator 1 and 2. Ring modulation combines two waveforms, and outputs the sum and difference of the frequencies present in each waveform. This produces a signal rich in overtones, suitable for producing bell-like or otherwise metallic sounds.
- Master Tune/Scale: The Master Tune knob determines the global frequency offset of all oscillators. The menu above the knob transposes the frequency by full octaves. The Master Tune knob is a modulation target so you can modulate the frequency of all oscillators at once.
- Portamento: The Portamento knob adjusts the amount of portamento or glide, which makes the oscillators slide from one note to another. A little portamento is great for smooth keyboard solos, or when playing large bass intervals. Good portamento is one of those things that give your play just that little bit extra: it's magic. So we have fine-tuned it until it was just perfect! The best thing is: we have created a unique portamento because it also works in polyphonic mode, so you can play sliding chords. You can choose between two different portamento modes.

1. When using Normal portamento (high/low key portamento), gliding will happen always when holding down keys.
2. When using Legato portamento (also called fingered portamento), gliding will not be applied for the first key pressed, only for the keys after that.

Portamento is also a modulation target.

Tips

- Disabling an oscillator reduces CPU usage. Also, the oscillators operate more efficiently when the Sync knob is set to 1 and is not modulated (disabling hard sync).

- The section presets button next to the level/pan knobs for the oscillators lets you store and retrieve all oscillator parameters as a section preset, which is very useful for accessing often-used combinations of parameters.

Oscillator component buttons

Each oscillator is visually represented as a component button. By tapping on the component button and dragging, you can directly alter the most important oscillator parameters:

- Detune: drag horizontally
- Scale: drag vertically
- Sync: Double tap and hold your finger on the button and drag horizontally
- Wave Form: Double tap and hold your finger on the button and drag vertically

Tap on a button to open the full oscillator interface that provides access to all parameters.

Filters

FabFilter Twin 2 has four multi-mode stereo filters. The two main filters are put right after the oscillators and the other two are used within the delay section. The filters can be routed in three modes: serial, parallel and per oscillator/delay. 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.

For every filter the following parameters are available:

- Frequency: The filter frequency is adjustable over the entire audio range. The Frequency controls the center or cut-off frequency of the filter and can be controlled in real time, either manually, via modulation, or via external devices using MIDI.
- 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 and thus it makes for a great modulation target.
- 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 with most filter characteristics.
- Characteristic : FabFilter Twin 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
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 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...
- Routing : The routing of the filters can be changed. The different settings are clearly graphically represented in the interface. There are three different setups:

1. Serial: the sound passes through both filters serially. So the sound of all oscillators are led through filter 1 and then through filter 2.
2. Parallel: the sound is led through both filters in parallel and the result is then mixed together after the filters.
3. Per Osc / Delay: the output of oscillator 1 or delay 1 goes through filter 1, the output of oscillator 2 or delay 2 goes through filter 2. For the main filters, the output of oscillator 3 is routed through both filters.

- Tip: The section presets button next to the routing button lets you store and retrieve all filter parameters as a section preset, which is very useful for accessing often-used combinations of parameters.

Filter component buttons

Each filter is visually represented as a component button. By tapping on the component button and dragging, you can directly alter the most important filter parameters:

- Frequency: drag horizontally
- Peak: drag vertically
- Pan: Double tap and hold your finger on the button and drag horizontally
- Response: Double tap and hold your finger on the button and drag vertically

Tap on a button to open the full filter interface that provides access to all parameters.

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 correspond to the actual filter frequencies.

- To open the filter display, tap on one of the filter component 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.
- Double tap a filter dot to toggle between the different filter slopes.

Tips

- Of course, all changes made in the filter display can be automated!
- You can connect a MIDI controller to either the frequency or the peak adjustment of the link dot with the MIDI Learn feature.

Delay section

FabFilter Twin 2 comes with a very useful delay section with two filters. The delay section makes all kinds of flanging and chorus effects possible, as well as typical delay effects, to give your sounds more stereo depth and rhythmical subtleties. Delay can be a very important aspect of sound design and the ability to modulate all the parameters will generate innovative and new sounds. The delay section consists of two separate delay lines and two multimode filters.

The following delay parameters are available:

- Delay Time/Offset : The Delay Time parameter sets the delay time given to a signal passing through. The time display just above/below the Delay Time knob shows the current delay time in milliseconds.

If Delay Sync is set a different value than Free or Ultra Short, this knob changes into the Delay Offset parameter, which sets a time offset relative to the synchronized delay time. More on this below.

- Delay Sync: There are different ways to set the delay time. These different modes are set by the Delay Sync parameter next to the time display.

The delay time can be locked/synchronized to the tempo of your sequencer host. When this is activated the delay time knob controls the sub-multiples of this tempo.

The small dots that appear around the knob make it easier to get precise and quick access to those values that are related to your sequencer tempo like triplets and dotted notes.

Or you can use the delay in a 'free mode'. Now the delay time is not synchronized to the host tempo and you can use the knob to set the delay time at any desired time. In this mode it is possible to 'tap' the tempo of the delay by taping on the number-display next to the delay time knob. The display will turn into a purple TAP button. The next time you tap here the time between the taps is calculated and used as delay time. Just tap it a few times to get some values you want to work with.

The 'Ultra Short' mode is designed to get the typical flanging, chorus and comb-filter effects. Now delay time can be precisely set between 0.1 and 50 milliseconds and feedback controls are disabled. Read the tips for more details on this.

- Feedback, Cross Feedback: You can vary the feedback to produce more than one repeat from a single sound. All the feedback control does is 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 knob on the interface. Cross-feedback is used to mix different delay times and can create 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.

- Link: When the little lock button between the two delays is activated, the controls for the first delay will also control all parameters of the second delay. Therefore the controls of the second delay are disabled.
- Dry, Wet: The large dry and wet level/pan knobs let you control the amount of dry (undelayed) and wet (delayed) signal that comes out of the plug-in. They are both modulation targets, but you must be aware that these are global volume settings, applied after all voices have been mixed together. Modulating this with source that is voice-dependent, such as MIDI velocity or an envelope generator, will not give good results. Instead, modulate the output level found in the bottom bar of the interface, because this level is applied for each voice separately, before they are mixed together.
- Enabled: Each delay has a separate on/off switch. But we made another option available: the whole delay section can be turned on and off using the general buttons in the right bottom corner. 'I' means the delay is working and audible. 'O' means delay is turned off. Then there is

one more option that always turns the delay section off regardless what the preset dictates. This setting cannot be saved in a preset and therefore will not be altered when browsing presets. This makes it possible to browse presets with the delay section permanently turned off. Don't forget to listen to the delay settings used in those presets. It sometimes is an important part of the sound design and can give great depth and groove to a sound.

Tips

- By setting a delay time of between 30 and 100ms and adding a little gentle modulation with no feedback, you get the classic chorus effect.
- At very short delay times, (10 to 50ms in Ultra Short mode) 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 it can create the illusion of ring modulation or phase sync.
- Short delays of between 30 and 100ms are used to create slap-back echo effects.
- Delay times in excess of 100ms, will give you the familiar tape echo type of sound, and this creates more rhythmical effects.
- The section presets button next to the delay filter routing button lets you store and retrieve all delay and delay filter parameters as a section preset, which is very useful for accessing often-used combinations of parameters.

Delay component buttons

Each filter is visually represented as a component button. By taping on the component button and dragging, you can directly alter the most important filter parameters:

- Time/Offset: drag horizontally
- Feedback: drag vertically
- Sync: Double tap and hold your finger on the button and drag horizontally
- Cross Feedback : Double tap and hold your finger on the button and drag vertically

Tap on a button to open the full filter interface that provides access to all parameters.

Filters

Two filters are put after the delay lines in the signal path. The two filters are the same multimode filters as the main filters used after the oscillators.

- Modulation: Modular sound design employs modules as the basis of design. The modules in FabFilter Twin 2 are the three oscillators, the multimode filters, the delays, the XY Controller, the XLFO, the Envelope Generator, the Envelope Follower and the MIDI source. Each of these modules have parameters that can be changed by both knob movements and modulation. These parameters are called modulation targets. The modules that can send control signals to the modulation targets are called modulation sources. The modulation signal always flows via a modulation slot that allows you to vary the extent of modulation.

FabFilter Twin 2 offers very flexible modulation possibilities which make this plug-in capable of many sound design applications. Ever-changing parameters can make your music become more alive because an exact reoccurrence is very unlikely to happen.

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 tap 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. Twin 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. And the XY Controller lets you modulate two targets using horizontal and vertical finger movements.

To add a modulation source, tap the + button in the source selection bar.

To delete a modulation source, tap 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 Twin 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 tap 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.
Now drop the icon on the highlighted knob of the parameter that you would like to modulate, for example the Master Tune 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 tap 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.
- To delete the slot, tap the Remove button to the right of the Level slider.

Our what-you-use-is-what-you-see interface makes complex programming very easy. Twin 2 uses dynamic slot highlighting to visualize all the sources that modulate a specific target. When a parameter is modulated a small modulation indicator appears:

- Tap the 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 tap anywhere on the interface background or tap the Modulation Indicator again.
When a modulation indicator appears next to a component button or envelope generator it means one or more parameters are modulated. When you tap 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! (Any suggestions? Please let us know.)

To add an XLFO as a modulation source, tap the + button in the source selection bar and tap 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. Tap 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.
- MIDI sync: The XLFO can be restarted at any point using MIDI if the MIDI sync option is enabled at the top-right corner of the frequency knob. When activated a note-on MIDI message (e.g. pressing a key) will restart the cycle of the waveform (to the point set by the Phase offset slider).
- 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 oscillator detune, master tune, or filter frequency, turn the slot level to maximum, and the total amount of modulation will exactly correspond to 2 octaves. If you modulate a detune knob, it will sound like a classic arpeggiator. 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.

- Tap a step to select it.
- Tap next to a step to deselect all steps.
- Tap 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.
- Tap 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 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 an 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 some aspect of the instrument's sound (often its loudness) over time.

Since you need at least one EG to make a sound, you cannot delete the first EG, called Main EG. The Main EG differs from the other EGs because it doesn't have trigger, threshold and delay parameters and always has a range from 0 to 1. The Main EG also has a component button counterpart in the top synth section of the interface, enabling you to tweak attack, decay, sustain and release directly without having to open the modulation section.
To add an envelope generator as a modulation source, tap the + button in the source selection bar and tap New Envelope Generator.
The following EG parameters are available:

- Trigger: The EG can be triggered by two different kinds of input: MIDI note events (MIDI) or the signal from the side-chain (Side Chain). When MIDI is selected, the Threshold knob is hidden. Depending on the type and amplitude of the incoming side chain 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. See also Using the side chain.
- Delay: The time it takes for the attack to start after the key is pressed (or triggered when the side-chain signal exceeds the threshold).
- Attack: The Attack portion of the envelope is the time taken for the amplitude to reach maximum value. Slow attack is commonly part of sounds called pads. But for percussive sounds 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 a level known as the Sustain level at a rate 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.
- Range: Normally, the EG works with a range from 0 to 1 (the top option), but you can also choose to change the range so EG outputs 0 at the sustain level. It will attack from a negative value to a positive value, and finally release to a negative value. The EG "centers" around the 0 value, which can be useful for modulation.

Tips

- When adjusting the EG control points, you can hold down the Shift key to fine-tune a setting, just like with regular knobs.
- To let a sustain pedal control the EG, you can hook it up to the Hold setting. The recommended way to do this is via a MIDI source.
- 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 will output an envelope similar to the side-chain signal. The amplitude of the positive peaks of the input signal is measured and the outer shape (= envelope) is the output signal of the modulation source. You can set the Attack and Release parameters to 'smooth out the bumps'.

To add an envelope follower as a modulation source, tap the + button in the source selection bar and tap New Envelope Follower.

Note: The envelope follower is only useful if the plug-in is set up in the host application to receive a side chain signal. More about this in Using the side-chain.

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.

MIDI source

The MIDI source is a powerful modulation source if you want more control using a MIDI keyboard or MIDI control knobs.

To add a MIDI source as a modulation source, tap the + button in the source selection bar and tap New MIDI Source.

MIDI Input

Normally the MIDI source lets you use MIDI input such as velocity, pitch bend and modulation wheel to influence any parameter that can be modulated.

Controller number

When you set the input selection to Controller, it lets you use any MIDI controller as a modulation source.

Response curve

The response curve can be adjusted to get the desired control over the MIDI source output. For example, when used with velocity as MIDI source the linear, exponential, logarithmic, square, square root or sine curves make great dynamic differences.

Using a MIDI source is different from MIDI Learn because there is no direct control of a knob via MIDI but it uses a modulation slot instead. This way you can add modulation to an already modulated destination. Or you can use the full rotation of a knob while actually modulating a

smaller range. This can be a good way to control say, the filter cut-off frequency or EG attack for which you sometimes want to make small changes with great precision.

At the top right of the source interface, the Presets button provides access to the MIDI source section presets. The Remove button deletes the MIDI source. You can customize the default MIDI source settings (used when creating a new MIDI source) by overwriting the predefined Default section preset.

Tips

- When modulating oscillator master tune or detune with pitch bend, the default slot level value corresponds to +/- one note of. The maximum slot level corresponds to +/- one octave.
- To let a sustain pedal control the Main EG, use a MIDI source. Set the input to Controller, and the controller number to 64. Now, drag the source drag button for the MIDI source to the hold control point of the Main EG, and set the slot level to maximum. Of course, you can also set up more sophisticated behavior of the sustain pedal, for example by modulation the release setting instead of hold: the possibilities are endless.

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 finger 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, tap the + button in the source selection bar and tap 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.

Tips

- With MIDI learn, you can set up a hardware MIDI controller to control the XY controller. So if your MIDI controller has XY functionality you can directly control Twin 2.

Polyphony and unison

FabFilter Twin 2's polyphony and unison settings are located at the bottom of the interface.

As you would expect, you can use FabFilter Twin in either Mono or Poly mode. In Poly mode, the Voices setting determines the number of voices that can be played simultaneously. Depending on

the complexity of the current patch, you should set this to a value that your computer can comfortably handle to avoid glitches while playing.
In both modes you can specify the number of unison voices. When using unison, multiple voices will be activated for each key that you play. Those voices will be panned from left to right, and can be slightly detuned relative to each other using the Spread parameter. Using unison will make you patches sound much more stereo and fat. Try it!

Tips

- The mono mode of FabFilter Twin 2 uses last-note priority, so the last key pressed determines the frequency that you hear. When you release a key, the frequency slides back to the second-last key pressed, and so on.
- The Voices setting always sets the maximum number of simultaneous voices. If Unison is higher than 1, more than one voice will be used at a time from the maximum number. For example, if you have Voices set to 16, and Unison to 4, you can play four different keys at a time. Of course, using a high number of voices, unison or not, will cause high CPU usage.
- Believe it or not, but the Unison Spread parameter is a modulation target. You could, for example, increase the spreading gradually while a note is played by modulating it with an envelope generator.

Output controls

The output controls are located in the bottom-right corner of Twin 2's interface.

This is where the output level and panning settings are controlled. The dry (unprocessed) signal and the output of the delay lines have their own level and panning knobs.

From left to right:

- The Audition setting (with the small headphones icon) is normally set to Output. You can set it to Side Chain to be able to listen to the incoming side chain signal. See Using the side chain.
- The Stereo setting sets the stereo spreading of the output signal. By default, it is a 100% or 'normal stereo', which means that it does not alter the signal. By reducing the setting towards 0%, the signal will become more mono. By increasing it towards 200%, the differences between the left and right channel are enlarged, which makes the sound more stereo.
- The Out setting controls the main output volume and panning.

Notes

- Beware that making the signal more stereo can cause some undesirable phase cancellation.
- Both the output level/pan and stereo settings are modulation targets. Volume is commonly modulated by the velocity modulation source. The Stereo knob is a global modulation target, meaning that it affects all voices simultaneously. When modulating a global target like Stereo

with a per-voice modulation signal such as an envelope generator, the highest signal value of all active voice envelopes is used.

- Like the Stereo setting, the Dry and Wet level settings for the delay section are applied globally. Modulating them with per voice sources such as velocity or an envelope generator is possible, but doesn't give perfect results. Instead, modulate the output level in the bottom bar, described above.
- The resize button next to the Out setting allows you to make the interface wider if desired.

Tips & tricks

Fabfilter Twin 2 has been designed to make sound designing fast, easy and fun. Here are some tips and tricks that will further help you to create a good preset and to get just that sound that you are looking for.

Drag-and-drop modulation

Obviously, the best tip we can give you, is to make use of the drag-and-drop modulation connection method. There is no need to go searching through the drop down menus for sources and targets: just drag a source icon of any modulation source directly to any knob that can be modulated to fill a modulation slot.

Section presets

The settings for an XLFO, EG, EF, MIDI source, and the oscillators, filters, and delay sections can be saved in section presets, to be re-used for future sound designing. FabFilter Twin 2 comes with some general factory section presets, but we encourage you to save your own frequently used settings. Use the Presets button in a section to access the presets menu.

Using an XLFO as an arpeggiator

The XLFO has a Snap setting (looks like a small piano keyboard). If selected, a small piano keyboard appears and the sliders snap to "note" positions. Also, if you modulate a filter frequency, the master tune, or oscillator detune with this XLFO and turn the slot level to maximum, the note positions in the XLFO will correspond with actual notes. So, you can use this to turn the XLFO into an arpeggiator!

Modulating an XLFO frequency

You can modulate the frequency of an XLFO with any modulation source other than itself. You could, for example, modulate the frequency with an envelope generator, which can be really cool.

Take advantage of the XLFO phase offset parameter

The XLFO has a phase offset parameter, which is controlled by the large triangular > slider underneath the step sliders. You can also modulate this! In addition, you can set the frequency to zero, so the XLFO just stays at its current position, and use the phase offset to ‘sweep’ back and

forth through the steps. If you modulate the phase offset you can sweep through the steps multiple times (try this with an XY controller and the slot level at maximum).

Modulating a modulation slot level

FabFilter Twin 2 lets you modulate the levels of other modulation slots. For example: use the first slot to modulate master tune with an XLFO and use the second slot to modulate the first slots level with a slow envelope. This gets you a swelling vibrato. You can nest modulation in this way as deeply as you want.

Portamento

We have put a lot of effort in creating a very good portamento, because this really gives a magical feel to your play. Even adding just a little bit works great. We have also implemented a unique polyphonic portamento which enables you to play sliding chords. Try it!

Inverted keyboard track frequency modulation

When you use the keyboard track in a MIDI source to modulate the Master Tune or Osc 2 Detune parameter, with the modulation level set to the maximum, and the slot switch to inverted, the oscillator will produce the same frequency for every note you play.

Play the self-oscillation

Our award-winning filters self-oscillate like no other digital filters. When you modulate the cut-off frequency of a filter with the keyboard track modulation source at full range, and set the filter frequency correctly with maximum peak, your filter actually becomes an oscillator and you can play notes with it!

7

AUDIO WARPING – PRO ONLY

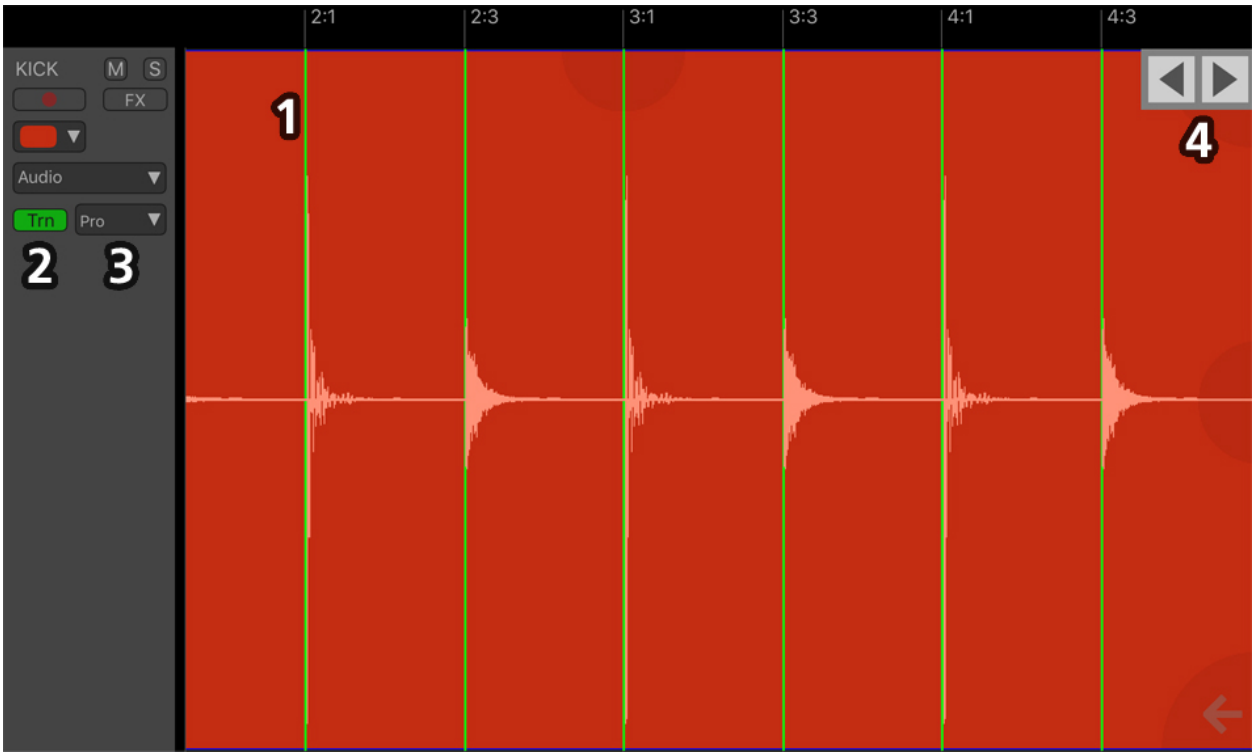
New to Auria Pro is the ability to quickly and easily "warp" audio tracks in time. Auria Pro provides movable warp markers on audio tracks that can be moved in time and changes the rhythm or timing of an audio track. This is accomplished through Auria Pro's all new élastique time stretching algorithms from zplane. Individual snare hits can simply be slid earlier or later in time using a finger and the results heard instantly. Entire audio tracks can be quantized like they were MIDI. And all of this can happen in real time.

Definition of Audio Transient: a short duration, high amplitude, and complex sound which occurs at the onset of an individual acoustic event, such as a musical note. Imagine the exact moment a drumstick strikes a snare head and begins the many different vibrations which make up the sound of a snare drum. Or the first consonant sound when a voice begins singing. This initial transient can now be detected within Auria Pro, allowing individual musical events (notes) inside an audio track to be recognized and manipulated.

Auria Pro adds two new selectable modes to audio tracks, Transient View mode and Warp View mode.

Transient View

When new tracks are recorded, or existing audio files are imported into an Auria Pro project, the default action is to scan these files automatically in order to detect transients and create subsequent transient markers. These markers can be viewed and edited by switching a track view to transient mode.



Transient Marker View

1. Transient Marker (in green)
2. Toggle track view between OFF, Transient Mode(TRN), and Warp Mode (WARP)
3. Time-stretch modes: Pro, Efficient, Mobile
4. Tap arrows to jump forward or back through transient markers

To view a track's transient markers, tap the button labeled #2 above until it shows TRN in green. Green vertical lines will appear indicating wherever a transient has been detected.

Selecting Transient Markers

To select an individual transient marker simply tap the green vertical line representing the transient in question so that it changes color to white.

To quickly jump between transient markers use the two arrow buttons in the upper-right corner of the Editor (labeled #4 in the above illustration). This is exceptionally useful when the editor is horizontally zoomed in far enough that only one transient marker is visible at a time.

Moving Transient Markers

To move a transient marker simply tap and hold on the individual marker until its vertical line turns light blue, then keeping the finger pressed to the screen slide it left or right to move the marker in time.

NOTE – transient markers only need to be moved to correct when they are not placed at precisely the onset of an audio event (note, drum hit, etc). Moving a transient marker will not affect the underlying audio, see the section below on Warp Markers to learn how to change the timing and rhythm of an audio track.

Deleting Transient Markers

To delete a transient marker first tap the marker in question to select it (it will turn white), then choose DELETE TRANSIENT from the Edit Menu. To delete all the transient markers inside a region first make sure the particular region is selected (outlined in blue), then choose CLEAR TRANSIENT MARKERS from the TRANSIENTS sub-menu inside the Edit Menu.

Creating transient markers

To create a new individual transient marker position the vertical cursor in the desired position in time and select ADD TRANSIENT MARKER from the TRANSIENTS sub-menu inside the Edit Menu.

To automatically create all new transient markers for a region select CREATE TRANSIENT MARKERS from the TRANSIENT sub-menu inside the Edit Menu.



Create Transients dialog

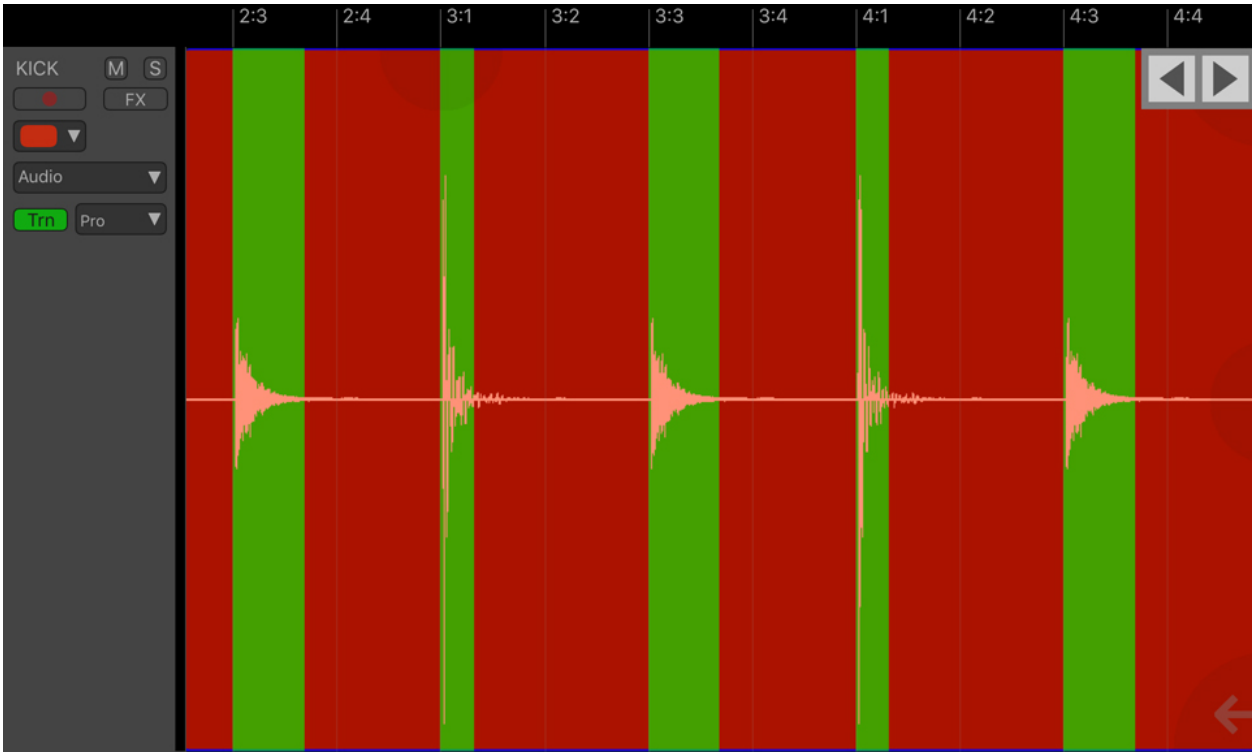
- Auto – automatically scans audio region for transients, requires no further adjustment
- Manual – enables manual selection of threshold settings used for detecting transients
- Start Threshold – amplitude value in dB above which transients must occur in order to be detected
- End Threshold – amplitude value in dB below which the audio must fall for the ends of notes/drum hits to be detected. See End Markers below

If AUTO mode either detects too few or too many transients, switch to MANUAL detection. Try to set the START THRESHOLD value close to the track's noise floor, so that only desired notes/drum hits will be detected. END THRESHOLD should normally also be set close to the noise floor so that the entirety of each note/drum hit is captured, but it may be adjusted independently if warranted.

End Markers

In addition to detecting the transients which signify the beginnings of individual musical/audio events (notes, drum hits, etc), Auria Pro will also determine when those detected events end. These end markers are useful when using Auria Pro's transient slice and separate functions. See the later section for more information on Slicing/Separating at Transient Markers.

To view End Markers, select the option SHOW END MARKERS from the TRANSIENTS sub-menu inside the Edit Menu.



Transient View with SHOW END MARKERS enabled

When End Markers are enabled, individual detected events will appear in green highlighting, where the left edge corresponds to the initial transient and the right edge to the end of the event.

End Markers can be moved in the same fashion as Transient Markers: tap and hold the right edge of the green highlighting until it turns light blue and then slide to the left or right while keeping the finger pressed to the screen.

Warp View

Used for performing audio warping, these markers are first automatically created from detected transients but they differ from transient markers in that when they are moved left or right in time the audio directly underneath the marker will move right along with it. This is accomplished by time stretching the audio around the warp marker; in essence the track simply speeds up or slows

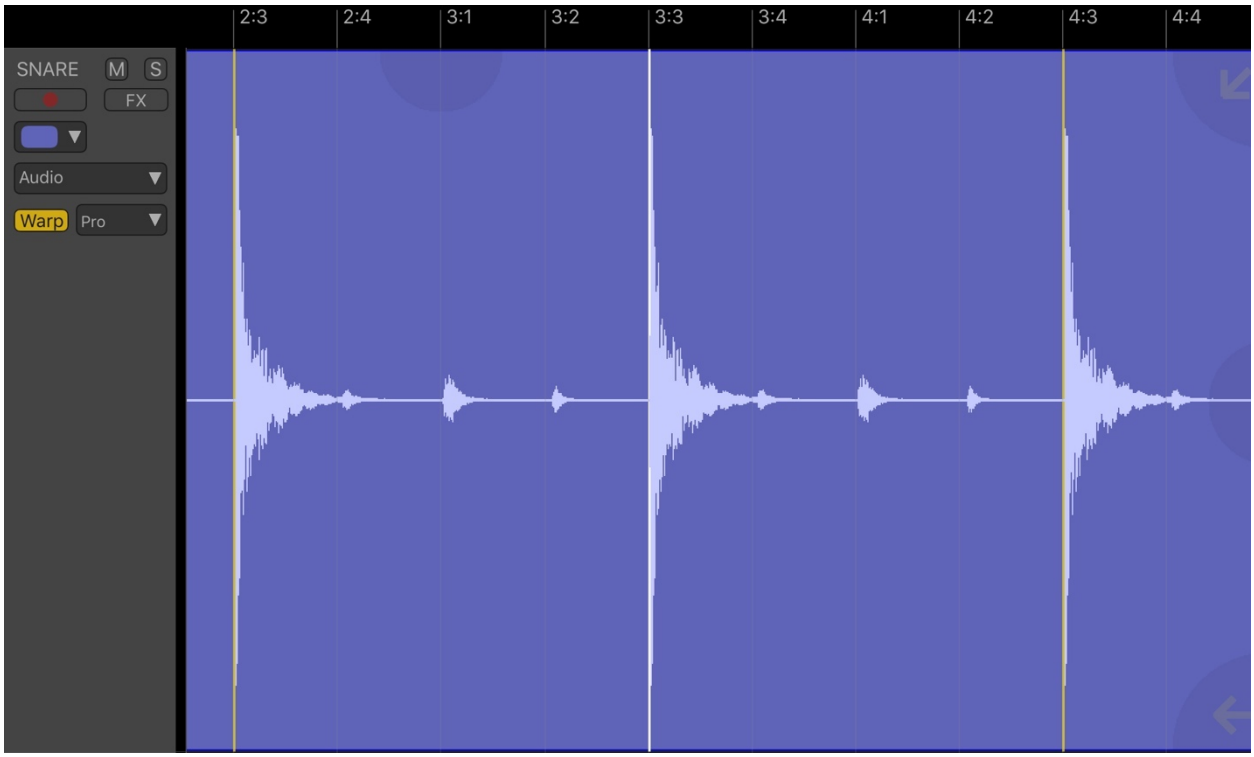
down as needed so that the warp markers occur at new points in time, without changing the pitch of the audio or significantly altering its quality.

To view a track's warp markers, tap the button labeled #2 above until it shows WARP in yellow. Yellow vertical lines will appear indicating wherever a warp marker exists.

Moving Warp Markers

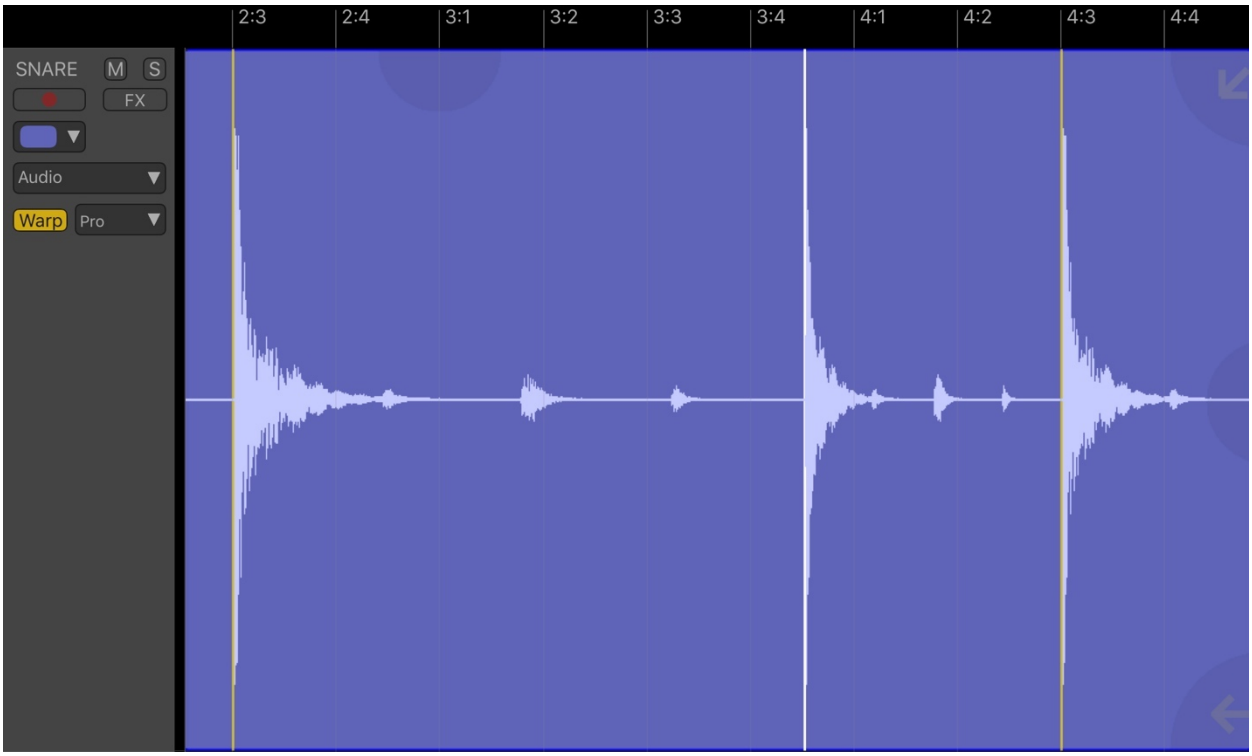
To move a warp marker simply tap and hold on the individual marker until its vertical line turns white, then keeping the finger pressed to the screen slide it left or right to move the marker around in time. The underlying audio will be "stretched" along with the marker as it is moved.

To illustrate, here is a typical snare track showing three individual snare hits, where each snare hit occurs on beat three. Note the middle marker is already selected:



Before: three snare hits with corresponding warp markers

And here is that same section after moving the middle warp marker about one and a half beats to the right:



After: warped snare track

Note that the first and third snare hits stayed exactly in place, while the middle snare moved over to the right. The audio between the first two hits has stretched out to accommodate the new timing while the audio between the last two has squeezed together.

Auria Pro uses real-time algorithms when audio warping so that moving Warp Markers can be immediately heard. However, as this increases the CPU load on the iPad, Auria Pro implements a unique caching system to minimize CPU load when warping audio, intelligently switching between real-time and offline processing as needed. A new clock icon will appear in Auria Pro's Menu Bar whenever this caching system is performing necessary offline processing.

NOTE: As iPad models vary with regards to CPU power, older iPads may encounter difficulty when called upon to process warped regions in real-time. To help users with these less powerful iPads, Auria Pro has two methods to mitigate the strain of real-time audio warping:

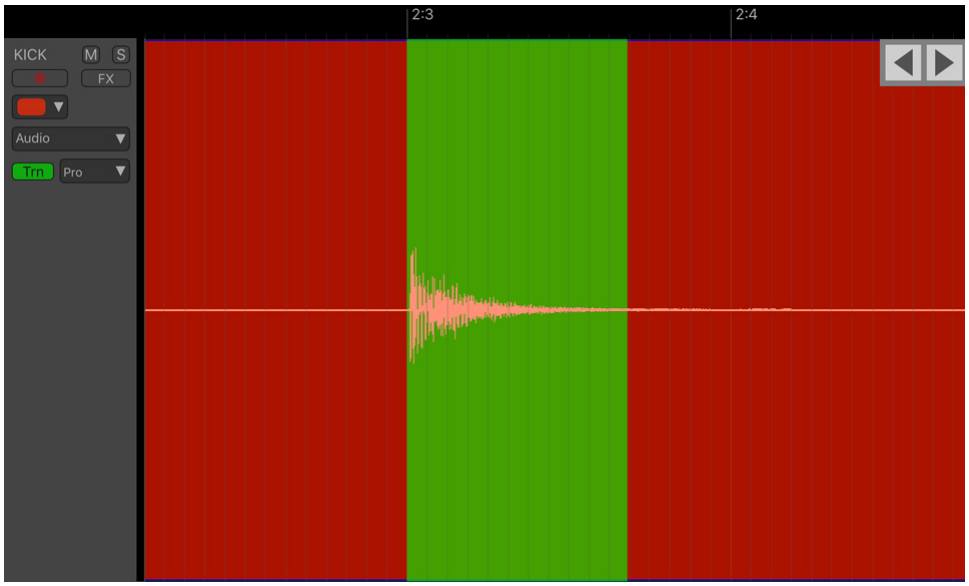
- Multiple time stretching modes. Auria Pro defaults to the élastique Pro algorithm, but other less CPU intensive algorithms are also available. See the section on the different modes.
- Disabling real-time warp mode. For iPads or CPU-intensive projects where real-time time stretching is not practical, this mode can be disabled in the Settings.

Slice/Separate at Transient Markers

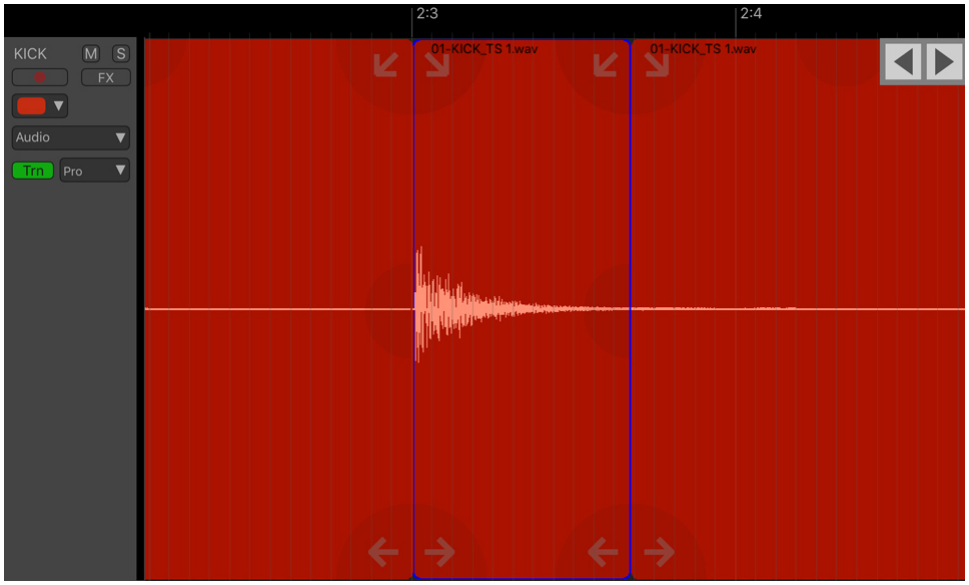
Auria Pro supports two similar methods to automatically edit audio regions based on its Transient Markers, either Slicing or Separating the audio at the markers. Both functions utilize both the Transient and End Markers, and can be accessed from the TRANSIENTS sub-menu inside the Edit Menu.

Slice at Transient Markers

This function splits the selected audio region into multiple smaller regions, making a cut at both the Transient and End Markers.



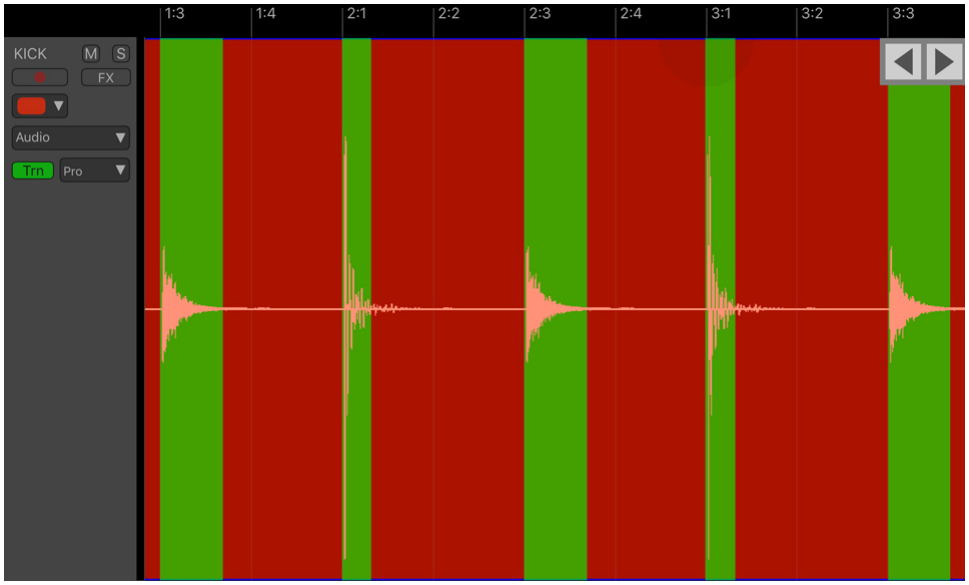
Before slicing at transient markers



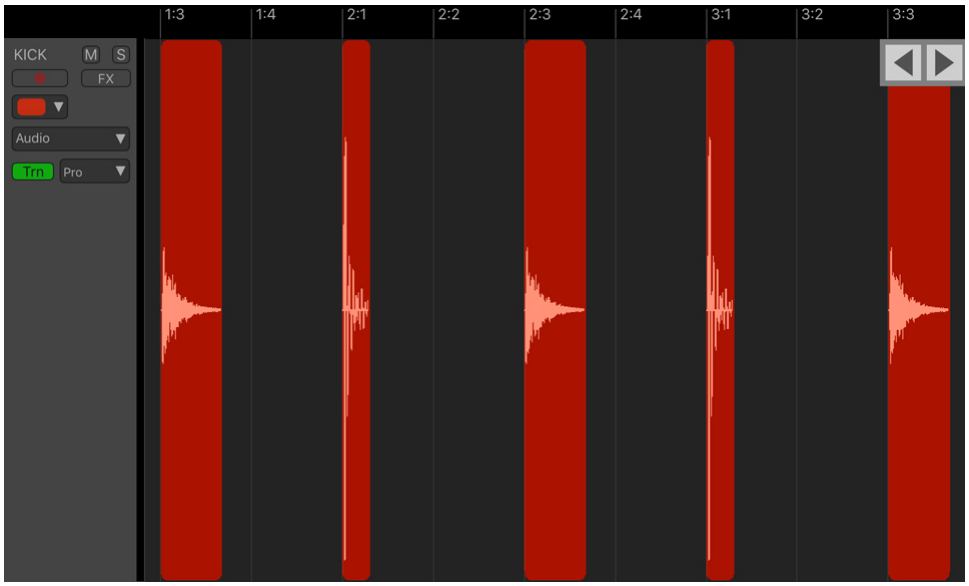
And after slicing at transient markers

Separate at Transient Markers

Similar to the above Slicing function, Separating at Transient Markers goes a step further and removes the audio found in between the End and Transient Markers.



Before separating at transient markers



And after separating at transient markers. Note the now empty space between each event

One example of using Separate at Transient Markers would be eliminating the background noise on a drum track, such as in the above illustration; any bleed from the other instruments will automatically be stripped out of the recording.

Audio Quantize

Another new function available in Auria Pro is audio quantizing, or the ability to automatically move audio events in time for the purpose of correcting unsteady timing or applying a different musical feel to a performance.

To begin quantizing an audio region, select it with a single tap and then open the AUDIO QUANTIZE dialog found inside the TRANSIENTS sub-menu of the EDIT Menu.



Audio Quantize dialog

The individual controls, such as SWING, STRENGTH, GRID vs. GROOVE, control the exact same quantization parameters as found when quantizing MIDI. Please see the MIDI Quantizing section for more information on using these specific parameters.

There are two options found in this dialog box unique to audio quantizing, whether to quantize the audio via warping or by slicing:
Audio Quantize: Warp – This mode uses a region's existing Warp Markers, and automatically moves each individual warp marker according to the quantization settings selected.
Audio Quantize: Slice – This mode will first slice the selected region at every Transient Marker, creating multiple smaller regions from the selection. Then these smaller regions are arranged in time according to the quantization settings selected.

Time-Stretching Modes

Auria Pro comes with four popular élastique time-stretching algorithms designed by zplane. Three of these algorithms are capable of real-time processing, while the Soloist algorithm is only available when real-time warping is turned off in the Settings.

The following descriptions come directly from the zplane website:

élastique Pro

élastique Pro is a general purpose time-stretching engine that offers unmatched quality and easily fulfills the high quality demands of professional productions and broadcast applications. The time-stretching engine is based on state-of-the-art psychoacoustic models and signal processing theory. élastique Pro is based on a completely new approach to time-stretching, making stretching artifacts obsolete and thus providing sharp transients and crystal clear vocals without phasing artifacts. élastique Pro offers stable timing, inter-channel phase coherence and sample accurate stretching. As a special feature, élastique Pro allows you to do formant preserving pitch shifting not only for monophonic, but also for polyphonic input files to avoid the well-known mickey-mouse effect when pitching up or down. Last but not least, all of this can be done in real-time on current CPUs.

élastique efficient

élastique efficient gives you great time-stretching quality at a stunning workload efficiency. The algorithm is targeted at complex polyphonic signals like complete mixes etc. efficient uses specifically developed technologies to efficiently detect tonal and transient components with high accuracy in the frequency and time domain. It furthermore offers different mechanisms to improve performance even more: the processing can be split into several parts in order to avoid workload peaks at low latencies and the overall bandwidth can be reduced in order to save more processing power (at the price of a slightly decreased quality level).

élastique mobile

élastique mobile is an even more efficient algorithm specifically designed for mobile processors.

élastique SOLOIST

élastique SOLOIST is targeted at monophonic input signals, and is only available when using offline warping. Thanks to this restriction, it offers incredible quality for this type of input. It allows pitch shifting with formant preservation to leave the sound's timbre/formant characteristics untouched, and has a transient copying feature to preserve crisp attacks. SOLOIST also allows splitting the analysis and the synthesis part of the processing to implement an incredibly efficient synthesis when the analysis is done offline.

Note: The default mode of élastique Pro should work well in most scenarios and is the recommended starting point when warping. If using this mode results in CPU Overload warning messages when warping then try stepping down to efficient and then mobile modes. Offline SOLOIST mode is the best option when the best sounding time-stretching is required with strictly monophonic tracks, such as solo voice. To use this mode first check that Offline Warp mode is set.

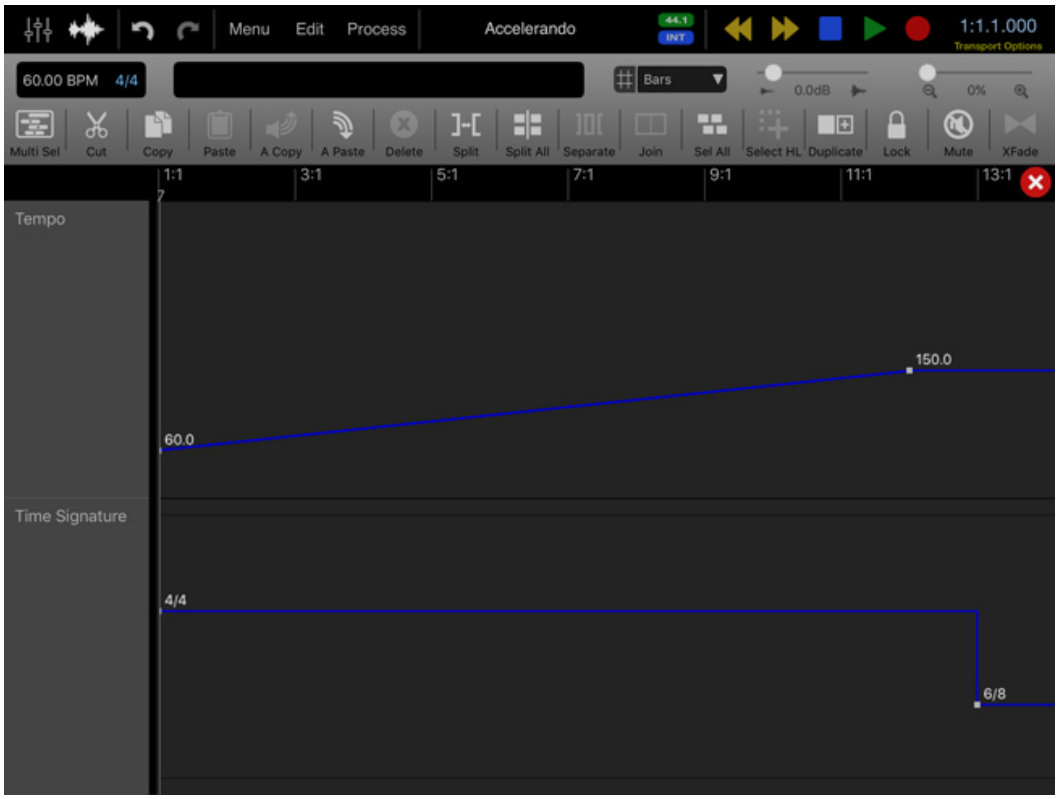
8

TEMPO & TIME SIGNATURE TRACK- PRO ONLY

Auria Pro can now work with projects that contain both changes in tempo and time signature. In addition to the existing Time and Tempo window, which allows for specifying a song's initial static tempo and time signature, there is a new view available for working with multiple changes to tempo or time signature throughout the course of the project.

The Tempo and Time Signature Track uses Auria's familiar system of Control Points (like the Automation system) to designate where specific tempo and time signature events should occur. Control points can be added and deleted and dragged to specific values.

To open the Tempo and Time Signature track (which is only available from the Edit window) open the Edit menu and select TEMPO TRACK from the list. The following window will open showing both tempo and time signature information:



The Tempo and Time Signature Track

This view is split into two halves; the top half containing tempo information, and the bottom half containing time signature information.

Editing Tempo and Time Signature Changes

To add a new Tempo control point, tap and hold at the corresponding point in time inside the top half of the window. Adjust the control point up and down until it matches the desired tempo, and move it left and right until it occurs at the right moment in time.

The lines connecting tempo control points can have their shapes defined using the pop-up window in the upper-right corner. Choose between straight linear changes, curves, and immediate jumps between points. Time signature control points can only be connected using immediate jumps between points.

To add a new Time Signature change, tap and hold at the corresponding point in time inside the bottom half of the window. Adjust the control point up and down until it matches the correct new time signature, and slide it left and right so it occurs at the desired moment in time.

Note: The Tempo and Time Signature Track defaults to a time base of Bars:Beats for the Timeline Ruler, and cannot be changed to any other time base.

Also note: Adding or altering Time Signature changes will have an affect on the Timeline Ruler display, as the Ruler's base units (Bars:Beats) is being modified whenever the time signature changes. For example, if the time signature changes from 4/4 to 3/4 at bar 14, every bar afterwards now has one less beat, so the Ruler display will shift accordingly.

9

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.

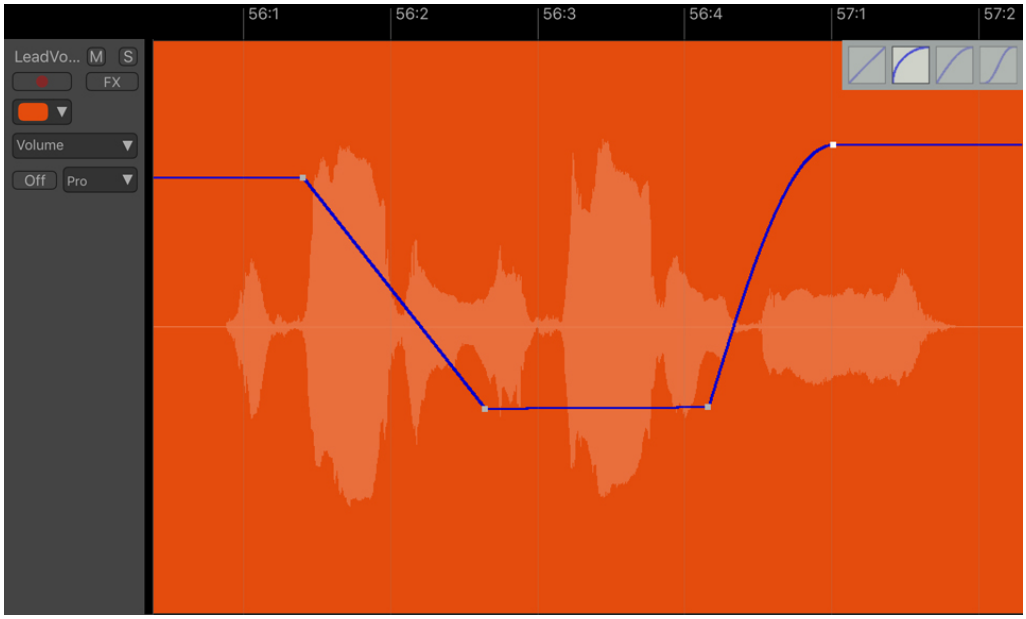
To edit automation data from the Mix window, simply overdub new automation over existing data as needed. For more fine control switch to the Edit window, which is detailed in the next section.

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.

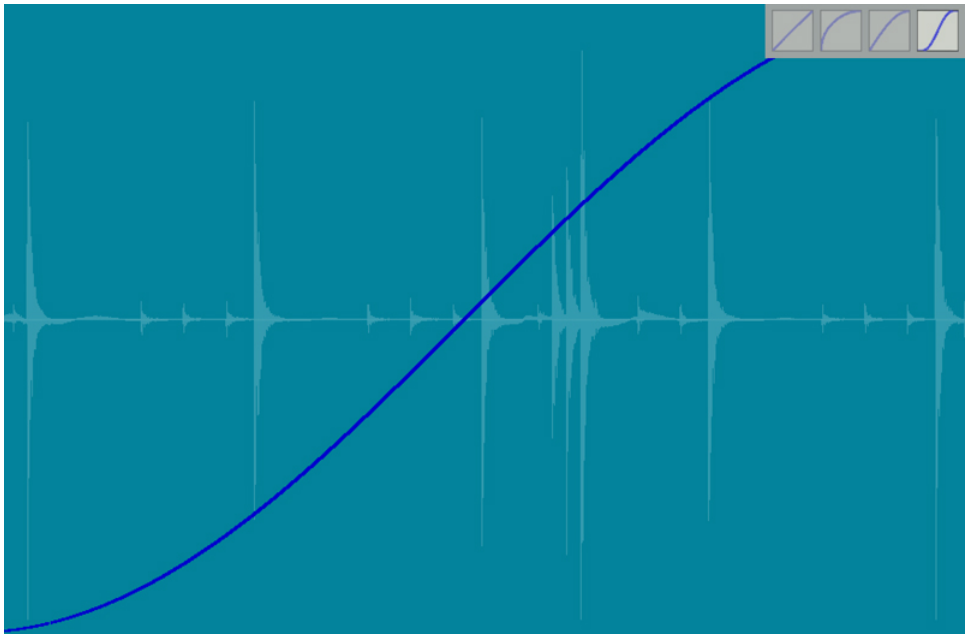


Example of volume automation, with one Control Point selected and Curves window visible

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.

10

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).

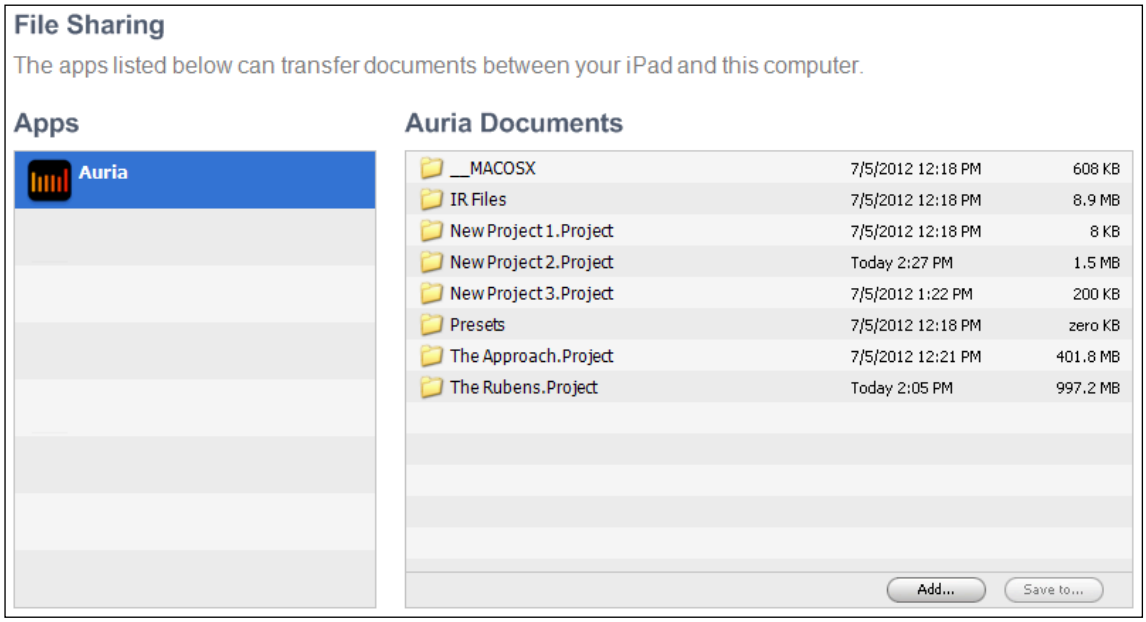
All necessary parts of a project (the project file, all audio files, and region overviews) are stored inside its bundle, making project transfers back and forth from the iPad very simple: only the project bundle itself needs to be moved (or the project directory if viewed from Windows).

Auria can 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 cleanup process can save critical space on the iPad’s internal storage. This function can be run 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

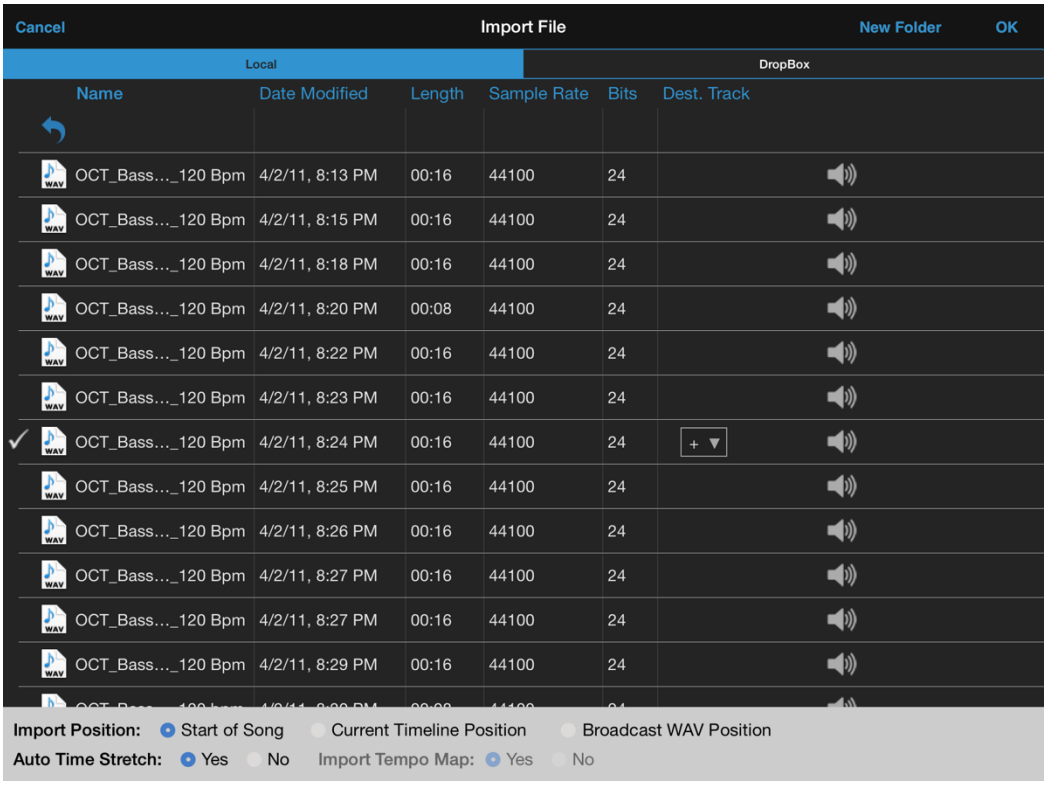
All files that are used within Auria can be maintained by connecting the iPad to a Mac or Windows system and using the iTunes File Sharing system, found in the main iTunes application. Auria will appear in the File Sharing section of the Apps tab in iTunes, and audio files, projects bundles, AAF projects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

In addition to recording audio directly into Auria (plus MIDI in Auria Pro), outside media files can be brought into projects directly. Loops, samples, or even entire other multi-track recordings can be brought in through importing.

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



The Import File dialog

All audio and Standard MIDI files, including those that have been loaded into Auria via iTunes File Sharing, will be listed here. Tap a file 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.

If importing a Standard MIDI File (only available in Auria Pro) there is an option to choose whether or not to use the outside MIDI file's own tempo map, which will overwrite the current project's tempo information.

Note: Importing large audio files can take some time, especially if the Auto Scan Transients Import option is enabled under Settings. If the particular audio file will not require any sort of audio

warping, transient slicing, or audio quantizing, then disabling this setting will greatly speed up audio file importing.

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: The AAF format only supports audio data, and will ignore any MIDI data inside a project. To transfer MIDI tracks between DAWs a second step of exporting a Standard MIDI File will need to be performed.

Additional 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.

Project 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.

Save Project to Other App – Pro Only

Using this option, found under the SAVE PROJECT sub-menu, saves the current project to another compatible app. This can be used to transfer Auria Pro projects via AirDrop to a Mac, or to backup the project using a compatible iOS storage drive, such as the SanDisk iXpand Flash Drive.

Auria Pro will automatically add the entire project to a .zip file and copy it to the selected destination. Use this to quickly backup larger projects to compatible iOS drives or other destinations.

The Apple Files App

Starting in iOS 11, Apple introduced the Apple Files App (commonly referred to as 'Files'). Files gives the iOS User much greater control and flexibility to move files around to different apps as well as on and off of the iPad.
Root app folders are known as 'Document' folders on iOS. These are the main application folders that contain all of an app's contents. The Auria Pro documents folder can be found in the Files app by navigating to 'On My iPad' (left column).

Here are a few ways that Files can be used in conjunction with Auria and Auria Pro:

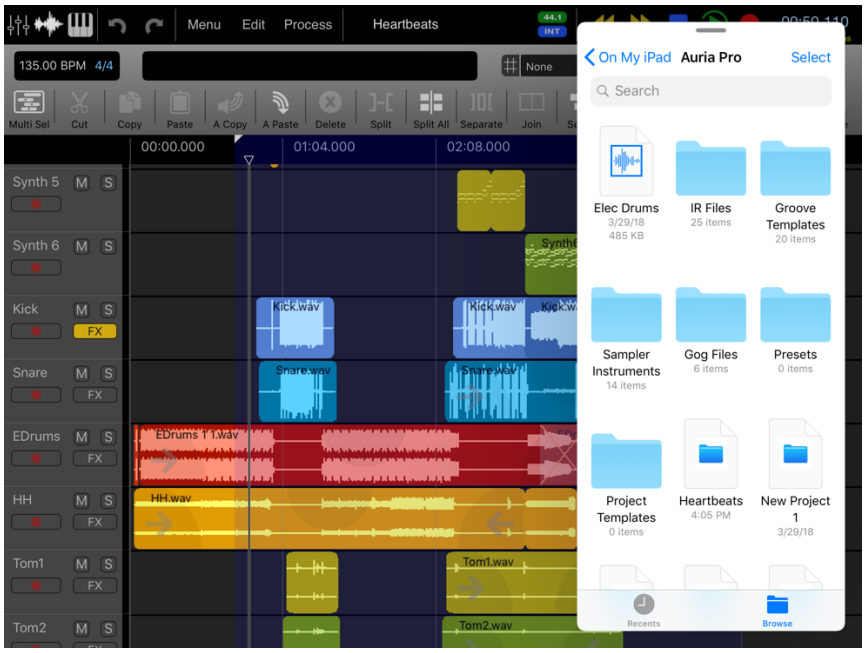
Drag and Drop

1. Make sure you have recently opened the Apple Files App so that its app icon appears in the 'recently used' app bar on the bottom of your iPad
2. Open a project in Auria and switch the Edit view
3. Using one finger, quickly flick up from the bottom of the iPad screen to bring up the recently used app bar



Recently used app bar shown in Auria Pro

4. Tap, hold and then move a finger on the Apple Files App icon to 'pop' it out. Drag and release anywhere on your screen to see the Files app float on top of Auria.



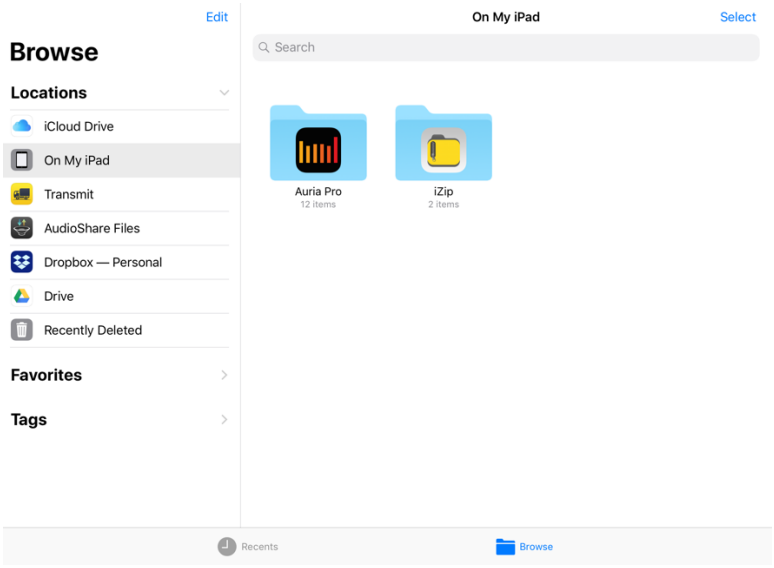
Apple Files App shown floating on the Auria Pro screen

5. You can now choose any compatible audio or MIDI file from any location in the Files app and tap, hold and drag it into the Auria Edit window. Importing files made easy!
6. To move the floating window, tap and hold on the grey bar found on the top of the window. To get rid of the window completely, move it all the way to and past the right side of the screen.

Backup and Restore

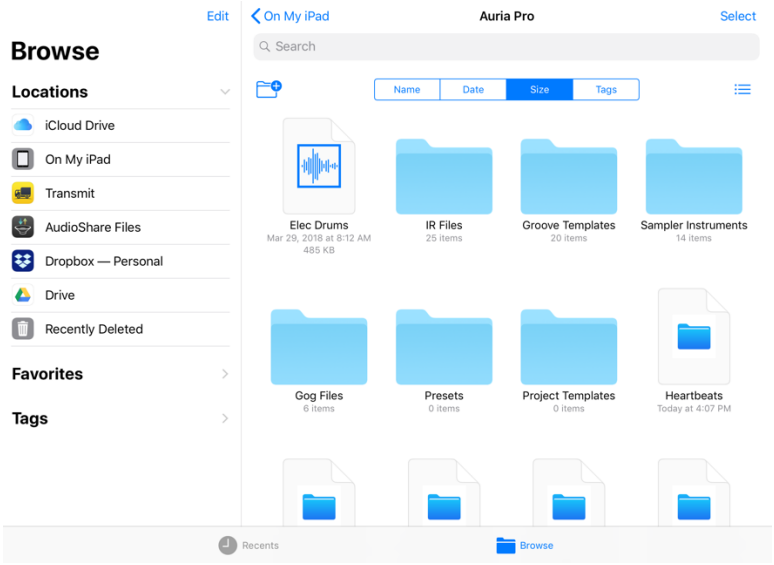
Previously, backing up and restoring files and projects from Auria was limited to only DropBox as a cloud service and having to make a physical connection to a Mac or PC and using iTunes File Sharing. The Files App has now widened the playing field as well as making transfers to a Mac much easier.

1. Open the Apple Files App on your iPad and tap on 'On My iPad' to find the Auria Documents folder.



Auria Pro Documents folder shown in the File App

2. Tap on the Auria Documents folder to see all of the contained content



Inside the Auria Documents folder in the Files App

3. To move a file from the Auria Documents folder, start by tapping on the 'select' button on the upper right
4. Tap to select the files and/or folders that you would like to work with
5. Once any file or folder is selected there will be new selections at the bottom of the screen- Share, Duplicate, Move and Delete

- **Share-** Use this option to send a file via iMessage, email or send to any app available in the Share menu. *If you would like to move/back up files to your Mac, you can AirDrop wirelessly from the Share menu.
- **Duplicate-** Use this option if you would like to retain a copy of your file/folder in the Auria Documents folder and have an additional copy to send/backup elsewhere.
- **Move-** Use Move to be able to select any location in the Files App to move your files/folders. This is the most common option to backup to cloud services. *Remember, this option will remove your file/folder from the Auria Documents folder.
- **Delete-** Permanently delete the selected files/folders

6. Use the above method to backup and restore any folder, project, audio and MIDI files to and from Auria.
7. Any compatible audio or MIDI file placed in the Auria Documents folder will be accessible in Auria via the Import Menu function

Tips

- The Files App allows you to use any compatible cloud app of your preference (Google Drive, iCloud, DropBox, etc…) to backup and restore Auria files. Keep in mind that DropBox is still the only app directly integrated with Auria, so in order to restore a project or audio file you must first move it from your cloud folder back into the Auria Documents folder.
- Direct access to the IR Files, Sampler Instruments and Gog Files (if you own Drumagog 5 for Auria) folders. This allows you to easily add new files to each of these respective folders or temporarily take some out if you are running out of room on your iPad. Some sample instruments can be a few GB in size!

11

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.

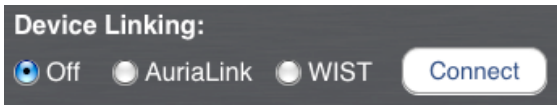
Note: Since synchronization depends on a Bluetooth connection, there will be some unavoidable latency between the two devices. This latency is quite small (usually under 20 ms), so for sync with an outside sequencer or virtual synth will probably not be noticeable. However, with AuriaLink, and especially 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:



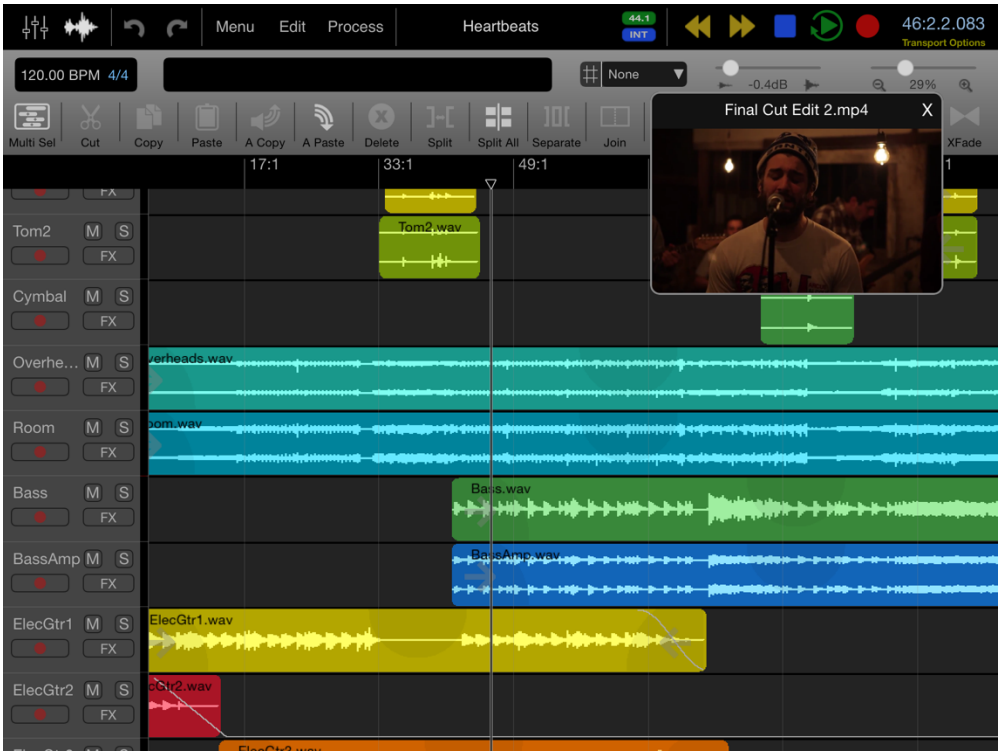
1. On the first device, running Auria, tap open the Settings window (this device will become the slave)
2. Under Device Linking select WIST
3. Tap Connect
4. The iPad will attempt to connect to any nearby iOS devices through Bluetooth.
5. A list of available devices will open, and then tap the second 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.

12

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)

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

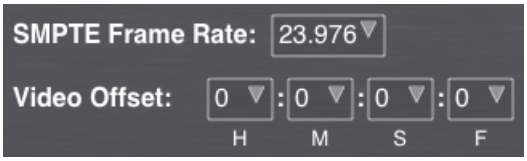
Tap OK to finish the import process.



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



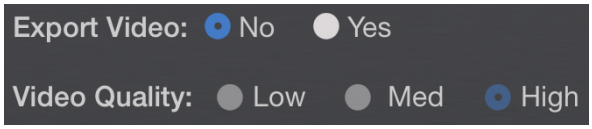
Video section of the Settings window

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.



Bottom of 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.

13

AUDIO UNITS, INTER-APP AUDIO, and AUDIOBUS

Audio Units (AU)

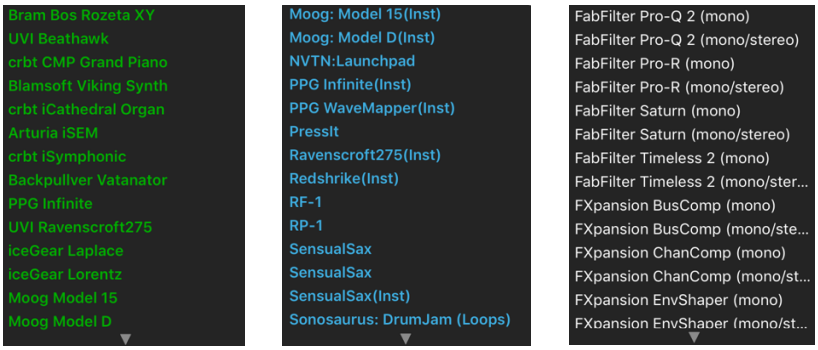
Starting in iOS 9, Apple added the plugin format commonly seen on macOS called, Audio Units to iOS. Audio Units are a major advancement over IAA on iOS and function more like the native plugins available in the Auria Store. AUs are selectable directly from the Auria inserts list (shown in green). Unlike IAA, multiple instances of the same AU can be run in the same Auria project. AUs also save their parameters and are fully automatable.

When selecting an AU plugin, the GUI will pop up and float natively in Auria much like the native plugins. Unlike IAA, there is no background app switching or hassle over syncing the transport.

Virtual instrument AU plugins can be controlled by an external keyboard or Auria Pro's built in MIDI Global Keyboard (Pro only). See the MIDI Global Keyboard section.

Audio Unit Requirements:

- Auria 2.06 (or newer)
- iOS 9 (or newer)
- 3rd party apps that support the AUv3 format



The Auria FX Inserts list: AU show in green, IAA show in blue and native Auria plugins appear in white

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 using 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 instruments on MIDI tracks and as 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 newer)
- iOS 7 (or newer)
- 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.

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

Tap Record followed by Play on the IAA transport bar to start recording in Auria
Play the instrument app
Tap the Play/Pause button again on the IAA transport to stop recording
If needed simply rewind and re-record, and if not…
…Tap the Auria icon in the transport bar to switch back to Auria
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 Sync Chapter for more information on how to set this up.

Note: Auria Pro users have an additional option when working with IAA Instruments, as they will be selectable on MIDI tracks via the INSTRUMENTS Menu. Please refer to the MIDI Sequencing chapter for more information.

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).

To use Inter-App Audio on an insert:

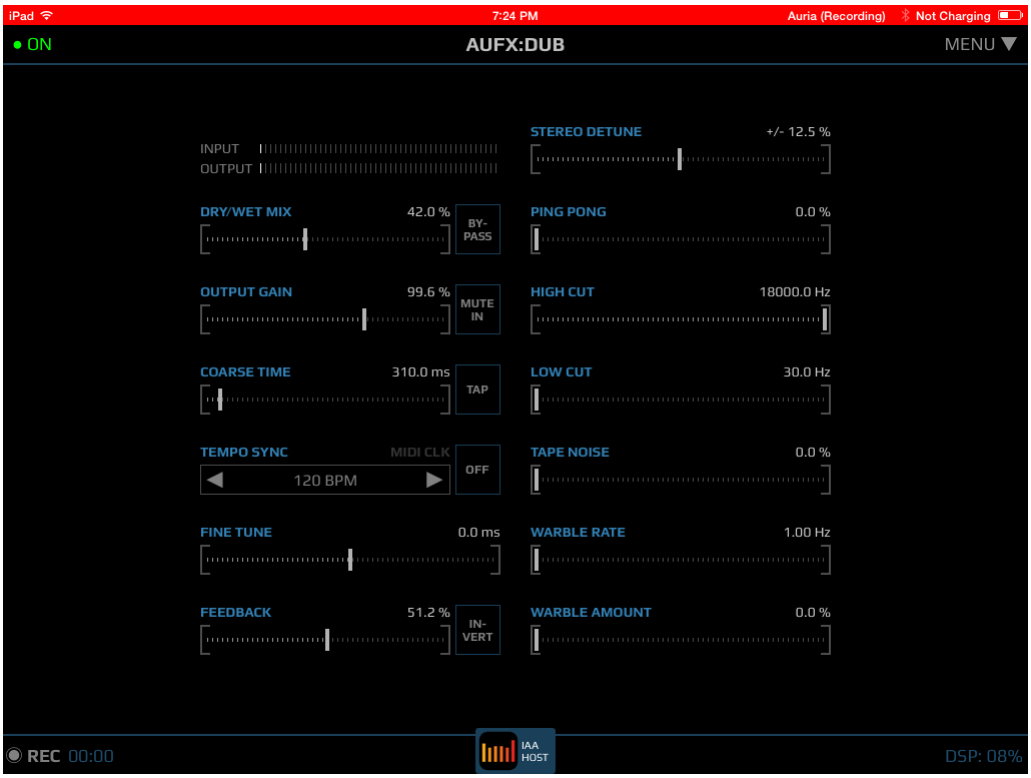
1. On any active audio track, subgroup, or master channel, open the Insert section by tapping the FX button

In the list of available plug-ins any IAA-compatible apps will appear in blue, so select the desired effect

The display will switch over to the effect's display

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

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

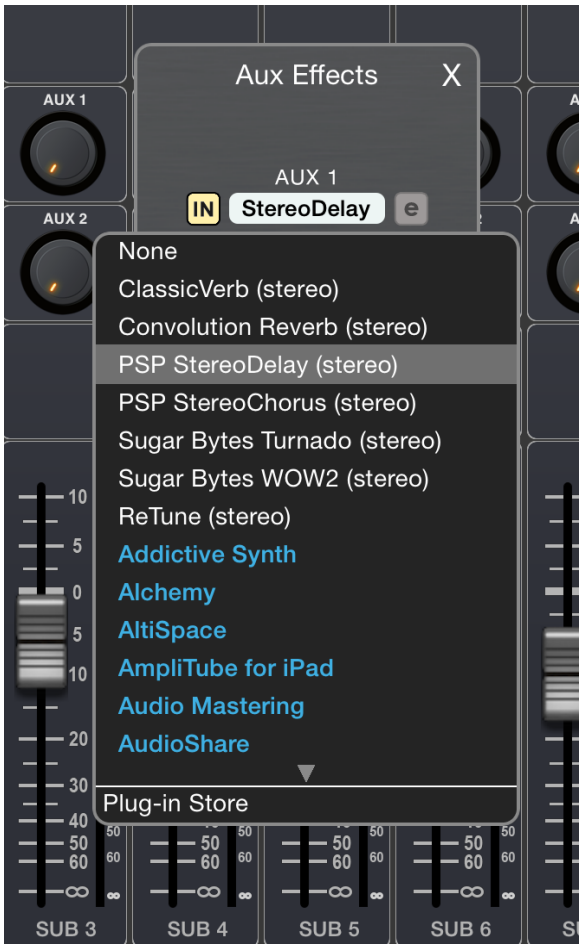
At this point the effect could be left active to run in real-time, and performing a mixdown would include this Inter-App Audio effect as it sounds in the mix. However, just like with regular plug-ins, leaving the IAA running will use valuable system resources, like ram and CPU. Additionally, as only 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 (while Auria Pro has six total aux 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 on the desired AUX channel

In the list of available plug-ins any IAA-compatible apps will appear in blue, so select a desired effect

The display will switch over to the effect's display

Adjust the effect's initial parameters as needed

Switch back to Auria by pressing the Auria icon in the effect's IAA bar

Turn up at least one corresponding aux send knob on an active track or subgroup

Tap Auria's play button in the transport

Toggle back to the effect by tapping the e button found in the Aux Effects window which corresponds with the active slot

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

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

Mute the other aux return

At this point when playing back the project only the necessary aux return should be heard

Either Mixdown to a new track, or…

…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

Un-mute all the other parts and remove the effect from the AUX FX section

Adjust the level of the new effect track as needed

AudioBus Support in Auria

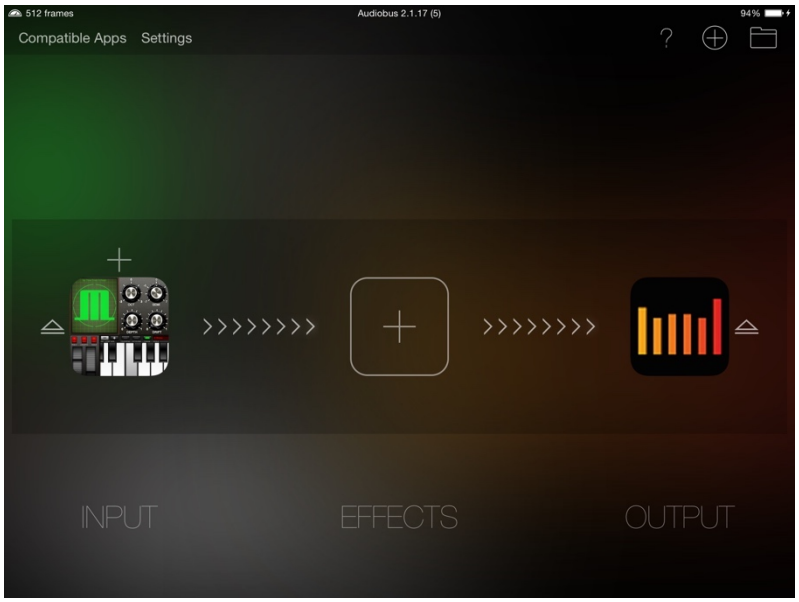
Auria supports both Audiobus input and output modes, meaning Auria can both record other Audiobus compatible apps directly onto audio tracks, or Auria can route its own outputs through Audiobus to other apps for recording/processing.

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

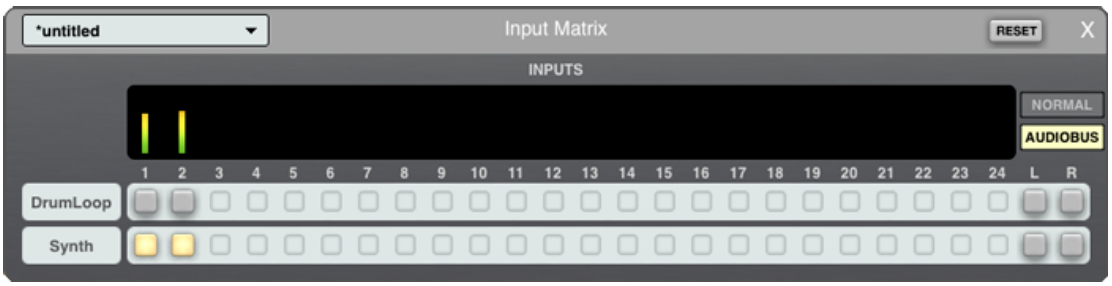


The Audiobus app setup for recording into Auria, using Magellan in the input slot

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.

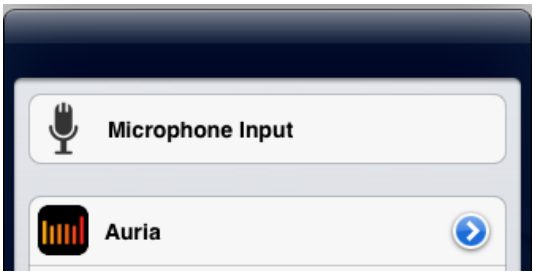
To eliminate that additional latency when recording, Auria has a mode called “Audiobus Live Monitoring” (found in the Settings Menu). If enabled, Auria will let Audiobus handle the input monitoring and bypass Auria’s mixer, lowering the amount of total latency when recording Audiobus.

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.

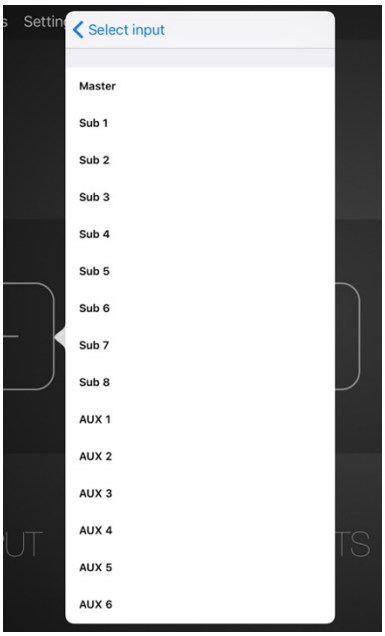
When inserting Auria in the Input (left-hand) slot Audiobus should display a blue arrow, indicating that Auria has multiple choices of outputs available.



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

As running multiple audio applications at once can greatly tax even the fastest of iPads, a few tips can be helpful when using Audiobus. If recording into large 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.

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MIDI SYNC & REMOTE

Auria can send and receive many kinds of MIDI control information, including notes, controllers, sync, transport, and even remote mixing parameters. In addition to recording and playing back musical notes, these other options 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 Protocols

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 (but not receive it) 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 recording/playback of 3rd party MIDI instruments, either virtual or external hardware. Can also be used to control 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. This is on by default.

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 CoreMIDI, sometimes referred to as 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. This mode can also be used with 3rd party iOS apps which do not properly display their

own virtual MIDI port information via CoreMIDI, simply enable Network MIDI in both apps to make the connection.

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

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

Tap both Send and Receive Remote for that port (remote control requires 2-way communication)
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.

Transient Paste MIDI Note – Used with Auria Pro's Audio to MIDI system found under the Transient menu, where an audio region's Transient Markers can be turned into MIDI data. This option sets the specific MIDI note to use when pasting the MIDI data.

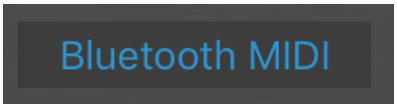
Transient Paste Fixed Velocity – Assigns the specific MIDI velocity to use when pasting Transient Markers.

MIDI Export Type – Used when exporting a project's MIDI data as a Standard MIDI File (.mid) from Auria Pro. Type 0 is a single track format, Type 1 is the multi-track format.

MIDI Base Octave – Customizes which MIDI octave is displayed in Auria Pro. There is currently no agreed upon standard for labeling octaves with notes (such as C1, C2, C4, C6, etc) so use this setting to lower the octave numbering.

Bluetooth MIDI Support

Added in Auria version 2.05, there is now full support for external, wireless Bluetooth MIDI keyboards and Control Surfaces.



Bluetooth MIDI connect button

To connect a wireless Bluetooth device:

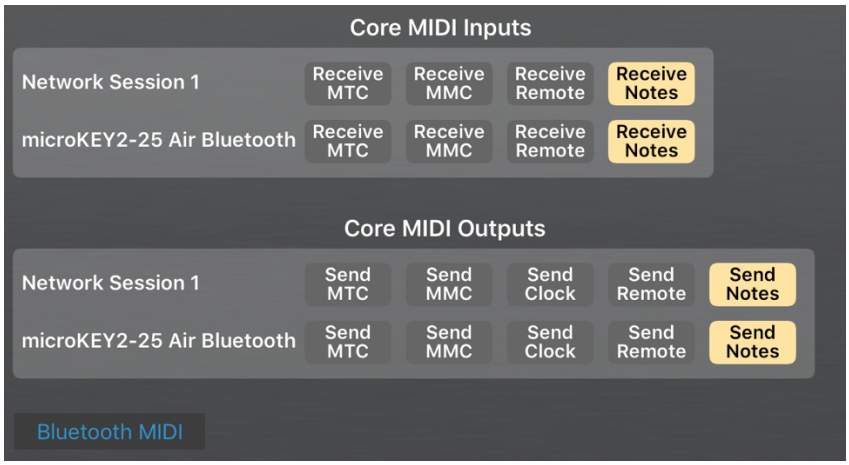
1. Ensure Bluetooth is enabled for your iPad (this can be found in the iPad Settings app)
2. In Auria Pro navigate to the Auria Settings (Menu> Settings> MIDI tab)
3. Tap the Bluetooth MIDI button (found in 'blue' below the Core MIDI i/o) to open the Bluetooth MIDI Devices pop-up window

4. Power up your Bluetooth MIDI device and make sure it is in pairing mode (refer to your device's user manual for assistance)
5. If the device is found, it will show up in the Bluetooth MIDI Devices window and be shown as 'Not Connected'
6. Tap on your MIDI device name in the list to connect the to your iPad



Bluetooth device successfully connected

Once your Bluetooth MIDI device is connected to your iPad and Auria Pro, you can access the Core MIDI
Inputs and Outputs in the Auria MIDI Settings.

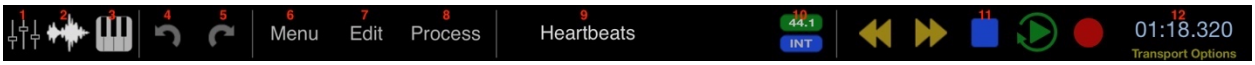


Core MIDI i/o routing will be available once your Bluetooth device is connected

Part Three – Reference

15

MENU BAR



The Top Menu Bar

1. Mixer Window
2. Edit Window
3. MIDI Global Keyboard
4. Undo
5. Redo
6. Main Menu
7. Edit Menu (only available in Edit Window)
8. Process Menu (only available in Edit Window)
9. Project Name
10. Sample Rate/USB Interface Indicator

(not shown) Grouping (only available in Mix Window)

(not shown) Locator In/Out Point (only available in Mix Window)

11. Transport
12. 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.
- **Snapshots: Load Snapshot** – Loads a previously saved mixer snapshot.
- **Snapshots: Save Snapshot** – 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: Save Project as Template** – Saves an empty template based on the current project's track structure
- **Save Project: Save Copy of Project** – Saves a copy of the current project locally under a different name.
- **Save Project: Save Project to DropBox** – Saves a copy of the project to DropBox.
- **Rename Project** – Changes the name of the current project.
- **Import File** – Imports audio files (WAV, AIFF, M4A) and MIDI files (.mid). 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 (Mono/Stereo) or MIDI tracks.
- **Mixdown** – Creates a final mixdown of the project. For more information see the Mixdown section.
- **Export MIDI File** – Saves a Standard MIDI File of the current project's MIDI data. Pro Only.
- **AAF File: Import 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.
- **AAF File: Export AAF file** – Exports project as an AAF file and saves it locally.
- **AAF File: Export AAF to DropBox** – Exports project as an AAF file and saves it to DropBox.
- **Reset: Reset Mixer** – Resets all parameters on the mixer.
- **Reset: Clear All Groups** – Clears all grouped channels.
- **Reset: 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 a link to Auria's User Guide.
- **Auria Forum** – Link to Auria's official discussion board.
- **Auria Store** – Access the optional In-App Store, with purchasable plug-ins, loops, convolution IR files, samples 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.

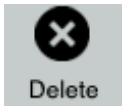
- **Undo** – Undo the most recent action.
- **Redo** – Redo the previous action.

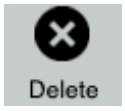
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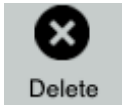
Cut – Cuts the region or highlighted selection.
- 

Copy – Copies the region or highlighted selection.
- 

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

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

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.
- 

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

- **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.
- 

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

Mute/Unmute Region – Mutes/unmutes the selected region(s).
- **Rename Region** – Renames the selected region.
- 

Share: 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.
- 

Share: 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.
- **Share: 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.
- **Share: Import from AudioShare** – Pastes audio from the AudioShare app into the current Auria project.
- **Transients: Create Transients Markers** – Creates new transient markers, see Audio Warping chapter for more information.
- **Transients: Add Transients Marker** – Manually adds transient marker at the Cursor.
- **Transients: Clear Transients Markers** – Deletes the selection's transient markers.
- **Transients: Audio Quantize** – Automatically quantizes the selection's audio, either via Warp Markers or slicing its transients. Please see the relevant MIDI information for help with the fundamentals of quantizing.
- **Transients: Slice at Transients Markers** – Splits the selected audio at every transient marker, creating multiple regions.
- **Transients: Separate at Transients Markers** – First splits the selected audio into multiple regions at every transient marker, and then deletes the audio after End Markers and the before the following transient marker.
- **Transients: Copy Transients Markers** – Copies the selected regions' transient markers to the Clipboard as MIDI.

- **Transients: Paste as MIDI** – Pastes the Clipboard's content as MIDI data. Set specific MIDI note and velocity in Settings.
- **Transients: Show End Markers** – Toggles end markers on/off, which indicate the moment the audio following a detected transient falls into silence. Used when Separating at Transient Markers or Quantizing.
- **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.
- **Tempo Track** – Opens the Tempo and Time Signature Track.
- **Link Locators and Highlight** – When enabled setting the Locator points will also automatically highlight the same section, and vice-versa. Turned off by default.
- **MIDI Record Overdub Mode** – Determines whether to combine old and new MIDI recordings when overdubbing, or to replace the old recording with the new.
- **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. The Process Menu will differ depending on whether audio or MIDI data is selected.

Audio processing functions:

- **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.
- **Pitch** – Pitch shifts the selected audio, includes separate controls for adjusting both pitch and formant (harmonic content). For more information on the available algorithm types please see this section regarding élastique.

- 

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.
- **Bounce Track in Place** – Mixes-down selected track and applies any current non-destructive effects or processes, making them destructive. The newly processed track will take the place of the original. Can be used to bounce instruments on MIDI tracks.
- **Use as Groove Template** – Copies the selected events (like Warp Markers or MIDI notes) to the Groove Clipboard, making it available when utilizing Groove Quantizing.

MIDI processing functions:

- **Crescendo** – Creates a gradual change of the selected notes' velocities, use the FROM and TO controls to specify the relative starting and end volumes.
- **Delete Controller** – Deletes the specified MIDI controller data (CC#) found in the selection.
- **Delete Notes** – Deletes the specified range of notes contained inside the FROM and TO range.
- **Fixed Length** – Changes the durations of the selected notes to the length specified.
- **Fixed Velocity** – Changes the velocity value of the selected notes to the amount specified.
- **Humanize** – Adds a selectable amount of randomness to the timing and velocities of the selected notes.
- **Legato** – Adjusts the selected notes' lengths so that each note is sustained until the start of the following note.
- **Optimize Controller Data** – Removes un-needed controller data while maintaining an identical-sounding performance.
- **Pedal to Length** – Converts sustain pedal messages into longer MIDI notes.
- **Quantize** – Destructively quantizes the note timings based on the selected settings. For more information please see the section on MIDI Quantizing.
- **Restrict Polyphony** – Decreases the number of simultaneously playing notes by shortening note length, based on the selected maximum polyphony value.
- **Reverse** – Reverses the order of selected MIDI notes.
- **Transpose** – Raises or lowers the pitches of selected notes by the selected number of semitones.

- **Velocity Compressor** – Compresses the dynamics of MIDI notes in a similar fashion as an audio compressor.
- **Velocity Gain** – Adds or removes the selected value to the note's velocity.
- **Velocity Limiter** – Limits the dynamics of MIDI notes in a similar fashion as an audio limiter.
- **Velocity Range** – Limits the absolute velocity values of the selected notes to the specified range.
- **Velocity Rescale** – Scales the velocities of the selected notes by the percentage indicated.
- **Bounce Track in Place –** Mixes-down selected track and applies the current instrument plus any current non-destructive effects or processes. The newly processed track will take the place of the original.
- **Use as Groove Template –** Copies the selected events (like Warp Markers or MIDI notes) to the Groove Clipboard, making it available when utilizing Groove Quantizing.

Project Name – Displays name of the current project. Double-tapping opens Rename Project dialog.

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. Global Solo light appears here, lighting up whenever any channel is soloed; tap the flashing Solo indicator to turn off all solos. Auto-Punch indicator light also appears here, indicating whenever the mode is enabled.

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. Only available in Mixer window.

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. Only available in Mixer window.

Transport

- **Rewind** - Double tap to rewind the cursor to beginning of project
- **Fast Forward** - Double tap to move the cursor to the end of the last region
- **Stop -** Double tap to rewind the cursor to beginning (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
- **External Sync** – When enabled Auria will chase external MTC sync.
- **Set Marker 1-4** – Places a marker at current cursor position.
- **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.

16

MIX WINDOW



The Mix Window

The mixer window consists of:

1. Channels

Subgroups

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


FX
1
2
AUTO
R
W
3
INPUT
--
4
OUTPUT
--
5
AUX 1
AUX 2
6
PAN
7
L
R
10
5
0
5
10
20
30
40
50
60
8
0
5
10
20
30
40
50
60
9
2
M
1
0
S
11

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. Input – Assigns track input, depending on whether track is Record Enabled:

- Record Enabled – Lists available hardware inputs.
- Record Disabled – Includes TRACK, for normal playback from currently recorded track; or available Buses, used to return a particular bus to a channel (Pro Only).

5. Output – Assigns track output to either:

- Main L/R mix
- One of eight stereo subgroups
- Buses – multiple buses can even be assigned (Pro Only)
- Direct Outs – Assign the track to a specific pre-fader hardware output (Pro Only)

6. AUX – Adjusts the level of audio being sent to the currently selected AUX FX.

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

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

9. 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.

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

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

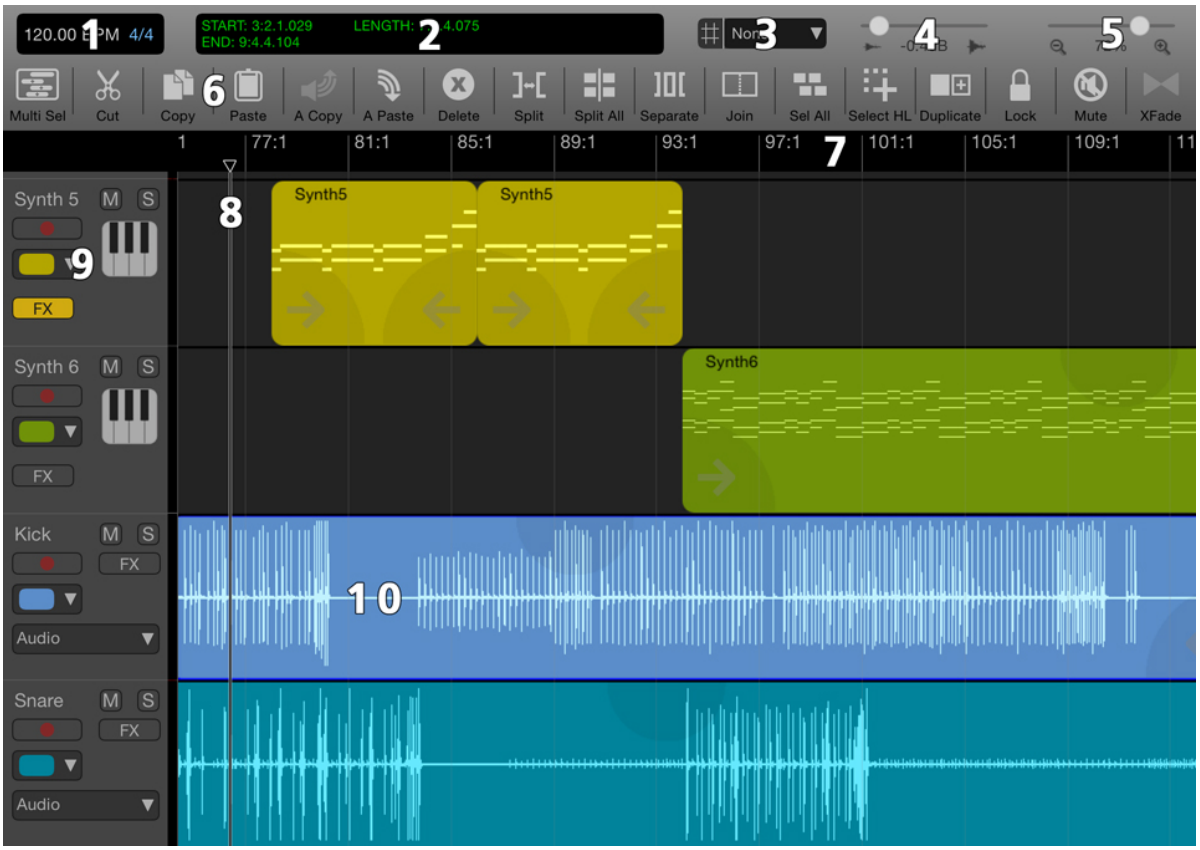


Master Channel

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 Effects window, used to add/remove Insert Effects.
4. AUX Master – Adjusts the overall (master) level of all audio being processed through the specific AUX FX to the Master Channel.
5. AUX M/S –Mute/Solo for the AUX returns.
6. AUX Selector – Toggles between which of the three banks of Aux Channels are currently being displayed (Pro Only).
7. Fader – Adjusts the signal level of the Master Channel.
8. Volume Meter – Displays the signal level. Meter Type (Peak/RMS) can be selected in Menu > Settings.
9. Meter – Opens the PSP MasterMeter.

17

EDIT WINDOW



The Edit Window

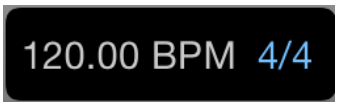
The Edit Window consists of:

1. Time and Tempo Settings
Region Info Box
Snap
Waveform Display Gain
Zoom
Icon Toolbar
Timeline Ruler
Cursor
Track Display
Regions

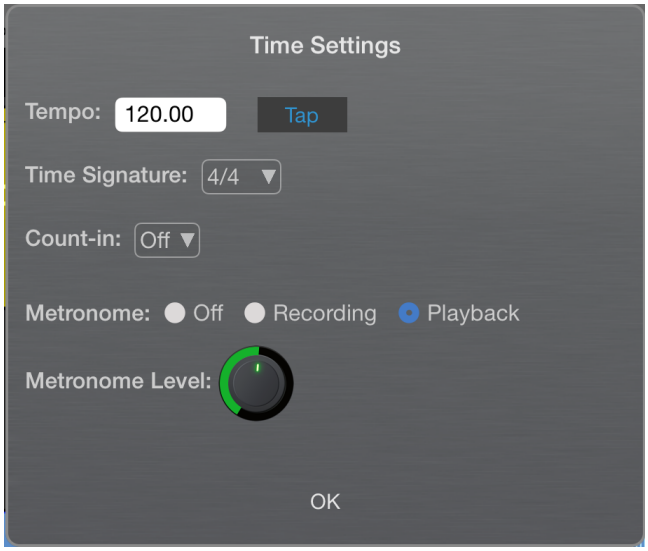
Swiping left or right with one finger will scroll to bring off-screen areas into view, swiping up or down will scroll between tracks. Pinching with two fingers will zoom either vertically or horizontally, though only one direction at a time.

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:

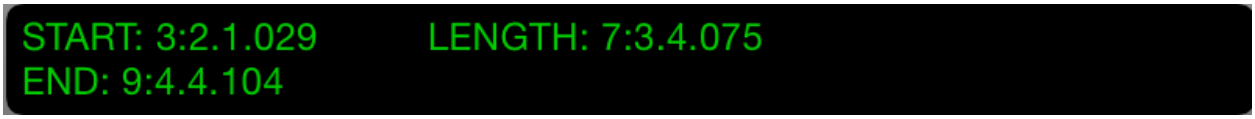


Time and Tempo Settings Dialog

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.

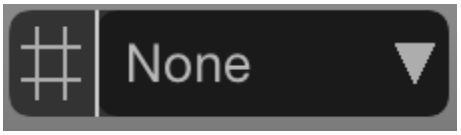
Use Tempo Track – Determines whether to follow the project's tempo track or ignore it. If enabled the above tempo and time signature settings will be unavailable. Pro Only

Region Info Box



Displays information pertaining to either the selected region, highlighted selection, or MIDI data (Pro Only)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
- Transients

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

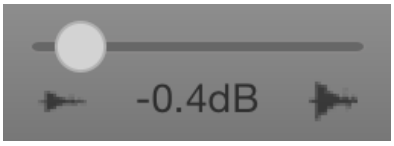
Time Format: Bars:Beats

- Bars
- Beats
- 1/4 Beat
- 1/8 Beat
- 1/16 Beat
- Triplet versions of above
- Duplet versions of above
- 9 Tuplet
- 7 Tuplet
- 5/4 Tuplet
- 5/8 Tuplet

Time Format: SMPTE

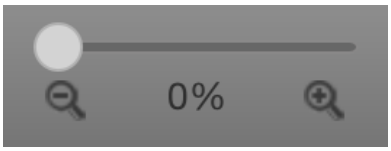
- Frame
- 1/2 Frame
- 1/4 Frame
- 1/8 Frame
- 1/16 Frame

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



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 (double-tap and slide along the timeline ruler) they will appear on the timeline, and can also 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.

Audio Tracks



1. **Track Name**

M – Mute

S – Solo

Rec – Record Enable button.

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

Color – Changes the color of regions displayed within the track.

Waveform/Automation – Selects which automation parameter is displayed. 'Audio' displays the waveform only.

Transient/Warp View – Toggles between displaying Transient Markers, Warp Markers, or None. Pro Only

élastique Mode – Toggles between the available real-time warping algorithms. Pro Only

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.

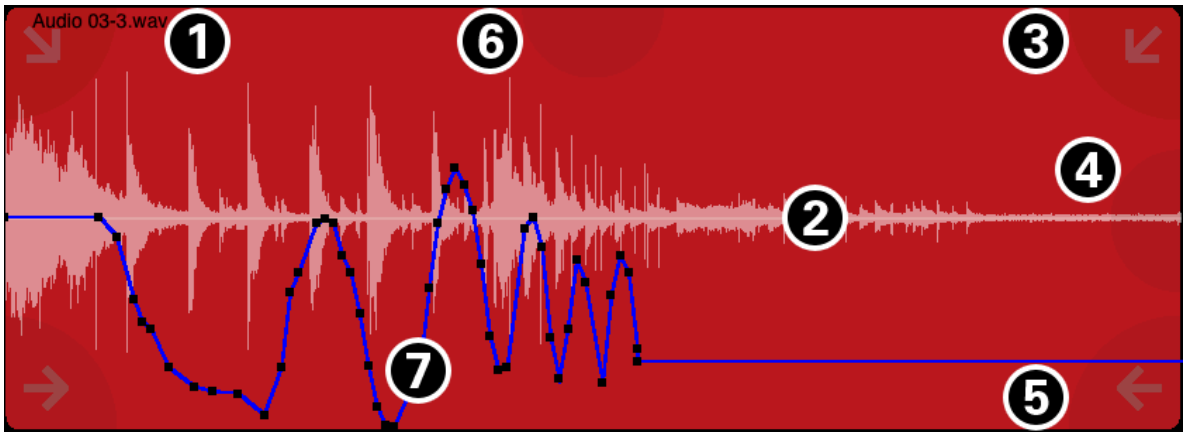
To move a track, touch and hold the track name. It will pop up, indicating it is ready to be moved, and can be dragged to its desired position.

MIDI Tracks – Pro Only



1. **Track Name**
2. **M** - Mute
3. **S** - Solo
4. **Rec** - Record Enable button
5. **Piano Roll** – Open Piano Roll editor
6. **Color** - Changes the color of regions displayed within the track.
7. **FX** – Opens the real-time MIDI CONTROL parameters
8. **Instrument** – Selects the desired MIDI instrument and patch
9. **e** – Opens the selected instrument's GUI
10. **MIDI/Automation** - Selects which automation parameter is displayed. 'MIDI displays MIDI notes only.
11. **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.

Regions



1. **Region Name** – Corresponds to region's filename.

Waveform

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.

Region Duplication Handle – The arrow on the far-right is the automatic duplication handle. See the Automatic Region Duplication section for more information.

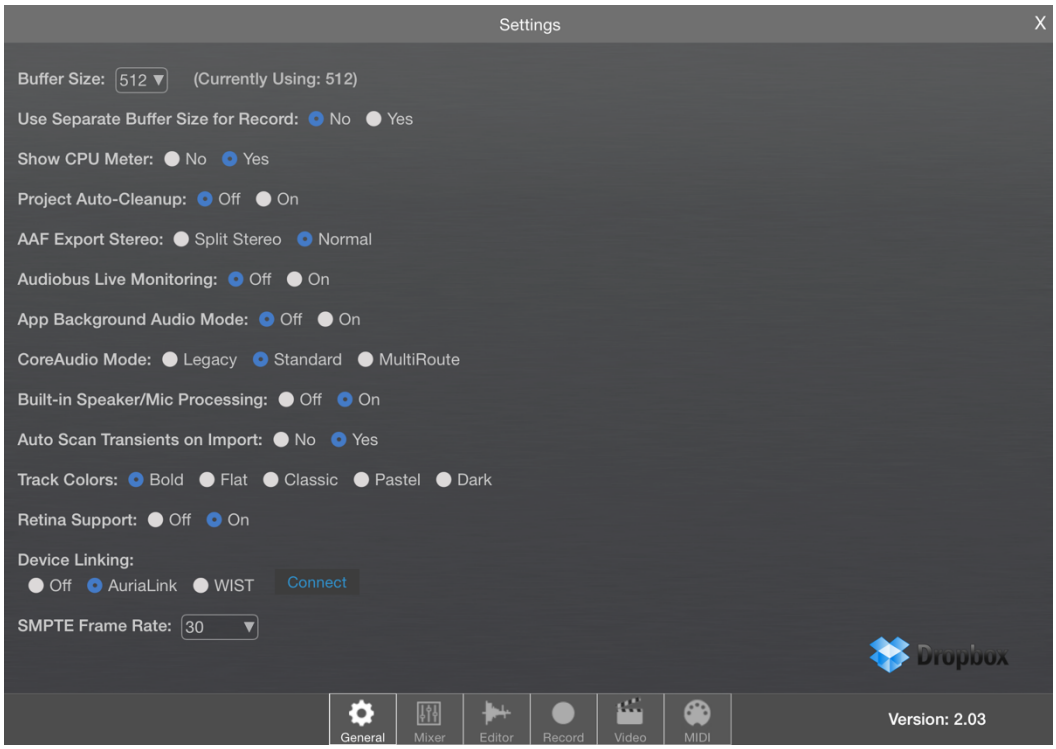
Trim Handles – The arrows in the lower left and right are the trim handles. Tap, hold, and swipe to adjust. See the Handles section for more information.

Gain Handle – The arrow at the very top in the middle is the gain handle. See the Gain Handle section for more information.

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.

18

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

Buffer Size – Drop-down box that selects the size of the audio 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. If another app has priority it may set a different buffer size, in this case the actual buffer size will be displayed in parenthesis.

Use Separate Buffer Size for Record – If no the same buffer sizes are used for both recording and playing back. If yes, then a separate selectable recording buffer size is available.

Record Buffer Size – Drop-down box that selects the size of the audio buffer when recording. Allows for using a low latency buffer size for recording and a larger, more stable buffer for playback.

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, and the BATT/SPACE meter, which shows remaining battery charge and storage space.

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 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.

Auto Scan Transients on Import – Determines whether new audio files are scanned for transients and have transient markers created. Enabled by default.

Track Colors – Switches between the following track color schemes:

- Bold
- Flat
- Classic
- Pastel
- Dark

Retina Support – Toggles between Retina Mode's higher resolution graphics or a resource-saving lower resolution mode. Older iPads may see better performance with Retina Support disabled. Enabled by default.

iPad Pro Screen Optimization (iPad Pro models only) – Toggles the graphics optimization to fill the extra screen space *only available when using an iPad Pro. See the iPad Pro Screen Optimization chapter.

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.

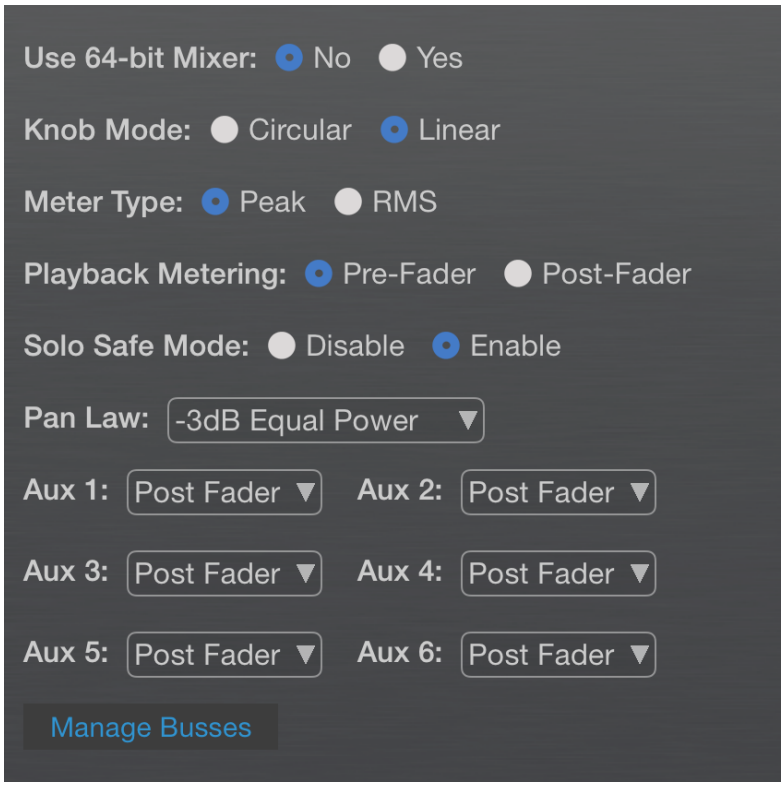
SMPTE Frame Rate – Sets the project's frame rate, influences the Frames time display in the Counter, Timeline, and Snapping features. Also used when sending MIDI Time Code.

Cleanup Project – Scans the current project and deletes unused audio regions from disk. This is a destructive process so before cleaning be sure to backup critical projects.

Transfer Purchases from Auria - Transfers any Auria plug-in purchases over to the Pro version. See this section for much more information.

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.

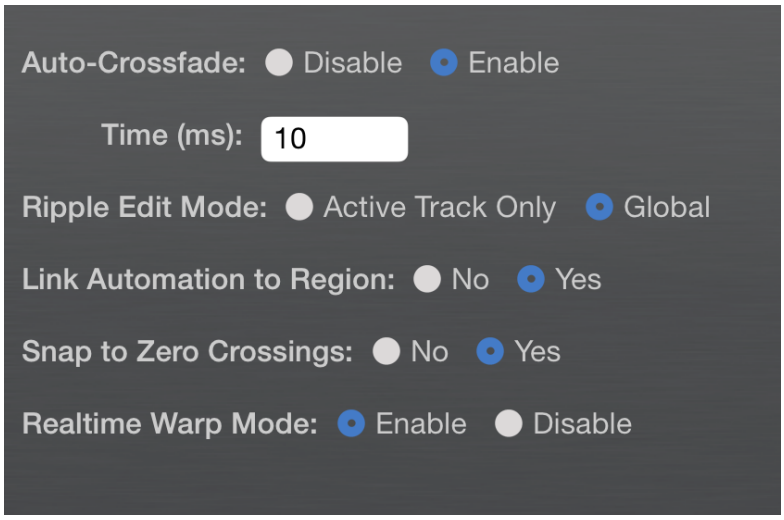
Solo Safe Mode – When enabled, soloing a track will also solo the AUX returns.

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

Aux Send Mode – Assigns either pre or post fader operation to Aux Sends. Default is post-fader. Pro version will have total of six auxes.

Manage Buses - Allows the creation/deletion and naming of available buses, up to 32 total. (Pro Only)

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:

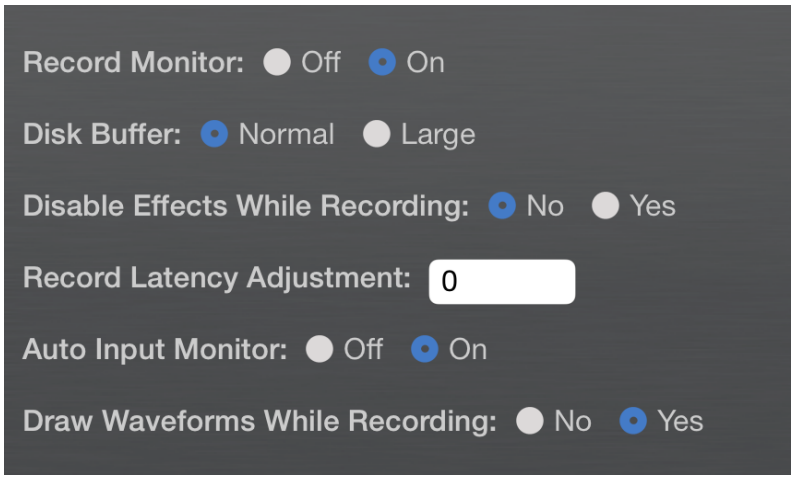
- Active Track Only – Only the specific track being edited will ripple
- Global – All project tracks will be affected by the edit

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.

Snap to Zero Crossings - Toggles whether any edits performed on audio regions automatically snap to the nearest zero crossing, preventing unwanted clicks in the audio. Disable when sample-accurate editing is needed. Enabled by default.

Real-time Warp Mode – Toggles whether Auria Pro will process moving Warp Markers in real-time or offline. Older iPads may see performance improvements with this disabled. Enabled by default.

Record Settings



Record Settings

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

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.

Auto Input Monitor – Toggles whether Auria will automatically monitor both inputs and outputs when recording. Enabled by default.

Draw Waveforms while Recording – Toggles whether Auria Pro should draw audio waveforms in the Editor while recording that same audio. Slower iPads may benefit from disabling this. Enabled by default.

Video Settings

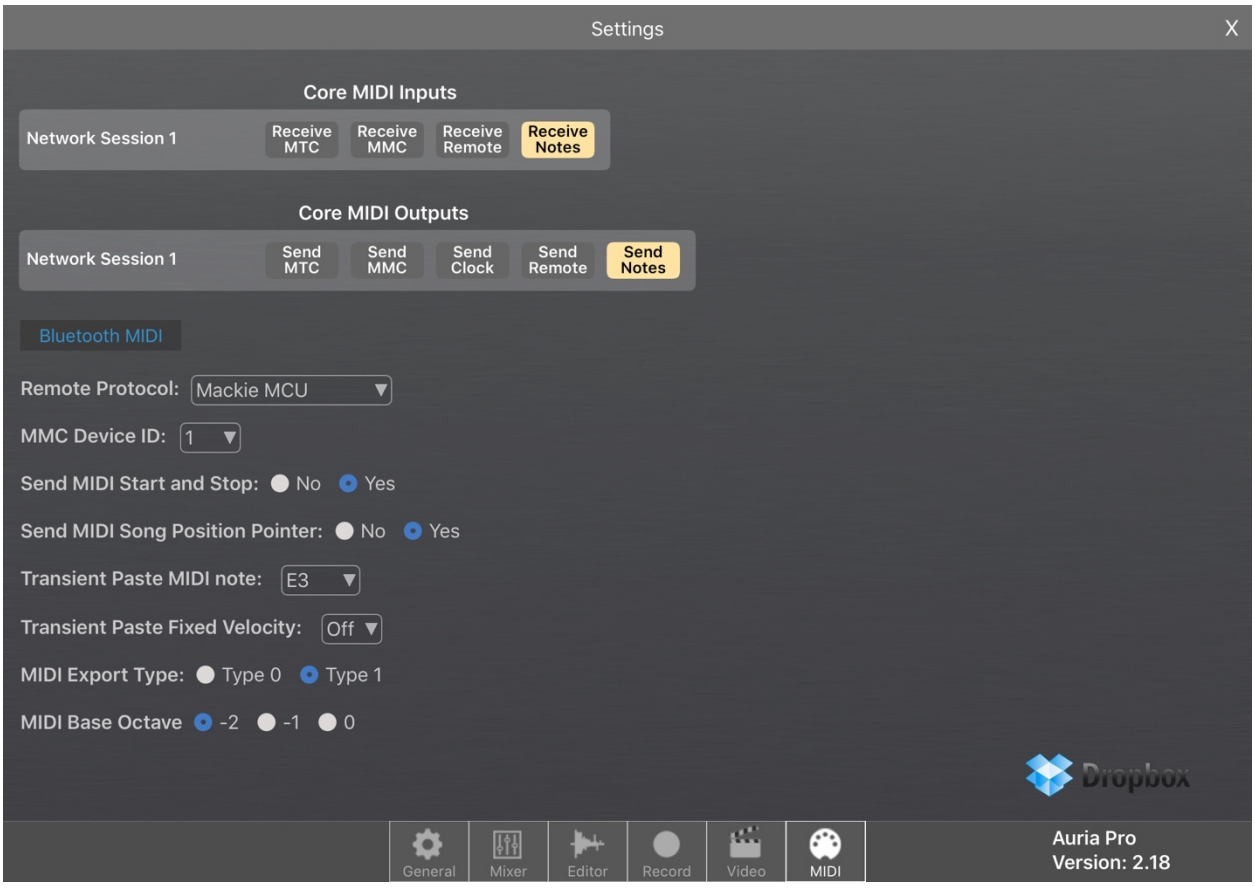


Video Settings

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 different uses of MIDI Sync and Remote options please refer to the MIDI Sync 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.

Bluetooth MIDI- Used to open the Bluetooth MIDI Devices window to connect a wireless Bluetooth MIDI device. Please refer to the Bluetooth MIDI Devices section

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.

Transient Paste MIDI Note – Used with Auria Pro's Audio to MIDI system found under the Transient menu, where an audio region's Transient Markers can be turned into MIDI data. This option sets the specific MIDI note to use when pasting the MIDI data. Pro Only.

Transient Paste Fixed Velocity – Assigns the specific MIDI velocity to use when pasting Transient Markers. Pro Only.

MIDI Export Type – Used when exporting a project's MIDI data as a Standard MIDI File (.mid) from Auria Pro. Type 0 is a single track format, Type 1 is the multi-track format. Pro Only.

MIDI Base Octave – Customizes which MIDI octave is displayed in Auria Pro. There is currently no agreed upon standard for labeling octaves with notes (such as C1, C2, C4, C6, etc) so use this setting to lower the octave numbering. Pro Only.

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MIDI CONTROL PARAMETERS – PRO ONLY



Auria Pro's MIDI CONTROL window

Every MIDI track in Auria Pro has its own real-time controls available for modifying MIDI playback on the fly. The controls found in this window can be used during playback and will affect the sound of the particular track, much like tweaks to the ChannelStrip will be immediately heard.

Instrument Panel

Instrument – Selects a specific MIDI instrument and patch/sample. Auria Pro's own instruments will appear on this list, as will any 3rd party compatible Inter-App Audio instruments installed on the iPad. Default is Lyra's Grand Piano sample.

e – Opens the selected instrument's graphic interface.

Channel – Assigns the MIDI track to a specific MIDI channel. Default is All (Omni).

Input – Assigns the track's MIDI input source, such as a specific IAA instrument or external MIDI interface. Default is All (Omni).

Output – Assigns the track's MIDI output destination, such as a specific IAA instrument or external MIDI interface. For use only when the track's output is an external instrument or device. Default is Off .

Parameters Panel

Velocity Shift – Adds or subtracts the selected amount to the velocity value of every MIDI note.

Velocity Compression – Multiplies every note's velocity value by the percentage selected. Values above 100% increase note volumes, values below 100% quiet note volumes.

Length Compression – Multiplies every note's duration by the percentage selected. Values above 100% lengthen the notes, values below 100% shorten the notes.

Random 1 – Adds randomness to the selected parameter, including note Velocity (loudness), Length (duration), and Position (timing).

Random 2 – Adds randomness to a second parameter, with the same options as above.

MIDI Delay – Delays the notes the selected amount.

Transpose – Raises or lowers the pitch of every note the selected amount, by half step (semitone).

Legato – When switched on this lengthens the note durations so each note sustains until the start of the very next note.

Bank – Assigns a specific MIDI Bank Number to the track. Primarily for use with external MIDI instruments to recall a desired patch.

Patch – Assigns a specific MIDI Patch Number to the track. Primarily for use with external MIDI instruments to recall a desired patch.

Quantize Panel

Quantize Mode – Selects between a strict Grid-based quantizing engine or a pattern-based Groove engine, and will determine which of the following controls are displayed.

Grid – Sets the desired grid value for the real-time quantizing engine. For example, selecting 1/4 Note results in every MIDI note being shifted in time towards the nearest 1/4 note, while selecting Whole Note moves every note to the nearest measure downbeat.

Length Grid – Quantizes the ends of notes to the selected note value. Default is Off.

Swing – Adds a swing rhythm feel (i.e. syncopation) to quantized notes.

Random – Adds some randomness to the timing of the quantized notes.

Strength – Determines, by percentage, how far to move the quantized notes from their original position towards the selected Grid value. For example, 100% Strength will move every note to exactly the nearest grid value, while 50% Strength will move them half-way towards the grid. Default is 100%.

Window – Determines a time window where only those notes already inside that window are to be quantized. Percentage based value related to the selected Grid value. For example, a 50% Window only moves those notes which are already close to the selected Grid value, while a 100% Window results in every note being moved.

Groove template select (Groove mode) – Available when using the Groove Quantize tab, selectable at the top of the quantizing section, this selects an available groove pattern to use as the template for the real-time quantizing engine. Includes any installed DNA Groove Templates, custom saved patterns, plus any MIDI data copied to the Clipboard (in connection with the USE AS GROOVE TEMPLATE process)

Save (Groove mode) – Saves the groove pattern currently on the Clipboard as a new template, making it available for use in other projects. Stored inside the Groove Templates directory.

Resolution (Groove mode) – Assigns a new note value reference to the current pattern. Only needs to be changed when creating new custom Groove Templates from the Clipboard, and should be set equal to the smallest note duration contained in the pattern.

Velocity Strength (Groove mode) – Determines how much of the template's velocity data should be imparted to the quantized notes. 100% Velocity Strength will completely replace the quantized notes' velocities with the corresponding values in the selected template, while 50% Velocity Strength will only shift the quantized notes' velocities half-way from their current value to the pattern value.

Preserve Note Lengths – Available in both quantizing modes (Grid and Groove), this determines whether the quantized notes' lengths should be preserved or affected by the Grid value. The default value is On, which keeps the original note durations.

Levels Panel

MIDI Volume – Assigns a MIDI volume message to the track, for use primarily with external MIDI instruments.

MIDI Pan – Assigns a MIDI pan message to the track, for use primarily with external MIDI instruments.

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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/Gate



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.

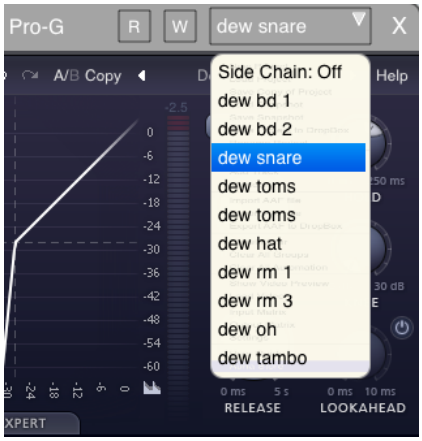
One common example would be when working with kick drum and bass guitar the two parts often compete for the same low-frequency range and don't "sit" together easily. So, a compressor is inserted on the bass guitar track, and the kick drum track's signal is then fed into that compressor via a side-chain 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.

- Applies soft bypassing to avoid clicks. While the plug-in is bypassed, the EQ display dims and a red light glows in the bypass button itself.
- 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.