Mixbus v6

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Harrison Consoles

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1. Introduction

Mixbus – Bringing Console Style Mixing to the DAW!

Welcome to Mixbus! Mixbus is an “analog-style console” that has been incorporated into a Digital Audio Workstation. The Mixbus engine was developed at Harrison Consoles, where we have been designing high-end digital and analog consoles for over 40 years.

Many great-sounding, influential, and legendary works were recorded and/or mixed on Harrison consoles, including:

- **Michael Jackson** “Bad” and “Thriller” (the best-selling album of all time).
- **AC/DC** “Back in Black”
- **Janet Jackson** “Rhythm Nation”
- **Queen** “The Game”
- **Sade** “Diamond Life” including the hit single “Smooth Operator”
- **Paul Simon** “Graceland”
- **Supertramp** “Breakfast in America”
- many, many recordings on Manfred Eicher’s **ECM Records**
- much more by artists such as **Led Zeppelin**, **ABBA**, **U2**, **Madonna**, **Mike Oldfield**, and **Frank Zappa**.

Harrison continues to be a major force in the ultra-high-end professional audio community. World-class facilities like Sony Pictures use Harrison consoles to mix top-tier films.

Image:
Harrison MPC4-D Console at
Sony Pictures
Sound Stage 6
Culver City, CA

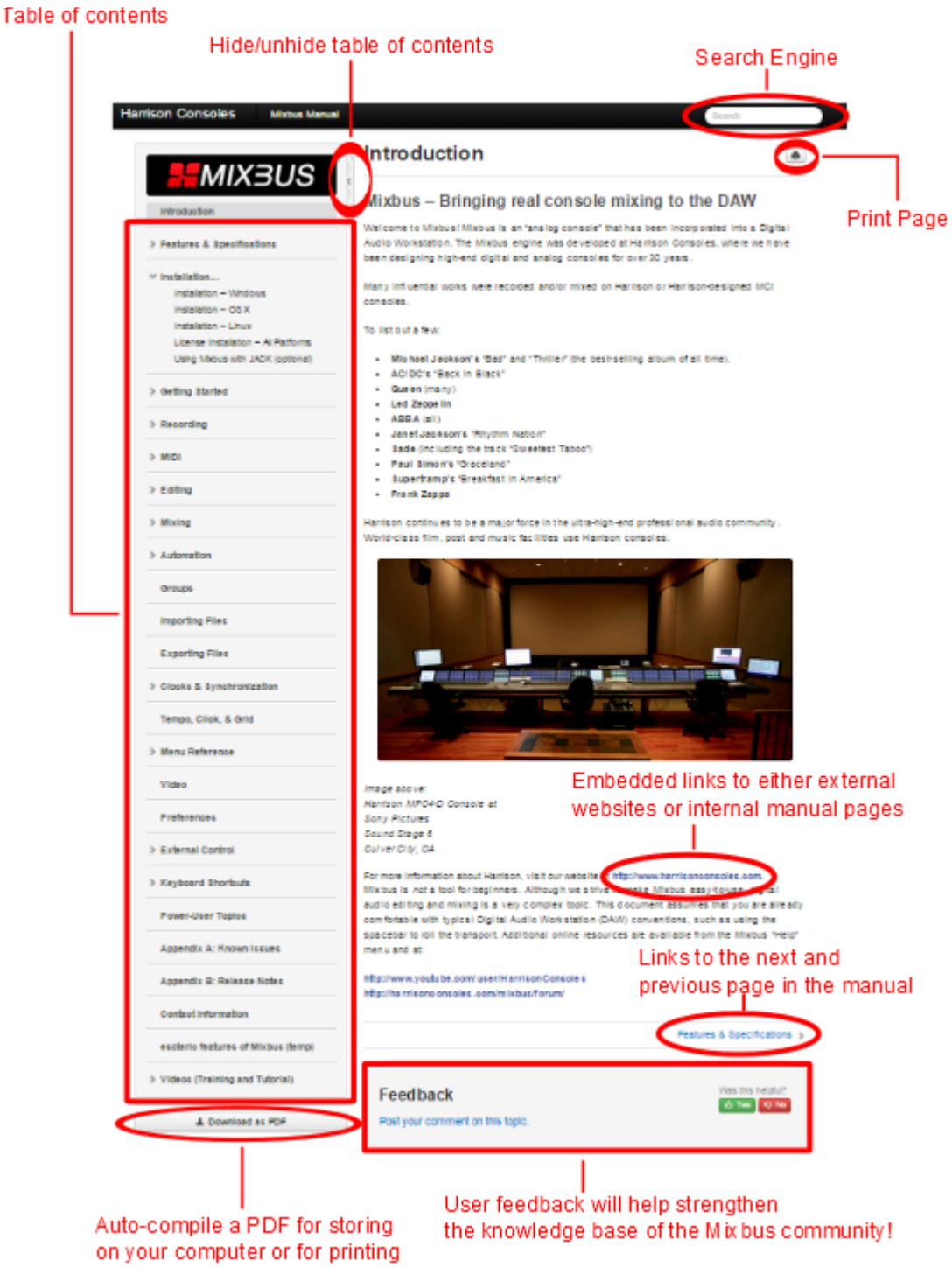
For more information about Harrison, visit our website at **http://www.harrisonconsoles.com.**

Mixbus is *not* a tool for beginners. Although we strive to make Mixbus easy-to-use, digital audio editing and mixing is a very complex topic. This document assumes that you are already comfortable with typical Digital Audio Workstation (DAW) conventions. Additional online resources are available from the Mixbus “Help” menu and at:

http://www.youtube.com/user/HarrisonConsoles
http://harrisonconsoles.com/mixbus/forum/

1.1. About This Manual (online version and PDF download)

The following image will help you understand a few conventions used in the online version of the Mixbus manual. **A PDF version has been made available for download via a link in the lower left corner of the page**. Don't forget to use the "Feedback" feature near the bottom of each page. (Feedback is intended for manual specific items only. If you need tech support or would like to make a feature request please email us or visit our user forum.)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Mixbus – Bringing real console mixing to the DAW
Embedded links to either external websites or internal manual pages
Links to the next and previous page in the manual
Feedback
Post your comment on this topic.
Auto-compile a PDF for storing on your computer or for printing
User feedback will help strengthen the knowledge base of the Mixbus community!

2. About Mixbus

Mixbus is a DAW (Digital Audio Workstation) created by console designers, not computer people. We uniquely present a DAW mixer that looks and operates just like an actual console. This approach has helped Mixbus be enthusiastically adopted by engineers of all specialties including those who worked with or are still working with mixing consoles.

Mixbus is renowned for preserving the “Harrison sound” from our consoles in a uniquely flexible DAW format. Superior sound plus comprehensive DAW features make Mixbus the premier choice for any project.

2.1. Difference between Mixbus and Mixbus 32C

Mixbus 32C is a separate product from Mixbus and is purchased and licensed separately. It is possible to have both installed on your computer at the same time.

Mixbus 32C is the premium version of Mixbus. While it retains all of the superior features of Mixbus it adds an important difference: the channel signal path models the analog circuitry of our classic 32 Series console from 1975. It also adds additional mixing buses and has a more advanced channel EQ which is a model of the EQ circuit in the original analogue console.

2.2. What's New in Mixbus v6

Mixbus v6 adds the following new features and improvements:

- 5th generation processing engine including extensive latency compensation and cue-monitoring.
- Graphical operation improvements, largely suggested by forum users, that improve Mixbus's usability on smaller monitors.
- Foldback buses allow the building of multiple custom cue mixes for artists.
- Region List is redesigned and a new Source List that helps you manage recording takes and organize imported files such as loops.
- "Cut" Tool provides a faster and clearer way to split regions with a dedicated tool mode.
- MP3 import/export on Mac, Windows, and Linux.
- Virtual MIDI keyboard window allows you to "play" MIDI notes into a track when you don't have a physical device available.

5th Generation Mixbus Engine

The v6 release involves several deep improvements in the **5th generation Mixbus engine**.

The 5th generation engine incorporates almost 3 years of continuous behind-the-scenes development. Many subtle behaviors, like the ringing-out of reverb and delay tails after transport stop, are greatly improved.

Mixbus v6 launches faster than v5, and the transport (play, stop, locate) is much smoother and more responsive. You should also notice improvements in looped recording, looped playback, and varispeed.

Some of the specific improvements are listed below:

Latency compensation:

- In addition to the Mixbus console (which has always been latency-compensated), it is now possible to add additional buses alongside the tracks, which can be used for additional grouping, effect returns, and other tasks. And these buses are now fully latency-compensated just like the regular mixbuses>master path.
- Plugin Delay Compensation (PDC) can now be disabled. This might be useful when recording live tracks while monitoring prerecorded tracks.
- To disable PDC, you must first click Preferences->Toolbar->"Display Latency Compensation Info" to expose the latency-compensation toolbox.

Monitoring improvements:

- By default, MIDI tracks will now monitor both the "input" and "Disk" when they are in automatic mode.

Users can override this on a per-track basis and choose to monitor the input or Disk, only.
- Audio tracks can now monitor both the input and disk simultaneously.
- The SDIO (separate disk I/O) will also allow the addition of the “record-wet” feature (reserved for a future release)

Operational Improvements

- Slates now show the track number and use a larger, higher-contrast font.
- Track number turns red if the track is record-armed.
- Renaming a mixbus will automatically display the new name inside the channel strip bus enable buttons.
- Channel VCA and phase buttons are now always visible at the bottom of the channel strip.
- The EQ knobs now show “freq” rather than percent. This is a popular request from users on our forum.

Source List + Region List

- The “Region List” has been redesigned to show only the regions actually in use in the current snapshot. The secondary function of the region list (showing the source files for every region) has been moved into its own dedicated section, the “Source List”

Region List

- The “Region List” shows only the regions that are used in the current snapshot. Selecting a region in the list will select the corresponding region on the editor canvas. This allows you to reverse-interrogate your edits by, for example, sorting regions by their start time and then selecting them.

Source List

- The “Source list” shows the file sources used by the session, whether they are used in the current snapshot or not.
- Sources have a “capture time” (i.e. Take) stamp, so the user can find other regions captured at the same time

Tagging Regions and Sources

Regions and Sources may be “tagged” together, which can help in session management. For example, you can tag all the regions that were part of a good overdub; or you can tag regions with the particular artist who recorded them. See Edit->“Tag selected regions” in the main menu.

Separately Pannable Utility (“Aux”) buses

- In addition to the Harrison console signal flow structure, you may add an unlimited number of audio buses (sometimes called “Aux” or “Utility” buses).
- When a track has a “Send” to a utility bus, you can double-click the send to access the Send’s level and pan-control (if any) from the channel’s fader area.

- Click “Show Sends” on the bus, and the mixer window will “spill” the attached tracks into view, and provide the send-level and pan from the channel’s fader area.
- Learn more here: DAW Style Mixing Buses

Foldback buses

- Foldback buses can be used to create headphone mixes for your talent (artists). Foldback buses represent a send to a headphone mix. The mix for a foldback bus can be the combination of any mixer strip, either pre- or post-fader; the foldback bus has its own plugins and fader, allowing you to create a complete custom sound for each artist. Furthermore, it is possible to remotely control foldback mixes from an OSC-enabled tablet or phone application.

Enhanced transport syncing

- Mixbus can now record when an external transport sync source is varispeeding (playing at a speed other than 100%). This improves the ability of Mixbus to transfer tracks from analog tape, among other benefits. It will also allow us to emulate traditional analog recording studio tricks. We plan to expose controls for those new features in the 6.x update series.

A new “Transport Masters” dialog is provided to display all current sync sources, and their speed and location. You can select which external synchronization source to chase on-the-fly. This is a huge advantage for demanding tasks like on-site live broadcasts or providing backing tracks for a live show.

> ! Due to unsolved issues in the v6.0 release, the external sync selection, and Transport Masters window, will be disabled until an upcoming hotfix release.

“Cut” tool

- Additional to the built-in region splitting capability, a Cut (scissor) tool is added similar to those used in other DAWs.

Virtual MIDI Keyboard

- An on-screen keyboard is available to enter rhythms and melodies when you don’t have a physical MIDI keyboard available. The virtual keyboard maps your QWERTY keyboard into a note-input device.

- The Virtual MIDI Keyboard can be accessed in the “Window” menu. By default, the Virtual MIDI Keyboard is connected to the input of the currently-selected MIDI track. For more details, see: Virtual MIDI Keyboard

MP3 Import

- In the past, only the MAC version could import mp3's using the Apple built-in decoder. Now Windows and Linux can also import mp3 directly.
- On import, MP3 files are converted to a .wav file (inside the session folder) for internal use.

Default preferences and behavior changes:

Waveforms now Logarithmic

- Many of our users preferred the "Logarithmic" waveform display option. The default value for waveform displays has been changed to "Logarithmic". If you were using that option in the past, then you might notice your region waveforms now appear visually larger. (you will not hear any difference, though)

Auto-Input behavior

- "Tape Machine Mode" has been replaced with "Auto Input does Talkback", which is defaulted OFF. In the past, if you had Auto-Input engaged, the default behavior was to monitor the inputs when transport was stopped. In a multi-room studio, this can help performers hear each other and rehearse between takes. But this is not so useful in a typical DAW studio when everybody is in the same room, and it might lead to feedback. So we changed the default operation of Auto Input. If you want the old operation, then enable the option "Auto Input does Talkback" in Preferences->Signal Flow.
- Auto-Input is now defaulted "ON" for new users.

Elimination of Separate Sidechain Bussing

- The discrete sidechain bus of earlier versions is eliminated in V6. Sidechain controlled compression is replaced with a plugin-based approach.

3. Features & Specifications

Mixer Features:

- Mixbus uses precision DSP algorithms developed by Harrison for our world-renowned large format digital mixing consoles. Mixbus also incorporates elements from the rich history of Harrison analog designs. Mixbus has been optimized to provide a classic Harrison sound.
- Straightforward “what you see is what you get” mixer layout based on Harrison’s renowned 32-series and MR series music consoles.
- Unlimited stereo or mono input channels (subject to available CPU power of course) featuring EQ, filter, compression, mix bus sends, and mappable plug-in controls on every channel.
- Like a hardware mixer, the DSP horsepower for EQs and compressors is preallocated. Enabling a compressor or EQ will not increase the CPU usage of the computer.
- Dedicated stereo mix buses (can be used for groups or aux buses) featuring tone control, compression, and tape saturation.
- Stereo Master that features tone control, compression, tape saturation, and final limiting.
- K-meter on master output helps you make dynamic, polished-sounding mixes.
- Stereo phase correlation meter indicates mono-compatibility of your mix.
- Plug-in delay compensation to support effects such as parallel compression.
- Plug-ins may be pre or post fader in the channel strip signal flow.
- VCA faders allow arbitrary grouping of fader and mute/solo between channels. Multiple VCAs may be attached to any channel.
- Spill buttons are provided on each mixbus and VCA to show only the tracks that feed it.
- Monitor Section provides dedicated plugins, solo options and speaker-management features.
- Foldback section provides an unlimited number of cue feeds for your artist headphone mixes.
- Comprehensive metering with peak, peak hold, and compressor gain reduction visible on every track and bus.
- The engine is internally dithered, ramped, and gain staged so that sound quality is preserved as close to analog as possible.

Editor/DAW Features:

- Unlimited audio, MIDI, and virtual instrument tracks.
- Every track/bus has unlimited pre- and post-fader inserts, sends, and plug-ins.
- MIDI tracks with MIDI output to external devices OR virtual instruments.
- Non-destructive, non-linear editing with persistent undo.
- Comprehensive project management with Snapshots, Playlists, Templates, Mixer store/recall, and more.
- Fast and intuitive audio editing with “smart” mouse editing modes.
- Dynamic automation of fader, pan, and plug-in parameters.
- Pitch/time stretching, transient detection, and varispeed playback.
- Object-oriented, per-region volume, dynamic volume, crossfade in/out, and mute functions.
- Standard file formats (BWF, WAV, WAV64, AIFF, CAF, MP3 & more…).
- Supports industry standard plug-in formats and I/O devices on each supported platform.

- Transport control via Level 2 MMC or MTC (master or slave).
- LTC (SMPTE) and MTC chase and generation.
- Assign MIDI CC knobs to mixer controls with just 1 click.

General

- Refined workflow and graphics, including scaling and support for HiDPI (Retina) displays
- Multicore DSP processing allows more simultaneous tracks and plug-ins
- 64 bit builds provide extended memory for virtual-instrument sample libraries
- 32 bit and 64 bit version available for all three platforms (OS-X, Windows, Linux)
- Synced video-playback window, with import and export
- Lua scripting engine allows automated workflows

System Requirements:

- Mac: 10.6.8 or newer, 32bit or 64bit (PowerPC is no longer supported) with any CoreAudio device.
- Windows: XP Pro or newer, 32 bit or 64 bit with MME or ASIO device
- Linux system (x86 or x86-64) with ALSA
- Two or more processors/cores

*See our website for the latest details on OS compatibility, plug-in compatibility, and system requirements:

https://mixbus.helpscoutdocs.com/article/49-system-requirements

3.1. What's Different About Mixbus?

"True Analog" Sound

Harrison is unique among audio companies: In 1985, we launched digitally-controlled *analog* consoles which were installed at many world-class music and film mixing facilities. A few years later, digital recording and playback became the new standard, and our customers requested a complete digital signal path in their facilities. Harrison made a bold proposal: we would replace the existing analog processing racks with a digital system, and leave the control surface in-place so the mix engineers could continue to work with the same workflow and sound that had served them well. Of course, to make this practical we had to make the digital engine "sound" exactly like the analog engine it was replacing.

During this process of replacing an analog console processor with a digital version, we made many discoveries about how to make digital sound less like digital, and more like analog. We've been refining these technologies for over 35 years since the launch of those first digital consoles. Harrison has made several generations of analog and digital consoles since that time, and has been continuously improving our DSP. This unique development path is what gives Mixbus a different sound than traditional workstations.

Analog-style Workflow

Most DAWs were first introduced in the stone-age days of computers. Processors were rated in megahertz and memory was measured in kilobytes.

At that time, the standard tool was the studio console; it provided nearly every feature that was needed for day-to-day recording, editing, and mixing tasks. With the addition of a few outboard boxes (such as reverbs, delays, and maybe a final compressor) it was possible to create a polished album.

When computers were introduced, they were unable to recreate the features of then-standard tools such as a 48-track tape machine and console. Those early computer systems could only play back 4, 8 or 16 channels; and because CPU power was so desperately limited, you had to choose which of those channels would be assigned an EQ. In this model, even the most basic console features had to be treated as "outboard" gear. And so was born the "Plug-In" !

Fast-forward to the present, and you know that computers are fast enough to run hundreds of channels, each with a half-dozen plug-ins. As a new-generation workstation, Mixbus takes advantage of today's fast computers to take the load off of the engineer, and return to the fun and creativity of mixing music.

Open Source, Open Standards, Open Community

Mixbus is largely open-source and is the collaborative effort of a worldwide team including musicians, programmers, and professional recording engineers. Many workstations provide "open" control or plugin protocols… Mixbus goes far beyond that. Like a good piece of vintage hardware, you can open the box and

look inside. This transparency encourages Mixbus development to happen with integrity. We spend our time on the issues that users want and need; not just items that look good in an advertisement. In the crazy world of DAW software, it is nice to know that a sane option exists for your business or personal use.

3.2. Operational Differences from Other DAWs

If you have some background in audio, but are new to computer recording, then you'll find Mixbus very easy to learn and navigate. If you are coming to Mixbus from another DAW, here are some of the conventions that might seem different to you:

Console-like signal flow

Most DAWs allow the user to create tracks and buses, but the user is expected to assemble the tracks and buses into a sensible signal flow. To do this, you create one or more buses as-needed, and then assign the track output(s) to the bus inputs. If you later realize that you want another track assigned to that bus, you'll have to make that connection; and at the end of the day you might have lots of nearly-invisible and arbitrarily-ordered connections all over the place!

Mixbus provides a fixed signal flow, modeled after an analog console. You don't have to "choose" how many buses you will need, or where they are assigned. This provides several benefits. The bus assignments/ sends are displayed in a fixed order, so you can quickly determine what channels are feeding the reverb bus, for example. Additionally, the assignment and/or send-level to a bus can be automated.

Flexible signal flow in the channel makes lots of tasks easy!

Mixbus allows you to drag&drop the order of the plugins in the signal flow. This includes the channelstrip inline EQ, compressor, and *fader* as separate elements. If you'd like to add a reverb "post fader" (so the reverb continues to ring-out when you turn down the volume of the instrument…), you can do that directly in the mixer strip. This saves you from creating a "bus" for adding effects to a single track.

Advanced users can right-click on a plugin and choose "pin connections" …. this opens up additional possibilities (depending on plugin capabilities) such as sidechaining, mono->stereo effects, mid-side, or similar tasks. This is another way that Mixbus can save you from adding additional "aux buses", with all of their complexity and overhead, that is normally required in other DAWs.

Plug-ins are encouraged, but are not considered part of the core workflow.

Most other DAWs serve as a "container for plug-ins". They provide little or no signal processing or bussing, by default. They expect third parties to develop the actual DSP processing.

Mixbus is different! Harrison provides our own high-quality DSP processing for the channel strip EQ and compressor. We've tweaked the various EQ, compressor, saturation and limiter stages to complement each other. You may still add plug-ins if you wish, but it is no longer necessary to purchase, and choose between, multiple plug-ins each time you want to change your sound. You can make a great mix without any plug-ins at all!

Regions may be stacked in layers

Each audio “region” (or “clip”) is a discrete sound that exists on the timeline track, and it may have other regions stacked above or below it – *on the same track*. Deleting the topmost region layer does not create a hole; instead, you will see&hear the audio that was hidden underneath it. You might think of this like the layers of an image editing program. Like an image editor, it is even possible to make regions “transparent” so that the regions underneath it are still visible (or in this case, *audible*).

This provides 2 benefits. Firstly, it allows you to record an overdub, or part replacement, on top of an existing track. You can then move, trim & fade (or even delete) the replacement part without having to worry about the audio layer(s) underneath it. Secondly, you can use “transparency” to stack lots of sounds (such as bird chirps) on a single track, and effect them together, without having to allocate multiple tracks so each individual sounds is heard.

Region Fades are Crossfades

The start and end of each region has a “fade”, which gradually raises or lowers the volume. When you listen to the region by itself on a track, you hear it fade from silence into the audio, and then back to silence. But if the region is on top of another region, then something even more interesting happens: you will hear a ***crossfade*** from the underlying region, into the top region, and then back to the underlying region. This allows you to stack regions on top of one (or more!) regions underneath it, and set your crossfade to match perfectly. Later, even if the underlying audio is changed, you’ll still preserve the fade of the topmost region.

Because digital audio streams don’t like abrupt changes in the audio stream, EVERY region has a start and end fade. Mixbus’s default fades make it easier to edit with confidence that you won’t introduce a click.

Less emphasis on Tools, more emphasis on Actions

Most DAWs have a set of “tools” (or “mouse modes”) which modify how the mouse behaves. For example, you select the “scissor” tool to make cuts, and the “trim” tool to trim region ends. In Mixbus, we have very few tools. You can do most of your work using the “Grabber” (hand) tool. Point where you want to perform the action, and then use the keyboard shortcut to initiate an action such as “S” for split, or “J” to trim a region’s start (you can also grab the edge of the region to trim it, but the keyboard actions are faster).

For more helpful details about Mixbus’s operation, please visit: Mixbus Conventions & Tips in the Getting Started section.

4. Installation

Platform-specific information

Mixbus works on 3 different desktop operating systems (Mac, Windows, and Linux). Each system has its own conventions for installation. Rather than force the user to learn an unconventional installation procedure, we have made Mixbus follow the conventions of the OS you are using.

- On Mac OSX, Mixbus is provided as an application “bundle” inside a DMG file. Just open the bundle, and drag Mixbus to your Applications folder.
- On Windows, Mixbus is provided with a step-by-step installer that prompts you for the installation location and then installs Mixbus.
- On Linux, Mixbus is provided as a zip file that, when unzipped, provides an installation script.

The following pages explain the details for installing Mixbus on various operating systems:

- Windows
- OS X
- Linux

When you first run Mixbus, you will be prompted to enter the license for your purchase. You can learn more about that here: License Installation

System Requirements

Mixbus runs on nearly any desktop or laptop computer running Windows, Mac, or Linux. There is a free “demo” version to see if Mixbus works with your setup.

Mac OS:

- OS version: 10.6.8 or newer, v5.3 and newer includes 10.15 (Catalina).
- Any CoreAudio-supported audio device.
- Supported plugin formats: AudioUnit, VST, VST3.

Windows:

- OS version: Windows XP or newer.
- ASIO audio devices are preferred for high performance. But Mixbus can also use built-in soundcards (choose the MME driver).
- Supported plugin formats: VST, VST3

Linux:

- Kernel version 2.6 or higher. An audio-based distribution such as AVLinux, UbuntuStudio, or CCRMA is recommended.
- ALSA or JACK-supported audio device.

- Supported plugins: VST, VST3

All platforms:

- 2+ cpu cores/processors are highly recommended.
- 2Gig or more RAM is highly recommended (or much more, if you use sample-based virtual instruments).
- A 3-button mouse with scrollwheel is highly recommended. (see note about Mac touchpads and magic-mouse)
- A full-size keyboard with number pad is highly recommended.
- 250 MB available hard disk space (for program installation only).
- For Mixbus: A monitor with a height of 900 pixels (or more) is highly recommended.

4.1. Installation – Windows

Mixbus on Windows uses the Windows standard installation process. Double-click the installer to launch the process, and follow the directions for a “Typical” installation. This will install Mixbus to the default location. As of v4.0, the Mixbus installer is not signed by Microsoft, so you may have to explicitly allow the installer to run. This is normal operation.

Virus Protection Software

If you have real-time virus shield software it might need to be disabled temporarily to allow Mixbus installation. Depending on which virus product you have, it may also put up notices saying the installer is suspicious and it wants to check for viruses. Don’t be alarmed, just allow the Mixbus installer to run and allow your virus software to do its thing.

In Windows 10 you also have Windows Defender protecting your system. When installing you will first get the following:

Click on “More Info”. You will then get this:

Click on “Run Anyway” to install Mixbus.

After the program installation, complete the installation process by installing your license files, see Install the Mixbus License file

> ! If you are upgrading from an earlier version of Mixbus (version2.x or 3.x) it is advisable to keep your previous version of Mixbus installed on your computer in the event you need to revisit old sessions. It is possible that opening a session created with a previous version of Mixbus may lose some functionality.

> ! Windows systems, and particularly laptop systems, often require some system optimizations to operate without clicks and pops. Most importantly, you should visit the System->Advanced settings dialog, and tell Windows to optimize for “background tasks”. This will dramatically improve the stability of audio I/O. For laptops, we highly encourage you to leave your laptop plugged-in, and select the high-performance CPU mode when it is

plugged-in. There are many other optimizations (such as disabling virus-checks) that may be beneficial. You can find some of these choices here: optimizing windows for audio recording.

A note about “Audio Setup” on Windows:

On Windows, Mixbus will detect sound cards with a MME, WASAPI, or ASIO driver. This means that you will likely see sound cards appear multiple times in the audio setup menus. If provided, the ASIO driver option is preferable for stable, low-latency operation.

Many built-in desktop sound devices will not provide an “Input device” if a microphone is not physically plugged in. If you plan to record with Mixbus, you must plug a device into the sound card’s input before you launch the audio engine. If you would like to use Mixbus without a capture device, choose “None” for the Input device. Mixbus does not provide a “None” option for the playback device.

When you select an ASIO device, the “Buffer size” setting will be automatically changed to the current preferred buffer size of your ASIO device (normally set in the device’s manufacturer-supplied control panel). It is important to use the device’s preferred buffer size. Otherwise you might incur a “buffer size adaptation” process which reduces efficiency and may affect latency (delay) through Mixbus.

Non-ASIO devices will default to a buffer size of 1024 for stable operation. Changing this setting is possible but is unlikely to have much effect on the performance because of the Windows sound infrastructure.

A note about VST plug-ins on Windows:

After the initial installation, Mixbus will not have any knowledge about your third-party VST plugins. You must add the path(s) to your VST plugins in the Edit->Preferences->Plug-in tab, and then initiate a “Scan”. For more information about plugins, click here.

Continue to Download and Install the License File

Uninstalling Mixbus

You can use the uninstaller provided in “ Start Menu -> Programs -> Mixbus”, OR you can uninstall the program from the Windows “Control Panel” menu.

Remember you can have multiple versions of Mixbus installed on your computer at the same time.

4.2. Installation – OS X

Installing Mixbus:

Mixbus provides 2 packages for Mac OSX:

64 bit (for Intel Mac computers running 10.7 or newer)
32 bit (for Intel Mac computers running 10.6.8)

> ! If you are upgrading from an earlier version of Mixbus (v2.x or v3.x) it is advisable to keep your previous version of Mixbus installed on your computer in the event you need to revisit old sessions. It is possible that opening a session created with a previous version of Mixbus may lose some functionality.

To install Mixbus, download and double-click the .dmg archive (Mixbus_Intel_4.0.dmg or similar). Drag the Mixbus icon to your Applications folder (or anywhere that you like to store your applications; Mixbus doesn't care which folder it is started from). If you would like Mixbus to be shown in the Dock, then you should drag Mixbus from your selected location to the Dock.

We recommend that you temporarily disable all anti virus software when installing Mixbus. Don't forget to re-enable your antivirus software after installation.

> ! As of v4.0 Mixbus is not digitally signed by Apple, or sold on the Apple App Store. On Mavericks and newer, you may have to **right-click** on Mixbus and choose "Open" the first time you launch it.

> ! Do NOT drag Mixbus directly from the .dmg archive to the Dock; this would create a shortcut to the Mixbus application inside the DMG file. If you do this, then when you click Mixbus in the dock, OS X will have to extract and mount the .dmg file before launching Mixbus. This will make Mixbus run very slowly.

Audio Input and Output:

Mixbus can use any CoreAudio-supported interface. Many newer soundcards work without a dedicated driver; but some older interfaces (such as the Digi 00x series) may require a special driver from the manufacturer before they appear in the list of CoreAudio devices.

Newer Mac computers have separately-clocked input (mic/line-in) and output (headphone/line-out) devices. Prior versions of Mixbus required that you create an "Aggregate" device to combine the internal soundcard input and output into a single device. Mixbus v3, and newer, provide a method to select separate input and output devices without making an aggregate device; Mac will create an invisible "aggregate" for you. When

possible, it is always best to use a professional I/O card to avoid any internal sample-rate conversion that may be performed by OSX.

You can combine multiple devices into an “Aggregate” device which provides all the I/O as if it were a single device.. For example, you can “aggregate” 2 different 8-channel devices so that you can record 16 channels into Mixbus. In a few cases, we have found that troublesome devices that don’t work well in Mixbus can be improved by creating an “aggregate” device, selecting the device’s input and output for the aggregate. This provides an extra level of separation from the device’s driver into Mixbus. To create an “Aggregate” device, consult the Apple documentation for your version of OSX.

A note about AU plug-ins on Mac OSX:

After the initial installation, Mixbus will not have any knowledge about your third-party AU plugins. You must visit Edit->Preferences->Plugins to scan your AU plugins and make them available in Mixbus. For more information about plugins, click here.

To finish your installation, please continue to Download and Install the License File

4.3. Installation – Linux

A note about Linux distributions (distros): While an expert system administrator can adapt nearly any distro for use with Mixbus, it is easier for most users to start with a distro that is designed for use in high-performance audio.

Some distro's are designed for audio (such as Ubuntu Studio). A further list of distros is at: http://www.linux-sound.org/distro.html. It is beyond the scope of this document to compare these distros.

Sound card (I/O) support:

Support for professional sound I/O is limited on Linux. However there are some excellent options available at various price ranges. Normally, no driver or configuration is needed to use a sound card if it is supported by your distro. It is important to choose a distro that has good pro-audio support.

Unfortunately, support must be determined on a per-device basis. The ALSA project provides a fairly comprehensive list of USB, PCI, and built-in devices. There is a separate list of FireWire devices. When in doubt, check with your distribution's documentation, user forums, or support.

Downloading and Installing Mixbus:

If you are upgrading from an earlier version of Mixbus (v2.x or v3.x) it is advisable to keep your previous version of Mixbus installed on your computer in the event you need to revisit old sessions. It is possible that opening a session created with a previous version of Mixbus may lose some functionality.

After your purchase, you will receive a link to download the installation package. You must first unzip the package using your desktop's archive utility.

Once unzipped, a folder will appear with the installer (Mixbus_something.run). On most systems, you can double-click the .run file in your file browser. If that doesn't work, then you can launch Terminal, navigate to the download directory, and run the installer manually by typing: ./Mixbus_(whatever).run

The installer will prompt you to uninstall previous versions. The installer will check the suitability of your system for real time audio, and report any problems that it finds.

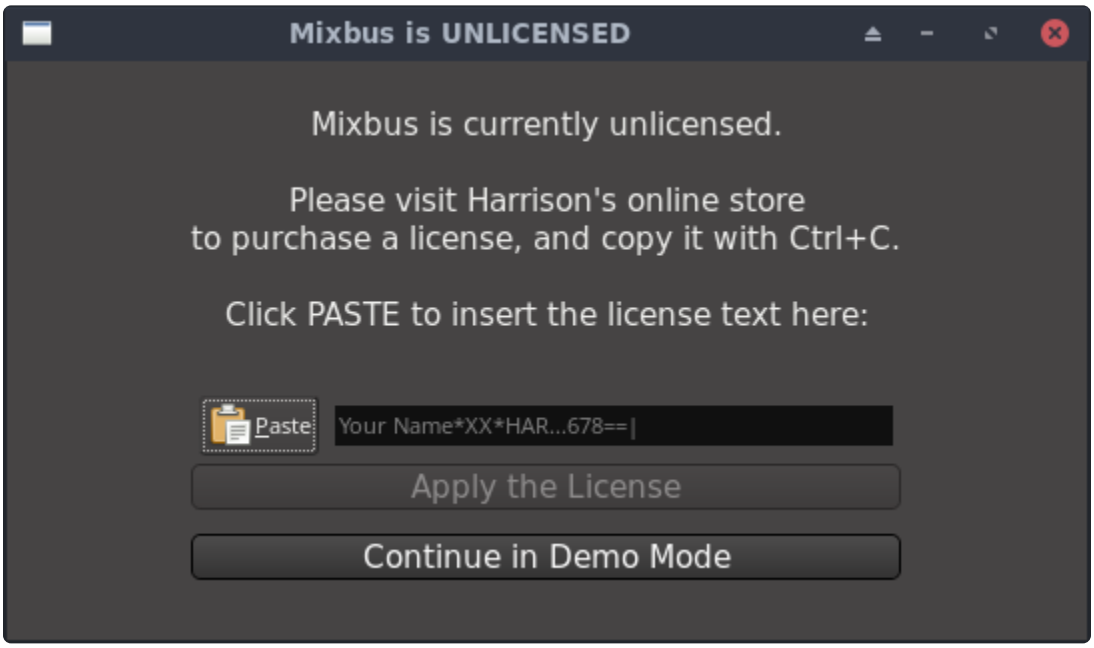
Continue to Download and Install the License File

4.4. License Installation – All Platforms

License Installation

> ! Our licensing method has changed as of version **5.3**. Please pay careful attention to the instructions below.

At startup, Mixbus checks to see if a license is installed. If no license has been installed, you will see the following “UNLICENSED” dialog:

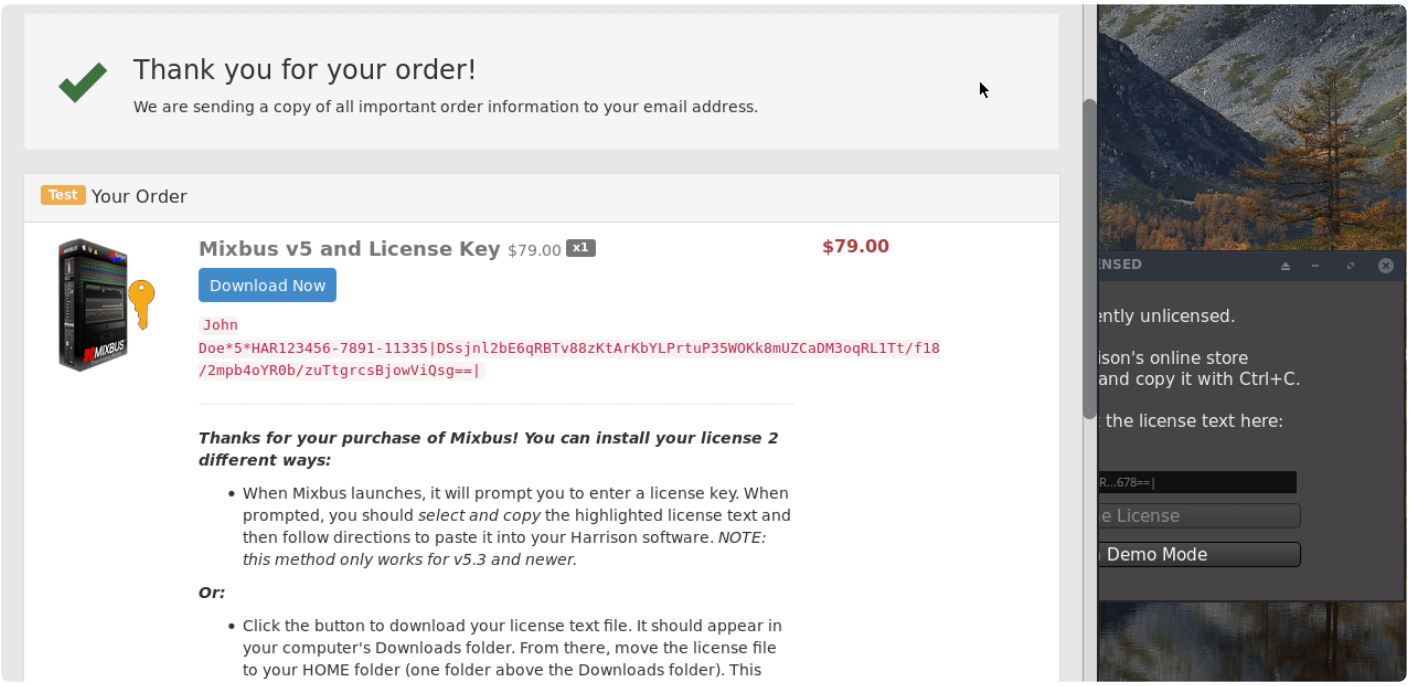


The dialog provides 2 options:

Paste and Apply License

After purchasing Mixbus you will receive, both on the final page of the checkout process, and in your email, a cryptographically signed license generated by us. This is the license that unlocks the full program. To use this license:

1. Highlight the license text from beginning to end and then copy it.
2. Click the **Paste** button in the dialog.
3. Mixbus will verify the license’s signature and then allow you to apply the license to the program.
4. Click the **Apply** button.



Your license has your name encrypted in it so you should never share it with anybody!

Continue in ‘Demo Mode’

Upon pressing this, Mixbus will immediately launch your web browser and prompt you to buy a license. However it will only do this once per program run and you can close your browser tab if you choose. Mixbus will then continue loading in Demo Mode and allow you to use unlimited features. However, there is an occasional, quiet “hiss” in the audio.

—

Alternative methods for licensing

After purchasing Mixbus you will receive, both on the final page of the checkout process, and in your email, a link to download your license generated by us. This link is good for **seven days** after your purchase date. Once you’ve downloaded this text file, you may keep a copy for your own use.

1. Download the license anywhere on your system.
2. Move the license file to your computer’s HOME folder.
3. Start the program.

Most licensing problems we encounter involve the user putting their license files in the wrong place. The following should help determine the correct folder.

Alternative locations for the Mixbus license file:

On MacOS,Mixbus searches the following locations (first found is used):

- $HOME/Library/Application Support/harrisonconsoles/
- $HOME/.
- /usr/local/share/ (intended for multi-user institution-wide licenses)

On Windows, Mixbus searches the following locations (first found is used):
You can type these in the address path bar of most modern versions of Explorer.
Simply drag your license file(s) into one of these folders.

- $HOMEDRIVE$HOMEPATH
- localappdata – typically C:\Users\NAME\AppData\Local\
- userprofile – typically C:\Users\NAME\
- allusersprofile + \harrisonconsoles\ – typically C:\ProgramData\harrisonconsoles\

The correct location for the license files is dependent on how your computer is set up. Typically in Windows you can do the following (this works most of the time):

- open your C:\ folder
- open the USERS folder
- In the USERS folder you will see a folder with your computer's name, or your name, or some name that was chosen when your computer was set up.
  Open this folder and drag or copy your license files into it.

Another way is to use the Window -> Licence Manager pulldown in Mixbus, then click on the "Open Home Folder" button. This will open a window to the folder where Mixbus thinks your licenses should be. Drag or copy your license files into it.

On Linux, Mixbus searches the following locations (first found is used)

- $HOME/.
- $XDG_CONFIG_HOME/harrisonconsoles/
- $HOME/.config/harrisonconsoles/
- /usr/local/share/ (intended for multi-user institution-wide licenses)

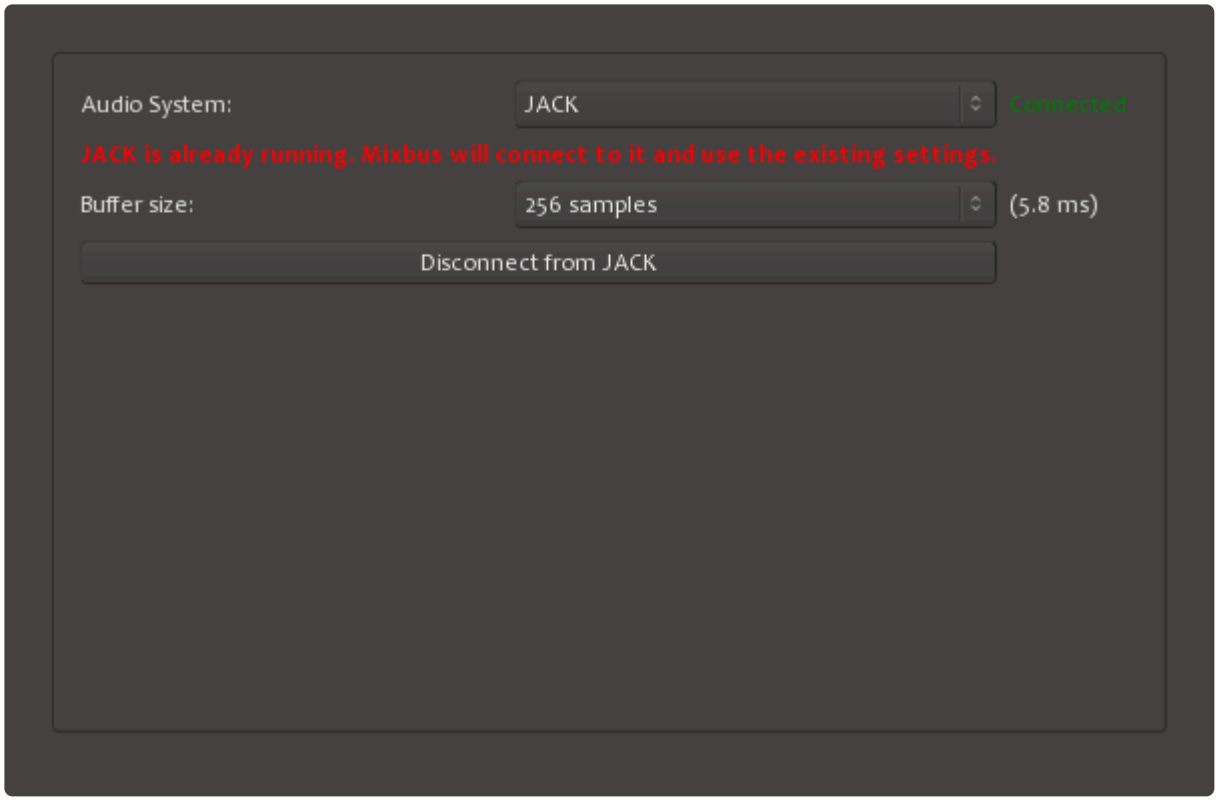
4.5. Using Mixbus with JACK (optional)

JACK is a system service that allows for the interconnection of audio between applications. Early versions of Mixbus required that JACK be installed on your system. **Mixbus no longer requires or expects JACK to be installed**. Mixbus uses your sound card directly via ASIO (Windows), Core Audio (Mac) or ALSA (Linux).

If you would like to interconnect the audio of different applications into Mixbus, then you can download and install JACK for your system. JACK is not a Harrison product, and we cannot provide email support for JACK users. However you will find a lot of JACK users on our forum, because JACK is a powerful utility.

JACK creates virtual audio and MIDI ports that can be connected and shared between nearly any application on your system. If the application is JACK-aware (i.e. it was compiled with the JACK libraries) then it will have some additional features such as sample-accurate transport control and sync. There are many third-party apps that can control JACK connections, save and restore your connections, or even route the audio or MIDI using Ethernet. Mixbus is fully JACK-enabled, and it can save all the JACK connections that are made to it. Mixbus can also use JACK as a sync source, or it can be the master sync source.

JACK control from within Mixbus is straightforward:



When you select JACK as the Audio System, there is only one parameter: the buffer size option. Buffer size allows you to adjust the latency of the recording system. The 256 samples shown here is equivalent to 5.8ms, which creates less latency that having a sound source 6’ away from you (sound takes about 6.6ms to travel 6’). You can also disconnect from JACK from this window.

A full description of JACK timecode sync is beyond the scope of this manual. You can learn more about JACK at JackAudio.org

4.6. CPU Usage

Why does Mixbus require more CPU usage than my other DAW?

Because Mixbus is emulating the operation of an “analog console”, it requires significantly more CPU resources than a typical DAW. Mixbus’s EQs, compressors, and other features are always active. The benefit of this operation is that you can always confidently enable the channelstrip features without worry of overloading your system. Once the mixer is running, it does not increase the load when you operate the controls.

Although Mixbus requires a high CPU usage by default, it is actually quite efficient compared to other DAWs, once you’ve added plugins equivalent to the high-quality EQ, compression, bussing, tape saturation, limiting, metering, and the other built-in features of Mixbus.

How do I increase the number of tracks or plugins that I can use?

- Increase the “buffer size” in the Audio Setup dialog
- On Windows and Linux, you must take steps to optimize your system for audio (see our Installation section)
- Use high quality audio devices and drivers
- And of course using a faster computer, with more CPU cores, is desirable.

General comments about computer audio performance:

The CPU meter on your computer averages the cpu usage over a very long period (perhaps one second). In digital audio, the timing is much more sensitive. When the soundcard passes us a buffer, then we have to wake up, process the audio buffer, and return it to the soundcard before the soundcard needs to play it out. If we don’t wake up in time, or we don’t get finished in time, then you hear a “click” caused by the dropout (we call these xruns, short for over-run or under-run).

In the case of a 1024 buffer size, this has to be done within (1024/44100) 25ms, or about 1/40th of a second.

This is complicated by the fact that it’s the OS’s job to “wake us up” and tell us that the soundcard has some data for us. Some OS’s and drivers are better at this than others. But let’s say it takes 5ms before we are even alerted that some data is available. We now have 20ms left to process the audio. If it takes us 10ms to process the audio, then we are finished within 50% of the allotted time and that’s what we display in the meter. This is a very accurate indication of your computer’s ability to process audio.

This also explains why the selected buffersize is so critical to the DSP load. A 256-sample buffersize is about 6ms, so if your computer sometimes waits 5ms after receiving the soundcard interrupt before it “wakes us up”, we only have 1ms remaining to do our work.

For more information on factors impacting the performance of your computer please see this video:

CPU Performance vs. Real-Time Performance in Digital Audio Workstations

5. Getting Started

Getting Started (Overview)

Mixbus is an editor that works on a document format called a **Session**. A session is a folder on your hard drive that stores all of your recordings, edits, and mixer settings together. Unlike a word processor, Mixbus does not allow you to start an empty document and then “save” it later. Instead, when you first launch Mixbus, it will guide you through creating or selecting a session before you can start to record, import, edit or mix your project.

A session is a collection of **tracks**. Use the “Add Track/Bus” dialog which can be found in the Session and Track menus. You can also create tracks by right clicking in the gray track area under the track headers, or by right-clicking in empty mixer-window space, or by clicking the blue “+” button in the top of the mixer window’s left-side panel.

Alternatively, you can create new tracks by **Importing** existing audio from the Session-Import menu.

If you find yourself starting the same way for each session, you can create a session **template** which will be available as a starting point next time you start a new session. You’ll find the list of available templates when you create a new session. Click Session Templates to learn more.

You can save multiple **snapshots** of your settings inside that session. **Snapshots** conveniently save your entire editing/mixing session for later recall. Saving a snapshot is instantaneous and takes up essentially no extra disk space (snapshots all share the same audio data in the session folder). Click Snapshots to learn more.

If you make a mistake while recording or editing, you can click “Undo” to revert your changes. Only a few editing operations (such as adding or removing a track) cannot be undone. One of the cool features of Mixbus is the fact that your “Undo” history for each session snapshot is stored when you save the session; This means that you can return to a session after a few days, and still undo the last changes you made, if you decide that they weren’t a good idea.

When you are finished, you can **Export** your session to an audio file using the Session->Export menu item. The exported file can be shared online or burned to a CD.

> ✻ Mixbus uses the **right mouse button** for many features. Try right-clicking on tracks, regions, mute buttons, solo buttons, group panes, regions, selected ranges, markers, mixer redirects, clocks, the region list, and the marker bar area. Each will launch a context menu that provides many features.

Need Help?

Getting started with a new DAW can be a challenging task. While Mixbus shares conventions with many

other popular DAWs, it can still be a little difficult at the beginning. Rest assured that it is worth your time and effort to learn Mixbus. For technical issues, bug reports, and feature requests, email support is available from Harrison Consoles: mixbus@harrisonconsoles.com.

If you'd like to interact with other users of Mixbus, you can visit our forum:

http://harrisonconsoles.com/mixbus/forum/

> **"Once you have reached the level of a Harrison Console, you are ready for new flights of engineering imagination. Unlimited. Unhaltered. Uninhibited by the bounds of technical obstructions. YOU BECOME THE STUDIO. You are the living extension of the electronic link between the perfect sound and the perfect master. You are the ultimate conductor in the universe of recorded music"** – Harrison 4032 brochure, circa 1975

5.1. Mixbus Conventions & Tips

General Conventions

Over 40 years, Harrison developed a methodology for our user interfaces. We aim to make the easy things very easy, while keeping the hard things possible. If you are going to access a feature hundreds of times each day (like: boost the bass, or start playback), then we assign it a very prominent space. If you use a feature only a few times a day, then we make it slightly less prominent: for example you might have to right-click to access it. And if it's a feature that you only access once per week or once a year, then you might have to open a preferences dialog, or even edit a text file somewhere.

The following tips will help you navigate Mixbus fluidly. These tips generally apply throughout Mixbus and may not be documented in the individual sections where they apply. New users should learn these conventions quickly, in order to avoid developing "bad habits" that impede fast operation.

> ✻ Advanced Users: In addition to this information, advanced users might want to visit Operational Differences and Power User Topics. If you are experienced with several other DAWs and audio engineering conventions, these 3 pages will likely provide you with everything you need to master Mixbus in just a few hours.

Menus and keyboard actions:

Mixbus follows your OS's convention for the main menu. On Mac OSX, the main menu will appear at the top of your monitor. On Windows, the main menu will appear on the top of the main window.

Every menu item can have a keyboard shortcut assigned to it; you will see the assigned keyboard shortcut next to each menu item, if one is assigned. You can change the keyboard shortcuts, but if you are a new user we suggest that you learn the existing shortcuts before making changes. The existing shorcuts have been painstakingly developed to be easily learned, be ergonomic, and provide access to the most-used features of Mixbus. One exception to this rule: if you are using a non-US, QWERY keyboard, then you might find the need to remap some actions for your keyboard layout.

Keyboard shortcuts are MUCH faster than accessing the menu item; so if you find yourself using a menu item frequently, you might consider memorizing the action.

Selections:

- A Mixbus "selection" can be one of:
  - One or more Tracks (or their equivalent Mixer Strips, in the mixer)
    - Optionally in addition to the track selection, there can be a "time" selection on those tracks.
  - One or more Objects on the timeline (Regions and/or Automation Control Points)
  - One or more MIDI notes inside a MIDI region

- One or more plugins in a mixer strip

However, these selections are **mutually exclusive**: you can select several tracks, but you can't select a track *and* a region. This avoids some common causes of confusion. For example, if you have selected a region in the editor, and a plugin in the mixer, then what should happen when you press "Delete"? Mixbus prevents you from selecting unlike items to avoid this possibility. The selection might change when you select tools. For example if you have a Range selection with the range tool, and then you switch to the "Object" tool, the range selection will disappear. (in the very rare case when you didn't want this, there is an "undo selection change" in the Edit menu)

In almost every case, a selection will be identified with a "red border" around the selected items. And in most cases the items will change to a red color to indicate selection.

✻ The **ESC (Escape) key** will clear the current selection, so there will be no selected tracks, regions, notes, or automation control points.

Mouse wheel:

Mixbus uses the mouse-wheel (or 2-finger drag, on a touchpad) extensively. Here are just a few of the things you can do with the mouse wheel:

- Zooming the editor canvas timeline by mouse-wheeling in the ruler-bar, or anywhere by holding Ctrl. ("Cmd" on a Mac).
- Adjusting knob and fader values.
- Adjust the metronome volume, by mouse-wheeling on the metronome button.
- Nudge the playhead forward & backwards in the Mini-Timeline.
- Advancing a clock value (such as bars) by mouse-wheeling over the appropriate number in the clock.

Right-click:

Mixbus uses the right mouse button extensively. Nearly every item has a context menu that can be accessed with a right-click. Try right-clicking on tracks, regions, mute buttons, solo buttons, group panes, regions, selected ranges, markers, mixer redirects, clocks, the region list, and the marker bar area.

Right-click actions are those items that aren't necessary to see hundreds of times a day, but perhaps dozens of times in a day. So for example the Solo button is activated with a click; but if you want to "isolate" the solo (so the channel always plays even if other channels are solo'ed), then the Solo Isolate function is accessed from the right-click menu on the Solo button.

! macOS users: make sure to enable the "secondary click" functionality of your touchpad, in System Preferences, so you can access the right-click functions. Cmd+Click is NOT a right-click!

Shift+Right-click: (Delete shortcut)

Shift+Right-click is a shortcut to Delete objects. Some examples include regions, plugins, and automation control points. Holding shift while right-clicking on one of those objects will delete it.

Shift+click on mixer controls: (Group Override shortcut)

When two or more channels are in a “group”, and the group has “gain” enabled, many mixer functions will function together. For example: if you drag a fader, then the grouped channels will all move together. But if you hold Shift when you click the fader, then only the clicked fader will move; not the grouped channels. This is called a *group override*. This also applies to “momentary” mutes and solos as described below.

Shift+Ctrl-click: (applies to ALL tracks)

Holding Shift and Ctrl (Shift+Cmd for Mac users) while clicking on a track button will apply that operation to “all” tracks. Some examples include the Mute, Solo, and Record-arm buttons. Clicking on those buttons with Shift+Ctrl held will apply your click to ALL tracks. This is a fast way to un-solo all tracks, or to rec-arm all the tracks in a session simultaneously. Other functions also respond to Shift+Ctrl+Click: for example if you hold these keys while selecting the “meter point” for a channel, the menu selection will apply to all channels.

Middle-click: (non-latching mute/solo)

Clicking a Mute or Solo button with the middle mouse button will trigger a “non-latching” button push; this means that when you release the mouse button, the solo/mute button will also release. This provides a quick way to temporarily mute or solo a track while you hold the mouse down.

Mixer Knobs and Faders:

- When a knob is in the “default” position, the indicator is very dim. This allows you to scan the mixer for settings that are not at default.
- Mixer strip knobs can be double-clicked to instantly return them to their default value.
- Holding Ctrl/Cmd while turning the knob provides a fine-adjust mode, while Shift+Ctrl/Cmd allows super fine adjustments.
- Hold Shift to override any actions that might be grouped; (see Group Override, above)
- Mixbus knobs and sliders incorporate an “in-on-change” feature. When any knob or fader is clicked and changed, it will automatically enable the control by also setting the “in” or “on” control for that section.

Floating-Point processing:

Mixbus uses a floating-point internal architecture, it is possible to exceed 0 dBFS (decibel full scale i.e: the loudest sound in the digital domain) on the channel and mix bus strip meters without ill effects. For this reason there is no colored indication of overly-loud signals on input and mix bus strips.

Master Bus Volume:

The master bus is not a “volume” knob for your speakers. The master fader level is applied to the exported file; so if you turn it down to make your speakers quieter, you will likely make a very quiet mix, on export.

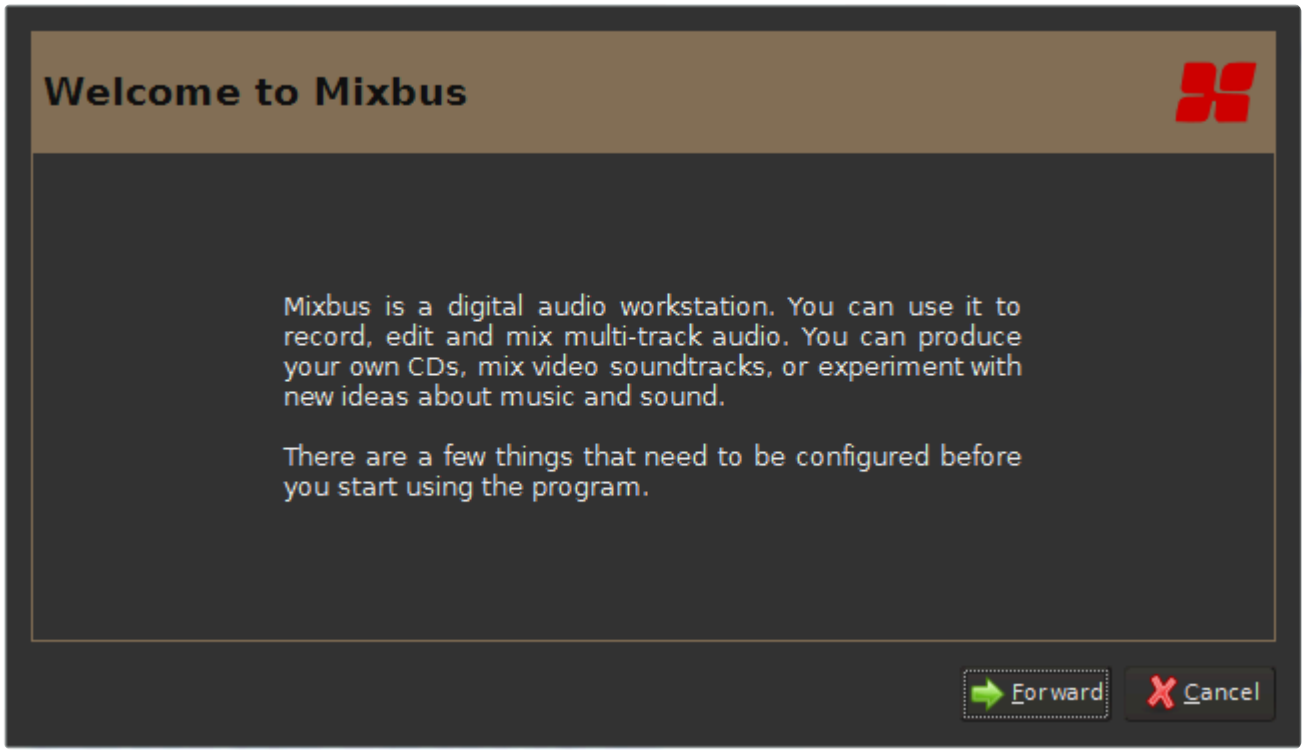
If you are not using the monitoring section built into Mixbus (Session>Properties>Monitoring tab) you will use the volume knob provided by your computer or I/O device to control the loudness of your speakers. If you are using the fader of the Master strip to control speaker volume, remember to set the fader to the correct level before you export your project to a .wav file. Use the K-14 meter to verify that the export volume is at a reasonable level.

> “If you asked a hundred engineers, mixers and producers to define the “perfect console” you would probably get a hundred answers. The answers, however, would all contain the same salient points. The “perfect console” would have totally transparent sound, capturing the true musical quality of all performances. It would have unlimited features, facilities, and functions. It would perform all required tasks with a minimum of operator effort. It would be totally reliable and require no preventive or corrective maintenance…
> You can spend the rest of your life looking for the “perfect console”. You’ll never find it. You will find, however, that we have what you need.”- Harrison MR20 brochure, circa 1982

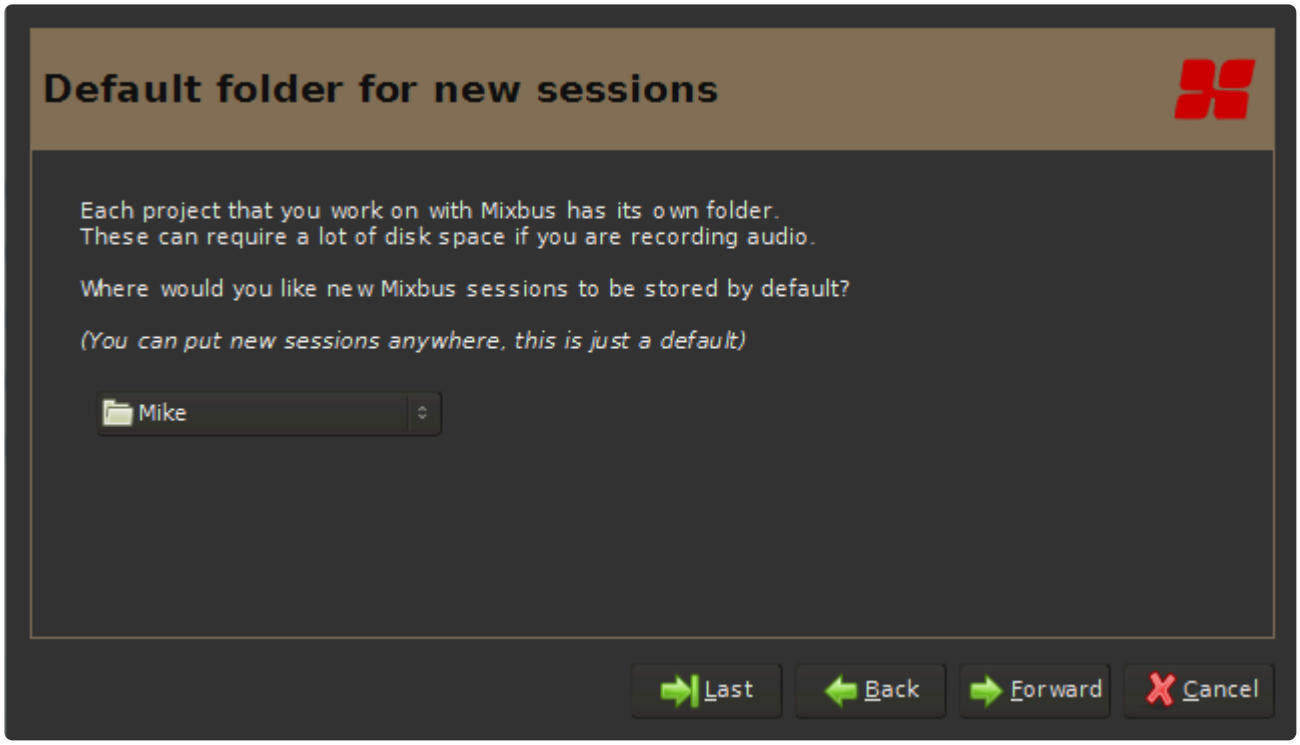
5.2. First Time Configuration

First Time Configuration Dialog boxes:

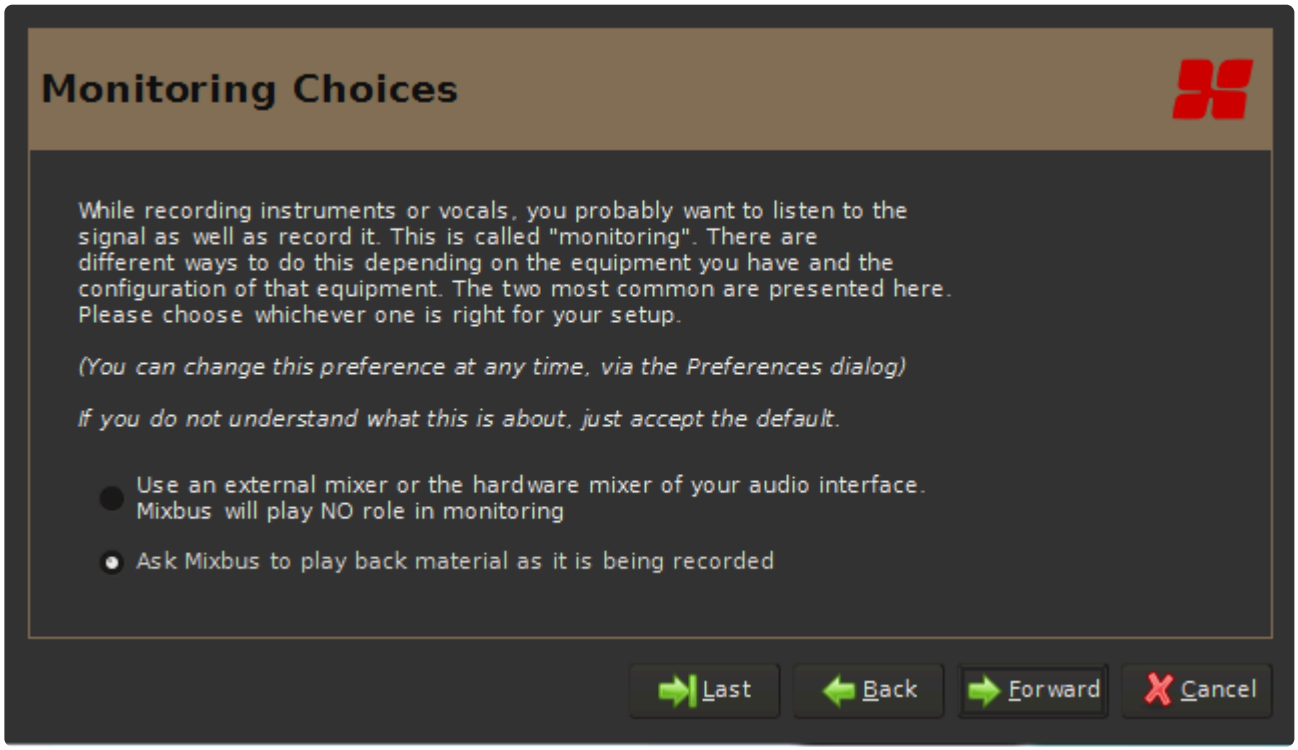
The first time you launch Mixbus on your machine, Mixbus will prompt you for a few basic setup options. You can change these preferences later.



A welcome message will appear. Click the“Forward” button to continue

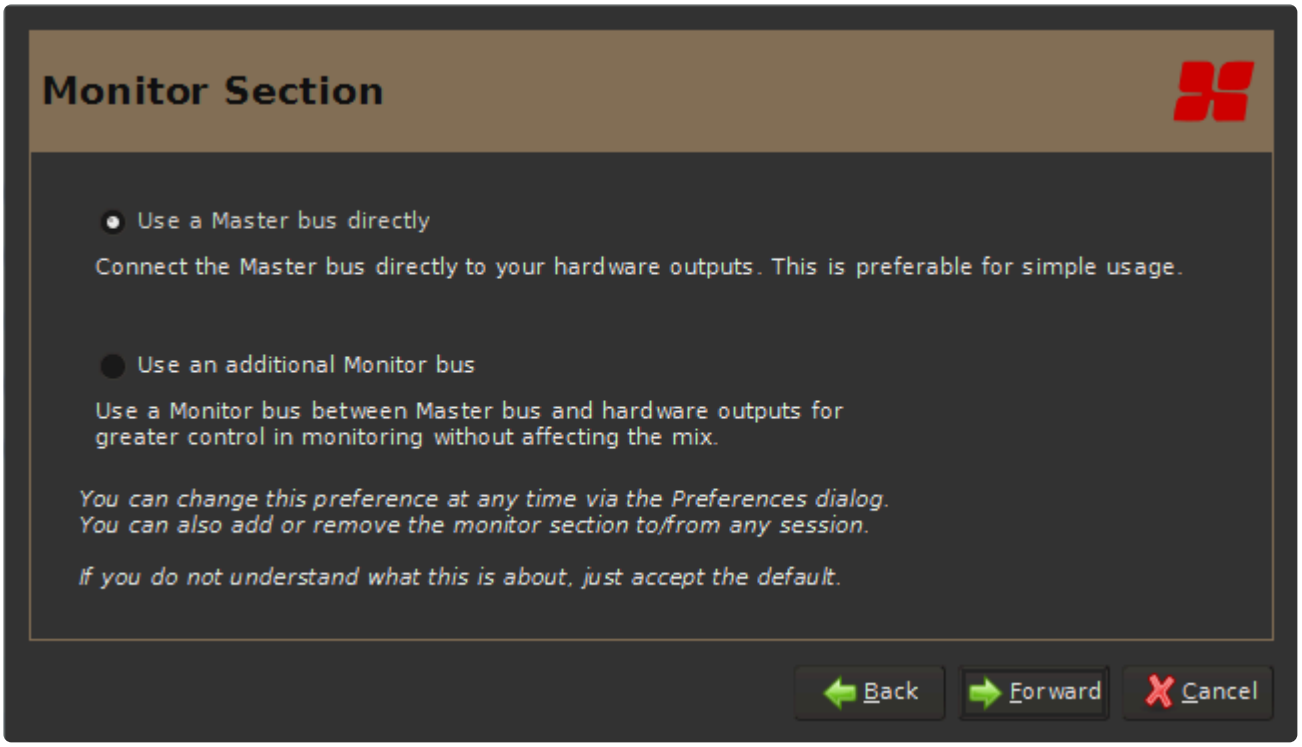


An option to change the default session folder is given. If you have a preferred hard drive you would like to use, the dropdown box will allow you to browse to it. Choosing a drive other than the one that your OS or the Mixbus application resides on may increase performance (this is dependent on your computer configuration).



Some soundcards provide a low-latency mixer interface which can be used to monitor yourself while you are recording. If you want to use your interface to monitor yourself, choose “Use an external mixer or the

hardware mixer of your audio interface". Otherwise choose "Ask Mixbus to play back material as it is being recorded". You can find more information about monitoring, click here



The monitoring section provides an additional volume control between your master mix, and your soundcard. You will want to enable this option if your interface doesn't provide a monitor level knob. It also provides some additional solo-ing features. For more on the monitoring section click here

5.3. Starting a session

What is a Session?

A “session” is the document type that Mixbus opens and edits. When a new project is started Mixbus will create a session folder where it stores all information regarding that particular project. When a session is saved and later opened, Mixbus will reload the saved settings and information so that all work previously done is restored.

A “snapshot” is a copy of the session that you create to save the specific settings of the session. You might create a Session for your band’s live recording, a song, or a compilation of songs for a CD, and have a snapshot for each song or part of a song where the settings change.

Session setup:

3 options are given in the session setup window:

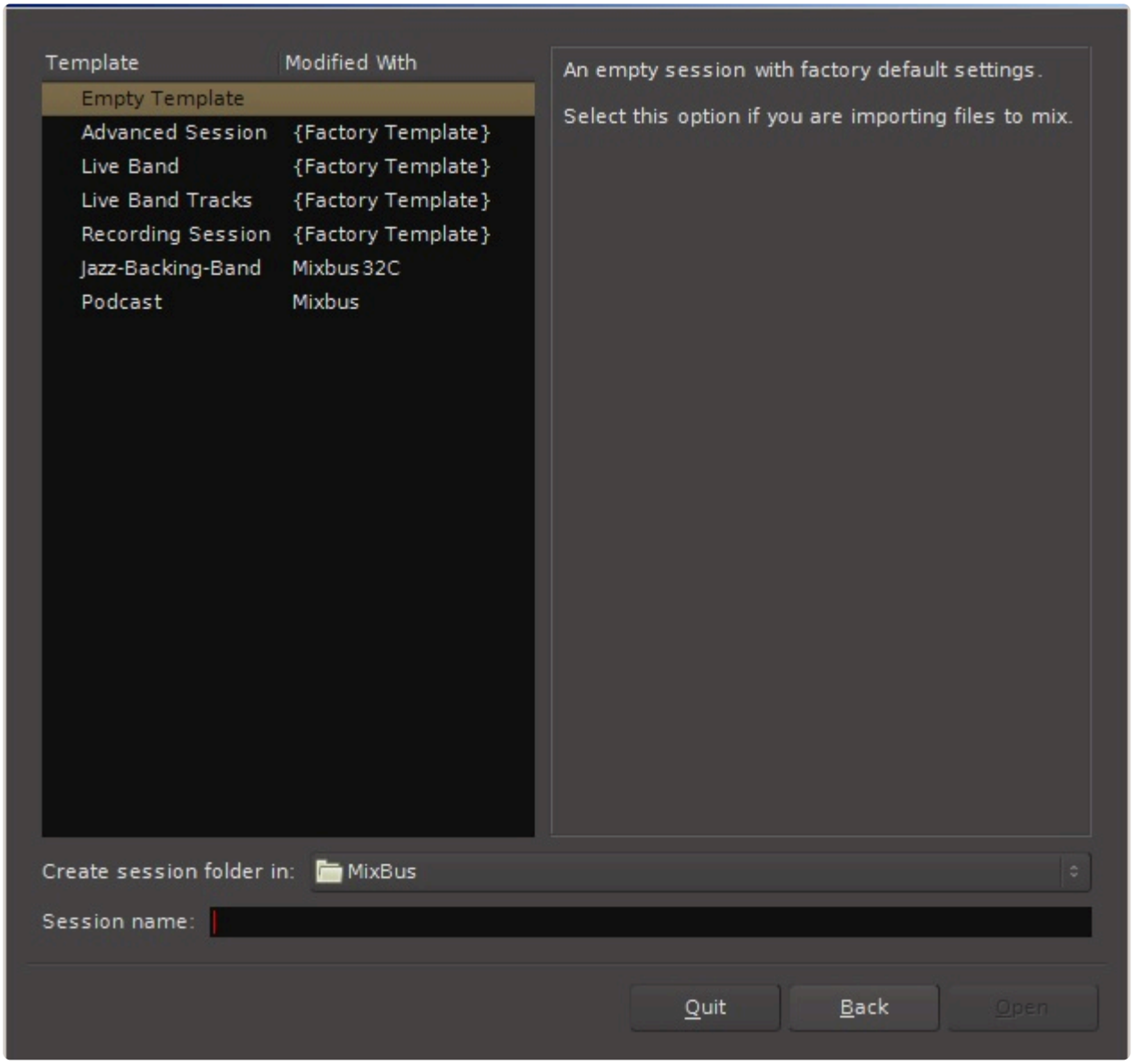
1. Open a recent session:(this list will be blank for first time users). The black window with in the dialog box displays a list of a saved session’s name, sample rate, the program it was created by, and the date it was last modified.

If more than one snapshot is available in the session, an arrow next to that session will appear. Notice the arrows next to the sessions “BCRK140718_raw” and “BCUP110419_1 but no arrow next to “WB090115_M6”. Clicking on the arrow will show all snapshots available in this session, as displayed for “BCRK140718_raw”. In this case the engineer chose to name snapshots with their song names, “easycome” and “milkyway”.

You can click on a column header to sort the session list by date or by name. Click again to reverse the sort order (oldest to newest, or newest to oldest).

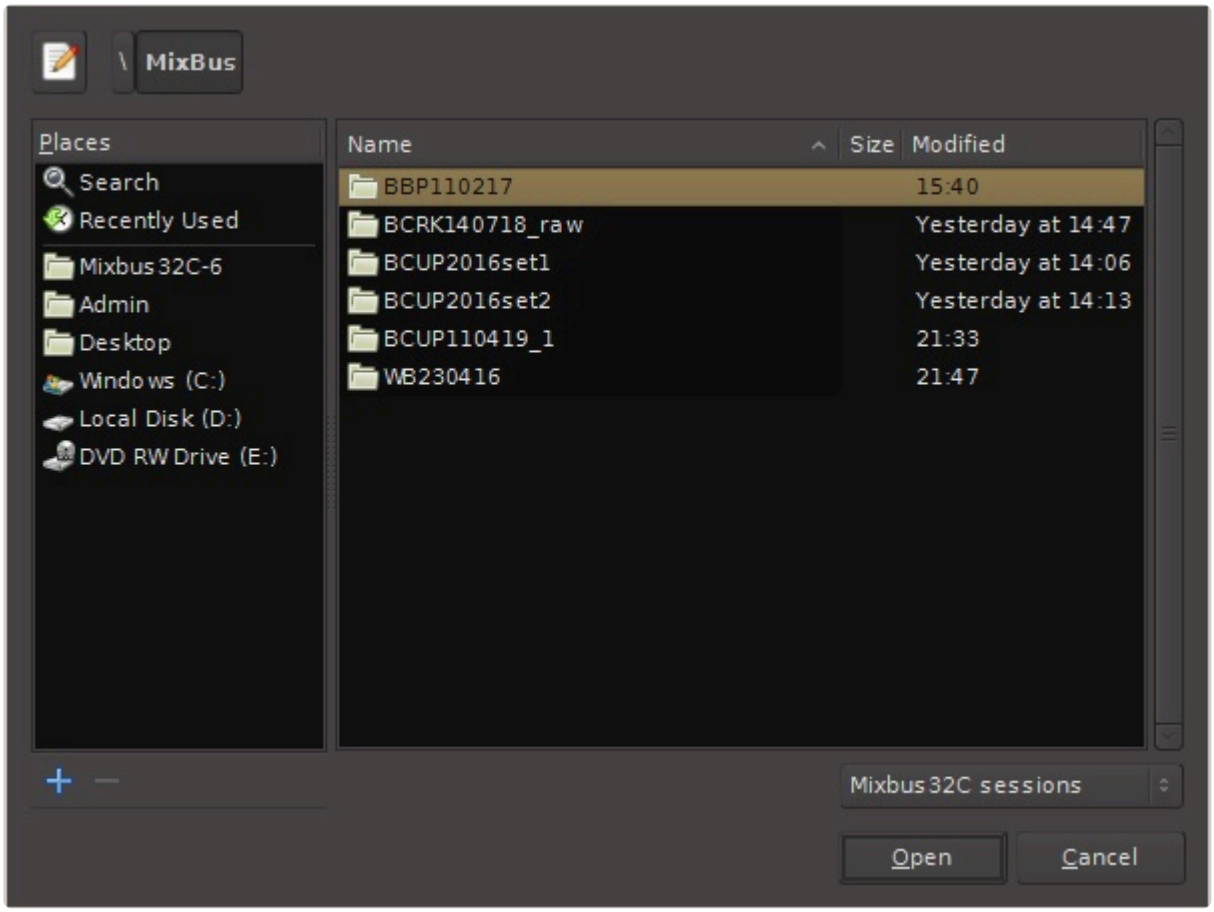
> If you hover the mouse over a session, it will report the full path to the session folder on your system.

2. “Click to create a New session”: When creating a new session a dialog box will allow you to name the session, choose a location, and choose a Template to use for the new session.

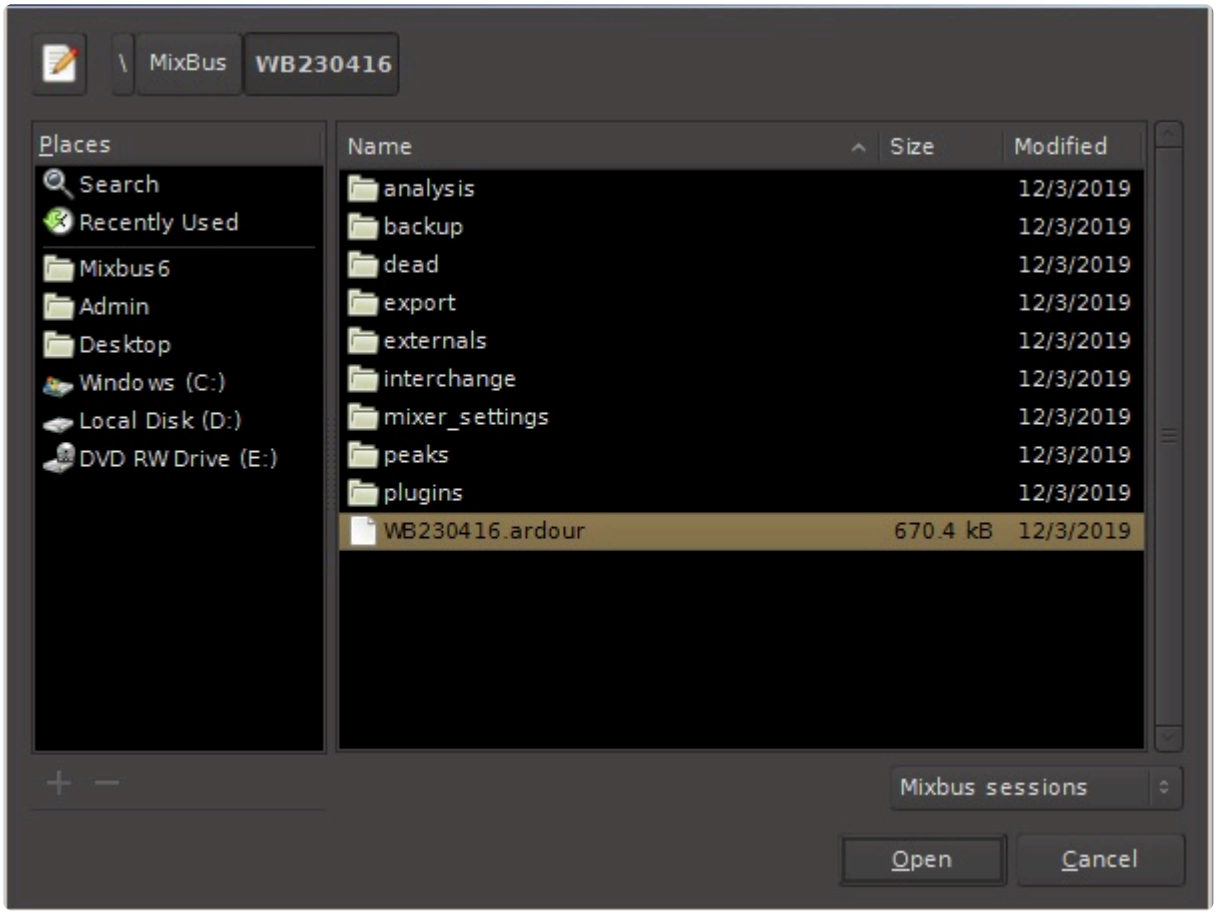


> ! A session's sample rate is determined when it is created. You cannot change the session's rate after it is created. Music projects are typically created at 44.1 kHz, while most audio-for-video projects utilize 48 kHz.

3. "Click to open an Existing session": If a previously created session does not show up in the session window, for example if previous sessions have been stored in a different location from other sessions, you can use the browser function to the left to locate and open an existing session.



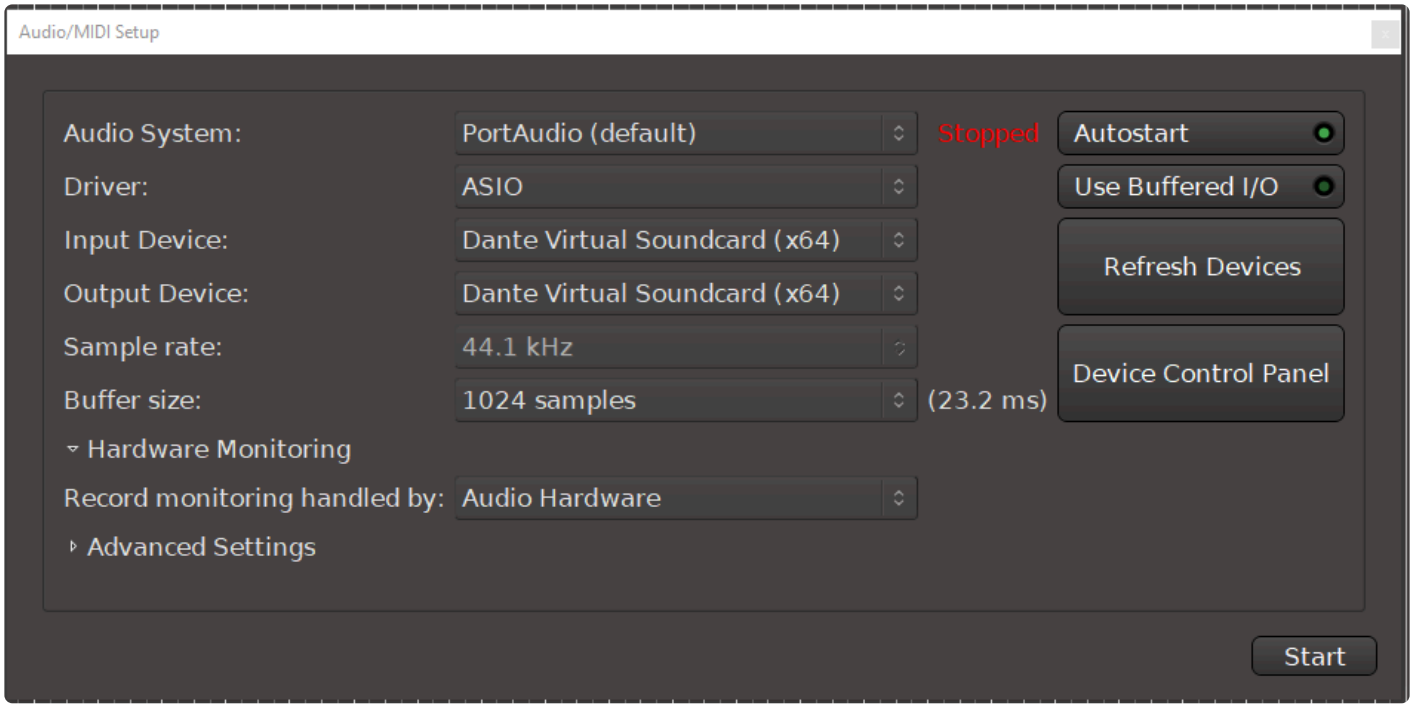
A session file will have a file name suffix of ".ardour". In the image below the session folder "WB230416" is opened. The session file is "WB230416.ardour" and its location as shown by the folder tabs across the top. Select (highlight) the .ardour file and click on the "OPEN" button to open this session.
For more on Session File Management click here.



If you hold Ctrl (Cmd on Mac) when clicking “OK”, the session will be opened in “SAFE” mode. Mixbus will ignore the last-used playhead position and zoom; and no plugins will be loaded. The plugins will appear as placeholders which you can edit or remove, and then re-open the session.

Audio Setup

After selecting a session to open, Mixbus will present you with the Audio/MIDI Setup Dialog.



If you are opening an existing session, the audio system will be configured for that session's sample rate. The other settings (I/O devices, buffer sizes, etc) will be configured with your most-recently used settings.

If you are creating a new session, this is your chance to define the sample rate for the session. For more on the Audio/MIDI setup window click here.

5.3.1. Session File Management

Session Folders

Every session is stored on your disk as a folder. A session folder contains everything pertinent to the session: Audio files (or links to them), snapshot files, the waveform display cache, your exported mixes (unless the default location has been changed) and many other things. Deleting a session's folder will completely remove the session and all of its recorded audio (if any).

> ✻ If you move the session folder to another location, Mixbus will no longer be able to find it, and it won't show your session in the "Recently Opened" list. But you can navigate to the session folder and open one of the snapshots to re-add it to your history.

In the main menu "Session", you will find several options to open a new session; or close, save and copy the currently open session.

New: Calls up the Session Setup dialog and allows a new session to be created. This will save and close any currently open session.

Open: Allows the user to open an already saved session. A navigation dialog will appear so that you can locate your session.

Recent: Contains a list of recently opened session that can be opened.

Close: Exits the current session. If any changes have been made to the session a dialog box will present a save option.

Save: Saves the current session

Save As: Saves the **entire current session's *folder*** with a new name and/or new location

- Switch to newly-saved version: Closes the session you are currently working with and opens the new session.
- Copy media to new session: All files within the interchange folder (audiofiles, midifiles, and videofiles) will be copied to the new session/interchange folder. If this option is not checked Mixbus will refer back to the previously named session's interchange folder.
- Copy external media into new session: Same as above but also moves any files that may be located in other directories or hard drives. See topic clean-up, sub topic real world example for similar functionality and why you may need it.

Rename: Renames the **current session's *folder***

Snapshot (& keep working on current version) …: Makes a copy of the opened session snapshot in its current state for later recall. (See: Session Snapshots)

Snapshot (& switch to new version) …: Saves a copy of the opened session snapshot in its current state with a new name and/or location. (See: Session Snapshots)

Save Template: Session templates are a fast way of recalling common session “layouts” such as hardware connections, plug-ins, and number of tracks. If you find your current session is a common one, you can save the it as a template (no audio will be saved).

5.3.2. Session Snapshots

What is a session snapshot?

A “snapshot” is a file that stores the current session settings including the track layouts, audio & midi regions, markers, tempos, faders, EQs, and plugins. Snapshots all share the same audio data in the session folder.

Snapshots require very little disk space, so you can store snapshots for multiple mixes or edit sessions without worry.

Using snapshots:

Snapshots are typically accessed via one of 3 methods:

1. In the main “Session” menu you can create new snapshots.
2. In the editor sidebar, you can select from a list of this session’s snapshots.
3. In the New Session dialog, you can click a triangle next to your session to display the list of snapshots in the session.

> Each snapshot has its own *persistent undo history*. If you open an old snapshot and click UNDO, you will undo the last editing operation that was applied to that snapshot.

Backup Snapshots

By default, Mixbus automatically overwrites a backup snapshot every few minutes. The backup snapshot changes its name every hour. So there is a backup snapshot stored for every hour that you have worked on a session; in addition to the most-recent backup file.

To recover from a “backup” file in the backup folder, you must first drag it up one level into the main “session” folder. Then you can open it normally. If you try to open it from within the “backup” folder, it won’t find the audio files.

Here are the steps to recover a backup snapshot from the backup folder:

1. Quit Mixbus
2. Open the session folder and find the “backup” subfolder.
3. Choose the last-created snapshot file (the file ending in .ardour)
4. Copy the file up to the main session folder
5. Launch Mixbus
6. In the “recent sessions” list, you should see a triangle next to your session; click the triangle and choose the backup snapshot.

5.3.3. Session Templates

What is a Session Template?

Once you've created a session and set up your track routing, plug-ins, etc, you may want to reuse that setup for a future project. A template contains everything *except the actual audio and midi data*: track layouts, plug-ins, and your channelstrip EQ/Compressor/Bussing setup.

Any session can be stored as a session "template" for future sessions.

Using Session templates:

To save a template, choose Session->Save Template from the main menu, and name the template something meaningful so you can find it in the future.

To use a template, you have to create a NEW session. In the New Session dialog, there is a list of your templates, in addition to the factory-supplied templates.

Managing & Sharing session templates:

Session templates can be later renamed, edited, or shared using the Template Manager: see Templates

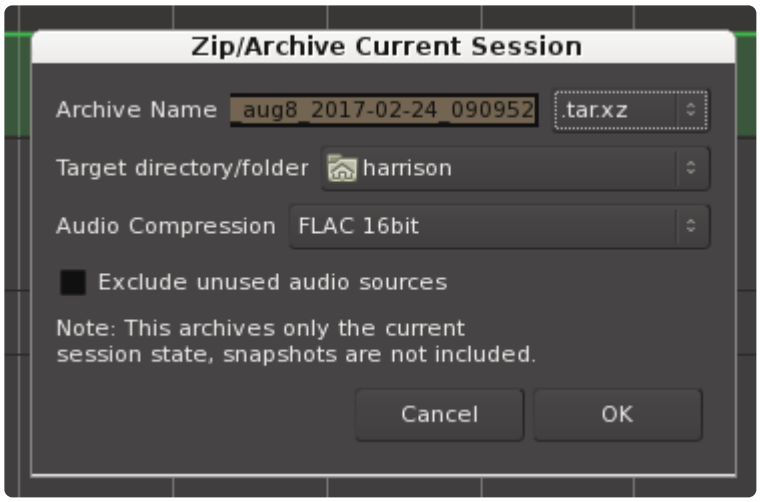
Where are my templates stored on disk?

Templates are stored in the user config area, under a folder called "templates". You can find more details about the user's config folder in Power-User Topics.

> Some plugins can store additional data that won't fit in the session file. For example, a convolution reverb might include the user-selected impulse response files. For that purpose, there is also a "plug-ins" folder inside the template folder which stores this additional information.

5.3.4. Session Archive

Using the “Archive” utiliity, you can create an “Archive” of a session snapshot which automatically compresses the snapshot, the audio+midi files, and other necessary files into a single compressed file.



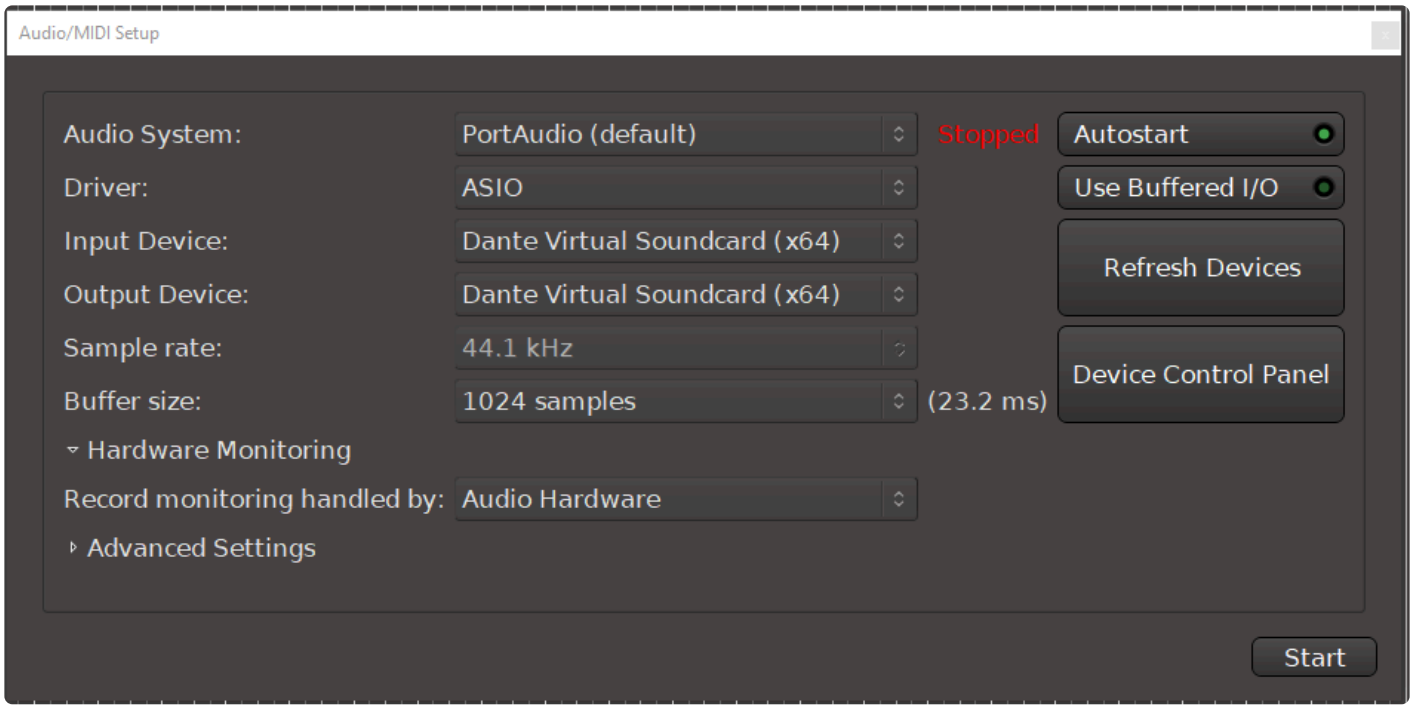
- Archive Name: the name of the resulting archive file.
- Target directory/folder: the folder where the resulting archive file.
- Audio Compression: Choose FLAC to reduce the size of the audio files using “lossless” compression. This requires additional time for the archive process, but results in a smaller archive file. Choose 24bit for full-quality archive, or 16bit for slightly lesser quality, and significant disk-space savings.
- Exclude unused audio sources: if this is enabled, then the archive will not include any audio or midi files that aren’t currently in use by the snapshot. (for example: unused takes)

5.4. Audio Midi Setup Dialog

The ***Audio Midi Setup dialog*** appears automatically when you create a “New” session. This step allows you to choose the session’s sample rate, as well as the soundcard settings and other details.

> The sample rate that is selected during the initial session setup ***cannot be changed*** later. However you can later change the soundcard setup, buffersizes, and other I/O details.

You can also launch the Audio Midi Setup dialog from the main menu, under “Window”.



Start/Stop

As seen in the figure there is an indicator showing “STOPPED” in red while you are setting up these parameters. When you are done with the settings, press the “START” button to connect to the selected audio device.

> It is sometimes useful to click the “Stop” button while you have a session open. This will disconnect Mixbus from the audio device, and therefore use no cpu power. The audio device must be re-started if you want to save your session.

Audio System:

The “Audio System” selection tells Mixbus which way you will connect to your devices.

- To use your audio devices on **Mac OS X**, select “**Core Audio**” (this option is only available on Mac).
- To use your audio devices on **Windows**, select “**PortAudio**” (this option is only available on Windows).
- To use your audio devices on **Linux**, select “**ALSA**” (these options are only available on Linux).
- “**JACK**” uses the JACK server to talk with your sound card and other software on your system. It is only available if you have already installed JACK on your system. If the JACK server is already running, then Mixbus will connect to the JACK server. If not, then Mixbus will start the JACK server for you (the JACK option is available on Mac, Windows and Linux). You can learn more about JACK here.

Driver

This menu specifies the driver software protocol for the selected input and output devices detected. For example you might set this to ASIO if an external I/O device is detected.

Input Device & Output Device:

- Select your preferred audio interface from the pulldown list. For best results, you should use Mixbus with a high-quality audio interface.
- Some combinations of input & output devices will not work. Normally, you should choose the same or similarly-named device for both input and output.

Sample rate:

- A drop down list with all sample rates supported by your selected audio interface will be displayed.

Buffer size:

In conjunction with sample rate, buffer size is used to control latency and to manage hardware resources. See more here.

Hardware input * output latency:

When audio is routed from outboard equipment and back to Mixbus it must undergo a conversion from analog to digital (ADC), the time it takes for this conversion can be entered to compensate for latency so that all audio remains time aligned (This process can be automated, see “Calibrate Audio” below).

> Most users will find that it is unnecessary to calibrate the input & output latency. The converter latency is very small when compared to the delay incurred by the computer buffer-size. Furthermore, small system changes (such as unplugging and re-plugging a USB device) can cause this value to change.

MIDI System

This selection will default to the MIDI framework supported by your OS. You can select “None” if you think the MIDI system is unneeded, or is delaying startup.

Calibrate Audio:

This calls up a latency measurement tool with step-by-step instructions to compensate latency introduced by external equipment and the digital to analog, and analog to digital conversion process.

Device Control Panel:

This opens the control panel of the audio interface that has been selected on the “Device” list, when one is provided.

Autostart

When enabled, Mixbus will skip the “Audio Setup” dialog, whenever possible (for example: when opening an existing session, and the last-selected audio device is capable of the session’s sample rate).

When creating a New session, the Audio Setup dialog will always appear.

Refresh Devices

This re-scans your audio hardware to provide all the available devices in the Input/Output device menus.

Use Buffered I/O (Windows-only)

The “Buffered I/O” button tells Mixbus to use an older ASIO protocol which works better on some devices, including Roland “Capture” devices. If you have trouble launching the audio system, and/or you are unable to set the buffersize and sample rate, you might try enabling “Use Buffered I/O”

5.4.1. Latency and Buffer Size

How do I choose a buffer size? And how does computer audio (really) work?

Although Desktop computers are blindingly fast inside, they are optimized to interface with the world at “human speed”… a user sends dozens of keyboard commands every second, at best; and the screen is updated perhaps 30-50 times per second. But audio events (or more accurately, “samples”) typically occur 44,100 times per second, or more! Computers are not intended to be ‘interrupted’ by the outside world this often; and if you did, it would severely compromise the amount of work that the computer can attend to.

When a computer soundcard (whether internal, USB, Thunderbolt, or something else) gathers audio data from the Analog-to-Digital converters, it can’t “wake up” the computer CPU 44,100 times per second. So it has to gather (or: “buffer”) the audio for a while, and then pass this larger buffer to the computer for processing. This buffer size can be changed by the user to balance the system’s stability and power, versus latency and responsiveness.

If you choose too low a buffer size, you will likely encounter ‘clicks’ when your system gets busy. If you choose too high a buffersize, then you might hear a delay through the system, or notice that the computer is not very responsive to you.

This video by Richard Ames provides an overview of your computer’s audio system, and why the buffer size is such an important factor in your computer’s performance.

Latency

“Latency” is the amount of time it takes for digital audio to pass from the computer’s input to output. The buffer size has a direct effect on latency: Larger buffer sizes will incur a longer latency. In many cases, latency doesn’t matter much. If you just need to play your tracks for mixing, then a fairly large buffersize will provide the best system stability.

Due to the nature of computers and digital audio, there will always be some latency in a DAW. Even if the soundcard operated with no buffer (a buffersize of “1”) then the process of converting Analog to Digital, and then back to Analog, will incur about 2-3 milliseconds.

If audio arrives at the output later than desired then it is considered to have high latency. If the latency is minimal, we may not be able to perceive it. Low latency is desirable for recording, tracking, overdubbing, etc. But there is a lot of debate about how much latency is too much. And it varies depending on the situation and the performers.

Buffer

A buffer is an area in the sound-card’s memory that is set aside to temporarily store digital audio. By

temporarily storing digital audio we can prevent any interruption in the flow of sound. Interruptions will be heard as clicks and pops. We can set the size of the buffer to achieve a specific delay, usually measured in milliseconds. The memory stores samples of audio, and the more samples that are “buffered”, the more latency will be exhibited. Hence, the size of the buffer also determines the latency. Setting the buffer too small can cause buffer underruns with audible pops and clicks. Setting the buffer too large might cause a noticeable delay in the sound or operation.

In digital audio, buffer sizes are almost always defined in blocks of powers-of-2. Your soundcard driver will define the available buffer sizes for you to choose. The most commonly available values are: 64, 128, 256, 512, 1024, and 2048.

Generally speaking, a buffer of 1024 is a good balance of stability and responsiveness when you are mixing. That’s the default value that we choose for Mixbus. But if you are recording yourself and you want to play through the computer, you might choose a smaller buffersize. Just be aware that you might cause pops & clicks in your recording if you exceed the capabilities of your system.

Sample Rate Vs. Buffer size

If you choose to run at 96 or another high rate, you’ll notice 2 things:

1. Your soundcard’s buffer now “fills up” faster, and it wakes the computer more often.
2. Your computer is now processing twice as many samples, and therefore working harder.

For those 2 reasons, you will likely need to increase the buffersize when you use higher sample rates, to maintain the same performance.

If you want to absolutely minimize the latency through Mixbus, you could choose the lowest possible latency and the highest possible rate. This will minimize the latency from input to output. But it will likely exceed your computer’s capabilities very quickly, and you might get clicks&pops in your recording.

Buffer size while mixing or editing

While mixing or editing, a large buffer will prevent pops and clicks at the expense of a slightly less responsive system. This lack of responsiveness may be evident when pressing the play button and a slight delay occurs before audio can be heard. Many times this delay is so small you may barely notice it. So while mixing or editing, a larger buffer is advisable. A buffersize of 1024 samples is typical.

Buffer size while recording

If monitoring through a DAW and an artist needs to listen to themselves while recording (such as a vocalist listening to a headphone mix) then low latency audio will be required. Adjust the buffer size to achieve a latency of 10 ms or so. A value of 128 samples is typical. Furthermore it is important to avoid using plugins that cause additional latency.

Our friends over at Sound on Sound have some great information on this topic

http://www.soundonsound.com/sos/jan05/articles/pcmusician.htm

Hardware Vs. Software Monitoring

If you plan to record with Mixbus, one of the most important options is the Monitoring mode.

If you choose “Mixbus does monitoring” then:

1. Mixbus can control which channels you are hearing (subject to the Monitoring rules … see section “Monitoring”). So, for example, Mixbus will only “monitor” the inputs that are currently record-armed.
2. The inputs will automatically be mixed into the master output of Mixbus, so you can hear them in context with your mix.
3. You will be able to hear any effects (eq, compression, and/or plug-ins) that are applied to the tracks.
4. HOWEVER: the signal path through your converters, Mixbus, and plug-ins can add up to a significant latency. The latency can be minimized by using a small buffer size, and avoiding the use of plug-ins with long latency

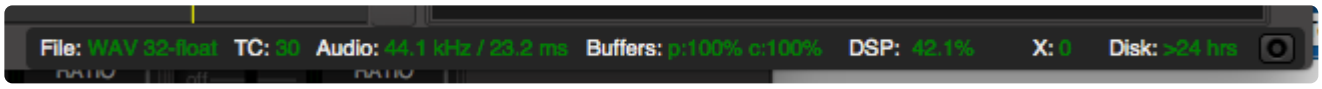
If you choose “Hardware does monitoring” then:

1. You will not incur the latency penalty of getting into and out-of the computer. However if your interface or mixer uses digital signal processing, there may still be some latency.
2. You must use your external I/O controls, and/or mixing console, to build monitor mixes for your talent.
3. Mixbus cannot control which channels you are hearing. Record-arming a track in Mixbus will not automatically route that track into the monitor mix.

✻ Some artists may be more sensitive to latency than others. If the artist is not performing as expected, one thing to check is latency; even if the artist does not perceive or complain of latency. In some anecdotal cases, singers have been bothered by the “comb filter” (frequency cancellations) caused by a very short delay between their headphones and internal vocal cavities. In this case you can try decreasing the latency to see if that helps. (sometimes increasing it will also help). If the singer finds this objectionable then you should consider a all-analog, hardware-monitoring setup instead.

! On Windows, using ASIO drivers, you must sometimes choose the buffer size from the device’s control panel. In a few cases, Mixbus does not have control over ASIO buffer sizes.

5.5. Status Bar



The status bar provides information about your Mixbus's status including current settings and realtime performance indicators. Right-click on the status bar to enable or disable the available displays.

- File: Audio file format setting (from Session->Properties->Media)
- TC: Time code format (from Session->Properties->Timecode)
- Audio: Sample rate / Buffer latency (from Window->Audio/MIDI Settings)
- Buffers: p (playback) and c (capture) – these numbers indicate the ability of your disk to keep up with Mixbus's playback and recording buffers. If either number is reduced to zero, then Mixbus will stop playback and report an error to you.
- DSP: Time required to process an audio block / duration of the audio block, compared to the available buffersize timing. This displays the worst-case maximum (not average).
- X: Number of xruns : an xrun (*under* -run or *over* -run) is caused when Mixbus cannot keep up with your soundcard. (see below)

Disk: Estimated remaining recording time, given the current settings and number of record-armed tracks.

Of these reports, the Xrun counter is the most important. If the xrun counter is increasing while you are recording or playing, it means that you have exceeded the capabilities of your system, and you are likely hearing clicks and pops.

The typical methods to minimize the CPU load are:

- increase the "buffer size" in the Audio Setup dialog
- optimize your system for audio performance; such as choosing "optimize for background tasks" in Windows.
- use a lower sample rate, reduce the number of tracks
- reduce the number of plugins in your session.
- And of course using a faster computer, with more CPU cores, is desirable.

The CPU meter on your computer averages the cpu usage over a very long period (perhaps one second). In digital audio, the timing is much more sensitive. When the soundcard passes us a buffer, then we have to wake up, process the audio buffer, and return it to the soundcard before the soundcard needs to play it out. If we don't wake up in time, or we don't get finished in time, then you hear a "click" caused by the (we call these xruns, short for over-run or under-run).

In the case of a 1024 buffer size, this has to be done within (1024/44100) 25ms, or about 1/40th of a second.

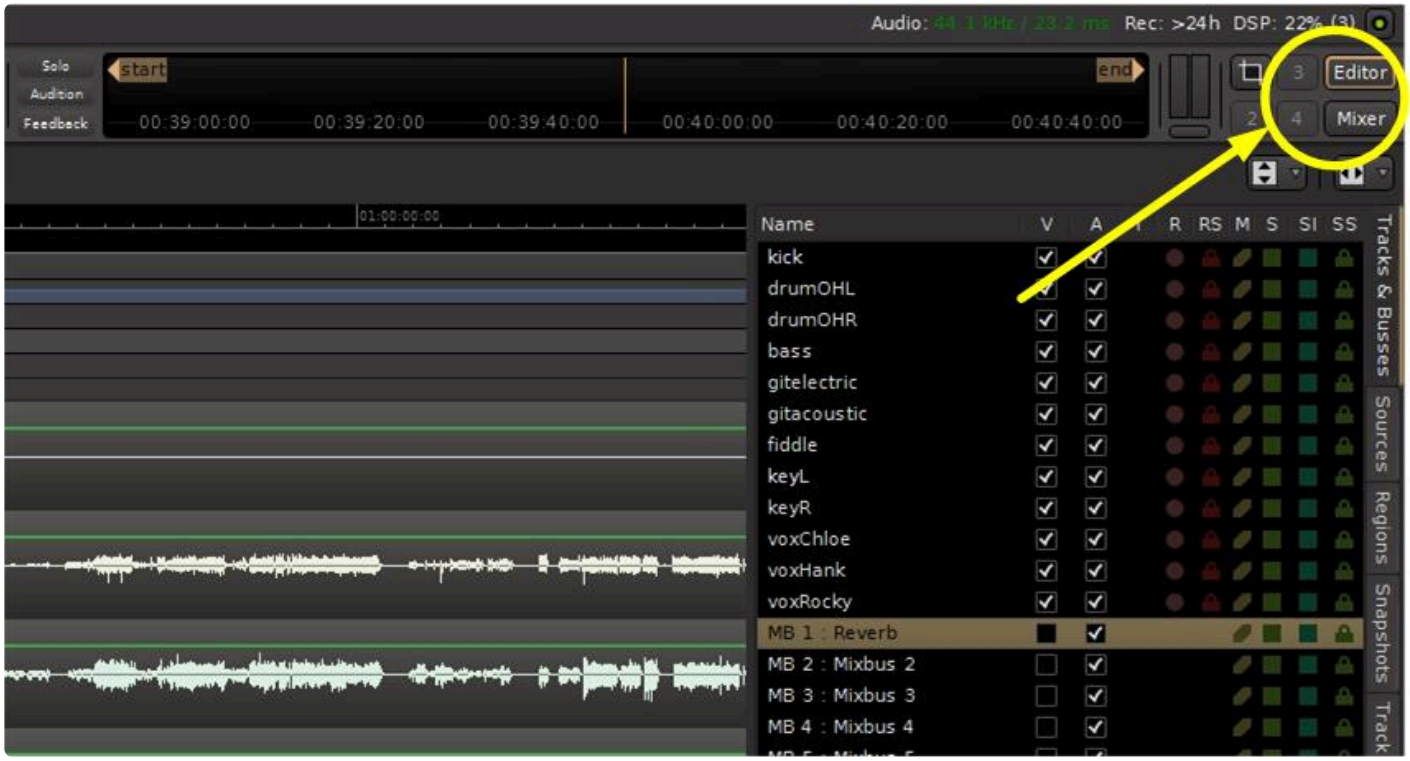
This is complicated by the fact that it's the OS's job to "wake us up" and tell us that the soundcard has some data for us. Some OS's and drivers are better at this than others. But let's say it takes 5ms before we are even alerted that some data is available. We now have 20ms left to process the audio. If it takes us 10ms to process the audio, then we are finished within 50% of the allotted time and that's what we display in the meter. This is a very accurate indication of your computer's ability to process audio.

This also explains why the selected buffersize is so critical to the DSP load. A 256-sample buffersize is about 6ms, so if your computer sometimes waits 5ms after receiving the soundcard interrupt before it "wakes us up", we only have 1ms remaining to do our work.

For more information on factors impacting the performance of your computer please see this video:
CPU Performance vs. Real-Time Performance in Digital Audio Workstations

5.6. Primary Windows: Editor and Mixer

This section will reference the Editor and Mixer windows. These are the two primary windows that are used to operate Mixbus. You can navigate between the two windows using the Editor and Mixer buttons at the top right corner of Mixbus:



You can also detach either screen by right clicking either button and choosing the Detach option, or by dragging either button to the desktop . This is useful in dual monitor setups.

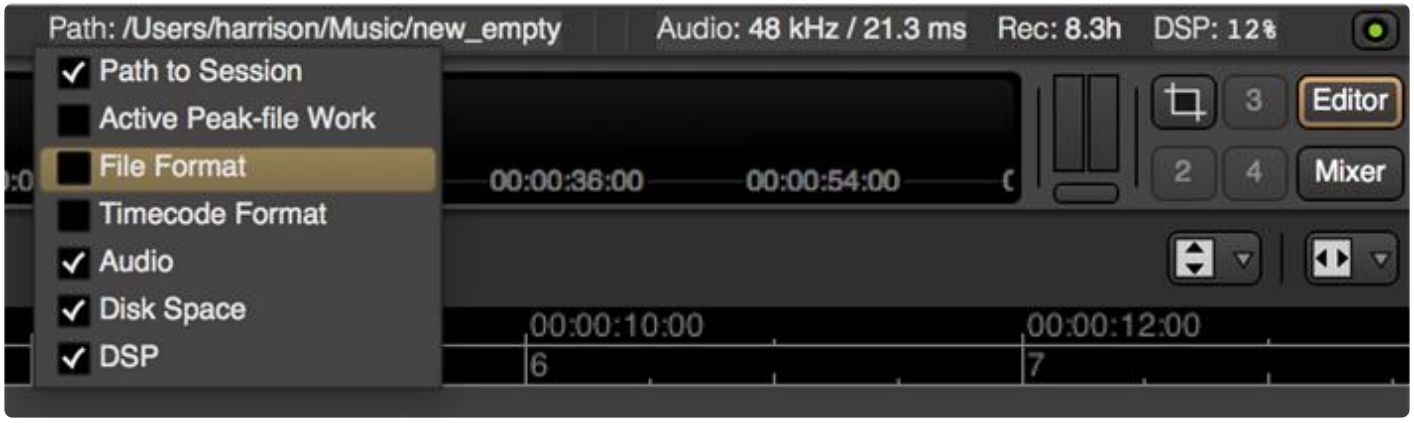
The Transport Toolbar

The top portion of the Mixbus main window is dedicated to the transport toolbar. The transport toolbar provides transport buttons (Play, Stop, Rewind), the main Clock Displays, and the mini-timeline. You may add or remove elements from the transport toolbar in Preferences->Appearance->Toolbar. For example, when recording you might wish to show the “Punch” and “Monitoring” controls; but then you can remove those when need more room for the tasks of editing and mixing.

The Info Bar

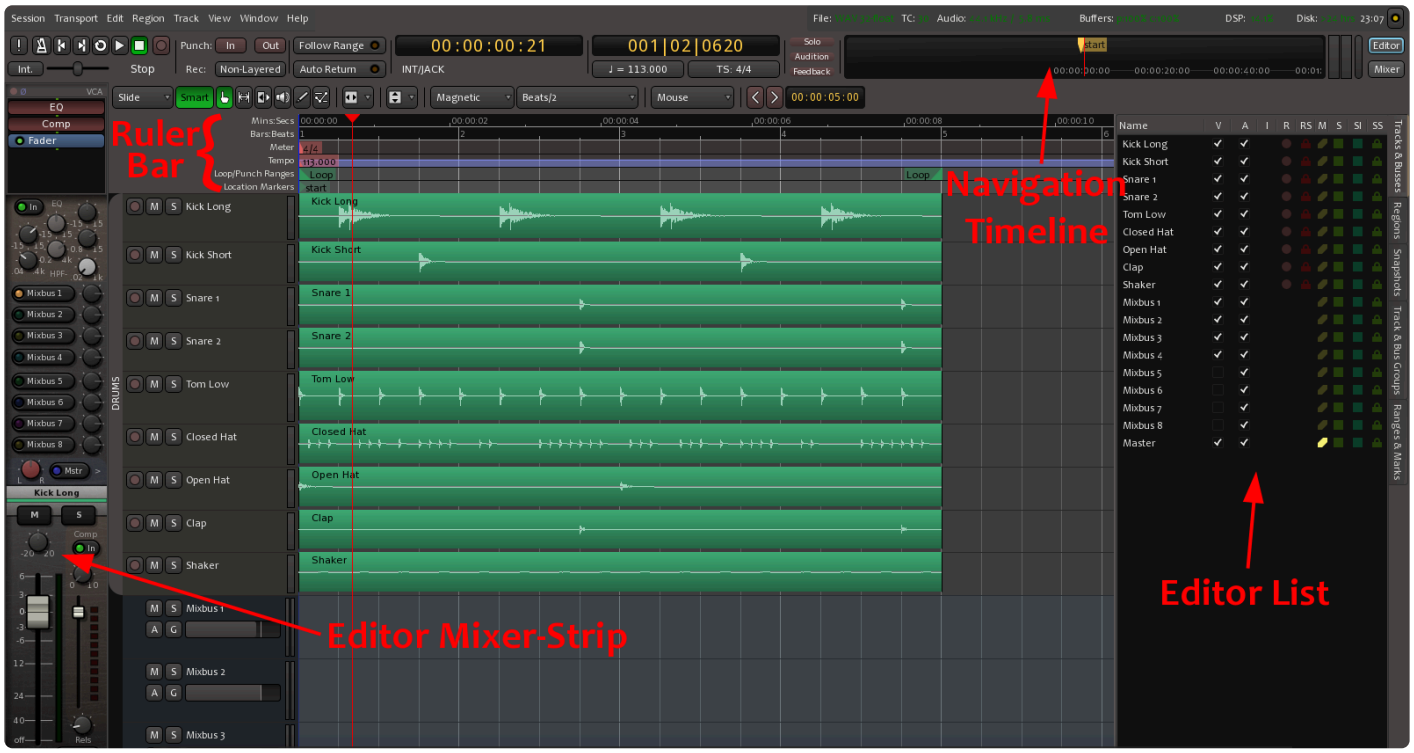
At the top right end of the transport toolbar is the Info Bar. It displays the soundcard settings, how much disk space you have, and how much CPU you are using. You can right-click on it, and select what items you

want to show. It also has the ability to show the path to the session:



The Editor Window

When you first launch Mixbus, you will be presented with the **Editor** window. The editor window presents the Audio and MIDI tracks that make up your session. You can reorder, resize, and rename the tracks from the Editor window. Generally speaking, the Editor window provides all the features you need to edit your arrangement, while the Mixer window allows you to balance the sounds in the arrangement so they can all be heard.



The **top** of the editor screen provides the **transport buttons** (play, stop, etc) as well as the main clocks that display the location of the playhead in the session. Just below the transport buttons are the **editor tools** which are used to edit your arrangement.

The **top right** of the editor screen provides the **Mini-Timeline**, which allows you to jump around the timeline. Left click to jump to a specific place, or scroll the mouse wheel while hovering over the timeline to jump around. You can hold down Control for finer jumping, and Control+Alt for still finer jumping. Right-clicking on the Mini-Timeline allows you to select a time display range. If you create Location markers, they will appear on the Mini-Timeline. You can click on a marker to jump to that location.

Just below the editing tools, you'll see the **ruler bar** which helps to indicate your timeline position using Min:Sec, timecode, samples, bars/beats, as well as providing a place to set Markers, Time Signature or Tempo changes, and Loop and Punch ranges. Learn more about the rulers here.

The main editor **canvas** shows your tracks, and any regions (audio clips) that are present.

On the **right** of the editor is "Editor List". The Editor List may be shown and hidden via the "View" menu. Learn more about the Editor list here.

On the **left** of the editor is the "Editor Mixer". The Editor Mixer shows the mixer channelstrip for the currently-selected track or bus. The Editor Mixer may be shown and hidden from within the "View" menu or the associated keyboard shortcut (Shift+E by default).

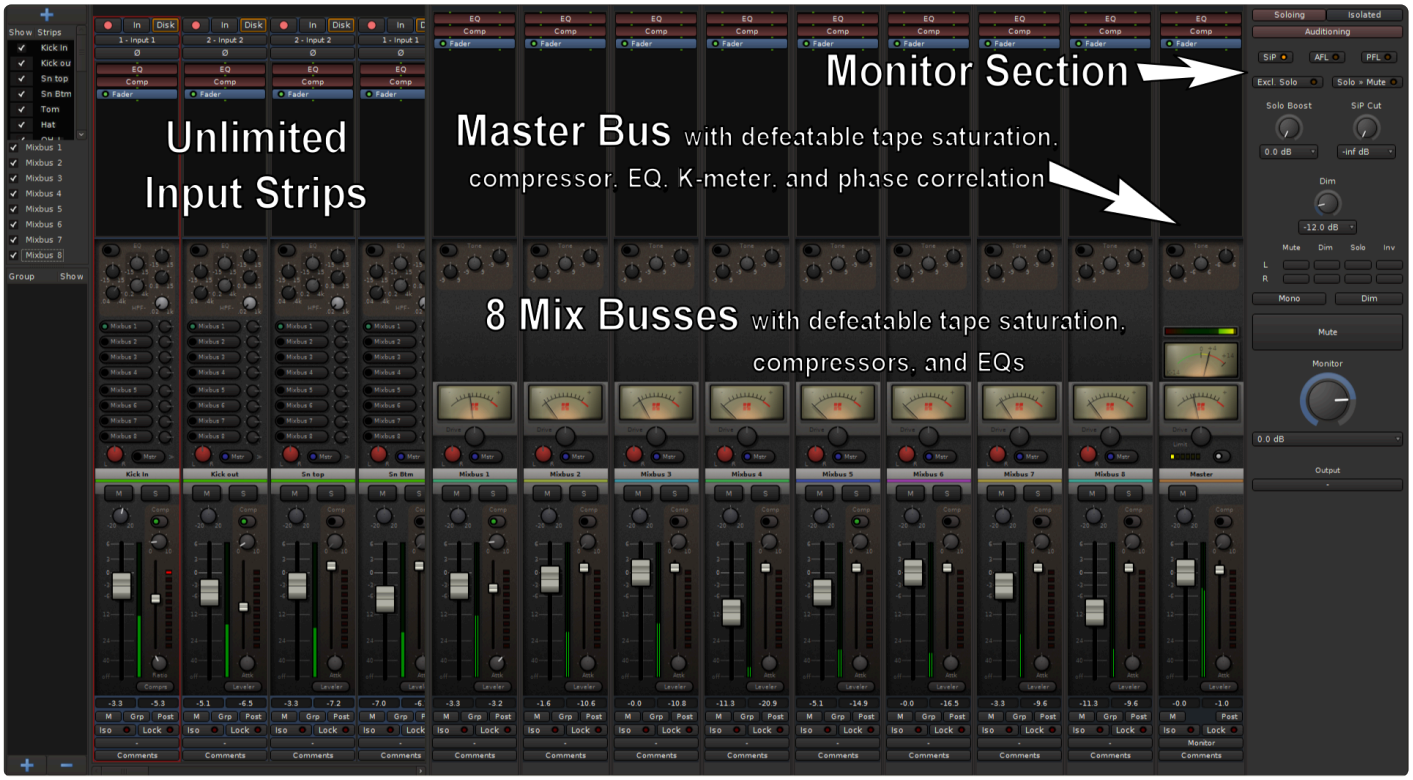
The **bottom** of the editor is the Summary view. The summary view shows a light grey box that encompasses your current view, with the entire session behind it. You can move the grey box around your session to quickly move your view. The Summary may be shown and hidden via the "View" menu.

Mixer Window

The Mixer Window may be shown using the "Window" menu, or the keyboard shortcut (ex: Alt/Opt+M) which toggles between the editor and mixer windows. If you have the editor and mixer windows on different monitors (screens), then the "Toggle Editor/Mixer" shortcut does not have an effect.

When viewed in the Mixer Window, Mixbus resembles a traditional analog console. Input Channels are on the left, while the busses and master are located on the right side.

The audio signal flow is generally left-to-right. The signal starts at the input channels (which might be live inputs, playback tracks, MIDI tracks, or virtual instruments), progresses through the mix buses, and into the master. The signal path's EQ, Filter, Compressor, panning, bus summing, and master bus processing are performed inside Harrison's proprietary signal path, are always present, and do not require user-supplied plug-ins.



In Mixbus, like other audio workstations, each recording track has its own mixer strip. These mixer strips resemble the input channels of a traditional analog mixing console. Input channels can be configured to be mono or stereo and by default include all the DSP processing that makes up a real console channel. This is the major difference between Mixbus and other DAW mixers. No time is wasted searching for and adding the necessary high-quality mixing tools.

Just to the right of the input channels are the 8 mix bus channels. Mix bus channels are used to collect the outputs from the input channels into groups. For example, in a typical music project, you might send the drums to mix bus 1, the vocals to mix bus 2, instruments to mix bus 3, and use mix bus 4 as a parallel effects bus, like reverb. This allows you to easily adjust the relative levels of these groups and apply processing to them.

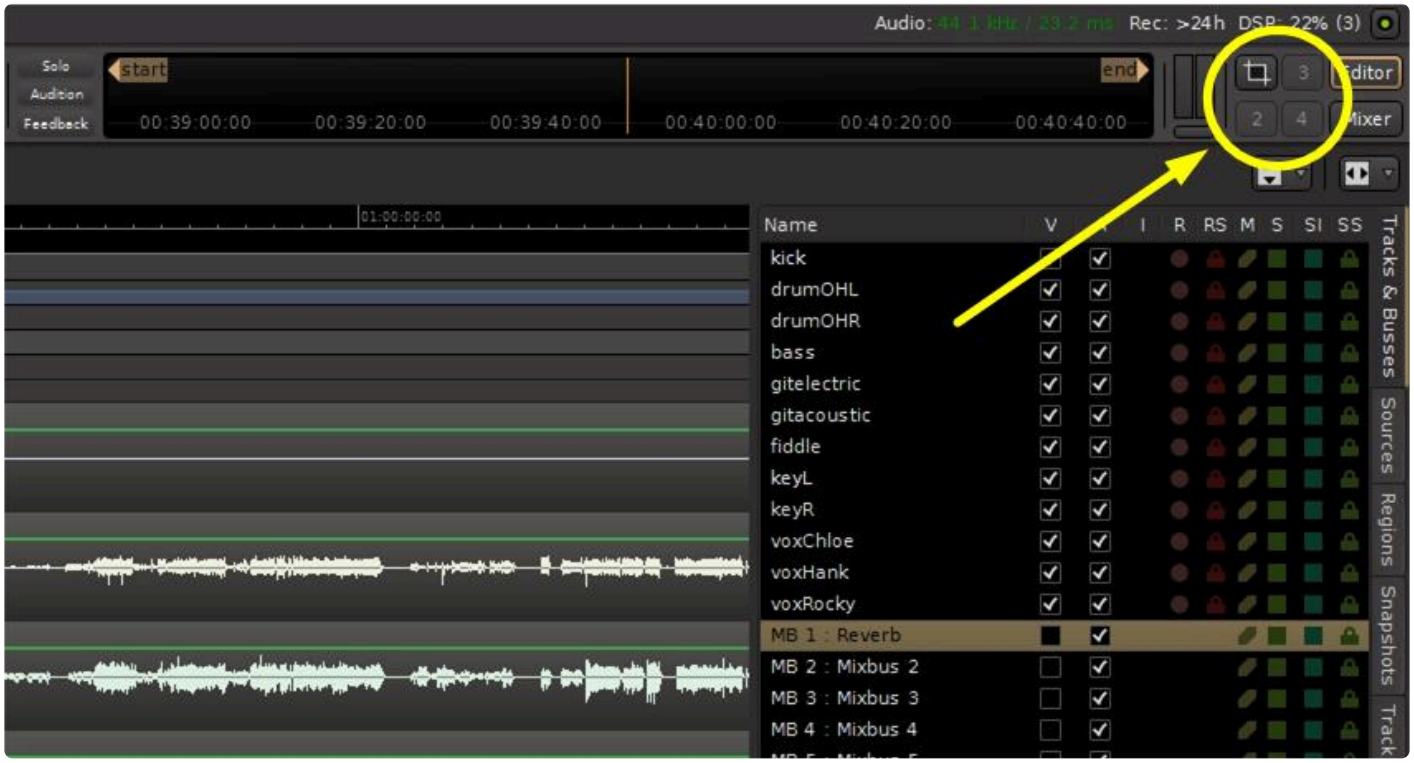
> ✻ You can double-click on a strip's "name" (we also call it the strip's "slate") to toggle between wide and narrow mode. Shift+double-clicking a strip will toggle ALL of the strips together. Toggling all of the strips to a "narrow" view will allow more channels to fit on the screen.

The master channel is the final mixer strip on the right. Typically, the master channel will be connected to your monitoring system (speaker outputs). The output of the Master channel is the final output of your project i.e. your 2 track master file. This output is rendered to the final .wav file when you export the session.

The monitor section on the far right is has been made available for those who do not have access to a dedicated external monitoring section.

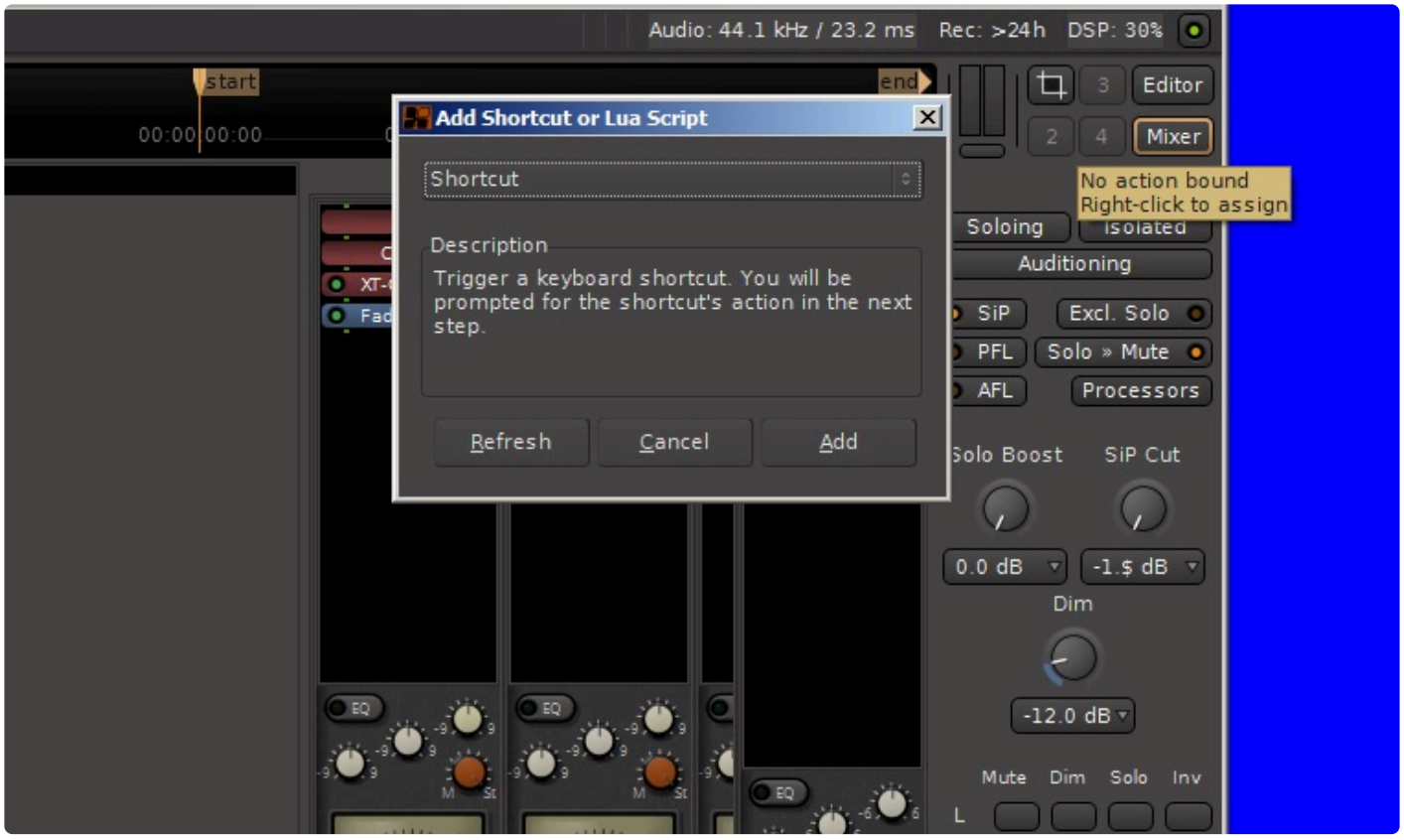
Shortcut Buttons

On both the Editor and Mixer windows there are 4 shortcut buttons in the upper right corner:



These buttons may be customized for many functions. In the picture button #1 happens to be configured to do a Mixer screen shot. The buttons can be configured as a single-click trigger of a keyboard shortcut, as a trigger of a LUA script which you might write, and many other functions.

Assigning these buttons is done by right clicking. Here right-clicking was done on button #2:

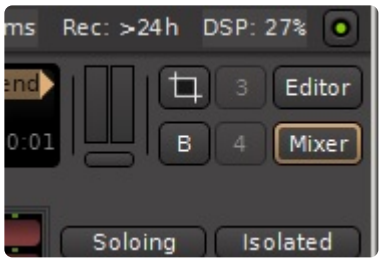


When you hover your mouse cursor over an unassigned button, the “No Action Bound” box will pop up to let you know its function is unassigned. Right clicking on the button will bring up the “Add Shortcut or LUA Script” box shown.

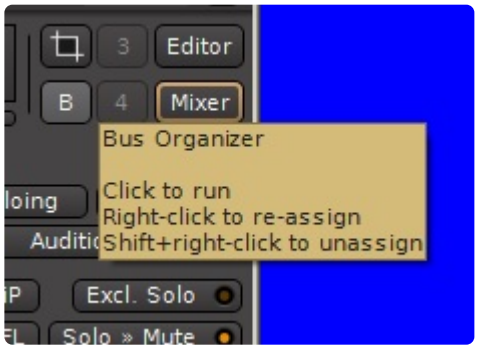
“Shortcut” is the default function in the select tab but is not the only function you can select:

If you select “Bus Organizer” for example and click on the “Add” button, button #2 will now control the Bus

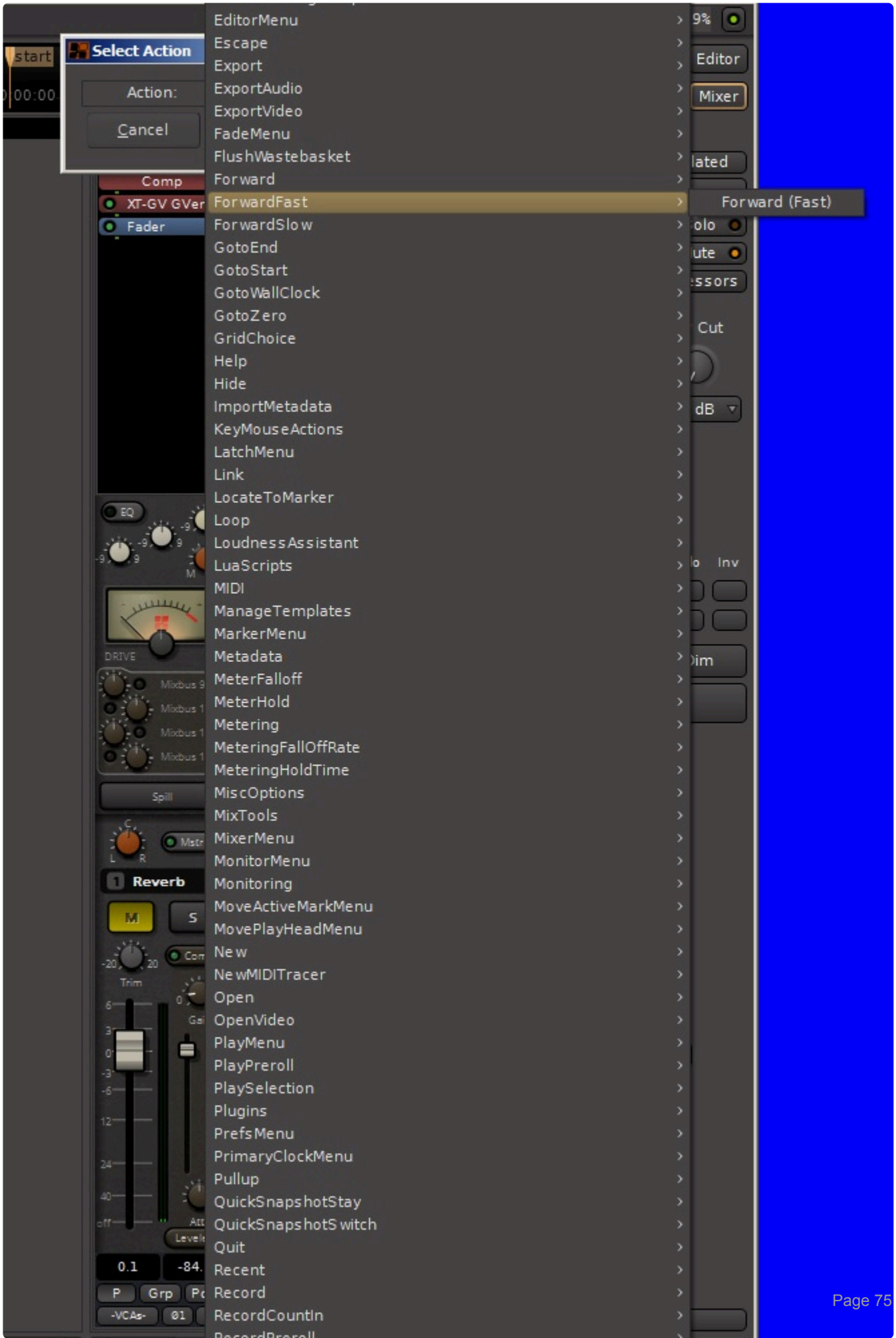
Organizer function:



Putting the mouse cursor over the button, the popup box will tell you how to change or clear the button function:



If you select “Shortcut” and click on the “Add” button, you can select from a HUGE number of functions you can assign to this button:


Select Action
Action:
Cancel
EditorMenu
Escape
Export
ExportAudio
ExportVideo
FadeMenu
FlushWastebasket
Forward
ForwardFast
ForwardSlow
GotoEnd
GotoStart
GotoWallClock
GotoZero
GridChoice
Help
Hide
ImportMetadata
KeyMouseActions
LatchMenu
Link
LocateToMarker
Loop
LoudnessAssistant
LuaScripts
MIDI
ManageTemplates
MarkerMenu
Metadata
MeterFalloff
MeterHold
Metering
MeteringFallOffRate
MeteringHoldTime
MiscOptions
MixTools
MixerMenu
MonitorMenu
Monitoring
MoveActiveMarkMenu
MovePlayHeadMenu
New
NewMIDITracer
Open
OpenVideo
PlayMenu
PlayPreroll
PlaySelection
Plugins
PrefsMenu
PrimaryClockMenu
Pullup
QuickSnapshotStay
QuickSnapshotSwitch
Quit
Recent
Record
RecordCountIn
Forward (Fast)
Editor
Mixer
Cut
Reverb

Maximize Mixer/Editor space

The “Maximize” commands can be found under menu> View. Enabling this function causes the editor window to occupy all screen real estate. The title bar will be hidden as will the task bar. The exact behavior of this function depends on your OS’s desktop “fullscreen” conventions. The same function is available for the Mix window. To exit this mode uncheck Maximize Editor in Menu>View. (if you are unable to see the main menu, try moving your mouse to the top of the screen, and wait for the main menu to appear)

5.7. Adding Tracks and Buses >>

5.7. Adding Tracks and Buses

Adding Tracks, Buses and VCA's

This section describes how to add tracks, buses, and VCA controls to your session. If you wish to import existing media, you should visit the Importing Files section.

What is the difference between Tracks, Buses and VCA's?

Mixbus supports several kinds of objects that can appear in the editor and mixer windows:

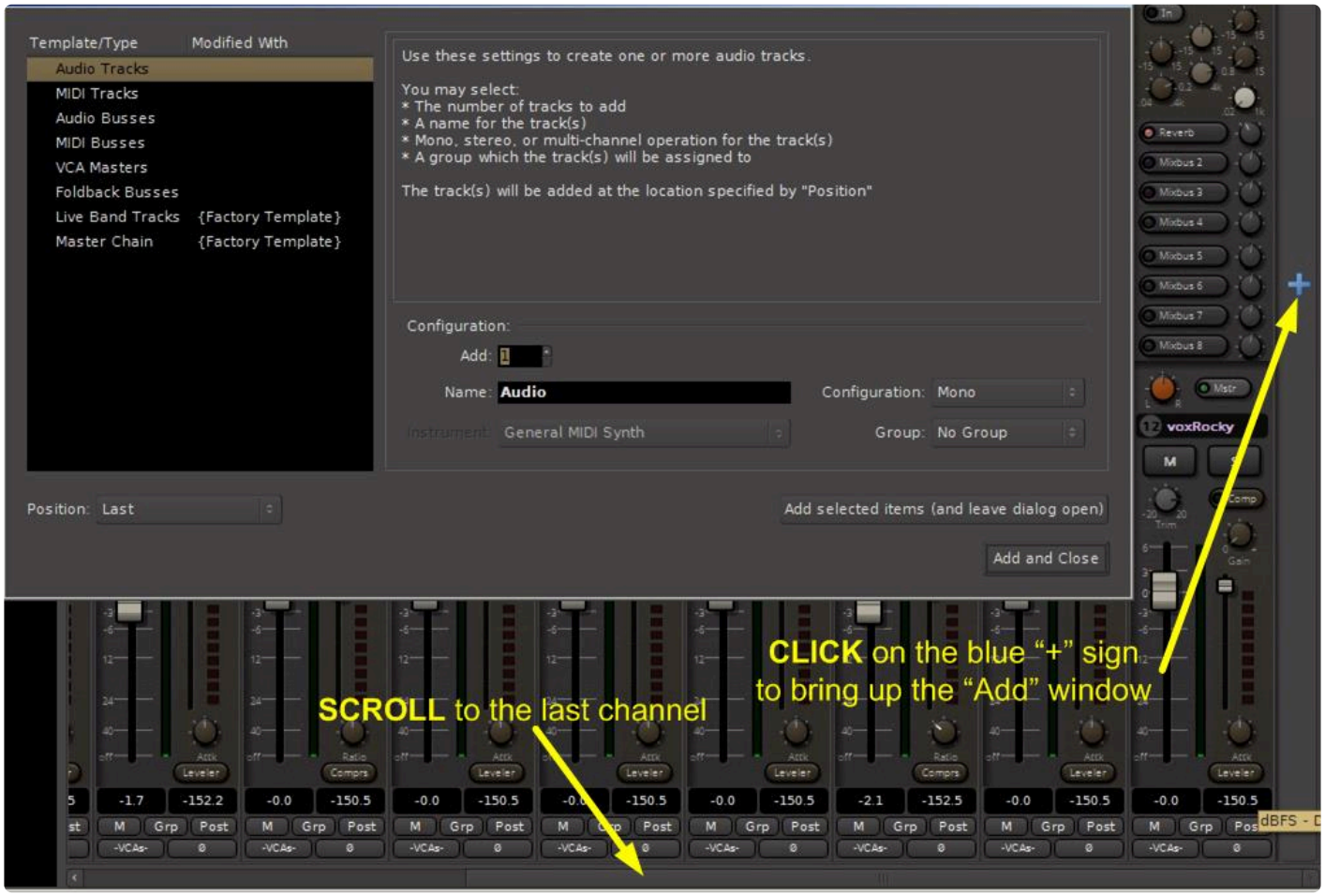
- **Audio tracks** contain a sequence of audio "regions" on the timeline. Audio tracks also provide the mixer features such as fader, panning, bussing, eq, compressor and plugins.
- **MIDI tracks** contain a sequence of midi regions on the timeline, so they can be recorded, edited and played. MIDI tracks sometimes have a Virtual Instrument (VI) plugin, which converts the MIDI regions into audio. When a VI is used on the track, MIDI tracks provide the mixer features such as fader, panning, bussing, eq, compressor and plugins. If no VI is provided, then the MIDI data is sent directly to an output; either to a MIDI Bus, or to an external gear.
- **Audio buses** are just like audio tracks, but they have no playlist of audio regions; the bus is only used to process audio from another track or live input.
- **MIDI buses** are just like midi tracks, but they have no playlist of audio regions; the bus is only used to process midi from another track or live input. Sometimes, a MIDI serves as a good location for a single VI that is fed from multiple MIDI tracks.
- **VCA channels** are described here: VCA's
- **NEW** in v6 – **Foldback Buses** which are typically used to create a cue (headphone) mixes for the artist. These are audio mix buses, not recorded, which are compeletely separate from the main mix buses. Foldback buses are described here: Foldback Buses

> **A note about buses, and Mixbus's mixing workflow**: Unlike typical DAW workflow, where track outputs are connected to other track(bus) inputs, the Audio Buses added here are not intended to be used as part of the mixer panning & summing. The goal of Mixbus is to revert to the buss-style mixing that is used in "hit-making" studios. It's a subtle difference but very important. Audio Buses are best suited for utilitarian tasks such as "live" inputs from an external reverb, or grouping 2 similar tracks together for processing, before they are assigned to a mixbus or master bus.

There are 4 ways to open the "Add Track/Bus" dialog:

- Use the "Add Track/Bus" menu item, which can be found in the Session and Track menus.
- Right-click in the empty gray area in the editing canvas.
- Right-click in the empty gray area of the mixer window.
- Clicking the blue "plus" button to the right of the last channel strip (you may need to scroll to the last channel or resize your window to see it).

The “Add Track/Bus/VCA” dialog



The “Add Track/Bus/VCA” dialog provides a list of the track types and templates that are available to be created.

At the top of the list, there are generic entries for New Audio Track, New MIDI Track, etc. When one of these items is selected, some of the options below will be provided:

- Add: Multiple audio tracks can be added at one time via the “add” setting.
- Name: A custom name may be assigned to the audio track before it is added. If multiple audio tracks are added they will be sequentially numbered.
- Configuration: (for Audio tracks only) Allows for stereo or mono tracks with mono being the default.
- Instrument: (for MIDI tracks only) – this allows the user to choose the VI (Virtual Instrument) that will be inserted in the MIDI track.
    - Selecting a synth for the track will change the audition synth.
- Group: Assigns the new audio track to a group. Whether a group is selected or not you can change the group assignment at anytime after the audio track is added.

Session Templates

If a Template is selected, instead of the generic “Add Track/Bus” selections, then you can view the

description of the template, and any unused controls are greyed-out.

Mixbus supports 2 different kinds of templates, which both appear in this list:

- Scripted Templates: these templates are developed as Lua Scripts, and can prompt the user for more input when they are run. It is possible for users to add their own scripts. These scripts currently report “{Factory Template}” in the *Created With* column.
- User Templates: these templates are created by the user, and are created when the user clicks Session->Save As Template in the main menu. These templates report the version of Mixbus (or Mixbus32C) that was used to create the template. It is not guaranteed that templates made in one version of the program will work perfectly in a different version.

At the bottom of the dialog, you’ll find 3 buttons:

- Insert: Determines where in the Mix/Edit window the new audio track will be placed.
  - Before Selection: In the Mix window this will place the new track to the left of the selected track. In the Edit window this will place the new track above the selected track.
  - After Selection: In the Mix window this will place the new track to the right of the selected track. In the Edit window this will place the new track below the selected track.
  - Last: This will place the new track in the last position in the Mix/Edit window.

- Add (and leave dialog open): this adds the selected track(s) and leaves the dialog open, so more tracks can be added

- Add and Close: this adds the selected track(s) and closes the dialog

NOTE: You can re-order tracks anytime, using either the Editor List, or by selecting tracks and then using the Track menu { “Move Tracks Up/Down..” } to move the selected tracks together.

5.7.1. Track Templates

Track templates store the settings of a track so it can be re-used in another session. Track templates store the track's name, plugins, and mixer settings (such as panning, EQ, compressor, and mixbus/master sends)

Creating Track templates:

To create a track template, right-click on the channelstrip and choose "Save as Template"

Using Track templates:

In the "New Track" dialog, you will see your saved templates, along with the factory-provided templates, in the left column of the New Track dialog.

Managing & Sharing track templates:

Track templates can be later renamed, edited, or shared using the Template Manager: see Templates

5.8. Managing Tracks

On the left side of the editor canvas, you'll find the Track Headers for each track. From the track header, you can mute, solo, or otherwise affect the track.

Mixbus provides several methods to rename and re-order your tracks to help your workflow. You can rename and reorder tracks from the editor view or the mixer view.

The editor list is a great place to manage your tracks, because it typically shows many more tracks at the same time, and you don't have to scroll the main editor canvas to rename or re-order tracks.

Naming Tracks

> ✻ No two tracks or buses can share the same name. If you try to rename a track to something that already exists, Mixbus will show an error dialog (when practical), or append a number to the end of your newly named track.

- In the editor window, double-click on the track name to edit it. Press TAB to advance to the next track.
- In the editor list, "Track/Bus" tab, double click the track name to change it. Press TAB to advance to the next track.
- In the mixer sidebar, right-click on the mixer-strip name, and choose "rename" from the popup menu.

> ✻ You can't rename a track while it is record-enabled, because the track name defines the resulting audio wavefile name.

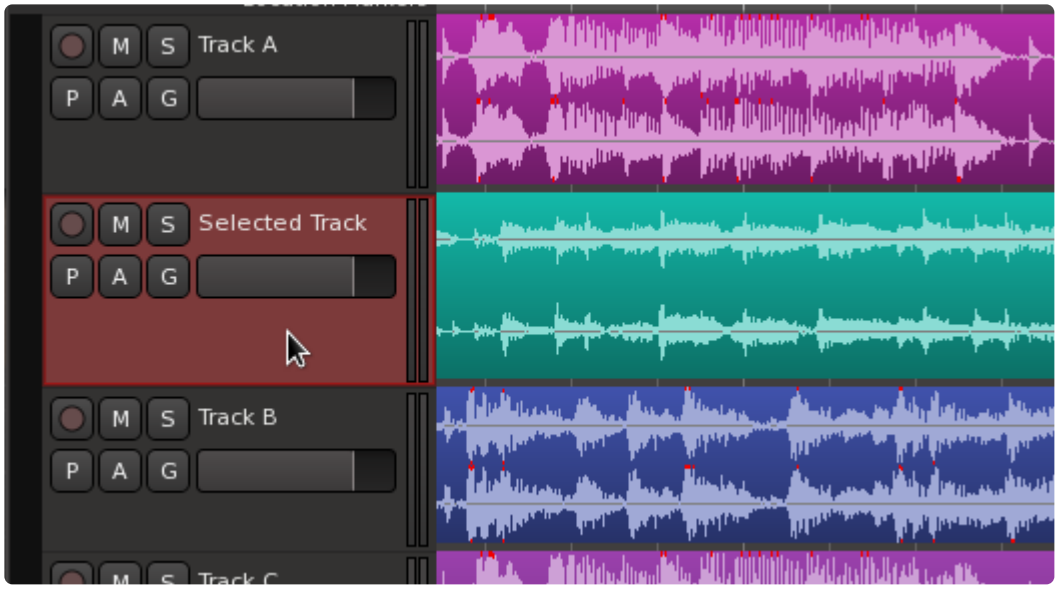
Re-ordering tracks

There are 3 ways to change the order of tracks:

- In the editor window, you can select tracks by turning them red (clicking on them to select them), then use the menu item Tracks->Move Tracks up/down (or the associated keybindings) to move the selected tracks up and down together.
- In the editor list, you can drag and drop tracks to reorder them.
- In the mixer sidebar, you can drag and drop tracks to reorder them.

Selecting tracks

Click on the track "header" (the area which holds the mute, solo, and track name) to select a track. The track should highlight, with a red border, as shown below:



When you “select” a track, Mixbus will make an effort to scroll that track into view, in both the editor and mixer windows.

> Unlike some DAWs, selecting a track does not select the “regions” in the track. In Mixbus, you select tracks and time-ranges with the Range tool, and Objects (Regions) with the Grabber tool. Those are distinct operations and they are used for different tasks. You can’t have both selected at once.

You can multi-select tracks by using standard modifier keys:

- Hold Shift while selecting to select a contiguous range of tracks.
- Cmd+Click on a track to add or remove it from an existing selection.

What can I do with selected tracks?

With tracks selected, you can perform a wide range of operations including:

- Call operations from the Track menu, like “Remove” or “Move Up/Down” … this will apply to all selected tracks
- Right-click on a VCA, and choose to “assign” (or de-assign) the selected tracks to a VCA
- Right-click in the Group box, and choose “make a new group -> from selection”
- Use items in the toolbar, or the View menu, to Fit the selected tracks in the editor

Other ways to select a track:

- Clicking anywhere in a mixer strip will select the mixer strip, and this also selects the associated track in the editor.
- Initiating a “Range” selection on a track will select the track. See: Range Editing
- Clicking on the track in the editor sidebar (Tracks/Buses tab) will select the track, and also scroll it

into view if it is not currently shown.

- By default, “Alt + up&down arrow keys” will advance the selection up or down through the tracks.

5.9. Audio Input & Output

Input

Mixbus uses an internal audio routing system to make audio connections between Mixbus, your sound I/O, and other parts of Mixbus.

With a new session, Mixbus makes these connections:

- The master bus is connected to the first two physical output ports.
- Track/Bus inputs are assigned sequentially from the physical inputs whenever a new track/bus is created.
- The Auditioner and the Click are connected to the first two physical output ports.

However it is possible to change any or all of these connections using buttons in the mixer strip, or Window->Audio Connections.

Track I/O Selections

Track inputs can also be assigned directly from the channel strip, by clicking on the switcher near the top of the channel strip to reveal the channel's redirect page. See it's individual page for more details.

*The image above shows a channel strip **before** its switcher has been activated*

*The same channel strips **after** its switch has been activated showing its redirect page.*

A gif showing how to change the inputs of a channel

Output

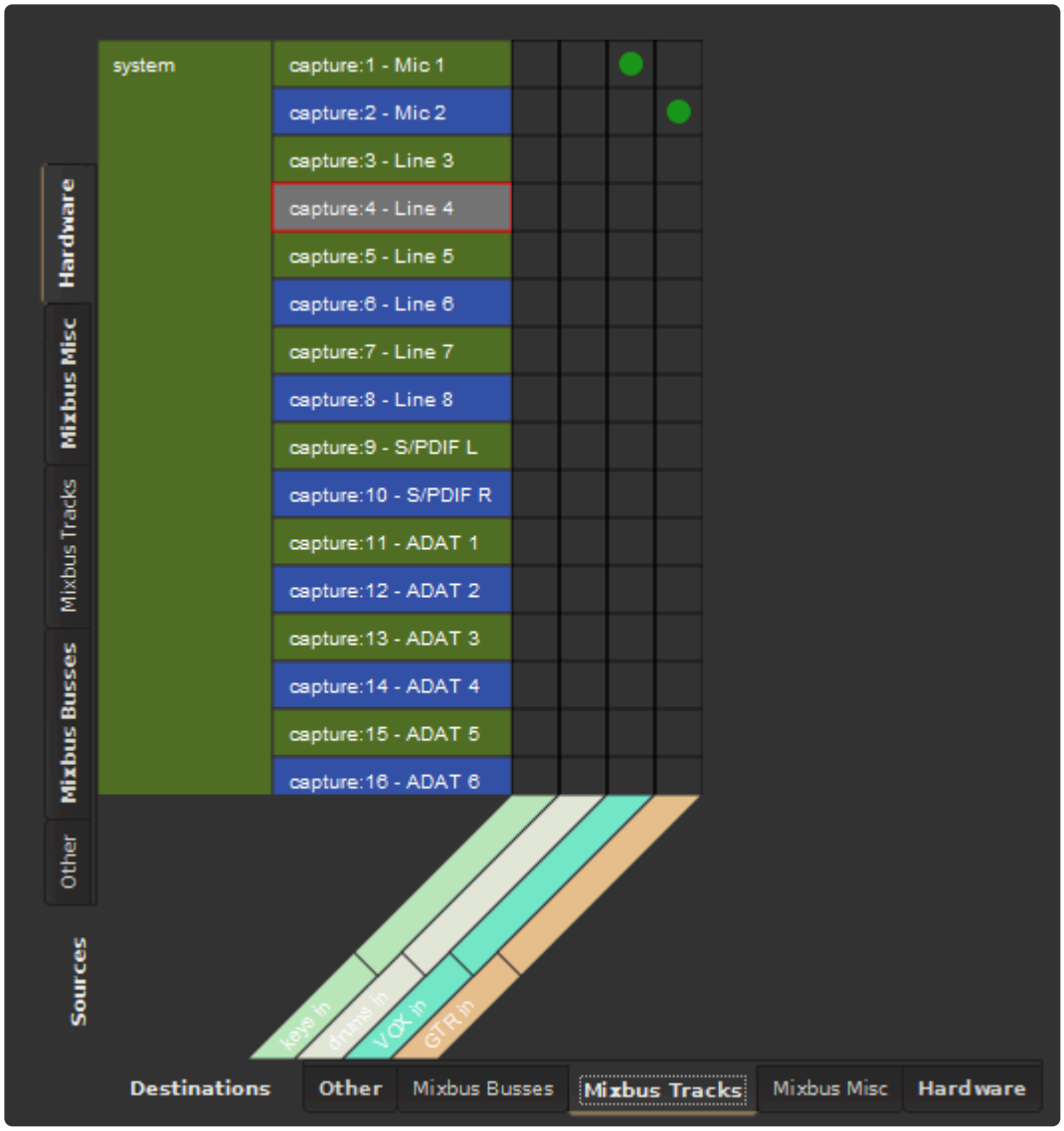
For convenience, the Master Bus has an “output” button on its redirect page, which can be used to select the ports that the master bus output will feed. Normally, you connect the Master Bus output to your audio interface outputs.

In the above image the master bus has both of its outputs connected to the audio interface 1/2 stereo output.

> Note: The redirect page may not display a connection until you select it from the menu. The redirect input and output points are provided as a quick way to override the current setting, not a method to see what the track is connected to.

Audio Connections Manager

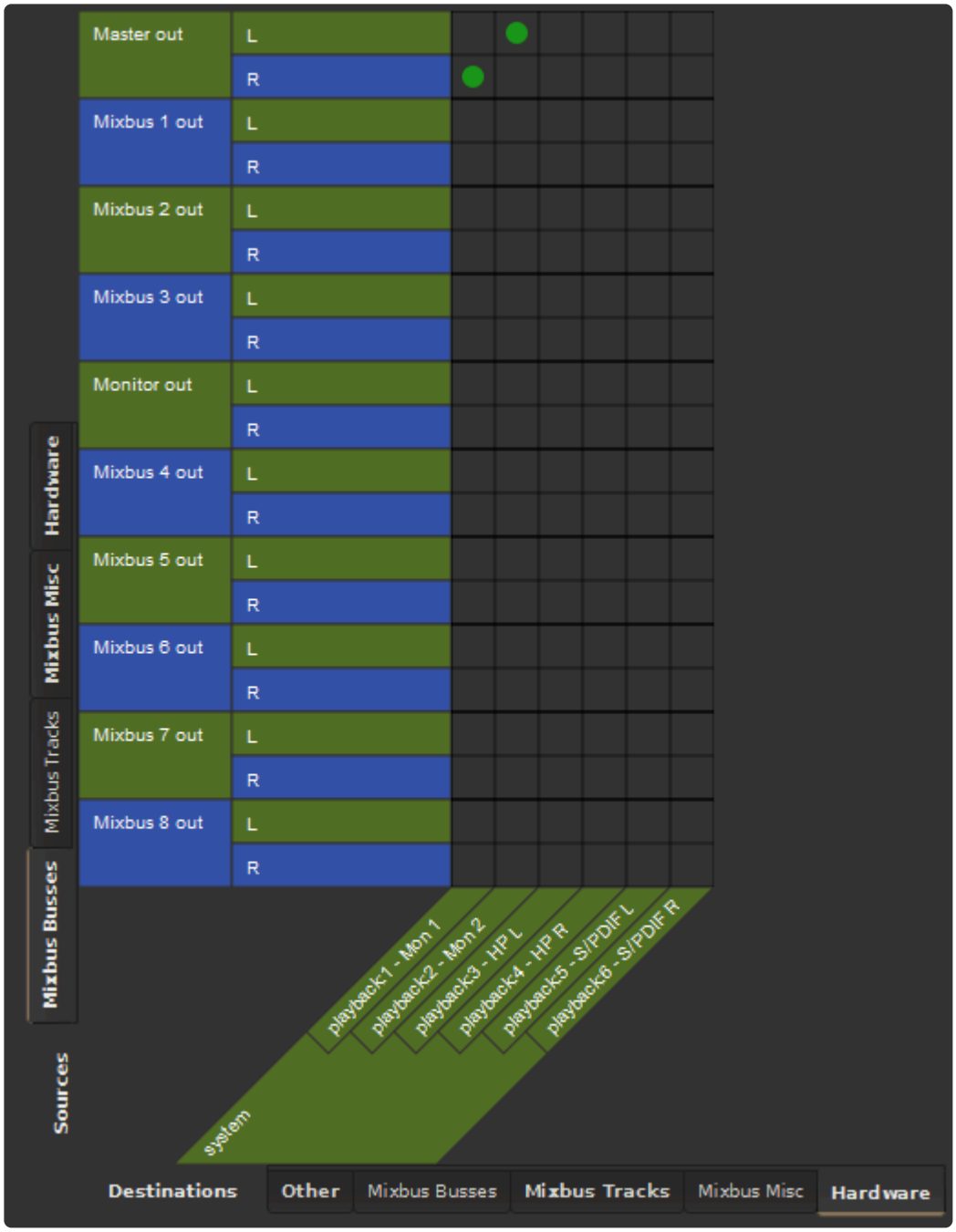
The Audio Connections Manager window allows you to connect any output port to the track/bus input:



In the above image an audio interface with 16 inputs is shown (viewed via the hardware tab on the left side). The bottom tab labeled "Mixbus Tracks" shows a total of 4 tracks in the session. The green dots indicate Mic1 is connected to the Mixbus channel strip input labeled VOX and Mic 2 is connected to the Mixbus channel strip input labeled GTR. The other two channels labeled keys and drums have no connections because in this example those tracks had been recorded previously. This is an example of having drums and keys already recorded and a vocalist and guitar player coming to the studio for an overdub session.

Note: audio connections use a floating-point 32-bit format, which means that headroom is essentially unlimited. You don't have to worry about "clipping" the sound when routing from one application to another. You may route a single output port to several inputs, and an input port can be routed from several different outputs (the summing of these signals is done automatically).

The image below shows the output of the Master bus. You can use the grid to connect the master bus to your speakers, and a stereo bounce track, and any other connections:



5.10. Transport (Play, Stop, etc) >>

5.10. Transport (Play, Stop, etc)

“Transport” is the term used to describe the motion of the playhead. Mixbus can play forwards, backwards, and at various speeds. Furthermore, you can instantly “locate” the playhead wherever you like. All of these functions are considered Transport functions.

Transport is a fundamental operation, so there are many different ways to control the transport:

- Buttons in the Transport Toolbar
- The Transport menu
- Keyboard Shortcuts
- Remotely, from an external device

…and probably many more!

> All of the operations in the Transport Menu can be assigned to a keyboard shortcut, so advanced users will most often use keyboard shortcuts to navigate the timeline and control the transport. But you can use the menu actions, and the buttons in the toolbar, until you’ve mastered the keyboard shortcuts

5.10.1. Transport Toolbar

Transport Toolbar:

The Transport Toolbar resides at the top of the main window. The transport buttons (far left) are always available; many of the other elements can be shown or hidden from Preferences->Appearance->Toolbar

Transport buttons:

The transport buttons control the playhead's location, speed and record-enable.

Transport controls are (from left to right):

MIDI Panic: Send note off and reset controller messages on all MIDI channels
Metronome: Enable/Disable audio click (see The Click to learn more)
Start: Go to session's Start marker
End: Go to session's End marker
Loop playback: Plays loop range
Play: Activates "play", starting from the current playhead location (shortcut: space bar)
Stop: Stops the transport (shortcut: space bar)
Record-enable: Toggles mater record-enable (you must also have tracks rec-enabled, to record on them)

Jog/Shuttle control: Allows for increased or decreased playback speed.

Right-click to adjust the speed display, mode, and max speed settings.

- Units:
  - Percent or Semitones.
- Mode:
  - Sprung: control returns to center position when mouse is released.
  - Wheel: control remains at its last location.
- Maximum speed:
  - adjustable from 1.5 X to 8X of "play" speed.

Time Sync Button: This button toggles between Mixbus's internal transport control, and synchronization to an external device (LTC (SMPTE), MTC (MIDI Time Code), or JACK). To learn more, visit: Clock

Synchronization

Record & Punch options:

To enable the “recording options” section of the toolbar, visit Preferences->Appearance->Toolbar and enable “Display Record/Punch Options”.

- “In” enables the punch-in marker (see Punch Recording)
- “Out” enables the punch-out marker
- “Non-Layered” is a special recording mode: when a new region is recorded, then all of the regions that existed below that range will be removed, and the last-recorded audio will be the only region in the layer stack. See: Object Editing

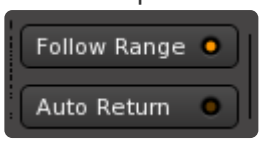
Monitoring options:

To enable the “monitoring options” section of the toolbar, visit Preferences->Appearance->Toolbar and enable “Display Monitor Options”.

- “Auto Input” toggles the status of the Auto Input mode. Learn more in the Monitoring chapter.
- “All In” implicitly forces all tracks to monitor the input, regardless of any other options or settings.
- “All Disk” implicitly forces all tracks to monitor the disk playback, regardless of any other options or settings.

Transport modes:

The Transport toolbar has 2 *mode* buttons which change the behavior of the playhead:



Follow Range Mode (previously called “Follow Edits” or “Join Playhead”)

Follow Range Mode is enabled via the horizontal button right of the transport control buttons. Follow Range Mode automatically moves the playhead to follow the mouse when it is clicked on a track, and/or to allow quick auditioning of the selected time range.

Behaviors with Follow Range Mode on:

- Starting playback (with the spacebar or any other means) will play the Range selection, if there is one.
- A single click in Range (or Smart-Range) mode will locate the playhead to that location.
- When a Range selection is made, the playhead will be located to the beginning of the range.

Behaviors with Follow Range Mode off:

- Editing clicks (such as object grabs or range selections) will have no effect on the playhead location.

Auto Return Mode

- If Auto Return is **on**, then the playhead will always return to the location where playback started.
- If Auto Return is **off**, then when the playhead is stopped, it will remain in the location when “stop” was pressed.

> Tip: The combination of Follow Range Mode and Auto-Return mode provide a wide variety of possible operations. If you are familiar with another workstation’s conventions, you should experiment with these settings to see if one of them matches your expectations.

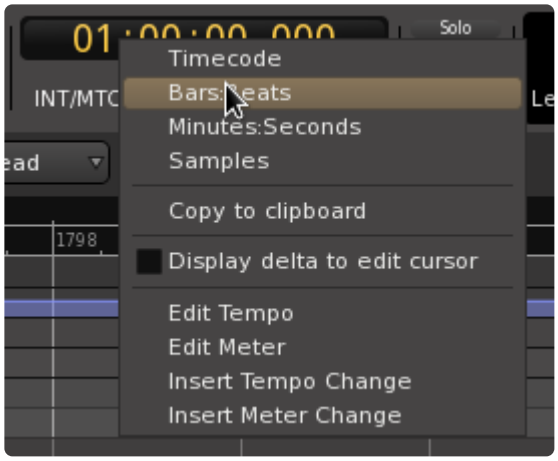
Clocks

By default, Mixbus provides 2 clocks at the top (Note: if your screen is small, Mixbus will automatically omit one of the clocks). You can manually disable the secondary clock by visiting Preferences->Appearance->Toolbar

The clocks have identical operation, and each clock may operate in one of 4 different modes:

- Timecode
- Bars:Beats (musical time, or sometimes called BBT for Bars-Beats-Ticks)
- Minutes:Seconds
- Samples

To select a clock’s mode, you right-click on the clock to present the clock context menu:



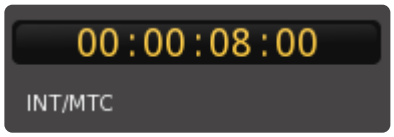
- Copy contents to clipboard: copies the clock text to the clipboard, for pasting elsewhere
- Display Delta to Edit Cursor: when this is enabled, the clock will show the “difference” in time between the playhead and the mouse cursor. This is a useful tool for measuring distances on the timeline.
- Tempo and Meter selections will launch the appropriate dialog. Learn more here: Tempo, Click & Grid

> In all clock modes, you can use the mouse-wheel to adjust the subdivisions of the clock. For example, in the Bars:Beats mode, you can mouse-wheel on the “Bars” value to increment or decrement the Bar value.

> You may also click on a clock to enter the timecode manually. In this case, a Right-to-Left convention is used. To locate to 1-hour in the Timecode clock, you would enter “1” followed by 7 zeroes.

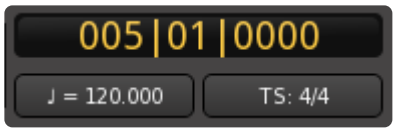
> You may also enter “relative” times, by first typing a “+” or “-” before entering the value to offset the current clock position.

Timecode Clock Mode:



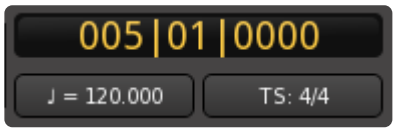
In Timecode mode, the display shows timecode in Hours, Minutes, Seconds, and Frames.

Bars:Beats Clock Mode:



In Bars:Beats mode, the clock shows position in Bars (i.e. Measures), Beats, and Ticks. “Ticks” are sometimes called “pulses per quarter note” or PPQN, and Mixbus uses 1920 Ticks per beat.

Minutes:Seconds Clock Mode:



In Minutes:Seconds mode, Mixbus shows the position in “wall clock” time.

Samples Clock Mode:

In Samples mode, the clock simply displays the sample-offset from time 0.

Pull-up/down values are displayed underneath the Samples clock. To change the pullup/pulldown value for the session, visit Session->Properties->Timecode

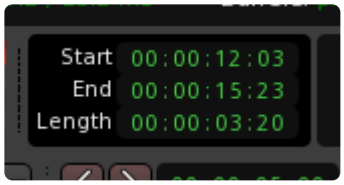
Alerts

These 3 buttons will blink “red” when one of the following occurs:

- **Solo** will blink red when any channel or bus is Solo’ed
  - This is sometimes called the “rude solo indicator”
  - Click the Solo button to clear all solos in the session
- **Auditioning** will blink red when an “audition” is occurring
  - An “audition” is when Mixbus is playing a file directly to the speakers, bypassing the mixer engine
  - Auditioning can be triggered (for example) in the Import dialog, if the user chooses to audition a file before importing it.
  - Click the Audition button to cancel the audition.

- **Feedback** will blink red when Mixbus detects a feedback loop in the mixer processing order
  - Each time the transport is “stopped”, Mixbus recalculates the latency offsets and other information for the mixer engine
  - If, when calculating the graph, Mixbus finds a “loop” or otherwise can’t determine the necessary latency offsets, this button will blink
  - If Mixbus can’t determine the engine offsets, then it will not play audio.
  - If this happens, you should consider the last few routing assignments, and consider whether you should revert one of your most-recent changes.

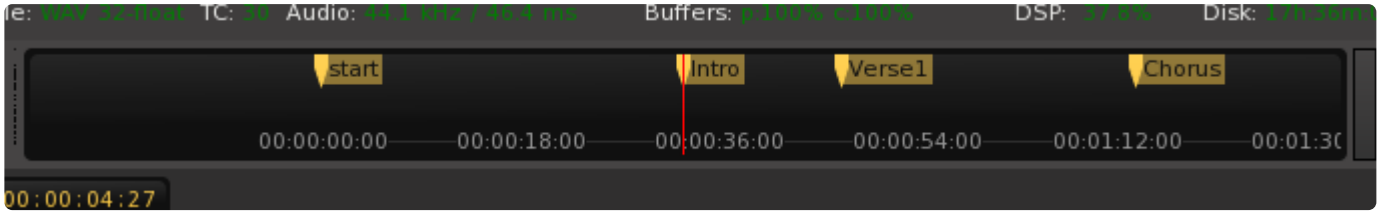
Selection Clock



By default, the selection clock is not shown. You can enable it in Preferences->Appearance->Toolbar.

If enabled, the “Selection clock” shows the start, end and length of the current selection. The selection clock works for a “Range” selection as well as a series of Region selections.

Mini-Timeline



The mini-timeline provides a compact and convenient way to navigate the session.

Learn more here: mini-timeline

Master bus meter

This simplified meter gives an indication of the master bus level. By default, this meter is a K-14 meter, using the same scale as the K-14 on the master bus, but shown in a different format.

You can adjust the “scale” of this meter in Preferences->Meterbridge->Default Type for Master Bus.

The vertical button on the far right will turn “red” when clip (0dBFS) is reached on the master bus. Click the button to clear the red indication.

Scripted Action Buttons

By default, the Scripted Action Buttons are not shown. You can enable them, in pairs, in Preferences->Appearance->Toolbar.

These buttons can be used to directly trigger one of:

- Actions (anything that can be bound to a key-binding). Select “Shortcut”, and then click “Add”. You will be prompted for which action you’d like to assign to the button.
- “Scripted actions”. Learn more in the Scripting chapter. Select from the available scripts in the menu. Some scripts even provide their own icon!

When a button has been assigned a scripted action, you can float your mouse over the button to display the action(s) in a tooltip.

If an action is not assigned to a button, it will appear dim and inactive.

Editor/Mixer page selection buttons

These buttons are used to switch “tabs” between the mixer and editor pages of the main window.

You can drag the Editor or Mixer button to your desktop to detach the page into its own window.

You can right-click on these buttons to detach or re-attach the editor/mixer windows to the main interface.

5.11. Navigating the Editor Canvas

Mixbus provides several ways to navigate the editor canvas.

Keyboard commands, buttons, the mouse-wheel (or 2-finger scroll on a touchpad), and scrollbar-like widgets are supported, and each method has its own strengths (and weaknesses).

Why is this so complicated?

Editing digital audio is not an easy task. Relatively few people have the interest, or need, to investigate all the details that go into a high-end professional workstation. It's a multi-dimensional problem with unlimited tracks, which might be audio, midi, or even video. And it's all changing on a timeline: a concept you won't find in a word processor or spreadsheet. Screen space is limited and there is a LOT of information that can be presented.

Users have different expectations, different workflows, and different devices (mice, touchpads, etc). Nearly every workstation has a different set of conventions, and if you've learned the conventions for one DAW, you might expect those same conventions to work on Mixbus; but alas there is not a widely-accepted way of working. (the same problem occurs in image-editing and other software … there are very few conventions that transfer from one system to another)

Furthermore, conventions that work on some software don't apply well to an audio workstation. Consider a scrollbar: you've probably found that a scrollbar works great for a spreadsheet until the number of items in the list becomes several hundred screens worth of data; then the scrollbar jumps around too fast to be useful. In an audio workstation, the timeline is potentially 24 hours long or more, and you can zoom in to see a screenful of individual samples. This can require upwards of 100,000 screens of information, which doesn't fit well on a scrollbar.

Learning to navigate the editor canvas

The best way to learn about Mixbus's conventions is by watching an experienced user – in person or via video. An experienced user will navigate the timeline effortlessly, using each of the available tools for their specific purpose.

These chapters provide an overview of the most common navigation features:

Scrolling and Zooming in the Editor Window

Basic Transport Features

Summary View

Mini Timeline

5.11.1. Scrolling and Zooming in the Editor Window

https://www.youtube.com/embed/3JY4mTboOow?rel=0

Scrolling

The editor canvas may be scrolled up/down and left/right to display a different selection of tracks or time on the timeline.

Vertical Scrolling

- Mouse Wheel: The primary method of vertical scrolling is the mouse scroll wheel (or the 2-finger drag, on a touchpad).
- Keyboard: The up/down arrow keys will scroll the view vertically
- Selections:
  - Alt+up/down arrow (Ctrl+arrow on a mac) will move the selection up & down, which will cause the editor to scroll to keep the selection in view.
  - Selecting a track header will cause the track to be scrolled fully into view.

    - Selecting a track name in the “editor list”, or the mixer window, will cause the selected track(s) to be scrolled into view.
- Dragging an object:
    - Dragging a region to the top or bottom of the screen will cause the canvas view to be scrolled
- Using the Summary View: Scrolling can be accomplished by grabbing the gray box in the summary view window and dragging it, or by clicking the arrow buttons at the left and right sides of the window.

Horizontal Scrolling:

- The primary method of horizontal scrolling is to move the Playhead (these operations require that “Follow Playhead” is engaged)
    - Click and release in the ruler-bar area to locate the playhead there;
    - Click & drag in the ruler-bar area to drag the playhead. When you approach the left or right edge, the screen will scroll to keep it in view.
    - The editor canvas can be scrolled horizontally by locating in the Mini-Timeline.
    - Various transport functions like “to start”, “to end”, or next/previous marker will move the playhead and therefore scroll the view.
- Keyboard Shortcut: the menu item View->Scroll Timeline can be assigned a keyboard shortcut to allow scrolling from the keyboard.
- Mouse Wheel: Hold “Shift” while turning the mouse wheel to scroll the timeline left/right.
- Mouse with side-scroll wheel (or Touchpads with a 2-finger side-to-side drag.)
- Dragging an object:
    - When you try to drag a region, MIDI note, or automation control point off the left or right edge of the canvas, the screen will scroll to keep it in view.
- Using the Summary View: Scrolling can be accomplished by grabbing the gray box in the summary view window and dragging it, or by clicking the arrow buttons at the left and right sides of the window.

> ✻ Power-User tip: zooming with the mouse-wheel is a ***very*** fast way to horizontally navigate the timeline. First zoom far out, until you can see the entire timeline. Then point the mouse where you’d like to zoom, and zoom in to the desired zoom scale at the new location.

Following the Playhead

In most cases, you’ll want the screen to automatically scroll sideways to keep the playhead in view. Mixbus has 2 settings in the Transport menu which control this feature:

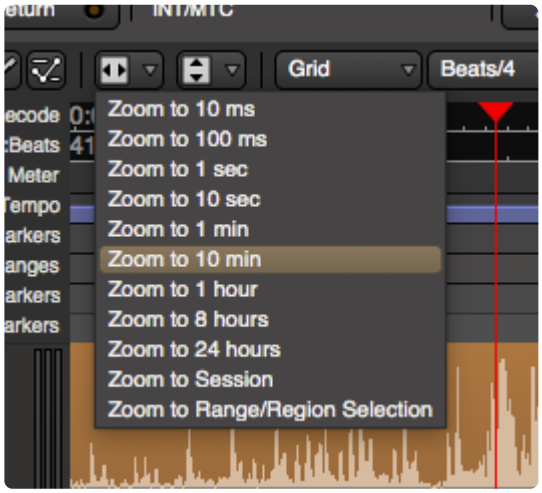
- Follow Playhead: when this is enabled, the screen will scroll sideways to keep the playhead visible, each time the playhead leaves the one side of the screen, the screen will move to show it again.
- Stationary Playhead: when this is enabled, *and* Follow Playhead is enabled, the playhead will remain in the center of the screen, and the waveforms will scroll underneath it.

Sometimes, you might prefer to scroll to a location, and prevent the playhead from moving your view. For example you might zoom in very close to make an edit, and you want to remain looking in the same place while the playhead plays over a selection. In that case, you can disable “Follow Playhead”.

Zooming

“Zooming” adjusts the scale of the timeline to show a greater or lesser portion of the timeline.

- Mouse Wheel: zooming with the Mouse Wheel is the preferred method:
  - Hold “Ctrl” (“Cmd” on Mac) to zoom in or out on the mouse’s location
- Zooming with the Summary view
  - Dragging up/down in the” summary view”:#summary-view will zoom the session.
- Zooming with the ruler-bar (playhead)
  - If the option “Preferences->Editor->zoom with vertical drag in rulers” is engaged, then dragging up/down in the ruler-bar will zoom around the playhead.
- Keyboard: + (plus) and – (minus) keys increase or decrease the zoom around the **Zoom Focus**. (see below)
- Zooming on a Range
  - Select a Range with the range tool (or smart tool) and then click “z” to fill the screen with the selected range
- View>Zoom Menu items:
  - Zoom to Session: Brings the entire sessions audio clips into view.
  - Toggle Zoom State: Alternates between the last 2 zoom states
- Zoom Presets widget: this button allows the user to select from predefined zoom scales:



Zoom Focus

Menu: View>Zoom Focus

Zoom Focus provides an anchor point by which keyboard zooming occurs around. For example, some users expect the screen to zoom towards and away from the playhead (this is the default).

Additional zoom focus options are available under View>Zoom Focus. They include:

- **Left** : leftmost side of the summary view is anchored
- **Right** : rightmost side of the summary view is anchored

- **Center** : center of the summary view is anchored and zooming occurs around it
- **Edit Point** : edit point is anchored and zooming occurs around it

Track Size & Waveform Scale

Track Size

Track heights can be adjusted using one of several methods:

- Grab the bottom edge of a track header to adjust its height with the mouse.
- Right-click on a track header, and choose the Height submenu
- Select a track and use View->Fit (shortcut: “F”) to fill the screen with the track contents
- Select a track, and use Track->Height from the main menu
- Track Height presets: Use the button in the editor to select predefined track views:

Waveform scale

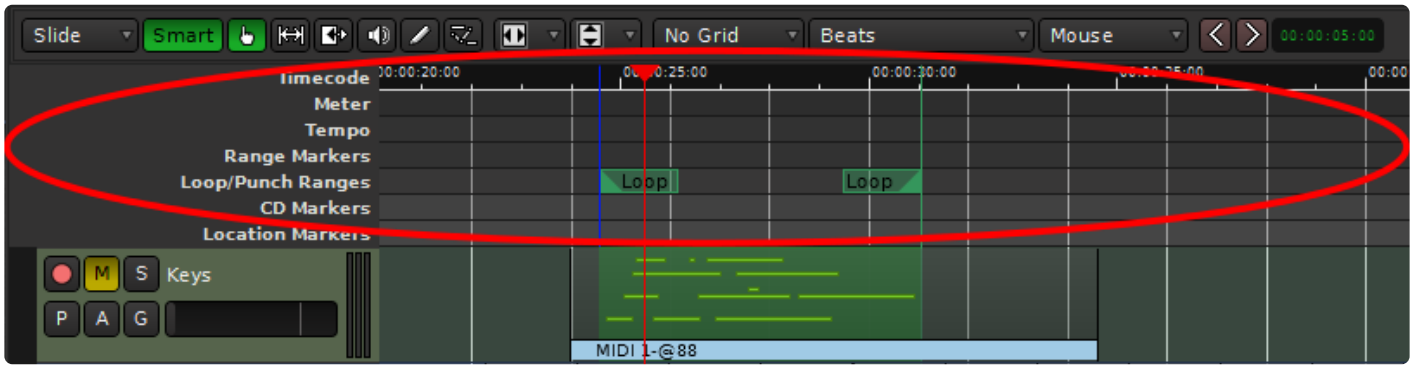
The waveform scale can be globally increased, so that quiet signals are more easily visible. Visit Preferences->Appearance->Editor:

- Linear: Displays the audio waveform in a linear fashion.
- Logarithmic: Displays the audio waveform so that quiet sounds can be more easily seen on the screen. The display is disproportionate. This setting does not change the gain of the signal, it is strictly a visual aid.

Additional waveform display options can be found in Menu>Edit>Preferences>Editor. Click here for more.

5.11.2. Transport, Markers, and Rulers

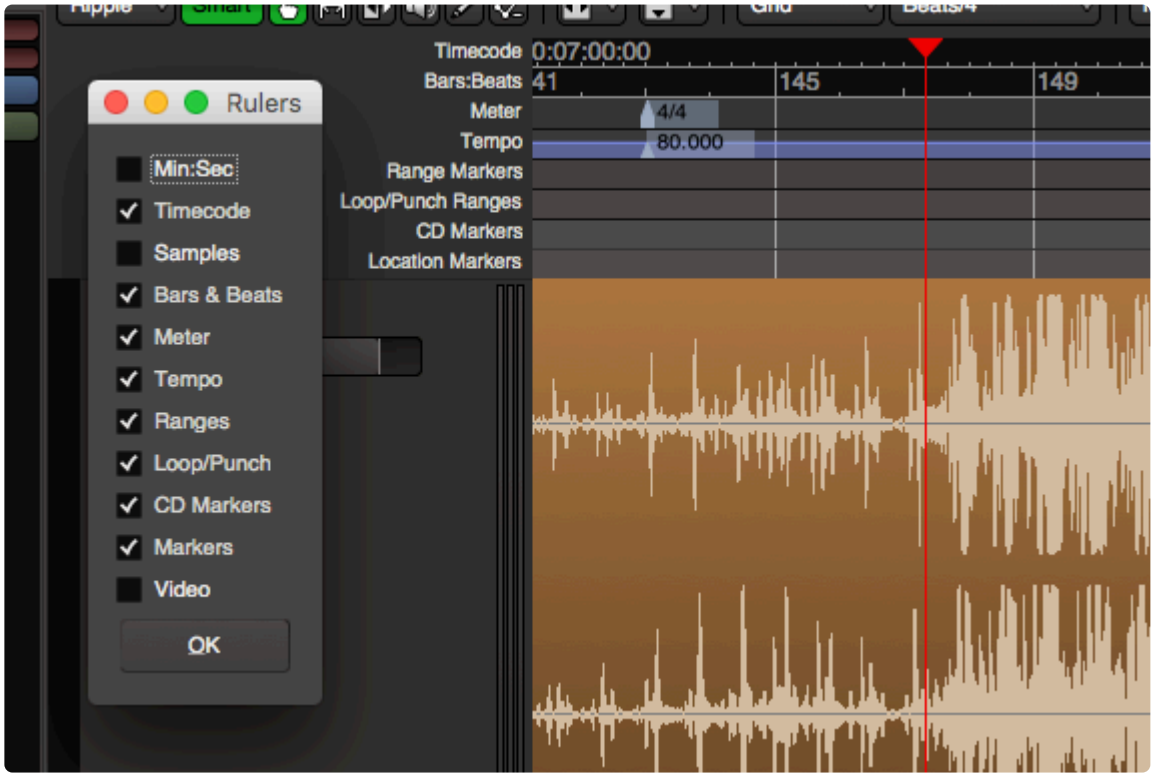
Ruler Bar



The ruler bar runs along the top of the edit canvas. A number of timelines can be shown via View>Rulers, or by right clicking in the blank area just to the left of the marker labels.

Available Rulers include:

- Min:Sec : Minutes and Seconds elapsed since time “0”
- Timecode : Video timecode in SMPTE formats. Choose the display format in Session->Properties->Timecode.
- Samples : raw digital samples
- Bars & Beats : musical time



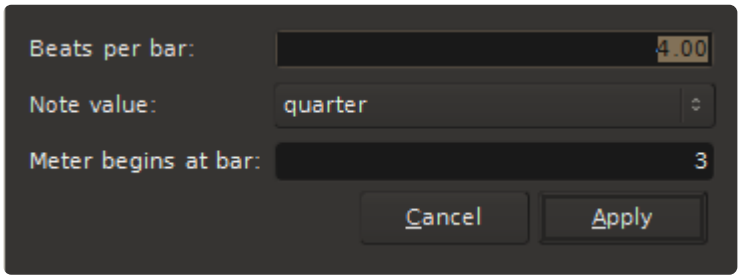
Marker bars in the ruler area

Marks can be set in the following ruler areas:

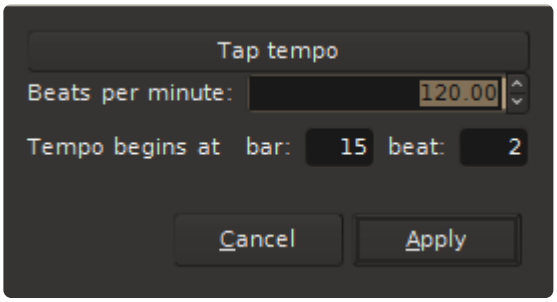
- Meter (a meter marker initiates a new musical timebase)
- Tempo (a tempo marker initiates a new musical tempo)
- Range Markers (Ranges can be exported simultaneously in the Export dialog)
- Loop/Punch Ranges
- Markers (Location markers are typically used to quickly locate to a new time in the project, such as “verse 1”).
- CD Markers (CD markers are used in the Export dialog to define CD track markers)

To set a marker in any of the rulers, simply right click to pull a dialog box.

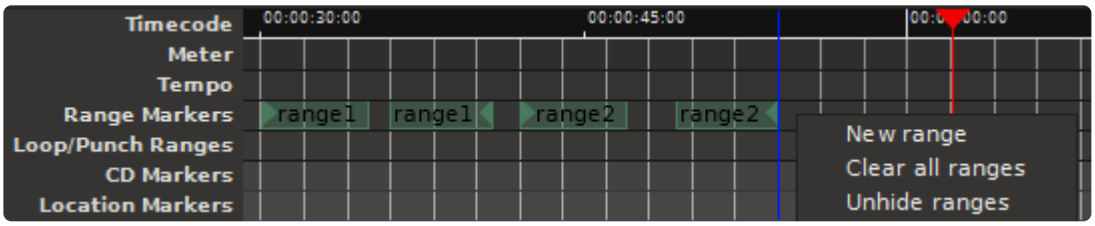
The Meter Marker:



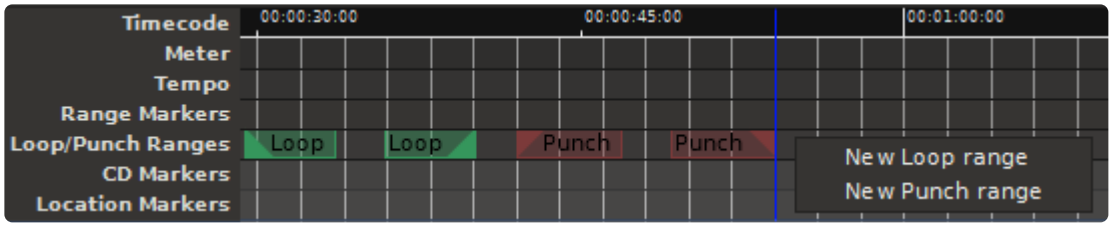
The Tempo Marker Dialog Box:



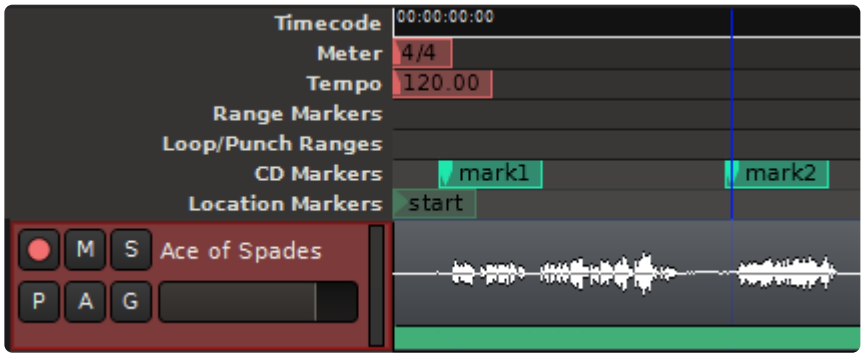
The Range Marker Dialog Box:



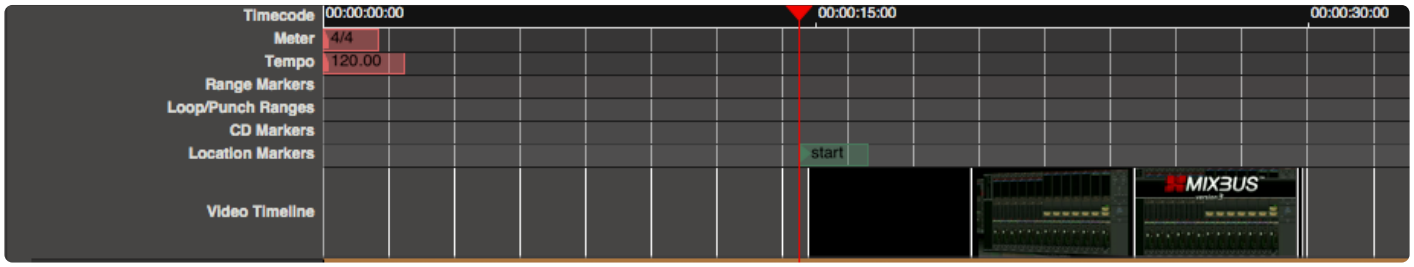
The Loop/Punch Ranges Marker Dialog Box:



The CD Markers Dialog Box:



Video Timeline Ruler



The “video timeline” ruler can show a thumbnail of a video, if you have imported a video file.

5.11.3. Summary View >>

5.11.3. Summary View

The “Summary View”, located at the bottom of the edit window, is a condensed view of the entire session between the “start” and “end” markers.

By default, the Summary view is not shown. But you can enable it by visiting the “View” menu.

Within the Summary view you can see a small representation for each audio/midi region on the timeline, and a lightly shaded box that indicates your current edit canvas view. You can grab the light shaded box to pan up, down, left, or right. And you can grab the edges of the light shaded box to stretch or shrink (zoom) in either dimension. Additionally, arrow buttons are provided at either end of the view box for maneuvering the shaded box if you prefer.

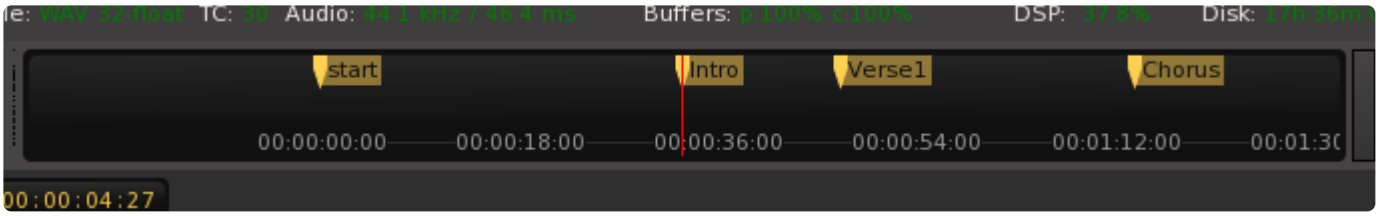
The height of the view box can be made larger or smaller by grabbing its top border and dragging.

In the image above three regions are shown with the shaded box fully encompassing the region contained in track 2, indicating it will be fully visible in the edit canvas. The start of regions 1 and 3 will not be visible but the regions ends will be.

5.11.4. Mini-Timeline >>

5.11.4. Mini-Timeline

The mini-timeline appears in the transport toolbar, and it shows a condensed view of the timeline including markers.



One of the benefits of the mini-timeline is that the scale remains fixed, regardless of the scale of your editor window. This allows you to zoom very close or very wide in the editor, and still view the markers and make 20- or 30-second jumps in the timeline.

Click on a marker to locate to the marker.

Click anywhere in the mini-timeline (except on a marker) to locate to that time.

> NOTE: unlike locating from the editor, the mini-timeline does NOT apply the Grid/Snap settings.

Right-click anywhere on the mini-timeline to change the zoom scale between several presets.

6. Recording

Quick-Start for recording: following these steps to record some audio tracks!

1. First, add one or more tracks via the main menu: Track>“Add Track or Bus”.
2. Record-arm the track by clicking the button with the red circle on the new track
3. Record-arm the transport by clicking the big button with the red circle, at the top of the main window (both the track(s) AND the transport must be armed before anything will be recorded)
4. Now press “Play”, and you will be recording! Rewind to the beginning of the track and press “Play” again to hear your recording.

https://www.youtube.com/embed/2LnjZC4UMAY?rel=0

You may arm tracks and/or the main transport *anytime*; so if you prefer, you can start playing first, and then press the record buttons to start recording.

Once you’ve recorded a track, you might investigate some ways to refine your recordings further:

1. Set the tempo (optional). Click on the Tempo button (in the main Clock) to change the current tempo. By setting the tempo, you can play the Click, and this will make editing the track much easier.
2. Change the way you hear yourself by visiting Monitoring.
3. You may choose any input to feed the track. Expose the “Input” button and choose the soundcard input you’d like to record to this track.
4. Use “Punch Recording” to pre-select a recording range, and Mixbus will automatically punch-in the

transport Record-arm when the punch-in point is reached.

5. Use “Pre-Roll Recording” to provide some count-in before your recording actually starts.

Recording Modes:

Mixbus can record in 2 different modes:

- Layered (default) : in this mode, any new recordings will be added as regions on a “layer” atop the existing audio regions in the track.
- Non-Layered: In this mode, the underlying audio will be replaced with the resulting new region.

Both modes are “non-destructive”, meaning that you can undo your changes, or re-layer and re-trim the regions to expose the parts that were hidden.

The mode switch is provided in the transport toolbar. If you don’t see this button, visit Preferences->Appearance->Toolbar and enable “Record/Punch Options”

To learn more about layers, visit Region Editing

6.1. Audio Tracks

What is a track? How do I create one?

A **track** is the fundamental building-block of a Mixbus session. You can create new tracks by creating empty tracks (using the New Track/Bus dialog) or by Importing existing audio files as new tracks. Mixbus supports audio tracks and MIDI tracks. The track type (audio or MIDI) defines what kind of data you will see in the editor.

What is an Audio Track?

An **audio track** is a series of audio regions, sequenced in a Playlist, that receive the same channel processing (the eq, compressor, plugins, panning, bussing, etc)

Each **audio region** (sometimes called “clip”) is representative of an audio file on disk. The region might represent only a small part of the audio file, because you can edit the region *non-destructively*: this means that whatever you do to the audio *region* in Mixbus, it doesn’t actually change the audio in the *file*.

You can record new audio into an audio track at any time. Because Mixbus is *non-destructive*, the previous audio on the track will be preserved, so you can always revert to the previous recording if you want to.

How do I edit an audio track?

The track’s Playlist defines the order of audio regions (files), as well as properties such as their mute state, volume levels, fade in/out, etc. So when you are editing a track with the editing tools, you are actually editing a Playlist. The various editing tools and features are described in the Editing chapters:

- Object (Region) Editing
- Range Editing

6.2. Monitoring

Monitoring

“Monitoring” is the process of hearing the prerecorded tracks (and/or hearing yourself) while the transport is playing. There are several options that control how Mixbus will monitor the inputs of each track.

The top of each channel in the mixer, when clicked, toggles between two views: signal path (inputs, outputs, monitoring, etc) and redirects (plugins, sends, inserts). Monitoring is controlled from the signal path view. When the signal path is showing, you’ll find selection buttons for “In”, and “Disk” at the top of each channel:

If the “In” button is selected as in the leftmost channel above, then the input to the track will be heard. If the “Disk” button is selected as in the second channel, then you will only hear the playback of the track. If neither button is selected, then Mixbus will use *automatic* monitoring to switch between In (as shown in the 3rd channel) and Disk (as shown in the 4th, rightmost channel). Automatic monitoring is represented with an outline around the currently-selected option.

Monitoring preferences

Mixbus has 2 global preferences that control *automatic* monitoring. You select the mode in Preferences->Signal Flow:

> **Audio Hardware Does Monitoring** This option is best used with audio interfaces that have the ability to monitor the inputs to the interface. Some interfaces come with a software application to control this while others may have a knob or button on the interface to allow direct monitoring. Other interfaces may need to be configured using the operating system’s volume control application. In this case Mixbus will *never* automatically monitor the input, because you are going to use your hardware device to hear yourself. When Hardware monitoring is available, it is generally a better option because it provides lower-latency monitoring. However it does not allow you to hear the effect of Mixbus’s processing or plugins, because you are bypassing the computer in your monitor path.

Mixbus Does Monitoring This setting is also known as “software monitoring”. With this setting, when Mixbus is recording audio it will route the inputs of the record-enabled tracks through the tracks instead of playing back the previously recorded audio. This is best used on systems that work well with very low buffer sizes (128 at 44 or 48 kHz or 256 at 96 kHz). Very small buffer sizes are used to minimize the monitoring latency.

Automatic monitoring is also affected by the “Auto Input” mode, which is found in the Transport menu. “Auto Input” is enabled by default. When Auto-input is enabled, then Mixbus will monitor record-armed tracks any time that you are aren’t playing.

Finally, there is an option to enable “Auto Input does Talkback”. If “Auto Input does Talkback” mode is enabled, then Mixbus *will monitor the input of an all tracks, even unarmed ones, during stop*. In a multi-room setup, this feature allows easier setup of your instrument or communication among members of a band, because all the live mics are passed through to the master bus output so everyone can hear themselves through the monitoring system (which might consist of foldback buses). This option only takes effect when Auto Input is enabled in the Transport menu.

Note: in versions prior to v6, the option was called “Tape Machine Mode” and it had the reverse effect as “Auto Input does talkback”. The option was changed to (hopefully) make it less ambiguous, and to change the default value so that Auto Input does *not* monitor all tracks, only the rec-armed ones.

Monitoring Toolbar

In Preferences->Appearance->Toolbar, you can enable the “Monitoring Toolbar” which provides quick access to the Auto-Input toggle, as well as buttons to globally force all unassigned tracks to Input or Disk monitoring:

If a track is already assigned to Disk or Input (i.e. the button is colored solid) then the monitor toolbar will have no effect on that track.

Mixbus Monitoring Modes Chart

The following chart shows how the monitoring interacts with various settings in Mixbus.

Monitor Mode	Is The Track Rec Enabled?	Is The Master Rec Enable On?	Auto Input Status	Transport Mode	Meter	Monitor

Normal	No	No	On	Stop	Input	Input
Normal	No	No	Off	Stop	Track (silence)	Track (silence)
Normal	No	No	On	Play	Track (Audio)	Track (Audio)
Normal	No	No	Off	Play	Track (Audio)	Track (Audio)
Normal	No	Yes	On	Stop	Input	Input
Normal	No	Yes	Off	Stop	Track (silence)	Track (silence)
Normal	No	Yes	On	Play	Track (Audio)	Track (Audio)
Normal	No	Yes	Off	Play	Track (Audio)	Track (Audio)
Normal	Yes	No	On	Stop	Input	Input
Normal	Yes	No	Off	Stop	Input	Input
Normal	Yes	No	On	Play	Input	Track (Audio)
Normal	Yes	No	Off	Play	Input	Input
Normal	Yes	Yes	On	Stop	Input	Input
Normal	Yes	Yes	Off	Stop	Input	Input
Normal	Yes	Yes	On	Play	Input	Input
Normal	Yes	Yes	Off	Play	Input	Input
Tape Machine	No	No	On	Stop	Track (silence)	Track (silence)
Tape Machine	No	No	Off	Stop	Track (silence)	Track (silence)
Tape Machine	No	No	On	Play	Track (Audio)	Track (Audio)
Tape Machine	No	No	Off	Play	Track (Audio)	Track (Audio)
Tape Machine	No	Yes	On	Stop	Track (silence)	Track (silence)

Tape Machine	No	Yes	Off	Stop	Track (silence)	Track (silence)
Tape Machine	No	Yes	On	Play	Track (Audio)	Track (Audio)
Tape Machine	No	Yes	Off	Play	Track (Audio)	Track (Audio)
Tape Machine	Yes	No	On	Stop	Input	Input
Tape Machine	Yes	No	Off	Stop	Input	Input
Tape Machine	Yes	No	On	Play	Input	Track (Audio)
Tape Machine	Yes	No	Off	Play	Input	Input
Tape Machine	Yes	Yes	On	Stop	Input	Input
Tape Machine	Yes	Yes	Off	Stop	Input	Input
Tape Machine	Yes	Yes	On	Play	Input	Input
Tape Machine	Yes	Yes	Off	Play	Input	Input

6.3. Loop Record

https://www.youtube.com/embed/VOiyDFv8zt0?rel=0

This short video demonstrates how to loop record.

6.4. The Click (Metronome)

The Metronome (or click) can be activated by toggling the Metronome button in the upper left of the edit window, or the associated keybinding.

Enabling this function allows Mixbus to generate a click that musicians can play along to. If a song is recorded to the “click” then editing can be eased with “snap” settings that automatically select snap to measures, beats, or subdivisions of a beat. (see Editing with the Grid 6.5. Punch Recording >> for more details)

Once enabled, a click sound will be produced when the transport is engaged. An audible click with an emphasis will be produced at the first downbeat of every bar.

> If you are recording to the click, you should take care to set the tempo for the song before you start.

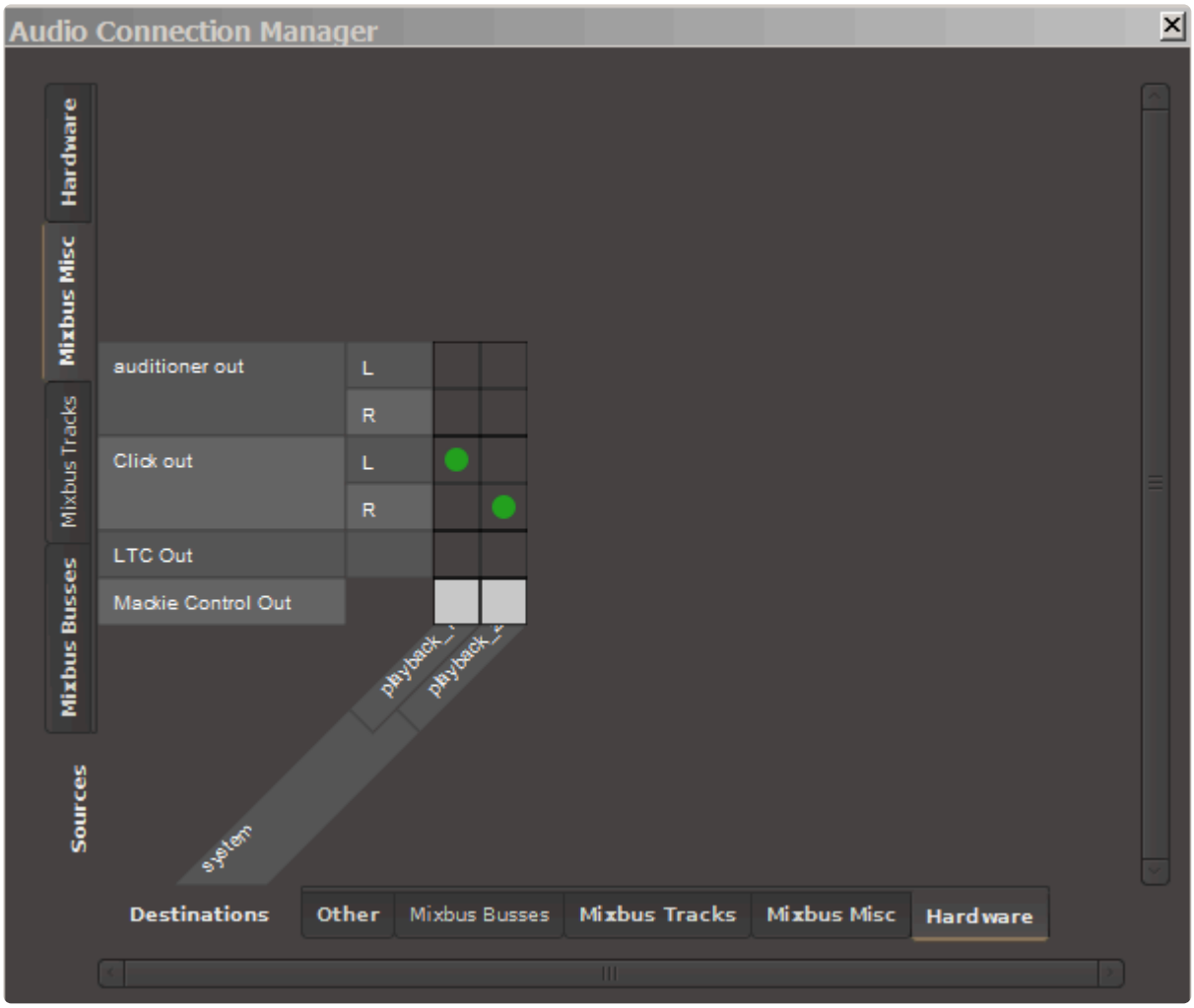
Right clicking on the metronome button is a shortcut to open the Preferences dialog box, where more Click settings can be found including the specific sound and volume of the click.

Mouse-wheel (or vertical scroll, on a touch-pad) on the Metronome button as a shortcut to adjust the click’s level.

> An audio file can be substituted for the default click sound.

By default, the metronome connects itself to the first 2 soundcard outputs, or into the Monitor if you are using a monitor section.

Sometimes (particularly if you switch the “monitor” section on and off), the metronome will lose its connection. To re-connect it, open Window->Audio Connections and restore the connection from the metronome to your speaker outputs, like this:


Audio Connection Manager
Hardware
Mixbus Misc
Mixbus Tracks
Mixbus Busses
Sources
auditioner out
L
R
Click out
L
R
LTC Out
Mackie Control Out
playback_1
playback_2
system
Destinations
Other
Mixbus Busses
Mixbus Tracks
Mixbus Misc
Hardware

6.5. Punch Recording

What is Punch Recording?

Punch Recording (sometimes called “Auto Punch”) is a method to let Mixbus automatically initiate recording during a part of the timeline.

A common usage of punch recording is the case of a person recording themselves, during an instrumental solo. The player would like to hear the audio playback up to the punch point, and then have Mixbus begin recording automatically when the solo section is reached.

How do I punch-record?

To enable punch recording, you must define a punch-range and enable the Punch In/Out buttons:

1. Pre select a time-range to record (example: the guitar solo) by dragging the mouse in Range mode.
2. Enable the punch markers by right clicking in the range and selecting “Set punch from selection”
3. Enable the Punch In/Out buttons in the transport toolbar. (If you don’t see these, you can enable them in Preferences->Appearance->Toolbar
4. arm the track(s) and the master record on the transport. As the transport rolls, recording will only occur as the playhead rolls over the defined area protecting existing audio before and after the range.

- If the “In” button is enabled, then Mixbus will initiate the transport Record-arm when the punch range is entered.
- If the “Out” button is enabled, then Mixbus will stop recording when the punch range is exited.

> It is a good idea to set your “punch” points slightly before and after the desired range; this way you will capture slightly more audio in case you start playing slightly early or late. You can trim the region(s) later.

Combining Punch Ranges and Loop ranges

It is sometimes useful to create a Loop range that is starts a few measures before the Punch range, and continues a few measures after it. Then when the “Loop” is initiated, you will be auto-punched with each pass through the loop, with some pre-roll and post-roll.

6.6. Pre-Roll Recording

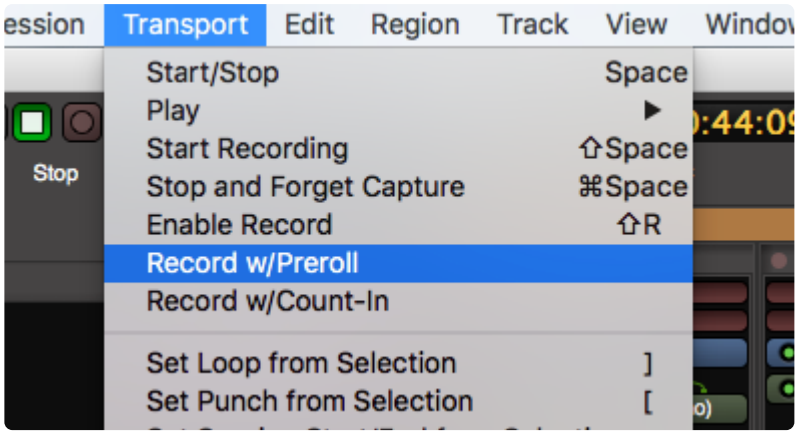
What is Pre-Roll Or Count-In Recording?

These recording modes allow the user to hear some prior material, or a count-in, before recording is initiated.

> Another benefit of recording with pre-roll is that Mixbus is actually capturing your signal during the count-in period. So if you start to record slightly early, you can advance the region's start time until your full performance is revealed.

How do I Record with Pre-Roll?

Record-with-preroll is an *action* that may be triggered from the Transport menu (you can also assign a keyboard shortcut to it).



> You can adjust the Pre-Roll time in Preferences->Transport. You can set the pre-roll time in either Seconds, or Bars (Measures)

When this action is initiated with the playhead at location "A", Mixbus will rewind the specified preroll amount, and play through it, before reaching the playhead location "A" where recording is to begin. The resulting region will start at "A", but you can trim the region back to expose more of the recording, if needed.

How do I Record with Count-In?

Record With Count-In is very similar to record-with-preroll, but there are 2 differences:

- You hear only the metronome click before recording starts, you don't hear the previous material
- The tempo of the metronome is that of the recording start-point, not the prior material

Use ***Record With Count-In*** if you want to record at the very start of the session, so you can hear a count-in before the music starts. Alternatively, you can use record-with-countin if you don’t want to hear the prior material and you want to “punch-in” after a countoff of the previous tempo

7. Using MIDI

What is MIDI, anyway?

MIDI is a wire-protocol that was developed in the 80's to allow synthesizers to send musical performance data between devices. So a MIDI keyboard can send performance data (i.e. "notes") to an instrument that actually creates the sound. It is beyond the scope of this manual to explain MIDI in detail. But a MIDI Primer has been provided here.

What can Mixbus do with MIDI?

Here are some of the tasks that are typically implemented with MIDI:

1. Play a multitimbral MIDI file through a General MIDI instrument, as a backing track. Learn more here: General MIDI
2. Add a Virtual instrument to a track, and record MIDI data to play through it. Learn more here: Virtual Instruments
3. Send MIDI to an external device. Learn more in Setting up an External MIDI device
4. Arranging tracks by assembling and triggering MIDI loops and phrases.
5. Using a MIDI as a controller (for knobs and faders). You can learn more about that in the External Control section

✻ Currently Mixbus does not have a musical-staff "notation" view. That might appear in a future version of Mixbus.

7.1. MIDI Tracks

https://www.youtube.com/embed/nMPqBaQOW-o?rel=0

What is a track? How do I create one?

A **track** is the fundamental building-block of a Mixbus session. You can create new tracks by creating empty tracks (using the New Track/Bus dialog) or by Importing existing audio files as new tracks. Mixbus supports audio tracks and MIDI tracks. The track type (audio or MIDI) defines what kind of data you will see in the editor.

What is a MIDI Track?

MIDI tracks are just like audio tracks, but they store MIDI performance data on the timeline, rather than audio regions.

Otherwise, MIDI tracks are identical to audio tracks; they can have MIDI inputs and outputs, audio inputs and outputs, plugins, and they have the full complement of Mixbus EQ/Compressor/bus sends. MIDI tracks are most often used to host virtual instrument (VI) plug-ins, which will render the MIDI data into audio which is then available to mix within the Mixbus mixer. It is also possible to route the MIDI data directly to a MIDI synthesizer device, and then patch the audio (if desired) back into the same MIDI track, thereby incorporating the external device as if it were a VI.

MIDI data may be generated by importing a MIDI file, by playing a MIDI instrument into the track, or by drawing notes with the Draw tool. Each time you draw or record MIDI notes and controls, Mixbus creates a MIDI region with a corresponding MIDI file inside the session folder.

MIDI editing was designed to operate with the same conventions as Mixbus's audio region editing. You can cut, copy, move and trim MIDI regions. And your operations apply to a Playlist which can later be reverted or changed, or even shared with other tracks.

Because MIDI data is always recorded into a MIDI file, this has some implications for MIDI continuous control data (CC's). You can only draw controller automation underneath an existing MIDI region. These controller moves are part of the MIDI region and will move with it. Note that you can still write fader, EQ, plugin and other mixer-strip automation linearly on the track, as these exist outside of the MIDI realm.

7.1.1. Adding MIDI tracks

Adding MIDI tracks

Importing MIDI files to a new track:

See Importing Files.

Creating a new MIDI Instrument track:

See Adding Tracks and Buses.

Listening to a MIDI track.

With MIDI tracks, it is fairly common for multiple tracks to be fed from a single MIDI keyboard or device. This adds an additional level of complexity to the task of “monitoring” the MIDI input.

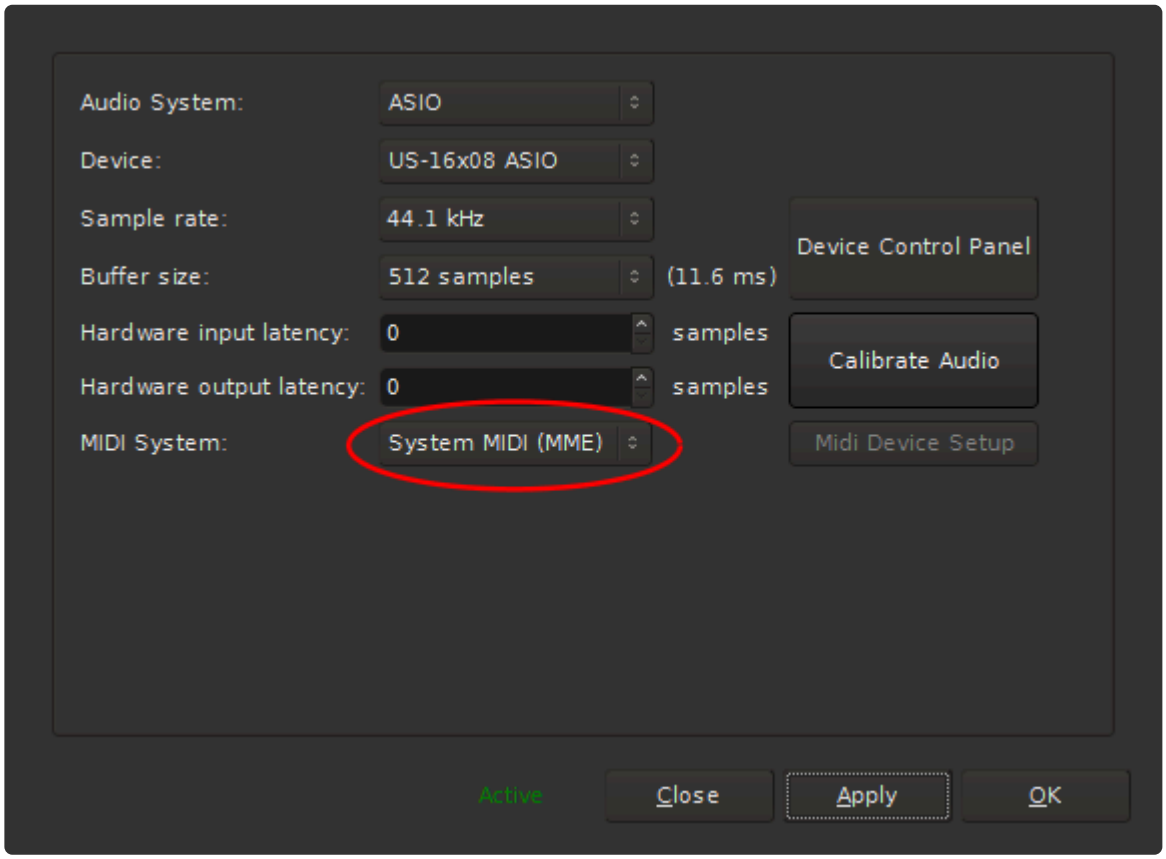
Once a MIDI track is added you will find a MIDI input Enable/Disable button near the top of the channel strip. If you want MIDI to be delivered to the track, the MIDI Input button must be enabled for that track.

7.1.2. Recording MIDI

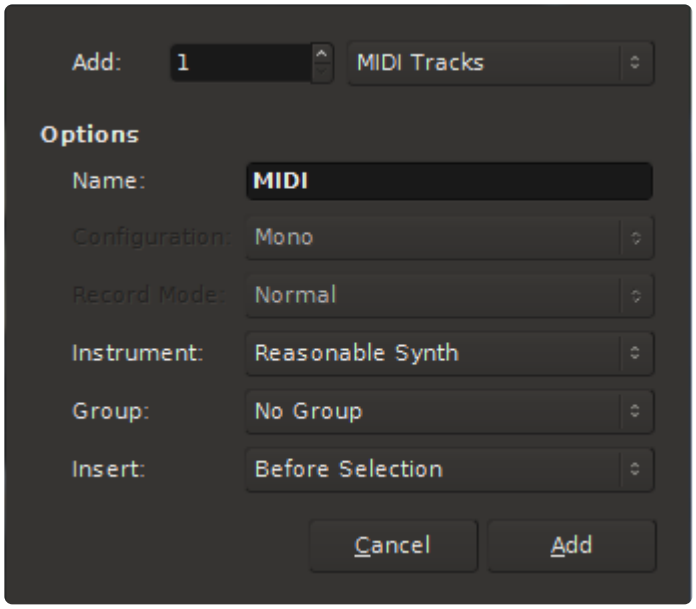
A step by step guide to record MIDI performance data.

1. Plug in your external MIDI equipment into the input of your MIDI interface or USB connector on your computer.

2. During session set up be sure system MIDI is enabled (CoreMIDI for Mac and MME for Windows).



3. Add a MIDI track(s) to your session. Click here for more on adding tracks. By default “Reasonable Synth” plug-in will be inserted into the redirect. You can select a different plug-in at this point or it can be changed after the track has been created. More on Plug-ins here



4. Route the system MIDI to the track input. In the image below the MIDI device happens to be called “Launchkey 25 capture”. You should look for the name of your device here.

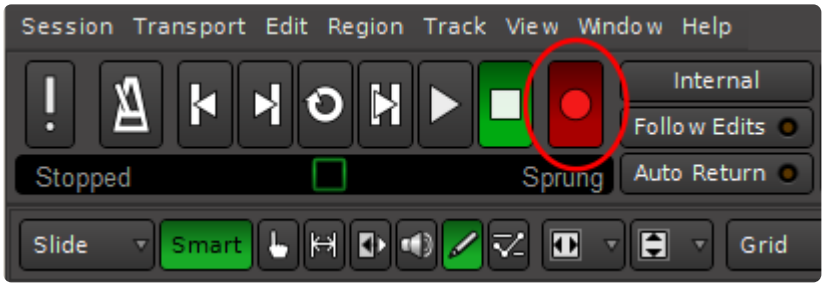
A convenient set up is to switch to the Edit window and turn on the Mixer Editor (Menu>View>Show Editor Mixer). This way you can manage the MIDI channel strip and view your performance as you record it. More on Editor Mixer here

5. Enable MIDI input. The default is on as shown in the image below. Click on the green MIDI connector to disable if you don’t want to hear the MIDI input.

6. Be sure to set the channel strip to monitor input.

7. Record arm the MIDI track

8. Record arm the master record button in the Edit window, roll the transport and record your performance.



> A short cut to arming the master in the edit window: rec arm the channel strip then press 3 on the number pad. This will record arm the master and roll the transport simultaneously. This is especially useful if you are managing the session while recording yourself.

9. Once a performance is recorded (or perhaps you have imported MIDI data to the track) you can send the audio output to a track to be recorded (bounced). Click the output connector (find the Output connector by switching the switcher at the top of the mixer strip) and route the MIDI output. In this image it has been routed to a stereo audio track (audio 7). Click here for more on recording.
Additionally you can route MIDI performance data to outboard equipment via your MIDI interface (many audio interfaces have a MIDI interface built-in).

You can also visit “Window->MIDI Connections” to see a grid of your track outputs (on one side) and hardware outputs (typically on the bottom right tab)

7.1.3. MIDI Track Operations

Right-click on a MIDI Track Header to initiate a pulldown menu of actions. Many of the actions are the same that are available in audio tracks and buses. But some items are specifically for MIDI:

Note Range:

- Show Full Range: The midi “scroomer” will be scaled to show the full range of MIDI notes 1..128
- Show Contents: The scroomer will be scaled to fit all of the existing MIDI notes in the track

Note Mode:

- Sustained: Notes will be represented as a rectangle, where the rectangle’s width displays the note start and end.
- Percussive: Notes will be represented as a “diamond” shape, showing only the note “start”. Percussive mode is useful when showing drum tracks and there is no “length” for each drum hit.

Channel Selector:
The MIDI Channel Selector Dialog will appear:

Select Patch:
Click the menu to send a Bank/Patch message for the selected channel. For more information, see: MIDI Banks and Patches

Color Mode Selection:

- Meter Colors: (*default*) The note color indicates velocity” of the MIDI notes.
- Channel Colors: The CHANNEL of the note event will be used to color the MIDI notes
- Track Colors: All notes will be colored with the track’s color

7.1.4. MIDI Filters

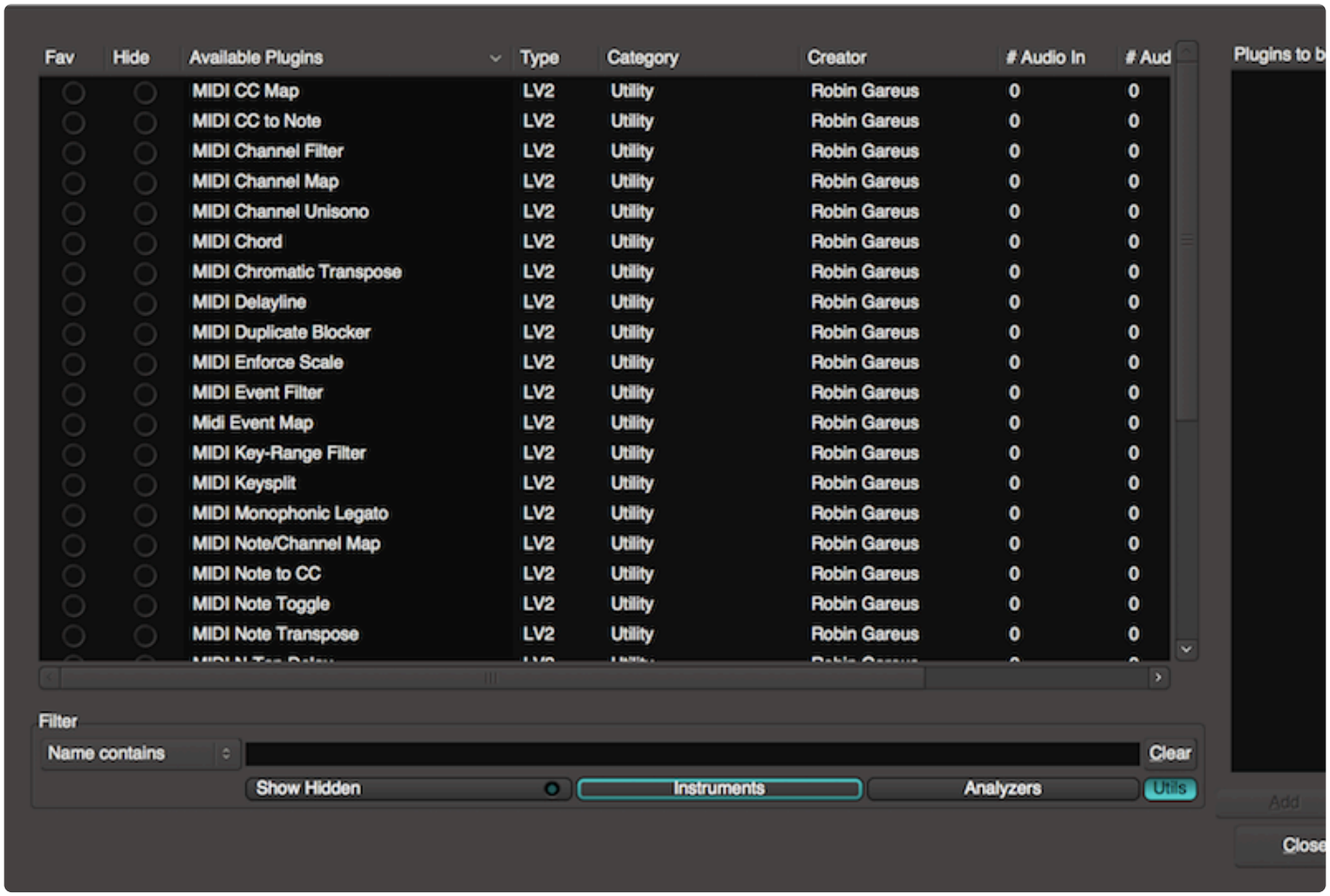
In addition to Virtual Instruments which accept MIDI notes and render audio, Mixbus also supports “MIDI plugins” or **MIDI filters**.

What is a MIDI Filter?

A MIDI Filter is a plugin that has only MIDI I/O, and no audio I/O. A MIDI filter can change the incoming MIDI stream (forcing all events to a given velocity, for example) or a MIDI Filter can generate its own MIDI stream (such as an arpeggiator or algorithmic-music-creation plugin)

How do I add a MIDI Filter to a track?

MIDI Filters appear in the Plugin Manager just like plugins. But by default they are not shown. To display MIDI filters, click the small “Utils” button in the plugin manager twice, so it is highlighted & filled:



What MIDI filters are supplied with Mixbus?

See List of Inlcuded MIDI Filters

It is also possible to create your own cross-platform MIDI filters using Lua scripts

7.2. Virtual Instruments

What is a Virtual Instrument?

Virtual Instruments (VI) are plug-ins that convert MIDI performance data (notes) into sound. Otherwise Mixbus treats them like a normal plug-in. They can be inserted anywhere in the channel's signal flow. This allows you to insert a MIDI-generating plug-in, or a MIDI "filter" plug-in before the VI. But normally, you should ensure that the VI is the first plugin at the top of the redirect list.

Mixbus supports four Virtual Instrument formats:

- VSTi (using the VST plug-in specification, the VST format has been extended to include MIDI note I/O).
- VST3 instruments
- Audio Unit Instruments – this is an Apple technology and is only supported on Mac OS X. Most plugins on Mac are available in both AU and VST formats.
- LV2 instruments – LV2's can be compiled for Mac, Windows and Linux. Currently there are no commercial LV2 instruments available, except those included with Mixbus.

How do I create a Virtual Instrument track?

When you create a new MIDI track, there is a menu selection for the Instrument you would like to use on this channel. By default, the instrument will be inserted at the top of the redirect list.

> ✻ It is also possible to create a MIDI track without a virtual instrument. In this case you will likely send the MIDI output to a hardware synthesizer, and return the signal into the MIDI track's audio inputs (or whatever track(s) you choose). See Setting up an external MIDI device

What can I do with a virtual instrument?

Just like a plugin, you may double-click on the VI to display its custom user interface.

You may add additional plugins before or after the VI. And you may drag the order of instruments and plugins. The signal flow through a channel is shown in the redirect box; Red lines indicate the MIDI path, and green lines indicate the audio path(s). Advanced users can investigate Pin Connections to further refine the signal flow through the channelstrip.

> ! In most cases, the fader, EQ, and Compressor elements should come AFTER (below) the Virtual Instrument in the channelstrip, so they can operate on the audio stream created by the VI.

> Note that because the VI is in the plug-in window, after the channel's input, the input trim is not effective on the VI. Most VI's have their own internal volume control which can be used as a trim.

Using a virtual instrument with multiple outputs (such as virtual drum instruments)

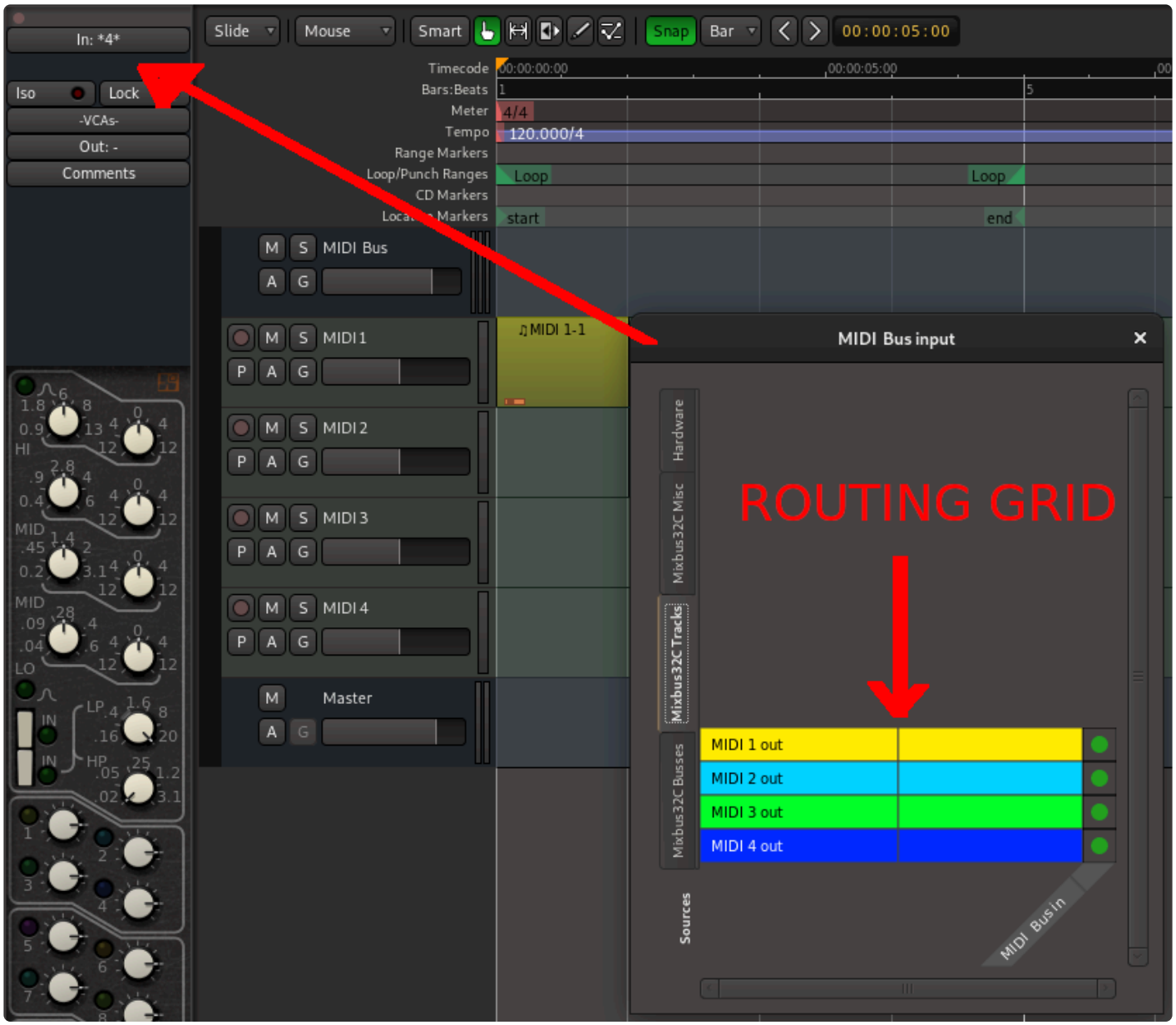
By default, Mixbus creates virtual instruments in their "stereo output" mode (or "mono" if that's the only option they provide). Some plugins provide multiple outputs; for example you can configure the kick drum to appear on a separate output than the snare drum. This is a fairly complicated topic for advanced users. See: Multichannel instruments

Feeding a virtual instrument from multiple tracks

In some cases, you might have multiple tracks with MIDI data, and you want all of the tracks to feed a single instrument. To do this, you should create multiple MIDI tracks, with no instrument. Then create a single MIDI "Bus" with the desired instrument. Use this image as a reference

In this image the 4 MIDI tracks are 'feeding' the MIDI Bus track at the top. The MIDI tracks themselves do not have instruments on them, but have MIDI data in regions on the track that will flow to it

Finally, connect the MIDI outputs of the tracks to the input of the MIDI bus. As shown in this image.



The MIDI "bus" will sum (merge) the inputs of several MIDI tracks, and provide the resulting MIDI stream to a virtual instrument.

> ✻ If your VI is multi-timbral, each MIDI Track can be set to output to one or more channels of MIDI by right clicking the MIDI Track and choosing the 'Channel Selector' menu option. There you can chose from one or more of sixteen MIDI channels.

7.2.1. Multichannel instruments

Some virtual midi instruments provide more than stereo outputs. This allows (for example) a drum instrument to provide a separate output for each virtual “microphone”.

Mixbus provides 2 multichannel “drum” instruments: “Black Pearl Drumkit Multi” and “Red Zeppelin Drumkit Multi”.

Some examples of third-party multichannel drum instruments include Native Instruments Kontakt, Toontrack’s EZDrummer, or Melda’s MDrummer.

The different plugin formats behave differently for multichannel outputs. AU instruments can have a variable number of outputs; while VST instruments, and Mixbus’s bundled LV2 instruments, pre-define how many channels they can support.

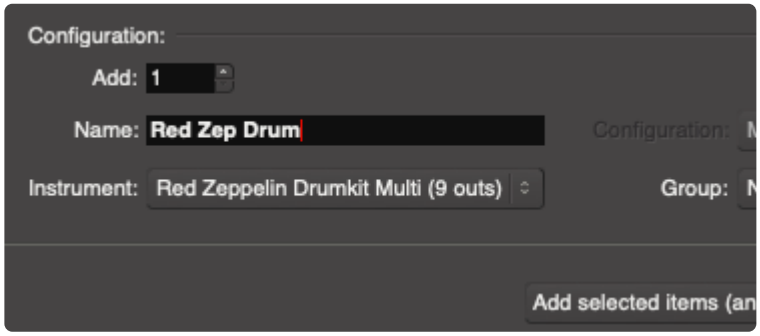
When a multichannel instrument is created, you want each sound to appear on its own mixer strip in Mixbus. Rather than make these connections yourself, it is much faster to let Mixbus do these connections for you. Mixbus calls this feature “Fan Out”.

✻ There is a conventional among VST and AU plugins that the “Fanned out” buses should be stereo pairs. Starting with v5.3, Mixbus creates stereo buses and connects them to each pair of instrument outputs. (the bundled Black Pearl and Red Zeppelin kits are smarter, and can create a mix of mono and stereo outputs)

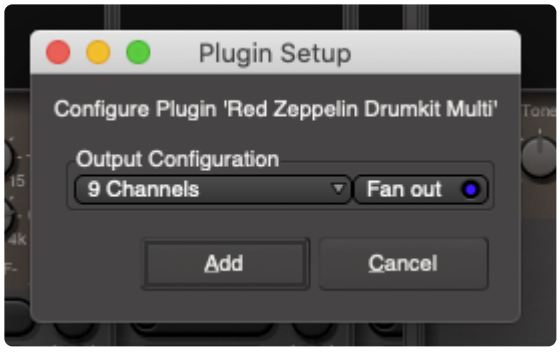
! If your instrument does NOT use stereo outputs (Addictive Drums is one example) then you’ll have to visit “Window->Audio Connections” and assign your outputs to both the left&right inputs of the desired track. Perhaps in the future we will add an option to create mono tracks instead of stereo tracks.

Setting up a multichannel instrument with fanned-out tracks:

To utilize these plugins in multichannel mode, it is necessary to initiate the operation from the “Add Track” dialog:



And then in the “Plugin Setup”, select ’9 Channels’ and ‘Fan Out’ (these are enabled by default)



…this will result in 9 additional tracks. In the bundled drum plugins, each resulting bus will be labeled with the drum it represents (kick, snare, etc). In third-party plugins, the resulting buses will be numbered 1….N.

> ! You’ll notice that the instrument track’s “MSTR” master-bus-assign button has been disabled. This prevents the first 2 outputs of the instrument being double-assigned to the master bus, because the first 2 outputs of the instrument are still playing through the strip.

> ! In versions before v5.3, Mixbus created “buses” instead of tracks. But it is desirable to “print” the drum tracks to audio, once the major arranging is done. This future-proofs the session, in case the instrument is no longer available. It also allows fine-grained editing, just like real drums.

Other operations and their effect on multichannel instruments:

- If an instrument plugin is drag-copied from one channel to another, it will be configured as a stereo-out plugin.
- If you drag an instrument from the “favorites” list in the mixer window, it will be configured as a stereo-out plugin.
- If the default stereo “fan out” isn’t desired, then disengage the “fan out” checkbox. The instrument will still be created multichannel outputs, but no resulting buses will be created. The user must manually create buses and/or tracks to receive the separate outputs of the instrument track.

Printing your tracks:

Once the arrangement is complete, you can print the instrument outputs by record-arming the tracks, record-arming the master rec-enable and rolling through your song. Once you’ve printed the tracks to audio, you’ll need to switch the track monitoring from “IN” to “DISK”. The fan-out process forces the monitor to INPUT so you can hear the tracks without having to rec-arm them.

(old version shown; some details have changed)

https://www.youtube.com/embed/VsKZQTa6W30?rel=0

7.3. Sending to an external MIDI device

External MIDI devices

In some cases, you might prefer to send MIDI from Mixbus's MIDI tracks to an external MIDI sound module ("synthesizer") rather than use a software Virtual Instrument inside Mixbus.

How do I send my MIDI to an external device?

Click the "Switcher" at the top of the mixer strip to expose the channel's input and output buttons. Click the "output" button and choose the physical MIDI port that you'd like to send out to.

> Alternatively you can set up the MIDI patching via the main menu, Window->MIDI Connections

How do I change the sound preset ("patch") of the external device?

See: MIDI Banks and Patches

7.4. Multitimbral and “General MIDI”

Mixbus can record, edit and render multitimbral MIDI files, often used for backing tracks or “sweetening” tracks to fill-out a live recording with additional instruments.

What is “Multitimbral”?

MIDI performance data (such as notes from a MIDI file) may be provided in up to 16 different channels on a single MIDI connection. Typically, each of the channels represents an instrument. A channel can change instruments anytime by sending a “patch change” message, which the receiving synth may (or may not) respond to.

At the start of the file, a “patch change” MIDI message will typically be sent for each of the channels in use, and these patch messages will define what instrument should be sounded on that channel … perhaps a saxophone, guitar, or piano.

What is General MIDI?

General MIDI is an agreed-on set of patches to render performance data with a variety of common instruments from various ethnicities. As MIDI can only send messages like “Channel 7, patch 21”, it was found necessary to define some instruments for these generic patch messages. In this way, a .mid file that follows the “General MIDI” specification can be reliably played in a synthesizer that is also General MIDI compliant.

For example, it is generally agreed that drums will be played on Channel 10.

> ✻ It is beyond the scope of this document to describe all of the conventions that have been developed for MIDI and Generic MIDI.

How does Mixbus implement Multitimbral and General-MIDI synthesis?

Mixbus is optimized for high-quality mixing of audio. Support for General MIDI, multitimbral MIDI, and backing tracks is not a high priority for us. Nevertheless, it is a feature that is sometimes desirable, and with Mixbus v4 we have added some mechanisms to import, edit, and render multitimbral MIDI data.

Mixbus provides 2 multitimbral synths:

- *a-FluidSynth*, a multitimbral MIDI-capable synthesizer that loads .sf2 soundfonts.
- *General MIDI Synth*, a simplified version of FluidSynth that automatically loads the *GeneralUser*

soundfont by S. Christian Collins. GMSynth provides a very thorough implementation of the General MIDI specifications.

What is a soundfont?

A “soundfont” is a file that contains audio samples and a “mapping” of each sample which defines the MIDI notes it will be played. Fully-featured soundfont players, such as Fluid Synth, provide full support for many MIDI features such as pitch-bend and modulation; as well as a built-in chorus and reverb to improve the sometimes lackluster sounds of General MIDI.

How do I use the General MIDI Synth?

To use the General MIDI Synth, simply add a new MIDI track (or Bus, if you intend to feed it from multiple tracks), and leave General MIDI Synth as the Instrument selection.

How do I choose a different patch/sound in the GeneralUser soundfont?

See: MIDI Banks and Patches

How do I use other soundfonts?

There are many Soundfont (.sf2) files online, from both commercial and “free” vendors.

You may download any .sf2 file, and use the selector menu inside *a-Fluid Synth* to navigate to, and load, the desired soundfont file. Note that the quality of sound fonts can vary widely, and you might need to check the soundfont’s documentation to learn which patches and note ranges are implemented.

Some soundfont files are optimized for General MIDI, and multitimbral operation. While other soundfonts are intended to provide only a single instrument, or several instruments, and do not implement the General MIDI specification. For example you can download a soundfont file that provides only a high-quality piano, a virtual orchestra, or a vintage electronic drum machine. A full investigation of .sf2 files is outside the scope of this manual, but you can find more information by searching the internet for “sf2” or “soundfont”.

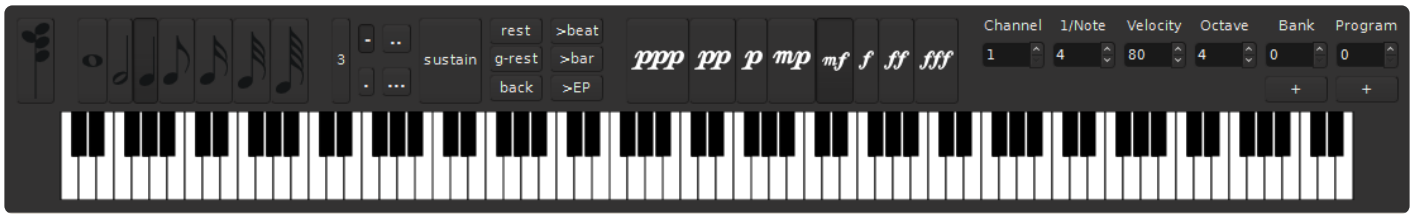
! NOTE: while searching the internet, you might find a wide range of free soundfonts. In our experience, the provided GeneralUser soundfont provides an excellently curated set of drum and instrument sounds. In many cases, we’ve found that free online soundfonts are only a subset of the patches that GeneralUser already has included. Furthermore, the curator has made an effort to verify that every sample is freely licensed for your use. For casual users, we highly encourage you to use the provided General MIDI Synth for free drum and instrument libraries. Professional users might want to purchase commercial players and libraries online from established vendors such as Kontakt, UVI, VSL…

7.5. Step Entry

Step entry of MIDI data is an alternative method of entering MIDI arrangements. In step editing, you set the length of notes, and the transport is automatically moved forward each time you play a note (or chord) on the keyboard. This is a fast way to enter a long sequence of chord progressions or a drum pattern. Or it can be helpful for users that are not proficient in keyboard performance, but would like to enter notes using the keyboard.

To initiate step entry, right-click on the track's "REC" button. A keyboard will be displayed.

Use your externally-connected keyboard to enter notes and chords; the step-entry dialog will automatically advance the playhead with each event. This is a fast way to enter a chord progression or drum parts.



7.6. MIDI Banks and Patches

What is a MIDI Bank or Patch?

A MIDI **patch** is a preset sound in a sound-generator (synthesizer). For example a keyboard might have a patch for “piano”, and a drum machine might have a patch for “jazz drum kit”. MIDI Patches are numbered from 1 to 128. Mixbus can send a “patch change” message to a device, to request the patch at that index.

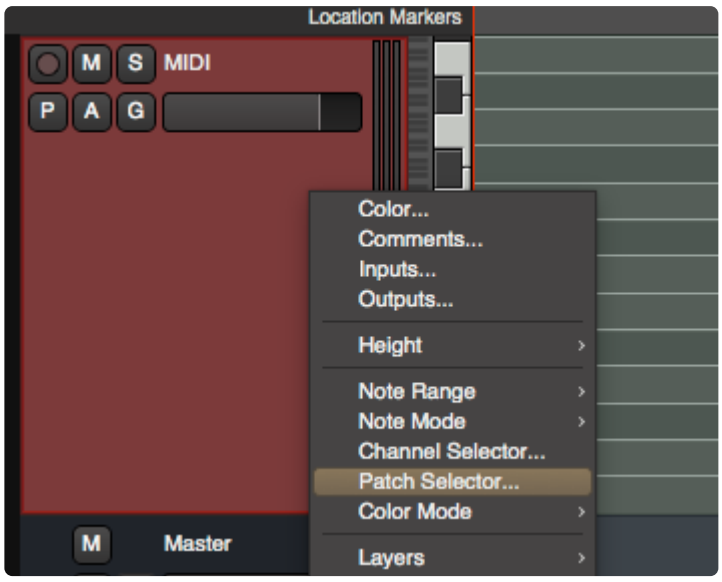
A MIDI **bank** is a mechanism to access additional sets of 128 patches. Banks were added to support devices with more than 128 sounds, or patches. Mixbus allows the user to select from 128 banks, and each bank allows up to 128 patches.

The MIDI specification allows up to 16 “channels” on a single connection, and each channel can be assigned a “patch” from one of the “banks”. However very few devices have a full set of banks and patches available on all 16 channels. You must consult your device’s manual to determine what banks and patches are supported, and how many channels can be used, and how many notes may be sounded simultaneously. (Mixbus’s built-in synth, GMSynth, has quite a wide range of patches & banks. Learn more here: General MIDI)

> With the advent of “virtual instruments” rather than external MIDI gear, bank/patch changes are much less common. Virtual instruments typically use their own GUI or preset mechanism to change sounds, not MIDI events. So if you are using Virtual Instrument plugins, you can likely ignore MIDI banks & patches.

How do I audition the patches in my external MIDI synth or VI plugin?

Right-click on a MIDI track header, and you will be presented with a list of actions:



Click “Patch Selector” to launch the patch selection dialog:

The patch selector has these controls:

- Channel selection: Each Mixbus track provides 16 MIDI channels
- Bank: If a patch map is provided for the track/instrument, you’ll see a list of available banks here.
- MSB / LSB : if no patch map is provided, use these selectors to manually select the MIDI bank values that are sent to the device.
- Patch buttons: if a patch map is provided, the available patch names will be shown here. If no map is provided, or if the map doesn’t specify a patchname for a position, then the button will show the patch number. * Audition on Change: when enabled, selecting any patch will trigger a series of MIDI notes to the synth.
- Start Note: If “Audition on Change” is active, this is the first note that is played.
- End Note: the last note that will be played.
- Velocity: select the MIDI “velocity” (loudness) of the notes sent to the plugin
- MIDI Keyboard: the keyboard can send MIDI notes directly to the instrument for auditioning (these notes are *not* recorded into the track)

Where do I specify the patch names that should appear in the patch dialog?

Mixbus’s built-in multitimbral synths (GMSynth and a-FluidSynth) automatically publish their patch information to Mixbus when they are instantiated on a channel.

Other commercial plugins normally provide their own patch management inside the GUI, and do not publish their patch settings to Mixbus.

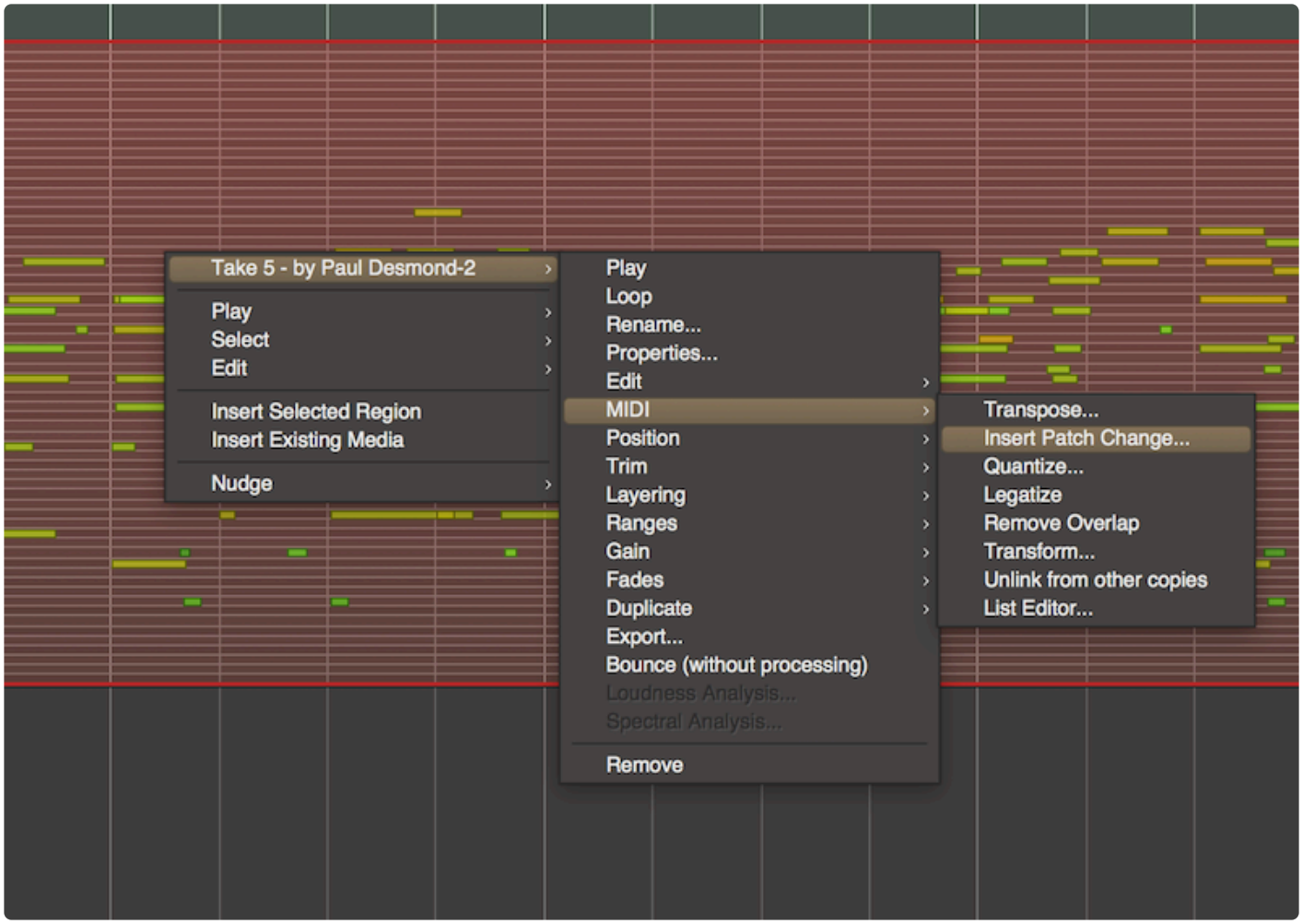
For tracks with *no* plugin instrument on the track, the “External MIDI Device” menu allows the user to define

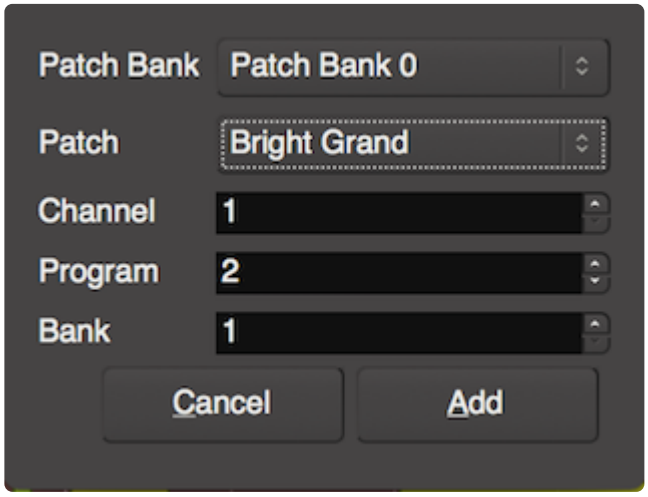
what external device is connected to the MIDI output:

Advanced users can add additional patch maps (.midname files) for their device. Editing .midname files is outside the scope of this document. Contact Mixbus support for more details.

Inserting a patch change during playback

In some cases, you might want to introduce a “patch change” during playback of a track. For example this might be desirable to change your “violin” track to switch to the “plucked violins”, temporarily. To insert a patch change event, right-click on a region, and choose “Insert Patch Change” from the MIDI submenu:





The dialog will prompt you for the bank and patch number. NOTE: if the patch doesn't appear where you expect it, select "Playhead" for the Edit Point, and use the playhead to define the patch change location.

Once inserted, you can use the Edit ("E") tool to move the patch change. Scrolling the mouse wheel over the patch change will increment the patch number. Use the shift+right-click convention to delete a patch change.

! In the current version of Mixbus (4.2), patch changes are only sent when they are “played through”. In other words, locating to a time after a patch change does not automatically send the patch. This feature might be implemented in a future version.

7.6.1. Adding MIDI Patch files

What is a .midnam file?

The .midnam file specification is defined by the MMA (MIDI Manufacturers Association) and is part of the official MIDI specification, to allow interchange of a device's patches & controller settings between different sequencers.

You can learn more information about these files here: https://www.midi.org/specifications/item/midi-in-xml

Where are Mixbus's .midnam files located?

The files are located in the Mixbus application folder, in a subfolder called "patchfiles". The location of this file varies with each operating system.

1. On MacOS, right-click on the Mixbus application, and select "Show Contents". Then navigate to Contents->Resources to find the patchfiles folder.
2. On Windows, Navigate to Program Files\Mixbus\.. and find the patchfiles subfolder.

How can I find a .midname file for my device?

In many cases, you can find a .midname file for your device by searching online. These files are sometimes provided by the manufactuer. Download the file and drop it into the patchfiles subfolder. Once installed, you'll see the list of patches in the patch-selector.

! NOTE: keep a copy of your new .midnam file, because when you re-install Mixbus, the .midnam file in the application folder will be overwritten.

* You can also remove unnecessary .midnam files, to reduce the size of the list in the menu. But the .midnams will be replaced when Mixbus is reinstalled.

If I can't find an existing .midname file for my device, how do I create one?

If you can't find a device file online, then you can make a copy of an existing device file, and edit it manually. The files are plain text, using the XML text format, and are fairly easy to read and edit.

1. Locate a .midnam file that is most similar to your desired device. (such as "Korg_Volca_Bass.midnam", if your device is a Korg)

2. Open the file in a **plain text editor**: most common errors occur when the editor tries to re-save the file in a different format.
3. Save the file with a new name. (such as, “My Korg Device.midnam”)
4. Edit the internal values of the file to meet your needs. For example, change the text of the author to reflect your own name.

```
<?xml version="1.0" encoding="UTF-8"?>
<!DOCTYPE MIDINameDocument PUBLIC "-//MIDI Manufactu
"http://www.midi.org/dtds/MIDINameDocument10.dtd">
<MIDINameDocument>
  <Author>David Robillard</Author>
  <MasterDeviceNames>
    <Manufacturer>Korg</Manufacturer>
    <Model>Volca Bass</Model>
```

The most important parts of the file are:

The device’s manufacturer and model name:

```
<?xml version="1.0" encoding="UTF-8"?>
<!DOCTYPE MIDINameDocument PUBLIC "-//MIDI Manufacturers Associ
"http://www.midi.org/dtds/MIDINameDocument10.dtd">
<MIDINameDocument>
  <Author>David Robillard</Author>
  <MasterDeviceNames>
    <Manufacturer>Korg</Manufacturer>
    <Model>Volca Bass</Model>
    <CustomDeviceMode Name="Default">
      <ChannelNameSetAssignments>
        <ChannelNameSetAssign Channel="1" NameSet="Names"/>
      </ChannelNameSetAssignments>
    </CustomDeviceMode>
    <ChannelNameSet Name="Names">
      <AvailableForChannels>
        <AvailableChannel Channel="1" Available="true"/>
```

The patches that are provided in the device:

```
        <ControlChange Control="32" Value="0"/>
      </MIDICommands>
      <UsesPatchNameList Name="Patches"/>
    </PatchBank>
  </ChannelNameSet>
  <PatchNameList Name="Patches">
    <Patch Number="Patches-32" Name="" ProgramChange="0"/>
    <Patch Number="Patches-0" Name="Grand Pno 1" ProgramChange="1"/>
    <Patch Number="Patches-1" Name="Grand Piano 2" ProgramChange="2"/>
    <Patch Number="Patches-2" Name="Stage Piano 1" ProgramChange="3"/>
    <Patch Number="Patches-3" Name="Stage Piano 2" ProgramChange="4"/>
    <Patch Number="Patches-4" Name="Bright Piano" ProgramChange="5"/>
    <Patch Number="Patches-5" Name="Sustain Pno" ProgramChange="6"/>
    <Patch Number="Patches-6" Name="Piano Str 1" ProgramChange="7"/>
    <Patch Number="Patches-7" Name="Piano Str 2" ProgramChange="8"/>
    <Patch Number="Patches-8" Name="Tack Piano" ProgramChange="9"/>
    <Patch Number="Patches-9" Name="Tite E Grand" ProgramChange="10"/>
    <Patch Number="Patches-10" Name="Bright E Grand" ProgramChange="11"/>
    <Patch Number="Patches-11" Name="Warm E Grand" ProgramChange="12"/>
```

The names for the MIDI controllers (CC's) that are supported by the device:

```
      <PatchNameList Name="User Patches"/>
    </PatchBank>
  </ChannelNameSet>
  <ControlNameList Name="Controls">
    <Control Type="7bit" Number="5" Name="Slide Time"/>
    <Control Type="7bit" Number="11" Name="Expression"/>
    <Control Type="7bit" Number="40" Name="Octave"/>
    <Control Type="7bit" Number="41" Name="LFO Rate"/>
    <Control Type="7bit" Number="42" Name="LFO Int"/>
    <Control Type="7bit" Number="43" Name="VCO Pitch 1"/>
    <Control Type="7bit" Number="44" Name="VCO Pitch 2"/>
    <Control Type="7bit" Number="45" Name="VCO Pitch 3"/>
    <Control Type="7bit" Number="46" Name="EG Attack"/>
```

7.7. Editing MIDI Regions

A MIDI region is exactly like an audio region, except that the information inside the region is MIDI performance data, not audio.

> In the same way that audio regions have underlying audio files that are actually stored on disk, a MIDI region has a MIDI file in the session that it represents.

MIDI regions may be cut, copied, pasted, and trimmed in much the same way as audio regions. Region editing is accomplished with the same tools: the Grabber (Hand) tool and the Range tool.

Grabber Tool

Range Tool

Editing features like the Grid, Sync Points, and Ripple Mode apply to MIDI regions just like audio regions.

There are 3 major differences with MIDI regions vs audio regions:

1. MIDI regions do not have fade in/out handles.
2. MIDI regions are “transparent”. This means that if you layer 2 regions atop each other, you will hear the notes of the region below. This allows you to overlap regions so you can sustain one note through the start of a new region.
3. By default, a copy of a MIDI region is a “linked” copy. So if you move the note in the source region, the same note will be moved in the copy (and vice versa).

MIDI regions do not only contain MIDI notes; they also contain performance information such as pitch bend, aftertouch, controllers, etc. This has one side-effect: in order to draw controls (with the Draw tool), there MUST be an underlying region. MIDI data cannot exist on a track without an underlying MIDI file, and this is represented by a midi region in the track. If you record MIDI data, then a region is created to hold any notes or controls that you play into the track. If you want to draw notes or controllers, you must first use the Draw tool to drag a region, and then you can draw notes or automation into the region.

Using the Grabber (“G”) tool, you can right-click on a MIDI region and navigate to the MIDI context menu:

- Transpose – transposes all the notes by a given octave/semitone offset.
- Insert Patch Change – Inserts a patch change event in the region.

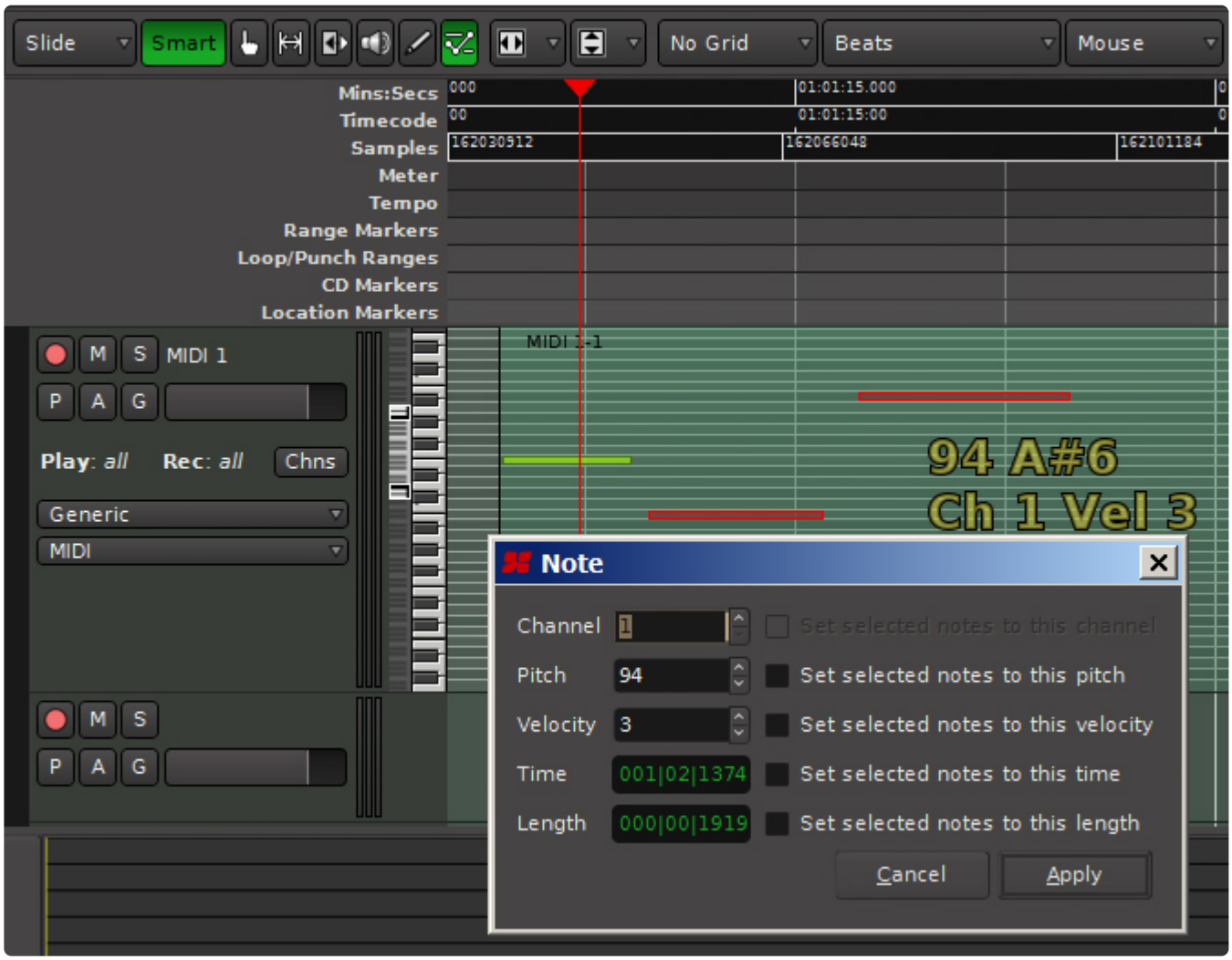
- Quantize – Quantizes the note start/ends to a defined grid.
- Legatize – Extends note ends to meet the next note's beginning.
- Remove Overlap – After this operation, each new note onset will cut off any notes that were held on before the new note was struck.
- Unlink from other Copies – unlinks this MIDI region from its copies. Any edits done to this region will no longer apply to the region(s) it was copied from.
- List Editor – shows MIDI notes in a text list. Each note can be individually edited in the list.

It was a design goal of Mixbus to edit MIDI regions "just like audio regions". To some users this is going to feel inelegant. To others it will be a breath of fresh air.

7.7.1. Editing MIDI Notes and Controls

In the image below: Highlighted in green is the Draw (“pencil”) tool, and to the right is the Edit (“points and lines”) tool. These tools are used to edit MIDI notes inside MIDI Regions.

In the image below: 2 MIDI notes have been selected for editing.



> Remember, the Grab and Range tools are used to edit Regions, while Draw and Edit operate on the contents of regions.

Draw tool	Edit Tool
• Click anywhere to start dragging a new MIDI	• Shift+right-click on a MIDI note to delete it

note; drag to extend the length of the note ◦ Newly-created notes get the average “velocity” of other nearby note velocities • Click in the middle of a note to select it (the note will turn red). Ctrl+click (Cmd+click on Mac) to add/remove notes from the selection • Shift+right-click on a MIDI note to delete it • Click the left or right edge of the MIDI note to trim the start and end times of the note • Click in the middle of a note to drag it • Shift+right-click on a MIDI note to delete it • Scroll the mouse wheel on a note to increase/ decrease the velocity	• Click in the middle of a note to select it (the note will turn red). Ctrl+click (Cmd+click on Mac) to add/remove notes from the selection • Click the left or right edge of the MIDI note to trim the start and end times of the note • Click in the middle of a note to drag it • Scroll the mouse wheel on a note to increase/ decrease the velocity • Click an empty space in the region to initiate a “box select” rectangle. With multiple notes selected, you can:

Once a note (or multiple notes) is selected (red), you can:

- Click Backspace or Delete key to delete the selected note(s)
- Click the up/down arrows to transpose the note pitch up/down
- Point at the selected note, and use the mouse wheel to adjust all of the selected note velocities
- Right-click on a selected note to open the MIDI context menu

MIDI context menu:

- Delete – Requires one or more notes to be selected. Deletes the selected note(s)
- Edit – Operates on a single selected note. Allows changing the length, pitch, velocity, etc.
- Legatize – Operates on a multi-note selection. Extends note ends to meet the next note’s beginning.
- Quantize – Operates on one or more selected notes. Quantizes the note start/ends to a defined grid.
- Remove Overlap – After this operation, each new note onset will cut off any notes that were held on before the new note was struck.
- Transform – Allows transforming the note pitch, velocity, and other parameters using logical operations.

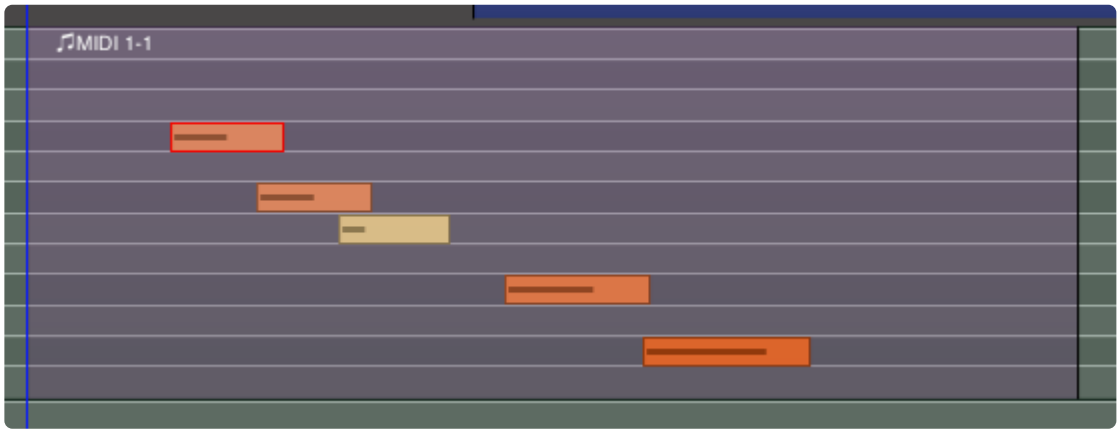
> If you want to apply these MIDI operations to *all* the notes in a region, you can use the Grabber (“G”) tool, and right-click on the region(s). The MIDI submenu allows you to apply these operations to all notes in the selected region(s), without having to select the notes separately.

MIDI Control (CC) Automation

MIDI Controls use the same conventions as automation, except that you can only have MIDI controller automation inside a MIDI region. You can only click to create new control points if there is a MIDI region on the timeline at that location.

MIDI Velocity bars

The note “velocity” may be displayed inside MIDI notes as a “bar”. See the image below:



Velocity is displayed as both a “color” change (increasingly saturated color with higher velocity) … and also with a bar indicator inside the note.

This feature can be enabled or disabled in **Appearance->Editor->MIDI Regions**

7.8. Virtual MIDI Keyboard

NEW in Mixbus v6 (both standard Mixbus and Mixbus 32C) is a virtual keyboard which can be used to provide MIDI triggers and controls. We doubt you could seriously play any music on it, but if you can you've got real talent.

To turn on the virtual keyboard, go to the Window pulldown menu and select "Virtual Keyboard". The keyboard will open in a new window, and is always on top of the Mixbus window (meaning, you can't hide or cover it).

Virtual Keyboard Controls:

- MIDI channel selector
- pitch bend slider
- 8 parameter controller knobs, each with selectable MIDI channel
- Transpose selector, with a range of +/- 12 notes
- keyboard layout button to select various optional mappings of the computer keyboard to the virtual keyboard (QWERTY, DVORAK etc). In the image above the mapping is set to QWERTY.

- Config menu providing the following controls:
  - velocity parameter settings
  - Octave and Range controls to select which part of the 0-127 MIDI range is covered by the keyboard.
    - Octave is self explanatory, which octave is the keyboard playing in, the possible range is

C-1 to G9
        - Range adjusts the size of the keyboard, from 2 octaves (25 keys) to 11 octaves (128 keys)

- Bank/Patch menu providing the following controls:
    - fill-in

8. Plug-ins

Plug-ins

> ✻ This chapter covers the general principles of plug-ins in Mixbus. To learn about using plugins in your mix, visit Using Plug-ins

Mixbus can load plug-ins in several formats, depending on your platform. Mixbus can only use plug-ins that meet these criteria:

- The plug-in must be a supported format on your OS (see below)
- The plug-in must be compiled for the same platform as the Mixbus you are running. (for example, 64bit Mixbus loads 64bit plug-ins, but not 32bit plug-ins)

> ! It is Harrison's intent to support every plug-in available, whether commercial or free. However there are many reasons why a third-party addon might not work in Mixbus, or might crash, or might behave unexpectedly. This is true of any DAW.

A summary of plug-in support in Mixbus:

Apple™ AU plug-ins (supported on OS X only)

AU (Audio Unit) is the native (Apple-developed) plug-in format for OS X. Most commercial plug-in developers make AU plug-ins for Mac, because this is the native format for Apple's Garageband and Logic.

Unlike other formats, AU plug-ins normally provide a "variable" number of inputs and outputs. The host (Mixbus) must tell the plugin how many inputs and outputs we want, and then the plugin can respond whether that is a valid choice. This can get quite complicated. Mixbus chooses the I/O based on the number of signal "pipes" at the insertion point of the plug-in.

Scanning of AU plugins must be initiated manually in Mixbus->Preferences->Plugins. If a plugin causes Mixbus to crash during the "scan" procedure, then this plugin will be placed on a "blacklist" which prevents that version of the plugin from scanning in the future.

Steinberg VST plug-ins (supported on all platforms)

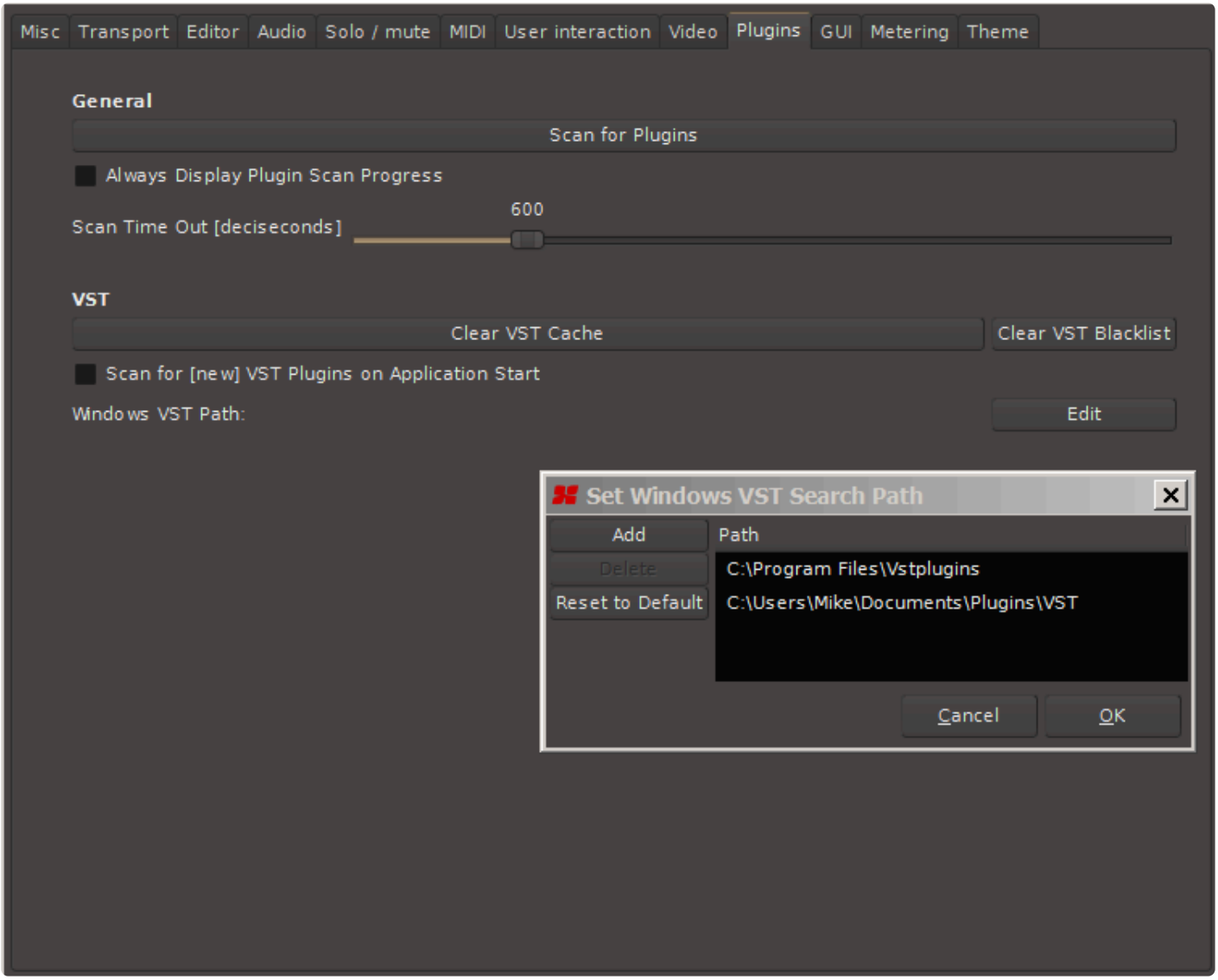
VST (Virtual Studio Technology) plug-ins were developed by Steinberg ™ and are the most common plug-in format on Windows. VST plug-ins can be developed for Windows, OS X, and Linux.

VST plugins typically come in 2in/2out versions (stereo I/O). Mixbus channels are typically mono or stereo; but the signal width can vary in the redirect sections. Mixbus will try to accommodate the audio and MIDI

“pipes” that are present in the signal flow at that location. For more details, see Processor Signal Flow.

Scanning of VST plugins must be initiated manually in Edit->Preferences->Plugins. If a plugin causes Mixbus to crash during the “scan” procedure, then this plugin will be placed on a “blacklist” which prevents that version of the plugin from scanning in the future.

Unlike AU and LV2, the VST specification does not specify where plugin files should be installed. When using VST plugins, you must take note of the folder location that the installer reports. You may find it easier to redirect the installer to one of the default locations that Mixbus uses. You can find these locations in Preferences>Plugins>VST>Path.



Or you can create a new folder of your liking and install all of your plug-ins there. This way Mixbus has to scan fewer folders. For every new folder created, Mixbus has to be made aware of its existence. If you have 20 different folders, then it places a burden on the user to inform Mixbus of their locations. Simplifying the plugins location upon installation may prove useful.

Steinberg VST3 plug-ins (supported on all platforms)

VST3 plug-ins were developed by Steinberg ™ as a successor to the VST format. VST3 plug-ins can be developed for Windows, OS X, and Linux.

VST3 improves on the VST2 spec with more flexible I/O choices, sample-accurate automation, logical parameter organization (for automation). https://www.steinberg.net/en/company/technologies/vst3.html

Unlike VST, it is not necessary to define the installation folder for VST3's. The installation path is predefined for all manufacturers.

LV2 plug-ins (supported on all platforms)

LV2 is the "next generation" open-source plug-in format. LV2 plug-ins have several advantages over other formats. For example, LV2 plug-ins do not have to be instantiated to be scanned, so startup is much faster and less likely to crash. For developers, LV2 is completely unencumbered by licensing restrictions. Harrison's bundled plug-ins currently use the LV2 format, and the same plug-ins operate on Windows, Mac, and Linux. This allows you to share sessions with users on other platforms, and have your sound remain unchanged.

LADSPA plug-ins (supported on all platforms)

LADSPA is the "Linux Audio Developer's Simple Plug-in Architecture". LADSPA plug-ins are simple to develop and are freely available. LADSPA plug-ins do not support customized GUIs. The "generic" GUI will provide sliders and knobs on a simple window. Being very portable, most LADSPAs are available for all platforms (OS X, Linux, and Windows). LADSPA makes an excellent starting point for developers that want to learn plug-in development. The quality can vary significantly, but there are a few good plug-ins available.

Mixbus can load a mono LADSPA plug-in into a stereo channel (by duplicating the plug-in and assigning the same controls to both sides of the signal). This works well for some plug-in types (such as EQ) but less well for compressors, which might introduce a shift in the stereo image if they aren't truly stereo. In these cases, you should choose a stereo LADSPA, not a mono one.

Digidesign AAX/TDM/RTAS plug-ins (no support currently).

AAX, RTAS and TDM plug-ins are proprietary formats for Pro Tools ™. Mixbus does NOT support RTAS or TDM plug-ins. It is unlikely that Mixbus will ever support AAX, RTAS or TDM plug-ins.

Windows DirectX plug-ins (no support currently)

The DirectX plug-in API was developed by Microsoft and is the native plug-in format for Windows. Few professional plug-in manufacturers provide DirectX plug-ins, so Mixbus does not currently support DirectX plug-ins.

> Note: While we have taken care to follow the specifications for each plug-in format, plug-ins are developed by third-party vendors. This means that Harrison cannot take responsibility for plug-ins that don't work as expected. Please use caution when trying a plug-in for the first time. Verify that all the functions work as expected before committing to using a particular plug-in in a session.

Latency Compensation

Some plug-ins, particularly plug-ins that use hardware "accelerators" or convolution-type processing, can cause significant latency (time delay) from the input to the output. This can cause problems if the delayed signal is mixed with a signal that is not delayed. A long delay can cause the tracks to sound "out of time", while a very short delay can cause comb filtering ("phasiness"). Mixbus can automatically accommodate plug-in and insert latencies for you. To do this, Mixbus delays the non-affected tracks to match the track with the longest latency. Please be aware of these facts:

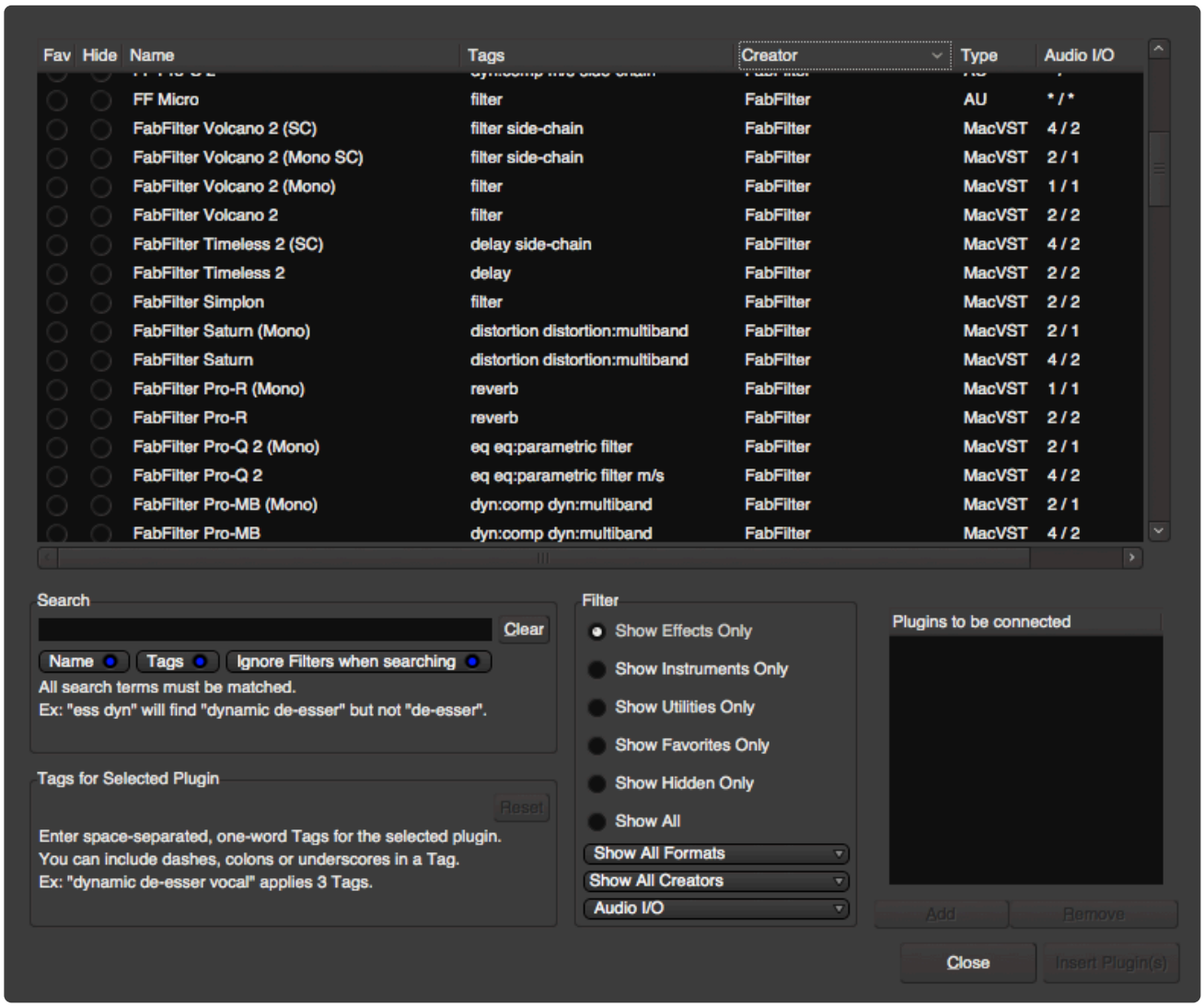
- Adding or bypassing a plugin can change its latency. Some plugins have internal controls (like "lookahead", "oversampling", or "quality") that change their latency.
- Latency offsets are recalculated when you "stop" the transport. So if you hear a delay or phasing caused by plugin latency, try stopping and restarting playback.
- Tracks can accommodate "unlimited" latency by buffering the audio from disk earlier than it is needed.
- Mix buses are limited to 8192 samples of latency compensation. If the total latency of all plug-ins in a mix bus exceeds 8192 samples, then a time offset will occur.
- "Buses" created with the Add Track/Bus dialog (we call these "utility buses") now support latency compensation.

8.1. Plug-in Tags

Tagging Plug-ins

Introduced in the version 5 update, plug-in tagging allows the user to append their entire collection of plug-ins with an arbitrary amount of tags to make plug-ins easier to find across all of Mixbus' interfaces.

The new Plug-in manager:



The new plug-in manager allows for plugins to be tagged, and searched via their tag, name, or creator (or a combination of all three) making it very quick to find plugins you need for your session. By default we take whatever the plugin reports as its category. However, for most plugin formats, these aren't very useful.

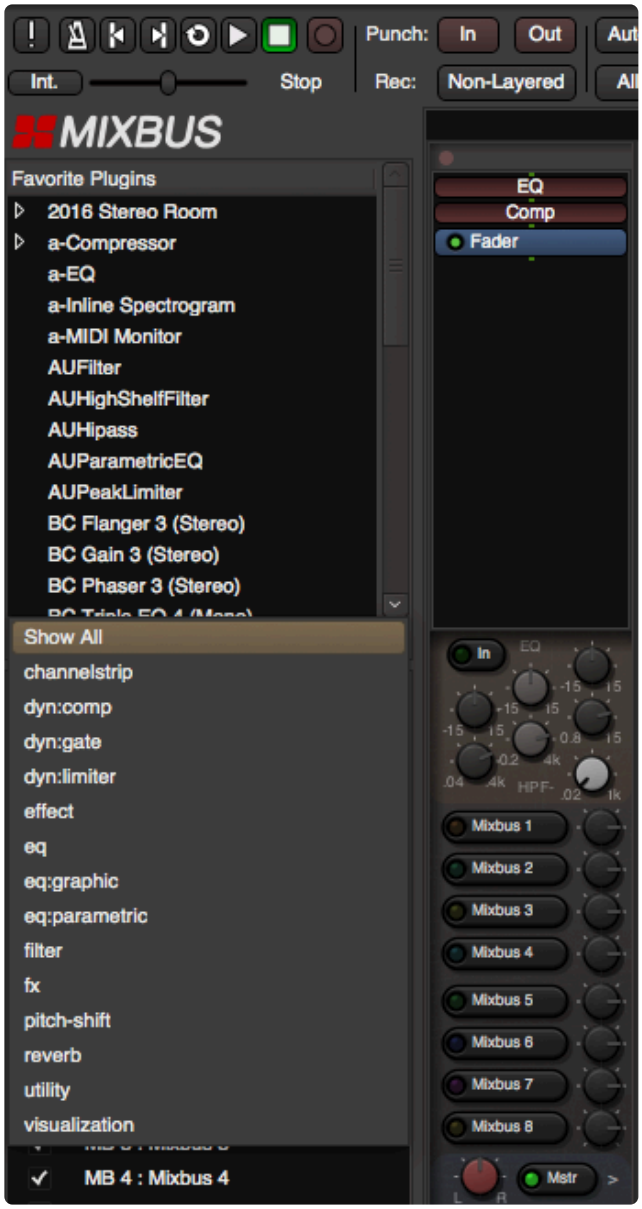
> We recommend tagging plugins in a style similar to that shown in the image above however, you can tag them in whatever way best suits your workflow.

To tag a plugin all you need to do is:

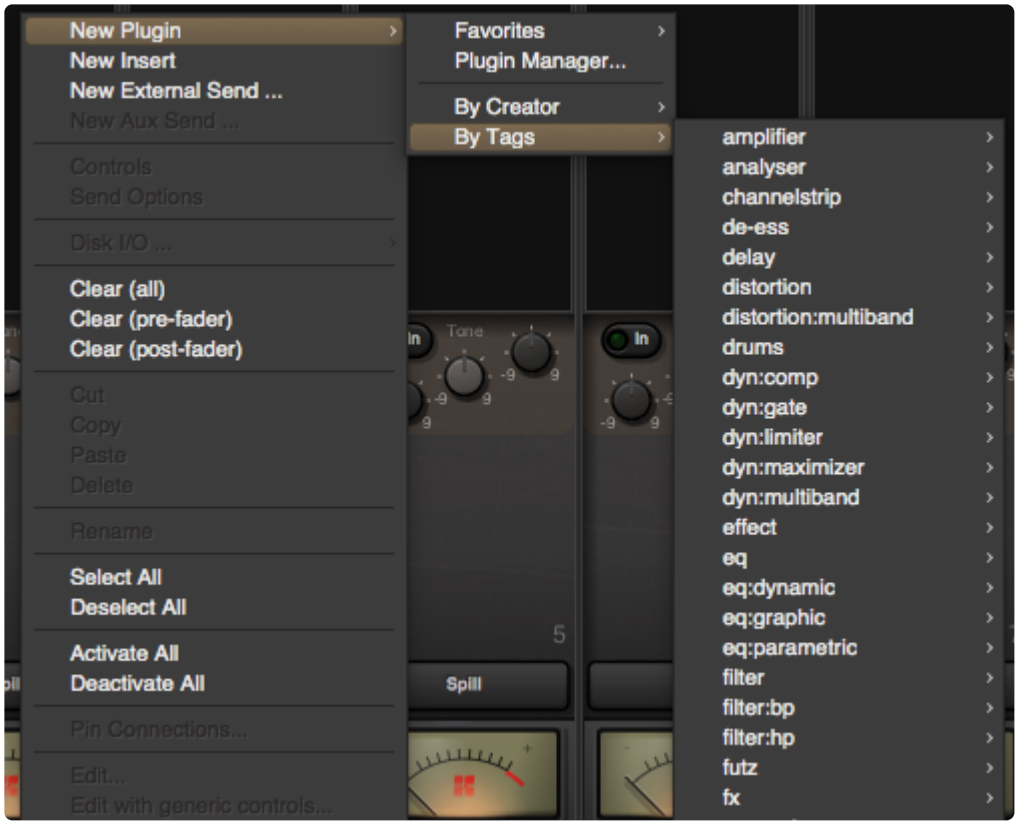
1. Select it in the plugin in the plugin manager
2. Erase it's contents
3. Replace it with whatever you want it to be searched by

Multiple tags can be included by adding a space after every word. For instance, "bass eq parametric" is three separate tags. Now that plug-in will appear in three separate locations *simultaneously*.

Additionally, plug-ins that appear in the favorites pane will now be able to be sorted by tags:



And they will appear in a strip's right click menu under plug-ins:



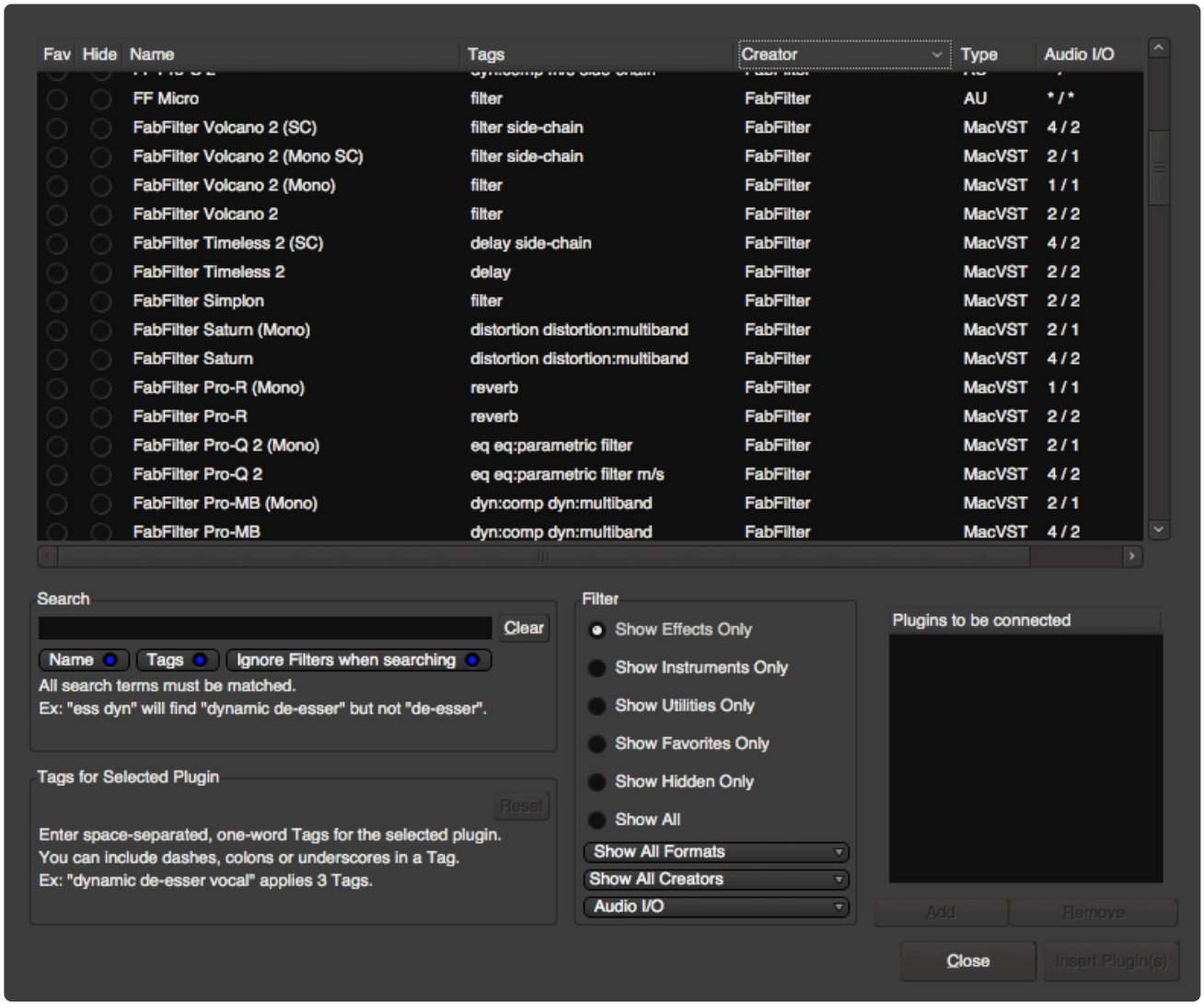
For information on Plug-ins click here, or on how to use them in Mixbus click here.

8.2. Plug-in Manager

The Plug-in Manager window provides a rich set of controls for browsing your installed plug-ins, and making them quickly available when needed.

To access the Plug-in Manager, *double click* in any channel strip's redirect box (it's the empty black box near the top of a mixer strip)

Alternatively, *right-click* in the redirect area, and choose "New Plugin->Plugin Manager".



Main Plugin Window

- Fav: If this column is checked, the plug-in will appear in the "favorites" shortcut in the redirect menu, and also in the Favorites list on the top of the mixer sidebar.
- Hide: If this column is checked, the plug-in will NOT appear in the "by creator" or "by category" shortcut menus.
- Name : the name of the plugin

- Tags : arbitrary tags (categories) which can be applied to the plugin. See: Plug-In Tags for more information.
- Creator : the developer of the plugin
- Type : the type (format) of the plugin (VST, VST3, AU, LV2, Lua, or LADSPA)
- Audio I/O : the number of audio inputs and output channels that the plugin allows. Asterisk (*) means “any number of channels”
- MIDI I/O : the number of midi inputs and output channels that the plugin allows. Asterisk (*) means “any number of channels”

Click on a column header to sort the list by that column.

Plugin “Search” Box

- Search: the search entry bar allows you to instantly find plug-ins by searching for a string of text
- Clear: this clears the text in the search entry bar.
- Name: if enabled, the plugin’s “Name” will be matched to the text in the Search entry bar
- Tags: if enabled, the plugin’s Tags will be matched to the entry in the Search entry bar.
- Ignore Filters when searching: When you are searching, you probably want “all” available plugins that match your search text. If you want the filters to continue operating when you search, then you should disable this option.

Try using the Search bar when looking for a plug-in. Just type the first few letters of the name of your plug-in, and your plug-in will be sorted to the top window instantly.

“Filter” section

- There are 6 “filter” selections that control what appears in the list. “Show All” will disable the filters and show all available plugins. The default is “Show Effects Only”.
- Format selection: select a format here to show plugins that match a specific format (“Type”).
- Creator list: select a creator to show only plugins from that creator.
- I/O list: select an option to show only plugins with a specified audio or midi I/O configuration.

“Tags for Selected Plugin” section

When a plugin is selected, this field will become enabled, and you can add or change the “tags” for the selected plugin. See: Plug-In Tags for more information about plugin tags.

If you like to use a particular plugin for vocals, you can add a tag like “vocal-chain” and then search for that term when you want to find the plugin quickly.

For your convenience, Mixbus comes with a collection of tags for over 3,000 popular plugins.

Plugins to be connected

This window indicates the plugins that will be added to the channel. It is possible to add several plugins here, before clicking ‘Insert Plugins’ to actually instantiate the plugins on the track.

Once Installed, plug-ins will be visible within the redirect box of the channel strip. Click here and here for more information.

For information on Virtual Instruments click here.

9. Editing

In Mixbus, audio and MIDI data is presented as “regions” on a timeline. All editing operations apply to the regions. Some of the things you can do to a region: move it, trim the start/end, set the fade-in/out length, and adjust the gain. Additionally there are many, many ways that you can manipulate those simple features of a region to accomplish your task:

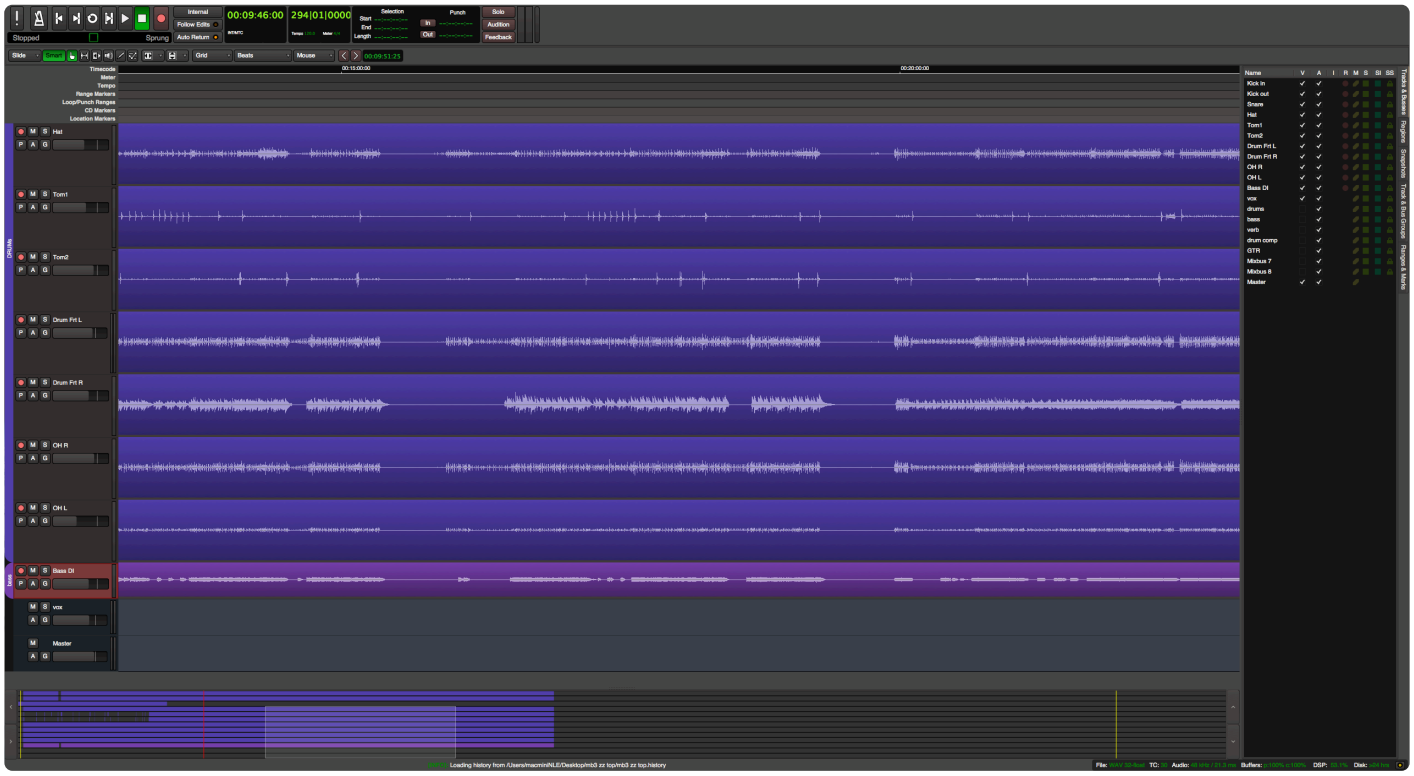
- Using editing **tools**, like the “Range” tool: A “range” is a selection on the timeline that can span over multiple regions, and make cuts or moves as if they were a continuous piece of audio.
- Using editing **modes**, like Ripple mode: if you make a deletion or insertion using Ripple mode, then the later regions (to the right) will respond to your moves so you don’t have to re-shuffle them if you make cuts earlier in the session.
- Using the **Edit Point**: You can choose to make edits on the location of the playhead (“what you hear”) versus the mouse (“what you point at”). Sometimes it makes more sense to edit with one point, and sometimes it makes more sense to edit with the other. Learning when to use each mode is an important skill to develop.
- Using the **Grid**: You can use the “Grid” to limit your edits so they only happen on musical times (such as measure, or quarter notes)

Combining these techniques gives you a nearly limitless opportunity to work with regions on the timeline. Like the tools and modes of a drawing application, you have to learn the combinations of the tools to get your work done most efficiently.

UNDO: with all of these ways to edit, it is unavoidable that you will make mistakes. Mixbus provides an “undo” feature that allows you to revert your changes. Perhaps more interestingly, Mixbus supports *persistent* undo. Each time you save your editing session, the “undo” information is saved alongside the session in a separate file. Upon reopening the session, you can still “undo” the edits that happened prior to your closing it. The main session file, and each of the snapshot files in the session folder, have their own persistent undo storage file.

9.1. Edit Window Overview

Basic Editing Concepts



Users familiar with other DAW conventions will quickly adapt to the recording and editing features of Mixbus.

> LOGIC users: you will notice that dragging a region to the blank area below the editor tracks will automatically create a new track for the region. This is a quick way to separate out a region and make a new track for it.

Track Management

On the far left of the edit window, the **Editor Mixer** can be displayed via Menu>View>Show Editor Mixer. The Editor Mixer is a single mixer channel strip that “follows” the selected track. To see all of the mixer strips for the tracks and buses on one window, show the **Mixer** window. Alt+M (or Ctrl+M for Mac) toggles between the Mixer and the Editor windows. Located near the top of the mixer strip is an Input button for selection of the physical input that feeds the track. For example if you have 8 tracks but only a 2 input sound card, you must route the inputs to the tracks as-needed.

The right side of the screen is a pull-out window which accommodates several different functions. Most panels have a right-click context menu for additional features:

- Tracks/Buses – drag & drop to **reorder tracks**, or click to **show/hide** them.
- Regions – shows all of the recorded or imported regions. Drag & drop from here into the editor.
- Snapshots – Click a snapshot to recall all of the settings in the snapshot.
- Edit Groups – Click the +/- buttons to create **edit groups**, and enable them.

> ✻ Note: Normally “snapshots” are used as backup states. You store the snapshot and continue working on the main session. Future “save” operations will apply to the selected session, not the snapshots. If you prefer that future “save” operations go into the newly-created snapshot, then you will want to invoke the “Save As” command.

> ✻ Tip: You can also select tracks in the editor and use the “Move tracks Up/Down” command from the track menu.

Using Tracks

To the left of the editing canvas are the track headers.

You can select or deselect a track by clicking on the empty area. You may shift-click to select a range of tracks, or Ctrl/Cmd-click to select and deselect individual tracks. Right-clicking the empty track area will open the **track context menu**. Certain functions such as mute, solo, and record-arm can be applied to ALL tracks simultaneously by Ctrl-Alt-Shift-clicking (Cmd-Opt- Shift-click). You can manipulate multiple tracks at once by selecting them and then using operations from the “Track” menu in the main application. Hovering the mouse near the bottom of the header causes 2 horizontal lines to appear allowing the track to be **resized** vertically.

The “P” key on each track allows selection of the track’s playlist. **Playlists** can be used to enable multiple takes in a single track, or the sharing of playlist data across multiple tracks. Some users create a new

playlist for each recording “take” while others use **Layers** to manage takes.

Tracks also support **automation** of the fader, panner, and other processing parameters including those of plug-ins. The “A” button in each track opens the automation menu which allows editing of automation data. Each parameter has multiple dynamic automation states: Manual, Play, Write, and Touch. These control whether dynamic automation will play back or be recorded while playing. Automation lines can be created and edited with the Draw Mode (green box in the image below).

You may also select a Range of automation for deletion, or to raise/lower the selected range.

Groups

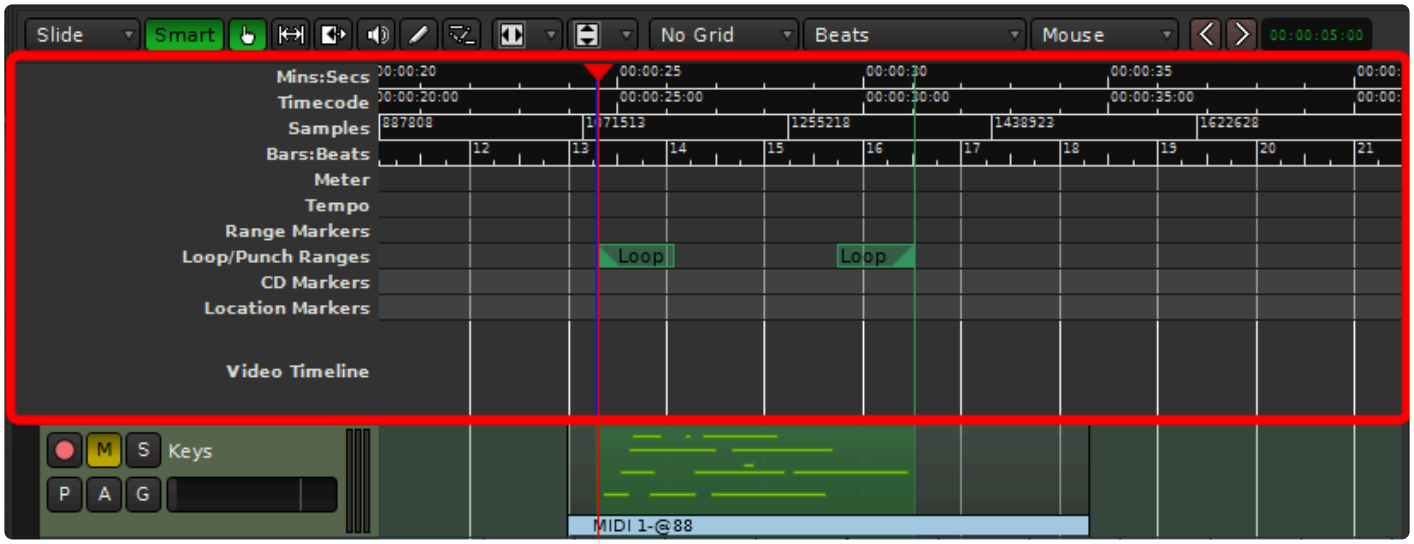
Tracks may be assigned to “edit groups” using the G button. When tracks are assigned to an edit group, and the regions align exactly on the start and end times, then all changes to one track will be applied to all other tracks in the edit group, regardless of whether the grouped tracks and regions are selected for editing. This is very convenient for editing multitracked instruments. Another convenience is that some playlist operations (such as “New”) will apply across the edit group. Edit groups are created and enabled in the Mixer List to the right of the screen. To enable the Mixer List go to menu>View>Toggle Mixer List See more on Groups here.

> Note: In addition to edit groups, you can create mix groups from the mixer window. Mix groups allow the same sort of features as edit groups, but they apply to mixing functions such as fader, mute, solo and record-arm. This is another convenient feature for multitracked instruments.

> Tip: Holding “Shift” will temporarily override a mix group assignment for the fader and mute. For example, if you have 2 tracks in a mix group, the faders will move together if you move one. But if you would like to move a single fader without disabling the mix group, you can hold “Shift” while you move the fader, and it will not cause the other grouped fader(s) to move.

Recording, Loops and Punch Ranges

Above the recording tracks is the **“ruler bar”** area (highlighted by the red box in the image below).



Left-click anywhere in the ruler bars to **locate** the playhead. You can also locate the playhead to the current mouse location with the “p” key. Right-clicking in each of the “ruler bar” areas will allow you to create location markers (cue points), CD markers, tempo markers, or meter markers. Clicking and dragging in the “range” or “loop punch” ruler bars will allow you to define **song ranges, loop ranges**, or **punch in/out ranges**. Right-clicking on the ruler bar names will allow you to show or hide the various rulers.

Navigating in the Editor window: Zooming and Panning

The area of the editor where the regions is shown is called the “canvas”. There are several ways to navigate (pan and zoom) the editor canvas.

Go to Menu>View>Zoom Focus to select the **zoom focus**. The “zoom focus” affects the operation of the zoom buttons and the zoom key commands by changing the focal point for the zoom.

Zoom focus can be one of:

- Left: Zoom focus is the left side of the summary view gray box.
- Right: Zoom focus is the right side of the summary view gray box.
- Center: Zoom focus is the center of the summary view gray box.
- Playhead: Zoom operations will center on the playhead.
- Mouse: Zoom operations will center on the mouse position (only makes sense when using the zoom keyboard commands; when buttons are clicked the Playhead is used as zoom focus).
- Edit Point: When the Playhead is the Edit Point, it is used for zoom focus.

In addition, there are several scroll-wheel commands that can be used to navigate the editor canvas:

- Scroll: Scrolls vertically.
- Shift+scroll: Pan the editor canvas left/right.
- Ctrl/Cmd+scroll: Increase/decrease the zoom (note: this always uses the Mouse as zoom focus).
- Alt/Option+scroll: Increase/decrease the height of the track that is currently under the mouse.

The Transport Menu and Toolbar

Many items in the Transport Menu and Toolbar are self-explanatory. For more details click here

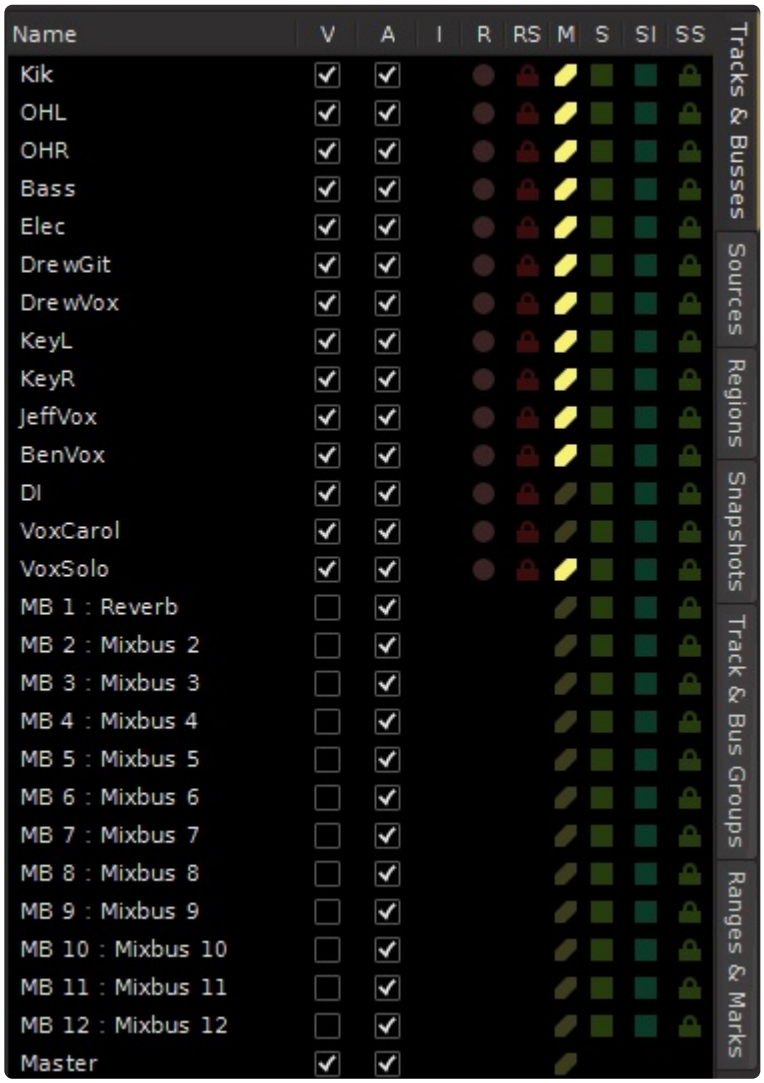
9.1.1. The Editor List

Track and Bus Management in the Editor

To the rightmost side of the editor window you can display the **Editor List** (Menu>View>Show Editor List).

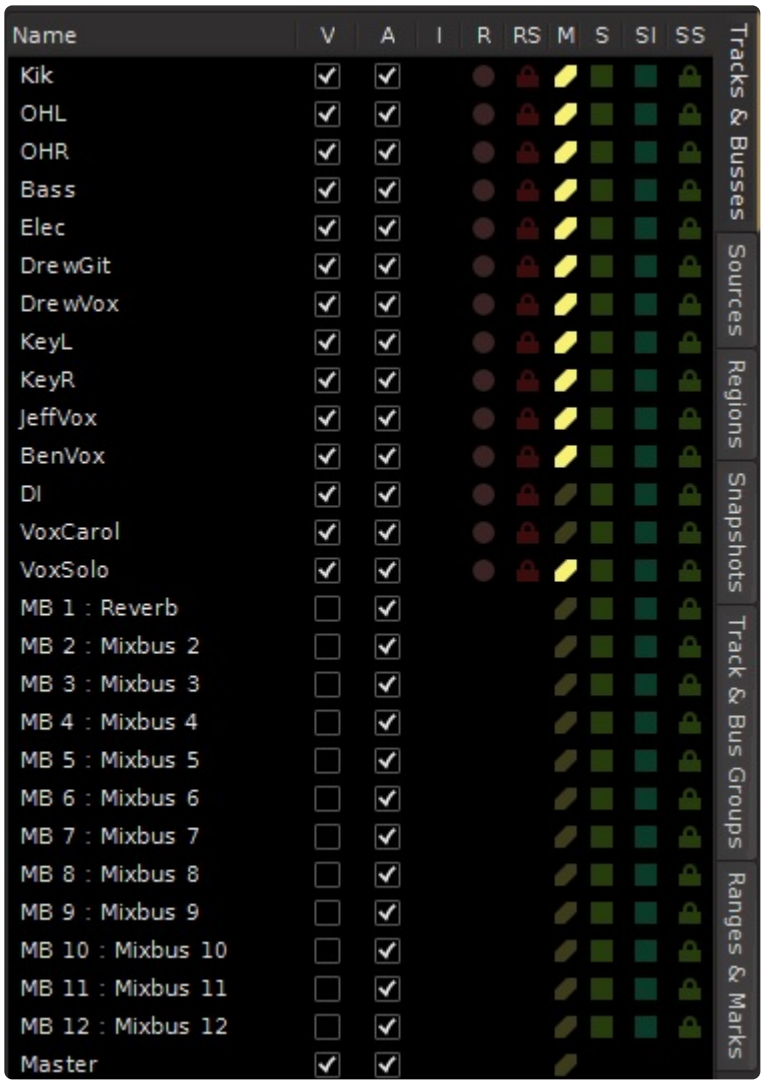
On the right side of the Editor List there are several tabs. The tabs that may be selected are:

- Tracks & Buses
- Sources
- Regions
- Snapshots
- Track & Bus Groups
- Ranges & Marks



9.1.1.1. Tracks & Buses Tab

Selecting the “Tracks & Busses” tab will show the status of all of the tracks and buses in use. The Tracks/Buses tab is used to manage tracks. You can click on the checkboxes to hide/show tracks, and you can drag/drop the tracks to change their order.



Features of the “Tracks & Busses” tab

- Name: Clicking on name will show that track/bus at the top of the Edit Canvas. Double click to rename the track/bus
- V: Toggles the track or bus to Visible/Invisible in the edit canvas
- A: Toggles the track/bus to Active/Inactive
- I: MIDI input Enable/Disable
- R: Rec enabled button/indicator
- RS: Record/Safe button/indicator
- M: Mute button/indicator
- S: Solo button/indicator

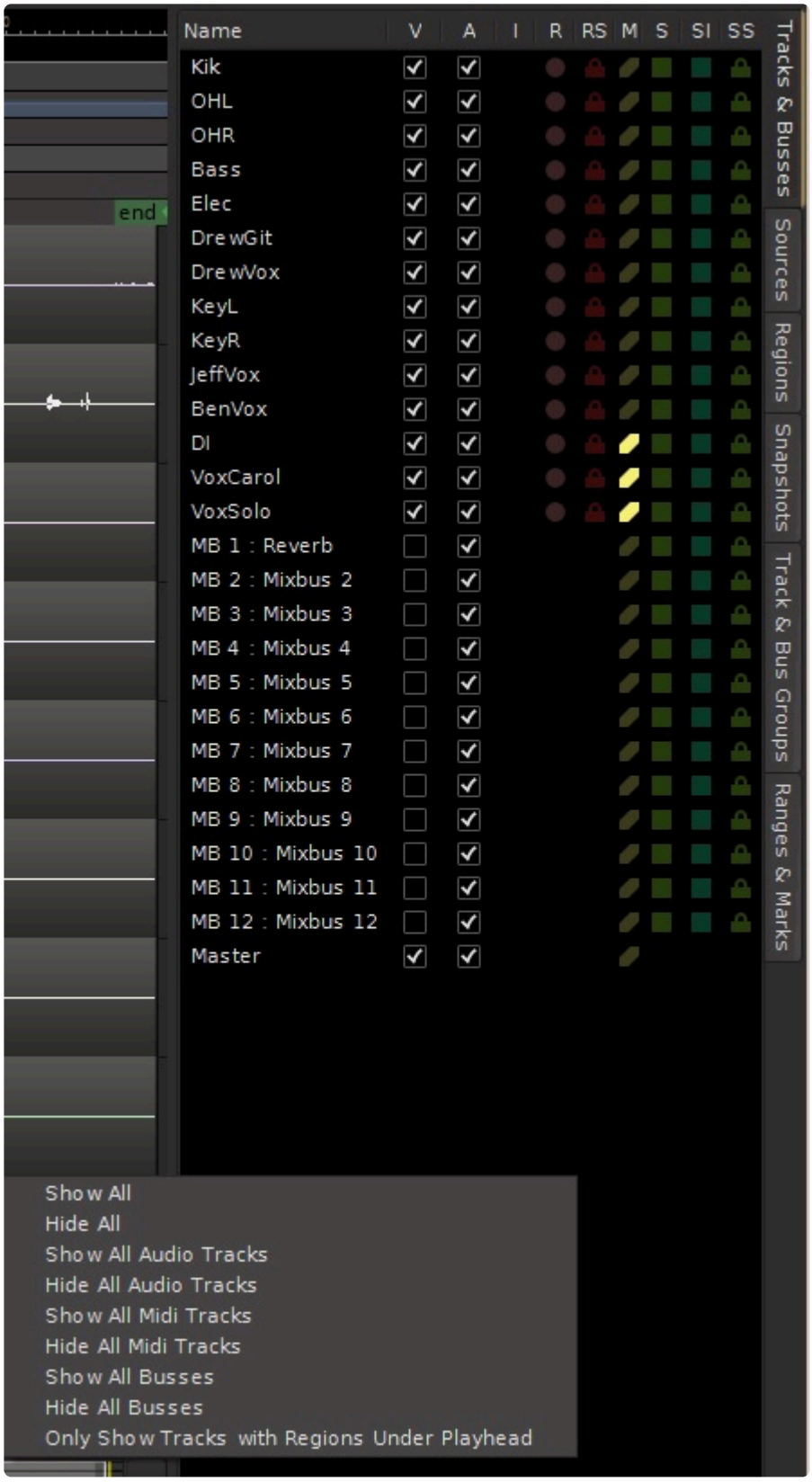
- SI: Solo Isolate button/indicator
- SS: Solo Safe Lock/Unlock

Tracks/busses can be reordered by clicking while dragging the track to the desired location.

> Right clicking anywhere within the tab displayed calls up a context menu. “Show Tracks With Regions Under Playhead” is a method which allows the users to quickly maximize the editing canvas space.

A right-click menu provides these options:

- Hide/Show all.
- Hide/Show all tracks.
- Hide/show all buses.
- Hide/show all MIDI tracks
- Show tracks with regions under playhead (only shows the tracks that have active audio regions right now).

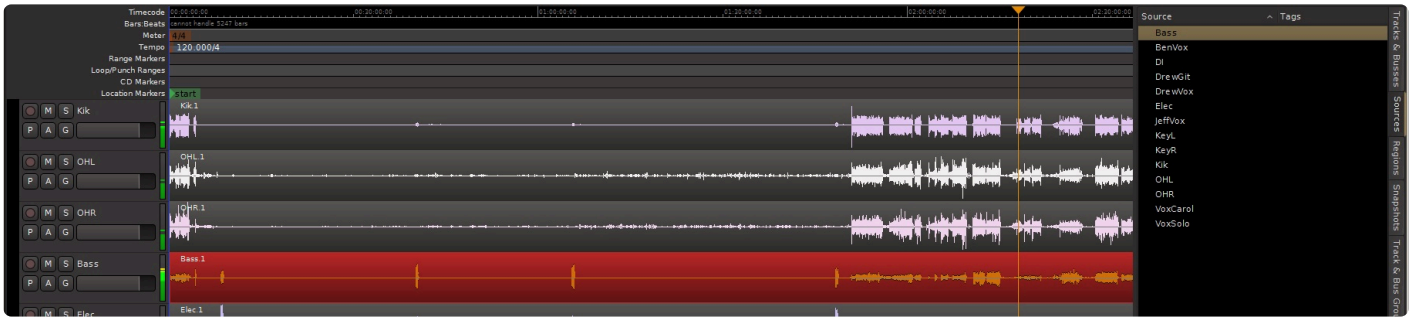


Double-click on a track to jump the editor to show that track.

9.1.1.2. Sources Tab

The “Sources” tab shows the file sources used by the session, and whether they are used in the current snapshot or not.

You can click on the sources in the list to select an entire track (or set of tracks). The example below shows the bass track highlighted.



Right clicking anywhere within the tab displayed calls up a context menu. On this tab this displays the source removal control.

9.1.1.3. Regions Tab

This shows the audio regions that have been recorded or imported into the session. The full-length audio file is shown and the underlying “regions” are the different excerpts from it. You can grab a file, or a region excerpt, and drag it into the timeline.

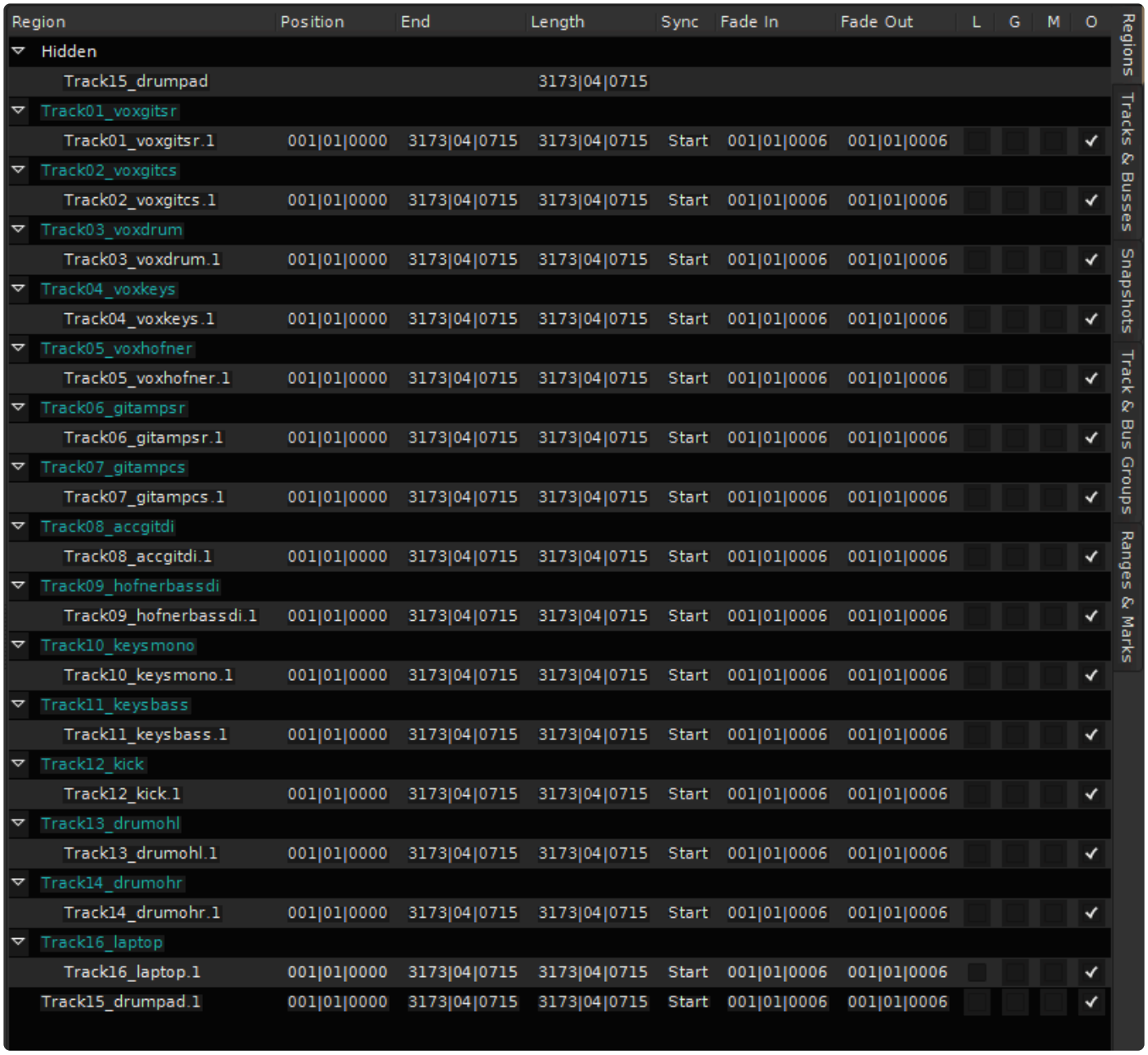
Pointing at a file shows the path to the file on your disk(s)

Right-click on a region entry to:

- audition (simply plays the wave file, outside of any context in the mix)
- hide
- Import a new file to the list (see: Import)
- Delete any regions that aren’t used in your session (see: cleanup)

For convenience, the table shows you the region’s start time, length. In the current version, these properties cannot be modified.

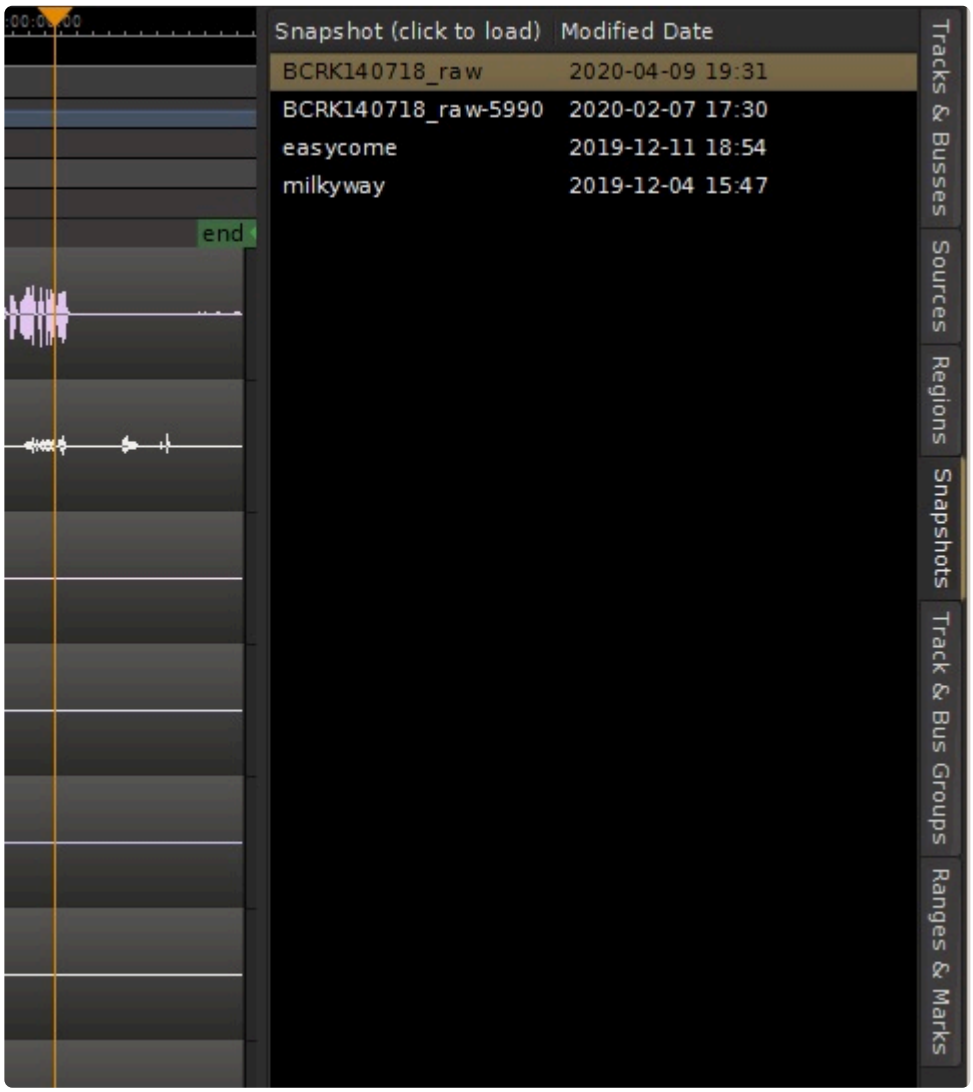
It also shows properties such as muted, locked, etc. (see Object Editing) which CAN be modified.



Right clicking anywhere within the tab displayed calls up a context menu. Various controls such as layering and gain controls are presented.

9.1.1.4. Snapshots Tab

The snapshot list:



This lists shows the available snapshots in the current session.

A “snapshot” is a complete recall all of the track, region, mixer strip, and plug-in information.

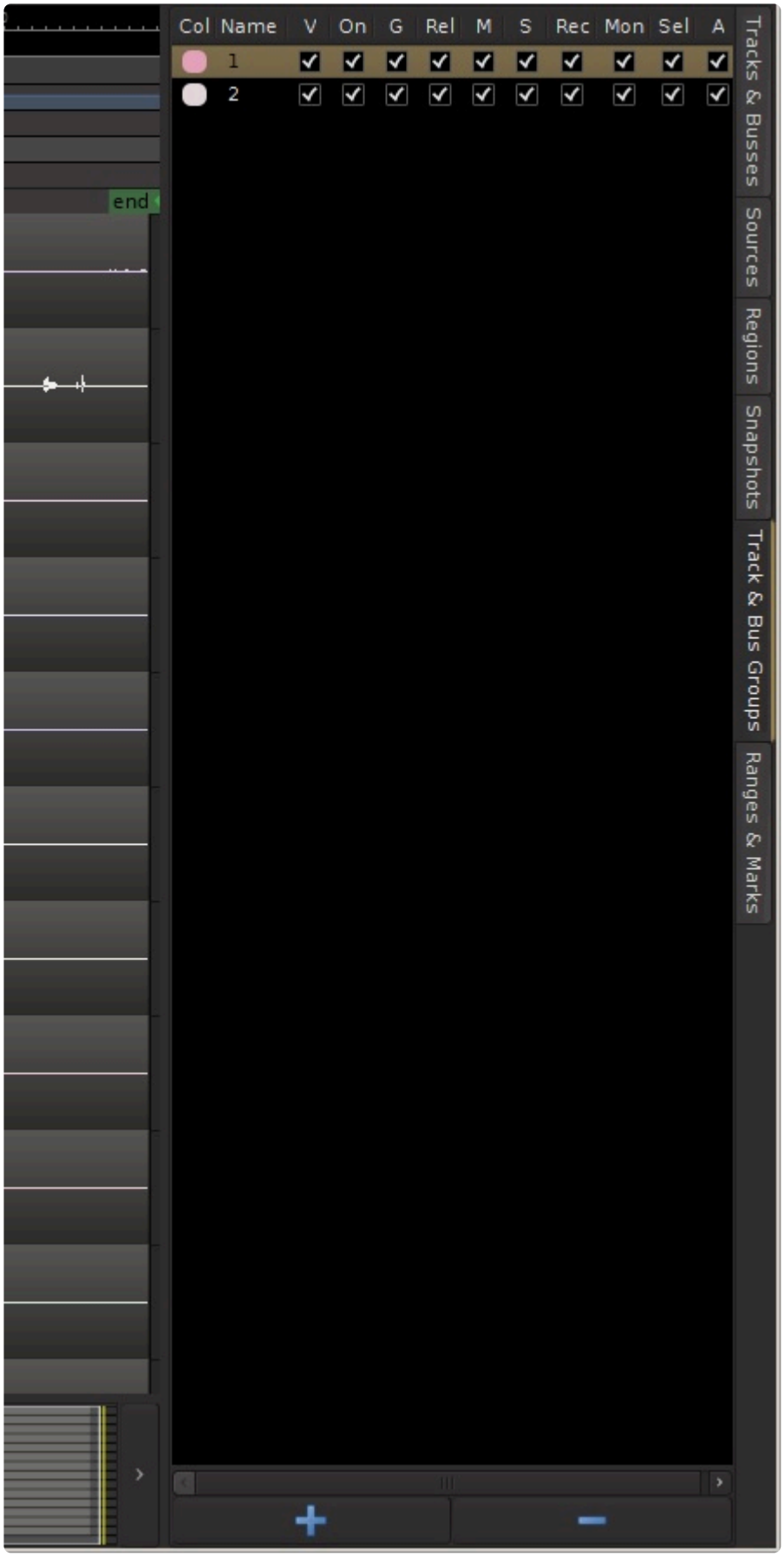
All of the snapshots share the same session audio data; this means you can have multiple “snapshots” (i.e. songs, mixes, or remixes) without recreating copies of all the audio data.

Single Click on a snapshot to switch to that snapshot.

Snapshots are also accessible in the New Session Dialog, when launching Mixbus. See “Starting a Session”

9.1.1.5. Track & Bus Groups Tab

The Track & Bus Groups tab provides control over the grouping of tracks and buses.



The blue + and – buttons at the bottom of the tab allow quick addition or deletion of groups.

The selector boxes for each group allow control of the following:

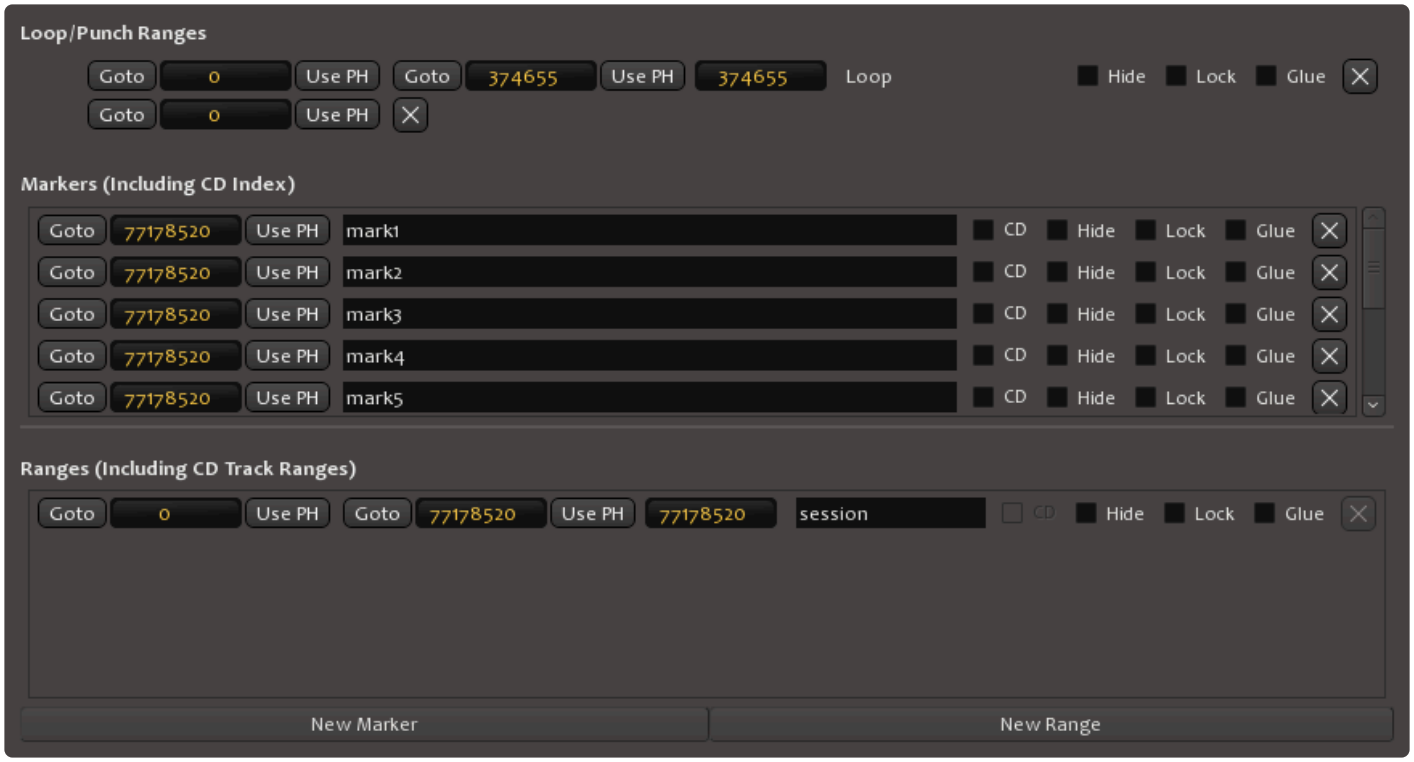
- **Col**: Color
- **Name**: the group's name
- **V**: VCA Assign – permits a group's assignment to a VCA
- **On**: Group on or off control
- **G**: Gain – syncs all faders to the *same* level
    - **Rel**: Relative – preserves fader differential between tracks
- **M**: Muting
- **S**: Soloing
- **R**: Record Enable
- **Mon**: Monitoring
- **Sel**: Selection – links selections made in the editor canvas
- **A**: Active Status – makes the group active/inactive

Right clicking anywhere within the tab displayed calls up a context menu. Various controls such as creating groups, assignment to a VCA and global group enable/disable are displayed.

9.1.1.6. Ranges and Marks Tab

The Ranges & Marks tab summarizes 3 editor functions:

- Loop/Punch ranges: time ranges where looping or punch-in recording are active
- Markers: times where a marker has been placed
- Ranges: time ranges that have been defined in the editor



9.1.2. Track Headers

The “Track Header” provides quick access to key functions

Red Dot: record-arm
M: Mute Button
S: Solo Button
P: Playlist Button
A: Automation Button
G: Group Selection Button

The horizontal slider servers as a fader.

Right Clicking in the track header pulls up a context menu.

Selected tracks can be muted, solo’ed, record-armed or gain-adjusted directly from the keyboard.

Track Height

The waveform displayed in the editor canvas area can be made to appear larger or smaller by simply hovering the cursor near the bottom of the track header. A double arrow and 2 lines will appear, at this point click and drag the cursor downward to increase the size, or click and drag upwards to decrease the size. You can perform this action to multiple tracks at one time by shift clicking a range of tracks or ctrl clicking for discontiguous selection.

Alternatively the menu item Track>Height will provide additional methods.

This action has no effect on the gain of the track, it is simply a visual aid.

For additional information about track size and zoom, click here

For additional information about track reordering, naming, or selecting, click here

9.2. Auditioning

Before you can edit, you must be able to hear what you are editing. Mixbus supports 2 different kinds of auditioning:

Auditioning a raw sound file or region with the “Auditioner Path”:

Mixbus provides a secondary audio path called the “Auditioner”. Whenever the Auditioner path is in use, the “Audition” button will blink in the top of the Editor.

Typically, you will audition files through the Import dialog. On the right side of the dialog, you will find a fader and meter that control the level of the auditioned file.

You can also audition regions that are shown in the editor by right-clicking on the regions and choosing “audition”. This will start playback from the beginning of the region. You may also audition the regions in the region list, to the right of the editor window. This is useful if you have a list of often-used soundfx imported to the session already.

By default, the “auditioner” path is routed directly the first 2 outputs of your soundcard. You can choose a different destination for the auditioner path.

If you have a “monitor” bus enabled, then the auditioner will feed the monitor bus automatically.

MIDI files, of course, must be rendered to audio before they can be auditioned. You can set the default MIDI-file renderer (for auditioning) in Preferences->MIDI.

Auditioning a sound in-place with the “Audition Action” in the editor:

https://www.youtube.com/embed/VAEx2MPPCX8?rel=0

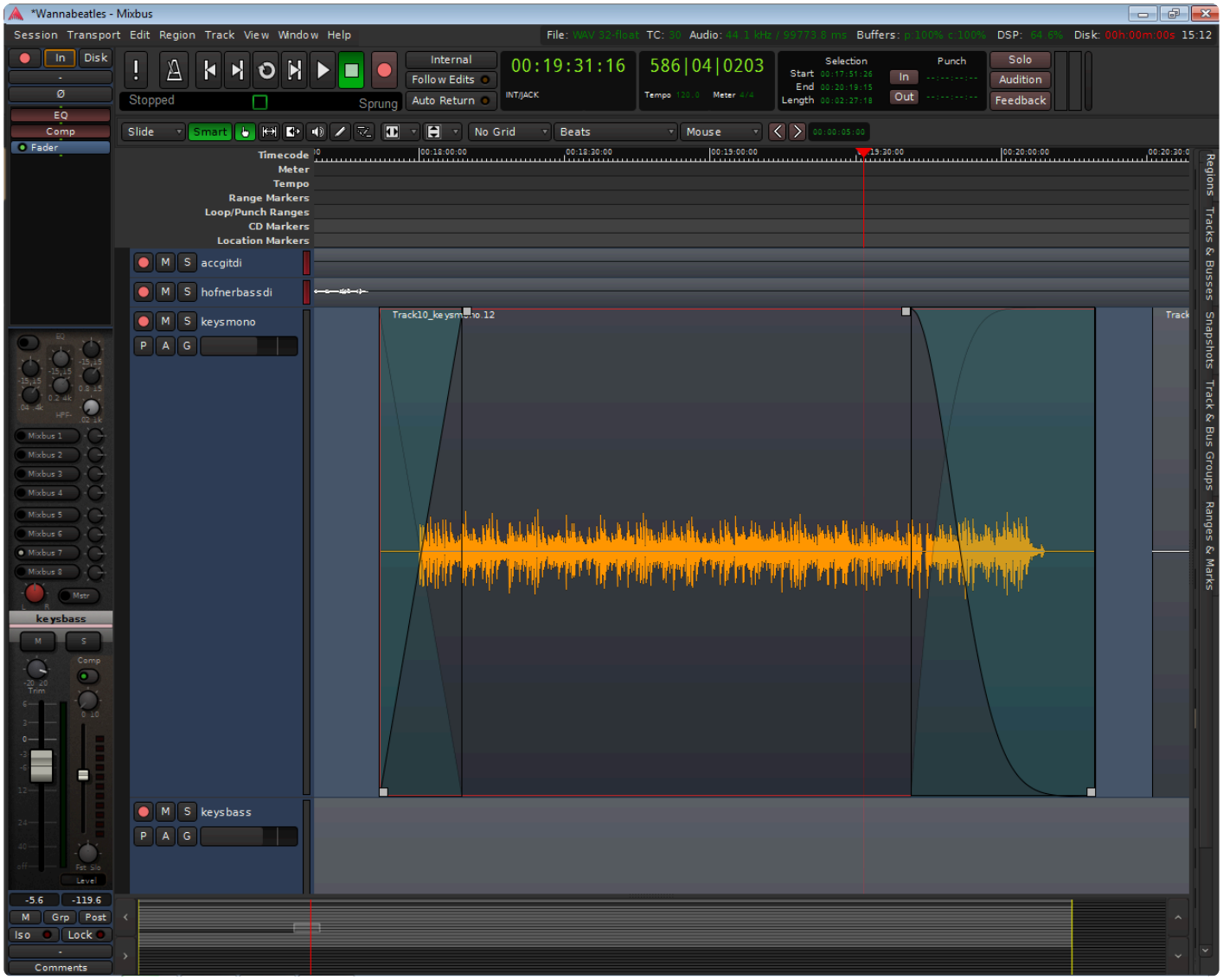
9.3. Object (Region) Editing

https://www.youtube.com/embed/aFWfLfrn_vU?rel=0

Click above for video instructions.

Object (Region) Editing

The basic element of audio editing in Mixbus is the Region. Another common name for a Region is a “clip”. A region represents an audio file (or a portion of an audio file) on disk. Regions can be moved, stacked, trimmed, copied, deleted, and otherwise edited. The maximum length of a region is the length of the file that it uses as a source, but it can be much shorter. A region might represent just a small segment (such as one syllable or drum hit) from the source file. Editing or deleting a Region does not change the source file in any way. This is called non-destructive editing. Audio regions can only exist on audio tracks. You cannot drop an audio region onto an audio bus, or a MIDI track. Similarly, a MIDI track can only hold MIDI regions.



> The fastest way to learn about region actions is to **right-click** on a region. This will provide a list of operations that you can apply to the selected region(s). The items in the Region menu (in the main application menu) may also be applied to any selected regions. If you don't have any regions selected, then most of the items in the Region menu will be disabled.

Regions have several "hot spots" that can be modified using the left mouse button when in Object or Smart-Object mode. Object edits follow the Grid selection, so operations will snap to the nearest Grid location if it is turned on.

Move	**Click in the waveform area to drag the region in time or across tracks.**
Start/End Trim	**Near the start/end of a region, the mouse will change to a "trim" tool, allowing you to trim the start/end of the region.**
Fade In/	**Grab the handle in the top corner on either end to change the region's fade.**

Out	
Fade Length	**Grab the handle in the lower corner to lengthen or shorten the region's fade time, while leaving the endpoint of the fade fixed.**

Tip: When you move/drag a range, it becomes visibly transparent. This allows you to see the regions underneath. This sometimes helps if you are recording an overdub or replacing one drum sound with another.

Tip: You can select multiple regions at the same time using Ctrl/Cmd-click or selecting them with a rubberband box. Regions selected this way will edit together: you can move, trim, or fade them all at the same time.

Tip: Combining the Trim, Fade In/Out, and Fade Length, you have complete control when assembling a crossfade between two regions. With these operations you can find the perfect location and length of a crossfade with a minimum of effort.

Additional mouse commands on regions include:

Middle-click drag	**Drag the region, overriding the edit mode. If the mode is Slide, then the region will be locked in time and will only move up/down across tracks. If the edit mode is Lock, then this will override the lock and allow the region to slide side-to-side in time**
Ctrl-click drag	**Drag the region, while making a copy**
Ctrl/Cmd-left-click on region start/end (slip edit)	**Slide the contents of the audio file *inside* the region, leaving the region start and end times intact**
right-click anywhere in a region	**Open the Region Context menu (see below)**
right-click in the crossfade area	**Change the shape of the fade in/out**

For more keyboard shortcuts view Keyboard Shortcuts

Note: For many region operations, you don't have to actually select the region. Just hover

the mouse over the region, and use the shortcut key. The shortcut keys are listed in the Region menu. This is a fast way to mute/unmute regions in a session, for example.

Region Layers

https://www.youtube.com/embed/iLmHPtUiU5Y?rel=0

Regions may be stacked on the same location in the timeline. This is called “layering”. Normally only the top region is heard; it covers the audio regions below it. However there are cases where the region below can be heard. One example is the top region’s fade-in or fade-out. Region fades operate as crossfades. If a region is stacked on top of another region, then the region underneath will audibly “fade out”, while the topmost region “fades in” across the length of the crossfade. The second way to hear a region below the topmost region is if the topmost region is not *opaque *(the Opaque flag can be turned off in the Region menu and/or the right-click Region context menu).

Region layering can be changed on-the-fly by the user. Try right-clicking on a region to change its layering. Layering is one way to manage multiple “takes” of a performance. Each time a track is recorded, a new region is written onto the topmost layer. But the layers underneath still exist. There are global options that define the default behavior of crossfades and layers. While the defaults are suitable for most users, understanding these options can be very important for advanced users (see Crossfade and Layer Modes in

Section 26.0 Mixbus Options).

You can see the stacked “takes” by right-clicking on a track header and choosing Layers->Stacked.

Mixbus can create a loop and record stacked takes. Just define your loop range, then define a “punch” range inside it, and start rolling. Each successive recording pass will generate a layer on top of the previously recorded regions.

Audio region fades

Every Audio Region has a fade-in and fade-out. By default, the region fade is very short, and serves to de-click the transitions at the start and end of the region. By adjusting the region’s fade length, a more gradual transition can be accomplished. Important: In Mixbus, each region’s fade is a crossfade with the audio below it. In other words, if the beginning of a region overlays a region(s) below it, you will hear a gradual fade from the underlying region(s), and a fade-in of the topmost region.

You may right-click on a region’s fade-in or fade-out to choose the shape of the fade.

Deactivate: The fade is completely deactivated. The audio in the region will be heard exactly as-is.
Linear: Cuts off abruptly at lower levels. Used as a crossfade between 2 audio regions. The signals are each -6dB attenuated at the midpoint. This is the correct crossfade to use with highly-correlated signals (such as test tones) for a smooth transition. This mode is rarely used on “real” material because you will hear a distinct dip in the middle of the crossfade.
Constant Power: Used as a crossfade between 2 audio regions. The signals are each attenuated -3 dB at the midpoint. This is the correct crossfade to use when you want to smoothly fade from one sound into another.
Symmetric: This is not mathematically correct like the Equal Power or Linear curves, but it provides a slower fade-out at low volumes. This is sometimes useful when editing 2 entire music works together so that the transition is more gradual.
Slow: Normally used as a region fade-in or fade-out. This curve is similar to gradually lowering a fader. It starts very gradually but then rapidly fades to silence. It is well-suited to fading out the end of a song.
Fast: Normally used as a region fade-in or -out. The level is reduced at a constant-dB rate, so it sounds very smooth.

While it can be tempting to manipulate the crossfade shape, the **location** of a crossfade is far more important than the crossfade shape. Placing a short crossfade before a loud transient (like the downbeat of the new material) is generally the best and easiest way to make a crossfade, and the “shape” of the fade is essentially moot.

NOTE: MIDI regions do not have have fades

Region Context Menu

The region context menu (launched by right-clicking a region) has many of the same operations as the main Region menu. The Region context menu always operates on the single region that was right-clicked on, while the Region menu operates on all selected tracks. Many items in the Region Menu are self-explanatory.

Region Gain

Every region in Mixbus has a level control (called Region Gain in some menus) that applies to the whole region. The Region's level can be modified by the Region menu, or from the right-click context menu:

- Increase/Decrease Region Gain (Increase is limited to +10 dB)
- Reset Region Gain to Unity
- Normalize (Raises or lowers gain so that the highest value in the region is 0 dBFS, limited to +10 dB max gain)

The Normalize entry will say "Denormalize" if the region is already normalized. This will return the region gain to unity.

> Note: If you normalize a region, it will raise the region gain so that the waveform shown in the region reaches 0 dBFS. If the region is extended to represent more of the underlying audio file, this audio may reach above 0 dBFS.

> Note: Mixbus has 5 built-in gain stages in a track: Region Gain, Region Envelope, Input Trim, Fader, and Makeup Gain (when compressor is enabled). Each gain stage serves an important purpose, and the combination of these stages makes it easy and fun to mix in Mixbus.

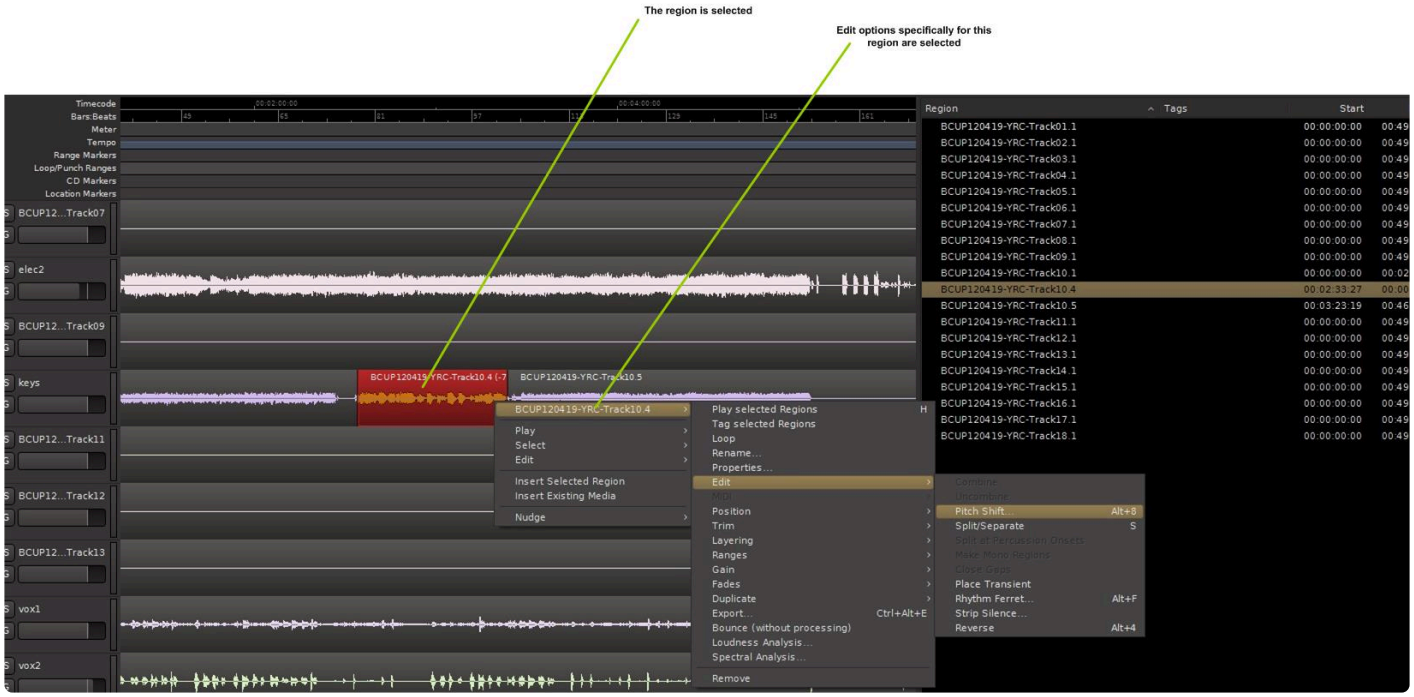
> Tip: Simply point the mouse at the region and use the gain up/down key to quickly adjust region gain. You can also select multiple regions at once and then use the region gain key commands to raise or lower all the regions simultaneously.

Description of Region Properties:

- Mute
- Opaque
- Lock
- Lock to Video: this is set when a video is imported with sound. The resulting regions have the "video lock" enabled so you can't accidentally nudge them out of time with the video. This is separate from the normal region Lock function. You must disable this lock before you can move the region on the timeline.

Region Specific Editing Options

In the picture below a particular region (called 10.4) has been selected. Notice that there are a considerable number of editing options available by right-clicking on the selected region. “Gain” as discussed above is just one of the available options. Many of these options are only available by right-clicking directly on the region and will not be displayed on the “Region” pulldown at the top of the screen, even if a particular region is selected. The same editing options are made available when right-clicking on the desired region in the Editor List which is also shown here. Notice how region 10.4 is highlighted in the Editor List when it is selected on the waveform.



In this picture the user has selected the pitch change option.

9.4. Editing Tools & Modes

https://www.youtube.com/embed/uwOwtYJDSZ0?rel=0

Click above for video instructions.

Edit Modes

The edit mode selector is to the far left, directly underneath the transport controls. There are 3 different editing modes when moving regions:

circled in red in the image above, you may select:

1. **Slide** is the normal mode that allows regions to be moved without any limitations.
2. **Lock** allows trimming, fades and track shuffling, but does not allow the region to be moved in time.
3. In **Ripple** mode, changes to a region are “rippled” to other regions on that track in the future. For example, deleting a Range or Region will move the regions after (to the right of) it, to the left. Thereby closing the gap.

https://www.youtube.com/embed/x7L6gc2lmHQ?rel=0

Editing Tools

The Edit Toolbar allows selection between several different editing tools. Each tool also has an associated shortcut key, and can be changed via the Edit menu.

Object Tool (see Region Editing)

The object tool appears as a “pointer finger” icon while hovering over regions. Just a few of the operations of the Object tool include:

- Click on a region to select it for editing, or drag a region to move it.
- Click inside the colored bar at the bottom of a region to trim the start or end points.
- Click on a region’s fade-in or fade-out handle to adjust the fade length

> If the “Smart” button is selected, the Object tool will also gain many features of the Range tool; see “Smart Editing” below for more details.

Range Tool

The Range tool allows the user to select and drag across a range of time. The time-range can span across multiple tracks and regions, as if the range were a contiguous piece of audio tape. Tracks that are selected in the current range are highlighted red in the edit window. You can add or remove tracks to the selection by clicking with the mouse.

- Drag across a range of time to create a contiguous selection.
- Grab the edges of the selection to lengthen or shorten each end of the selection.
- Right-click on the selection to launch a menu that allows looping, bouncing, exporting, consolidating, or analyzing the selection.
- Use the functions in the Edit menu to cut/copy/paste or otherwise manipulate the selection.

(see: Range Editing for more details about the Range tool)

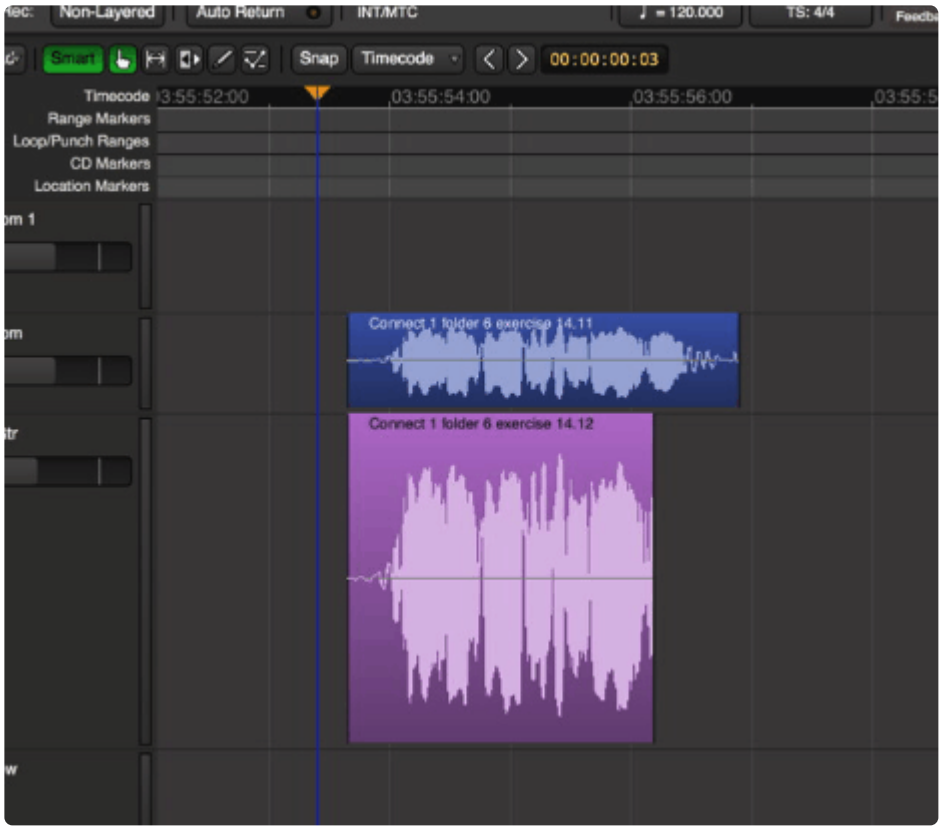
> How to “Scrub”: “Scrubbing” is a term used for auditioning a short piece of audio – perhaps by dragging the cursor over the desired audio. Mixbus doesn’t have a dedicated “scrub” tool, but you still scrub effectively by starting playback, and then holding down the “P” shortcut while pointing the mouse where you want to listen. The “P” shortcut will locate the playhead to wherever you point the mouse.

Cut (scissor) Tool

The Cut tool allows you to split any region(s) under the cursor.

TimeFX Tool

The TimeFX (“stretch”) tool allows the user to stretch regions to be longer or shorter, without changing the region’s pitch.



To use the TimeFX tool, drag on a region to lengthen or shorten the region. This will launch a dialog:

- Strict Linear: Click this to force Mixbus to use a simple resampling method (not typically used with musical content).
- Contents: Choose an advanced resampling mode that best matches the sound of the region.
- Stretch/Shrink: Initiates the stretch operation.

Draw Tool

The Draw tool allows the user to draw midi notes and automation control points.

- Click inside a MIDI region to create a MIDI note event. Drag to set the length of the note
- Click on a region to create a new gain automation point. For more details, see “Region Gain and Automation”.
- Click on an automation track to create a new automation control point

Contents Tool

The Contents tool allows the user to edit the “internal” contents of a region. For example you can use this tool to select midi notes inside a region and move or delete them.

- Click and drag inside an audio region to select region gain control points
- Click and drag on a MIDI note to move it
- Click and drag inside a MIDI region to select one or more MIDI events
- Click and drag inside an automation track to select automation control points, and then move or delete them

“Smart” Editing

When Smart Mode is engaged, the “Object” Mouse mode takes on some of the features of the Range mode. The top-half of a track operates in range mode (allowing the user to drag-select a time range on one or multiple tracks). The bottom-half of the track operates like Object mode, allowing the user to grab a single region, or rubberband-select a group of regions. Other features of Object mode (such as trimming regions or adjusting fade lengths) are still possible in this mode. The cursor will change to show either Range or Object tools when the user transitions from the top of the track to the bottom of the track. “Smart” mode is only lighted when the Object (hand) tool is engaged, but the Smart setting is remembered at all times.

Selection conventions

Selected items (Regions, Tracks, MIDI Notes, Plug-ins, or Automation Control Points) will be outlined in RED.

Selection categories (tracks, plugins, regions) are mutually exclusive. In other words, you can only have Region(s) or Track(s) selected at one time. For example, if you have a Region selected, and you select a Track, then the Region will be de-selected.

A Range selection implies a single time range selection and one or more Track selections. Range and Track selections work together, they are not mutually-exclusive. You can’t have a Range selection without a track. A Range selection implies that one or more tracks are selected. However you CAN have one or more Tracks selected without a time range.

This convention makes it clearer, for example, what will be deleted when you click the “Delete” key.

Pressing the ESC (Escape) key will clear all selections.

Operation Priority

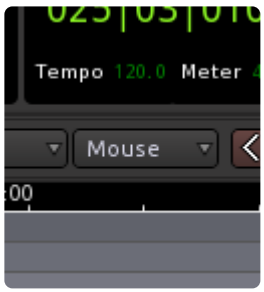
When an editing operation occurs, Mixbus must determine where the operation applies. This depends on many factors including the type of operation, the location of the Edit point, the existence of a selected Range, and Track Selections. Most of the time Mixbus will do exactly what you expect it to. If not, then it might be due to an unexpected operation priority. See the table below.

Delete Functions:	Object (Region) Editing Functions:	Edit Point Functions:	Range Editing Functions:	Nudge Functions:
Example: Delete, Backspace	Example: Mute, Normalize, Reverse, Transpose	Examples: Split, Trim, Align	Examples: Separate, Crop,	Examples: Nudge

Priority: 1. Any Selected Range, 2. Any Selected Regions, 3. The region directly under the mouse (even if the region isn't selected), IF the Edit Point is "Mouse" 4. Selected Plug-ins 5. Selected MIDI Notes 6. Selected Automation Control Points	Priority: 1. Any Selected Regions, 2. Any regions on selected tracks under the edit point (even if the regions aren't selected) 3. A region directly under the mouse (even if the region isn't selected), IF the Edit Point is "Mouse"	Priority: 1. Any Selected Regions at the Edit point, 2. Any regions on selected tracks under the edit point (even if the region isn't selected), 3. The region directly under the mouse (even if the region isn't selected), IF the Edit Point is "Mouse"	Priority: 1. The selected Range	Priority: 1. Any Selected Regions 2. The Playhead (if nothing else selected)

The Edit Point

Functions on regions such as splitting, trimming, and pasting will happen at the currently defined edit point. The edit point can be the mouse cursor or the playhead. Mixbus allows you to choose between the two modes on the fly using the edit point button.



in the above image the Edit Point is set to "mouse".

The settings for Edit Point are:

- **Playhead** : Mixbus will use the Playhead position to determine the location of an edit.
- **Mouse** : Mixbus will use the Mouse cursor position to determine the location of an edit.

Note: In order to use the Mouse edit point, the mouse must be located inside the editing area. This means you can't use commands from the "menu", because Mixbus doesn't know

where you are trying to apply the edit. So if you want to use the Mouse as your edit point, you'll need to learn some of the basic editing keybindings (such as "S" for split, and "J"/"K" to trim the start/ends of regions)

Tip: Another advantage of using the mouse is that Mixbus can guess which track you want to edit without having the track explicitly selected. In other words: in Playhead mode, Mixbus has to know which tracks (or regions) are selected before it can edit. In Mouse mode, if no tracks or regions are selected, then Mixbus will assume you want to edit the region you're currently pointing at. For these reasons, the Mouse mode can be faster for experienced users.

Tip: However some users like to edit with their "ears", and in this case choosing Playhead as the Edit Point might be the preferred choice. Using Playhead mode, you can use your ears to start fades, trims, and selections while the playhead is rolling. Selecting the track(s) that you want to work on means that you can operate the editor using only the keyboard and your ears, without referring to the screen at all!

Tip: In both Mouse and Playhead mode, all of the keybindings work. For example, the ","/"." keybindings can be used to mark the start and end of a range selection while playing through a section in Playhead mode.

9.5. Range Editing

https://www.youtube.com/embed/gWs9TWv91fg?rel=0

Click above for video instructions.

Range editing allows you to select and delete contiguous selections of times, on multiple tracks, without worrying about the individual regions in the track. Once a range is selected, the most common operations are:

- Delete everything inside the Range by clicking Delete or Backspace
- Cut or Copy the Range to the paste buffer
- Use the Edit menu to operate on the range
- Right-click on the Range to perform one of many tasks

> The fastest way to learn about Ranges is to create a range on the timeline, and then **right-click** on it. This will provide a list of features that you can accomplish with the selected range. Most of the items in the “Edit” menu are also applicable to the selected range.

- Play Range : Locates the transport to the start of the range, plays through it, and stops at the end.

- Loop Range: Sets loop makers in the ruler bar and loops the range.

- Zoom to Range: Re-sizes the canvas to accommodate the range.
- Spectral Analysis: Performs spectral analysis on the regions within the range.
- Optimize Polarity click here for more
- Move Range Start to Previous Region Boundary, etc: these functions help to fit the range to the boundaries of nearby regions (you can also use the Grid for this task)
- Separate : Splits the Region(s) inside the range, on each of the start/end boundaries of the Range.
- Convert to Region in Region List : Combines all the selected regions in the range, and creates a single audio file that will appear in the Region List of the edit window.
- Select All in Range : Selects all the regions that are touched by the Range selection. (Note: there's also "Edit->Select->Select all within Edit Range" which only selects the regions completely inside the range)
- Set Loop from Selection : Sets the Loop Start/End markers to match the Range selection
- Set Punch From Selection : Sets the Punch In/Out markers to match the Range selection
- Set Session Start/End from Selection : Sets the Session Start/End markers to match the Range selection
- Add Range Markers : Creates a new "Range Marker", which is a persistent pair of start/end markers that can be used to re-select later, name CD track indexes, or stored as an export range (for example, a song in a multi-song project, or an excerpt of a song).
- Crop Region to Range : Trims the region start and end points to fit within the Range
- Fill Range with Region : Using the selected Region in the Region list, this fills the range with enough copies of that region to fill the range.
- Consolidate Range : Removes all the region data in the Range, and renders all the regions into a single region (and audio file). This file incorporates all region gain setting, region gain automation, and fade in/out settings for the included regions.
- Bounce Range to Region List : Combines all the selected regions in the range, and creates a single audio file that will appear in the Region List of the edit window.
- Export Range… : Launches the Export Dialog with the selected range defined as the export time

9.6. Automation Range Editing

Select a range of ***automation*** or ***region-gain***, and drag the whole range of control points up/down.

https://www.youtube.com/embed/DjlBbao5gMQ?rel=0&start=135

- Click the **"R"** key to select the Range tool.
- Sweep a range across a Region, OR on an automation track.
- Click **"D"** to switch to the Draw tool.
- Click in the selected range, and drag up/down to move the automation line up/down during that time.

9.7. Ripple Editing

https://www.youtube.com/embed/x7L6gc2lmHQ?rel=0

https://www.youtube.com/embed/x7L6gc2lmHQ?rel=0

9.8. Snap and Grid

What is “Snap” and “Grid”?

“Snap” will cause mouse drags to jump to the nearest snap setting.
Snap can be set to multiple options: markers, region start/ends, and the grid. (those are all enabled by default. change them in prefs)

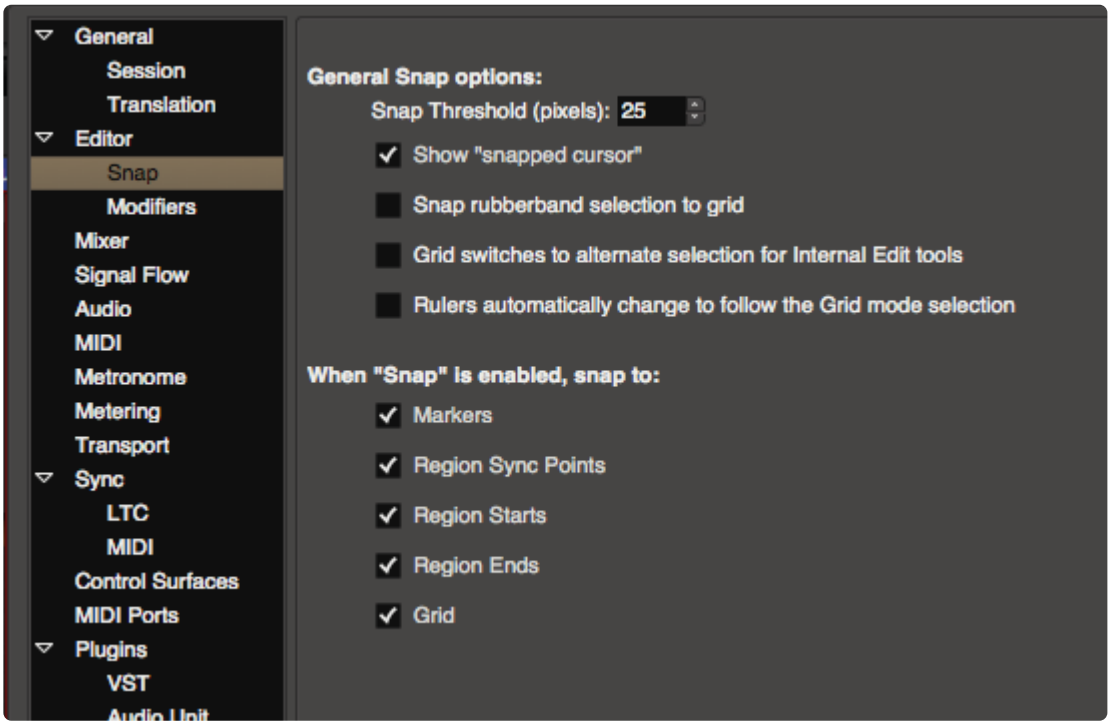
Grid can be enabled, and it will draw lines at selected intervals; which can be musical, like 16th notes, or can be timecode based (minutes and seconds) . You can leave the Grid enabled, but snap disabled, if you just want to see the lines but not snap to them.

For example: if the Grid is set to “beats” and Snap-to-grid is enabled, then any operations such as split, paste, or range-select will happen exactly on a beat, according to the musical timeline and tempo.

Alternatively, you can leave “Snap” enabled (so your mouse actions can snap to Markers, or region edges) but disable the Grid.

! The grid consist of lines running vertically in the edit canvas. If you zoom too far out, you might see a coarser grid than you expect. Mixbus tries not to show “too many” or “too few” grid lines depending on the zoom level. You might find that items snap in-between the grid lines sometimes. That’s expected behavior. If you can’t see or snap to the grid you’d like to use, you may have to zoom in or out.

Snap Options:



General Snap Options:

- **Snap Threshold (pixels)** (default 25) : when you are within this distance of a snap-able location, the mouse will jump to the snap-able location.
- **Show “snapped cursor”** : when enabled, *and you are using the Mouse as Edit Point*, a blue line will appear where the “snap” is going to occur.
- **Snap rubberband selection to grid** : when you select a “box” with the grab tool, this determines whether the box edges will Snap, or not.
- **Grid switches to alternate selection for Internal Edit tools** : when you switch from a regular tool to an “internal editing” tool (Draw or Edit), the Grid will automatically switch modes. In the past, this was used to increase the Grid precision when drawing MIDI notes. With the current “Scalable Grid” implementation, this is no longer very useful so it is defaulted “off”.
- **Rulers automatically change to follow the Grid mode selection** : If enabled, when you select a Grid, the appropriate ruler for that grid will appear.

> ! If the Grid setting seems to be changing arbitrarily, it’s probably because you are switching between Region-editing and Content-editing tools. You should disable the option “Grid switches to alternate selection for Internal Edit tools”

“When Snap is Enabled, Snap To”

It is possible to select multiple “Snap” options simultaneously.

- **Markers**: When enabled Mouse Operations will snap to the nearest timeline location marker

- **Region Sync Points** : Mouse operations will snap to the nearest region Sync Points
- **Region Start** : Mouse drags will snap to the nearest region “start”
- **Region End** : Mouse drags will snap to the nearest region “end”
- **Grid**: Mouse operations will snap to the visible grid lines, which correspond to the selected grid unit.

> To use Region Starts/Ends/Syncs as “Snap to” resolutions, you must have either
> • No tracks selected, which allows “Snap to” regions on any track.
> • One or more tracks selected, which allows “Snap to” regions on all selected tracks.

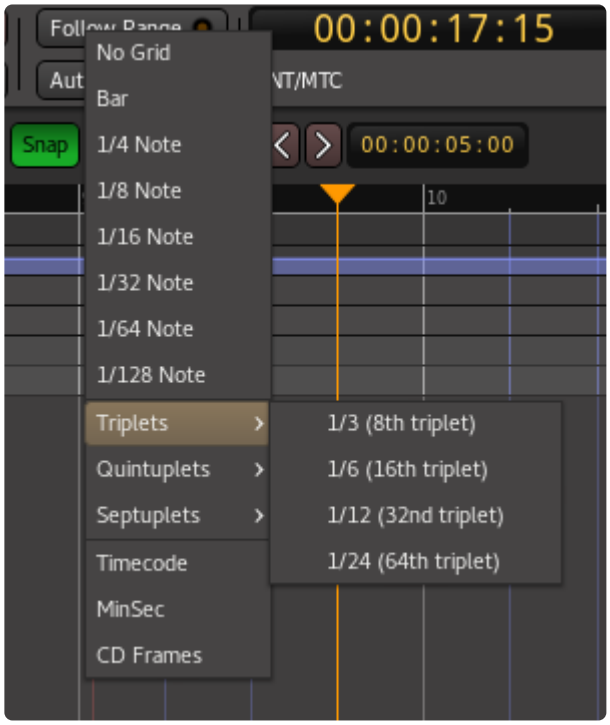
Grid override:

To temporarily disable the Grid while dragging, hold “ALT” while dragging. (this option can be changed in “Preferences->User Interaction”)

Relative Snap:

Sometimes you want a region to move in units of the grid (eight-notes, quarter-notes, or whatever), but you want it to maintain the same *offset* from the grid. In this case, hold “Shift+ALT” while dragging, and the region or note will retain it’s relative offset from the grid. (this option can be changed in “Preferences->User Interaction”)

Grid Units: (Menu>View>Snap to)

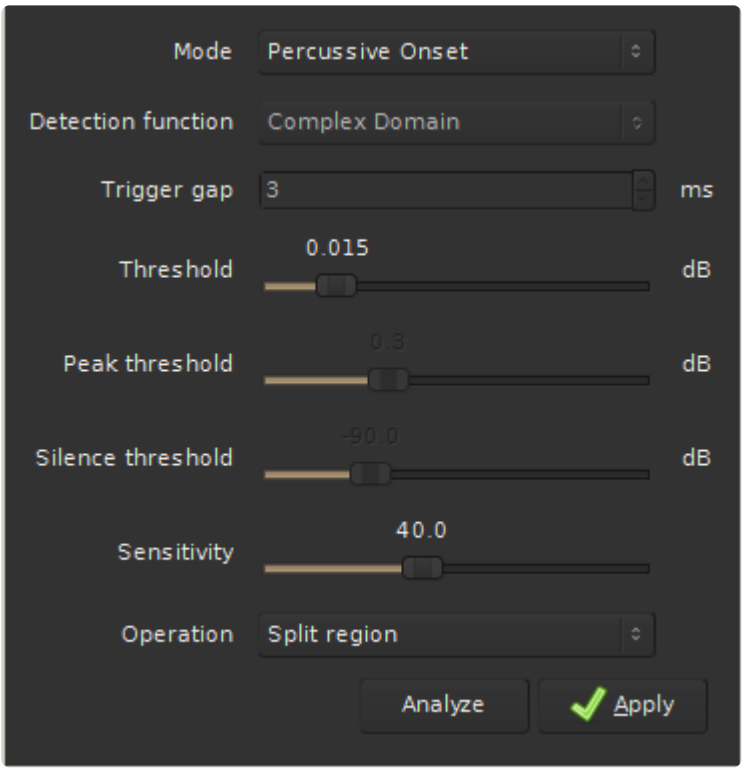


No Grid	If this is selected, then no grid will be visible or effective.

Bars	Whole bars.
Beats (i.e. quarter notes)	Grid units consisting of various note-lengths (quarter-notes, sixteenths, triplets, etc)
Timecode; Frames/Seconds/ Minutes	The timecode duration refers back to the session settings.
MinSec	Hours, Minutes, Seconds.
CD Frames	For CD mastering. The CD Frame is the smallest data entity in the Red book standard.

9.9. Rhythm Ferret Window

The Rhythm Ferret dialog provides a method to quickly “split” an audio region (or selection of regions) based on events. This can be useful when separating a recorded performance into its component strikes or notes for further editing.



- Mode: Determines whether the Ferret will trigger on percussive onsets for unpitched percussion instruments, or note onsets (for pitched instruments).
- Threshold, Sensitivity: adjust these parameters to refine the detection of events
- Trigger gap: the shortest distance between events that the Ferret will allow. Increase this number if you get multiple triggers on a single note.
- Analyze: displays the results of the analysis on the selected track(s).

Once you are satisfied with the points that the Ferret has chosen to split, you can click “Split on Onsets” to automatically split the selected regions on the marks.

9.10. Region Sync and Alignment

What is Aligning?

Aligning regions means moving one or more regions based on the edit point. An alignment operation moves the region(s) so that some part of the region is positioned at the edit point.

For example, you can select 10 regions in the timeline, and select “Align Regions”. (typically these regions will be on *different* tracks, but that’s not necessarily so) . The regions will be moved so they all start at the same time on the timeline.

Available alignment commands include:

- Align Region starts
  Selected region(s) are moved so that their start is located at the current edit point
- Align Region ends
  Selected region(s) are moved so that the end is located at the current edit point
- Align Region sync points
  Selected region(s) are moved so that their sync point is located at the current edit point
- Align Region starts relative
  Selected region(s) are moved so that the start of the earliest region is located at the current edit point, and all others maintain their relative position relative to that region

What is the region’s “Sync Point” ?

By default, a region’s “sync point” is the start of the region.

However, consider the case where here is some blank space at the start of a region which should not be trimmed away, but causes the region to align incorrectly with other regions. For example, imagine the case where a drum hit is not at the immediate edge of the region’s start, but slightly delayed after it. If you “snap” the region to the grid, you’d like the drum hit to land perfectly on the grid, not slightly off.

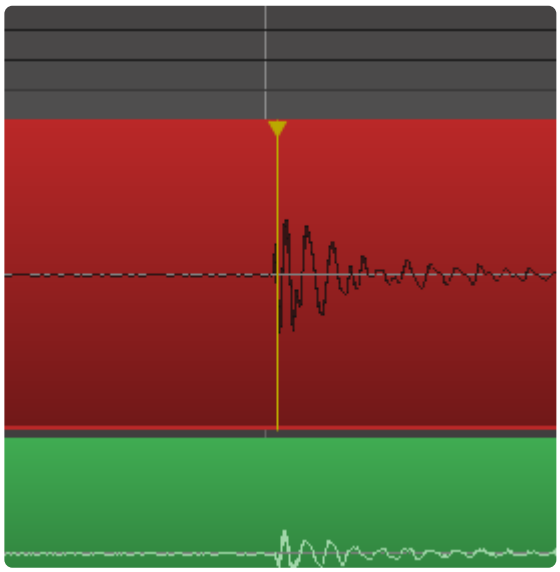
By defining a region’s “sync point”, you can align and snap regions with each other sensibly, while freely trimming the starts of the regions for crossfades and such.

Sync Point:

By default, a region’s leading edge will snap to the grid but often this edge is not relevant to musical timing. By setting a sync point within a region (ex: on the downbeat) the region can be snapped to the grid in a musically meaningful way.

To set a sync point:

Select the region and place the cursor where you would like the sync point. Press the “v” key and a yellow line will appear. Now the region can be snapped to the grid using the sync point as the reference.



To move a sync point:

Simply repeat the procedure above (there can be only one sync point per region).

To remove a sync point:

Select the track with sync point then choose: Menu>Region>Position>Remove sync. (This effectively sets the sync point back to the beginning of the region)

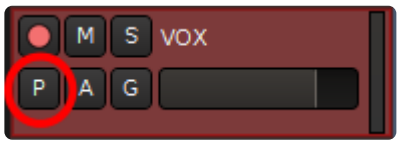
9.11. Playlists

Playlist Overview :

A track has a series of Regions on a timeline. Each Region represents a part of a sound file (or MIDI file). The order of these regions, and additional region properties such as gain/controller curves, fades, and mute/unmute status, are stored in the track's **playlist**.

Tracks can have multiple playlists, and you can create a new playlist, copy a playlist from another track, or link 2 tracks to use the same playlist. This provides some interesting possibilities. Some users like to use playlists to separate the different "takes" of a tracking session. Some users like to link playlists so that the tracks can be processed separately, but any edit on one track will be applied to the other.

Clicking the "P" button on a track header opens the Playlist menu:



9.12. Polarity Optimization

https://www.youtube.com/embed/f_f8G5tnkfk?rel=0

What is Polarity Optimization?

Polarity optimization is a way to take multi microphone recordings and automatically determine the best polarity setting for each of the tracks. The resulting analysis uses RMS level as the criteria for deciding the “best sound”. This mathematically optimized solution saves the user from comparing the many combinations generated by individually toggling the polarity switches of every track, in every possible combination.

Polarity optimization can be used in any situation where you have multiple microphones recording the same source, and it is important to find the polarity switch positions that generate the best sound.

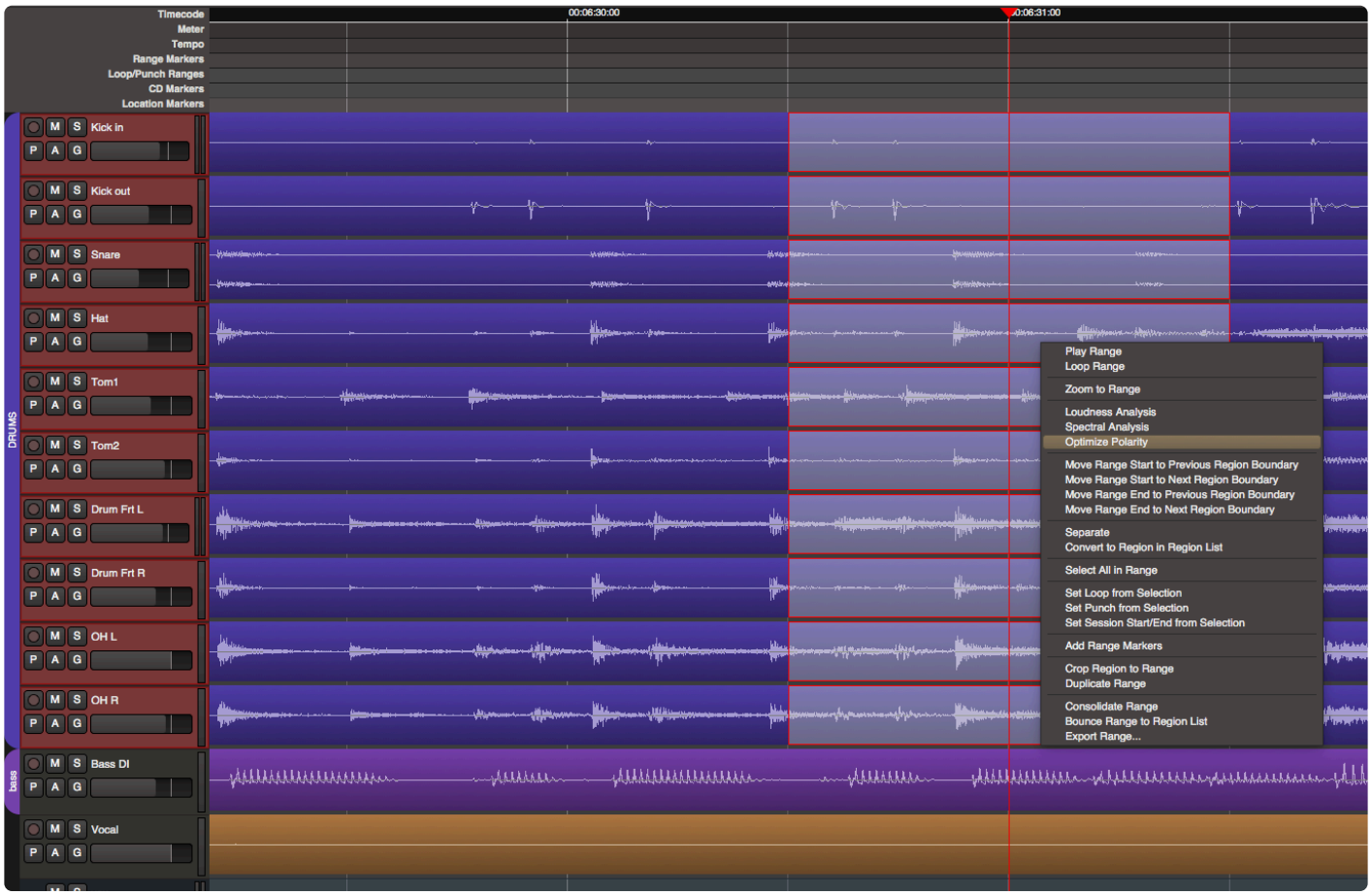
The Polarity Optimizer finds the optimum polarity setting for your tracks, and then it utilizies the channel’s polarity invert switches within the Mixbus mixer strip to implement the optimum polarity settings. After the Optimizer runs, you can further view and edit the polarity selections by using the mixer-strip polarity buttons.

Using Polarity Optimizer

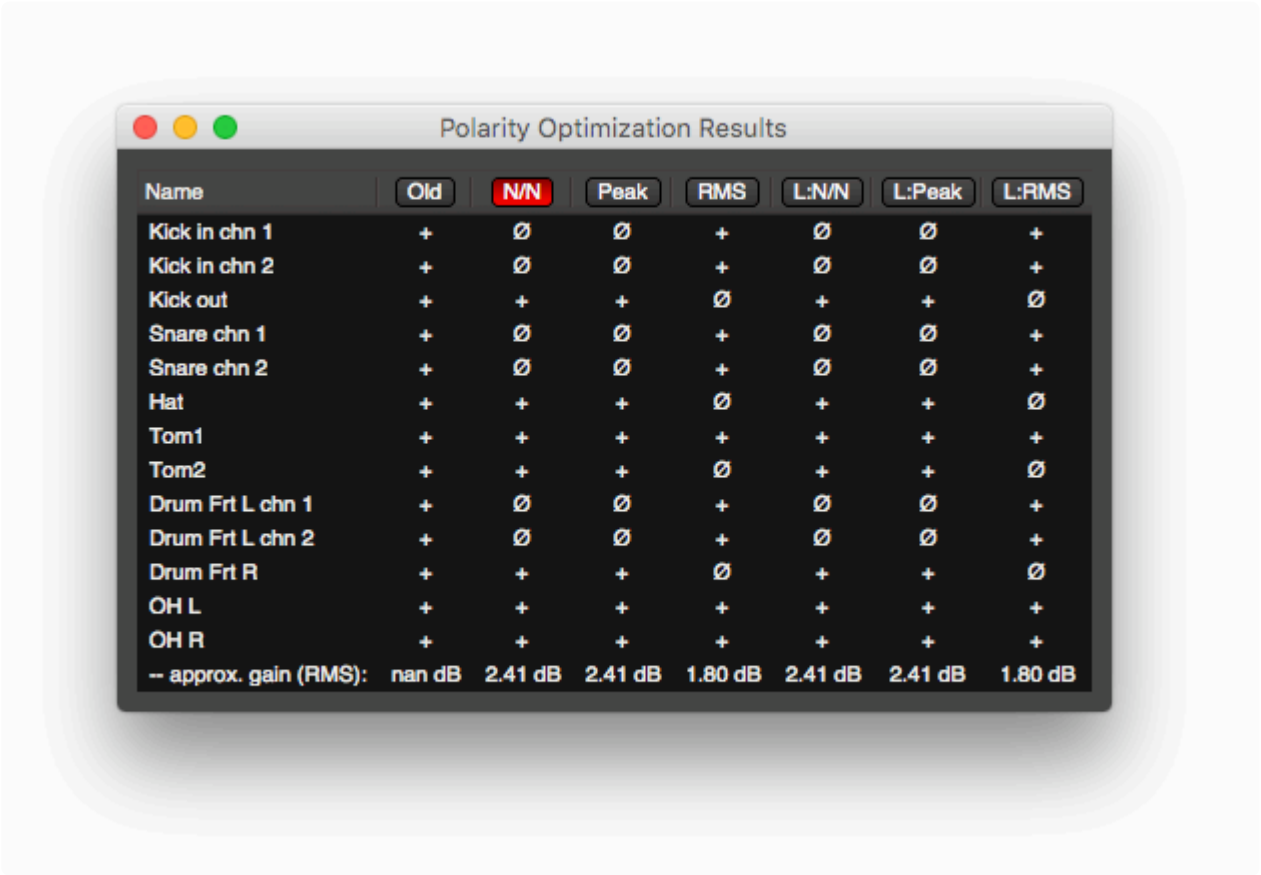
To initiate the polarity optimizer, use the Range tool (or Smart Tool) to select a time-range. It is important to select a period of time that represents all of the instruments being played.

> A mixture of mono and stereo tracks may be used in the analysis. Left/right sides of a stereo track are treated as 2 separate mono sources.

Right click in the range, and choose “Optimize Polarity” from the context menu .



Results Dialog Box:



Preprocessing Options:
The top row of the dialog box lists the 7 preprocessing options. Click on any of these buttons and roll the transport to audition the results. To accept the results simply close the dialog box.

> The preprocessing options are applied only to the signal as it is being processed. The processing isn't actually applied to the tracks, it is just applied while the the Polarity Maximizer is performing its calculations.

Explanation of the preprocessing options:

The preprocessing options help you make sure the the optimum polarity is found.

- The "normalization" options are provided to accomodate unusually quiet or loud tracks, which cause the tracks to be over- or under- represented. Normalization can be implemented as "peak" normalization, or "RMS" average normalization.
- A low shelf boost emphasizes the low end response (+6 dB at 160 Hz) and biases the analyzer to prefer alignment of low frequency content which may result in a more obvious low-end boost.

Optimization Results:

The Results dialog shows a grid of the selected tracks and the applied preprocessing options. The buttons across the top can be used to set (and audition) the calculated polarity settings, for each of the selected pre-processing modes.

- Old: The polarity of the selected tracks before Polarity Optimization was applied
- N/N: Polarity optimization applied with no emphasis, and no normalization of the tracks
- Peak: Peak normalized. Normalizes the peak volume before processing.
- RMS: RMS normalized. Normalizes the RMS volume before processing.
- L:N/N: Low emphasis, not normalized.
- L:Peak: Low emphasis, peak normalized.
- L:RMS: Low emphasis, RMS normalized.

When “Old” is selected and the transport is rolled you will audition without the results of the Polarity Maximizer. When any other option is selected and the transport is rolling, you can audition the results of the Polarity Maximizer with the corresponding preprocessing options applied.

In practice, you will find that most of the different preprocessing methods result in the same settings. In most cases, you should select one of the options that has the most “agreement” with the other selections. This indicates that the polarity settings are robust enough to apply even when subsequent level and equalization is applied.

Track name results list:
The name of all tracks used in the analysis are listed and whether the track has had its polarity inverted as a result of the analysis.

> The Polarity Maximizer applies a high pass filter (DC offset removal) to attenuate signals below 5Hz to remove it from the computation. This processing, like the processing options listed above, is not applied to the audio tracks.

The underlying technology of the Polarity Maximizer was provide by Jim Davis Ph.D. MIT, 2000 Dept. of Computer Science and Engineering at Ohio State University.
http://web.cse.ohio-state.edu/~jwdavis/

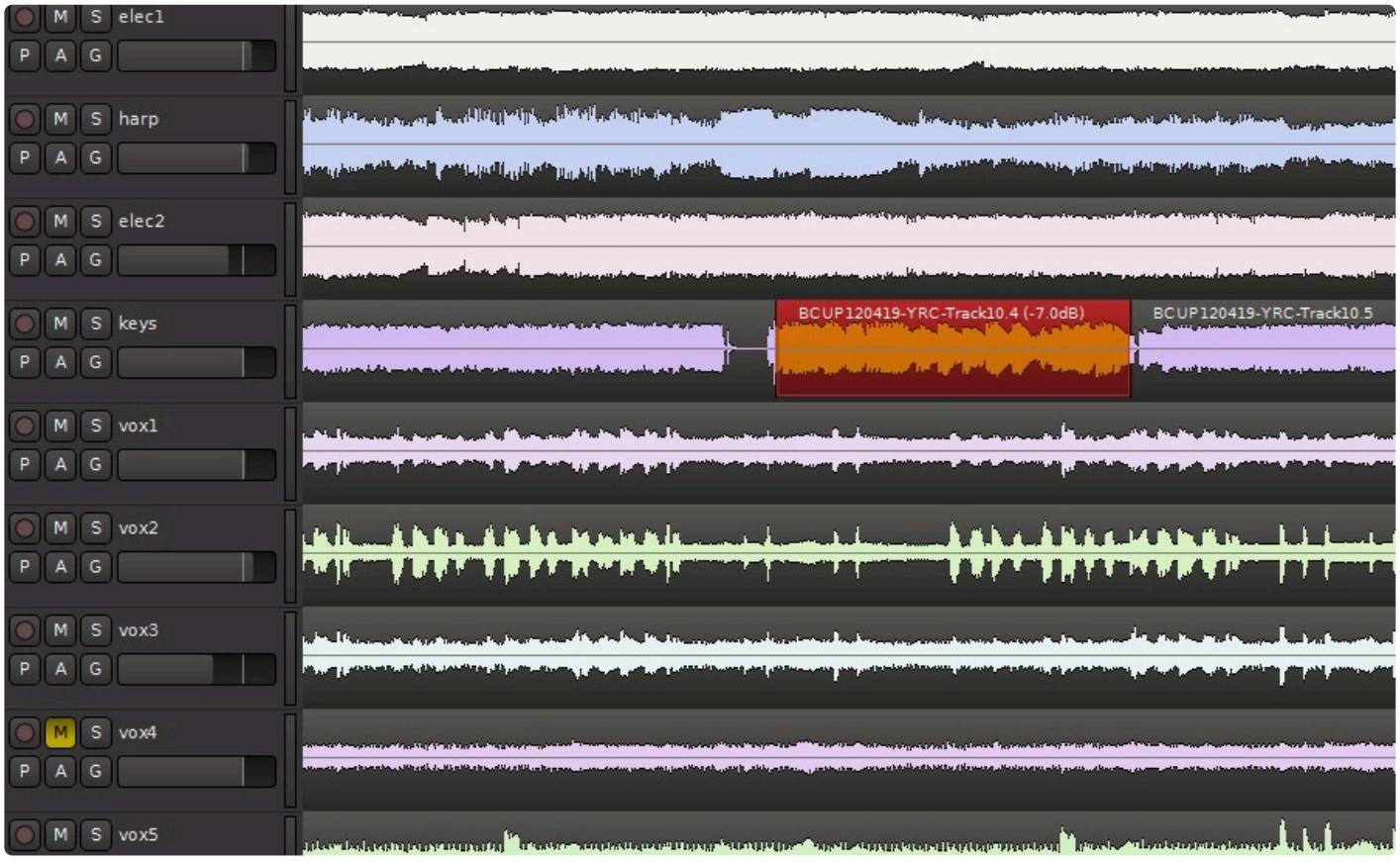
9.13. Regions and Ranges

Editing in Mixbus is based on two closely related but different ways to select audio in the Editor window: the “Region” and the “Range”.

Regions

A Region is also commonly called a “clip”. It is a piece of an audio track spanning a defined period of time. A track can comprise one Region, or a track can be composed of multiple Regions. A Region can be edited and manipulated in many different ways, which will be presented in the next topic of this manual. Regions can be created by recording a track, importing a file, pasting a selection into an existing track, separating part of a track into a new region, or drawing a new MIDI region.

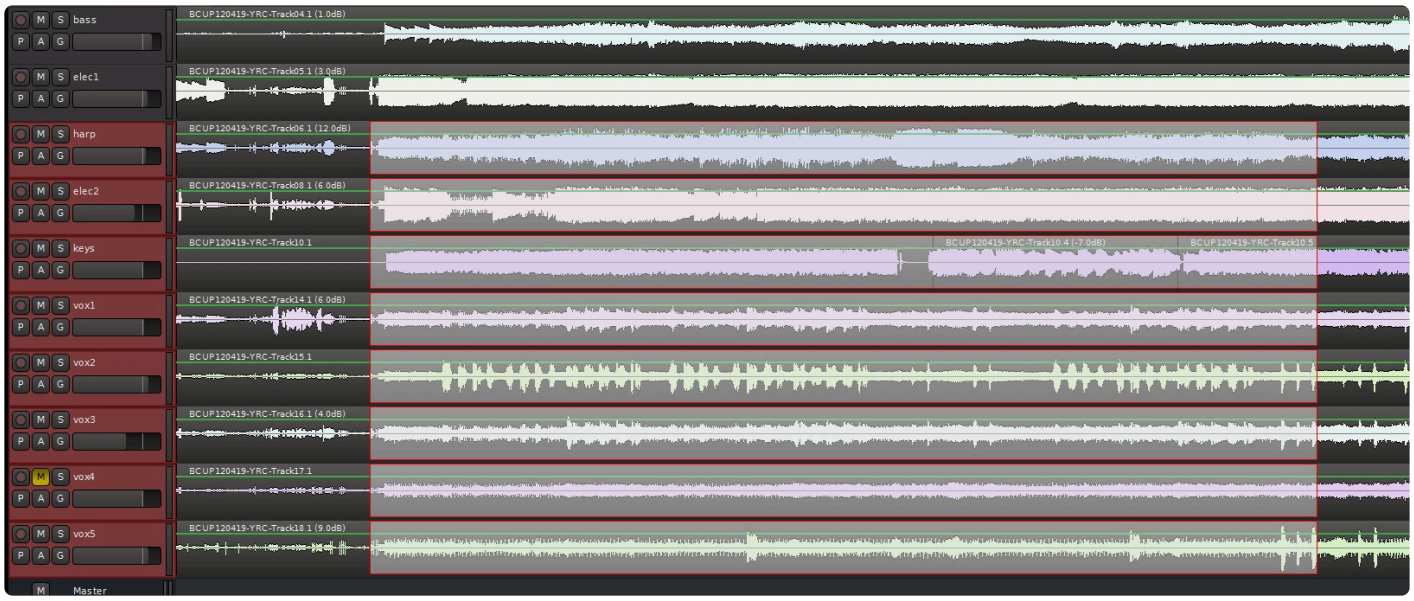
This picture shows that one track in this session is comprised of 3 Regions. The middle Region in this track is highlighted in red. The other tracks in this session are comprised of only one Region.



Ranges

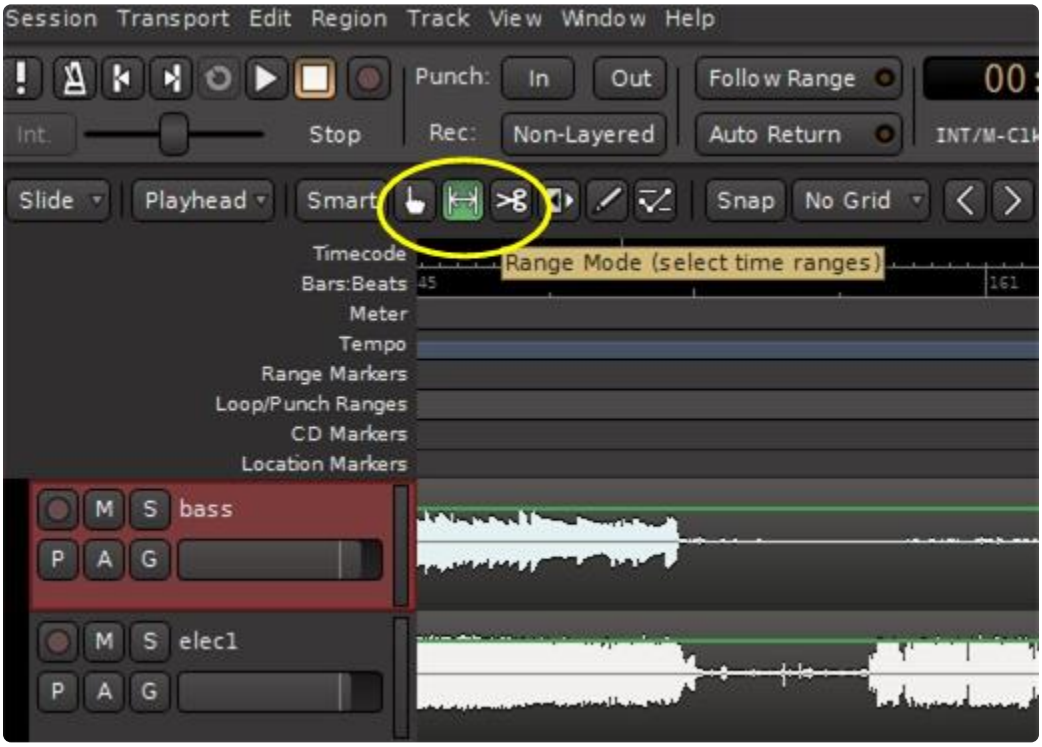
A Range is a selection of one or more audio tracks spanning a defined period on the timeline. A Range can also be edited and manipulated in many different ways. The track selections within a range can contain portions of or entire Regions of a track.

This picture shows a Range selecting a time slice of several tracks. As most of the tracks are made of only one Region, the Range is selecting only part of their Region. However note that as in the previous picture the keys track has 3 regions. The Range selection is selecting the entire second Region, but only part of the track's first and 3rd Regions (which in this case extends to the end marker of the session).

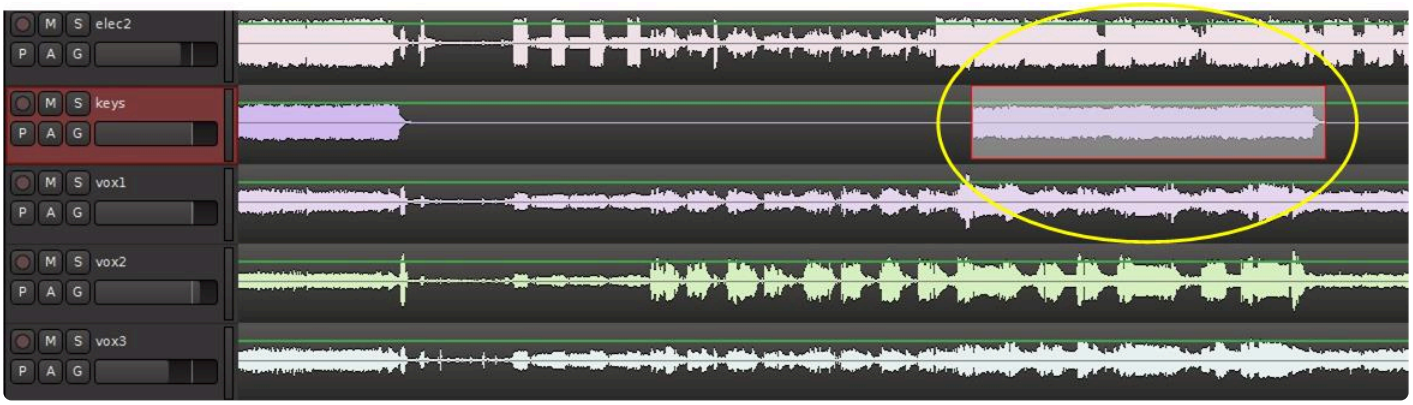


Separating a Region Out of an Existing Track

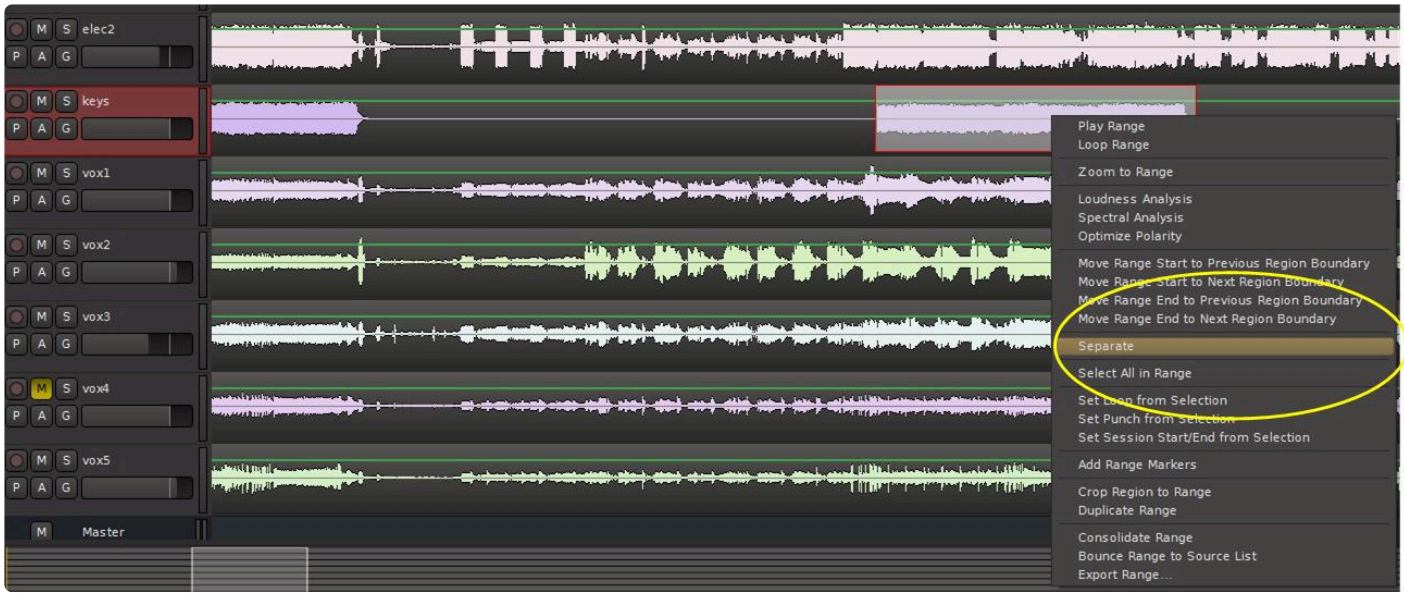
Let's say you need to tweak part of an existing track without affecting the other parts of the track. To separate out a new Region on a section of an existing track (which may itself have one or several regions), first select the Range tool (or click the keyboard shortcut 'R'):



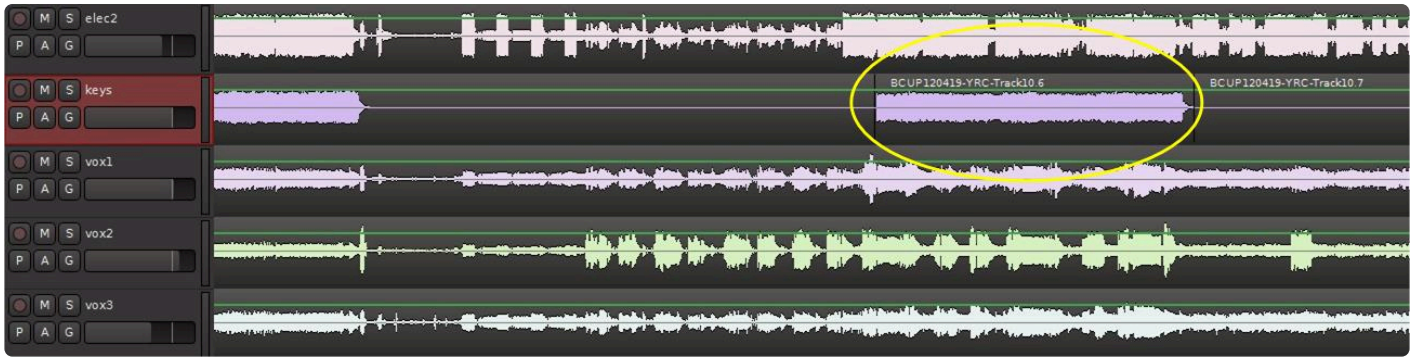
Next, use the range tool (by left-click and drag) to select the desired timespan on the track that will be made into a new region:



Hover the cursor over the highlighted timespan and right click, then choose “Separate” (or click the keyboard shortcut ‘S’):



The selected timespan will now be in a separate Region from the rest of the track. This Region can be edited and manipulated separately from the rest of the track, for example fading it in and out at its boundaries, giving it a different gain scaling from the rest of the track, and numerous other options.



10. Mixing

What is mixing?

“Mixing” is the process of combining multiple sounds (usually “tracks”) into a final, deliverable sound.

In the case of music mixing, the goals of mixing might be:

- To make the overall timbre of the mix convey the feeling that the creator has chosen.
- To make each element clearly heard, so that a listener can focus on a single instrument if s/he wishes.
- To make the volume level consistent enough that the user isn’t required to constantly adjust the volume knob.

Of course all of these rules are subject to the artistic wishes of the creator.

What are some mixing techniques?

Here are some of the techniques that we use to create a “mix”:

- Balance the volumes of each element (instrument) so they don’t occlude each other.
- Dynamically adjust the volume of elements to accentuate parts of interest at different times of the piece.
- “Pan” the signals into the available speakers so they can be separated by position.
- Create a “space” around each element so they can be perceived as close or distant.
- Tweak individual elements to accentuate their desirable character, or attenuate undesirable character.
- Limit the frequency range of elements, so that more elements can be distinguished in the same space.

You’ll find that Mixbus provides tools for all of these tasks right underneath your fingertips.

10.1. Metering

Mixbus provides a wide range of meters. Each meter is suited to its particular task.

Some DAWs try to put a lot of different scales on a single meter, and/or they expect the user to switch modes to see different values. Mixbus provides several “different” meters in the places where they are useful.

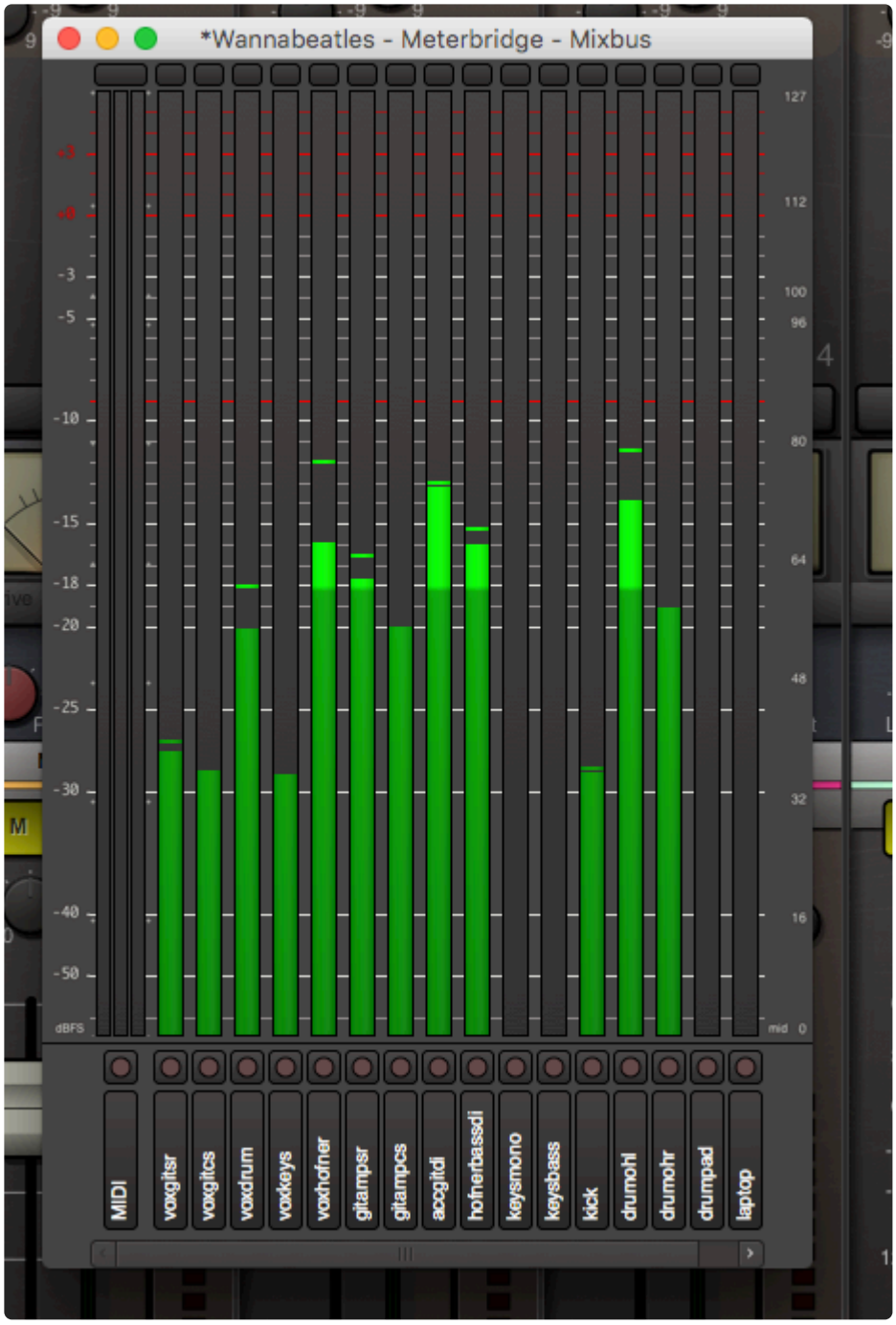
Track Meters

In the editor, to the right of each track’s name, you’ll find a small meter that indicates the channel’s level. This meter grows in height when you increase the size of the track. This meter follows the settings in “Preferences->Meterbridge->Default track meter type”. The colors for peak-hold time, among other options, can be applied to these meters. The meter follows the selection for “meter point” in the bottom-right corner of the associated mixer strip. When a channel is record-armed, the meter point will automatically move to meter the channel’s input. **Due to their scaling, and their location in the editor window, these meters are well-suited to checking your recording levels while recording**

Master Meter (editor toolbar)

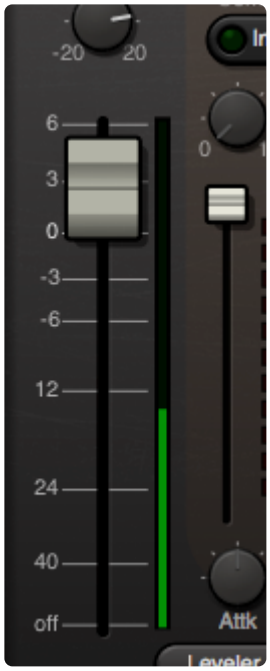
In the top of the transport toolbar, you will find a meter indicating the master bus. This meter can be hidden in Preferences->Appearance->Toolbar. This meter follows the settings in “Preferences->Meterbridge->Default meter type for master bus”. By default, Mixbus uses a K-14 meter (the same scaling that is used for the master bus K-14 meter). In addition, there is a peak-hold indicator with adjustable decay time. The colors for peak-hold time, among other options, can be applied to this meter. **This meter serves to provide a condensed view of the master bus output when you are editing**

Meter Bridge Window



The Meter Bridge window can be accessed from the main menu, under “Window”. The meters in this window follow the settings in “Preferences->Meterbridge”. The colors for peak-hold time, among other options, can be applied to these meter. **The meter-bridge is well-suited to tracking a live band, as you can see several high-resolution meters in a condensed view. You can also show the track rec-arms and monitor switching from this window**

Channelstrip meters



To the right of each channel's fader, you'll find a small meter that indicates the channel's level. The meter follows the selection for "meter point" in the bottom-right corner of the associated mixer strip. When a channel is record-armed, the meter point will automatically move to meter the channel's input. The scaling and ballistics of these meters are fixed, so that they are directly comparable across all channels of the mixer. **The channelstrip meters are well-suited to balancing your mix. The scaling is optimized to show the upper range (-20dB to 0dBFS), and it is scaled to match the fader and the compressor threshold sliders.**

Channelstrip Compressor Gain-Reduction meter

This meter indicates the gain-reduction applied by the channelstrip compressor (see Compression Techniques). Each light indicates 1dB of gain reduction.

Mixbus & Master Bus – Tape Saturation Meter

A Tape Saturation meter is provided on each mixbus, as well as the master bus. This meter indicates the amount of *distortion* applied by the tape-saturation stage. If the meter is fully left, then no distortion is applied. If the meter is fully right, then the signal is heavily distorted. You can use the “Drive” knob to adjust the amount of tape saturation / distortion.

Master Bus K-14 Meter

The K-14 meter is an RMS (averaging) meter, originally specified by mastering engineer Bob Katz as a means to compare and help control the dynamic range of pop music mixes. (see : http://www.digido.com/how-to-make-better-recordings-part-2.html) **Use this meter to monitor the perceived “loudness” of your music mix, and compare the loudness of your mix to other mixes of the same genre**

The K-Meter displays the level of the signal directly before the output of the master bus. The K-Meter comes after the fader, EQ, plugins, and final limiter in the master bus.

If the K-14 meter indicator is straight up, at 0, this indicates that the sound’s average level is at -14dBFS. This was selected as a good target for most pop/blues/jazz/rock recordings. At this level, a song will sound quieter than most contemporary pop music, but when the listener turns the volume up to match, the increased dynamic range of this mix will allow the song to sound much better than a typical pop song played at the same volume.

Increasing the volume beyond the “0” indicator will result in a louder mix, at a cost of decreased fidelity and impact when the user turns the volume up.

Master Bus Phase Correlation Meter

The stereo correlation meter provides an indication of the “mono compatibility”, or stereo width, of your mix. The indicator uses a bright bar which simultaneously indicates the range of in-phase (highly correlated) and

out-of-phase (decorrelated) sounds in your mix.

- If the indicator is fully to the right, and **yellow**, this indicates a completely “mono” mix. (the signals in the left and right channels are the same)
    - many signals (such as a spoken-word recording with one microphone) should be mono.
- If the indicator is fully to the left, and fully **red**, this indicates a mix that is completely “out of phase”, in other words the left and right channels are equal, but one is inverted with respect to the other. This almost certainly indicates a technical fault which should be investigated.
- If the indicator is largely to the right, and **green**, then it indicates a mix with some stereo width, but largely mono compatible. This is optimal for pop music recordings that might be played in a variety of playback systems.
- If the indicator is in the center, showing a range of red and green, then the mix is very wide stereo. This is probably not optimal for pop/rock recordings, but can be acceptable for natural or orchestral recordings. Mono compatibility of this mix is questionable.

Mono compatibility is largely defined by low frequencies. If your mix indicates poor mono compatibility (red bar on the left), then you can probably solve this by reducing the number of bass instruments in the mix, high-pass filtering any non-bass instruments, and high-pass filter any stereo effects, such as reverb, in the mix.

Master Bus Limiter

Each segmented LED represents 1 dB of gain reduction in the Master Bus Limiter. The master bus limiter is a brickwall, lookahead limiter that prevents the digital signal from exceeding -1dBFS. *When mixing, use this bright yellow meter to alert you when you “would have” exceeded the -1dBFS threshold. *

10.2. Gain Staging

https://www.youtube.com/embed/DjlBbao5gMQ?rel=0&start=135

https://www.youtube.com/embed/DjlBbao5gMQ?rel=0

Click above for video instructions.

What is Gain Staging in Digital Audio?

Digital audio represents the audio waveform as a series of numbers, each number indicating a voltage in the electrical signal when it was converted from (or to) an analog signal. More numbers (voltage levels) represents more accuracy in the signal's representation. Too few numbers introduces "quantization noise".

Quantization noise happens whenever you represent the signal with fewer steps than you had before. This happens quite often inside the computer, where the bit depth changes depending on many factors. Quantization noise is slightly more offensive than "regular" (random) noise. For these reasons, we must make sure there's enough random noise to cover up any quantization noise that we've introduced. This technique is called Dithering, and careful use of dither is one of the reasons that Mixbus sounds great.

For our purposes, the “noise floor” of a signal is the noise level in the quietest part of the recording; this might include noise that was picked up by the microphone, or dither noise that was used to cover up quantization noise inherent to digital maths. If the noise floor is less than the noise in your environment, then you won’t even notice it.

Generally speaking, 16-bits worth of numbers can recreate everything between a typical quiet-room noise and a very loud sound (i.e.“CD quality sound”). A 24 bit number can store signals from an infinitesimally small noise floor (lower than a very quiet studio), all the way up to a roaring jet-engine volume.

Regardless of the number of voltage steps, there will be a value that indicates “maximum volume”, and if you exceed that value, then the system must “clip” the excessive values to the largest possible value.

A very short “clip” is almost always unnoticeable (just at the peak of a snare-drum hit, for example). But sustained clipping, when the signal is significantly outside the bounds of the “real world”, creates lots of ugly distortion that we want to avoid.

So our goal is to prevent signals from clipping, but keep them well clear of and above the noise floor. Luckily with modern 20+ bit converters, this is very easy to do.

With modern A/D converters, we recommend that you record your signals with the peak meter never exceeding -15 dB. This indicates that the signal was far from the clipping level when it was converted to digital. With the increased low-level resolution of modern converters, it is not necessary to record at high levels.

Floating-point processing

Internally, Mixbus uses a technique called floating-point math. For our purposes, this means that some extra bits are reserved to store numbers outside the range (both louder and quieter) than the range of the analog world. This means that it is essentially impossible to “clip” the math inside Mixbus. In practice, this means that you can boost a signal very loud – even louder than would normally be allowed in the analog realm – and as long as you reduce it by the same amount later, it won’t be clipped when it is heard at the output.

To this end, Mixbus provides many levels of gain adjustment which will allow you a wide range of creative options.

Region Gain and Region Gain Curves

- **Region Gain and/or Gain Automation**: Increasing a region’s gain in the editor will provide:
  - Each region has a volume setting (accessed with the Cut Gain and Boost Gain shortcuts). The gain increase/decrease visibly affects the waveform display
  - Region gain curves allow the gain to change through the length of the region

Channel Strip Gain stages

- **Channel Input Trim**: This stage is used to bring the overall level of the track to a usable level. The

trim knob is located near the fader for easy access.
- **Fader**: Used to achieve an overall balance of all tracks in a session. This is also a good place to make artistic changes such as bringing an instrument forward during a solo.
- **Make up gain within the In-line compressor**: Make up gain is used to restore perceived level loss during compression. If the compressor meter is hitting -3 dB during transients, a make-up gain of 1 or 2 dB may be useful. It can also be used as a fast and easy way to adjust the level post fader.

Mix bus

- **Input Trim**: This knob is used to adjust the volume at the input of the master bus.
- **Tape saturation**: The tape saturation meter does not monitor level, rather it monitors the amount of saturation effect. This effect will add harmonic density which can result in an increase in perceived loudness. Turning the tape saturation knob off will allow signal to pass unaffected.
- **Fader** Used to achieve an overall balance of all tracks in a session.
- **Make up gain within the In-line compressor** Make up gain is used to restore perceived level loss during compression. It can also be used as a fast and easy way to adjust the level post fader.

Master bus

- **Input trim** This knob is used to adjust the volume at the input to the master bus. This is a useful tool if you've balanced the mix with the track faders, but the overall level is still too low or high.
- **Tape saturation** The tape saturation meter does not monitor level, rather it monitors the amount of saturation effect. This effect will add harmonic density which can result in an increase in perceived loudness. Turning the tape saturation knob off will allow signal to pass unaffected.
- **Fader** Used to achieve an overall volume of all tracks in a session.
- **Make up gain within the In-line compressor** Make up gain is used to restore perceived level loss during compression. It can also be used as a fast and easy way to adjust the level post fader.
- **Final Limiter** The final limiter is a brickwall, lookahead limiter with a very fast attack and release. The threshold is preset at -1 dBFS. If a signal peak is going to exceed -1 dBFS, the lookahead limiter will reduce the volume slightly before and after the peak. Yellow indicators show gain reduction in increments of 1 dB.

Monitor Section (if enabled)

- **Attenuator** provides an output level for the speakers, as well as mute and a configurable "dim" control. Use these controls to reduce the volume in your speakers WITHOUT affecting the level of the master bus output. For more details, visit the Monitoring Section

VCA Faders (if enabled)

- A VCA Fader can be used to adjust the level of multiple faders from one master fader. For more details, visit the VCA Strip section.

10.3. Mixer Window Overview

Mixer Strips – Inputs, Mixbuses, Master, Monitor, and VCA's.

When viewed in the Mixer Window, Mixbus resembles a traditional analog console. Input Channels are on the left, while the busses and master are located on the right side.

The audio signal flow is generally left-to-right. The signal starts at the input channels (which might be live inputs, playback tracks, MIDI tracks, or virtual instruments), progresses through the mix buses, and into the master. The signal path's EQ, Filter, Compressor, panning, bus summing, and master bus processing are performed inside Harrison's proprietary signal path, are always present, and do not require user-supplied plug-ins.

In Mixbus, like other audio workstations, each recording track has its own mixer strip. These mixer strips resemble the input channels of a traditional analog mixing console. Input channels can be configured to be mono or stereo, virtual instrument or MIDI, and by default include all the DSP processing that makes up a real console channel. This is the major difference between Mixbus and other DAW mixers. No time is wasted searching for and adding the necessary high-quality mixing tools.

Just to the right of the input channels are the mix-buses. Mix bus channels are used to collect the outputs from the input channels into groups. For example, in a typical music project, you might send the drums to mix bus 1, the vocals to mix bus 2, instruments to mix bus 3, and use mix bus 4 as a parallel effects bus, like reverb. This allows you to easily adjust the relative levels of these groups and apply processing to them.

The master channel is the final mixer strip on the right. Often, the master channel will be connected to your

volume control, which is then connected to your speakers (or headphones). The output of the Master channel is the final output of your project (i.e. your 2-track master file). This output is rendered to the final .wav file when you export the session.

The monitor section on the far right has been made available for those who do not have access to a dedicated external monitoring section.

Finally, on the far right you will find the VCA Faders. You may create 0, 1 or unlimited VCA faders. VCA faders may be attached to multiple channels, and they give you the ability to mute, solo, or adjust the level of all the attached channels simultaneously.

You may adjust the size of the Mixbus and VCA “Panes” by clicking and dragging the vertical bar:

Tip: Shift+Ctrl+Click (Shift+Cmd+Click on Mac) on a track’s Mute, Solo, or Rec enable to toggle ALL the tracks in a session.

Tip: You can Ctrl+click (Cmd+click on Mac) on the track’s name slate, and make the track wide or narrow. Shift+Ctrl+click (Shift+Cmd+Click on Mac) to make ALL the tracks or ALL the buses narrow/wide.

Tip: You can also control the mixer page via the keyboard. Select a channel by clicking on it with the mouse, then you can use the ‘S’ to solo, ‘M’ to mute, ‘R’ to record-arm the track. The up/down arrow keys nudge the fader up and down, while the left/right keys scroll the mixer left & right.

10.3.1. Mixer Sidebar

On the left side of the mixer window, there is a sidebar that allows you to manage your Plugin Favorites, Tracks, and Groups. You can hide&show this list using “V*iew->Mixer List*” in the main menu.

Favorite Plugins

The “Favorites” bar allows you to store a selection of plugins, and their presets.

Mode Selector

- **Favorite Plugins:** this selection shows your favorite plugins in the list. You can add items to the Favorites list by dragging them out of the mixer strip and into this area. Alternatively you can enable and disable the “Favorites” checkbox in the Plugin Manager. The search field, at the bottom of the list, allows you to filter the plugins for their name and/or tags (just like the plugin manager search field)
- “Recent Plugins:” this option shows the 10 plugins that you’ve used most recently
- “Top-10 Plugins:” this option shows the 10 most often-used plugins in your workflow

Track/Bus List

The “track/bus” list shows a list of available tracks and buses.

- Click the ‘show’ box to hide or show a strip in the mixer page.
- Click and drag an item to reorder the strips in the mixer page.

> Reordering the mixer strips will also reorder the tracks in the Editor window.

A right-click menu provides these options:

- Hide/Show all.

- Hide/Show all tracks.
- Hide/show all buses.
- Show tracks with regions under playhead (only shows the tracks that have active audio regions right now).

Below the track/bus area is a list of the mix buses which can be shown or hidden.

Group List

Below the mixbus area is the Group section. For more information about grouping, visit the Groups topic.

10.4. Signal Flow

What is signal flow?

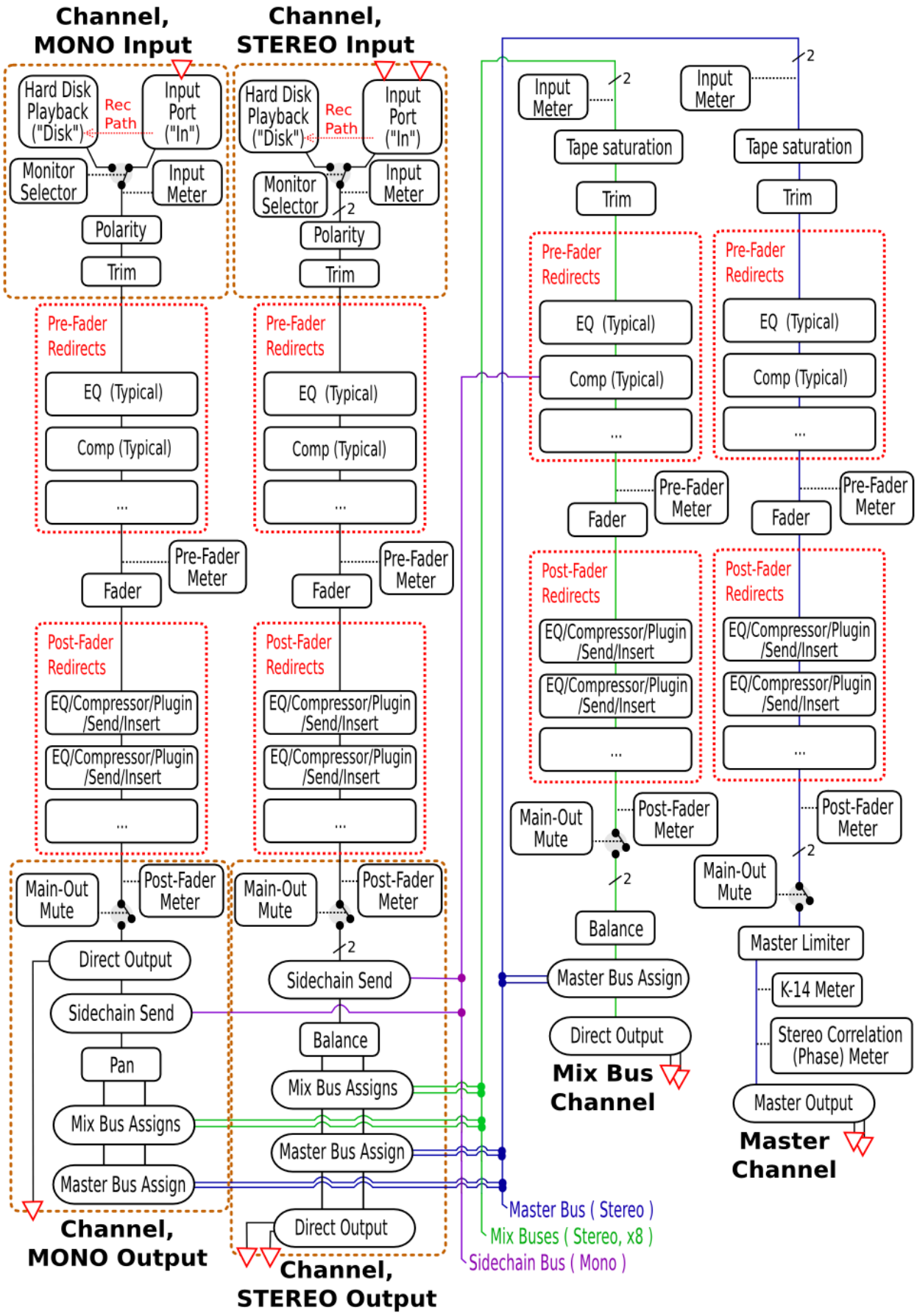
“Signal Flow” is the path that audio follows through an electronic system of wires, and/or its equivalent inside your computer. The “signal flow” of a recording starts with the performer, and ends with the listener. This incorporates the microphones, converters, computers, mixers, effects, amplifiers and speakers that make up the signal path. In this manual we refer to ‘signal flow’ of Mixbus itself. The signal flow starts with the A/D converter of your soundcard’s input, and proceeds to the D/A converter on the output. Mixbus’s signal flow consists of input tracks, mixer-strip processors, buses, and monitor stages that execute the various processes of recording, mixing, and exporting.

Mixbus’s signal flow

The Mixbus signal flow is modeled on an analog console.

Mixbus may have an unlimited number of mono and stereo channelstrips (Tracks). Each channelstrip may be assigned to any of the mixbuses, and/or directly to the master bus.

> ✻ In one small way, Mixbus differs from traditional mixing console: each of the mixbus assigns have level controls, so each of the mixbuses can be used as either an “aux send/return path” (using the level controls to adjust the send level) or a “grouping bus”, (using the hard assign buttons with the level control at unity) depending on the user’s needs. Typically in an analog console, a separate selection of aux sends and grouping buses are provided. If you are a live-sound engineer familiar with stage monitor consoles, then you already know about having a level control on each bus send, and Mixbus is exactly the same.


Channel, MONO Input
Hard Disk Playback ("Disk")
Rec Path
Input Port ("In")
Monitor Selector
Input Meter
Polarity
Trim
Pre-Fader Redirects
EQ (Typical)
Comp (Typical)
...
Fader
Pre-Fader Meter
Post-Fader Redirects
EQ/Compressor/Plugin /Send/Insert
EQ/Compressor/Plugin /Send/Insert
...
Main-Out Mute
Post-Fader Meter
Direct Output
Sidechain Send
Pan
Mix Bus Assigns
Master Bus Assign
Channel, MONO Output
Channel, STEREO Input
Hard Disk Playback ("Disk")
Rec Path
Input Port ("In")
Monitor Selector
Input Meter
2
Polarity
Trim
Pre-Fader Redirects
EQ (Typical)
Comp (Typical)
...
Fader
Pre-Fader Meter
Post-Fader Redirects
EQ/Compressor/Plugin /Send/Insert
EQ/Compressor/Plugin /Send/Insert
...
Main-Out Mute
Post-Fader Meter
2
Sidechain Send
Balance
Mix Bus Assigns
Master Bus Assign
Direct Output
Channel, STEREO Output
Input Meter
2
Tape saturation
Trim
Pre-Fader Redirects
EQ (Typical)
Comp (Typical)
...
Fader
Pre-Fader Meter
Post-Fader Redirects
EQ/Compressor/Plugin /Send/Insert
EQ/Compressor/Plugin /Send/Insert
...
Main-Out Mute
Post-Fader Meter
2
Balance
Master Bus Assign
Direct Output
Mix Bus Channel
Input Meter
2
Tape saturation
Trim
Pre-Fader Redirects
EQ (Typical)
Comp (Typical)
...
Fader
Pre-Fader Meter
Post-Fader Redirects
EQ/Compressor/Plugin /Send/Insert
EQ/Compressor/Plugin /Send/Insert
...
Post-Fader Meter
2
Main-Out Mute
Master Limiter
K-14 Meter
Stereo Correlation (Phase) Meter
Master Output
Master Channel
Master Bus (Stereo)
Mix Buses (Stereo, x8)
Sidechain Bus (Mono)

The “redirect” areas are represented by the black box at the top of the mixer strip. Sends, Inserts and Plugins are “redirects”, and they may be placed in any order, before or after the fader. If a redirect is above the fader, then it is considered a pre-fader redirect.

An important tip about signal flow: The Mixbus “bussing” structure (mix buses + master bus) is stereo-only. Typically a channel has one (mono) or 2 (stereo) inputs. This defines the channel width through the channelstrip. However, it is possible to change the channel width via plugins. For example you can have a mono-input, stereo-output effect (such as a chorus or autopanner effect). The last redirect (plugin) in the post-fader redirect box will define the signal width just before it goes into the panning/bussing section. If the last plugin has a single (mono) output channel, then the mono panner will be used. If the last plugin has 2 (stereo) output channels, then the stereo “balance” panner will be used. If the last plugin has 3 or more channels, then the stereo “balance” panner will be used and the remaining channels will be ignored by Mixbus.

The signal flow through each plugin may be modified with Pin Connections.

Not shown in the diagram is the Pre-Fader and Post-Fader send mutes. Sometimes when you mute a track, you still want a Send to remain active (for a headphone feed, for example). It is possible to disable the mute for pre- and post- fade sends separately. To enable or disable the pre- and post-fader send mute, right-click on the Mute button in the editor window.

10.4.1. Mono vs Stereo

When you create a new Mixbus track (or bus), you have the option to make the track either mono or *stereo*.

What is the difference between *mono* and *stereo*?

A *stereo* signal is typically recorded with 2 (or more) microphones such that a single sound (like a drum) is captured with both microphones; but because of their separate locations, each microphone will record the sound with slightly different timing & reflections. It is this difference (similar to the difference detected by our 2 separate ears) that causes *stereo* recordings to have additional width & depth beyond simple mono left/right "panning".

In this document we will always use the word *stereo* in *italics*, if we mean a true *recorded-in-stereo* signal. Otherwise we will use the term stereo or "stereo" (in quotes) to indicate 2-channel audio with left and right sides.

2-channel audio is not necessarily *stereo*

Mixbus is designed to support 2-channel playback throughout. If you feed a true *stereo* signal into Mixbus, then Mixbus (by default) will maintain the _stereo_-ness of that recording throughout the signal flow.

You can also take multiple mono signals, and "pan" them to the 2 channels (left and right) by adjusting the level of each channel. This is sometimes called "pan-pot stereo" or even more commonly just "stereo". But this is not strictly a *stereo* signal.

2-channel files are not necessarily *stereo*

You might receive a file that has 2 channels ("stereo") but in fact it does not represent a *stereo* signal.

For example, you might have a 2-channel file recorded with a handheld recorder. The first channel is the host interviewer, and the second channel is the guest. In this case, the file has 2 channels but the information is not *stereo* and should not be treated as such.

Some other examples might be:

- a vocal track is provided in a "stereo" file. The singer is louder in the left side. But it's the same (mono) signal in both tracks, just panned slightly left.
- a snare-drum track might be provided in the upper (head) and lower (snare) mics, but this is not a *stereo* signal.
- a kick-drum might be provided with "inner" and "outer" (sound hole) mics, but this is not a *stereo* signal.

Some workstations (like Apple Logic) will export tracks in 2-channel (“stereo”) by default, even if the information is exactly the same in both the left and right tracks (i.e. center-panned mono). In this case, you might want to import the track as 2 mono tracks, and then discard the redundant track from the editor.

In Mixbus, what is the difference between a mono and stereo CHANNEL (i.e. TRACK)?

Mono channels/tracks

Mono channels use a **panner** – this means that the mono signal is sent to the right and left channels simultaneously, and then one side or the other is attenuated to make the sound appear farther to the left or right of the stereo field.

Stereo channels/tracks

Because *stereo* is a special case, Mixbus’s *stereo* channels operate differently. A *stereo* channel uses a **balance** knob instead of a pan knob. A balance knob preserves the left channel in the left speaker, and the right channel to the right speaker. Then one side or the other is attenuated to “balance” the signal to the left or right. But the left signal is never mixed into the right side, or vice versa. This preserves the *stereo* phase information in the signal.

When you mix a combination of mono-panned and *stereo* signals into a stereo mixbus, then the mixbus applies a “balance” control as well, and this maintains the phase relationship of the *stereo* signals throughout the mix.

Mixbus also treats *stereo* signals differently in the channel’s compressor: it is important that the compressor’s gain-reduction works on both left and right sides equally. This prevents the stereo image from shifting left or right when the signal is compressed. Mixbus’s stereo channelstrip compressor does this correctly. This is another reason that, when you have a true *stereo* signal, you should assign it a *stereo* channelstrip. Similarly, if 2 channels are together in a file but not really “stereo”, then you should create 2 mono strips instead of a stereo one.

How do I resolve the problem if I’m given tracks/files that don’t match my needs?

The easiest way to solve your mono/stereo questions is to resolve it during file “import”.

When you import a file, in the import dialog, there is a “mapping” pulldown menu which allows you to either:

a) select “one track per channel” to split selected stereo files into mono tracks.

b) select “one track per file” to import stereo files as stereo tracks.
c) select “sequence files” to import all the selected files into one track, sequentially.
d) select “merge files” to combine 2 mono files of identical length into a stereo track. (this option ONLY appears when 2 files of identical length are selected).

How do I convert a mono signal to *stereo*?

Strictly speaking, a mono signal cannot be converted to *stereo*. It wasn’t recorded with *stereo* phase information from a spaced pair of microphones.

However there are several ways to make the mono signal occupy both channels in a left/right “stereo” mix:

- Every mono channelstrip has a “pan” knob that allows you to pan the signal left or right.
- Insert a plugin that changes the channel’s signal from mono to stereo. Normally Mixbus uses only the “left” side of a plugin, if you insert a stereo plugin on a mono channel. But you can right-click on a plugin and choose “pin connections”, “add output” to split the channel into stereo for a mono-in, stereo-out plugin.
- Assign the channel to a stereo mixbus, and apply an effect that enhances the left/right panning, such as a reverb, haas delay, auto-panner, chorus or doubling plugin.

How do I convert a *stereo* signal to a mono signal?

Generally speaking, if you have a true *stereo* signal, you shouldn’t just sum the 2 channels together and make a “mono” signal. The different time/phase relationship between the left and right sides will cause frequency cancellations and other artifacts.

If you have a truly *stereo* signal, it is rarely necessary or desired that you would want to make it mono. One exception might be the case where the *stereo* mic selection/location was poorly chosen. In this case, perhaps the best solution is to use only the “left” side, or the “right” … whichever sounds better for your needs.

If you have a *stereo* signal and you feel you must make it mono, some solutions might be:

- Assign the stereo track to a mixbus, and use the mixbus’s width control to make the signal “mono”. (this mono-summing happens at the top of the channelstrip, before any plugins)
- Insert a plugin that provides a stereo-width control. (optionally: right-click on the plugin and choose “pin connections” if you want the plugin to have a single (mono) output)

How do I convert a *stereo* track to 2 *mono* tracks?

If you are “importing” a track to mix, the best way is to make this decision during the import operation. Select the option “one track per channel”, and your stereo file will be split into 2 mono tracks which you can manipulate separately.

If you have recorded a stereo track, and later want to split it into 2 mono tracks, you should first ask yourself why you did this recording in stereo, at all :)

Mixbus provides a Lua script which can split a stereo track into 2 mono tracks. The stereo track's Playlist (including all edits) will be split into left and right components, and assigned to 2 new tracks. Each track has the same processing applied as the source track. To access the script, click Window->Scripting->Actions->Stereo To Mono->Run

How do I convert 2 *mono* tracks to a *stereo* track?

If you've read the prior information on this page, you might understand why converting 2 mono track to "stereo" is not as easy as it sounds. Consider the case where each of the mono channels has slightly different plugins, EQ, edits, and other details. We can't trivially convert 2 mono tracks into a single stereo track.

The only sensible way to convert 2 mono tracks into stereo is to solo the 2 tracks, export the mix to stereo, and re-import the resulting file as a stereo track. (then de-activate or remove the original mono tracks) We might make a convenience function that does this automatically in the future.

In many cases, you don't want to convert the tracks to stereo at all; instead you should consider making a "group" for the 2 tracks. This will "tie the tracks together" so that selections, fader levels, colors, and other parameters will be changed together, as if they were a single track. But you'll still have individual panning, plugins, EQ's and such. Since you can't "really" make 2 mono signals into *stereo* anyway, this is usually the best choice.

10.5. Bussing

https://www.youtube.com/embed/cuInwhnrtlk?rel=0

What is a Bus?

A *bus* is a signal path onto which one or many audio signals are combined to arrive at a common place for further processing. In the earliest days of analog circuitry, a “bus” was simply a wire onto which a number of signals were summed together, and then directed to the next stage of processing.

In digital audio, the electrical wire is replaced by a series of numbers representing a signal that is the arithmetic sum of a number of individual signals. This final value is then passed to the next processing stage.

Buses are typically used for these fundamental mixing tasks:

1. Combining multiple signals onto the “master bus”, or *main mix* which feeds your speakers or headphones.
2. Making a *monitor* mix, similar (but not necessarily identical) to the master bus, in which an artist can hear themselves and other musicians.
3. Combining a subset of similar signals into a *subgroup* or “group bus”: for example you can make a bus that combines the drum tracks into a Drum Bus.
4. Developing a secondary (*effects* or *aux*) mix, which is then processed with an effect such as reverb. For example you can “send” multiple signals to a single reverb effect, with varying levels.

Rather than adding a plugin to every channel, using buses can be less taxing on your computer. Inserting a plug-in, such as a reverb, on a bus saves the need of inserting a reverb across many individual tracks. This also ensures the plug-in applies the same effect across all the tracks being bussed for a more coherent and realistic sound.

A “bus” has no recorded audio stored in it like an audio track does. A bus serves to sum and processes audio signals “live”. Buses do, however, have a representation in the editor timeline, so their controls can be ‘automated’.

How do I use buses in Mixbus?

Mixbus has a built-in bus summing structure that is very similar to an analog mixing console. Visit Sends Section to learn more.

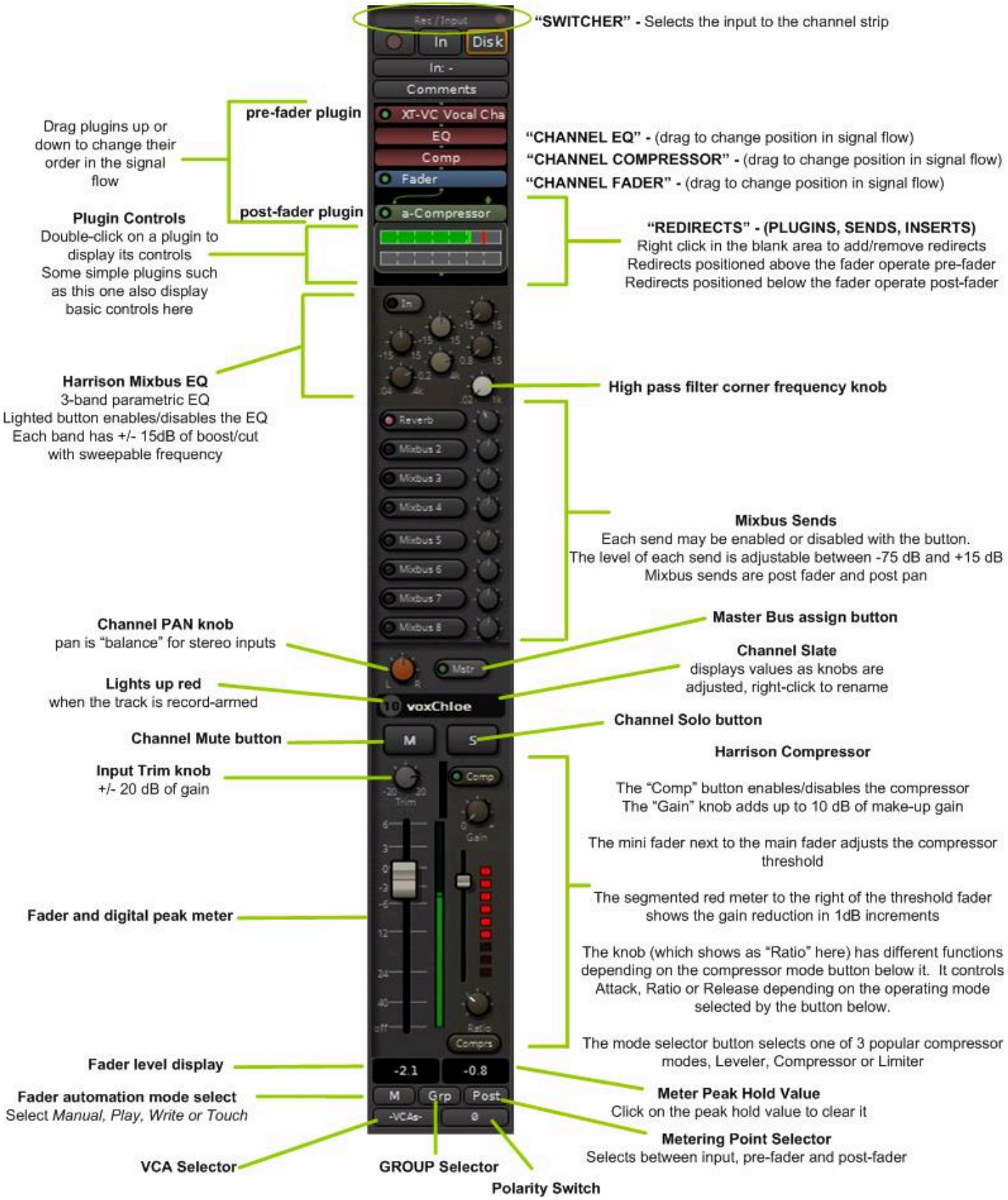
Mixbus *also* has the ability to create an arbitrary number of “buses” (some DAWs call these Aux Tracks, or Inputs) from the “New Track/Bus” dialog. You can connect these buses in various ways via the the channel input, outputs, and “sends” mechanism. You can learn more about this in the DAW-Style Mixing Buses section.

However, we encourage users to focus on the provided mixbus summing architecture, which operates like an analog console. In most cases, this will result in the easiest, fastest, and best-sounding operation. This bussing structure is what makes Mixbus special.

10.6. Input Strip

The input channel strip is the fundamental element in the mixing window. This is where an audio signal is routed, processed, and then sent to either a mix bus or the master channel. An input channel strip can be mono or stereo.

- New in Mixbus v6 are the positions of the Polarity and VCA assign buttons. They are now always displayed at the bottom of the channel strip.



The features of the input strip include:

- The Rec/Input switcher – see section Rec/Input ("Switcher") Section

- The redirect box – see section Redirects – Plugins, Sends & Inserts

- The EQ section: see section EQs
  - EQ “IN” – Enables the EQ section
  - 3 EQ bands, each with +/-15 dB gain:
    - low and high bands are shelving with adjustable corner frequency
    - the midrange band is parametric with proportional-Q peaking (Q increases with additional boost/cut) with frequency and cut/boost variable

- highpass filter knob – changes the the highpass filter corner frequency
  - filter is second-order (12 dB/octave) and always enabled when the EQ is enabled

- Mixbus Sends Section: see section Bussing
  - 8 mixbus sends, each with On/Off switch, and level control

- Stereo Pan Knob
- Master Bus Assign (“Mstr”) : when this is lit, the signal will be sent directly to the Master bus (*in addition to* any mixbus assignments) post panpot

- Channel Mute and Solo

- Input Trim knob : the input trim value is applied immediately after track playback, and BEFORE any plugins in the redirect box

- Fader
- Meter : the channel meter is a digital peak-reading meter; the scale is the same as the fader and the compressor threshold

- The Compressor Section: see section Compression Techniques
  - Comp “IN” – Enables the compressor section
  - Makeup Gain – provides up to 10dB of Makeup Gain; this gain is only active when the compressor is enabled
  - Threshold – the threshold indicates when the compressor will begin working on the signal; reduce the threshold slider to compress more
  - Gain-Reduction meter – indicates gain reduction, in 1dB increments
  - “Speed” & “Mode” knobs : see Compression

- Fader level : this displays the curent fader position, in dB. Click to enter a value.
- Meter peak-hold : this displays the loudest digital-peak value that has been detected by the meter. It turns red when the value is over 0dBFS. Click to reset the value.

- Fader Automation selector : select between Manual, Play, Write, and Touch. This control *only* affects the channel’s fader
  .
- Group selector : click this button to assign the channel to a Group (see: Groups)

- Meter Position selector : click this button to select the meter position (Input, Pre-Fader, Custom, Post-Fader, Output)
- Polarity switch
- VCA Assign

View Pulldown

There is an additional method for displaying the Rec/Input menu on the channel strips. Rather than clicking the button on the channel strip, the View pulldown menu provides either a menu click or keyboard shortcuts for switching the display. The display changes for ALL channels, not just a selected channel. The figure below shows how to access this control and also shows the associated keyboard shortcut.

Other Buttons and Controls

Polarity

- Click this button to invert the signal's polarity. On a stereo channel, you will see 2 polarity switches, one for each channel.

VCA

- Click this button to assign one or more VCA faders to control this channel's fader and mute. See the section VCA Strip for details.

Grp

- Grouping links faders together for common control. Click this button to assign this channel's fader to a group. See the section Groups for details.

M

- Click this button to select the desired fader automation mode. The choices are Manual, Play, Write, Touch and Latch. See the section Fader, Pan and Plug-in Automation for details.

General Operation

Channel strip operation is the same as everywhere in Mixbus.

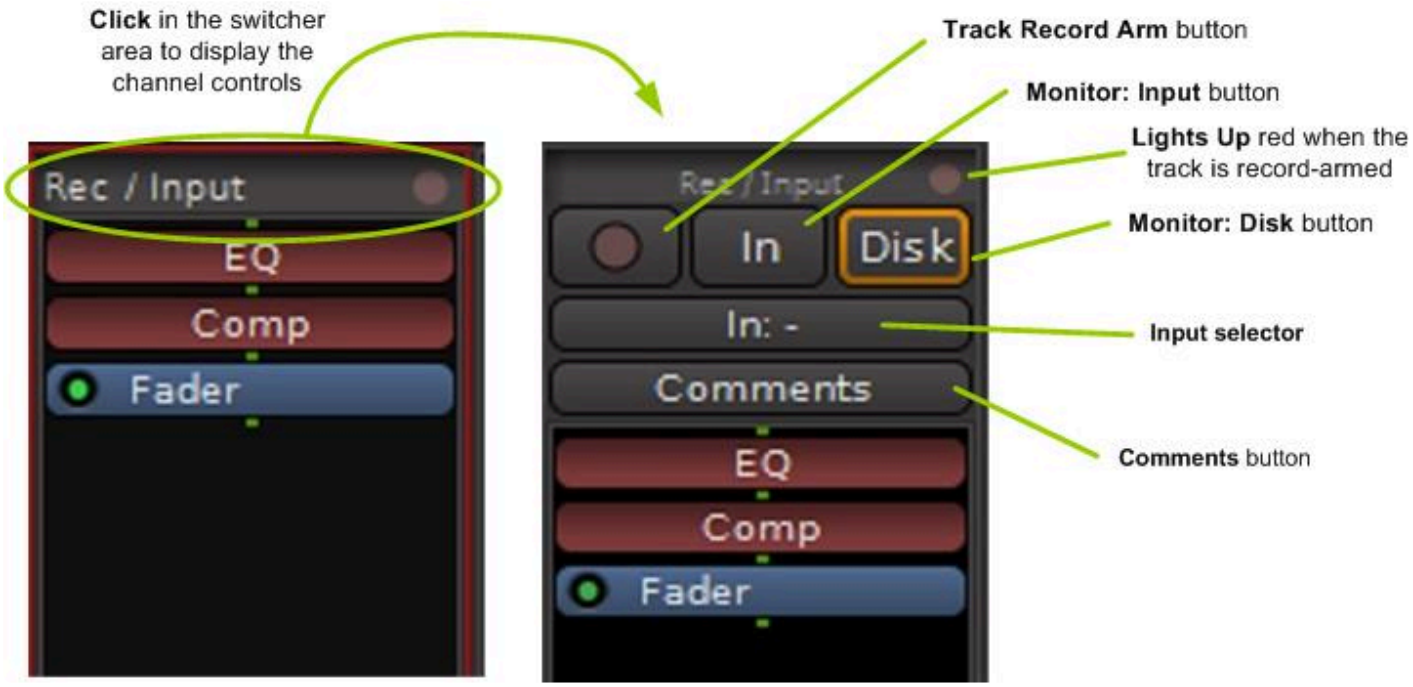
- mouse left-click on a button to activate it
- to turn a knob or move a fader, click and hold on it and move the mouse up and down
- double clicking on most knobs will snap it to its centre position, 0dB position (3 o'clock) on a mixbux level knob
- right-clicking on the channel slate will bring up a menu where you can change the name, colour, input or output routing, and other options:

- right clicking on the compressor button allows you to select between leveller, compressor or limiter operation:

10.6.1. Rec/Input (“Switcher”) Section

At the top of any mixer strip, you’ll find a “Switcher” control which allows you to toggle the visibility of the Rec/Input section, which provides controls for the Record-arm, Input selection, Monitor modes, and Comments. When the internal controls are hidden, the Rec/Input switcher provides an overview of the channel’s record-arm status.

> ✻ Click on a “switcher” to change ALL channelstrips to show the same controls. This is a quick way to view all of your channel I/Os and other settings while preparing for your mix. Then Click again to switch all the channels back to show the signal flow.



The track controls vary slightly between Tracks, Buses, Mixbuses, and the Master strip. The buttons on an Input strip include:

- **Record arm**: Click this button to arm the track for recording. An armed track shows a solid red dot.
- **Monitor: Input** : This button forces the channel to monitor it’s input. See monitoring
- **Monitor: Disk** : This button forces the channel to monitor it’s disk playback. See monitoring
  - In version 6 both monitor input *and* monitor disk can be enabled
- **In: –** : Use this button to select the soundcard input(s) or other sources to feed this channel.
- **Comments Button** : Click this button to enter comments for the channel.

On Mixbuses and the Master strip, only the **Comments Button** is presented. It will displayed or hidden when the switcher on **any** channel strip is selected.

10.6.2. EQ Section

Channel Strip EQ

The Channelstrip EQ provides a very wide range of frequency and gain changes (up to 15dB!) to help tame your tracks. The high and low bands are shelving filters for general tone sculpting, while the middle band is a “proportional Q” bandpass design.

The proportional-Q design was used in the original Harrison consoles to allow an engineer to use the EQ as either a subtle tone shaping tool, or a steep filter problem-solver, with only 2 knobs. As the gain is progressively increased (or cut), the bandwidth becomes progressively narrower (“Q” increases). A typical technique to eliminate an unwanted tone or resonance is to increase the gain to maximum cut, sweep the frequency to find the offending sound, and then turn the gain to the minimum cut required to “notch” out the unwanted frequencies (remembering that reducing the amount of cut widens the bandwidth and attenuates a wider range of frequencies).

Specifications:

Gain Range +/- 15dB

Frequency Ranges (adjustable):

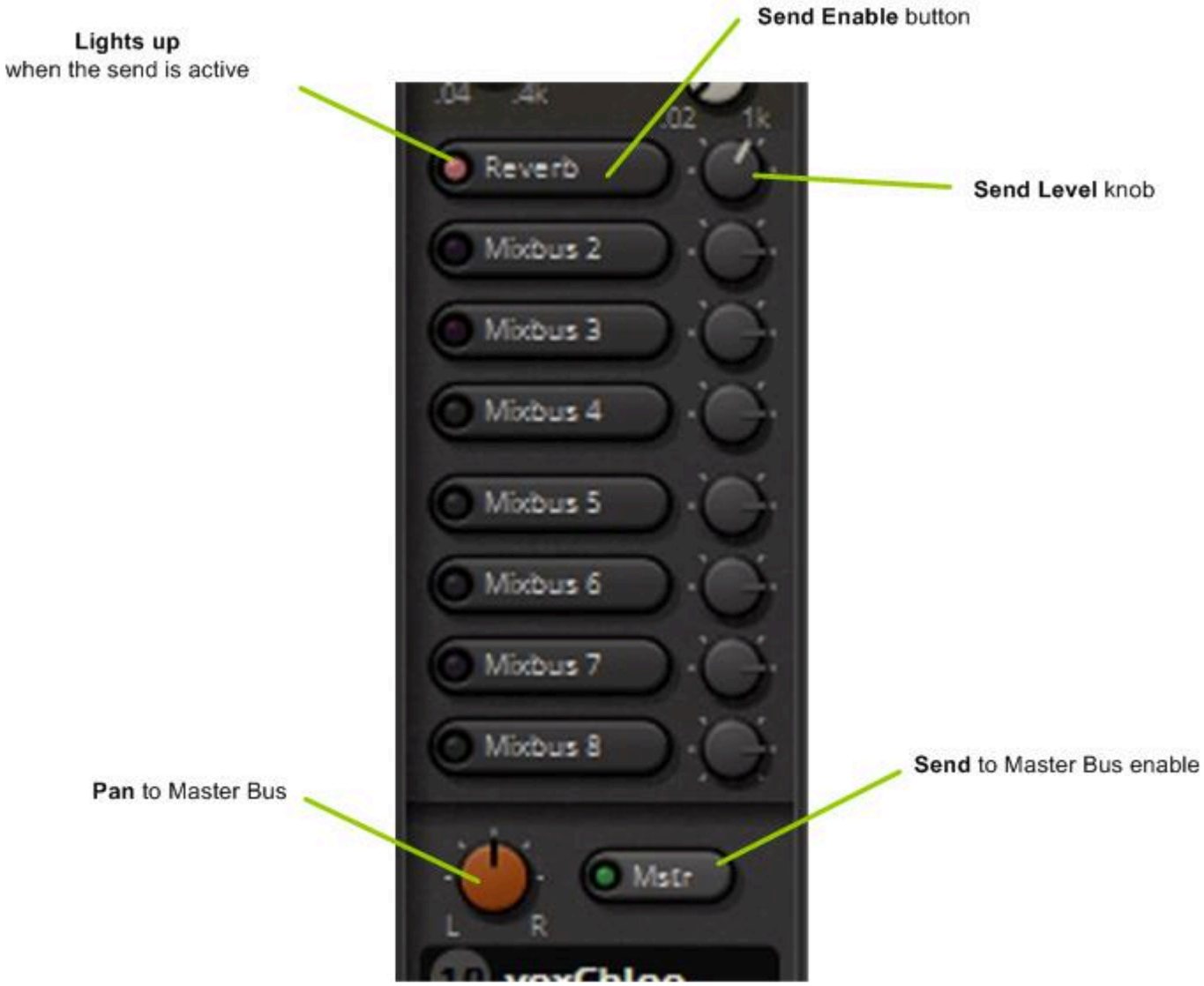
- HI: 800 Hz to 15,000 Hz Q=1.0 Shelf
- MID: 200 Hz to 4000 Hz Proportional-q bell
- LO: 40 Hz to 400 Hz Q=1.0 Shelf

HPF 12 dB/octave, frequency ranges 20 to 1,000 Hz

You can copy EQ settings from one track to another by dragging the EQ processor (in the processor list at the top of the channelstrip) onto another channel's EQ processor.

10.6.3. Sends Section

The picture below shows the Mixbus v6 send controls:



Mixbus provides these bus send features from each input strip:

- 8 stereo mixbus “sends” which correspond to the 8 available Mixbus Strips in the mixer window.
  - Each mixbus send has an on/off indicator, and a level control.
  - Mixbus sends occur at the end of the processing redirect list, and therefore are post-fader and post-pan (by default, they follow the master bus pan setting)
  - In the image above Mixbus send “1” is enabled, and the name corresponds to the name set on Mixbus strip 1

- Master bus, with on/off indicator and stereo pan control

From an input channel, you may send to any combination of the multiple mixbuses and/or master bus simultaneously. Latency compensation is provided in the mixer so that these signals will reach the master bus with coherent timing.

Bus Assignment

To assign to a mix bus, click on the button or the label for the desired mix bus. A lighted button will indicate an active assign to the mixbus. All of the lit buttons have slightly different colours to help distinguish them. To adjust the level of the bus send click and hold on the knob, moving the mouse up or down will turn the knob in the desired direction. Double clicking on the knob will restore it to the default 0dB position at approximately the 3 o'clock position. If the bus send is OFF, double clicking the knob will also turn it ON.

As all mix buses are stereo, the channel strip signal is panned into the selected mix buses. By default a channel is panned to a mixbus according to the position of the master bus pan knob.

Utilizing a mixbus as a "Group Bus":

To use the mixbus send as a "group bus", you will typically click the Bus assignment button, and leave the bus send knob at the default position of "unity" or 0dB.

For example, you might assign all of the "drum" tracks to mixbus #4, and name it "Drum Bus". In this case, you have a single volume control (fader), EQ, compressor, and series of plugins that can be applied to the drum group overall.

By leaving the master send on and engaging a Mixbus send, you may hear both audio feeds simultaneously. The result will be an increase in volume. For that reason, you might want to use the "exclusive assign" shortcut, by holding Ctrl (or Cmd on Mac) which will assign the channel to the desired bus, and immediately de-assign from any other buses, including the master bus.

Utilizing a mixbus as an "Effect Send"

To use the mixbus send as an "effect bus", you will typically click the Bus assignment button, and then adjust the "send" knob to the desired level.

For example, you might assign the snare drum to mixbus #4, where you've placed a reverb plugin. You can now adjust the level of the snare in the reverb bus, as compared to other instruments that are also feeding the reverb bus.

Bussing shortcuts

- Control clicking (Command on Mac) a bus on/off will exclusively-assign the track to that bus (other buses will be turned off, so the track ONLY goes to the selected bus)

- Shift+Control clicking on a bus will assign ALL tracks to that bus. This is a quick way to send (or remove!) every track from a bus.

10.7. Mixbus Strip

There are 8 “mixbus” buses. All 8 are fed from the mix bus sends of the input channels.

"MIXBUS EQ" - (drag to change position in signal flow)
"MIXBUS COMPRESSOR" - (drag to change position in signal flow)
"MIXBUS FADER" - (drag to change position in signal flow)

"REDIRECTS" - (PLUGINS, SENDS, INSERTS)
Right click in the blank area to add/remove redirects
Redirects positioned above the fader operate pre-fader
Redirects positioned below the fader operate post-fader

The example here shows a reverb plugin inserted pre fader

3 band EQ
with on/off button, +/- 9 dB
range at fixed frequencies

Stereo Width knob
Variable blend control of the stereo input to the mixbus, from mono sum (full counterclockwise) to fully separated stereo (full clockwise)

Tape Saturation Distortion Meter and Drive knob
This is an emulation of saturation characteristic of recording tape. The Drive knob controls the amount of additional distortion added to the signal.

Spill Button
Click this button to show ONLY the channels assigned to this mix bus in the channel strip area

Mixbus Master Bus Send, Slate, Fader, Compressor
These controls all work exactly the same as on the channel strip.

Mixbus EQ

Each of the 8 mixbuses provides a 3-band “Tone” control.

The Mixbus EQ provides carefully chosen center frequencies, and wide tone-shaping shapes, to provide overall tone control of the signal on the bus. The gain range is slightly lower than the channelstrip EQs and the bandwidth is wider, to provide more nuanced controls.

Specifications:

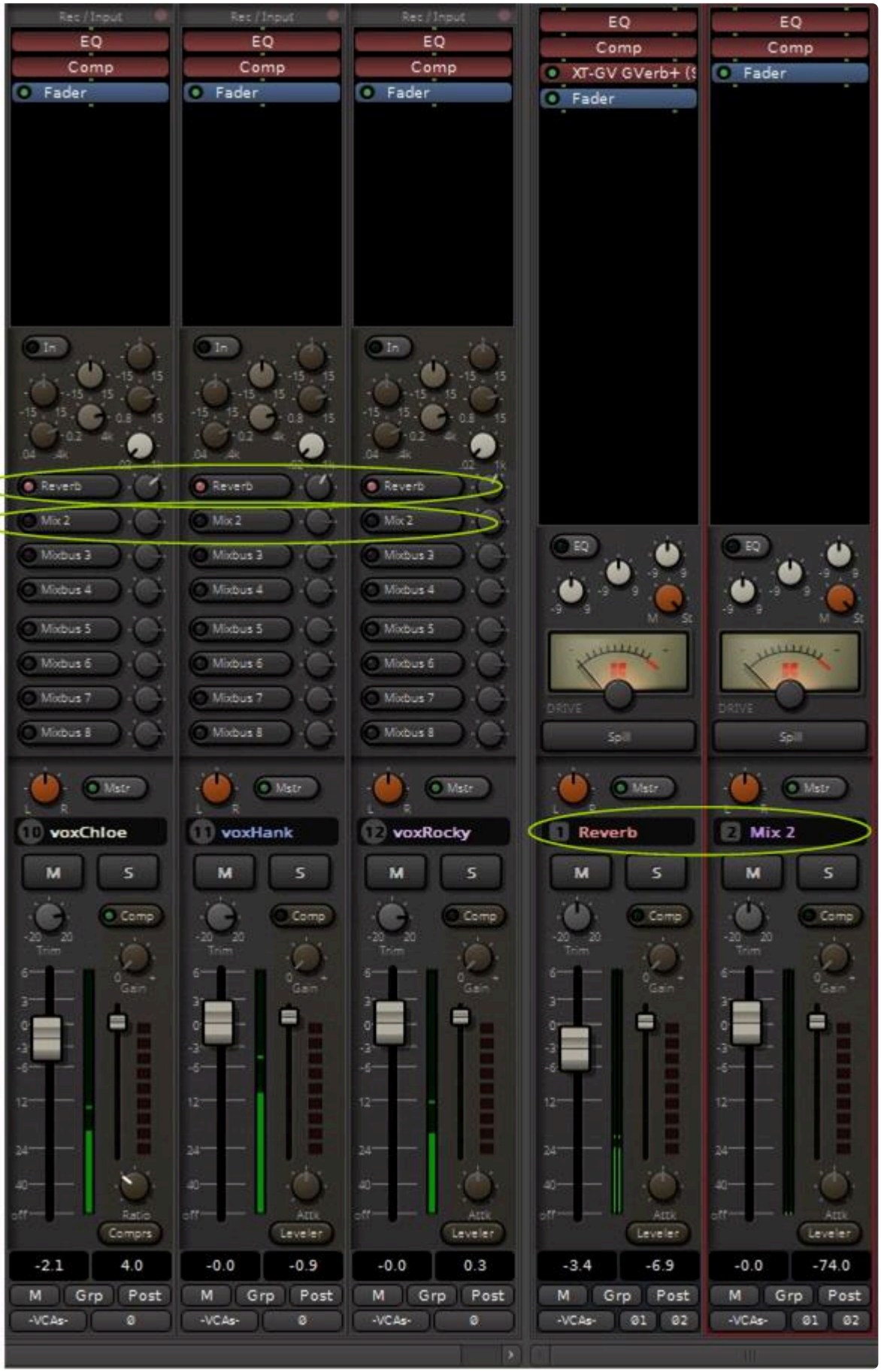
Gain Range +/- 9dB

Frequency Ranges (fixed):

- HI: 2000 Hz Q=1.0 Shelf
- MID: 800 Hz Q=0.7 Bell
- LO: 300 Hz Q=1.0 Shelf

Additional Feature

- If the Mixbus is renamed, the new name will be propogated beside that mix bus’s send knob on all channel strips:


Rec / Input
EQ
Comp
Fader
In
Reverb
Mix 2
Mixbus 3
Mixbus 4
Mixbus 5
Mixbus 6
Mixbus 7
Mixbus 8
Mstr
10 voxChloe
11 voxHank
12 voxRocky
1 Reverb
2 Mix 2
M
S
Comp
Trim
Gain
Ratio
Comprs
Attk
Leveler
-2.1
4.0
-0.0
-0.9
-0.0
0.3
-3.4
-6.9
-0.0
-74.0
M
Grp
Post
-VCAs-
0
01
02
XT-GV GVerb+
DRIVE
Spill

10.8. Master Strip

Mixbus provides a single Master channel. The Master channel provides the final output signal for listening via speakers or exporting to wav files.
The Master channel has a built-in transparent signal limiter (with look-ahead) to avoid the distortion that occurs in overly loud digital signals.
The various controls on the Master are very similar to those on a channel strip and all operate exactly the same way.

"MASTER BUS EQ" - (drag to change position in signal flow)
"MASTER BUS COMPRESSOR" - (drag to change position in signal flow)
"MASTER FADER" - (drag to change position in signal flow)

"REDIRECTS" - (PLUGINS, SENDS, INSERTS)
Right click in the blank area to add/remove redirects
Redirects positioned above the fader operate pre-fader
Redirects positioned below the fader operate post-fader

3 band EQ
with on/off button, +/- 6 dB
range at fixed frequencies

Tape Saturation Distortion Meter and Drive Knob
This is an emulation of saturation characteristic of recording tape. The Drive knob controls the amount of additional distortion added to the signal.

Master Limiter and Gain Reduction Meter
The meter shows the gain reduction in 1 dB increments

Stereo Correlation Meter
Indicates the stereo "spread" and mono compatibility of the master bus output.

K-14 Loudness Meter
indicates the average perceived loudness of the stereo output signal, using the K-14 meter scale

Mixbus Master Bus Send Fader, Compressor
These controls all work exactly the same as on the channel strip except that there is no "solo" button.

Master Bus EQ

The Master Bus EQ provides very fine control of the final output tone. The band frequencies are carefully chosen to address the problem areas of a typical pop music mix: very low frequencies, mid-low frequencies, and “very high” frequencies (leaving the primary mid-band relatively unchanged, as the reference) The limited gain range (+/- 6dB) provides for very fine adjustments.

Specifications:

Gain Range +/- 6dB

Frequency Ranges (fixed):

- HI: 4000 Hz Q=1.0 Shelf
- LO/MID 300 Hz Q=0.7 Bell
- LO 90 Hz Q=1.0 Shelf

Monitor Panel Button

Optionally, a monitor section can be added between the Master Bus output and the physical audio interface (soundcard input, etc) so that monitoring can be controlled without affecting the actual Master Bus output.

- When the Monitor section is added, another button is added to the Master strip which controls display/ hide of the monitor strip.
- The monitor panel is added or removed by selecting Session -> Monitor Section -> “Use Monitor Section”

See the details of the monitor panel and its functions here: Monitor Strip

10.9. Monitor Strip

The “monitor section” is a signal path that may be inserted between the master bus in Mixbus and the input of your interface, particularly useful if you do not have a good hardware monitor controller in your setup. The Monitor Section allows you to control what you hear, without affecting the main mix which will be exported. It is important to avoid using the master bus fader as your “volume control” while mixing, because changes to the master fader will be applied to your final mix, which not only changes the volume of the mix, but can also affect any post-fader processing (such as the Master Bus Limiter or the Tape Saturation on the output bus). Instead you should use the master bus to set the level of the final mix, and use a separate volume control to adjust your “listening level”. The Monitor section in Mixbus provides a way to do this, and more.

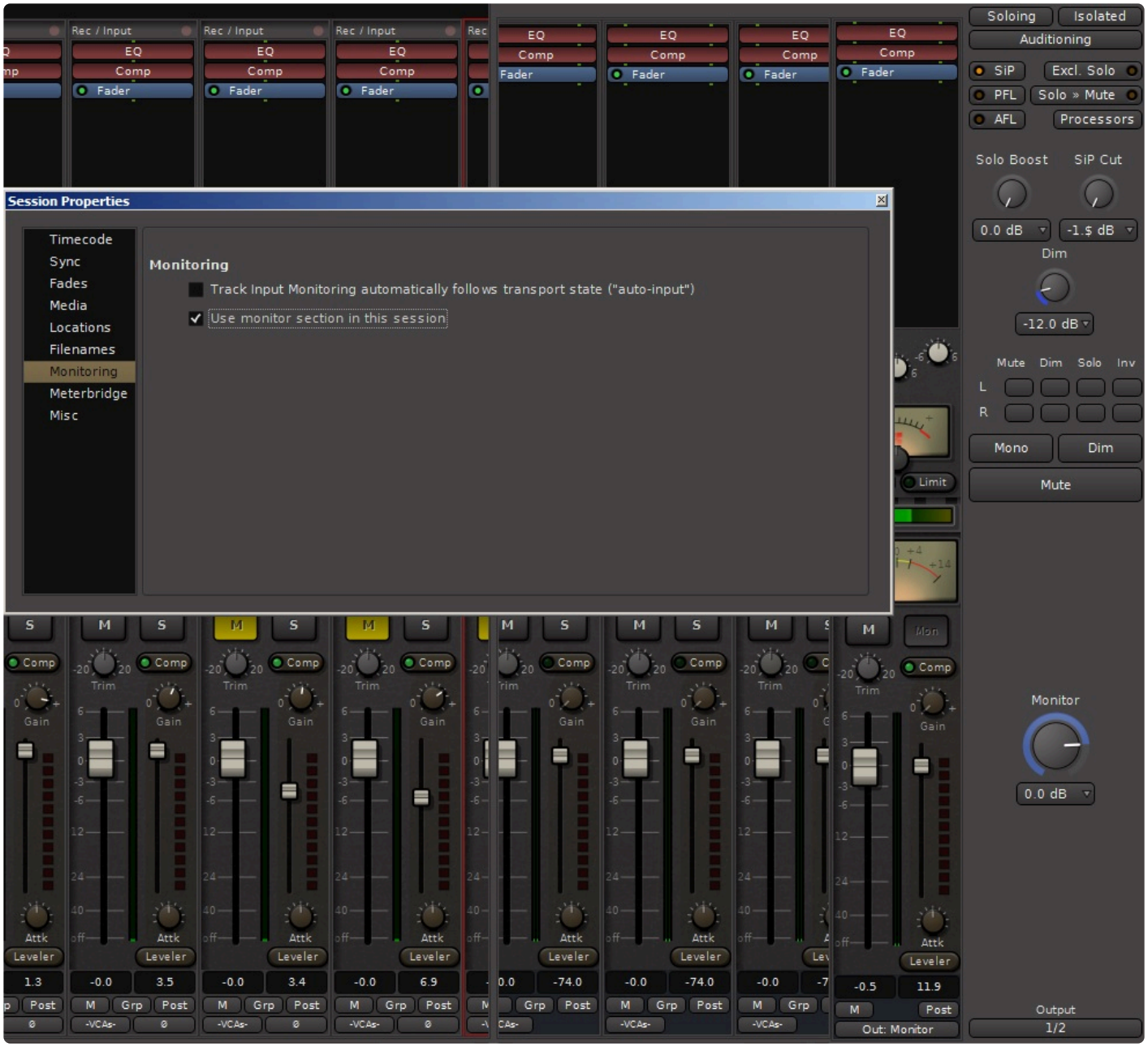
This “listening level” can be provided by your laptop speaker volume, your powered speaker volumes, a mixer, or a dedicated monitoring solution. In many cases, such as mixing on-the-go with headphones and no external gear (among many other examples), the Mixbus “monitor” section might be the preferred option for controlling your listening volume.

The monitoring section serves these functions:

1. It provides a “volume control” that can be used to adjust your listening level, without affecting the level of your actual mix.
2. It allows some conveniences such as “mono”-izing your mix, or muting speakers, so you can audition your mix in mono.
3. Because the monitor path provides a separate listening path outside of the mixer, it is possible to send Solo’ed tracks to the monitor, without affecting what is happening to the main mix. This is called AFL or PFL soloing. (more details below)
4. It allows the insertion of plugin processors. These processors can be used for room EQ or other corrective measures. The processors only affect the listening path, not the main mix.
5. It allows the selection of any physical output ports on your D/A converters. This can be used to switch between alternative monitoring systems.

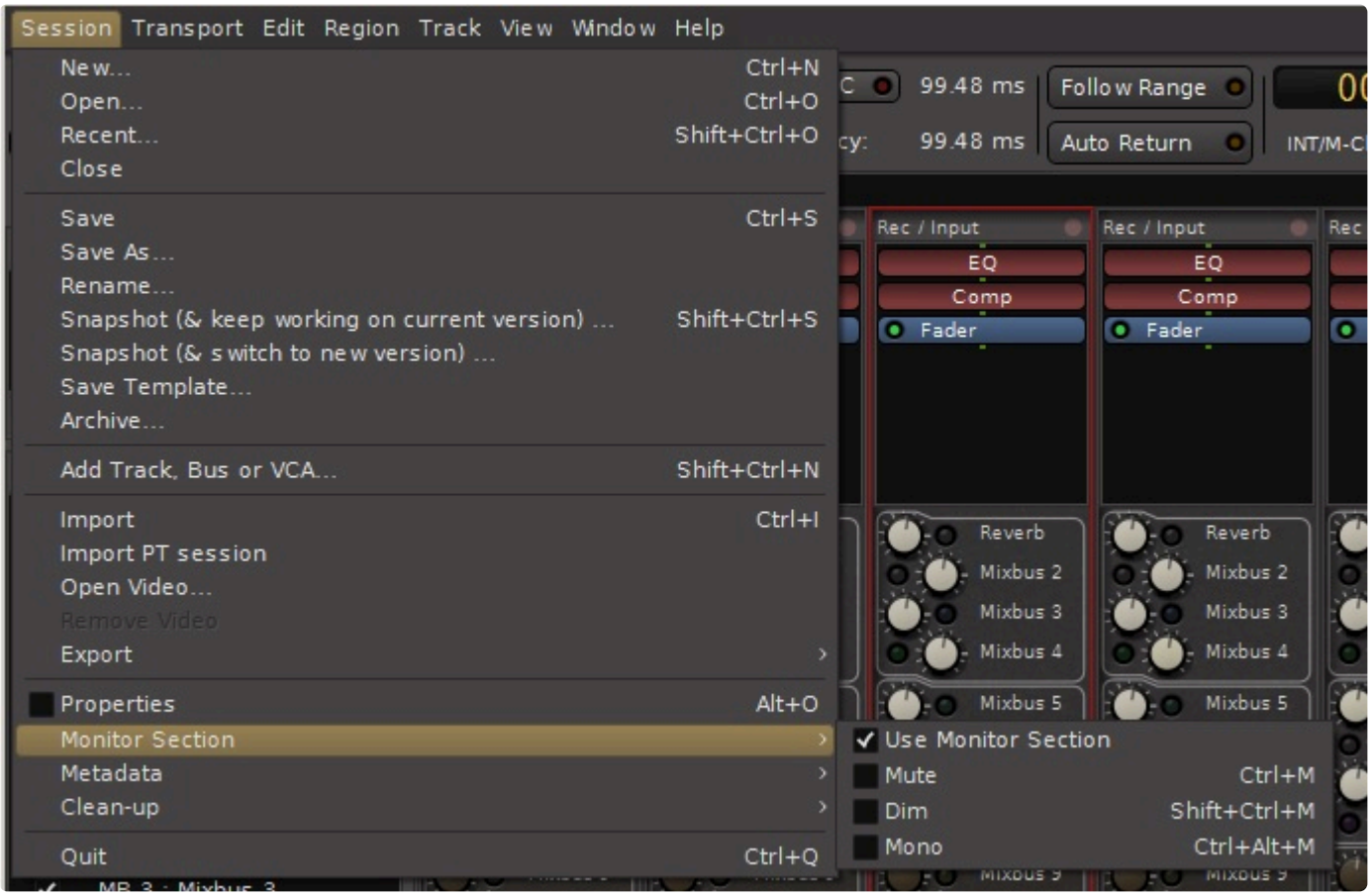
The monitor panel is added or removed in 2 ways:

- selecting Session -> Properties – > Monitoring -> “Use Monitor Section in this session”:



OR

- selecting Session -> Monitor Section -> “Use Monitor Section”:

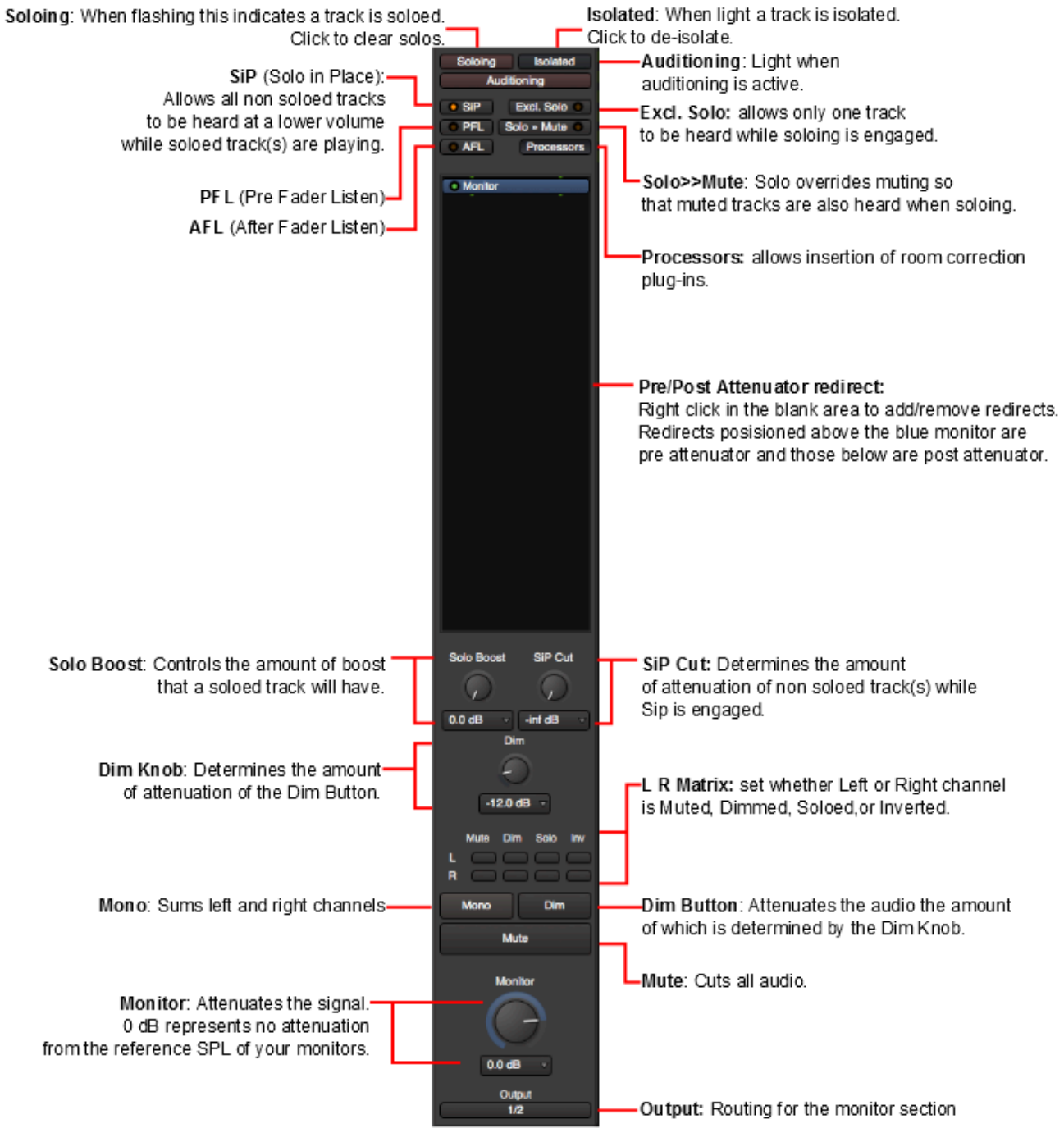


Once activated, the monitor section will be visible on the rightmost side of the Mix window as shown.

> The Monitor volume knob does not adjust the volume of the Metronome (“click”). By default, the outputs of the click are connected directly to the first 2 outputs on your soundcard, which typically feed your speakers directly. You can adjust the volume level of the click by visting Preferences->Misc->Click. And you can change the outputs of the click by visiting Window->Audio Connections.

The Mixbus Monitoring Section:

Monitor Section



Examining the Various Volume Controls in a Typical System

The Volume Control on Your Monitors

Usually you can find a volume control on your monitors (if they are active) or on an outboard amp (if they are passive). This volume control is suited to calibrating your monitors to a reference SPL. Once calibrated you can forget about this control other than periodic re-calibrations (SPL calibration of monitors is beyond the scope of this document but much information is available on the internet).

The Audio Interface, or external Monitor Controller

Some people prefer to use a Monitor Controller to control the DAW output, as well as a few other sources, that can also drive multiple sets of speakers and headphones. These devices have various features that allow you to control exactly what you are listening to. Sometimes, audio interfaces can also provide this functionality with either a hardware or software based volume control that accesses the internals of the interface. Some are elaborate and some are simple; consult the documentation for your interface to find details.

The Master Fader

The pitfall of using your master fader as a monitor level control:
Lets say you adjust your master fader to achieve the ideal monitoring level, and later you attenuate it to give your ears a break. If you forget about this attenuated setting it could easily lead you to make some poor mixing decisions later. Additionally, you may print your mix at this attenuated level (bad mix decisions and all) and not realize it. And, the dynamic tonality of the mix can be modified if you lower the master fader and the output “hits” the Master Bus Limiter and Tape Saturation differently, providing more or less limiting and saturation.

> Whether your monitors are calibrated to a reference SPL or not, a monitoring section is still vitally important in achieving a consistent sound.

10.10. VCA Strip

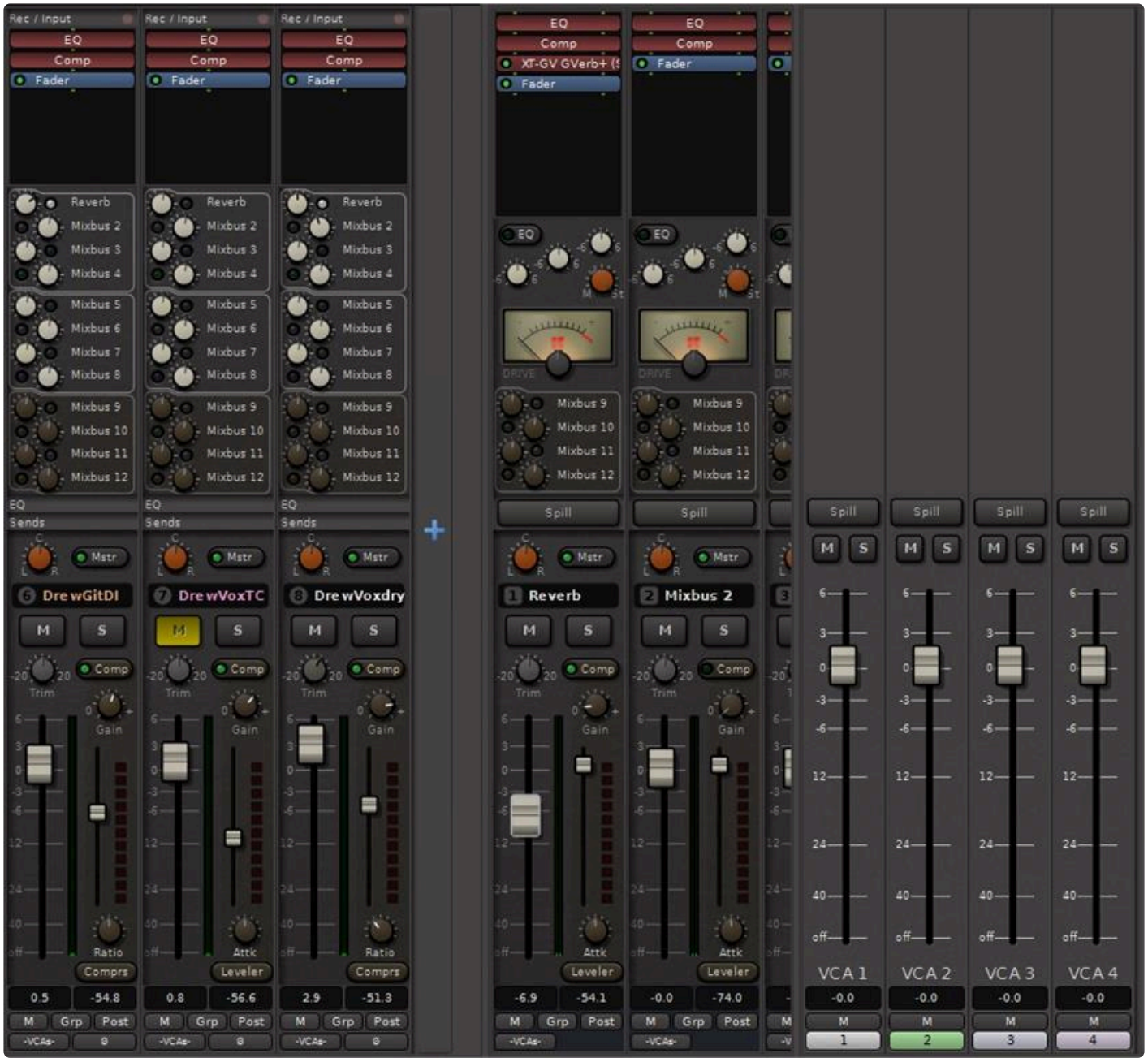
What is a VCA Strip?

A VCA, or VCA Master, is similar to an input strip; it has a fader, mute and solo button; but it does not pass audio itself. Instead, it controls the value of the **mute**, **solo** and **fader** of the channels that are “attached” to it.

Watch this video to learn more about Mixbus’s VCA implementation:

https://www.youtube.com/embed/4yr-eodbdvs?rel=0

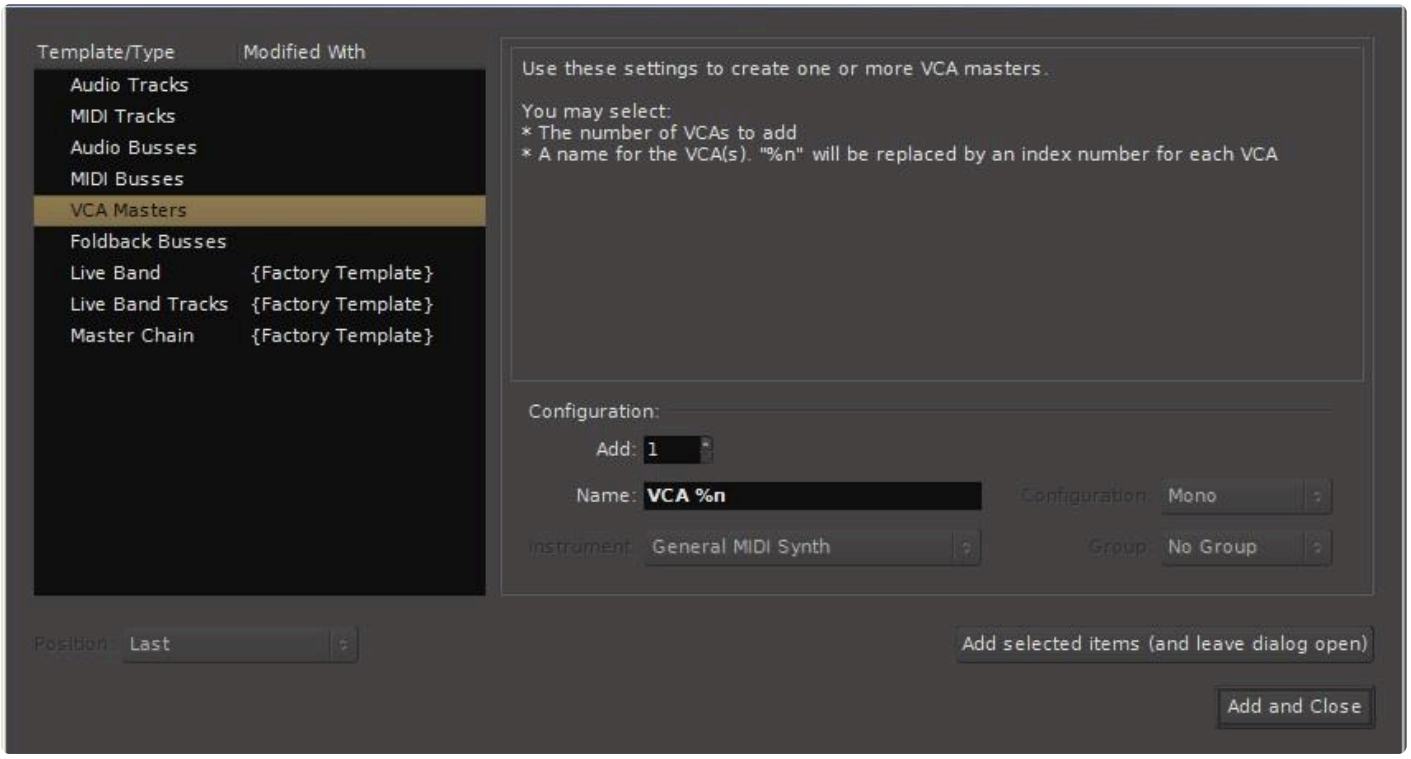
In the screenshot below, 4 VCA masters are visible on the right side of the mixer:



Mixbus allows a channel to be attached to “unlimited” VCA masters. You can add VCA masters in the “Track->Add Track/Bus” dialog from the main menu.

Creating a VCA Master

To create a VCA master, select Track -> Add Track,Bus or VCA -> VCA Masters:



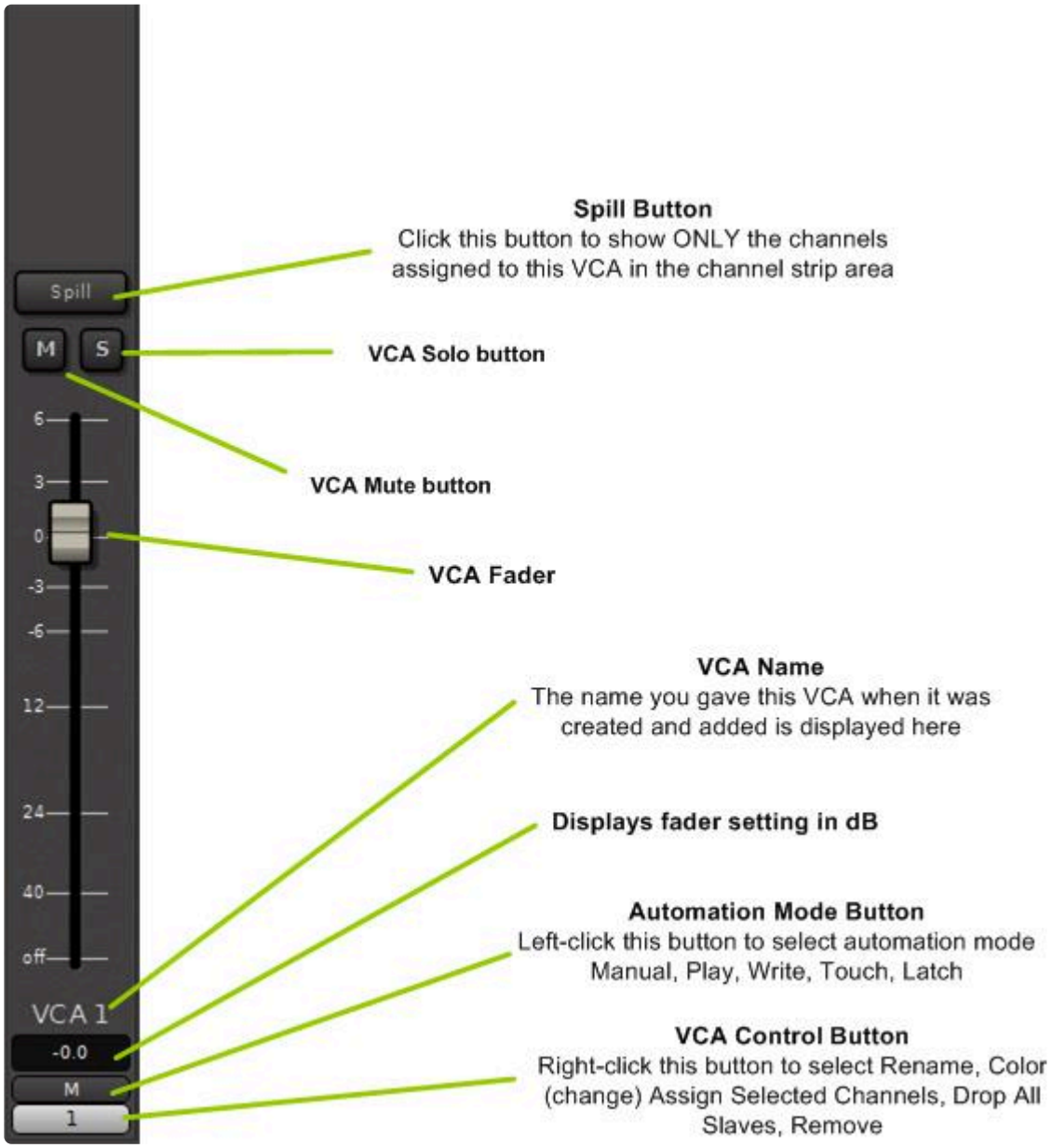
Assigning a track to a VCA Master:

To assign a track to a VCA master, click on the “VCA” button at the bottom of the Channel or Mixbus strip, and click the boxes of the the VCA you wish to assign to control the strip.

After VCAs are assigned, you will notice that the “VCA” button is replaced with indicators showing which VCAs are assigned to control this strip:

Alternately, in the editor window select the tracks, and then right-click in the “Group Bar” on the left side, and choose “Assign selected channels to VCA->”

VCA Strip Operation:



The “Spill” button, when activated, will hide all channels that are NOT attached to the VCA Master. If you have a group of channels assigned to a VCA master, this is a very quick way to display those channels in the main mixer area. Only one “Spill” (either VCA or Mixbus) can be active at one time.

The Mute button will mute ALL channels and buses that are controlled by this VCA.

The Solo button will provide a solo which is the pre-fader sum of all channels and buses that are controlled by this VCA.

The VCA name displayed is the name you entered when creating and adding the VCA.

Fader Automation mode button. This control *only* affects the VCA’s fader. Select between

- Manual
- Play
- Write
- Touch
- Latch

The VCA Control button always displays the VCA number. Right-clicking on the button provides these features:

- Rename
- Color
- Assign Selected Channels
- Drop All Slaves
- Remove

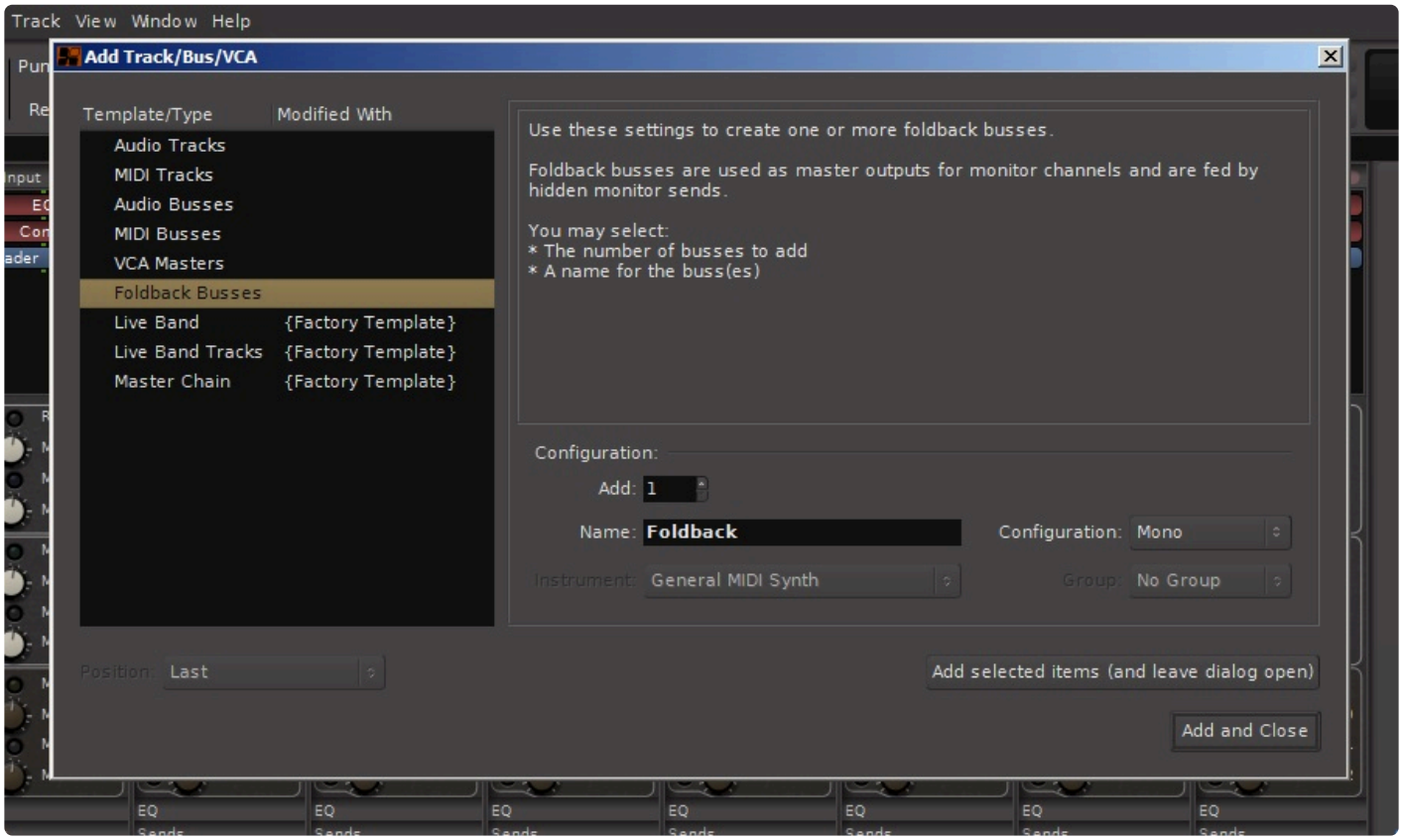
The VCA control button always displays the VCA number.

10.11. Foldback Buses

New in v6 are “foldback” buses whose primary purpose is to create summing buses specifically for cue/ headphone mixes.

Creating Foldback Buses

On the Track menu select “Add Track, Bus or VCA -> Foldback Busses



You may select:

- the number of foldback buses to add
- the names of these buses
- whether they are mono or stereo

The Foldback Strip

The image shows the Foldback strip when displayed. Unlike mix buses or channels, there is only one foldback strip displayed. You can select which of the foldback buses you are looking at by clicking the arrows at the top of the strip. The entire foldback strip is hidden by pressing the “hide” button. The foldback strip may be redisplayed by going to the View tab and selecting “Mixer: Show Foldback Strip”.

Foldback Display
Use the arrows to choose which Foldback bus to display

Hide Button
Clicking this button will hide the entire foldback strip

Foldback Control Button
Click this button to show the list of Foldback buses created. Right-click to access a menu for output routing, renaming, commenting or removing

Polarity button
Inverts the polarity of the Foldback bus. If it is a stereo bus, there will be a button for each channel as shown here

Show Sends button
When engaged this button will flash. The level indicator at the bottom of the channel strips assigned to this foldback will indicate the send level, rather than the channel fader level

Sends Area
Sends to this foldback bus will be displayed in this area, along with their individual send level and pan controls

Output Balance
If the Foldback bus is stereo, this balance control will appear. It can be deactivated by right clicking and checking the "Bypass" box

Plugin Area
Insert and order plugins here

Output Fader

Routing Button
Click to access the output routing controls

Comments Button
Click to open and edit the comment box

Some Foldback Strip operations are as follows:

- Foldback Control Button
  - Click this button to show and select from the list of Foldback buses created. Right-click to access a menu for output routing, renaming, commenting or removing

- Polarity Button(s)
  - Inverts the polarity of the Foldback bus. If it is a stereo bus, there will be a button for each channel as shown here.

- Sends Area
  - this area will display the sends to the displayed foldback bus. A slider sets the send level from each source. If the foldback bus is stereo there will also be a pan knob displayed. As elsewhere in Mixbus, to adjust the pan click and hold on the knob and move the mouse up and down.
  - The “Show Sends” button will flash when engaged. The level indicator at the bottom of the channel (or Mixbus) strips assigned to this foldback will indicate the send level, rather than the channel fader level.
  - If the foldback bus is stereo a balance control will display above the plugin area. Click and hold to move the yellow arrow left or right to adjust the balance. This control can be disabled by right-clicking and selecting the “Bypass” box. The balance control will remain visible but grayed out if disabled. The right-click also provides a few other controls such as balance law selection and numerical balance entry.

- the plugin area works the same as elsewhere on Mixbus. Plugins can be inserted and ordered relative to each other and the rotary fader.
- at the bottom there is the standard comment box, and the output routing button.

Routing to the Foldback Buses

To route a channel or Mixbus to a foldback send, right click in the plugin area of the strip, select “New Monitor Send” and choose the foldback bus to be routed to. The foldback bus must have been created already in order for you to be able to route to it.

- on the foldback bus the send level slide control described earlier will appear, and if the foldback bus is stereo, a pan knob as well.

Getting It Out

At the bottom of the Foldback strip is the standard Mixbus routing button. You can access the Routing Matrix feature to send the foldback bus output to your audio interface. If the foldback bus is stereo, you will have to route both outputs explicitly in the routing grid.

10.12. “DAW Style” Mixing Buses

Mixbus was created to specifically look and operate like an analogue mixing console. The built-in Master bus, Mixbuses and grouping and VCA features supplemented by the new Foldback bus feature should facilitate 99% of your workflow requirements. However some users may want to retain the workflow used by other DAW products i.e. creating your own summing buses (sometimes called “Aux” buses). Mixbus provides this operating flow should you choose to use it.

“Aux” or “Utility” buses are audio channels, just like tracks, but they do not record or play back audio. Instead, they accept audio at the input, process it, and send it to the mixing buses. (In ProTools, these are called Aux Tracks)

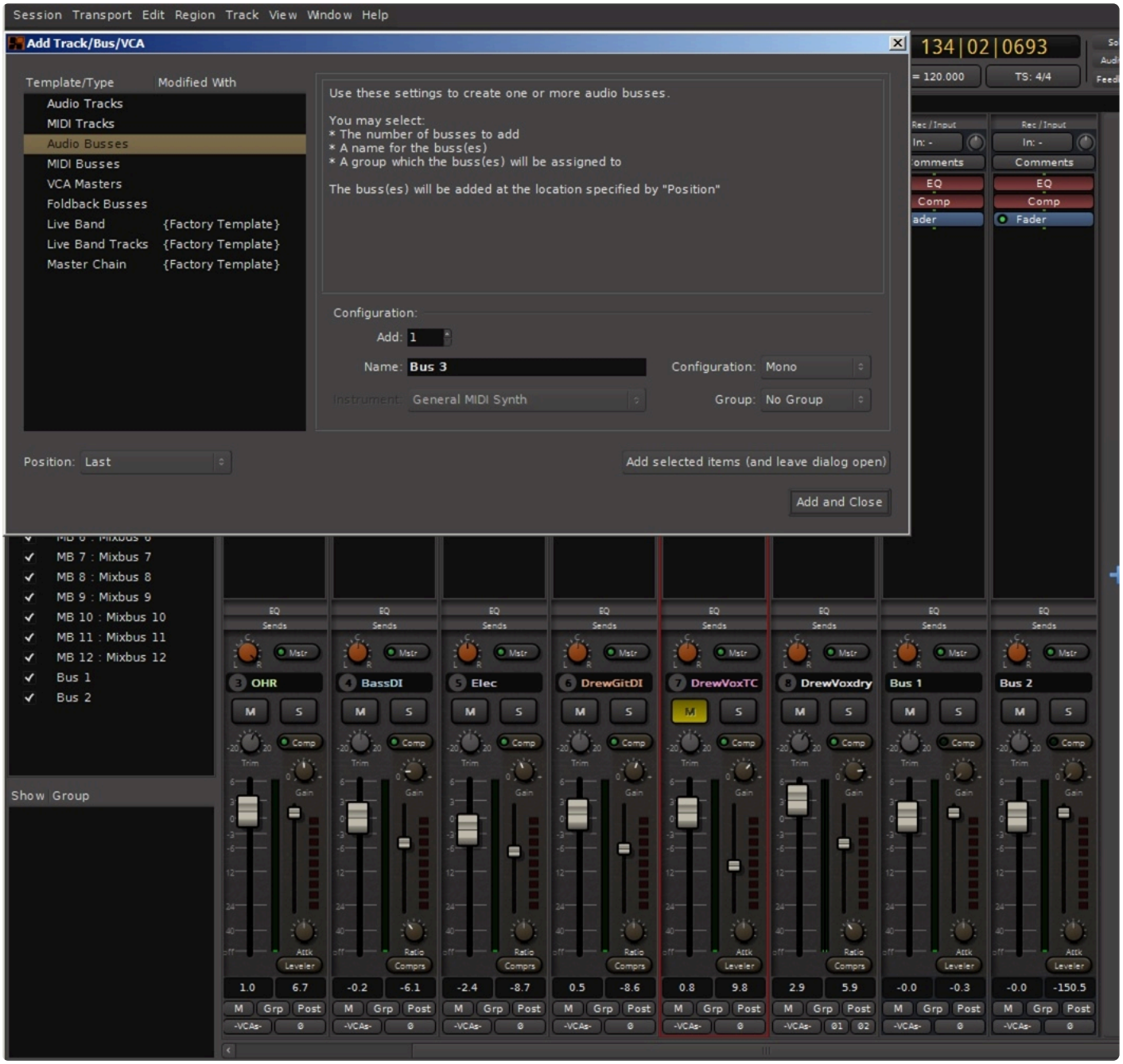
> An Aux bus, like a mix bus or master bus, has no recorded audio stored in it. A utility bus serves to sum and processes the audio.

> These buses are provided to support DAW-like workflows, and exist **outside** of the Harrison mix engine. These kinds of buses are good for utilitarian tasks, but they do not have the same controls as a “True Analog” Mixbus bus.

Creating a Mixing (“Aux”) Bus

To create a “DAW Style” mixing bus (also sometimes called an “Aux” bus), go to the “Track” pulldown and select “Add Track, Bus or VCA”. Then select “Audio Busses”. You can select the number of buses to add, whether that are mono or stereo, their names, and whether they are assigned to a group.

The picture shows the addition of a third mono audio bus:



Note that the “Aux” bus strip looks and operates exactly like a channel strip, except that its input is the sum of whatever audio is routed to it, and its audio content is not recorded. It’s output can be routed in the usual way to the master stereo bus, the 8 (12 in Mixbus 32C) mix buses, or anywhere else desired using the Mixbus routing grid. For more on routing a bus using the Mixbus routing grid see the section “Sends and Inserts”.

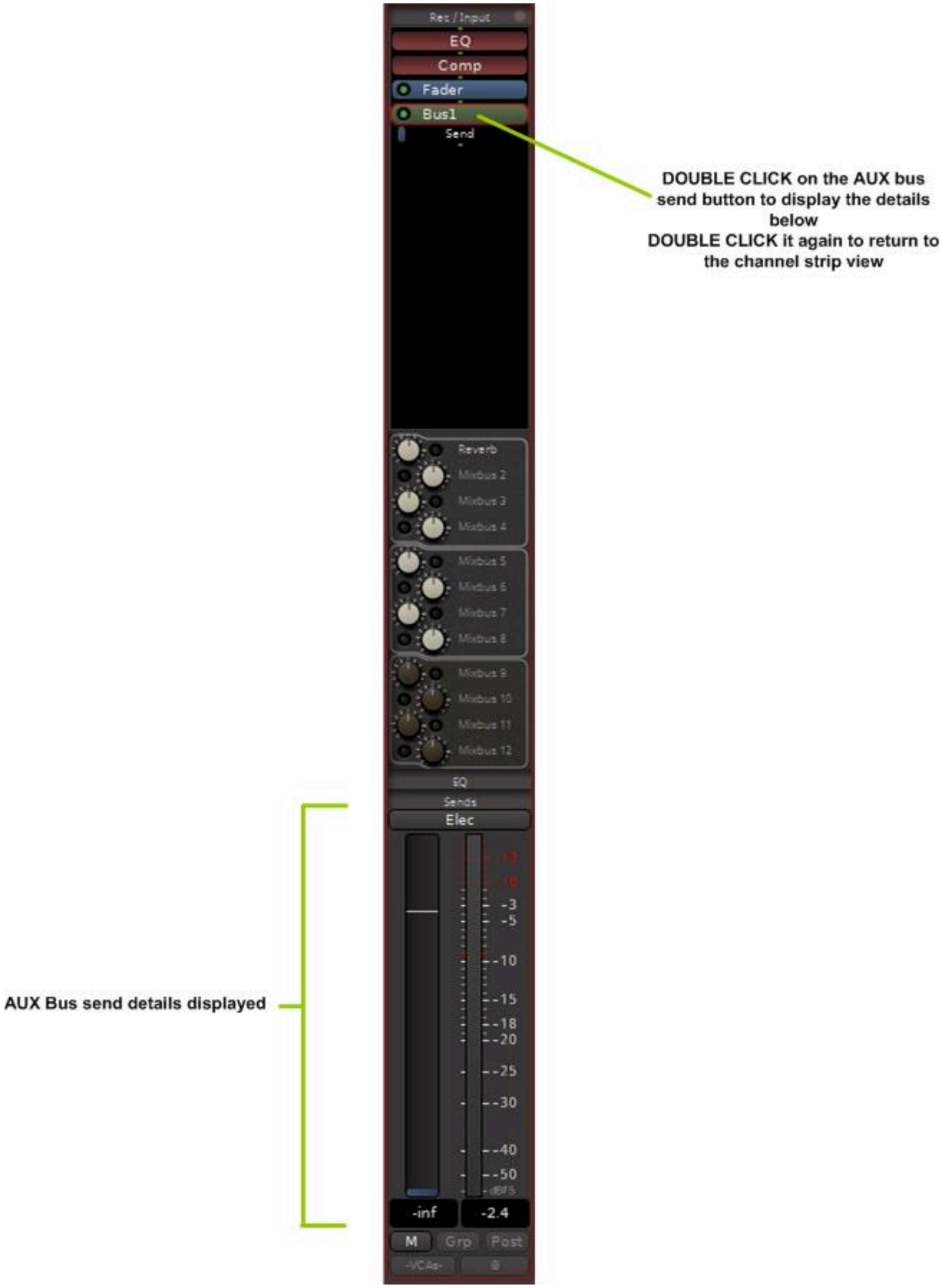
Routing to an Audio (“Aux”) Bus

To route to an audio bus of this type, right-click in the Redirect section of any channel or Mixbus. Select “New Aux Send” and the bus you wish to route to.

> If “New Aux Send” is disabled (grayed out), it probably means that you haven’t yet created a bus to send to. Create a new Audio Bus first, and then the option for “New Aux Send” will appear.

In the Redirect section of the sending channel or Mixbus an AUX bus send button and a level slider will appear as shown in the picture above. As with all other items in the Redirect area, dragging the bus send up or down will permit the send to come from anywhere in the signal path.

More details of the AUX bus send can be viewed by double clicking on the AUX bus button. Double click the button again to return to the regular channel strip view:



In this picture you see the AUX send level slider and level send meter (which measures POST the send

level slider). The regular EQ and bus send sections remain above (this picture is a Mixbus32C example with the regular mix bus sends displayed).

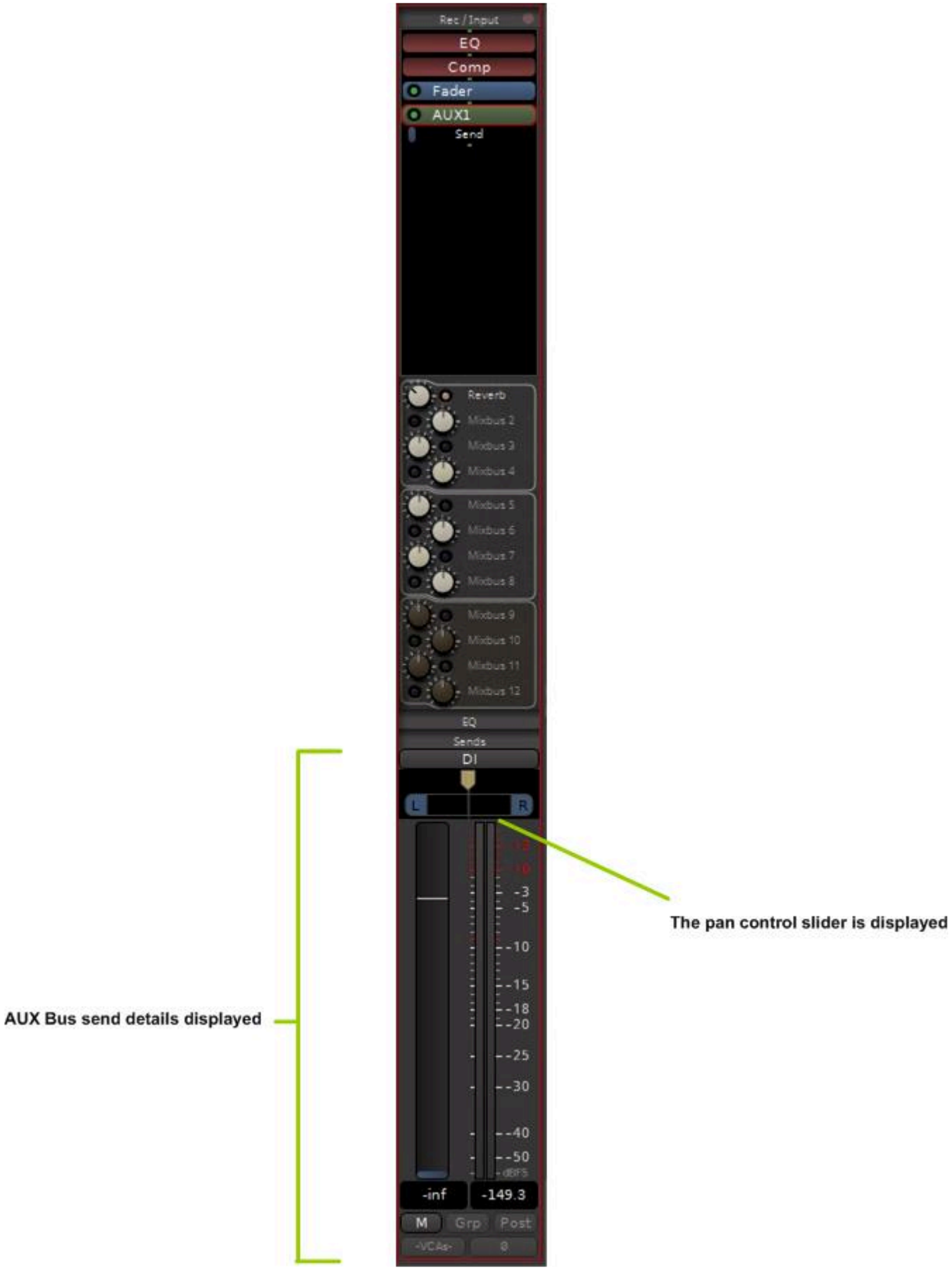
You can delete a bus by right-clicking on it, in the editor or mixer view, and choosing “Remove”.

Panning Options

Pan operation to an AUX bus depends on the type of AUX bus you created.

- If the AUX bus you created is MONO, the channel pan control has no effect on the send to the AUX bus. The AUX strip’s main pan control controls panning to the main stereo bus and mix buses in the usual way. The AUX bus send detail will appear as in the picture above.

- If the AUX bus you created is STEREO, there are additional options that can be selected:
    - by default, pan to a stereo AUX bus follows the channel main pan knob (which controls panning to the mixbuses and the master stereo bus). The pan slider will be shown when displaying the AUX bus details, as shown in the next picture:


Rec / Input
EQ
Comp
Fader
AUX1
Send
Reverb
Mixbus 2
Mixbus 3
Mixbus 4
Mixbus 5
Mixbus 6
Mixbus 7
Mixbus 8
Mixbus 9
Mixbus 10
Mixbus 11
Mixbus 12
EQ
Sends
DI
L
R
-3
-5
-10
-15
-18
-20
-25
-30
-40
-50
dBFS
-inf
-149.3
M
Grp
Post
-VCAs-
0
AUX Bus send details displayed
The pan control slider is displayed

The pan slider follows the channel strip main pan knob, and vice versa. Panning to all buses can be adjusted with either one.

- To pan the AUX bus send independent from the main channel pan knob:
  - right-click on the AUX bus send button, go to “Send Options”, and UNCHECK the “Link Panner Controls” box:

This feature gives you independent pan control for each stereo AUX bus you create, independent of the

main pan knob to the master stereo and mix buses.

Feedback Option

Note the “Allow Feedback Loop” check box in the figure above.

By default Mixbus prevents you from creating signal routing paths where the output somehow ends up at its own input, a bad thing that should always be avoided. This is especially possible with AUX buses when you aren’t paying attention to what you are doing. However if you need this capability, checking this box will allow you to do it at your own risk.

10.13. Mute and Solo

What are Mute and Solo?

The simplest explanation of Mute and Solo are:

Mute: When enabled, the mute button prevents the track from being heard.
Solo: When enabled, the solo button mutes all the tracks *except* this one.

….but there are many subtleties, especially for Solo, that are called upon to solve vexing problems that we encounter as audio engineers.

Mixbus provides Mute and Solo buttons, and a wide range of options, to meet these needs.

By default, when you solo a track, the other (non-soloed) tracks are drawn with a yellow frame. This indicates they are “implicitly muted”. (i.e. muted by someone else)

You can disable this “implicit mute” display in Preferences->Mixer->Solo

Solo Operations

Exclusive Solo:
ctrl + alt + click on a Solo button (OSX: command +control + click). Normally, if more than one track/bus is solo’d, they will all play, but this allows you to listen to only one solo’d track when more than one have been solo’d.

On the solo/mute controls left-button (button 1) mouse clicks do the following, depending on holding these keyboard modifiers:

- no mod: toggle solo on/off
- Primary mod (usually ‘Shift’): override track group
- Primary + Secondary modifiers (usually ‘Shift’ and ‘Ctrl’): exclusively solo this track
- Primary + Tertiary modifiers + click: apply to all

Middle-click (button 2):

- no mod. : momentary toggle (until release)
- Primary mod: assign generic MIDI. (if generic midi control surface is enabled)

Right-click (button 3) brings up a context menu:

- Solo Isolate
- Solo Safe
- Mute-points

> These shortcuts (group override, apply to all) are also available for other controls where applicable (e.g meter-point,..). See Mixbus Conventions & Tips

Solo Isolate

“Solo Isolate” is a mode which *prevents the channel from muting when other channels are soloed*. This mode is used, for example, on a reverb return where you’d like to always hear the reverb, even when a channel is soloed which would normally mute this channel.

You can Ctrl-click (Cmd on Mac) on a channel to Solo isolate it. Or Right-click on a Solo button to see a context menu with this option.

When a channel is Solo-Isolated, the word “ISO” will appear in the “switcher” area at the top of the strip.

Solo Safe

“Solo Safe”, when enabled, will “lock” the current state of the Solo button so the user cannot accidentally change it by clicking on the Solo button. If the channel is Solo’ed, it will not be possible to un-solo the channel by clicking the Solo button.

To activate this option, Right-click on a Solo button to see a context menu.

Solo’ing with a Monitor section

If the session has a monitor section (Session > Properties > Monitoring), there are additional controls for Exclusive Solo, Mute/Solo overrides and Pre/After Fader Listen available. Click here for details.

Solo’ing Mixbuses

Like a real console, the mixbuses are a separate mixing stage. Imagine the 8 (12 in 32C) mixbuses are like a second, standalone mixer. So when you solo a mixbus, the other mixbuses are “implicitly muted”. There’s no interaction between the inputs and the mixbuses, so soloing a mixbus does not implicitly mute or solo any tracks.

Utility buses aren’t really a mixing stage at all; they are just a signal path where you can combine other paths (or split them out from a multichannel plugin, for example). In this case, when you solo a utility bus then it also has to solo all the things feeding into it. Similarly, if you solo anything, it has to make sure that the “utility buses” that it feeds are soloed and this causes other stuff to get implicitly muted.

The best plan is to keep your signal paths simple, and you’ll get the best results. Just because you “can” add a bunch of buses and options, doesn’t mean that you “should”.

10.14. Redirects: Plug-ins, Sends & Inserts…

The black box near the top of each channel strip, mix bus, and master bus is the “redirect box”. Mixbus uses the term “redirect” for the items that can be inserted into a channel strip: **plug-ins**, **sends**, and **inserts**. In the image below, the track on the left has the “XT-BC Bass Character” plug-in, the track on the right has an insert and a send. The XT-BC Bass Character plug-in and the send are pre-fader (because they are above the fader entry) while the insert is post-fader. The signal flow of Mixbus is very flexible: you can order sends, inserts, plug-ins and the fader arbitrarily simply by click and dragging up or down (note: the XT-BC plug-in shown is for display purposes and may not be included in your purchase of Mixbus).

- Plug-ins
- Sends and Inserts

Note: By default, plug-ins and inserts will be created pre-fader, while sends will be post-fader. But you can drag them into any order you choose.

Channel strips are created preconfigured with an EQ and Compressor plug-in already installed (the top two reddish plug-ins in the above image). Signal routing can be changed by reordering here. The controls for these are found within the GUI of each channel strip.

Plug-in Control Sliders

Below each plug-in, you may add one or more “Plug-in Control Sliders.” Each Slider controls a single control parameter for the plug-in it is under. You may show or hide plug-in control sliders by right-clicking on a plug-in and selecting “Controls.” In the above image the “XT-BC” plug-in has control sliders shown for “Body – Freq” and “Clarity -In.”

The Redirect Menu

Right-clicking in a redirect box will launch the redirect menu. These choices are available in the menu:

- New Plug-in>
  - Favorites: a list of favorite plug-ins, pre-selected from the Plug-in Manager (more here).
  - Plug-in Manager: launch the Plug-in Manager (more here).
  - By Creator: a list of plug-ins, sorted by their creator.
  - By Category: a list of plug-ins, sorted by type (note: this is only useful for plug-in formats that support this).

- New Insert: create a new Insert (Send and Return).
- New External Send: calls up routing matrix
- New Aux Send: create a new Send.
  ————————————————-
- Controls: Allows the addition of one or more “Plug-in Control Sliders” to be displayed.
  ————————————————-
- Clear (all): Remove all redirects from this track/bus.
- Clear (pre-fader): Remove all redirects from this track/bus pre-fader.
- Clear (post-fader): Remove all redirects from this track/bus post-fader.
  ————————————————-
- Cut/Copy/Paste/: Cut/Copy/Paste/ the selected redirect.
- Delete: delete the selected redirect (you can also shift-left-click any redirect to delete it).
  ————————————————-
- Rename: Rename the redirect (yes … even inserts and sends)
  ————————————————-
- Select All: Selects all redirects
- Deselect All: Deselects all redirects
  ————————————————-
- Activate/Deactivate All: Activate/deactivate all redirects on this track/bus.
  ————————————————-
- Pin Connections: Click here for details.
  ————————————————-
- Edit: Pop open the editor for the selected redirect (you can also double-click on any redirect to edit it).
- Edit with generic controls

Redirect Box Signal Routing

Within the redirect box, and in between the plug-ins, are a series of green and/or red dots called “pips” that represents the signal routing between plug-ins.

Mono audio = one green pip: In the image above the channel strip on the left is a mono audio track. A single green pip representing one channel of audio is routed to the EQ, then to the compressor, and on to the fader.

Stereo audio = two green pips: In the image above the channel strip in the center is a stereo audio track. Two green pips representing two channels of audio is routed to the EQ, then to the compressor, and on to the fader.

MIDI data = red pip(s): In the image above the channel strip on the right is a MIDI track. One red pip representing MIDI data is entering the MIDI plug-in. The MIDI plug-in then generates audio based on that data. In this example the audio is stereo as can be seen by two green pips passing to the EQ, on to the compressor, and then to the fader. MIDI data is also passing thru the MIDI plug-in. The EQ, compressor, and fader are not processing this MIDI data, just passing it thru.

10.14.1. Processor Signal Flow

Signal Flow through the Processor List:

The “processor list” is the black box at the top of every Mixbus channel strip. You can imagine the “processor list” as a container for a series of guitar-pedal processors; each processor must be connected to the processor before and after it.

Often, the signal flow is very simple. For example, a mono track will most likely have a series of mono effects applied. In this case, you’ll see a picture like this:

In this example, the signal is processed by the channelstrip EQ, the Compressor, the Fader, and finally a Gate plugin. The signal flow proceeds from top to bottom, and the green “pips” between each plugin show the audio signal connections between each redirect.

But what happens if you want to use a plugin that has stereo I/O? Just like a guitar pedal, you might choose to use only the left-side signal for the mono track. But there are other possibilities as well. The plugin might have 1 input, and create a stereo output (such as an auto-panner, or reverb/delay effect). Or the plugin might have an extra “sidechain” input to trigger a gating or compression effect. Or the plugin might be an instrument that has 16 individual outputs for different drum sounds.

Starting with Mixbus version 3.3, the signal path between redirects is highly flexible, while also using sensible defaults.

> ! Sessions created with Mixbus versions before 3.3 will have tracks that preceded these rules. For backwards-compatibility, we will load those sessions with the old rules. In some cases you might want to use “Pin Connections” (described later on this page) to improve the routing of those old sessions.

https://www.youtube.com/embed/XCTGs2p3P74?rel=0

Why is this so complicated?

Mixbus tries to do the “right thing” by default. Mono channels will get mono plugins, and stereo channels will get stereo plugins. Virtual instruments will appear with stereo outputs, by default. But if you want to change these settings, there are quite a lot of variables that must be considered.

There are several challenges inherent to the signal flow through a channel using plugins:

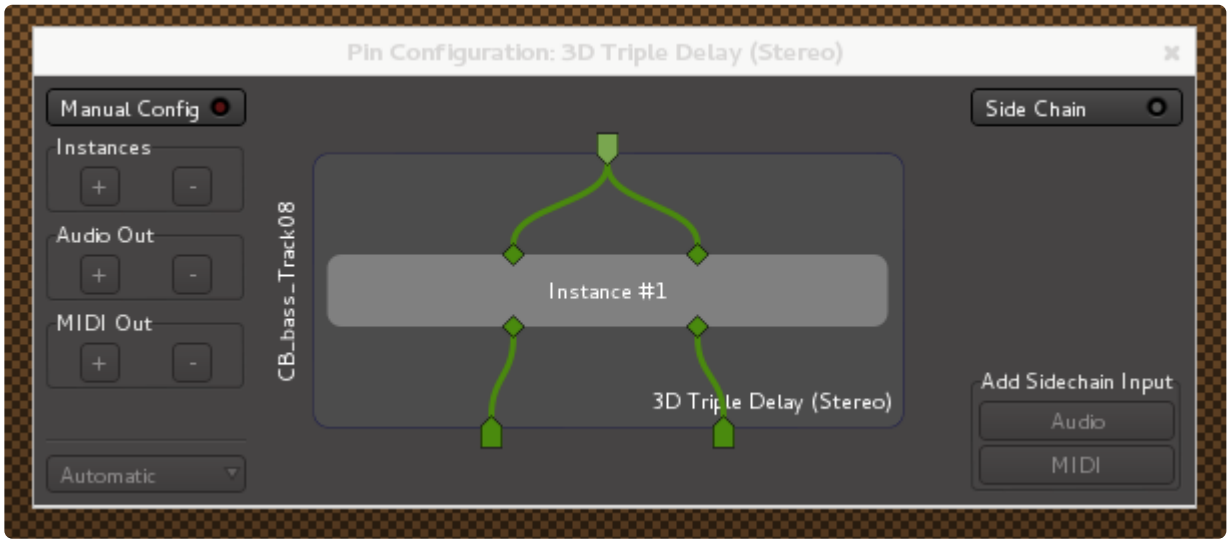
- Plugins may be drag/dropped into any order
- Some plugins can be fed a “sidechain“signal which should allow the possibility of being sourced from any other track.
- Often you need to use a plugin in ways that it wasn’t intended (only using one side, for example)

Additionally, each of the Mixbus-supported different plugin formats provide different rules:
1) LV2 and LADSPA plugins typically have fixed I/O such as 1->1, (1 input, 1 output) or 2->2
2) VST plugins typically have a fixed I/O configuration, but can be told that some signals aren’t connected.
3) AudioUnit plugins (Mac) allow the HOST to define how many connections they will have. So Mixbus can request a mono (1->1) or stereo plugin as-needed. But the user also has the option to change the configuration, if they want to do something unusual.

Finally Mixbus can “replicate” plugins. If you try to insert a mono plugin on a stereo strip, Mixbus will automatically create 2 plugins, and send the controls from the first plugin to the second. This allows you to process both sides of the stereo signal using the same settings of an EQ, for example. You can add or remove replicated plugin instances using “Pin Management” (see below)

Pin Connections

Right-click on a plugin to reveal the “Pin Configuration” dialog:



By clicking on the pin connections, you may drag a connection between the channelstrip signal flow, and the embedded plugin processor. Right-click on a pin to remove all connections to that pin.

One can also bypass plugin instances with a ‘thru’ connection by dragging an input pin directly to an output pin. This connection is latency compensated to match the delay through the plugin.

The “Pin Configuration” dialog provides these additional features:

- Manual Config : click to enable the configuration buttons, below:
  - Instances : increase/decrease the number of plugin instances inside the processor
  - Audio Out : increase/decrease the number of audio outputs that the plugin can access
  - MIDI Out : increase/decrease the number of MIDI outputs that the plugin can access
- Output presets (virtual instruments only) : click to choose how many outputs the instrument will provide
- Side Chain : click to enable a sidechain input for the plugin.
  - When Side Chain is enabled, the sidechain source may be managed with the buttons below it.

10.14.2. Plug-ins

What is a Plug-In?

Plug-ins are small pieces of signal-processing software that Mixbus can load inside itself and assigned to channels. Channels may have an unlimited number of plugins, inserts and sends on a channelstrip.

Mixbus comes with a selection of Plug-Ins installed. Some are free, while others require an additional purchase to use fully. You can learn more about Mixbus's included plugins here.

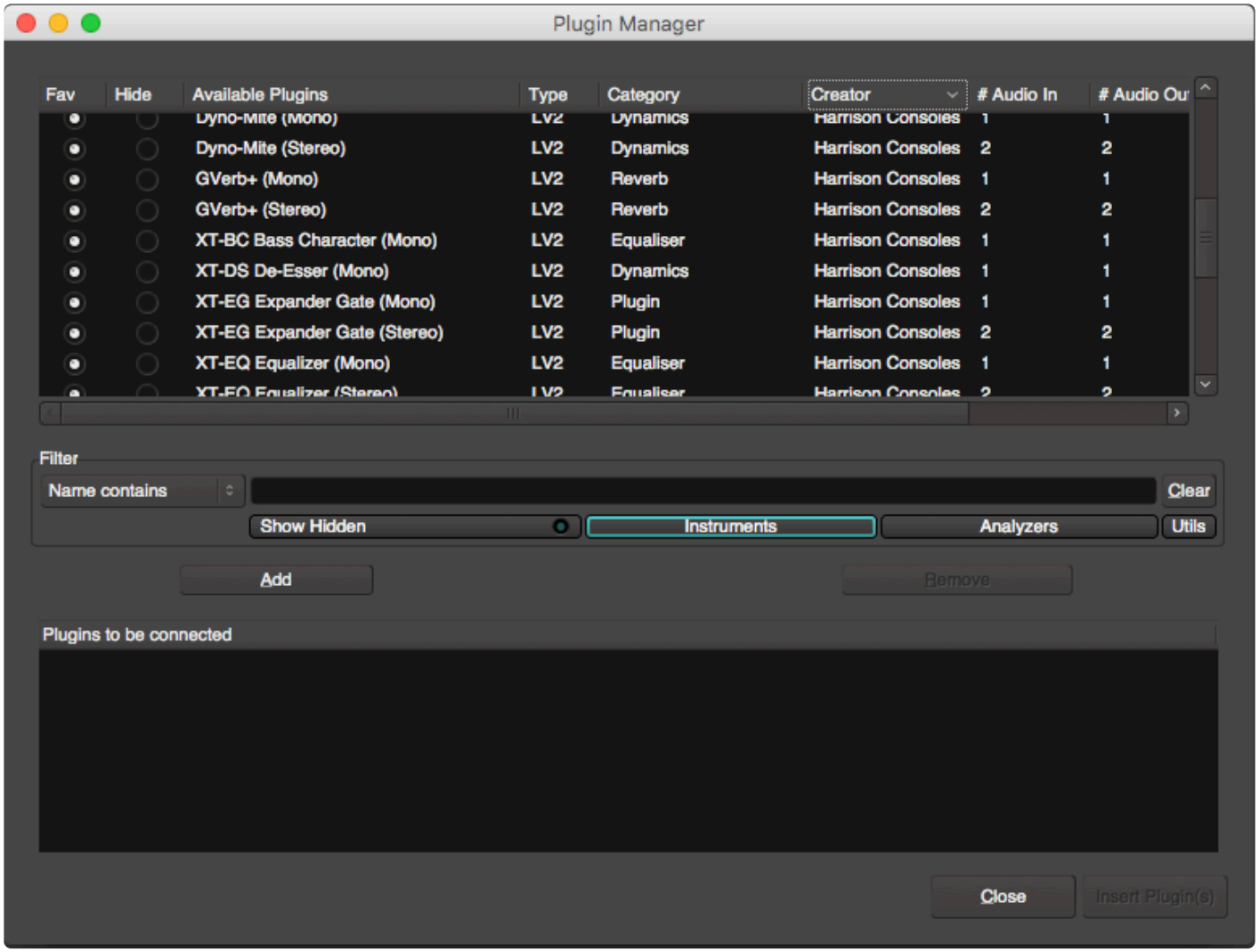
Additional plugins may be installed on your system, and these will be provided by third-party developers.

> When you first install Mixbus, it does not recognize your third-party plug-ins. To scan your plug-ins (and tell Mixbus to scan for plug-ins on startup in the future), please visit Edit->Preferences->Plug-ins and click the "Scan" button.

How do I add a Plug-In to a channel?

You can add plug-ins in several ways:

- double-click in the black "redirect" area to launch the Plug-in Manager
- right-click in the black area, and choose New Plug-in, then:
  - Favorites
  - Plug-in Manager
  - By Creator
  - By Category



> A note about categories: some plug-in formats (such as LV2) provide a very well-defined method to categorize plug-ins. Most older formats (including VST2 and AudioUnits) do not provide good support for categories.

Adding, Removing, Reordering, and Disabling Plug-ins

Each plug-in is displayed as a rectangle in the redirect box. Plug-ins are red if they are pre-fader (above the fader), and green if they are post-fader (underneath the fader)

- double-click the plug-in to show the plug-in control panel
- click the plug-in to select it (it will turn red). Now click “Delete” to delete it.
- click & drag the plug-in to change its order in the signal flow
- click the small green indicator on the plugin to enable/disable the plug-in
    - (you can also middle-click on a plug-in to enable/disable it)
- right-click on a plug-in to see the plug-in context menu
- shift+right-click a plug-in to delete it

Plug-in Favorite Window

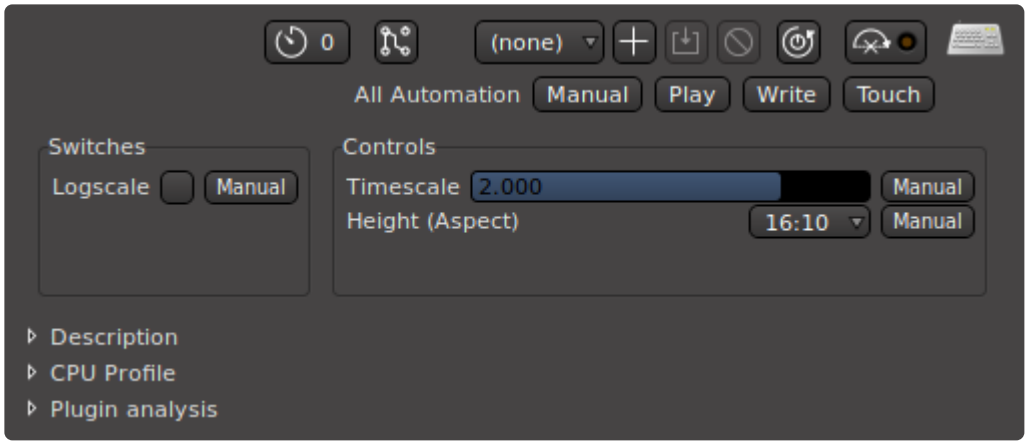
In the upper left side of the mix window is the Plug-in Favorite Window. You can drag and drop plug-ins to and from a channel strip.
A new favorite can be save and even a preset for the plug -in can be saved and recalled from this window.
Multiple channel strips can be selected and a plug-in can be inserted.
Left clicking on a plug-in or preset name is another way of adding plug-ins to the channel strip.

10.14.2.1. Generic Plugin Controls

In some cases, it is desirable to see a generic display of a plugin's controls, rather than the plugin-provided UI. Some cases might be:

- A generic UI is provided when the plugin doesn't provide its own control panel
- If a plugin's control panel is incompatible with your setup, this provides a way to control the plugin without showing its panel
- The generic UI can access hidden parameters or settings that are not accessible from the plugin's control panel
- The generic UI allows you to type values directly into a slider, which might not be available in the plugin's control panel.
- The generic UI provides direct access to Mixbus's automation (Read/Write/Touch)

A typical generic plugin ui is shown below:



The topmost portion of the control panel provides access to the latency compensation, pin-connections, presets, and global bypass.

Switches are provided on the left side of the dialog.
Sliders and selector menus are provided on the right side of the dialog.

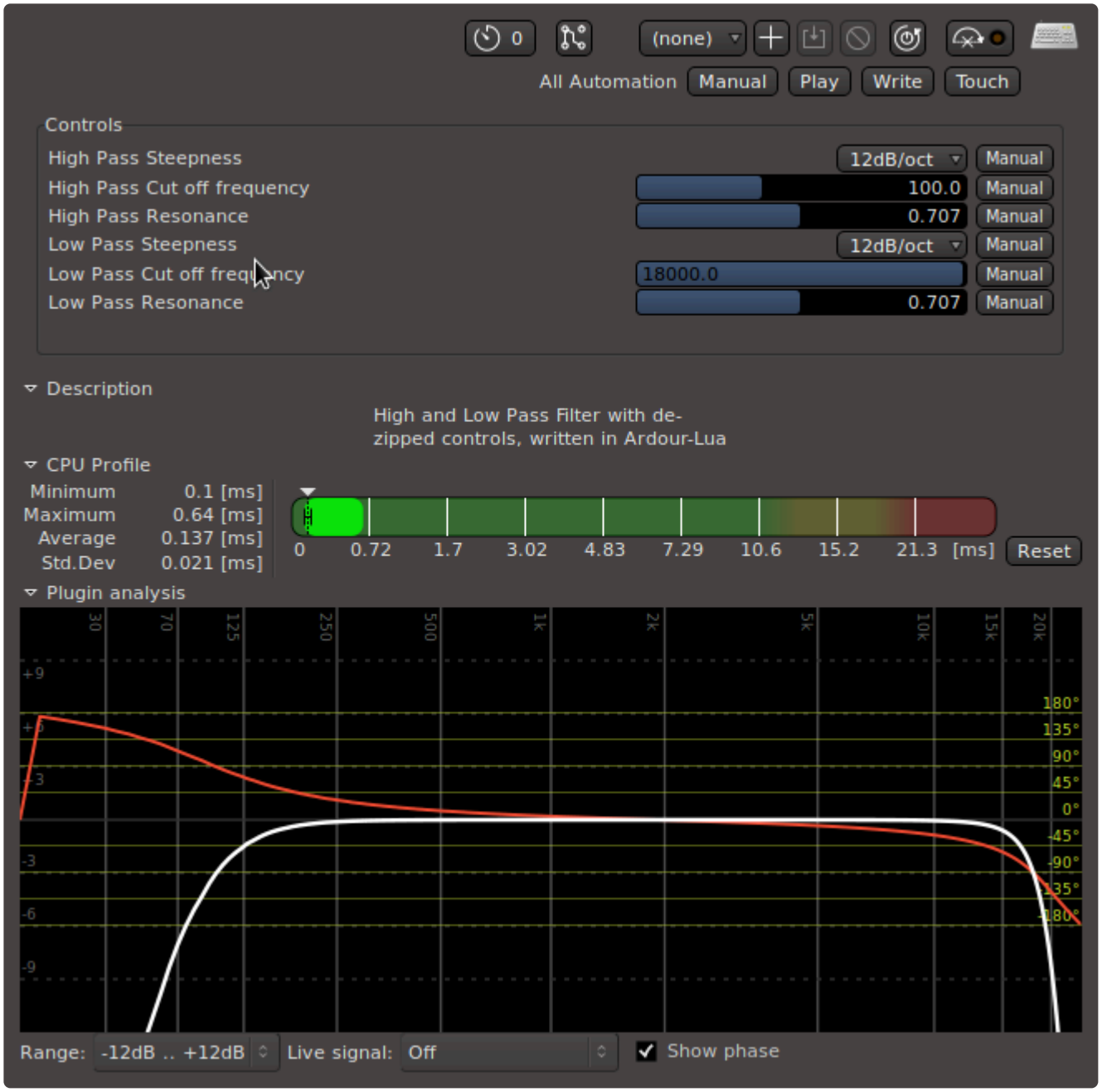
To the right of each parameter is an automation button to access Read/Write/Touch automation modes for that plugin parameter. The "All Automation" buttons near the top of the dialog can be used to put all plugin parameters in Read/Write/Touch

Double-click inside a slider to type a value into the control.

Plugin Analysis

At the bottom of the dialog, you will find 3 additional features.

- Description: expose this section to see a longer description of the plugin (this is only available for LV2-format plugins)
- CPU Profile: expose this section to see a visual indication of the plugin's average and peak (worst-case) CPU usage.
- Analysis: expose this section to see a "live" display of the plugins magnitude and phase response. This can help to expose the behavior of a plugin that you don't understand, or suspect is malfunctioning.



10.14.3. Sends and Inserts

What is a “Send”?

A “send” is a connection point that is created inline with the channel’s signal flow; it allows you to send the signal out of the channelstrip, and into something else. You might “send” to another channel, or a plugin sidechain, or a physical output on your soundcard.

Sends are in addition to the mixbus-send knobs in the channelstrip. They provide more flexibility (for example, sends can be moved “anywhere” in the channel’s signal flow, among the other plugins and sends/inserts.) However, they operate outside of the Harrison “True Analog” summing & panning engine. These sends are similar to features found in other DAWs and we sometimes call these “DAW Sends” instead of “console sends” or “console buses”.

Aux Sends

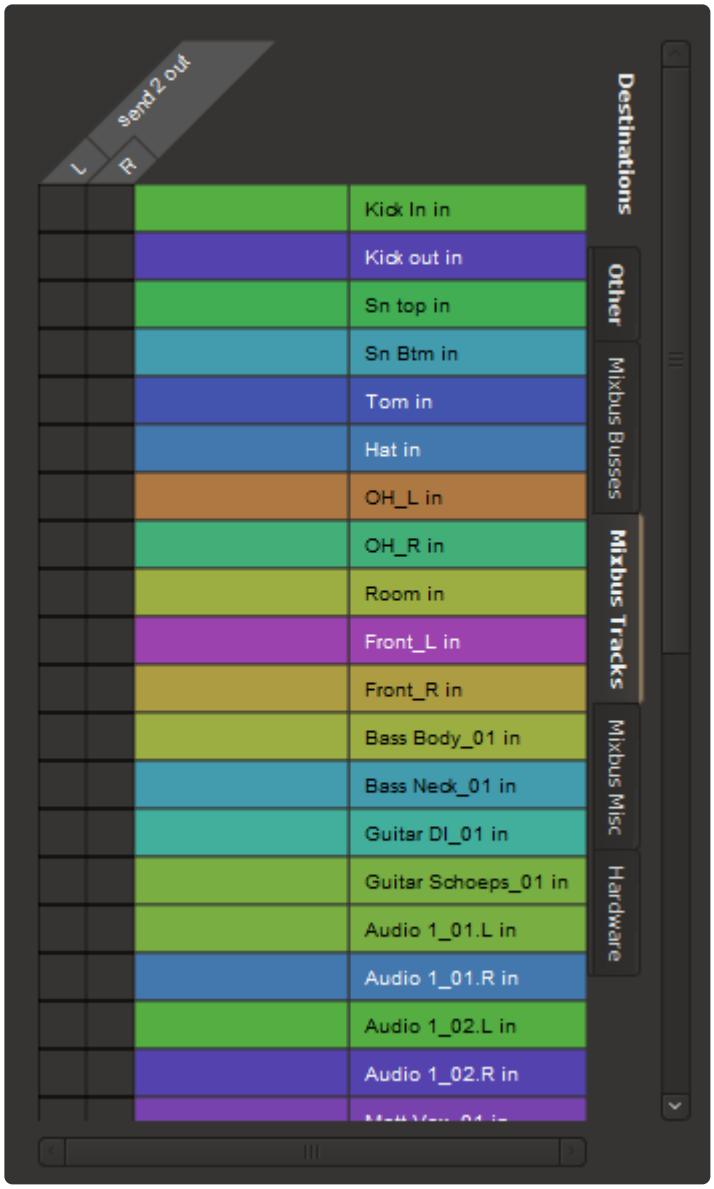
Create an “Aux send” by right-clicking in the plugin area, and choosing “new Aux Send”, then choosing the bus you’d like to send to.

Aux sends can only be connected to the input of a “bus input strip” and cannot be sent to an external port, such as a soundcard output.

> ! As implemented in v4.3, “Aux Sends” from the channelstrip do not have pan/balance controls.

External Sends

When a new send is created, a window for connecting the Send’s output ports will immediately appear. You may connect a Send’s outputs to any connections available in the system.

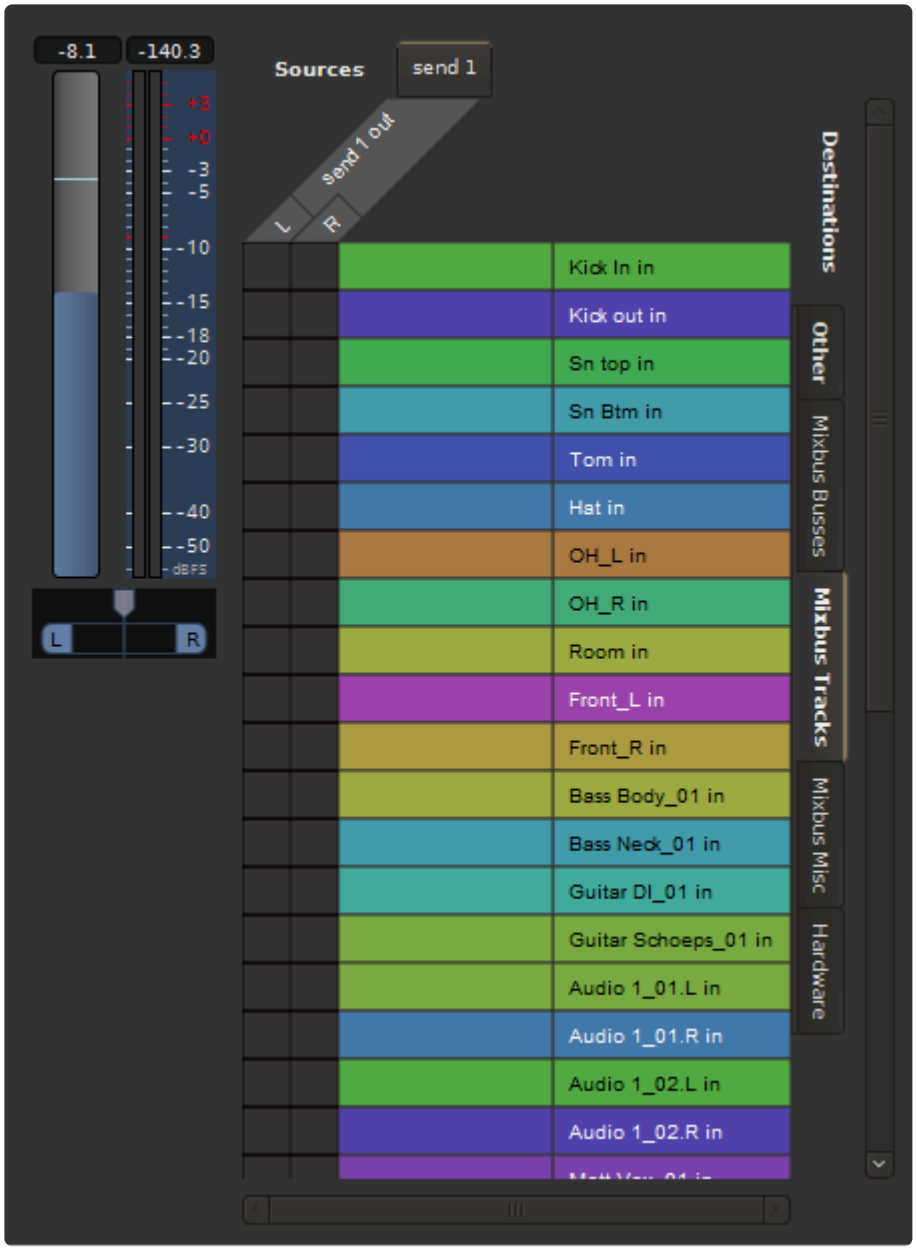


By default, Sends have no outputs and must be configured as needed. Click “add” to add ports, and connect them to the destination that should receive the output of the send.

- Add/Remove: adds/removes a port to the send.
- Disconnect All: disconnects all ports (does not remove ports).
- Available connections: This is a list of all the available connection ports, with one tab for each application, and one for the system sound I/O (“system”).

Note: The number of “ports” created will define the channels of the send. Sends can have 1 (mono), 2 (stereo) or more (multichannel) ports assigned.

After a send has been configured, a double-click on the send will launch the Send Editor:



In addition to the output management, The Send Editor also has a volume slider and a panner (if the number of outputs is > 1).

> Sometimes when you mute a track, you still want a Send to remain active (for a headphone feed, for example). It is possible to disable the mute for pre- and post- fade sends separately. To enable or disable the pre- and post-fader send mute, right-click on the Mute button in the editor window.

What is an “Insert”?

An “insert” is similar to a Send, but it both sends the signal out, and then provides a method to return the signal immediately back into the channel. The most common usage of an “insert” is to send the channelstrip

signal out of Mixbus and into a hardware device for processing. Then the insert provides a return path from the device, so you can hear its effect. You can enable and disable the “insert” to compare the signal with and without external processing.

Inserts

An Insert incorporates a “send” output and a “return” input. The Insert send/return allows the audio signal to be routed out of a channel and back into it, after performing some external processing. Inserts are often used to “patch in” a piece of outboard gear that is not available as a plug-in (just like you might do with a patch bay on an analog console)

- Measure Latency: Once your insert is connected to an external piece of gear, then click this button to measure the latency. If the Insert is used on a track or mix bus, this delay will be accommodated automatically.

Outputs

- Add/Remove: adds/removes a port to the insert “send”.
- Disconnect All: disconnects all sending ports (does not remove ports).
- Available connections: This is a list of all the available connection ports, with one tab for each application, and one for the system sound I/O (“system”).

Inputs

- Add/Remove: adds/removes a port to the insert “return”.
- Disconnect All: disconnects all return ports (does not remove ports).
- Available connections: This is a list of all the available connection ports, with one tab for each application, and one for the system sound I/O (“system”).* Available connections: This is a list of all the available connection ports, with one tab for each application, and one for the system sound I/O (“system”).

10.15. EQ Techniques

The following videos cover some common uses of equalizers. Remember your ears are the final judge!

https://www.youtube.com/embed/wn1SvzR2Nv4?rel=0

https://www.youtube.com/embed/HM8XSfaGJuw?rel=0

https://www.youtube.com/embed/Xb-prKFUdi4?rel=0

10.16. Compression Techniques

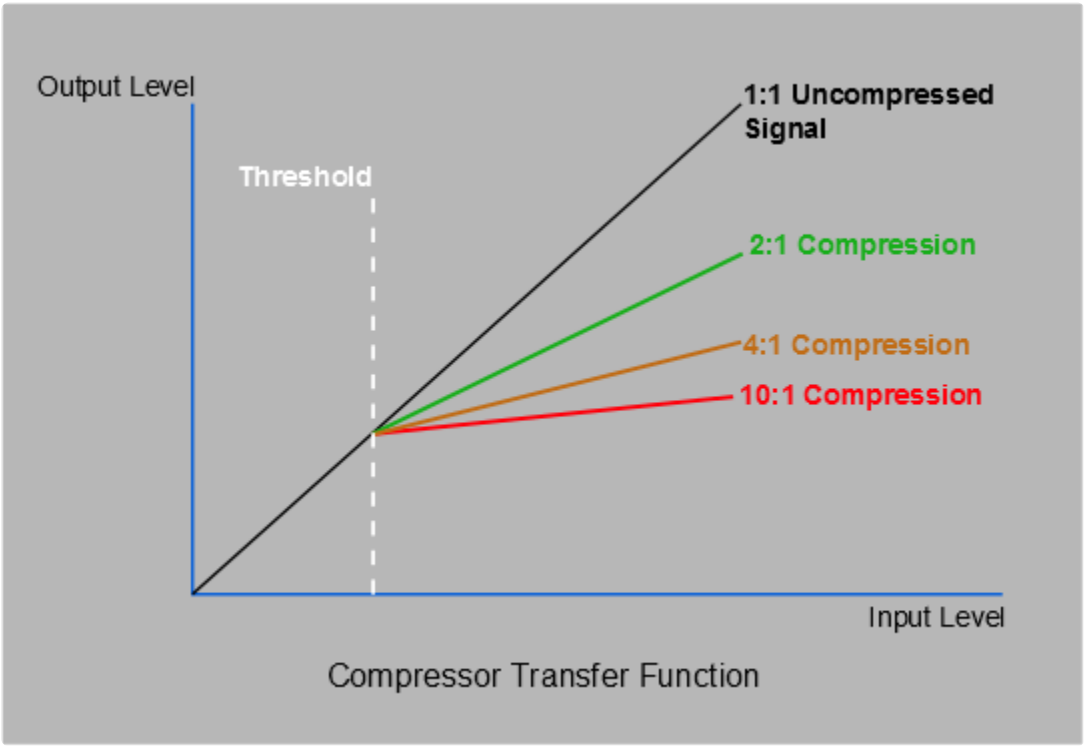
https://www.youtube.com/embed/Sa3-ibczoK4?rel=0

What is a compressor?

A compressor is a device that automatically reduces the volume of a signal when it exceeds a threshold volume level. Knob settings determine how fast it will decrease the signal, as well as how strong the effect is. Compression can serve the technical purpose of reducing the dynamic range of a signal; or it can be used as an artistic effect by changing the sound's volume envelope.

Basic controls of a compressor

- Threshold: The "threshold" defines a level at which the compressor will start to activate. Signals quieter than the threshold will not be affected by the compressor. For more dramatic compression, drag the threshold "down" so that it overlaps more with the input meter.
- Ratio: Ratio defines the strength of the compression once the signal exceeds the threshold. With a ratio of 10:1, exceeding the threshold by 10 dB will only result in an increase of 1 dB at the compressor's output.
- Attack/Release: the attack and release speed affect how fast the volume change can happen. You can adjust these settings to avoid objectionable artifacts of the compression effect (if used for dynamic-range control), or to accentuate the effect (if used for artistic purposes).
- Makeup gain: A compressor reduces the loudest signals of a track. Makeup gain is used to bring the complete track back to the level before it was compressed. The makeup gain applies equally to the loud and quiet signals, so the quiet parts are effectively made louder after compression and makeup are applied.



Mixbus provides a compressor on each channel strip. Each compressor has 3 modes (Leveler, Compressor, Limiter) which can be selected. In the section below, we explain the most common compressor techniques and how to achieve them using the Mixbus channel strip compressors.

Compression Techniques

It is commonly touted that ‘a great compressor makes everything sound better’…. but what does that mean, and is it really true? Ignoring the crosstalk, noise, frequency-response irregularities, and distortion inherent to these boxes, why are compressors desirable? In this section we will try to objectively evaluate why some mixers like to use compression on their tracks. It’s worth mentioning that some of the best mixes were made with little or no audio compression. But if we know what a compressor does, then we can make sensible decisions whether to apply compression or not.

1. Leveling: The “leveler” mode has a very low ratio, a very fast release, and an adjustable attack (via the “Speed” knob). Due to mic technique or other elements out of our control, sometimes a signal has a very wide dynamic range. The “leveler” with a *fast attack* can be used to transparently reduce the level of the loudest parts, so that the track has a more consistent level. Reducing the dynamic range of our tracks can make them easier to mix, although it might rob some of the performance’s natural dynamics. The leveler with a *slow attack* allows the initial sound (the first syllable of a word or phrase, for example), to be passed through at full volume, while reducing the “body” of the sound slightly. This retains and enhances the character of the recording, and it can help improve intelligibility of words and instruments in a busy mix.
2. Compression: The “compressor” mode is more aggressive than the Leveler. It has a fixed attack/ release time, and an adjustable “ratio” (compression strength). Compression dramatically reduces the dynamic range of the signal, and it is less transparent than the leveler. Compression is a natural effect

of amps driven to their limits, or vocal shouting. Therefore our brains perceive compressed signals as “louder”, regardless of their actual volume. Because the loud signals are brought much lower, it makes the quiet signals sound as loud as the loud parts. This means that mouth noises, valve noises, and other ancillary sounds will sound more pronounced. This can impart a “nearby, in-your-face” effect.

3. Limiting: The “limiter” mode has a nearly-instant attack, and a variable release. A limiter is a very fast-acting compressor, and these were initially designed to capture signal peaks before they were passed on to sensitive broadcast transmitters, disc-cutting lathes, or speaker systems. Limiters have a very fast attack and a very high ratio, making them destructive to most signals. A limiter can be used to blunt the very short peaks of drum hits, and therefore allow the overall level to be increased. It must be used very sparingly though, or the signal will sound distorted.

Prior to Mixbus v6 the Mixbus strips provided a 4th mode, “Sidech”. The built-in sidechain feature has been **removed** from Mixbus v6. See Sidechain compression for more details.

In addition to the channelstrip compressor, the master bus provides a final limiter. This is a look ahead brickwall limiter set at -1dB, and it protects your final mix from exceeding full-scale digital. The “lookahead” feature is a delay that reduces the distortion it applies your mix, at the cost of some latency. See Mastering Techniques for more details on how (and when) to use this limiter.

The Harrison Mixbus Compressor Modes:

1. Leveler: The attack is adjustable and the ratio/release are preset. Turning the knob clockwise increases the attack time (makes the attack slower).

2. Compressor: The ratio is selectable and the attack/release are preset. Turning the knob clockwise makes the ratio higher.

3. Limiter: The release is selectable and the attack/ratio are preset. Turning the knob clockwise increases the release time (makes the release slower).

These 3 modes are designed to provide the most common compressor settings for music.

> ✻ You can copy compressor settings from one track to another by dragging the compressor processor (in the processor list at the top of the channelstrip) onto another channel's compressor processor.

Additional Harrison developed compressor plug-ins and Compression Techniques:

https://www.youtube.com/embed/GL99xDmKoDU?rel=0

https://www.youtube.com/embed/HqTcM-m0QsU?rel=0

https://www.youtube.com/embed/iESeTOSvX80?rel=0

10.16.1. Sidechain Compression

> ! NOTE: The built-in sidechain feature has been **removed** from Mixbus v6. This feature was provided in Mixbus version 3,4, & 5 because we had limited support for sidechain-able plugins. Since most sidechain tasks benefit from precise control over the threshold, ratio, attack and release, we decided to remove the limited sidechain bus that was incorporated in the channelstrip compressor. Instead, you should use the full-featured “plugin” sidechain buses described below.

What is Sidechain Compression?

Compression is an automated way of reducing a signal’s level, dependent on the signal’s own loudness. The louder the signal is, the more the compressor will reduce it. **Side-chain** compression reduces the level of the signal using a *different* signal to trigger the volume reduction.

In pop/rock music, this technique is sometimes used to “duck” the bass guitar on each kick-drum hit, making the bass and kick drum sound like a single instrument. In dance/EDM styles, it is often used to generate a “pumping” effect using the kick drum or a “ghost” kick drum track that is not actually heard. In this case the release is set very slow; after each beat the signal slowly rises, reaching full intensity just before the next beat.

Other Uses of Sidechain

Alternative uses of a sidechain include feeding a gate or vocoder. A sidechain is any input that affects the plugin processing, but isn’t directly heard by the user.

Sidechains Using Third-party Plugins and the Plugin Pin-management Dialog

Mixbus can feed “any” audio track or input source to a side-chainable plugin. To access the plugin’s input connections, you should use the pin manager as described in this video:

https://www.youtube.com/embed/XCTGs2p3P74?rel=0

https://www.youtube.com/embed/XCTGs2p3P74?rel=0

Mixbus provides 2 side-chainable plugins by default: a-Compressor and a-Expander

Using Sidechains in Mixbus6

As noted above, the built-in sidechain feature has been removed from Mixbus6. The following is a general description of how to connect the sidechain control on plugins.

In this example we will show a popular third party VST plugin product called FabFilter Pro-G. The instructions presented are intended to be generic and should be similar for other kinds of plugins with a sidechain control from other vendors.

First, go to your plugin list and select the plugin you want to use:

Once inserted the pin connections will look like this:

Next click on the Sidechain button. NOTE: getting at this may be different depending on the plugin you are using:

Next select the source that you want to use for the side chain signal:
(Remember this process may be different on the plugin you are using).

Next click on the blue box on the right side of the pin connection box and drag a connection to one of the green side chain inputs. If this is a stereo plugin repeat this a second time to connect the side chain signal to both signal paths of the plugin.

NOTE: This will differ depending on what plugin you are using, but the procedure should be similar.

NOTE: Most plugin formats will will identify sidechain inputs with a blue connection point. VST plugins like this one do not explicitly show these as sidechain inputs, but merely as additional inputs as in this example. However you will not be able to route signal to these inputs until you select with the Sidechain button.

The sidechain control inputs of this plugin are now connected. The signal path inputs and outputs were already connected by Mixbus when you installed the plugin into the redirect area.

10.17. Mastering Techniques

What is Mastering?

“Mastering” is the process of *preparing a sound track for distribution*. It might involve, but is not limited to:

- Critical listening, to verify that there are no technical errors in the final product.
- Fixing equalization problems that were caused by poor acoustics in the original production/mixing room.
- Applying final tone control to match the overall tone of other tracks in the collection, or comparable tracks in the same genre.
- Applying final loudness adjustments to match the level of other tracks in the collection, or comparable tracks in the same genre.
- Sequencing of tracks into a collection (i.e. sequencing songs on a CD).
- Applying frequency-limiting and dynamic-range adjustments, to match the media that will be distributed.
- Creation of metadata (such as track markers, song name(s), performance rights, etc) for the distribution media.
- Exporting/packaging the distributable in a format that is suitable for the next stage of duplication/ distribution.

Although many of these tasks are outside the scope of Mixbus, Mixbus provides several tools that can help with the mastering process.

Master Bus Processing and Metering

The master bus provides a suite of processing and metering that helps achieve the overall tone and loudness that the artist wants.

Master bus strip tone controls

The master bus provides a 3-band EQ (“tone”) control for the final output. The tone control has a fairly limited range (+/-6dB) for fine control. The center frequencies are optimized to solve problems that are commonly used in the final stages of mixing:

- Low: a low-frequency shelf at 90Hz : the Low band is provided to balance the low bass content of the track, without having to re-mix the bass instruments.
- Low Mid: a wide bell shape at 300Hz : many instruments have strong output in the low-midrange, and they can combine to make too much energy in this range. This range is also a site for tape-saturation harmonics. The “Low Mid” band is provided to increase or decrease the main band of music.
- High: a high-frequency shelf at 4,000Hz : A high-frequency shelf at 4 kHz allows you to brighten the entire track without unduly affecting individual instruments.

Master bus strip compressor

The master bus strip compressor operates like the channelstrip compressors. (see : Compression Techniques)

Because the master bus strip compressor is applied before the Master Limiter, this provides a chance to level-match parts of a song (using the leveler mode), or to clip outstanding transients before they reach the master limiter (using the limiter mode).

Typically a very small amount of compression is needed on the master bus, therefore the master bus limiter threshold has a very different “scaling” than the channelstrip compressors. You have to pull the threshold down much lower before it has any effect.

The order of master bus compressor, fader, and EQ may be manipulated to achieve a wide range of desired effects before the signal is finally sent to the master bus limiter.

Master Limiter

The master bus limiter. It has no controls other than an on/off indicator.

The Master Limiter is a lookahead limiter with a very fast attack and release time.

The “lookahead” feature imparts a short 5ms delay on the input signal. This delay allows the limiter to “look ahead” and begin reducing the gain before a loud peak occurs. This dramatically reduces the distortion that is caused by hard limiting.

The limiter’s threshold is fixed at -2dB. This value is intentionally less than digitally clipping so that your final processing (sample-rate conversions, or lossy encoding like mp3) are provided extra headroom for their interpolation. This can result in better sounding mixes. If you want your exported file to peak at a higher value, you can implement that in the “normalization” setting for each of the file formats you are exporting.

Each segmented LED represents 1 dB of gain reduction.

Plug-ins

There are many plug-ins that are developed specifically for the master-bus output, for the purpose of increasing the master output level. These may be placed in the channelstrip signal flow, and used instead of, or in conjunction with, the Mixbus master bus processors.

Stereo Correlation (“Phase”) Meter

The stereo correlation meter provides an indication of the “mono compatibility”, or stereo width, of your mix. The indicator uses a bright bar which simultaneously indicates the range of in-phase (highly correlated) and out-of-phase (decorrelated) sounds in your mix.

- If the indicator is fully to the right, and **yellow**, this indicates a completely “mono” mix. (the signals in the left and right channels are the same)
    - many signals (such as a spoken-word recording with one microphone) should be mono.
- If the indicator is fully to the left, and fully **red**, this indicates a mix that is completely “out of phase”, in other words the left and right channels are equal, but one is inverted with respect to the other. This almost certainly indicates a technical fault which should be investigated.
- If the indicator is largely to the right, and **green**, then it indicates a mix with some stereo width, but largely mono compatible. This is optimal for pop music recordings that might be played in a variety of playback systems.
- If the indicator is in the center, showing a range of red and green, then the mix is very wide stereo. This is probably not optimal for pop/rock recordings, but can be acceptable for natural or orchestral recordings. Mono compatibility of this mix is questionable.

Mono compatibility is largely defined by low frequencies. If your mix indicates poor mono compatibility (red bar on the left), then you can probably solve this by reducing the number of bass instruments in the mix, high-pass filtering any non-bass instruments, and high-pass filter any stereo effects, such as reverb, in the mix.

K-14 Meter

The K-14 meter is an RMS (averaging) meter, originally specified by mastering engineer Bob Katz as a means to compare and help control the dynamic range of pop music mixes. (see : http://www.digido.com/how-to-make-better-recordings-part-2.html)

The K-Meter displays the level of the signal directly before the output of the master bus. The K-Meter comes after the fader, EQ, plugins, and final limiter in the master bus.

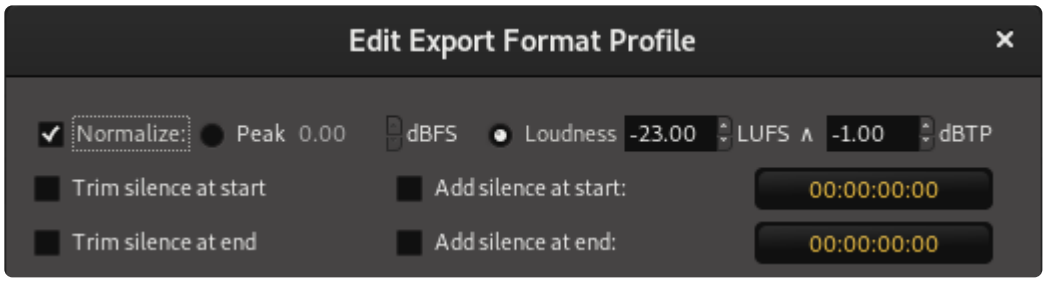
If the K-14 meter indicator is straight up, at 0, this indicates that the sound’s average level is at -14dBFS. This was selected as a good target for most pop/blues/jazz/rock recordings. At this level, a song will sound quieter than most contemporary pop music, but when the listener turns the volume up to match, the increased dynamic range of this mix will allow the song to sound much better than a typical pop song played at the same volume.

Increasing the volume beyond the “0” indicator will result in a louder mix, at a cost of decreased fidelity and impact when the user turns the volume up.

> In the past, it was highly encouraged for mastering engineers to create music at the highest possible average level, because this made the songs noticeably louder when they were played adjacent to other performer’s songs on a CD or mp3 player. However, in the case of modern radio broadcast, online channels such as YouTube, and internal processing in iTunes and iPods, it is now likely that your music will be level-matched (based on average loudness) with other songs. In this case, it is advantageous to mix at a lower volume, and leave more headroom for short musical peaks which will make your mixes sound louder and better than the more heavily-processed tracks that were popular in the early digital era.

10.17.1. Normalization

Mixbus' export page allows for easy customization of loudness targets that help you meet the loudness standards of the medium you will eventually end up in, whether it be a Redbook CD, Streaming, or other client specifications.



The Normalization Process

The normalization process will apply a single "gain" value to the whole resulting export file, so that the loudest part of the file reaches the desired target peak or loudness value.

Traditionally, the most common usage of this feature is to set the peak to 0dBFS. That means the loudest sample of the resulting file will reach exactly 0dBFS.

However, a single "peak" digital sample has very little to do with the perceived loudness of the file. For example you can have a single loud peak in a 10-minute file, and the normalization function will decrease the level of the file so the peak fits in without clipping. This is pedantically useful in many cases, but not really important step for a deliverable mix.

Normalizing to LUFS is a *better* solution, because it is a much better indication of "perceived loudness". You can have a very peaky file that still sounds quiet; whereas LUFS takes the "average" level into account. If we normalize a file to -9 LUFS, we know that we have a very loud mix indeed!

But: there is a limit with LUFS normalization: if Mixbus increases the loudness too much, it will push your "peaks" into clipping. (side note: if the LUFS normalization would result in a signal above your peak target, then Mixbus normalizes to the peak, not LUFS … this is why your resulting file is not reaching the LUFS you want)

The important point is: normalization only provides clean gain (either increasing or decreasing the volume for the entire file). It does not change the peak-to-average ratio of your song.

Said another way: normalization has no effect on the sound or quality of your mix, or its perceived loudness. That's up to you. Normalization can only adjust the final measurements to prevent exceeding your maximums.

(another side note: Mixbus' K-14 meter is an instantaneous RMS meter, not LUFS; but if you mix to the

RMS level it should get you in the ballpark of desired LUFS)

Best Practices

This is highly subjective and can change with the industry, but as it stands now, there are a few practices most mixing engineers follow and we make that easy to adhere to in Mixbus!

You should first *develop your mix* so that the master bus's "K-14" meter is reaching near the '0' meter during the loudest part of your mix. This will help ensure that your average-to-peak ratio is high enough to meet your medium/broadcaster's expectations.

Reaching the '0' mark on the "K-14" meter means that your mix is something near -14LUFS; and so when you run it through the normalization during export, Mixbus can then reduce it to meet your deliverable needs.

11. Groups

https://www.youtube.com/embed/cuInwhnrtlk?rel=0

https://www.youtube.com/embed/cuInwhnrtlk?rel=0

Tracks (and busses) may be assigned to a “group” which imparts some special behaviors. For example, you might want to put the drums in a “group”; in this case, adjusting the fader on one track will apply to the others.

What is a Group?

A group is a mechanism of connecting several tracks together when they have a strong inter-channel relationship. By default, tracks in a group will be ‘selected’ together whenever one track is selected; they will be edited together; they will share the same color; moving one fader will move all of the group faders; and assigning a track to a mixbus will assign all the group’s tracks to a Mixbus.

The most common usage for “groups” is when the tracks are separate, but they actually represent a single “recording”, such as a stereo pair, or a drum kit that was recorded live, or a 5.1 multichannel track. The “group” helps to preserve the togetherness of the tracks.

A track can only be assigned to a *single* group, whereas a track can be assigned to multiple mixbuses or VCA Faders.

A group of tracks should not be confused with a mix bus. Confusion sometimes comes from the fact that much of the time a mix bus has more than 1 track bussed to it. Likewise, a group has more than 1 track associated with it. However, a group of tracks will not sum their audio, but will group certain software

functionality which allows for **better session organization and more efficient mixing and editing.** That's not to say you can't have all tracks in a group bussed to its own mix bus. It's quite common in fact.

For instance, a session may have 6 drums tracks and after finding the right balance between those tracks you might group them so that moving one fader moves all in that group, causing all the drum tracks to change level. Or soloing one track in the group solos all in that group. Clicking on the group header will disable the group allowing individual changes to the tracks, after which the group can be re-enabled.

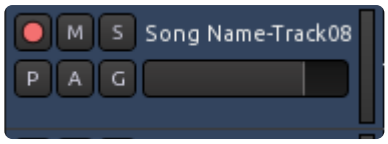
Click here to view a video on groups and bussing, and here for a video on subgroups.

How do you create a group?

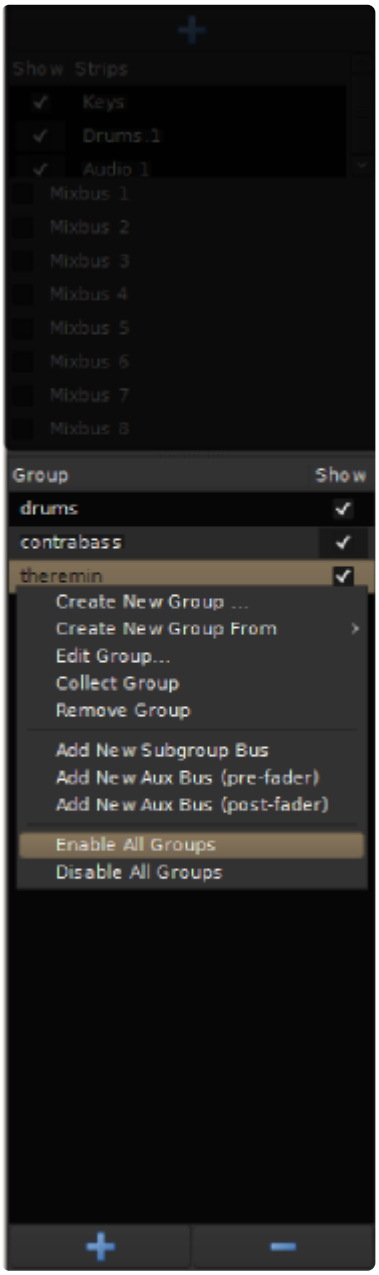
By clicking and dragging in the Group Bar, you can quickly assign multiple consecutive channels to a new Group. See: Group Bar

A group may be created or altered by clicking on the "Grp" button below the track fader. Once the group is named the "Grp" will be replace with the groups new name or a shortened version of it.

Alternatively, a group may be created or altered by clicking on the "G" button located in the track header.



Creating a group from the mixer window “Group” area.

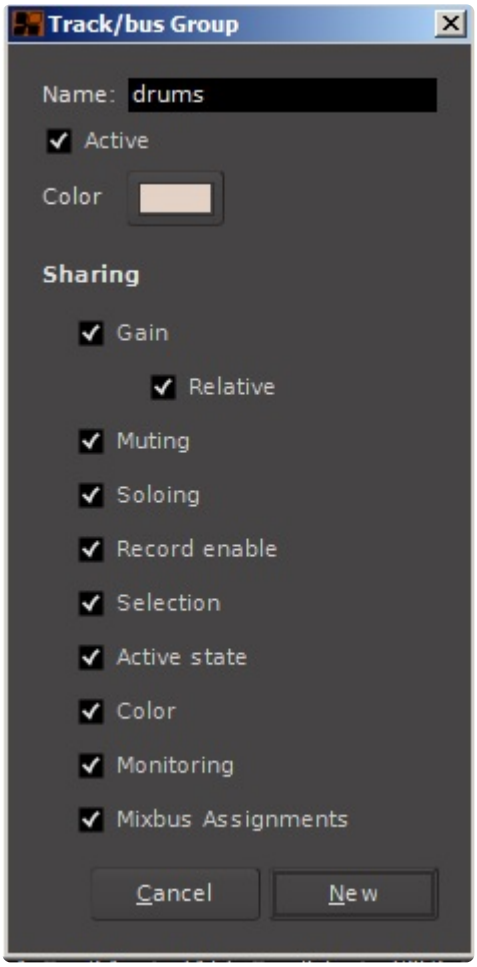


The Group section of the mixer window, showing the +/- and Enable/Disable for each group

Here all groups that have been created for the session are listed. A check box labeled “Show” is provided. Un-checking this box will hide a group from the edit and mix window. **Hiding a group is a convenient way of un-cluttering your sessions**. Groups can also be added or deleted clicking the + or – buttons at the bottom of the window. Clicking on the name of the group allows you to rename it.

Options within a Group

Grouping tracks allows the sharing of certain functionality and the functions available within a group are customizable. Right-clicking on a group tab calls up these options:



- **Active**: makes the group active/inactive
- **Gain**: syncs all faders to the *same* level
  - **Relative**: preserves fader differential between tracks
- **Muting**
- **Soloing**
- **Record Enable**
- **Selection**: links selections made in the editor canvas
- **Active state**
- **Color**
- **Monitoring**
- **Mixbus Assignments**: Channel assignment to a mix bus will follow the group assignment. If checked, assigning one channel of the group to a particular mix bus will also assign all channels in that group to the same mix bus.

Left-clicking on a group will enable or disable the group.

Grouping behaviors when the “Gain” is enabled

- Dragging a fader will affect all the others in the group
  - By default, this a “relative” move. (the faders will retain whatever dB offset they have prior to the move)

    - You can disable the “relative” checkbox, and all the faders will be forced to the exact same value, whenever you move one fader in the group
- Muting or soloing will affect all the others in the group
- Assigning or de-assigning a mixbus will affect the others in the group

Group Override

Holding “Shift” will override the group, resulting in these behaviors:

- Dragging a fader will affect only the current channel, not the others in the group
- Muting or soloing will affect only the current channel, not the others in the group
- Assigning or de-assigning a mixbus will affect only the current channel, not the others in the group

11.1. Group Bar

What is the Group Bar?

The Group Bar is an empty area to the left of the track headers (in the editor window) or above the mixer channels (in the mixer window).

The group bar provides convenience functions to create groups from your channels, or assign them to a grouping master bus, or assign them to a VCA master fader.

Operations from the Group Bar:

Right-click in the black (empty) group-bar area to display the Group-Bar context menu:

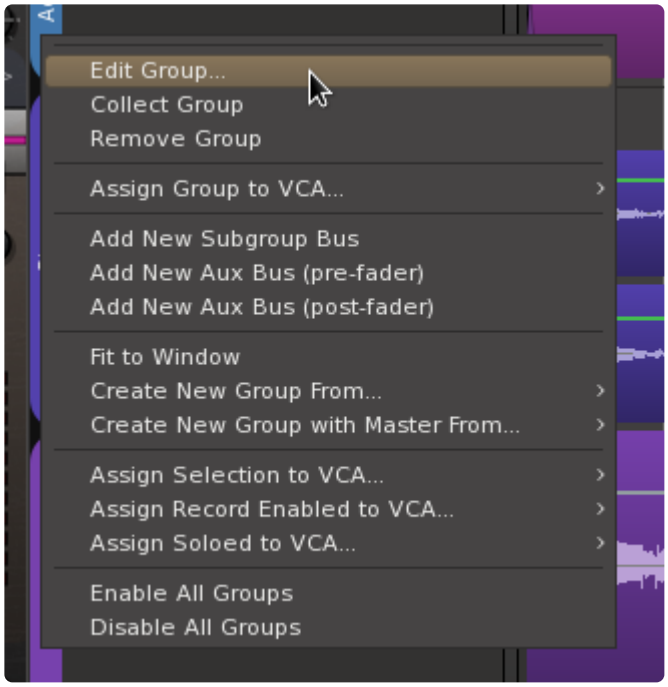
- Create New Group From : Select either “Selection, Record-Enabled, or Soloe’d” to create a new Group with those channels.
- Create New Group With Master From : Select either “Selection, Record-Enabled, or Soloe’d” to create a new Group with those channels. A new “bus” will be created, and your channels will be assigned as the input for that bus.
- Assign Selection to VCA: The selected channel(s) will be assigned to the VCA selected from a submenu
- Assign Record-Enabled to VCA: Any record-enabled channel(s) will be assigned to the VCA selected from a submenu
- Assign Soloed to VCA: Any solo’ed channel(s) will be assigned to the VCA selected from a submenu
- Enable all groups
- Disable all groups

Creating an Edit Group with the Group Bar:

In the edit window, a group selection can be be made by clicking in the black bar to the left of the track header. While holding the left click, slide the mouse vertically along-side the tracks you wish to group. Right click on the name of the group to rename it. To add tracks to that group just left click and slide the mouse alone-side the track you would like to add. This will create a group tab that extends the width of the tracks assigned to that group. Many times it is convenient for a group to contain tracks adjacent to one another but it is not absolutely necessary, a group of tracks does not have to be contiguous.

Learn more about the grouping functions here: Groups.

Right-clicking on a group tabs calls up a context menu for groups:

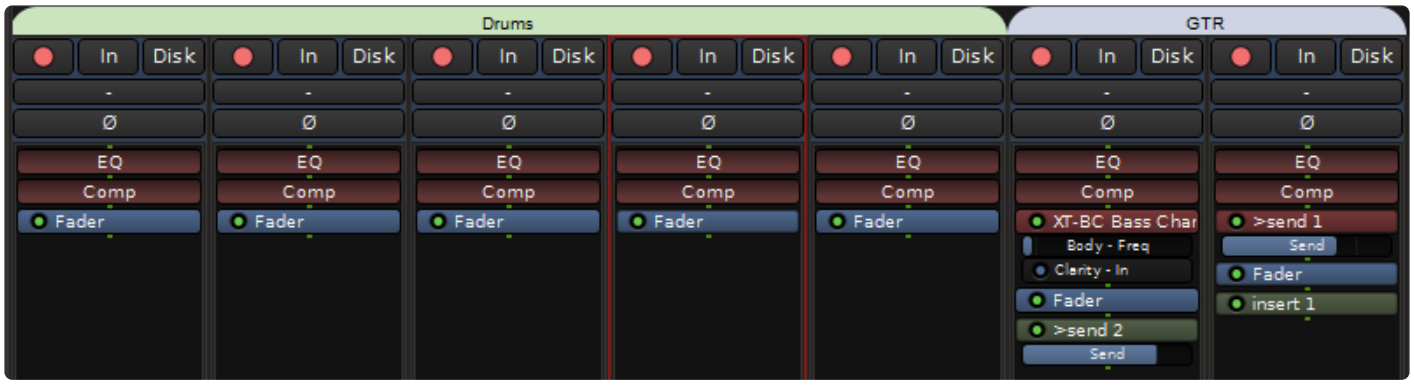


- Edit Group : launches the Group editor
- Collect Group : Moves the grouped tracks together into a contiguous series
- Remove Group : Deletes the Group (but not the tracks in it)
- Assign Group to VCA: The channel(s) in the selected group will be assigned to the VCA selected from a submenu
- Add new Subgroup Bus : Creates a New Bus, and assigns the output(s) of the grouped tracks to it
- Add New Aux Bus (pre-fader) : Creates a New Bus, and creates a pre-fader Send to it, from each of the grouped tracks
- Add New Aux Bus (post-fader) : Creates a New Bus, and creates a post-fader Send to it, from each of the grouped tracks
- Fit to Window : the channel(s) in the selected group will be sized to fill the editor window (vertically)
- Create New Group From : Select either “Selection, Record-Enabled, or Soloe’d” to create a new Group with those channels.
- Create New Group With Master From : Select either “Selection, Record-Enabled, or Soloe’d” to create a new Group with those channels. A new “bus” will be created, and your channels will be assigned as the input for that bus.
- Assign Selection to VCA: The selected channel(s) will be assigned to the VCA selected from a submenu
- Assign Record-Enabled to VCA: Any record-enabled channel(s) will be assigned to the VCA selected from a submenu
- Assign Soloed to VCA: Any solo’ed channel(s) will be assigned to the VCA selected from a submenu
- Enable all groups
- Disable all groups

The mix-window group bar.

In the mix window, a group can be be created by dragging in the black bar above the channel strips. While

holding the left click, slide the mouse horizontally over the channel strips you wish to group. This will create a group tab that extends the length of the tracks being assigned to that group. Many times it is convenient for a group to contain tracks adjacent to one another but it is not absolutely necessary. You may have any tracks in a group.



In all other ways, the Group Bar in the Mixer window operates identically to the Group Bar in the Editor window.

12. Automation

https://www.youtube.com/embed/UJbjPbpoWlk?rel=0

Region Gain and Automation

Mixbus has 2 separate types of automation: Region Gain Envelopes and Processor Automation.

Region Gain envelopes are often used for repair work, such as reducing unwanted breaths, accentuating syllables, or reducing noise. The “region gain envelope” is part of the audio region, and is applied before any other processing is applied in the mixer strip. Region Gain only adjusts the volume (level) of the region across time.

Learn more about Region Gain automation in this video: Mixbus Gain Staging

Processor Automation is often used for creative changes such as fading elements in and out during a song. Automation can be applied to many different mixing parameters including the input trim, fader, panning, muting, or plug-in parameters.

12.1. Fader, Pan and Plug-in Automation

Automation

The short video below demonstrates how to open an automation lane:

https://www.youtube.com/embed/B0DKjABQ6MA?rel=0

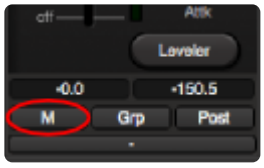
Automation is used to automatically change the setting of a knob, based on the time of the timeline. For example you might want to increase the volume of a delay effect during the “chorus” part of a song, or automate the master fader to make a smooth fade-out at the end of a track.

Many mixer parameters may be automated: fader, mute, pan, mix bus sends, eq, compressor, and plug-in settings.

Real-time (“Touch”) Automation:

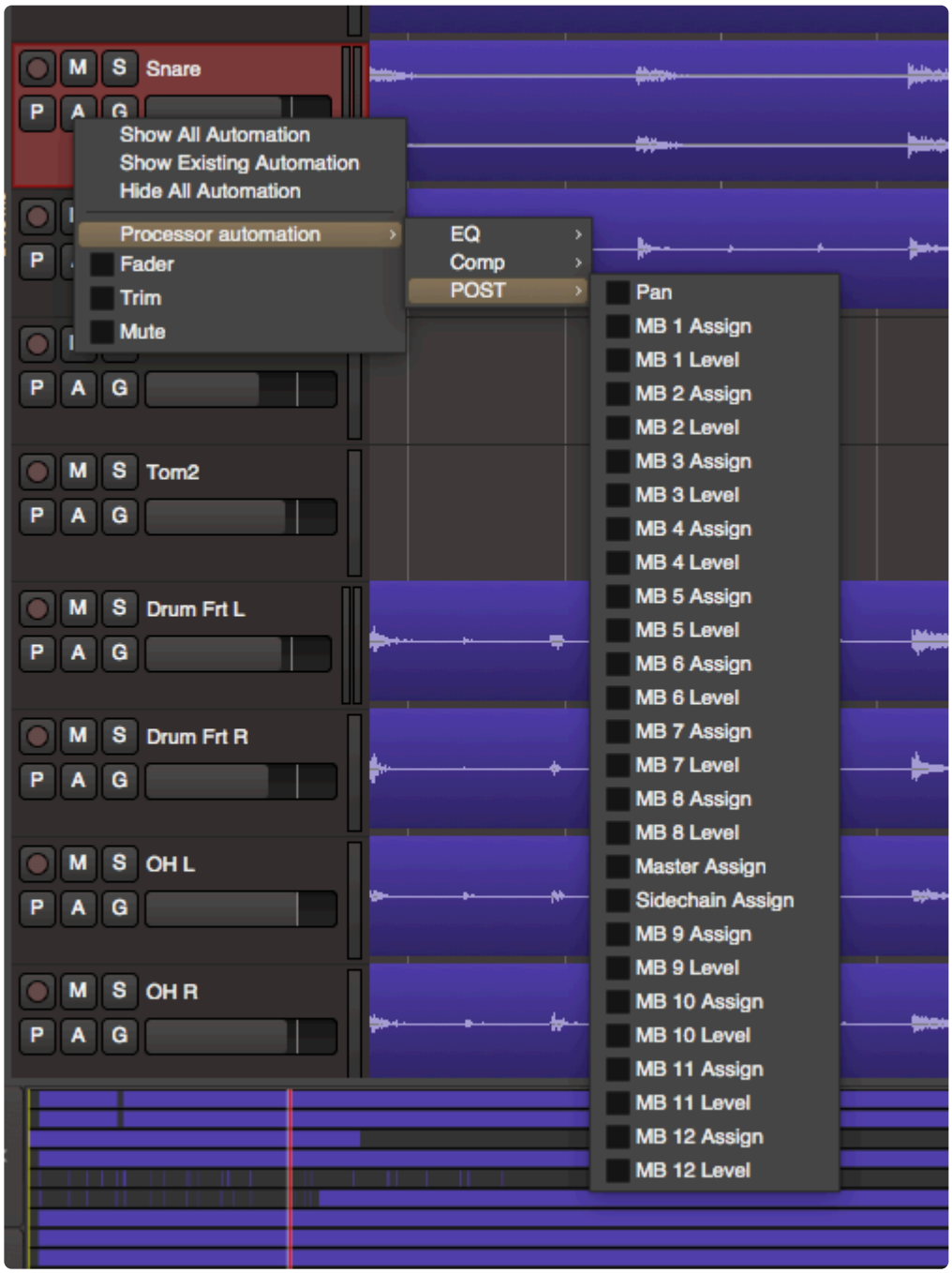
In Real-time automation the parameter is put into a “record” mode. While the transport is rolling, the user moves a knob, and these moves are recorded.

In the Mixer window below every fader there is an automation button dedicated to that fader because this is a very commonly-automated control (circled in red, below).



(this button *only* affects the automation mode of the ***fader***)

Additional parameter automation can be accessed from the the track's header: click the "A" button to view fader and mute automation. Under "Processor Automation" are EQ parameters, Compressor parameters, and plugin parameters. You'll find the Pan and bus sends under "POST".



Here is a description of the modes:

Manual	**Automation is ignored.**
Play	**Whenever you locate, the parameter will jump to the value that is recorded in the automation. You cannot change the value manually.**
Write	**When locating, the value will stay as it is (no playback of automation). During playback (only), the current value will be written to automation. You can move the control manually and your move will be written to the automation.**
Touch	**The parameter will act like it is in Play, unless you "touch" it with the mouse. Then it will record your moves until you release the mouse button, then will pop back to the saved value at that time point.**
Latch	**Operates like "Touch" mode except it will retain the last value of your moves when you released the mouse button.**

The mouse wheel has a slightly different automation mechanism. It works in "Latch" mode even if a different automation mode is selected. When the wheel is turned, it "touches" the parameter and begins recording the move. The automation continues to record until the transport stops.

Automation Thinning:

When automation is recorded in real time, the automation data points are "thinned" so that only enough control-points are created to adequately capture the moves. The points will appear very sparse if the control is moving slowly, and will be created very close together during fast moves. The aggressiveness of thinning can be adjusted if necessary via menu Edit>Preferences>Misc.

Tip: "Write" automation is a quick way to copy settings from one place to another. Locate the playhead while the parameter is in "play" to pick up the value you want to apply somewhere else. Now change the mode to Write, and locate to the new place. Then play through the desired section to apply the setting.

Punching in your automation with "Touch" mode:

When you punch-in, 2 points are created: one to hold the previous value, and one to start the new changes. Similarly, when you release the control (or stop a pass in Write mode) then 2 points are also created: one which ends the new changes, and one to hold the previously-existing value.

Automation is only recorded during forward playback. However it is possible to record automation while the transport is looping! Using this feature you can audition several different passes of the automation. When playback is stopped, the most recently-recorded moves will be kept. For example, if you loop 3 times and then stop halfway through the looped section, the resulting automation will be the first half of the 4th pass and the last half of the 3rd pass.

Graphical Automation Editing: Automation tracks

You hide and show automation lines using the “A” button on the track header, or right-click on the track header and select the parameter for automation.

Automation appears in a separate track from the audio. Multiple automation tracks may be shown simultaneously. If the automation track is associated with an audio track, a “ghost” of the region’s audio will appear in the track. Buses do not show waveforms in the automation tracks.

Automation may be graphically edited using either the Draw, Grabber or the Edit (Contents) tool. The rules for each are slightly different (and may change in the future)

Grabber tool

- Click a control point to grab it and move it
- Click a line to move the gain points on either side of the line.
- Right-click on a control point to Edit or Delete it.

Range Tool

- Drag a Range, then you can:
  - Click Delete (or Backspace) to delete the automation in that range (with guard points that prevent you from adjusting data outside your range selection)
  - Switch to the Draw tool, and drag up/down to move all the points in that range (with guard points)
  - To view a demonstration, see this video clip: Region Gain editing with the Range tool. … this technique also works with automation lanes.

> This is a case where Mixbus is much smarter than some other DAWs you might be familiar with. When you select a “range” of automation and delete the range, Mixbus doesn’t just delete the automation ‘points’ inside the range. Instead, Mixbus creates a “guard point” at the start and end of your range, and preserves the automation timeline before and after the range selection. In other words, it only deletes the automation “inside” your range selection. Sounds obvious, eh? But strangely that is not how some other DAWs work.

Draw tool

- Click anywhere in the track to create a new control point, and connect it to the points on either side.
- Click a control point to grab it and move it. (now it is selected so you can delete it)
- Click a line to move the gain points on either side of the line.
- Right-click on a control point to Edit or Delete it.

Edit Tool

- Click a control point to grab it and move it.

- Shift+right-click a control point to delete it.
- Click anywhere in the track to start a “box select”, then:
  - Click Delete (or Backspace) to delete the selected points
  - Grab one control point, and move them all together
- Right-click on a control point to Edit or Delete it.

> ✻ Note: The Grabber tool can grab an automation point, and move them up or down, but it can’t move their location in time.

Clicking with the Draw tool will create the first control point, and also draw a line to the very beginning of the track.

- Click anywhere in the track to create a new control point, and connect it to the points on either side.
- Click a control point to grab it and move it.
- Click a line to move the gain points on either side of the line.
- Use the Range tool (or Object+Smart mode) to create a “range”. Now switch to the Draw tool, click inside the range, and you can drag the entire range up/down.

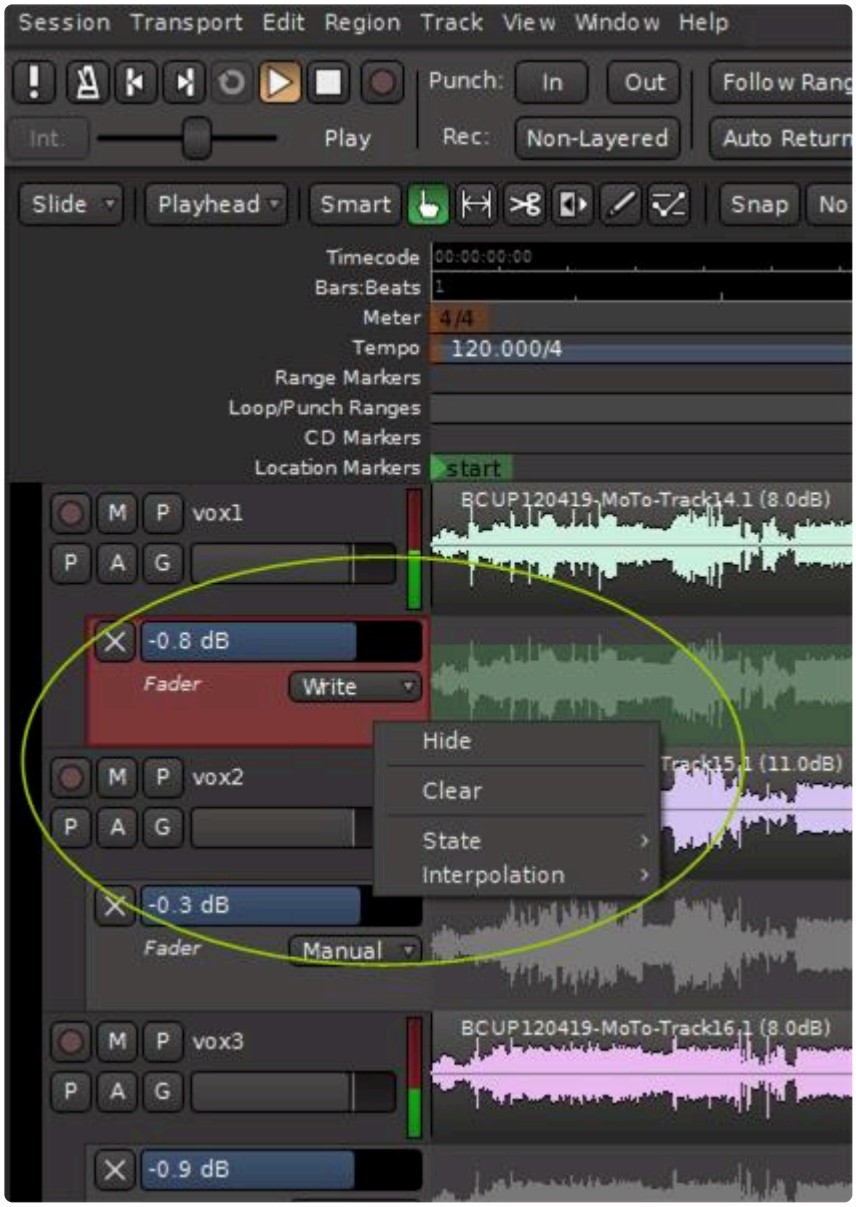
> ✻ Note: Using the Edit tool, you can click in empty space, and begin a “box selection” to select multiple points at once. You can then click on a point to move them all, or you can press Delete(or Backspace) to delete them together.

> ✻ Note: Remember that in order for the automation to be auditioned, the parameter must be in “Play” mode. You do NOT have to be in write or Touch mode to graphically edit the automation though.

> ✻ Note: Past the extents of the automation line, the “Play” value is the value of the last control point.

Clearing Automation

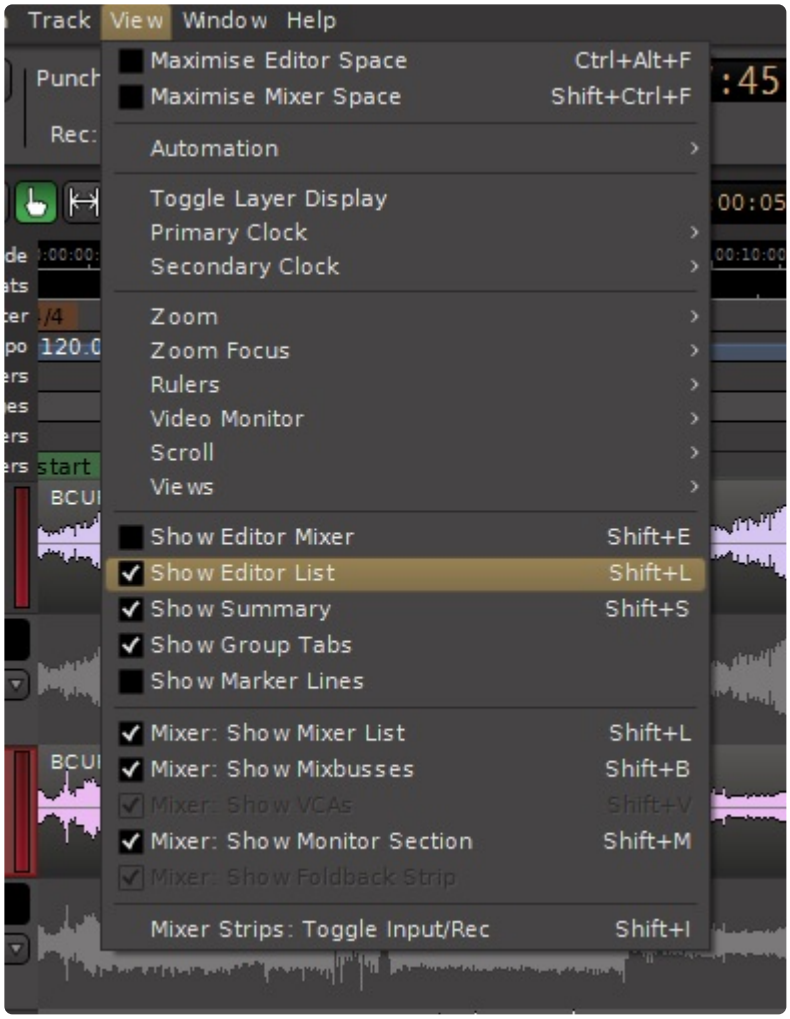
To clear saved automation in an individual Automation lane, right click on the automation lane (NOT the audio track) header and select “clear”. All of the automation saved in that lane will be cleared.
Note that if you are automating multiple controls on a particular track, each automated control will have their own automation lane. So for example you could clear and re-do your fader moves without affecting the other automation controls on the track.



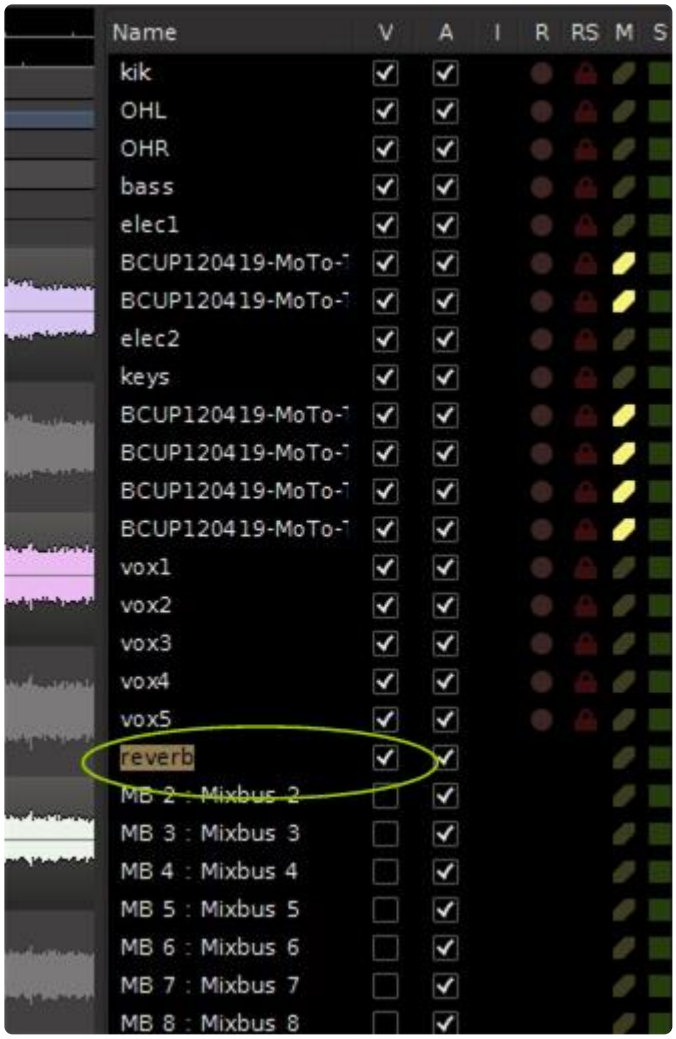
Automating Mix Buses

Normally the mixbuses are not shown at the left of the screen with the track headers. This is because mixbuses do not store audio. While automation of the mixbus fader can be selected on the mixbus strip in the Mixer panel, it is sometimes necessary to automate other functions on the mixbus.

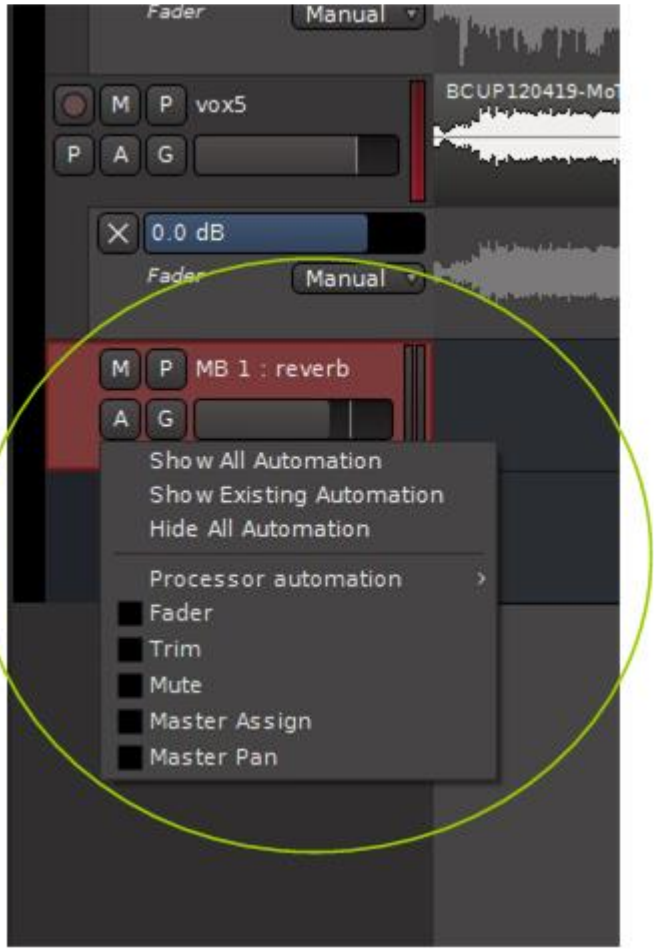
To display a mixbus at the left along with the track headers so that its parameters may be automated, first open the “Editor List”:



In the Editor List make the desired mix bus(es) visible by checking the “V” box:



The mixbus header will now be displayed at the left along with the other track headers. Clicking on the “A” automation button allows you to set up any parameter on the mixbus to be automated.


Fader
Manual
M
P
vox5
P
A
G
BCUP120419-Mo
0.0 dB
Fader
Manual
M
P
MB 1 : reverb
A
G
Show All Automation
Show Existing Automation
Hide All Automation
Processor automation
Fader
Trim
Mute
Master Assign
Master Pan

12.2. Region Gain Automation

Region Gain

https://www.youtube.com/embed/DjlBbao5gMQ?rel=0&start=135

https://www.youtube.com/embed/DjlBbao5gMQ?rel=0

https://www.youtube.com/embed/DjlBbao5gMQ?rel=0

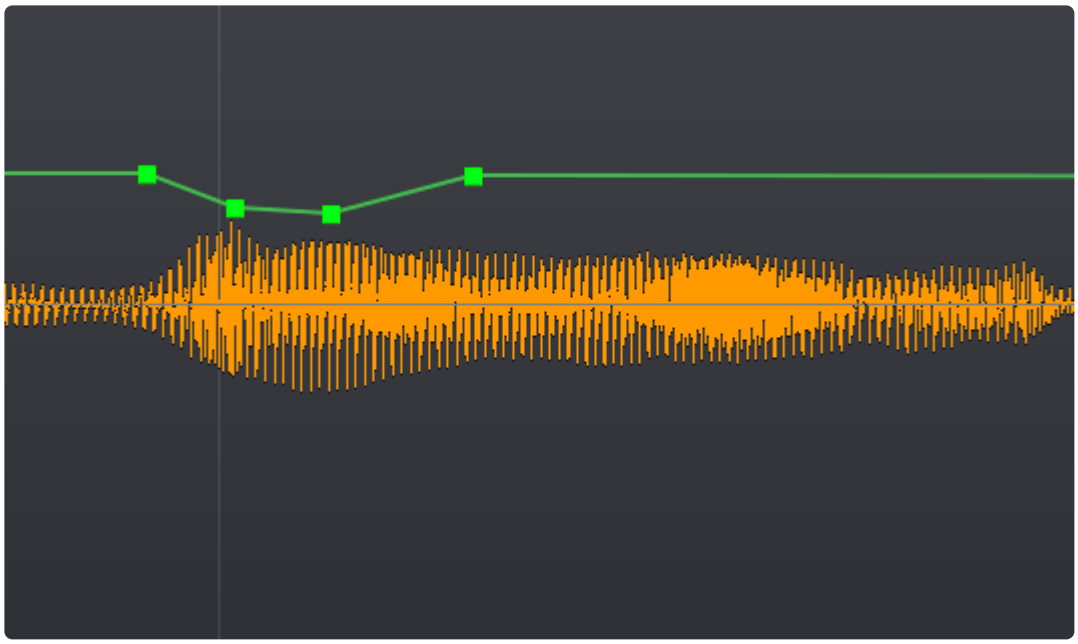
Click above for video instructions.

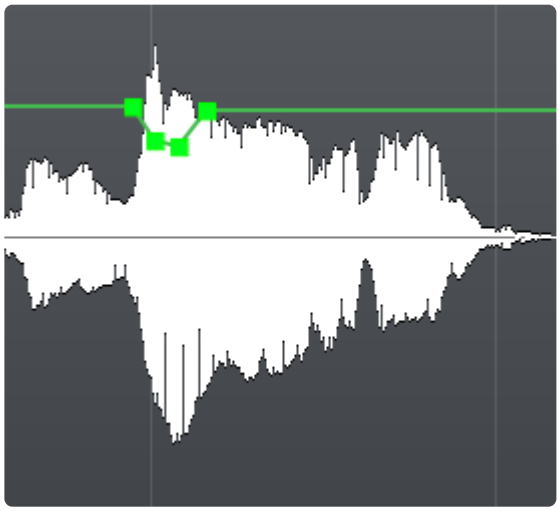
Region Gain is a volume envelope that extends for the length of the region. By default, the region envelope is “unity” (i.e. no change) across the entire region and it is hidden. To adjust region gain with the mouse, you must use the Draw tool.

In many situations you may find it useful to turn grid mode off. Clicking on a region with the Draw tool will cause a control point to be placed. By default, a region has 2 control points, one each at the start and end of the region.

Tip: If you want to quickly change the gain over the entire region, then click on the region to show the line, and then grab the line and move it up/down. This is an alternative to Region Level.

Tip: This is an excellent way to tweak a syllable or reduce a section of noise, and it is often more transparent than deleting it entirely. Also, looking through a waveshape and finding a few of hot the spots and reducing the gain will allow the compressor to be set more conservatively, providing a more transparent sound.





Mouse operations using the Range, Draw, and Edit tools inside a Region:

Range Tool

- Drag a Range, then you can switch to the Draw tool, and drag up/down to move all the points in that range (with guard points)
    - To view a demonstration, see this video clip: Region Gain editing with the Range tool.
- Drag a Range, then click Delete to remove that range

> This is a case where Mixbus is much smarter than some other DAWs you might be familiar with. When you select a “range” of automation and delete the range, Mixbus doesn’t just delete the automation ‘points’ inside the range. Instead, Mixbus creates a “guard point” at the start and end of your range, and preserves the automation timeline before and after the range selection. In other words, it only deletes the automation “inside” your range selection. Sounds obvious, eh? But strangely that is not how some other DAWs work.

Draw tool

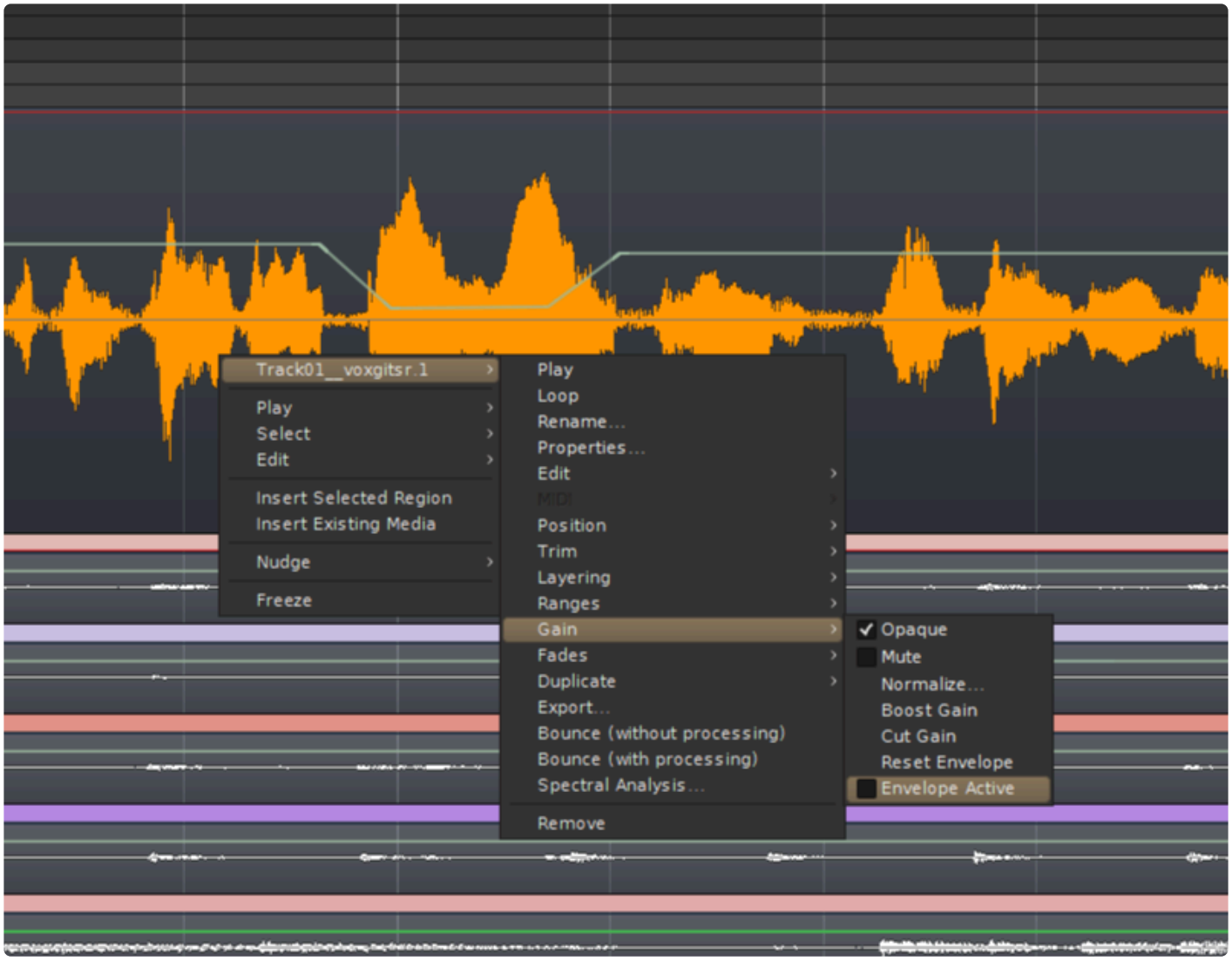
- Click anywhere in the region to create a new control point, and connect it to the points on either side.
- Click a control point to grab it and move it.
- Shift+right-click on a control point to delete it.
- Click a line to move the gain points on either side of the line.
- Right-click on a control point to Edit or Delete it.

Edit Tool

- Click a control point to grab it and move it.
- Click in empty space to make a rubberband-selection (box selection). This will select the enclosed automation *points* which can then be moved or deleted together.
- Shift+right-click a control point to delete it.
- Right-click on a control point to Edit or Delete it.

Temporarily bypassing region gain

Menu>Region>Gain>Envelope Active
or
Right click on the region and click on the track name>Gain>Envelope Active


Track01__voxgitsr.1
Play
Select
Edit
Insert Selected Region
Insert Existing Media
Nudge
Freeze
Play
Loop
Rename...
Properties...
Edit
MIDI
Position
Trim
Layering
Ranges
Gain
Fades
Duplicate
Export...
Bounce (without processing)
Bounce (with processing)
Spectral Analysis...
Remove
Opaque
Mute
Normalize...
Boost Gain
Cut Gain
Reset Envelope
Envelope Active

13. Scripting

What is a “script”?

A “Script” is a text that follows a specific format, and operates like a small program inside the main Mixbus program. This text (or “script”) can be typed inside Mixbus, and immediately used by Mixbus, rather than first being “compiled” as part of Mixbus’s source code.

Mixbus can have an “unlimited” number of scripts provided by Harrison, or developed by the user.

We expect the number of provided scripts to grow quickly, as our users develop new scripting ideas.

Mixbus uses “Lua” for its internal scripting language.

What is Lua?

Lua is a simplified programming language that is specifically designed for programs like Mixbus to use internally.

Here is a very simple example of a Lua command:

Session:set_transport_speed (1.0)

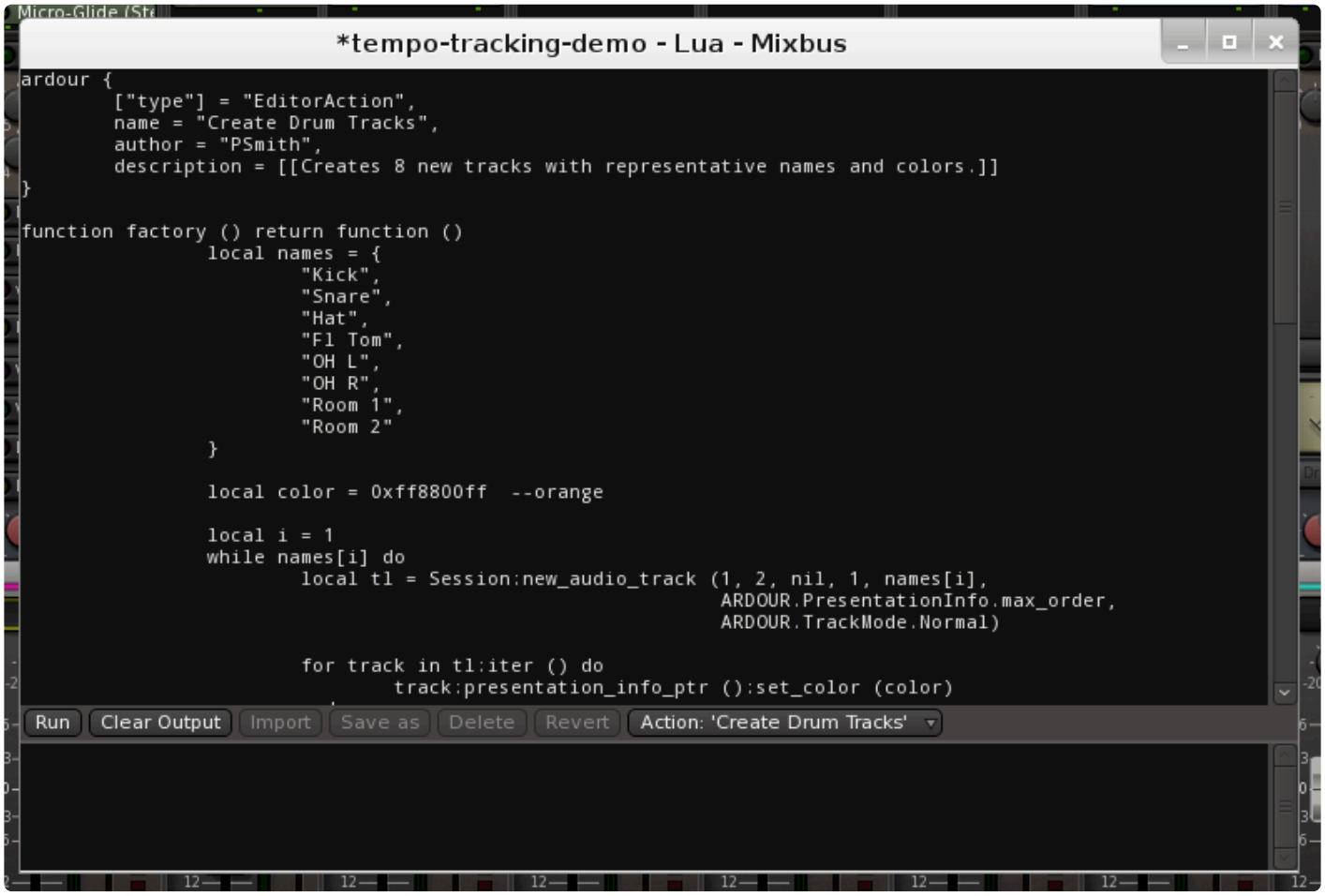
This command would cause Mixbus to start the transport at speed 1.0. (i.e. 100% or “Play”)

There is some additional text at the top of the script file that defines the script’s name and other details to Mixbus.

How do I use Lua scripts?

Several example Lua scripts are provided for your use. There are 3 common ways to initiate a script.

Lua scripts initiated directly from the “Scripting” window:



In Window->Scripting you can view a text editor where Lua scripts may be viewed, edited, and executed. You may type Lua text directly into the text editor, and then initiate the script by clicking ‘Run’ in the bottom-left corner.

In the bottom-right corner, you’ll find a menu which provides a selection of available scripts. Some of the available Action scripts are ready-to-use. Another set of scripts, called “snippets” are provided as starting points for your own Lua scripts. For example there are “snippets” that advance through each of the session’s tracks.

The most common usage of scripts is to recall an existing script, and change it slightly for your immediate needs.

For example: the “rename tracks” script will remove the text “track” from all of your tracks. But perhaps you need to remove the word “session_song_” from all of your track names. In that case, you will load the “rename tracks” script, but modify the text that it is removing or replacing.

You may “save” your scripts for later recall.

Lua scripts triggered as an “action” in the toolbar, or by keyboard shortcuts:

In Edit->Script Manager, you’ll find a dialog that allows you to assign scripts to action buttons, keyboard shortcuts, or session events:

In Preferences->Appearance->Toolbar, you can enable up to 8 action buttons, and assign a Lua script to each of the buttons. The script can even generate its own icon, as shown here for the “create Drum Tracks” script:

Right-click on an action button to assign an action to it. You may also assign a Lua script directly to a keyboard shortcut.

Lua scripts triggered by Session events:

Lua cripts can be triggered by events inside the session.

For example, a script can be triggered “when the transport is stopped”, or “when a marker is reached”.

The details of session-event scripts are currently beyond the scope of this manual.

Lua scripts as DSP Plugins:

A Lua script can be loaded as a DSP plugin. Users will find the provided Lua DSP plugins in the plugin manager, with type “Lua”.

Developing DSP plugin scripts is beyond the scope of this manual.

Developing your own Lua scripts:

Lua Scripting is developed by the Ardour open-source community, with Harrison’s funding and support. Developing your own Lua scripts is beyond the scope of this manual, but you can find regularly-updated details on this page.

14. Templates

Mixbus allows you to create “Session Templates” and “Track templates” to partially automate your workflow:

Session Templates

A “session template” is a file that stores all the settings for a session, *except for* the actual audio, midi, and automation region data. When creating a new session in the “New Session Dialog”, you may choose to select a template. This will restore the settings of the templated session including track layout, track names, plugins, mixer settings, and preferences.

To create a session template, select “Save as template” from the main Session menu.

Learn more about Session Templates here: Session Templates.

Track Templates

A “track template” is a file that stores the settings for a track. When creating a new track in the “New Track Dialog”, you may choose to select a template. This will restore the settings of the templated track.

To create a track template, right-click on the mixer strip and choose “Save as template”

Learn more about Track Templates here: Track Templates.

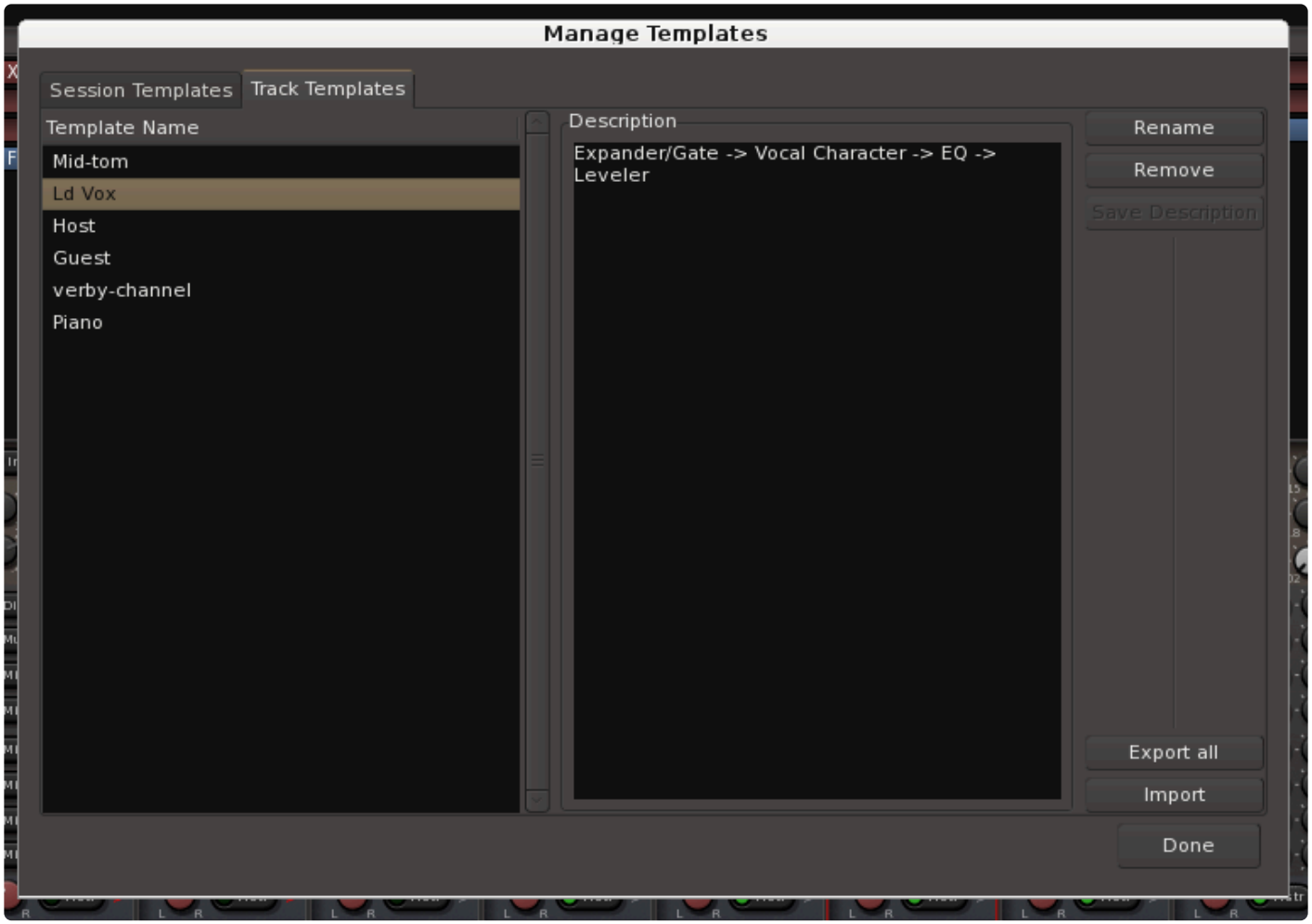
Scripted Templates

It is possible to create “scripts” which can automate certain tasks, and can act on further input from the user. The Mixbus package includes examples of scripts that operate as Session and Tracks. Creating scripted templates is beyond the scope of this manual. Contact mixbus@harrisonconsoles.com or visit the Mixbus forum for more details.

For more details about the Lua scripting language used in Mixbus, please see Scripting

Template Manager

The Template window (Window->Templates) allows you to view, manage, and share your “template” files. Both Session templates and Track templates can be shared in the Template manager window:



Select a template item on the left, and you can choose to Rename or Remove the selected item.

“Export All” – this creates a compressed archive file (with the extension .tar.gz) which includes all of your template files. You may send this file to other users who might want to use your templates

“Import” – this button allows you to select a compressed archive file (with the extension .tar.gz) which will then be extracted into your session and track template folder.

15. Importing Files

https://www.youtube.com/embed/otZgZs3CDYM?rel=0

Importing via Drag&Drop

Audio & MIDI Files may be imported to Mixbus by dragging and dropping them from a file browser into the editor window. The file will be imported to the track where it is dragged; if you want to create a new track for the file, then drag the file to the empty area at the bottom of the editor canvas.

Importing Audio files with the Import dialog

Using the import window (Session->Import) is a much more flexible and powerful way to import files to Mixbus.

The Import dialog allows you to add existing media and has 2 tabs at the top: “Browse Files”, and “Search Tags”.

- **The Path Bar**: Shows your current path and allows you to navigate back “up” the directory tree.
- **The Location Bar**: Allows you to directly type in the name of a file in this directory or enter the entire path via text (the “notebook” button shows or hides the Location bar).
- **Places**: A list of common locations on your computer. The top two items, “Search” and “Recently Used”, are always there. Clicking “Search” allows a text entry to search your system. Warning: This can take a LONG time. The lower section provides shortcuts to commonly-used locations. You can Add and Remove these locations using the buttons underneath. You can also remove or rename locations by right-clicking on them.

NOTE: On MacOS, your external hard drives will be found under “Volumes”.

The main area shows a list of files in the current directory. They can be sorted by Name, Size, or Modified Date by clicking on the headers at the top of the columns. Underneath is a selector to show “Audio Files, MIDI Files, Audio and MIDI Files” (files with recognized audio file extensions such as .wav), or it can show “All Files”.

To the right of the file selector is the Sound File Information area. When a file is selected in the browser, you get a report of the number of channels, sample rate, and in the case of broadcast wave files, the timestamp

when it was recorded. There is also an area to enter metadata “tags” for the audio files. Tagging files this way makes it very easy to find them in the future, using the “Search Tags” tab. If a MIDI file is selected you get a report of number of channels, Format, Length, and Timestamp.

Finally, there are options to Audition the files. You can click “Play” to play the file, Stop to stop playing, or choose auto-play so that files automatically start playing when you click on them. By default files will play when you double-click on them. There is also a level-fader on the right side that allows you to trim the audition level.

> ! The most important buttons are underneath. These buttons control the modes that will dictate how files are imported.

Add Files …:

- As New Tracks: Each file will be imported to a new track.
- To Region List: The imported files will be imported to the region list but no tracks will be created. This is useful if you are importing individual hits or effects into the region list, where you will use them to replace existing sounds.

Insert at:

- Use File Timestamp: After importing the file will be located at the file’s time stamp (if it is a bwav file).
- At Edit Point: After importing the file will be located at the edit point.
- At Playhead: After importing the file will be located at the playhead location.
- At Session Start: After importing the file will be located at the session’s start marker.

Mapping:

- One Track Per File: Each file will be appear on one track. If the file is stereo, it will be a stereo track. If there are more than 2 channels then they will be ignored.
- One Track Per Channel: Each channel of each file will appear as a track. If a file is stereo, it will be converted to 2 mono tracks. An 8-channel wav file will be converted to 8 mono tracks.
- Sequence Files: If this option is selected, then the files will be imported to a track, and will be imported sequentially on the timeline.
- Merge Files: This option ONLY appears when 2 files of identical length are selected. “Merge files” will load the 2 mono files as a stereo file.

MIDI Track Names:

- By track number : this will use the MIDI file’s track numbers, as the name for the newly-created track(s)
- By Track Name : this will used the MIDI file’s embedded “track name” information to name the newly-created track(s)
- By Instrument name : this will used the MIDI file’s embedded “instrument” information to name the newly-created track(s)

Conversion quality:

If the file has a different sample rate than the current session, it is necessary to do sample rate conversion. This determines the quality and it is always recommended that “Best” is used.

Instrument:

If MIDI data files are imported this option will install a basic synth plugin on those tracks called “ReasonableSynth”. This built-in utility allows you to audition your MIDI performances without any plug-in parameters to set incorrectly. It may be deselected and substituted with another plugin via the drop down list or after the track has been added to the session.

Use MIDI Tempo Map:

If enabled, Mixbus will import the MIDI Tempo markers from the first selected MIDI file, and apply it to the session’s tempo map.

Copy Files to session:

! IMPORTANT!! If you want to use a file as-is, you can uncheck “Copy Files to session” box. This saves disk space by not duplicating the file into the session. No sample-rate conversion is performed. Your files can sound sped-up or slowed-down if their sample rate does not match the sample rate of the session. By default, files are always copied to the session with appropriate sample rate conversion. If “Edit> Preferences>Misc->Always Copy Imported Files” is enabled, then this option will be grayed-out.

* Note: If a file is “linked” into the session without being copied into the session, then the sound won’t be available if the file is moved or deleted. For example, if the file is on a removable hard drive then it might not be available next time the session is used. Use this at your own risk.

Tags:

If you have used the Tag feature to “tag” files that are used in the past, then you can use the Search Tags tab to find those files quickly. In this case, we searched for the term “shout” which was added to the files on the previous page. They instantly appear in the file area to be auditioned and imported.

16. Exporting Files

Please watch the following video for an overview of exporting features in Mixbus:

https://www.youtube.com/embed/NT5Zo1OHKaQ?rel=0

Differences between Bounce, Consolidate, and Export

Mixbus has multiple ways of “rendering” audio into contiguous audio files.

Bounce

The least-used method, “Bouncing” is a way to store a drum hit or sound effect that will be reused later in the project. A new region will be created in the region window. The bounced region will have the region’s gain, gain curve, and fades applied to the audio. You can later drag this region back into the project wherever you need it.

To initiate a Bounce action, right-click on a region and choose “Bounce”. A new region will appear in the Editor List, inside the “Region” tab.

Consolidate

“Consolidate” allows you to select a time-range consisting of many regions, and create a single region on the timeline. This can simplify future edits or exporting operations, because the various layers, gain curves, fades, and other region properties are made into a single contiguous audio region which can be manipulated more easily.

To initiate a Consolidate action, you will select a time-range with the Range tool (this can consist of multiple tracks), right-click on the Range, and choose Consolidate. The regions under the selection will be consolidated into contiguous audio regions.

Exporting

“Exporting” is used to generate one or more files, outside of the Mixbus session, in the file-location and format that the user defines. “Exporting” is how you create the final stereo output file of a song, or a series of songs and markers for a CD, or a video for posting on YouTube.

There are many ways to initiate an “Export”, and Mixbus supports many, many options. A single Export event might generate multiple files, in multiple formats, with multiple different , and with or without metadata such as CD markers, song names, etc. Exporting is probably one of the most complicated (but powerful) features of Mixbus.

You can cancel an Export operation at any time by closing the dialog and/or clicking Cancel.

Exporting to an Audio File

Once a mix is ready for delivery, you will need to generate a final output from Mixbus. There are several ways to initiate an Export from Mixbus; each method helps to define the tracks that will be exported, the start/end times of the exported file, and the format of the files to be created.

Initiating an Export

There are 5 common ways to initiate an export:

- Session>Export>Export to Audio File(s)
- Session>Export>Stem Export
- Session>Export>Export to Video File
- (Selected Region)->Right Click->Export
- (Range Selection)->Right Click Menu->Export

Export to Audio File(s)

This method initiates the most common export. The Export dialog will appear, prompting you for details such as the exported filename and location, the file format, and the time-ranges that you would like to export.

> When choosing “Export to Audio File(s)”, Mixbus will automatically choose the master bus output, and the session start/end markers for an export. But you can override the settings to choose a different time span, in the Export dialog window. If you choose multiple timespans, then you will receive one audio file per time span. This allows you to export ranges that are non-contiguous, overlapping, or even nested.

Stem Export

This initiates an export of individual tracks. In the “Channels” tab, you may select whether to apply the channel processing, or to leave the track contents unaffected. Each track or bus will generate a separate audio file. If you choose multiple timespans, then you will receive a separate audio file per time span.

Export to Video File

This initiates an export similar to the main session export, but it generates a video file by combining the video track with the master-bus outputs.

Region export

This simply exports the selected audio region into any format that the user chooses. This does ~~not~~ include any mixer-strip or plugin effects.

Range export

This initiates an export, just like the session export, but it automatically selects the selected time-range as the start/end times for the exported file.

Setting the format for the export:

In the Export window are 3 tabs across the top:

- File format
- Time Span
- Channels

File Format tab

The **File format tab** allows control of settings such as bit depth, sample rate, compression type, location to be saved, and file name structure.

Additionally,options for generating a “Loudness Report”, as well as automated uploading to SoundCloud is available (must first have a SoundCloud account via www.soundcloud.com).

Prior to version v5.2, exporting to MP3 required a post-export command. Now, you may select mp3 directly in the Format dialog.

Multiple format exporting:

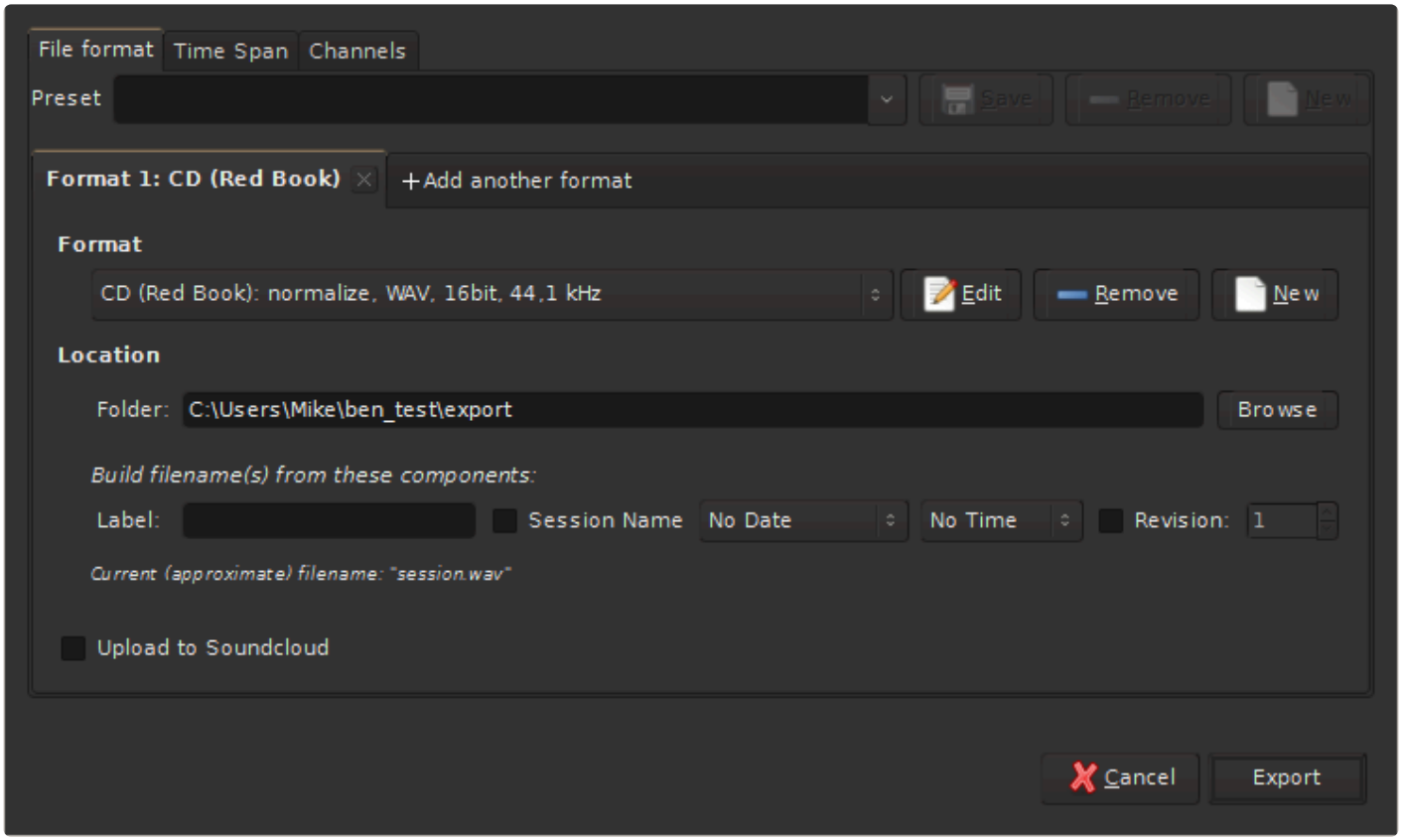
Many files formats may be generated and exported at one time by clicking “+ Add another format”.

- Export Session to Audio File (the most common option): This exports the entire range from the Start marker to the End marker.
- Export Selected Range to Audio File: This exports the current Range selection to an audio file (Note: this can also be triggered by right-clicking on the selected range and choosing “export range”).
- Export Selected Regions to Audio File: This exports a length of time starting at the first selected region, and continuing to the end of the last selected region.
- Export Range Markers to Multiple Audio Files: This very powerful feature allows you to export several ranges (songs) at one time. The resulting files will use the region names as the file names.

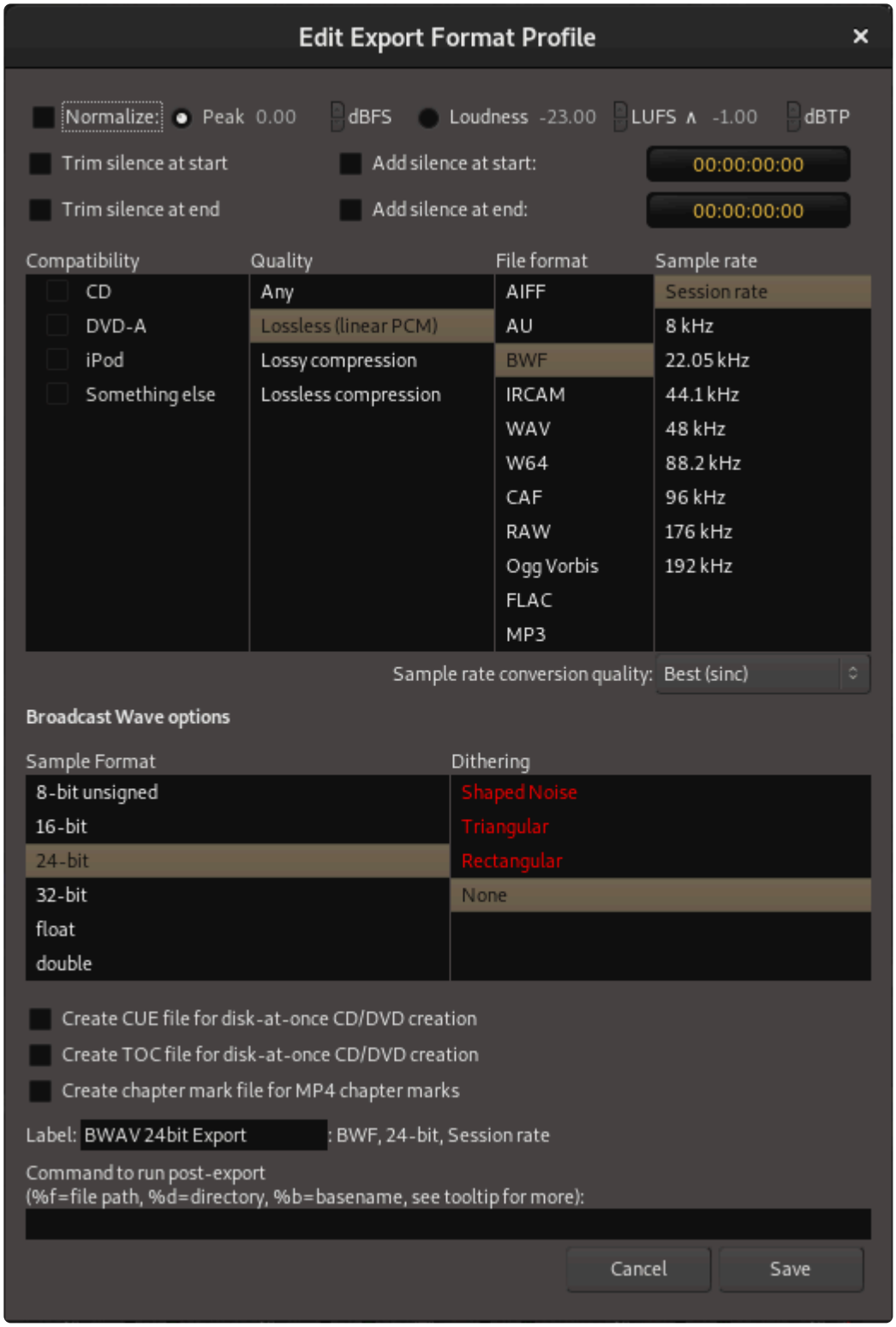
Before the export is initiated, you must choose the location and format of the exported file.

Export to File: Enter the path and file name that you wish to create. This defaults to the sessions “export” folder. Click “Browse” to see a file browser and navigate to a different folder.

- Export Source: Under the “Channels” tab choose “Master Output” to export the master bus output of Mixbus. additionally you can export individual tracks separately, or control the Left/Right assignment of tracks to the file.
- Channels: Chooses whether the resulting file will be Stereo or Mono.



Once a format(s) is chosen it may be edited.
The formats are editable using this dialog box:



Edit Export Format Profile:

- File Format: (the list depends on the system's available options). For CD quality, choose WAV.
- Sample Format: The "32-bit" setting means 32-bit integer format. This is ultra-high resolution but does not accommodate signals over 0dBFS. The "float" setting is 32-bit floating point which accommodates signals over 0 dBFS which is preferred in most cases. For CD quality audio choose 16-bit.

- Sample Rate: For CD-quality audio choose 44.1 kHz.
- Conversion Quality: This is only enabled if the export sample rate is not the same as the session. The setting “best” is recommended.
- Dither Type: None, Rectangular, Shaped Noise, or Triangular. The “None” or “Shaped” settings are recommended.
- CD Marker File Type: CUE & TOC. These file types may be utilized by CD-burning software to define track indexes, track names, and other metadata.

If a project is intended to be published as a CD, you will want to create CD markers so Mixbus will generate a TOC (table of contents) text file for the CD.

Label: this is the name that will appear in the list of formats.

Post-export command: this feature can be used to launch a scripted operation on the exported file. More details here: (blank)

Time Spans tab

This tab allows you to choose one or more time-spans. Each selected time-span will be exported, using the settings in the “Format” and “Channels” tabs. For example if you have 2 different formats chosen, and 2 different time spans, then 4 files will be created during the export.

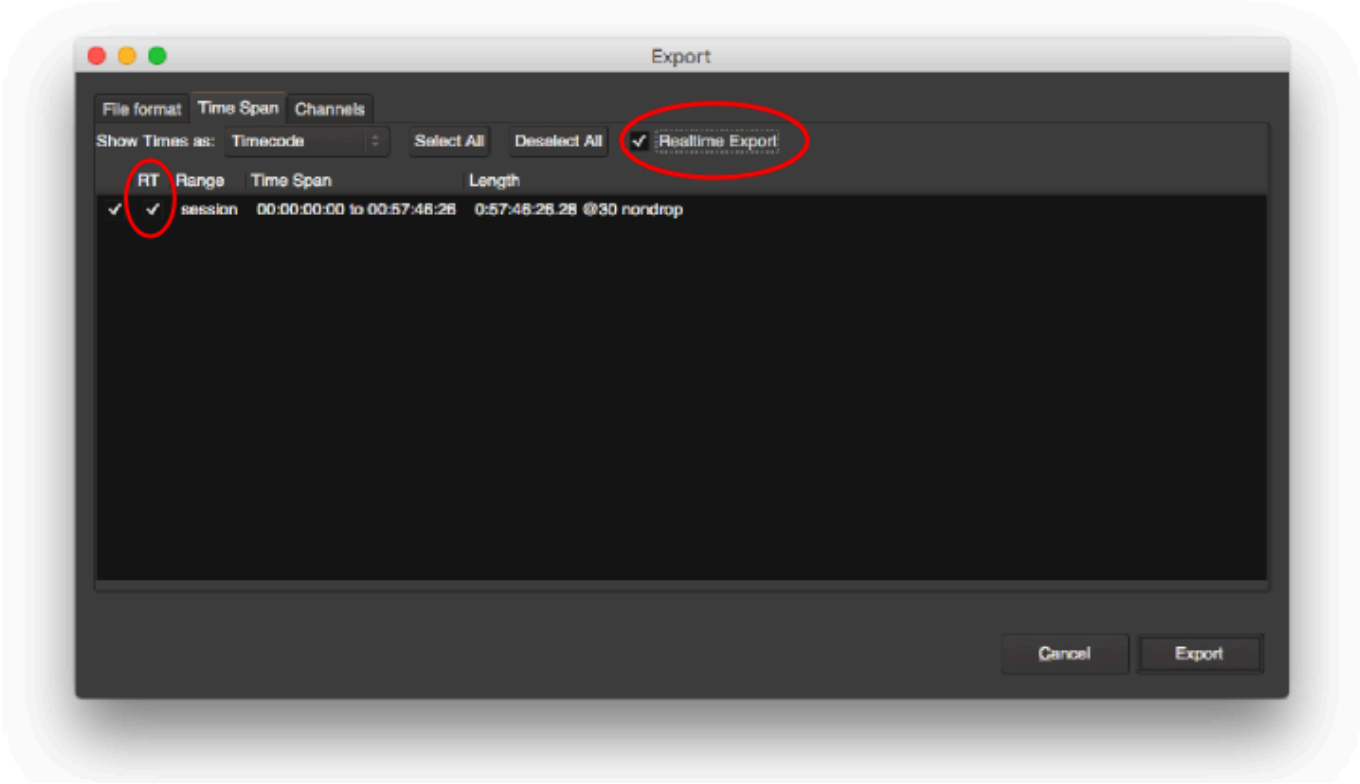
Time-spans include:

- Session Start/End (this is selected by default if you trigger a Session->Export to Audio File)
- Range Selection (this is selected by default if you trigger the export by right-clicking on a Range). If no range is selected, then this does not appear
- Named ranges: If you have created named ranges (for example, your session has multiple songs on the timeline, and each is assigned a named range) then they will appear here.

Realtime Export

Use this option if your session makes use of external hardware audio processing.

> Normally, Mixbus maximizes your CPU to perform the fastest-possible export. However, some users have external devices, such as rackmount analog gear or external MIDI synthesizers, which cannot be used at faster-than-realtime speed. In the past, we required these users to manually create a bounce track and mixdown to it, rather than using the normal Export features. Realtime Export provides all the same features as a regular export, but it limits the engine to run in “real time” so external gear can be utilized during the export.



Channels Tab

The Channels tab allows the user to choose which tracks or buses will be exported. The operation is slightly different, depending on the way the export was initiated.

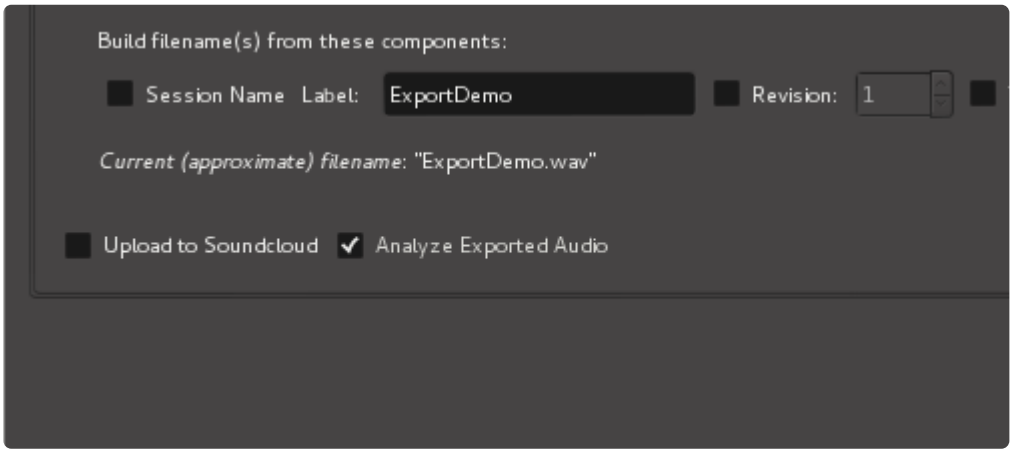
When “Session->Export to Audio Files” is selected, Mixbus will select the Master bus for export. But the user may change this selection to choose any tracks and/or buses. Mixbus will “sum” the track outputs into a single file.

When “Session->Stem Export” is selected, Mixbus will not select any tracks or buses by default. The user must select which tracks or buses to export. Each selected track will result in one exported file. The track’s output is used to define the number of channels in the exported file. If a track is mono throughout, then it will result in a mono audio file. If the track has 2 outputs (stereo) then the resulting file will be stereo.

When stem exporting, the user may choose whether to apply the track’s processing (fader, mute, eq, compressor, and plugins) or to export the raw audio without processing.

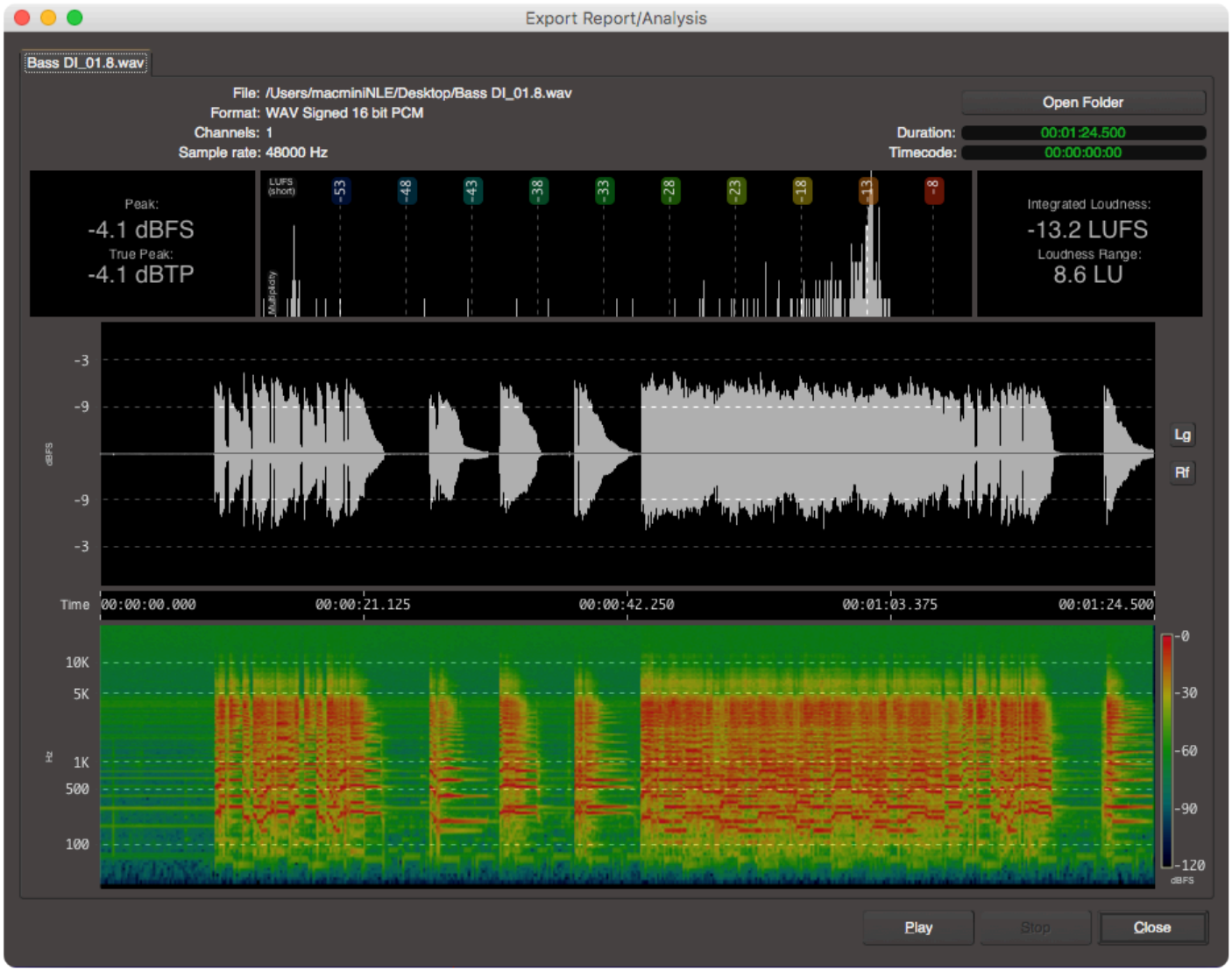
The selection of the “Channels” tab is combined with the “Format” and “Time Range” tabs. If you are doing a “stem export”, with 8 tracks selected, and 2 formats, in 2 different time spans, then this will result in 8×2×2 = 32 different files on export!

Export Analysis:



If you enable the “Export Analysis” checkbox, then Mixbus will display a summary of the exported file:

Export Report and Analysis



The Export Analysis dialog may be displayed after an export The export analysis window provides these features:

- LUFS loudness measurement
- LU (Loudness Unit) histogram
- LU Range
- Waveform display, with peak indicators
- Peak Sample value + True Peak value
- Spectrogram view

These tools provide you an extra level of confidence that your track sounds great and meets common deliverable requirements for distribution and broadcast.

The following video describes the export report and analysis tool in more detail.

https://www.youtube.com/embed/UL1s5oubozs?rel=0

The following video describes how to “bounce” the master bus down to a new stereo track:

https://www.youtube.com/embed/1tCTd6JmqXY?rel=0

The following video describes how to export multiple ranges into separate audio files:

https://www.youtube.com/embed/rWURWqJVWcg?rel=0

The following video shows the “Export” features of Mixbus v2; this additional information is still sometimes useful for v3 users:

https://www.youtube.com/embed/wddiGR-Y1bk?rel=0

16.1. Post-Export Command Reference

The “Post Export Command” can be used to operate on files after the export process has completed.

command arguments

Arguments may be passed to the command interpreter using these aliases:

%f Full path and filename of the exported audio file
%d Directory containing the exported audio file (including trailing directory separator)
%b Basename of the exported audio file (without extension)
%s Path to the current session file
%n Name of the current session file
%% A literal percent sign

Any part of the command-line enclosed in double-quotes (") will be used as-is.

For example, exporting an mp3 file can be done by inserting lame -b320 %f which will convert the exported audio file (‘%f’) to a 320 kbs mp3 using the lame encoder (provided lame is installed first on the system).

Example: File copy to a backup drive on windows:

cp %f D:/backup

Example: converting to mp3 using a third-party encoder (such as LAME)

NOTE: Since version 5.2, you can select mp3 as a format in the Export dialog.

To export MP3’s you will need to edit one of your *export profiles* (or make a new one). For details on how to do that, visit the Exporting manual page. Additionally you will need access to the “LAME” encoder (despite the name, the “lame” encoder is widely regarded as one of the best-sounding mp3 encoders), as a program that can be launched from a command-line.

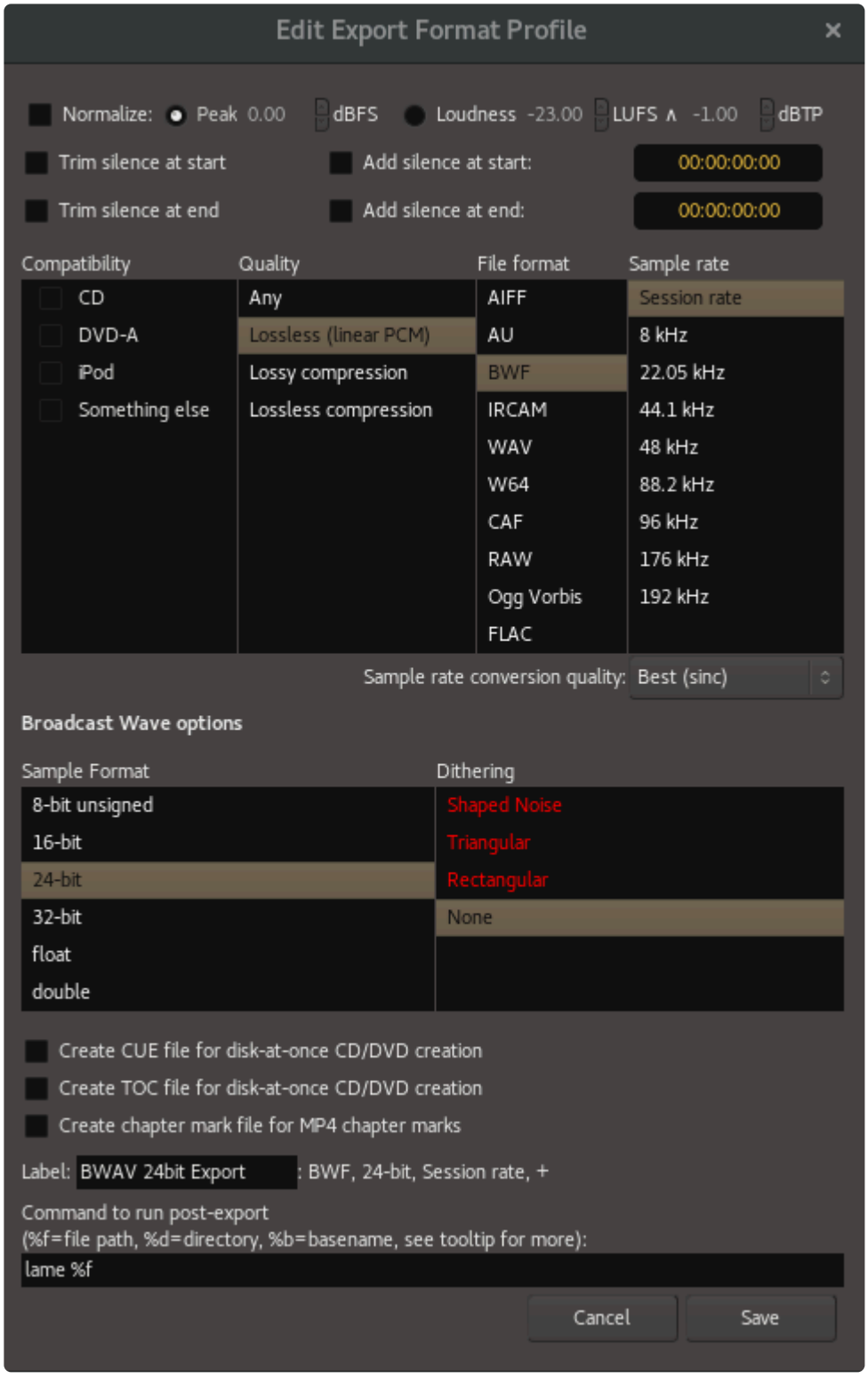
The LAME utility can be downloaded from multiple sources.

- For Windows: the one we use on our testing machine is RareWares LAME Bundle.
- For mac, we use Thalictrum’s LAME (this will make it available from the terminal)
- For Linux machines (debian based at least) you can probably install lame directly from your package manager. For example: “sudo apt-get install lame”

Use these at your own risk.

Finally, set up your command post export to run the lame encoder:

1. Edit your existing export format
2. In the section “command post export” (located at the bottom of the format page) you can choose to run a system command after your export has finished, and we link to some variables that you can use like %f being the absolute file-path of your newly exported file.
3. Set your post export command to run the lame encoder on your newly acquired file. See the picture for an example of our setup.



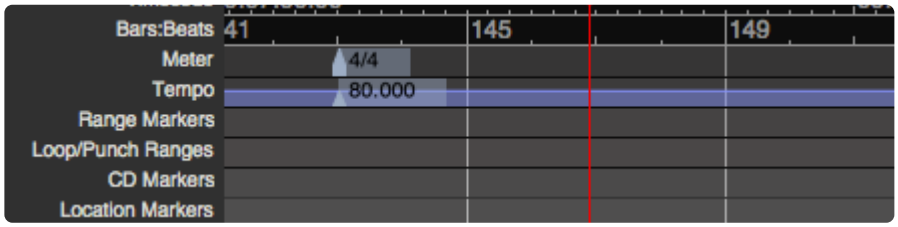
If the executable file isn't available in your command-line PATH, then you will need to tell Mixbus the absolute path to the executable. Instead of just putting "lame" you would need to do: "path\to\lame.exe %f"

Hit export and now you will have an MP3 file exported *automatically*!
(note that some LAME implementations will need additional commands to export at higher bit rates, and some might require a slightly different syntax to the one provided)

17. Tempo, Metronome, & the Grid

Defining a Song's Tempo and Meter

Mixbus allows simple creation and editing of a musical time/tempo map for each session. The meter and tempo map is controlled with the Meter and Tempo rulers. To show these rulers, right-click in the ruler area underneath the transport buttons and check the boxes for the Meter and Tempo rulers. Alternatively, you can use the View menu to hide/show the rulers.



Here are some reasons that you might want to set the Meter/Tempo for your song:

1. To record with a "click" or metronome to help maintain consistent time among different performers and parts.
2. To use the Grid to facilitate editing (if the audio follows the tempo map of the song).
3. To add MIDI segments or loops, and have them follow the tempo changes.

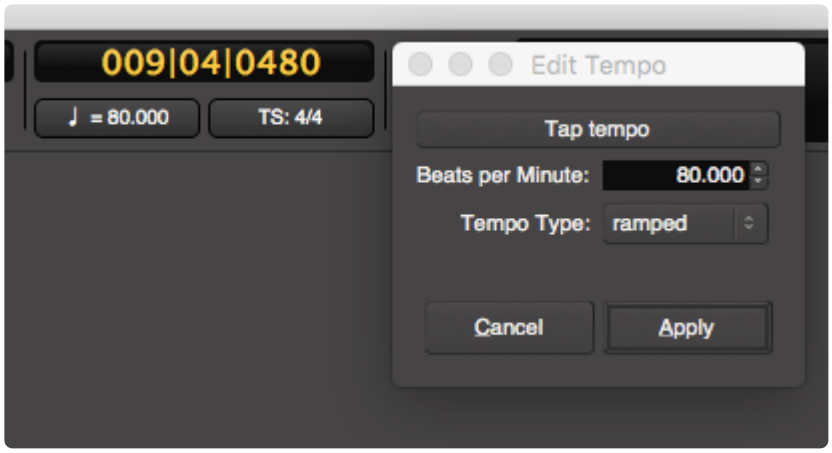
By default, Mixbus starts with a single Tempo and Meter marker, set to 120 bpm and 4/4, respectively. If the session has a fixed tempo (or doesn't use tempo at all), then it is not necessary to create additional markers; and clicking on the Tempo button will adjust the session's initial tempo marker.

In other cases, you may want the tempo to change at different times during the session. In this case you will right-click on the existing markers and set the tempo to the desired tempo and time signature. Right-clicking in the Meter/Tempo rulers allow you to create new Meter or Tempo markers.

> ✻ You may select the initial Meter Marker, and move it to match the start of your song. This allows you to align the first measure (bar) line with your song's start. Use the tempo-mapping tools in Tempo Maps to match the editors bar|beat clock with your natural performance.

Tempo Controls

You can click on the "tempo" button underneath the clock, and this will launch the Tempo dialog:



The tempo dialog provides these options:

- Tap Tempo: click this button repeatedly to set a tempo
- Beats per Minute: type a new tempo here, or use the +/- buttons to increment the tempo
- Tempo Starts at: this indicates the measure/beat where the tempo change occurs; edit these fields if needed.
- Tempo Type: tempo markers can be Ramped or Constant. Learn more about tempo ramps here: Tempo Maps
- Lock Style: learn more about music-locked tempo markers here: Audio Time vs Music Time (NOTE: the initial tempo marker is always locked to music time, so this option does not appear)

When you click the “tempo” button under the clock, you are always adjusting the most-recently passed tempo marker (when the dialog was launched). If the song only has one tempo marker, then you’re adjusting the global tempo.

Tip: Here’s an easy way to set the tempo for a song: record a portion of the song, and then select one bar with the Range tool. Now click “Edit->Tempo->Set Tempo from Edit Range” to set the tempo.

Time Signature (Meter) Controls

You can click on the “TS” (Time Signature) button underneath the clock, and this will launch the Meter dialog:

The tempo dialog provides these options:

- Beats Per Bar: the numerator of the time signature
- Note Value: the denominator of the time signature
- Meter Starts at Bar: this indicates the measure where the meter change occurs; edit these fields if needed.

- Lock Style: learn more about music-locked meter markers here: Audio Time vs Music Time (NOTE: the initial tempo marker is always locked to music time, so this option does not appear)

> Note: Meter changes can only occur at the beginning of a new measure..

> Note: the first “Meter” marker may be dragged away from time 0:00. This allows you to start the musical timeline (bar 1, beat 1) anywhere in the session. You can drag it to the start of your song so the bar|beat readout will match your musical score.

Using the Metronome (click)

To play a click sound during playback and recording, enable the “Metronome” button near the top-right of the edit window:

The rate of the click is defined by the session’s Tempo and Meter rulers.

You can visit Preferences->Metronome to edit the click’s sound; or just right-click on the metronome button to immediately view the properties.

> You can mouse-wheel (or 2-finger scroll, on a touchpad) to adjust the metronome’s volume directly from the transport window.

> Tip: It is common practice to leave some time at the beginning of a song, rather than starting from the zero indication. Some common choices are 8 measures, 1 minute, 10 minutes, or 1 hour. This allows space at the beginning of the song to add an intro, place a sync click or test tones, or just make editing easier by allowing some unallocated space at the beginning of the track.

Using the Grid

Editing with the “Grid” allows you to make edits perfectly aligned with other events. If the grid is set to musical time (and the audio was recorded to a click) then editing can be done very quickly without zooming in, because you know all edits will happen on a musically-sensible time.

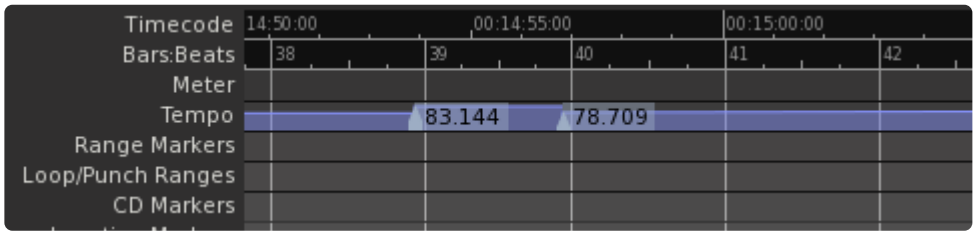
To learn more about Snap + Grid settings, visit Snap and Grid

17.1. Tempo Maps

What is a Tempo Map?

By default, every Mixbus session starts with an initial Meter marker of 4/4, and an initial Tempo marker of 120 bpm.

However, the user may add additional Tempo and Meter “markers” which change the tempo on the timeline. See: Tempo, Metronome & the Grid



Constant vs Ramped tempo markers

A “constant” tempo marker will set a steady tempo which continues until the next tempo marker is reached.

A “ramped” marker will cause the tempo to gradually change to meet the tempo value of the marker that succeeds it. This allows you to track the natural speedup/slowdown of a live performance, or create a natural ritardando at the end of the song. The tempo “ramp” is shown with a sloped line, indicating the difference between the two tempos:

Creating a Tempo Map from a recorded performance.

This video demonstrates how to create a tempo ramp from a live performance:

https://www.youtube.com/embed/rrr9lr_Pbkg?rel=0

Editing tempo markers & the tempo lane

There are several ways to manipulate the tempo markers on the tempo ruler:

- Click & drag the tempo marker to change its location on the timeline
- Right-click on the tempo marker to change its properties
- Shift+Right-click to remove a tempo marker
- Shift+drag up/down on a tempo marker to change its tempo value
- Hold “Ctrl” (“Cmd” on Mac) while dragging a tempo marker to make a copy with the same tempo.

Additionally, you can edit the Tempo Line itself

- Shift+Click on the tempo-line between 2 tempo markers. Drag it left/right to adjust the tempo of the previous marker.
- Shift+Ctrl-click on the tempo line to create a “pinch”. A “pinch” adjusts the tempo line between 2 markers, and adjusts them in such a way that it does not move the succeeding markers.

! If you are unable to Shift+Drag on the Tempo Line, please visit Preferences->Editor-

>Modifiers and click “Reset to Recommended Defaults” (or make sure that “When Beginning a Drag -> Constrained Drag Using:” is set to “Shift”).

Importing a tempo map from MIDI

Mixbus can import a “tempo map” from an existing MIDI file. To do that, use the Import dialog to choose the MIDI file. And enable the checkbox for “import tempo map”.

MIDI files do not allow gradual “tempo ramps”. Each tempo marker in the file will indicate a discrete change to a new tempo. For that reason, MIDI tempo changes will not be as ‘smooth’ as Mixbus’s tempo ramps.

17.2. Audio Time vs Music Time

What is the difference between *Audio* time and *Music* time?

Audio time (or “real” time) is the length and tempo of a performance as executed and recorded. Whereas “music” (BBT, for Bar-Beat-Tick) time is a series of arranging data (notes and rests) that can be executed very quickly, or very slowly, and still be considered the same.

In “music” time, a quarter-note is a quarter note, regardless of tempo. MIDI consists only of arrangement data, and so it is possible to arrange a MIDI sequence and later increase or decrease the tempo. Just like musicians playing from a score, the arrangement’s tempo will increase, but the musical expression and timbre will remain largely unchanged.

When recording audio, the performers are recorded in real-time subject to the players’ own tempo and interpretation of the music. And once recorded, you cannot easily change the tempo of the recording. Once audio is recorded, it requires quite extensive processing and manipulation to change the tempo … and even more so if you don’t want the pitch to change as well (i.e. the “chipmunk” effect)

The problem is that Mixbus (and other DAWs) are expected to show both MIDI and Audio data on the same timeline. And there’s some ambiguity: one can imagine audio data (such as a single drum hit) that has no inherent tempo, and can be considered arrangement data, and subject to tempo changes in the musical timeline; or one can imagine a MIDI event that is related to the audio timeline and should not move if (for example) the tempo is changed.

Why is this so hard?

Attempting to shorten or lengthen an audio performance, after it is recorded, requires the sound to be digitally manipulated which will incur some loss of quality. Because Mixbus is intended for high-quality editing, mixing, and mastering tasks, we have chosen to give priority to the “audio” timeline. But there are equally valid reasons why a user might be more interested in the “music” time, and so both music and audio-timelines must coexist.

Consider these examples:

- A live performance is recorded, and the musicians have varied widely in tempo and meter. The user would like to develop a “grid” which follows the varying tempo, and allows editing to the “grid”, just as if the performance were recorded strictly to a metronome. Or perhaps the user would like to add MIDI arrangement using a musical score editor, and these notes must somehow be synced to the performance’s tempo.
- A MIDI performance is generated with a fixed tempo. Later, a vocal and instrumental tracks are added to the MIDI arrangement, and these tracks are edited with crossfades to assemble the best-sounding mix. Now the arranger would like to gradually decrease the tempo at the end of the song (i.e. ritardando). The MIDI events, being tied to the Music timeline, will stretch farther apart, but will stay

in tune. But what will become of the audio regions?
- A session which mixes spoken-word sections with musical sections. During a musical portion (implemented with MIDI), the tempo is increased, so the musical portion finishes faster than it did, when the audio was recorded. Should the audio voiceover move left to accommodate the shorter musical segment?

Mixbus has some mechanisms to accommodate these situations. But it can get quite complicated.

We have chosen to focus our efforts on the “audio” timeline, rather than the “music” timeline, which is more common among those workstations that began as MIDI sequencers.

How do you use Audio Time and Musical Time?

Every item on the timeline (regions, markers, and automation control points) must be either connected to the “audio” timeline, or the music timeline. If an item is attached to the musical timeline, then any changes to the BBT (bar-beat-tick) timeline will be moved along with the musical timeline. For example, if you increase the tempo then the regions will move to occur earlier.

If you are assembling “samples” of a kick drum alongside MIDI-sequences, then it is appropriate to attach the regions to the musical timeline.

What are the default settings for the different timelines?

By default, audio regions use the audio timeline. You can right-click on an audio region to “glue” it to the BBT musical timeline.

By default, MIDI regions are “glued” to the BBT timeline, while audio regions are not. You can right-click on an audio region (or selection of region) to “glue” it to the BBT musical timeline.

Tempo markers are glued to the BBT timeline by default, but they may be unglued (and therefore attached to the audio timeline), by right-clicking on them.

Automation control points are attached to the audio timeline, and cannot (currently) be changed.

Known issues.

Currently, even if a MIDI region is attached to the audio timeline, and it won’t move when the tempo map is moved; its CONTENTS will still stretch. That is not desired, so we will likely fix it in a future version.

This is complicated stuff. Investigating other DAW manuals and videos will reveal that there are no easy, non-destructive ways to solve these issues.

So what is the preferred workflow for managing a project that

uses both the audio and BBT timelines?

Generally speaking, Mixbus is optimized to work in one of these 2 ways:

1) Record a live performance on the audio timeline, (optionally) map the tempo, and then record MIDI instruments, but never move the tempo/BBT-map after you've started adding MIDI.
2) Arrange your song in MIDI, and adjust tempos/timeline as needed. Then record audio, and never again move the tempo/BBT-map after you've started recording audio.

18. Menu Reference

This menu reference documents selections available to the user. In many cases menu selections are self explanatory and require no elaboration. Other items may require explanation, in which case a brief passage will be provided. In other cases links to specific pages of this manual will be provided.

Most menu actions have an associated keyboard shortcut. You can change the default shortcuts by visiting Window->Keyboard Shortcuts.

Why are some menu items disabled or greyed-out?

Some menu items, particularly the items in the “Edit”, “Track” and “Region” menus, will be disabled if there is nothing selected which might apply the operation. For example, the item “Track->Remove” will only be available if one or more tracks are selected.

18.1. Session Menu

The Session menu provides the main functions of session management:

- New: More here
- Open: More here
- Recent: More here
- Close: More here

- Save: More here
- Save As: More here
- Rename: More here

- Snapshot (& keep working on current session): More here
- Snapshot (& switch to new version): More here

- Save Template: More here

- Archive: NEW in v4. This allows the user to store the current snapshot in a compressed format, using only the needed audio and (optionally) converting the audio files to a lossless compressed format. The resulting archive (zip file) can be reopened from the Mixbus “open” dialog.

- Metadata: More here

- Add Track or Bus:More here

- Import: More here

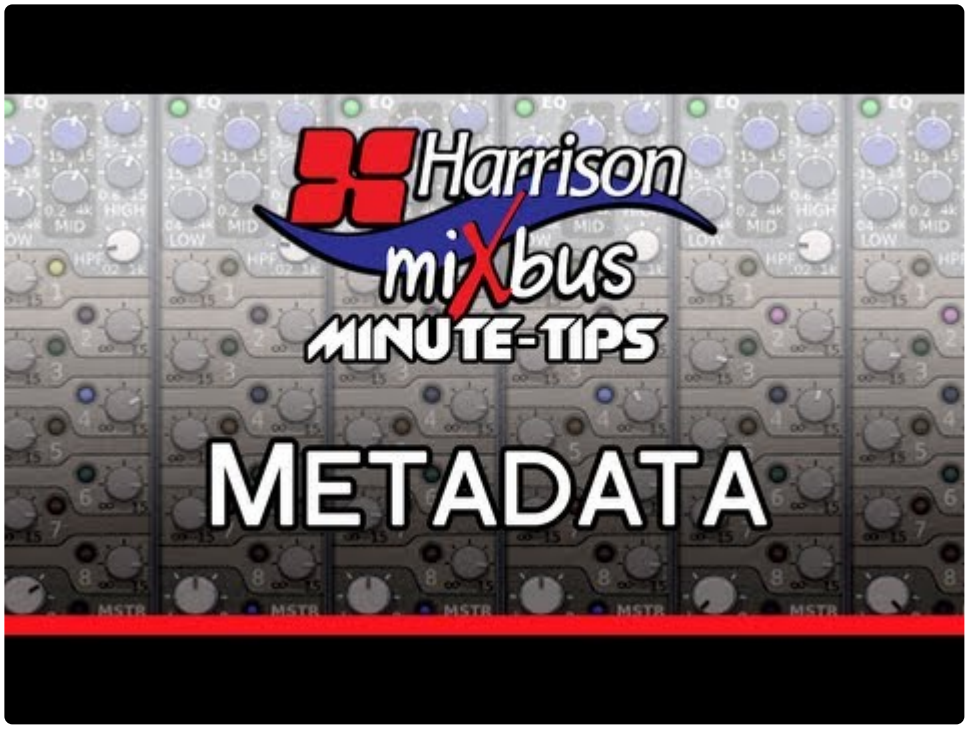
- Open Video: Will import video with audio (the audio embedded within a video can also be extracted). See Video
- Remove Video: Removes video from the timeline but not from the session folder.

- Export: More here
- Clean-up: More here

- Properties: More here

- Quit: Saves the currently opened session and closes Mixbus.

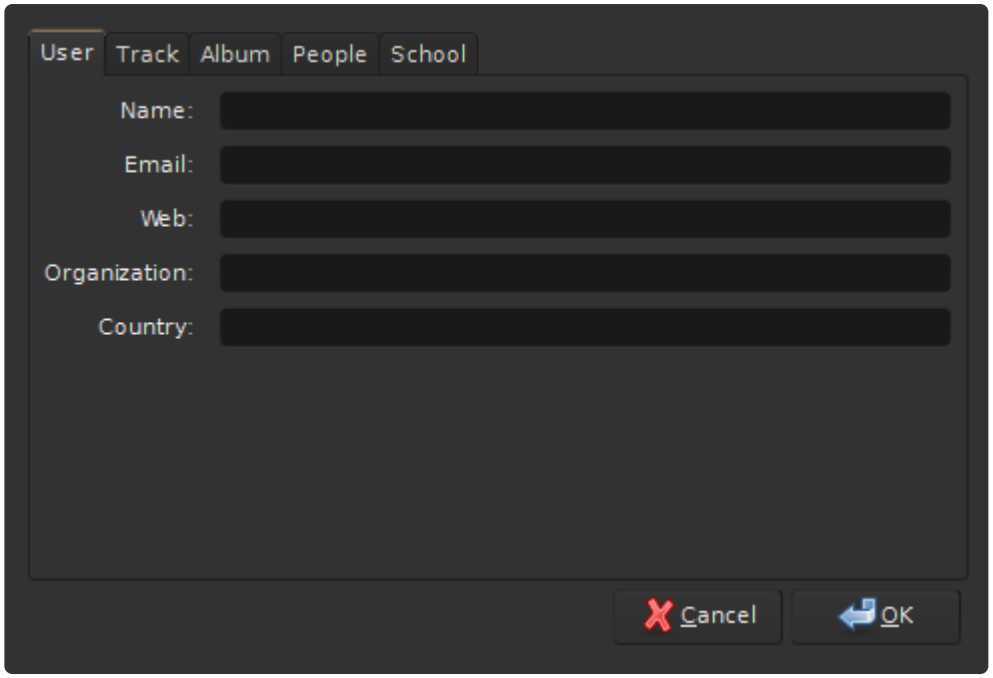
18.1.1. Metadata



https://www.youtube.com/embed/anlRLGZxjIs?rel=0

Session files include “metadata” which can include information about the project including the composer(s), engineers, etc. Metadata is stored inside the session file in a simple XML format.

The Metadata dialog is launched from the Session menu:



The User tab includes information about the primary user of Mixbus. This includes the Name, Email, Web, Organization and Country information. This is a global setting and will be applied to future sessions. This information is purely for convenience and for documentation purposes.

There are 3 specific ways that this information is used within Mixbus:

Email: automatically entered into the Freesound entry box in the Export dialog.
Organization: This text is stored in broadcast wave file metadata.
Country: This text is stored in broadcast wave file metadata.

The User information is stored outside the session file and is persistent across sessions.

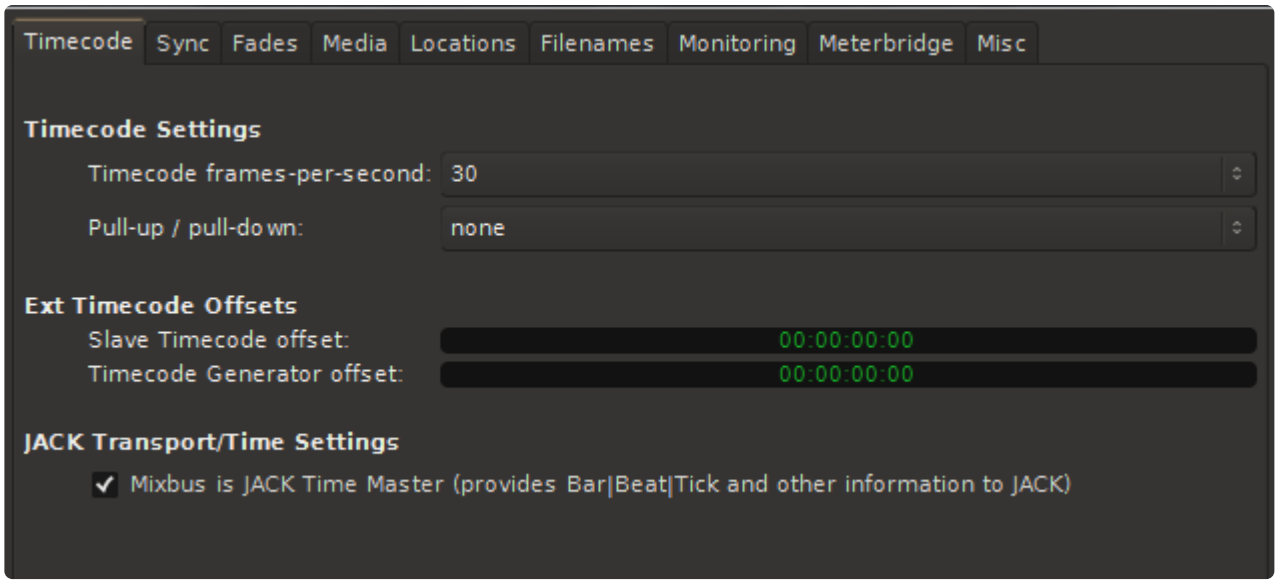
The remaining tabs (Track, Album, People, and School) record data that is stored in the session file. The data is stored in an industry-standard XML format, and can be easily read and manipulated by 3rd party applications or scripts. Contact Harrison if you are interested in further development of Mixbus metadata.

18.1.2. Session Properties

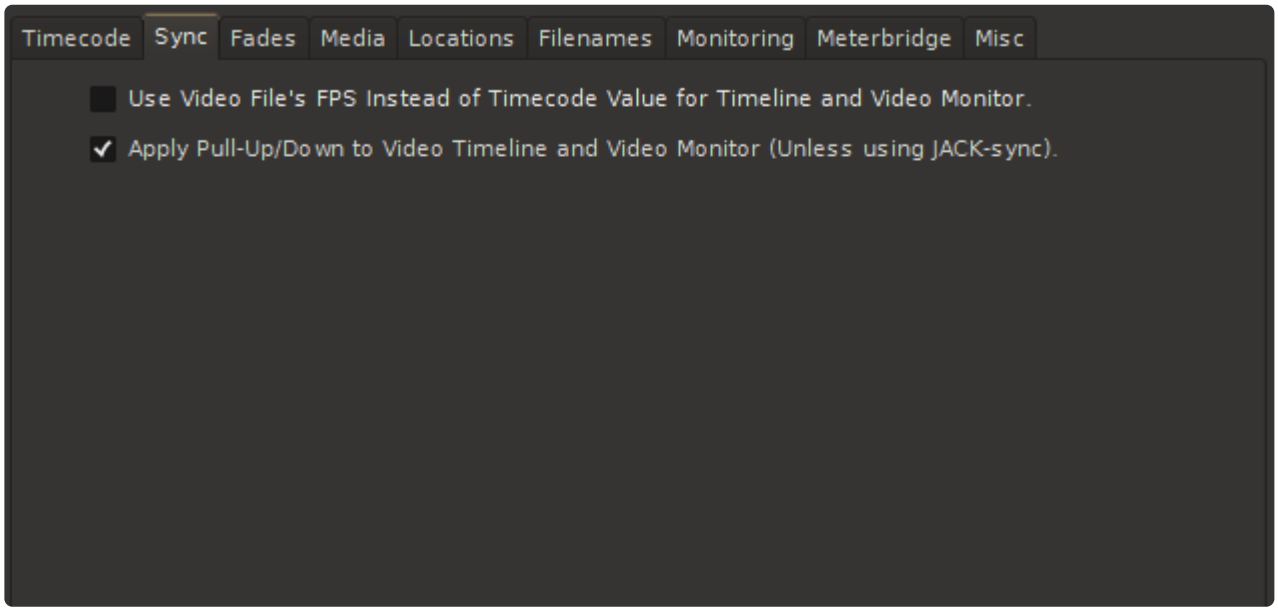
Session properties are settings for the currently loaded session. These are session-specific controls such as the timecode format, audio file format, and monitoring setup for this session.

Menu Options

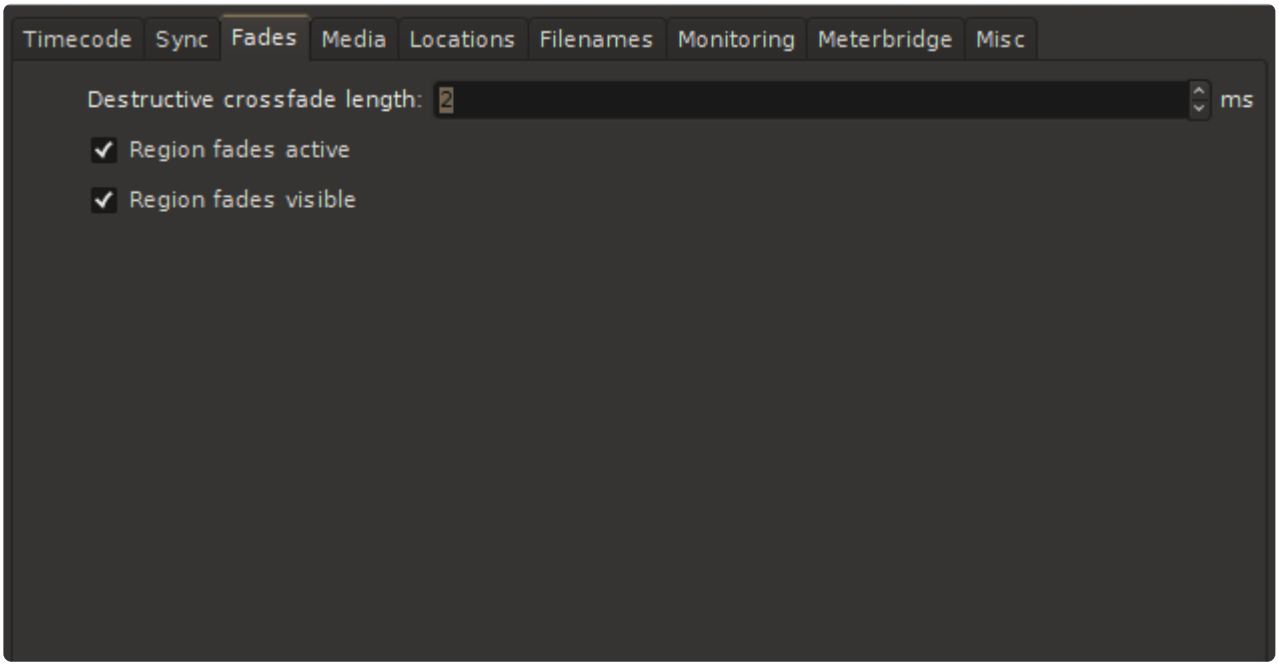
Timecode Tab



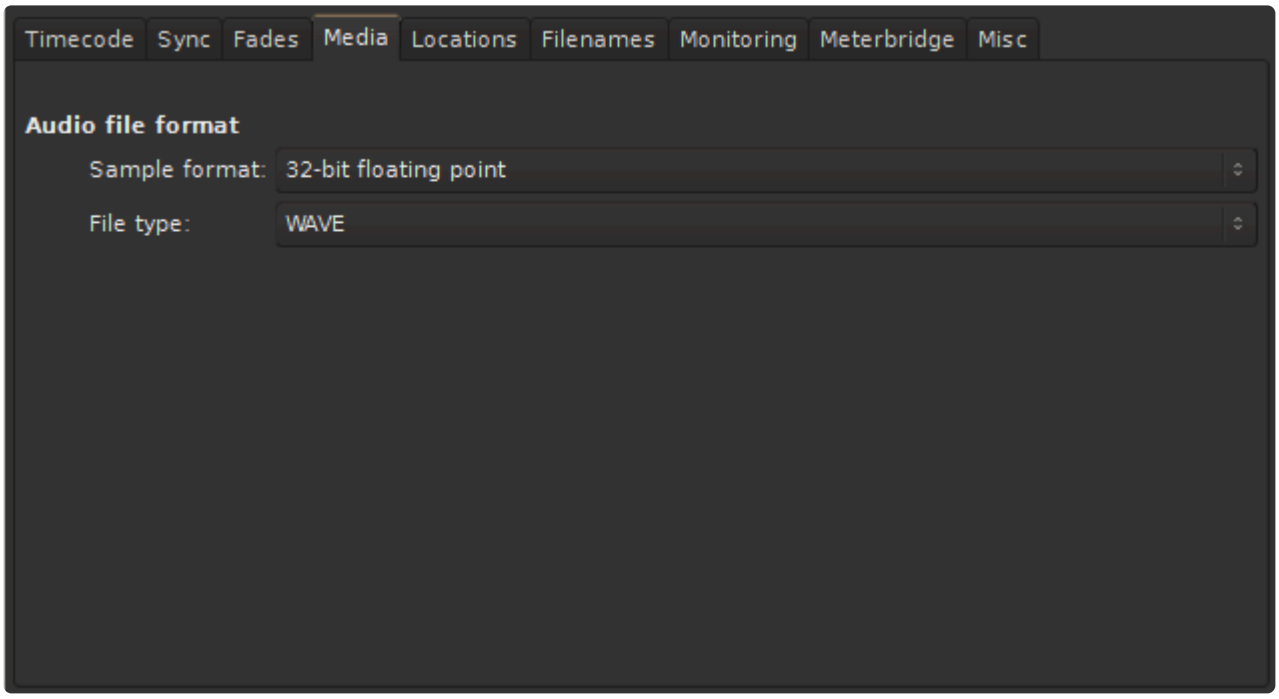
Sync Tab



Fades Tab



Media Tab



The audio data format that Mixbus uses internally. New audio recordings will be stored in this format, and files will be converted to this format when imported.

- Float (floating point 32 bit) (default value) Allows extra headroom above 0 dBFS for JACK-enabled connections and/or bounces within Mixbus.
- 24 bit: If you plan to record live sources from a A/D converter and do not use internal bouncing-to-

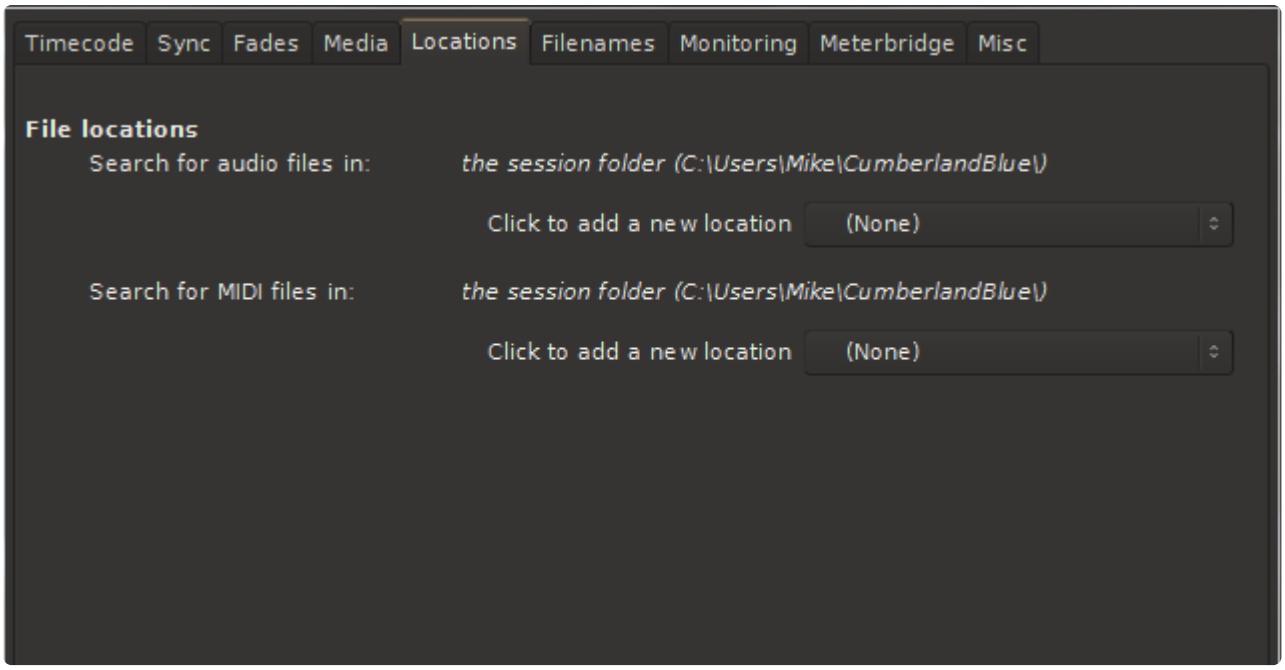
disk, then 24 bit will save some disk space.
- 16 bit: If working with audio sourced from a CD, this uses half the storage space of “float” option.

Audio File Format: File Type

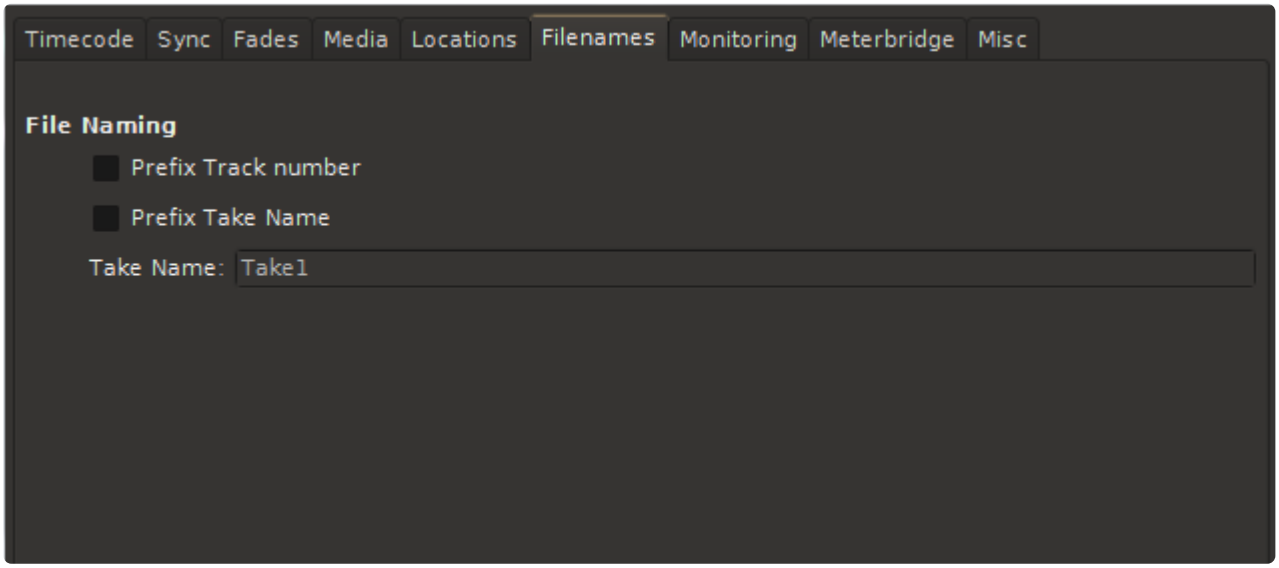
The file type that Mixbus uses internally. New audio recordings will be stored in this format, and files will be converted to this format when imported.

- Broadcast Wave: Industry standard .wav files with an additional time-stamp that may be used in some software. File size is limited to 4GB.
- Wave: Industry standard PCM file. File size is limited to 4GB.
- Wav64: Industry standard PCM file. Uses 64 bit length value in the header allowing nearly limitless recording length at any sample rate.
- CAF: Same as Wav64, but this format is only common on OS X.
- FLAC: Open source lossless codec. FLAC’s algorithm can typically be reduced to between 50 and 70 percent of its original size while still retaining full lossless quality.

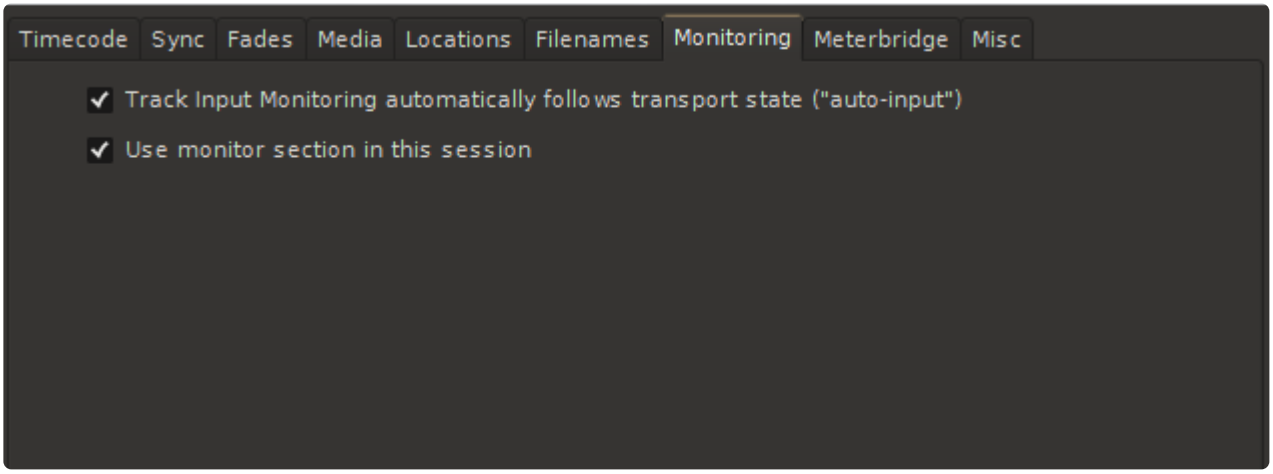
Locations Tab



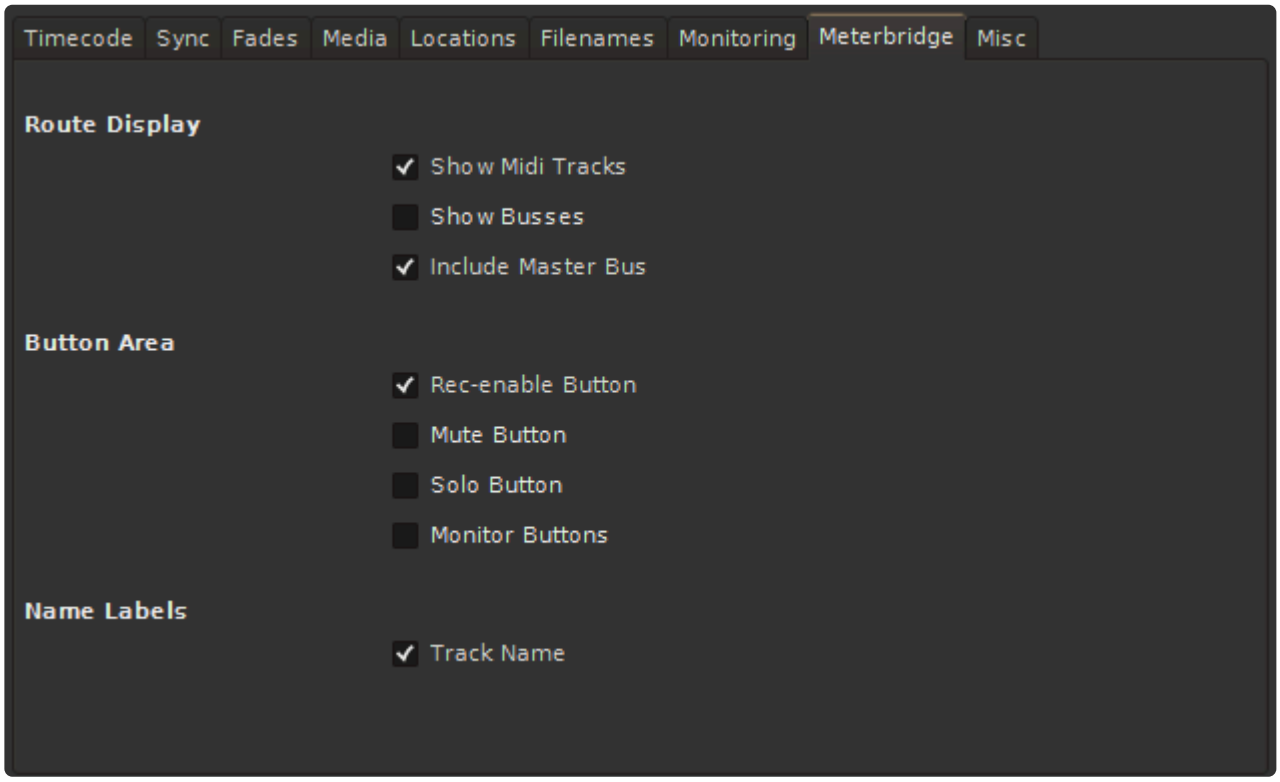
Filenames Tab



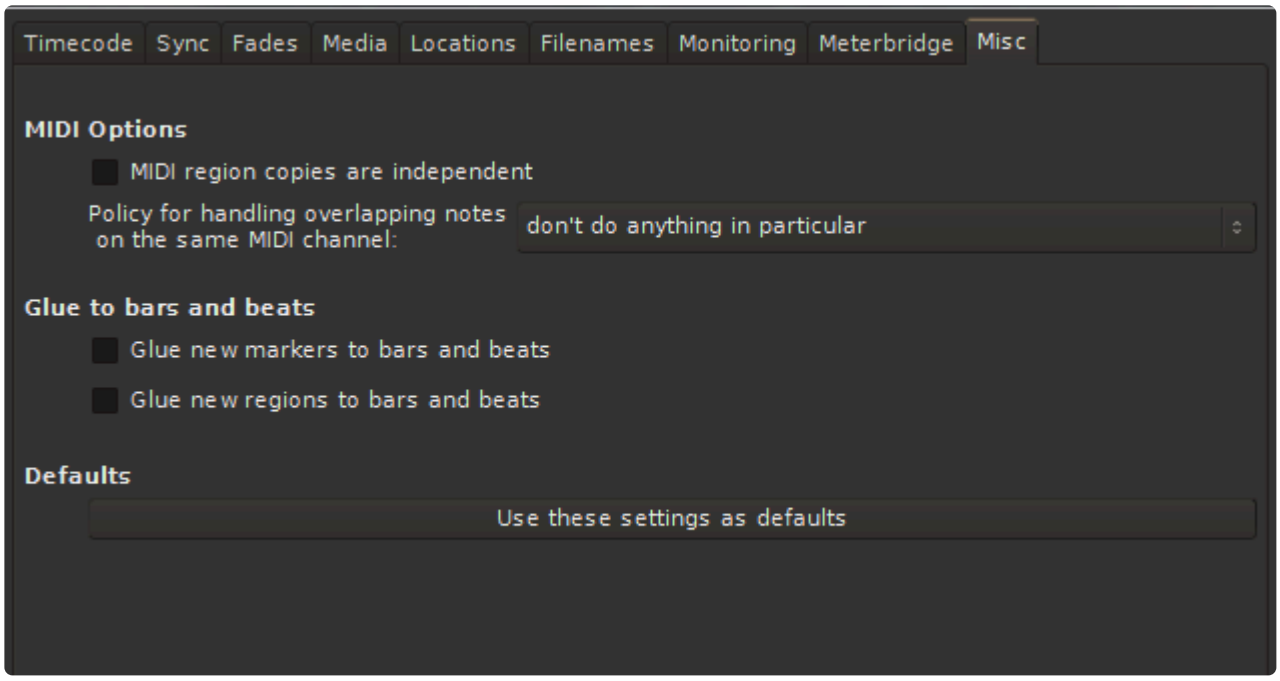
Monitoring Tab



Meterbridge Tab



Misc Tab



18.1.3. Clean-up

Bring all media into session folder:

> ! Located under Session>Bring all media into session folder. This **important function** serves 2 purposes:
> • Allows portability of sessions
> • Allows for the archiving of sessions

This function will locate all audio and MIDI data that may be scattered across multiple folders, or even multiple hard drives, and copies or moves all audio and/or MIDI associated with that session into the session folder. Once this is done the user can be assured that the session folder will contain all the necessary audio/MIDI in one convenient location.

A real world example:

While working with a session you may pull new material from many different locations on your hard drive, CD drive, or network. After saving the session, Mixbus will know where to find all of these bits of audio the next time the session is opened *on that particular computer*. But if your session file is on a portable hard drive and most (**but not all**) of your audio files are also on that portable hard drive and you take that hard drive to a different computer, or studio, expecting to open the session, you may be confronted with a dialogue box indicating the session was unable to located certain audio files. Running the “Bring all media into session folder” alleviates this problem.

The same is true for archiving of a session. If the user does not invoke the “Bring all media into session folder” function before archiving then the archive may be incomplete.

> ✻ It maybe useful to run the clean up function located under Session>Clean up in conjunction with “Bring all media into session folder”. There maybe no need to transport or archive unused material.

Clean-up unused sources:

Clean-up will move all unused files to a folder labeled “dead” within the session folder. Clean-up is a destructive operation and all undo/redo history will be lost.

Flush Waste Basket:

Deletes audio files located in the “dead” folder.

A note about Session Cleanup.

There are a few reasons why cleanup might not save much space:

1. Because all of the session snapshots are inside the session folder, Mixbus knows when a region is used by a different snapshot. So it may not “clean up” as much as you expect.
2. If you recorded a very long file, but only use a very small snippet, this reference to the file will keep the file from being cleaned up. If you “bounce” the regions (making a new, short file of exactly the length of the region) then you might see a bigger space savings. But note that this means you can no longer extend the fade handles for that region; there’s no longer any information left outside the region.

18.2. Transport Menu

- Start/Stop
- Play >
    - Play Selection
    - Play w/Preroll
    - Start/Continue/Stop
    - Play from Edit Point and Return
    - Play Loop Range
- Start Recording
- Stop and Forget Capture: this stops the current recording, and permanently deletes any audio files that were recorded in the current pass. This cannot be undone.
- Enable Record
- Record w/Preroll: see here for more details

- Set Loop from Selection
- Set Punch from Selection
- Set Session Start/End from Selection

- Forward
- Rewind
- Transition To Roll
- Transition To Reverse

- Playhead >
    - Playhead to Mouse
    - Playhead to Active Mark
    - Center Playhead
    - Nudge Playhead Forward
    - Nudge Playhead Backward
    - Move to Next Transient
    - Move to Previous Transient
    - Playhead to Next Grid
    - Playhead to Previous Grid
    - Playhead to Next Region Boundary
    - Playhead to Previous Region Boundary
    - Playhead to Next Region Boundary (No Track Selection)
    - Playhead to Previous Region Boundary (No Track Selection)
    - Playhead to Next Region Sync
    - Playhead to Previous Region Sync
    - Jump to Next Mark
    - Jump to Previous Mark
    - Goto Zero
    - Goto Start

    - Goto End
    - Goto Wall Clock

- Active Mark >
    - To Next Region Boundary
    - To Previous Boundary
    - To Next Region Sync
    - To Previous Region Sync

- Markers >
    - Add Mark from Playhead
    - Remove Mark at Playhead
    - Toggle Mark at Playhead
    - Locate to Mark 1
    - Locate to Mark 2
    - Locate to Mark 3
    - Locate to Mark 4
    - Locate to Mark 5
    - Locate to Mark 6
    - Locate to Mark 7
    - Locate to Mark 8
    - Locate to Mark 9
    - Set Session Start from Playhead
    - Set Session End from Playhead

- Time Master: (for JACK sync only) : when this is enabled, Mixbus will serve as master to report the Bar|Beat|Tick information to the JACK clock
- Punch In/Out
- Punch In
- Punch Out
- Auto Input : Learn more in the Monitoring section.
- Follow Range : Learn more in the transport section.
- Auto Play : The transport will automatically play whenever you locate the playhead to a new position.
- Auto Return: Returns playhead to last starting position, when the transport is stopped.
- Click: Toggles the Click (Metronome) sound on & off.
- Follow Playhead : The edit window will scroll to follow the playhead when the playhead reaches the edge of the editing canvas.
- Stationary Playhead : If enabled, AND Follow Playhead is enabled, then the timeline will scroll behind the playhead. This requires significant CPU and disk resources, so it is not recommended for large sessions.
- Panic: Sends MIDI note off to all channels. Details here

18.3. Edit Menu

- Undo
- Redo
- Undo Selection Change
- Redo Selection Change
- Cut
- Copy
- Paste

- Select >
    - Select All Objects
    - Select All Visible Lanes
    - Select All Tracks
    - Deselect All
    - Invert Selection
    - Set Range to Loop Range
    - Set Range to Punch Range
    - Set Range to Selected Regions
    - Select All After Edit Point
    - Select All Before Edit Point
    - Select All Overlapping Edit Range
    - Select All Inside Edit Range
    - Select All in Punch Range
    - Select All in Loop Range
    - Move Range Start to Previous Region Boundary
    - Move Range Start to Next Region Boundary
    - Move Range End to Previous Region Boundary
    - Move Range End to Next Region Boundary
    - Start Range
    - Finish Range
    - Select Next Track or Bus
    - Select Previous Track or Bus

- Delete

- Crop: This splits regions under the selected range, and then removes all regions outside of the selected range.

- Split/Separate : If a “range” selection exists, the regions under the range will be split at the start and end of the range. If no selected range exists, then the regions will be split at the edit point. Learn more about the Edit Point here.

- Separate >

    - Separate Under
    - Separate Using Loop Range
    - Separate Using Punch Range

- Align >
    - Align Start
    - Align Start Relative
    - Align End
    - Align End Relative
    - Align Sync
    - Align Sync Relative

- Fade >
    - Fade Range Selection
    - Set Fade In Length
    - Fade In
    - Set Fade Out Length
    - Fade Out

- Remove Last Capture

- Edit Point >
    - Change Edit Point
    - Change Edit Point Including Marker

- Snap Mode >
    - No Grid
    - Grid
    - Magnetic
    - Next Snap Mode
    - Next Snap Choice
    - Previous Snap Choice
    - Next Musical Snap Choice
    - Previous Musical Snap Choice

- Snap to >
    - Snap to CD Frame
    - Snap to Timecode Frame
    - Snap to Timecode Seconds
    - Snap to Timecode Minutes
    - Snap to Seconds
    - Snap to Minutes
    - Snap to One Twenty Eighths
    - Snap to Sixty Fourths
    - Snap to Thirty Seconds

    - Snap to Twenty Eighths
    - Snap to Twenty Fourths
    - Snap to Twentieths
    - Snap to Sixteenths
    - Snap to Fourteenths
    - Snap to Twelfths
    - Snap to Tenths
    - Snap to Eighths
    - Snap to Sevenths
    - Snap to Sixths
    - Snap to Fifths
    - Snap to Quarters
    - Snap to Thirds
    - Snap to Halves
    - Snap to Beat
    - Snap to Bar
    - Snap to Mark
    - Snap to Region Start
    - Snap to Region End
    - Snap to Region Sync
    - Snap to Region Boundary

- Tempo
    - Set Tempo from Region = Bar
    - Set Tempo from Edit Range = Bar

- Smart Object Mode: More here

- Lua Scripts
    - Script Manager: see Scripting
    - (presets configurable from Script Manager)

- Preferences: More here

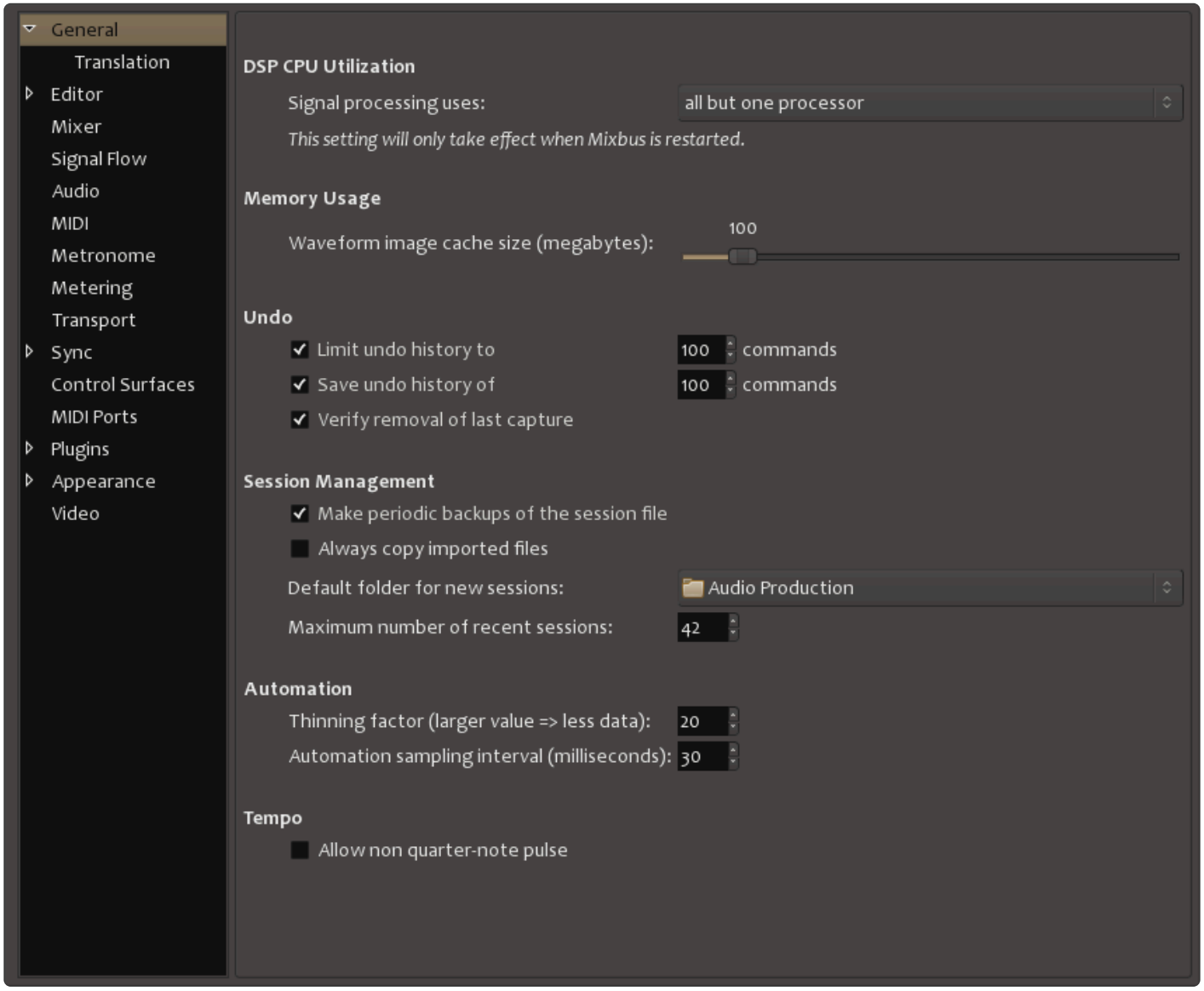
18.3.1. Preferences Window

The Preferences window can be found under the menu item Mixbus->Preferences in OS X, and under the menu item Edit->Preferences in Windows and Linux.

> The Preferences window defines your global preferences, that apply to all of the sessions on your machine. Global preferences define the way you like to work, and the way your system is set up (such as your MIDI control surfaces or timecode sources). If you want to change settings for a particular session, go to Session Properties.

The preferences window is organized by category in the left hand column of the window. Some categories have subcategories that you can navigate by clicking on them:

General



DSP CPU Utilization

Signal processing uses

This option defines how many “threads” Mixbus will create to process audio signals. If you use a lot of tracks and plug-ins, then Mixbus will require significant CPU resources. This setting allows Mixbus to use multiple processors in parallel, thereby taking advantage of modern computers with 2, 4, or more CPU cores.

- **all but one processor** is the default setting. This setting reserves one processor so your computer does not feel sluggish when Mixbus is heavily loading your CPU.
- **all processors** might allow Mixbus to run more tracks/plug-ins with the risk of causing other desktop applications feeling sluggish.

Memory Usage

Waveform image cache size (megabytes)

This slider allows you to set the size, in Megabytes, of the waveform image cache. For large projects with a lot of tracks, setting this higher may improve screen rendering times to improve performance. For smaller projects, or for systems with smaller amounts of RAM, the cache size can be reduced.

Undo

Limit Undo History

This option should be enabled to limit the memory and disk space that Mixbus will use to store Undo information. Select a number of operations that you'd like Mixbus to remember within a given session.

Save Undo History

This option specifies how many commands are available to undo after a session is saved and re-opened. Previous commands from prior work on the session will still be available to undo even when a session is first re-opened.

Verify removal of last capture

When this option is enabled, Mixbus will display a confirmation message when the "stop and remove last capture" action is used.

Session Management

Make periodic backups of the session files

If this is enabled, Mixbus occasionally creates a date-stamped snapshot in the session's "backup" folder.

> ✻ Regardless of the "periodic backup" setting, Mixbus stores a "pending" file whenever you start a new recording. This makes it very likely that, even in the case of a crash or power outage, a recording in process can be recovered.

Always Copy Imported Files

If this is enabled, then (a) the "copy to session" checkbox will be automatically selected, and cannot be changed. (b) Files that are drag & dropped into the session will be copied into the session, rather than "referenced" from their existing location, wherever that may be (such as on a removable hard drive).

Default Folder for new Sessions

This option specifies the default location that appears in the New Session dialog pulldown.

Maximum Number of Recent Sessions

This option defines how many sessions will be displayed in the New Session dialog list.

Automation

Thinning factor

This option removes data points within the automation draw points. Excessive draw data may reduce system responsiveness with little to no audio benefit. Larger numbers means more thinning, and therefore fewer draw points but more responsiveness.

Automation sampling interval

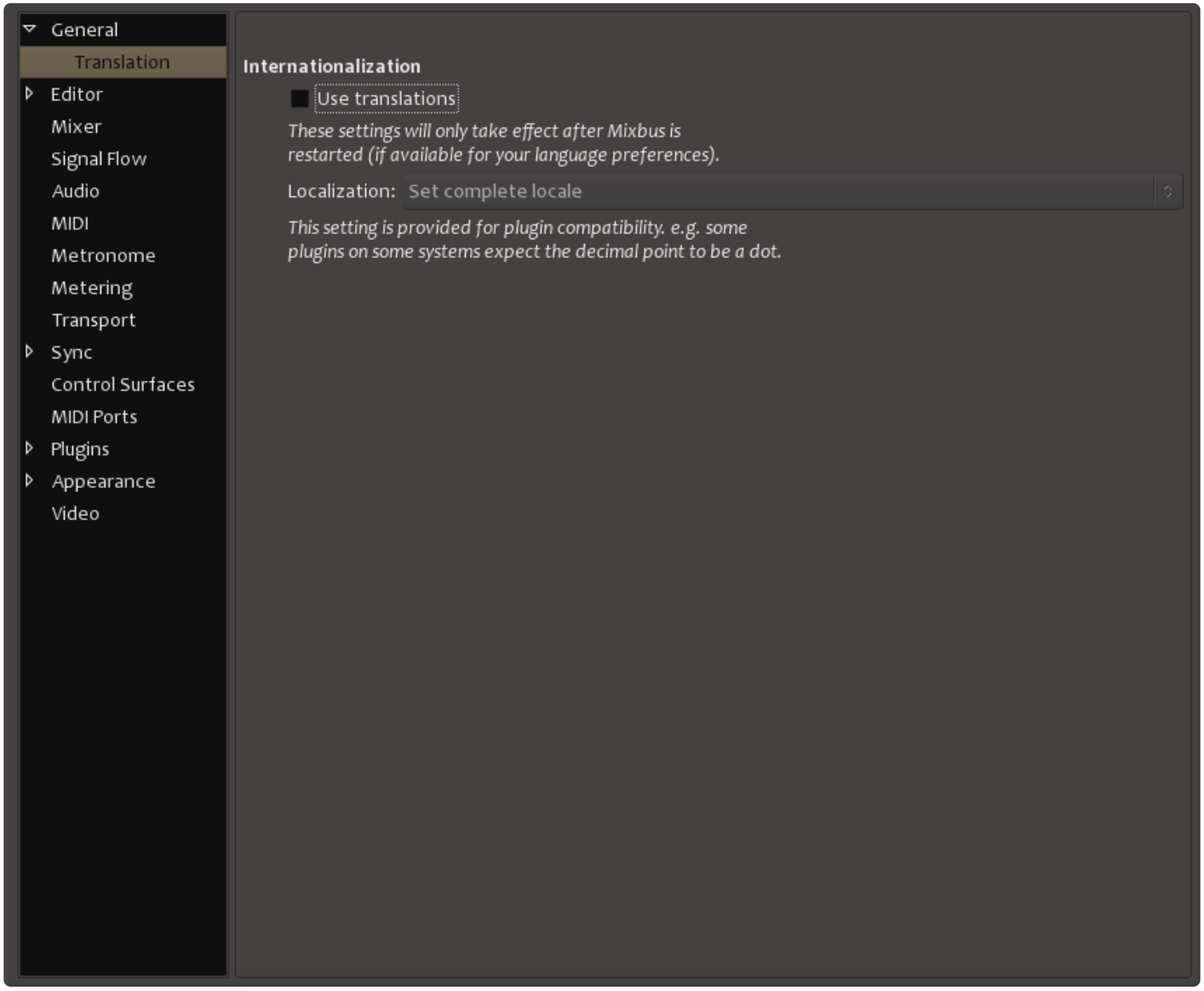
This option determines the rate (in milliseconds) at which automation points are acquired. If the density of automation points is too high system responsiveness may be reduced.

Tempo

Allow non-quarter note pulse:

Normally, Mixbus assumes that the “tempo” is based on quarter-note pulse. Disable this option to use a non-quarter note pulse. For example a time-signature of 5/8 will use the eight-note pulse to meet the tempo.

General > Translation



Internationalization

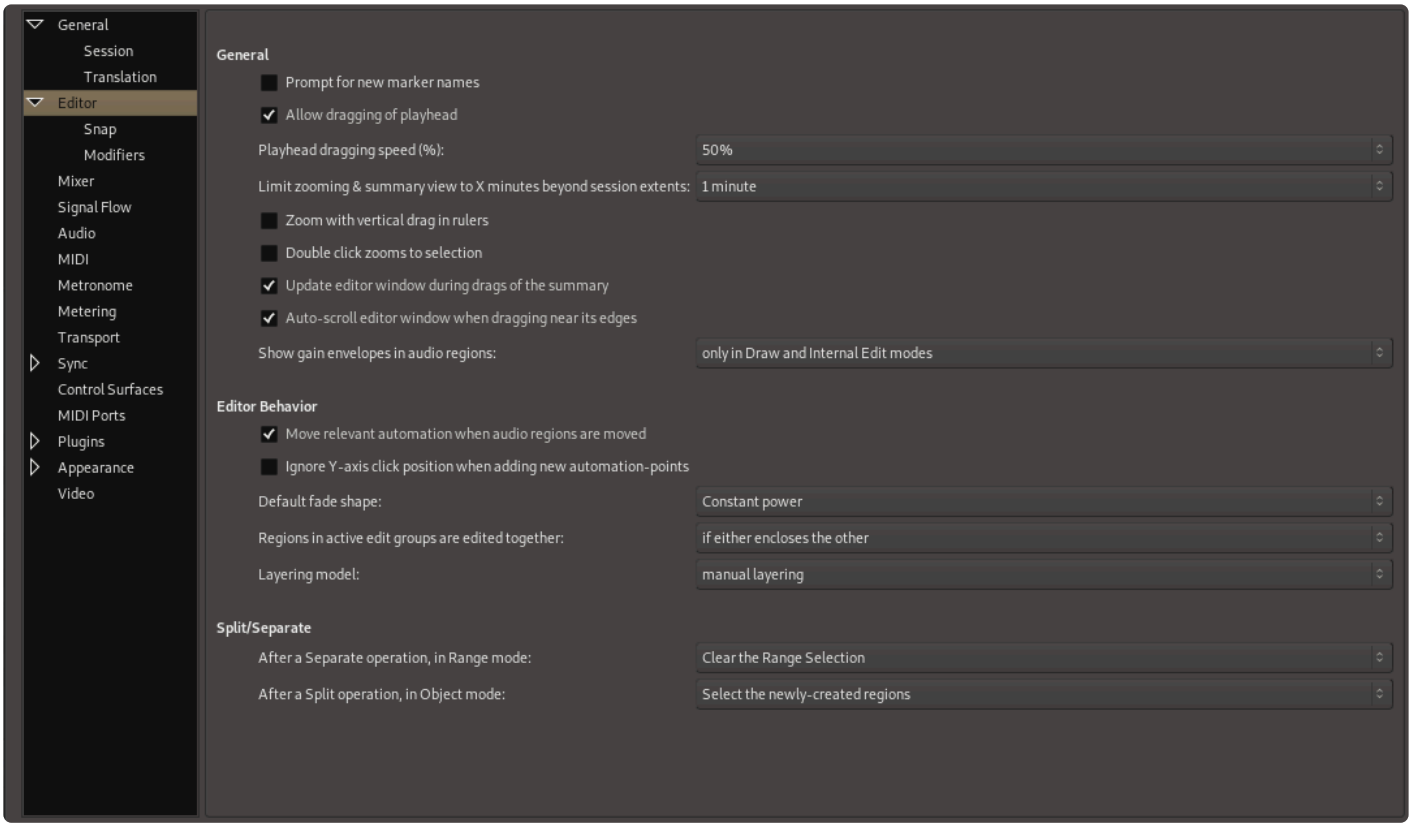
Use Translations

This option allows Mixbus to accommodate multiple languages and conventions, such as a decimal point being . in some languages and , in others.

Localization

This drop-down menu allows you to **set complete locale**, **enable only message translation**, or **message translation and numeric format**. This setting is provided for plugin compatibility. Some plugins on some systems expect the decimal point to be a dot.

Editor



Snap:

General Snap Options:

- Snap Rubberband to grid
  In the editor, when doing a “rubber-band select” (drawing a box to select multiple regions or control points), the box edges will snap to the grid.

Prompt for new marker names

When this option is enabled, creating a new marker will prompt for a name immediately. This allows you to organize your markers from the moment you create them, but can also slow down the workflow if you are adding several markers in real time. If you’d prefer to rename the markers later, de-select this option. You can always rename markers later by right-clicking on them.

Allow Dragging of Playhead

(default: ON) Allows clicking on the red playhead to drag it. Disable this if you want your clicks to bypass the playhead and click on the thing behind it.

Zoom with vertical drag in rulers

When this option is enabled, you can click and drag vertically in the ruler of the Editor to zoom in & out.

Dragging up zooms out, dragging down zooms in.

Double click zooms to selection

Enabling this option allows you to quickly zoom to whatever regions you have selected by simply double-clicking on one of the selected regions.

Update editor window during drags of the summary

By default, when dragging the “summary” view, the editor will be updated live. If your system is unable to redraw the editor window quickly enough, you can disable this option.

Auto-scroll editor window when dragging near its edges

When enabled: if you drag a region close to the edge of the editor, it will begin scrolling to accommodate the move.

Show gain envelopes in audio regions

This option allows you to specify whether gain envelopes in audio regions are shown only in Draw and Internal Edit modes only, or in all modes.

Editor Behavior

Move relevant automation when regions are moved

(default: ON) When you drag a region, this will also move any automation points that are within the region’s bounds. NOTE: this will overwrite any existing automation in the destination location.

Default fade shape

This option allows you to specify the default shape for crossfades, between Linear, Constant power, Symmetric, Slow, or Fast.

Regions in Edit Groups are edited together

This option allows you to specify the behavior of grouped regions in the editor window. The grouped regions can be edited together either only if they have identical length, position and origin (indicating that they are likely from the same source material and have been edited in sync), or whenever they overlap in time (if a region has any overlap with the currently-selected region, then it will move with the region).

Layering Model

This option allows you to specify the layering behavior of regions. The default is **Manual Layering**, where the user is expected to manually control each region’s layer. Or, **Later-is-higher** defines region layering so that later regions will be assigned a higher layer.

After splitting selected regions, select

This option allows you to specify which regions, if any, are selected after a split is performed. You can choose from no regions selected, newly-created regions selected, or existing regions and newly-created regions selected.

Split/Seperate

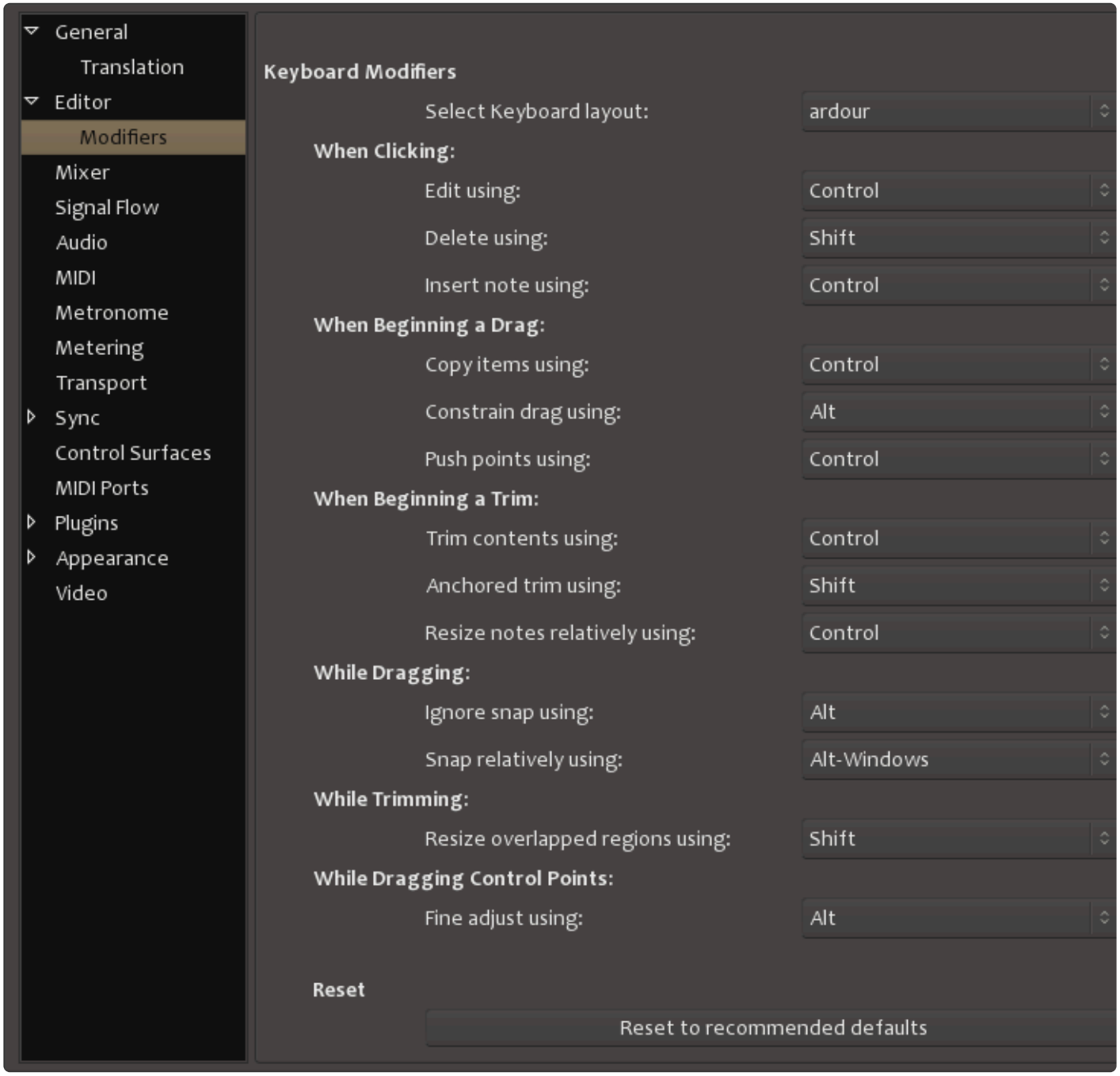
In Range Mode

Controls what happens to regions after split in Range Mode

In Object Mode

Controls what happens to regions after split in Object Mode

Editor > Modifiers



Keyboard Modifiers

This section allows you to change the keyboard modifiers **Shift**, **Command**, **Control**, **Alt**, as well as the lesser-used **Windows** keys, and combinations thereof, when using the mouse to click, drag, and trim in the Editor window. Using these keyboard modifiers you can adapt Mixbus to the editing workflow you prefer. Several parameters are offered:

Select Keyboard Layout

Currently Mixbus provides a single set of predefined keyboard shortcuts. If we later provide additional keybindings, they will appear here.

When Clicking

You can change which modifier(s) activate **Edit**, **Delete**, and **Insert Note**.

When Beginning a Drag

You can change which modifier(s) activate **Copy items**, **Constrain drag**, and **Push points**.

When Beginning a Trim

You can change which modifier(s) activate **Trim contents**, **Anchored trim**, and **Resize notes relatively**.

While Dragging

You can change which modifier(s) activate **Ignore snap** and **Snap relatively**.

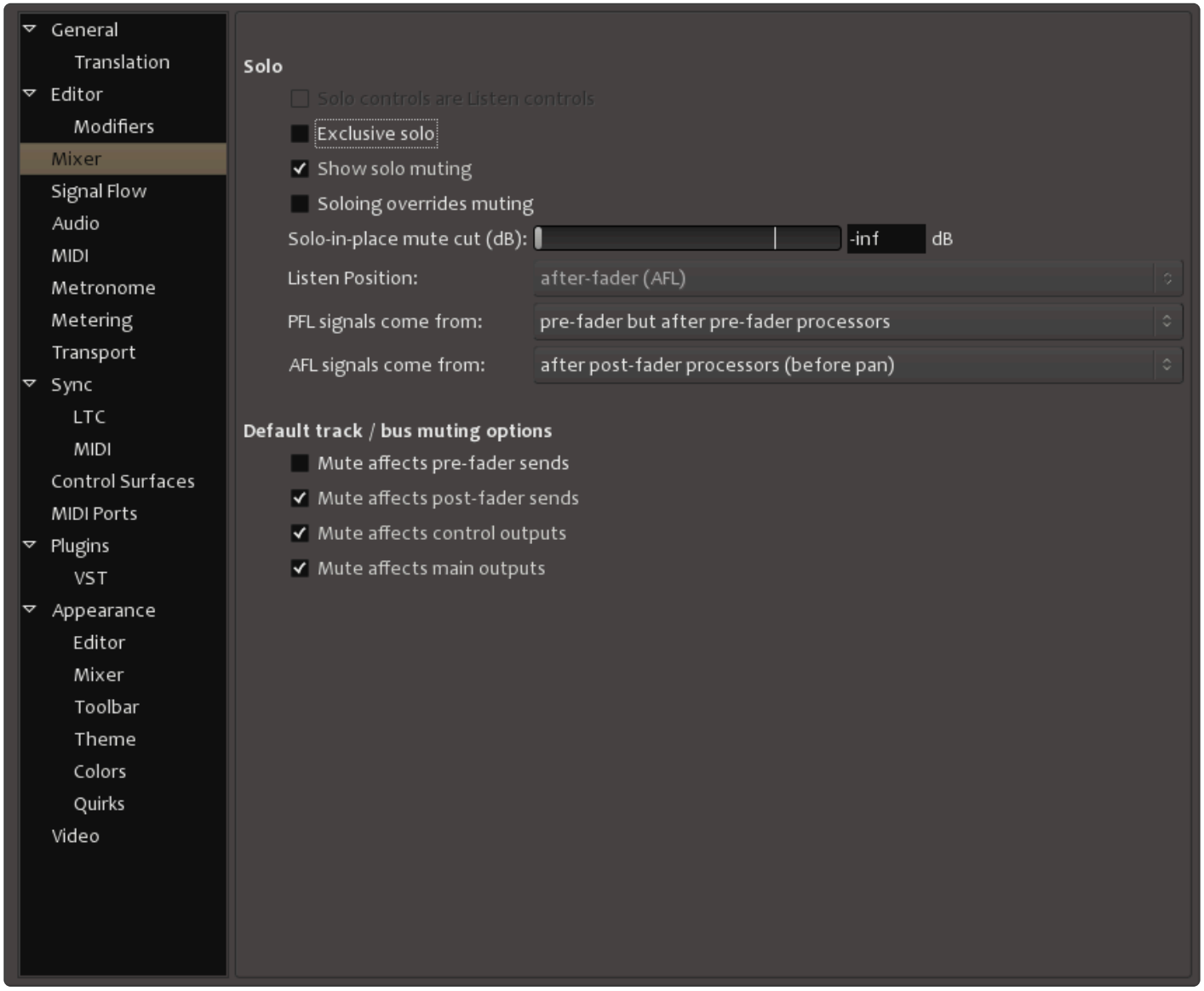
While Trimming

You can change which modifier(s) activate **Resize overlapped regions**

While Dragging Control Points

You can change which modifier(s) activate **Fine adjust**.

Mixer



Solo

Solo controls are Listen controls

When this is enabled, Solos will appear on the “monitor” output, but they do not mute the master bus output. This allows the engineer to listen to a solo, without affecting the master bus output in those cases where the master bus is being recorded “live” and should not be interrupted.

Exclusive Solo

When this option is enabled, only one track or bus can be in solo at any given moment. If another track or bus is selected, then it will become the only track or bus that is in solo mode.

Show solo muting

When this option is enabled, soloing one track or bus causes a yellow outline to be drawn around all the other tracks or busses which are muted by the solo selection. Every track or bus that is not solo'ed will be effected in this way, to remind you why the track is not audible.

Soloing overrides muting

When this option is enabled, pressing the solo button will allow the audio on that track or bus to be heard, even if the mute button is also pressed. When this option is not enabled, then the mute button still continues to function, even if the track is solo'ed.

Solo-in-place mute cut (dB):

Listen Position

PFL signals come from

AFL signals come from

Default Track /bus muting options

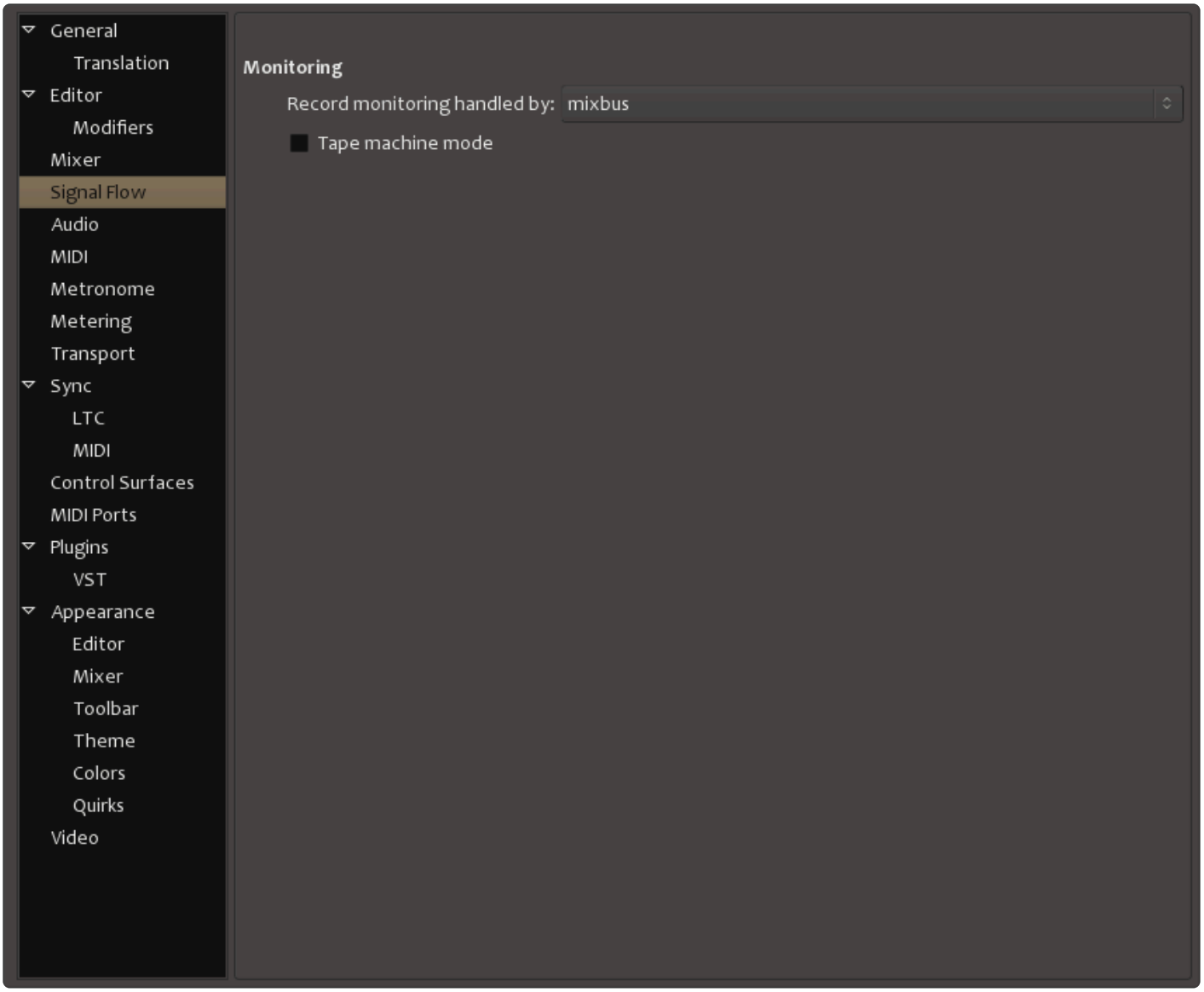
Mute affects pre-fader sends

Mute affects post-fader sends

Mute affects control outputs

Mute affects main outputs

Signal Flow



Monitoring

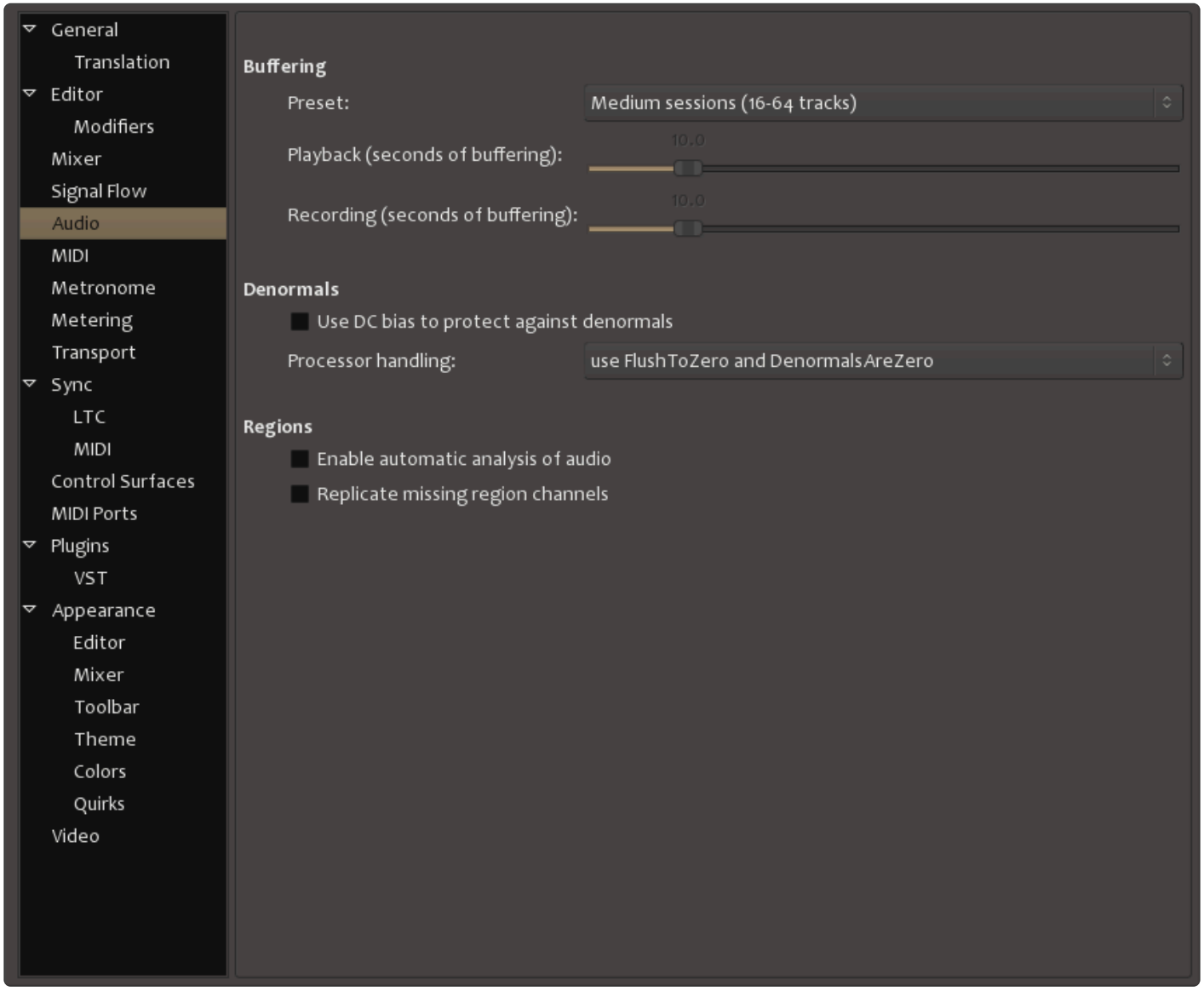
Recording monitoring handled by

See Monitoring

Tape machine mode

See Monitoring

Audio



Buffering

A smaller buffer allows mixbus to load sessions and locate faster. If you receive a message like “your disk was not able to keep up with Mixbus”, then you should increase the buffer size. If you have fewer tracks (8-16), and/or a very fast disk (SSD), then you can use very low buffer sizes, and Mixbus will feel extremely responsive. Generally speaking, Mac requires larger buffers than Windows and Linux.

Preset

There are four preset options: **small sessions**, **medium sessions**, **large sessions**, and **custom**, which allows you to set your own buffer times using the sliders for playback and recording.

Playback (seconds of buffering)

Increase this value if Mixbus reports “your disk was not able to keep up with playback”. Increasing the buffer might cause a slight delay when Mixbus locates, because it is buffering more audio before playback can start.

Recording (seconds of buffering)

Increase this value if Mixbus reports “your disk was not able to keep up with recording”.

Denormals

Denormals relate to 32 bit floating point processing used by Mixbus. Some plugins, when used in Mixbus, generate “denormals” which is an invalid processor state that causes random high CPU spikes. If you believe a plugin is generating denormals, you can enable this checkbox for the track, and see if it helps the problem. For more on denormals click here: http://en.wikipedia.org/wiki/Denormal_number

Use DC bias to protect against denormals

This option adds a miniscule amount of DC offset to prevent the signal from reaching denormal levels. In a few cases, plugins that cause CPU spikes can be improved by enabling this option, or one of the other “denormal” options

Processor Handling

This option attempts to use internal CPU settings to avoid denormals. In a few cases, plugins that cause CPU spikes can be improved by enabling this option, or one of the other “denormal” options.

Regions

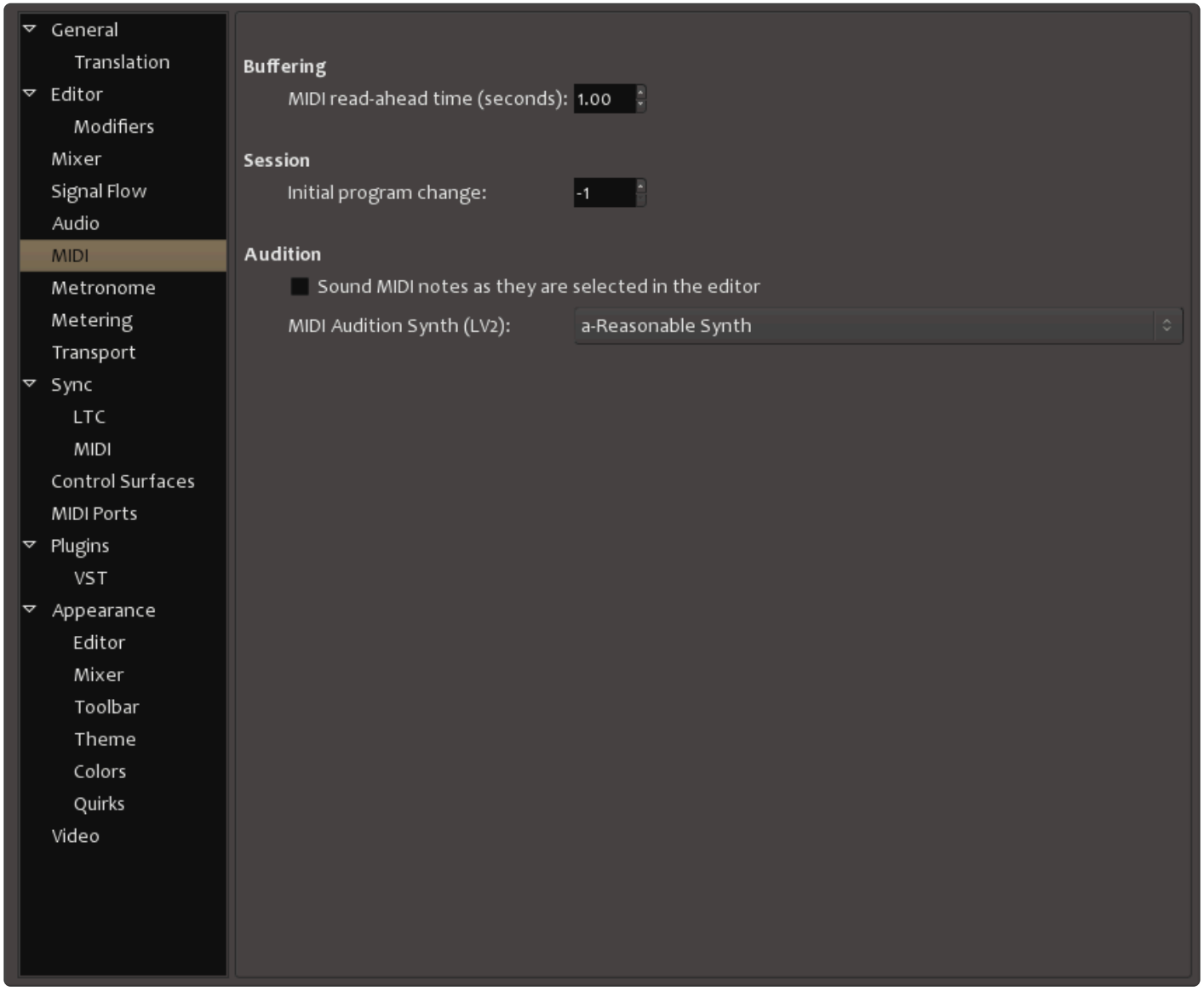
Enable automatic analysis of audio

If this is enabled, Mixbus will always analyze audio (to find transients) when files are imported are recording. This might cause imports to take longer, but it saves time when you try to “skip to next transient”

Replicate missing region channels

If you send a mono region on a stereo channel, this option will cause the mono signal to appear on both left and right channels. If this option is disabled, then the mono region will only be heard from the left side of the channel.

MIDI



Buffering

MIDI read-ahead time (seconds)

Increase this value if you have very dense & complicated MIDI arrangements.

Session

Initial program change

Mixbus will send this MIDI program-change to the MMC port, when a new session is opened.

Audition

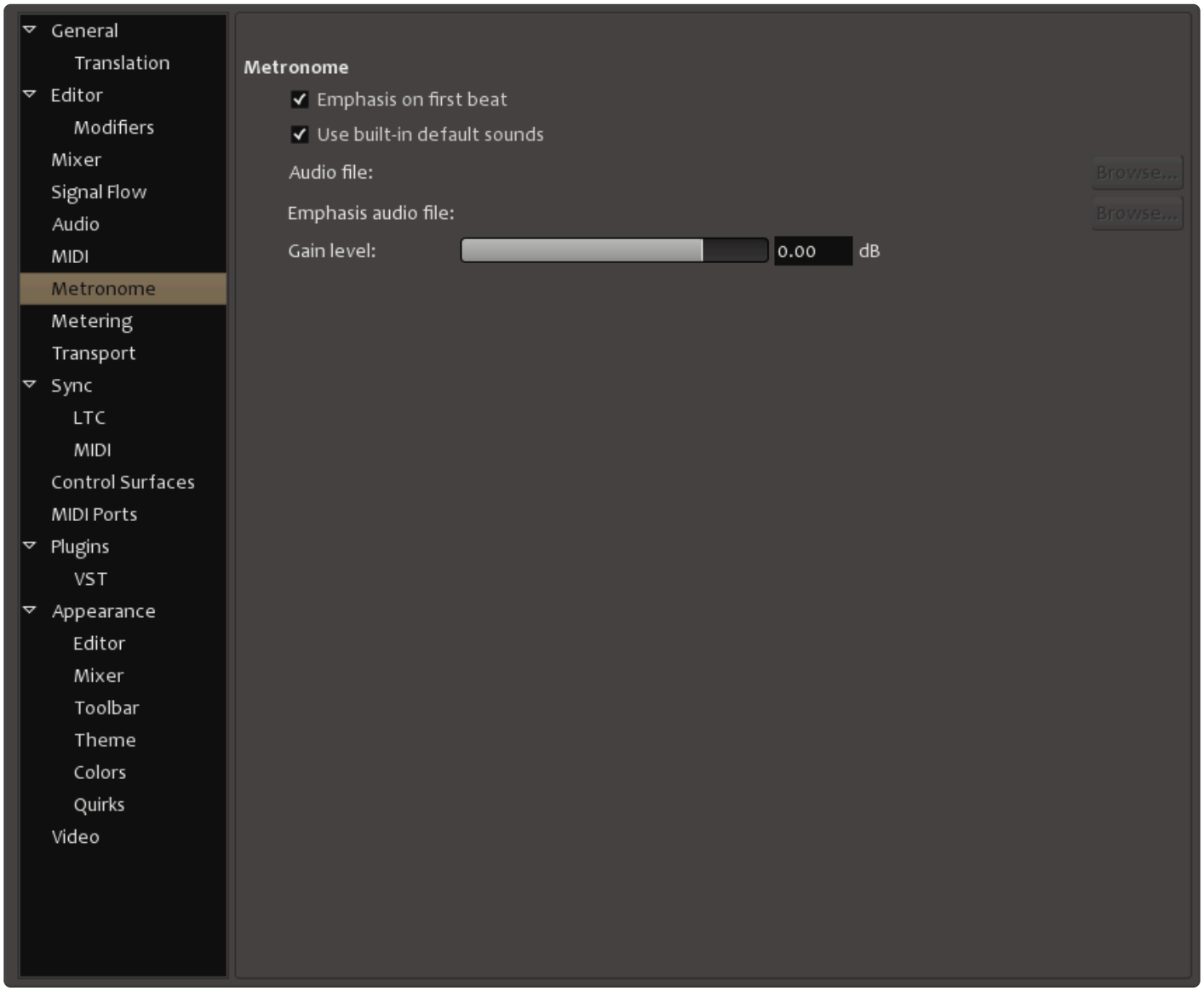
Sound MIDI notes as they are selected in the editor

When this option is enabled, clicking on a MIDI note in the editor will cause it to play.

MIDI Audition Synth (LV2)

When a midi file is auditioned (in the Import dialog, for example) … this defines the virtual instrument that is used for the audition path.

Metronome



Metronome

Emphasis on first beat

When enabled, this option uses a different tone, with a higher pitch, on the first beat of each measure to make it easier to hear where each bar begins. When disabled, the same tone is used on each beat.

Use built-in default sounds

Enables a simple option to automatically generate a useable click track with the metronome using built-in default sounds. If you prefer to use your own sounds, de-select this option and load your sounds below.

Audio File

If the default sounds are disabled, browse to define the sound file for the “click” used by the metronome.

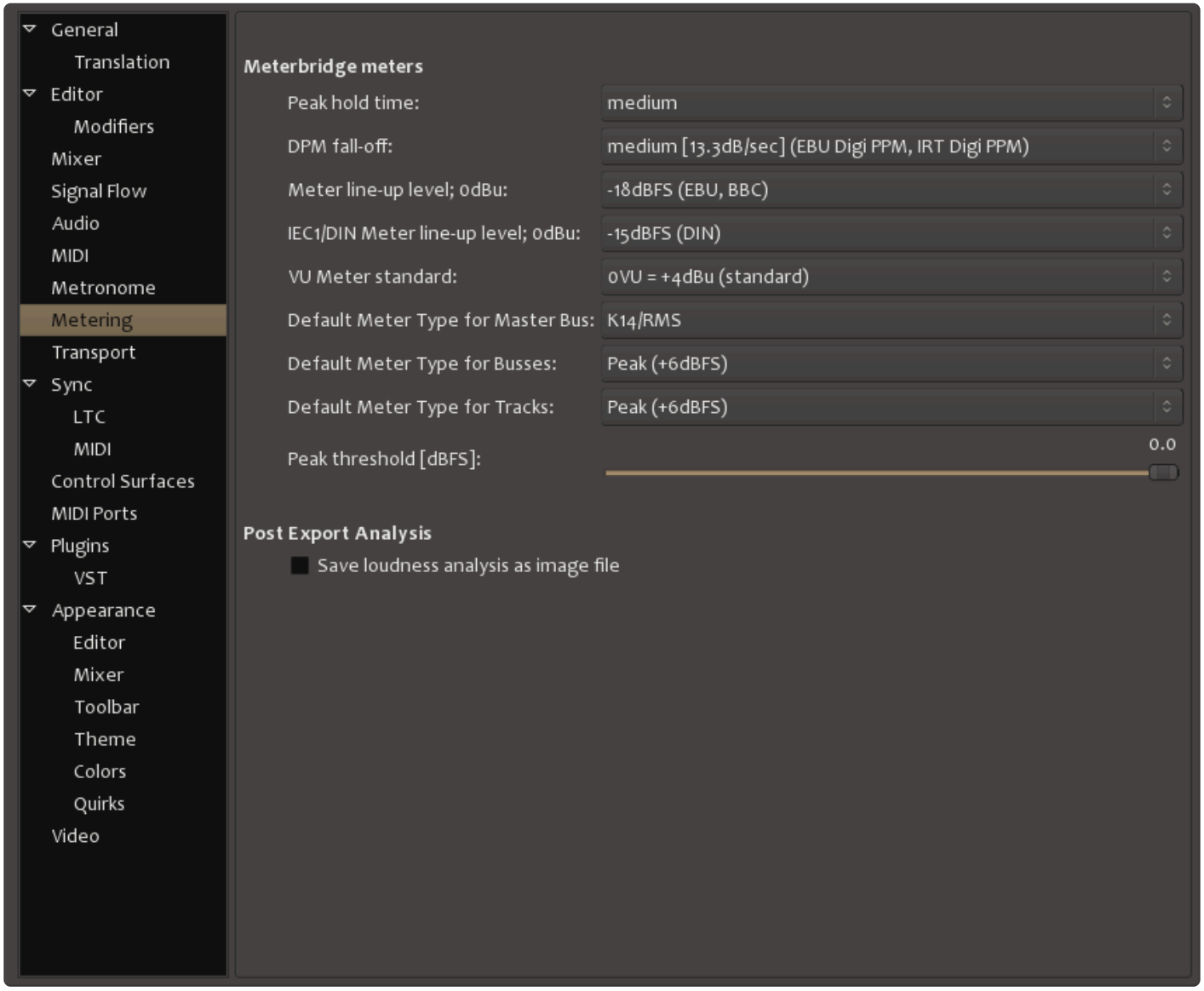
Emphasis audio file

If the default sounds are disabled, browse to define the sound file for the “emphasis click” used by the metronome. The emphasis click occurs on the first beat of each measure.

Gain level

Adjustable from – infinity to +6 dB

Metering



Meterbridge meters (NOTE: these options only apply to the meters in the Meterbridge window)

Peak Hold Time

The hold time for the peak-hold indicators. Ranges from **off** to **short**, **medium**, and **long**.

DPM fall-off

The decay rate of audio meters, ranging from Off (instant decay) to Slowest (6.6dB/sec decay rate).

Meter line-up level; 0dBu

IEC1/DIN Meter line-up level; 0dBu

VU Meter standard

Default Meter Type for Master Bus

Default Meter Type for Busses

Default Meter Type for Tracks

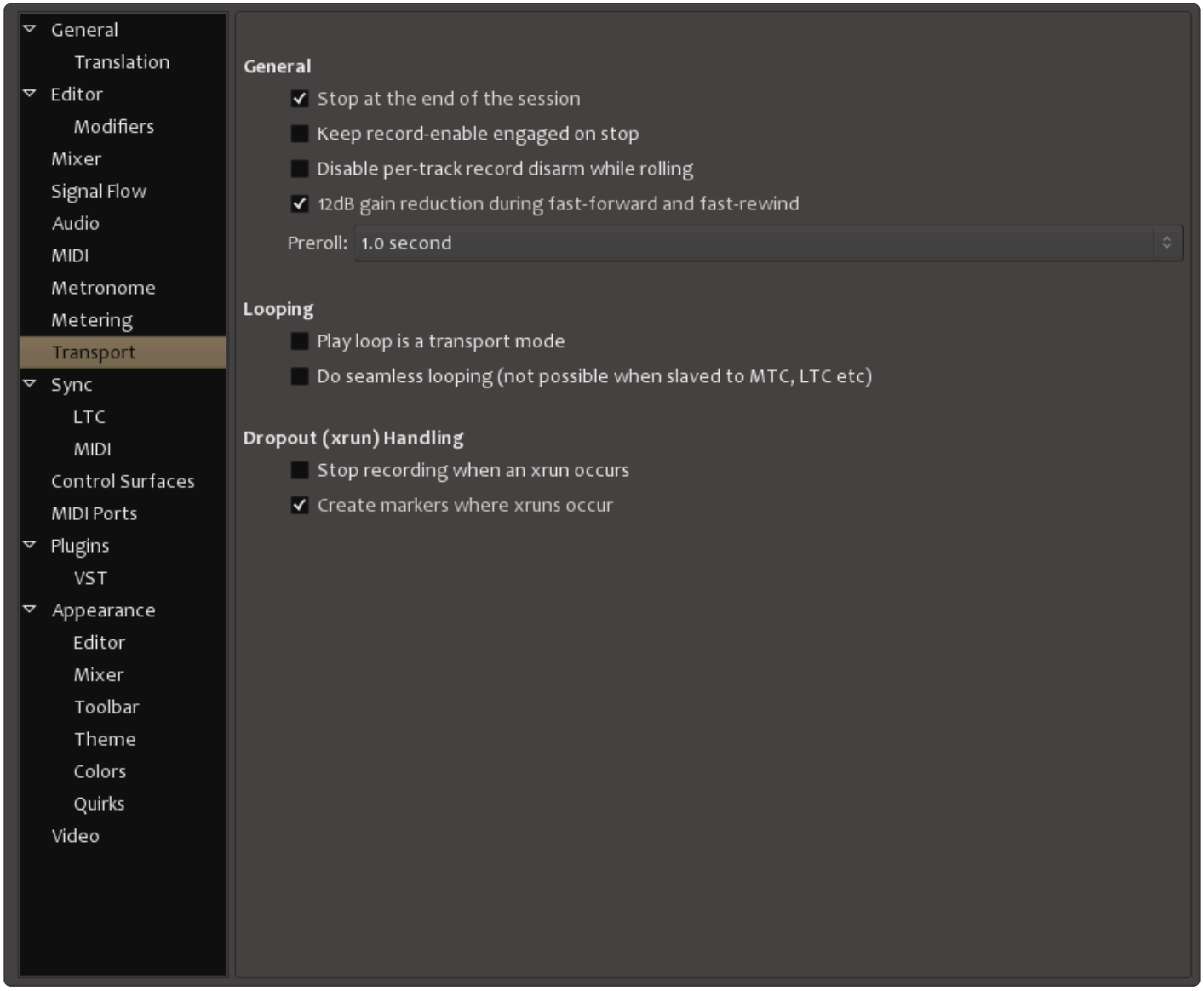
Peak threshold [dBFS]

Post Export Analysis

Save loudness analysis as image file

When an “export” is performed, and the “analysis” checkbox is enabled, this option will cause Mixbus to *also* save an image file of the resulting loudness analysis. This can be useful record to keep with your exported master file.

Transport



General

Stop at the end of the session

When enabled if Mixbus is not recording, it will stop the transport when it reaches the current session end marker. When disabled Mixbus will continue to roll past the session end marker at all times.

Keep record-enable engaged on stop

To prevent accidental overwriting this is off by default.

Disable per-track record disarm while rolling

When enabled this will prevent you from accidentally stopping specific tracks recording during a take.

12dB gain reduction during fast-forward and fast-rewind

This will prevent the unpleasant increase in perceived volume that occurs when fast forwarding or rewinding through certain kinds of audio.

Preroll

There are several options here including Bars (1, 2, or 4), Zero (no preroll at all), and Seconds (0.1 to 2.0)

Looping

Play loop is a transport mode

When enabled the loop button does not start playback but forces playback to always play the loop. When disabled the loop button starts playing the loop, but stop then cancels loop playback.

Do seamless looping (not possible when slaved to MTC, LTC etc)

When enabled this will loop by reading ahead and wrapping around at the loop point, preventing any need to do a transport locate at the end of the loop. When disabled looping is done by locating back to the start of the loop when Mixbus reaches the end which will often cause a small click or delay.

Dropout (xrun) Handling

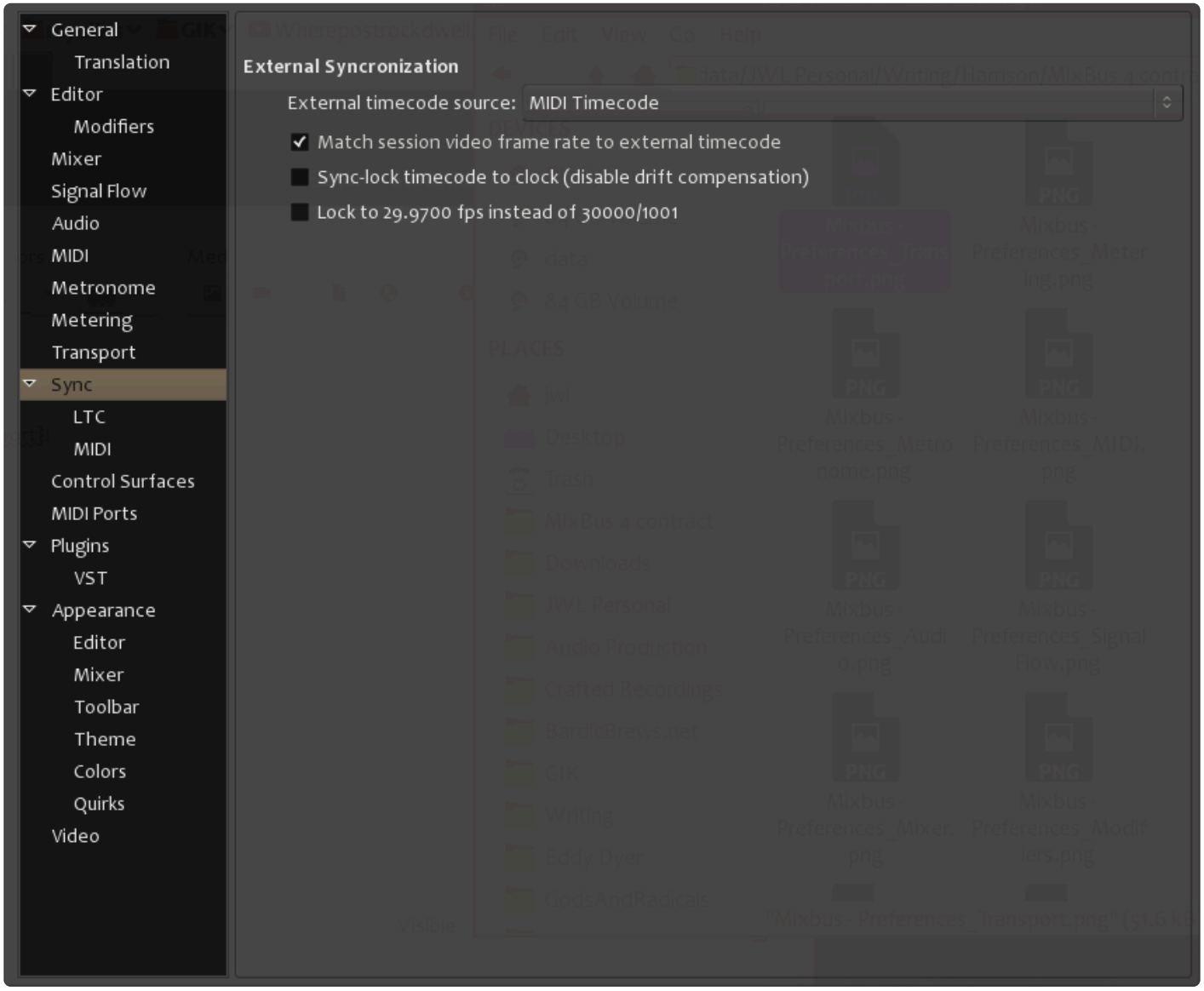
Stop recording when an xrun occurs

A buffer xrun will produce noise. Checking this option prevents this noise from accidentally being recorded.

Create markers where xruns occur

When enabled, this option creates new markers on the timeline whenever an xrun occurs.

Sync



External Synchronization

External Timecode source

This controls what protocol Mixbus synchronizes to. Options are Jack, MIDI Timecode, MIDI Clock, and LTC.

Match session video frame rate to external timecode

This option controls the value of the video frame rate while chasing an external timecode source.

- When enabled the session video frame rate will be changed to match that of the selected external timecode source.
- When disabled the session video frame rate will not be changed to match that of the selected external

timecode source. Instead the frame indication in the main clock will flash red and Mixbus will convert between the external timecode standard and the session standard.

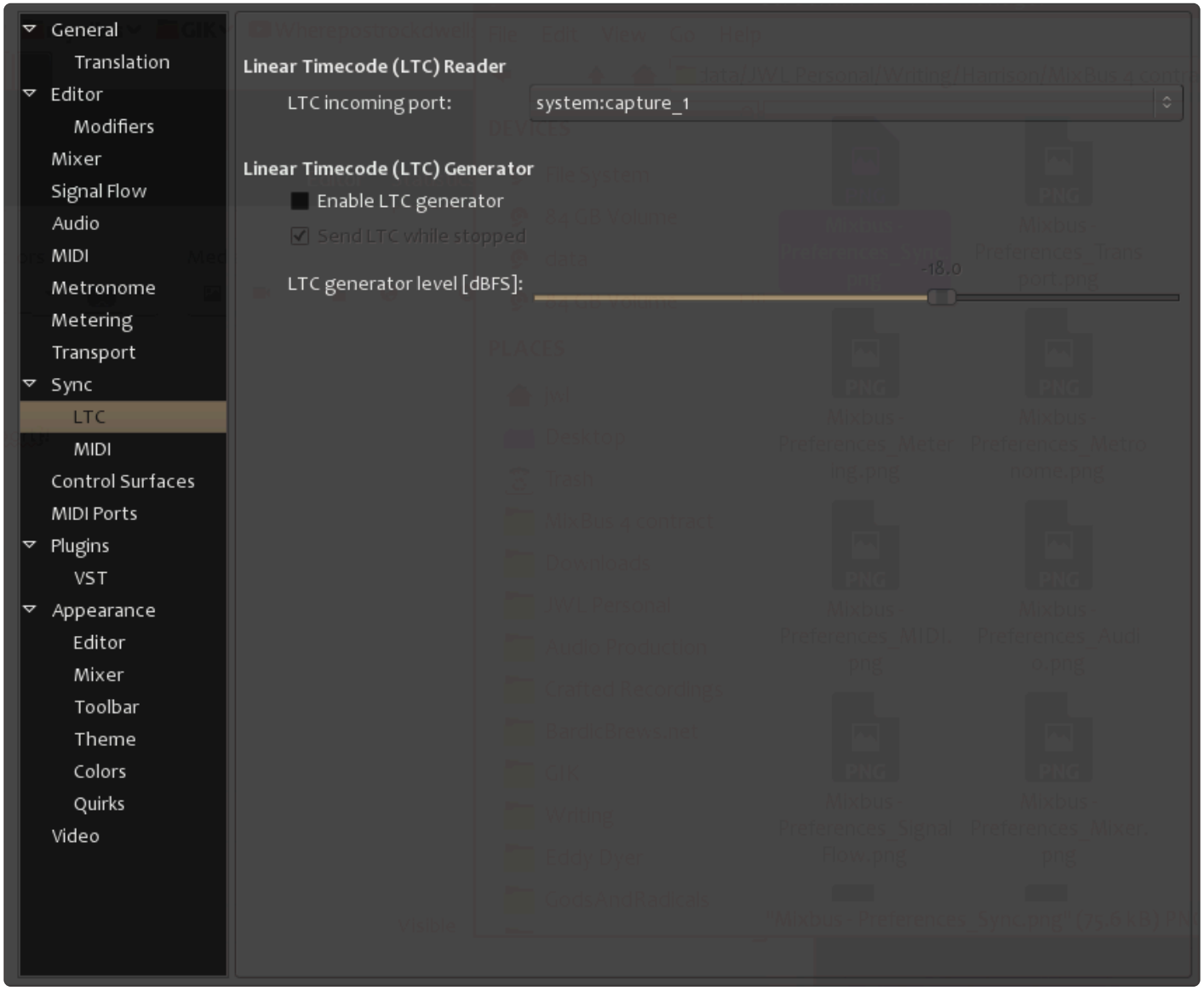
Sync-lock timecode to clock (disable drift compensation)

- When enabled Mixbus will never varispeed when slaved to external timecode. Sync Lock indicated that the selected external timecode source shares clock-sync (Black and Burst, Wordclock, etc) with the audio interface. This option disables drift compensation. The transport speed is fixed at 1.0. Varispeed LTC will be ignored and cause drift.
- When disabled Mixbus will compensate for potential drift, regardless if the timecode sources share clock sync.

Lock to 29.9700 fps instead of 30000/1001

When enabled the external timecode source is assumed to use 29.97 fps instead of 30000/1001. SMPTE 12M-1999 specifies 29.97 fps as 3000/1001. The spec further mentions that drop-frame timecode has an accumulated error of -86 ms over a 24 hour period. Drop-frame timecode would compensate exactly for a NTSC color frame rate of 30 * 0.9990 (i.e. 29.970000). That is not the actual rate. However, some vendors use that rate – despite it being against the specs – because the variant of using exactly 29.97 fps has zero timecode drift.

Sync > LTC



Linear Timecode (LTC) Reader

LTC incoming port:

Choose an audio input port (on your soundcard) which will be used to interpret LTC timecode.

Linear Timecode (LTC) Generator

Enable LTC generator

When enabled, Mixbus will send LTC from the LTC port.

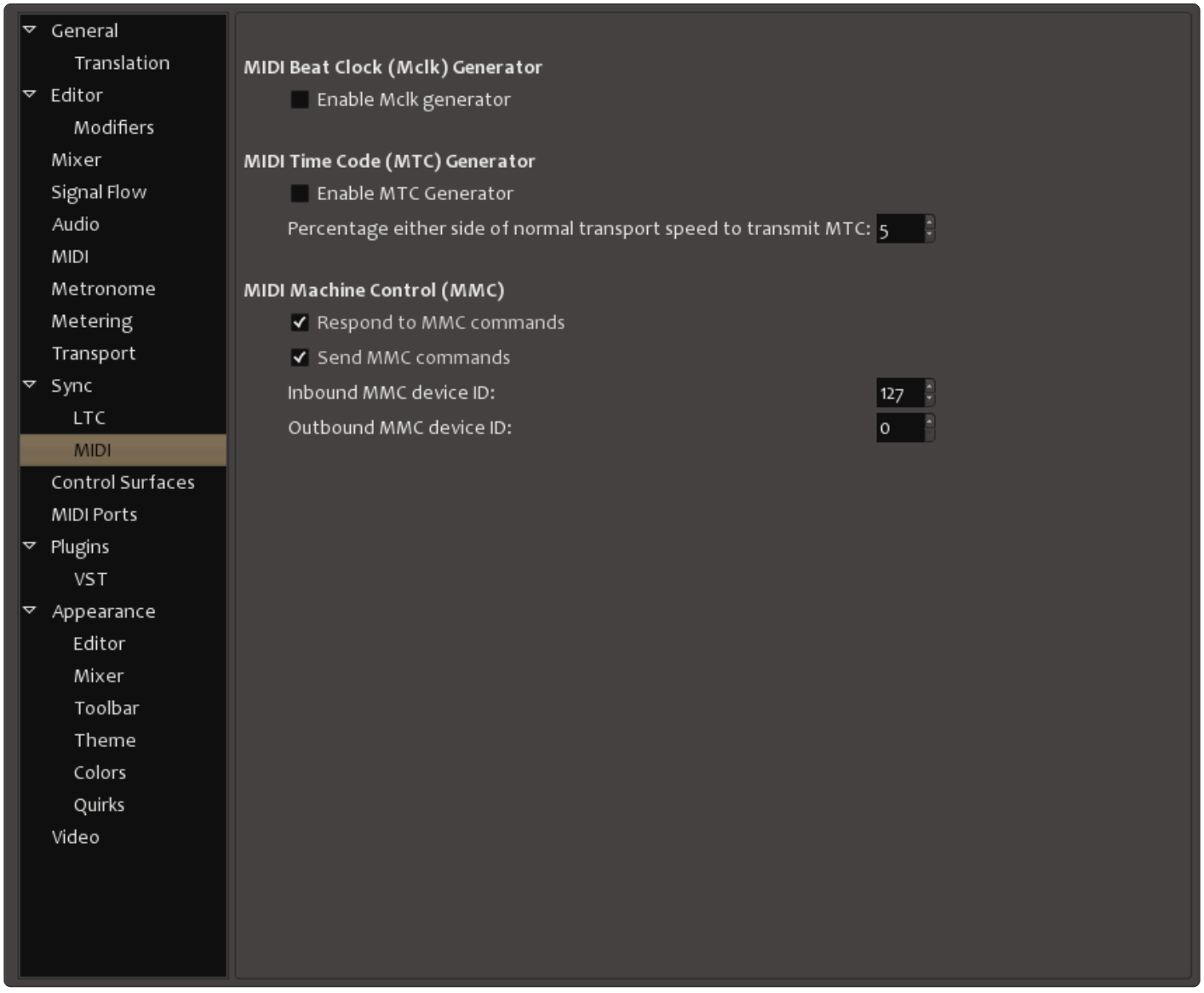
Send LTC while stopped

When enabled Mixbus will continue to send LTC information even when the transport (playhead) is not moving.

LTC generator level [dbfs]

Specify the peak volume of the generated LTC signal in dBFS. A good value is 0 dBu^= -18 dBFS in an EBU calibrated system.

Sync > MIDI



MIDI Beat Clock (Mclk) Generator

Enable Mclk generator

When enabled, Mixbus will send MIDI Beat clock through the MTC port.

MIDI Time code (MTC) Generator

Enable MTC Generator

When enabled, Mixbus will send MTC clock through the MTC port.

Percentage either side of normal transport speed to transmit MTC:

Some devices cannot maintain sync when playback is too fast. This option can limit the range that Mixbus will send MTC to slaved devices.

MIDI Machine Control (MMC)

Respond to MMC commands

If enabled, Mixbus will respond to MMC (Play, Stop, etc) commands on the MMC MIDI port.

Send MMC commands

When enabled, Mixbus will send MMC commands via the MMC MIDI port.

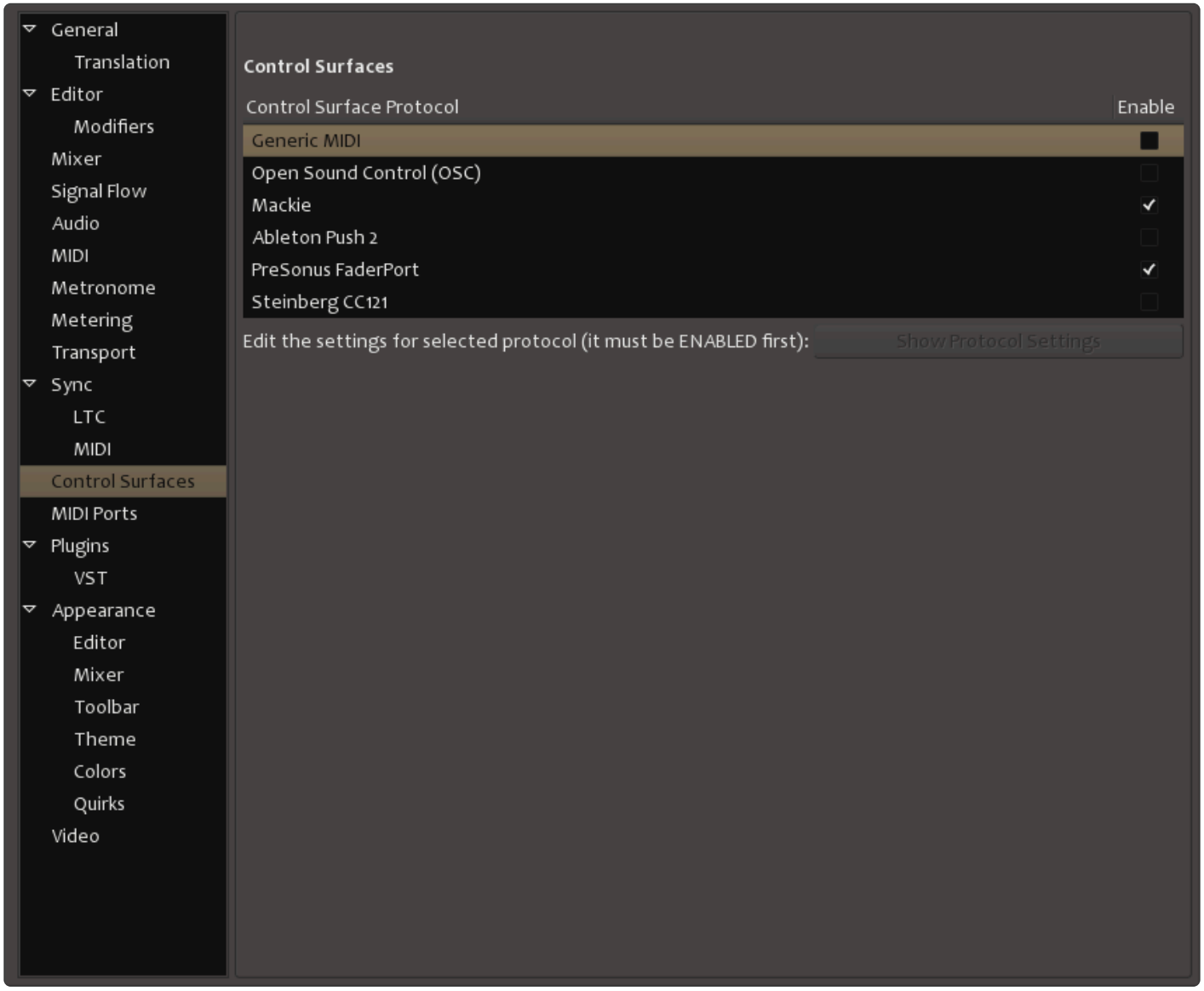
Inbound MMC device ID

MMC machines can be assigned an “ID” for controlling multiple machines

Outbound MMC device ID

MMC machines can be assigned an “ID” for controlling multiple machines

Control Surfaces



Control Surfaces

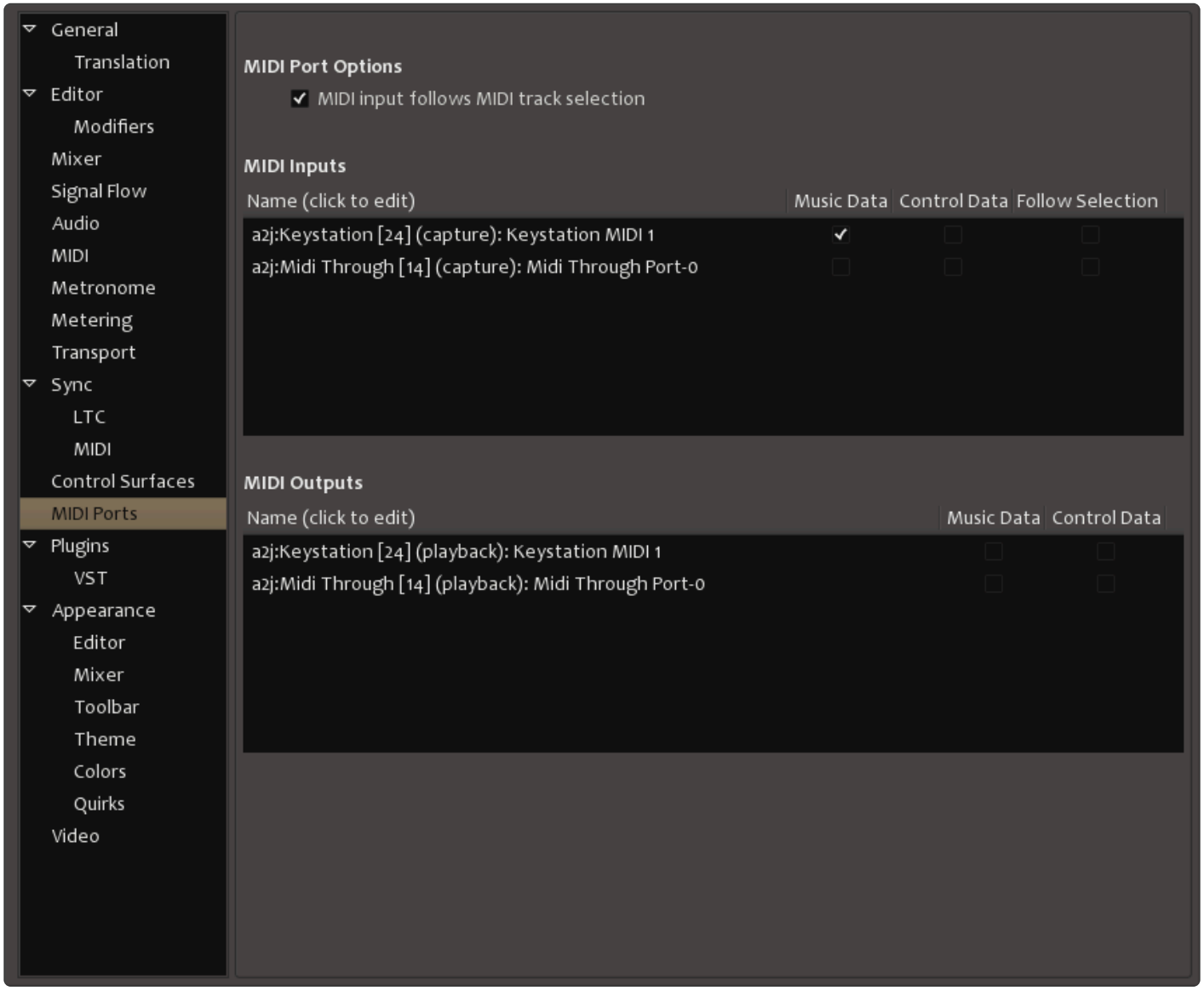
Control Surface Protocol

This option allows you to specify the Control Surface Protocol you'd like to use. Some options, like Generic MIDI, OSC, and Mackie are common standards that may be used by a variety of control surfaces. Others, like Ableton Push 2, PreSonus FaderPort, and Stenberg CC121 are Mixbus' implementation of specific controllers available. If you'd like to use one of these products, enable it here, and tweak the various options below.

Edit the settings for selected protocol (it must be ENABLED first)

Once you have enabled a protocol above, you can edit the settings for that protocol by clicking the button, which will launch a new dialog box.

MIDI Ports



MIDI Port Options

MIDI input follows MIDI track selection

If enabled, then MIDI will be delivered to the selected track, and no other tracks.

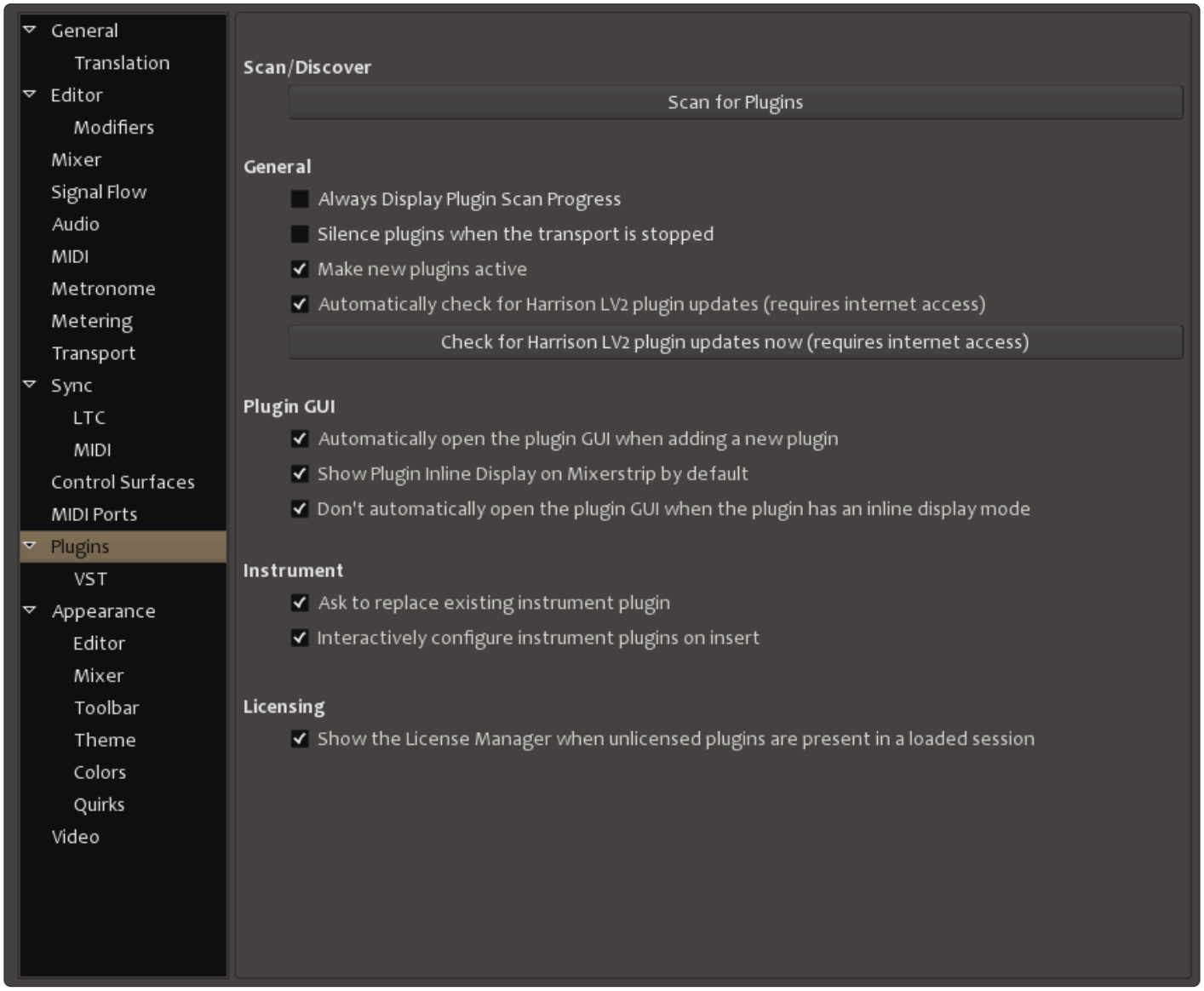
MIDI Inputs

This area allows you to enable various MIDI input devices connected to your system, and activate **Music Data**, **Control Data**, and **Follow Selection** for each device.

MIDI Outputs

This area allows you to enable various MIDI output devices connected to your system, and activate **Music Data** and **Control Data** for each device.

Plugins



Scan/Discover

The Scan for Plugins button initiates a new scan for any new plugins you may have installed along the paths listed in the VST path box. A pop up dialog window shows the progress of the scan operation.

General

Always Display Plugin Scan Progress

When enabled, Mixbus will display a dialog window showing each plugin being scanned. This is ON by default.

Silence plugins when the transport is stopped

When enabled, Mixbus will stop processing plugins when the transport is stopped. This can save battery life, but it causes effects like reverb or delay to “mute” when the transport is paused. This option is OFF by default.

Make new plugins active

By default, Mixbus will immediately enable a plugin when it is added. If you disable this option, then newly-added plugins will be “disabled” by default.

Automatically check for Harrison LV2 plugin updates (requires internet access)

This option, when enabled, will automatically update Harrison plugins. You can also manually update them by clicking the provided button.

Plugin GUI

Automatically open the plugin GUI when adding a new plugin

When a new plugin is added to the strip, the plugin’s UI will be shown. This is ON by default. Note that this only applies when a single plugin is added. If you add multiple plugins at once, the GUI will not appear automatically.

Show Plugin inline Display on Mixerstrip by default

When a plugin with an “inline display” is added, this option will cause the inline display to be shown automatically

Don’t automatically open the plugin GUI when the plugin has an inline display mode

When a plugin with an “inline display” is added, this option will cause the inline display to be shown *instead of* the plugins main UI.

Instrument

Ask to replace existing instrument plugin

When an instrument is added to a track that already has an instrument, Mixbus will prompt you to replace the existing instrument. This is ON by default.

Interactively configure instrument plugins on insert

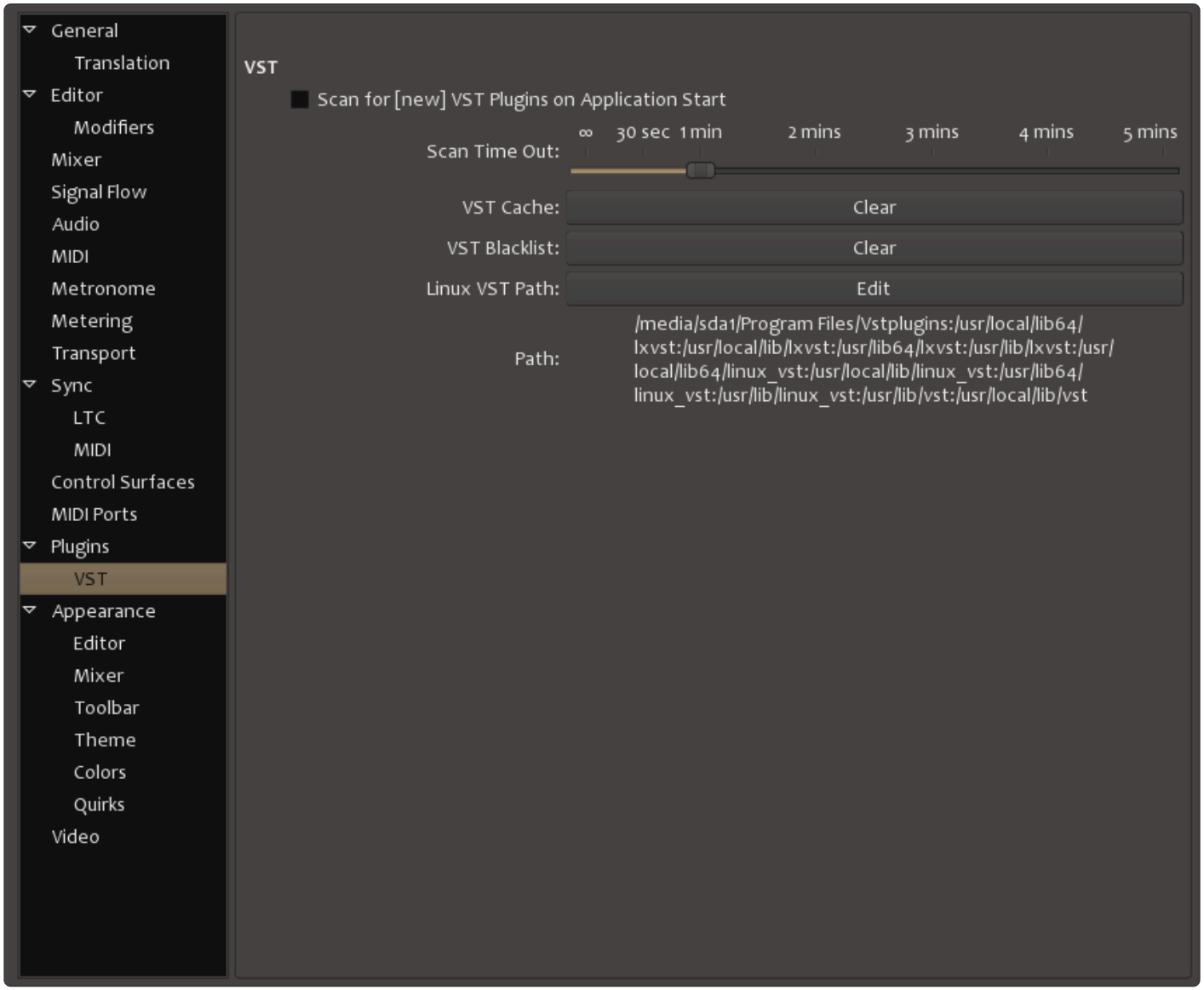
When a plugin with multi-outputs is added to a track, this option will cause Mixbus to prompt the user for the desired channel configuration: mono, stereo, or multichannel.

Licensing

Show the License Manager when unlicensed plugins are present in a loaded session

When not enabled, you could miss that unregistered plugins are not behaving as you'd expect. Enabling this option allows you to activate licenses quickly and easily to resume your session as soon as possible.

Plugins > VST



VST

Scan for [new] VST Plugins on Application Start

New VST plugins are searched, tested and added to the cache index on application start. When disabled new plugins will only be available after triggering a ‘Scan’ manually.

Scan Time out

If a plugin exceeds this time when scanning, this might indicate a bug in the plugin. In this case the scan will be aborted and the plugin will be put on the “blacklist”. Plugins sometimes take a long time to scan, because they are checking online for licensing information or updates. Adjust this value to accommodate your plugin’s needs.

VST Cache

Clicking this button will clear the VST “cache” which is a list of plug-ins that have been scanned. Mixbus will only load plug-ins that have been scanned and are in the cache. So clicking this button will VST plug-ins in Mixbus until paths have been scanned.

VST Blacklist

When a plug-in crashes on scanning, it is placed in the “blacklist” and will not be scanned in the future. Click this button to clear all blacklisted plug-ins so they can be scanned.

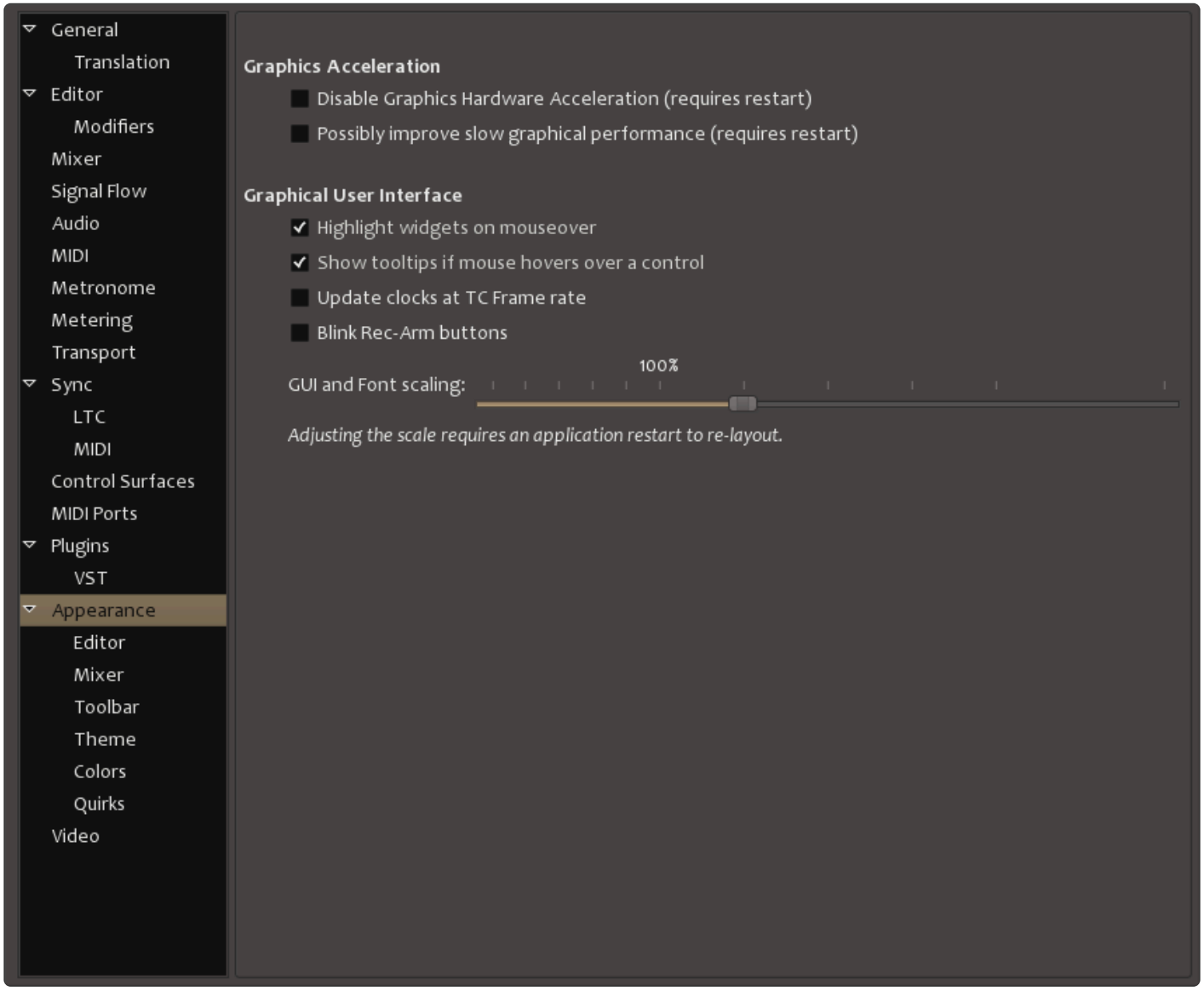
Linux VST Path

Clicking this button launches a dialog box that allows the user to specify which folders contain VST plug-ins. **Add** opens a dialog box that allows you to add new folders. Highlight the desired folder and click “OK” to add it to the VST path. Pressing **Delete** will remove the selected path from the VST scan. **Restore to Default** will set the VST path to those folders that are defined as VST paths in the Windows system. This includes paths set in the registry as well as the environment variable VST_PATH.

Path

The VST Path is a list of directories that Mixbus will use to scan for plug-ins. This path may not be directly edited by the user, but can be cleared, added to, and set to defaults with the buttons above.

Appearance



Graphics Acceleration

Disable Graphics Hardware Acceleration (requires restart)

(this option is only available on linux) . This option can improve compatibility with older video hardware, but might cause slower screen updates.

Possibly improve slow graphical performance (requires restart)

(this option is only available on linux) . This option can improve compatibility with some video hardware.

Graphical User Interface

Highlight widgets on mouseover

If enabled, widgets will be “highlighted” when the mouse passes over them.

Show tooltips if mouse overs over a control

Update clocks at TC Frame rate

Normally clocks update approximately 10 times per second. This option will cause the clocks to get updated at the selected video frame rate, for those users who need to watch the frame counter more accurately.

Blink Rec-Arm buttons

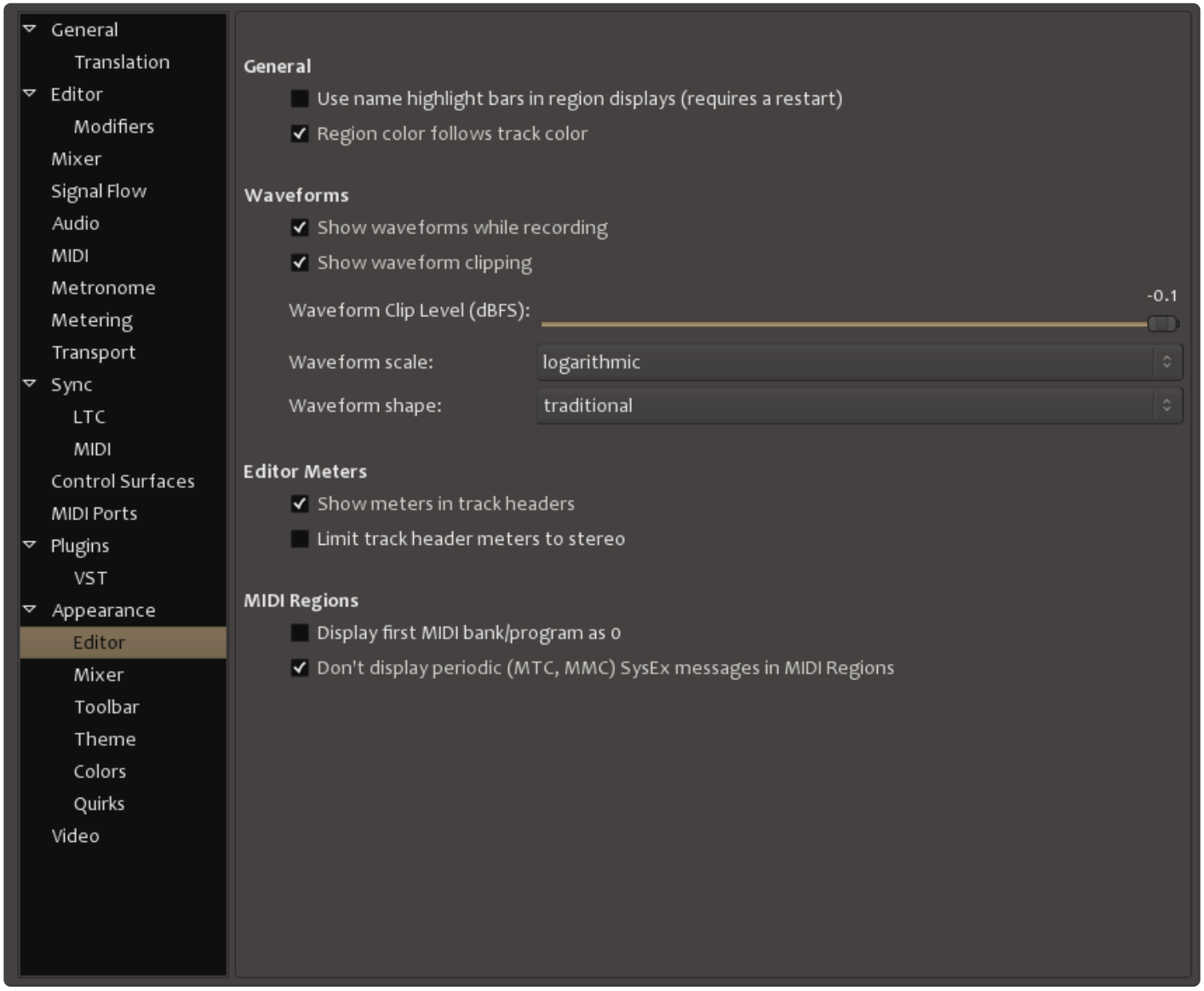
This option blinks the track rec-arm buttons; this can be a more obvious indication when a track is armed.

GUI and Font scaling

Adjust this value to increase or decrease the text used in menus, buttons, and dialogs. 100% is the default value. Settings below 80% may be difficult to read. Adjusting the scale requires a restart, so buttons can resize themselves to fit the new text size.

> ! Due to the way that MacOS handles fonts (due to retina, etc), this slider does not appear on macOS. Instead, you should visit MacOS System Preferences->Displays, and increase the scaling.

Appearance > Editor



General

Use name highlight bars in region displays

This option toggles the display of the highlight bars with names at the bottom of each region, as was default in earlier versions of Mixbus.

Region Color Follows Track Color

When enabled, regions are drawn the same color as the track. If disabled, all regions will be the same color, not affected by the track's selected color.

Waveforms

Show waveforms while recording

This option toggles whether to display region waveforms during recording. It is a useful reference, but does use more system resources. It's ok to leave on in most modern computers.

Show waveform clipping

This option toggles the indication of waveform clipping in the region display. If disabled, regions will not show red indicators when a region exceeds 0dBFS.

Waveform clip level

This option gives you a horizontal fader to adjust where the waveform clip level is calculated, from 0 to -50dBFS.

Waveform scale

This setting does not change the gain of the signal, it is strictly a visual aid that makes the waveforms appear "fatter" with two different waveform rendering scales. The two options are:

- Linear: Displays the audio waveform in a linear fashion. This is the most common display scale in most DAWs.
- Logarithmic: Displays the audio waveform so that quiet sounds can be more easily seen on the screen. The display is disproportionate, giving more visual resolution to low-level signals and less visual resolution at higher-level signals.

Waveform Shape

This setting does not change the gain of the signal, it is strictly a visual aid. The two options are:

- Traditional (Default): Waveforms are drawn around a "zero" axis in the middle of the region.
- Rectified: Waveforms are drawn with "zero" on the bottom of the region, and the wave values are shown in an absolute-value (rectified) magnitude offset

Editor Meters

Show meters in track headers

This option toggles whether meters are shown in the track headers in the editor window. Disable this to hide the small meters at the right edge of track headers in the editor. Enabling this option is useful to help monitor & identify tracks as you listen, but uses more system resources.

Limit track header meters to stereo

If this option is enabled, then Mixbus will only show stereo meters in the editor's track header, even if the plugin has many outputs.

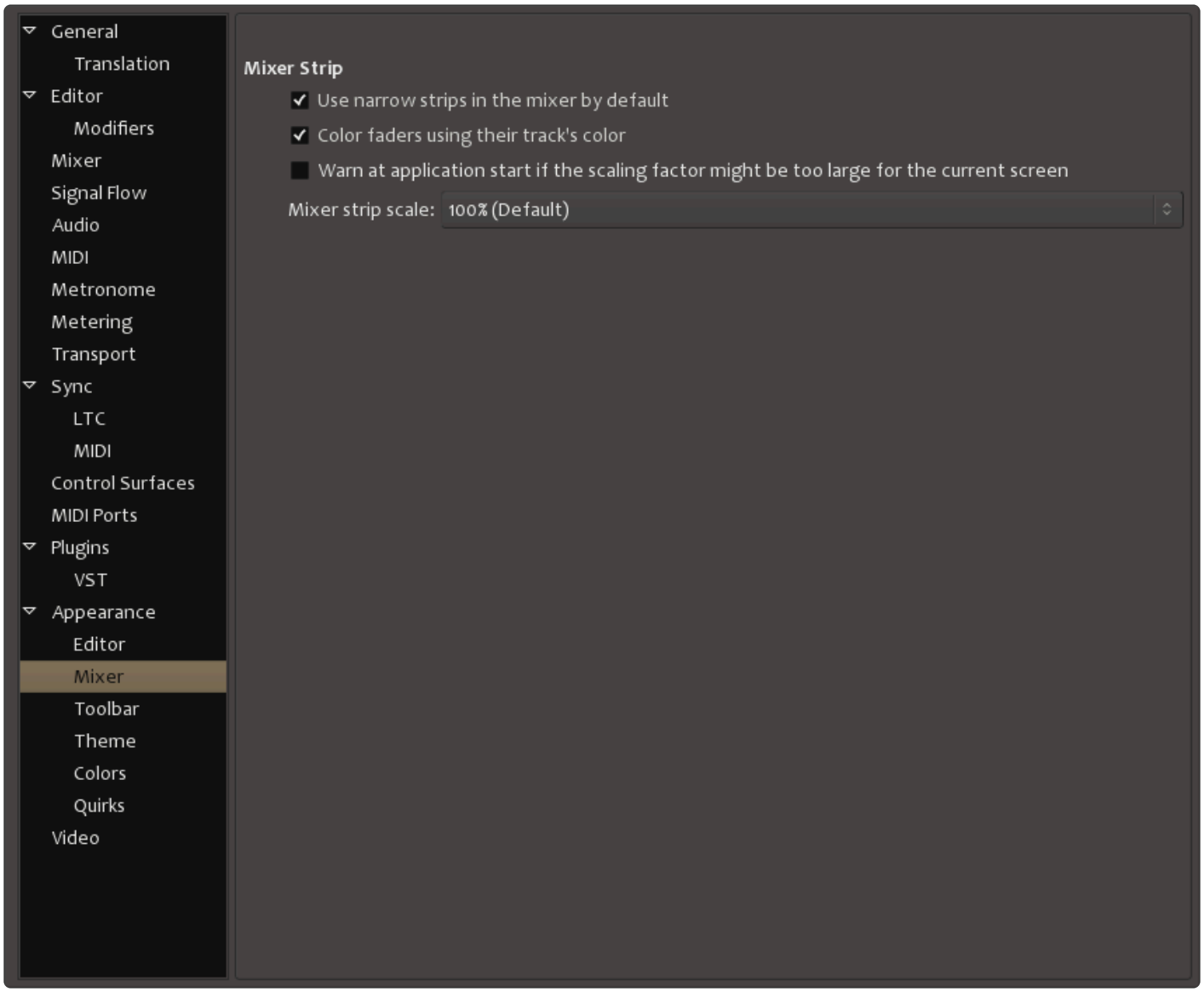
MIDI Regions

Display first MIDI bank/program as 0

By default, Mixbus shows banks&programs starting with “1”. This option shows the more technically-correct numbering starting with “0”.

Don’t display periodic (MTC, MMC) SysEx messages in MIDI Regions

Appearance > Mixer



Mixer Strip

Use narrow strips in the mixer by default

When enabled, the mixer will default to narrow channel/bus strips, to give room for more tracks visible in the

display.

Color faders using their track’s color

When enabled, the mixer fader “knob” will be colored in accordance with the track’s color.

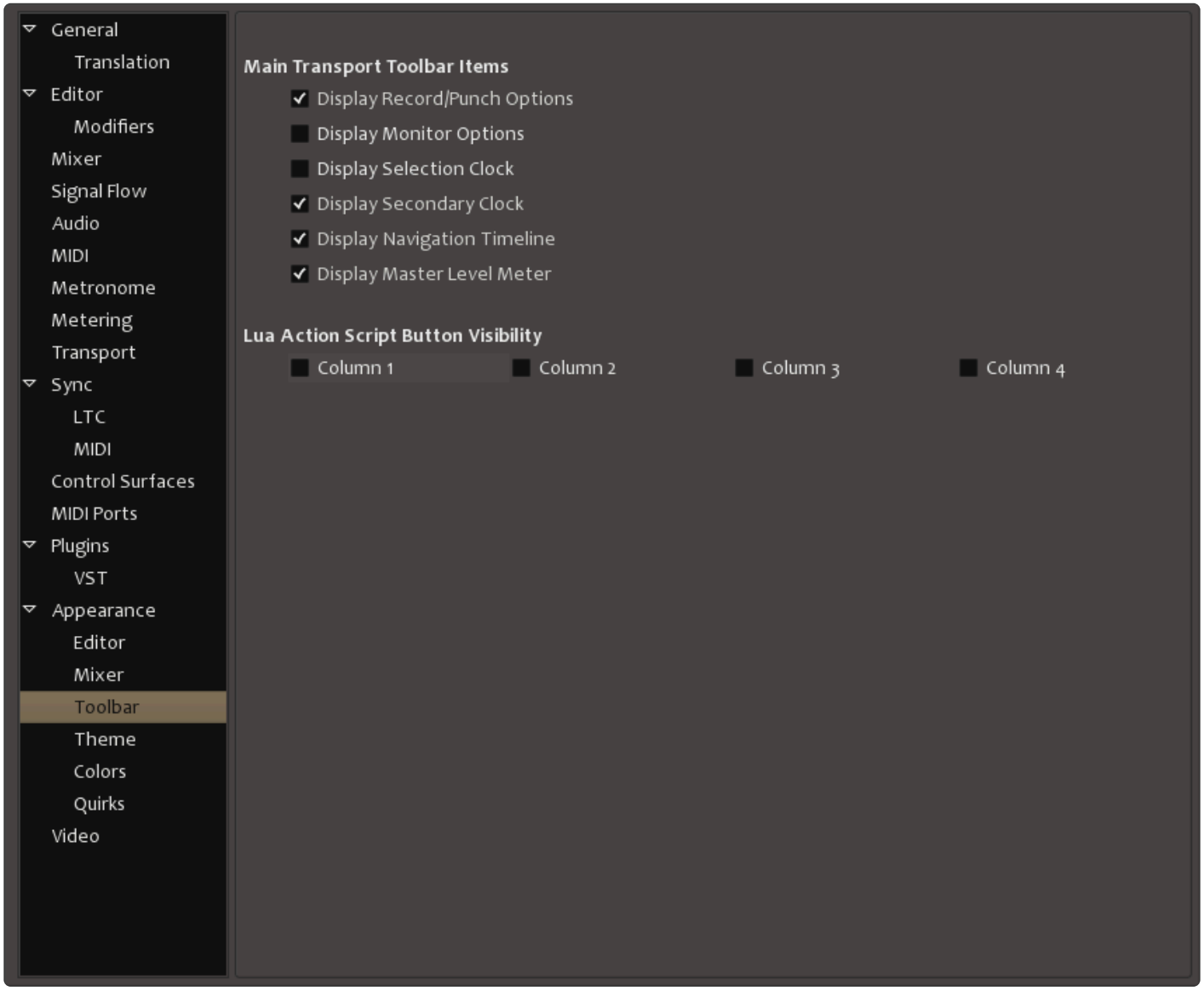
Warn at application start if the scaling factor might be too large for the current screen

When enabled, this will alert you that you will not be able to view the entire screen at once because the scaling factor (below) is set too high.

Mixer strip scale

Adjustable from 70 to 140 percent. This allows you to tweak the display of Mixbus to suit your taste at whatever resolution you are running.

Appearance > Toolbar



Main Transport Toolbar Items

Display Record/Punch options

Display Monitor Options

Display Selection Clock

Display Secondary Clock

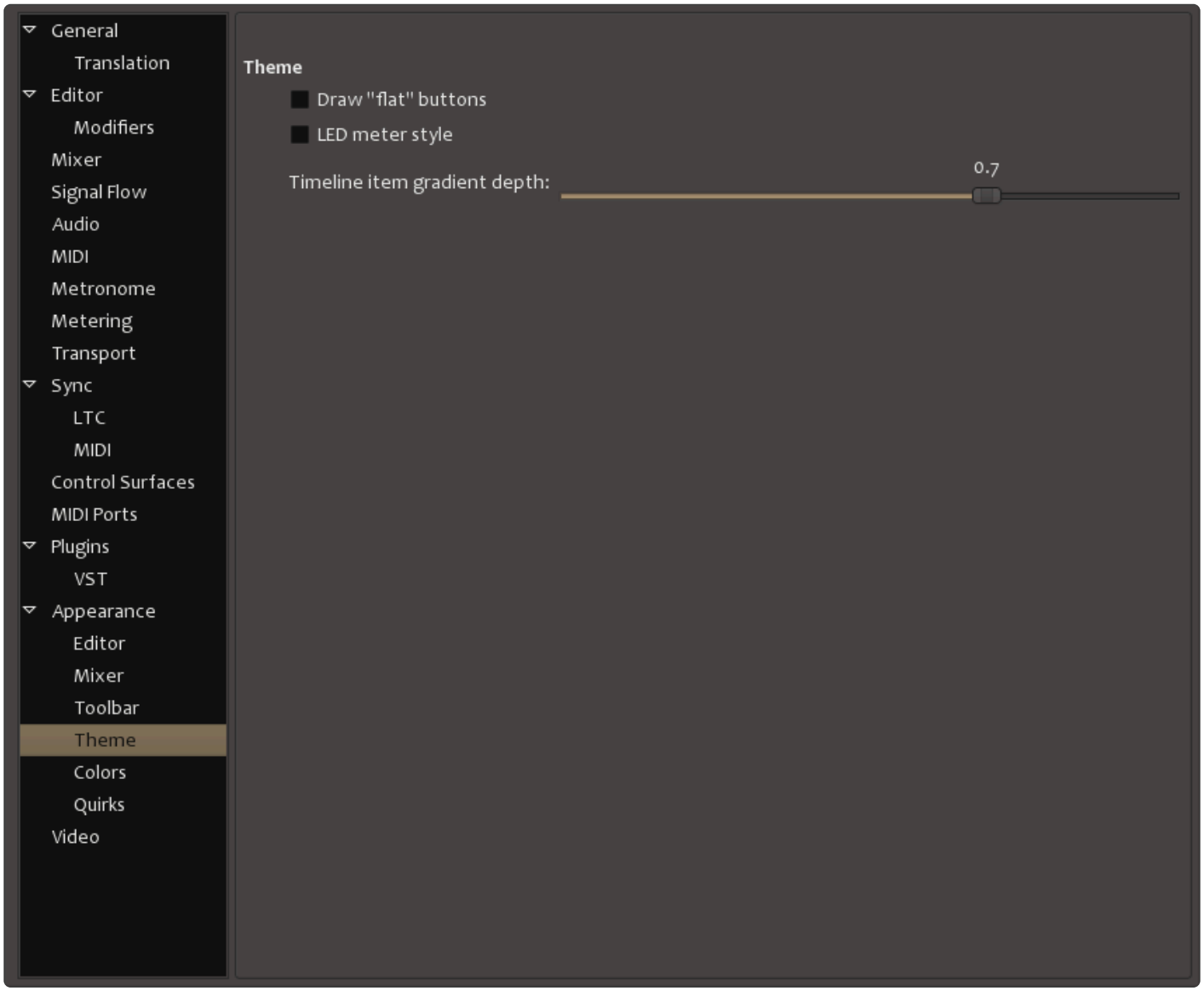
Display Navigation Timeline

Display Master Level Meter

Disable this to hide the master level meter that appears in the top-right corner of the editor window.

Lua Action Script Button Visibility

Appearance > Theme



Theme

Draw “flat” buttons

Removes gradients from the buttons (may result in easier viewing on some monitors).

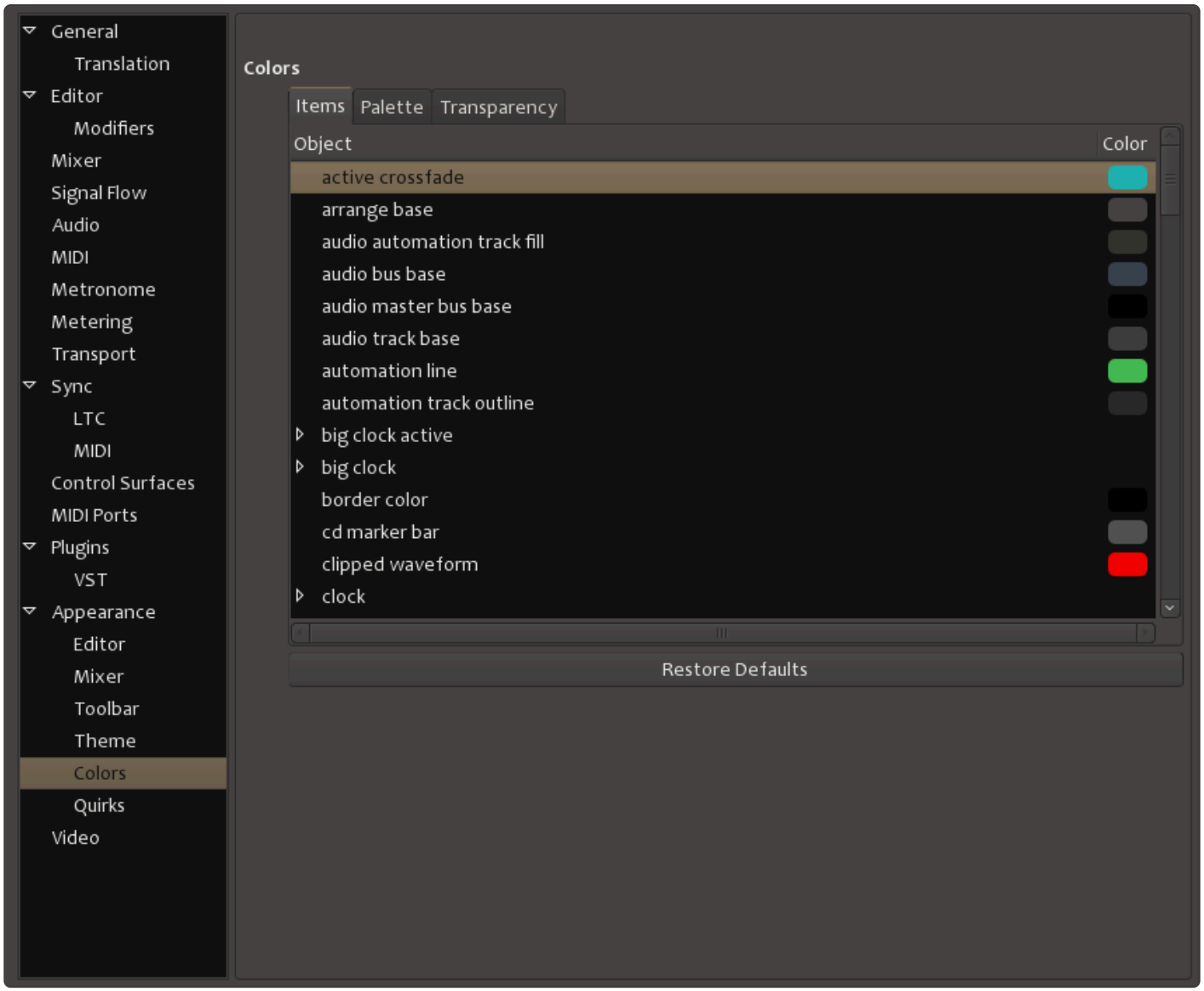
LED meter style

If this option is enabled, the meterbridge meters will appear with segmented “LED style” meters instead of a solid bar.

Timeline item gradient depth

Defines the amount of “color gradient” applied to the audio regions in the timeline. Reducing this value will result in a flatter, less 3d-look.

Appearance > Colors



This window provides a way to customize every color used by Mixbus. There are three tabs.

Items

This tab lists each Object, alphabetically, and displays the color associated with that object. Click on the color to modify.

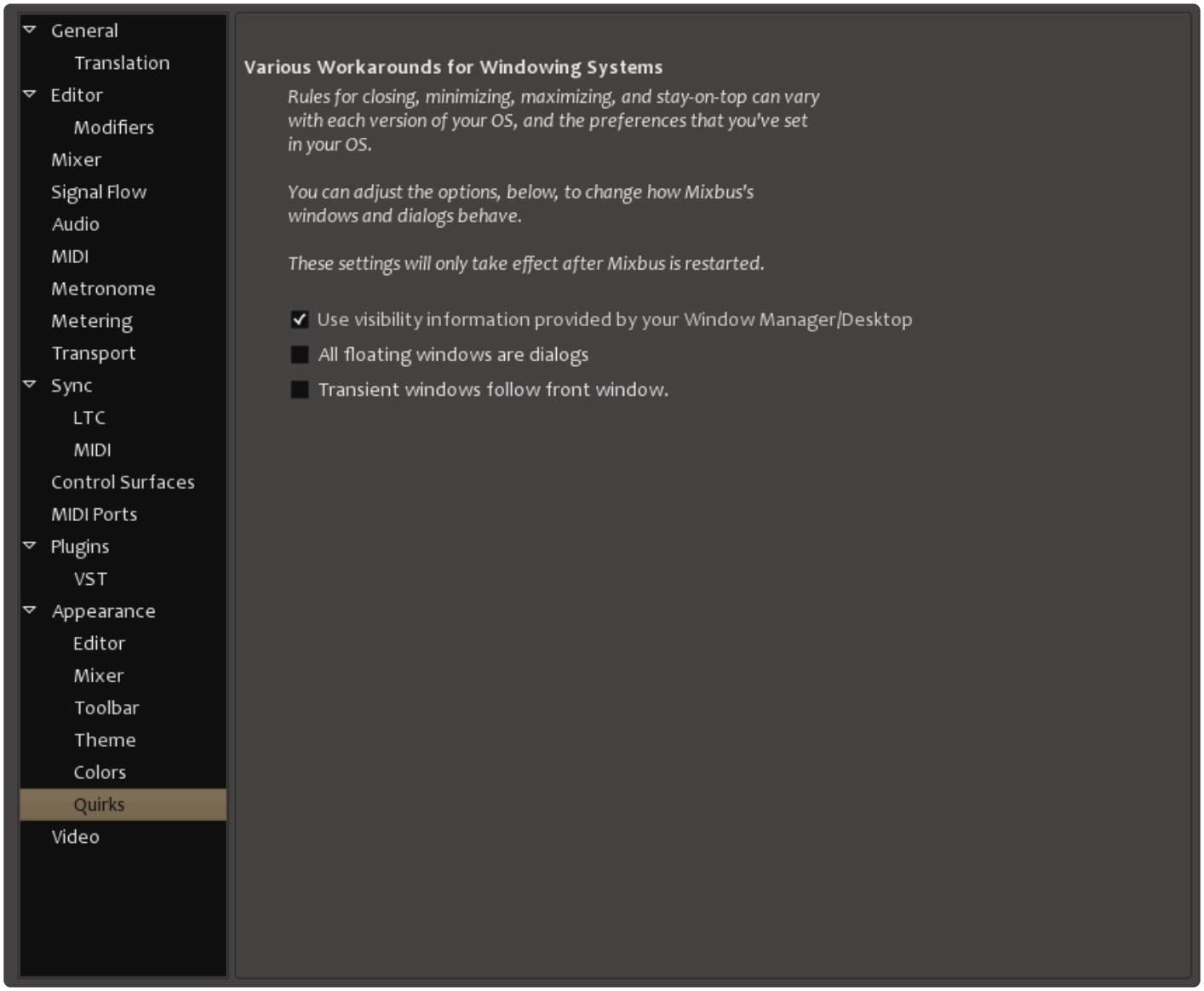
Palette

This tab shows the color palette used by Mixbus, when selecting colors. You can click on each color to modify it, which will become part of the new available palette.

Transparency

This tab lists everything in Mixbus that can be rendered with Transparency. Each object has a slider to adjust the amount of transparency.

Appearance > Quirks



Various Workarounds for Windowing Systems

Rules for closing, minimizing, maximizing, and stay-on0top can vary with each version of your OS< and the

preferences that you've set in your OS. You can adjust the options below to change how Mixbus's windows and dialogs behave. These settings will only take effect after Mixbus is restarted.

Use visibility information provided by your Window Manager/Desktop

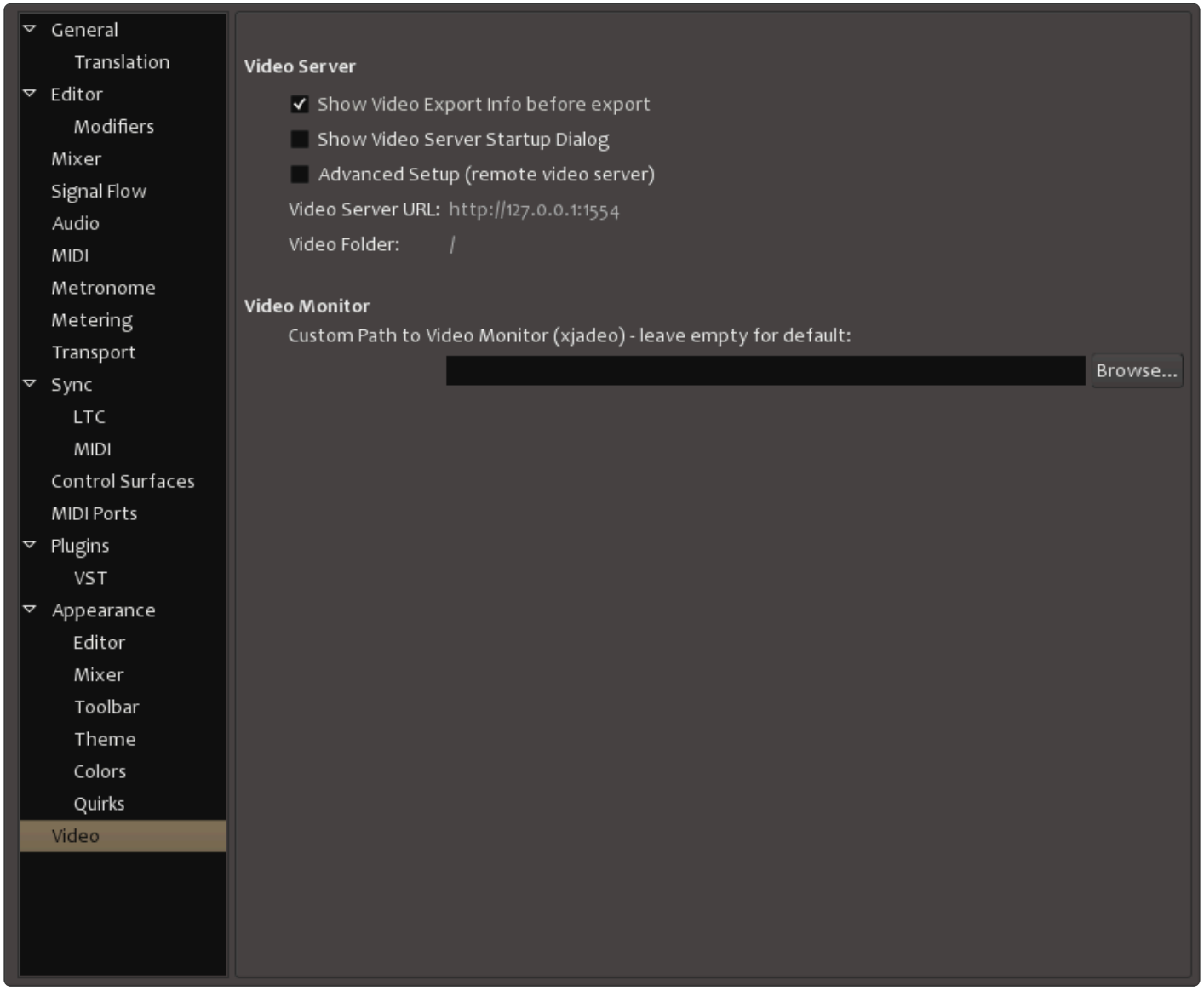
All floating windows are dialogs

Mark all floating windows to be type "Dialog" rather than using "Utility" for some. Requires restart of Mixbus. This can help if you find that some popup dialogs are appearing behind the main window, or otherwise not being managed correctly by your desktop.

Transient windows follow front window

Makes transient windows follow the front window when toggling between the editor and mixer. Requires restart of Mixbus.

Video



Video Server

Show Video Export info before export

When enabled, an information window with details is displayed before video export dialog.

Show Video Server Startup Dialog

When enabled, the video server is never launched automatically without confirmation.

Advanced Setup (remote video server)

When enabled you can specify a custom video server URL and docroot.

Video Server URL:
h4. Video Folder:

See Video

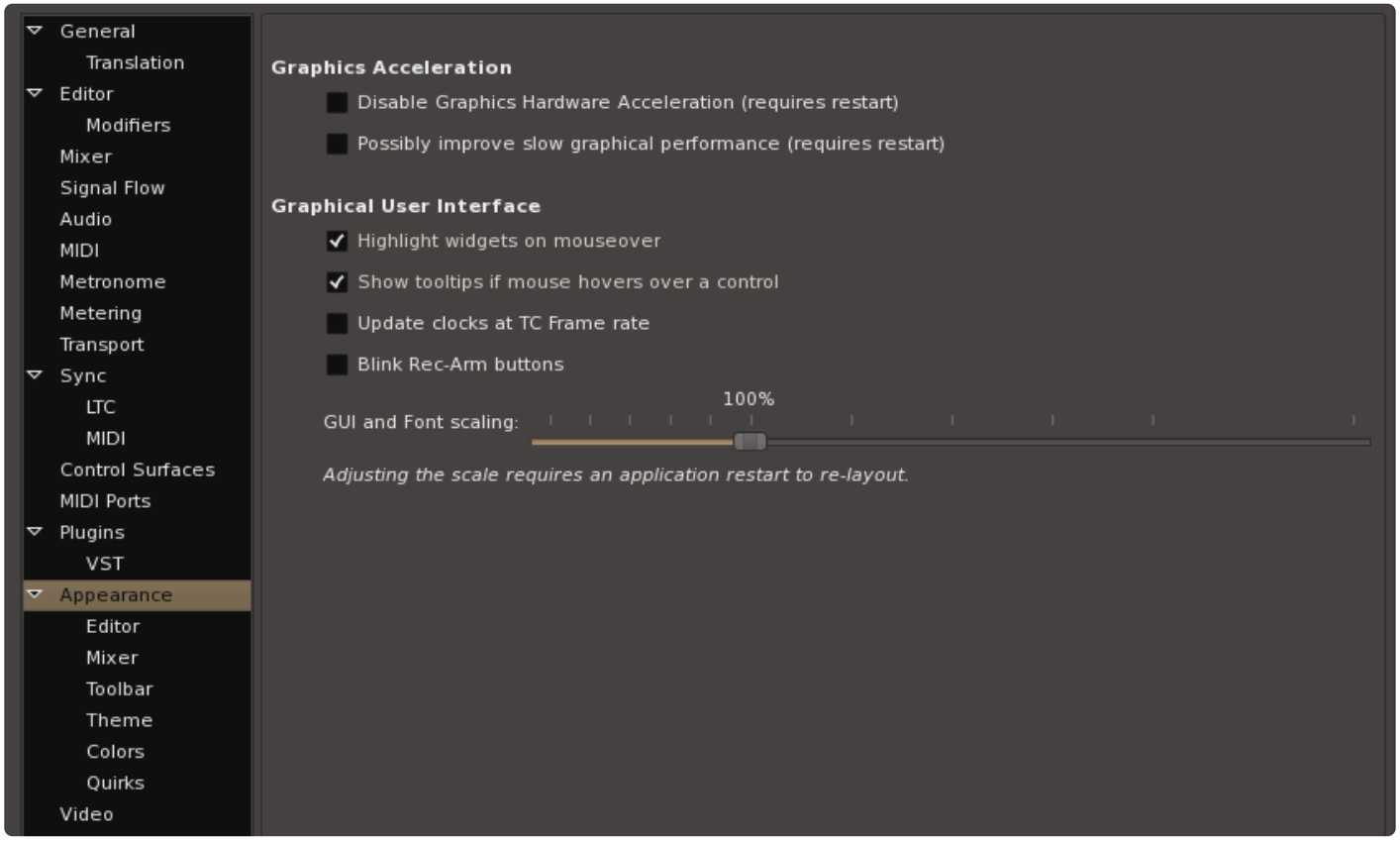
Video Monitor

Custom Path to Video Monitor (xjadeo)

Leave empty for default.

18.3.1.1. Appearance

Appearance:



18.4. Region Menu

This menu give the user access to many editing features for regions or tracks in the canvas.
The region you intent to work with must first be selected or the menu selections with be grayed out.

Region menu includes:

- Insert Region From Region List : If a region is selected in the “Region List” (in the editor sidebar) then this action will insert it at the Edit Point.
- Play
- Loop : sets the “Loop” range to match the region, and starts looping the transport.
- Rename : allows the user to change the name that appears in the region’s top corner.
- Properties : launches a dialog showing the region’s name, position, length, gain, and other properties.
- Loudness Analysis : Performs a loudness analysis on the region, reporting its dBFS and LUFS loudness, as well as spectral energy
- Spectral Analysis : Performs a spectrum analysis on the selected region(s). Use this to compare the spectrum of sound in multiple selected regions.

- Edit >
  - Pitch Shift : initiates a dialog which allows you to shift the region’s pitch up or down, either as a percentage or in semitones.
  - Split at Percussion Onsets : performs a “percussive note onset” analysis, and then splits the regions on the detected hits
  - Make Mono Regions : splits a stereo (or multichannel) region into multiple mono regions. These mono regions appear in the editor sidebar list, “Regions”
  - Close Gaps : removes any space between the selected regions, creating a butt joint between each region
  - Place Transient : manually places a “transient” marker inside the region, at the Edit Point. Transient markers can be used for the grid or other editing actions.
  - Rhythm Ferret
  - Strip Silence…
    select track(s) or region(s) you would like to detect silence. Batch processing possible by selecting multiple tracks or regions. Once the analysis has completed the areas that have met the criteria for silence will be highlighted.
    - Threshold: This sets the audio level for detecting silence and is adjustable by 1 dBFS increments. Any audio below the set threshold will be considered silence.
    - Minimum length (in milliseconds, default =1000 (1 second)): Allows control of the shortest time period that will be considered silence.
    - Fade Length (in samples, default =64): This setting adjusts the length of fades at the transitions of detected silence and detected audio.
  - Reverse : this reverses the audio region’s sound, so it plays “backwards”.

- Layering >
  - Raise to Top : raises the selected region to the top of the layer stack.

    - Raise
    - Lower
    - Lower to Bottom

- MIDI>
    - Transpose : launches a dialog, allowing the user to transpose all MIDI notes up or down
    - Insert Patch Change : Inserts a patch change at the Edit Point. See this section on MIDI Patches
    - Quantize : launches a dialog which edits the midi note start and/or end times to exact grid locations on the music timeline.
    - Legatize : extends the selected note(s) so the end of each note reaches the beginning of the next note (legato)
    - Remove Overlap : trims each MIDI note so that it ends before another note begins (portamento)
    - Transform : launches a dialog with various MIDI operations
    - Unlink from other copies : by default, a copied MIDI region will be “linked” to the source region. Any change to either region will affect all copies. This action will unlink the selected region(s) so they may be edited with no changes to the copies.
    - List Editor : this launches a dialog that allows the user to see and edit MIDI events.

- Gain >
    - Opaque : when disabled, the region will not obscure the regions below it; instead you will hear the region combined with the regions below it.
    - Mute : when enabled, the region will not play, and will be treated as if it were “transparent” to regions below it
    - Normalize : adjusts the selected region(s) gain to reach a specified value. (a dialog allows you to choose *peak* or *rms* normalization)
    - Boost Gain : increases the region gain by 1dB
    - Cut Gain : decrease the region gain by 1dB
    - Reset Envelope : this resets the region’s gain envelope to “unity”
    - Envelope Active : if unchecked, the region’s gain envelope will be ignored.

- Position >
    - Move to Original Position : if the file was originally recorded in Mixbus, or if it has a file timestamp (broadcast wave file), then this action will move the selected region(s) to the original location.
    - Snap Position To Grid : moves the region to the nearest grid (snap) location.
    - Lock : if enabled, the region cannot be moved on the timeline.
    - Glue to Bars and Beats : If enabled, the region is attached to the “music” timeline. See “Audio Time vs Music Time”:”#audio-time-vs-music-time
    - Lock to Video : if locked to the video, then the region will move when the session’s video file is moved on the timeline.
    - Set Sync Position : Sets the region’s sync point to the Edit Point (when initiated from a menu, the sync point is always the playhead) : see “”:#align-regions
    - Remove Sync : Removes the region’s sync point. It will align (or snap) to the region’s start
    - Nudge Later : moves the region by the amount shown on the “nudge” clock in the editor toolbar.

    - Nudge Earlier
    - Nudge Later by Capture Offset : moves the region by one buffersize; this is sometimes useful to correct mis-aligned recordings due to soundcard latency
    - Nudge Earlier by Capture Offset
    - Sequence Regions : the selected regions will be sequenced in a series, with no space between them.

- Trim >
    - Trim Start at Edit Point : trims the selected region(s) so they start at the *edit point*.
    - Trim End at Edit Point : trims the selected region(s) so they end at the edit point.
    - Trim Loop : trims the selected region(s) to the Loop range.
    - Trim Punch : trims the selected region(s) to the Punch range.
    - Trim Previous : trims the selected regions to meet the previous region's end.
    - Trim Next : trims the selected region to meet the next region's start.

- Ranges >
    - Set Loop Range : Set the Loop start/end markers to this region's start/end.
    - Set Punch : Set the Punch start/end markers to this region's start/end.
    - Add Single Range Marker : create a Named Range marker corresponding to the selected region(s)'s start/end.
    - Add Range Marker Per Region : create a Named Range marker corresponding to each selected region.
    - Set Range Selection : Set the Range Selection start/end to the selected region(s)'s start/end.

- Fades >
    - Fade In : if unchecked, the region's fade-in will be disabled, resulting in a hard cut (not recommended)
    - Fade Out : if unchecked, the region's fade-out will be disabled, resulting in a hard cut (not recommended)
    - Fades : toggles the selected region(s)'s fade-in and fade-out simultaneously

- Duplicate >
    - Duplicate : copies the region, and pastes the copy immediately at the end of this region.
    - Multi-Duplicate : duplicates the region multiple times (a dialog will prompt for the number of times to duplicate)
    - Fill Track : duplicates the region enough times to fill the region to the Session End marker.

- Export : the region can be exported (including region gain, gain envelope, and fade in/out) to a new file, to any disk location, and in the format of the user's choice.

- Bounce (without processing) : the region will be bounced (including region gain, gain envelope, and fade in/out) to a new file, which will appear in the editor sidebar Region list.

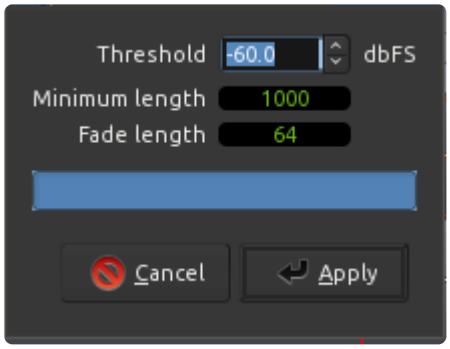
- Remove : the region is deleted.

18.4.1. Strip Silence

The Strip Silence command can be found under Region>Edit>Strip Silence. You must first select the track(s) or region(s) in which you would like to detect silence. Batch processing is possible by selecting multiple track or regions. Once the Strip Silence dialogue has been opened Mixbus will begin analysis on all selected tracks or regions (allow time based on CPU speed and track/region length). Once the analysis has completed the areas that have been determined to contain silence will be highlighted.

There are 3 adjustments available:

- **Threshold**: This sets the audio level for detecting silence and is adjustable by 1 dBFS increments. Any audio below the set threshold will be considered silence.
- **Minimum length** (in milliseconds, default =1000 (1 second)): Allows control of the shortest time period that will be considered silence.
- **Fade Length** (in samples, default =64): This setting adjusts the length of fades at the transitions of detected silence and detected audio.



18.5. Track Menu

A track header must first be selected to access many of the items in this menu.

Most of these items can also be found in the track header context menu, which is accessed by right-clicking on a track header. Generally speaking, right-clicking on a track will only affect the single track, while the Track menu will affect all selected tracks.

- Add Track, Bus, or VCA : Adds additional channels to the editor and mixer. Learn more here.
- Mix Tools : Initiates a scripted action which an affect multiple tracks. Learn more here.
- Import : Learn more in the Import section.
- Duplicate Tracks/Busses : creates a “duplicate” of selected Track(s), including EQ, bussing, plugins. A dialog prompts whether to include the regions (playlist) in the duplicate tracks.
- Toggle Record Enable : toggles record enable for the selected track(s).
- Toggle Solo
- Toggle Mute
- Insert Time : Inserts blank time at the selected time, and optionally move all markers to the right.
- Remove Time: Deletes selected range and markers (from ALL tracks). Everything to the right moves left to close the gap.
- Move Selected Tracks Up
- Move Selected Tracks Down
- Height >
    - Fit Selection (Vertical)
    - Largest
    - Larger
    - Large
    - Normal
    - Small

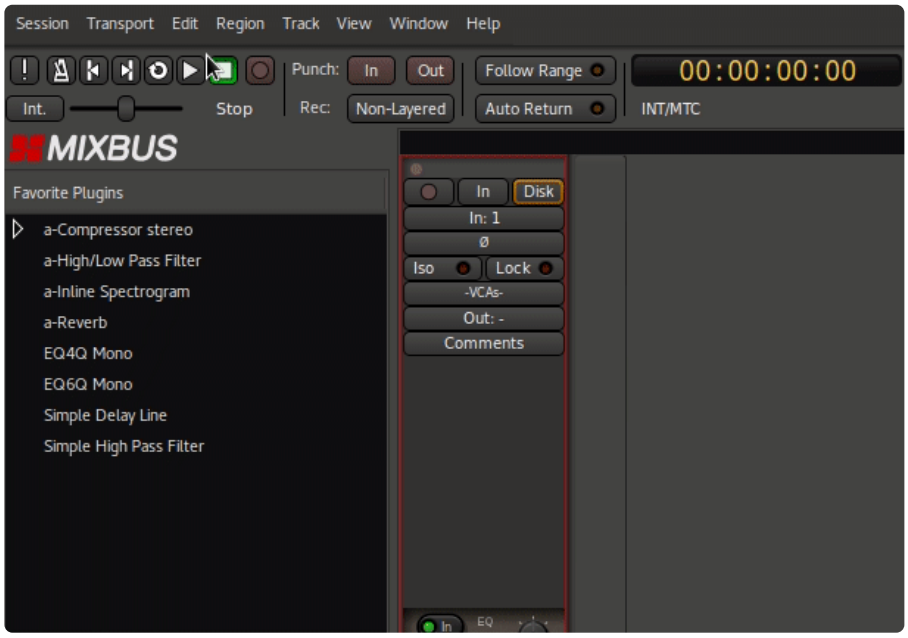
- Toggle Active : Toggles the state of all selected tracks; inactive tracks will become active, and vice-versa. Inactive tracks utilize no CPU or disk access; they are completely disabled. This option cannot be changed while Mixbus is playing.

- Remove

18.5.1. Mix Tools

What are Mix Tools?

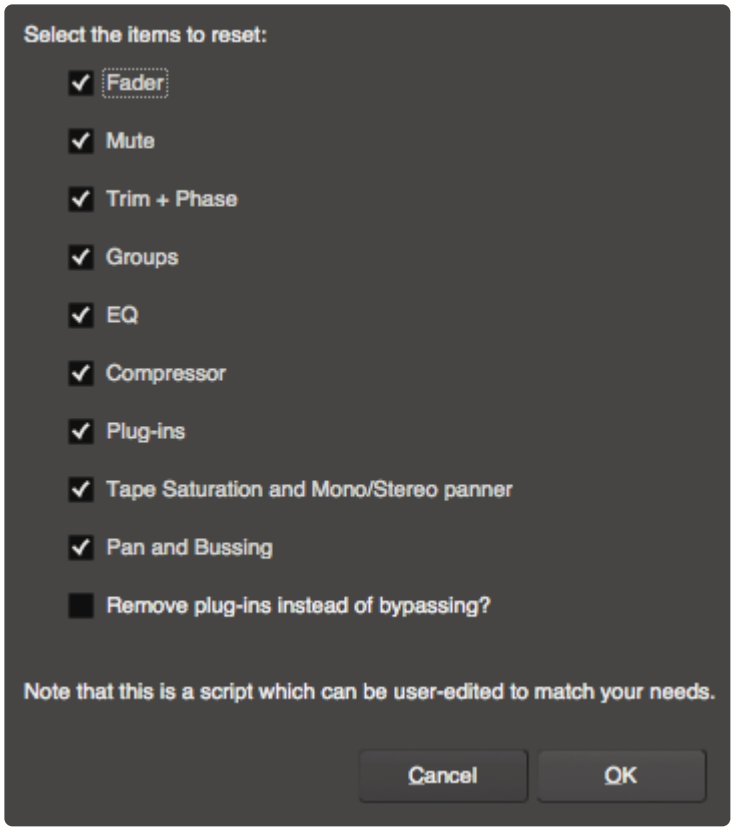
In the main menu, under the “Track” menu, you will find several operations which can automate certain tasks that are commonly encountered while mixing.



Currently, 4 Mix Tools are provided:

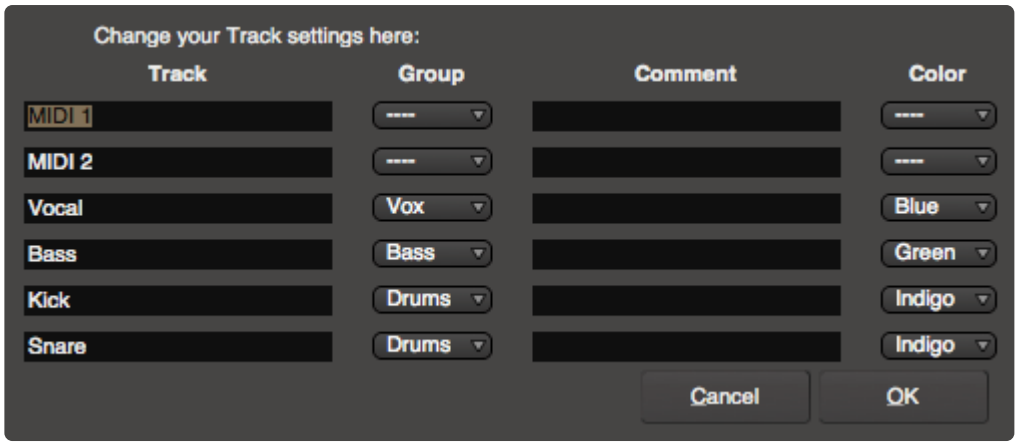
Reset Mixer

This tool applies to all the mixer channels, and it resets the knobs to their default positions. You may choose from the list if you’d like to save certain mix settings. Note that this CANNOT be undone.

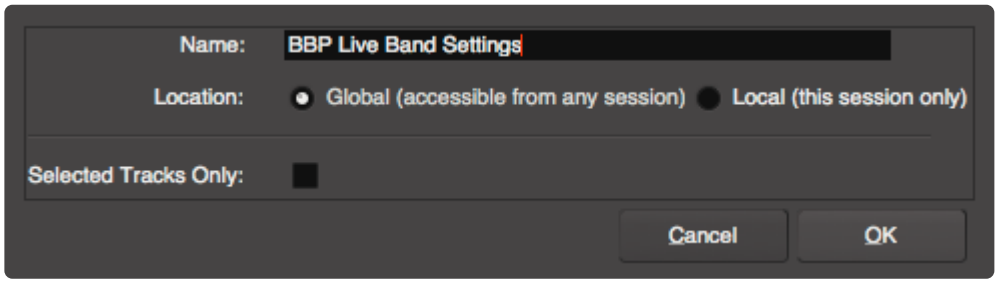


Track Organizer

This tool provides a list of all the tracks in the session, along with simplified interface to rename the tracks as well as assign a color, comment, or group to each track:

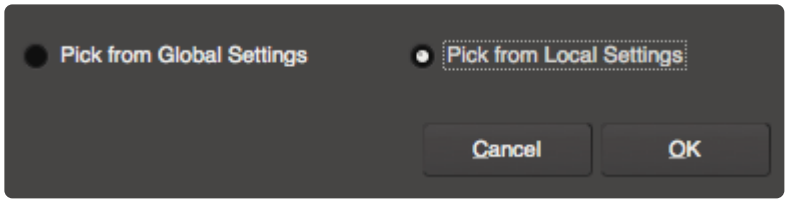


Store Mixer Settings



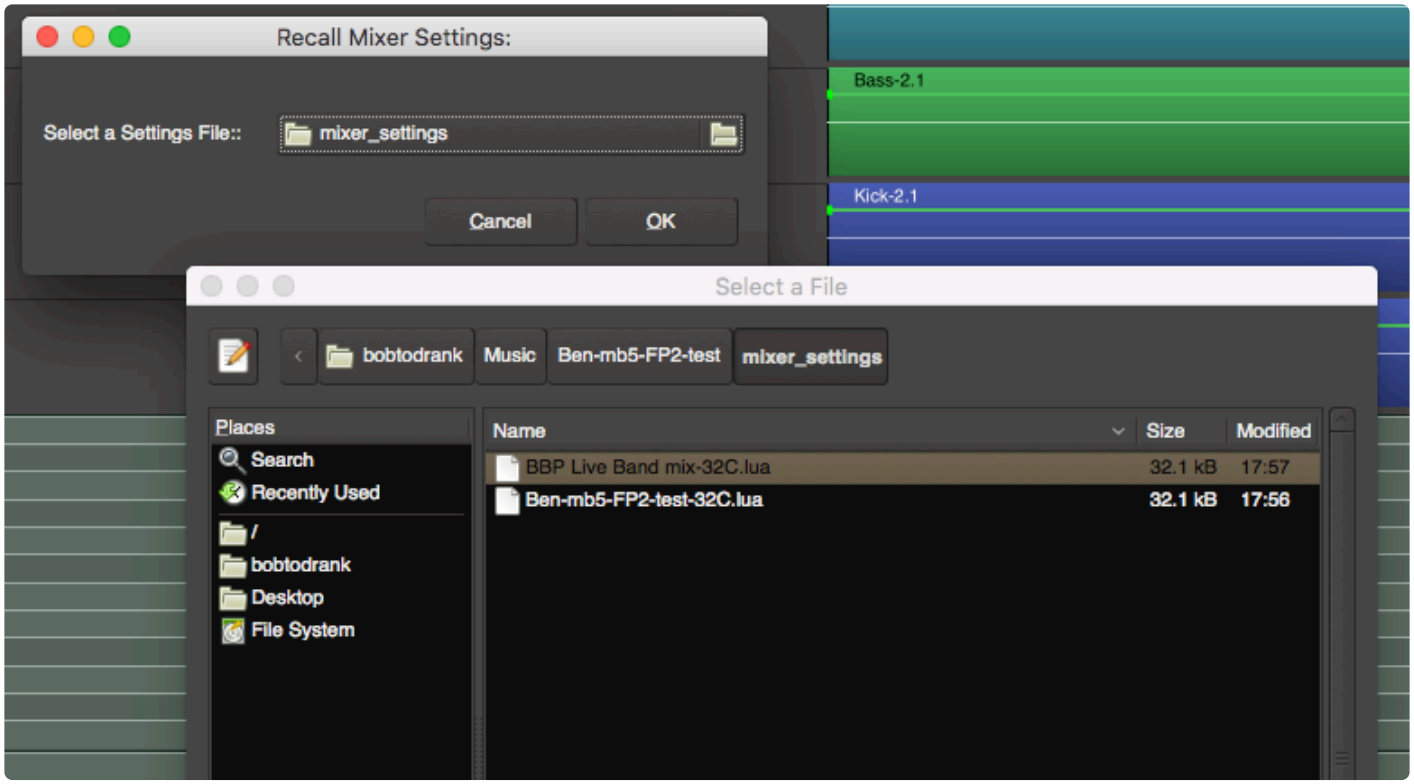
- **Settings Name** : enter the name of the file here.
- **Settings Location** : you may store the file in the Session folder, or in your globally-accessible preferences folder. Choose “Global” if you think you might want to use these settings on a different session, someday.
- **Selected Tracks Only** : if this is enabled, then only the selected (red-colored) tracks will be saved to the file.

Recall Mixer Settings

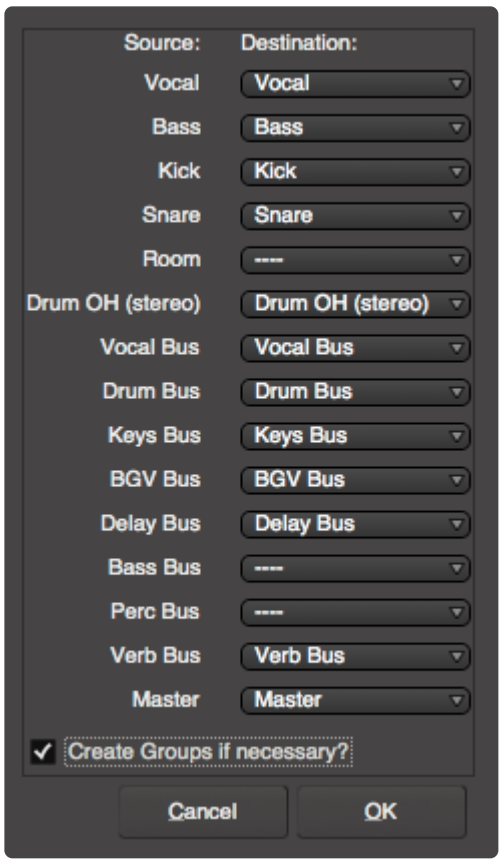


First, choose whether to load a file from your Global settings, or from within the current session.

> ! Mixer files contain the mixer-strip Input Trim, Polarity, EQ, Compressor, Panning, Mixbus Sends, Master Bus Sends, and Plugins. They do **not** include Sends or Inserts in the channelstrip.



Select the file that you would like to import.



The “Source” column displays the channels that were found in the mixer snapshot file. The “Destination” column displays the closest-matching tracks that were found in your current session. If a track with the same name isn’t found, the Destination will show “——”. If Mixbus can’t automatically find a match, you can choose the tracks that you’d like to assign from the Source.

Since tracks can be assigned to groups, there is an option to “Create groups” for any group that doesn’t already exist in your session. If you un-check this box, then Mixbus won’t create any new groups to accommodate the selected mix file.

Each of the provided actions in the “Track Tools” menu is implemented as a Lua script. It is possible to edit the script manually, to change or improve the script’s behavior. This is an advanced topic for customization. To chat with developers about this feature, visit the Mixbus user forum.

18.6. View Menu

- Maximise Editor Space : when this is enabled, the Mixbus “editor” window will try to make itself fullscreen. The exact behavior will differ depending on your OS version.
- Maximise Mixer Space
- Primary Clock >
    - Focus On Clock : the primary clock will get “focus” for text entry, so you can type in a time location.
    - Timecode : switches the primary clock to show “Timecode” format. Learn more here.
    - Bars & Beats
    - Minutes & Samples
    - Samples

- Secondary Clock >
    - Timecode : switches the secondary clock to use “Timecode” display format. Learn more here.
    - Bars & Beats
    - Minutes & Samples
    - Samples

- Zoom >
    - Zoom In : zooms the timeline to show a smaller time interval.
    - Zoom Out
    - Zoom to Session : zooms the timeline to show the entire length between the Session *Start* and *End* markers.
    - Zoom to Selection : zooms the timeline to show the length of the selected Range, AND vertically to show the track selection.
    - Zoom to Selection (Horizontal) : zooms the timeline to show the length of the selected Range
    - Fit Selection (Vertical) : zooms the editor so the track/region selection fills the vertical space of the screen.
    - Toggle Zoom State : reverts to the previous zoom state
    - Expand Track Height : increases the height of all selected tracks
    - Shrink Track Height : decreases the height of all selected tracks

- Zoom Focus >
    - Zoom Focus Left : when this is selected, Mixbus will zoom to the “left” side of the screen. (this only applies when using the -/+ shortcut keys)
    - Zoom Focus Right
    - Zoom Focus Center
    - Zoom Focus Playhead
    - Zoom Focus Mouse
    - Zoom Focus Edit Point
    - Next Zoom Focus

- Rulers >

    - Min:Sec
    - Timecode
    - Samples
    - Bars & Beats
    - Meter
    - Tempo
    - Ranges
    - Loop/Punch
    - CD Markers
    - Markers
    - Video

- Video Monitor >
    - Original Size
    - Letterbox
    - Always on Top
    - Fullscreen
    - Timecode
    - Frame Number
    - Timecode & Background

- Scroll >
    - Scroll Tracks Down
    - Scroll Tracks Up
    - Scroll Forward
    - Scroll Backwards

- Views >
    - Save View 1 : Saves the current editor “viewport” – including the timeline and track zooming.
    - Goto View 1 : recalls the “viewport” stored in View 1
    - Save View 2
    - Goto View 2
    - Save View 3
    - Goto View 3
    - Save View 4
    - Goto View 4
    - Save View 5
    - Goto View 5
    - Save View 6
    - Goto View 6
    - Save View 7
    - Goto View 7
    - Save View 8
    - Goto View 8
    - Save View 9

    - Goto View 9
    - Save View 10
    - Goto View 10
    - Save View 11
    - Goto View 11
    - Save View 12
    - Goto View 12

- Show Editor Mixer : Toggles visibility of the mixer-strip on the Edit window
- Show Editor List : Toggles visibility of the editor sidebar list, on the right side of the editor window.
- Toggle Mixer List : Toggles visibility of the mixer sidebar list, on the left side of the mixer window.
- Toggle Monitor Section visibility : Toggles the “monitor section” visibility, if one is enabled in Session->Properties->Monitoring.
- Toggle Mixbus Pane: Toggles visibility of the scrollable & reisizable “Mixbus” pane in the mixer window.
- Toggle VCA Pane: Toggles visibility of the scrollable & resizable “VCA” pane in the mixer window.
- Show Summary : Toggles the visibility of the “summary” (world) view at the bottom of the editor.
- Show Group Tabs : Toggles visibility of the “group” tabs at the top of the mixer, and the side of the editor.
- Show Marker Lines : Toggles visibility of lines below each “marker”

18.7. Window Menu

Click on an item to visit the topic

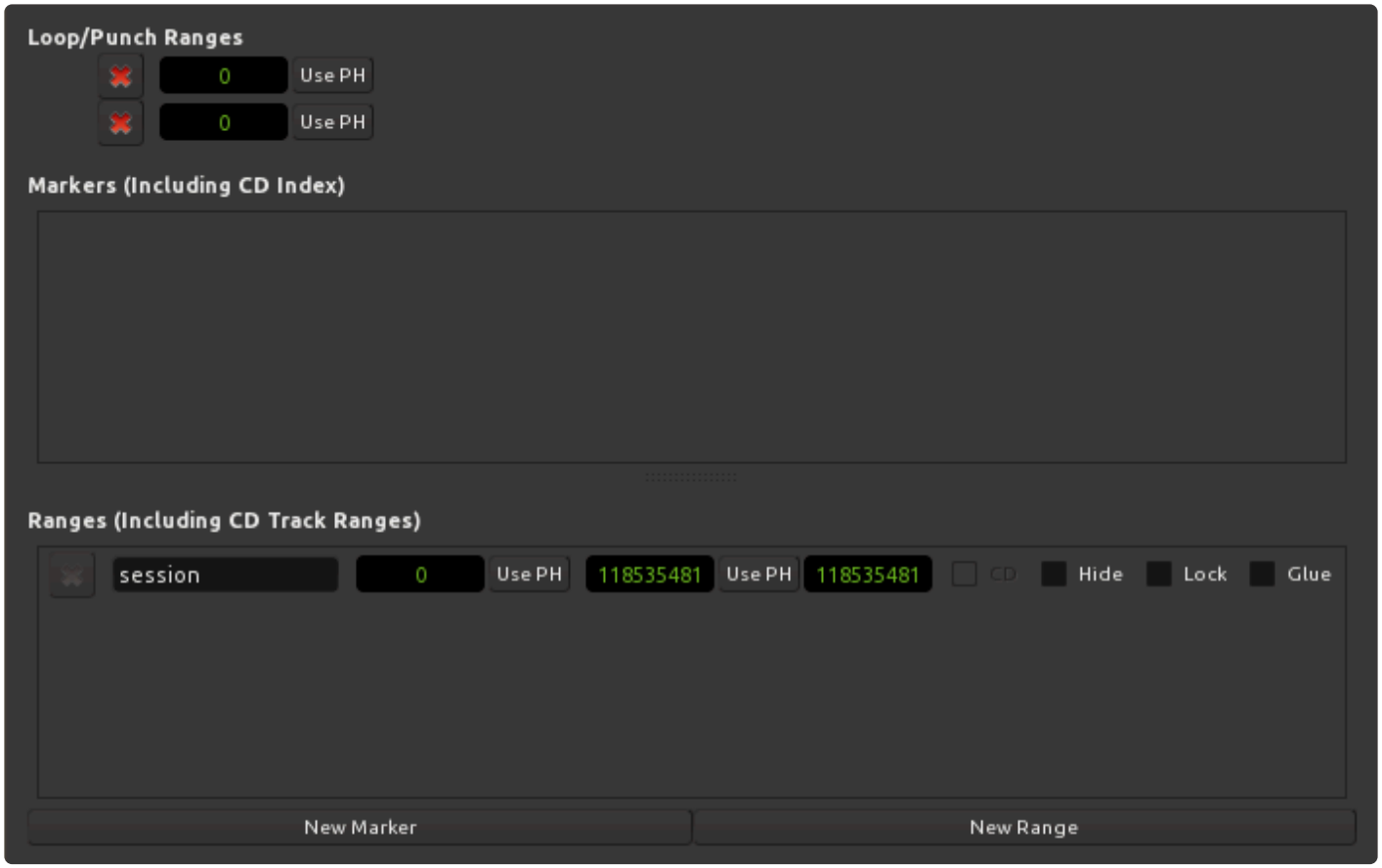
- Audio/MIDI Setup
- Editor
- Mixer
- Preferences
- Meterbridge
- Scripting
- Toggle Editor+Mixer : a shortcut to switch between the editor and mixer windows (or swap between them, if they are attached to the main window)
- Locations
- Keyboard Shortcuts
- Big Clock: A scalable clock window for distance viewing. Right Click for formats.
- Transport Controls: a resizable window that shows transport (play/stop/rewind) buttons; suitable for touchscreens
- Transport Masters: a window that displays all the available timecode inputs, their current status, and allows the user to choose which one Mixbus will chase.
- License Manager: Displays the status of your Mixbus and Harrison LV2 plugin licenses.
- Video Monitor: If the session has a video file loaded, this will display the video window for playback with Mixbus's timeline.
- MIDI Tracer: This window allows the user to view incoming and outgoing MIDI messages, for troubleshooting
- Audio Connections: More here
- MIDI Connections: A grid that allows making MIDI connections between channels, sends, and hardware I/O
- Log: The LOG window provides a history of warnings and errors which might be useful to troubleshoot problems.

18.7.1. Locations Window

The Locations window provides another view of the Location Markers, Loop Range, Punch Range, Ranges, and CD Ranges that appear on the “rulers” of the Editor window. This window allows you to see all of the existing markers in one window with all of the parameters displayed.

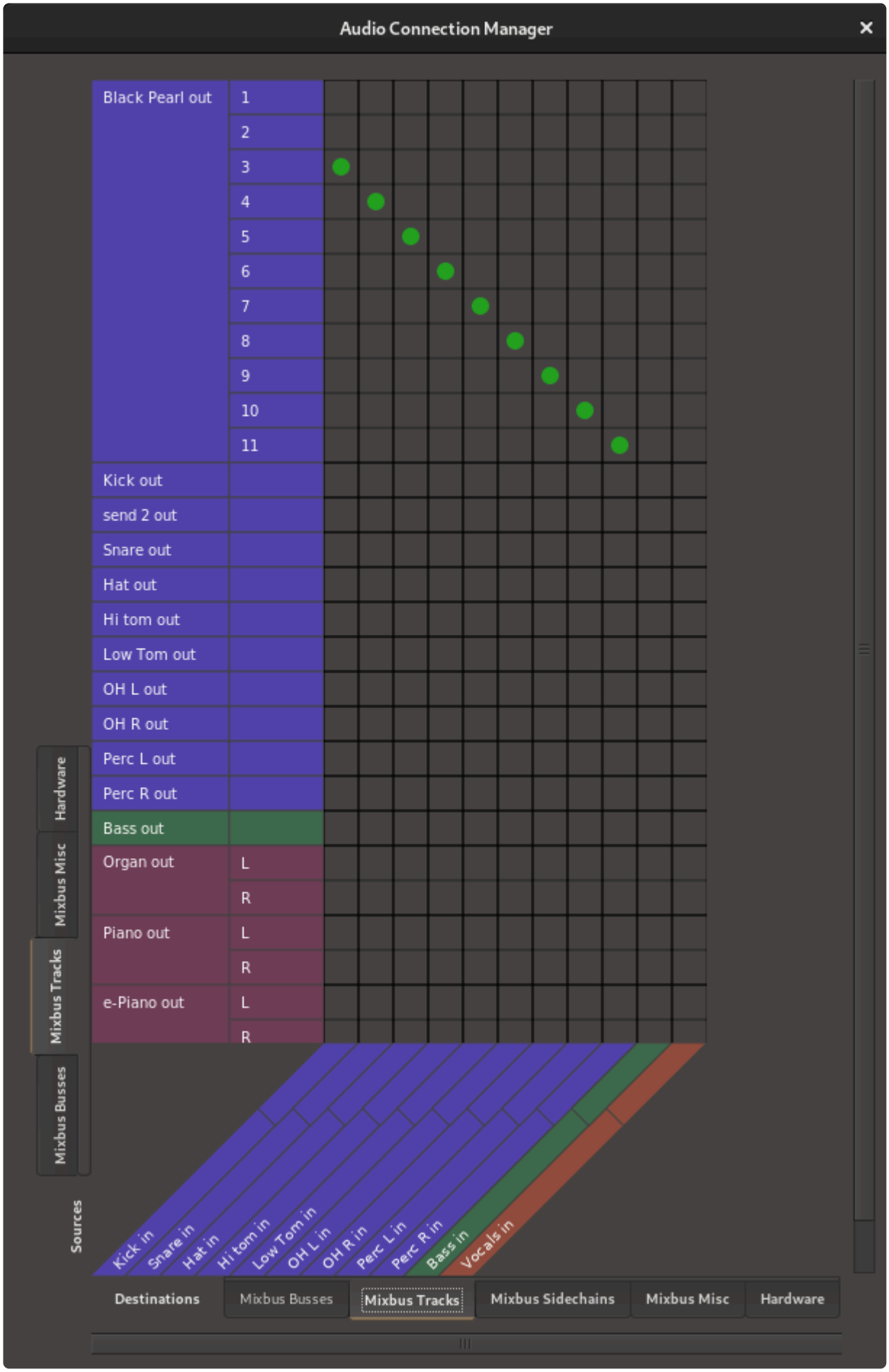
- Go: Moves the Playhead to the associated Marker or the start/end of the associated Range.
- Use PH: Assign the current Playhead location to the marker or the start/end of the associated Range.
- Hidden: Markers and ranges may be shown or hidden.
- CD: For locations, this is CD Indexes. For ranges, this indicates that the range represents a CD Track.

Items with the CD box checked will also get the ISRC, SCMS, Pre-emphasis, Performer, and Composer entry boxes which will be incorporated into the CUE/TOC file during session export and can be imported by CD burning software.



18.7.2. Audio Connections Manager Window

The Audio Connections Manager window allows you to view all active Audio port connections in your session. This is a global overview allowing you connect any source to any destinations.


Audio Connection Manager
Black Pearl out
1
2
3
4
5
6
7
8
9
10
11
Kick out
send 2 out
Snare out
Hat out
Hi tom out
Low Tom out
OH L out
OH R out
Perc L out
Perc R out
Bass out
Organ out
L
R
Piano out
L
R
e-Piano out
L
R
Hardware
Mixbus Misc
Mixbus Tracks
Mixbus Busses
Sources
Kick in
Snare in
Hat in
Hi tom in
Low Tom in
OH L in
OH R in
Perc L in
Perc R in
Bass in
Vocals in
Destinations
Mixbus Busses
Mixbus Tracks
Mixbus Sidechains
Mixbus Misc
Hardware

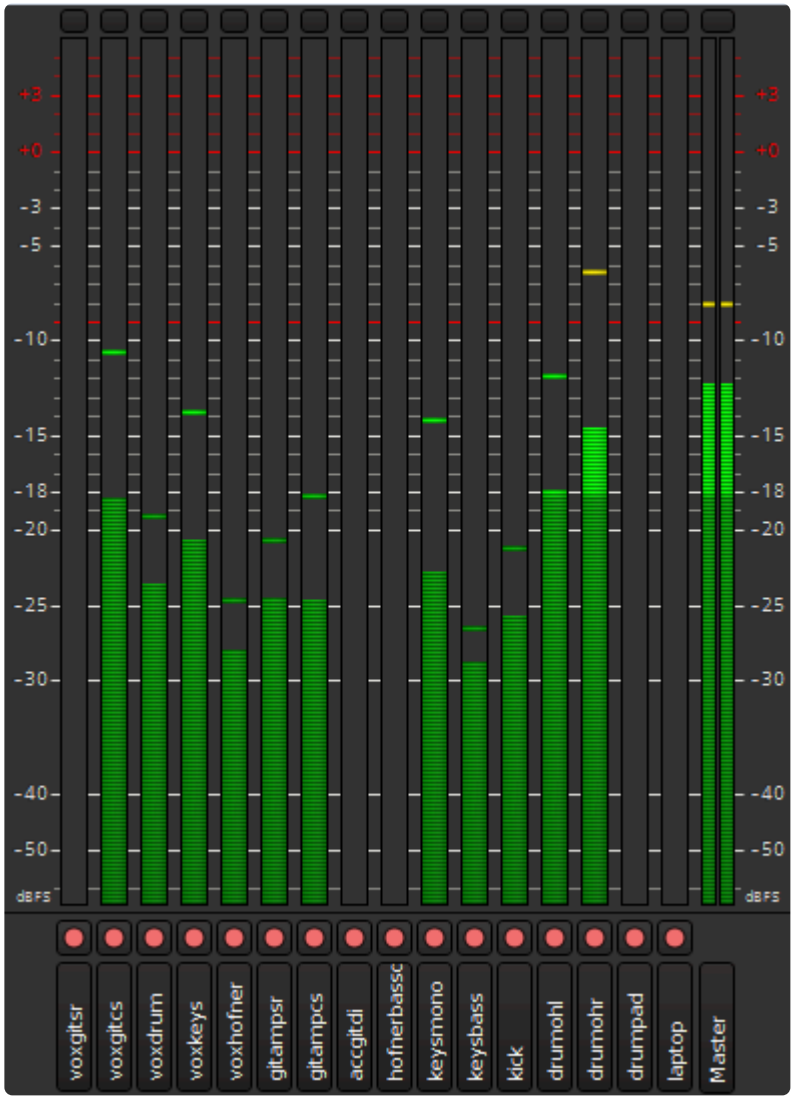
Destinations and Sources include:

- Hardware.
- Audio Tracks
- MIDI Tracks
- Audio and MIDI Busses
- Master and Monitor
- Sidechains (if any are in the session)
- Misc. tracks (Auditioner, LTC, CLick Track)

In this patch-bay-like overview of the session, click on any grid pair to connect them, and enjoy a seamless and intuitive way to connect audio throughout your session and hardware!

18.7.3. Meterbridge

The Meterbridge can be found under Window>Meterbridge and can be toggled via a check box.



The Meterbridge window is a compact and re-sizable view of all meters for all tracks, mix buses and master bus levels available during a session. 10 metering standards are available and are user selectable by right clicking on each meter.

Available metering:

- Peak
- RMS + Peak
- IEC1/DIN
- IEC1/Nordic
- IEC2/BBC
- IEC2/EBU

- K20
- K14
- K12
- VU

Most users involved with music production will find Peak metering a good choice during tracking. This meter type helps the user avoid digital clipping. This is because peak meters are calibrated to 0 dBFS (decibel full scale). When the meter reads 0 no more bits are left to capture dynamics.

> ! It is possible for a recorded signal to be clipped before being digitized and displayed on a meter. As a result, a safe level can be displayed on any meter and in any DAW. No DAW meter will show if a signal has clipped before being digitized.

Most music production users may find Peak+RMS Metering, K Metering, or VU metering useful on the master bus.

> ✻ The K system is extremely useful and is a new standard that has gained traction. K14 is suitable for the vast majority of music production. As the name suggest it allows for 14 dB of headroom. K20 allows for 20 dB of dynamic headroom and is useful for less compressed “audiophile” recordings or large movie theater soundtracks. K12 is useful for Television broadcast etc. The K metering system has been devolved by Bob Katz.

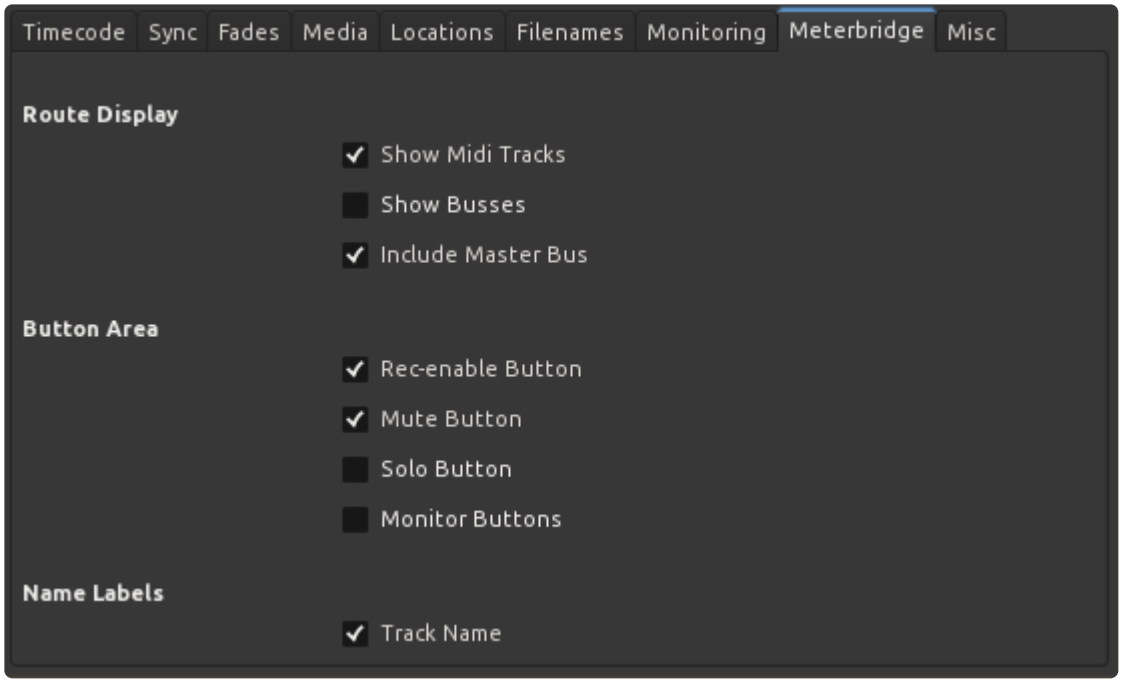
The 4 remaining metering standards are of the IEC type and are typically used for broadcast work and allows for consistency of volume within the broadcast industry.

Located at the bottom of each meter is a rec-enable button. Under the rec-enable button is the track name. Right clicking on the track name allows the user to change the height of this area. Advanced configuration allows the addition of Solo, Mute, Monitor Playback and Monitor Input.

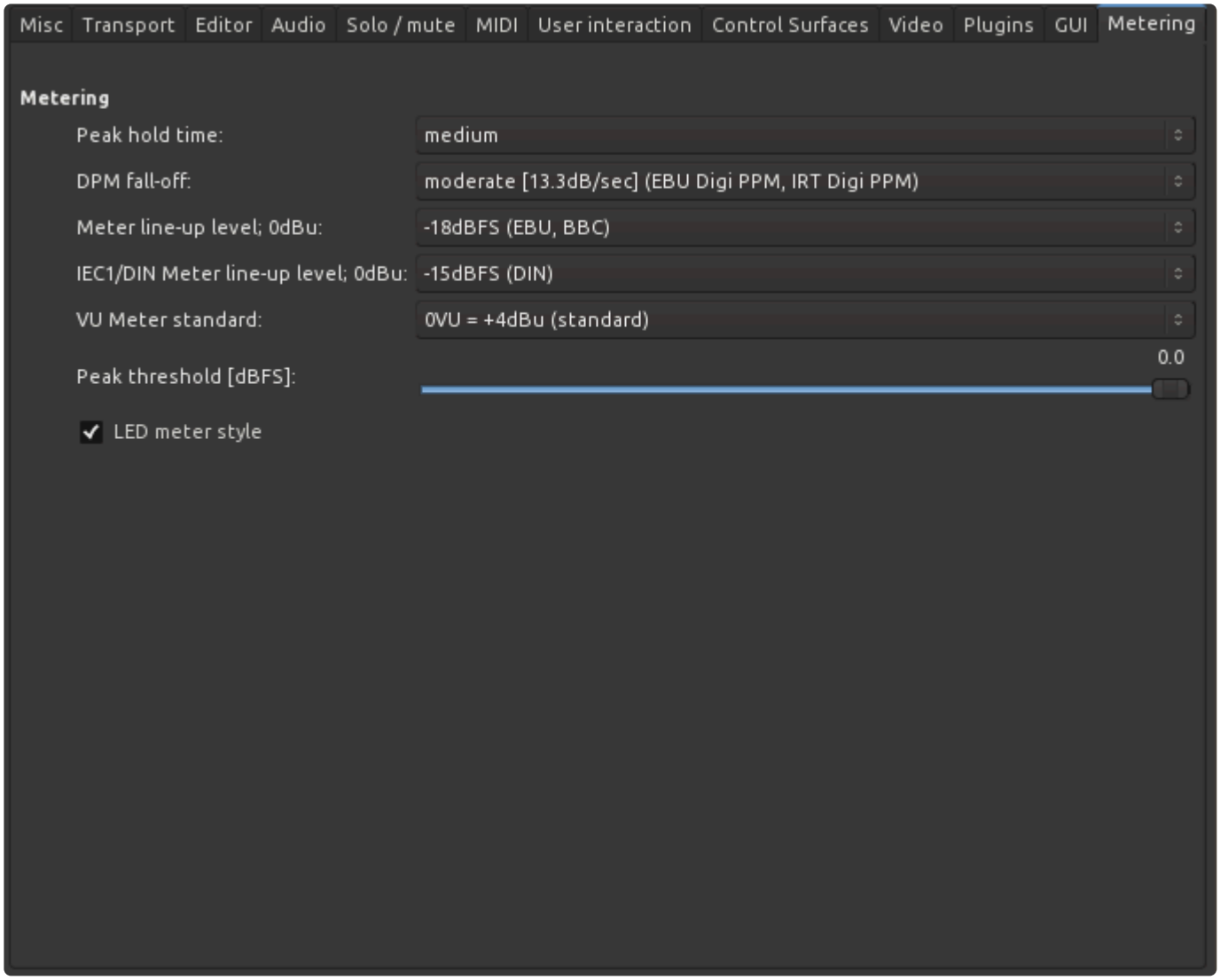
At the top of each meter is a button to reset peak warning indicators. You can Ctrl+left click to reset the group or Ctrl+Shift+left click to reset all peak warning indicators.

Meterbridge properties:

The Meterbridge is user customizable by going to to the menu Session>Properties>Meterbridge

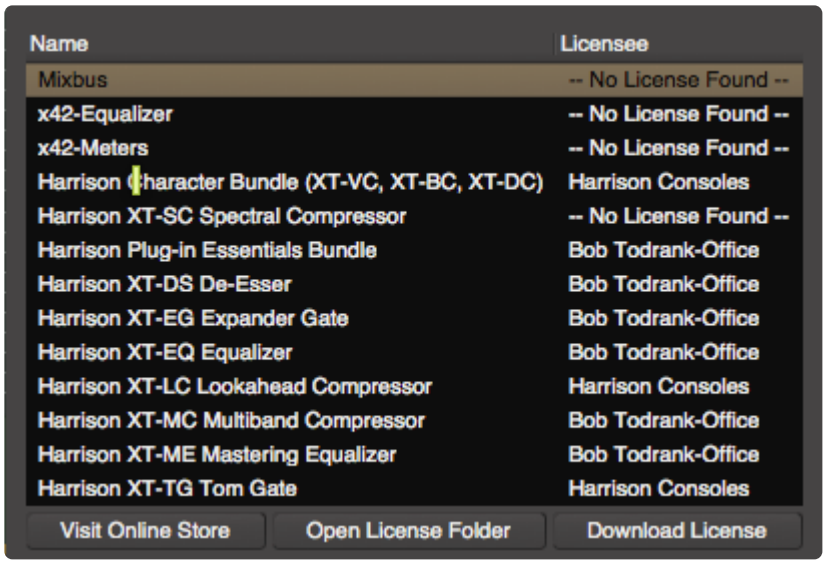


Advanced configuration of the meters is available in Edit > Preferences > Metering.


Misc
Transport
Editor
Audio
Solo / mute
MIDI
User interaction
Control Surfaces
Video
Plugins
GUI
Metering
Metering
Peak hold time:
medium
DPM fall-off:
moderate [13.3dB/sec] (EBU Digi PPM, IRT Digi PPM)
Meter line-up level; 0dBu:
-18dBFS (EBU, BBC)
IEC1/DIN Meter line-up level; 0dBu:
-15dBFS (DIN)
VU Meter standard:
0VU = +4dBu (standard)
Peak threshold [dBFS]:
0.0
LED meter style

18.7.4. License Manager

The “License Manager” dialog allows the user to view which Harrison features are currently licensed, and it also allows the user to download a new license once purchased.



Each of the available items is listed on the left, and if a license is installed, then the name of the licensee is provided on the right side (typically, you should see your own name for each product that is licensed)

Three buttons are provided to help manage licenses:

- **Visit Online Store** : if an unlicensed row is selected, this button will open a web-browser page where you can purchase a license.
- **Open License Folder** : this will open a file-browser window with the user’s HOME folder where the license files are stored.
- **Download License** : this opens a dialog. Paste any Harrison license link into the field, and Mixbus will download and save the license file to your HOME folder. (more details here)

18.7.5. Plugin DSP Load

The Plugin DSP Load Window enables the user to see how much or how little their thrid-party plugins are affecting the session's overall alloted DSP usage in milliseconds.

It can be enabled by ticking it's menu entry in Window->Plugin DSP Load. Once loaded it can be moved freely.

The DSP Load window features statistics for each plug-in covering it's:

- Minimum DSP calculation time.
- Maximum DSP calculation time
- Average DSP calculation time.
- Standard Deviation (σ).
- A meter showing where it lies between 0 and the session's current max DSP calculation time (1000 x (buffersize / samplerate))

18.8. Help Menu

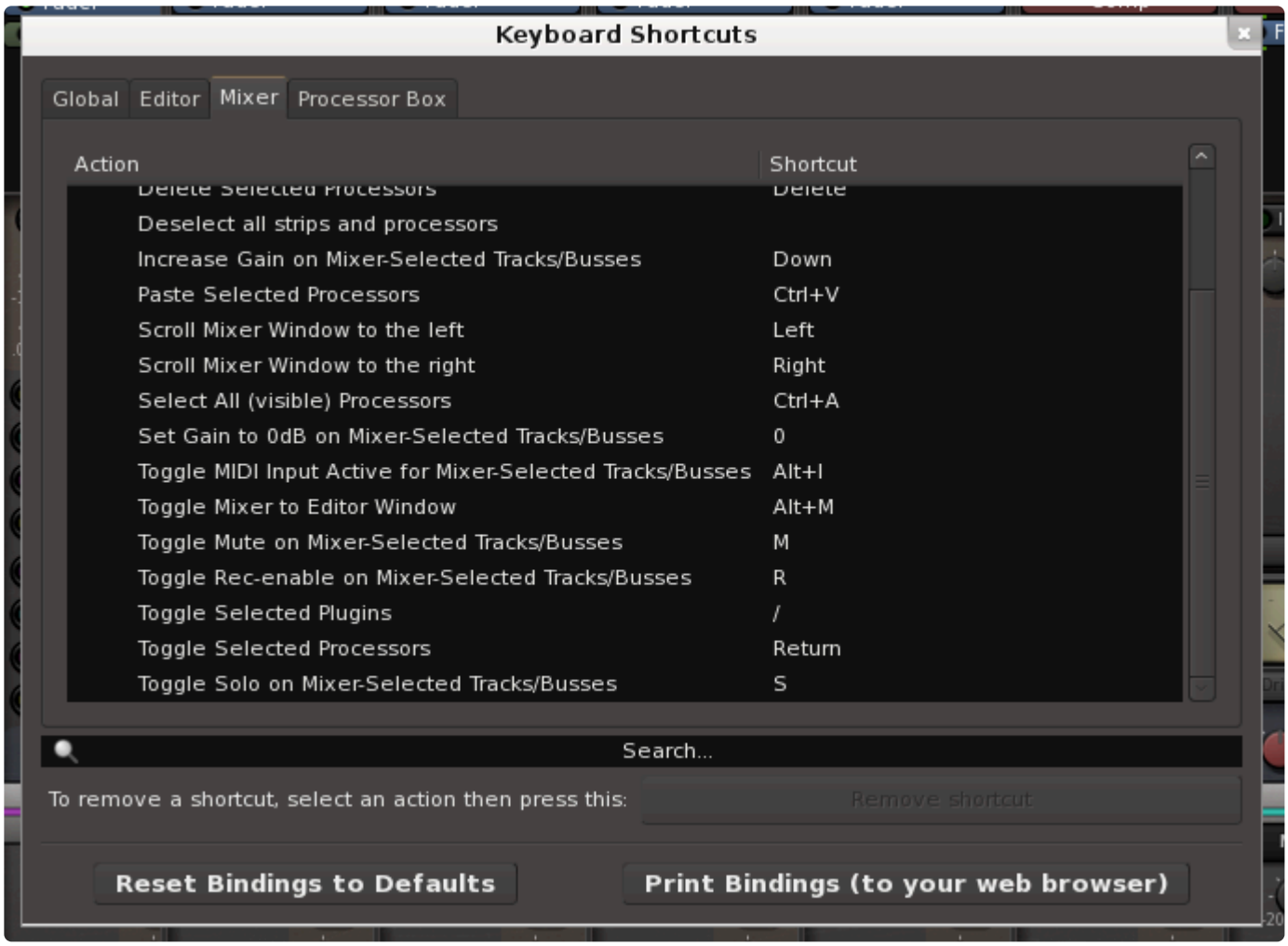
- About
- Manual: Links to this manual
- Videos: Links to Mixbus Instructional videos
- User Forums: Links to The Mixbus user forum
- Developer Chat: Links to IRC (Internet Relay Chat)

19. Keyboard Shortcuts

Any “action” found in the Mixbus main menu can be initiated from a keyboard “shortcut”.

By default, Mixbus provides a selection of keyboard shortcuts that provides a good balance between easy-to-remember mnemonic actions (such as “L” for looping) and ergonomic actions for power users (such as J and K for trimming a region’s start/end).

You can change the key shortcuts via Window->Keyboard Shortcuts in the main menu. If you change your keyboard shortcuts, your shortcuts will be stored in the user preferences folder.



Mixbus provides a set of “Global” shortcuts that work from any window. These global shortcuts generally control the Transport (which you’ll want from any window) and the window-switching functions. The Global shortcuts take precedence over the window shortcuts.

- Tabs : This selects between the “Global” shortcuts, and the window-specific (Editor, Mixer, etc) shortcuts
- Shortcut browser : Use this area to find shortcuts (categorized by their normal menu location)
    - To assign a shortcut: select an action with your mouse, and then press the keyboard shortcut

(with modifier) that you'd like to assign to that action
    - If you try to assign a keyboard shortcut that is already in use, Mixbus will tell you which action is already assigned, and prevent you from "stealing" it.
- Search bar: use this field to display only the shortcuts that match the search term. For example "play" will hide any actions that don't have the word "play" in their name.
- Remove Shortcut : click this to clear the currently-selected shortcut.
- Reset Bindings : click this button to reset ALL of your shortcuts to the factory defaults for your platform (Mac, Win or Linux)
- Print Bindings: click this to "print" a complete list of your shortcuts to a web page. (you can then make a hardcopy print of this page from your browser, if you like)

Viewing your keyboard shortcuts

Because keybindings change slightly between Mac and Windows, and because we are regularly adding new features that might require keybindings, we no longer provide the keybindings in this manual. You can see the current keybindings for any action beside the corresponding item in the main menu.

> If you'd like to print your current keyboard shortcuts, visit Window->Keyboard Shortcuts and click the button "Print Keyboard Shortcuts". This will open your internet browser window, with a list of currently active keyboard shortcuts.

The "number pad", which appears on some full-size keyboards, has some special behavior. You can read more about it here: Using a Number Pad

19.1. Using the Number Pad

Many full-size keyboards provide a “number pad” on the right side.

Locating to a marker, using the number pad

To locate to a marker, click the “**.**” (period), then type the number of the marker (for example, “12”) and then press “**.**” again. This will locate to the designated location marker (“12”, in this case). Markers are designated **by their location on the timeline**, not by their creation time or their name. So when locating with the number pad, jumping to “3” will always be later on the timeline than “2”, regardless of what the markers are named.

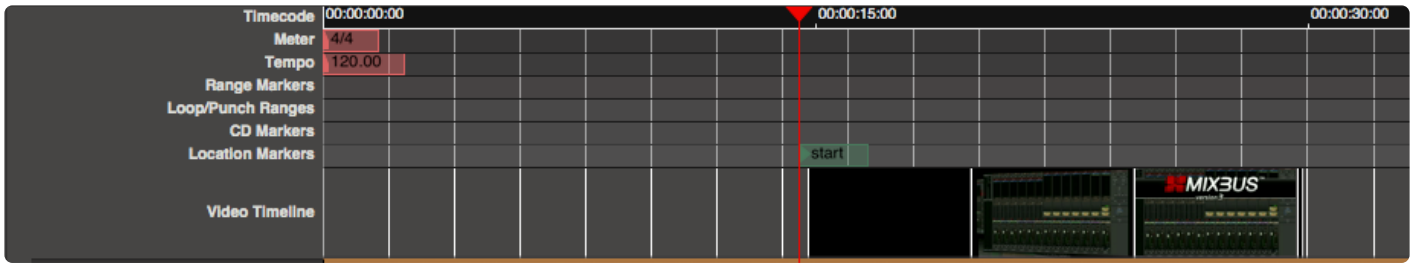
Operating the transport from the number pad

Number Pad	Action	notes
0	**play/stop**	Performs just like the spacebar.
1	**reverse play**	Click again to increase the rewind speed.
2	**forward play**	Click again to increase the forward speed.
3	**record-roll**	Arms the master record button, and initiates Play.
4	**loop-roll**	Starts playing the selected loop range (if any).
5	**loop record**	
6	**Toggle Punch On/Off**	
7	**Toggle Metronome On/Off**	
8	**Toggle “Auto Return”**	
9	**Toggle “Follow Edits”**	

The remaining Number Pad keys are reserved for future use.

20. Video

Mixbus can load a (single) video file to associate with the session. The video reference is displayed in a ruler above the editor, and (optionally) also shows moving video in a separate video window.



Mixbus does not allow editing of the video. However, you can export the video, with your audio mix, into a new video. And you can set start & stop time for that video.

Video is largely developed by the Ardour open-source community. You can learn more about the video-display features in the relevant sections of the Ardour manual: http://manual.ardour.org/video-timeline/

Video playback requires an external “server” called Harvid, and (optionally) an external video window. These utilities are included in the Mixbus package, but installation steps are different for each OS:

Installation on Windows

On Windows, the Mixbus installer provides Harvid (the video server) and Jadeo (the video playback window). To install them, select the “video tools” checkbox in the installer.

Installation on Mac

On Mac, the Mixbus application already includes Harvid (the video server). The Mixbus package provides the Jadeo video viewer as a *separate* application in the DMG. If you want to use the video window, you must drag the included XJadeo video viewer into your Application folder, so Mixbus can find it.

Installation on Linux

On Linux, the Mixbus installer provides Harvid (the video server) and Jadeo (the video playback window). No additional steps are required.

Video Formats Supported:

D = Demuxing supported
E = Muxing supported

- E 3g2 3GP2 (3GPP2 file format)
- E 3gp 3GP (3GPP file format)
- D 4xm 4X Technologies
- E a64 a64 – video for Commodore 64
- D aa Audible AA format files
- D aac raw ADTS AAC (Advanced Audio Coding)
- DE ac3 raw AC-3
- D act ACT Voice file format
- D adf Artworx Data Format
- D adp ADP
- E adts ADTS AAC (Advanced Audio Coding)
- DE adx CRI ADX
- D aea MD STUDIO audio
- D afc AFC
- DE aiff Audio IFF
- DE alaw PCM A-law
- D alias_pix Alias/Wavefront PIX image
- DE amr 3GPP AMR
- D anm Deluxe Paint Animation
- D apc CRYO APC
- D ape Monkey’s Audio
- DE apng Animated Portable Network Graphics
- D aqtitle AQTitle subtitles
- DE asf ASF (Advanced / Active Streaming Format)
- D asf_o ASF (Advanced / Active Streaming Format)
- E asf_stream ASF (Advanced / Active Streaming Format)
- DE ass SSA (SubStation Alpha) subtitle
- DE ast AST (Audio Stream)
- DE au Sun AU
- DE avi AVI (Audio Video Interleaved)
- E avm2 SWF (ShockWave Flash) (AVM2)
- D avr AVR (Audio Visual Research)
- D avs AVS
- D bethsoftvid Bethesda Softworks VID
- D bfi Brute Force & Ignorance
- D bfstm BFSTM (Binary Cafe Stream)
- D bin Binary text
- D bink Bink
- DE bit G.729 BIT file format
- D bmp_pipe piped bmp sequence
- D bmv Discworld II BMV
- D boa Black Ops Audio
- D brender_pix BRender PIX image
- D brstm BRSTM (Binary Revolution Stream)

- D c93 Interplay C93
- DE caf Apple CAF (Core Audio Format)
- DE cavsvideo raw Chinese AVS (Audio Video Standard) video
- D cdg CD Graphics
- D cdxl Commodore CDXL video
- D cine Phantom Cine
- D concat Virtual concatenation script
- E crc CRC testing
- E dash DASH Muxer
- DE data raw data
- DE daud D-Cinema audio
- D dds_pipe piped dds sequence
- D dfa Chronomaster DFA
- DE dirac raw Dirac
- DE dnxhd raw DNxHD (SMPTE VC-3)
- D dpx_pipe piped dpx sequence
- D dsf DSD Stream File (DSF)
- D dsicin Delphine Software International CIN
- D dss Digital Speech Standard (DSS)
- DE dts raw DTS
- D dtshd raw DTS-HD
- DE dv DV (Digital Video)
- D dvbsub raw dvbsub
- E dvd MPEG-2 PS (DVD VOB)
- D dxa DXA
- D ea Electronic Arts Multimedia
- D ea_cdata Electronic Arts cdata
- DE eac3 raw E-AC-3
- D epaf Ensoniq Paris Audio File
- D exr_pipe piped exr sequence
- DE f32be PCM 32-bit floating-point big-endian
- DE f32le PCM 32-bit floating-point little-endian
- E f4v F4V Adobe Flash Video
- DE f64be PCM 64-bit floating-point big-endian
- DE f64le PCM 64-bit floating-point little-endian
- DE ffm FFM (FFserver live feed)
- DE ffmetadata FFmpeg metadata in text
- D film_cpk Sega FILM / CPK
- DE filmstrip Adobe Filmstrip
- DE flac raw FLAC
- D flic FLI/FLC/FLX animation
- DE flv FLV (Flash Video)
- E framecrc framecrc testing
- E framemd5 Per-frame MD5 testing

- D frm Megalux Frame
- DE g722 raw G.722
- DE g723_1 raw G.723.1
- D g729 G.729 raw format demuxer
- DE gif GIF Animation
- D gsm raw GSM
- DE gxf GXF (General eXchange Format)
- DE h261 raw H.261
- DE h263 raw H.263
- DE h264 raw H.264 video
- E hds HDS Muxer
- DE hevc raw HEVC video
- E hls Apple HTTP Live Streaming
- D hls,applehttp Apple HTTP Live Streaming
- D hnm Cryo HNM v4
- DE ico Microsoft Windows ICO
- D idcin id Cinematic
- D idf iCE Draw File
- D iff IFF (Interchange File Format)
- DE ilbc iLBC storage
- DE image2 image2 sequence
- DE image2pipe piped image2 sequence
- D ingenient raw Ingenient MJPEG
- D ipmovie Interplay MVE
- E ipod iPod H.264 MP4 (MPEG-4 Part 14)
- DE ircam Berkeley/IRCAM/CARL Sound Format
- E ismv ISMV/ISMA (Smooth Streaming)
- D iss Funcom ISS
- D iv8 IndigoVision 8000 video
- DE ivf On2 IVF
- D j2k_pipe piped j2k sequence
- DE jacosub JACOsub subtitle format
- D jpeg_pipe piped jpeg sequence
- D jpegls_pipe piped jpegls sequence
- D jv Bitmap Brothers JV
- DE latm LOAS/LATM
- D live_flv live RTMP FLV (Flash Video)
- D lmlm4 raw lmlm4
- D loas LOAS AudioSyncStream
- DE lrc LRC lyrics
- D lvf LVF
- D lxf VR native stream (LXF)
- DE m4v raw MPEG-4 video
- E matroska Matroska

- D matroska,webm Matroska / WebM
- E md5 MD5 testing
- D mgsts Metal Gear Solid: The Twin Snakes
- DE microdvd MicroDVD subtitle format
- DE mjpeg raw MJPEG video
- E mkvtimestamp_v2 extract pts as timecode v2 format, as defined by mkvtoolnix
- DE mlp raw MLP
- D mlv Magic Lantern Video (MLV)
- D mm American Laser Games MM
- DE mmf Yamaha SMAF
- E mov QuickTime / MOV
- D mov,mp4,m4a,3gp,3g2,mj2 QuickTime / MOV
- E mp2 MP2 (MPEG audio layer 2)
- DE mp3 MP3 (MPEG audio layer 3)
- E mp4 MP4 (MPEG-4 Part 14)
- D mpc Musepack
- D mpc8 Musepack SV8
- DE mpeg MPEG-1 Systems / MPEG program stream
- E mpeg1video raw MPEG-1 video
- E mpeg2video raw MPEG-2 video
- DE mpegts MPEG-TS (MPEG-2 Transport Stream)
- D mpegtsraw raw MPEG-TS (MPEG-2 Transport Stream)
- D mpegvideo raw MPEG video
- DE mpjpeg MIME multipart JPEG
- D mpl2 MPL2 subtitles
- D mpsub MPlayer subtitles
- D msnwctcp MSN TCP Webcam stream
- D mtv MTV
- DE mulaw PCM mu-law
- D mv Silicon Graphics Movie
- D mvi Motion Pixels MVI
- DE mxf MXF (Material eXchange Format)
- E mxf_d10 MXF (Material eXchange Format) D-10 Mapping
- E mxf_opatom MXF (Material eXchange Format) Operational Pattern Atom
- D mxg MxPEG clip
- D nc NC camera feed
- D nistsphere NIST SPeech HEader REsources
- D nsv Nullsoft Streaming Video
- E null raw null video
- DE nut NUT
- D nuv NuppelVideo
- E oga Ogg Audio
- DE ogg Ogg
- DE oma Sony OpenMG audio

- E opus Ogg Opus
- D paf Amazing Studio Packed Animation File
- D pictor_pipe piped pictor sequence
- D pjs PJS (Phoenix Japanimation Society) subtitles
- D pmp Playstation Portable PMP
- D png_pipe piped png sequence
- E psp PSP MP4 (MPEG-4 Part 14)
- D psxstr Sony Playstation STR
- D pva TechnoTrend PVA
- D pvf PVF (Portable Voice Format)
- D qcp QCP
- D qdraw_pipe piped qdraw sequence
- D r3d REDCODE R3D
- DE rawvideo raw video
- D realtext RealText subtitle format
- D redspark RedSpark
- D rl2 RL2
- DE rm RealMedia
- DE roq raw id RoQ
- D rpl RPL / ARMovie
- D rsd GameCube RSD
- DE rso Lego Mindstorms RSO
- DE rtp RTP output
- E rtp_mpegts RTP/mpegts output format
- DE rtsp RTSP output
- DE s16be PCM signed 16-bit big-endian
- DE s16le PCM signed 16-bit little-endian
- DE s24be PCM signed 24-bit big-endian
- DE s24le PCM signed 24-bit little-endian
- DE s32be PCM signed 32-bit big-endian
- DE s32le PCM signed 32-bit little-endian
- DE s8 PCM signed 8-bit
- D sami SAMI subtitle format
- DE sap SAP output
- D sbg SBaGen binaural beats script
- D sdp SDP
- D sdr2 SDR2
- E segment segment
- D sgi_pipe piped sgi sequence
- D shn raw Shorten
- D siff Beam Software SIFF
- E singlejpeg JPEG single image
- D sln Asterisk raw pcm
- DE smjpeg Loki SDL MJPEG

- D smk Smacker
- E smoothstreaming Smooth Streaming Muxer
- D smush LucasArts Smush
- D sol Sierra SOL
- DE sox SoX native
- DE spdif IEC 61937 (used on S/PDIF – IEC958)
- E spx Ogg Speex
- DE srt SubRip subtitle
- D stl Spruce subtitle format
- E stream_segment,ssegment streaming segment muxer
- D subviewer SubViewer subtitle format
- D subviewer1 SubViewer v1 subtitle format
- D sunrast_pipe piped sunrast sequence
- D sup raw HDMV Presentation Graphic Stream subtitles
- E svcd MPEG-2 PS (SVCD)
- DE swf SWF (ShockWave Flash)
- D tak raw TAK
- D tedcaptions TED Talks captions
- E tee Multiple muxer tee
- D thp THP
- D tiertexseq Tiertex Limited SEQ
- D tiff_pipe piped tiff sequence
- D tmv 8088flex TMV
- DE truehd raw TrueHD
- D tta TTA (True Audio)
- D tty Tele-typewriter
- D txd Renderware TeXture Dictionary
- DE u16be PCM unsigned 16-bit big-endian
- DE u16le PCM unsigned 16-bit little-endian
- DE u24be PCM unsigned 24-bit big-endian
- DE u24le PCM unsigned 24-bit little-endian
- DE u32be PCM unsigned 32-bit big-endian
- DE u32le PCM unsigned 32-bit little-endian
- DE u8 PCM unsigned 8-bit
- E uncodedframecrc uncoded framecrc testing
- DE vc1 raw VC-1 video
- DE vc1test VC-1 test bitstream
- E vcd MPEG-1 Systems / MPEG program stream (VCD)
- D vivo Vivo
- D vmd Sierra VMD
- E vob MPEG-2 PS (VOB)
- D vobsub VobSub subtitle format
- DE voc Creative Voice
- D vplayer VPlayer subtitles

- D vqf Nippon Telegraph and Telephone Corporation (NTT) TwinVQ
- DE w64 Sony Wave64
- DE wav WAV / WAVE (Waveform Audio)
- D wc3movie Wing Commander III movie
- E webm WebM
- E webm_chunk WebM Chunk Muxer
- DE webm_dash_manifest WebM DASH Manifest
- E webp WebP
- D webp_pipe piped webp sequence
- DE webvtt WebVTT subtitle
- D wsaud Westwood Studios audio
- D wsvqa Westwood Studios VQA
- DE wtv Windows Television (WTV)
- DE wv raw WavPack
- D xa Maxis XA
- D xbin eXtended BINary text (XBIN)
- D xmv Microsoft XMV
- D xwma Microsoft xWMA
- D yop Psygnosis YOP
- DE yuv4mpegpipe YUV4MPEG pipe

Syncing Mixbus to Non-Linear Editing software

See: (Clocks & Synchronization)

21. Preferences

This page discusses the preference folder. If you would like to learn about the preference window please visit Preference Window

Your Mixbus preferences are stored in 2 places:

Session->Properties

- This stores the preferences for the current session
- For more details see Session Properties Window

Edit->Preferences (or Mixbus->Preferences on Mac OSX)

- This stores your global preferences which are applicable ALL sessions
- For more details, see Preferences Window

Preferences file location, and clearing the preferences:

Mixbus preferences are stored in the Mixbus preference folder. The location of the folder varies with each OS (Mac, Windows, or Linux)

You can clear your preferences by deleting a single file or the entire Mixbus preferences folder. If Mixbus cannot find a preferences folder or file, it will create a new one with default contents. More details about the Preferences folder can be found in Power-User Topics.

22. Clocks & Synchronization

Clocks

Mixbus provides two clocks which display the current position of the playhead. Having 2 clocks is a convenience: for example you can leave one clock showing “musical” time (such as bars/beats) while the other clock shows time in hours:minutes: seconds.

Right-click on a clock to change the mode of the clock. A menu appears with the available options.

> To learn more about clocks in the transport toolbar, visit the Transport Bar section.

> Note: There are several other clocks in Mixbus, such as the Nudge clock and the SMPTE offset clock. These clocks can also be set to any mode, so you can (for example) set the nudge clock to advance in bars:beats instead of minutes:seconds.

Some clock modes have smaller sub-displays which show the current meter/tempo, SMPTE format, or sample rate (in applicable modes).

It is possible to click on a clock and directly enter a time code location directly. Pressing “Enter” will locate the playhead to the new time.

> Tip: you can point at a clock element (minutes, seconds, beats, or whatever) and use the mouse-wheel (or 2-finger scroll on a touchpad) to quickly tweak the value without entering it manually.

Synchronization

Mixbus allows selection of several synchronization sources: Internal, LTC, MTC, and JACK

- Internal: The playhead position is defined by Mixbus; all transport commands come from Mixbus.
- LTC: The playhead position and transport state will be determined from incoming LTC (SMPTE) timecode.
- MTC: The playhead position and transport state will be determined from incoming MIDI Time Code.
- JACK: The playhead position and transport state will be determined from JACK. Transport commands initiated from Mixbus will be applied to all JACK-synchronized programs. (JACK is a third party program, and is not included with Mixbus 3, however, MB3 can utilize JACK if it is installed in your system.)

Note: A list of JACK-aware programs is listed at: http://jackaudio.org/applications

- Time Master (JACK only): Only one JACK-synchronized program can be the “Time master”. When JACK sync is used the timelines of the synchronized programs are located using the sample-clock offset. Since different programs may have different tempo maps, or methods of handling musical time, the “time master” identifies the machine that will define the musical time (bars/beats) position of the song. This is for advanced users only.

22.1. External Synchronization

Synchronization

Mixbus can synchronize to external hardware or software in three ways:

- LTC (Linear Time Code, sometimes called SMPTE) fed from an audio input.
- MTC (Midi Time Code) fed from a MIDI input
- Midi Clock (sometimes called Midi Beat Clock) fed from a MIDI input
- JACK

In all 3 cases, Mixbus can be the Master (it sends the timecode to the other devices or software) or a Slave (in which case Mixbus will “chase” the other source). In most cases, the digital workstation should be the Master device, so Mixbus will normally be the Master. But there are cases when you need to do it the other way, too.

Each method has different requirements and capabilities.

- **LTC** (sometimes called SMPTE) uses a channel of audio to encode a continuously-running timecode position. The master device generates this signal, and the slave devices use this stream to determine the master’s position and speed. LTC has the benefit of using everyday audio signals for sync, but the underlying software has to be pretty smart to decode the position. Implementations vary from very good to very bad.
- **MTC** is similar to LTC, but uses a MIDI channel to encode the position. It is much easier for developers to interpret, and it uses widely available MIDI cables and connections. However it is slightly less robust because of the decoupling between the MIDI and the audio clocks in most computers.
- **JACK** allows 2 or more software applications on your computer to share their timecode position and speed, without any audio or MID connections required by the user. JACK synchronization has several benefits over MTC & LTC, but it is not likely to be available on most commercial software, or external gear like a drum machine.

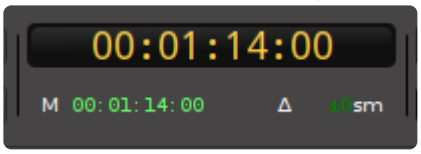
The “Internal/External Sync” button enables or disables external chase. If external sync is disabled, the button shows “Internal”, and Mixbus will use its own internal clock and transport controls.

Regardless of the “Internal/External Sync” button, Mixbus always sends sync, so other

machines can follow it.

You select the synchronization source from the Preferences. Sync source is globally selected, it doesn't change with each session you have open. This matches most people's needs, since most people have a studio setup that doesn't change depending on the session they are using.

When syncing to external timecode, you will see a display showing the current external timecode value, and a "Delta" value indicating Mixbus's offset from the master device:



Mixbus's timecode sync functions are largely developed by the Ardour open-source community. For more details, see the applicable chapters in the Ardour manual: http://manual.ardour.org/synchronization/timecode-generators-and-slaves/

22.2. Transport and Timecode Sync

External transport is composed of 2 different mechanisms:

1. **Transport position** (for example: measure 4, beat 2, or SMPTE location 00:05:01.5)
2. **Transport speed** (for example: playing, stopped, or rewinding)

Methods of Transport and Timecode Sync using MIDI

MMC (“midi machine control”) sends commands like Play, Stop, Locate, and Rewind when they are initiated by the master.

MTC (“midi transport control’”) sends a continuous stream of the master’s position. From this, the playback speed can be derived.

Often MMC and MTC are combined in a single MIDI wire, to provide actions like “locating”, as well as continuous transport position of the master.

It is also common to send MMC commands from multiple machines to the Master machine. in this way, pressing the “Play” button on a remote machine can tell Mixbus to start playing. Mixbus then sends MTC timecode to all slave machines so they sync to the master’s position.

Methods of Transport and Timecode Sync using audio

LTC (“Linear Time Code”) typically uses the SMPTE protocol to provide the current position, like MTC. It is sent at a higher rate than MTC and can therefore provide more accurate time sync. LTC requires an audio port on the master & slave devices. LTC is common among professional video equipment, but not very prevalent among musical devices such as drum machines.

Factors that affect quality of synchronization

It is recommended that the digital recorder (usually Mixbus) is the master device, and sends a sync signal to the slave device(s).

To maintain “perfect” sync between 2 devices, you should use digital devices that are driven by a common master sample-rate clock. If 2 devices are using the same digital clock, then once sync is achieved, both devices will play at the same rate. This is why it is important to use a master sample-clock when possible.

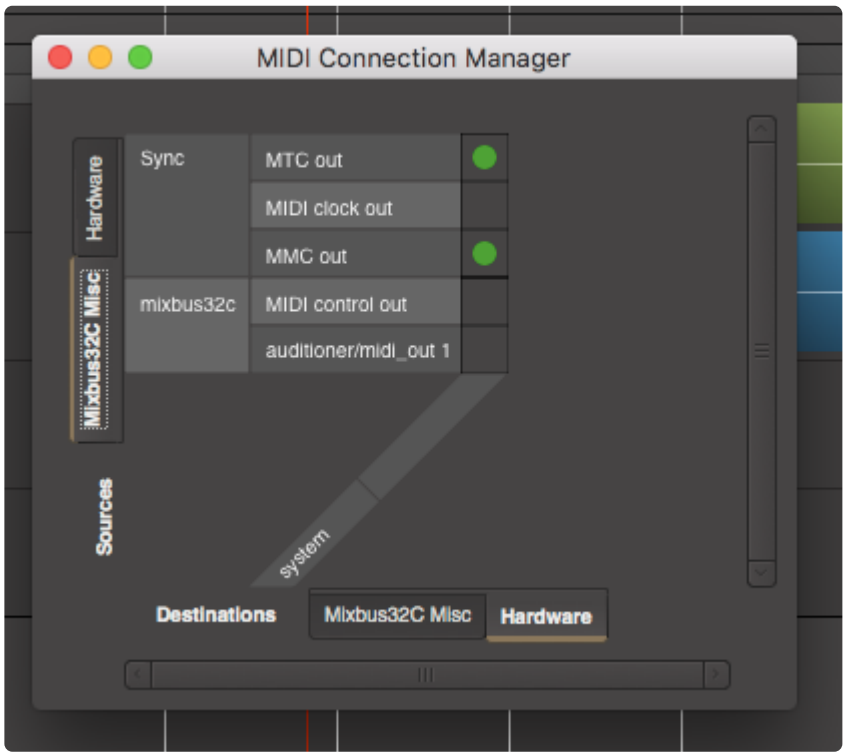
The quality of the synchronization will be heavily dependent on the quality of the sync signal from the master machine, and the ability of the slave machine to follow the sync signal.

MIDI input&output from a computer suffers from problems of “latency” (delay) & “Jitter” (changing delay). This is worst with USB-connected devices which are subject to the USB protocol buffering. For this reason,

Generating an external clock from Mixbus (Mixbus is Master)

This method is preferred when you'd like to sync an external device (like a drum machine) to Mixbus. This is the most common use for MTC/MMC synchronization.

By default, Mixbus is already configured to send MTC/MMC. To send the sync signal to an external device, the user must only connect Mixbus's MTC+MMC ports to the appropriate MIDI wire. To do that, click on Window->MIDI Connections. Then select "Mixbus Misc" on the left tab, and "Hardware" on the bottom tab. Then click in the grid to make connections, as shown below:



Syncing Mixbus to an external clock (Mixbus is Slave)

Mixbus can respond to external timecode control (via MMC or OSC) or timecode sync (via MTC or LTC/SMPTE audio)

External Sync is initiated with the "Internal/External" sync button, to the right of the transport buttons on the editor.

Synchronization sources and options are set in the Preferences Window

When slaved to an external clock, Mixbus uses a high quality DLL (delay-locked-loop) to stabilize the incoming signal. This allows Mixbus to synchronize closely with an external signal, even if it has a lot of jitter.

If the sync source uses the same digital master clock as Mixbus, then once sync is achieved, they are guaranteed to continue playing & recording at the same rate.

If the 2 sources do NOT have the same sample-rate clock, then Mixbus's behavior changes between playback and recording:
1) When playing back material, Mixbus will speed up & slow down to match the incoming clock position.
2) When recording material, Mixbus will achieve sync when recording starts, and recording will continue at the sample-clock rate, regardless of the changes in the incoming clock position.

Also see External Synchronization

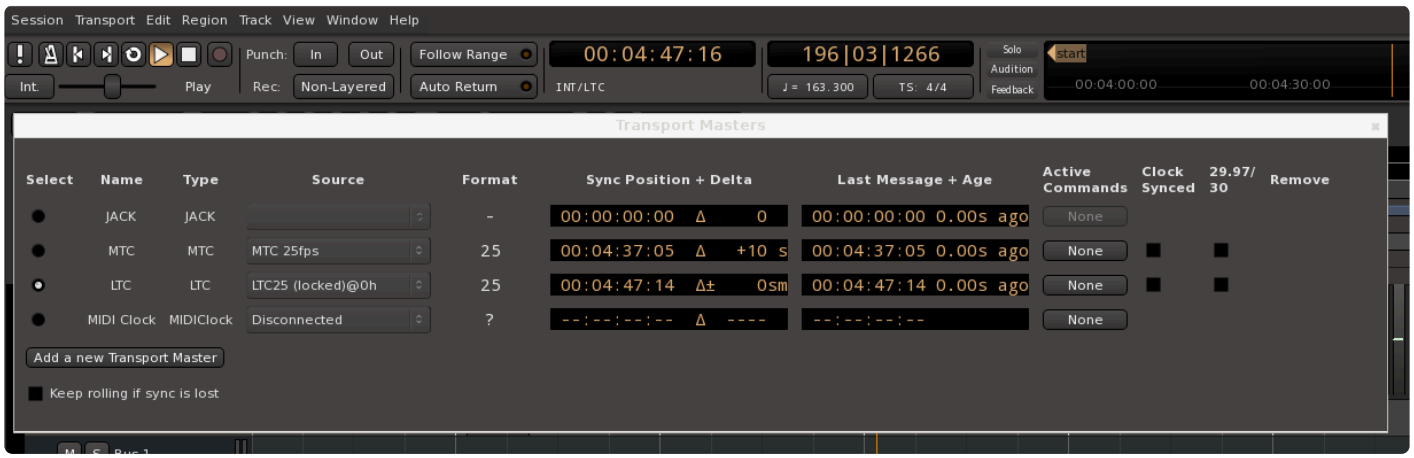
Mixbus's transport and timecode sync functions are largely developed by the Ardour open-source community. For more details, see the applicable chapters in the Ardour manual:

- http://manual.ardour.org/synchronization/timecode-generators-and-slaves/
- http://manual.ardour.org/synchronization/overview-of-timecode-related-settings/

22.3. Transport Masters

v6 adds extra functionality to clock synchronization from external sources.

To access the Transport Masters dialog, go to the Window menu and choose “Transport Masters”.



In the window the user can select the Mixbus clock source. Four choices are available:

- JACK
- MTC (MIDI Time Code)
- LTC (Linear Time Code) (SMPTE)
- MIDI clock

Controls

- **Select:** select which of the timecode inputs that Mixbus will chase
- **Name:** in the case of multiple timecode sources of the same type, the user may assign a name here.
- **Type:** one of the available sync options: LTC, MTC, or Midi Clock
- **Source:** Select the appropriate audio (for LTC) or MIDI (for MTC/Midi Clock) input
- **Format:** a display of the incoming timecode format. Determined automatically. This cannot be changed.
- **Sync Position + Delta:** the current timecode position of the sync engine, and its delta with respect to this timecode source (in seconds, milliseconds, or samples)
- **Last Message + Age:** the value of the last message received at the input, and the time it was received (if this isn’t changing then it suggests your input is not clean)
- **Active Commands:** some TC formats can send transport commands embedded in the timecode. Set the options you’d like to activate here.
- **Clock Synced:** if the timecode source uses the same digital clock as Mixbus, then you should select this option. It allows Mixbus to sync more quickly and closely since we know the speeds of the 2 devices are identical.
- **29.97/30 :** check this box to indicate that the timecode source is 29.97
- **Remove** (only available for user-added timecode sources)

Add a new transport master: Click this button to create a new LTC, MTC, or Midi Clock source. You may have multiples of each source, and choose which one to chase.

Keep Rolling if sync is lost: in an important recording, you might prefer for Mixbus to continue playing/recording even if the sync input is lost or no longer matches Mixbus's transport state.

> Previous versions of Mixbus would not permit recording during varispeed operation. Starting in v6, Mixbus will varispeed to chase the clock source selected, and allow recording while varispeeding.

23. External Control

Mixbus can respond to external controllers using 4 different protocols:

Generic MIDI (also called “MIDI Learn”)

Mixbus can accept MIDI control-messages from any MIDI-enabled device that sends CC (continuous controller) or NRPN messages. Many MIDI keyboards include MIDI controller knobs and switches that can be bound to Mixbus mixer-strip controls or transport controls.

Mixbus can use controls directly by using “MIDI Learn”. With this method, you tell Mixbus exactly what control you want to adjust on-screen, and then you turn your physical controller to “assign” it to that knob.

Alternatively, you can use a pre-defined “Binding map” which preconfigures your device’s controls to work with Mixbus. You can choose from a predefined list, or you can create your own binding file, and share it with other Mixbus users.

Mackie MCU

The “Mackie Control” protocol allows controllers with motorized faders (typically 8 or more) to control the Mixbus mixer.

Presonus Faderport

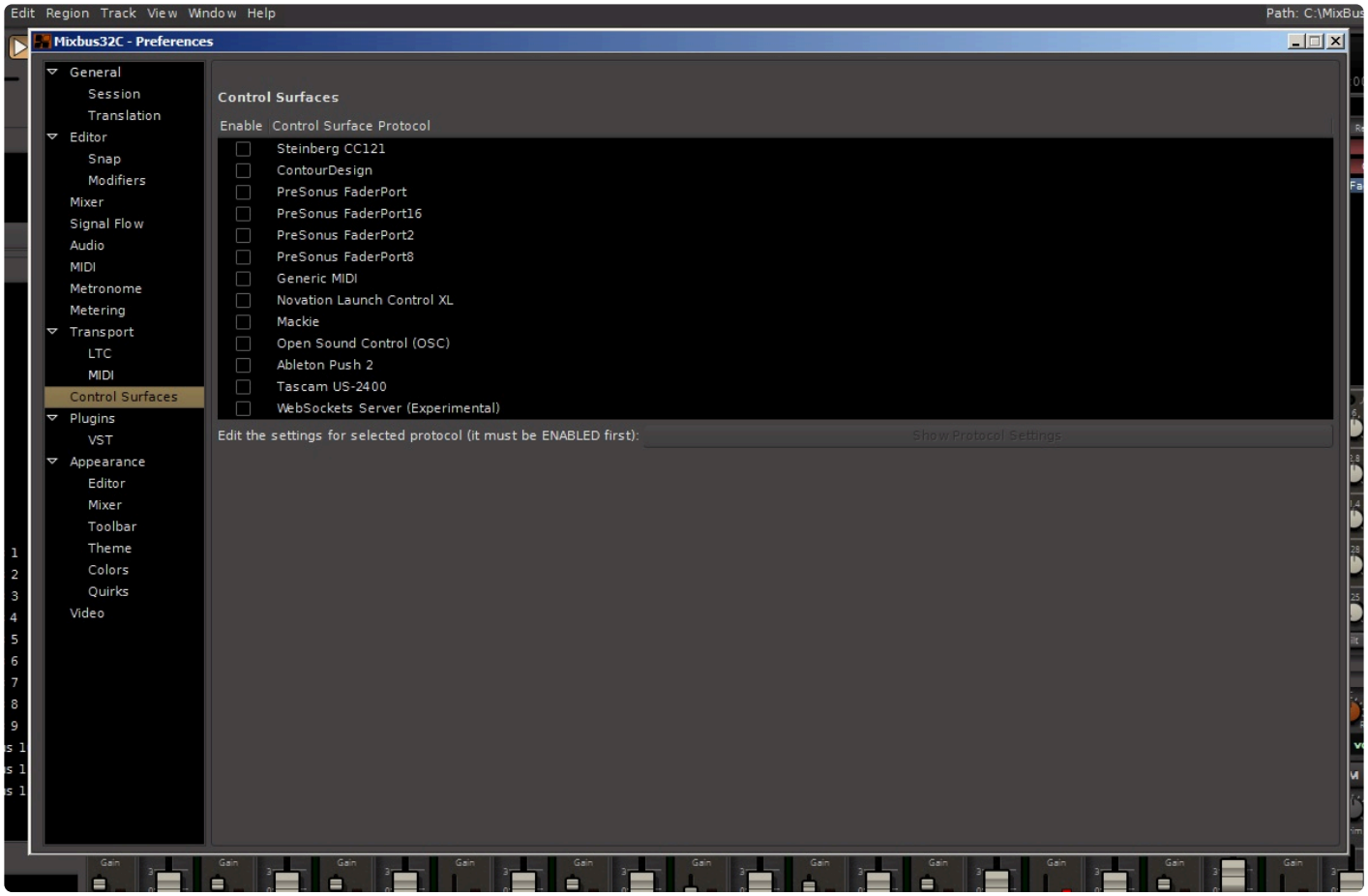
The Presonus Faderport is an inexpensive controller with 1 motorized fader, one knob, and several buttons. Mixbus provides a dedicated control mode for the Faderport.

OSC control

OSC (Open Sound Control) is a network protocol that allows you to control Mixbus from an ethernet-connected controller.

Enabling External Controllers

Mixbus will recognize a number of external controller protocols, but the ones you need must be enabled in Edit -> Preferences -> Control Surfaces.



Enable the controller protocol you need by checking the box for the controller you have.

23.1. Mackie MCU-compatible fader controllers

https://www.youtube.com/embed/ywl_YMh8j9A?rel=0

https://www.youtube.com/embed/RoNzXnUNrDQ?rel=0

Mixbus supports fader controllers that use the Mackie “MCU” protocol

To use a Mackie control device, you must follow these steps:

1. Consult the device’s manual, and configure the device for “Mackie Control” protocol (not HUI)
2. Open Preferences->Control Surfaces, and click the checkbox to enable “Mackie”
3. Double-click the Mackie line to open the configuration panel.
4. Choose your device, or the closest equivalent, from the “Device Type” list
5. Choose the MIDI input and output ports for the device

…. after step 5, the faders should immediately jump to their correct position.

Because the capabilities of fader controllers vary widely, you may find that some buttons don’t exist, or don’t work as expected, on your device. The documentation below applies to the original Mackie MCU, and very-similar devices such as the Behringer X-Touch.

Encoder Assign Buttons:

The Encoder Assign buttons define what the knobs above each fader do.

- Track: To use the Track controls, you must first “select” a channel using a channel Select button. The 8 knobs will switch to show the channel’s Input trim, solo isolate, solo lock, and polarity switches.
- Pan/Surround: This is the default setting. When this is selected, each channel’s knob will control its pan. Clicking the knob will toggle the master bus assign on/off.
- EQ: To use the EQ controls, you must first “select” a channel using a channel Select button. The 8 knobs will switch to show the selected channel’s EQ settings (this varies for channels vs mixbuses vs master bus)
- Send: To use the Send controls, you must first “select” a channel using a channel Select button. The 8 knobs will switch to show the channel’s mixbus send levels. You can adjust the send level with a knob, or click the send to enable/disable it. You may also click the “Flip” button to control the mixbus send levels from the fader.
- Plug-In: this feature is currently unimplemented.
- Inst: For Mixbus, “Inst” is used to control the channel compressor. To use the Compressor controls, you must first “select” a channel using a channel Select button. The 8 knobs will switch to show the selected channel’s compressor settings.

! The EQ, Send, and Dyn (“Inst”) modes will follow the selected channel. Selecting EQ mode, and then selecting a different channel, will continue to show the EQ knobs, but for the new channel. This is a very nice way to advance through your channels and adjust each channels EQ, in turn.

Global View Buttons:

The Global View buttons select which subset of available channels will be visible on the fader surface.

- Global View : this resets the view to showing channels in sequential order, with the first available mixer channel on the far left of the surface.
- MIDI Tracks: this shows only the MIDI (instrument) tracks on the surface. Audio tracks and buses will be skipped over, if you adjust the Channel or Bank buttons.
- Inputs: This shows “buses” that were created with the New Track dialog. Buses are often used as live inputs to the mixer. MIDI tracks, audio tracks, and mixbuses will be skipped over, if you adjust the Channel or Bank buttons.
- Audio Tracks: this shows only the Audio tracks on the control surfaces. Midi tracks and buses will be skipped over, if you adjust the Channel or Bank buttons.
- Audio Inst: This operates the same as MIDI Tracks button.
- Aux: This operates the same as the Inputs button.
- Buses: This shows the mix buses.
- Outputs: Currently unimplemented; this will show the Click (metronome), Auditioner (solo bus), and Monitor buses in the future.
- User: This shows the first 8 channels that are “selected” in the mixer GUI. This allows you to select a discontiguous collection of channels, and control them side-by-side on the control surface.

Function Buttons:

The 8 Function buttons can be assigned any “action” (keyboard shortcut) in the Macke setup dialog.

Modify:

The Modify buttons are used, similar to modifiers on your keyboard, to add additional features to the Mackie protocols. For example, Shift+Undo is “Redo”. Note that the Mackie modifier keys do NOT affect what you are doing with the keyboard/mouse. The “Shift” key is largely used for default functions. The “Control, Command, and Alt” buttons are reserved for users to assign their own actions.

Automation:

The Automation buttons apply to the selected track’s fader. The “Group” button currently turns automation OFF. “Latch”, and “Trim” are currently disabled, but these features will be added in the future.

Utility:

- Save: Saves the currently-opened session snapshot
- Undo: un-does the most recent editor operation (such as a recording). Shift+Undo to “redo”.
- Cancel: Deselects all tracks and objects.
- Enter: Selects all tracks.

Transport:

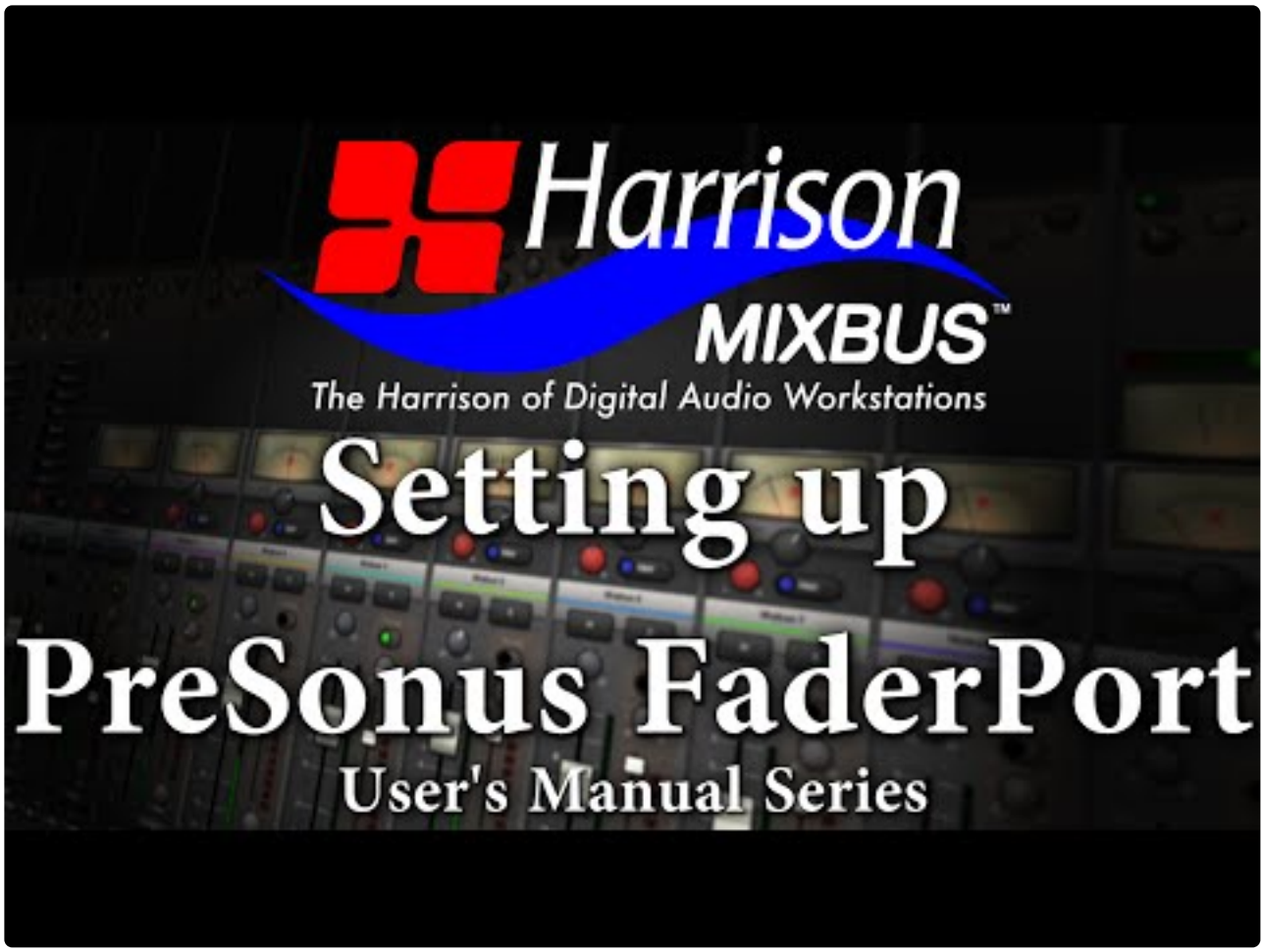
- Marker: Creates a new marker. Shift+Marker deletes the marker (if one is currently under the playhead). Marker+ffwd = next marker; Marker+rewd = last marker
- Nudge: Combined with the ffwd/rewind buttons, this will nudge the playhead forward or back.
- Cycle: This initiates a “loop” playback between the Loop markers, if you are currently playing. Shift+cycle = use currently selected range as the Loop range.
- Drop: sometimes called “Mark In”: This uses the current playhead location to set the selection start time. Shift+Drop = Toggle Punch IN.
- Replace: sometimes called “Mark Out”: This uses the current playhead location to set the selection end time. Shift+Replace = Toggle Punch OUT.
- Click: this toggles the Click (metronome) on and off. Shift+Click = Set Punch range to current range selection.
- Solo: This clears any track solo’s that are currently engaged. Shift+Solo =
- Fader Bank: this advances the faders by 8
- Channel: this advances the channels by 1
- Scrub: currently no function is assigned to this button.
- Zoom: this modifies the directional arrows and Jog Wheel to “zoom” rather than move the view.
- Directional arrows: the directional arrows move the editor view. If the Zoom button is engaged, they zoom the x and y axes.
- Jog Wheel: adjusting the wheel will move the playhead. If the “Zoom” button is engaged, then the wheel zooms in&out on the timeline

> The Mackie remote-control protocol talks to the Mixbus session directly, and isn't directly related to what you see on the screen. However it does follow the order of tracks in the mixer page, and it does not show "hidden" tracks.

The Mackie MCU protocol support is largely developed by the Ardour open-source community. To learn more about fader controllers using the Mackie MCU protocol, please visit the Ardour manual pages: http://manual.ardour.org/using-control-surfaces/devices-using-mackielogic-control-protocol/

> Mixbus does not currently support the older Mackie "HUI" protocol. Many devices provide the "HUI" protocol because it is the only 3rd-party protocol that can be used with ProTools. However, the HUI protocol is very limited. Nearly every device implements the newer "Mackie Control" protocol, which provides higher-resolution faders, more buttons, and other benefits. Mixbus uses this better protocol to talk with your device. Check your device to see which modes it provides.

23.2. Presonus Faderport (original)



https://www.youtube.com/embed/tht2eiJIjiA?rel=0

Presonus Faderport

The Presonus Faderport is a one-channel controller with a single touch-sensitive motorized fader, pan knob, and transport controls. Mixbus has a special mode for the FaderPort which provides dedicated features for this hardware.

When attached, the Faderport follows the selected (red-bordered) channel. You can change the selection with the Faderport's > and < buttons.

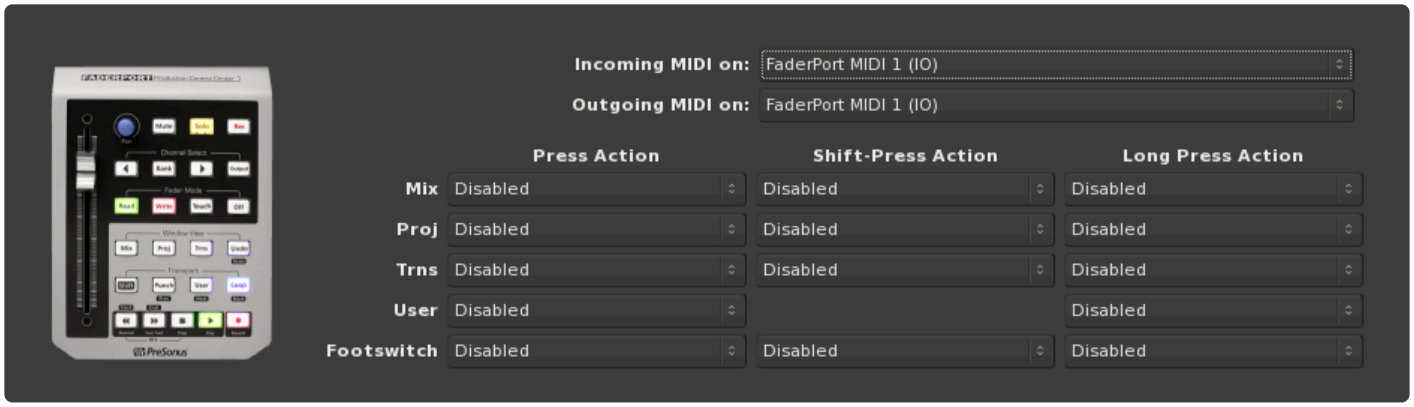
The Faderport has these controls:

- Fader
- Pan knob
- Mute / Solo / Rec-Arm
- Channel Select, including Output(*)
- Fader automation mode: Read, Write, Touch, Off

- Undo (shift+Undo to redo)
- Punch : blinks when auto-punch (in or out) is enabled
- Transport Controls including Loop, Markers, and Goto Start/End, as shown.
- *Mix*, *Proj*, *Trns*, and *User* are user-assignable buttons, see below.

The “Bank” button is currently unused.

The Faderport’s control panel provides a means to assign functions to the various buttons:



> Clicking the “Output” button will cause the Faderport’s fader and mute to control the Master bus. Holding Shift+Output will cause the Faderport to control the MONITOR bus fader and mute.

> It is not currently possible to assign the Faderport to a VCA fader, but we expect to add this feature immediately after v4.0.

Remote-control protocols are largely developed by the Ardour open-source community, and you can find documentation about the FaderPort on the Ardour website: http://manual.ardour.org/using-control-surfaces/Presonus_FaderPort/

23.3. Presonus Faderport, Faderport8 and Faderport16

Presonus Faderport8

The Presonus Faderport8 is an eight-channel controller with eight touch-sensitive motorized faders, a multi-purpose encoder knob, and transport controls. Mixbus has a special mode for the FaderPort8 which provides dedicated features for this hardware.

The Faderport8 may be used as an MCU device, but most users will prefer to use the custom “Faderport8” mode. In this mode, the Faderport8 closely matches the behavior described in the Faderport8 manual when controlling Presonus StudioOne. (Presonus and StudioOne are trademarks of Presonus).

To enable the Faderport8, visit the settings in Preferences->Control Sufaces:

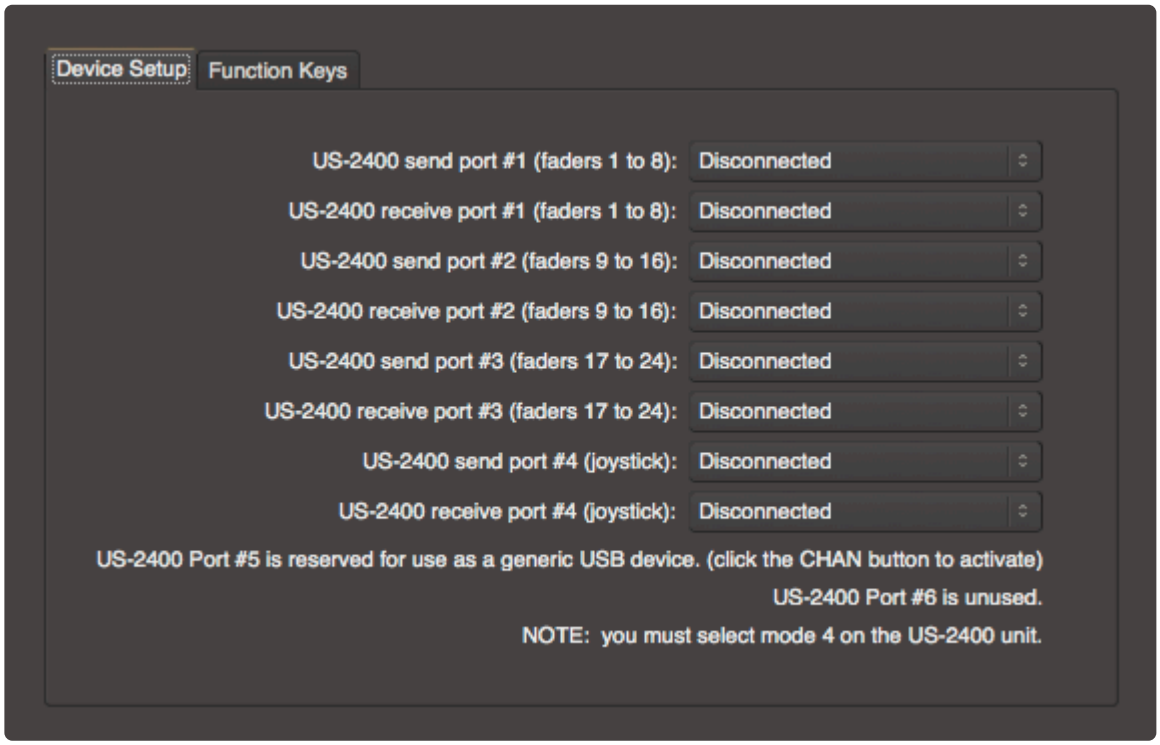
The Faderport16 operates identically to the FP8, but it provides 16 faders.

Mixbus’s Faderport8 operation was funded by Harrison, and developed largely by the Ardour open-source community. You will find up-to-date documentation for the Faderport8 operation at: http://manual.ardour.org/using-control-surfaces/PreSonus_FaderPort8

23.4. Tascam US-2400

The Tascam US-2400 was produced from roughly 2007 – 2010. It provides 24 channels, each with a fader, mute+solo+select switches, and a knob with led indications. It also has a master section with a master fader, transport controls, and function keys.

Connecting the US-2400:



To utilize the US-2400, first make sure to set the unit to mode “4”. Refer to the US-2400 manual to set modes.

The US-2400 provides 6 separate MIDI input and output ports. Select the 4 input and 4 output ports as described in the control panel.

The fifth pair of ports may be used as a “Generic MIDI controller” or for some other task.

The sixth pair of ports is unused. We believe this is likely used to light the “bank indicator LEDs” on the unit, but no documentation has been found for this feature.

Operating the US-2400:

https://www.youtube.com/embed/QHJwTncc2e4?rel=0

Each channelstrip provides a touch-sensitive fader, Mute, Solo, and Select switch. By default, the channelstrip knob provides left/right pan control of the channels. When the Select switch is lit for a channel, the channel's parameters are mapped to the knobs.

The **Master fader** has 3 special-function switches:
1) Select: this selects the Master bus and assigns its controls to the knobs, like any other channel.
2) Clr Solo: this clears any soloe'd channels and buses
3) Flip: This switches the first 8 faders (or 12 for 32C) to operate as the mixbuses instead of input channels.

Global section:

Click the "Pan" button to return to default operation, showing input channels and their associated pan knobs.

The "Meter" button switches the channelstrip knobs to show the meter level of the associated channel.

The "Chan" button switches the knobs above the faders to operate as 24 generic MIDI knobs which can send MIDI CC values.

The "Send" buttons have no operation by default. Hold "F" and press one of the 6 Send keys to activate a Function-key action. F-key actions are configured in the control panel.

Transport controls provide rewind, play, stop and master-record enable.

Bank +/- buttons are provided to increment the faders in banks of 24.

In & Out buttons are provided to trigger “Range select start” and “Range select End” function.

23.5. OSC Control

https://www.youtube.com/embed/jhr5BLH8AnM?rel=0

The OSC protocol can be used to remotely control Mixbus via Ethernet.

OSC Control provides the ability to initiate any “action” (i.e. anything that can be activated by a keybinding), as well as direct access to editor, track and bus settings. OSC can be used to “script” events in Mixbus using an external scripting platform. A complete description of OSC is beyond the scope of this manual.

There are 3 common programs used to remotely control Mixbus:

- TouchOSC – see Harrison’s TTC-1 Control Layout for TouchOSC
- Lemur – see Harrison’s LTI Control Layout for Lemur
- Open Stage Control

The following is a summary of the features that are available from OSC:

- (s) denotes a string argument

- (i) denotes an integer argument
- (f) denotes a float argument

Sending this message will trigger the associated keybinding “action” in Mixbus:

- /access_action (s)

Transport Functions:

- /add_marker
- /add_marker (f)
- /access_action s
- /loop_toggle
- /loop_toggle (f)
- /loop_location (ii)
- /goto_start
- /goto_start (f)
- /goto_end
- /goto_end (f)
- /scrub (f)
- /jog (f)
- /jog/mode (f)
- /rewind
- /rewind (f)
- /ffwd
- /ffwd (f)
- /transport_stop
- /transport_stop (f)
- /transport_play
- /transport_play (f)
- /transport_frame
- /transport_speed
- /record_enabled
- /set_transport_speed (f)

Editor functions:

- /locate (ii)
- /save_state
- /save_state (f)
- /prev_marker
- /prev_marker (f)
- /next_marker
- /next_marker (f)
- /undo

- /undo (f)
- /redo
- /redo (f)
- /toggle_punch_in
- /toggle_punch_in (f)
- /toggle_punch_out
- /toggle_punch_out (f)
- /rec_enable_toggle
- /rec_enable_toggle (f)
- /toggle_all_rec_enables
- /toggle_all_rec_enables (f)
- /all_tracks_rec_in (f)
- /all_tracks_rec_out (f)
- /cancel_all_solos (f)
- /remove_marker
- /remove_marker (f)
- /jump_bars (f)
- /jump_seconds (f)
- /mark_in
- /mark_in (f)
- /mark_out
- /mark_out (f)
- /toggle_click
- /toggle_click (f)
- /midi_panic
- /midi_panic (f)
- /toggle_roll
- /toggle_roll (f)
- /stop_forget
- /stop_forget (f)
- /set_punch_range
- /set_punch_range (f)
- /set_loop_range
- /set_loop_range (f)
- /set_session_range
- /set_session_range (f)
- /toggle_monitor_mute
- /toggle_monitor_mute (f)
- /toggle_monitor_dim
- /toggle_monitor_dim (f)
- /toggle_monitor_mono
- /toggle_monitor_mono (f)
- /quick_snapshot_switch
- /quick_snapshot_switch (f)

- /quick_snapshot_stay
- /quick_snapshot_stay (f)
- /fit_1_track
- /fit_1_track (f)
- /fit_2_tracks
- /fit_2_tracks (f)
- /fit_4_tracks
- /fit_4_tracks (f)
- /fit_8_tracks
- /fit_8_tracks (f)
- /fit_16_tracks
- /fit_16_tracks (f)
- /fit_32_tracks
- /fit_32_tracks (f)
- /fit_all_tracks
- /fit_all_tracks (f)
- /zoom_100_ms
- /zoom_100_ms (f)
- /zoom_1_sec
- /zoom_1_sec (f)
- /zoom_10_sec
- /zoom_10_sec (f)
- /zoom_1_min
- /zoom_1_min (f)
- /zoom_5_min
- /zoom_5_min (f)
- /zoom_10_min
- /zoom_10_min (f)
- /zoom_to_session
- /zoom_to_session (f)
- /temporal_zoom_in (f)
- /temporal_zoom_in
- /temporal_zoom_out
- /temporal_zoom_out (f)
- /scroll_up_1_track (f)
- /scroll_up_1_track
- /scroll_dn_1_track (f)
- /scroll_dn_1_track
- /scroll_up_1_page (f)
- /scroll_up_1_page
- /scroll_dn_1_page (f)
- /scroll_dn_1_page
- /bank_up
- /bank_up (f)

- /bank_down
- /bank_down (f)
- /use_group (f)

Controls for special strips:

- /master/gain (f)
- /master/fader (f)
- /master/db_delta (f)
- /master/mute (i)
- /master/trimdB (f)
- /master/pan_stereo_position (f)
- /master/select (f)
- /monitor/gain (f)
- /monitor/fader (f)
- /monitor/db_delta (f)
- /monitor/mute (i)
- /monitor/dim (i)
- /monitor/mono (i)

Controls for the Selected strip:

- /select/recenable (i)
- /select/record_safe (i)
- /select/mute (i)
- /select/solo (i)
- /select/solo_iso (i)
- /select/solo_safe (i)
- /select/monitor_input (i)
- /select/monitor_disk (i)
- /select/polarity (i)
- /select/gain (f)
- /select/fader (f)
- /select/db_delta (f)
- /select/trimdB (f)
- /select/pan_stereo_position (f)
- /select/pan_stereo_width (f)
- /select/send_gain (if)
- /select/send_fader (if)
- /select/send_enable (if)
- /select/master_send_enable (i)
- /select/send_page (f)
- /select/plug_page (f)
- /select/plugin (f)
- /select/expand (i)

- /select/pan_elevation_position (f)
- /select/pan_frontback_position (f)
- /select/pan_lfe_control (f)
- /select/comp_enable (f)
- /select/comp_threshold (f)
- /select/comp_speed (f)
- /select/comp_mode (f)
- /select/comp_makeup (f)
- /select/eq_enable (f)
- /select/eq_hpf/freq (f)
- /select/eq_hpf/enable (f)
- /select/eq_hpf/slope (f)
- /select/eq_lpf/freq (f)
- /select/eq_lpf/enable (f)
- /select/eq_lpf/slope (f)
- /select/eq_gain (if)
- /select/eq_freq (if)
- /select/eq_q (if)
- /select/eq_shape (if)

Track Controls:

- /strip/mute (ii)
- /strip/solo (ii)
- /strip/solo_iso (ii)
- /strip/solo_safe (ii)
- /strip/recenable (ii)
- /strip/record_safe (ii)
- /strip/monitor_input (ii)
- /strip/monitor_disk (ii)
- /strip/expand (ii)
- /strip/select (ii)
- /strip/polarity (ii)
- /strip/gain (if)
- /strip/fader (if)
- /strip/trimdB (if)
- /strip/pan_stereo_position (if)
- /strip/pan_stereo_width (if)
- /strip/plugin/parameter (iiif)
- /strip/plugin/parameter/print (iii)
- /strip/plugin/activate (ii)
- /strip/plugin/deactivate (ii)
- /strip/send/gain (iif)
- /strip/send/fader (iif)

- /strip/send/enable (iif)
- /strip/name (is)
- /strip/sends (i)
- /strip/receives (i)
- /strip/plugin/list (i)
- /strip/plugin/descriptor (ii)
- /strip/plugin/reset (ii)

Misc:

- /refresh
- /refresh (f)
- /strip/list
- /strip/list (f)

23.5.1. Lemur

What is Lemur?

Lemur is a very advanced application that lets you build custom OSC and MIDI controller interfaces using a powerful networking protocol implemented by your favorite digital audio work stations. These interfaces have numerous applications from live performance to writing and, now, mixing. It runs on both iOS and Android, allowing users of both platforms to share interfaces and run in tandem.

What is the LTI *Lemur Touch Interface*?

“LTI” is Harrison’s Lemur interface layout, developed in-house as a remote control surface for Mixbus, using the Lemur application.

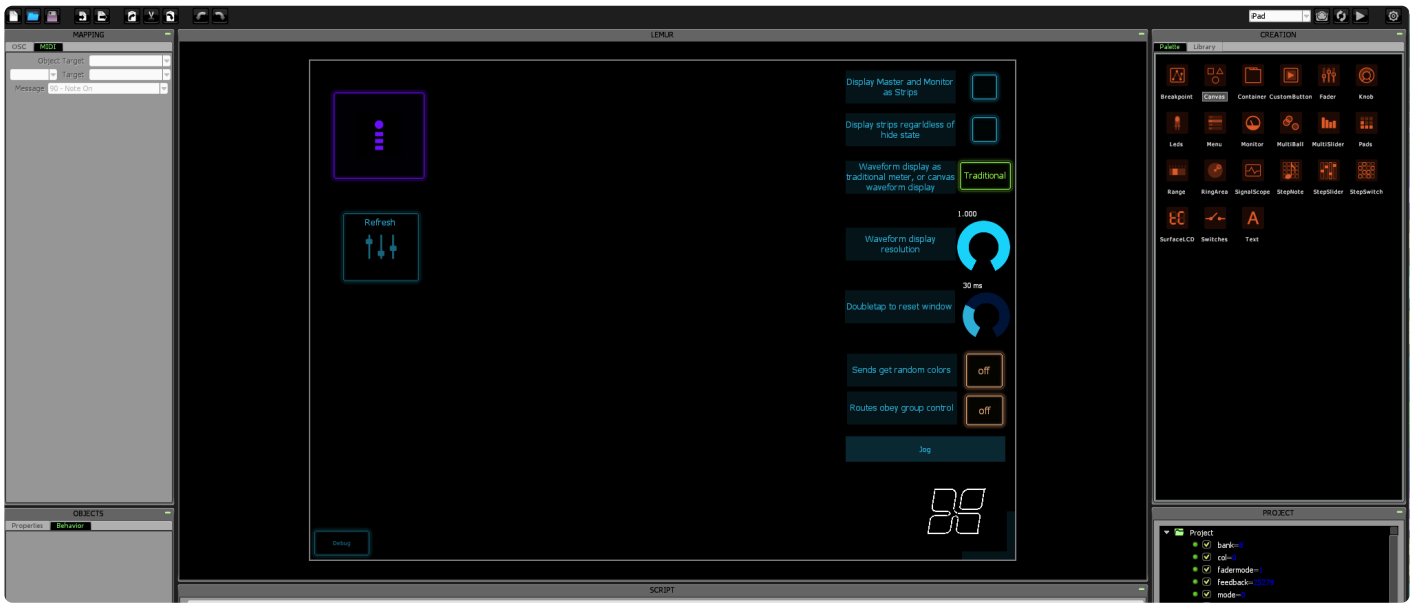
https://www.youtube.com/embed/yDde6f1eWQ4?rel=0

Because the LTI uses the Lemur editor application, the Harrison LTI controller layout may be modified by you, the end-user.

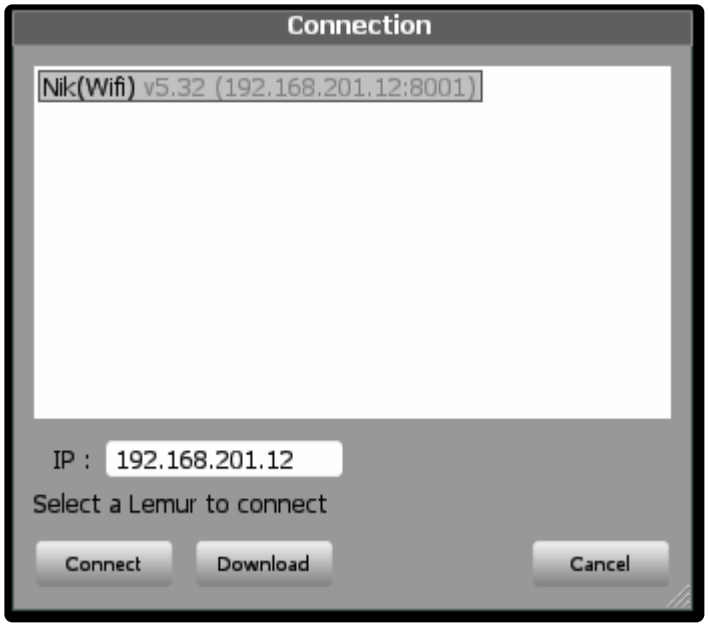
Instructions

Here are some instructions for use:

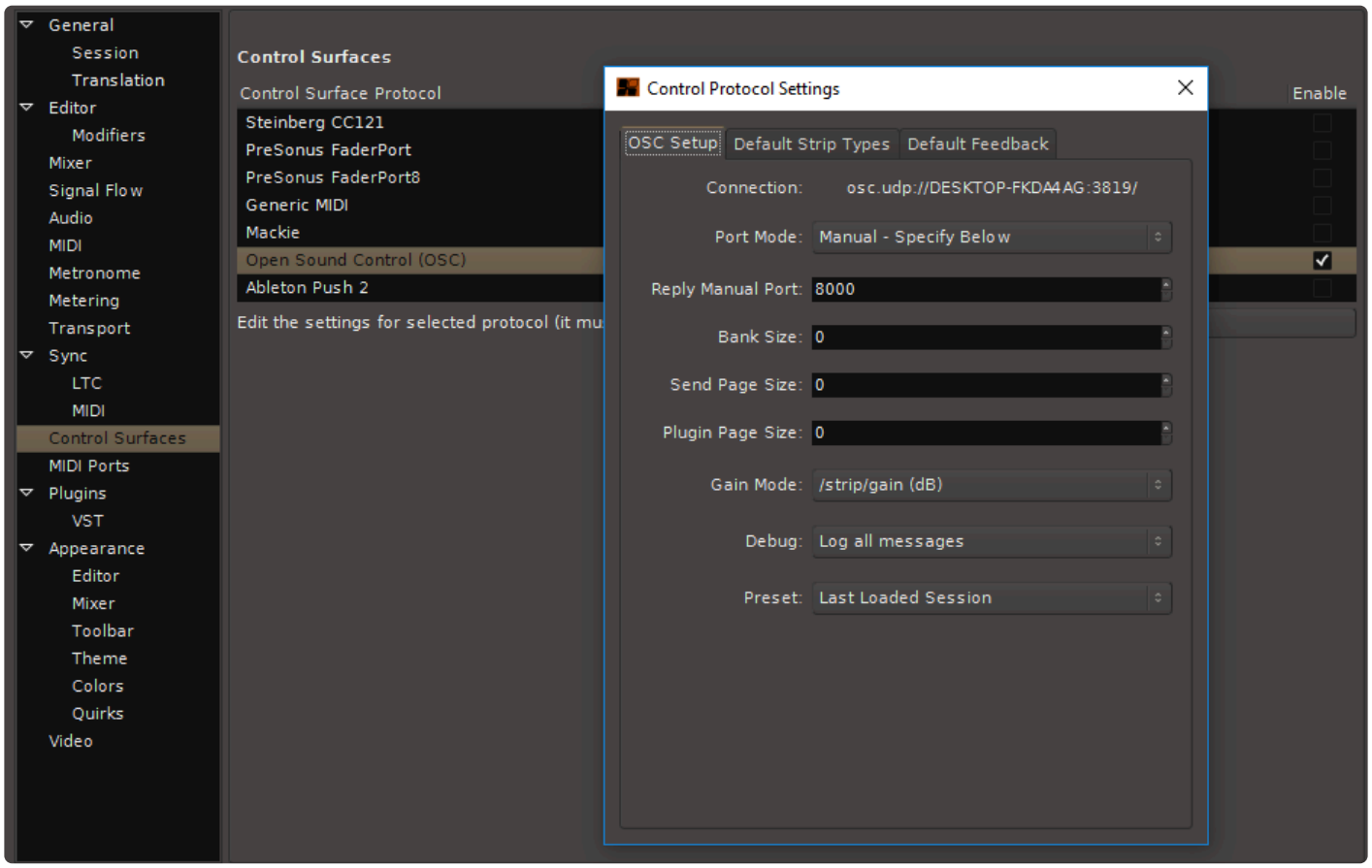
1. If you haven't already, purchase the Lemur application from your relevant App Store. (Android and iOS are both supported,)
2. Install the application and make sure it runs correctly on your device
   (Generally speaking, we wouldn't recommend running the LTI on a screen with a resolution lower than: 1024×768)
3. Go to the Lemur downloads page and download the editor application to your computer or laptop.
4. Install the application to your computer and then open the application.
5. Download the LTI template here and then open it in the editor. It should look a lot like *this*:



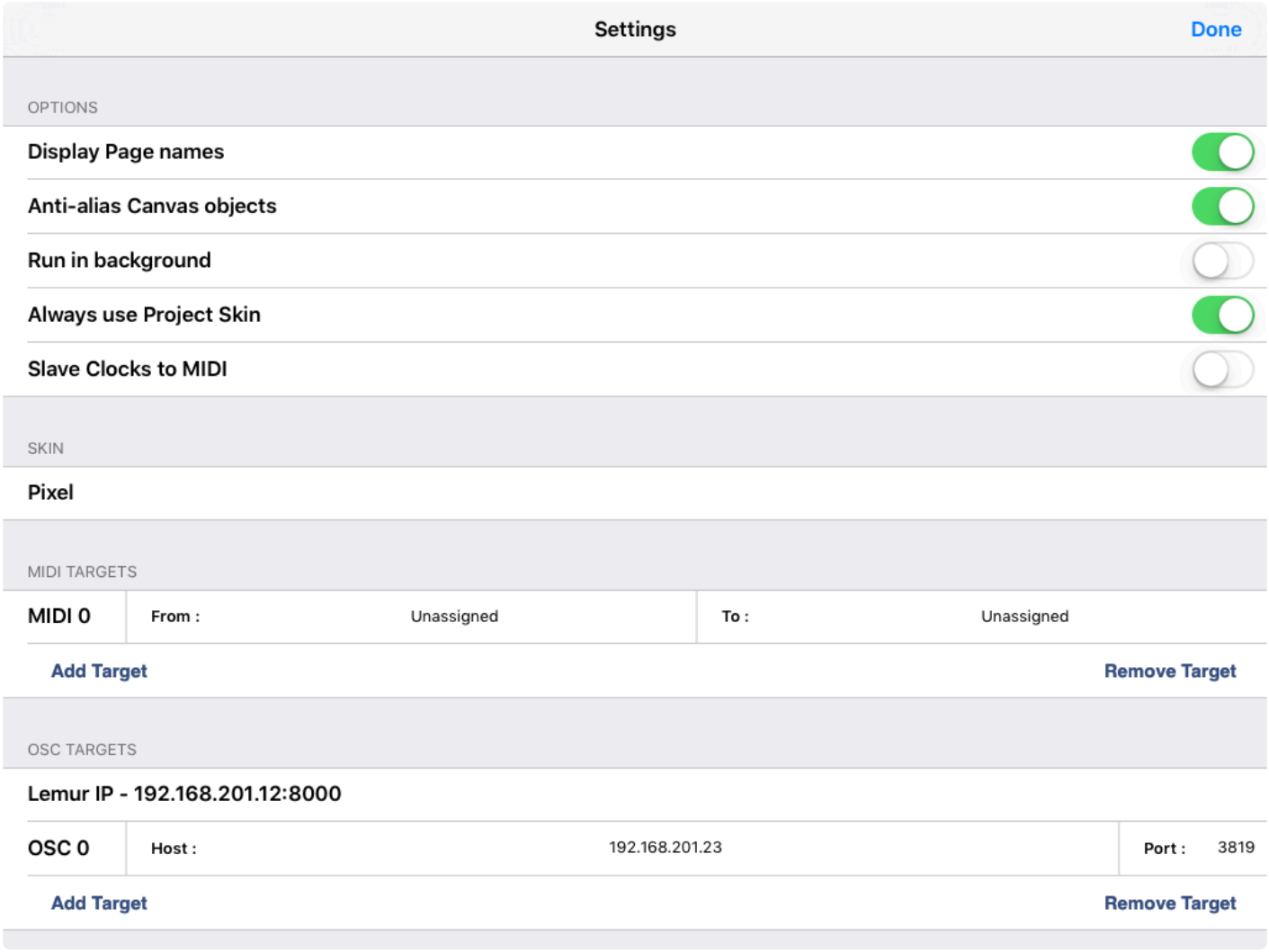
6. Open up the Lemur application on your device and then connect your device to the template editor of your computer using the play button at the top right of the editor and then verify that it has loaded correctly on your tablet. This works over **WIFI ONLY**. You should see a connection like this:



7. In Mixbus, go to Edit->Preferences->Control Surfaces and enable OSC via the check-box. Additionally, make sure that “Port Mode:” is set to **Auto Reply** and not Manual Mode.



8. On your lemur application, open up the settings and make sure that OSC 0: points your computer’s *private* wifi address *with the port of 3819*.



9. Finally, Press the master connect (the big purple on on the connections interface) button and you should see the ring around the button start to flash, like a heartbeat. This indicates that Mixbus is sending signals back to the Lemur interface.

23.5.2. TouchOSC

What is TouchOSC?

TouchOSC is a remote-control application that runs on iOS and Android phones and tablets.

TouchOSC sends “Open Sound Control” messages to any receiving app. Mixbus has a rich OSC-compatible remote control API.

TouchOSC is not developed by Harrison; it is developed by Hexler, and it has its own community forums and website. In keeping with Mixbus’s open-ended development model, users may view, modify, and edit the TouchOSC layout that we’ve provided for Mixbus.

If you add a great new feature to the TTC-1 layout, be sure to share it with us, and we will incorporate your changes so other users will get the benefit as well.

What is the TTC-1 layout?

TouchOSC can load different “layouts” which define the location of the buttons and widgets, and what messages will be sent to Mixbus over the OSC protocol. We’ve developed a layout file that allows you to remotely control Mixbus’s transport features. Because it needs a name, we called it TTC-1, or “TouchOSC Transport Controller #1”

Installing and launching the TTC-1 controller layout for Mixbus:

1. Purchase the TouchOSC application from your phone/tablet App Store. (Android and iOS are both supported,)
2. Also download the TouchOSC *editor* to your Desktop computer, from the TouchOSC web site
3. Download Harrison’s TTC-1 tablet layout here: ttc-1 download
4. On your desktop computer, launch the TouchOSC editor, and “Open” the layout file.
5. In the editor, click “Sync”.
6. In the TouchOSC application (on your phone/tablet), click “Layout -> Add From Editor”.
7. You should see you computer name appear in the “Found Hosts” list.
8. Select the new layout for the TouchOSC app, and you should see the TTC-1 control panel appear.
9. Determine your computer’s IP address on the wireless network that is shared with your phone/tablet.
10. Enter the computer’s IP address in the TouchOSC application.
11. In Mixbus, enable the OSC protocol in Preferences->Control Surfaces.

Using the TouchOSC TTC-1 Controller:

These 2 videos demonstrate the TTC-1 transport controller:

https://www.youtube.com/embed/ve8ewSUMvCg?rel=0

https://www.youtube.com/embed/jhr5BLH8AnM?rel=0

https://www.youtube.com/embed/jhr5BLH8AnM?rel=0

23.6. External Transport Control

Transport Control

Mixbus transport (Play, Stop, Locate, etc.) can be controlled by an external input in 3 ways:

- MMC (Midi Machine Control)
- OSC (an Ethernet network protocol)
- JACK (using the JACK server, if it is installed on your computer)

Mixbus's external transport functions are largely developed by the Ardour open-source community. For more details, see the applicable chapters in the Ardour manual: http://manual.ardour.org/synchronization/overview-of-timecode-related-settings/

23.7. MIDI Control Surfaces

Generic MIDI Control

https://www.youtube.com/embed/xfdSe2aPKDI?rel=0

https://www.youtube.com/embed/k0UBT3MScZE?rel=0

Mixbus can respond to MIDI control messages such as transport control, knobs, sliders and switches.

MIDI remote-control protocols are largely developed by the Ardour open-source community, and you can find documentation on the Ardour website.

Mixbus uses 2 different mechanisms for MIDI control surfaces:

- **MIDI Learn:** http://manual.ardour.org/using-control-surfaces/generic-midi/midi-learn/
- **MIDI Binding Maps:** http://manual.ardour.org/using-control-surfaces/midi-binding-maps/

24. Bundled Plugins (unlicensed)

Mixbus comes with a number of plugins pre-installed. A few basic plugins do not require any further licensing. The “XT” plugins do require a separate license for each to enable their control panels. When you purchase a license for one of these plugins the license file goes in your computer’s Home folder, the same place as your Mixbus license.

These included plugins require a separate license for each.

XT-3D Triple Delay
XT-BC Bass Character
XT-DC Drum Character
XT-EG Expander/Gate
XT-EQ Equalizer
XT-DS De-Esser
XT-DM Dyno-Mite
XT-GV Gverb+
XT-LC Lookahead Compressor
XT-ME Mastering EQ
XT-MG Micro Glide
XT-MC Multiband Compressor
XT-SC Spectral Compressor
XT-SP Stunning Phaser
XT-TG Tom Gate
XT-VC Vocal Character
x42 EQ
x42 Whirl
x42 Meter Bundle

Unlike most unlicensed third-party plugins, the sound of these plugins is still effective when they aren’t licensed. So if you share a Mixbus session with someone who doesn’t own a license for these plugins, the sound as per your settings will remain correct.

24.1. XT-3D Triple Delay

XT-3D Delay is a creative delay effect. Utilizing 3 separate paths for the Left, Right, or Center(sum), you can create many different mono or stereo effects.

> This plugin requires a license to enable the control panel. To purchase a license, visit Harrison's XT Plugin Page

Best when used on: Vocals, Guitars, Drums and Keyboards.

Controls:

Input High Cut: Moving this slider to the left will decrease the amount of high frequencies that are fed to the delay effect.

Input Low Cut: Moving this slider to the *right* will decrease the amount of low frequencies that are fed to the delay effect.

L,C,R: the effect has 3 delay paths: one dedicated for the Left, one dedicated for the Right channel, and a third channel for the Center (sum of left+right). Each channel has an **In** knob (the level sent to the input of this delay path), **Out** knob (output level of this delay path) and a **Fine** adjust (a small additional delay for this path, which can be modulated.)

Wet/Dry knob: Turning this knob to the left will result in a 100% Dry sound (no effect is heard). Turning the knob to the right will result in 100% Wet sound (no dry signal is heard). A momentary button for 'Dry' and 'Wet' allows you to audition the effect without permanently changing the knob setting.

> When the 3D delay is used on a bus, it is conventional to use the effect in 100% "Wet" setting, and adjust the wet signal from the individual track "send" knobs. So the 3D Delay uses 100% "Wet" as a default value for this knob.

Wet Level Trim: This knob adjusts the level of the "wet" signal. You can trim the effect up to +/-24dB with this slider.

Modulation: Increasing this knob will slightly modulate the path delay times. This results in a subtle flanging or pitch-modulation effect.

Tempo: The tempo may be entered by tapping the Tap button repeatedly, or adjusting the knob. If the "Host" button is engaged, then the Tap and Tempo knobs will be disabled, and the plugin will use the host's tempo.

16th Snap: Engaging this button will lock the 3 delay icons to multiples of the tempo, when they are dragged. This makes it easier to generate beat-synced effects.

Delay Icons L, C, R: Each of the three delay paths is represented by an icon on the graph. A green signal-present indicator surrounds each icon, when the delay output has signal. Moving an icon to the left or right will pan that delay path's output. Dragging the icon lower will result in a longer delay time.

Presets: Each of these buttons will overwrite all of the current plugin parameters to match a predefined preset. The selected presets are starting points for the most popular delay effects.

Feedback section:

The feedback section has the most dramatic effect on the effect's decay. In many cases, you won't need to adjust these knobs because they are selected by the Presets. But a deep knowledge of the feedback knobs will allow an experienced user to realize many sonic possibilities.

LR>C<LR: this knob adjusts the level of the *left and right delayed output* that is fed to the input of the center delay path. Turning the knob "left" will feed the left&right output signals into the center's delay path. Turning the knob to the right will feed the output of the center path into the left&right paths. This knob is typically used for spatial effects and long delays.

L<>R: this knob adjusts how much of the left's output is sent to the right delay input, and vice versa. This knob is used for ping-pong effects.

Feedback: this knob adjusts the overall level of feedback paths. Increasing the knob will reesult in more pronounced feedback effect, and may result in howling feedback. For dramatic effects, you can automate this knob to drive your effects at or near 100% feedback.

Linear: This knob affects each paths' feedback from its input to its output. This is the most common feedback mode.

Damping: This knob defines the cutoff frequency for the feedback path. Increase the damping knob to emulate analog tape-style delays which result in successively less high-frequency content in each feedback repeat.

24.2. XT-BC Bass Character

XT-BC Bass Character is an effect intended for acoustic and electric bass instruments.

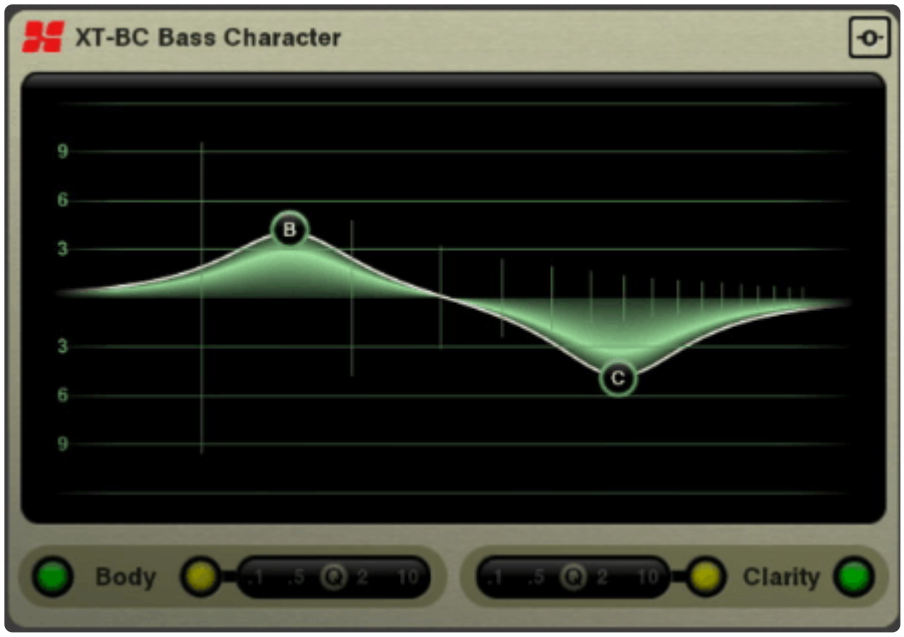
> This plugin requires a license to enable the control panel. To purchase a license, visit Harrison's XT Plugin Page

The Bass Character operates by detecting each musical note when it is played, and then applying an eq to make the note sound fuller and brighter (or weaker and duller, if needed).

You can use the Bass Character plugin to solve a problem like: an electric bass with too much fundamental and little harmonic content; or a bass amp that doesn't provide enough fundamental for the played notes on a 5-string bass.

Best when used on: Monophonic Bass instruments.

> NOTE: the XT-BC is for mono tracks only.



Scale + Spectrum Display:

The Bass Character plugin operates like an EQ, but with one difference: the EQ isn't set at fixed frequencies, but rather the eq is applied to each detected note.

The red "spectrum" curve shows the harmonic content of the detected note. The leftmost line indicates the "fundamental" or first harmonic of the note. Each vertical line to the right indicates an additional harmonic

(2nd harmonic, 3rd, etc). You can use this graph to determine the balance of an instrument: whether it has too much or too little fundamental, for example.

Controls:

The Bass Character plugin has 2 EQ bands: “Body” and “Clarity”. These bands are identical except that the “Body” control is a low-shelving EQ by default, while the “Clarity” band is a high-shelf EQ.

In/Out switch: The green toggle switch enables/disables each band.

Peaking switch: The yellow toggle switch changes the band to a peaking.

Q slider: when the Peaking switch is engaged, the Q slider engages, and adjusts the Q (sharpness) of the band’s filter.

Momentary Bypass: the top-right corner is a momentary-bypass button. Press this button to temporarily bypass the effect. Releasing the button will re-apply the effect.

24.3. XT-DC Drum Character

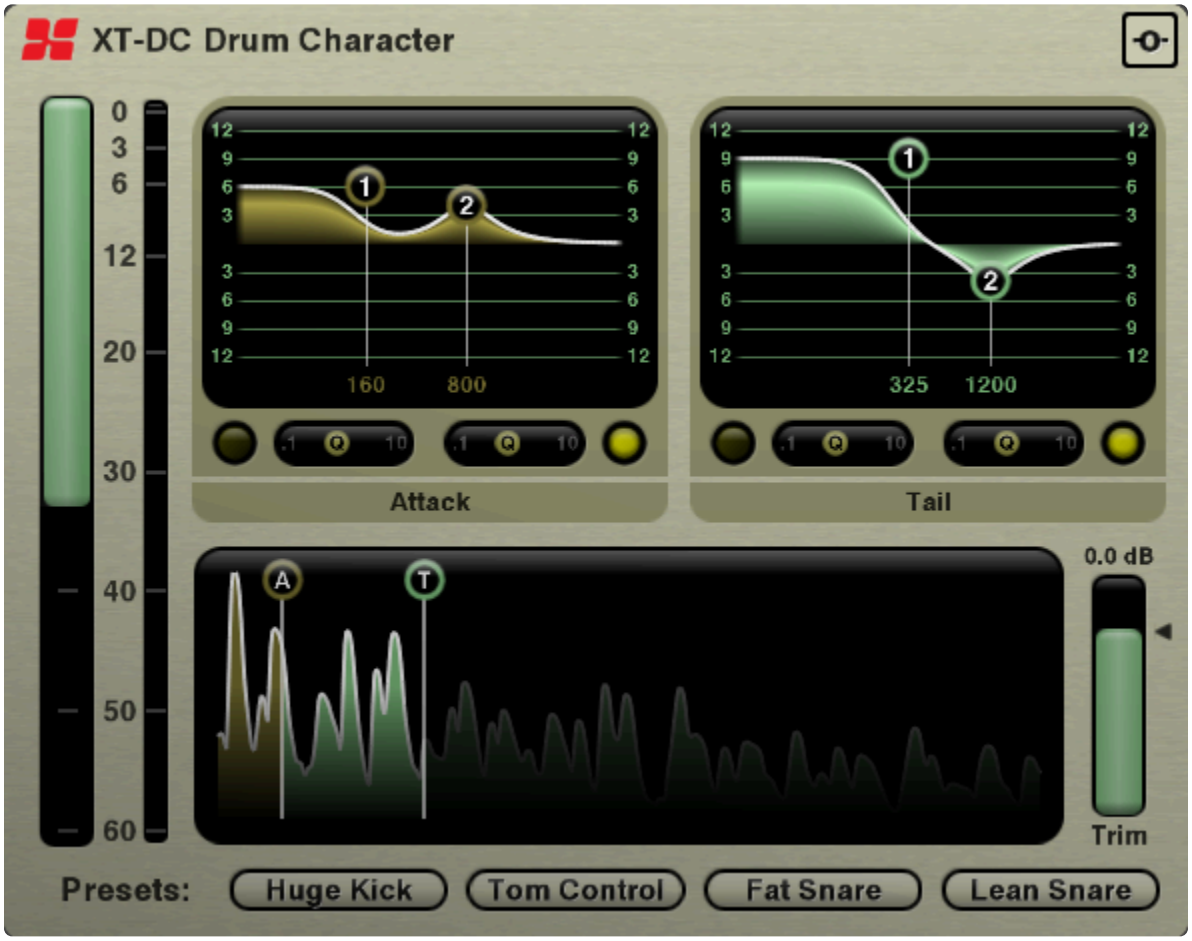
The XT-DC Drum Character is an effect intended for use on acoustic percussion instruments, particularly snare and bass drums.

> This plugin requires a license to enable the control panel. To purchase a license, visit Harrison's XT Plugin Page

The Drum Character plugins operates by producing two distinct EQ curves on each of the primary part of a percussion sound, the transient and the tail.

This allows you to control the exact sonic character of the initial hit of of the instrument (attack), as well as the roll off (or tail).

Best when used on: Individual drum/percussing instruments



Attack & Tail Parametric EQs: The Drum Character plugin features two parametric EQs allowing you to EQ both the attack and the tail different from one another

Attack & Tail Waveform Display: Here you can choose where on the transient the attack and the tail actually sit. This display is updated every time a new transient passes through.

Input Threshold: This vertical slider controls the amount of material that gets into the plugin.

Trim: This vertical slider allows you to trim the plugin output so the apparent volume after processing remains the same as the unprocessed signal.

Momentary Bypass: the top-right corner is a momentary-bypass button. Press this button to temporarily bypass the effect. Releasing the button will re-apply the effect.

24.4. XT-EG Expander/Gate

The **XT-EG Expander / Gate** is Harrison's versatile tool for gating noise or gently increasing the dynamic range of tracks.

> This plugin requires a license to enable the control panel. To purchase a license, visit Harrison's XT Plugin Page

Controls

Input Threshold: This vertical slider affects the onset of the Expander/Gate effect. You should set the threshold to match the level of signal you want to hear. Signals quieter than this threshold will be affected.

Depth: this affects the maximum gain reduction that the effect will apply. For a full gate, you should set this value to maximum. For gentle expansion, you should use a lower value.

Hold(ms): when the gate is "opened" by a loud-enough signal, the hold time ensures that the gate stays open for at least this time. *This knob is only operable when the "Expand" button is **not** engaged*

Hysteresis: when the gate is "opened" by a loud-enough signal, this prevents the gate from closing until the signal reduces by this much below the threshold. *This knob is only operable when the "Expand" button is **not** engaged*

Expand: when this button is engaged, the Hold and Hysteresis knobs are disabled, and the Ratio and Knee controls become enabled.

Ratio: The ratio defines the aggressiveness of the expansion mode. A low ratio will require the signal to be significantly lower than the threshold, before much gain reduction is applied. *This knob is only operable when the "Expand" button is engaged*

Knee(dB): this knob affects the onset of the gate. A larger knee means that the effect's onset will operate as if the threshold were lower, but the onset of the gain-reduction will be more gradual. *This knob is only operable when the "Expand" button is engaged*

Attack(ms): this adjusts how quickly the gate can "open" when a signal exceeds the threshold.

Release(ms): this adjusts how quickly the gate can close again, after the signal reduces below the threshold.

Lookahead: Using lookahead allows the gate to open "before" the onset of a signal above the threshold; this can prevent (for example) cutting-off the front of a drum hit. There are 2 levels of lookahead. When lookahead is engaged, the plugin will report some latency to the host, so the host can delay other tracks to match the drum sound back to its un-delayed timing.

Sidechain Controls

The "sidechain" is the signal-detector inside the expander/gate. Filtering the sidechain does not result in a change to the sound of the signal, but it does affect how the gate opens/closes. For example, if you want to gate a snare drum, it is often desirable to find and accentuate a strong frequency of the snare drum, so the gate is more sensitive to that sound.

> The XT-EG does not have an external input to feed a sidechain signal other than the main input. We might add this feature in the future.

Sidechain Filter: Click this button to enable the sidechain filter.

Listen: Click this button to listen to the sidechain signal, with the filter applied.

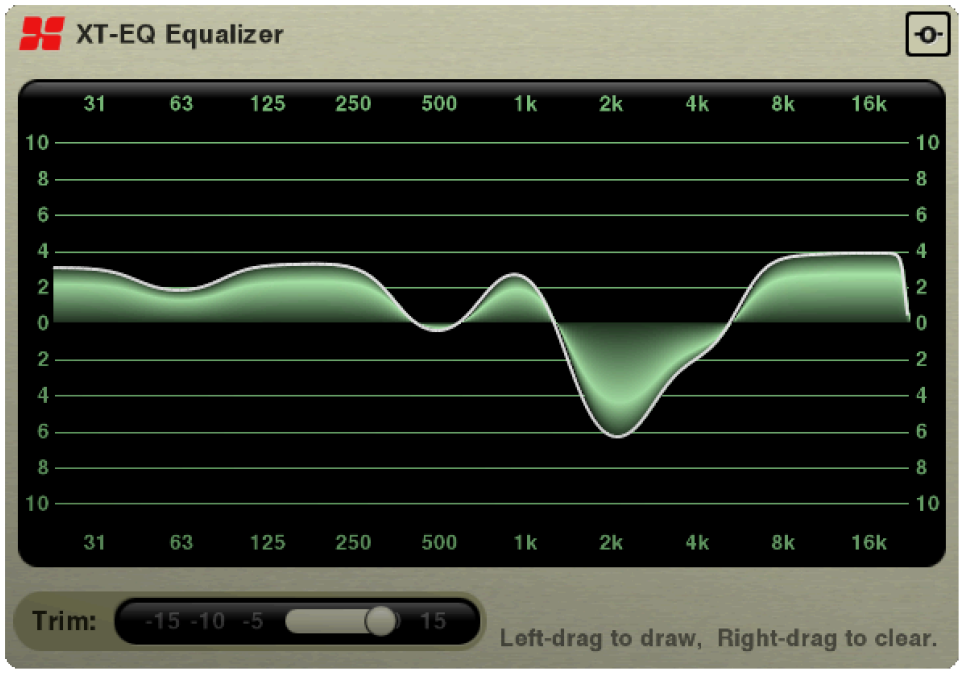
Q: This adjusts the sharpness of the filter band.

Frequency: This moves the frequency of the band to boost. Move the filter until the sound you want to expand/gate is accentuated, and other sounds are not.

24.5. XT-EQ Equalizer

Harrison's XT-EQ Equalizer is a cost-effective, smaller version of the XT-ME Mastering EQ for tracks and final mixes.

> This plugin requires a license to enable the control panel. To purchase a license, visit Harrison's XT Plugin Page



The XT-EQ has 10 bands of equalization. The filters provide a "flat top" response, meaning: if you boost 2 adjacent bands, the gain between the bands will be consistent. (unlike a traditional graphic eq, where 2 adjacent filters will create a camel-hump)

Each band has approximately 1 octave bandwidth. This makes the XT-EQ a very fast tool for focusing instruments on their representative frequency ranges, and reducing (but not completely filtering) the range outside their focus. The XT-EQ is an excellent mixing tool.

To adjust the eq, simply click anywhere and drag to "draw" the desired EQ curve. Because of the broad and even pass-band between bands, "what you see is what you get".

It is possible to adjust a single band (like a graphic eq) by clicking in the band and draggin up/down. Or point your mouse at a single band and use the mouse-wheel to adjust only that band.

Hold "Shift" and drag across the EQ to reset the bands to 0dB.

Trim: This is a clean volume control that adjusts the output level. If you use a lot of EQ "cuts" then you may

need to increase this to match the level. And vice versa.

Momentary Bypass: Click and hold this button to momentarily defeat the EQ. It is a good idea to use the Trim control to adjust the volume of the EQ'd signal so you can compare the effect of the EQ, and not just the level difference caused by the EQ.

24.6. XT-DS De-Esser

The XT-DS De-Esser is Harrison's transparent, powerful plugin for taming excessive sibilance of vocal tracks.

> This plugin requires a license to enable the control panel. To purchase a license, visit Harrison's XT Plugin Page

Input Meter, Threshold, and Ess meter:
The "Threshold" slider for the effect is overlaid on the input ("In") meter.
Lowering the threshold will increase the sensitivity of the de-esser effect.
The "Ess" meter indicates the signal level inside the ess band. If the frequency band is set correctly for the source material, then this meter will peak when Essy material is present.
Ess Freq and Ess Depth Control:
This section allows you to pick the low frequency cutoff (200 Hz – 8kHz) for the Ess, and the max attenuation for the signal (-12dB – 0dB).
Hi Freq and Hi Depth Control:
This section features a shelving unit to tame some of the higher frequency (2 kHz – 12 kHz) content after you've pulled out the 'Ess'. Max attenuation for the shelf is -12dB.
Ess Center and Ess Width Control:
The 'Ess Center' control allows you to adjust both the Ess Freq and Hi Freq controls together allowing you move the entire Ess Band in unison. The 'Ess Width' control allows you to control how wide or narrow the Ess bandwidth is.

Attack:
The ‘Attack’ knob controls how long it takes the signal to get to from zero to full attenuation. (200µs – 20ms)

24.7. XT-DM Dyno-Mite

XT-DM Dyno-Mite follows the dynamics of your track. It detects when the sound is “decaying” (gradually diminishing after a louder transient). Dyno-Mite affects the decay rate, either lengthening or truncating the sustain.

> This plugin requires a license to enable the control panel. To purchase a license, visit Harrison’s XT Plugin Page

Best when used on: *Drum loops; MIDI drums; Drum overheads; Individual drum sounds.

Controls:

Threshold: The leftmost bar shows in the input signal level, and a slider to adjust the processor threshold. The threshold should be set below the signal peaks shown on the meter. The lower the threshold is set, the more pronounced the effect will be.

Sustain: the central knob controls the Sustain.

If you turn the knob to the right, this will increasingly add gain while the signal decays, resulting in a longer sustaining sound. This is sometimes called “upward compression”. Room ambience will be increased along with the decay of the signal.

If you turn the Sustain knob to the left, Dyno-Mite will accentuate the decay by reducing the level faster than the unaffected signal. This results in a gated or “expanded” sound, with sharper transients and less room ambience.

The meter displays the amount of gain increase (in green) or decrease (in red). When the Sustain knob is set very low, an automatic makeup gain is applied. You will see this gain appear on the display.

Out Trim: the rightmost bar provides a trim control. You can use this control to adjust the final output level after the effect is applied.

Momentary Bypass: the top-right corner is a momentary-bypass button. Press this button to temporarily bypass the effect. Releasing the button will re-apply the effect.

24.8. XT-GV Gverb+

XT-GV Gverb+ is a creative reverb effect. Utilizing an algorithm that produces natural room sounds and lush reverb effects alike.

> This plugin requires a license to enable the control panel. To purchase a license, visit Harrison's XT Plugin Page The XT-GV Gverb+ used to be called simply "Gverb+" and was part of the Plugin Essentials pack, but has since been separated. You may still see literature that references its former status.

Best when used on: Vocals, Guitars, Drums and Keyboards.

Controls:

Input High Pass Filter: Moving this slider will decrease the amount of low frequencies that are fed to the reverb effect.

Input Low Pass Filter Moving this slider will decrease the amount of high frequencies that are fed to the reverb effect.

Early Mix Every reverb has two distinct characteristics: Early reflections, and the reverb's "tail". Early reflections define the shape characteristic of the room, and typically make it sound more natural as that is how our ears typically hear a reverberation in the real world. The tail is a lush wash of reverberations that

tend to make reverb sound unrealistic. This slider controls the balance of the two.

Size The size slider controls the overall feeling of the ‘size’ of the room. Turning this down can cause the reverb characteristic to become more personal or up-close. While turning it up can cause the characteristic to feel as if you are in the back of a cathedral.

Decay The decay slider determines how long a reverb’s tail takes to “dissipate” or “fall off”.

Pre-Delay The Pre Delay controls the amount of time between the dry sound, and the early reflections and reverb tail. Careful adjustment of this knob, or using multiple buses with different pre-delay values, can help bring out elements in a mix. Typically this works great on a vocal that you want to stand out.

Damping The damping control controls the amount of amount of high frequency content in the reverb. Dampening high frequency content helps promote a warmer reverb characteristic and sound.

Wet/Dry knob: Turning this knob to the left will result in a 100% Dry sound (no effect is heard). Turning the knob to the right will result in 100% Wet sound (no dry signal is heard). A momentary button for ‘Dry’ and ‘Wet’ allows you to audition the effect without permanently changing the knob setting.

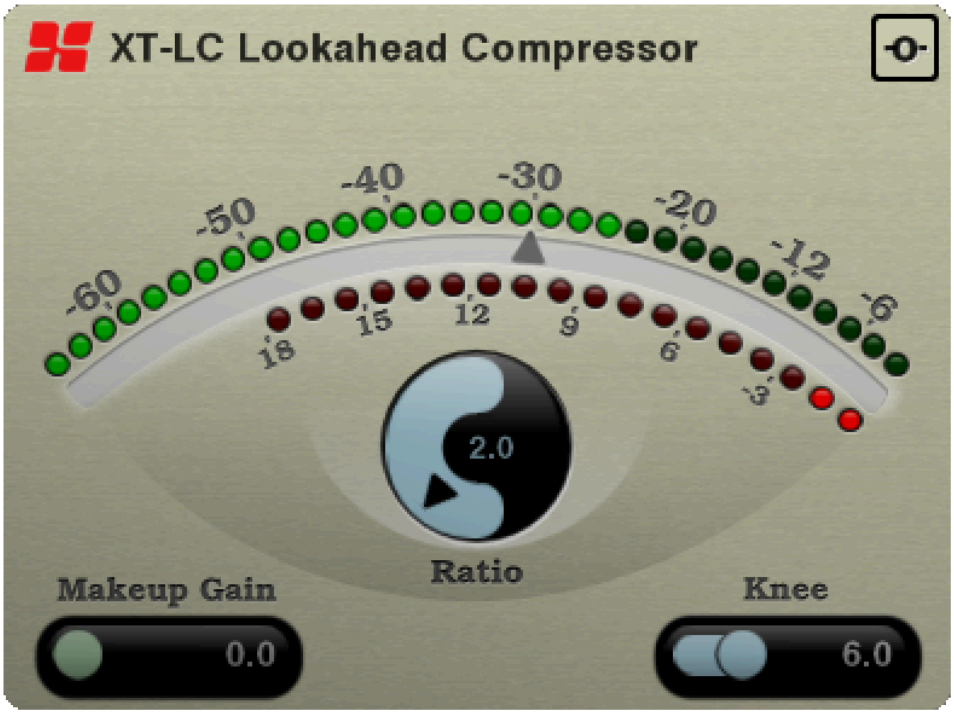
> When the Gverb+ reverb is used on a bus, it is conventional to use the effect in 100% “Wet” setting, and adjust the wet signal from the individual track “send” knobs. So the Gverb+ reverb uses 100% “Wet” as a default value for this slider.

Wet Level Trim: This slider adjusts the level of the “wet” signal. You can trim the effect up to +/-24dB with this slider.

24.9. XT-LC Lookahead Compressor

XT-LC Lookahead Compressor is a solution for very demanding compression tasks such as acoustic guitar, full mixes, and live vocals. The XT-LC's lookahead design can work where other compressors fall short.

> This plugin requires a license to enable the control panel. To purchase a license, visit Harrison's XT Plugin Page



Controls

Threshold: The triangle indication on the wide meter arc indicates the compressor threshold.

Ratio: The ratio knob controls the strength of the compression effect.

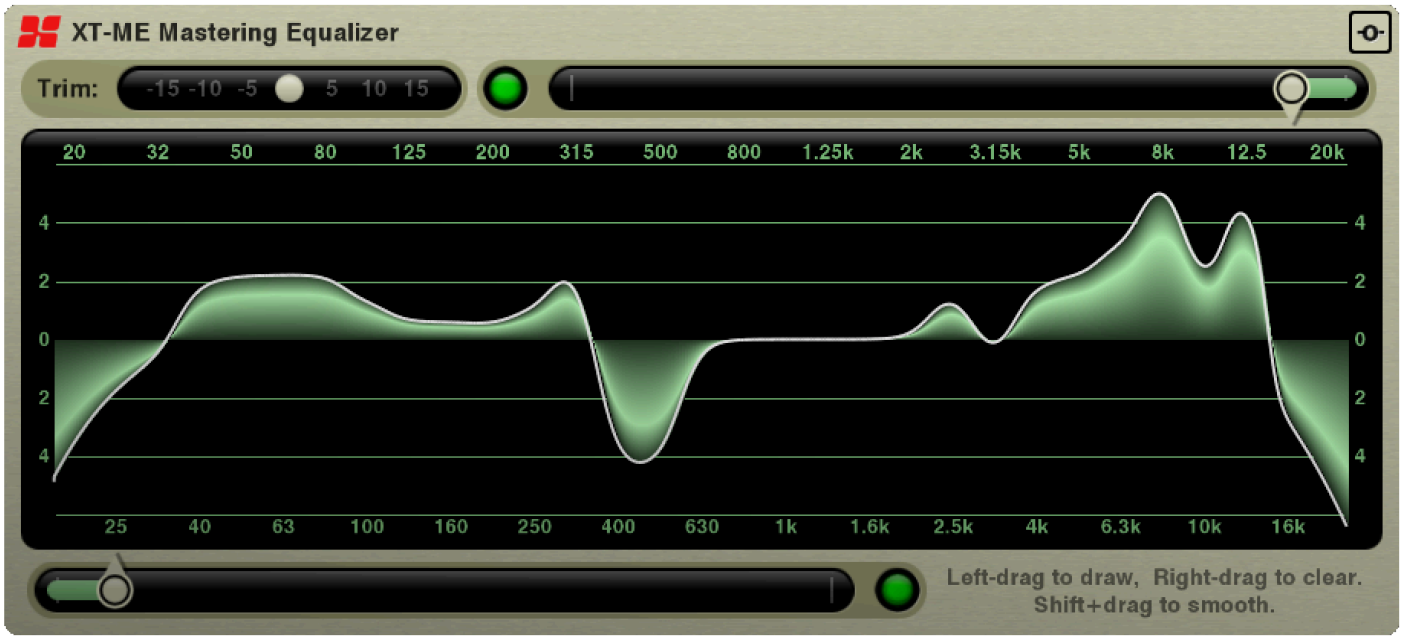
Knee: The 'knee' control defines how far the signal must exceed the threshold to achieve full compression. A smaller knee setting will result in more aggressive compression.

Makeup Gain: The 'Makeup Gain' control allows you to apply additional gain to the output signal, to 'make up' for the attenuation applied by the compression.

24.10. XT-ME Mastering EQ

The XT-ME Mastering Equalizer is Harrison's fastest, most precise, and most transparent EQ for finalizing your mixes.

> This plugin requires a license to enable the control panel. To purchase a license, visit Harrison's XT Plugin Page



The XT-ME has 30 bands of equalization. The filters provide a "flat top" response, meaning: if you boost 2 adjacent bands, the gain between the bands will be consistent. (unlike a traditional graphic eq, where 2 adjacent filters will create a camel-hump)

Each band has approximately 1/3 octave bandwidth. Combined with the flat-top passband between bands, this makes the XT-ME a highly desirable tool for mastering your final mix.

To adjust the eq, simply click anywhere and drag to "draw" the desired EQ curve. Because of the broad and even pass-band between bands, "what you see is what you get".

It is possible to adjust a single band (like a graphic eq) by clicking in the band and dragging up/down. Or point your mouse at a single band and use the mouse-wheel to adjust only that band.

Hold "Shift" and drag across the EQ to reset the bands to 0dB.

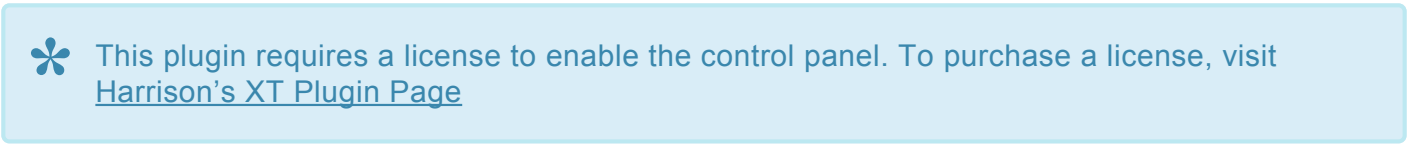
Trim: This is a clean volume control that adjusts the output level. If you use a lot of EQ "cuts" then you may need to increase this to match the level. And vice versa.

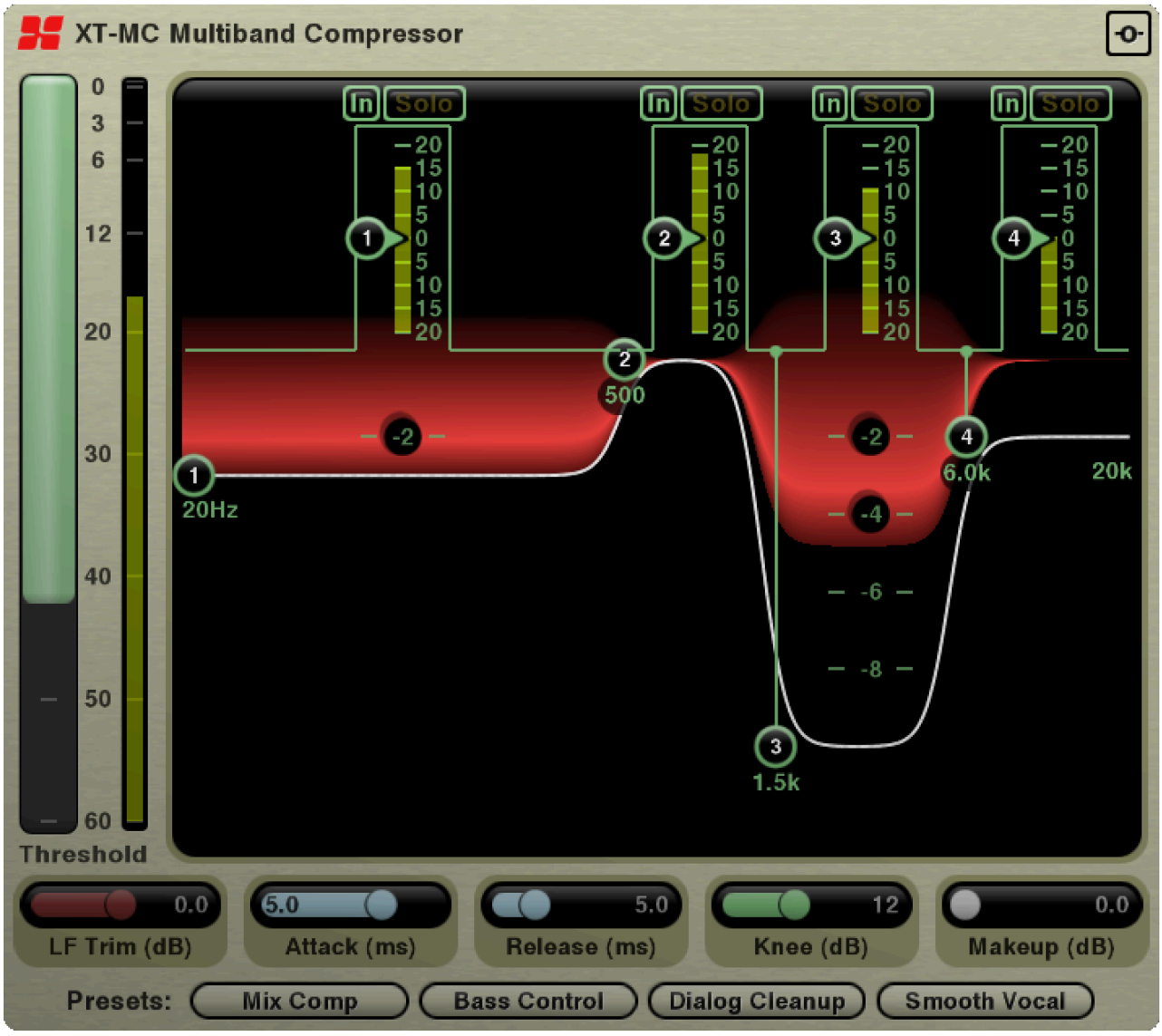
Filters: A low-pass filter and high-pass filter are provided. Click the green button to enable the filter, and adjust the slider to move the cutoff frequency.

Momentary Bypass: Click and hold this button to momentarily defeat the EQ. It is a good idea to use the Trim control to adjust the volume of the EQ'd signal so you can compare the effect of the EQ, and not just the level difference caused by the EQ.

24.11. XT-MC Multiband Compressor

XT-MC Multiband Compressor is a powerful 4-band compressor with phase-accurate crossovers.





Controls

Input Threshold: The primary control for the XT-MC is the input threshold slider. Pull the slider down until the threshold is matched to your signal level. Then use the individual band thresholds to adjust the effect of each band on your system.

Band controls (top): Each band has an In/Out Switch, a "Solo" button, a threshold slider (with

corresponding band input meter).

Band controls (graph): Inside the graph, you can grab a band edge to change the depth of each band, and the crossover frequencies between each band.

LF Trim: This slider provides makeup gain for the lowest compression band.

Attack (200µs – 20ms): The ‘Attack’ knob controls how long it takes the signal above the threshold to reach full compression.

Release (20ms – 200ms): The ‘Release’ knob controls how long it takes the signal, once it falls below the threshold, to return to an uncompressed state.

Knee (3dB – 24dB): The ‘knee’ control defines how far the signal must exceed the threshold to achieve full compression. A smaller knee setting will result in more aggressive compression.

Makeup Gain (0dB – 12dB): The ‘Makeup Gain’ control allows you to apply additional gain to the output signal, to ‘make up’ for the attenuation applied by the compression.

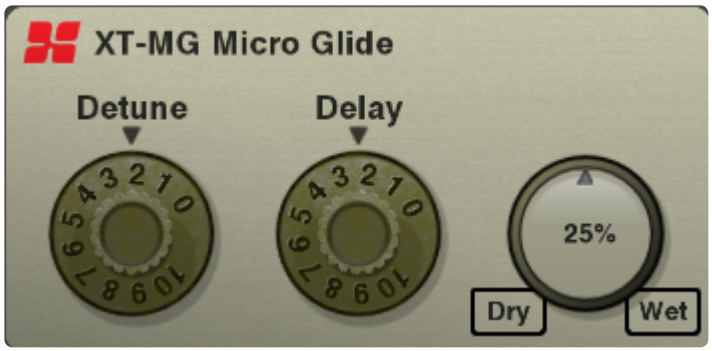
Presets: A few starting-points for common tasks are provided.

24.12. XT-MG Micro Glide

XT-MG Micro Glide is a chorus effect plugin designed to thicken up and liven-up vocals and instruments.

> This plugin requires a license to enable the control panel. To purchase a license, visit Harrison's XT Plugin Page

Best when used on: Vocals. Bass, Synths, Guitars



Controls:

The XT-MG Micro glide has three controls:

Detune The knob controls the spread of the voices

Delay The knob controls the delay between each voice to add depth to the signal

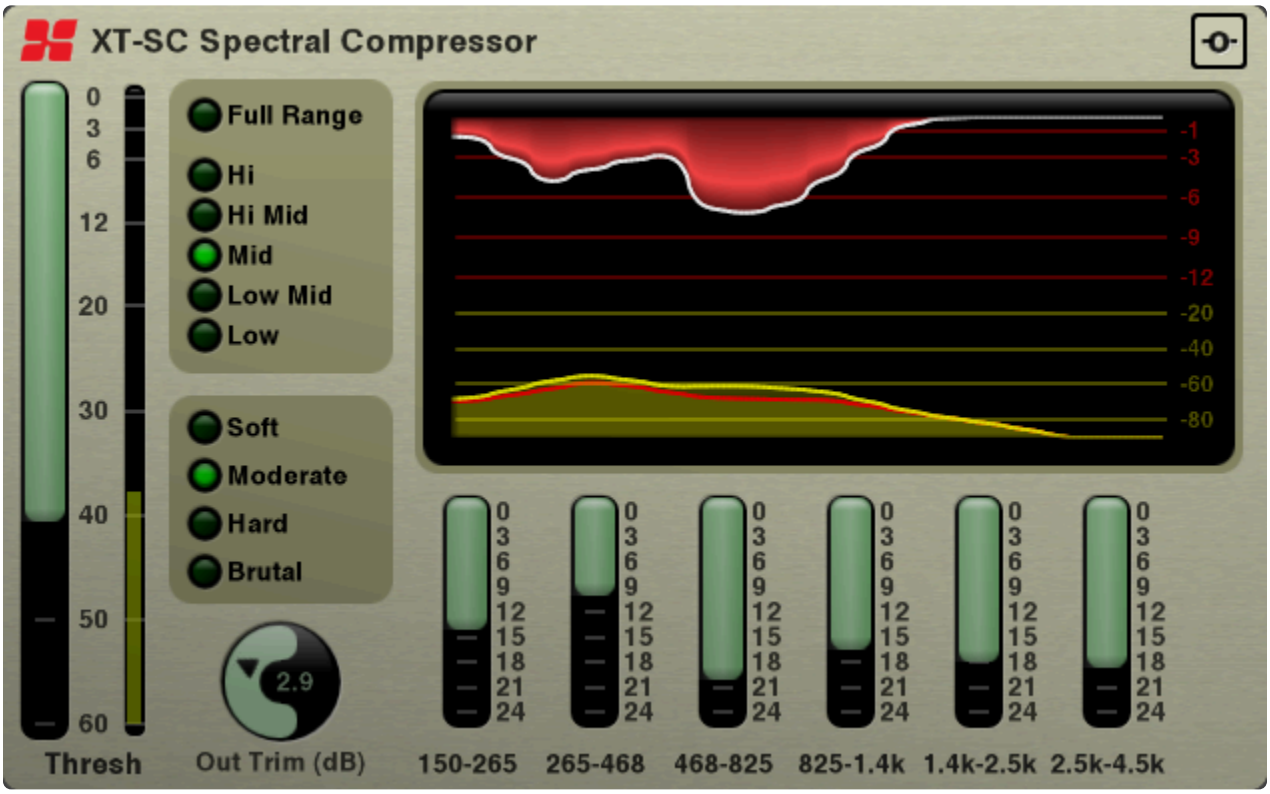
Wet/Dry knob: Turning this knob to the left will result in a 100% Dry sound (no effect is heard). Turning the knob to the right will result in 100% Wet sound (no dry signal is heard). A momentary button for 'Dry' and 'Wet' allows you to audition the effect without permanently changing the knob setting.

24.13. XT-SC Spectral Compressor

The AVA Spectral Compressor (AVA-SC) is a dynamics processor that allows you to apply compression selectively across 18 frequency bands. These compression bands can either be spread out across the full range of the spectrum, or focused on a specific frequency range for more precise control.

> This plugin requires a license to enable the control panel. To purchase a license, visit Harrison's XT Plugin Page

Best when used on: Practically anything!



Range:

Here you can select which frequency range you want to operate on. Typically, when using the Spectral Compressor plugin on a full mix, you want to stay on the "Full Range" mode.

When using the Spectral Compressor plugin on individual instrument, it is more useful to pick from the following ranges for utility:

High Mid (vocal compression plus de-essing, de-harshing, feedback clamping)
Mid (very focused vocal compression)
Low Mid (vocal compression plus rumble reduction)

Low (bass compression, full-mix bass smoothing, and/or rumble reduction)

Compressor Strength:

The compressor mode controls the *range* (in dB) which the input signal is allowed to go over the threshold before the gain reduction reaches “max”.

Mode	Range
Soft	24dB
Moderate	20dB
Hard	12dB
Brutal	6dB

Band Controls:

The main interface consists of six band “depth” sliders: Each of the 6 sliders represents 3 bands of compression.

The green bar indicates the **depth**: the maximum amount of gain reduction that will be applied in that band.

Using the “Full Range” mode as an example: the 12-35Hz range will have 3 separate compressor bands in that range. 12-35 Hz represents about 1.5 octaves, and each of the higher slider ranges contains about 1.5 octaves. So in the “full range” mode, there are 3 compressors in every 1.5-octave range, or said another way: there’s a compressor operating in every half-octave range.

The tightest range is the “high” mode which focuses all 18 bands into only 2.5 octaves: That mode is for taming squeaks and squeals in the upper ranges of hearing.

Controls:

Input threshold: Controls the main threshold for all materail entering the compressor.

Output Trim: Additional gain on the output of the compressor

Momentary Bypass: Press this button to temporarily bypass the effect. Releasing the button will re-apply the effect.

This Plugin is also available as an AVA variant! Click here to find out more information!

24.14. XT-SP Stunning Phaser

XT-SP “Stunning Phaser” is a creative modulation effect similar to the classic Phaser guitar effect.

> This plugin requires a license to enable the control panel. To purchase a license, visit Harrison’s XT Plugin Page

Unlike a traditional phaser, where the modulation is free-running, the XT-SP provides multiple methods to sync the modulation LFO to the session’s playhead position. This makes it particularly useful as a production & mixdown effect, because it always sounds the same.

Best when used on: Vocals, Guitars, Drums and Keyboards…. or anything that needs a bit of funk-ifying!

Controls:

Selecting various modes will enable/disable the associated controls, depending on their applicability.

Modes: Select one of 4 main modes:

- Vintage: A single sharp peak and 2 notches. Traditional, clean guitar-pedal sound.
- Wide: Several widely-spaced peaks and notches. A more complicated, but still clean sound.
- Tight: Several tightly-spaced peaks and notches. A dense and complicated sound.
- Sprung: In Sprung mode, the peaks and notches move faster as the frequency increases.

Oscillator Shape: select the modulation type:

- Sine: has a “dwell” time at the upper and lower limits. Adjust the vertical “shape” slider to increase the

steepness of the sine wave until it reaches a square wave.
- Triangle: the most common repeating-cycle shape for a phaser effect. Adjust the “shape” slider to morph the triangle shape.
- Tri-Saw: Similar to the Triangle wave, but adjusting the “shape” slider will tilt the shape into a sawtooth.
- Freeform: In freeform mode, the user is presented with sliders for 8 steps in the oscillator cycle. The “shape” slider is disabled.
- Stepped: in this mode, the user is presented with sliders for 8 steps in the oscillator cycle. The “shape” slider is disabled.
- Manual: in this mode, the user must adjust the “Base Freq” control manually, or automate it using Mixbus’s automation features.

Frequency Knobs:
Range: this is the sweep-range of the effect, in octaves.
Base Freq: this is the lowest frequency that the sweep will reach. Frequencies below here will not be strongly affected.

Barberpole modes: These modes use a sawtooth modulator shape, either climbing or descending. But there is no discontinuity when the sawtooth resets; the peaks and notches continuously cycle like a rotating barberpole. The resulting effect is good for applying a subtle downward or upward-spiraling effect durin mixdown. Or with more aggressive settings, it can apply an envelope-filtered sound. When a barberpole mode is selected, the Oscillator knobs will be disabled.

BPM: the BPM knob is used to set the rate of the oscillator.

- Free Run: in Free Run, the oscillator runs at the defined BPM, like a standalone guitar pedal.
- Sync Play: in this mode, the oscillator uses its own BPM setting, but it calculates the oscillator position from the host timeline; so repeated playback will always sound the same.
- Host Sync: in this mode, the oscillator always follows the exact position and tempo of the host’s timeline (“Hard Sync”)

Oscillator Display: this display shows the oscillator shape, as well as a vertical line representing the current position of the oscillator value. Click & drag on the oscillator display to adjust the “phase offset* of the oscillator. This allows you to synchronize the modulation to emphasize a particular rhythmic pulse in the plugin’s input signal.

Stereo Width: If the Stereo version is used on a track, then the display provides 2 different lines, and a “stereo width” knob which varies the distance between the 2 lines. For a dramatic stereo effect, use a very small distance between the 2 lines.

Divisions: Each LED indicator represents a different subdivision of the base tempo. Use these settings to increase or decrease the oscillator speed, while matching the tempo to the music. Select different ratios (1/12 or 2/3 on a 4/4 meter) to create polyrhythms in the modulation.

Wet/Dry knob: Turning this knob to the left will result in a 100% Dry sound (no effect is heard). Turning the

knob to the right will result in 100% Wet sound (no dry signal is heard). A momentary button for ‘Dry’ and ‘Wet’ allows you to audition the effect without permanently changing the knob setting.

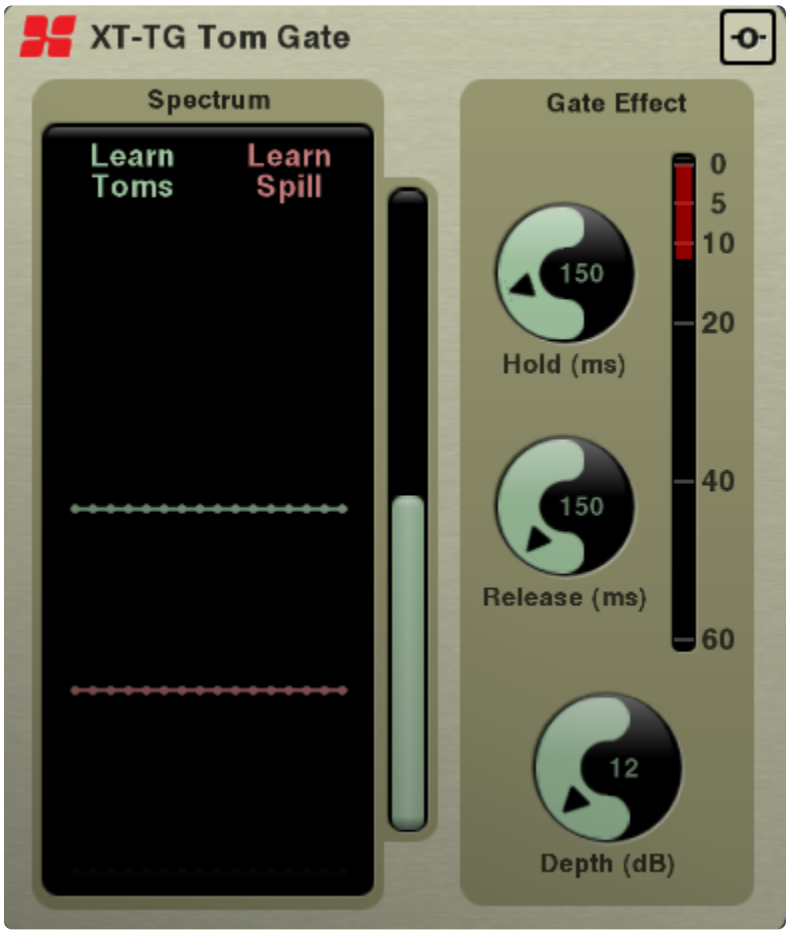
Regeneration: Perhaps the most important knob, the regeneration (sometimes called *feedback* or *resonance*) controls the sharpness of the filters by sending the signal through the filter multiple times. Increase the regeneration knob for wild synth-like effects. The red light will indicate when the phaser feedback path is saturated.

24.15. XT-TG Tom Gate

No, Tom Gate isn't somebody's name. XT-TG Tom Gate is a gate plugin that is specifically made to work on tom drums. It utilizes a special algorithm that is uniquely adapted to listen to a tom-tom drum and separate the individual drum sound from the drum kit spill into its microphone.

> This plugin requires a license to enable the control panel. To purchase a license, visit Harrison's XT Plugin Page

Best when used on: Single Drum Mics, like Tom, Kick, or Snare drums.



Controls:

Despite being a powerful effect, the XT-TG Tom Gate only has six controls:

Learn Tom While playing a tom recording on the track that this plugin lives, holding down the 'Learn Tom' button will train the XT-TG to the sound and transient quality of your tom.

Learn Spill While playing a tom recording on the track that this plugin lives, holding down the 'Learn Spill'

will train the XT-TG to the sound and transient quality of your microphone ‘spill’, i.e everything that isn’t a tom.

Threshold The vertical slider in the middle controls the threshold of the gating.

Hold, Release, and Depth Controls the typical gate features like: How long to hold the gate open, how fast the gate opens, and the level of gain reduction for the material that does not pass the gate.

In the top-right corner is a momentary-bypass button. Press this button to temporarily bypass the effect. Releasing the button will re-apply the effect.

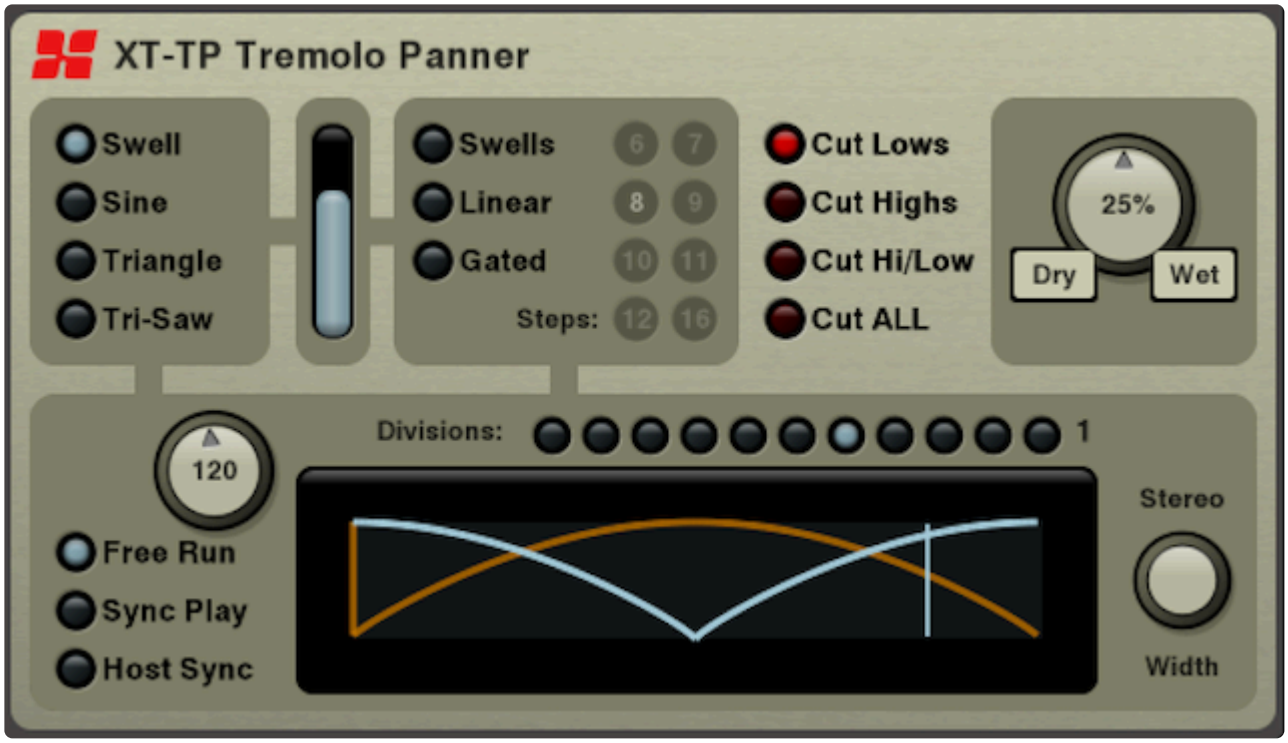
24.16. XT-TP Tremolo Panner

XT-TP “Tremolo Panner” is a creative modulation effect similar to guitar stompboxes, and including effects of “slicers” and “harmonic tremolo” pedals.

> This plugin requires a license to enable the control panel. To purchase a license, visit Harrison’s XT Plugin Page

Unlike a traditional tremolo, where the modulation is free-running, the XT-TP provides multiple methods to sync the modulation LFO to the session’s playhead position. This makes it particularly useful as a production&mixdown effect, because it always sounds the same on playback.

Best when used on: Guitars, Keyboards. Can also be used as a gating/pumping effect for drums.



Controls:

Oscillator Shape: select the modulation type:

- Swell: the traditional swelling tremolo effect, with a very short dwell. Adjust the vertical “shape” slider to increase the steepness of the swell until it reaches a square wave.
- Sine: has a “dwell” time at the upper and lower limits. Adjust the vertical “shape” slider to increase the steepness of the sine wave until it reaches a square wave.
- Triangle: the most common repeating-cycle shape for a phaser effect. Adjust the “shape” slider to morph the triangle shape.

- Tri-Saw: Similar to the Triangle wave, but adjusting the “shape” slider will tilt the shape into a sawtooth.

User Shapes: enabling one of these shapes will display controls for the user to draw their own oscillator waveform shape

- Swells: Up to 16 individual “swells” can be adjusted in level.
- Linear: 16 control points, connected with smooth ramp between each point. The “shape” slider adjusts a hold time around each point.
- Gated: up to 16 stepped control points. The “shape” slider adjusts the hold time after each point.

Shape Slider

- The shape slider affects the selected oscillator shape, typically making it steeper.

Modes: Select one of 4 main modes:

- Cut Lows: The tremolo will cut low frequencies, leaving higher frequencies intact.
- Cut Highs: The tremolo will cut high frequencies, leaving low frequencies intact.
- Cut Hi+Low: Alternately cut the high or low band (sometimes called a “harmonic tremolo” effect).
- Cut ALL: This cuts the volume of the signal, like a traditional tremolo pedal(Defaut).

BPM: the BPM knob is used to set the rate of the oscillator.

- Free Run: in Free Run, the oscillator runs at the defined BPM, like a standalone guitar pedal.
- Sync Play: in this mode, the oscillator uses its own BPM setting, but it calculates the oscillator position from the host timeline; so repeated playback will always sound the same.
- Host Sync: in this mode, the oscillator always follows the exact position and tempo of the host’s timeline (“Hard Sync”)

Oscillator Display: this display shows the oscillator shape, as well as a vertical line representing the current position of the oscillator value. Click & drag on the oscillator display to adjust the “phase offset* of the oscillator. This allows you to synchronize the modulation to emphasize a particular rhythmic pulse in the plugin’s input signal.

Stereo Width: If the Stereo version is used on a track, then the display provides 2 different lines, and a “stereo width” knob which varies the distance between the 2 lines. For a dramatic stereo effect, use a very small distance between the 2 lines.

Divisions: Each LED indicator represents a different subdivision of the base tempo. Use these settings to increase or decrease the oscillator speed, while matching the tempo to the music. Select different ratios (1/12 or 2/3 on a 4/4 meter) to create polyrhythms in the modulation.

Wet/Dry knob: Turning this knob to the left will result in a 100% Dry sound (no effect is heard). Turning the knob to the right will result in 100% Wet sound (no dry signal is heard). A momentary button for ‘Dry’ and

‘Wet’ allows you to audition the effect without permanently changing the knob setting.

24.17. XT-VC Vocal Character

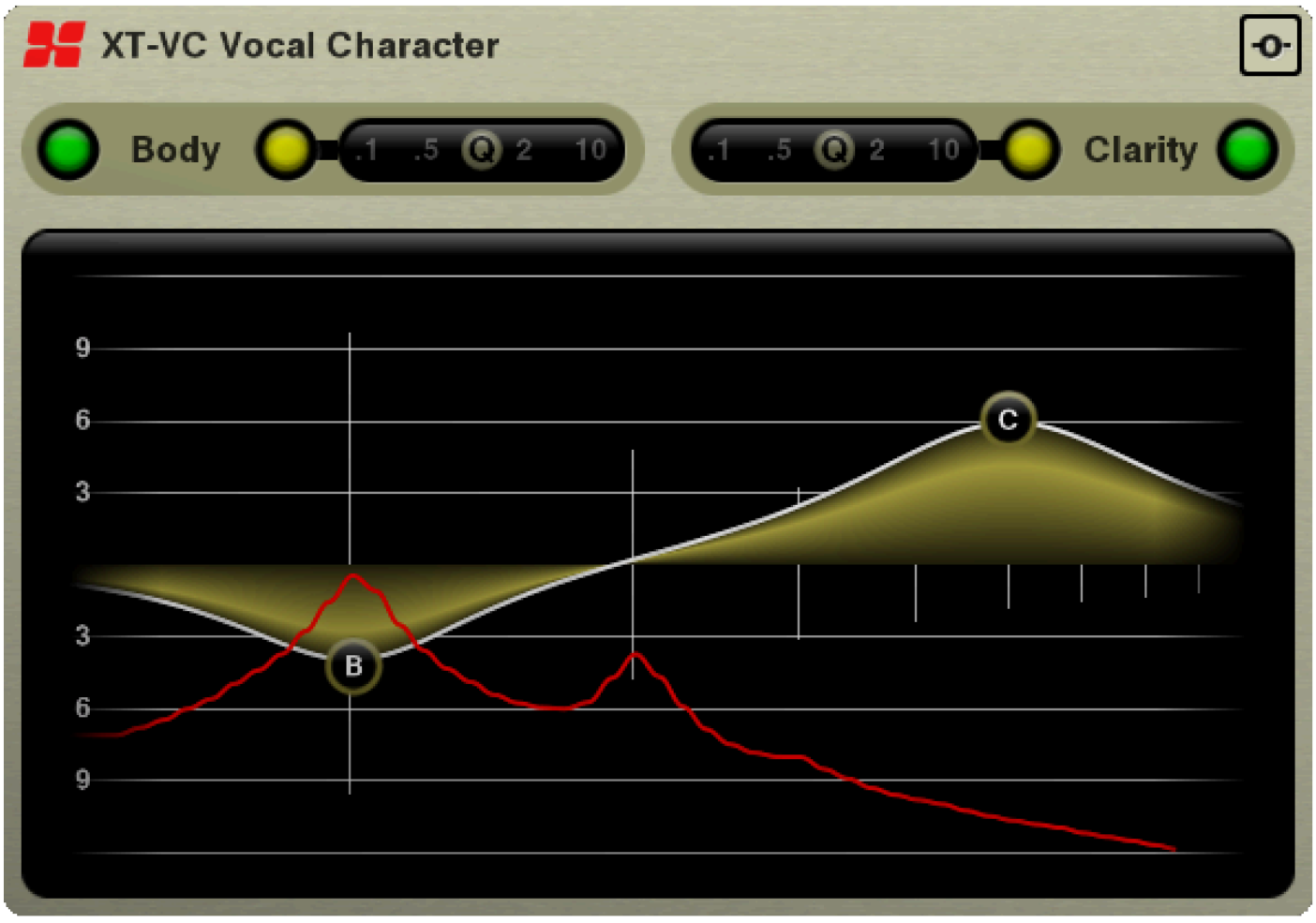
XT-VC Vocal Character is an effect similar to the XT-BC Bass Character except it is optimized for vocals.

> This plugin requires a license to enable the control panel. To purchase a license, visit Harrison's XT Plugin Page

The Vocal Character operates by detecting each musical note sung and then applying an eq to make the note sound fuller and brighter (or weaker and duller, if needed).

Best when used on: Vocals

> NOTE: the XT-VC is for mono tracks only.



Scale + Spectrum Display:

The Vocal Character plugin operates like an EQ, but with one difference: the EQ isn't set at fixed frequencies, but rather the eq is applied to each detected note.

The red "spectrum" curve shows the harmonic content of the detected note. The leftmost line indicates the "fundamental" or first harmonic of the note. Each vertical line to the right indicates an additional harmonic (2nd harmonic, 3rd, etc). You can use this graph to determine the balance of an instrument: whether it has too much or too little fundamental, for example.

Controls:

The XT-VC plugin has 2 EQ bands: "Body" and "Clarity". These bands are identical except that the "Body" control is a low-shelving EQ by default, while the "Clarity" band is a high-shelf EQ.

In/Out switch: The green button enables/disables each band.

Peaking switch: The yellow button changes the band to a peaking.

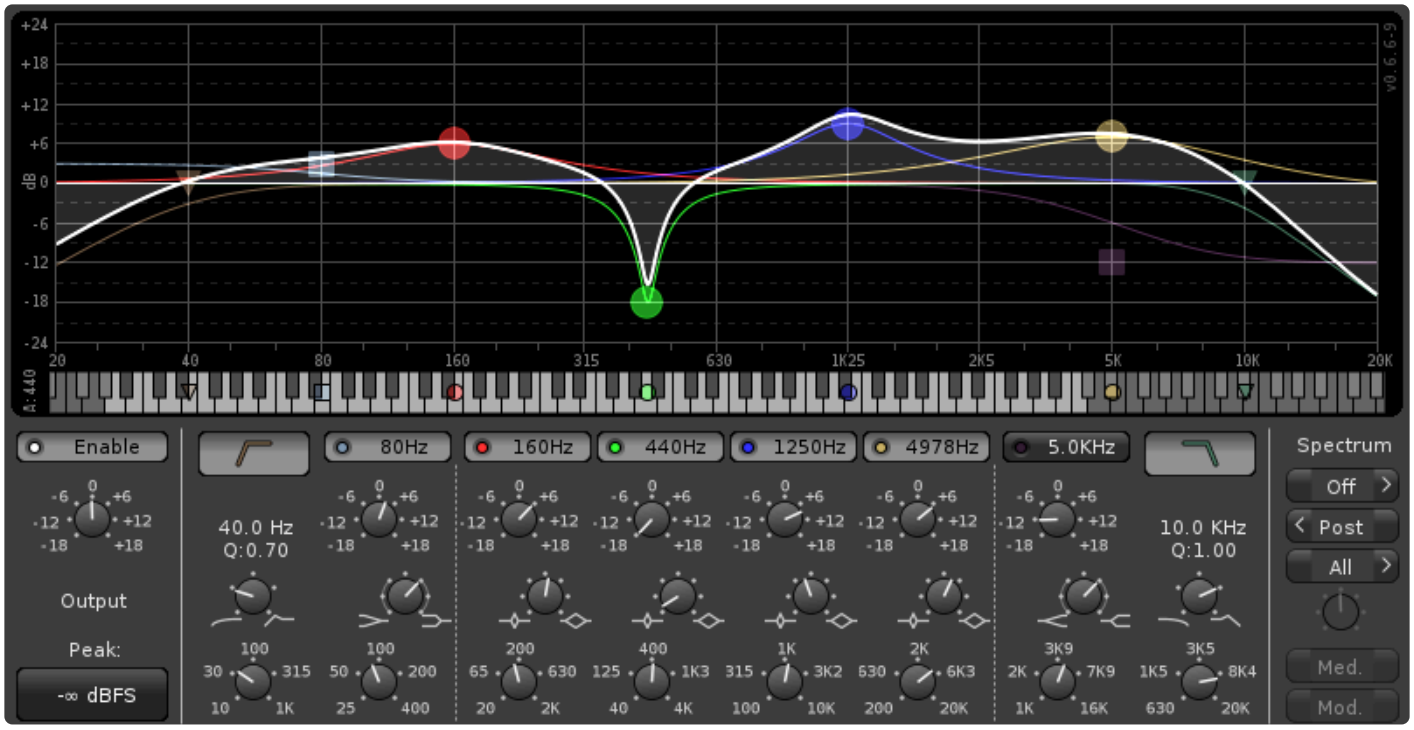
Q slider: when the Peaking switch is engaged, the Q slider engages, and adjusts the Q (sharpness) of the band's filter.

Momentary Bypass: the top-right corner is a momentary-bypass button. Press this button to temporarily bypass the effect. Releasing the button will re-apply the effect.

24.18. x42 EQ

x42 is a 4-Band Parametric EQ with Low/High Shelving, Filters, and pre- or post-EQ spectrum displays.

> This plugin requires a license to enable the control panel. To purchase a license, visit https://x42-plugins.com/x42/x42-eq



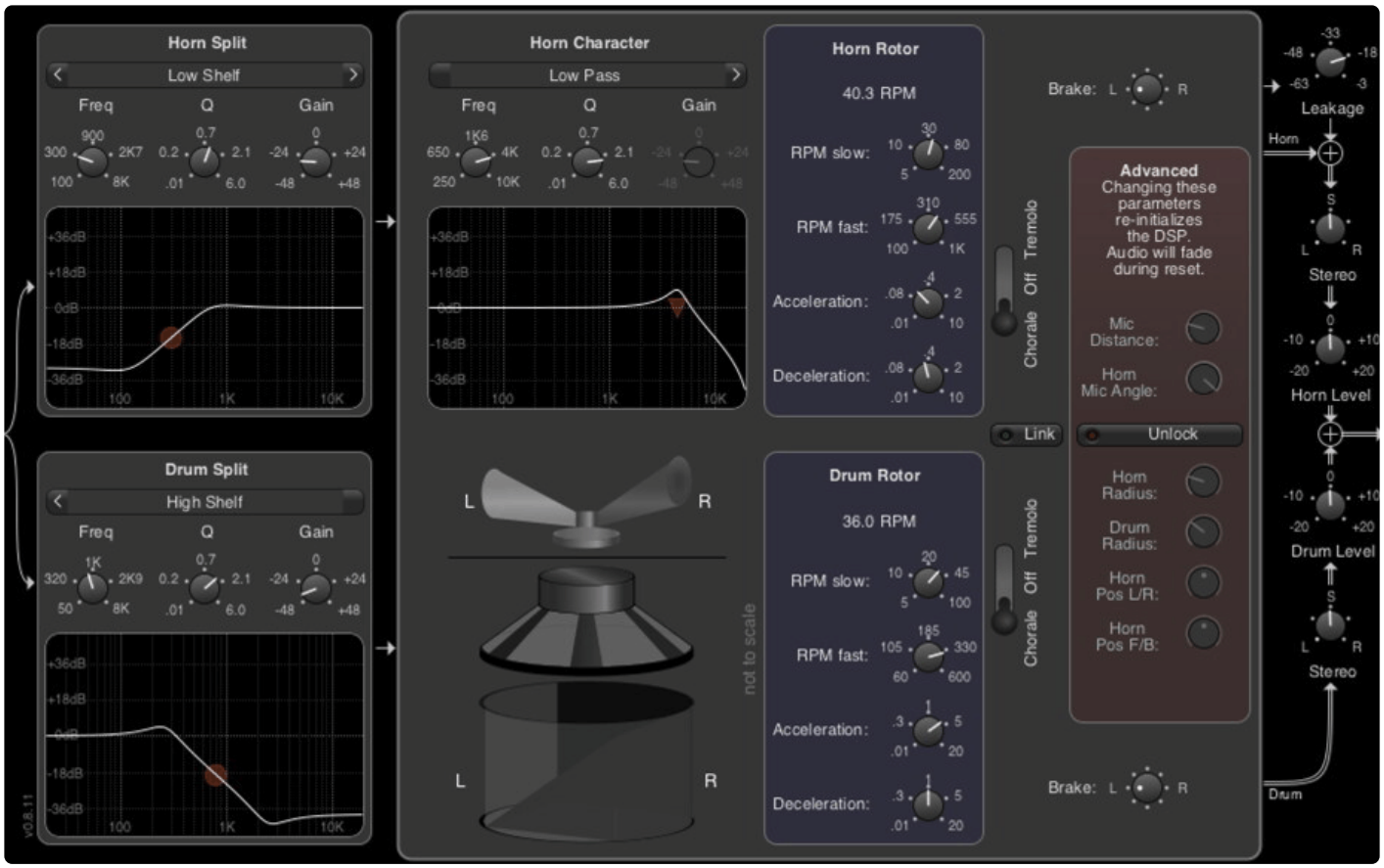
x42-eq features a “de-cramped” algorithm which accurately models analog gain, even when approaching the Nyquist frequency. It was recently updated with a piano-keyboard that snaps your EQ frequencies to the chromatic scale.

You can find more details about this plugin, and purchase a license, at the developer’s website: https://x42-plugins.com/x42/x42-eq

24.19. x42 Whirl

x42-whirl recreates the sound and properties of the electromechanical rotating speaker device that brought world-wide fame to the name and products of Don Leslie.

> This plugin requires a license to enable the control panel. To purchase a license, visit https://x42-plugins.com/x42/x42-whirl



The Whirl is a great mono-to-stereo effect which can add stereo spaciousness to keys, guitars, or even vocals.

> When inserted on a mono channel, whirl will automatically convert your mono track into a stereo output – no manual “pin connections” needed.

You can find more details about this plugin, and purchase a license, at the developer’s website: https://x42-plugins.com/x42/x42-whirl

24.20. x42 Meter Bundle

A collection of audio-level meters following established standards.

> ! By default, the Plugin Manger window only shows audio effects plugins. The x-42 meters are tagged “analyser”, and are categorized with the Utility plugins. To expose the list of meter plugins, you can click “Show Utilities Only” or type “analyser” into the search bar.

> ✻ This plugin requires a license to enable the control panel. To purchase a license, visit https://x42-plugins.com/x42/x42-meters

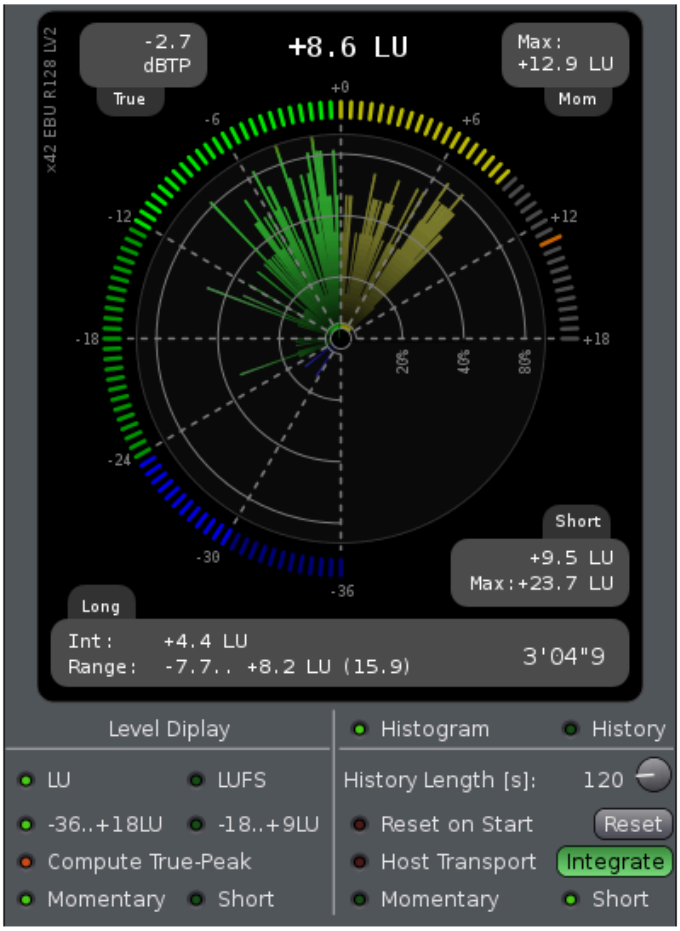
You can find more details about this plugin, and purchase a license, at the developer’s website: https://x42-plugins.com/x42/x42-meters

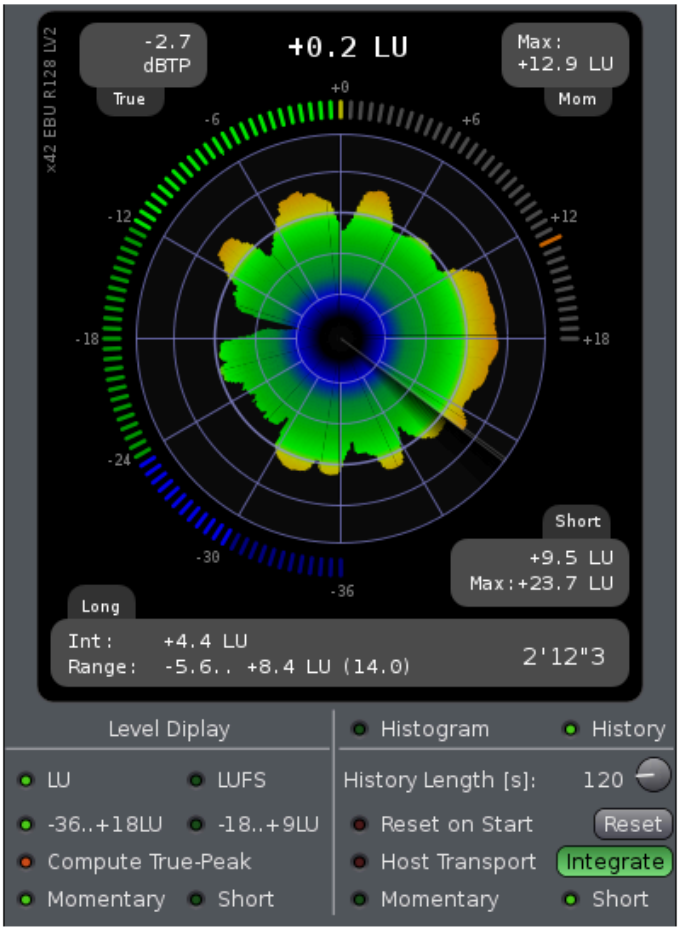
It includes needle style meters (mono and stereo variants):

- IEC 60268-10 Type I / DIN
- IEC 60268-10 Type I / Nordic
- IEC 60268-10 Type IIa / BBC
- IEC 60268-10 Type IIb / EBU
- IEC 60268-17 / VU

Stereo & Mono variants of bar-graph meters:

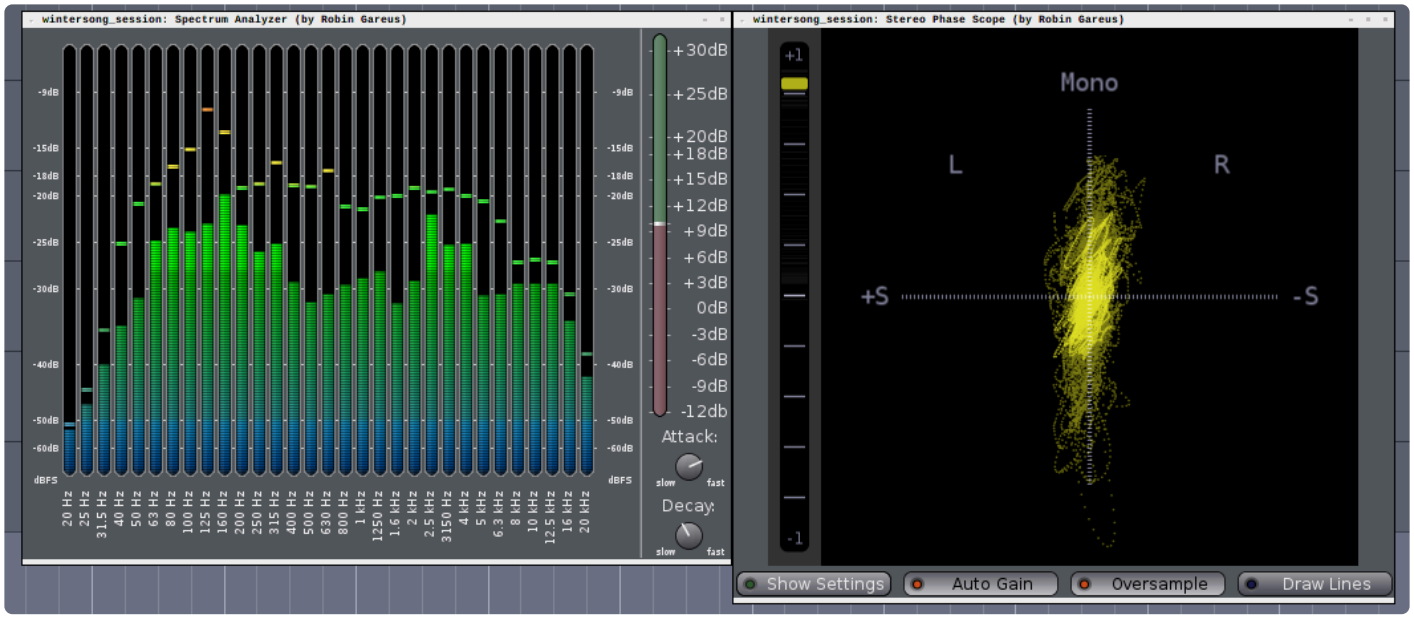
- 30 Band 1/3 octave spectrum analyzer IEC 61260
- Digital True-Peak Meter (4x Oversampling), Type II rise-time, 13.3dB/s falloff.
- True-Peak (4x Oversampling) + RMS (600ms integration time) combined with numeric readout
- K-12, K-14, K-20 / RMS type K-Meters according to the K-system introduced by Bob Katz
- DR-14 (crest factor / loudness range measurement method)

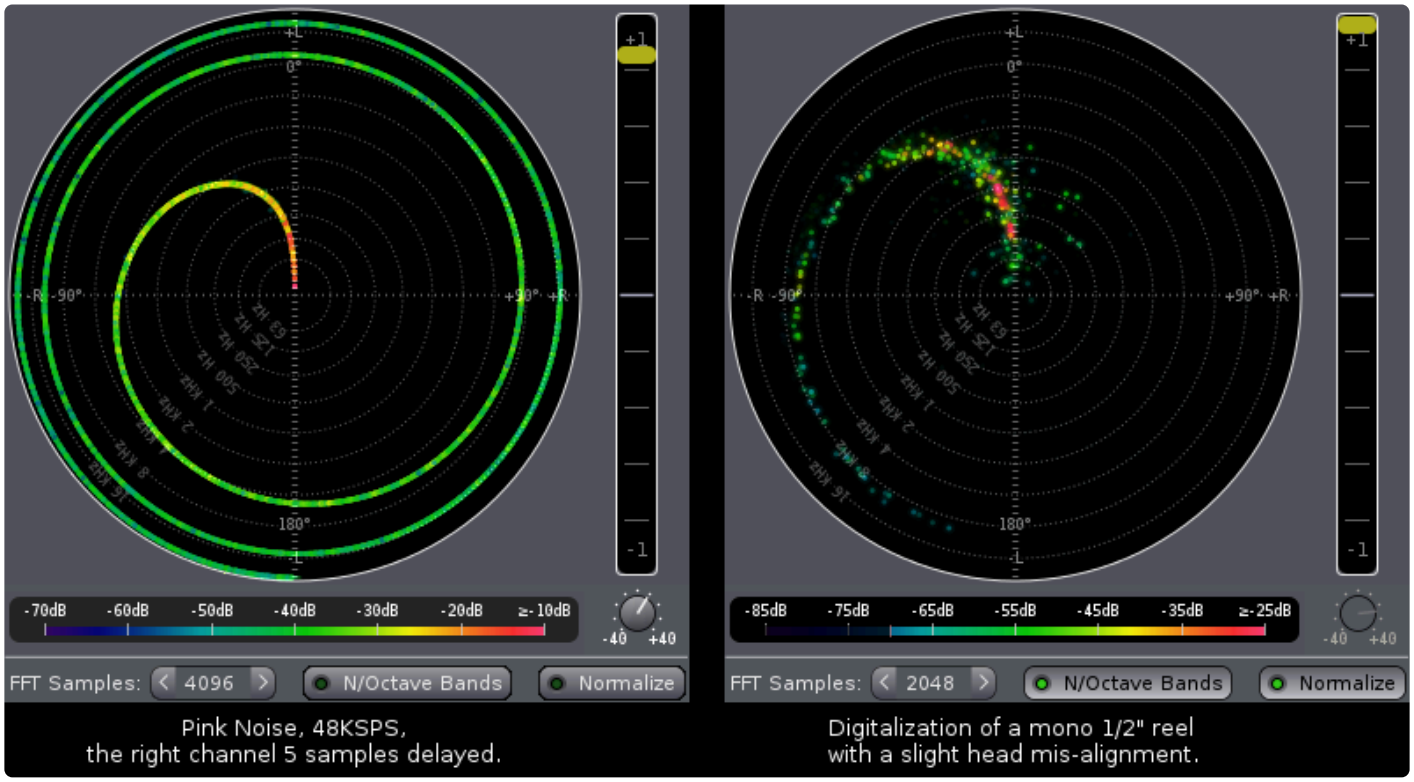




and the following stereo plugins:

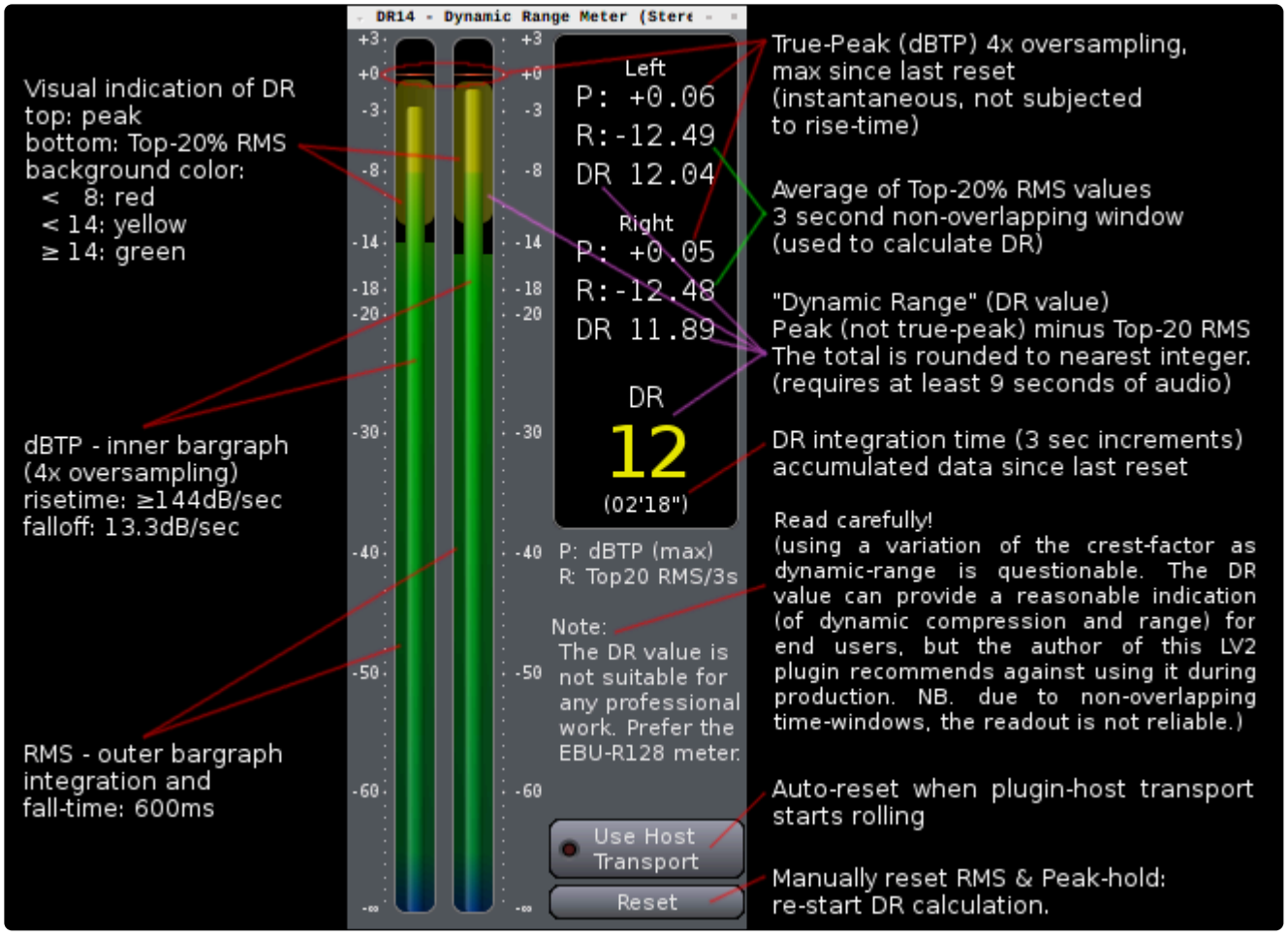
- EBU R128 Meter with Histogram and History
- Stereo Phase Correlation Meter (Needle Display)
- BBC Mid/Side M-6 (Needle Display)
- Goniometer (Stereo Phase Scope)
- Phase/Frequency Wheel
- Stereo/Frequency Monitor

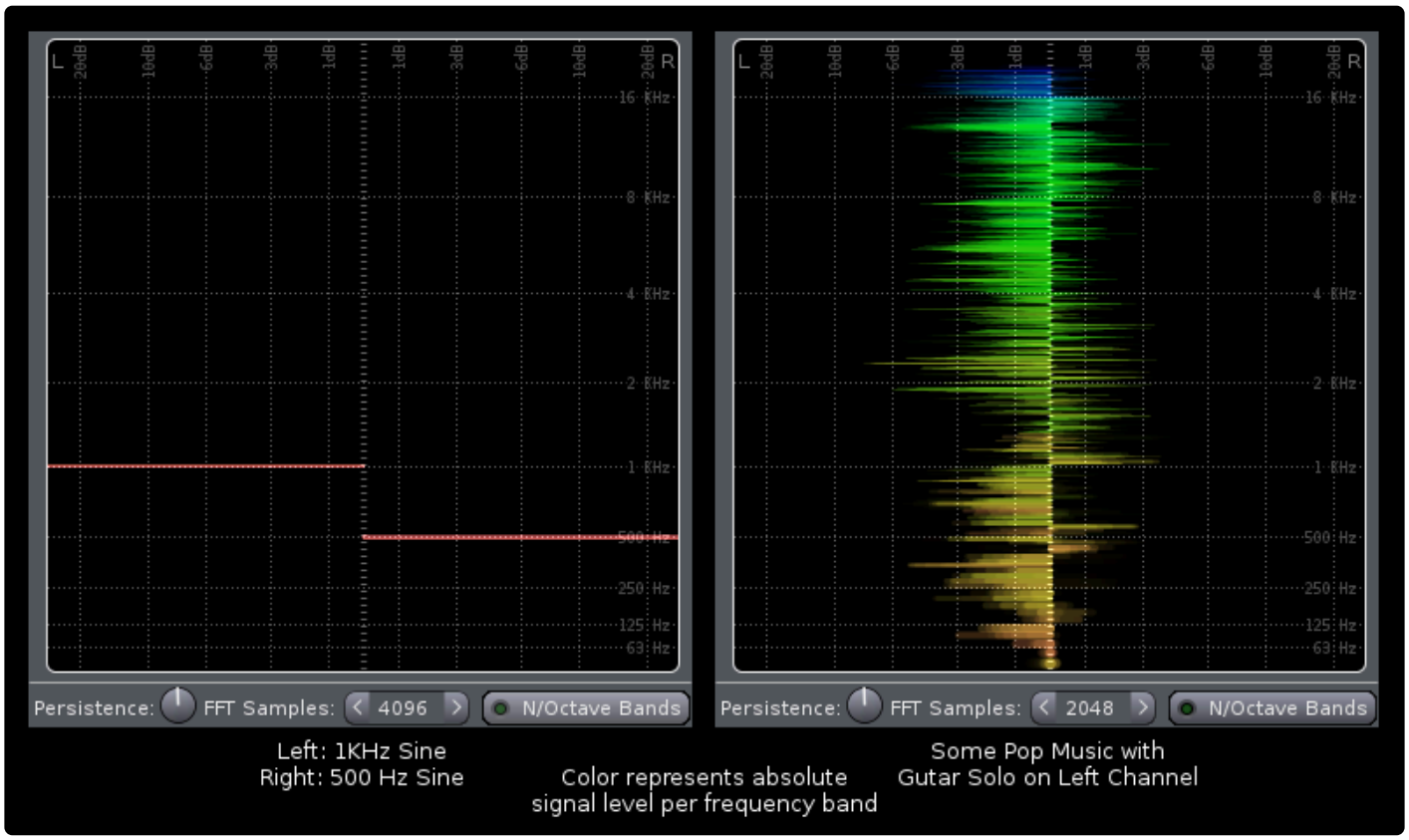




Pink Noise, 48KSPS,
the right channel 5 samples delayed.

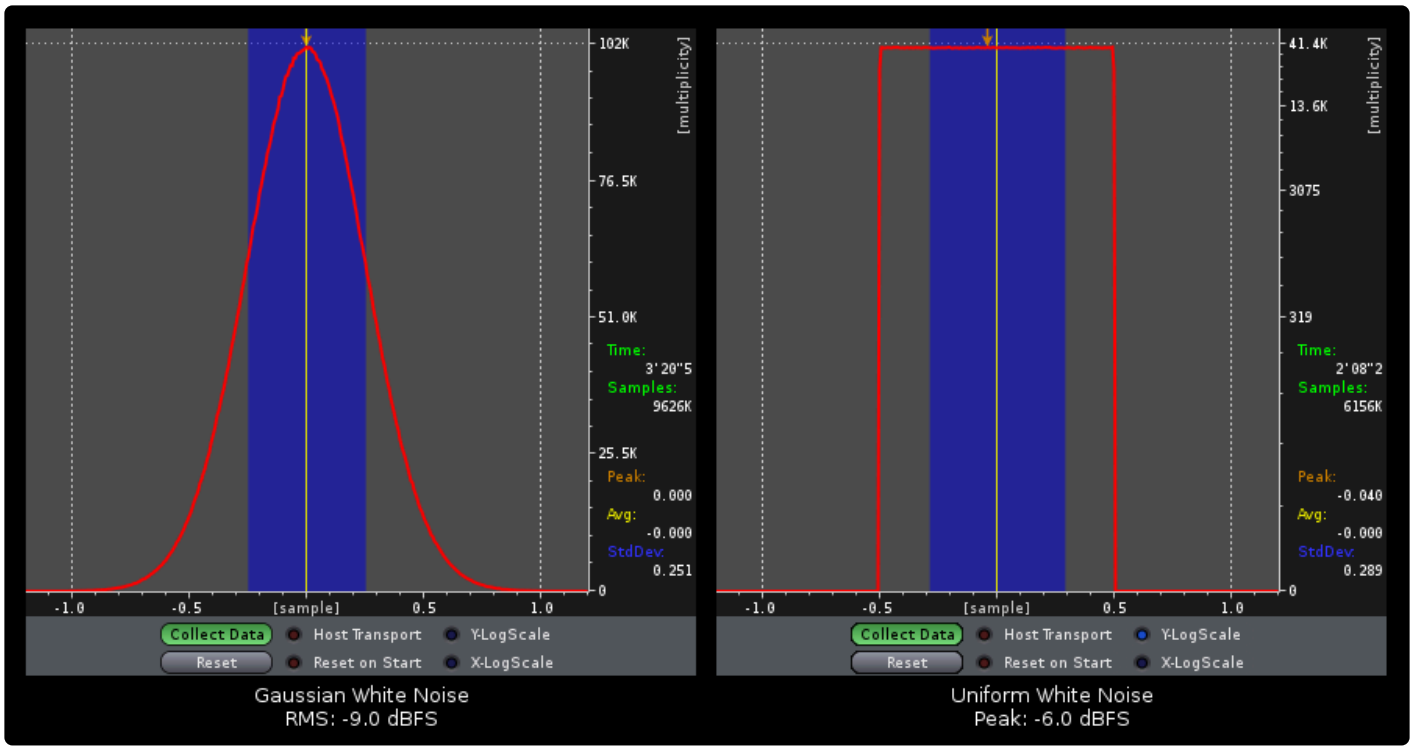
Digitalization of a mono 1/2" reel
with a slight head mis-alignment.

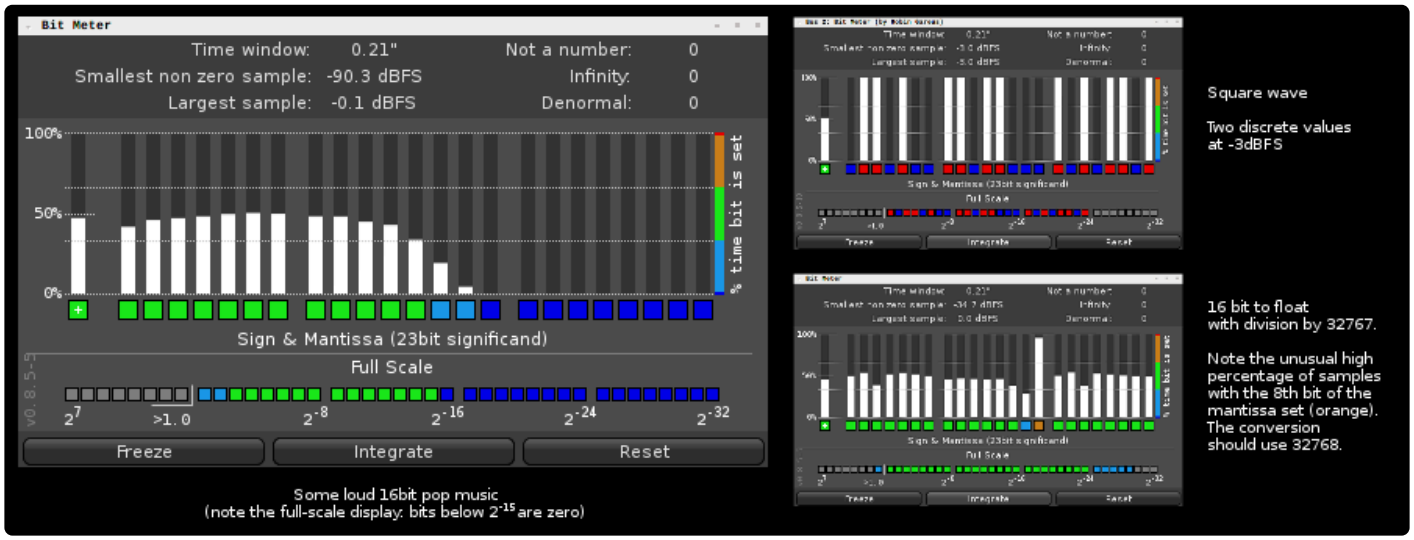

DR14 - Dynamic Range Meter (Stere
Visual indication of DR
top: peak
bottom: Top-20% RMS
background color:
< 8: red
< 14: yellow
≥ 14: green
dBTP - inner bargraph
(4x oversampling)
risetime: ≥144dB/sec
falloff: 13.3dB/sec
RMS - outer bargraph
integration and
fall-time: 600ms
Left
P: +0.06
R:-12.49
DR 12.04
Right
P: +0.05
R:-12.48
DR 11.89
DR
12
(02'18")
P: dBTP (max)
R: Top20 RMS/3s
Note:
The DR value is not suitable for any professional work. Prefer the EBU-R128 meter.
Use Host Transport
Reset
True-Peak (dBTP) 4x oversampling, max since last reset (instantaneous, not subjected to rise-time)
Average of Top-20% RMS values 3 second non-overlapping window (used to calculate DR)
"Dynamic Range" (DR value)
Peak (not true-peak) minus Top-20 RMS
The total is rounded to nearest integer.
(requires at least 9 seconds of audio)
DR integration time (3 sec increments) accumulated data since last reset
Read carefully!
(using a variation of the crest-factor as dynamic-range is questionable. The DR value can provide a reasonable indication (of dynamic compression and range) for end users, but the author of this LV2 plugin recommends against using it during production. NB. due to non-overlapping time-windows, the readout is not reliable.)
Auto-reset when plugin-host transport starts rolling
Manually reset RMS & Peak-hold: re-start DR calculation.


Persistence:
FFT Samples: 4096
N/Octave Bands
Persistence:
FFT Samples: 2048
N/Octave Bands
Left: 1KHz Sine
Right: 500 Hz Sine
Color represents absolute signal level per frequency band
Some Pop Music with Gutar Solo on Left Channel

as well as these mono, technical troubleshooting plugins:

- Signal Distribution Histogram
- Bitmeter





25. Included Plug-ins

Plugins

Mixbus comes with a suite of free plugins (in addition to the channelstrip EQ and Compressor that is built into every strip)

- ACE Amplifier (free, full-functioning)
- ACE Delay (free, full-functioning)
- ACE Reverb (free, full-functioning)
- ACE EQ (free, full-functioning)
- ACE Compressor (free, full-functioning)
- ACE Gate (free, full-functioning)
- ACE Inline Scope (source-code provided) with Inline display
- ACE Inline Spectrogram (source-code provided) with Inline display
- ACE MIDI Monitor (source-code provided) with Inline display
- x42 MIDI filter plug-ins (free, full-functioning)
- x42 Stereo Routing (free, full-functioning)
- x42-Digital Peak Limiter (free, full-functioning) with Inline display
- ACE High/Low Pass Filter (source-code provided) with Inline display

Included Instrument plugins

- x42 SetBfree modeled tonewheel organ (free, full-functioning)
- General MIDI Synth (free, full-functioning)
- RedZep drum instrument (free, full-functioning)
- Black Pearl drum instrument (free, full-functioning)
- ACE Fluid Synth (free, full-functioning)
- ACE Reasonable Synth (free, full-functioning)

25.1. ACE Plugins

The ACE (*A*rdour *C*ommunity *E*ffects) plugins are developed by the Ardour open-source community. These plugins provide a basic suite of effects including EQ, Compression, Reverb and Delay, as well as utilitarian tasks such as scrolling waveforms,

The source code for these plugins is available to be reviewed, or studied as the basis for your own plugins.

ACE EQ

The ACE EQ is a simple 6-band equalizer with 4 fully-parametric “bell” (peaking) bands, plus low-shelf and high-shelf filters.

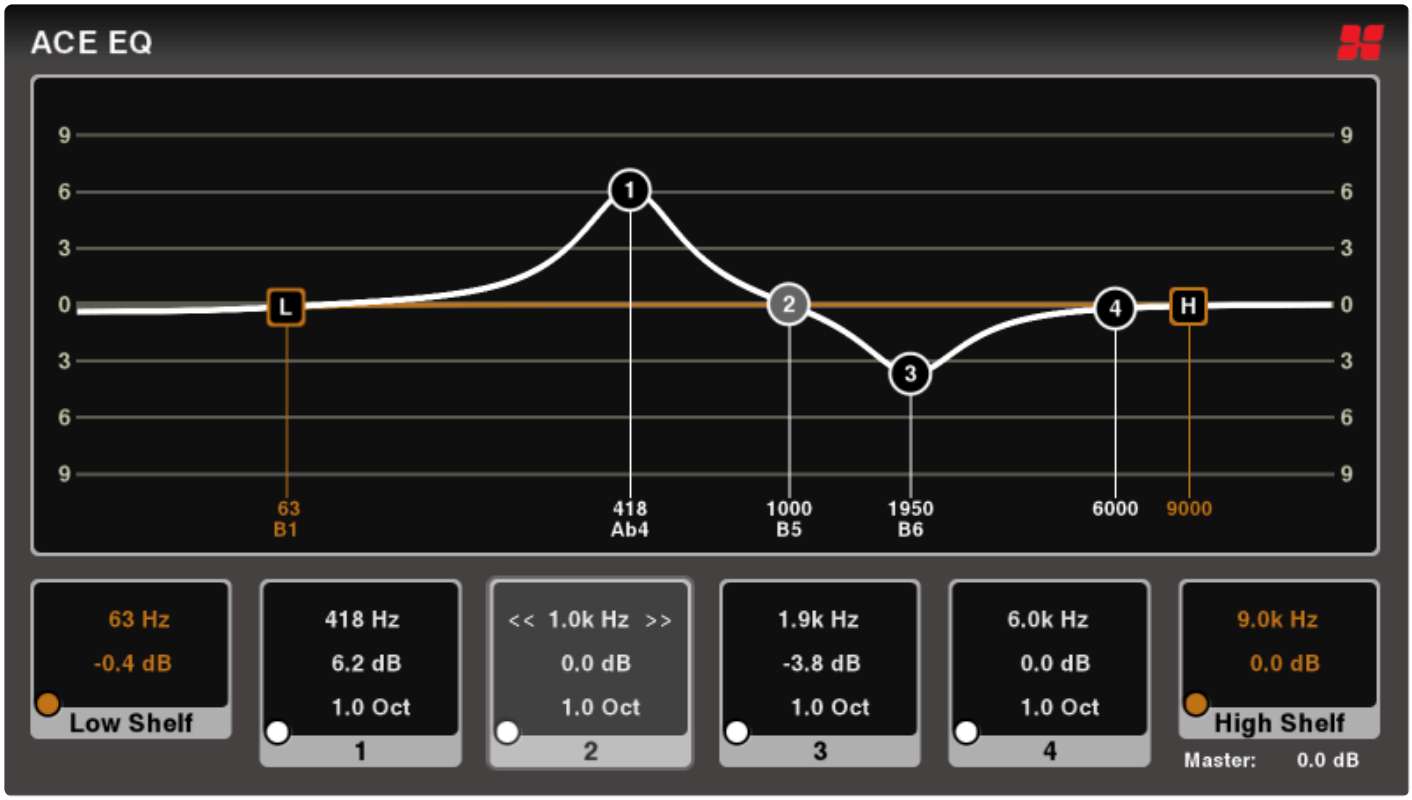
Each of the 4 bands has 3 controls:
Gain (the top knob): amount of boost or cut, in dB
Frequency: the center frequency for the boost or cut
Bandwidth: the width (in octaves) of the boost or cut. (this is the inverse of “Q” in some equalizers)

The High and Low shelves have 2 controls:
Gain: boost or cut, in dB
Frequency: the corner frequency of the shelf boost/cut.

There is an overall gain accessed by clicking on “Master” in the bottom-right corner.

The control panel has several notable features that are particularly well-suited to novice and/or student users:

- The EQ bands are always located perfectly on the curve. This is slightly different than traditional parametric EQs, and it results in a clean look.
- each EQ band displays the nearest note (on a chromatic 12-tone scale, based on A=440Hz) underneath the frequency value
- the individual bands will highlight (bi-directionally) when you enter a band’s control area. This helps to correlate the band graph with their control sliders.



Inline display: the ACE EQ plugin provides a condensed “inline” display that shows the EQ curve in the channelstrip.

ACE Compressor

A ‘compressor’ reduces the volume of an input signal when the signal exceeds the specified threshold.

There is also a *stereo* version which analyzes both left and right signals, and then applies the maximum calculated gain reduction to both sides.

ACE Compressor is a traditional compressor with the full suite of controls:

Sliders:

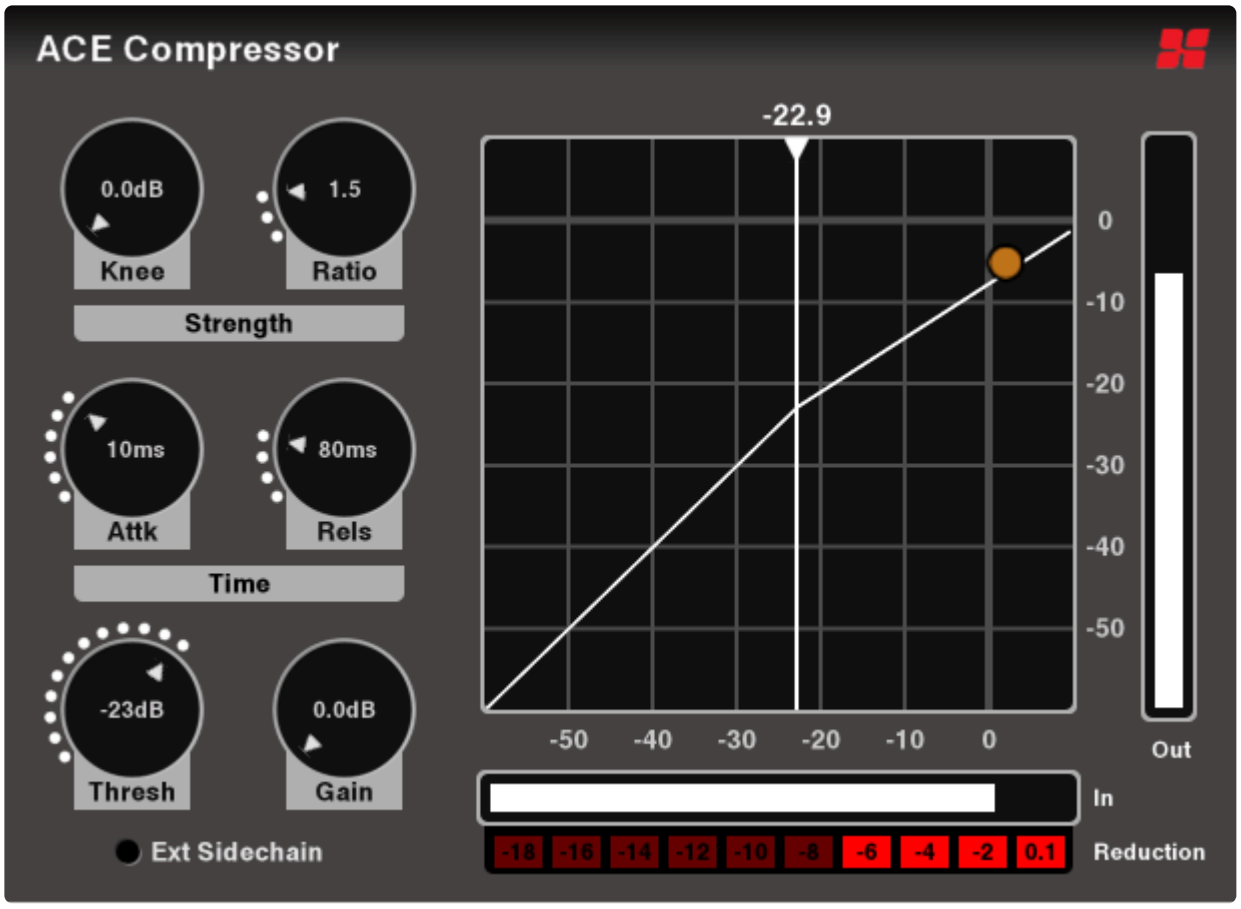
- Attack time: the time (in ms) to attain full gain reduction
- Release time: time (in ms) to return to 0dB after the signal is removed
- Knee: adjusts the aggressiveness of the gain-reduction element as the signal approaches the threshold. A larger “knee” effectively lowers the threshold but also lowers the ratio, so that the onset of gain-reduction occurs sooner but more gradually.

- Ratio: the ratio is the strength of the effect. At 1:1, no effect is applied. At 20:1, the threshold must be exceeded by 20dB before the output increases 1dB.
- Threshold: this defines the input level where the compression will begin to take effect. Decrease the threshold to compress more.
- Makeup Gain: since the effect of a compressor is to reduce the loudest signals, the result might sound quieter than before compression. Use the makeup gain to increase the level. (the overall result is to increase the level of the signal's quieter parts, therefore decreasing the dynamic range of the signal and making it easier to hear fine details)

Switches:

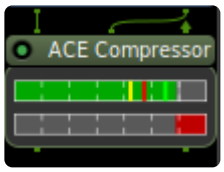
- Enable: this engages or disengages the plugin. This parameter is smoothed and may be automated.
- Sidechain: when Sidechain is engaged, the plugin ignores the input signal, and instead uses an externally-selected "sidechain" signal to feed the input detector (threshold, etc) see this topic for more details: sidechain compression.

> ! You must choose a release time that is longer than the attack time, or the behavior of the plugin will be erratic.



The compressor's transfer-curve is displayed on the control panel, displaying the input-to-output levels of the signal. A bouncing orange "dot" shows the current input-to-output ratio.

Inline display: the ACE Compressor plugin provides a condensed “inline” display when added to a mixbus channelstrip, so you can see the input signal, threshold, and resulting gain reduction:



(shown here in *stereo* mode, with the “sidechain” input disconnected)

ACE Expander

An Expander reduces the signal’s volume when the signal is below the threshold (as opposed to a compressor, which reduces the volume of a signal that is *over* the threshold). Set aggressively, an expander can serve the function of a “gate” which completely mutes the signal when it gets quiet. Sometimes called a “noise gate” because it is used to eliminate extraneous noise when the signal is below a desired threshold.

ACE Expander is a traditional expander with the full suite of controls:

Sliders:

- Attack time: the time (in ms) to attain full gain reduction
- Release time: time (in ms) to return to 0dB after the signal is removed.
- Knee: adjusts the aggressiveness of the gain-reduction element as the signal approaches the threshold. A larger “knee” effectively lowers the threshold but also lowers the ratio, so that the onset of gain-reduction occurs sooner but more gradually.
- Ratio: the ratio is the strength of the effect. At 1:1, no effect is applied. At 20:1, the threshold must be exceeded by 20dB before the output increases 1dB.
- Threshold: this defines the input level where the compression will begin to take effect. Decrease the threshold to compress more.
- Makeup Gain: since the effect of a compressor is to reduce the loudest signals, the result might sound quieter than before compression. Use the makeup gain to increase the level. (the overall result is to increase the level of the signal’s quieter parts, therefore decreasing the dynamic range of the signal and making it easier to hear fine details)

Switches:

- Enable: this engages or disengages the plugin. This parameter is smoothed and may be automated.
- Sidechain: when Sidechain is engaged, the plugin ignores the input signal, and instead uses an externally-selected “sidechain” signal to feed the input detector (threshold, etc) see this topic for more details: sidechain compression.

! You must choose a release time that is longer than the attack time, or the behavior of the

plugin will be erratic.

There is also a *stereo* version which analyzes both left and right signals, and then applies the maximum calculated gain reduction to both sides.

Inline display: the ACE Expander plugin provides a condensed “inline” display when added to a mixbus channelstrip, so you can see the input signal, threshold, and resulting gain reduction:

ACE Delay

The main display

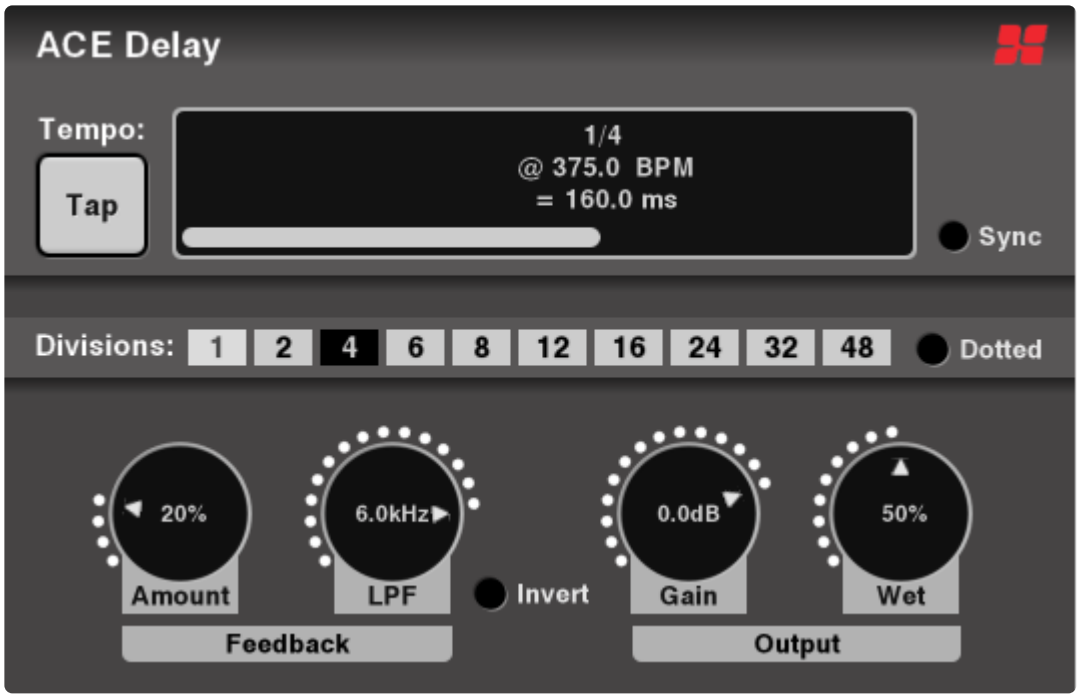
Sliders:

- Time: the length of the delay from the initial input sound. Adjusting the slider displays both the milliseconds and the BPM (referenced to the Divisor and Dotted selections)
- Divisor: (only when Sync BPM is enabled): sets teh delay time based on Mixbus’s current tempo
- Dry/Wet: when set to 0% (minimum), the input signal is passed through without any delay. When set to 100%, the input signal is muted, and only the delay signal is heard.
- Feedback: the output of the delay can be sent through the delay again. This results in multiple echoes. Setting this value very high can result in infinitely repeating echoes.
- LPF: A low-pass filter applied to the delay path. Reduce this value to recreate a vintage tape delay, where each repeated echo gets duller.
- Output Gain: increases or decreases the output level of the effect

Switches:

- Tap: successively click this button with your mouse to set the delay time (in pulses of quarter-notes)
- Invert: When enabled, the feedback path’s polarity (phase) is inverted, resulting in a slightly different character.
- Sync: When enabled, the plugin will use the Divisor selector to set the delay time, and ignore the Time setting.
- Dotted: When enabled, the delay-time will be 1.5x the value of the divisor: “dotted 8th” means an effective delay of “3 16ths”.

- Enable



The default settings for the delay are intended for for insertion directly on a track. If you want to use the delay as a send/return effect on a bus, you should set “Dry/Wet” to its maximum value.

The remaining ACE plugins do not provide a dedicated control panel. Instead they use Mixbus’s “Generic Plugin Control Panel”

ACE Reverb

ACE Reverb is a minimal reverb plugin that gives fast&easy results.

Sliders

- Blend: at 0 (minimum), the signal is dry. At 1.0 (maximum) the reverb is purely wet.
- Room Size: increase this slider to increase the perceived room size, and make the decay longer.

The default settings for the reverb are intended for for insertion directly on a track. If you want to use the reverb as a send/return effect on a bus, you should set “Blend” to its maximum value.

ACE Amplifier

This is a simple clean volume control. The only parameter is “Gain” which may be adjusted +/-20dB

ACE High/Low Pass Filter

This plugin provides a high-pass and low-pass filter.

- Steepness: filter order may be adjusted from 12 to 48dB/octave. Higher numbers indicate a steeper cutoff filter.
- Frequency: the cutoff frequency of the filter. frequencies out side of this range will be greatly reduced.
- Resonance: increasing the resonance will create a “bump” near the cutoff frequency, and further sharpen the filter steepness. Extreme settings of resonance can result in very large signal peaks.

Inline display: the ACE High/Low Filter plugin provides a condensed “inline” display when added to a mixbus channelstrip, so you can see the input signal, threshold, and resulting gain reduction:

ACE Inline Scope

The ACE Inline scope displays a scrolling waveform view inside the mixer strip:

Controls:

- Logscale: if the signal is quiet, then enabling Logscale will increase the size of the waveform and reveal quiet material, at the expense of less visual resolution for loud signals.

- Timescale: the display can show from 0.1seconds to 5 seconds at one time. Adjust this to scroll faster or slower.
- Height: this adjusts the height of the waveform display in the strip

ACE Inline Spectrogram

The spectrogram displays a scrolling view of the signal's frequency content, with low frequencies on the left, and high frequencies on the right.

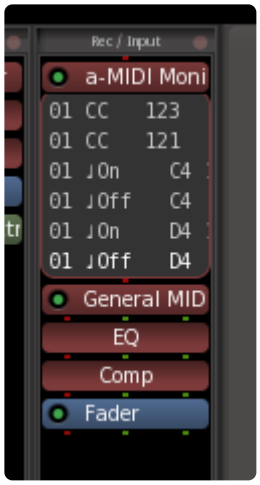
Controls:

- Logscale: if the signal is quiet, then enabling Logscale will increase the size of the waveform and reveal quiet material, at the expense of less visual resolution for loud signals.
- FFT Size: smaller FFT sizes will result in a faster-scrolling display, but doesn't show low-frequencies as well.
- Height: this adjusts the height of the spectrum display in the strip.
- Range, Offset: if the display is too saturated, or nothing shows up, you can adjust these values to better match your input signal level.

ACE MIDI Monitor

ACE MIDI Monitor is a troubleshooting tool that lets you see the flow of MIDI data in your tracks.

The plugin has an inline display that reports MIDI note "on/off" and MIDI controller messages directly in the mixer strip. In the example image below, ACE MIDI Monitor is inserted directly before the "General MIDI" synth and is reporting the controller and note-messages that are being sent to the synth.



ACE Notch Bank

This plugin is useful to remove noise with a harmonic spectrum such as mains power hum, or motor noise.

Select the base frequency to match the fundamental of the noise (60Hz for U.S. power main), and choose the number os stages to increase the number of harmonic filters above the fundamental. The Quality control is the sharpness or “Q” of the filters.

Additionally, this plugin serves as an example DSP plugin written in the Lua scripting language. The source code for this plugin can be found in your Mixbus application bundle, in Scripts/notch-bank.lua

ACE Slow-Mute

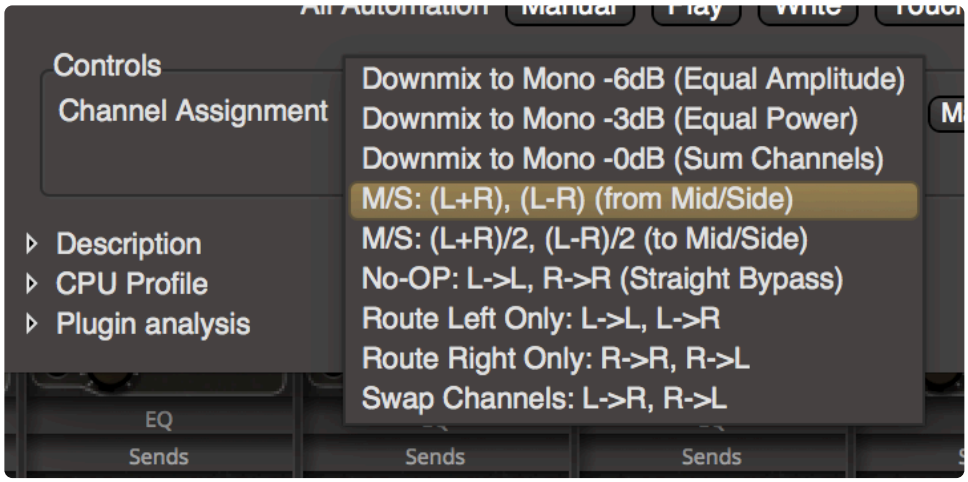
Another example DSP plugin written in the Lua scripting language. The source code for this plugin can be found in your Mixbus application bundle, in Scripts/ACE slow-mute.lua

This plugin slowly mutes the signal when “Mute” is enabled. De-selecting “Mute” will return the signal to full volume.

This plugin has no other controls.

25.2. Stereo Routing

This simple plugin provides various utilities for stereo tracks and buses. You can use this plugin to reverse left/right sides, and other utilitarian tasks.



The MId/Side converter is provided for tracks that were recorded using the mid/side techniques. Use the decoder to resolve a mid/side recording into normal stereo signals. A mid-side encoder is also provided, allowing you to convert the signal to mid/side format and back. Converting to mid-side before a processor, and then back to stereo, can sometimes provide a useful effect. You can use features like pin management to process only the ‘mid’ or ‘side’ channel.

25.3. x42 Instrument Tuner

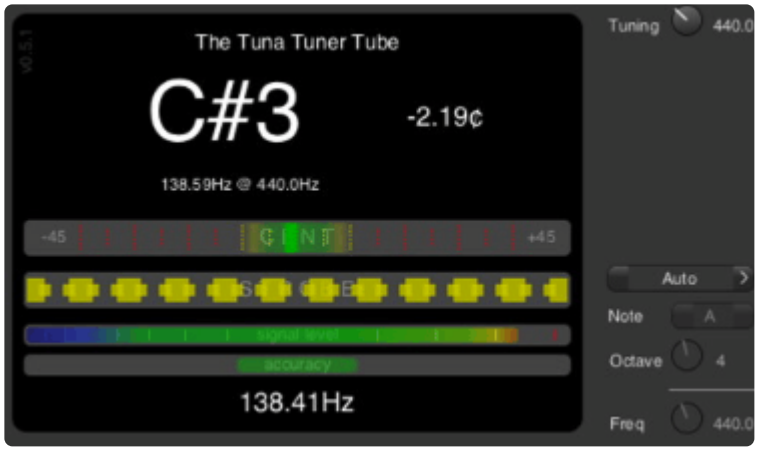
x42 Instrument tuner (or “Tuna”) is a high-quality tuner for acoustic and electric instruments.

> You can learn more about the x42 Instrument Tuner at the developer’s web site: x42

“Tuna” features an auto-tuner mode (detect closest note) with an idiosyncratic GUI and optional spectrum display. Unlike many other (FFT based) tuners, the algorithm follows the signal phase of the fundamental note, which makes this tuner suitable for stretch tuned instruments.

Tuna also features an “inline” channelstrip display which allows it to remain visible on the mixer window during recording or playback.

Best when used on: Monophonic instruments and vocals.



Tuner Display:

The tuner display is separated into 4 parts:

Text Display: The topmost text displays the pitch that is nearest the detected (input) note, and a numeric display of the tuning error from the reference.

Tuner bar: the colored bar (green in the middle) is a representation of a traditional tuner, where center indicates a perfectly in-tune note reference, and the left&right sides indicate when the note is flat or sharp.

Strobe: The third bar (moving boxes) is a visual indicator of the difference between the reference pitch and the input. Sometimes called a “strobe” display, this provides a strong visual indication of the incoming signal’s phase when compared to the reference…. this display often feels smoother & easier to match, than the traditional flat/sharp display.

Signal Level: this indicates the detected signal level. If the level is too quiet, then the tuner will not try to

detect the pitch. If the signal is too high, it might indicate clipping which could invalidate the pitch detection.

Accuracy: If the pitch of the input signal varies widely, then the tuner will report that accuracy is compromised.

Controls:

Tuning: This is the reference for tuning of note “A”. Default is A=440Hz.

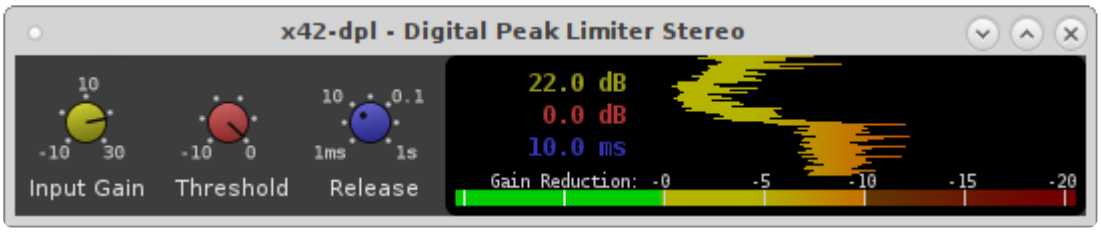
Auto/Freq/Note selector: Auto: the tuner will try to detect incoming pitch. Freq: user defines the reference frequency to tune to. Note: user defines the reference note. The latter options are provided in the rare case that the tuner cannot auto-detect the note you are trying to tune to.

Inline Mode

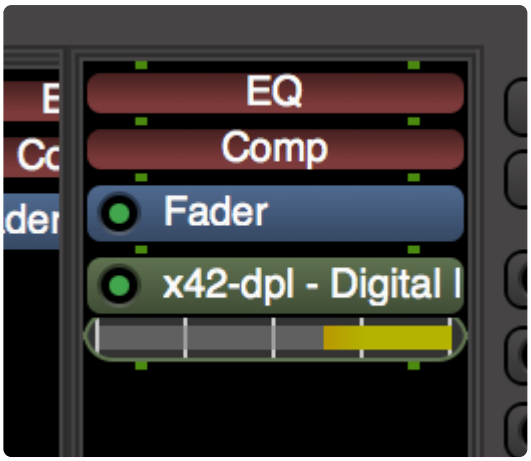
When placed on a channelstrip in Mixbus, the Tuna plugin will appear with an inline display. This provides a quick reference to check that your instrument input, or playback, is in tune.

25.4. x42 DPL Limiter

x42-dpl is a look-ahead digital peak limiter based on Fons Adriaensen's DPL-1. It is intended to be used as final step to avoid clipping when mixing or mastering, but can also be used on individual tracks as well. It comes in mono and stereo variants, the stereo version applies the same gain reduction to both channels.



A channel "inline" display is provided to give an indication when gain-reduction is being applied:



You can find more details about this plugin, and purchase a license, at the developer's website: https://x42-plugins.com/x42/x42-limiter

25.5. Instruments

Mixbus includes several Instrument plugins which can interpret your MIDI performance into sound.

25.5.1. ACE Fluid Synth

This is a plugin implementation of Fluid Synth, a popular SoundFont playback engine.

ACE Fluid Synth can open SoundFont (.sf2 and .sf3) patch files.

A SoundFont consists of a library of sampled sounds as well as accompanying information on note mapping, sound layering, velocity sensitivity, and other parameters.

While not as powerful as commercial sample libraries like Kontakt, SoundsFonts are readily available online and can be quite high quality. Generally speaking, a SoundFont package can match the capabilities of any state-of-the art “arranger” keyboard from 1995 through 2005.

Fluid Synth uses Mixbus’s generic control panel.

To use Fluid Synth, you must first download a Soundfont (.sf2 or .sf3) file, and then select it using the (SF2 File) browser in the control panel:

A built-in reverb and chorus effect can be utilized to audition sounds without assigning additional plugins on your mixer.

25.5.2. ACE Reasonable Synth

“Reasonable Synth” is a minimal instrument that makes a piano-like sound. RS responds to note inputs on every channel. It is useful to troubleshoot MIDI problems such as verifying that your MIDI keyboard is connected and working.

RS has no user-adjustable controls.

Because of its simplicity, RS is a great way to learn about MIDI and plugin signal flow in Mixbus. Add a chorus or phaser effect after RS, and you’ll get a reasonably good electric piano sound.

25.5.3. AVL Drumkits: Black Pearl and Red Zeppelin

The AVL drum kits were developed by Glen McArthur at AVLinux, and packaged by Robin Gareus for use in Mixbus.

When compared to the drumkits in Mixbus's "General MIDI Synth", the AVL drumkits utilize many more velocity layers, and therefore can deliver a more nuanced and "real" sounding drum kit, when this is desired.

The AVL drumkits may output each drum sound on a separate output, and they serve to demonstrate Mixbus's multichannel instrument outputs. To learn more, see: "

The "Black Pearl" drumkit uses a smaller kick drum and a tight snare for a traditional "jazz" drum setup:

The "Red Zeppelin" drumkit uses a larger, boomier kick and a brighter, more sustaining snare for a more aggressive"rock" sound:

RED ZEPPELIN
Crash Cymbal 1
Zildjian
AV
LINUX

25.5.4. General MIDI Synth

This synthesizer implements the General MIDI (GM) soundset, and various common extensions.

The synth responds to all 16 MIDI channels, as well as common control messages like pitch-bend, volume, and filter cutoff.

For more details about using the General MIDI Synth, see: Multitimbral Instruments and “General MIDI”

25.5.5. SetBfree Tonewheel Organ

SetBfree is a faithful recreation of the tonewheel organs popularized by the Hammond organ company.

Many instruments can be convincingly recreated by sampling each note; but the complex character and many controls of a tonewheel organ require a more complex emulation. SetBfree is a faithful emulation of every stage inside these iconic instruments.

Operation:

SetBfree responds to 3 different MIDI channels (1,2, and 3) for each of the 3 timbres (). This allows playing the instrument from 3 simultaneous inputs (2 manual keyboards plus foot pedals). The various switches and manuals may be controlled by your MIDI device controllers. (there are default config files for common keyboards such as the Kurzweil 2500 or Oxygen61 included)

When opened, the plugin's UI shows the most frequently accessed controls:

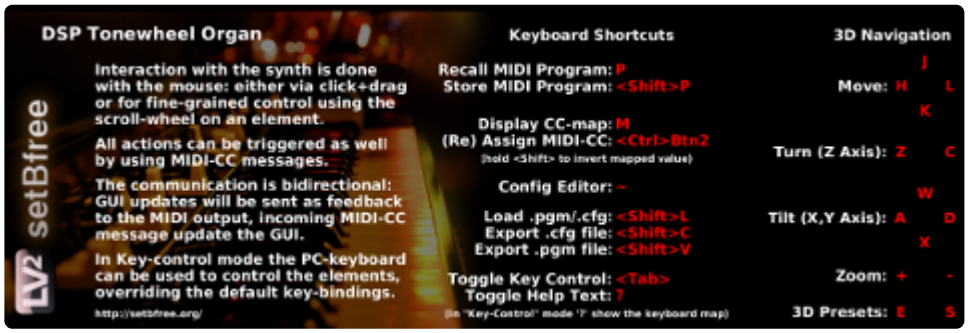
Controls
Users of a tonewheel organ will recognize the stops (sliders) and switches of the real thing.

- Stop sliders: (2 for the bass pedals, and 9 for each of the 2 manuals): adjust the overtones of the timbre.
- Overdrive/Gain: (not present on the real device): this adjusts the amount of tube distortion in the signal.
- Vibrato: these switches engage the vibrato for the 2 manual keyboards.
- V1/C1, etc … these switch positions select between various vibrato and chorus effects
- Tremolo (Drum and Horn) … these switches effect the speed of vibrato sent to the low (drum) and high (high) speakers (for a full implementation of an accompanying rotating speaker, see whirl)
- Percussion: enabling these options provides a "click" (either soft or pronounced) when a manual key is pressed.

- Reverb: a built-in reverb effect recreates the sound of the instrument in a room, as it was likely to be heard.
- Volume: master output level.

Detailed Operation:

Click the question-mark near the top-right corner to expose some additional functions that might be useful when performing with the instrument:



In addition to the sound modeling, the organ is displayed as a 3-dimensional model. This allows the organ to be manipulated in various ways, including playing the various manuals and changing the internal components (and their effect on the sound):

More details about this plugin can found at the developer's site: setBfree
Full source-code for the organ is provided at his site.

25.6. MIDI Plugins

Mixbus provides a rich suite of MIDI Effects, utilities, and generators.

25.6.1. MIDI Filters

Mixbus includes a free suite of open-source MIDI filters by developer Robin Gareus.

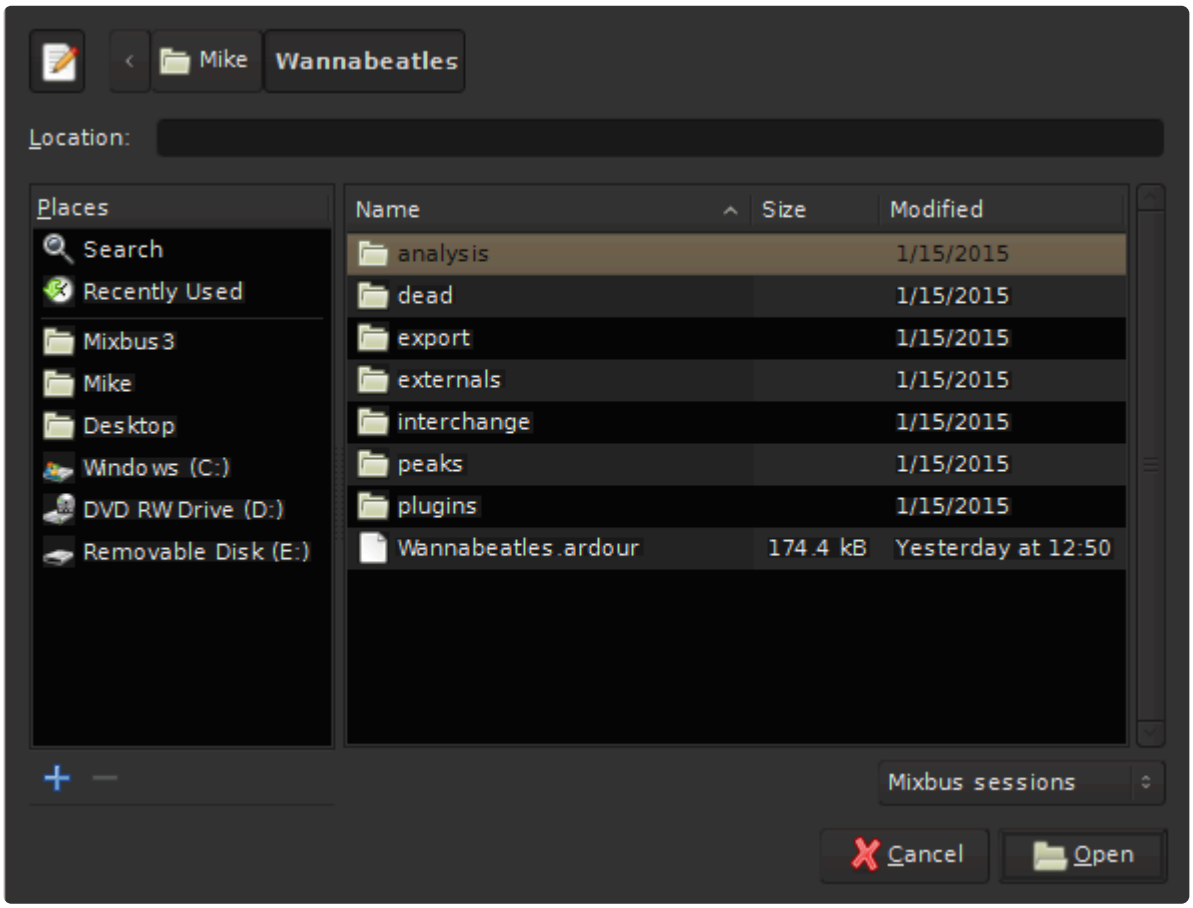
Name	Description
CC2Note	Convert MIDI control change messages to note-on/off messages. Note off is queued 10msec later.
MIDI Channel Filter	Simple MIDI channel filter. Only data for selected channels may pass. This filter only affects midi-data which is channel relevant (ie note-on/off, control and program changes, key and channel pressure and pitchbend). MIDI-SYSEX and Realtime message are always passed on. This plugin is intended for live-use, button-control. See also 'MIDI Channel Map' filter.
MIDI Channel Map	Rewrite midi-channel number. This filter only affects midi-data which is channel relevant (ie note-on/off, control and program changes, key and channel pressure and pitchbend). MIDI-SYSEX and Realtime message are always passed thru unmodified.
MIDI Enforce Scale	Filter note-on/off events depending on musical scale. If the key is changed note-off events of are sent for all active off-key notes.
MIDI Event Filter	Notch style message filter. Suppress specific messages. For flexible note-on/off range see also 'keyrange' and 'velocityrange'.
MIDI Key-Range Filter	This filter allows to define a range of allowed midi notes. Notes-on/off events outside the allowed range are discarded. If the range changes, note-off events are sent to currently active notes that end up outside the valid range.
MIDI Keysplit	Change midi-channel number depending on note. The plugin keeps track of transposed midi-notes in case and sends note-off events accordingly if the range is changed even if a note is active. However the split-point and channel-assignments for each manual should only be changed when no notes are currently played.
MapCC	Change one control message into another — combine with scalecc to modify/scale the actual value.
MIDI Keys Transpose	Flexible 12-tone map. Allow to map a note within an octave to another note in the same octave-range +- 12 semitones. Alternatively notes can also be masked (disabled). If two keys are mapped to the same note, the corresponding note on/events are latched: only the first note on and last note off will be sent. The settings can be changed dynamically: Note-on/off events will be sent accordingly.
MIDI Chord	Harmonizer – make chords from single (fundamental) note in a given musical scale. The scale as well as intervals can be automated freely (currently held chords will change). Note-ons are latched, for multiple/combined chords only single note-on/off will be triggered for the duration of the combined chords. If a off-scale note is given, it will be passed through – no chord is allocated. Note: Combine this effect with the 'MIDI Enforce Scale' filter to weed them out.

MIDI Delayline	MIDI delay line. Delay all MIDI events by a given time which is given as BPM and beats. If the delay includes a random factor, this effect takes care of always keeping note on/off events sequential regardless of the randomization.
MIDI Channel Unison	Duplicate MIDI events from one channel to another.
MIDI Strum	A midi arpeggio effect intended to simulate strumming a stringed instrument (e.g. guitar). A chord is ‘collected’ and the single notes of the chord are played back spread out over time. The ‘Note Collect Timeout’ allows for the effect to be played live with midi-keyboard, it compensates for a human not pressing keys at the same point in time. If the effect is used with a sequencer that can send chords with all note-on at the exactly time, it should be set to zero.
MIDI Chromatic Transpose	Chromatic transpose of midi notes and key-pressure. If an inversion point is set, the scale is mirrored around this point before transposing. Notes that end up outside the valid range 0..127 are discarded.
MIDI Monophonic Legato	Hold a note until the next note arrives. — Play the same note again to switch it off.
MIDI Remove Active Sensing	Filter to block all active sensing events. Active sensing messages are optional MIDI messages and intended to be sent repeatedly to tell a receiver that a connection is alive, however they can clutter up the MIDI channel or be inadvertently recorded when dumping raw MIDI data to disk.
MIDI Duplicate Blocker	MIDI Duplicate Blocker. Filter out overlapping note on/off and duplicate messages.
Note2CC	Convert MIDI note-on messages to control change messages.
MIDI Note Toggle	Toggle Notes: play a note to turn it on, play it again to turn it off.
MIDI N-Tap Delay	This effect repeats notes N times. Where N is either a fixed number or unlimited as long as a given key is pressed. BPM and delay-time variable allows tempo-ramps. On every repeat the given velocity-adjustment is added or subtracted, the result is clamped between 1 and 127.
MIDI Thru	MIDI All pass. This plugin has no effect and is intended as example.
MIDI Quantization	Live event quantization. This filter aligns incoming MIDI events to a fixed time-grid. Since the effect operates on a live-stream it will introduce latency: Events will be delayed until the next ‘tick’. If the plugin-host provides BBT information, the events are aligned to the host’s clock otherwise the effect runs on its own time.
MIDI Velocity Randomization	Randomize Velocity of MIDI notes (both note on and note off).
Scale CC Value	Modify the value (data-byte) of a MIDI control change message.

MIDI Sostenuto	This filter delays note-off messages by a given time, emulating a piano sostenuto pedal. When the pedal is released, note-off messages that are queued will be sent immediately. The delay-time can be changed dynamically, changes do affects note-off messages that are still queued.
MIDI Velocity-Range Filter	Filter MIDI note-on events according to velocity. Note-on events outside the allowed range are discarded. If a Note-off is received for a note that was previously filtered, it is also not passed though. If the allowed range changes, note-off events are sent to currently active notes that end up outside the valid range.
MIDI Velocity Adjust	Change the velocity of note events with separate controls for Note-on and Note-off. The input range 1 – 127 is mapped to the range between Min and Max. If Min is greater than Max, the range is reversed. The offsets value is added to the velocity event after mapping the Min/Max range.

26. Power-User Topics

Session folder layout



Navigating to a session folder will display a series of files and folders:

- instant.xml : this file stores the most recently-used mouse mode, playhead location, and other transitory information.
- {snapshot name}.ardour : each of these files represents a Snapshot, which stores all of the edits, settings, and properties of a session.
- {snapshot name}.bak : backup files are occasionally generated for each snapshot
- {snapshot name}.pending : pending files snapshots that get created when an audio recording is initiated. this allows us to recover audio data, even in the event of a crash or power outage. If a pending file is found when Mixbus is re-launched, you will see the Crash Recovery dialog, and you can choose to ignore or recover any lost information that was saved to the pending file.
- {snapshot name}.history : every snapshot file has a “history” file. The history file saves the UNDO history for that snapshot.
- analysis: This folder contains data relevant to any audio files that Mixbus has analyzed for transient-detection and note-detection. By storing this information, Mixbus doesn’t have to re-analyze audio when you initiate one of these functions.
- dead: This folder serves as a waste basket for unused audio. Menu>Session>Clean-up>Flush

Wastebasket: will delete audio files located in this folder. Click here for more
- export: This is the default location for exported files. Click here for more.
- externals: This folder contains a list of external files that are used in the session, but were not copied into the session
- interchange: This folder contains imported audio, recorded audio, and MIDI files.
- peaks: Contains waveform images for display in the editor. It is safe to delete the contents of this folder, and Mixbus will create new data when the session is re-opened.
- plug-ins: Contains plug-in settings. Some plugins (such as impulse-response based reverbs) need to store more information than is practical in the session XML file. The extra plugin data is stored here.

Session XML file format

Mixbus session files use the industry-standard “Ardour” session file format. These files are plain human-readable XML. Other workstations that use this format include Ardour and Waves Tracks Live.

A detailed description of the session format is beyond the scope of this manual, but here is an overview of a session file contents:

- Config : defines the Session Properties that were chosen for this session.
- Metadata : the metadata information for the session (personnel details, etc)
- Sources : a list of the audio and MIDI files that are used in this session
- Regions : a list of the regions (segments of an audio or MIDI file) used in the session
- Playlists : a list of playlists that are used in the session. A playlist is a sequence of regions, and their specific settings (trim, fade, etc)
- Locations : a list of locations (markers, cd markers, region markers, loop/punch markers) that are used in the session
- Bundles : a list of bundles (a bundle is a group of input (or output) ports that are associated, such as the 2 inputs of a stereo track)
- Routes : a list of “routes” (tracks and buses) that are used in the session. A Route has many sub-fields:
    - Currently selected playlist
    - Channelstrip fader, EQ, compressor, and pan settings
    - Plugins used in this route

Mixbus Preferences folder

The Mixbus “preferences” folder stores all of a user’s preferences, as shown in the Preferences dialog.

The preferences folder location varies with each OS, but it is always found inside the HOME (“~”) folder:

- Mac OS X: ~/Library/Preferences/Mixbus6
- Windows: ~/AppData/Local/Mixbus6
- Linux: ~/.config/mixbus6

Inside the Preferences folder, you will find a list of files and folders:

- config : this file stores your user preferences
- instant.xml : this stores transitory information such as the last playhead location and window locations
- bindings file : this file stores your customized keybindings
- recent : this file stores the list of recently-opened sessions that is presented to you when you launch Mixbus
- sfdb : this folder stores your metadata-info for any audio files that you've tagged in the Import dialog
- my-dark.colors : this file stores your personal Theme colors for the Dark theme. (Light theme is not yet implemented)
- plug-in cache (VST and/or AU) : this folder stores a list of plugins that have been identified on your computer. If you delete this folder or its contents, you will have to re-scan your plugins.

If you delete any of these files or folders, Mixbus will recreate a new file with default settings when it is re-launched.

More details about the "instant.xml" file

The instant.xml file is a "semi-persistent" preferences file. It serves this purpose:

When you *save* a session, many of your preferences for this session are stored in the session's local instant.xml, *and* the global instant.xml. When you create a NEW session, the global instant.xml file is used to create the new session's instant.xml

There are many parameters that are stored in the instant.xml file:

- Window positions (editor, mixer, meterbridge, plugins, etc)
- Action-Buttons assigned to the buttons in the toolbar
- Control Surface settings
- Clock mode selections
- Visibility of the Editor List and Mixer List panels
- Grid (snap) options
- Playhead position & zoom scale (*some instant.xml settings like this are ignored for a new session*)
- Selected mouse mode (tool) in the editor
- How the region-list is sorted (by date, name, or whatever)

These settings do not have any effect on the session's sound. They are not part of the session data.

A session can have multiple snapshot files, but it has only one instant.xml that is shared among all the snapshots.

What does this mean?

- When you create a NEW session, you'll be pleasantly surprised to see that some of your most-recent settings (window positions, action-button assignments) are already set-up for you in the new session.
- When you return to an OLD session, you'll find the session exactly as it was when you left it. Your more recent sessions & preferences will not be applied. The session's instant.xml will define the window positions, etc.
- NOTE: if you visit an old session and save it, you will overwrite your current global instant.xml, and those settings will be used in the next NEW session. This can be a nice feature, or a big annoyance.

This behavior is a compromise that was reached after many years of discussion & feedback. We are constantly re-visiting these issues to see which parameters should be stored globally (in preferences), in the *global* instant.xml, in the *session's* instant.xml, and the session snapshot file(s).

27. Appendix A: Common Support Questions

I purchased Mixbus, but it says it is “UNLICENSED” and I occasionally hear a burst of noise.

- Please check the Mixbus download email that arrived shortly after your purchase. The email should include a link to your Mixbus license file, and instructions for installing it.
- For security reasons, the license links will become inactive after 7 days. To reset the license links, please email Harrison support at: mixbus@harrisonconsoles.com Once you’ve downloaded the license file from the link, you may store it on a backup drive or USB key for later use. Your name is encrypted in the license file, so you should not share it with others.
- If you still have problems, please email Harrison support at: mixbus@harrisonconsoles.com

When I try to open the GVerb+ or 3D Delay, it says that the plugin is unlicensed. How do I get a license?

- If you’ve already purchased these plugins, please check the installation email that arrived shortly after your purchase. The email should include a link to your license file, and instructions for installing it.
- These plugins are part of the Plugin Essentials Pack which is available on our web store. The plugins are included in the Mixbus installation, but to utilize the on-screen controls you must purchase a separate license. The Essentials are available to all Mixbus users on all platforms (Windows, OS X, and Linux) and are automatically installed as part of Mixbus. This allows you to share Mixbus sessions with your friends and know that they will have the sound of the plugins available, even if they haven’t purchased them! Neat, huh?

How do I show Mixbus where to find my VST plugins? (Windows)

- In the main menu, click Window->Preferences->Plugins. Add the folder paths that point to your plugins and click “Scan”. More Here

How do I report a crash in Mixbus? (OS X)

- If you are using OS X and are still crashing at startup, the most likely cause is a plugin that Mixbus is not able to scan at startup. When Mixbus reports “Mixbus has closed unexpectedly”, please click “Generate Report”. This will open a window with the technical details of the crash report. Please copy/paste this information into an email to mixbus@harrisonconsoles.com for support.

How do I launch Mixbus from a command-line prompt? (for techies only)

- Windows: cd C:\Program Files\Mixbus4\bin Mixbus4.exe
- OS X: ARDOUR_BUNDLED=1 /Applications/Mixbus4.app/Contents/MacOS/Mixbus4
- Linux: cd /opt/Mixbus4/bin ./mixbus

Where does Mixbus store its preferences?

Mixbus creates a folder with several files

- On Windows, the folder path is something like ~\AppData\Local\Mixbus4\ (this varies with different versions of Windows).
- On Mac, the folder path is ~/Library/Preferences/Mixbus4
- On Linux, the folder path is ~/.ardour5

The Preferences folder contains these files, among others:
“config” stores the user’s preferences such as auto-backup and other items in the Options menu
“mixbus.bindings” stores the user’s selected keybindings (keyboard shortcuts) for menu actions
“ui.conf” stores your color, font, and other theme preferences
“announcements” stores the latest announcements from Harrison (usually this is empty unless a new version is available)
“instant.xml” stores the user’s last-used preferences such as snap mode, mouse mode, zoom mode, etc.
“recent” lists the sessions that appear in the recently-used dialog
“sfdb” stores the metadata database which the Import dialog stores metadata about files that you’ve “tagged” for later use.

If you delete the preference folder or any file inside it, Mixbus will create a replacement with default values when it is next launched. Sometimes this will cause a delay at startup; for example on OS X, Mixbus will have to re-scan the plugins when it is next launched.

Can I import Key bindings to another computer?

Yes, you have to manually copy the keybindings file from one computer to the other.

The file is found in your “preferences” folder:
~/Library/Preferences/Mixbus4 on Mac
~/AppData/Local/Mixbus4 on Windows

If your question is not answered here, feel free to email us at mixbus@harrisonconsoles.com for support.

28. Appendix B: Known Issues

- On systems with the minimum supported screen height (800 pixels for Mixbus, or 1200 pixels for Mixbus32C), you may not be able to fit the editor or mixer window on your screen. You can visit Preferences->Appearance and change the “mixer strip scale” to fit the mixer to your screen, but sizes less than 100% will be compromised.
- Some third-party plug-ins may cause crashes, strange noises, or other problems. We strive to support plug-ins as thoroughly as possible but due to the many variables (Computer platform, OS version, Mixbus version, plug-in version, licensing mechanisms, etc.) it is not possible to guarantee that all 3rd-party plug-ins will work as expected on your system.
- On Windows, anecdotal evidence shows that a few devices (including the Roland “Capture” series, Lexicon Lambda, Emu 0404, and Zoom R24) require that you enable the “Buffered I/O” setting. This tells Mixbus to use an alternative ASIO mode which works better on these devices.

For more details, contact Harrison via email (mixbus@harrisonconsoles.com).

29. Appendix C: Videos (Training and Tutorial)

The videos on the following pages are provided for your convenience and are a great resource for learning more about Mixbus and Mixbus32C. Some of these videos were produced using an earlier version of Mixbus, however the bulk of the information is still very relevant.

If you have read the manual, then some of these videos may be familiar, as some are placed throughout the manual, others are not.

These videos and more can also be found on our YouTube page. Don't forget to **subscribe** to our YouTube channel to be alerted to our **new videos coming soon**!

And don't forget our Mixbus user forum for more Mixbus information, or to just hang out and discuss how fun and easy it is to mix with Mixbus!

29.1. Quick Start Guides

For Mac Users

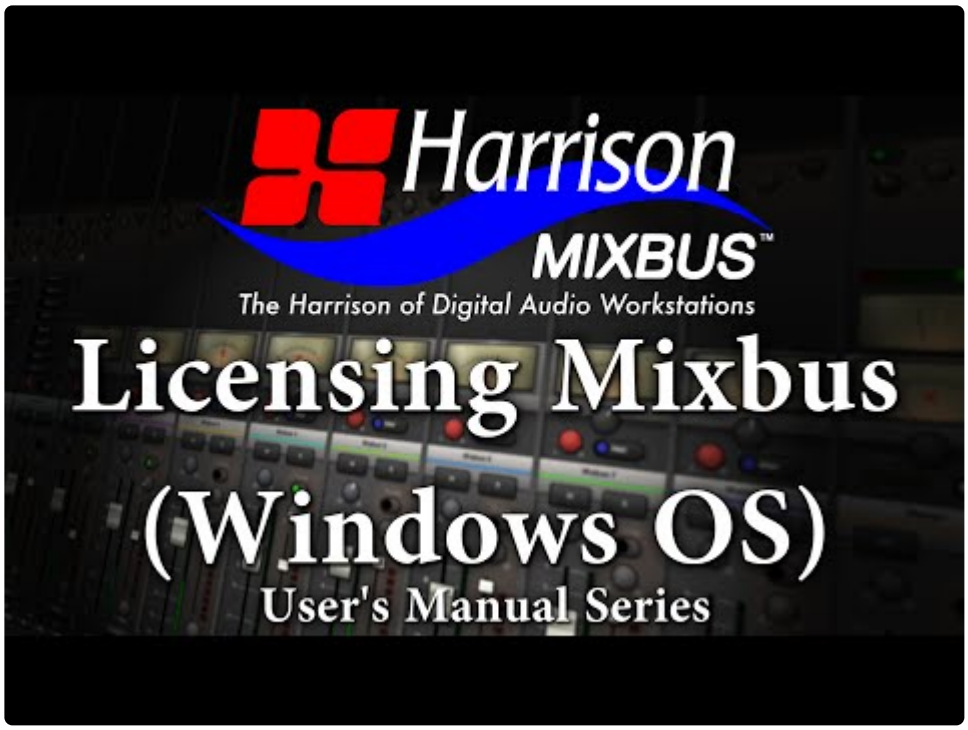
https://www.youtube.com/embed/vif6VaAncxc?rel=0

Licensing and the Home Folder for Mac

https://www.youtube.com/embed/83lsyy0ztKU?rel=0

Quick Start Guide for Mac

For Windows Users



https://www.youtube.com/embed/To5SOyd0Qw4?rel=0

Licensing and the Home Folder for Windows

https://www.youtube.com/embed/1ZU58nRzKI0?rel=0

Quick Start Guide for Windows

https://www.youtube.com/embed/nMPqBaQOW-o?rel=0

https://www.youtube.com/embed/nMPqBaQOW-o?rel=0

https://www.youtube.com/embed/jhr5BLH8AnM?rel=0

https://www.youtube.com/embed/jhr5BLH8AnM?rel=0

https://www.youtube.com/embed/jhr5BLH8AnM?rel=0

29.2. Mix Tips

https://www.youtube.com/embed/q1WoKW_TKYo?rel=0

Harrison’s Mix Tips Video Series. This video explains side-chaining in Mixbus 3.

https://www.youtube.com/embed/DjlBbao5gMQ?rel=0&start=135

https://www.youtube.com/embed/DjlBbao5gMQ?rel=0

https://www.youtube.com/embed/DjlBbao5gMQ?rel=0

https://www.youtube.com/embed/DjlBbao5gMQ?rel=0

Harrison's Mix Tips Video Series. This video explains gain staging in Mixbus 3

https://www.youtube.com/embed/cuInwhnrtlk?rel=0

https://www.youtube.com/embed/cuInwhnrtlk?rel=0

https://www.youtube.com/embed/cuInwhnrtlk?rel=0

This video demonstrates how to use grouping and bussing to enhance your session workflow within Mixbus 3.

https://www.youtube.com/embed/67TcjTd7LIs?rel=0

Harrison’s Mix Tips Video Series. This video demonstrates how to create a subgroup in Harrison Mixbus

https://www.youtube.com/embed/iESeTOSvX80?rel=0

https://www.youtube.com/embed/iESeTOSvX80?rel=0

Harrison’s Mix Tips Video Series. This video demonstrates how to set up parallel compression in Mixbus.

https://www.youtube.com/embed/ep5ietQjaq4?rel=0

Harrison’s Mix Tips Video Series. This video shows off some of the mouse shortcuts available in Mixbus.

https://www.youtube.com/embed/wn1SvzR2Nv4?rel=0

https://www.youtube.com/embed/wn1SvzR2Nv4?rel=0

Harrison’s Mix Tips Video Series. This video tells you everything you need to know about the EQ in Mixbus.

https://www.youtube.com/embed/Sa3-ibczoK4?rel=0

https://www.youtube.com/embed/Sa3-ibczoK4?rel=0

Harrison’s Mix Tips Video Series. This video details the compressors in Mixbus.

https://www.youtube.com/embed/cx_FqznRpbI?rel=0

Harrison’s Mix Tips Video Series. This video explains many different types of clipping, and how it applies to Mixbus.

https://www.youtube.com/embed/dPCUFIAsO8Q?rel=0

Harrison’s Mix Tips Video Series. This video shows off the meters in Mixbus.

https://www.youtube.com/embed/fs0m-wFXdZ8?rel=0

Harrison’s Mix Tips Video Series. This video demonstrates how to use reverb in Mixbus.

https://www.youtube.com/embed/ZbHwcsU7r2o?rel=0

Harrison’s “Shut Up and Mix” Video. Make sure annotations are turned on so you can see all of the mixing tips!

<< 29.1. Quick Start Guides | 29.3. Pro Tips >>

29.3. Pro Tips

https://www.youtube.com/embed/otZgZs3CDYM?rel=0

https://www.youtube.com/embed/otZgZs3CDYM?rel=0

Episode 1 of Harrison's Pro Tips video series for Mixbus. This video shows you how to quickly import tracks into Mixbus, and gives an explanation of the import dialog window.

https://www.youtube.com/embed/wddiGR-Y1bk?rel=0

https://www.youtube.com/embed/wddiGR-Y1bk?rel=0

Episode 2 of Harrison's Pro Tips video series for Mixbus. This video demonstrates all of the export options available in Mixbus.

https://www.youtube.com/embed/nJKSImri0vA?rel=0

Episode 4 of Harrison's Pro Tips video series. This video demonstrates how to route audio from Pro Tools into Mixbus.This is a power user function which Harrison does not provide technical support for.

https://www.youtube.com/embed/QxGhN8w5D1s?rel=0

A follow-up to Harrison's episode of Pro Tips on routing from Pro Tools into Mixbus. This video demonstrates how to export audio tracks from Pro Tools to import into Mixbus. This is a power user function which Harrison does not provide technical support for.

https://www.youtube.com/embed/aFWfLfrn_vU?rel=0

https://www.youtube.com/embed/aFWfLfrn_vU?rel=0

Episode 5 of Harrison’s Pro Tips video series. This video demonstrates the region editing and crossfade features of Mixbus.

<< 29.2. Mix Tips | 29.4. EQ Tips >>

29.4. EQ Tips

https://www.youtube.com/embed/wn1SvzR2Nv4?rel=0

https://www.youtube.com/embed/wn1SvzR2Nv4?rel=0

https://www.youtube.com/embed/wn1SvzR2Nv4?rel=0

Episode 4 of Harrison’s Mix Tips Video Series. This video tells you everything you need to know about the EQ in Mixbus.

https://www.youtube.com/embed/O9WnyWbMim4?rel=0

This Quick Tip demonstrates how to use the mid-band of the EQ to notch out a problem frequency on a track in Mixbus.

https://www.youtube.com/embed/Xb-prKFUdi4?rel=0

https://www.youtube.com/embed/Xb-prKFUdi4?rel=0

This Quick Tip demonstrates how to place your instruments in the sound field using the EQ in Mixbus.

https://www.youtube.com/embed/HM8XSfaGJuw?rel=0

https://www.youtube.com/embed/HM8XSfaGJuw?rel=0

This Quick Tip demonstrates how to create a “bump” for your bass guitar with the EQ in Mixbus. This can be done with a HPF and a Low Shelf.

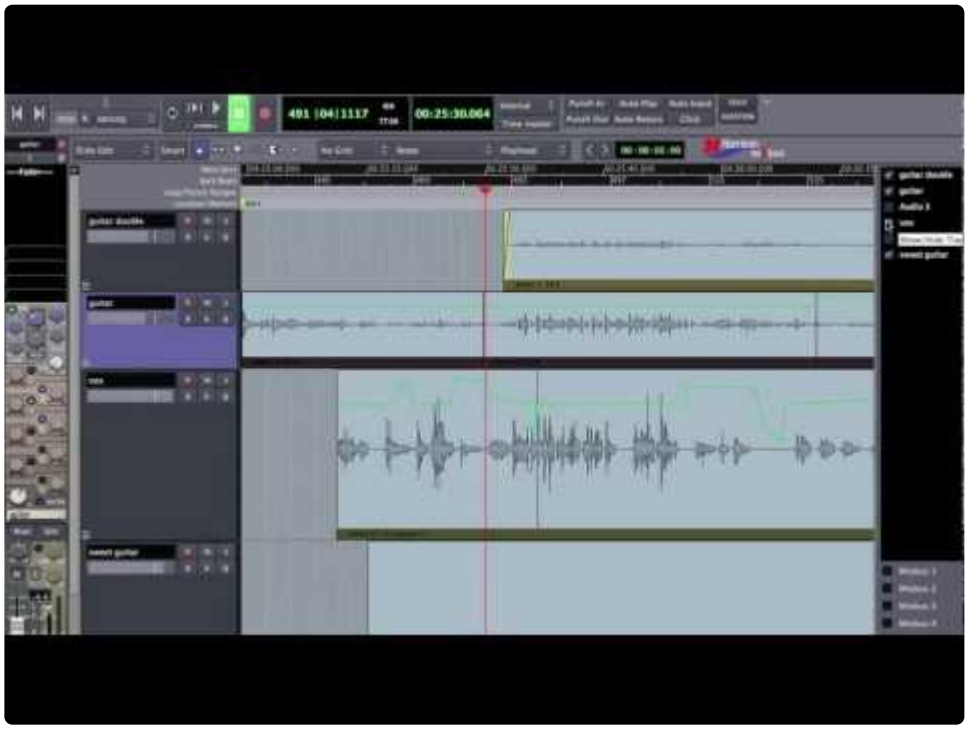
<< 29.3. Pro Tips | 29.5. Quick Tips >>

29.5. Quick Tips

https://www.youtube.com/embed/VOiyDFv8zt0?rel=0

https://www.youtube.com/embed/VOiyDFv8zt0?rel=0

This short video demonstrates how to loop record.



https://www.youtube.com/embed/bXLtAYwH7XM?rel=0

Demonstration of basic trim, fade, delete, copy, and paste operations including Smart Mode.

https://www.youtube.com/embed/7d33y5lVALg?rel=0

An example of sidechain compression used on a vocal track with a delay. Implemented using the built-in sidechain compression features of Harrison Mixbus.

https://www.youtube.com/embed/x7L6gc2lmHQ?rel=0

https://www.youtube.com/embed/x7L6gc2ImHQ?rel=0

https://www.youtube.com/embed/x7L6gc2ImHQ?rel=0

“Ripple” audio editing in the Mixbus workstation.

https://www.youtube.com/embed/XZU26CNDwmE?rel=0

A power users tip on using mutes and solos.

https://www.youtube.com/embed/O9WnyWbMim4?rel=0

https://www.youtube.com/embed/O9WnyWbMim4?rel=0

This Quick Tip demonstrates how to use the mid-band of the EQ to notch out a problem frequency on a track in Mixbus.

https://www.youtube.com/embed/Xb-prKFUdi4?rel=0

https://www.youtube.com/embed/Xb-prKFUdi4?rel=0

https://www.youtube.com/embed/Xb-prKFUdi4?rel=0

This Quick Tip demonstrates how to place your instruments in the sound field using the EQ in Mixbus.

https://www.youtube.com/embed/HM8XSfaGJuw?rel=0

https://www.youtube.com/embed/HM8XSfaGJuw?rel=0

https://www.youtube.com/embed/HM8XSfaGJuw?rel=0

This Quick Tip demonstrates how to create a “bump” for your bass guitar with the EQ in Mixbus. This can be done with a HPF and a Low Shelf.

<< 29.4. EQ Tips | 30. Appendix D: MIDI Primer >>

30. Appendix D: MIDI Primer

These pages will discuss basic MIDI messages, devices, and connections. It is intended to give enough information to get you started. Much of the information is presented in a generic way, not necessarily directly related to Harrison Mixbus. For specific MIDI for Mixbus click here.

What Is MIDI?

Musical Instrument Digital Interface, or MIDI, is a communication protocol that was published in 1983. The intent was to allow 2 instruments, from 2 different manufactures, to communicate with each other. One keyboard could be played while a second connected keyboard (with tone module) could sound the same note at the same time, doubling the performance.

It is important to know that the information being transmitted is **not audio!** It is “performance data” in the form of MIDI messages such as the note played, timing, velocity, etc.

Basic MIDI Devices

1. A MIDI **Controller** produces and transmits MIDI messages. Controllers can take on the functionality of keyboards or guitars, etc. They are designed to capture the performance of a musician. They do not produce sounds of their own.
2. A MIDI **Instrument**, or **tone module**, receives MIDI data and produces an audio output. It may be software or hardware based. A user can have one controller connected to one or more tone modules. Additionally, in the world of Digital Audio Workstations there are “plug-in” versions called virtual instruments (VI).
3. A unit that combines a controller and instrument in one package will be referred to as a **hybrid** for the purposes of this document. The most common form of this is often referred to as keyboard synthesizer. It can generate MIDI performance data and audio via an internal tone module. Sometimes the internal tone module can be switched off with a command known as Local Control. (If you seek out something called a “MIDI hybrid device” you may come up short).
4. A MIDI **Sequencer** is a device or software application that can record, edit, and play back MIDI events.

A single device can implement any combination of controller, instrument, and/or sequencer features. For example, before computer sequencers were popular, it was very common to buy a keyboard that provided the keyboard, internal tone module, and sequencing features in one device. These are sometimes called “arranger” keyboards.

You may find tone module, tone generator, and sound engine are used interchangeably.

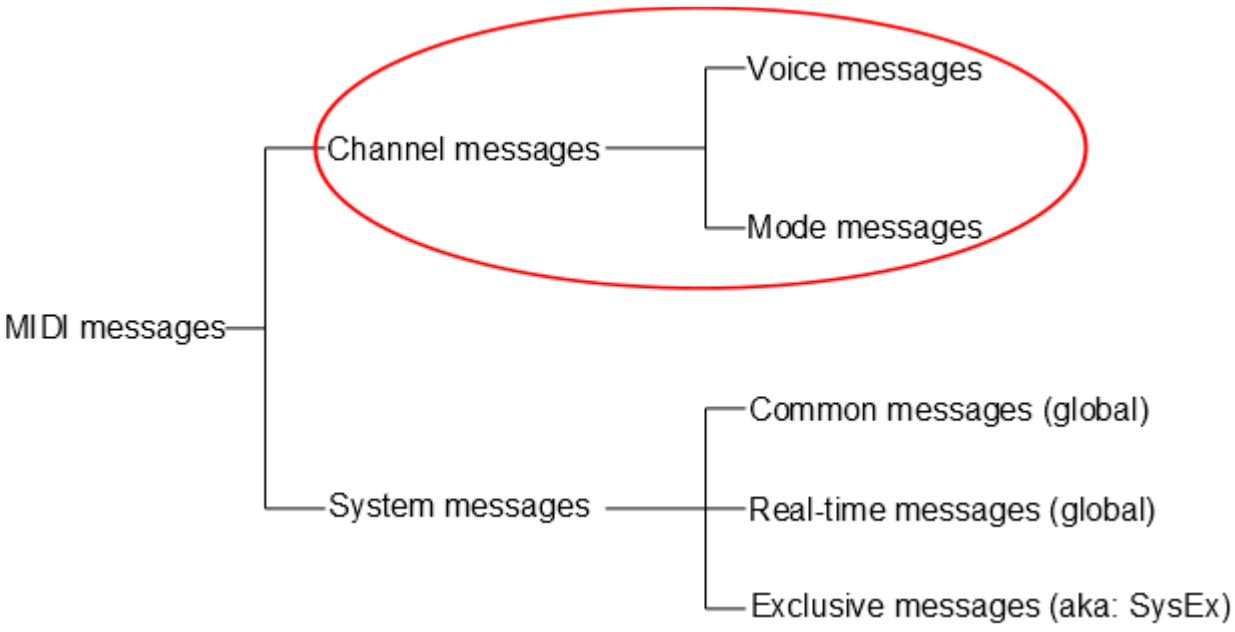
When a controller and tone module are combined together, there is often a function called **“Local Control”** which determines whether the keyboard will control the internal tone module directly. If you want to use a sequencer or DAW, you might use Local Control to

> disconnect the internal tone module from its keyboard. This way, when the controller sends performance data to the computer sequencer, and the sequencer returns the data back to the sound module, you don't hear the notes from the sequencer and internal controller at the same time.

MIDI Messages

The MIDI language is transmitted digitally. The information carried in the messages are note on, pitch, velocity, program change, note off, and more. There are 2 types of MIDI messages; Channel messages and System messages. These two sub groups can be further broken down and will be explained below. These MIDI messages are sent to other devices via a MIDI cable. More on the MIDI cable a little later.

Channels Messages: Each channel message contains a channel assignment and as a result are assigned to specific channels rather than sent globally. With the capability to transmit (Tx) and/or receive (Rx) on 16 channels, the user is offered some flexibility. Once a channel is establish the channel messages will deliver instructions such as Program change, Note on, Note off, and more. With this you can send performance data from one controller, on one specific channel, so that that one or more specific MIDI instruments on the same Rx channel responds.



Voice and Mode Messages

The "Voice" portion of the channel message is the musical performance part and contains information regarding the following:

- Note On: Identifies the note that will produce sound. The data sent is channel number, specific note number, and key on velocity.
- Note Off: Identifies the note that will stop producing sound. The data sent is channel number, specific note number, and key off velocity (notice that the note command has a discrete code for off and on.

More on this later).

- Polyphonic Key Pressure (Aftertouch): Some controllers have the ability to sense how much pressure was applied. The data sent is channel number, specific note number, and key pressure. If your controller does not have aftertouch, a default of medium pressure (64 of 128 possible) is assigned.
- Control Change: Specifies what function of the synthesizer is to be controlled. The data sent is channel number, controller number, and controller value.
- Program Change (sometimes called Program Select): Selects the type of sound that should be played. The data sent is channel number and program number. In the early days of MIDI, when a user wanted a custom sound he would need to patch cables. Today this cable patching process can be replaced with a stored file in memory, hence the name “patch”. You may also hear the terms voice or preset. These are essentially all the same thing. You can create your own patches or the manufacture can provide them for you. Because of the format of the MIDI message, only 128 patches would be recallable if it wasn’t for the Bank Message. Having multiple banks of patches, provides many more to choose from.
- Pitch Bend: Modifies, in a continuous manner, the pitch of the note being played. The data sent is channel number and pitch bend value.

The “Modes” portion of the channel message tells the MIDI instrument how to respond when it receives the performance part or the message.

Mode	Full Name	Common Name
1	Omni – On/Poly	Omni
2	Omni – On/Mono	no common name
3	Omni – Off/Poly	Poly
4	Omni – Off/Mono	Mono

Omni On/Off is used to determine the utilization of available channels. If Omni is ON then it will Tx or Rx data on all channels. If Omni is OFF then it will Tx or Rx on just one channel.

Mono/Poly tells the instrument whether to play 1 note at a time (Mono) or more than 1 note at a time (Poly). The Mono modes are the least commonly used modes (mode 2 and 4).

Polyphonic capability is merely the ability to play 2 or more notes simultaneously. So if you want to play a 4 note chord you will need 4 note Polyphonic capability.

Multi Mode

With Multitimbral we are talking about many tones to produce a complex sound like a piano part, a violin part, or a drum part. These are sometimes called “programs”,“patches”, or “presets”. You can have many parts playing at once, each assigned to its own channel. Before Multitimbral there was Monotimbral, where only one part could be played. Multitimbral can replace multiple notes with multiple programs so that we may produce the sound of an entire band with one MIDI performance track.

> Timbral, or timbre refers to tone quality. Timbre is how we determine the difference between the sound a bell makes and the sound a sine wave makes. When pitch and loudness are the same, much of the difference lies in the harmonics or timbre.

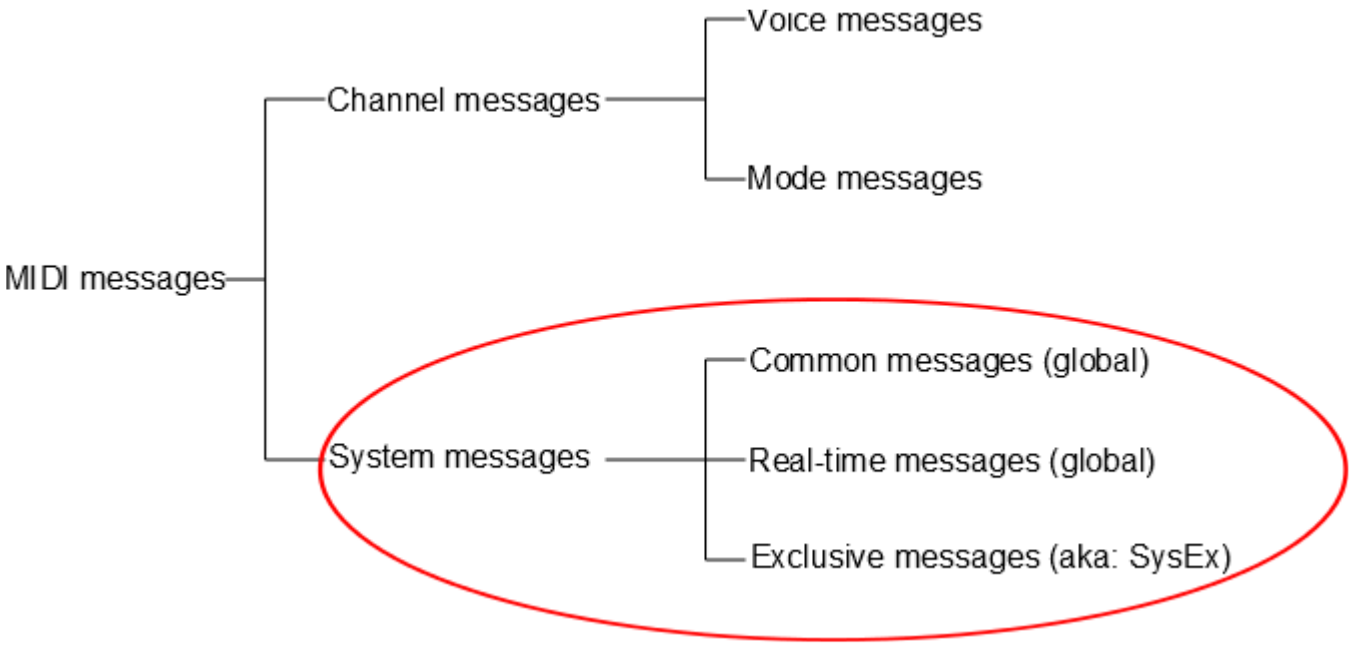
16 part Multitimbral playback capability is not uncommon today. Also be aware some manufactures may call Multi mode something different. For example, the classic Yamaha Motif refers to it as “song mode”.

There are other commands that can be invoked with a mode message, such as reset all controllers, local control, and all notes off.

Let’s address the “All Notes Off” message. As mentioned previously, the note on/off commands are discrete messages. So what happens if a note is turned on and for some reason the Rx equipment never gets the message to turn the note off? The “All Notes Off” message addresses this. It sends a Note Off command across all channels. In Mixbus there is a “MIDI Panic button” on the left side of transport section that sends the “All Notes Off” message. The button looks like an exclamation point. How appropriate!

above: the MIDI panic button circled in red

System Messages: These messages are mostly global and as such do not carry a channel assignment.



System Common Messages are for all MIDI devices in a network regardless of channel assignment. These messages are things such as MIDI Time Code (MTC), MIDI Clock, Song Select, and Song Position Pointer.

System Real-time Messages are used for synchronization with all devices in the network that are clock based such as sequencers and drum machines. These devices follow a tempo and need to be in sync. The resolution of the timing is expressed as Pulses Per Quarter Note (PPQN).

System Exclusive Messages are not global. They operate more as a point to point type of operation. It's kind of a catch all command that allows manufactures to use it for a whole host of things. For example a stored patch can be transmitted to and from equipment by the same manufacture. The manufacture can register their equipment with The MIDI Manufacturers Association so that they can obtain an exclusive SysEx ID so that only their equipment will respond to their commands. The code that one manufacturer uses to write a patch is going to be different from another manufacturer so sending a Yamaha patch to a Roland won't do much good. This is because each manufacturer has their own unique way handling information such as patches.

General MIDI (GM and GM1)

General MIDI is not actually a message but rather a standard that specifies certain functionality of MIDI equipment. Without General MIDI, each manufacturer can assign whatever patch they wanted, to whatever storage location they wanted. This would make it difficult to use different equipment from different manufactures. For example, one manufacturer may have a French Horn for Bank 1 program 4, and another manufacture may have that spot reserved for Viola. But General MIDI says that Bank 1 program 4 shall be a Honky-tonk Piano. With this, when a command is sent from one device to another the musician can hope for a sound closer to their expectation. General MIDI 1 elaborated on this and added even more specifications to the standard.

The MIDI Cable

The MIDI cable is unusual in that both ends have a 5 pin male connector.

Pin layout:

- Pin 1: Normally not used (proprietary systems may make use of)
- Pin 2: Terminated to a shield to eliminate noise
- Pin 3: Normally not used (proprietary systems may make use of)
- Pin 4 & 5: A twisted pair, and carries the 5 volt MIDI message

Sometimes this cable is called a MIDI DIN (Deutsches Institut fur Normung). The MIDI cable will carry MIDI signals in one direction only. If you need MIDI messages to return, then you will need a second cable.

The MIDI DIN cable characteristics are:

- Unidirectional transmission
- Serial transmission
- Maximum recommended length is 50 feet (to insure reliability)
- Transmission speed is 31,250 bits per second

USB, FireWire, and Ethernet have become popular alternatives when connecting to computers. However, if you need the full feature set that MIDI has to offer you will need MIDI cables.

Types of MIDI Ports

Often you will find 3 types of connections on MIDI equipment.

- MIDI out: Transmits the MIDI message
- MIDI in: Receives the MIDI message
- MIDI thru: Transmits a copy of the MIDI message found on the MIDI input (This facilitates daisy chaining and star configurations).

MIDI Connections

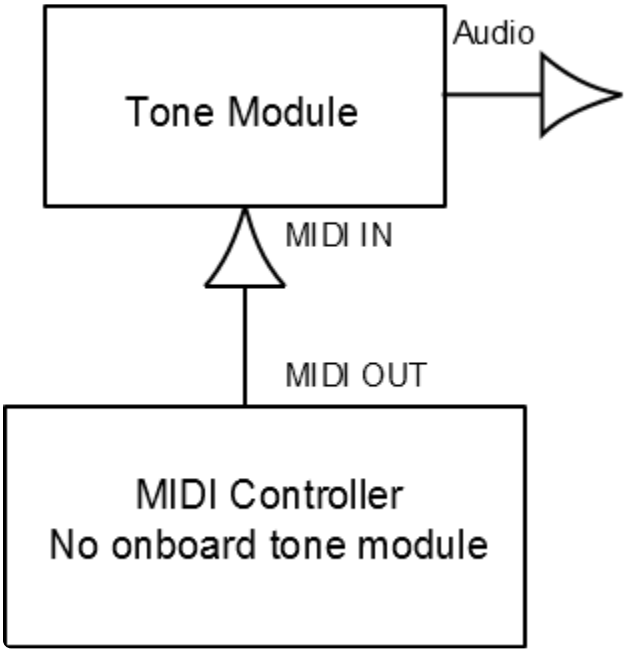


Fig. 1 above: A dedicated MIDI controller is connected to a tone module. This requires a MIDI cable. A USB or FireWire cable will not work.

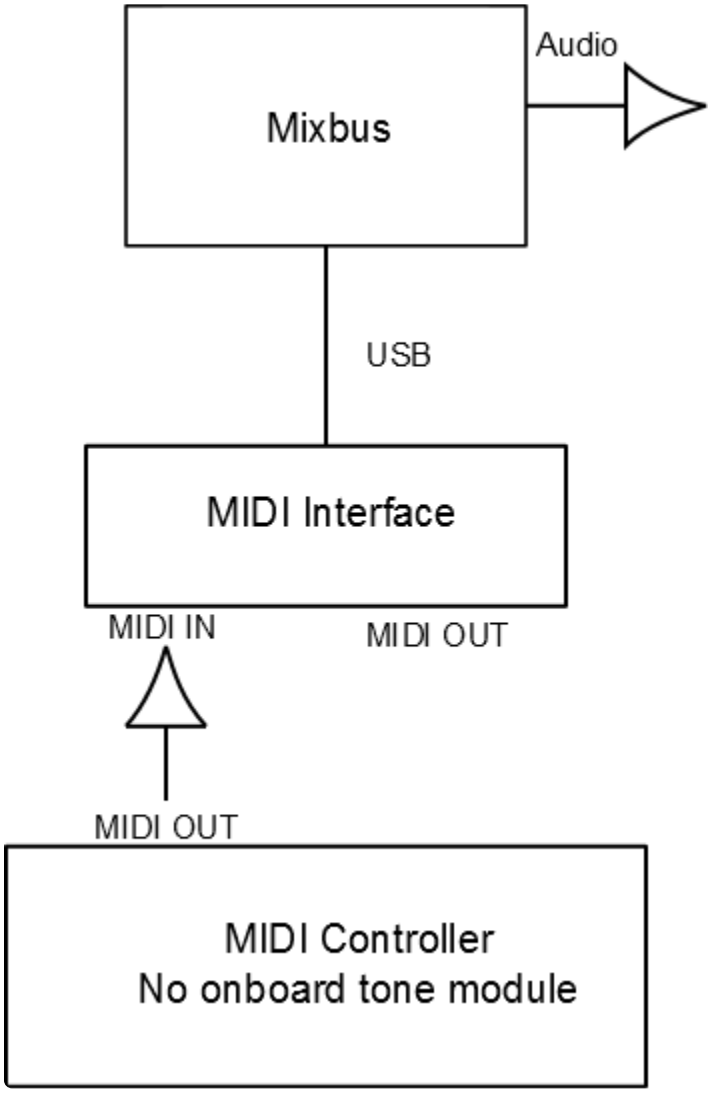


Fig. 2 above: A dedicated MIDI controller is connected to a DAW such as Mixbus via a MIDI interface. Running a Virtual Instrument plug-in within Mixbus replaces the dedicated tone module. The audio output from Mixbus can be routed to an internal DAC / sound card or to an external audio interface.

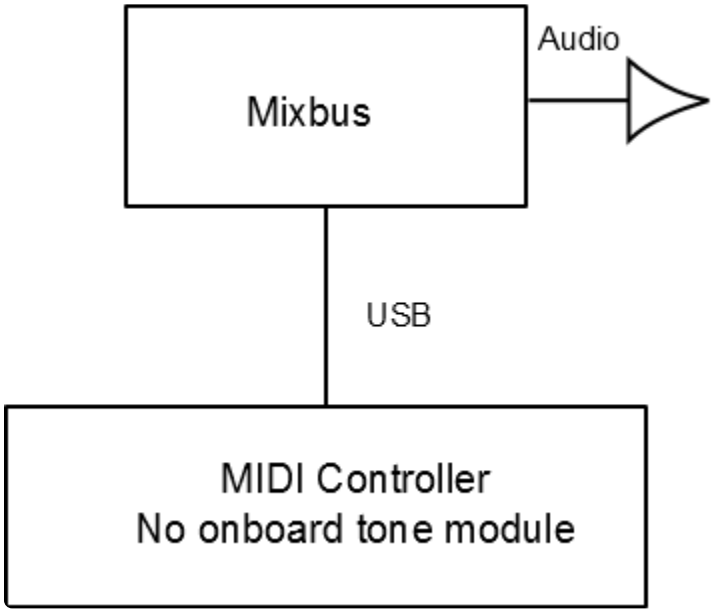


Fig. 3 above: A dedicated MIDI controller with USB MIDI out. In essence the MIDI interface is built-in to the the controller eliminating the need for an external MIDI interface.

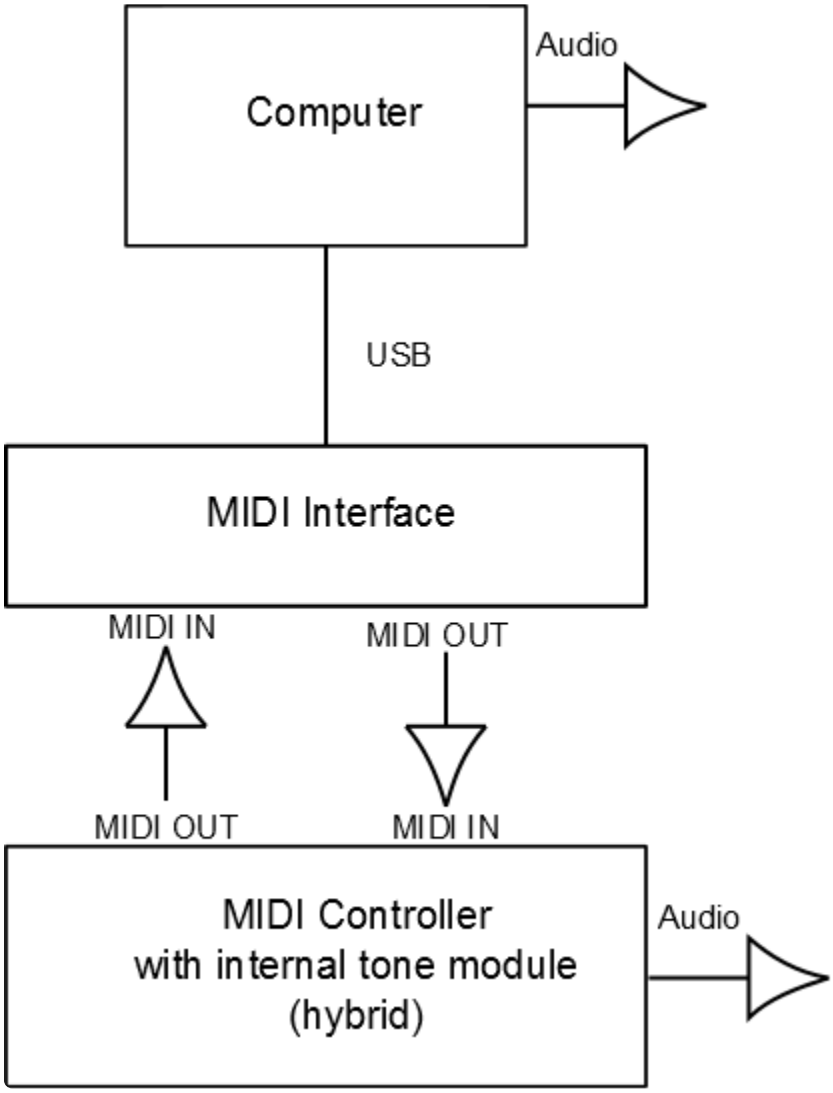


Fig. 4 above: In this example the MIDI controller has an on board tone module. Now MIDI messages can be sent from the computer to the MIDI controller so that audio can be produced with the on board tone module.

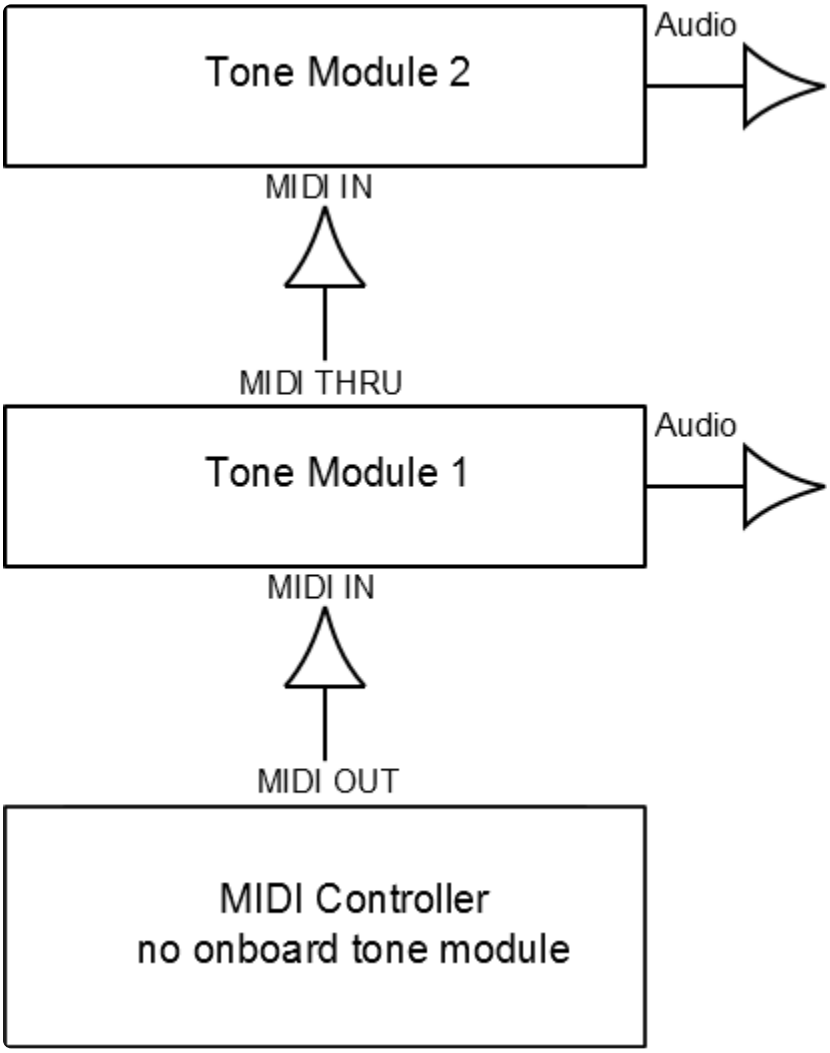


Fig. 5 above: This is an example of a **daisy chain configuration**. A dedicated controller sends MIDI messages to the first tone module which in-turn sends a copy of the signal to the second tone module. If the chain is carried out too far, the signal will degrade and become unreliable.

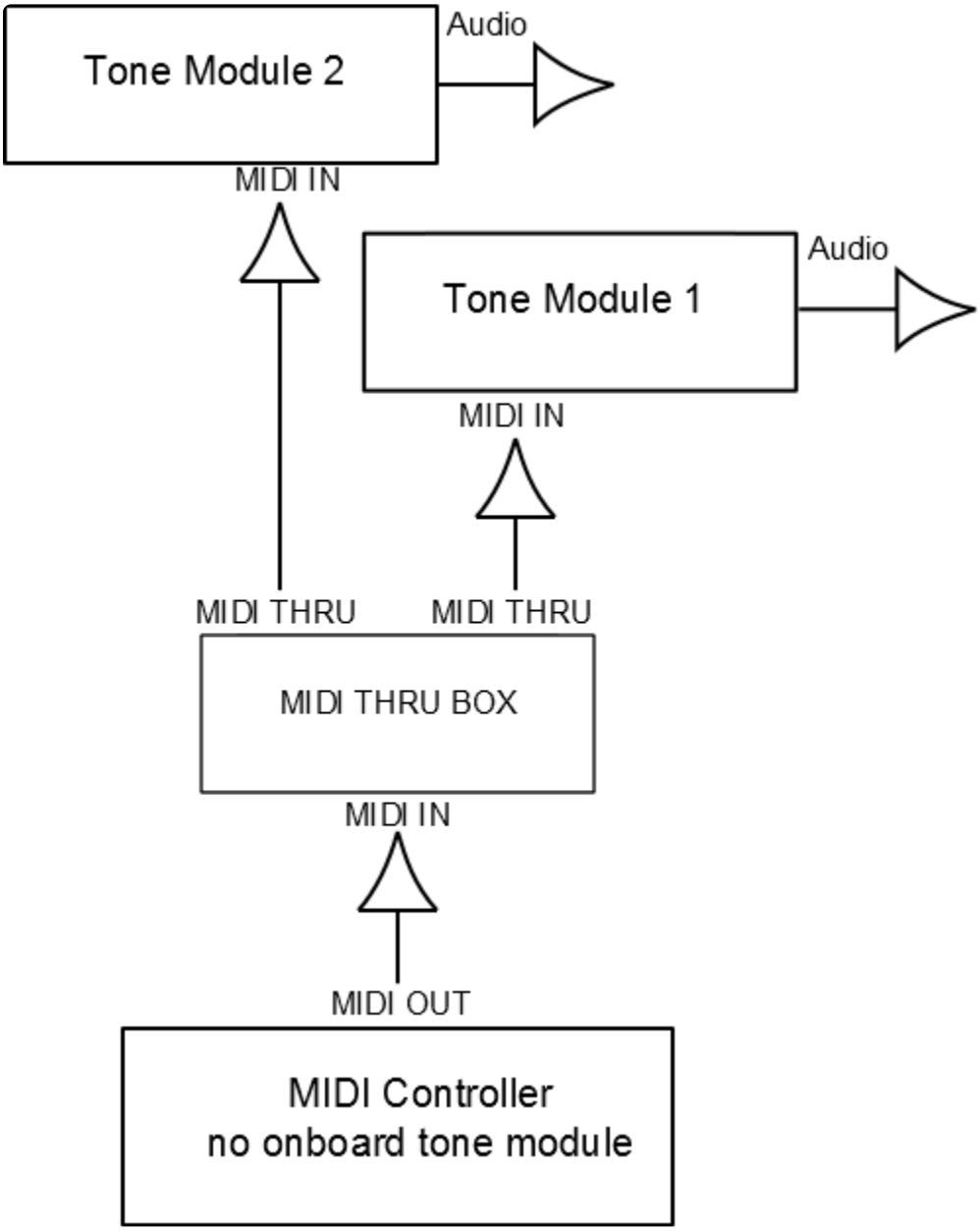


Fig. 6 above: This is an example of a **star configuration**. The Thru Box allows this configuration. Connecting many tone modules to a central point causes less chance of signal degradation. The advantages of this configuration increases with the number of devices on the network. For simplicity only 2 tone modules are shown. In practice many more may be connected causing the diagram to resemble a star pattern as the tone modules branch out around a central point.

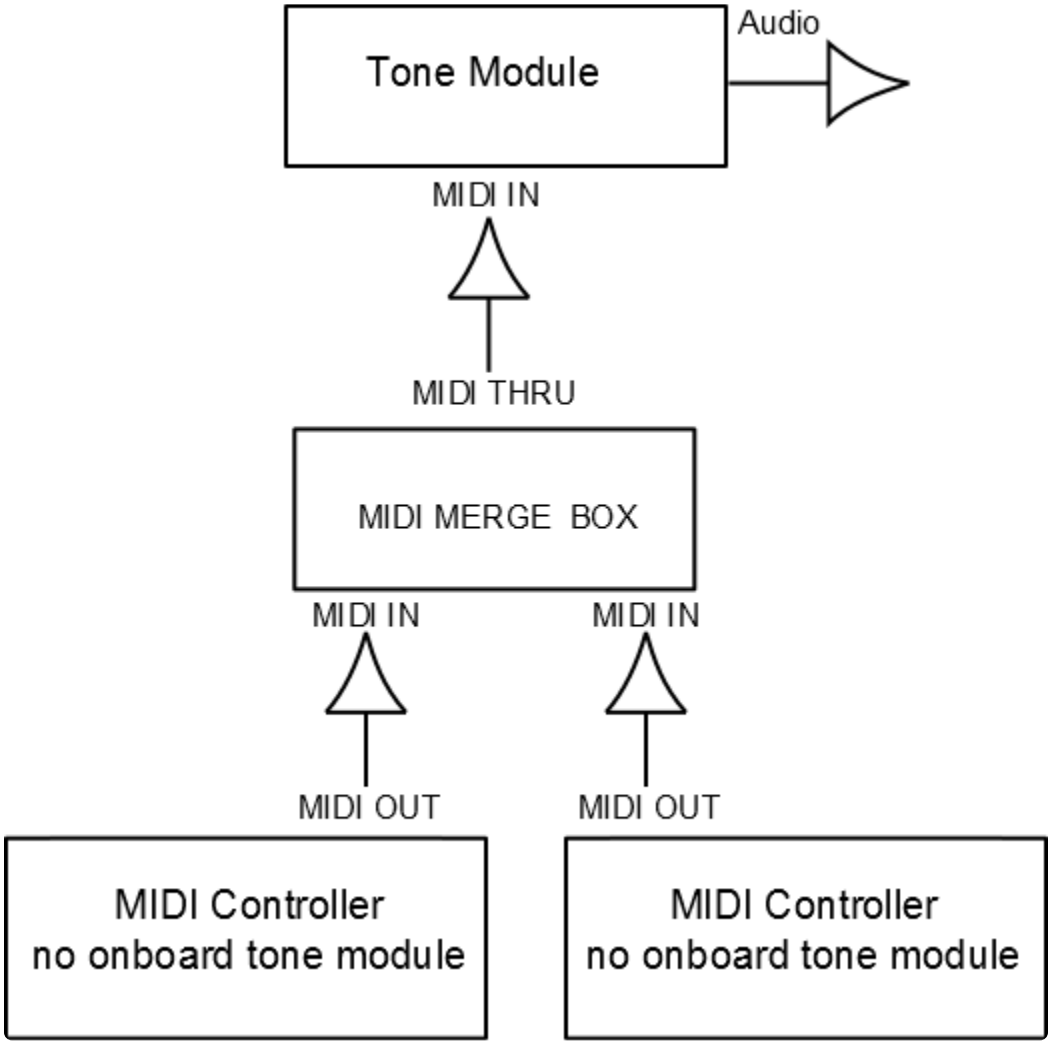
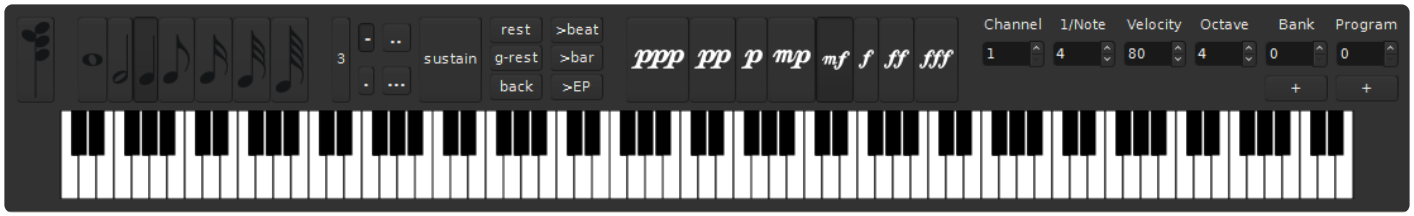


Fig. 7 above: A merge box allows 2 MIDI controllers to send MIDI messages to a single tone module. An example of this configuration could be that one of the controllers may be a keyboard and the other controller could be a string instrument both playing together into one tone module.

The Interface From MIDI To Mixbus

You will typically need an audio interface with MIDI I/O as in figure 2 or 4 above, or you may opt for a USB or FireWire connection as in figure 3. Of course you will also need need a MIDI controller. Although you can input data straight into Mixbus via the various tools provided in the software, it can be an inefficient method for recording an entire performance. These tools are best suited for making adjustments to an already captured performance or for step entry.

above: some of the tools offered in Mixbus
below: the on screen keyboard (right click the MIDI track rec button)



What MIDI Controller Should You Get?

Only you can decide what is best for you, but if you are new to MIDI a keyboard controller is a great choice for general purpose use and by far the most popular because of its flexibility. You will want to consider the number of octaves you would like the keyboard to cover. An alternative to a MIDI controller would be a synthesizer, one that generates MIDI messages and can also contains its own tone module. But because it can produce its own audio, it may cost more, take up more space, and be heavier. If you like the sounds a MIDI hybrid makes, and if it has line level outputs, you can record the audio straight into Mixbus via an audio interface as shown in figure 4 above.

Recording MIDI into Mixbus

To record a MIDI performance into Mixbus, first add a MIDI track. If you would like to hear your performance thru Mixbus then be sure a Virtual Instrument plug-in is in the redirect box. You can use Plug-in Manager to do this (by default a MIDI track will be created with the “Reasonable Synth” plug-in already in place).Then, in the MIDI Connection Manager, confirm that your MIDI input device is connected to the track you wish to record to. Record arm the track and click the master record button located in the transport section. Once the transport rolls you will be able to record your performance. MIDI messages will now be captured into Mixbus.

For more information click here

31. Contact Information

Visit our website to see the latest news about Mixbus and plug-ins for Mixbus:
http://mixbus.harrisonconsoles.com

For Mixbus support directly from Harrison, please email:
mixbus@harrisonconsoles.com

Visit our forum to chat with other users, and find common Mixbus questions & answers:
http://mixbus.harrisonconsoles.com/forum

Chat with other Mixbus users and developers using IRC:
http://webchat.freenode.net/?channels=ardour-mixbus

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