



fBox Console

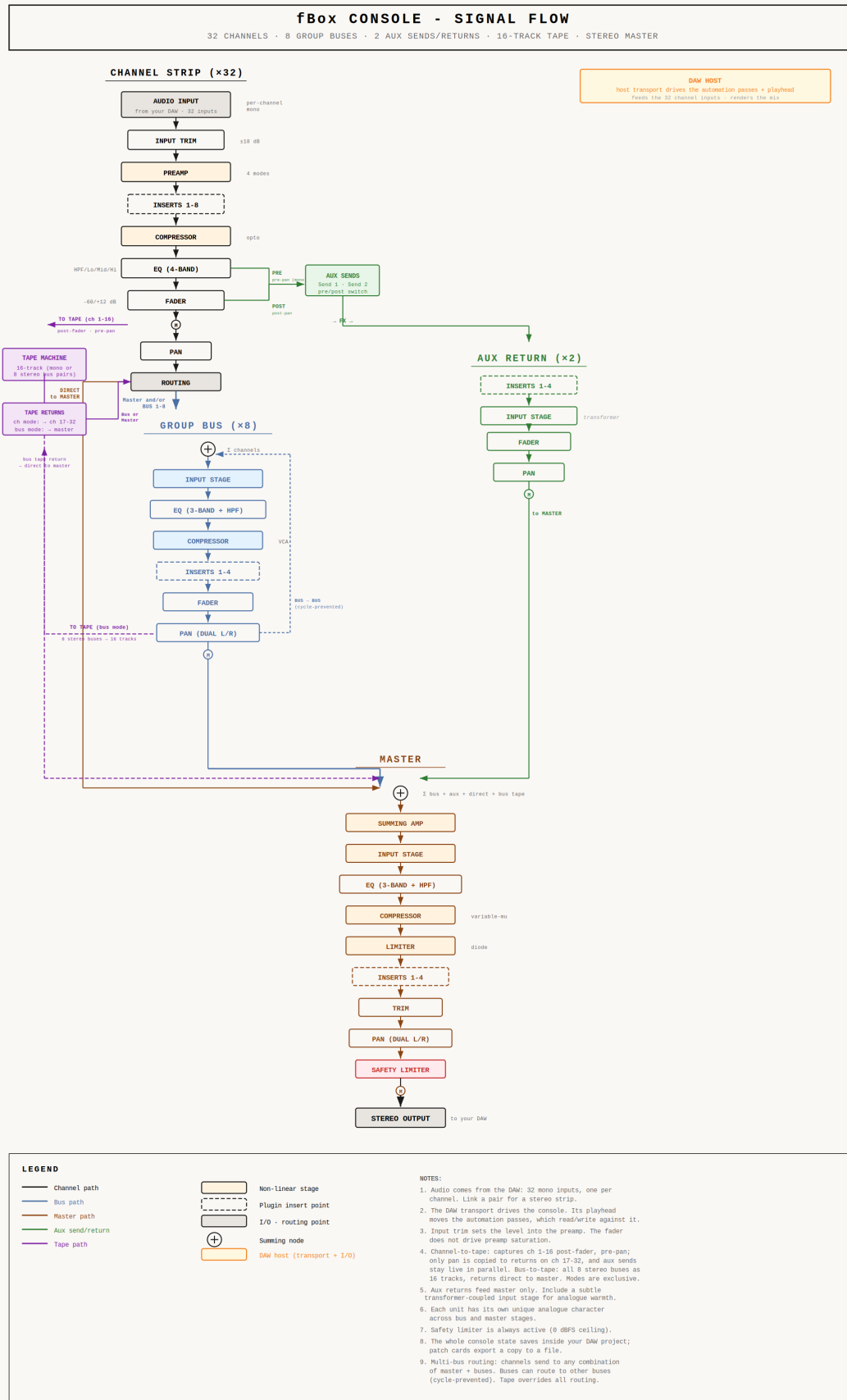
Virtual Analogue Mixing Console

Plugin

Product Manual

Build 1.0

by flakebelly
fboxaudio.com



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 Is

fBox Console is a virtual analogue mixing console that runs as a VST3 plugin inside your DAW. It is the plugin counterpart to fBox Console Standalone, which runs on its own as a Windows application: the same desk, hosted by your DAW rather than running by itself. fBox Console 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 plugin instance. Your DAW feeds the channel strips; you mix on the desk.

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, and almost every control is also available to your DAW's own automation lanes.

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 instance 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 DAW project.

As a plugin, fBox Console is driven by your DAW: the DAW's transport moves its automation, and the console's whole state is saved inside your DAW project. Its output is always stereo. Audio reaches its 32 channels through your DAW's routing - see [Getting Audio Into the Console](#).

Getting Started

Sound working in five minutes

1. Insert fBox Console on a track in your DAW.
2. Give that track enough channels for the console channels you need, and route your source tracks into the plugin's inputs. The console takes its audio from the DAW, one mono input per channel strip - see Getting Audio Into the Console for the full routing, including the recommended REAPER setup.
3. Click a channel strip to open it larger in the Focus Window on the left.
4. Use TRIM near the top of the strip to set working level. Aim for peaks around -12 to -6 dBFS on the channel meter.
5. Use the fader for balance, and start the DAW's transport to hear it.

The header shows "DAW Ins: N", the number of inputs the DAW is feeding, and each strip's SOURCE readout shows which input drives it. If a strip reads "No DAW Input" in red, nothing is routed to it yet.

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.

Getting Audio Into the Console

fBox Console takes its audio from your DAW, not from files. Inside the plugin it presents 32 mono inputs, one per channel strip: input 1 feeds channel 1, input 2 feeds channel 2, and so on up to 32. The SOURCE readout near the top of each strip shows which input feeds it - "In 5" means input 5. It turns green with a * when signal is arriving, and reads "No DAW Input" in red when nothing is patched to that channel. The header shows "DAW Ins: N", the number of inputs your DAW is sending.

To run the console as a full 32-channel desk, your DAW has to send each source to its own console input, which needs a host able to route audio into a plugin with many inputs. REAPER does this cleanly and is the recommended host. Most other hosts feed the plugin only a stereo pair, which is still useful but does not give you separate channel strips.

Important: *If you want the console at its fullest but your DAW cannot do this multi-input routing, use the fBox Console Standalone app instead. It loads audio files straight into its channels, needs no DAW, and gives you the whole desk with none of the routing below.*

DAW setup - REAPER (recommended)

REAPER routes any track channel to any plugin input, so it can drive all 32 console channels at once.

1. Insert fBox Console on a track. **2.** Open that track's Routing and raise Track Channels to the number of console channels you need, up to 32. **3.** Route each source to the console track with a send: a mono source goes to a single channel (the first to channel 1, the next to channel 2, and so on), and you can turn a source track's Master send off if you want it going only to the console. **4.** Open the plugin's pin connector (the ins/outs button at the top of the FX window) and confirm each track channel maps to the matching plugin input. **5.** Play, and watch the SOURCE readouts turn green as signal arrives.

A stereo source can go to any two channels: both sides play, and you can pan them left and right by hand. But STEREO LINK and LINK INSERTS only work on a fixed pair - 1-2, 3-4, 5-6, and so on up to 31-32 - so to link the two sides, send the stereo source to a proper pair, starting on its first channel (1, 3, 5...), then click STEREO LINK on that first channel. It links the two and pans them hard left and hard right. The standalone app does this for you on file load; in the plugin you set it by hand, because the audio comes from the DAW.

Tip: *Once the routing works, save it as a REAPER track template so you do not build it again every session.*

Other VST3 hosts

Most VST3 hosts feed a plugin only a stereo input, and some add a single stereo side-chain, so four channels at the very most. They cannot patch 32 separate sources into the console. Cubase and Nuendo, Studio One, Ableton Live and Bitwig Studio all sit in this group to different degrees, and most other VST3 hosts do too. Test your own, but expect a stereo input rather than 32. This is exactly why the 14-day free trial is worth taking before you buy: try the console in your own host first and see how it feeds the channels before you spend anything.

That does not make the console useless in these hosts. Put it on a group, a bus, or the master and run your whole stereo mix through it - see *Two Ways to Use the Console*, which covers this properly and treats it as a deliberate way to work, not a fallback. For the full 32-channel desk in a host that cannot route that many inputs, use the fBox Console Standalone app.

Hosts and systems that cannot run fBox Console

fBox Console is a Windows VST3 plugin. **Pro Tools** loads AAX only, not VST3, so it cannot host the plugin - on Windows you can still use the fBox Console Standalone app, or the workflow below. **Logic Pro** and **GarageBand** are Mac only and load Audio Units only; fBox Console runs on Windows only, so it does not run on those systems at all, neither the plugin nor the standalone.

How I work

I mix in Pro Tools, which cannot host this plugin, so here is what I actually do. I record and edit everything in Pro Tools, then bounce each part out as its own mono or stereo file. I load those files straight into the fBox Console Standalone app, mix the whole thing on the desk there, and render a stereo file. Then I master that elsewhere. The plugin is here for the hosts that can drive it; the standalone is here for everyone, and it is where I do my own mixing.

Two Ways to Use the Console

The console works in two quite different ways, and both are worth knowing, because the second is not a lesser version of the first. It is a deliberate way to work that many people will choose on purpose.

As a full mixing desk

Feed each source into its own console input and you have the whole desk: 32 channel strips, buses, aux returns, the tape machine and the master section, mixing entirely inside fBox Console. This is the fullest use, and it needs a host that can route many inputs into one plugin (see Getting Audio Into the Console). It is the closest thing to sitting at the standalone app inside your DAW.

As a mix-bus, master or mastering unit

Put a single instance on a group, a bus, or your master, and run a finished stereo mix, or a set of stems, straight through it. You are not using the 32 channel strips now, but you keep the parts of the console that colour the most: the summing, the master EQ, compressor and limiter, the tape machine across the whole mix, and the console's own serial-numbered character. One instance on the master turns your mix bus into an analogue-style console output stage.

This is the way to use the console in any host that cannot route many inputs, since it needs only a stereo path. But it is just as valid as a choice in a host that can. Plenty of engineers do not want to rebuild their channel mixing inside a plugin; they want their existing mix to pass through a console on the way out, and pick up the weight, glue and colour that a real desk's output stage gives. Placed on the master, that is exactly what fBox Console becomes.

On the master bus specifically

This is the placement worth reaching for first if you are not doing full multi-input routing. Your entire stereo mix passes through the console's summing, master processing and tape, so the whole thing is coloured together, as one, the way a mix sums through the master section of a hardware console. It is the simplest setup in the manual, works in every host that runs the plugin, and gives you a large part of what the console is for from a single instance.

In a mastering session

The same idea, one step further along. Drop the console on a mastering chain and run finished tracks through it for a touch of vintage console tone, tape warmth, or analogue-style glue on the stereo bus. Keep the moves small here, since you are working on a finished master, but the summing, the tape machine and the master compressor and limiter all earn their place at this stage.

Tip: Not sure which to use? If you want to mix a whole song inside the console, you need a host that routes many inputs, or the standalone app. If you want your existing mix to sound like it came off a desk, put one instance on your master and run everything through it. The second takes about ten seconds to set up and is where a lot of the console's character lives.

Activation and Licence

The first time you open fBox Console it 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 by every new instance - this is how you keep one desk for good.
Use a new unit for new instances	Go back to a fresh desk for each new instance. 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 Standalone app uses. A desk you save in the plugin turns up in the standalone'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.

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 top of the strip 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: Scan Plugins, Save Patch, Load Patch, Desk, Help and Reset Engine buttons, the channel-count selector, the "DAW Ins: N" input count, 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](#)).

Controls: the Channel Strip

Top to bottom, a channel strip runs: the name, the SOURCE readout, 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.

Name and source

At the top of the strip is the channel name; double-click it to type your own, and left alone it reads Ch 1, Ch 2 and so on. Directly below is the SOURCE readout, which shows which DAW input feeds this channel: "In 5" means input 5. It turns green with a * when signal is arriving, and reads "No DAW Input" in red when nothing is patched to that channel. SOURCE is a readout only - routing is done in your DAW (see Getting Audio Into the Console). A Reset Channel option returns every control to its default and removes every insert plugin from the strip; note it clears the inserts completely, it does not reset them in place.

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. Right-click a loaded slot for its menu:

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

Menu item	What it does
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.

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, use a fixed pair - 1-2, 3-4, 5-6 and so on - with L on the lower channel. **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.

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. On a tape return channel (17-32 while channel-to-tape is running) the pan knob is greyed out - it follows the source channel's pan and cannot be set independently.

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), 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 a return's pan is driven by its source, the return's own pan knob is greyed out and cannot be moved - by hand or by automation. It follows the source channel, including any automation on the source's pan. Everything else on a tape return is yours: its fader, EQ, compressor, inserts, routing and aux sends all work normally and can all be automated. Turn tape off and the channel becomes an ordinary strip again, pan included.

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, pan, fader and routing - so the tape-coloured signal is yours to shape further. 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), separate from the console's own processing quality. 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 your DAW's 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. This is the console's own engine, separate from your DAW's automation lanes, though every console control is available to those lanes too.

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. Start your DAW playing. Automation runs off the DAW's 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, roll the transport again, and watch it back - the controls move on screen.

True touch

Automation drives a control from the moment your first move on it begins. Record a pan move from 1:05 to 1:25 and the knob is automation's from 1:05 onwards - during the move it follows what you rode, and after it, it stays where the move left it.

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 project - 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 the transport stops, 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.

Adjusting an automated control

Because automation takes the control back on release, you cannot simply grab an automated fader and hold it at a new value to audition the change - it will return the moment you let go.

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, and set it where you want. When you have found the value you like, switch the pass back to AUTO ON and record it into a pass (or a fresh one) so it plays back every time. 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.

The value you settle on is remembered - see Hand-set values are remembered.

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.*

Hand-set values are remembered

The console keeps track of where you last set each control by hand, and that value is what it returns to whenever no pass is driving it.

This matters when a pass goes away. Switch a pass off, clear it, or delete it, and the control does not jump back to some earlier setting from before you recorded - it returns to the last value you put it at. That is almost always the one you want, because it is the one you chose most recently.

An example. A fader sits at -6. You record a pass riding it. Afterwards, with the pass off, you set it to -12 because you have decided that is where it belongs. Turn the pass on and off again and it returns to -12, not -6.

It works the same on a hosted plugin's knobs, so a plugin knob you set by hand after recording a pass on it keeps that setting.

Nothing you do while recording is remembered this way. Moves made during a take belong to the pass, not to the desk, so a pass can never quietly become the value the console falls back to.

The remembered values are saved with your project, and a control the DAW is automating is left out entirely - that one belongs to the DAW.

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. **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** is not recorded by the console's own passes. It is exposed to your DAW, so a DAW lane can automate it - but the console's automation cannot. To automate a plugin dropping in and out from a console pass, automate that plugin's own bypass control on its editor instead. A **tape return channel's pan** cannot be automated either. It mirrors its source channel, so there is nothing to record - automate the source channel's pan instead and the return follows it.

Important: *Plugin automation belongs to the slot, not the plugin. Move a plugin and its moves stay behind in the old slot; copying a plugin does not copy its automation. Build the chain first, then automate. Right-click a slot and Clear Automation to wipe a slot's data.*

DAW automation

Because fBox Console is a plugin, every one of its controls is also exposed to your DAW's own automation lanes, so you can drive the desk from the DAW timeline instead of, or alongside, the built-in passes. Alongside means on different controls: DAW lanes on some, console passes on others, all working at once. On any single control, one system drives it.

If both drive the same control, the DAW wins - everywhere. Draw a DAW lane on a fader and the console's own passes stand down on that fader completely, including inside a window they recorded. Only one system drives a control at a time, so there is nothing to fight over.

This is per control, not per position. A DAW lane anywhere on a fader claims that fader for the whole song.

Everything else is unaffected. Passes on controls the DAW is not automating carry on exactly as normal, so you can have DAW lanes on two channels and console passes on the rest, all at the same time.

Your console pass is never lost. It stays recorded, the gold ring still shows, and it plays again as soon as the control is yours.

Getting a control back. Delete the DAW lane, then either record on that control - arm it, set the pass to RECORDING and ride it - or move it by hand with the transport stopped. Either hands it back immediately. Reopening the project also does it.

A DAW lane's moves are never written into a console pass; the two stay separate.

A note on Latch and Write modes

In these modes your DAW records whatever arrives at a control - and console automation arrives at the control, because that is how it reaches the sound. So playing a console pass through a track armed in Latch writes those moves into the DAW's own lane.

That is useful if you want it: it is how you transfer a console pass into your DAW. But once the moves are in the DAW's lane, the DAW is driving that control, so the console stands down on it. Your pass is still there and still intact - it just is not what you are hearing any more. If you did not mean to do it, undo the lane in your DAW and record on the control again to take it back.

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.

Per-pass control	What it does
X	Deletes the pass. Disabled when locked.
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.

A console knob with recorded automation gets a gold ring; a fader with it turns its cap gold, so one glance tells you what is driven and what is manual. Hosted-plugin knobs do not show the gold ring - that marking is for console controls only. Grab a control while automation plays and your move takes over at once; automation steps aside for that control until you release. Move the DAW 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 DAW project.

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 a 512-sample buffer, about 23 at 1024. That is not a fault; it is simply how digital audio works, and it applies to your DAW's own automation exactly as it does to the console's.

In the plugin the buffer is your DAW's, set in its audio preferences, not something the console controls. For fader rides, EQ sweeps and anything that changes gradually, you will never notice the gap. 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 your DAW's buffer while you record that pass and put it back up afterwards. Halving the buffer 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 it just for that pass.

Saving Your Work

Saved with your DAW project

As a plugin, fBox Console saves its entire state inside your DAW project. Everything is held there: all channel, bus, aux, master and tape settings, routing, stereo links, names and colours, the channel count, your insert plugins and their state, every automation pass, and both serials. Reopen the project and the console comes back exactly as you left it, on the same desk. There is no separate session file to manage - your DAW handles it.

Patch cards

A patch card is a portable copy of the whole console, saved to a file so you can carry a setup between projects, between instances, or to another machine. Save Patch in the header writes the current console to a patch card (a .json file) - everything a project would hold, in one file you choose the home for. Load Patch reads one back in.

If the card was saved on a different desk, loading asks how to bring it in. **Recall Unit** brings back the exact desk the card was saved on - its character and all its settings, the unit as it was. **Apply Settings Only** keeps your current desk and drops the saved settings on top: the same knob positions, your machine's character. A patch card carries its identity, so a card saved for a different fBox product is refused with a note telling you which one it belongs to.

Important: *A patch card and a saved desk are different things. A desk (see Your Desk) is just the console's character and serial; a patch card is the whole console, settings and all.*

Rendering

There is no offline renderer in the plugin - your DAW does the bouncing. Everything the console does is in the DAW's render: every automation pass, all the character, and all your insert plugins. Freeze or bounce the console track in your DAW when you want the heavy processing to run offline (see Performance and CPU).

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.

Patch cards go wherever you save them - Save Patch opens a normal dialog and the file goes where you choose. The console's own state lives in your DAW project, not in a file of its own. 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 your DAW's buffer if you hear dropouts: 256-512 for mixing, 1024 for a final render. Freeze or bounce the console track in your DAW for the final render, so the heavy processing runs offline.

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 DAW's offline render 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 the DAW project.

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.*

fboxaudio.com