Mix Manual

LAMA Mix Guide

Software Manual, March 2023

DISCLAIMER OF WARRANTIES AND LIMITATION OF LIABILITIES

NOTE

For further information about Lean and Mean Audio BV, please use the contact information found on lamamix.com

SOFTWARE DESCRIPTION

LAMA Mix is a software audio broadcast console designed in close collaboration with A1's, to be a flexible and scalable alternative to dedicated hardware for critical broadcast production and live environments.

Benefits and Value

Values and benefits provided by the LAMA Platform are:

- It's a cost-effective mixer that can use regular computers as well as servers and cloud solutions to run as many audio processing as the CPUs can provide.
- It can be connected to any audio platform, as Mix is driver agnostic and thus can use any soundcard / audio over IP format.
- As it's a software solution it can easily be updated for future use cases.

Platform Requirements:

Windows 10/11, Windows Server 2016 or newer.

Minimal Recommended System:

AMD Ryzen 4600G, 5600G or better *
Intel 11600K or 12600k or better *
16 GB Ram

Workstation Recommended System:

AMD Ryzen 3900X, 5900X or better *
Intel 11900K, 12900K or better *
16 GB Ram

Tested audio solutions:

RME (several models, like the HDSPe MADI FX, HDSPe AES)
Focusrite (several models)
Dante Virtual Soundcard
Sonifex AVN-DIO10 Dante to 3G/HD/SD-SDI Embedder/De-Embedder
Merging (Using Ravenna ASIO, including 2202-7 mode)

*Keep in mind that the CPU needed is highly dependent on the use-case. As a general rule of thumb you can expect to run around 25-35 AutoMix instances on the minimal specs. On the Workstation spec you can run a number closer to 50-60 instances. Also keep in mind that you need an audio interface that supports the number of channels needed.

CONTENTS

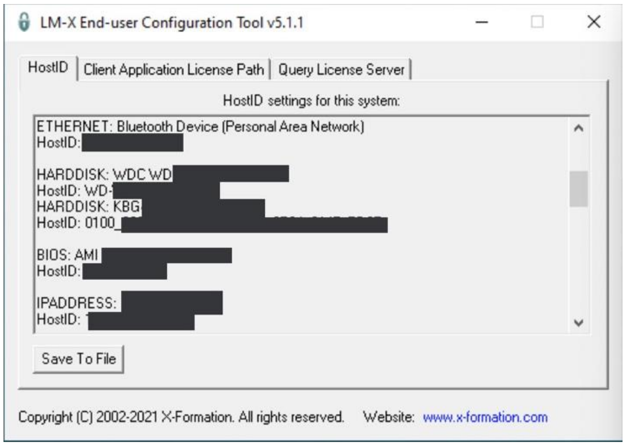
SOFTWARE INSTALLATION INFORMATION

Installation of LAMA Platform

The LAMA Platform is delivered as a stand alone .exe file. Place the .exe file in any folder.

Installation of LAMA License Center

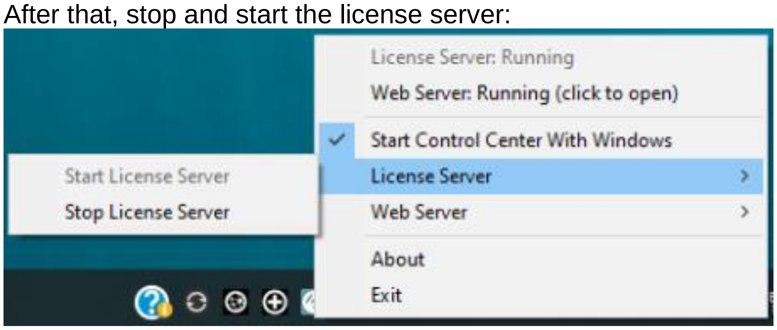
The LAMA license center comes as an installer. You install it on any computer on the network. The LAMA Platform will look for an instance of the license server on all possible computers it can access in the network.
Once installed you need to load the lmxconfigtool.exe that can be found in the installation folder.



Please send one of the HostIDs of one of the hard-disk to license@lamamix.com. We will send you a license file that contains all the bought licenses.

After receiving the license file, place it in the following folder:

After that, stop and start the license server:

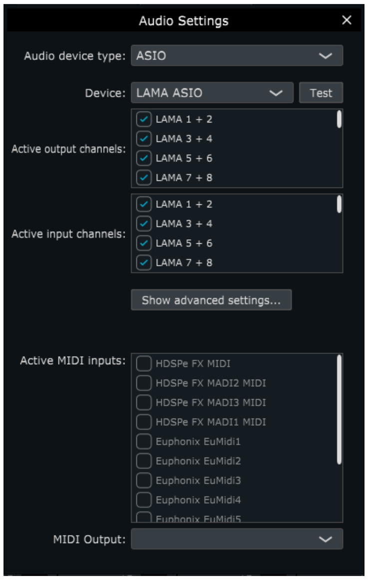


If you experience difficulties installing the application

If you experience difficulties loading or accessing the application after a standard installation on Windows 10, Windows Server 2016 or Windows Server 2019, please consult your IT department personnel to have proper access permissions setup for your use. If the problem cannot be resolved, please contact LAMA using the contact information found on lamamix.com

1
INSTRUCTIONS FOR SETTING UP THE MIXER

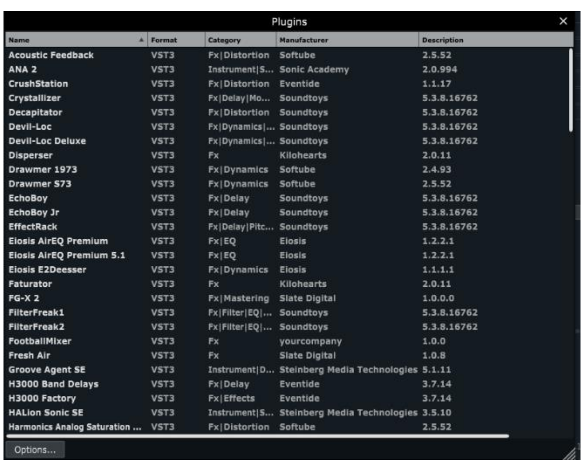
The Settings menu – Audio Settings



The Audio Settings screen is where you can set up and configure the audio and MIDI devices that Mix uses for input and output. The Audio Settings screen is divided into two sections: Device Type and MIDI Devices.
In the Device Type section, you can select the audio interface that Mix uses for processing audio signals. Mix supports any audio device that your computer can use, including built-in sound cards, USB audio interfaces, and Thunderbolt audio interfaces. We recommend selecting ASIO devices, as this standard has proven to be a stable and efficient standard for real-time audio processing.
In the MIDI Devices section, you can select the MIDI devices that Mix uses for input and output. MIDI (Musical Instrument Digital Interface) is a protocol that allows electronic musical instruments, controllers, and software to communicate with each other.
By selecting the MIDI devices that you want to use, you can control Mix using external MIDI controllers.

By customizing the settings in the Audio Settings screen, you can ensure that Mix is configured to work optimally with your hardware and software, and that you can use Mix to its full potential for your mixing and audio processing needs.

The Settings menu – Plugins Manager



The Plugins Manager allows you to manage and use VST plugins in your project. With the Plugins Manager, you can scan your system for available plugins, add them to your library, and use them on your channel strips and in the connectors screen.

To scan for plugins, simply click on the "Scan for new or updated VST3 plugins" button in the Plugins Options window. The system will search for VST plugins in the designated folders, and display a list of available plugins that you can add to your library.

Once you have added a plugin to your library, you can use it on your channel strips and in the connectors screen by selecting it from the “add plugin” menus..

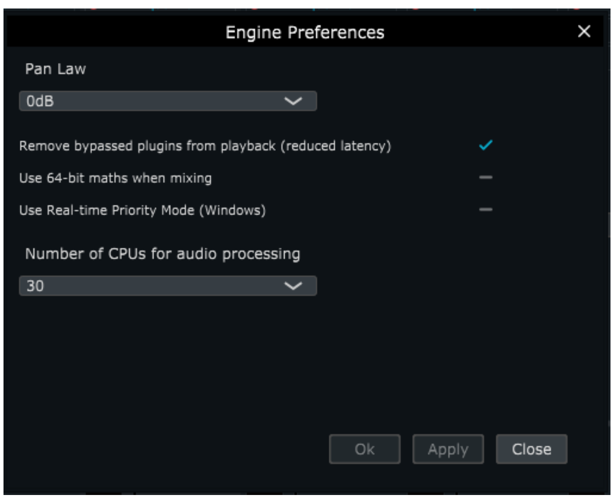
Using VST plugins in your project can enhance the functionality and creativity of your mixing and audio processing, and allow you to access a wide range of third-party effects.

The Settings menu – Midi Control

The MIDI Control tab allows you to configure the mixer to work with external MIDI controllers, such as Mackie Control devices. By setting up the MIDI control options, you can expand the capabilities of your mixer and enhance your workflow with tactile control over various parameters and functions.

In this tab, you can specify the MIDI input and output ports that your controller uses.
For example, if you have a Mackie Control Universal Pro controller, you can set the MIDI input and output ports to match its configuration, and then map the faders, knobs, and buttons to different mixer channels, sections, and functions. This allows you to adjust the levels, pan, EQ, and other parameters of your audio signals in real time, and to switch between different modes and views of the mixer with a single button press.
By using the MIDI Control tab, you can integrate your external controllers seamlessly with your mixer, and create a hybrid setup that combines the benefits of digital mixing with the tactile feedback and precision of physical controls.

The Settings menu – Engine Preferences



The Engine Preferences screen is where you can set all preferences related to the core engine of your project. This includes settings for audio processing, performance, and resource allocation, among others.

In this screen, you can customize various options, such as:

Pan law: This setting determines how panning affects the perceived loudness of audio signals. You can choose from different pan laws to adjust the balance between the left and right channels, and to match different mixing scenarios and standards.

Bypassed plugins: You can choose whether to remove bypassed plugins from the signal chain, to reduce the processing load and streamline the audio path.

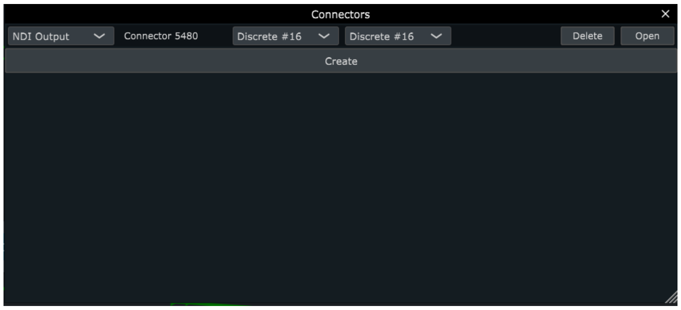
64-bit math: Enabling this option allows the engine to use double-precision floating-point calculations for mixing, which can improve the accuracy and clarity of audio processing.

Realtime priority: By setting the engine's priority to real-time, you can ensure that it has the highest priority for CPU and memory resources, which can reduce audio glitches, dropouts, and other performance issues.

CPU cores: You can adjust the number of CPU cores that the engine will use, to balance the processing load and optimize the performance of your system.

By customizing these preferences, you can fine-tune the behavior and performance of your project, and optimize it for your specific hardware and workflow requirements.

The File menu - Connector plugins

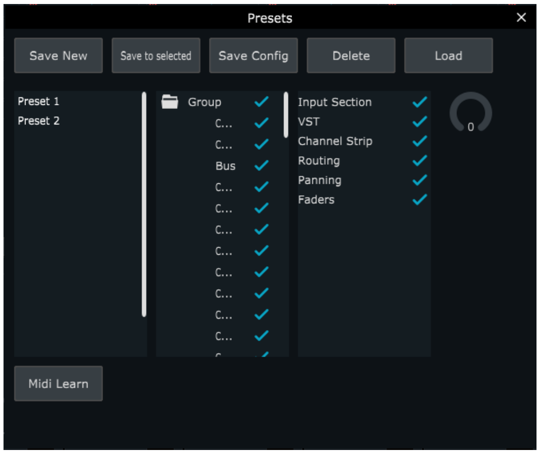


Connector plugins function as audio devices that allow you to add inputs and outputs to the routing lists on the channel view. When you load a connector plugin, it appears as a virtual soundcard output or input that you can use to route audio to and from your project. For example, if you load the official NDI plugin, you can send up to 16 channels of audio to it, as if it were a physical soundcard output.

To use a connector plugin, simply load the plugin view from the file menu and configure its settings as needed. Once the plugin is active, you can route audio signals to it from your channels and sections, and then use the plugin's output to send the audio to another destination, such as a remote client, a live streaming service, or a recording device.

By using connector plugins, you can easily expand the capabilities of your project and customize the routing of your audio signals. Whether you need to send audio to a specific device or application, or you want to create a complex routing scheme, connector plugins provide a flexible and powerful solution that can adapt to your needs.

The File menu – Presets

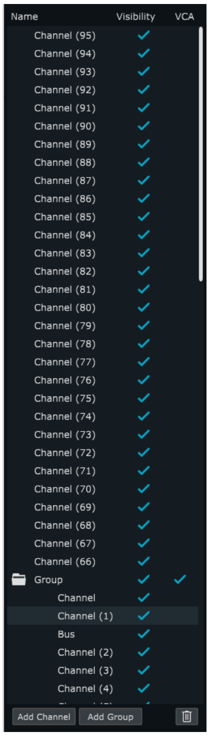


The Preset Menu allows you to navigate through different scenes or configurations of your project. You can prepare each scene by adjusting the settings for your channels and sections, and then easily switch between them using MIDI triggers that you can assign using MIDI Learn. To save a new scene, simply press the "Save to New" button, or to overwrite an existing scene, press the "Save to Selected" button.

The Save Config button is designed to save the current configuration of an existing preset. You may want to use this function when you have made changes to the fade speed, which determines the time it takes to transition to the new preset, or when you have deselected certain channels or sections that you don't want to load. By saving the updated configuration, you can easily recall the preset with your preferred settings in the future.

You can deselect specific channels and sections from loading when you recall a preset. This feature can be useful when you don't need to use all the channels and sections of a preset, or when you want to keep settings for specific channels or sections. For example, you might want to recall a preset that uses only the field microphones, while keeping the commentary mic unaffected. To deselect channels or sections, simply uncheck the corresponding boxes in the preset configuration, and then save the updated preset using the Save Config button.

The Tree section



The Tree section is where you can navigate between the channels and groups in your project. Every channel that you add to your project will be displayed in the Tree section, allowing you to quickly and easily access and manipulate your audio signals.
You can customize the display of the channels in the Tree section by changing the order, visibility, and group settings. This allows you to organize your channels in a way that makes sense for your workflow and makes it easy to access the channels you need when you need them.
In the case of a group, you can also choose whether to display a VCA fader for that group. A VCA fader is a type of control that allows you to control the volume of a group of channels with a single fader, making it easy to adjust the overall level of multiple channels at once.

In the lower part of the Tree section, you'll find three buttons:

Add Channel: This button allows you to add a new channel to your project. You can choose from a variety of channel types: Input Channels, Output Busses, and auxiliary channels.
Add Group: This button allows you to add a new group to your project. Groups are a powerful tool that allows you to organize and control multiple channels at once.
Delete Audio Process: This button allows you to delete an audio channel or group.

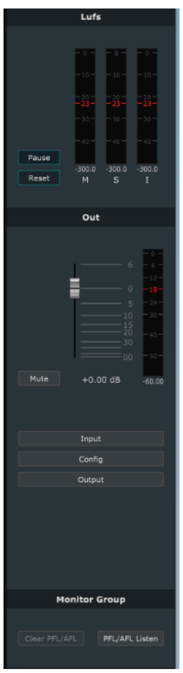
By using the Tree section and the buttons in the lower part of the section, you can quickly and easily navigate and customize your channels and groups, and create a workflow that works for you.

The Channel section

The Channel section is the heart of the mixer, where you'll find all the tools you need to process and mix your audio signals. It's divided into four main sections:

Faders: This section displays the faders for each channel in your project. You can use the fader section control the levels and pan positioning of your audio signals over time.
Routing: This section allows you to route your audio signals to various destinations, such as auxiliary buses or connector plugins. You can also use this section to create submixes and to route channels to busses for more efficient processing.
Channel strip: This section allows you to use native DSP processors in your processing chain, allowing you to add effects and other processing to your audio signals.
VST: This section allows you to insert VST plugins into your processing chain, allowing you to add effects and other processing to your audio signals.
Input: This section displays the input settings for each channel in your project. You can use this section to adjust the input gain and to apply processing to your input signals before they're processed by the rest of the mixer.

The Monitor section



The Monitor section is located on the right side of the mixer UI, and provides a range of tools for monitoring and controlling your audio signals. Here's a breakdown of the different elements you'll find in this section:
LUFS meter: This meter displays the loudness of the currently selected audio bus, using the LUFS measurement system. You can pause or reset the meter using the Pause and Reset buttons, respectively.
Monitor volume: This control allows you to adjust the overall volume of the monitor section, with a mute button for quick muting.
Input selector: This button allows you to select the audio bus you want to monitor. You can choose from a range of different buses, depending on your routing and processing setup.
Config button: This button opens a menu where you can configure the channels you want to listen to in the monitor section. This is useful if you only want to monitor specific channels on your bus.
Output selector: This button allows you to select the output of the monitor section itself.
Clear PFL/AFL button: This button allows you to clear all PFLs (pre-fader listens) in one go. This can be useful if you have multiple channels with PFLs enabled and you want to quickly reset them.
PFL Listen button: This button toggles the PFL (pre-fader listen) function on and off for the monitor section. When PFL is enabled, you can listen to individual channels in isolation, without the rest of the mix playing. This is useful for fine-tuning individual channel settings or for troubleshooting audio issues.

The Information bar

The Information bar contains 3 parts: the first sections shows the name of the project.
The CPU Load meter shows how long it takes to calculate the next audio buffer, compared to time the audio device driver gives the software to calculate the next buffer.
The last part contains the build information

2
TROUBLESHOOTING

How to handle clicks in the audio

Clicks (buffer underruns) are the enemy when working with live digital audio. There can be several causes of these buffer underruns. We will go over them and look at ways to prevent them.

Windows Schedular

Especially with older CPUs (we have not seen this on the recommended CPUs yet), there can be problems with the Windows Schedular that cause clicks: When you have as many worker threads selected in the settings screen (or more) than the amount of logical CPU cores in the system, windows is running other processes on cores that also need to run real-time audio. Because of context switching, these cores need time to switch between the real-time audio processes and things like storage indexing, virus scanners, memory swapping etc.

A solution for this is to set the worker threads 2-4 threads lower than the number of logical cores. So, if the system has 12 logical cores, try 8-10 worker threads.

Audio Device Buffer

The audio device buffer determines the time between each moment the audio device presents and needs a new buffer of audio information. The lower this number, the lower the latency the audio device produces. But like the windows schedular problem, the lower this number, the higher the probability that there is an event on the computer that locks the CPU for a time that is long enough to cause buffer underruns. As a general rule of thumb 256 or 512 samples buffer should be good options to try.

Sometimes it’s not possible to use these settings, for instance when using certain Ravenna (AES67) drivers with 2202-07. If the driver can only be set to a value much lower than the recommended value (and this causes clicks) you can try the multiplier settings in the settings menu. When the multiplier is set to a setting higher than 1, we enable an internal buffer that takes multiple audio buffers, then processes them in 1 go after which it gives the buffer back to the audio device in small parts. To a certain extent, this emulates a higher audio buffer inside the LAMA Platform.