



fBox Console

Virtual Analogue Mixing Console

Standalone Application

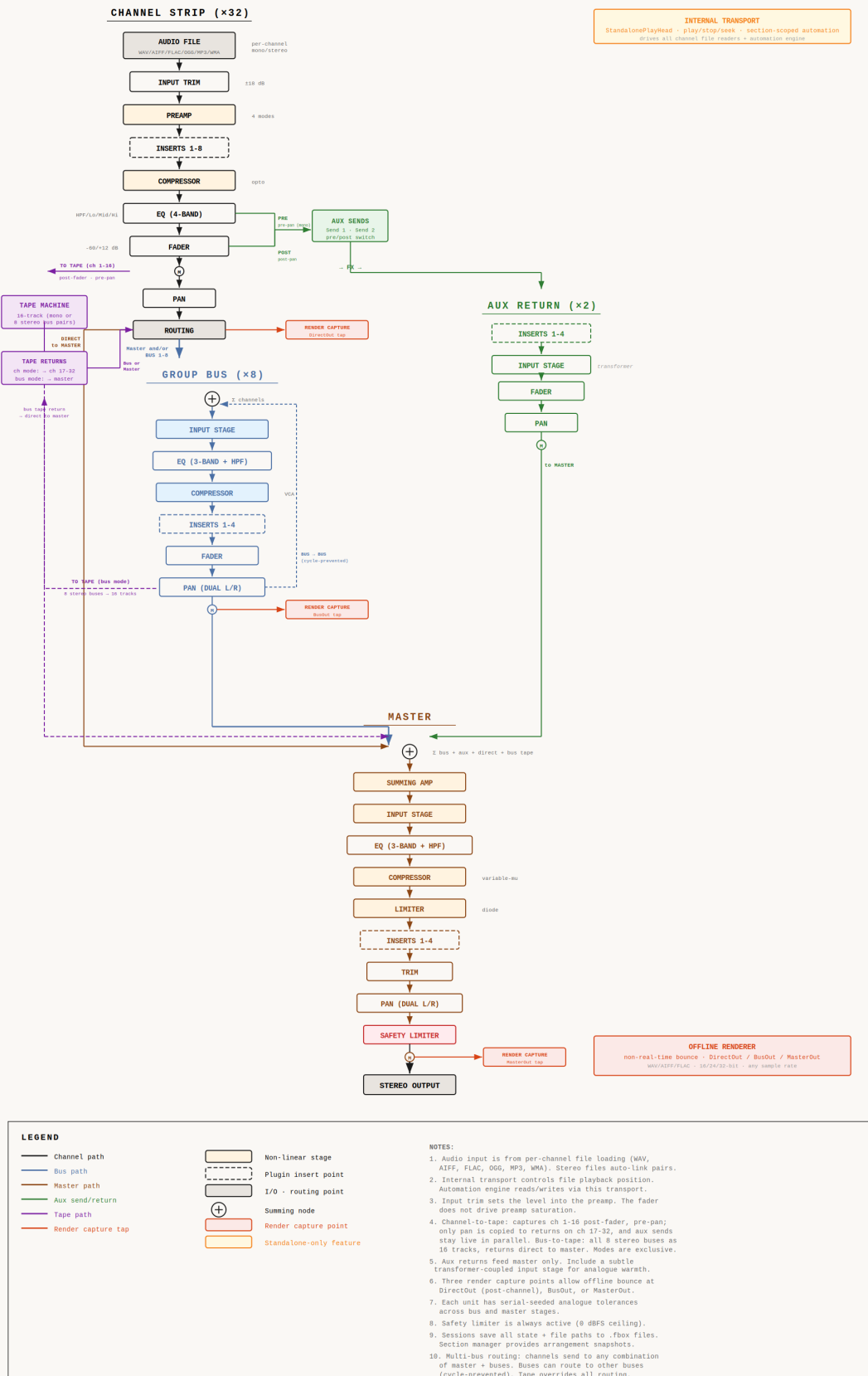
Product Manual

Build 1.0

by flakebelly
fboxaudio.com

fBox CONSOLE STANDALONE - SIGNAL FLOW

32 CHANNELS • 8 GROUP BUSES • 2 AUX SENDS/RETURNS • 16-TRACK TAPE • STEREO MASTER



What fBox Is

fBox is a complete analogue mixing ecosystem - individual plugins, a universal rack processor, and a full 32-channel mixing console with built-in tape machine - built around a single idea: every piece of hardware that ever left a factory floor had its own personality, and your plugins should too.

Most plugin companies model hardware. They pick one unit, capture it as accurately as they can, and sell that one model to everyone. Your copy is identical to mine. The compressor on your kick is the same compressor on your vocal. It has always worked this way - and it is the reason in-the-box mixes sound in the box.

fBox is built on the opposite principle. Every plugin instance carries a unique serial-seeded analogue character, generated the moment it is created. Load three fComps in one session and you have three different compressors, each with its own voice. That serial is fixed to the instance forever - it never resets, and it travels with the session.

The choice is yours. Open a fresh instance whenever you like and it is a different unit with its own voice - or find one you love and keep it. Save it, recall it, and mix through it for as long as you like: the same character, every time. Learn how it sounds and it becomes an instrument you know. And a unit you keep is yours alone - its serial exists on no one else's machine, the same way a piece of hardware you buy is one specific unit and no other. Bought once, owned permanently, no subscription.

This is not emulation. This is character at the instance level.

What fBox Console Standalone Is

fBox Console Standalone is a virtual analogue mixing console that runs as its own Windows application. It is the standalone counterpart to fBox Console, the plugin version that runs inside a DAW: the same desk, but running on its own with no host required. fBox Console Standalone gives you up to 32 mono channel strips, 8 group buses, 2 stereo aux sends with returns, a 16-track tape machine, and a stereo master section, all in a single window. You load audio files straight onto the channel strips, mix on the desk, and render the result to a stereo file with the built-in offline renderer. No DAW is required.

It is not a collection of unrelated processors. It is one integrated desk. Level, routing and gain staging are part of the sound, not an afterthought. The console's summing stages respond to how much signal is passing through them, so the mix gains density and glue as it fills out.

Every channel strip has a switchable preamp with four modes, 8 insert slots for fBox plugins, a single-knob channel compressor, a 4-band EQ, 2 aux sends, pan, mute, solo and a fader. The 8 group buses each have a 3-band EQ with high-pass filter, a VCA bus compressor, 4 insert slots and dual pan. The master section adds a summing amplifier, a vari-mu compressor, a diode limiter, and its own 3-band EQ with HPF and 4 insert slots. Almost everything on the desk can be recorded and played back by the console's own multi-pass automation.

Built into the console is a Tape Machine - a 16-track tape simulation on a classic multitrack recorder, sitting directly in the signal path, adding the character of analogue tape to your mix.

Every console is minted with a unique serial number that defines its hardware character - the small manufacturing differences that make this desk sound like this desk and no other. No two consoles sound identical, and the choice of how you work with that is yours. Roll a fresh console whenever you like and every session is a different desk with a different voice - or find one you love, save it and pin it, and it becomes the console you open every time. Kept that way its serial belongs to your machine alone, exactly as a real desk you bought would, and no one else has it. The same serial always produces the same console, and it is saved with your session.

The standalone application manages its own audio device, transport and rendering. You load files onto channels, mix, and bounce the result. The output is always stereo.

Getting Started

Sound working in five minutes

1. Launch fBox Console.
2. Click **Audio Set** in the header. Set **Device type** to **ASIO**, pick your audio interface, and choose a sample rate (44100 or 48000 Hz) and a buffer size (512 samples). If ASIO is not in the list, see Audio Settings below - getting this right is what stops crackles and dropouts.
3. Use the dropdown at the top of Channel 1 and choose **Load...** to browse for an audio file (WAV, AIFF, FLAC, OGG, MP3 or WMA). It loads onto that channel. Or drag a file straight onto the dropdown.
4. Repeat for as many channels as you need. Each channel holds one mono file.
5. Press **Play** on the transport bar. Spacebar plays and pauses from anywhere in the window.
6. Click a channel strip to open it larger in the Focus Window on the left.
7. Use **TRIM** near the top of the strip to set working level. Aim for peaks around -12 to -6 dBFS on the channel meter.
8. Use the fader for balance.

If you hear nothing, check that the audio device is selected, the transport is running, and the channel has a file loaded (its name shows on the strip).

Loading audio files

Each channel strip has a dropdown at the very top. Choose **Load...** to open a file browser and pick an audio file. The console reads it and the filename appears on the strip. You can also drag a file from your file manager onto that dropdown - the dropdown is the drop target, not the rest of the strip. All files should be at the same sample rate as your audio device for correct pitch and speed.

The console accepts mono and stereo files. Drop a stereo file on any channel and it is split across a pair for you: the left channel goes to the lower channel of the pair, the right to the channel just above it, the pair is stereo-linked, and pan is set hard left and hard right. Loading a stereo file on Ch 1 puts L on Ch 1 and R on Ch 2. Mono files load onto a single channel as normal.

Tip: Export stems from your DAW as WAV files at your project sample rate, then load each onto a console channel. Stereo stems split across adjacent pairs automatically. This gives you a complete mix environment outside the DAW.

Setting your channel count

Default: 8 channels. Click the **Ch** button in the header to cycle between 8, 16, 24 and 32 channels.

Important: Channels beyond the enabled count are not processed and not heard. If the count is set to 8, only channels 1-8 are shown and can hold files. The console expands automatically when you load a stereo file that needs the next channel, or when you enable tape. To load onto higher channels yourself, raise the count first with the Ch button so those strips are shown.

Audio Settings

Click **Audio Set** to open the audio panel. Four things to set, and it is worth getting them right once - almost every report of crackles, dropouts or "it sounds bad" comes back to this panel.

Device type - choose ASIO

This is how the console talks to your soundcard, and it matters more than anything else here.

ASIO is the one you want. It talks to your audio interface directly, which means lower delay, steadier timing, and a buffer size that means what it says. Every proper audio interface comes with an ASIO driver; if it is not in the list, download it from the manufacturer's website and restart the console.

The **Windows Audio** options go through Windows itself, which adds its own buffering on top of yours and shares your soundcard with everything else on the machine. They will work, but you will get more delay, less predictable performance, and the buffer setting will no longer reflect what the hardware is really doing. Use them only if ASIO is genuinely unavailable. **DirectSound** is older still and there is no reason to choose it.

If you have no audio interface at all and are using your computer's built-in sound, ASIO4ALL is a free stand-in that appears in the ASIO list once installed. It is better than Windows Audio, but a real interface is better than both.

Device - choose your interface

With ASIO selected, this lists the ASIO drivers on your machine. Pick your interface by name. You may also see entries belonging to software you have installed rather than hardware - anything that is not your interface is not what you want here.

Sample rate

How finely the audio is measured. 44100 Hz and 48000 Hz are the standard choices and either is fine; 44100 for music, 48000 if you are working to video. The console also offers 88200 and 96000, which cost roughly twice the processing for a difference most people cannot hear. Start at 44100 or 48000. Whatever you choose, all the console's processing runs at that rate, and it is shown in the header.

Buffer size

How much audio the console works on at a time. Smaller means less delay but more strain on your computer; larger means more delay but an easier ride.

The console plays back and mixes files rather than recording them, so you never need the very small buffers used for tracking. Set it comfortably high and leave it there: 512 samples is a good place to start, and 1024 is comfortable for large sessions and for rendering.

If you still hear clicks, crackles or dropouts at 1024, the session is asking more of the computer than it can give. There is no console-wide quality setting - quality is set per plugin and on the Tape Machine - so drop those to Eco, turn the Tape Machine off when you are not using it, or run fewer channels.

Tip: *New to this? Set Device type to ASIO, pick your interface, 48000 Hz, 512 samples, and get on with mixing. If it plays cleanly, it is set up correctly - there is nothing to be gained by fiddling further. Come back to the buffer size only if you hear crackles.*

New to mixing?

The short version: **TRIM** sets how hard the signal drives the channel, the fader sets how loud it sits in the mix, and you want level roughly right with TRIM before you start balancing with faders. Everything else is flavour you can add slowly once the balance is there. And keep every move small - the console's sound comes from many stages interacting gently, not from one big move.

Activation and Licence

The first time you launch, fBox Console asks you to activate. There are two ways in.

Free trial

Enter your email address and click **Send Verification Code**. A 6-digit code arrives in your inbox. Type it in and click **Activate Trial**. You get 14 days, with no payment details required.

Licence key

Paste your key (it looks like FBOX-XXXX-XXXX-XXXX) and click **Activate Licence Key**. Activation needs an internet connection. Once activated, the licence is stored on this machine and the console opens straight to the desk from then on.

Your Desk

Every console is minted with its own serial number and its own hardware character. Normally each new console you open is a different desk. The **Desk** button in the header lets you keep the one you fall for.

Menu item	What it does
New Unit	Roll a brand new desk - new serial, new character, new tape machine. The mix stays where it is; only the personality changes.
Save Unit...	Save the current desk so you can come back to it.
Load saved unit	Recall a desk you saved before. The submenu lists them all, and you can browse for a unit file from anywhere.
Pin this unit as my desk	Make this desk your default. It is adopted every launch and every new session - this is how you keep one desk for good.
Use a new unit each launch	Go back to a fresh desk each time. Your pinned desk is not deleted, it is just set aside.
Recall my pinned unit	Bring your pinned desk back - handy if you rolled a new one by mistake, or after unpinning.

Saved desks live in Documents/fBox Units/Console - the same place the fBox Console plugin uses. A desk you save in the standalone turns up in the plugin's Desk menu, and the other way round. Find a console you love while mixing in your DAW and you can carry it into the standalone, and back. A saved desk carries both serials, the console's and the tape machine's, so recalling one brings the whole desk back, tape and all.

Tip: *You cannot dial up a console's character to order, any more than you can with a real one - but you can learn it, and you can keep it. Mix on one until you know how it responds, then pin it. Your ear adapts to a desk you stay with.*

Interface Overview

Channel strips run across the centre of the window and scroll left and right. The master section is fixed at the far right. The transport bar and the arrangement strip sit along the bottom.

Focus Window (left)

Shows the selected channel, bus, aux return or master at a comfortable size. Click any strip to select it - the selected one gets a full outline. Scroll it with the mouse wheel to reach everything down to the fader. Automation records here just as it does on the main strips. The Focus Window is hidden while the tape panel is open, to give the tape machine room on smaller screens; close the tape panel and it returns.

Channel strips (centre)

Narrow, so you can see many at once. The mouse wheel scrolls the row sideways. Shift and the mouse wheel scrolls a strip vertically, from the dropdown at the top down to the fader.

Slide-out panels (right)

BUS, AUX and TAPE share the same space, so only one of the three is open at a time. Trying to open a second one does not close the first - the tab you clicked flashes red to tell you it is blocked. AUTO is separate: it can sit open alongside BUS or AUX so you can watch your passes while you work, but not alongside TAPE. Opening TAPE also hides the Focus Window, to give the tape machine room.

The AUTO tab changes colour with status: red when a pass is recording, green when a pass is playing back, dark when all passes are off. The TAPE tab goes brown when tape is running.

Master section (far right)

Always visible. The stereo master bus with its own EQ, compressor, limiter, inserts, fader and meters.

Header bar

Contains: Audio Set, Scan Plugins, New, Open, Save, Save As, Desk, Help, Reset Engine and Master Out buttons, the channel-count selector, and the watermark at bottom right, which shows flakebelly, the website, and this console's serial number beside its tape machine's serial number.

A word on that name. flakebelly is the one-person outfit that builds fBox - and it is also the name I make music under, which is, entirely objectively, best enjoyed at all hours and all volumes. Consider this your official notice to go and put it on. The mixing can wait ninety seconds. You will not find it on any streaming service - only on Bandcamp

and Subvert. Some things are worth going to fetch.

Diagnostics are written automatically as the console runs, so you need do nothing to capture them. They land in Documents/fBox/Diagnostics, with the previous run kept alongside the current one (see [What fBox Writes To Your Computer](#)).

Arrangement Strip and Sections

Along the bottom of the window sits the arrangement strip. It shows a composite waveform of everything loaded, section dividers, and the section controls. Drag its top edge upward to make it taller; the controls grow with it.

This is how you mix a song in sections. Instead of one static mix, you give the verse, the chorus and the solo each their own settings, and the console reconfigures itself as the playhead moves from one to the next - a bigger chorus, a stripped-back verse, a louder solo, all in one pass.

The waveform

Built automatically as you load and unload files. Click it to move the playhead. Zoom with Shift and the mouse wheel, or the plus and minus buttons. Scroll with the mouse wheel, or drag to pan when zoomed in.

Dividers and sections

Click +Div to drop a divider at the playhead. Or drag across the waveform to mark out a region in one go - a divider goes in at each end. Drag a divider to move it. Alt-click or right-click a divider to remove it, and the regions either side merge back into one. A region only takes a name once you have stored a snapshot into it; until then it just runs your current mix. Double-click a stored section to rename it (Intro, Verse 1, Chorus).

Snapshots - a different mix per section

A snapshot stores the whole desk for one section. With a section selected, click Store, and everything - every channel, all your routing and sends, every insert plugin, the buses, the master, and the tape setup - is captured into that section. Move the playhead into it during playback and the console becomes that mix; move out and it becomes whatever the next section says, or your normal mix if there is no section there.

That is the workflow: get the verse sounding right and Store it on the verse section. Change things for the chorus - push the drums, open the vocal, drop it onto tape - and Store that on the chorus section. Now the desk plays the song, switching mixes at each boundary.

Storing a section around an automated control

If you are changing a control that a pass drives, switch that pass to Off before you start. Automation takes a control back the moment you let go of it, so an automated fader will return before you get as far as pressing Store. An Off pass keeps everything it recorded and simply stops playing it back, which puts those controls in your hands. Play the section, listen, adjust, Store, and switch the pass back to Auto On when you are done. Off applies to the whole song rather than just the part you are working in, and if two passes cover the same control you will need both of them off. If a control keeps springing back while you dial a section in, that is a pass telling you it owns it.

Automation owns a control everywhere, not just during its moves. Record a move on a fader and from then on that fader follows automation across the whole song - inside a move it plays the move, outside one it sits where the move left it. A section's stored value for that fader is ignored while the pass is on. Sections place the controls automation is not driving, and only those.

That is what makes mixing possible: a fader you rode down stays down, instead of being pulled back to whatever it happened to be when you stored the section.

If you want a section to set a control that a pass drives, switch that pass Off first. An Off pass keeps everything it recorded and simply stops playing it back, which hands the control to the section. Store, then switch the pass back to Auto On.

A snapshot captures every channel control - trim, preamp, compressor, all four EQ bands, fader, pan and mute; all routing, the aux send on/off switches and the send-to-tape level; every aux send level; every insert on channels, buses, aux returns and the master, including each plugin's settings and its bypass; the stereo links; and the tape machine (on or off, the routing mode, the speed, and the per-track and output drives for both modes). Because tape is included, one section can run clean while the next drops the whole mix onto tape at its own speed and drive.

Load a plugin into an insert slot and every section learns about it straight away, including sections you stored long before that plugin existed. They all start out holding its current settings, so nothing changes until you deliberately make a section different and store it. Without that, a section stored before the plugin existed would have no opinion about it - and a plugin changed in one section would appear to follow you into the next.

Quality is never stored in a section. Not the tape machine's, and not any insert plugin's. Quality is a whole-session setting - you mix at Eco or Studio and switch everything to Master to render, and that switch has to hold across the entire song. A section that remembered a quality would undo it at the next boundary, or force you to set it again in every section. So changing quality anywhere changes it everywhere, and it is never something a section is waiting for you to store.

Solo is not stored. It is a monitoring control - a way of listening to something while you work - so it stays exactly where you left it as you move between sections. Mute is a mix decision and is stored like anything else.

Control	What it does
Store	Capture the console into the selected section. The button glows gold whenever the selected section needs storing - either it has nothing stored yet, or you have changed something since.
Clear	Remove the snapshot. That region goes back to running your normal mix and loses its name.
Lock	Protect a stored section. Click again to unlock. Only a section that already has a snapshot can be locked.

While a section is locked, its dividers cannot be dragged or removed, its snapshot cannot be cleared, and you cannot drop a divider or drag out a new region inside it - so a stray drag can no longer split a finished section. Locking does not freeze the sound:

you can still select a locked section, change it, and deliberately re-store it. It guards against accidents, not against you.

Status bars

Each section shows a coloured status bar. Red means nothing is stored and the region runs your normal mix. White means a snapshot is stored and the console matches it. Amber means a snapshot is stored but you have changed something since - re-store to keep the change. Green means locked. A locked section you have modified shows amber, not green - the warning wins, so a change can never hide behind a lock, and it returns to green once you re-store.

A section only turns amber when you have actually changed something. Touching a control and letting go without moving it leaves it white, as does clicking a strip to select it, or the playhead simply passing through on its way somewhere else. Amber always means a real difference between the desk now and the desk you stored.

Recording automation does not turn a section amber either. A control a pass drives is not one the section speaks for, so moving it while recording changes nothing the section stores. If you deliberately want a section to set such a control, switch its pass Off first - and then moving it does turn the section amber, as it should, because with the pass off the section is what will speak for it.

This includes the plugins on your inserts. Change a knob inside an fEq or an fComp sitting on a channel and the section goes amber exactly as it would for a fader, because that plugin's settings are part of what the section stores. Changing quality does not, since quality is not stored.

Crossfading, loop and section record

During playback the console eases between sections at each boundary rather than jumping. The **XFade** dropdown sets how long that takes: Instant, 10ms, 50ms (default), 100ms or 250ms.

Loop cycles through three states each time you click it, so you can choose what repeats:

Button	Colour	What loops
Loop	grey	Nothing - playback runs to the end and stops
Loop All	blue	The whole song, whatever section is selected
Loop Sec	green	The selected section

Loop All is the default, so a session you have just opened replays from the top rather than stopping. Switch to Loop Sec when you want to work on one part over and over - select the section and it repeats until you click again. Selecting a different section while it is running moves the loop with you. In Loop Sec with nothing selected, the whole song loops.

Sec Rec confines automation recording to the selected section, so you can build a detailed pass on one part of the song without touching the rest.

Recording round a loop, each lap replaces what it passes over. Stop half way round the last lap and the rest of the loop is still the lap before - every lap you record is kept, and at each point in the song you hear whichever lap wrote it last.

Two ways to change the mix as the song plays

The console gives you two systems, and most of the confusion around sections comes from reaching for the wrong one. They work together, they work alone, and which you use is entirely your choice - but they are good at different things.

Automation is for moves. A fader ride into the chorus. Opening a reverb send on the last word of a line. Sweeping a filter. Pulling the guitars down under the vocal for eight bars. Anything where one control changes, and especially anything that changes gradually. Automation touches only the control you moved and leaves the entire rest of the desk alone.

Sections are for wholesale changes. The chorus wants a different mix - not one fader, but the balance, the drums, the reverb, the tape, all of it, mixed as though it were its own piece of music. That is what a snapshot is for: it stores the whole desk and recalls the whole desk.

The question to ask: **would I mix this part of the song differently, or do I just want something to move?** Differently is a section. Move is a pass.

Reach for automation first

For most of what you will want to do, automation is the better tool, and by some margin.

It is **precise** - it changes one control and nothing else, so there is nothing to keep track of. It **layers** - record fader balance in one pass, EQ moves in another, panning in a third, and each stays separate and can be switched off or cleared on its own. It is **undoable** - Undo Rec puts the last thing you recorded straight back, and a locked pass cannot be wiped without the console asking you first. And it costs you nothing later: recording a pass does not commit any part of the song to a fixed snapshot, so a change you make afterwards still reaches everywhere.

A section is a bigger commitment. It stores everything, which is exactly what you want when a whole part of the song should sound different - and more than you want when one knob should move.

What this means in practice

Most songs need few sections and several passes. A typical mix might be:

- The whole song mixed once, properly. That is your base mix.
- One section on the chorus, because it genuinely wants a different balance.
- Three or four passes on top: fader rides, a send opening up here and there, a filter sweep into the drop.

That is a fully automated mix, and there is only one snapshot in it.

If you find yourself storing a section because you want one fader to be louder in the middle eight, that is the moment to stop and record a pass instead. It is quicker, it is easier to change your mind about, and it leaves the rest of the song free.

The order that works

1. Mix the whole song first. Gain staging, balance, channel EQ, your bus processing, the master chain, the tape machine. Everything that should be true from the first bar to the last. Do not make any sections yet.

This is your base mix, and while nothing is stored it plays everywhere. Anything you change while sitting in a region with no snapshot updates it - so at this stage, a change to the master compressor reaches the entire song, once.

2. Then mark out only the parts that should be mixed differently, and store those. A region with nothing stored keeps running your base mix, so you do not need a section for every part of the song - only for the parts that genuinely differ.

3. Then record your passes for everything that should move.

4. Then lock the sections you are happy with. It costs nothing and stops a stray drag reaching a finished part.

If you change your mind about the whole song later

You get to the end and decide the whole mix wants a touch more master compression. Two ways to handle it, and the first is usually better.

Ask whether it belongs in the base mix. If most of your song is unstored - which it will be, if you have kept sections for the parts that truly differ - then simply make the change. It reaches everywhere except your stored sections, and those are the parts you deliberately mixed differently anyway. Often they were already right.

If it does need to reach the stored sections too, select each one, make the change, Store, and move to the next. Do them one at a time and deliberately: do not make the change and then start clicking around, because arriving in a section recalls it and takes your change straight back off the desk.

If that feels like a lot of work, it is usually a sign there are more stored sections than the song needed - which is the argument for using passes for moves, and keeping sections for the parts that are genuinely their own mix.

Tip: *Mix the song first, then arrange the mix. Sections are for the parts that should sound different; passes are for the parts that should move; and the base mix underneath is for everything that should simply sound good all the way through.*

Tip: *Lock a section the moment you are happy with it. It is easy to nudge a divider or drag across the waveform while working on the part next door, and a lock means those slips cannot reach a mix you have finished.*

Controls: the Channel Strip

Top to bottom, a channel strip runs: the dropdown, the name, TRIM, the inserts, the routing button, the compressor, the EQ, the two aux sends, the pan knob, the mute/solo/arm row, the preamp, and the fader, with a colour strip along the bottom.

Dropdown and name

The dropdown at the very top loads audio and shows the filename once loaded. Its four options are **Load...** (browse for a file), **Unload** (remove the file, clear the name, break any stereo link), **Direct Out** (render this channel's processed audio to a file, up to but not including bus routing), and **Reset Channel** (return every control to its default and remove every insert plugin from the strip; the audio file and any stereo link stay). Note that Reset Channel clears the inserts completely - it does not reset them in place.

Directly below the dropdown is the channel name, filled in from the file automatically; double-click it to type your own. Hover the dropdown to see a long filename in full.

TRIM (Input Trim)

Range: -18.0 dB to +18.0 dB. **Default:** 0.0 dB. Double-click to reset to 0.0 dB.

TRIM is the first gain stage - it hits the audio before anything else on the strip. Moving the fader does not change the level feeding the preamp, the inserts or the compressor; only TRIM does. A small meter beside the knob lets you watch level up here without scrolling down to the fader.

Tip: Think of TRIM as the input volume for the channel. Set it so peaks land around -12 to -6 dBFS, then leave it - that is your gain staging done. From there, TRIM is also your drive control: the preamp and the other stages colour more the harder the signal hits them, so nudging TRIM up is how you push the channel into more character. Set level with TRIM, balance with the fader.

Inserts (8 slots)

8 slots for hosting fBox plugins (fEq, fComp, fDrone and the rest), in series, right after TRIM. The strip shows 2 by default; click the plus button to cycle 2, 4 and 8 visible slots. All 8 are always live regardless of how many you can see. Click an empty slot to load a plugin; click a loaded slot to open its editor. A plugin that syncs to tempo - a synced delay or LFO, the tempo-synced modes in fDelay, fMod and fVerb - locks to the session tempo on the transport bar (see Sessions, Rendering and Transport).

Right-click a loaded slot for its menu:

Menu item	What it does
Open Editor	Opens the plugin's own window.

Menu item	What it does
Bypass	Bypasses the plugin, and reads Un-bypass when it is already bypassed. A bypassed slot changes colour so you can see it at a glance.
Replace...	Swap a different plugin into the slot.
Clear	Remove the plugin from the slot.
Clear Automation	Wipe the recorded moves for that slot by hand. This item only appears when the slot has moves to clear.

If a plugin starts producing bad audio, the console faults it to protect the mix: the slot turns bright red with yellow text (normal bypass is dim red with white text) and the plugin is bypassed for you. Reset Engine in the header clears it. If the slot menu is empty, run Scan Plugins first - the console lists fBox plugins only. fTape is not available as an insert; for tape, use the console's Tape Machine.

Moving and copying inserts

Drag any loaded insert to any other slot - same channel, another channel, channel to bus, bus to master, anywhere. A plain drag moves it (and swaps if the target is full). Hold Alt and drag to copy it, settings and all.

Important: Automation lives with the slot, not the plugin. Recorded moves stay in the slot you made them in, and they never travel with a plugin. Because of that, taking a plugin out of a slot deletes that slot's moves. Clear, Replace, and dragging a plugin away all count, and the moves cannot be recovered afterwards - so if you are only pulling a plugin out to compare something, remember the slot's automation goes with it, and putting the same plugin back will not bring it back. Dragging clears both ends: the slot you dragged out of and the slot you dropped onto. An Alt-copy is gentler - it only affects where you drop, because the plugin you copied from is still sitting where it was, still doing its job, so its moves are left alone. Build your chain first, then automate it.

Section settings behave the other way round. Move a plugin to a different slot and its per-section settings travel with it, so a plugin set up differently in each of your sections keeps all of that when you rearrange your chain. Copy it and the copy arrives set up the same way. Only a genuine replacement - choosing a different plugin from the menu onto a slot that already holds one - starts clean.

So: your section settings follow the plugin, your automation stays with the slot. Rearranging a chain keeps the settings and loses the moves. Get the chain in the order you want, then record your passes.

The slot marker

A slot with recorded moves on it shows a coloured stripe down its left edge, so you can see at a glance what is automated before you touch anything.

Stripe	What it means
Gold	The slot has moves recorded on it. Replacing or dragging that plugin will delete them.
Blue	Slightly wider. Some of those moves belong to a pass you have locked. The console will ask before it destroys anything in a locked pass, so blue is your warning that a question is coming rather than a silent deletion.

The marker updates as you work, and it shows on the main strips and in the Focus Window.

Stereo linking

For a stereo source, put L on the lower channel of a pair and R on the one above it (1+2, 3+4). **LINK INSERTS** makes the pair share one insert chain, processed in stereo through the lower channel's inserts; the upper channel shows "INSERTS LINKED TO Ch N". Each channel keeps its own fader, pan, mute and solo - this links the inserts only, it does not touch pan. **STEREO LINK** is a full link: the lower channel drives every control on the upper one, and the pair is set hard left and hard right so the stereo image is held.

Routing

Click the **Routing** button to open the routing popup. A channel can feed any combination of destinations at once: **Master** (default on), **Bus 1** to **Bus 8** (click each to toggle; named buses show their names), and **To Tape** (channels 1-16 only). Ticked items are live. You can send a channel to Master and Bus 2 and Bus 4 at the same time - that is how you set up parallel processing, like a clean bus and a crushed bus side by side. The button label sums it up: "M+2+4", or "Tape" when tape is running. Routing is automatable, except to-tape routing.

Important: When tape is active it overrides all other routing. Master and bus entries grey out. Your routing is remembered and snaps back the moment tape is turned off.

Tip: Routing to a bus lets you group similar channels - all drums to Bus 1, all vocals to Bus 2 - and ride the whole group with one fader.

Compressor

Range: Off to 100%. **Default:** Off.

A single-knob channel compressor built on an optical gain element, after the inserts and before the EQ. It grabs quickly then settles, and lets go quickly then eases off - so it follows the material rather than working to fixed numbers: snappy on drums, smooth on vocals. Low settings level gently and almost invisibly; the middle gives you classic vocal compression; higher up it takes a firm grip on drums and bass, and at the very top it edges into limiting. There is harmonic colour in it too, and every channel's

compressor has a slightly different feel. Automatic make-up gain rides along, handing back about half of what it takes, so you keep your level while still feeling it work.

Tip: Compression narrows the gap between the loud and quiet moments, so a part sits more evenly in the mix. Start around 25-40% for gentle control and listen for the part steadying. Back it off if it starts to sound squashed. For detailed, knob-by-knob compression, load fComp on an insert instead.

EQ (4-band channel EQ)

A 4-band EQ after the compressor, laid out vertically: HIGH SHELF at the top, then MID, then LOW, with HIGH PASS at the bottom.

Band	Frequencies (stepped)	Use
High Shelf	8k, 10k, 12k, 16k Hz	Air and sparkle, or taming harshness. A shelf lifts or dips everything above the frequency.
Mid (bell)	700, 1.5k, 3k, 6k, 8k Hz	Presence, vocal clarity, honk removal. A bell boosts or cuts a band centred on the frequency.
Low (bell)	35, 60, 100, 220, 350 Hz	Warmth and weight, or clearing mud.
High Pass	Off, 30, 50, 80, 120 Hz	Removes rumble and low-end clutter below the frequency. Off means no filtering.

Gain on the three bell/shelf bands is -12.0 to +12.0 dB, default 0.0 dB. The EQ colours as you push it: the harder you boost or cut, the more harmonic character comes with it. The low bands stay smooth and round; the highs bite a little harder. This is part of the console's voice, and it is exactly the kind of thing that adds up across a mix, so keep the moves modest if you are also colouring elsewhere.

Tip: Try the HIGH PASS at 80 Hz on everything except bass and kick - it clears low-end mud across the whole mix at a stroke. Then keep boosts and cuts small, 2-4 dB. A shelf is a broad tilt; a bell is a targeted lift or scoop. For surgical work, load fEq on an insert.

Aux 1 and Aux 2

Two sends per channel, for feeding shared effects - usually a reverb and a delay. The knob sets send level (-60 dB to 0 dB). The small numbered button (1 or 2) switches the send on; yellow means on. The Pr/Po button sets where the send taps the signal: Po (grey) is post-fader, the default, and Pr (green) is pre-fader.

What pre and post mean

Post-fader means the send is taken after the channel fader, so when you pull the fader down, the amount going to the effect drops with it - the effect tracks the fader. This is what you want almost always for reverb and delay: fade a part out and its reverb fades with it. Post also follows the pan, so a hard-panned part sits in the same place in the reverb.

Pre-fader means the send is taken before the fader, so the amount going to the effect stays the same no matter where the fader is - you can even pull the channel fader all the way down and still hear it in the effect. Two creative uses: pull the dry fader right down on a pre-fader send and you get pure effect with no dry signal (an all-reverb ghost of a vocal, say, or a delay throw with the source gone); and pre-fader sends are how you build a separate headphone or cue balance that does not move when you ride the mix.

The send knob and the return fader

The aux knob on the channel sets how much of that channel goes into the effect; it only ever attenuates, topping out at 0 dB. How loud the effect itself sits in the mix is set elsewhere, on the aux return strip, which has its own fader (open the AUX panel to find it). So the send knobs decide how much of each channel goes into the reverb, and the return fader decides how loud that reverb is overall. Sending different channels at different amounts is how you build front-to-back depth: more send means more reverb on that part, which pushes it back and away; less send keeps a part dry, present and up front.

Tip: *One reverb shared across many channels via a send sounds more like one room than a separate reverb on every channel, and costs far less CPU. Put the reverb on the Aux 1 return, send a little from each channel, and set the overall wetness with the return fader. One thing to remember: turn the Mix knob on your aux fVerb or fDelay right up to 100% wet, so the return carries effect only. You then dial the amount in with each channel's send knob and the overall level with the aux return fader - not with the plugin's own mix.*

Pan

Range: -100 (hard left) to 100 (hard right). **Default:** 0. Display: "-50" is left 50, "50-" is right 50, "0" is centre. Equal-power, so the level does not dip when panned to the middle. Applied after the fader, before routing.

Mute, Solo and Arm

Three buttons in a row. **MUTE** silences the channel and turns red. **SOLO** isolates it and turns amber - solo as many channels as you like; solo a bus and everything feeding it is heard. **A** (the arm button) arms the channel for automation recording and turns red; only armed strips record. Solo is monitoring only and is never recorded by automation; mute can be. Hold Alt and click any Mute to mute every strip at once; Alt-click any Solo to solo everything; Alt-click any A to arm everything. This works from any strip and from the Focus Window.

Preamp

Near the bottom of the strip, just above the fader, sits the preamp - the channel's main colour stage. The modes are **Clean** (straight through, no colour), **Warm** (gentle transformer-style saturation with a lift low down, thickens and rounds), **Hot** (aggressive transistor-style clipping with mid presence and bite, grit and edge), and **Valve** (tube-style saturation with a soft squeeze, musical warmth). There is no drive knob - how hard the preamp works is set by how hard the signal hits it, which is TRIM's job: quiet signals stay clean, signals up near 0 dBFS start to colour, and pushing harder leans into saturation.

Tip: Set Warm and bring TRIM up until you hear the tone thicken. Hot and Valve are stronger and suit drums, guitars and sound design. Remember it stacks with everything downstream - a little here goes a long way once the buses and tape have had their turn.

Fader

Range: -60.0 dB to +12.0 dB. **Default:** 0.0 dB. Your balance control. Any colour from pushing it appears later, at the bus and master summing, not in the fader itself. Double-click anywhere on the fader to return it to unity (0 dB). Double-click the dB readout below the fader to type a value instead; type "-inf" for minimum.

Meter and colour strip

The meter is mono, pre-pan and post-fader. Click any dB readout to switch every meter on the desk between Peak and RMS. Along the bottom of the strip is the colour strip: click it to pick from 12 colours and group your drums, vocals, synths and returns. It shows in the strip and the Focus Window.

Buses

8 group buses. Click the **BUS** tab to slide the panel in and out. Each bus sums every channel routed to it. Top to bottom a bus strip runs: name, dual pan, the mute/solo/arm row, the routing button, the EQ, the compressor, the inserts, and the fader, with a colour strip along the bottom. EQ and compressor bypass buttons read ON or OFF, and both default to OFF so an unused bus starts clean.

Bus routing

Click the Routing button. A bus can send to **Master** (default on), **other buses** (any of the other 7 - build hierarchies, e.g. drum, bass and guitar buses all feeding one instrument bus before the master), **Direct Out** (render this bus's stereo output to a file), and **To Tape** (route all 8 stereo buses through the tape machine, greyed out if channel-to-tape is already on). Feedback loops are prevented: if Bus 1 feeds Bus 2, Bus 2 cannot feed back to Bus 1. Buses process in dependency order, so bus-to-bus routing adds no latency, and bus routing is automatable.

Bus compressor

A VCA bus compressor - the classic glue for holding a submix together.

Control	Range
Threshold	-30 to 0 dB (0 = off)
Ratio	2:1, 4:1, 10:1, 20:1
Attack	0.1, 0.3, 1, 3, 10, 30 ms
Release	0.1, 0.3, 0.6, 1.2 seconds, or Auto
Makeup	0 to 20 dB
Mix	0 to 100% (parallel compression)
SC HPF	Off, 60, 90, 150, 300 Hz

Auto release is the default and the place to start; it reads the material, snappy on drums and smooth on pads. It is not a clean compressor - it warms as it works, more so the harder it pulls, and each bus has its own feel. The horizontal meter shows the gain reduction.

Tip: Auto release, 4:1, then pull the threshold down until the meter shows 3-6 dB coming off, and bring the level back with makeup. If the bass makes it pump, raise the SC HPF so the low end stops triggering it. Mix lets you blend compressed and dry for parallel squash without losing punch.

Bus EQ

HPF (Off, 30, 50, 80, 120 Hz stepped), Low Shelf (40 Hz to 500 Hz), Mid Bell (200 Hz to 8 kHz) and High Shelf (1.5 kHz to 20 kHz), all plus or minus 12 dB. Every band has a frequency knob and a gain knob, all sweeping freely; at zero gain it is out of the way. A whisper of saturation shows only when you boost hard.

Tip: *Broad strokes on a group, not surgery - tilt the drums brighter, warm the vocal bus, take mud out of the guitars. Small moves matter here because they land on every signal in the bus at once.*

Bus inserts, input stage and dual pan

Each bus has 4 insert slots, after the compressor and before the fader (2 shown by default, expandable to 4), with the same drag, copy and slot-bound automation behaviour as channel inserts. Each bus also has a non-linear input stage that responds to how much signal hits it: quiet buses stay clean; as the level rises you get harmonic density, transients rounding, and cohesion; drive one hard and it saturates and gently compresses. Push the faders and the mix thickens on its own - which is where a lot of the console sound comes from. The dual pan (L PAN and R PAN, -100 to +100 each) places the left and right sides of the bus independently in the stereo field, default full width.

Aux Returns

Aux 1 and Aux 2 come back on their own stereo return strips. Click the **AUX** tab to open the panel. Each return has 4 insert slots, an input stage, pan, a fader (-60 to +12 dB), mute, an arm button and a meter. Returns feed the master.

The input stage gives the returns a little warmth as they come back, so effects sit in the track rather than on top of it. It is cleaner than the channel preamp - just enough iron to seat a reverb without colouring it. Returns have their own arm button, so their fader, pan and mute can be recorded, and Arm All and Disarm All include them.

Tip: *The return fader is your master wetness control. If a reverb sounds right per-channel but is too loud overall, pull the return fader down instead of chasing every send knob.*

Tip: *Using fDrone or another generator on an aux return. An aux return does not have to be an effect. Put fDrone - or fMod in ring mod - on an aux return and you have a drone or a texture available to the whole desk without giving up a channel. It works slightly differently from a reverb, because there is nothing being sent into it. Set its level with the aux return fader, not with the channel send knobs. The send knobs decide whether it is heard at all: with every send at zero the return is silent, and opening a send on any channel brings it in. So turn the return fader down to a level you like, then use any channel's send knob as an on switch. It stays silent until the transport is rolling, so a drone left on an aux will not sit there humming while you work.*

Master Section

The master sums every bus and aux return through the summing amplifier. It has a 3-band EQ with HPF and bypass, a vari-mu compressor with bypass and GR meter, a diode limiter with bypass and GR meter, 4 insert slots, a master fader (-60 to +12 dB), dual pan (L PAN and R PAN), mute, an arm button, and stereo meters. EQ, compressor and limiter all default to OFF - click ON to bring each in. The master has its own input stage with more headroom than the buses, for final glue before the output.

Summing amplifier

Everything passes through it on the way to the master. It is always on and has no controls. When several buses sum into one stage they interact in a way pure digital summing does not: at a couple of buses and moderate level it is transparent, but add buses and push level and the desk develops density and cohesion, with a subtle fullness low down and a softening up top. This is the glue - and the single biggest reason the whole mix adds up, so mind your levels into it.

Master EQ

Same shape as the bus EQ: HPF (Off, 30, 50, 80, 120 Hz), Low Shelf (40-500 Hz), Mid Bell (200 Hz-8 kHz), High Shelf (1.5-20 kHz), all plus or minus 12 dB, freely swept. Bypass is a true bypass. Small moves here land on the entire mix.

Master compressor (vari-mu)

A valve-modelled compressor for mix-bus glue. Its grip grows with signal level - gentle when things are calm, firmer as the mix pushes. It sits after the master EQ and before the limiter.

Control	Range
Threshold	-40 to 0 dB. Default: -10 dB.
Drive	0 to 10. Default: 3. How hard the signal is pushed into the valve stage - more drive is more compression and more of the valve's own colour. There is no ratio knob; a vari-mu's ratio is set by the music.
Attack	1, 3, 5, 10, 15, 30 ms
Release	0.1, 0.3, 0.6, 1, 2 seconds, or Auto (default)
Makeup	0 to 12 dB
Mix	0 to 100% (parallel)
SC HPF	Off, 60, 90, 150, 300 Hz

Tip: Auto release, then ease the threshold down until you hear it breathing gently with the music. It is forgiving - it leans in as things get louder rather than clamping transients. Use Drive for a firmer, more coloured grip. Best kept subtle: it is polishing a finished mix, not reshaping it.

Master diode limiter

A diode limiter modelled on the limiting stage in classic British broadcast desks, after the vari-mu - the compressor glues, the limiter catches what gets past and holds the top. Threshold is -20 to 0 dB (0 = off), Ceiling is -3 to 0 dB (default -0.3 dB), Release is Fast, Medium, Slow or Auto, and there is a Bypass.

The ceiling is not a brick wall. The signal runs clean until it comes within a few dB of the ceiling, then saturates up into it - leaning against the limit and colouring as it goes rather than hitting a hard stop. The ceiling holds, but it holds musically: push a lot into it and you hear character, not a clipped edge. Below the threshold it is transparent; pushed, it adds its own grit, edgier than the vari-mu. The horizontal meter shows how hard it is working.

Tip: Leave it off until your levels are set, then bring the threshold down to catch the peaks. Gentle, it is transparent peak control; pushed, it gets gritty - sometimes just what you want on a loud mix.

Master inserts

4 slots, after the limiter and before the master fader - mastering EQ, saturation, whatever the mix needs. 2 shown by default, expandable to 4.

Tape Machine

fBox Console includes a Tape Machine: a 16-track tape simulation on a classic 1-inch multitrack, with switchable 7.5, 15 and 30 IPS speeds. It sits in the console signal path and brings the character of analogue tape to the mix. Click the **TAPE** tab to open it; opening the tape panel closes the other panels and hides the Focus Window, to give the machine room.

The panel shows transport-synced reels, per-track drive knobs, LED meters, stereo output meters, an output drive knob, a quality selector, the speed selector, and the machine's serial number. The header shows the mode: CHANNELS (green) or BUSES (amber). The TAPE tab goes brown when tape is running.

Two routing modes (not both at once)

Channel-to-tape: all 16 channels go to tape together, all or nothing. Each channel returns on the channel 16 above it: Ch 1 comes back on Ch 17, up to Ch 16 on Ch 32. The console expands to 32 channels automatically. Only available when channels 17-32 are free, so if you use more than 16 channels, use bus-to-tape instead.

Bus-to-tape: all 8 stereo buses go to tape as 16 tracks. Bus 1 on tracks 1+2, up to Bus 8 on 15+16. The returns sum straight to the master and do not appear on channel strips. Engage it with To Tape on any bus.

Each tape track is mono. In channel mode the signal is captured after the fader, EQ and compressor, but before the pan pot. The source channel's PAN is copied to its return so the stereo image comes back where you put it; nothing else is copied. Because the return's pan is driven by its source, the knob on the return is greyed out and cannot be moved or automated. Turn tape off and channels 17-32 are ordinary channels again, with their pans back under your control.

Working with the tape returns

In channel mode, the tape returns on channels 17-32 are real channel strips, and this is where a lot of the fun is. Each one has its own EQ, compressor, inserts, fader and routing - so the tape-coloured signal is yours to shape further. Pan is the exception: a return's pan knob is greyed out, because it follows its source channel so the stereo image comes back off tape where you put it. Pan the source and the return follows. Roll off some top on a return to push it back, compress it for more of that tape squash, drop an fEq or fSat insert on it, or route a return to a bus to group and process your tape tracks together.

Your existing aux sends keep working when tape is on. The reverb and delay you set up carry on feeding the effects live, straight to master, in parallel with the tape signal. The tape returns have aux sends of their own, off to begin with, so you do not get the effect twice; turn them up if you want reverb on the tape-coloured sound instead. Stereo links follow too - a linked source pair returns as a linked pair. Returns are named after their source with "[Tape]" on the end, and soloing a source solos its return.

Tape on

The TAPE ON button is a bypass, not a mode switch: turn tape off and the panel stays as it is, turn it back on and you are in the same mode. While tape is running it overrides channel and bus routing - the signal goes to tape - and your original routing returns the moment tape is off.

Input drive (per track)

Range: -24 to +24 dB. **Default:** 0 dB. How hard the signal hits the record head. At 0 dB the tape runs at its natural level - gentle and largely transparent, but with the speed's own character already present. Up from there: warmth, then noticeable saturation and compression, and at the top the tape is being slammed. The LED meters show how hard you are hitting it. Double-click a readout to type a value. The two routing modes keep their own drive settings and swap over when you switch mode.

Output drive

Range: -24 to +18 dB. **Default:** 0 dB. Below 0 dB it is clean attenuation - just turning the level down, useful when the tape inputs are cooking and the signal off tape is already loud. At 0 dB it runs at unity with a touch of character. Above 0 dB it drives into saturation - grittier than the tape itself, with a thickening low down and the stereo field starting to move under load.

Tip: *The classic tape glue sound comes from moderate, even level onto the tape - not from slamming it. Bring each track's input drive up until the meters are peaking at the top of the green and just tipping into the yellow (around -9 to -6 dB on the dB scale down the right edge), and set the output drive to about the same. That is the sweet spot: enough level to warm and compress the signal, not so much that it grits up. Keep it in the yellow and out of the red. Push harder, or into the red, for obvious saturation; use the output drive above 0 dB when you want the amp's grit on top. Input drive for tape glue, output drive for amp grit - and if the inputs are hot, pull the output below 0 dB to tame the level while keeping the saturation.*

Tape speed

Three speeds, and they voice the tape quite differently - and the difference between them grows the harder you drive the tape, so audition them with some level going in, not just at a whisper. At **7.5 IPS** the sound is dark, warm and thick: the low bump sits low and full, the top rolls off early, saturation comes on soonest and hardest, and wow and flutter are most obvious - the lo-fi, characterful speed. At **15 IPS** (the default) you get the classic studio sound: a warm low bump, gentle softening up top, moderate saturation - the balance behind decades of records. At **30 IPS** you get maximum fidelity: the low bump moves up and becomes subtle, the top runs nearly flat and open, it takes real level to saturate, and wow and flutter are minimal - the mastering and clean-tracking speed. The reels spin at the speed you choose.

Tape character

The machine models the whole recording chain. High frequencies compress before low ones, so cymbals give way before the kick. There is warmth from the record head, the unmistakable colour of the tape itself, a bump at the bottom, a softening at the top, tape hiss, wow and flutter, and a little crosstalk between neighbouring tracks. All of it grows the harder you drive - a clean level shows the speed's basic voice, and pushing brings the character forward.

Tape quality and serial

The quality selector sets Eco (lightest on CPU), Studio (the sweet spot, and the default) or Master (the highest quality). There is no console-wide quality setting - quality is set per plugin and here on the Tape Machine. Drop it to Eco to save CPU in a large session, and switch it to Master before you render for the best possible tape sound (see Performance and CPU for the full workflow). Every Tape Machine also has its own serial number, separate from the console's, and its own character - head alignment, motor calibration, tape path. No two sound identical. The serial shows in the console footer beside the console's, and travels with your session and your saved desk.

Signal Flow

Understanding the order the signal passes through the desk helps explain why certain controls interact the way they do.

#	Stage	Notes
1	Input Trim	Clean gain, -18 to +18 dB
2	Preamp	Mode + signal level set the drive
3	Insert Slots 1-8	Hosted fBox plugins, in series
4	Compressor	Opto, single knob
5	EQ	HPF, Low bell, Mid bell, High shelf
6	Pre-fader aux sends	If set to PRE
7	Channel Fader	-60 to +12 dB
8	Post-fader aux sends	If set to POST (the default)
9	Metering	Pre-pan, post-fader
10	Pan	Equal-power pan law
11	Multi-bus Routing	Master and/or buses, in parallel
12	Bus Input Stage	Non-linear, level-dependent
13	Bus EQ	HPF + 3-band
14	Bus Compressor	VCA glue
15	Bus Inserts	4 slots
16	Bus Fader + Dual Pan	
17	Bus-to-Bus Routing	Cycle-prevented, zero latency
18	Bus-to-Tape	8 stereo buses to tape
19	Aux Return Input	Subtle warmth
20	Summing Amplifier	Analogue glue
21	Master Input Stage	Non-linear, higher headroom
22	Master EQ	HPF + 3-band
23	Master Compressor	Vari-mu
24	Master Diode Limiter	Saturating ceiling
25	Master Inserts	4 slots
26	Master Fader + Dual Pan	
27	Safety Limiter	Catches extreme peaks

The fader does not drive the preamp - preamp colour follows the level hitting it, which is set by TRIM. Any colour from riding a fader up appears downstream, at the bus and master summing.

Automation Pass System

The console records your mix moves against the transport and plays them back. You build the mix up in layers called passes, one at a time. The panel slides out near the master, with a transport readout, the arm controls, and your list of passes.

Recording your first pass

1. Click **+ New Pass**. It appears as "Pass 1" - click the name to rename. It starts OFF.
2. Arm what you want to record. Every channel, bus, aux return and the master has a small **A** button; click it and it turns red. Arm All arms everything, Disarm All clears it. The panel shows a summary like "8 ch / 4 bus / M armed".
3. Set the pass to RECORDING. The mode button cycles OFF (grey), AUTO ON (green), RECORDING (red). Click twice.
4. Press Play. Automation runs off the transport - nothing records or plays unless it is rolling.
5. Move controls. It only records while your hand is on a control: click and drag a fader and it writes; release and it stops. Everything else is left alone, so you can click around safely with RECORDING live. For a mute, a click is enough.
6. Stop, click the mode button round to AUTO ON, press Play, and watch it back - the controls move on screen.

True touch

Automation owns a control only for the window you recorded. Record a pan move from 1:05 to 1:25 and it owns that knob from 1:05 to 1:25 and nowhere else.

Outside that window the control still follows automation - it sits where the move left it, rather than springing back to some earlier setting. A fader ridden down to -10 dB and left there stays at -10 dB for the rest of the song, exactly as a real desk would.

Put your hand on a control and it is yours for as long as you hold it. Drag it and you hear the change straight away; automation steps aside for that control while you are moving it. Let go and it returns to whatever automation says for that point in the song.

That is deliberate. A move made by hand is not recorded anywhere, so there is nothing to bring it back the next time you open the session - it would only ever have been a value sitting on screen until something took it away. Rather than let that look permanent, the console hands the control back the moment you release it.

A control with no automation on it is not affected by any of this. Move it and it stays where you put it.

Where a move ends

A move ends when you let go of the control, or when you stop the transport, whichever happens first. Stopping mid-take finishes the move exactly as letting go does, and anything you do afterwards - moving the playhead, clicking elsewhere - is no part of the take.

If you stop moving a control but keep hold of it, that stretch is still yours. It plays back flat at the value you were holding, right up to the moment you let go, so what you heard while holding is what you get.

When a move ends, the control returns to whatever automation is underneath it immediately - it does not drift there. If there is nothing underneath, it simply holds where you left it for the rest of the song.

Dialling in a change on an automated control

Because automation takes the control back on release, you cannot simply grab an automated fader, set it where you want it, and store a section around it - it will return before you get there.

Switch the pass to OFF first. An Off pass keeps everything it recorded but stops playing it back, so the controls it drives are yours again: play the part, hear it, adjust it, and store your section. Switch the pass back to AUTO ON when you are done and it plays as before.

Off applies to the whole song, not just the part you are working on, and if two passes cover the same control you will need both of them off.

Tip: *If you find yourself fighting a control that keeps returning, it is telling you a pass owns it. Switch that pass off while you work, or record the change as a new pass instead - that way it is saved, and it plays back every time.*

What you can automate

Channels: fader, pan, trim, compressor, every EQ band, the two aux send levels, the aux on/off buttons, preamp mode, mute, and routing (except to-tape routing). **Buses:** fader, L and R pan, EQ, compressor, mute, EQ and compressor bypass, and routing. **Master:** fader, L and R pan, EQ, compressor, limiter, mute, and EQ and compressor bypass. **Aux returns:** fader, pan, and mute. **Plugins:** every knob on any hosted insert plugin, in any slot. Routing records as switches, so a channel jumping from one bus to another snaps cleanly where you did it and holds until the next change.

What you cannot automate

Solo is a monitoring control - the console never records it. The aux **Pr/Po** switch is not automatable, so set it where you want it before you record. **Insert-slot bypass** on any strip is not automatable either, though section snapshots do capture it. A **tape return's pan** cannot be automated. It follows its source channel, so there is nothing independent to record; automate the source's pan instead and the return follows. The **tape machine** cannot be automated either - not the TAPE ON button, the routing mode, the speed, the per-track input drives or the output drive - and to-tape routing is not automatable, so set your tape up before you record a pass. (Section snapshots do store the tape setup; that is a separate system.)

Important: *Plugin automation belongs to the slot, not the plugin. Take a plugin out of a slot and that slot's recorded moves are deleted with it; copying a plugin does not copy its automation. Moves in a locked pass are the exception - you are asked before those are destroyed. Build the chain first, then automate. Right-click a slot and choose Clear Automation to wipe a slot's moves by hand; that item only appears when the slot has moves to clear.*

Layering and managing passes

Make a second pass, set it RECORDING, and add moves on top - the first pass is untouched. Pass 1 might be fader balance, Pass 2 EQ, Pass 3 panning. If two passes hold the same control, the newer one wins; for everything else the older plays on. There is no limit on passes.

That is between passes. Within one pass, recording over a move you already made replaces it. From the moment you take hold of the control to the moment you let go, your new move is what is there - the old one does not survive underneath it. So if a fader ride is nearly right, just do it again over the same stretch; you will not end up with two moves fighting each other. Before and after your hand was on it, the earlier move plays on as it did.

If you do not like the new take, Undo Rec puts the old one straight back.

Per-pass control	What it does
Mode	Cycles OFF, AUTO ON, RECORDING.
Lock	Protects the pass - it cannot be cleared, deleted or recorded into. Lock a RECORDING pass and it drops to AUTO ON. It still plays back. This is how you make a good pass safe.
Clear	Wipes the pass's data. Disabled when locked.
X	Deletes the pass. Disabled when locked.

Per-pass control	What it does
Undo Rec	In the panel header. Removes what the last recording run wrote - and nothing else, leaving any earlier moves on the same controls untouched. It asks first, telling you how many controls it is about to clear and the stretch of song they were recorded over, so you can check it is pointing at the run you mean before anything happens. Single level; greyed until a run finishes. It covers faders and knobs, and the switched controls it records too - mutes, the aux on/off buttons, and the EQ, compressor and limiter bypass toggles. The one thing it does not undo is a routing change; put those back by hand.

Undo Rec stays available until you record again. You can stop, switch the pass to AUTO ON, listen back, work on something else, and come back to it an hour later - it will still be there, still pointing at the same run. It is only replaced when another recording run actually writes something, so rolling the transport with RECORDING armed and touching nothing leaves your undo intact.

A knob with recorded automation gets a gold ring; a fader with it turns its cap gold, and an insert slot with moves on it shows a coloured stripe down its left edge (see The slot marker), so one glance tells you what is driven and what is manual. Grab a control while automation plays and your move takes over at once; automation steps aside for that control until you release. Move the playhead, even stopped, and controls jump to their automated values at that point. Every pass, its data, your arm states and pass modes are saved with the session.

Locked passes and removing plugins

A locked pass cannot be cleared, deleted or recorded into - that is the point of locking one. It also cannot be quietly emptied out from underneath you.

Taking a plugin out of a slot normally deletes that slot's moves. If any of those moves are in a locked pass, the console stops and asks first: **A locked pass has moves on this slot. Going ahead deletes them.**

Choose to continue and only that slot's moves go - everything else in the pass is untouched, the pass stays locked, and it carries on playing back. Cancel and nothing happens at all: the plugin stays where it is and the pass is left alone.

The same question appears when you drag a plugin whose slot has locked moves. Dropping onto a slot that already holds a plugin swaps the two, so both ends can be carrying locked moves - the console tells you when both are affected, and going ahead clears both.

Tip: Lock a pass as soon as you are happy with it. It costs nothing, it still plays back, and from then on nothing can wipe it without asking you first.

Automation and buffer size

Automation records where the playhead is when your hand moves, and the playhead advances one buffer at a time. So there is always a small gap between the moment you move a control and the position your move is written to - about 12 milliseconds at 512 samples, about 23 at 1024. That is not a fault; it is simply how digital audio works, and every DAW behaves the same way.

For fader rides, EQ sweeps and anything that changes gradually, you will never notice it. Where it can show is on something sharp - a mute landing exactly on a beat, or a fast move you want tight against a transient. If a move feels slightly late when you play it back, drop the buffer to 512 while you record that pass and put it back up afterwards. That halves the gap.

Most of the time this is not worth thinking about. Set the buffer where the console runs comfortably and record your passes; if a particular move needs to be precise, lower the buffer just for that pass.

Which One Wins: Automation, Sections and the Base Mix

Three things can decide where a control sits: your normal mix, a section snapshot, and automation. They do not fight. Each one speaks only where it has something to say, and the order never changes.

At any point in the song:

Where you are	What drives the control
Inside an automation window	The recorded move.
Anywhere else, on a control a pass drives	Wherever its nearest earlier move left it.
Inside a stored section, on a control no pass drives	The section's snapshot value.
Everything else	Your normal mix.

The order matters: automation is asked first and answers everywhere, so a section only gets a say on controls no pass is driving.

The important thing is that this depends only on where you are in the song, never on how you got there. Click to a spot from before a move, from after it, or from inside a section, and the desk shows you the same thing every time. Play through and it does the same thing every time.

A worked example

You automate an aux send on the drums: a move ending at -17 dB. Later in the song you have a chorus section that stores that send at 0 dB. Playing through, you get:

- silence before the move - the send sits where it started
- the move, playing as you recorded it
- -17 dB after the move, holding
- -17 dB through the chorus as well - the send has a pass on it, so the section does not place it
- -17 dB after the chorus

Switch that pass Off and the same playthrough gives you the section's 0 dB through the chorus instead.

What this means in practice

Once you have automated a control, your normal mix no longer steers it - automation and sections do. That is deliberate: if your normal mix could reach in, it would drag the fader back the instant your move ended, and you could never leave a fader somewhere.

So if you want a control to be different somewhere, you have two tools:

- **Record another pass** on that part of the song. Best when the change is a move, or belongs with the music.

- **Store a section** there. Best when the change is a whole different setting for a whole part of the song.

Both are remembered and both play back every time. A bare hand move is neither - it is a try-out, and the next playthrough will tidy it away.

Tip: *If you find yourself repeatedly nudging the same control back to where you want it, that is the console telling you to record it. Drop a section on that part of the song and store it, or roll a short pass. Ten seconds of work and it stops happening.*

Sessions, Rendering and Transport

Session files

New, Open, Save and Save As in the header handle sessions, saved as .fbox files. A session holds everything: all channel, bus, aux, master and tape settings, routing, stereo links, names and colours, the channel count, the paths to your audio files, your insert plugins and their state, every automation pass, and all the arrangement data - dividers, names, snapshots, locks and crossfade setting. Both serials are saved too, so a session always reopens on the desk you mixed it on. New warns before clearing an unsaved session.

Open a session whose audio has moved or been renamed and the console lists every channel it cannot find, each with a Locate... button to point at the file's new home. Continue dismisses the dialog; anything you did not relocate simply stays empty and the rest of the session loads. On Windows you can associate .fbox files with fBox Console, so double-clicking a session in Explorer opens it.

Rendering and export

Master Out (in the header) renders the whole master output - your final mix. **Direct Out** (in a channel's dropdown) renders one channel's processed audio. **Bus Out** (in a bus's routing menu) renders a bus stem. Everything renders offline, faster than real time. Every pass, all the character, and all your insert plugins are in the file - it is the mix as you heard it. Formats are WAV 16-bit, WAV 24-bit, WAV 32-bit float, AIFF 24-bit or FLAC 24-bit; output sample rate is Device Default, 44100, 48000, 88200 or 96000 Hz (anything but Device Default is resampled). The save dialog warns before overwriting and remembers the last folder across all three render buttons. If a loop is active the renderer ignores it and renders straight through, then restores it.

Tip: Render to 32-bit float if you will master the file later, 24-bit WAV for general delivery, and 16-bit only for a final CD master.

Transport controls

Along the bottom of the window. **Spacebar** plays and pauses from anywhere.

Play starts from wherever the playhead is and toggles play and pause. If the playhead is sitting at the very end, it wraps to the start of whatever is looping so you are not left pressing play at a full stop.

To Start - the leftmost button - jumps straight to the beginning of the song, from anywhere, whether you are stopped or rolling. It ignores sections entirely.

Rewind is section-aware: the start of the current section on the first press, the previous section on the next, and zero once you reach the first one.

Stop stops and returns to the current section's start, or zero if there are no dividers.
FF skips forward 5 seconds.

Click the waveform in the arrangement strip to move the playhead anywhere; that also selects the section you land in. The readout shows the current time. Session length follows the longest loaded file, and the transport is the clock automation runs on.

Tip: *Play always starts from the playhead, so what you see is what you get. To hear a particular section, click it - that puts the playhead at its start and selects it in one action.*

Session tempo

Next to the transport buttons is the session tempo, shown in BPM. It reads 120.0 by default; click it and type a new value to change it (20 to 999). Because the standalone runs without a DAW, this is the master tempo for the whole session.

Any plugin you host that syncs to tempo follows this number - a delay set to note values, an LFO or modulation rate locked to the beat, the tempo-synced modes in fDelay, fMod and fVerb. They read the session tempo exactly as they would read it from a DAW, so set the BPM here and your tempo-synced effects fall into time with it. Change the tempo and every tempo-synced plugin re-locks to the new value.

Tip: *Set the session tempo before you dial in a tempo-synced delay or modulation, so the effect lands in time from the start.*

What fBox Writes To Your Computer

For transparency, here is every folder fBox creates and what goes in it. fBox writes nowhere else, and sends nothing anywhere unless you choose to.

Folder	Contents
Documents/fBox/Diagnostics/	Diagnostic logs, a folder per product. Small plain-text files recording what the console was doing, kept so a crash can be worked out afterwards. Every plugin running inside the console logs into the same folder, so it holds the whole session.
Documents/ fBox Units/	Your saved desks, a folder per product. Kept in Documents on purpose, so you can find, back up and share them. The console and the plugin share the folder.
AppData/Roaming/fBoxAudio/ (Windows)	Your licence, and a small settings file recording which desk you have pinned.

Sessions go wherever you put them - Save opens a normal dialog and suggests a folder, but the file goes where you choose. Nothing is filed away somewhere you did not pick, and nothing is written to your Desktop.

When something goes wrong

The log is written automatically as you work and survives a crash or freeze. If you hit a problem, zip the whole Documents/fBox/Diagnostics folder and email it to support@fboxaudio.com - send the entire folder, since what else was running alongside the console often explains the problem. Tell us what you were doing, what you expected, and what happened. The logs are plain text; you are welcome to read them, but you do not need to.

Performance and CPU

The console runs every channel, bus, tape track and master stage in one audio thread. A full 32-channel session with tape and full bus and master processing uses serious CPU - that is the cost of modelling the whole path. It skips work it does not need, though: silent channels with no inserts, silent linked pairs with no inserts, empty buses, and silent tape tracks. Channels with plugins loaded always process, even in silence, since generators like fDrone make sound from nothing. Aux returns always process too, whether or not anything is being sent to them, so a plugin sitting on a return stays alive and ready rather than freezing while it waits. No action is needed for any of it.

Keeping CPU down

Start at 8 channels and expand as needed; channels beyond the set count are not processed. Turn the tape machine off when unused. Leave bus EQ and compressor off on unused buses (they default off), and the master EQ, compressor and limiter off until the mix is ready. Close plugin editor windows you are not using. Raise the buffer if you hear dropouts: 256-512 for mixing, 1024 for a final render. Use Master Out for final mixes - the renderer runs flat out.

Session	CPU
8 channels, no tape, no bus processing	Light
16 channels with bus EQ and compression	Moderate
32 channels, tape, full bus and master	Heavy
The above with 100+ insert plugins	Very heavy

A desktop CPU from 2020 on handles a typical 16-channel session at a 256-sample buffer.

Mix at Eco, render at Master

The tape machine and every fBox plugin have a quality selector - Eco, Studio, Master. Quality costs CPU. Mix at Studio if your system can handle it, or drop everything to Eco if it cannot: the tape machine, and every plugin. The mix still plays, just at lighter quality. Then, when you are ready to bounce, switch everything up to Master and render.

Switching a big session to all-Master will probably make playback slow and glitchy, as the CPU struggles to keep up in real time. That is fine - you do not need it to play back smoothly to render it. The render is offline, so it takes as long as it needs and comes out correct at full Master quality, however much the live playback stuttered.

This is exactly how I work, on a genuinely ancient PC: mix the whole thing at Studio or Eco, flip everything to Master at the very end, and let the offline renderer take its time. Every one of my sessions plays back badly with everything on Master - and every one of them renders perfectly. You get a smooth mix and a full-quality file, with no need for a fast machine.

Tips

Starting a mix

Set TRIM so peaks sit around -12 to -6 dBFS. Rough balance with the faders near 0 dB. HIGH PASS at 80 Hz on everything except bass and kick. Aux 1 for a shared reverb, Aux 2 for a shared delay - sends around -18 dB to start, overall level on the return fader. Group similar channels to buses and ride the group. Small moves: the sound comes from everything interacting, not from one big move.

Remember the console adds up

Every stage adds a little colour and it stacks. Go gently across many stages rather than hard on one, especially with tape in the chain. If the finished mix feels hotter or more coloured than you meant, you are in good company - pull it back at the end. Push into the console and it pushes back.

Gain staging

Set TRIM to drive the preamp to the colour you want. Use the faders for balance. Hot channels push the bus summing stages harder, and that is where the density and the glue come from. Quiet channels stay clean while the loud ones do the colouring. This is how the desk was meant to be used, and it rewards good gain staging above everything else.

Console Character

Every console generates a unique serial number, shown in the bottom-right watermark. It defines the console's hardware character - the specific set of manufacturing tolerances that make this desk sound like this desk and no other.

No two units off a real production line sound identical. Capacitor tolerances drift the EQ's centre frequencies - one desk's presence peak sits slightly higher, another's slightly lower. Component variations shift gain staging so one channel breaks up a little sooner while another stays clean a little longer. Transformer differences give one desk a slightly warmer bottom and another a touch more air on top. These are not defects - they are the physical reality of analogue electronics, and they are what give a console its personality.

fBox Console models these variations across the whole signal path. Each of the 8 buses has its own EQ character, its own saturation threshold, its own compressor feel. The 32 channel strips each have their own subtle gain and harmonic variations. The master section has its own personality. Push two consoles with different serials to the same settings and one feels slightly warmer while the other has a touch more presence. The differences are musical, not technical.

The same serial always produces the same console, so a session sounds the same today as it did yesterday. And the desk you like is yours to keep and yours alone: save it, pin it, and its serial belongs to your machine and no one else's - the same way a piece of hardware you buy is one specific unit that exists nowhere else. The serial is saved with your session.

Tip: *You cannot choose your console's character any more than you can choose a real desk's - but you can learn it, and you can keep it. Mix on your console, get to know how it responds, and your instincts adapt to its personality. Pin the one you love and it is the desk you open every time.*

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