Open Labs



SoundSlate™

Production station (Gen5)

Operation Manual

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Using the SoundSlate Safely

Before using the SoundSlate, it is extremely important that you thoroughly read the following instructions. This will provide you with the information needed for safe operation of your SoundSlate.
Please continue reading the rest of this manual for specific information on the use of this unit.



Warning

To avoid permanent health damage, always adhere to these important safety advies:

- The SoundSlate, in combination with headphones or speakers, is capable of producing sound levels that can cause permanent hearing loss. Do not submit your ears to long periods of high volumes. If you experience any hearing loss or ringing in the ears, you should consult an audiologist.
- Do not open the SoundSlate or attempt to disassemble or modify the internal parts without prior guidance and/or instructions from an Open Labs Technician. If the SoundSlate does not seem to function properly, contact Open Labs technical support.



Caution

To ensure that the SoundSlate or any device connected to it is not damaged during usage or maintenance, it is important to adhere to the following instructions.

- Do not expose the SoundSlate to inclement weather conditions such as heavy rain, or use the unit in damp or wet conditions. It is unwise to place open containers of liquids near the unit, these could spill onto the unit. Do not allow any objects or liquids to penetrate the unit.
- Never use or store the SoundSlate in conditions that are subject to extreme temperatures. This mostly includes closed in environments where lots of heat and direct sunlight are present (example, enclosed vehicle).
- Do not place the SoundSlate in unsuitable locations where the chance of falling could happen.
- Before connecting the SoundSlate to other electronic components, turn off the power for all components. Before turning the power on or off for all components, set all volume levels to minimum. Gradually raise the volume control while playing the SoundSlate to set the desired listening level.

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P r e f a c e

This Product Manual gives information about the Open Labs SoundSlate instrument.

Manual Organization

The chapters in this User Manual are arranged as follows:

- Introduction: information and overviews of SoundSlate.
- Getting Started: information on preparing your SoundSlate for use.
- First time: instructions on what to expect from the software and hardware environment of your SoundSlate.
- Using SoundSlate as a VST Player using Open Labs Riff.
- Using SoundSlate as a Hardware DAW for recording.
- MimiK Manual: instructions on how to mimik keyboards and MIDI sound modules. ***Mimik is only pre installed on European models of SoundSlate.*** Mimik can be purchased and downloaded by calling Open Labs Sales.

Useful Suggestions

This manual has a fair amount of information, however, from time to time you may not find what you need to know in this manual. Every application that comes with a SoundSlate has a manual. These are usually in either a .PDF format or HELP file format, and can be located in their "c:\Program Files\Product Name" folder.

Also, Google or Wikipedia are both great resources for finding more detailed information on the application you require assistance with.

1

Chapter 1 – Introduction

Welcome to the Open Labs SoundSlate manual. Please read this manual thoroughly in order to familiarize yourself with the unit. Please keep this manual in a safe place for further reference.

SoundSlate redefines music production and performance by incorporating Open Lab's music operating environment, including the critically acclaimed virtual instrument host RIFF™, thousands of sounds and effects, a robust audio i/o, and a powerful computer core running WIndows XP into a 1U-compact rack case that packs a punch.

User Friendly: Open Labs SoundSlate enables you to change settings and access programs through an easy-to-use interface that shields you from the complexities of the operating system.

Power to Rock: Open Labs SoundSlate can accommodate the fastest processors available, more RAM, and more storage than any other music workstation.

Low Latency: SoundSlate gives you near-zero latency even under high processor loads.

Versatility: SoundSlate is so versatile, that they can run virtually any plug-in or application designed for the Windows XP operating system, including products from Steinberg, Digidesign, Native Instruments, IK Multimedia and many others!

Surfboard: An ethernet port allows you to directly access the Internet, and to quickly and easily download upgrades, sounds, and applications.

Extreme care should be taken when installing new hardware and software. Installing the wrong components could cause great instability to your SoundSlate. *Please contact support@openlabs.com before hardware changes.*

SoundSlate Components

DVD/CDRW Drive
SoundSlate comes standard with a DVD/CD-writer, so that you may use it as a backup device, or to create audio CDs of your music. The DVD/CDRW drive can also be used to play CDs, DVDs, or install software.

Hard Drive
The hard drive is the main physical storage device for all your data and applications. Because hard drives are delicate devices, always treat them with care.

Power Button/Reset Button
The power switch is located on the front left side area of the unit.

Audio Interface
The Open Labs SoundSlate includes a mutichannel audio interface. Please refer to Chapter 2 for information about the specific audio input/output features of your Open Labs SoundSlate.

USB Ports (8)
Two in the front, with six additional rear USB ports allow connections to USB devices.

External Monitor Port (2)
The monitor ports allows you to connect any monitor with a DVI connector to your SoundSlate.

Ethernet 10/1000
The Ethernet port will accept a single high-speed cable connection. This will allow you to share files, print, and access the Internet.

Touchscreen Monitor(Optional)
The Open Labs SoundSlate comes with an 8-inch touchscreen monitor. The touchscreen can be accessed using your fingertips, or a plastic stylus (not included). The touchscreen provides an alternative method of inputting commands. It can be used by itself as the sole source of command input, or it can be used in conjunction with the built-in keyboard and track ball, or a standard keyboard and mouse.

About the Touchscreen Monitor

If you chose to purchase SoundSlate with a touchscreen monitor, the following techniques will help you get the most out of the touchscreen:

- Rather than using the flat portion of your finger, use your fingernail. By using your nail, you make contact with a smaller portion of your finger and get greater accuracy.
- To have a "right-click" effect with your finger, tap and hold down on the screen with your finger. In a moment, the right-click options will pop up for you.

2

Chapter 2 - Getting Started

Unpacking and Setting up your SoundSlate

Select a clear area with stable power source, preferably one that is surge-protected, or an uninterrupted power supply (UPS) to unpack your SoundSlate.

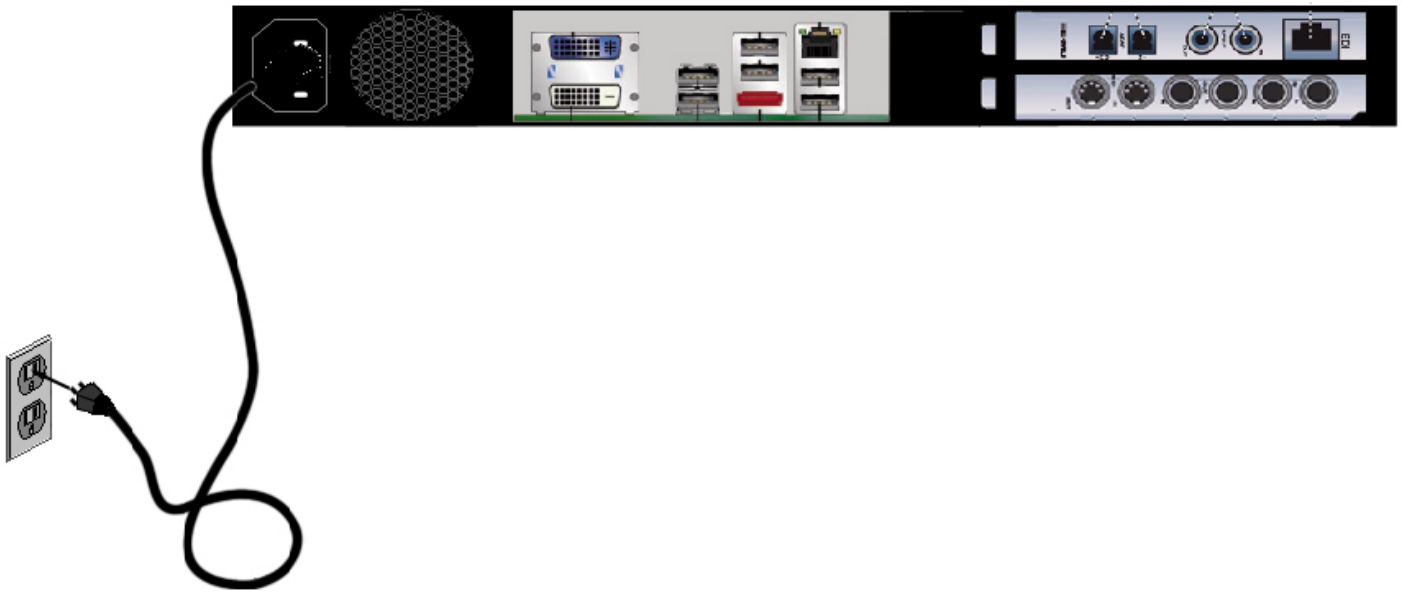
Please note:

- The SoundSlate has cooling vents on top and the back and left side of the chassis. These can not be blocked.

Connect all the cables (power and audio) prior to powering up the SoundSlate.

Connecting the Power Cable

Before powering up the SoundSlate, connect the system to an uninterrupted power supply.



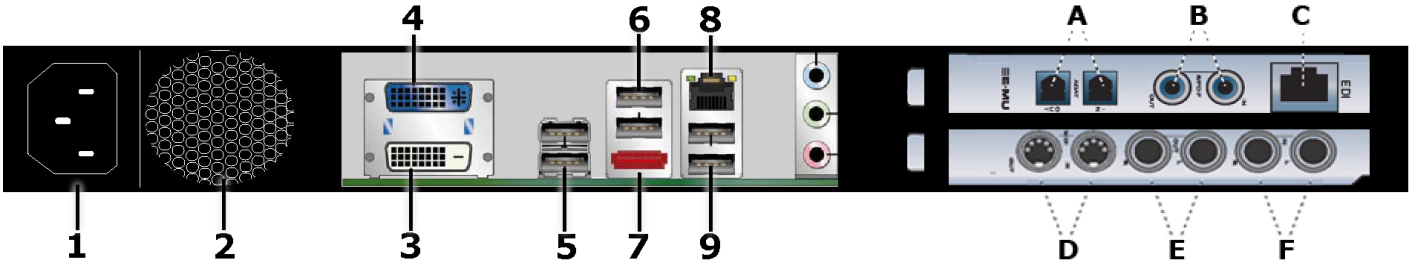
Front Panel Description



1. Power - On/Reset button
2. DVD Drive
3. USB 2.0 (2)

Rear Panel Layout

On a SoundSlate's back panel you will find:



1. AC power connector - Use the provided cable to connect to a power source.
2. Power supply fan opening. Please do not obstruct.
3. DVI-I Port 1 - (Use this to connect to an external monitor or projector.)
4. DVI-I Port 2 - (Use this to connect to an external monitor or projector.)
5. USB Ports (2) - Keyboard, Mouse, Hard-Drive, USB Camera
6. USB Ports (2) - Keyboard, Mouse, Hard-Drive, USB Camera
7. eSATA Port (1) - Use this to connect to lightning fast external hard drive
8. Ethernet Port - Connect to the a network or internet.
9. USB Ports (2) - Keyboard, Mouse, Hard-Drive, USB Camera

Audio & MIDI Connections

SoundSlate's back panel provides:

A) **ADAT Optical Digitial Input & Output:** The ADAT optical connectors transmit and receive 8 channels of 24-bit audio using the ADAT type 1 & 2 formats. The word clock contained in the input data stream can be used as a word clock source. Optical connections have certain advantages such as immunity to electrical interference and ground loops. Make sure to use high quality glass fiber light pipes for connections longer than 1.5 meters.

B) **S/PDIF Digital Audio Input & Output:** RCA phono jacks are standard connectors used for S/PDIF (Sony/Philips Digital InterFace) connections. Each jack carries two channels of digital audio. The E-MU 1010 receives digital audio data with word lengths of up to 24-bits. Data is always transmitted at 24-bits.

S/PDIF digital I/O can be used for the reception and/ or transmission of digital data from external digital devices such as a DAT external analog-to-digital converter or an external signal processor equipped with digital inputs and outputs.

The S/PDIF out can be configured in either Professional or Consumer mode in the Session Settings menu. The 1010 PCI card can also send and receive AES/EBU digital audio through the use of a cable adapter. See AES/EBU to S/PDIF Cable Adapter for details.

The S/PDIF input and outputs are usable at the 44.1kHz, 48kHz 88.2kHz and 96kHz sample rates, but are disabled for 176.4kHz and 192kHz. The word clock contained in the input data stream can be used as a word clock source.

C) **EDI Connector** (Not Used) - Connects to the MicroDock using the supplied EDI cable. This cable provides a a two-way data link between the E-MU 1010 and the MicroDock as well as supplying power to the MicroDock.

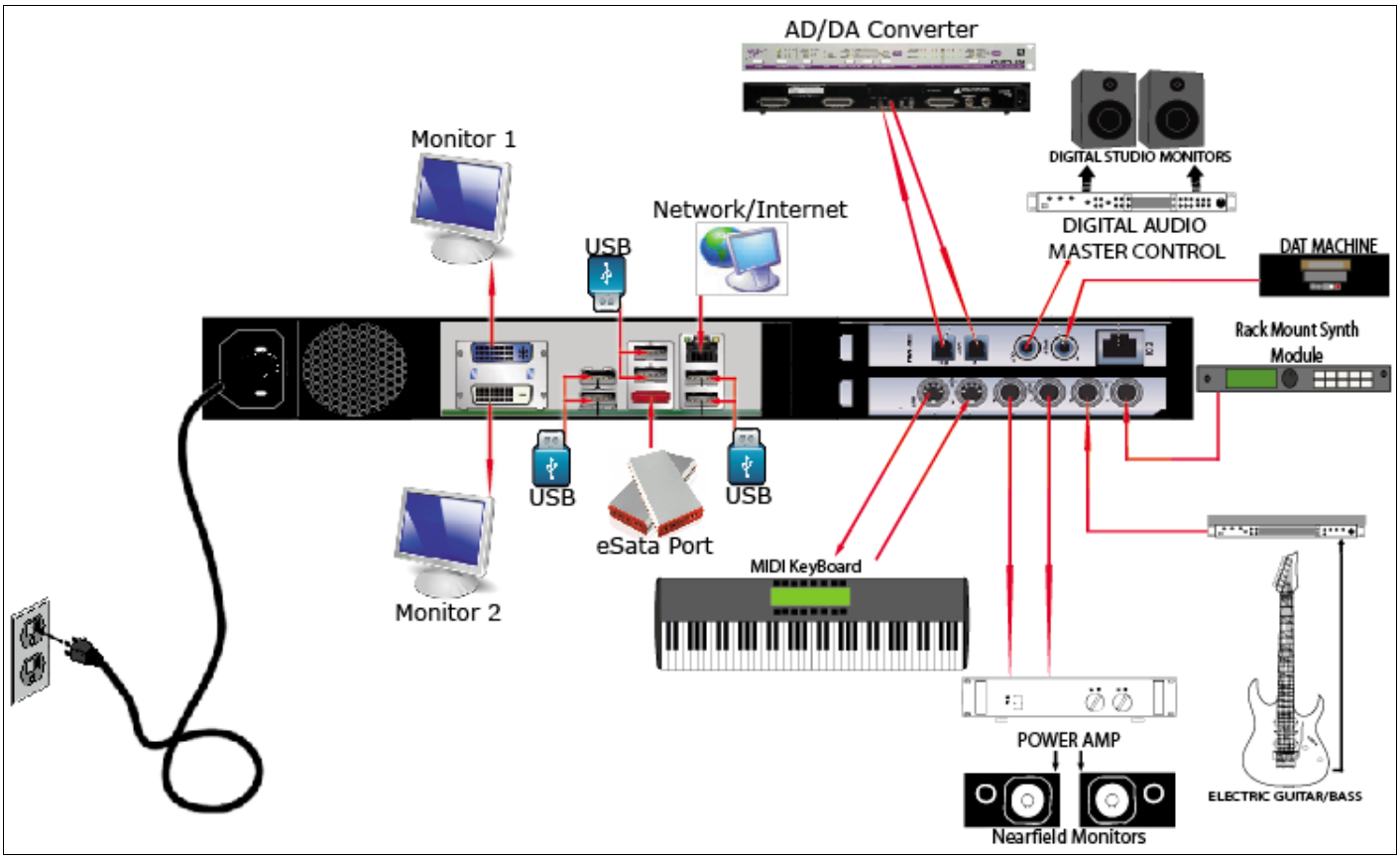
D) MIDI input and output

E) 2 line level, balanced analog outputs

F) 2 line level, balanced analog inputs

Basic Connection Diagrams

This page shows examples of connections to the various inputs and outputs.



There are plenty of places to connect devices to your SoundSlate. Contact support@openlabs.com if you would like to schedule a tech session to discuss using certain devices.

Here are some cables that work with SoundSlate. *The TRS cable is a balanced version of the quarter inch (1/4') cable.*



Turning on SoundSlate

The power switch on a SoundSlate is located on the front left side of the SoundSlate.



There is only one power switch on the SoundSlate. There is no seperate reset button.

Power Button Diagram

- Hard Drive Activity Light - Shows a orange light when there is disk activity.
- System Power Indicator - Shows a blue light when the unit is powered on.



Upon powering up, the SoundSlate will boot up in two phases: The first phase is the BIOS phase, or the pre-Microsoft Windows phase. This is when the hardware validates that all systems are functional within acceptable parameters.

Hardware devices, onboard memory, and other system devices run through a short diagnostic routine, and then the system will begin to load the operating system—Microsoft Windows XP. *It is normal for the BIOS screen to load upside down on your display. Once Windows XP loads, the screen will rotate to the correct orientation*.

Unless you plan to make immediate hardware changes, there is no need to access any of the BIOS configuration menus. If you do plan to make some BIOS modifications, you can access the BIOS by hitting the DELETE key when your SoundSlate first boots.

About the BIOS
System BIOS is a menu where you can configure your hardware settings. The settings pertain to hardware only, and will sometimes affect your software programs if configured incorrectly. If you plan to make modifications to the BIOS, it is recommended that you contact Open Labs technical support.

The second phase loads Microsoft Windows XP. This is an optimized installation of Windows XP, meaning that it contains all the tweaks that will enhance system performance.

Since this is the "Full Version" rather than the "Embedded" version of the Windows package, your SoundSlate has all the potential to be your most powerful sound production appliance without any hindrance to any computer-driven applications.

Most software supported under the Microsoft Windows XP platform can be used on your SoundSlate. Check with Open Labs support if you are unsure about a particular software.

After the system is fully booted, you should be in the Open Labs Graphical User Interface (GUI).

About the Open Labs GUI
The Open Labs GUI is not a standard part of Microsoft Windows XP. It is a customizable overlay to the standard Microsoft XP work environment. You can toggle between the two modes by clicking on the Open Labs icon (located in the lower left-hand corner of the screen), select Quit, and then select Go To Windows. You can always reinstate the Open Labs GUI by accessing the shortcut (named "Open Labs Shell") provided on the desktop.

For more information about the Open Labs GUI, see chapter 3.

3

Chapter 3 – Open Labs GUI

The first time you power up your Open Labs SoundSlate, it loads directly into Riff, once you close Riff, you may realize that the screen looks nothing like Microsoft Windows XP. The look is the Open Labs Graphical User Interface that will enhance your productivity, especially in combination with a touchscreen.

The Open Labs GUI In Detail

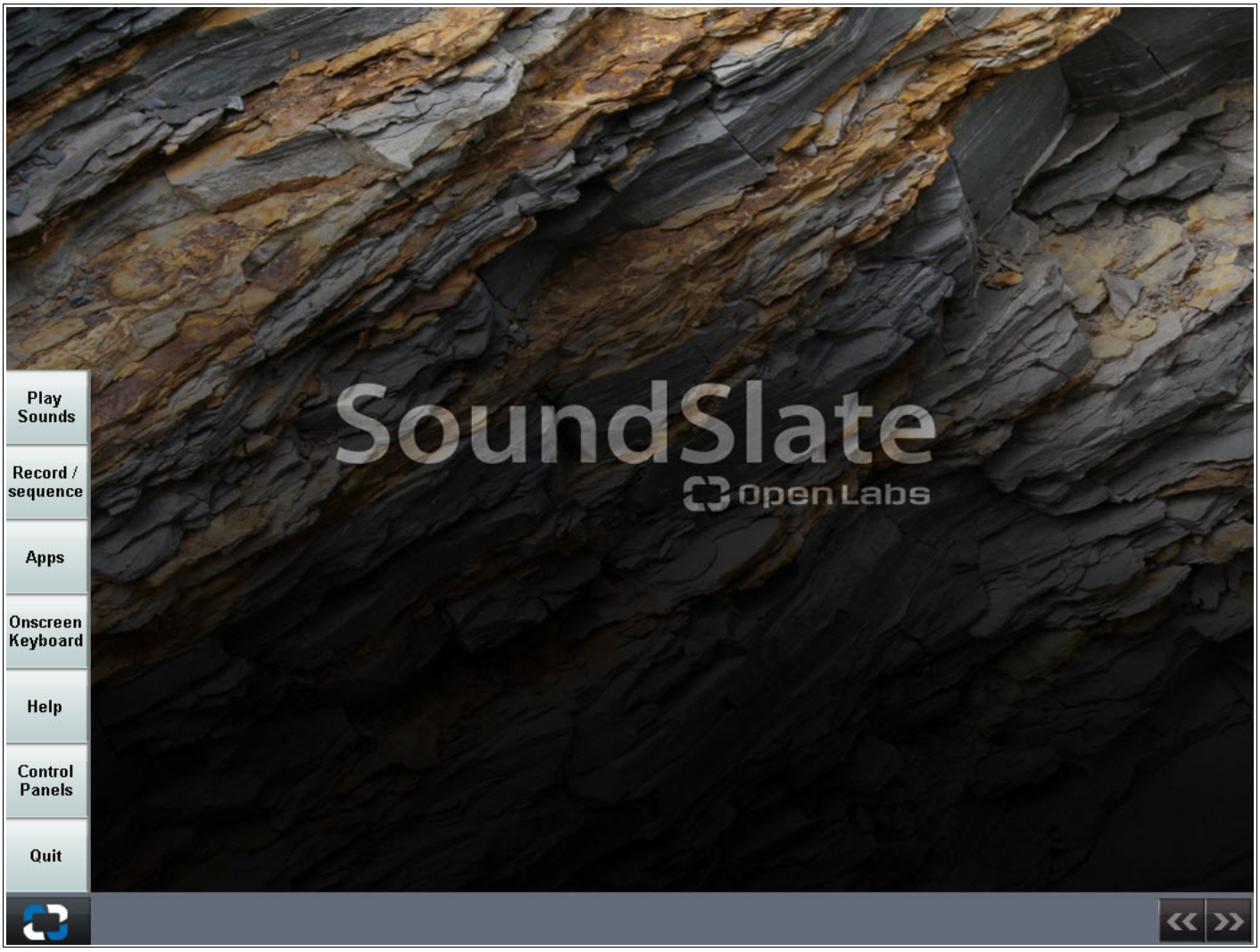
The GUI, or graphical user interface, is a customizable shell that allows the launching of applications from a single location. This GUI is optimized for use with a touchscreen and for musical performance. Traditionally, within the Microsoft Windows environment, you would have to make shortcuts for applications you use most often, and arrange them on your desktop or system tray for easy access. Because these items are often masked by open applications and windows, you may have to reposition your windows every time you wish to initiate a new application. Starting programs via the 'Start' button is also not ideal, because the program groups contain all installed programs, rather than just those few applications you use the most.

The Open Labs GUI is designed to allow access to the programs and tasks that you will use most frequently with your SoundSlate. After powering up your SoundSlate, Windows XP will launch, followed by the Open Labs GUI. You will notice right away how the GUI organizes and simplifies the tasks associated with making music.

Currently active applications will appear on the taskbar. If there are more applications open than will fit in the taskbar at once, you can scroll left and right using the arrows in the lower right-hand corner of the screen.

Clicking on the Open Labs icon in the lower left-hand corner will reveal shortcuts to a variety of applications and files.

The next page shows the GUI screen, and the first level of buttons displayed when you click the Open Labs icon.



Starting from the bottom, the icons will be explained.

QUIT
Clicking on this button will give you the following options:

- **Go To Windows:** This button will close the Open Labs GUI and take you to the standard Windows XP desktop.
- **Shut Down**: Use this to turn your SoundSlate off
- **Restart**: Use this to restart your SoundSlate.

CONTROL PANELS
The Control Panels button provides access to utilities that you might need while working on your SoundSlate. When you click on the Util button, you will see the following options:

- **Audio:** The Audio button directs the controls for your sound card. When you click the Audio button you will see the following options:
  - *Control Panel*: Launches the sound card control panel, from here you can adjust latency, sample rate, and boost input volume.
  - *Mixer:* This button launches the sound card mixer panel.
- **System Backup:** Opens Acronis True Image. (Please consult Tech Support before using).
- **File Browser:** Opens Windows explorer.

HELP

The **Help** button gives you access to various forms of information regarding your SoundSlate. When you click the Help button you will see the following options:

- **Remote Support:** Use this to connect to an Open Labs tech for support. **(Requires and appointment and active internet connection).** *Call 877-978-0888 and select option 2 to schedule a remote session.*
- **Tutorial Videos**: You will find video tutorials that will help you make recordings and play sounds.
- **Manuals**: Will take you to a folder containing user guides and manuals.
- **Open Labs Forum (Requires an Internet connection)**: To gain access to the Open Labs on line community, and technical support forums use this link: (http://forum.openlabs.com). Forum access requires registration.

ONSCREEN KEYBOARD

The **Onscreen Keyboard** button will display a graphical representation of a standard computer keyboard for touchscreen entry of alphanumeric input.

APPS

The Apps button is a way to access various programs installed on your SoundSlate. When you click the Apps button you will see the following options:

- **Burn CD-DVD:** Launches a CD/DVD burning program.
- **Play Media**: This button launches Windows Media Player, for the playback of media files.

RECORD/SEQUENCE

This button will launch one the MIDI/audio sequencers bundled with your SoundSlate. The options are:

- **Cubase LE** - Cubase LE is a digital audio workstation: a complete multitrack audio and MIDI recording, editing, processing, mixing, and mastering environment.

PLAY SOUNDS

You can use this button to launch Riff.

- Information on using Riff can be found in Chapter 4.

Using USB Devices

The SoundSlate comes with eight USB 2.0 ports. Two front USB connections, and six rear USB connections. If you require more USB ports, you can purchase a USB 2.0 hub.

Using the DVD/CD-RW Drive

The DVD/CD-RW drive can be opened by pressing the release button located below the ejection tray. Because Microsoft Windows supports bootable CDs, it is important that you do not leave any CDs that are bootable in the DVD/CD-RW drive so that you will not boot the CD accidentally. Adding CDs into the CD-RW Drive while the SoundSlate is running will sometimes initiate pop-up menus. This depends on how the CD was written. For example, typical software installation CDs have pop-up menus to assist in installation of its software when the CD is first inserted into the DVD/CD-RW drive. Alternatively, inserting an audio CD into the DVD/CD-RW drive will initiate CD-Player to play.

The CD drive can be used to burn files onto blank recordable CDs. Recordable CDs are sold at most computer stores and provide an inexpensive method of backing up your critical files.
You may use any number of programs to burn CDs with your DVD/CD-RW drive. Microsoft Windows XP will author CDs as well. To use the Microsoft tool, navigate to the "Start" button, and then double-click on My Computer.

Double-clicking on My Computer will open a window showing all your storage devices. Insert a black recordable CD (CD-R) into your DVD/CD-RW drive and your DVD/CD-RW drive icon should change to indicate the presence of a CD-R disc.

Double-clicking on this icon will open an empty window that is empty except for a legend to the left.

Using the drag-and-drop method, you can place multiple files into the empty area within the right pane of this window. Files should start to populate this empty area. Keep in mind that the maximum amount of data you can store on a single CD-R is approximately 700 MB, or 80 minutes of audio wav files.
When you are done, select: "Write these files to CD".

Another pop-up will appear and will provide a walk-through with step-by-step instructions to complete your CD. These step-by-step pop-ups are called wizards. Microsoft loves to use wizards, and you will encounter many wizards while using Windows XP.
Simply follow the Wizard to the end and your CD will be finished.

Using the Internet

The network adapter allows you to connect a high-speed network to your SoundSlate. Depending on your setup, the network could be local area only (just connects to other computers around you) or to the Internet. Sound Slates will support a 10 Mbits or 1000 Mbits network.
Anytime you connect to the Internet, it is vital that you use the same precautions as you would use with a normal computer. It is recommended that you keep Internet activities to a minimum to reduce risks of computer viruses, hackers, pop-ups, and other vulnerabilities inherent on the Internet.

Internet use for the SoundSlate should be restricted to the following:

- Connection to Microsoft to run updates of service packs. (Please check with Open Labs support for compatibility of Microsoft updates).
- Technical support with Open Labs.
- Updating virus definitions for your anti-virus protection.
- Collaboration with other musicians in sharing of music.
- Network printing and local file sharing.

Adding Software to your SoundSlate

Adding new software to your SoundSlate is not unlike adding software to your home computer. The process still requires that you have the software, either on a CD ROM or resident on your hard drive from an Internet download. System requirements of your SoundSlate should meet that of your software, or installation will fail. For example, you cannot install Macintosh software in a SoundSlate, or install Windows 98 compliant software. Your SoundSlate runs Microsoft Windows XP, and will only accept software that works on this platform.

As a guideline, most current software sold at your music supplier will work on a SoundSlate. It is only with older software applications that you should practice caution during installation. If you are uncertain whether software applications will work in the Microsoft Windows XP environment, please contact your software manufacturer.

Instructions for installing software are provided in the software package you wish to install. Please read the software's user guide for step-by-step instructions, or visit its website for support.
Be aware that there may be compatibility issues with some third party software. Your SoundSlate is extremely stable with the software installed as shipped. Open Labs does not recommend installing untested freeware, shareware or firmware to your SoundSlate. For questions about particular software products, please contact Open Labs support, or the software manufacturer.

New Software Installation Considerations

If you plan to add your own software, be sure you meet all the following criteria:

- The software you wish to install is compatible with Microsoft Windows XP.
- The software does not conflict with software that is already on your system. An example of this is having two brands of anti-virus software running at the same time.
- The software you are installing has support.
- Your SoundSlate meets or exceeds the recommended system requirements of the software you wish to install.
- Also ensure that you have the necessary disk space needed to install the program.
- System Recovery
- Should you install an incompatible program that compromises the stability of your SoundSlate, a hidden recovery partition has been set up for a complete system recovery. Call or email an Open Labs technician before attempting a system recovery. An Open Labs technician can be reached by calling (512) 444-6222, or by emailing **support@openlabs.com**.

4

Chapter 4 – SoundSlate as a VST Player



Riff is a multi-effect and multi-instrument rack for musicians and sound designers. It is also a VST host that runs as a standalone application.

Riff allows a user to stream the ASIO or VST input signal through chains of individual VST effects so you can use your SoundSlate as a real-time multi-effect processor.

It enables a user to play several VST instruments at the same time. You can layer several instruments to create complex sounds. Or, users may play instruments on different MIDI channels: It turns a computer into a multi-timbral sound generator.

To achieve very low latencies, the standalone version supports ASIO 2.0-compatible sound cards. It will enable users to use VST effects and instruments almost like real hardware devices.

The parameters of Riff and also parameters of the loaded VST plug-ins can be controlled through MIDI control change messages or VST automation.

Before you Begin

You should have a good working knowledge of an Open Labs workstation and a computer's operating system. For example, you will need know how to use the touch screen, touchpad, standard menus and commands. You should also know how to cut, copy, paste, open, save, and close files.

If this is your first time using any type of recording system, you may want to research the basics of Pro Audio, producing music, and MIDI recording. A good place to start is KVR Audio. (http://www.kvraudio.com/wiki/).

Also, be sure to register on the Open Labs user forum. (http://forum.openlabs.com). You can also refer to the Pro Audio Primer located in Appendix.

This next section will explain to you how to setup Riff after manually installing it. If Riff came stock on your Open Labs production station, you can skip this section as it will already be configured. However, if you have custom plug-in folders, you will want to read the "*Setting up your Plug-ins*" section on this page.

Setting up the Audio

To establish a connection between Riff and your audio output hardware, you will need to select the correct software driver.

1. Select the Audio Options from the Menu>Options>Audio Options tab.
2. In the Audio Device section of this window select your ASIO sound driver (if not already selected). This is the setting that allows Riff to communicate with your audio driver.
3. Hit "Apply", then "OK" to commit the changes and exit.

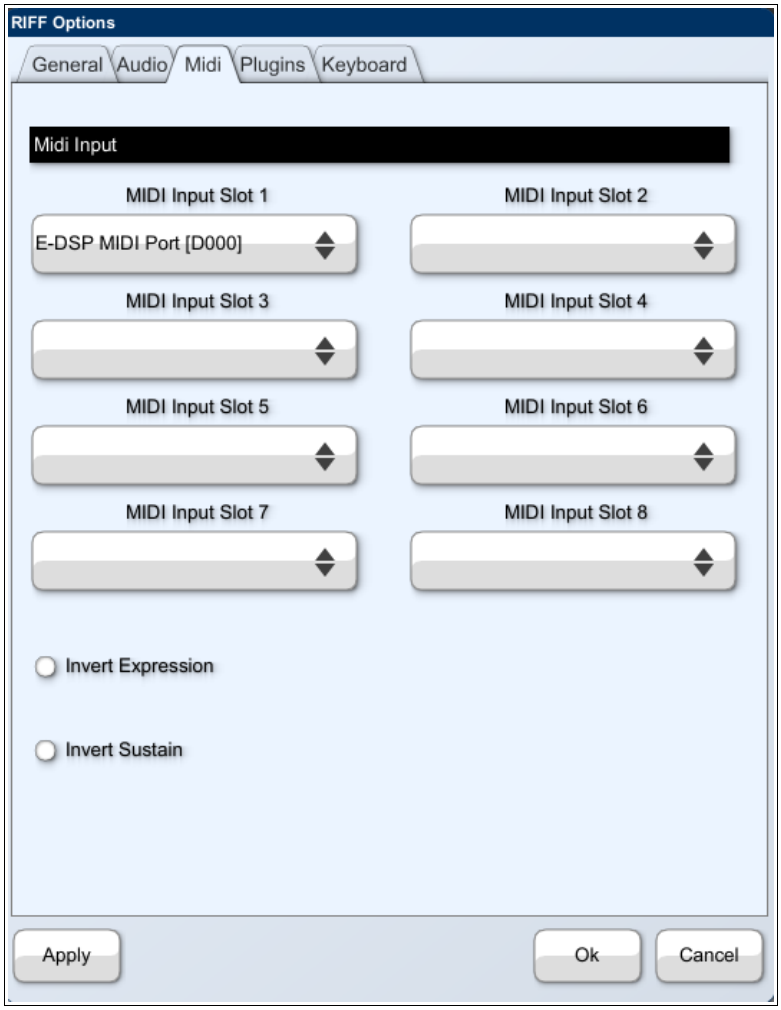


- *Audio Device*: Use this to select the primary ASIO sound driver.
- *Audio Input*: Use this to select your sound card's audio input. This application supports multiple audio input sources.
- *Audio Output*: Use this to select your sound card's audio output. This application supports multiple audio output sources.

Setting up the MIDI

To establish a connection between Riff and your MIDI driver, you will need to select the correct MIDI port. Riff allows you to select multiple MIDI-in and MIDI-out ports. All Open Labs systems use the "Open Labs Direct" driver.

1. Select the MIDI Options from the Menu>Options>MIDI Options tab.
2. There are four MIDI input boxes. Use the "MIDI Input 1" box to select your MIDI hardware driver.
3. If your system has more MIDI ports available, use MIDI input boxes two, three and four to select those MIDI-in drivers. If you have an 88-key controller for instance, you can select the MIDI driver that device is connected to.
4. If you intend to send MIDI to an external device, use "MIDI output" 1 to select that external MIDI hardware driver.
5. Hit "Apply", then "OK" to commit the changes and exit.

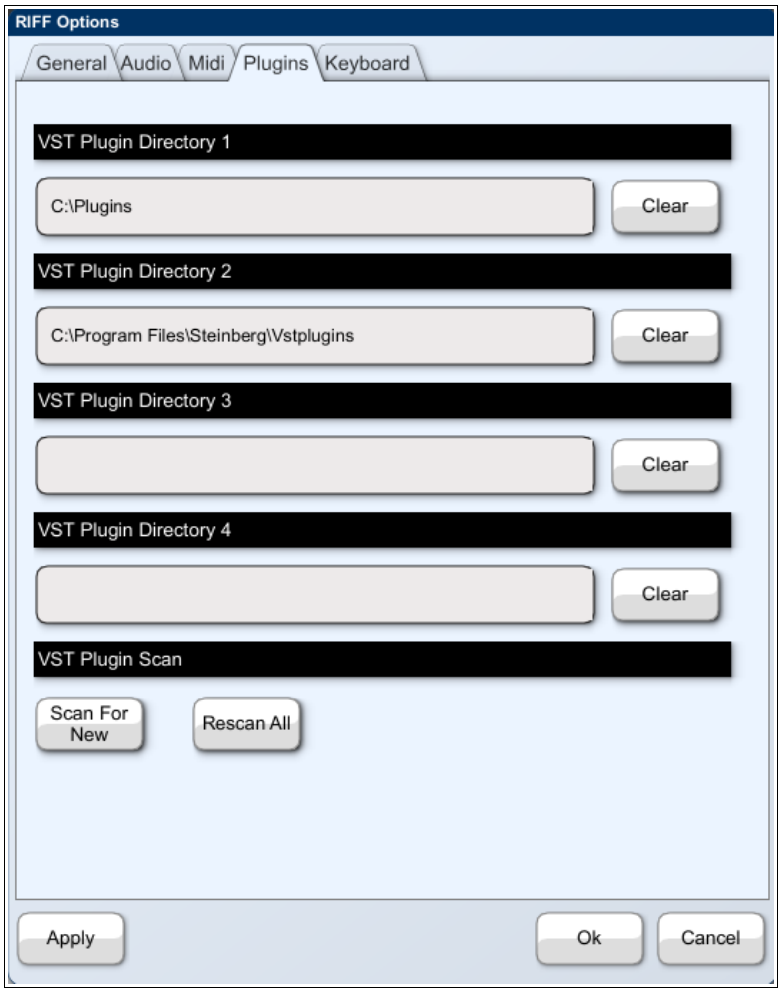


- *MIDI Input*: Use "MIDI input 1" to select your primary MIDI driver. The application also allows you to select three additional MIDI drivers.
- *MIDI Output*: Use this to select an available MIDI driver to send MIDI to an external source.
- *Invert Expression*: If you have an expression pedal connected, using this option will allow you to invert the phase of that pedal.
- *Invert Sustain*: If you have a sustain pedal connected, using this option will allow you to invert the phase of that pedal.

Setting up Your Plug-ins

By default, all Open Labs production station's plug-ins are located in "*C:\Program Files\Open Labs System (do not delete) Plugins*". With Riff, you can add three more additional locations for plugs. When you install your own plug-ins, please install them to "c:\plugins\". Then tell Riff where this folder is. To add a new plug-in folder:

1. Select the Plug-in options from Menu>Options>Plugins tab.
2. Click on the open space or directory to browse.
3. Browse to the folder where your plug-ins are located, and select it. *The default user plug-in folder is "c:\plugins".*
4. Now press the "Rescan All" button. Hit "OK".



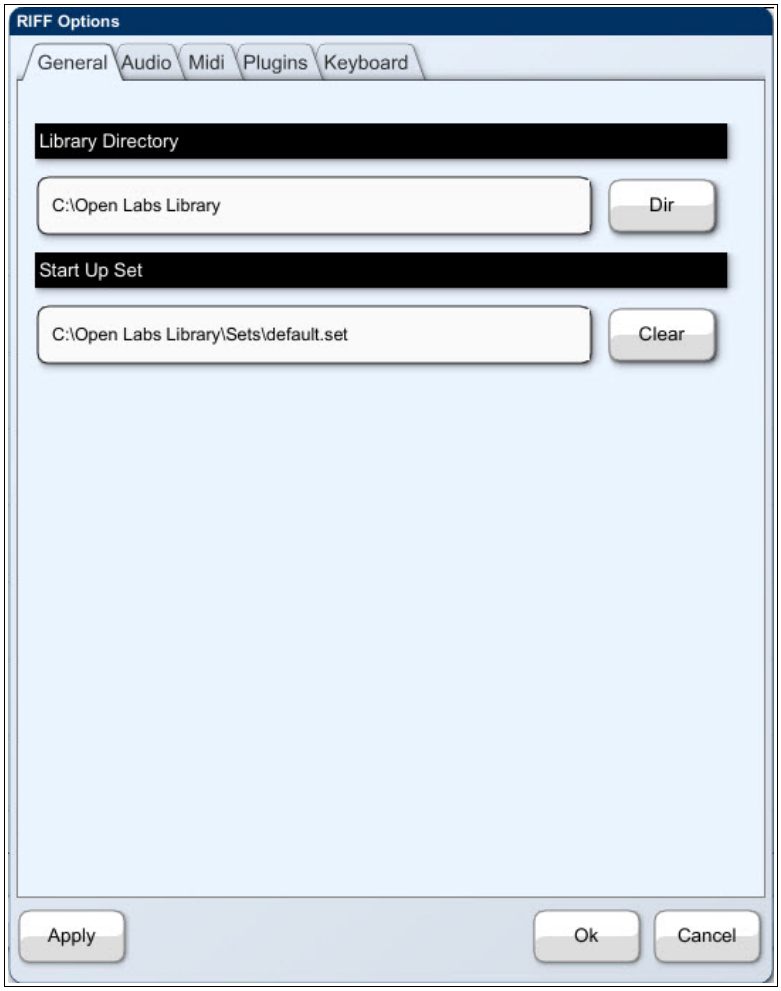
- *VST Plugin directory*: Use this window to select your main VST plug-in directory. Use the "Clear" button to reset this window. Also, we have provided you three more locations to use as VST directories.
- *VST Plugin Scan*: After adding a VST plug-in directory or directories, use either "Scan for New" or "Rescan All" to have Riff scan the folders you have added. All new plug-ins will be available after scanning. There is no need to relaunch the program.

Setting up General and Keyboard Options

The Options window contains a variety of important user-definable preferences used to configure Riff for your audio/MIDI hardware, plug-in locations, and to customize the way you work.

General

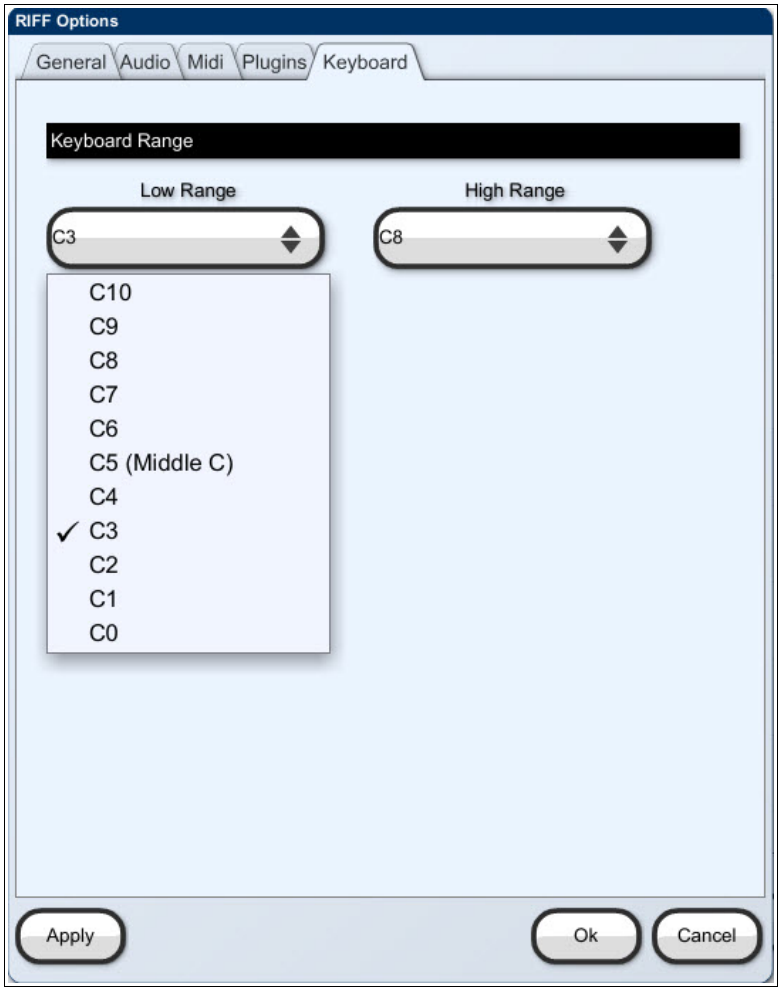
Select *Options* from the Menu>Options>General Options>General tab.



- *Library Directory*: Use the "Dir" button to browse over and select a directory to be used as the default save folder for Riff.
- *Start Up Set*: Click in this window to choose a project to be the default project that starts up with Riff.

Keyboard

To configure the Keyboard Range, select Keyboard from the Menu>Options>Keyboard Options>Keyboard tab.



- *Keyboard Low Range*: Allows you to select the low range of your keyboard. The range is C0-C10
- *Keyboard High Range*: Allows you to select the high range of your keyboard. The range is C0-C10

Connecting an External MIDI Keyboard

Since SoundSlate has no MIDI keyboard (piano keys), you will need to connect an external MIDI controller to play VSTi's. It is pretty easy to do, and this section will give you some examples of how to connect a USB MIDI controller and a standard MIDI controller.

While some USB MIDI controllers are *plug n play,* most USB MIDI controllers will require you to install drivers prior to connecting the device. On the otherhand, standard MIDI devices will not need any drivers installed, because SoundSlate's onboard MIDI ports are already installed.

Please make sure your devices drivers are installed before connected your controller otherwise it will probably not work.

Connecting an External Controller to Riff

Standard MIDI Device Connection

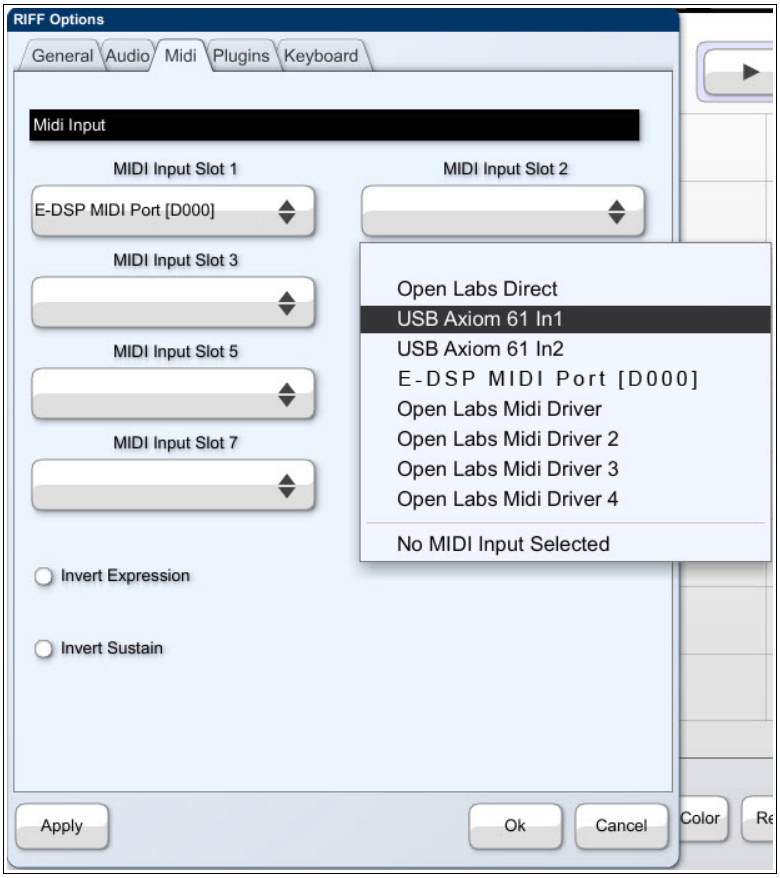
SoundSlate ships ready to receive MIDI signals from standard MIDI devices. Plug in the device, and start playing sounds. The default MIDI port is the "*E-DSP MIDI Port [D000]*".

USB MIDI Devices

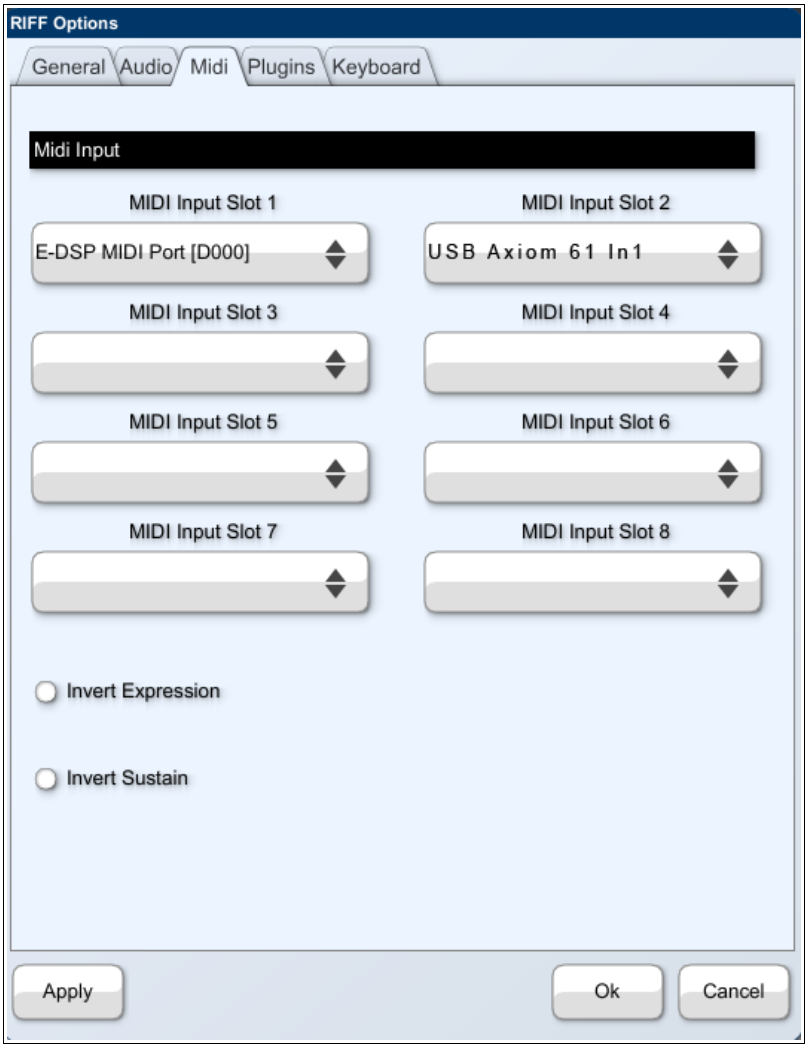
To connect a USB controller to Riff, make sure your controllers drivers are installed and the device physically connected. Once they are, open Riff and go to *Menu>Options>MIDI.*

Riff gives you eight MIDI input slots. Number one should always be reserved for the *Open Labs Direct* MIDI driver.

For this example, the M-Audio Axiom 61 is the controller. Go to "MIDI Input Slot 2" and select your USB controllers MIDI port.



Once the MIDI ports are assigned, the MIDI options will resemble the picture below.



Now press [Apply] the press the [Ok] button. This new ports are now allowing you to use your external MIDI controller with Riff.

That's all you have to do, pressing a key on the controller will trigger an instrument in Riff.

4 a – Riff Architecture

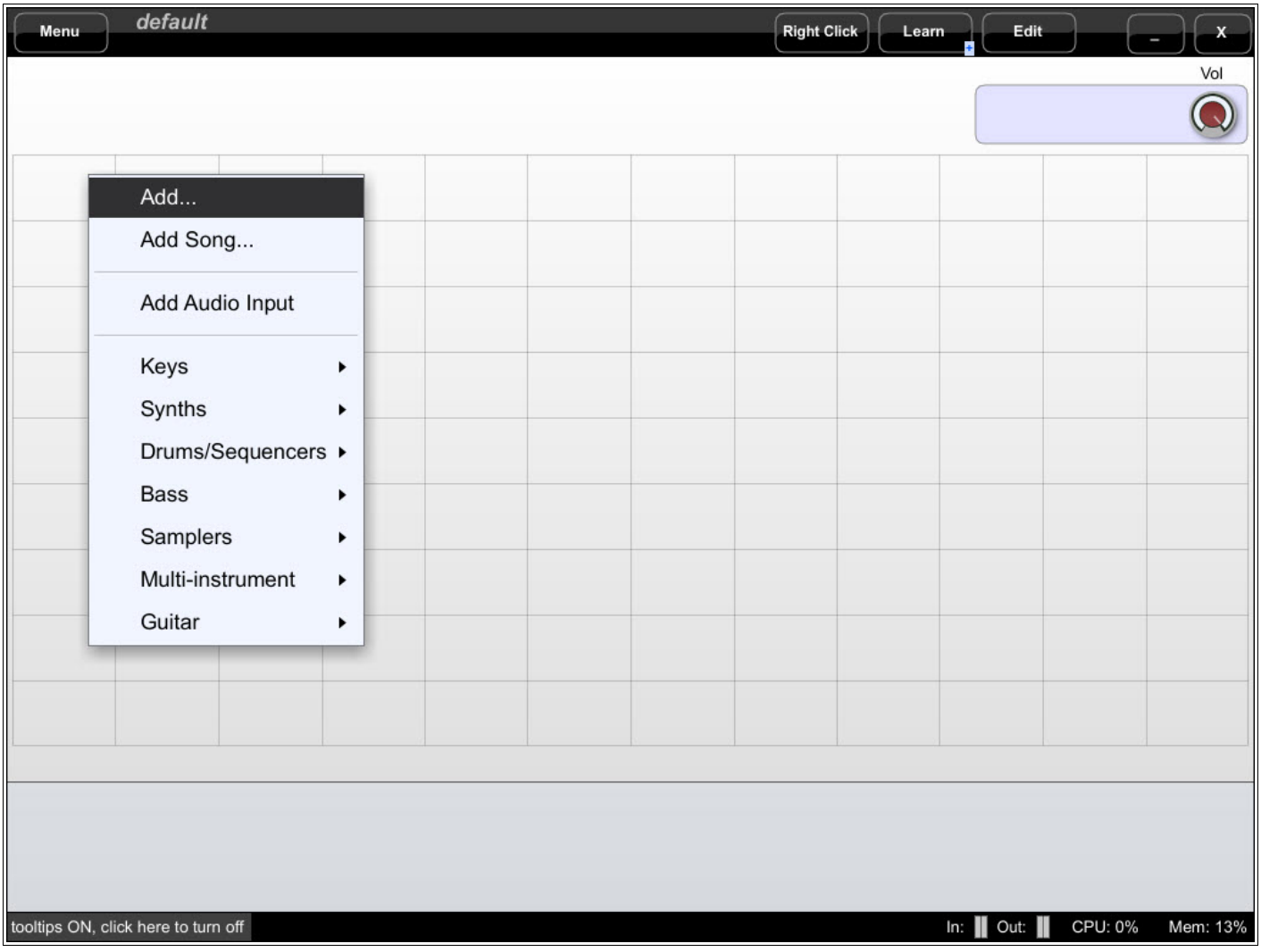
This section contains important information about the layout and organization of the various elements of Riff. This chapter contains no hands-on tutorials. However, the information here is vital to your understanding of this awesome application. *The tutorials start in Chapter 4.*

A Performance Environment

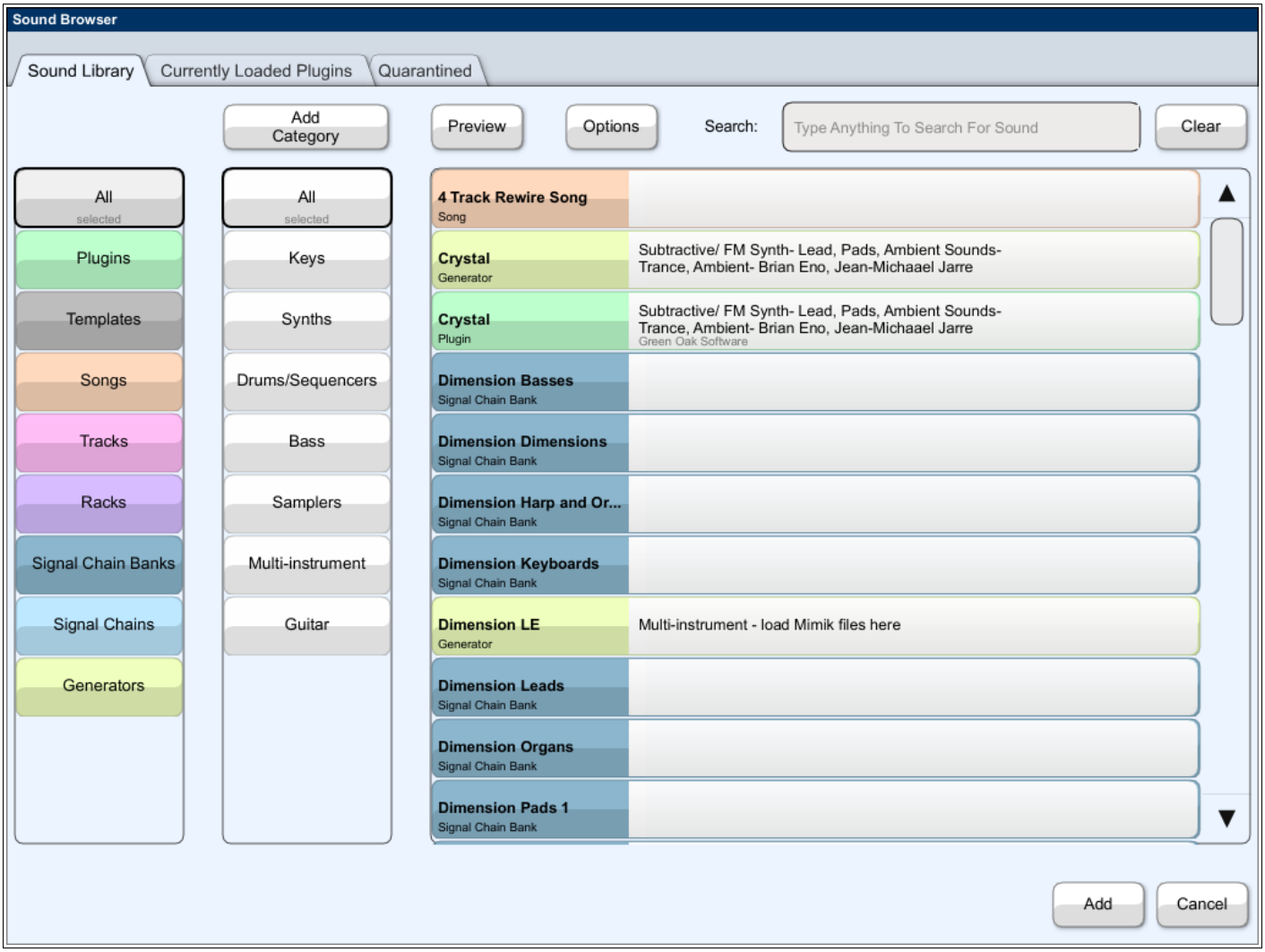
Riff is an environment to organize collections of VST sound modules, effects, and performance sets. You can think of Riff as a virtual live performance rack that allows you to create songs, set lists, signal chains, effect chains, and virtual live controls. By reading this section, you will gain knowledge of the Riff hierarchy starting with the Set List.

Set List Window

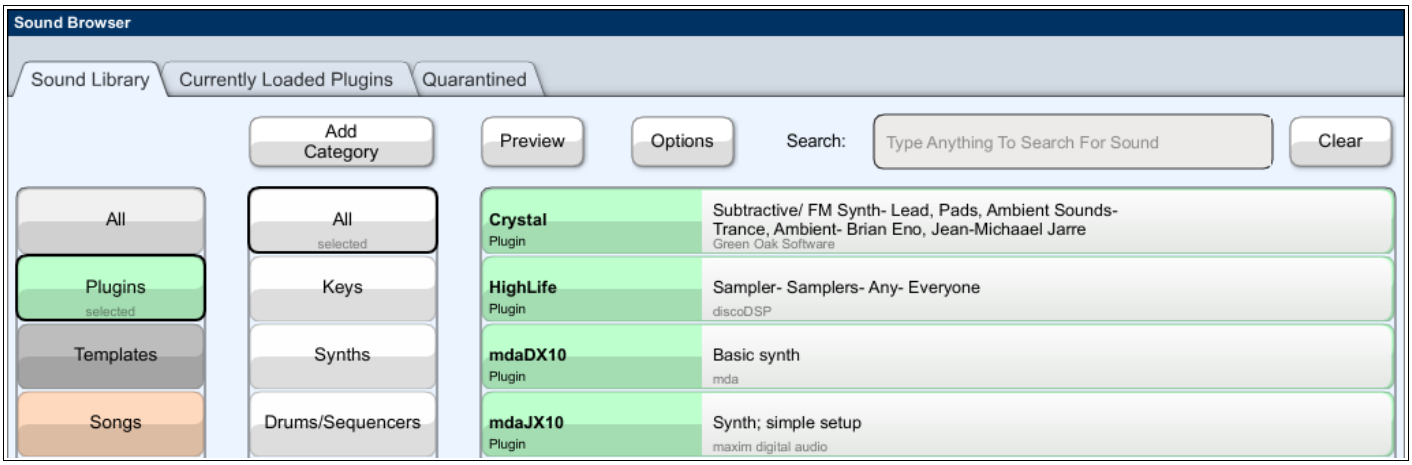
The Set List window is where you start, allowing you to add instruments. Once you right-click and press "Add" you will see the following windows. Each square of a Set List is called a Song.



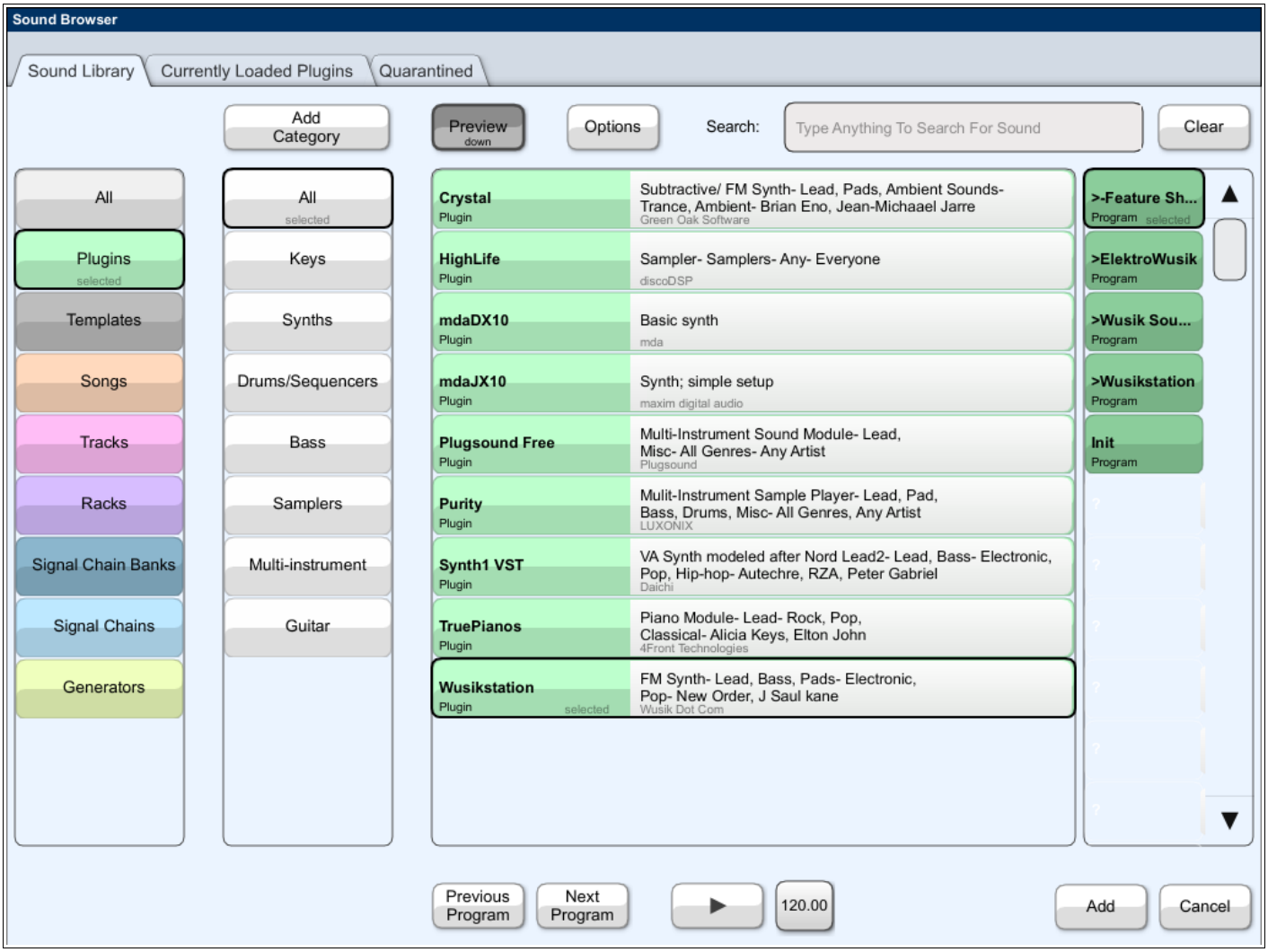
Sound Browser (Default View)



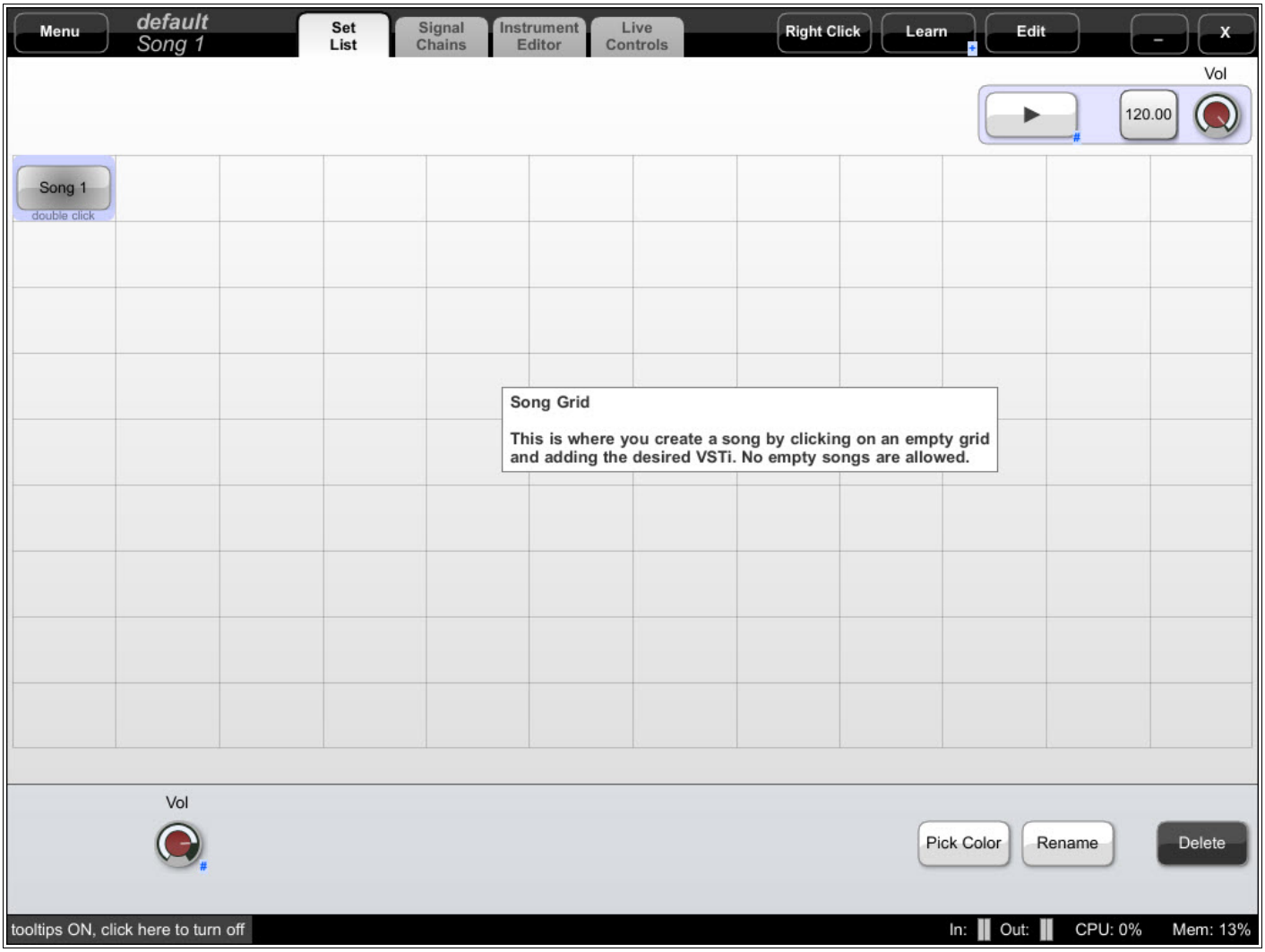
Clicking the [Plugins] *tab will show the plug-ins as well as a brief description of said plug-in.*



Sound Browser (Preview)



Pressing the [Preview] tab at the bottom of the screen will allow you to preview a selected instruments presets. Once you find you like, press [Add].

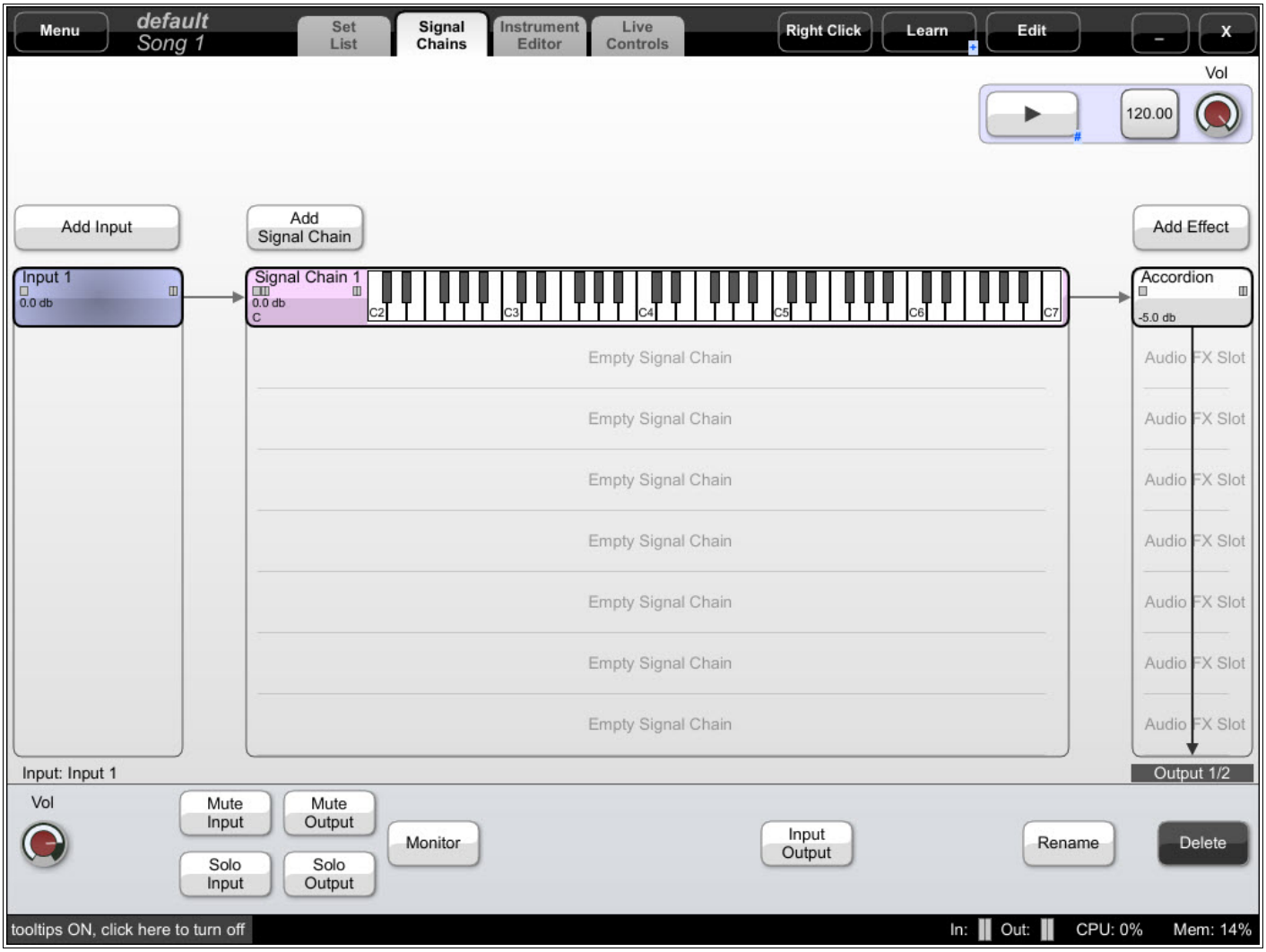


Before you can play a sound with Riff, you need to add a sound to the Set List. The Set List is grid-based and allows you to switch between Songs and sounds. You can easily see the status bar along the bottom.

There are VU meters along that bottom showing the following:

- Audio levels for input and output signal
- Meters that display the overall CPU and Memory usage of the ENTIRE system.
- MIDI activity light.

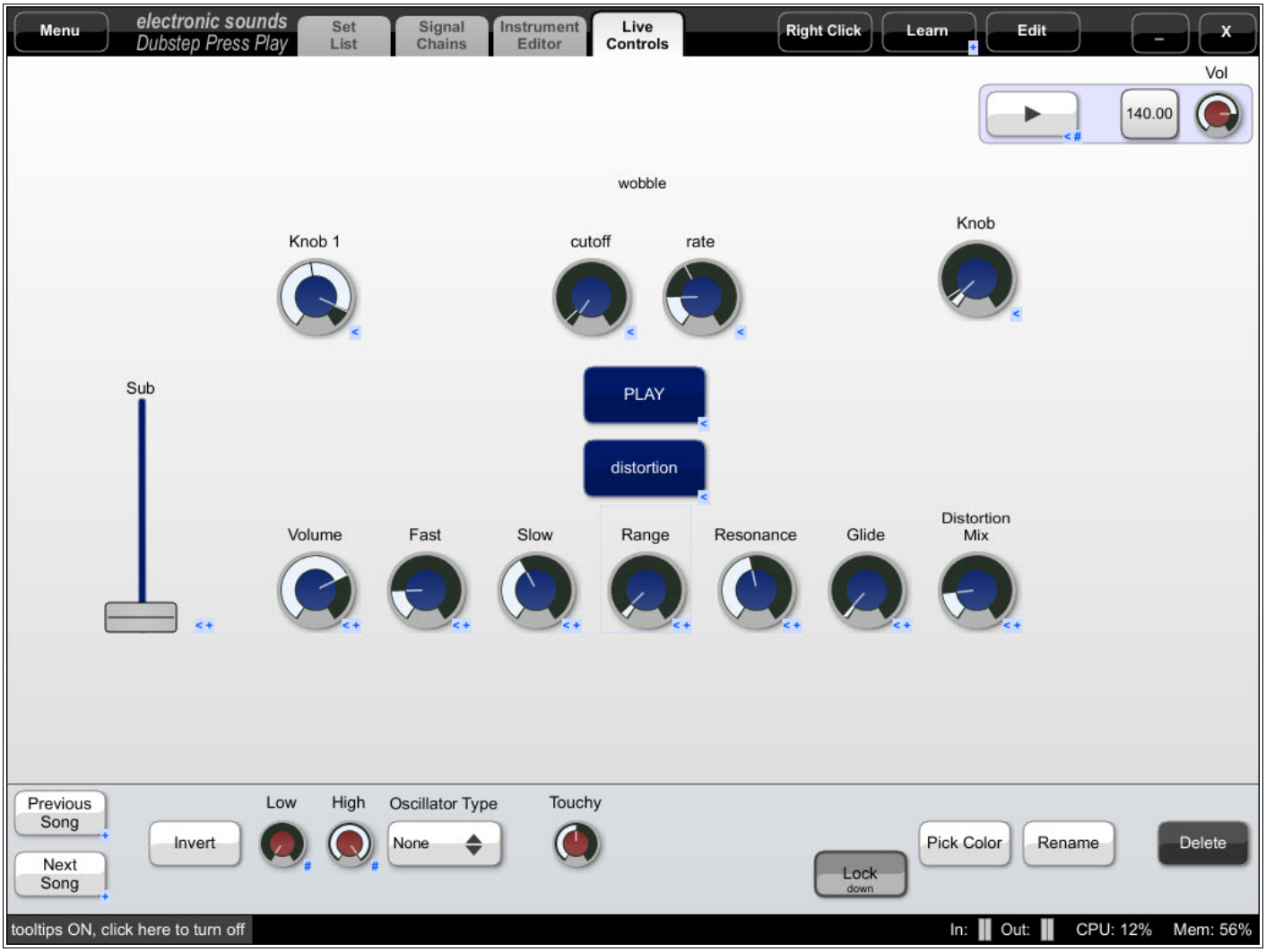
Since this window is scroll-free, you can literally play an entire show with this screen. Refer to Chapter 4 for more information on using the Set List.



The Signal Chain is a collection of plug-ins that start with an audio input stream and end with an audio output stream.



The Instrument Editor allows you access to a selected instrument/plug-in's graphical user interface (GUI). You can also reach the Instrument Editor screen by double-clicking the instrument in the Instrument Rack or in the Signal Chain.



This is one of the most powerful features of Riff. On the Live Controls screen, you can build a custom set of MIDI controllers that can be linked to any instrument, effect, or signal chain control contained within a Song. Live controls for a Song are saved within that Song and can be easily recalled, imported or exported.

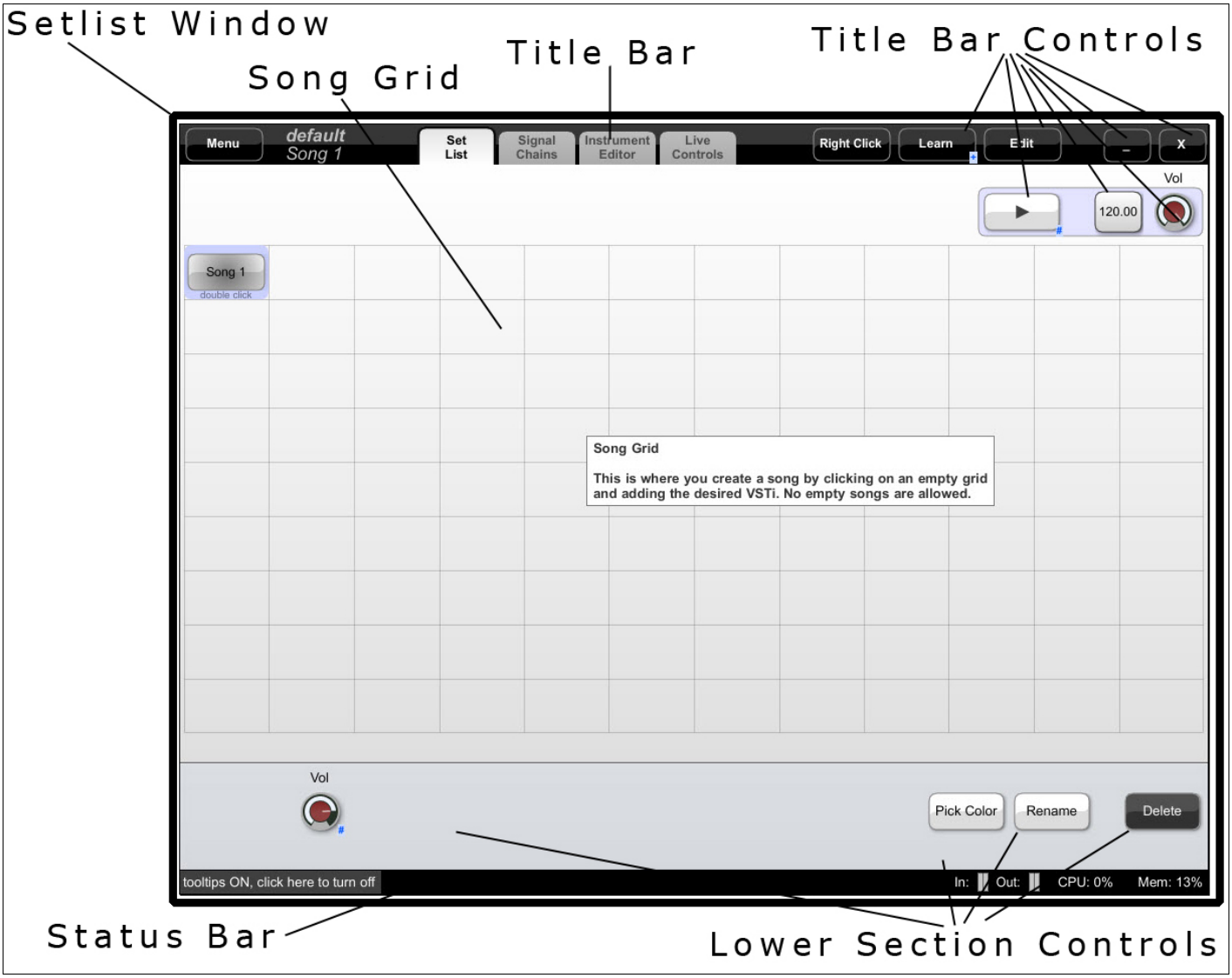
4 b – Understanding Riff

This section will begin to familiarize you with navigating around Riff. Initially, don't worry about being unsure with the controls - all will be explained.

Launch Riff

When you first launch Riff, the screen shown below appears. This Setlist is loaded with a default set. Other defaults will have different looks.

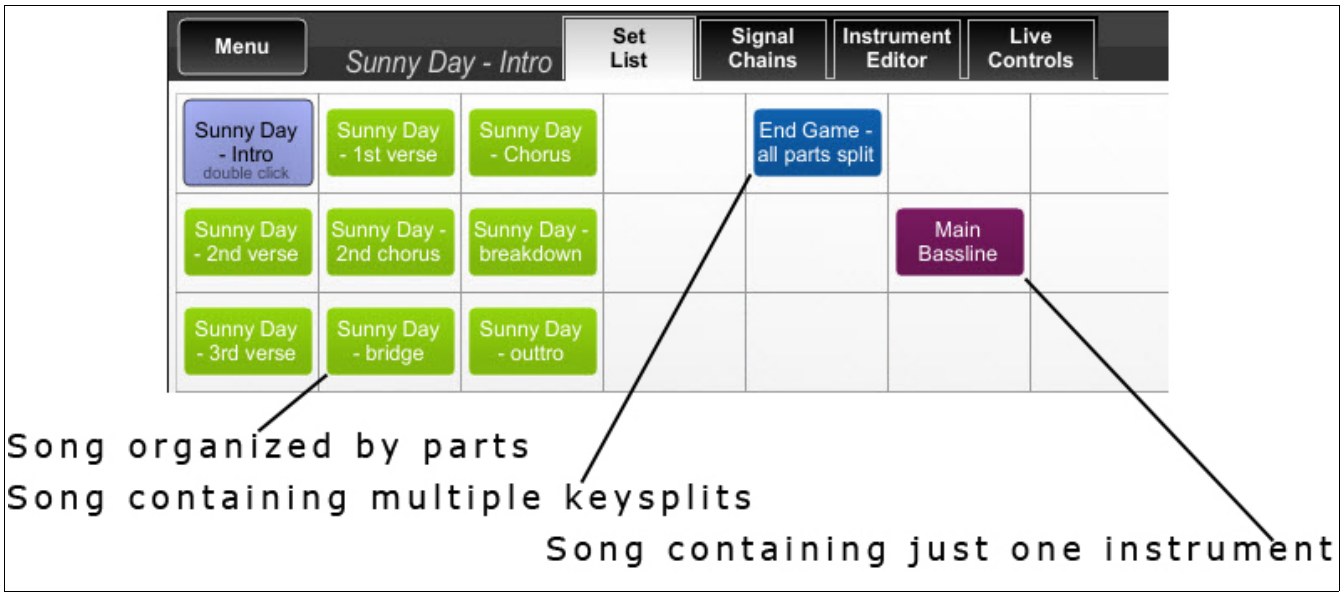
Set List



This screen is called the "Set List". **The Song Grid, Title Bar, Title Bar Controls, Status Bar,** and **Lower Section Controls** make up this screen. This window allows you to add and switch between virtual instruments. Each square of the grid is called a "Song".

Song

Use each square of the grid to create a song. A song can either be one square, or it can be a group of squares to create all the parts for the song you are playing. As shown in the example below, using the song grid you can arrange songs into a structure, rename and color code them for easy organization.



Each song is not just limited to one instrument. Once you go into the signal chain of each song square, you have tons of options, like creating keyboard splits and adding VST effects. *Refer to the section regarding the Signal Chain for more information.*

Title Bar

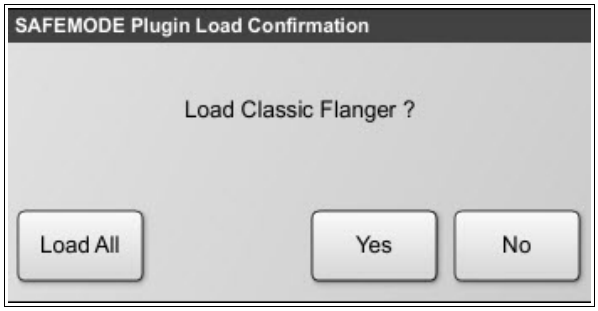
The Title Bar is the area of Riff that contains the program's control menu items. Items include: Menu, Set List, Signal Chain, Instrument Editor, and Live Controls.



Menu Bar

- File
  - New Set
    - Creates a new Set from scratch.
    - Launches a confirmation dialog, because creating a new Set will eliminate current one.
    - If you want to make a new Set, the current Set will be replaced with the new Set.
  - Open Set
    - Opens the Set List Browser.
  - Open Recent Set
    - Lists recently accessed Sets. Selecting from here will launch the selected set.

- Open Set in Safemode...
  - Safemode is a simple way of diagnosing a problematic Set. You can choose whether or not to allow an instrument or effect to load.



- Save Set
  - If no changes have been made to a Set, this is grayed out.
- Save Set As...
  - Choose this to save your Set with a custom name.
- Import (Only visible when you have the Instrument/Effect Editor tab selected)– *Allows the importing of the following items to a currently opened project.*
  - Import FXP (Program)
  - Import FXB (Bank)
- Export (Only visible when you have the Instrument/Effect Editor tab selected)– *Allows the exporting of the following items.*
  - Export FXP (Program)
  - Export FXB (Bank)
- Exit - *Exits Riff*

- Edit
  - Cut, Copy, & Paste items to and from the clipboard
- Options
  - General, Audio, MIDI, Plugin, Keyboard and Skin Editor options.
- Help
  - Open Help
    - Opens the Riff Help File.
  - About
    - Displays the information regarding the build version of Riff.

Title Bar Controls



Right-Click
When selected, your next left-click peforms the action of a right-click.

Learn
Switches the host into MIDI Learn Mode. Here, you can latch live controls to VSTi's. There are two types of Learn available in the host. See Chapter 4 for more information.

- Learn – Ties knob to a specific controller.
- Learn Relative – Enables you to map a controller to a common parameter, like the volume knob of multiple songs or signal chains.

Edit Mode/Live Mode
In Edit Mode you can move around Songs, arrange them, set the properties of the Banks and create tabs. Essentially, this is a way for you to lock down the interface.

When in Edit Mode here are a few things you can do:

- Create a new Song
- Import/Export sets
- Build a Signal Chain
- Create and Map Live Controls

When in Live Mode the following restrictions apply:

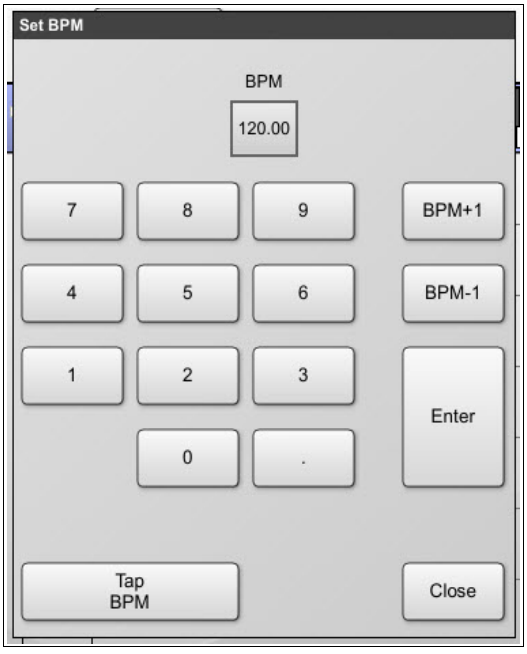
- No editing of live controls parameters except setting low and high
- No right-click menus anywhere
- No adding or removing Banks
- No adding or removing Instruments / Signal Chains

Minimize
Minimizes Riff

Close
Exits Riff

Play Button
This button toggles between play and stop. This is for instruments that have built in sequencers (Guru, Stylus RMX, or Reaktor for instance) and effects that are tempo synced.

BPM
This is the control for the global tempo. (*BPM = Beats Per* Minute) Pressing The box under "BPM" brings up the BPM input window.



Main - Volume Knob
Overall Master output volume knob of the host

- Can be dragged up or down to change value
- Ctrl + Left-click restores default (This applies to all knobs in Riff)
- *Right-clicking here only brings up "Learn" mode*

Lower Section Controls



Previous and Next Song
When you have more than one song on your setlist, the Previous and Next song buttons will appear. These allow you to go to the next song of your set. They can be MIDI learned to a surface control.

Vol(Volume)
This volume knob controls the volume for the individually selected song. It is independent from the Main Volume control.

- Right-clicking on the volume knob brings up Learn Mode.
- Learn – Ties the Vol knob to a specific controller
- Learn Relative – Ties the Vol knob to a controller but stays latched to the control volume on the selected preset. Whatever preset is in focus will have its volume controlled

Pick Color
Brings up the Color Picker dialog box. Here, the user can select the color choice for the preset.

Rename
This button allows you to rename the preset.

Status Bar



Tooltips
Tooltips will display info on what is being selected.

Audio VU Meter
The Audio VU meter shows the signal for the master output bus. By default, this will show outputs one and two. This can be changed to show whatever outputs you are using.

MIDI Indicator
The MIDI Indicator shows whether Riff is receiving MIDI from any MIDI controllers. It also displays the note/MIDI value.

CPU
CPU monitors the current processor usage of the entire system. Not just the program.

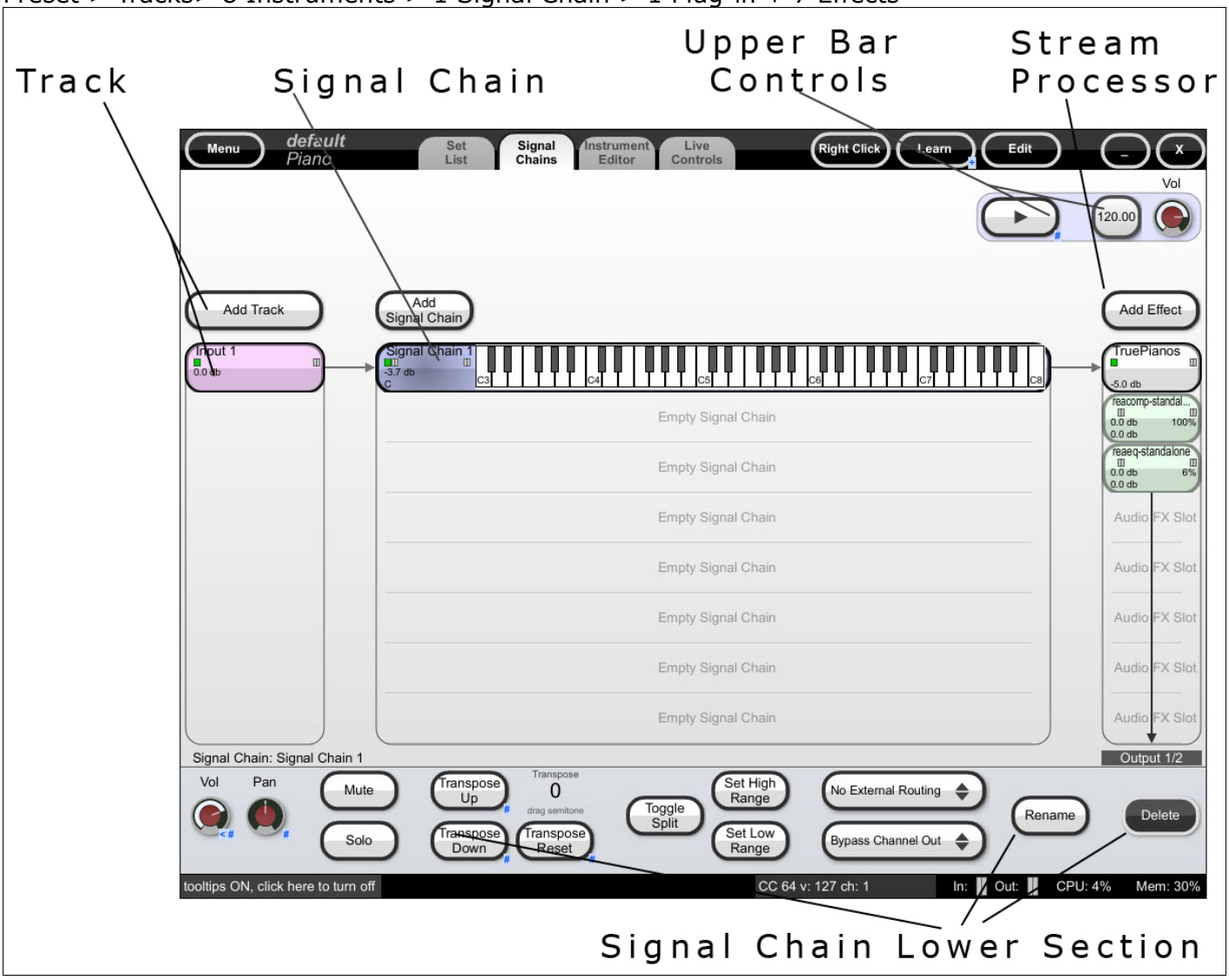
- Displays the usage in percentages.

MEM
MEM displays the physical memory usage of the entire computer system.

Signal Chain

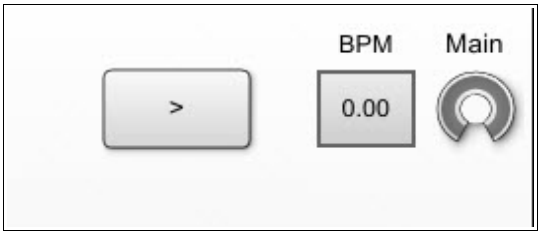
This is the main window where instrument racks and effect chains are created and managed. These are the components that comprise the Presets from the Song/Set grid pages. Each of the Song buttons on the Song/Set Grid page has a Signal Chain. Moving from left to right, each Track is composed of up to 8 instruments. Each Instrument can have up to seven effects added to it.

Preset > Tracks> 8 Instruments > 1 Signal Chain > 1 Plug-in + 7 Effects



Upper Bar Controls

The Upper Controls are located above the Stream Processor and below the Title Bar.



- *Play Button* – This button toggles between play and stop. This is for instruments that have built in sequencers (Stylus RMX, or Reaktor for instance) and effects that are tempo synced.
- *BPM* – This is the control for the global tempo. (*BPM* = *Beats Per* Minute) Pressing The box under "BPM" brings up the BPM input window.
- *Main* – Master output volume knob of Riff.

Track

- *Add Track*– This allows for a Track to be created with a signal chain containing a VSTi, or an empty signal chain with no VST. In order to use the VSTi's and play them, a Track must be selected with the desired plug-in.

Track Column Controls



The lower section of the screen shows the controls for the Track Column. Those controls are:

- *Vol* – Volume control for the Track. All instruments and their signal will be affected.
- *Mute MIDI* - Mutes any MIDI being sent on the selected Track.
- *Solo MIDI* - Solos MIDI being sent on the selected Track.
- *Mute Audio* - Mutes the selected Track.
- *Solo Audio* - Solos the slected Track.
- *Monitor* – Allows you to monitor multiple tracks simultaneously.
- *Input / Output** - Brings up the MIDI I/O window.
- *Rename* – Renames the currently selected Track
- *Delete* - Deletes the currently selected Track

* The [Input/Output] button will allow you to change the MIDI input devices and MIDI output devices for the selected Track. More information in Chapter 4a.

Double-clicking a Track shows the Rack Column. The lower section of the screen shows the controls for the Rack Column. Those controls are:

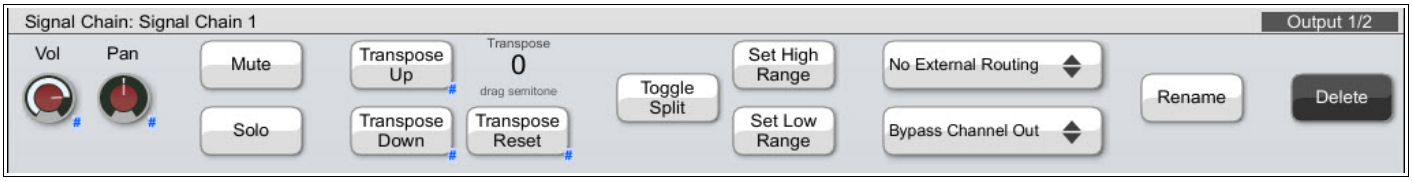


- *Vol* – Volume control for the Rack. All instruments and their signal will be affected.
- *Mute* – Mutes the Rack. All instruments and their signal chains will be affected.
- *Solo* - Solos the Rack and mutes all other Racks.
- *Audio Output* - Allows you to route a rack to a particular separate output.
- *Rename* – Renames the currently selected Rack.
- *Delete* - Deletes the currently selected Rack.

Signal Chain

- *Add Signal Chain* – Accesses the Sound Browser, where you can select a plug-in and add it to the Instrument rack. Up to eight instruments can be added per Track.

Signal Chain Column Controls



The lower section of the screen shows the controls for the Signal Chain Column. Those controls are:

- *Vol* – Volume control for the Signal Chain. All instruments and their signal will be affected.
- *Pan* – Pan control for the Signal Chain. All instruments and their signal chains will be affected.
- *Mute* – Mutes the selected Signal Chain. All instruments and their signal chains will be affected.
- *Solo* - Solos the Signal Chains and mutes all other Signal Chains
- *Transpose Up* - Transposes notes sent by one octave up.
- *Transpose Down* - Transposes notes sent by one octave down.
- *Transpose Reset* - Resets any transposed notes back to zero.
- *Toggle Split* – Allows you to open or close any keyboard splits you have created. This resets the split back to default.
- *Set High Range* - Selecting this button brings up the Learn High Range dialog box prompting the user to select the MIDI note that will be the highest note sent to the instrument. The user hits the key on the Ivory keyboard and the high ceiling is set. The change will immediately be seen on the keyboard layout on the Instrument rack.
- *Set Low Range* - Selecting this button brings up the Learn Low Range dialog box prompting the user to select the MIDI note that will be the lowest note sent to the instrument. The user hits the key on the Ivory keyboard and the low is set. The change will immediately be seen on the keyboard layout on the Instrument rack.
- *Signal Chain Router (External Routing)* – Quickly allows you to change MIDI channels and to inter-route between other Signal Chains. It also creates multiple key splits from one multi-timbral instrument.
- *MIDI channel selector (Channel Out)* – Use this to create a MIDI channel output for the selected signal chain. This is used when working with multi-timbral synths and external MIDI devices.
- *Rename* – Renames the currently selected Signal Chain.
- *Delete* – Deletes the currently selected Signal Chain.

Stream Processor

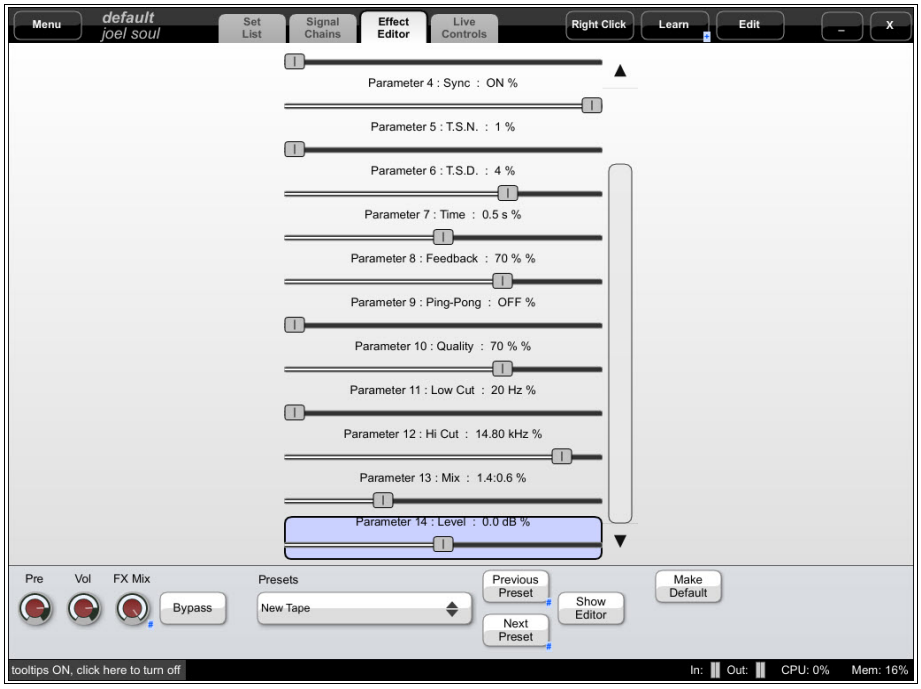
- Add Effect – Brings up the plug-in browser, where you can select an effect to add to the signal chain. Audio is processed from top to bottom, and effects can be switched around in order to change their routing. Typically, the first slot is used for an instrument, while the plug-ins that are below it affect its sound. The user can also load MIDI effect. These are plug-ins that output MIDI rather than audio. These are placed above the sound generating plug-in so that it can pass MIDI. Riff will not allow illogical configurations, (e.g. Sound generating plug-in, followed by a MIDI VST, followed by an effect followed by another sound generator).

Once you add an effect, you will get a new tab on the Title Bar called "*Effect Editor*".

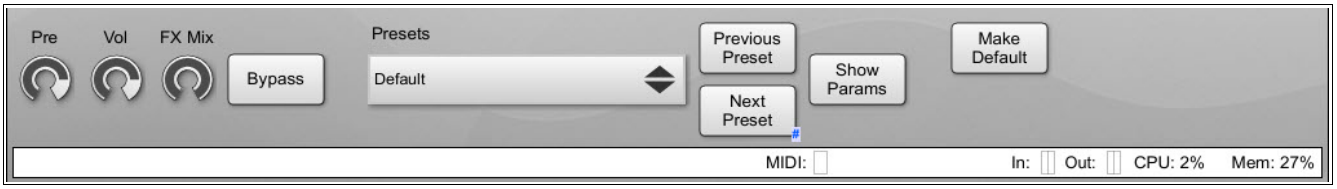
Selecting the “Instrument/Effects Editor” will bring up the GUI of that effect:



If an instrument or effect is selected, you can access the list of the parameters for the plug-in by clicking the “Show Params” tab on the Instrument Editor lower control screen. The last touched control on the plug-in will be highlighted on the Params page. This is helpful for locating hard-to-distinguish parameters.



This is a list of the raw parameters of an effect. This will also display parameters that might not be accessible from the plug-in GUI.



The lower section of the screen shows the controls for the Effect Column. Those controls are:

- *Pre – Pre control:* Controls the input into the selected Audio FX.
- *Vol* – Controls the output of the selected Audio FX.
- *FX Mix* – Controls the wet/dry mix.
- *Bypass* - Bypasses the output of the effect.
- *Presets* - A drop-down box that allows you to quickly browse through preset sounds. Displays the list of the Presets contained within a VST. Touching the Preset name will expand the list into a scrollable list where a new preset can be selected
- *Previous Preset* – Selects the previous preset in the list.
- *Next Preset* – Selects the next preset in the list.
- *Show Params* – Shows all the VST parameters without the GUI making it easier to find and change certain parameters. You can switch between a GUI and non-GUI.
- *Make Default* – Locks user created parameters as the default setting. This only applies to the selected Song. Loading another instance of the VSTi will not load with this as the default setting. The only exception to this of course is "Shared Instruments".

Instrument Editor

If an instrument or plug-in is selected, you can access the GUI by clicking the Instrument Editor tab on the Title Bar. Also, you can reach the Instrument Editor screen by double-clicking the instrument in the Instrument Rack or in the Signal Chain.

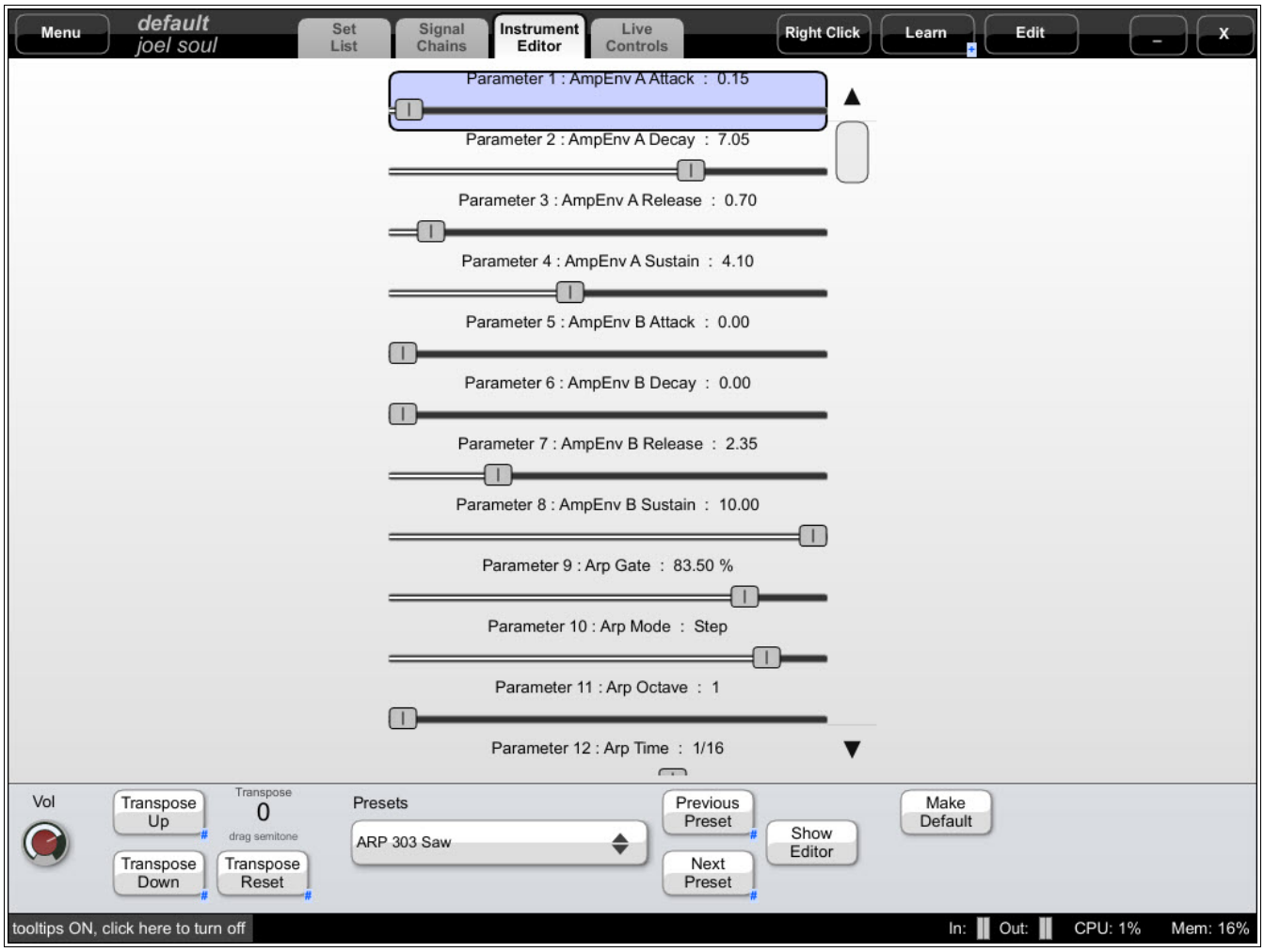


Instrument GUIs larger than 1021x727 will display horizontal and vertical scroll bars.

- *Vol* – The volume here basically controls the amount of output that is sent to the next effect in the Signal Chain. The volume is independent from the Track volume and the signal chain volume.
- *Transpose Up/Transpose Down* – Transposes notes sent by one octave up or one octave down.
- *Presets* – Displays the list of the Presets contained within the synth. Touching the Preset name will expand the list into a scrollable list where a new preset can be selected.
- *Prev Preset/Next Preset* – Navigates through the Preset list. Can be Learned or Relative Learned.
- *Show Params* – This button brings up the parameters view of the current plug-in. Clicking it again switches back to the plug-in's GUI. *Refer to the next page for an example.*
- Make Default - Locks user created parameters as the default setting for the selected song.

Instrument Parameters

If an instrument or plug-in is selected, you can access the list of the parameters for the plug-in by clicking the Show Params tab on the Instrument Editor lower control screen. The last touched control on the plug-in will be highlighted on the Params page. This is helpful for locating hard-to-distinguish parameters.



This is a list of the raw parameters of an Instrument. This will also display parameters that might not be accessible from the plug-in GUI. The Lower control section is the same as the Instrument Editor tab.

- *Show Editor* – Pressing this button will take you back to the Instrument Editor screen.

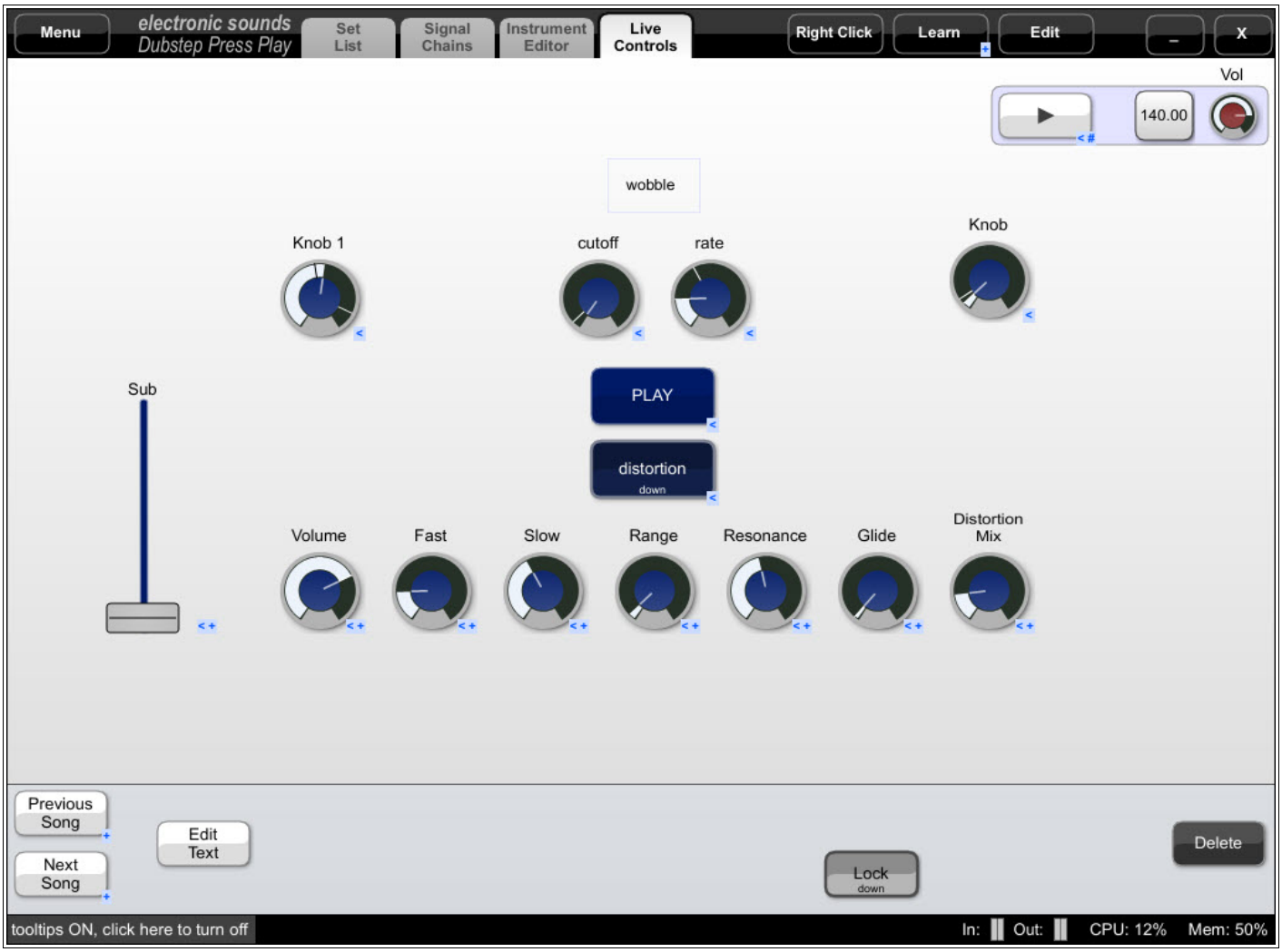
Exporting and Importing User Patches

If you tweak out a patch in an instrument, you can export that setting by going to *Menu>File>Export>Export FXP (Program)*. This also applies to Banks. Except you would pick *Export FXP(Bank)* instead. The import/export FXB/FXP function is only available when the instrument/effect editor tab is selected.

Live Controls

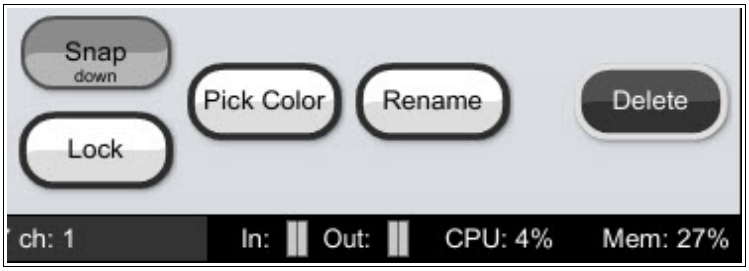
This is one of the most powerful features of Riff. On this screen you can build a custom set of MIDI controllers that can be linked to any Instrument, Effect, or Signal Chain Control contained within a Preset from the Song Grid. Live controls for a preset are saved within that preset and can be easily recalled.

The Edit Mode toggle button toggles the Live Control screen between Edit mode and Live mode. In Edit mode a user can add, remove, resize and edit the properties of the Live Control. Each Live Control can be mapped to a plug-in parameter, just like a real hardware control. In Live Mode a user cannot add, remove, resize and edit the properties of the Live Control.



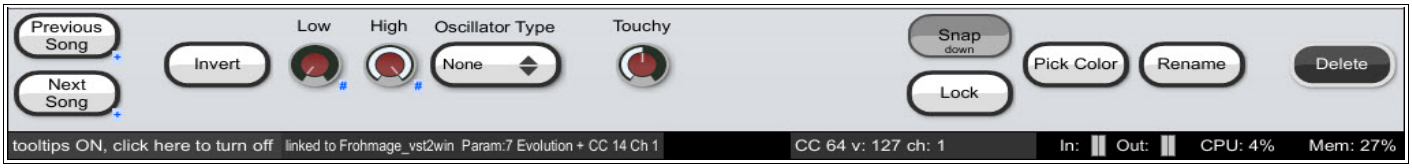
A left-click, right-click or a touch via the touchscreen on an empty space will bring up the "Add" menu. You can add a **Knob**, **Button, Note Button, Horizontal Slider**, **Vertical Slider**, **XY Pad, Crossfader,** and a **Text Editor**.

General Live Control Properties

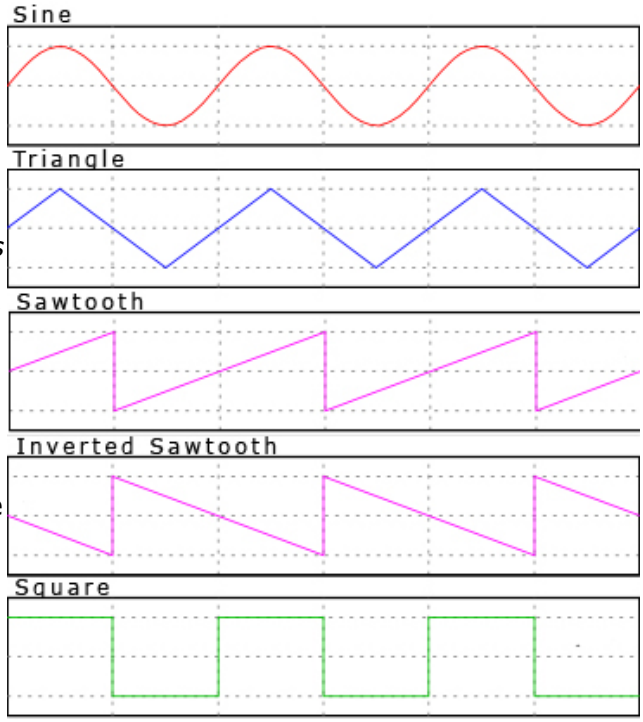


- *Snap* – Snap live controls to a grid
- *Lock* – This will lock down all the live controls making them operable. This is similar to live-mode but still allows you to edit the live controls, with the exception of resizing them.
- *Pick Color* – Brings up the Color Picker dialog box. Here, the user can select the color choice for the text.
- Delete – Deletes the currently selected Virtual Live Control.
- *Rename* – Renames the currently selected Live Control.

Knob Properties/ Horizontal Slider Properties/ Vertical Slider Properties



- *Invert* – This option allows you to invert the Virtual Live Control. This comes in handy when working with VST's that have reversed buttons to bypass.
- *Low* – Sets the low boundary of a knob or slider.
- *High* – Sets the high boundary of a knob or slider.
- *Oscillator Type* - The knob can be made to self oscillate. Below is a listing of the Oscillator types.



  - *Sine - A sine wave is often thought of as the most simple or pure wave because it possess only one harmonic and is a smooth wave.*
  - *Triangle – A triangle waveform that rises quickly to a particular amplitude, then quickly drops.*
  - *Saw - The sawtooth wave (or saw wave) is named a sawtooth based on its resemblance to the teeth on the blade of a saw. Sawtooth waves ramps upward as time goes by and then sharply drops. However, there are also sawtooth waves in which the wave ramps downward and then sharply rises.*
  - *Inverted Saw* – Inverts the Sawtooth wave type. Inverted Saw *waves ramps downward and then sharply rises.*
  - *Square - The square wave alternates quickly between two levels.*
  - *Random - This randomly selects oscillator parameter values.*

- *Time* – Choose the rate of oscillation by adjusting this knob.
- *Offset* - Offsets the starting point of the oscillation dependent on the clock of the host. This is used to make knobs/sliders hit their high/low ranges on different beats.
- *Use BPM* - This button allows you to sync the knob/slider with Riff's BPM.
- *Use Sequence* - You can latch the oscillators to start with the play button of the host and any other sequencer that has been synced to the host. When you hit play the oscillators start oscillating along with a sequencer (if you have put it to sync with the host). *This only applies to the selected live control.*

Button Properties



- *Button Type* - The options are Toggle (On/Off) and Momentary(Trigger)
    - *Toggle* – When set to toggle, a button remains on until you press it again. Hence On/Off.
    - *Momentary* – When set to Momentary, a button triggers when pushed and releases the trigger when released.
- *Invert* – This option allows you to invert the button function.

XY Pad Properties



- *Low X/ High X* – Sets the low and high ranges for the X axis on the XY pad.
- *High Y/ High Y* – Sets the low and high ranges for the Y axis on the XY pad.

Crossfader



- *Low A/High A* - Sets the low and high range for side A of the Crossfader.
- *Low B/ High B* - Sets the low and high range for side B of the Crossfader.

Text Properties

Creates a box the user can use for typing notes, text or labeling in. For instance, the name of the song or special tricks for that individual song.

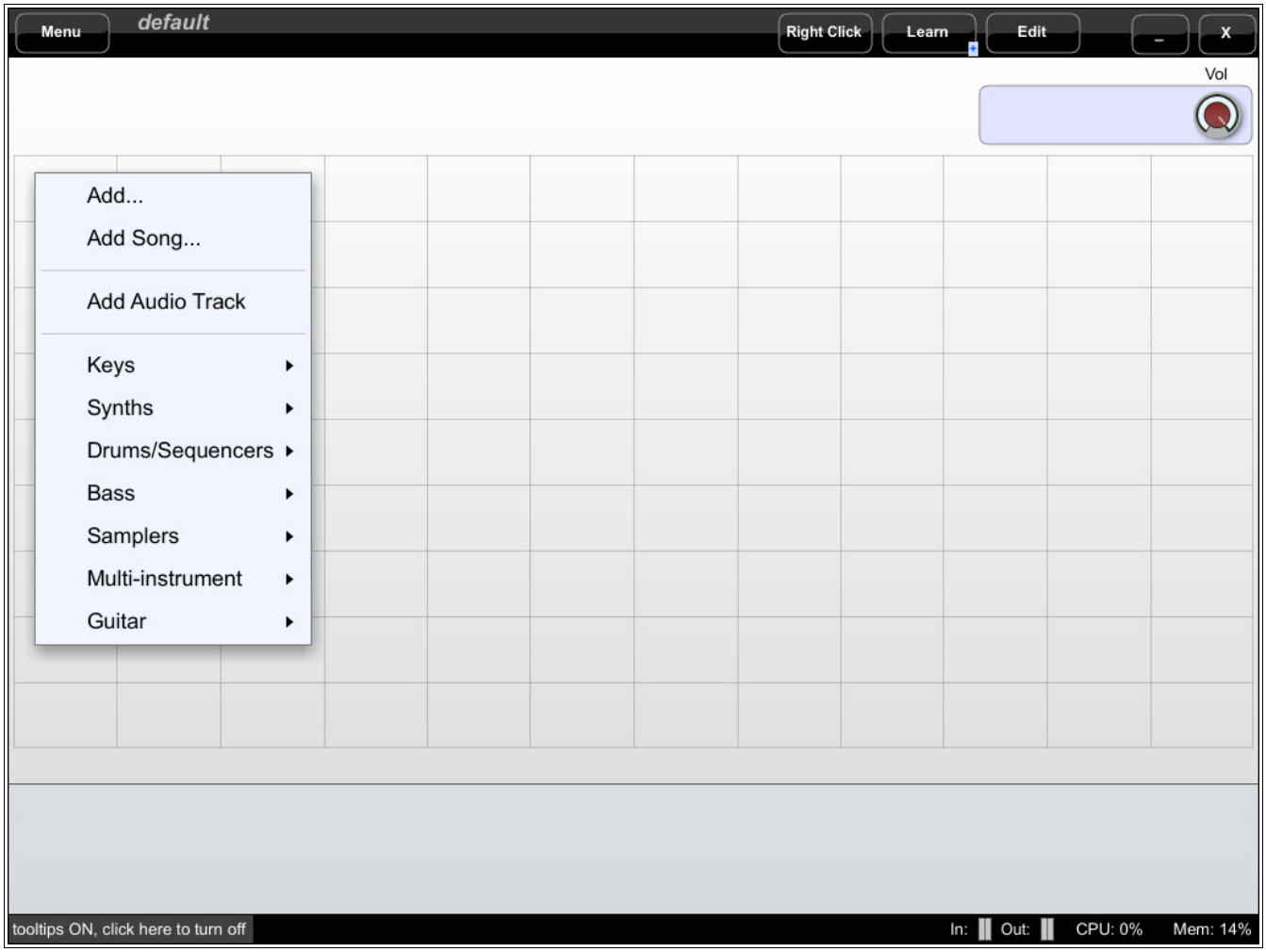


- *Edit Text* – This allows you to input the actual text to be displayed when in live mode.

Using the Song Grid

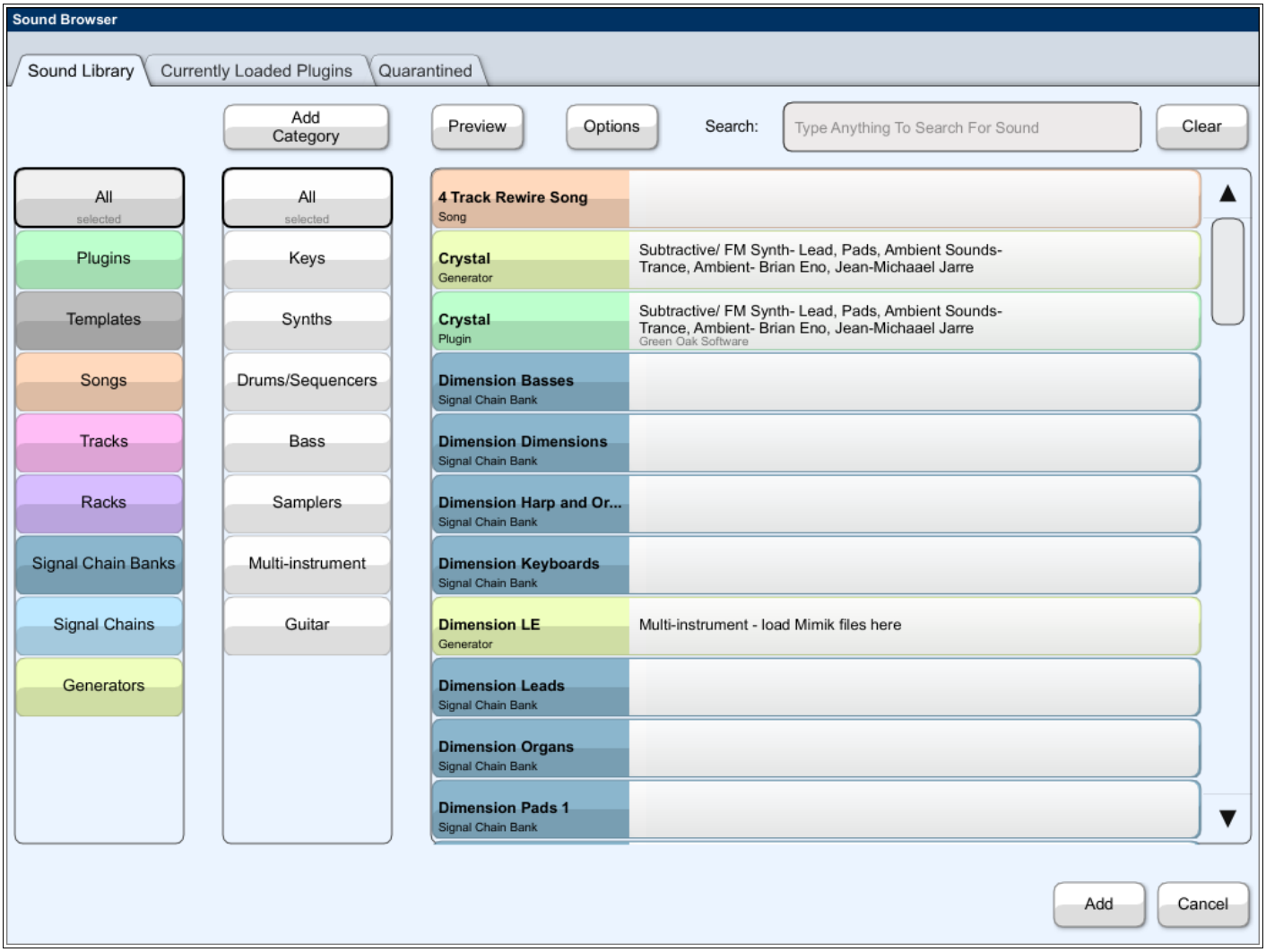
Creating a New Set and Adding an Instrument

To create a new Song; First, right-click in a grid square and select “Add”.



After right-clicking, you can choose from “Add...”, "Add Song..", "Add Audio Track", or you can choose from an instrument category. The instrument categories are: KEYS, SYNTHS, DRUMS&SEQUENCERS, BASS, SAMPLERS, MULTI-INSTRUMENT, and GUITAR. *Refer to Appendix B for a list of instruments.*

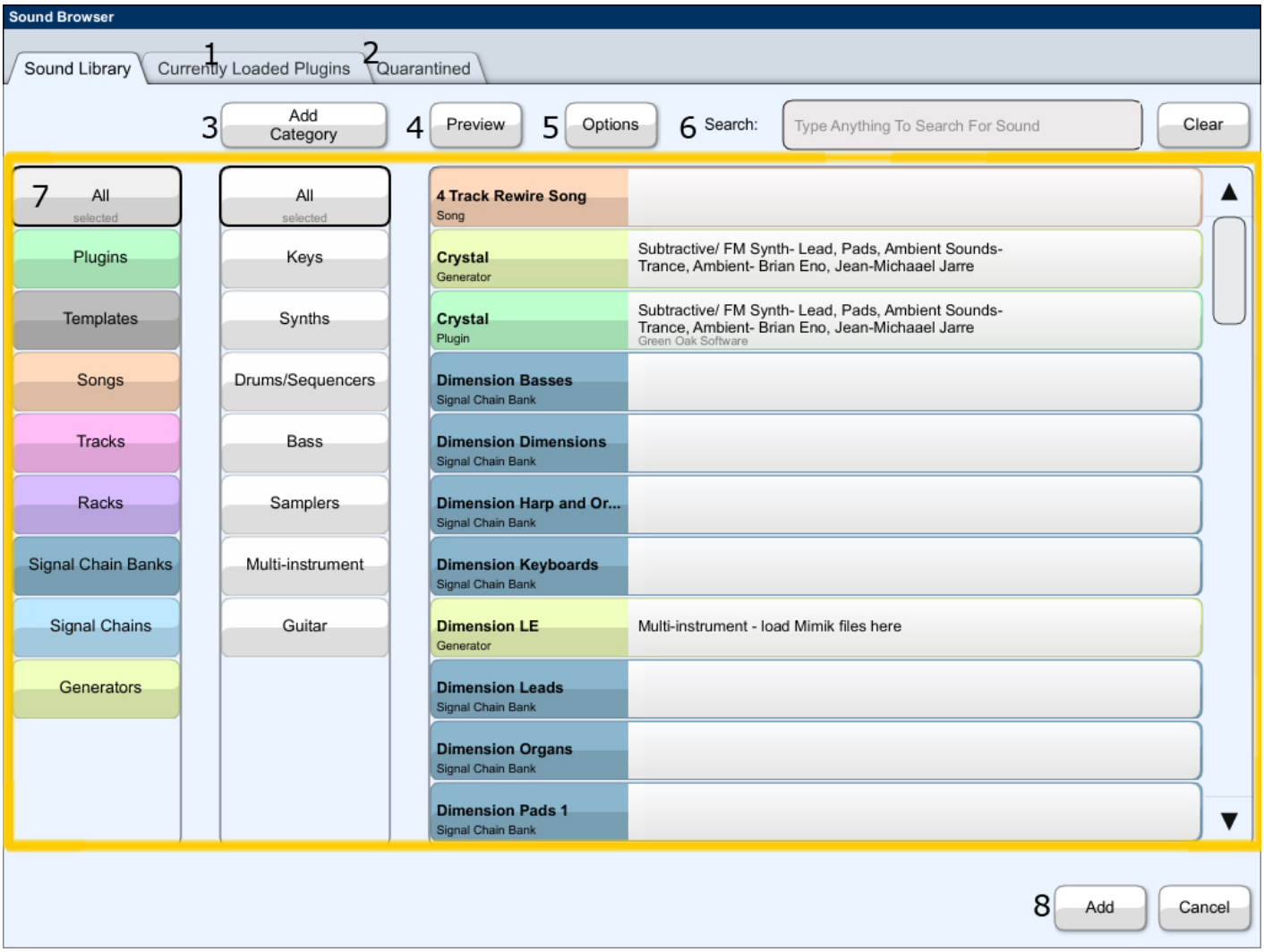
If you click “Add” you will see the Sound Browser. Pick an instrument from the list and press the [Add] button. This will insert this instrument into the Song grid for you to play.



The Sound Library is categorized by All, Plugins, Templates, Songs, Tracks, Racks, Signal Chain Banks, Signal Chains, and Generators.

About the Sound Browser

The following will give you a run down of the Sound Browser.



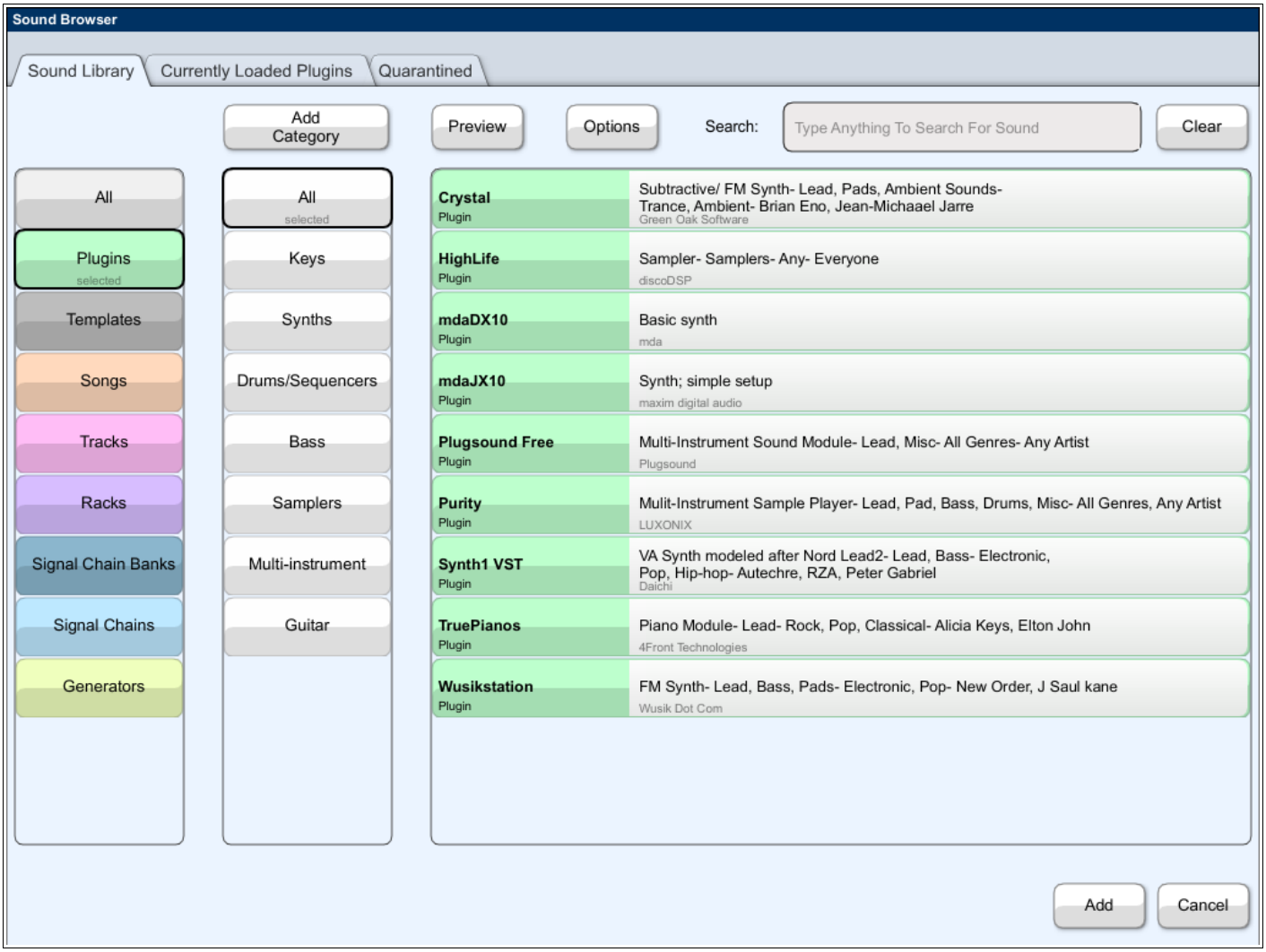
1. *Currently Loaded Plugins* – Use this to load a sound that is already loaded in another Song. This is how you can "share" instruments between songs.
2. *Quarantined* – If an instrument does not work properly with Riff, you can right-click it and have it sent to the quarantine list.
3. *Add Category* – Adds a new custom category to the category column. Instruments can be drag -n- dropped into a category. A Category can also be renamed to describe the type of instruments it contains. Once a certain number of catergories have been added the menu box will be full and not allow another categorie.
4. *Preview* – Allows you to preview certain instruments before adding to a song. Refer to the next section on how to use this feature.
5. *Options* – This launches Riff's options page as detailed in Chapter 2 “Plug-in Options”.
6. *Search* – Type in the name of an instrument on the keyboard to locate it. Use the [Clear] button.
7. *Categories* – Instruments are arranged by their category. Use this as a way to filter through various templates/songs/sounds.
8. *Add/Cancel* – When you select an instrument in the list, press the [Add] button to add it to the song. Selecting [Cancel] while in the browser, will exit the browser.

Previewing an Instrument

While in the sound browser, some VST's allow you to use the preview feature to audition any available presets. Songs and effects can also be previewed with this feature.

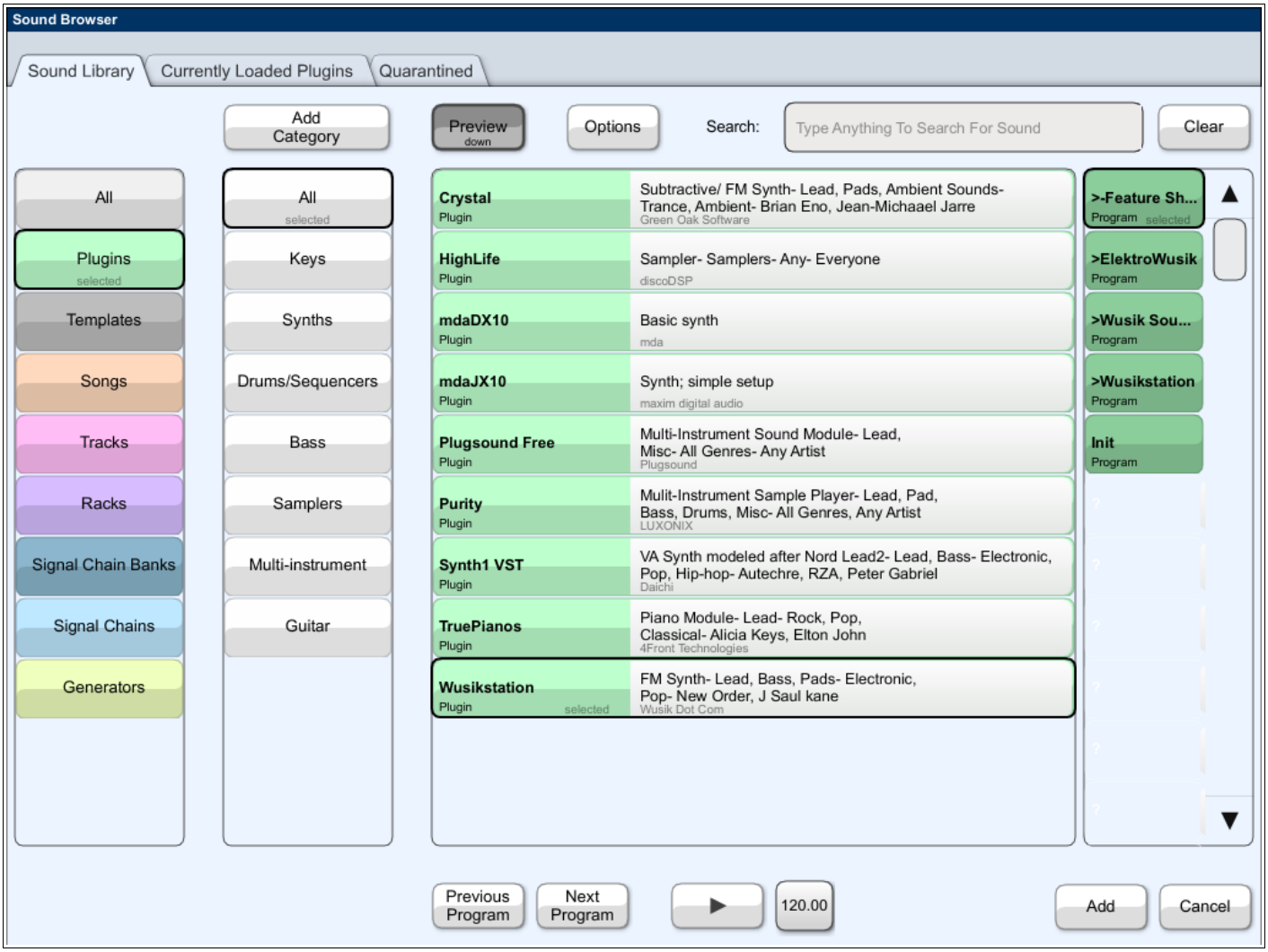
Let's try this using the instrument called "*Wusikstation".*

Click on an open Song from the Setlist and press the [Add] option. This takes you to the Sound Browser. Under the Sound Library Tab, select [Plugins] on the left column. Then, select Crystal from the VST instrument list.



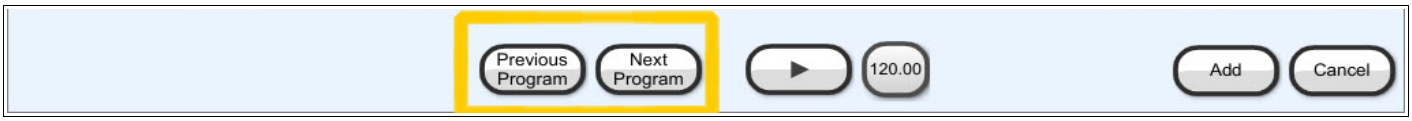
Now press the press the [Preview] button at the top the screen.

Once the [Preview] button has been pressed, new options open up to the right side of the screen. This list shows you the available presets for the selected instrument.



Use the [Previous Program] and [Next Program] buttons to scroll though the presets of the select instrument, or use the touchscreen to select a preset for previewing.

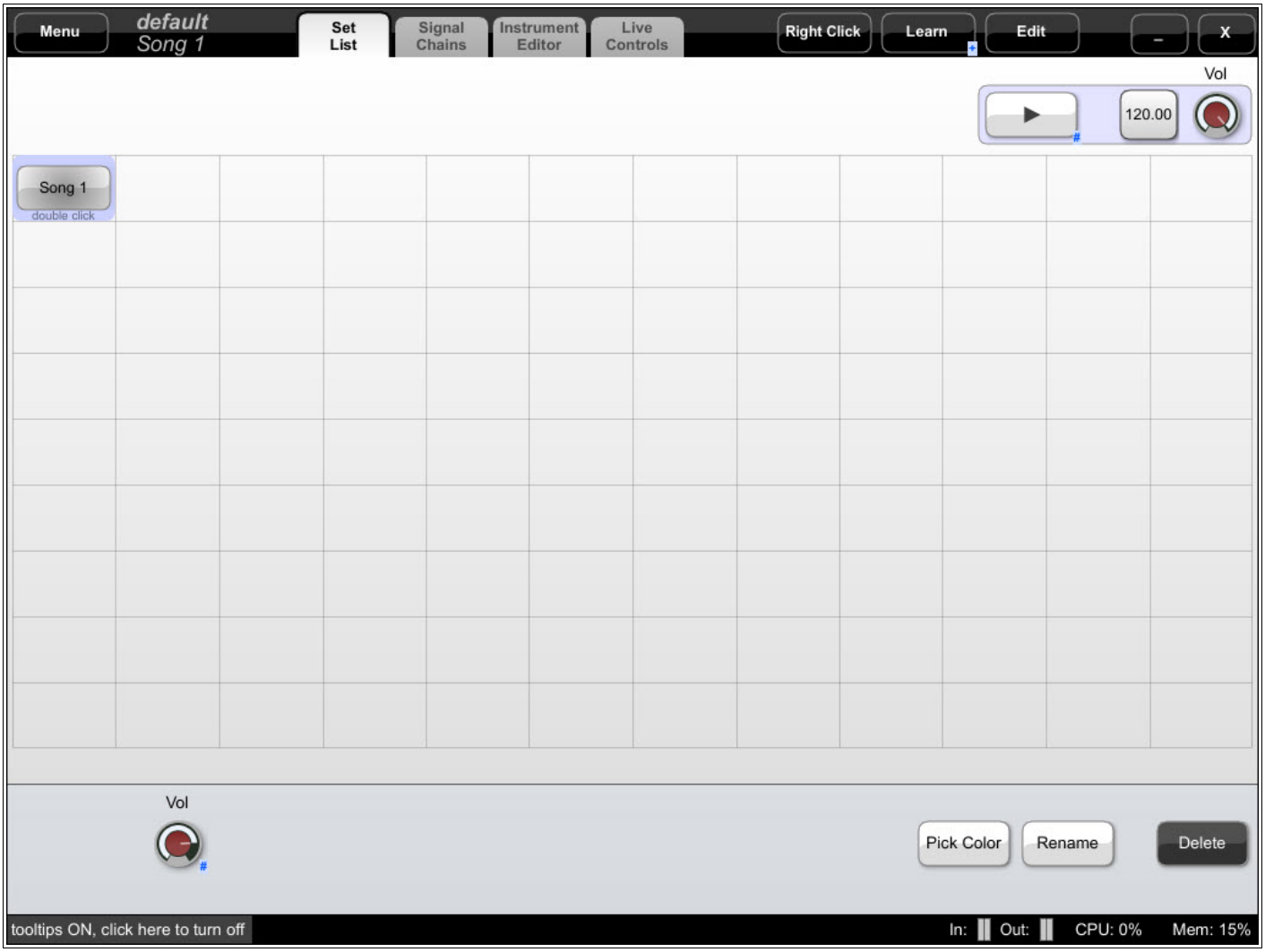
Also, there is a [Play] button. This is used to start any plug-in that has a sequencer.



To end previewing, press the [Preview] button, this will take you out of the preview mode. Once you find the sound you like press [Add] and the instrument will be added to the currently selected Song.

Note: Not all instruments support the Preview Feature.

Once an instrument is added to the Setlist, this is what you will see. Please refer to Chapter 3 for an explanation of this screen.

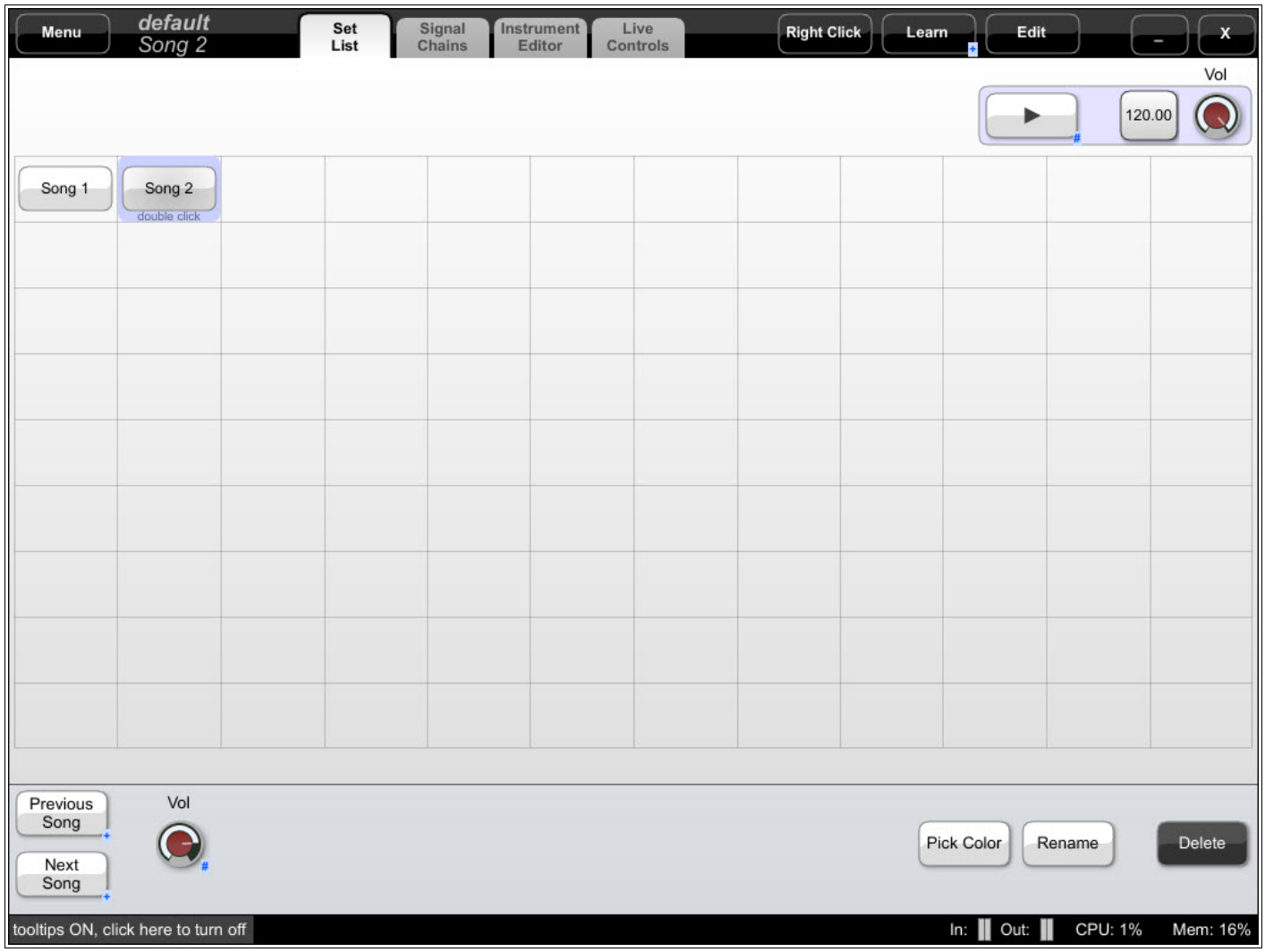


You can now play this sound with the ivory keys.

If you want to adjust the settings of the instrument in the first song, click the [Instrument Editor] tab, which is located on the Title Bar.

If you add another Song to the Set List, the [Previous Song] and [Next Song] buttons appear.

These two buttons will be in the lower left corner of the screen and will allow you to move to the next song in the Set List.



Saving a Set

To save the current Set List, go to Menu>File>Save Set As....
When the window pops up, type in a name and hit [OK].

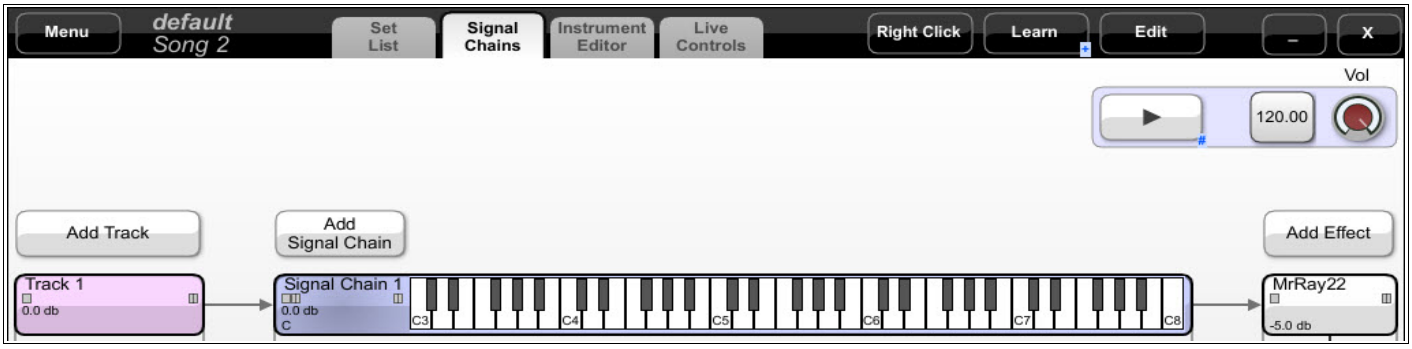
Rename a Song

Right-click a Song, then choose "Rename" and finally type in a name.

Using the Signal Chain

Adding an Additional Instrument to a Song

As stated in Chapter 3, a Song can contain multiple instruments. To add multiple instruments to one Song, select the Signal Chain Tab at the top of the screen.

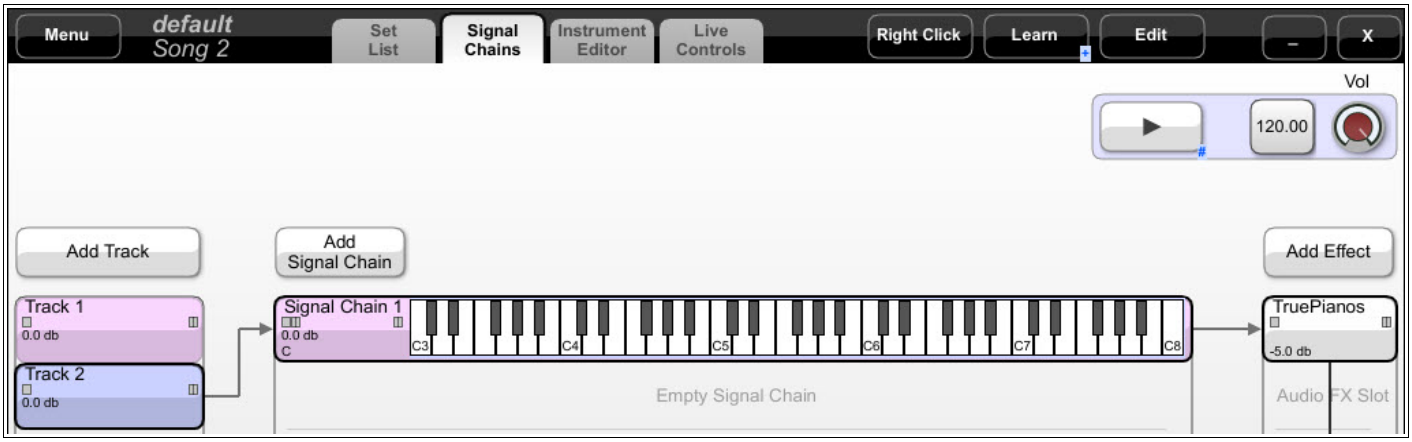


[Add Track] - This allows for a Track to be created with a signal chain containing a VSTi, or an empty signal chain with no VST. In order to use the VSTi's and play them, a Track must be selected with the desired plug-in.

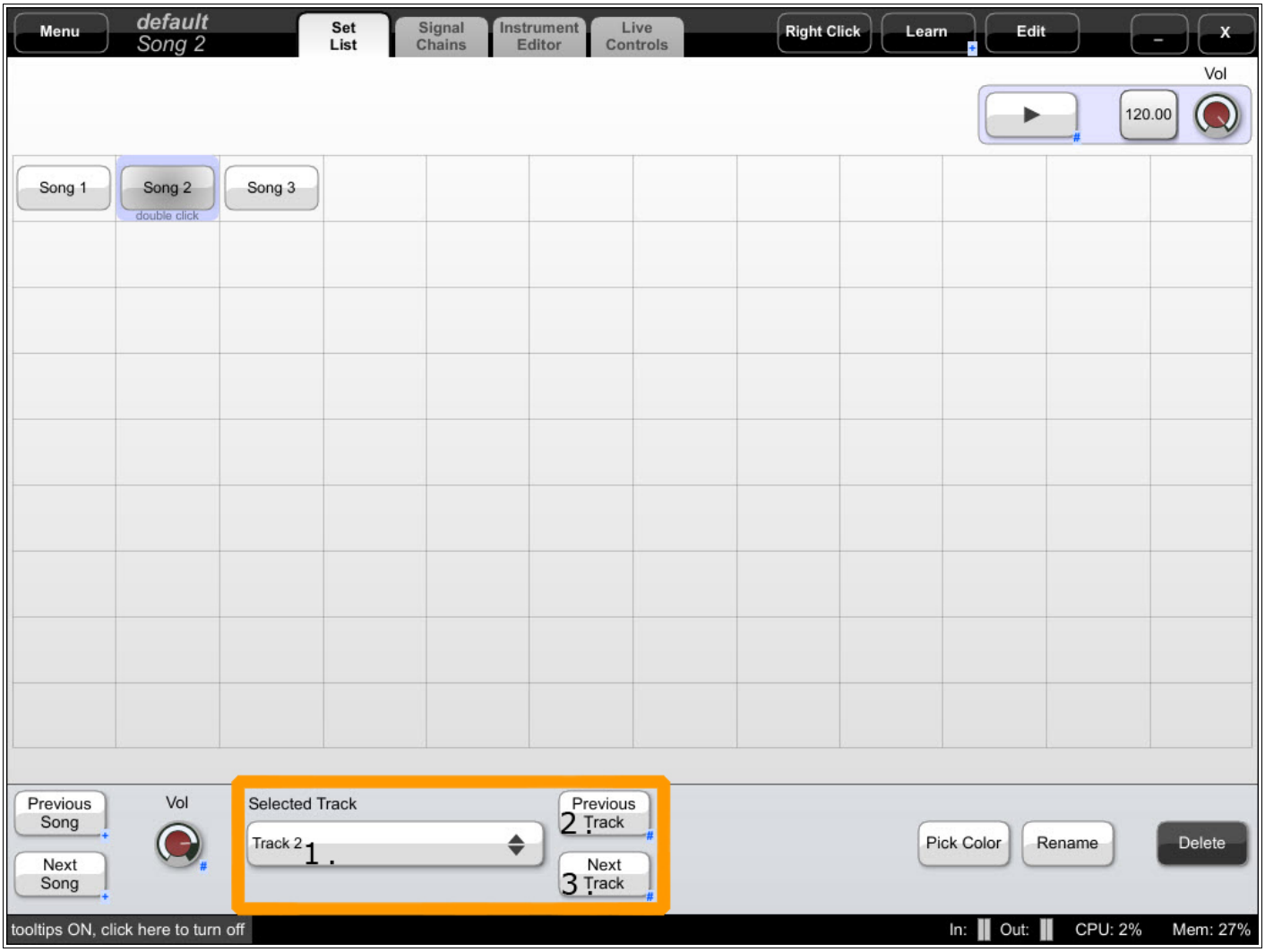
[Add Signal Chain] – Use this button to create a new signal chain with a VSTi, or create an empty signal chain for unique key splits. Touching the screen on a blank signal chain will give you the same add option.

To add that additional sound to this song go ahead and press the [Add Input] button. Again, you can decide to pick a sound by the category, or by using the Sound Browser.

The next image shows you that there are now two inputs in this song. Clicking on each input will play (solo) that sound.



When more than one Track has been added to the Signal Chains page, new options become available on the Setlist.



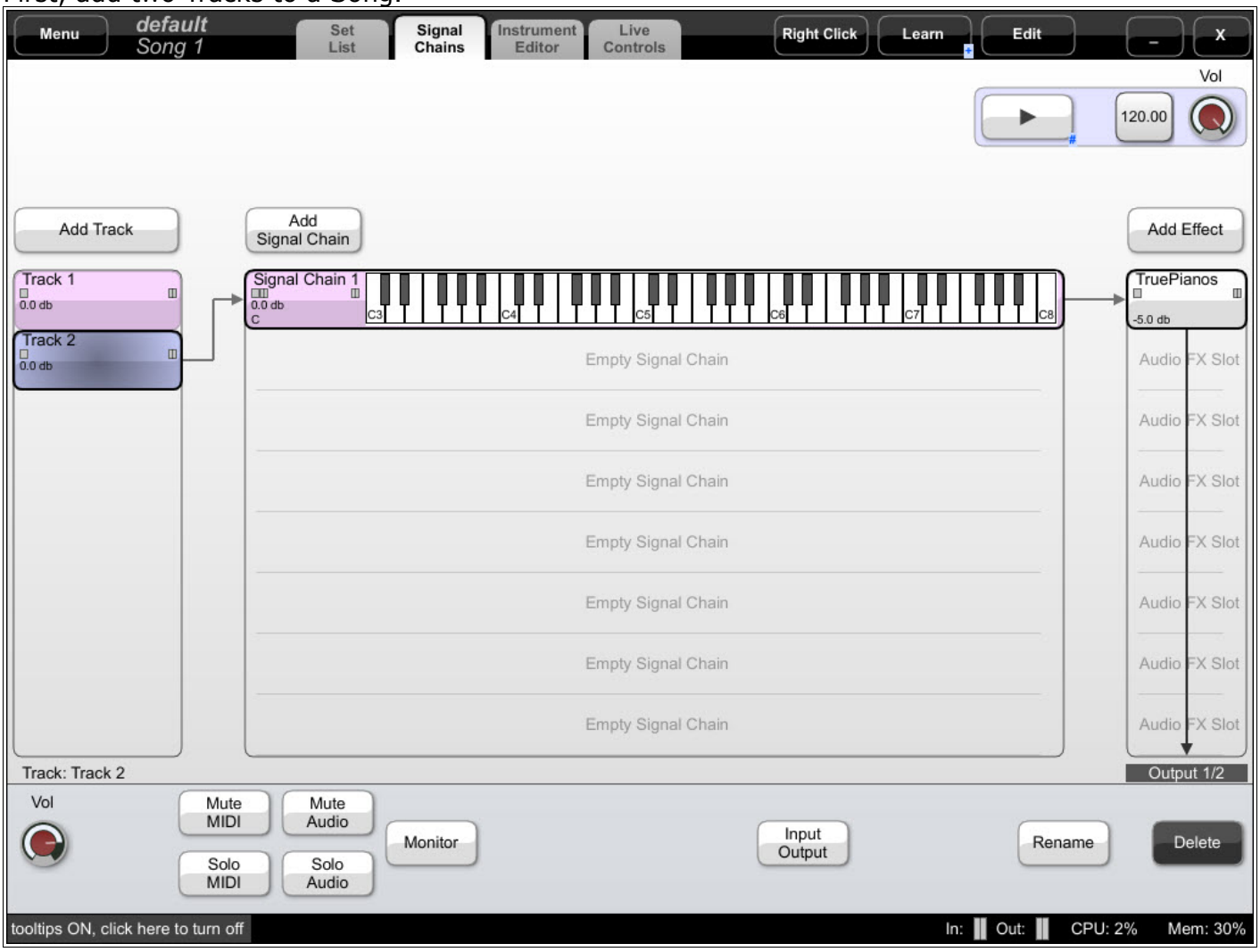
These new controls allow you to scroll the previous and next instrument in the Input Chain of the selected Song.

1. Shows which Track is selected.
2. A MIDI assignable button that allows you to select the previous Track of the selected Song.
3. A MIDI assignable button that allows you to select the next Track of the selected Song.

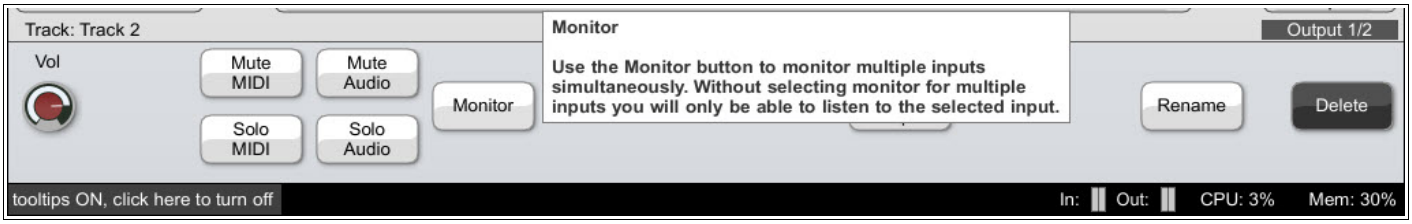
Monitoring Separate Tracks

Riff gives you the ability to monitor mutilple Tracks at the same time. Normally, when you select a Track, it mutes the other Tracks and only plays the one you have selected. The following directions will show you how to monitor multiple Tracks.

First, add two Tracks to a Song.

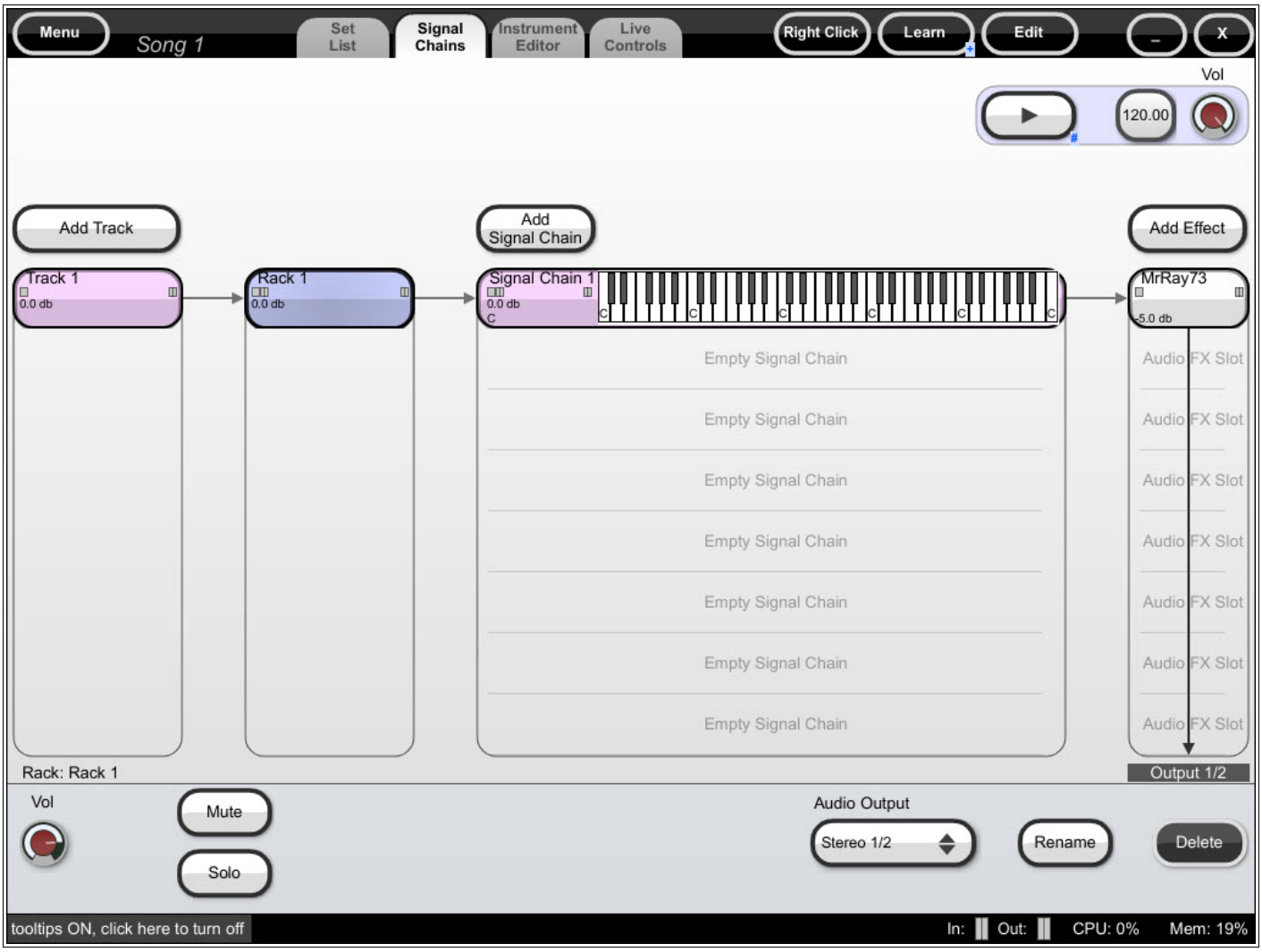


If you want to hear the second Track with the first Track, select Track 2 then press the [Monitor] button at the bottom of the screen.

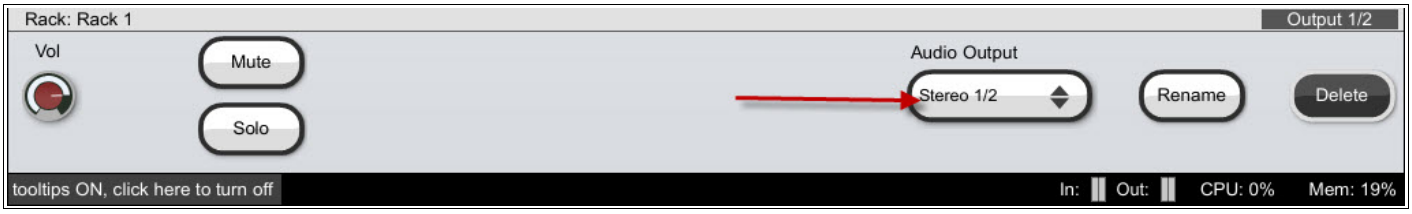


The Signal Chain Track Rack (Send Audio to Seperate Outputs)

Double-clicking on a Track will show the Rack. Also by pressing the small grey box at the bottom of the effect column will reveal rack parameters. You can use the Rack to assign a Track to a particular audio output. By default, audio is set to the main one and two outputs.



Selecting the Rack will present the "Audio Output" menu in the lower controls. You can Also Mute, Solo and Rename the Rack.

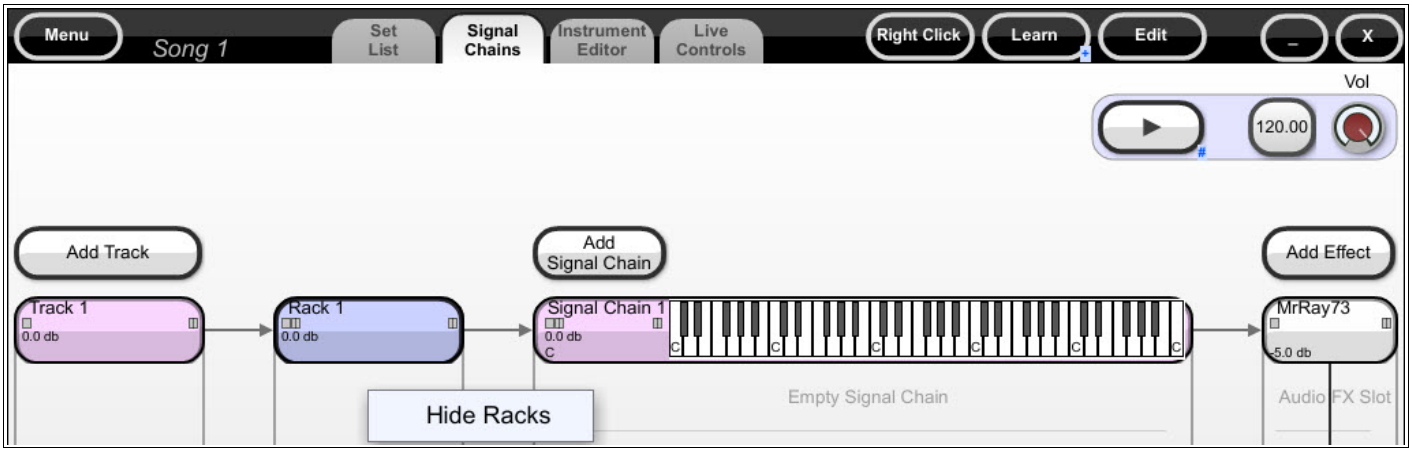


Be sure your audio options are properly setup. The image on the next page will show you what a properly configured options menu looks like.

If your audio outputs aren't assigned, go ahead and do that. Each output option should be assigned to an output buss.

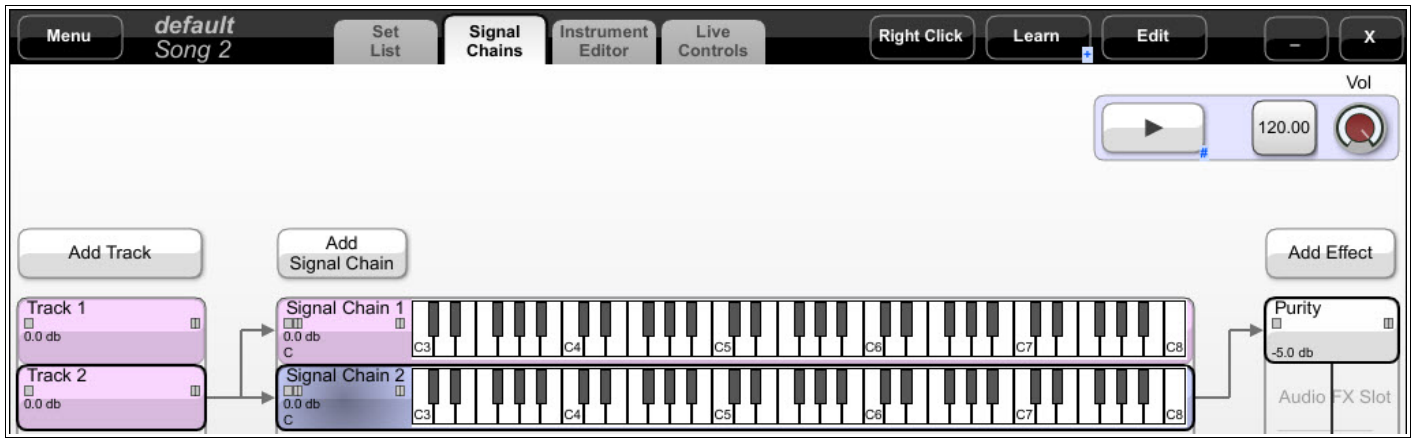


To hide the Rack, either right-click and select "Hide Rack", or just double-click on the rack to hide it.



Creating Keyboard Splits in a Song (Signal Chain)

Creating keyboard splits are now easier than ever! Select the input you want to create a split on. Now click the [Add Signal Chain] button. Select a new instrument. For this tutorial, True Piano is in the first signal chain, and Purity is in the second.



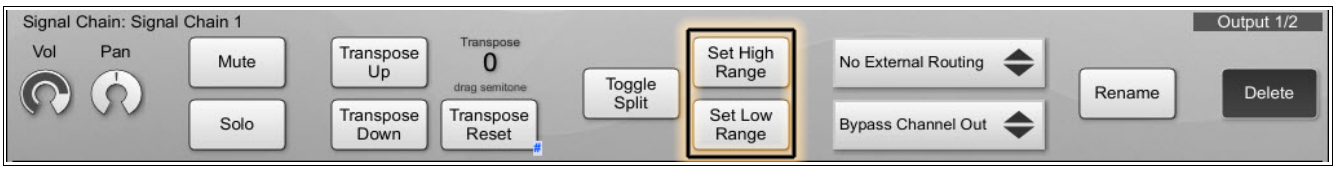
The above image shows that the second input now has two items in the Signal Chain. To adjust the settings of a Signal chain instrument, select it in the signal chain then either select the [Instrument Editor] tab, or double left-click the instrument to the right in the "Stream Processor" section. *Refer to Chapter 3.* Pick a sound for each Signal Chain instrument.

There are a couple ways to create splits. You can use the mouse and arrow pointer to create the split. Or you can use the touch screen. Select the signal chain instrument that you want to split and either touch screen/left-click and drag from the right or left of the keyboard. While you are dragging, the ranges will split. The other way is to use the "Set Range" buttons.

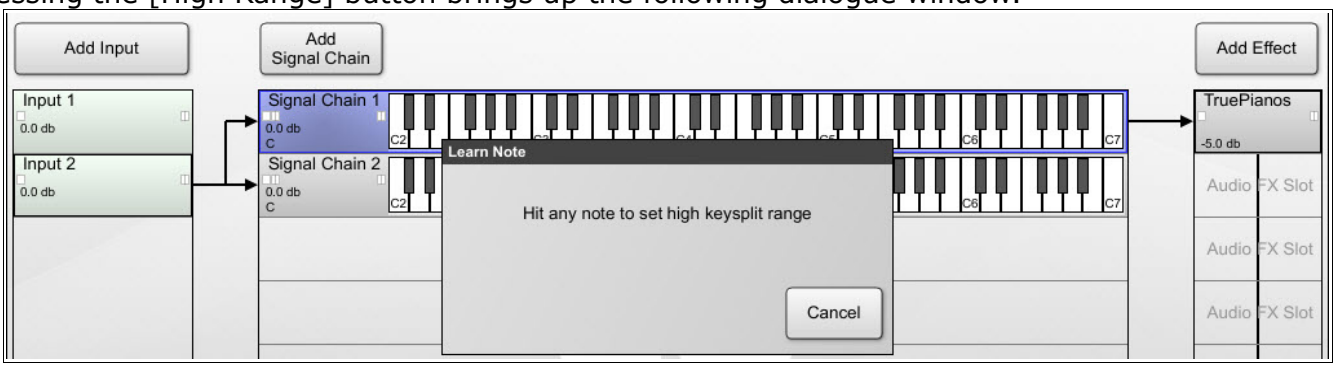
If you are not satisfied with a split and want to revert, select the Signal Chain instrument and press the [Toggle Split] button. This will spread the keyboard back to full range with no split.

Now let's create that split using the "Set Range" buttons:

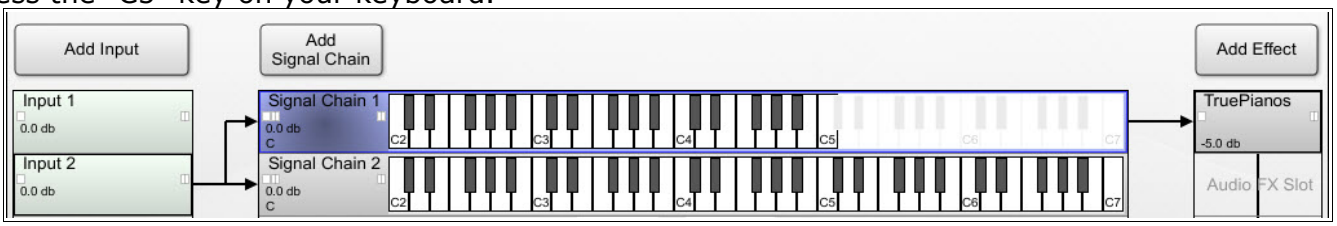
Select the first signal chain. Press the [Set High Range] or [Low Range] buttons. These are located at the bottom of the screen.



Pressing the [High Range] button brings up the following dialogue window.

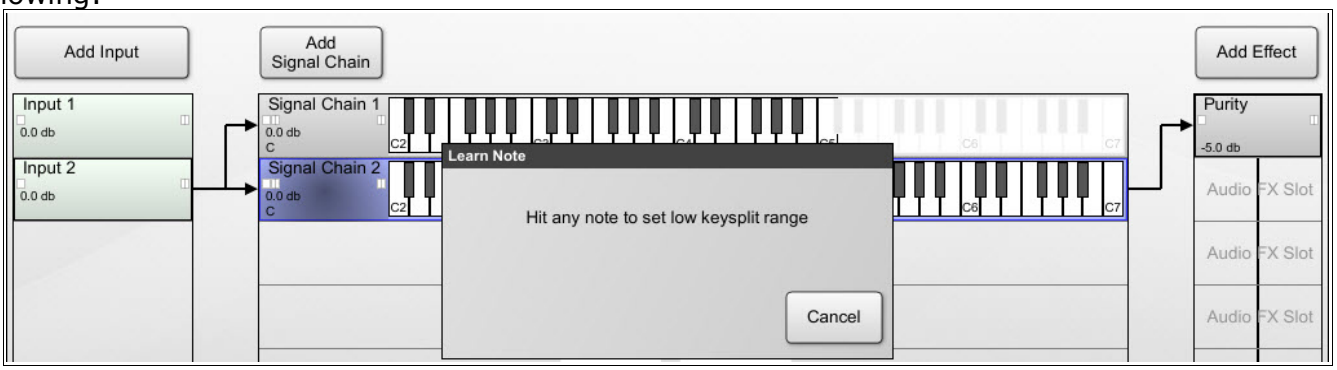


Press the “C5” key on your keyboard.

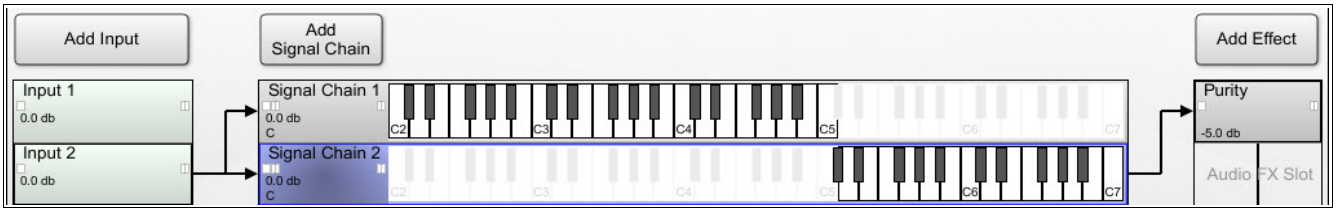


All notes above “C5” are now split off.

Now, select the second Signal Chain. And press the [Set Low Range] button. You will see the following:



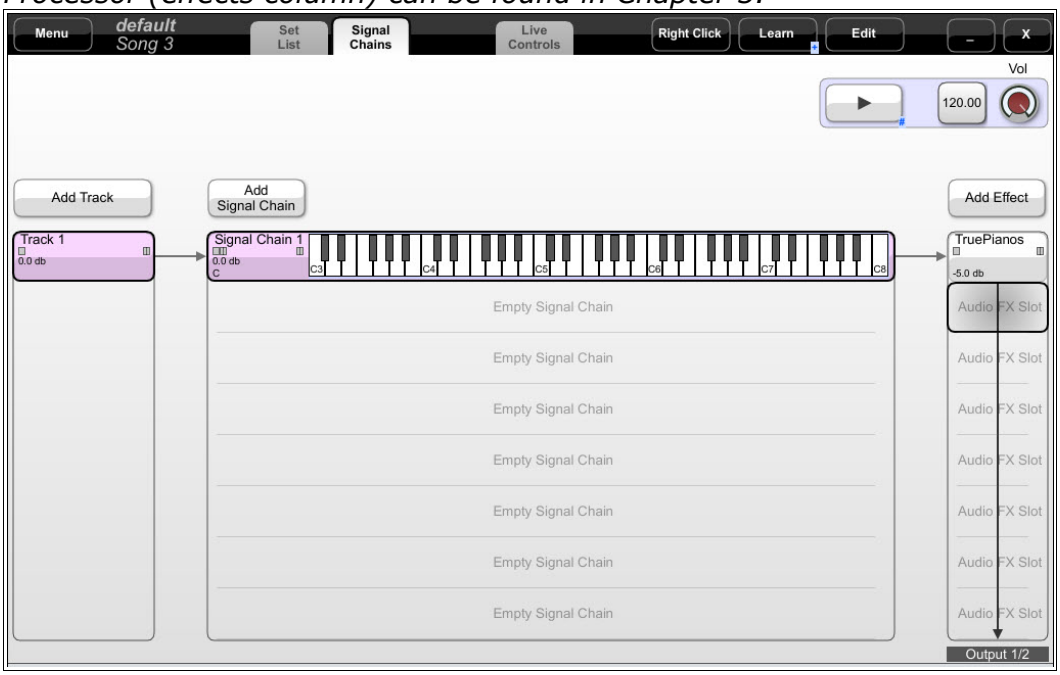
Press the “C#5” key on your keyboard.



As you can see you now have the lower section of True Pianos split from C5 down, and Purity split from “C#5” up.

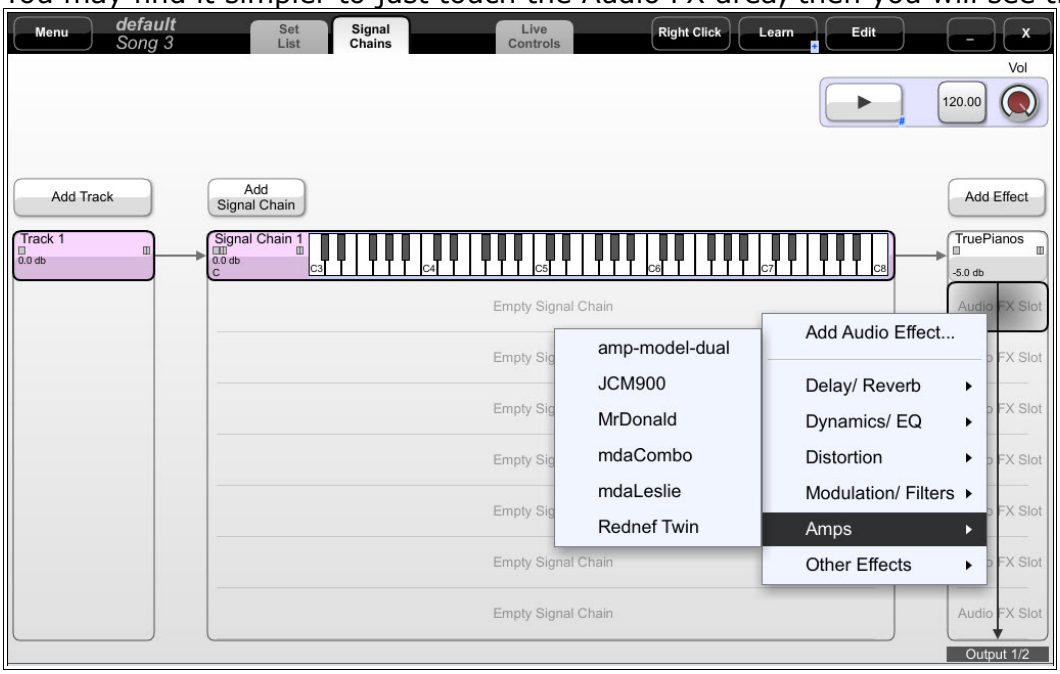
Adding Effects to an Input or Signal Chain

You can add a VST effect or MIDI effect to the desired signal chain. A maximum of seven audios, or seven MIDI effects can be chosen per signal chain. *More information regarding the Stream Processor (effects column) can be found in Chapter 3.*



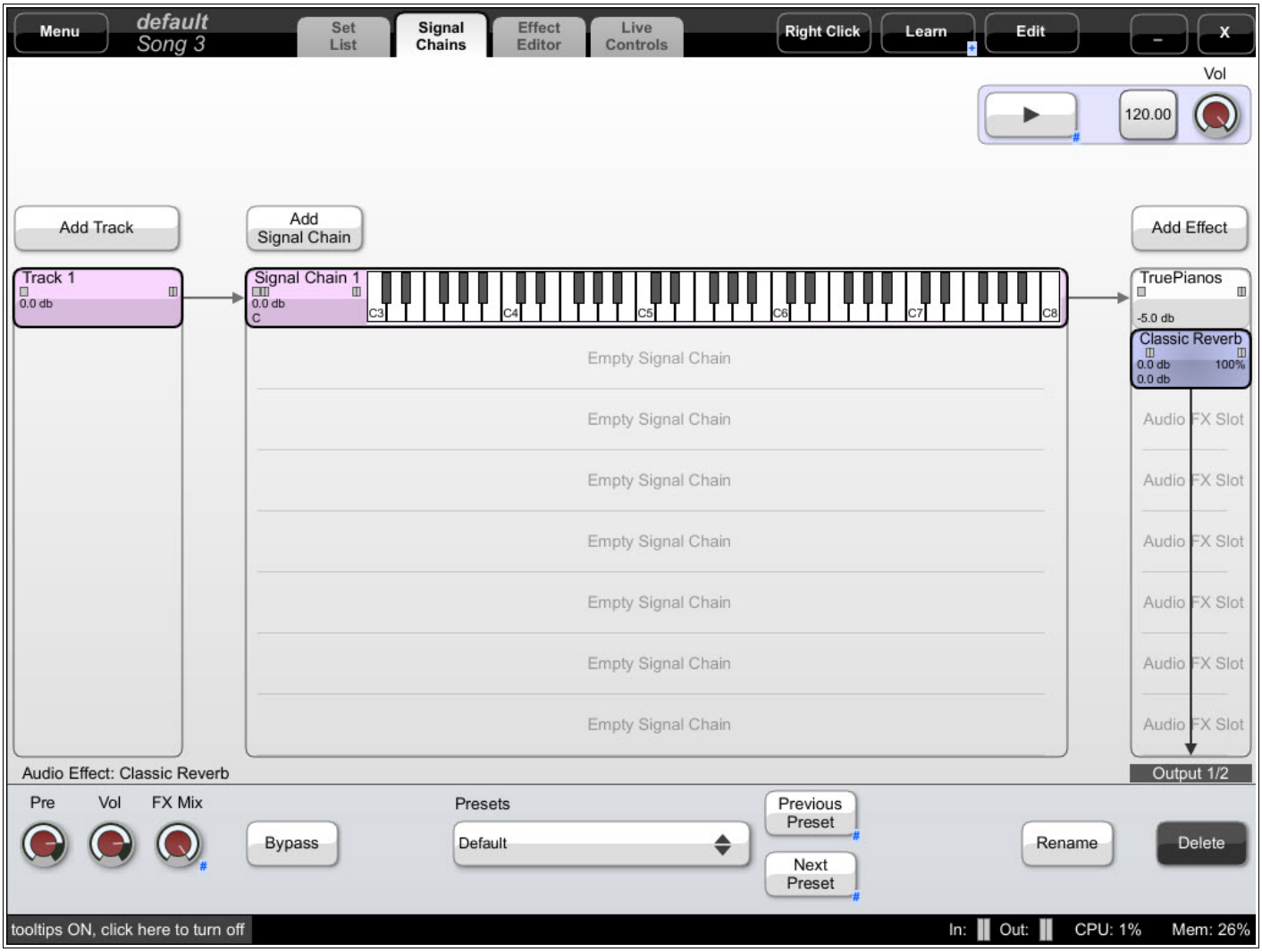
Audio FX Slot: This is where all the audio effects are stored for the chosen signal chain. To add an effect, use the [Add Effect] button located above or simply touch the screen on a blank slot under. Pressing [Add Effect] will bring up an effects browser very similar in workings as the Sound Browser.

You may find it simpler to just touch the Audio FX area, then you will see the following:

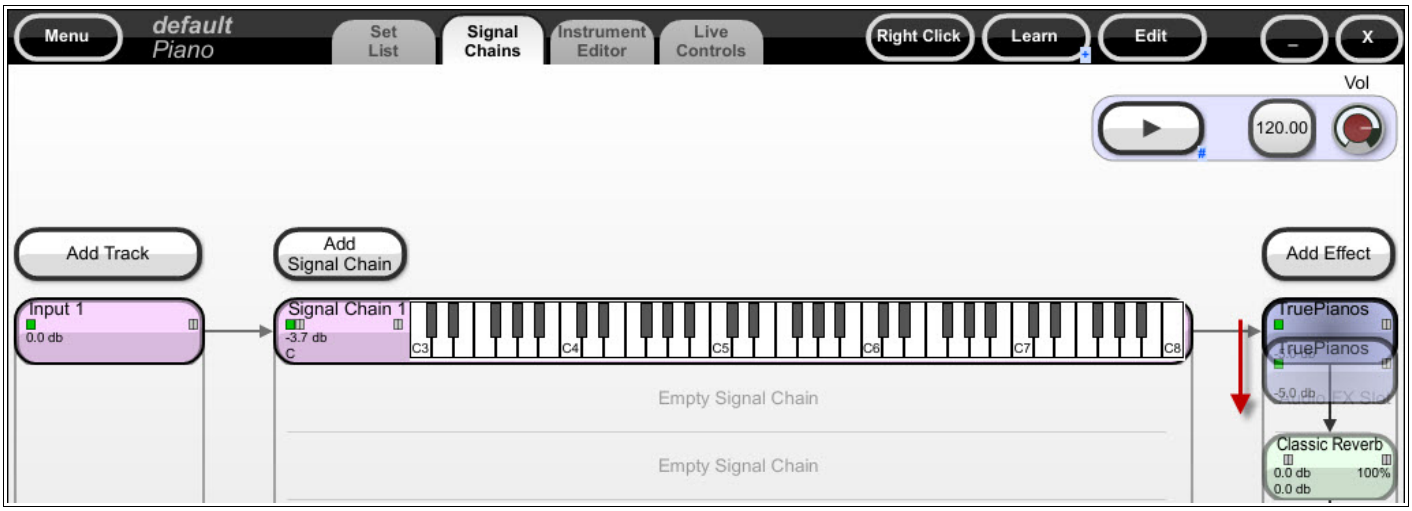


You can choose an effect by the category.

The next screen shot shows that there is an effect (in the Stream Processor) under True Pianos. When the effect is selected the “Effect Editor” will be present in the title bar.

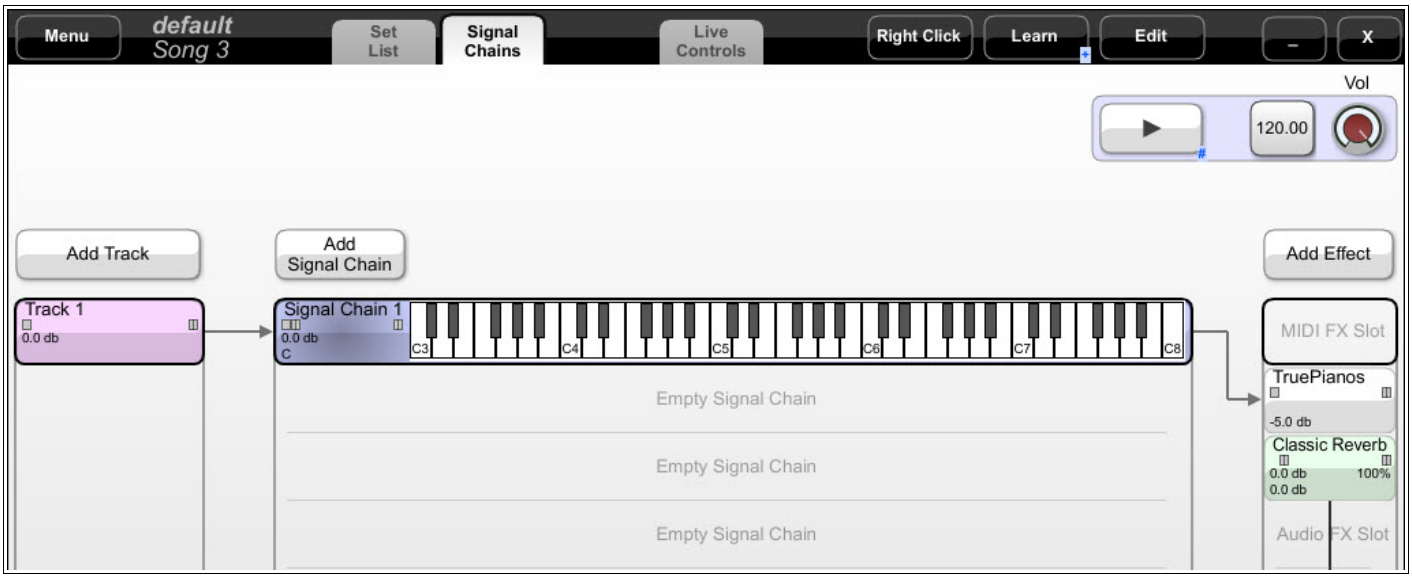


Effects can be reorganized by dragging up/down. Simply grab it and drag it up or down.

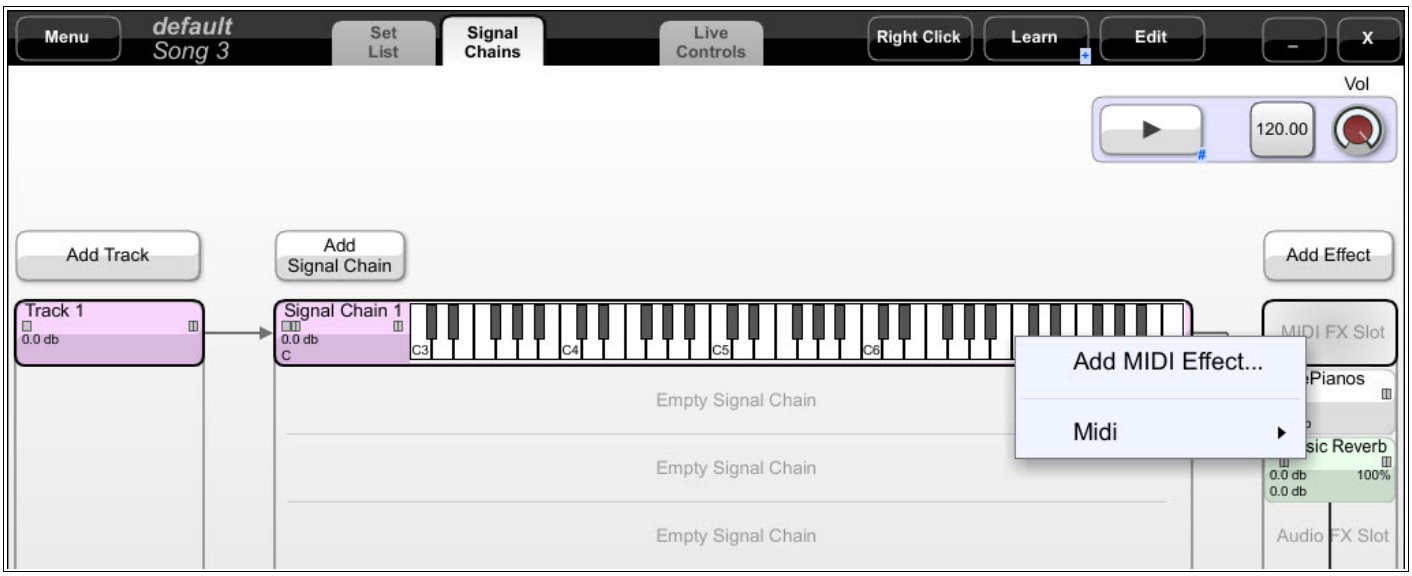


Adding MIDI FX

SoundSlate does not ship with any MIDI FX. However if you have any, left-click and drag the VSTi down by one slot, and this will open up the MIDI FX slot. If there is already an instrument and an audio effect already present, you will have to move each effect down one slot to open up the MIDI FX slot.



Right-click the "MIDI FX Slot" and select [Add MIDI Effect..].

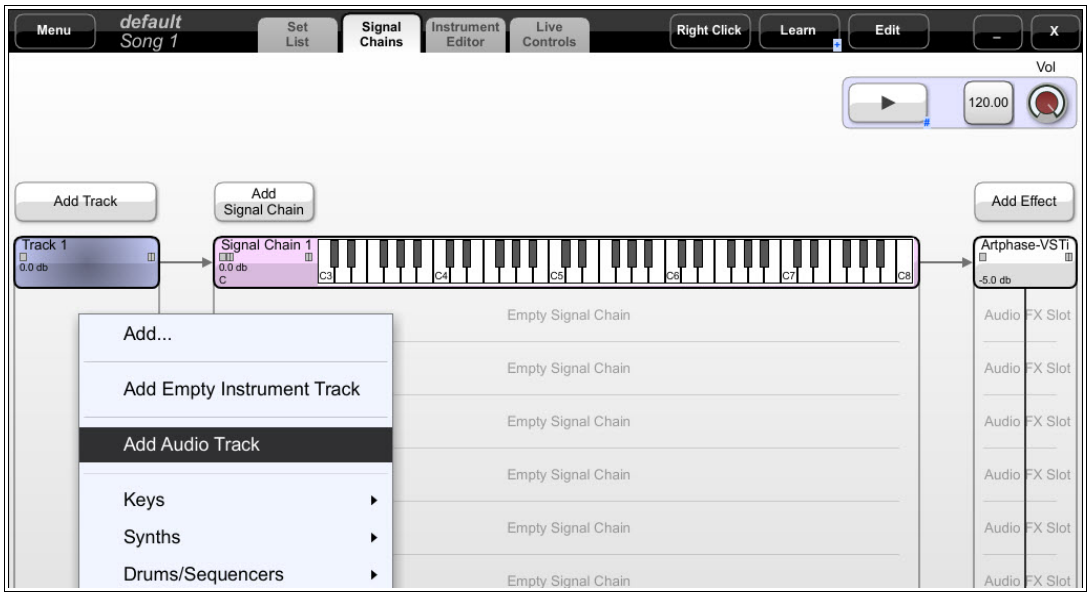


A screen similar to the sound browser will appear, here you choose the MIDI effect. The type of MIDI effects you can expect to find here in Riff are arpeggiators, MIDI note repeaters, basic MIDI step sequencers, humanizers, etc.

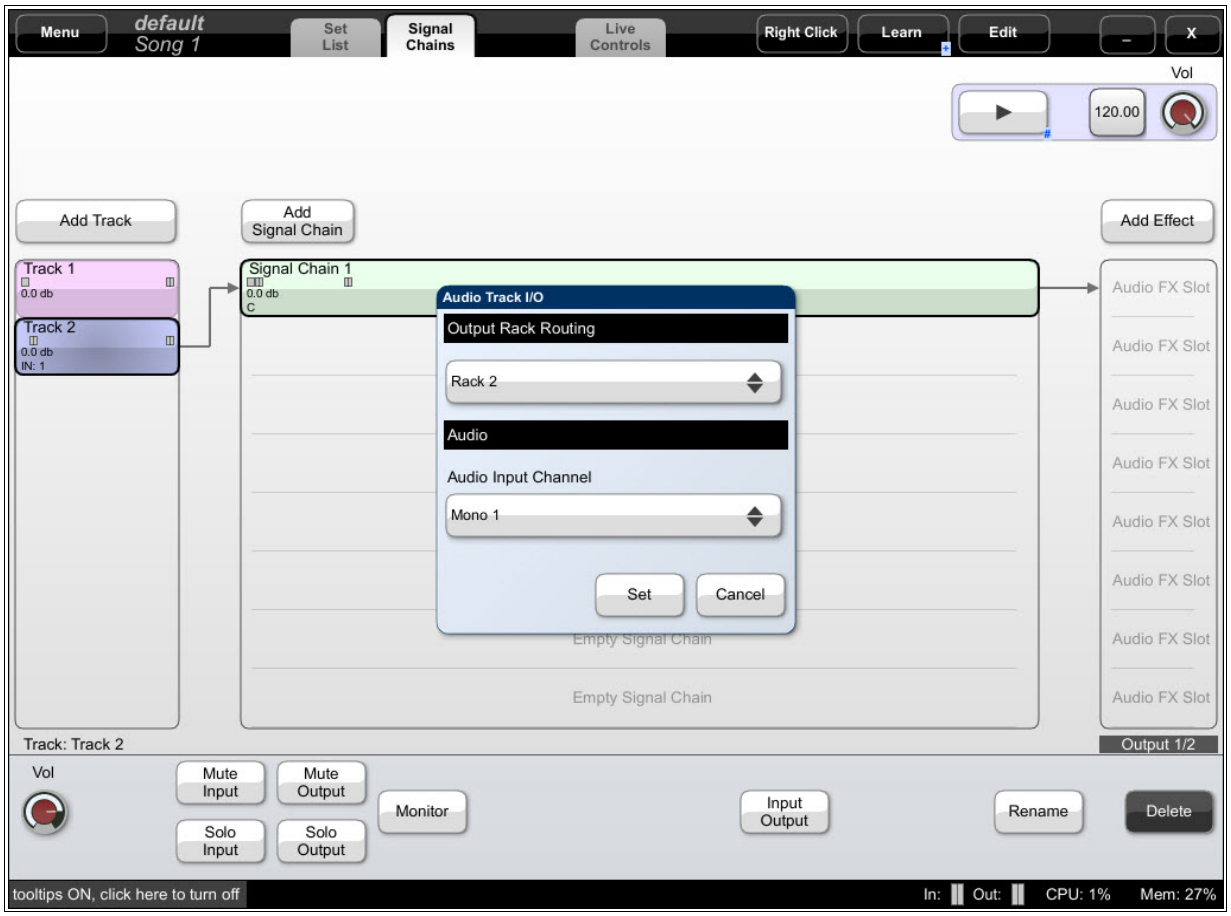
Creating and using Audio Inputs

Riff allows you to feed the signal chain a signal from a live input device, such as a guitar, bass, or microphone. *Refer to Chapter 1 for audio device setup options.*

Click on the [Add Audio Input] button to insert the audio input to the signal chain.

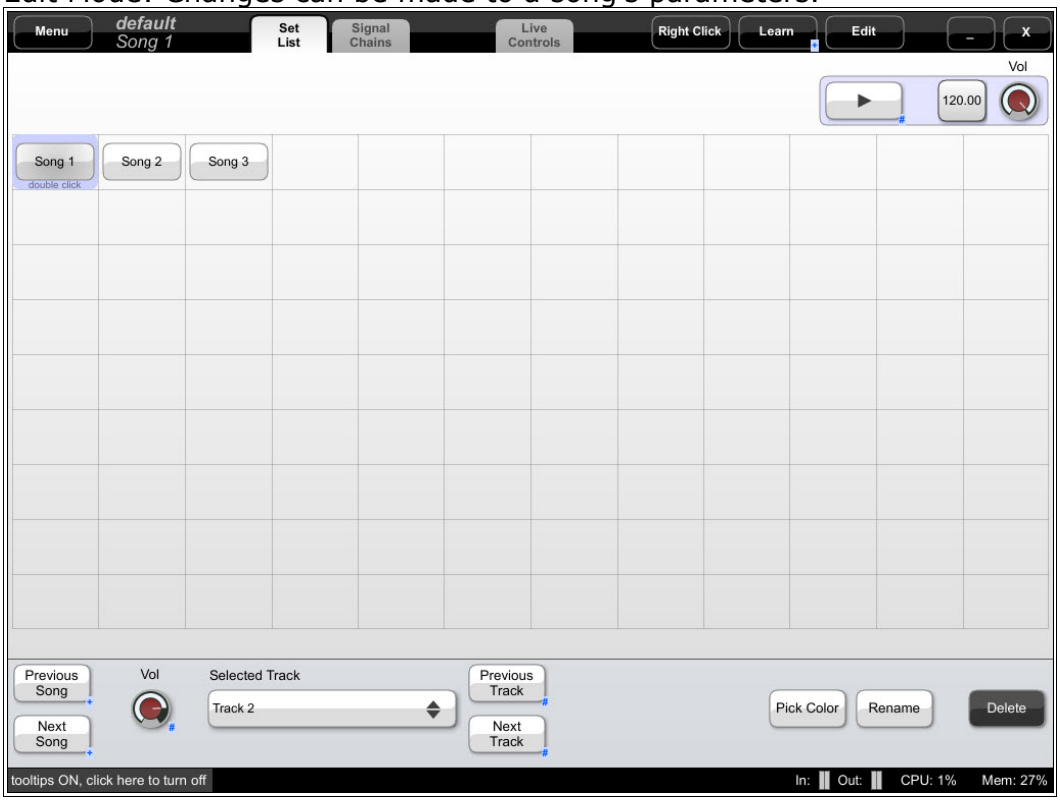


Once the input is inserted, click the [Input Output] button in the center of the lower control section. Use this to configure your audio input source.

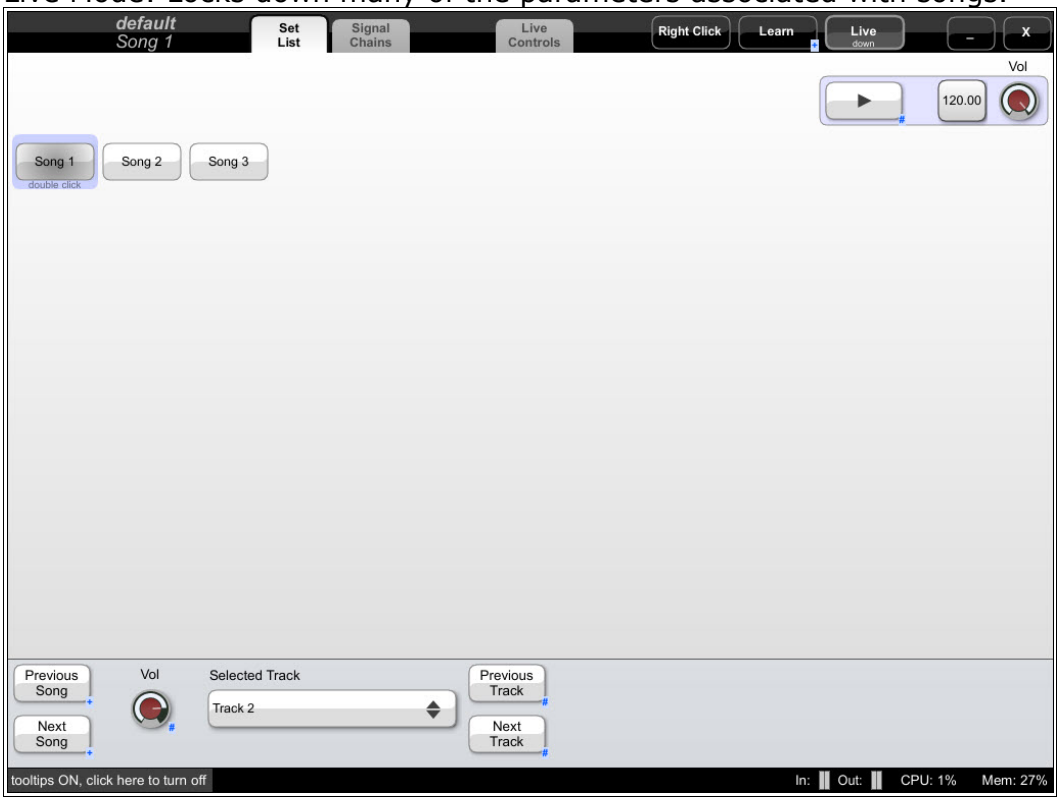


Edit Mode vs. Live Mode

Edit Mode: Changes can be made to a song's parameters.



Live Mode: Locks down many of the parameters associated with songs.



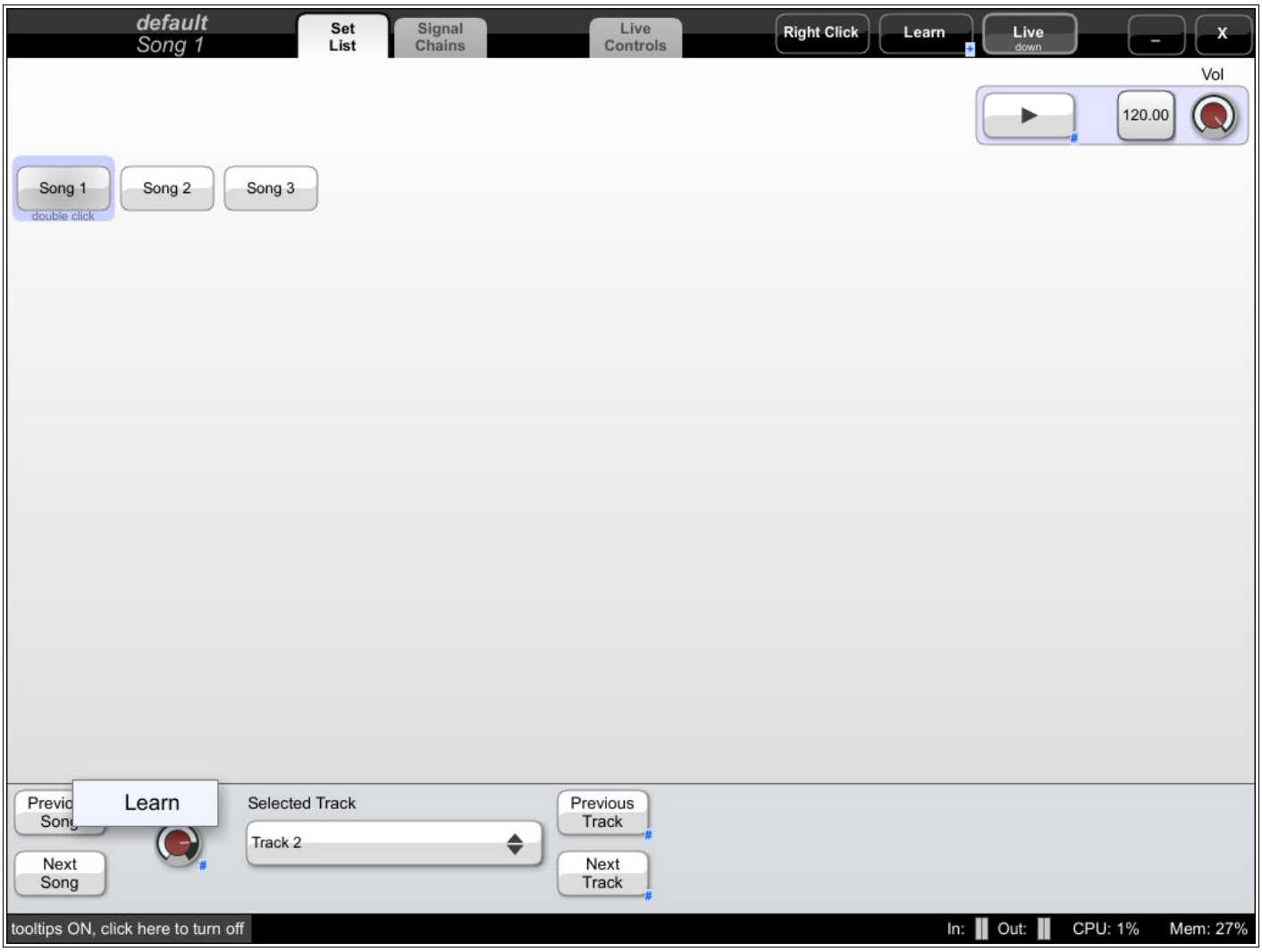
MIDI Learning Hardware Controllers

Most parameters in Riff can be MIDI "Learned".

The first tutorial will show you how to MIDI learn an Riff button to a hardware button on your controller.

First off, return to the Set List and add a couple of Songs. This will show you how to MIDI learn hardware buttons to control the Previous and Next Song command.

Right-click on the [Previous Song] button.



These are the items you can see in this window.

1. *Learn* – MIDI learn maps host buttons to MIDI controllers, virtual live controls to VST's, and virtual live controls to MIDI controllers.

Once you select “Learn” the screen will turn red which means Riff is waiting for you to press a button on your controller. If you look at the Title Bar you will see the [Learn] button now says “Learning”. To cancel learning, press the "Learning" button.

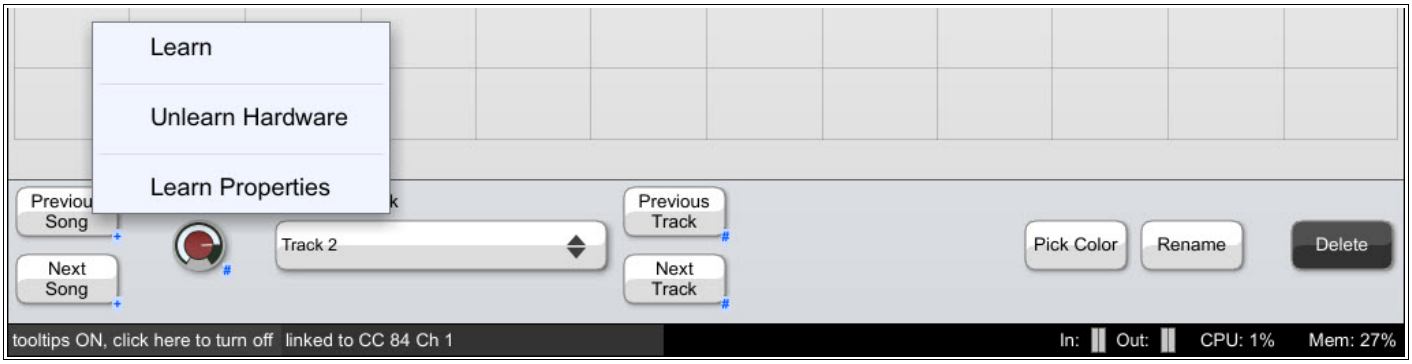


Select a button on youre controller, once you do this the control will be latched. The following picture shows a small icon in the lower right corner of the [Previous Song] button. This represents that this control has been latched to a controller.

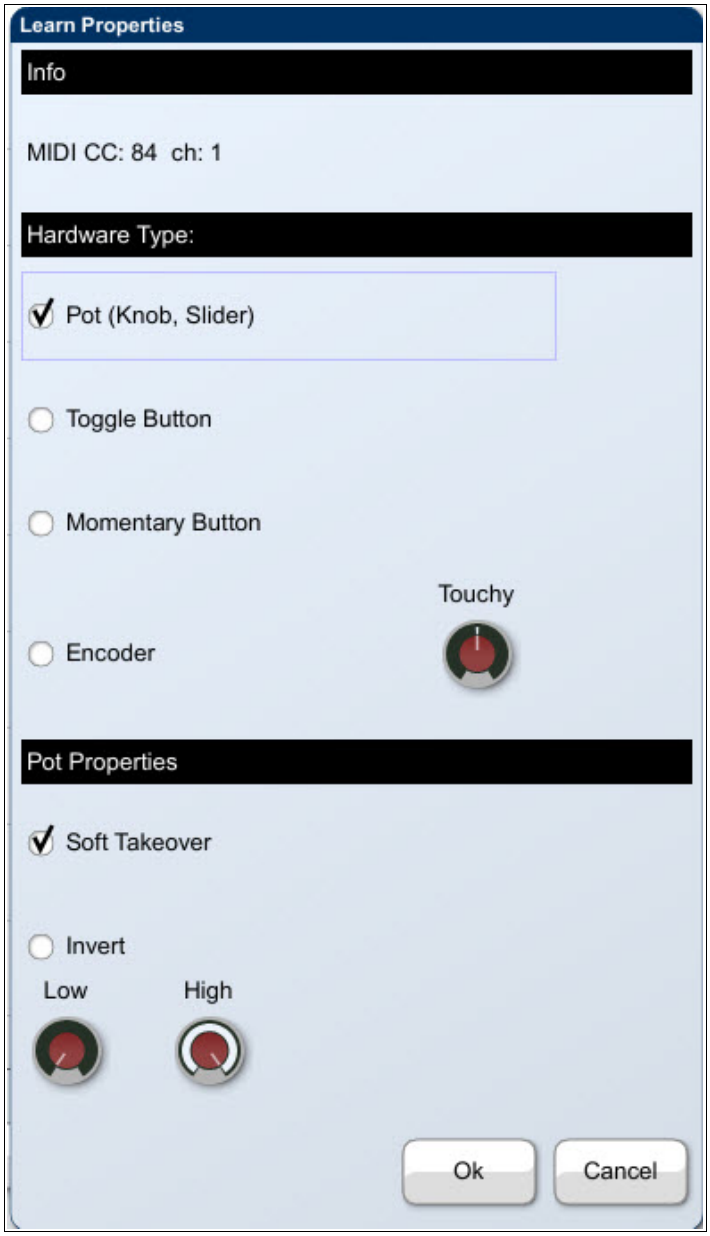


The [+] sign is for a control that has been learned, the [#] sign represents that a control has been learned relative.

Right-clicking a control that has been MIDI learned will yield new options.



1. *Learn* – MIDI learn maps host buttons to MIDI controllers, virtual live controls to VST's, and virtual live controls to MIDI controllers.
2. *Unlearn Hardware* – Use this to release a mapped item from it's MIDI controller. *This option is only available when you have an item mapped.*
3. *Learn Properties* – The following image shows the options in learn properties. *This option is only available when you have an item mapped.*



Info: this section will tell you the MIDI CC number, and the MIDI channel a particular control has been mapped to.

Hardware Type:

- *Encoder*: Informs Riff that an encoder hardware control is being used to perform as a regular knob. Instead of sending +/-1 it will scale from 1 to 127. *Sensitivity: Will change the rate at which the value is increased or decreased. It's similar to the "touchy" knob in live controls.*

Properties:

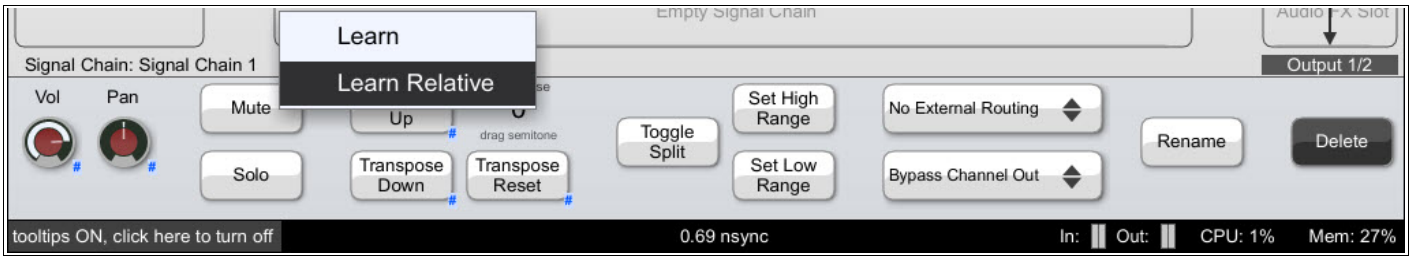
- *Soft Takeover*: Allows the user to realign a latched hardware controller to the corresponding VST parameter smoothly.
- *Invert*: Reverses the high/low or on/off settings of a Live Control.
- *Low/High*: Refers to the minimum and maximum value of a parameter's range.

Learn Relative

Various controls in Riff can be set to MIDI "Learn Relative".



Learning a parameter to "Learn Relative", keeps that mapping for all songs. For example, if you learn relative the signal chain mute:



It will mute the highlighted signal of any song (not just the particular one you are in). As long as you save this set, when you close Riff and reopen it, the learn realative mute will still function the same across all songs.

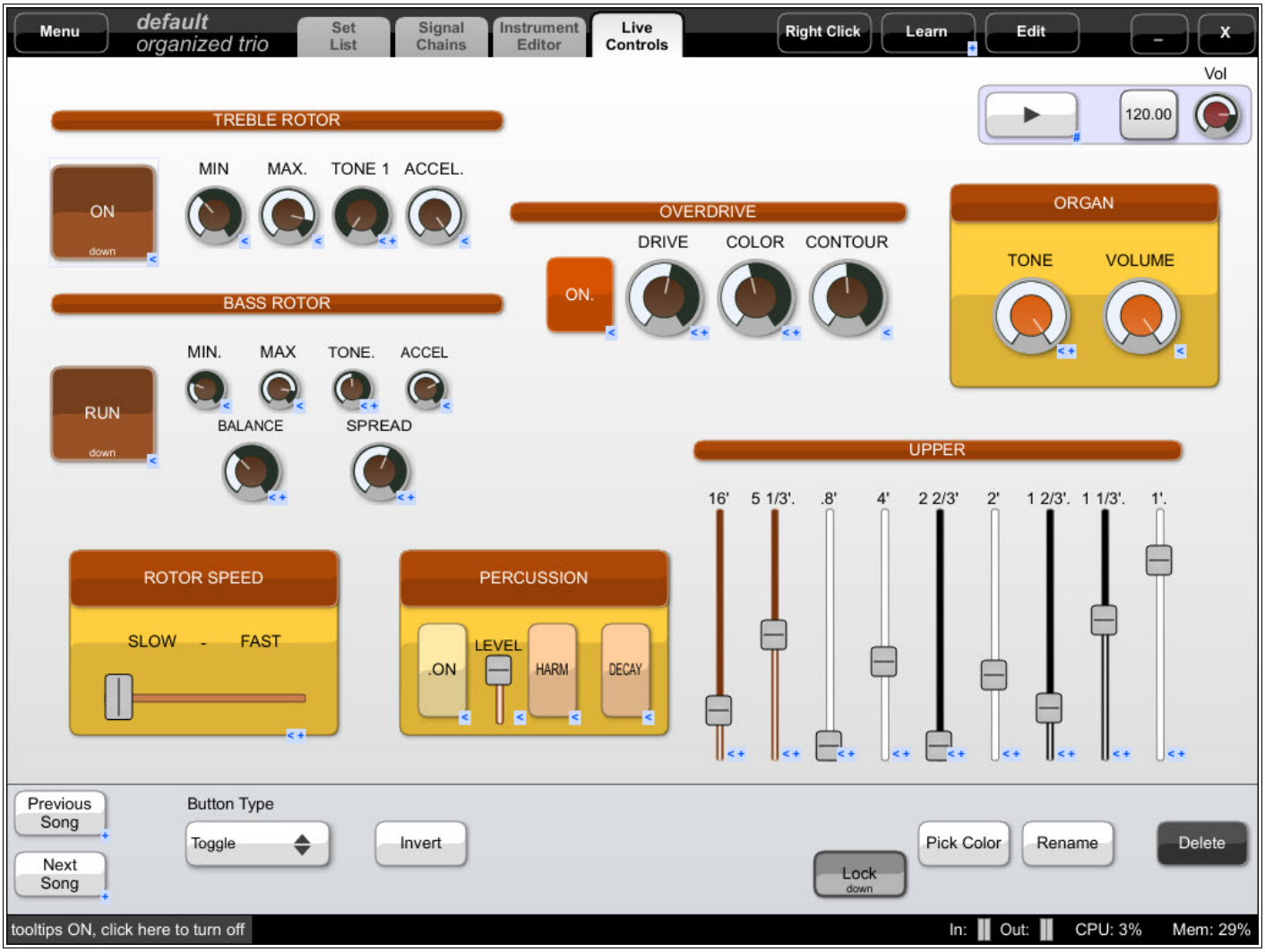
Here are some controls that can be learned relative:

- Setlist lower section Volume
- Setlist lower section play button
- Signal Chain page lower section volume
- Signal Chain page lower section Monitor button
- Signal Chain page upper control play button
- BPM Tap tempo button
- BPM +1/-1 buttons
- Signal Chain instrument lower section volume
- Signal Chain instrument lower section pan
- Signal Chain instrument lower section mute
- Signal Chain instrument lower section transpose up/down
- Signal Chain instrument lower section transpose reset
- Instrument editor lower section volume
- Instrument editor lower section transpose up/down
- Instrument editor lower section transpose reset
- Instrument editor lower section previous/next preset buttons
- Live controls lower section low/high knobs (all live controls Virtual Live Controls)
- Live controls lower section knob/horizontal properties time and offset knobs

Creating and Using Live Controls

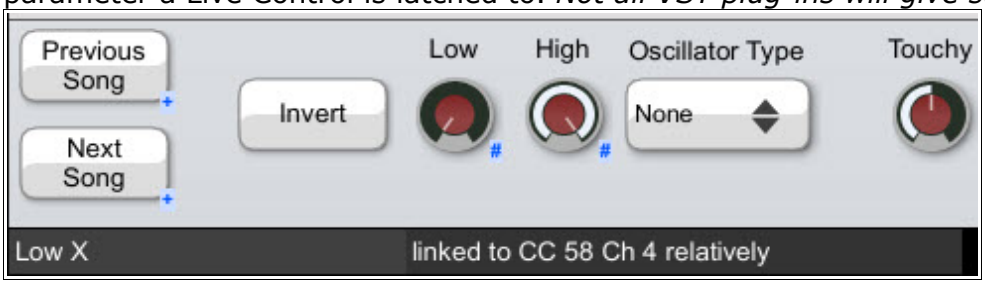
The Live Controls are where virtual knobs, sliders, buttons and XY pads can be created, imported, and mapped to MIDI controllers on VSTi plug-in parameters and even other applications. You can also create Live Controls for VST effect plug-ins. *For more information on the types of Live Controls refer to Chapter 3.*

Here is an example of a Live Control set made for a Vsti.



These controls can be directly linked to the parameters of a VSTi or VST.

Hovering over a latched Live Control like the volume for instance, will allow you to see what parameter a Live Control is latched to. *Not all VST plug-ins will give such detailed information.*



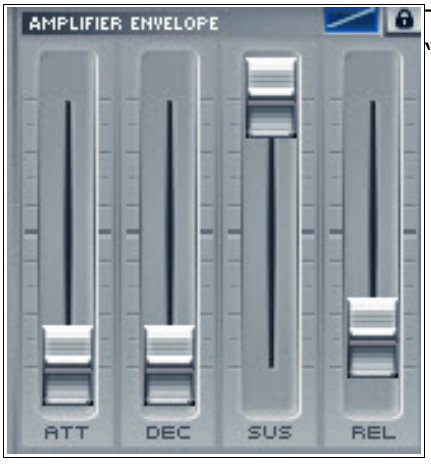
Create a New Live Control

To create and use a Live Control with an instrument, try the next tutorial.

First, create a new Song. For this tutorial insert the plug-in called “Wusikstation. Click on a grid, go to the “Keys” category and select “Wusikstation”.

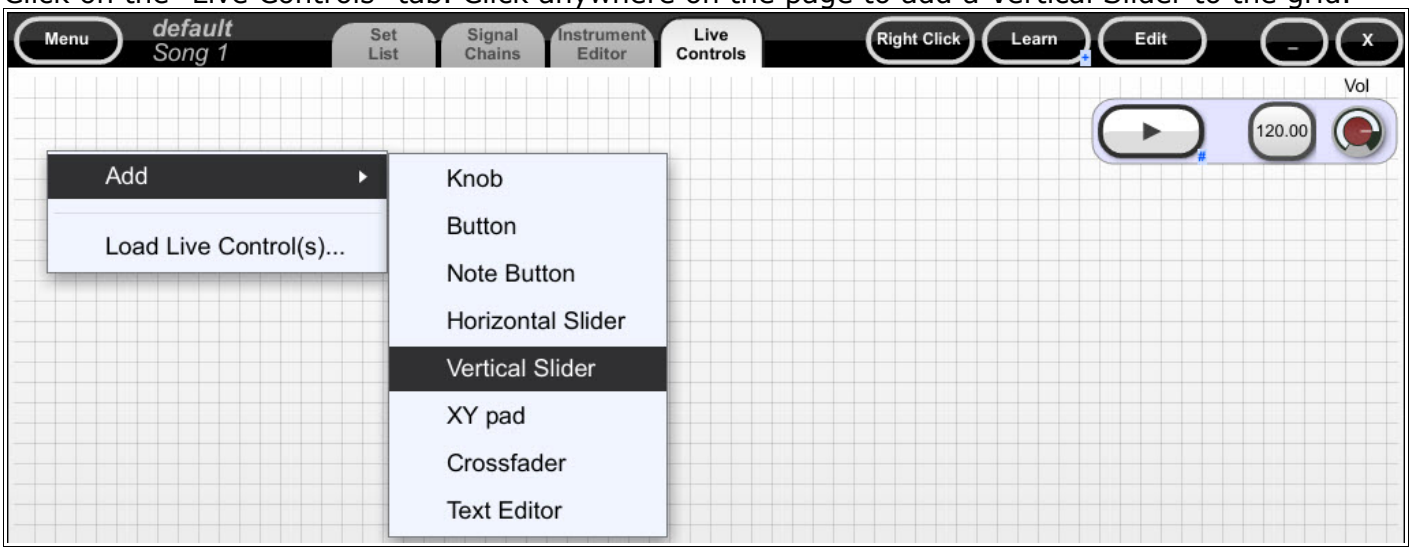
Click on the instrument editor tab to show the GUI of Wusikstation.



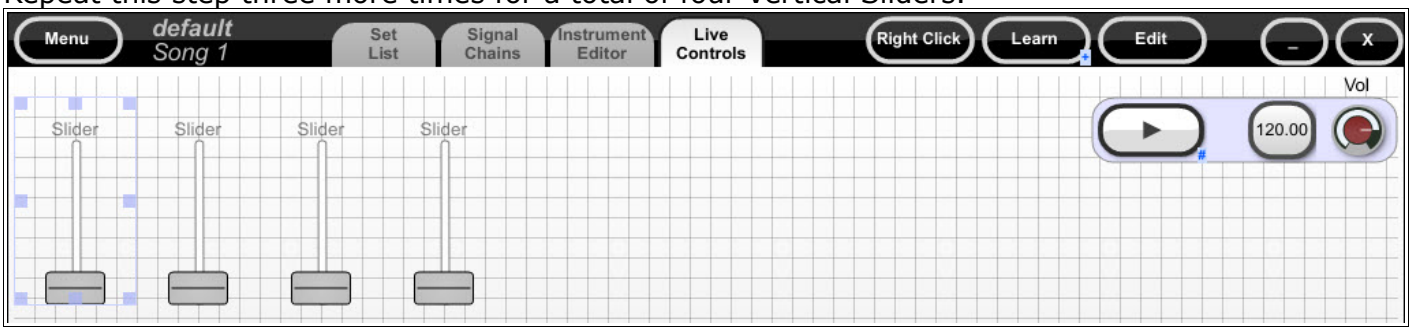


This Live Controls lesson will be to create and MIDI learn the “ATT”, “DEC”, “SUS”, and “REL” sliders in the Amplifier Envelop section.

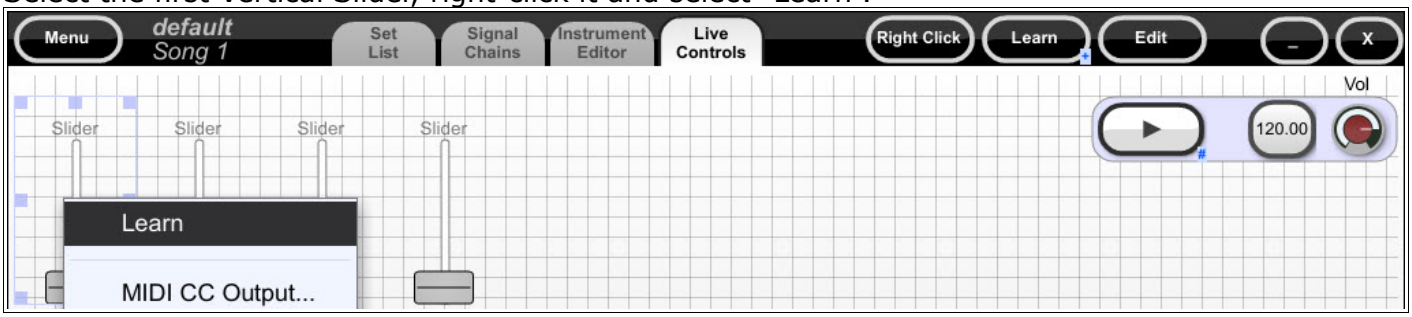
Click on the “Live Controls” tab. Click anywhere on the page to add a Vertical Slider to the grid.



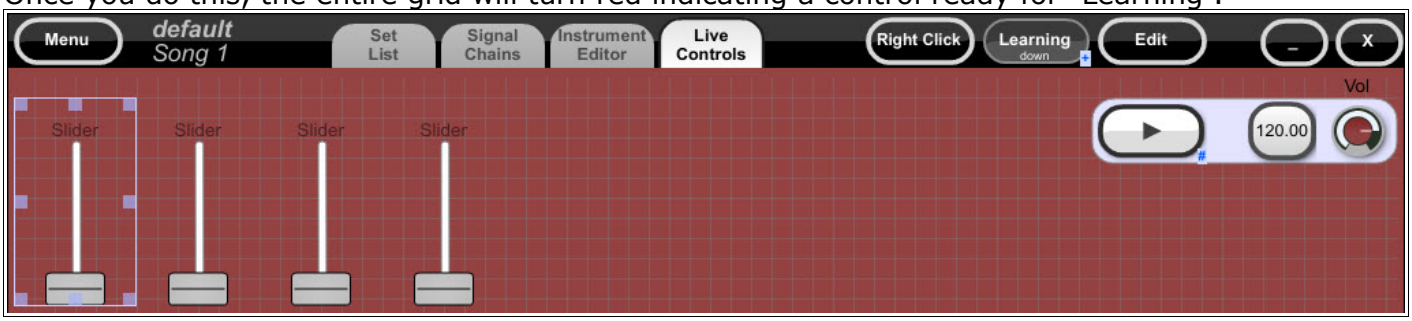
Repeat this step three more times for a total of four Vertical Sliders.



Select the first Vertical Slider, right-click it and select “Learn”.



Once you do this, the entire grid will turn red indicating a control ready for “Learning”.



When Riff is in Learning mode, the program awaits for you to select a parameter to latch this Vertical Slider to.

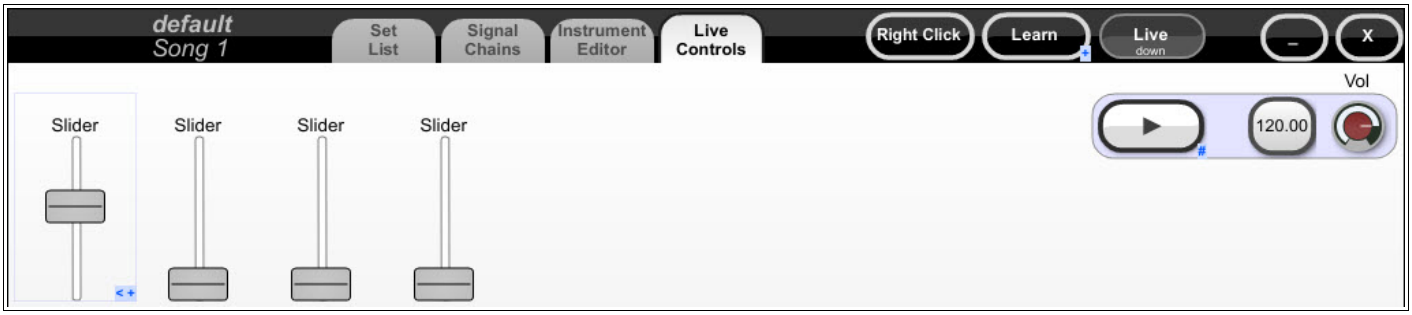
1. Click the destination control.
2. Select the Live control that will control it.

Click the “Instrument Editor” tab to get back to the GUI of Wusikstation. Touch the "ATT" slider using either the touchscreen or the touchpad.

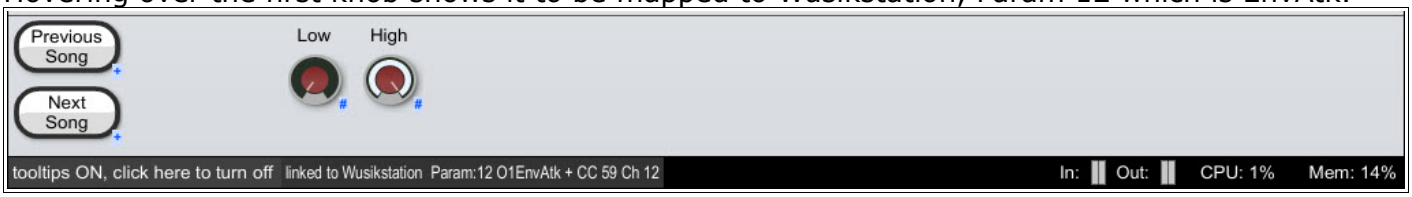


This screen will turn back to the default color and this control learned to the Live Control. Click on the “Live Controls” tab once again.

Switch Riff over to “Live” mode, play notes on your keyboard, and use the touchscreen to move the first fader.



Hovering over the first knob shows it to be mapped to Wusikstation, Param 12 which is EnvAtk.



To rename or change the color of a Live Control, switch back to “Edit Mode” select the Live Control and press the [Pick Color] or [Rename] button.

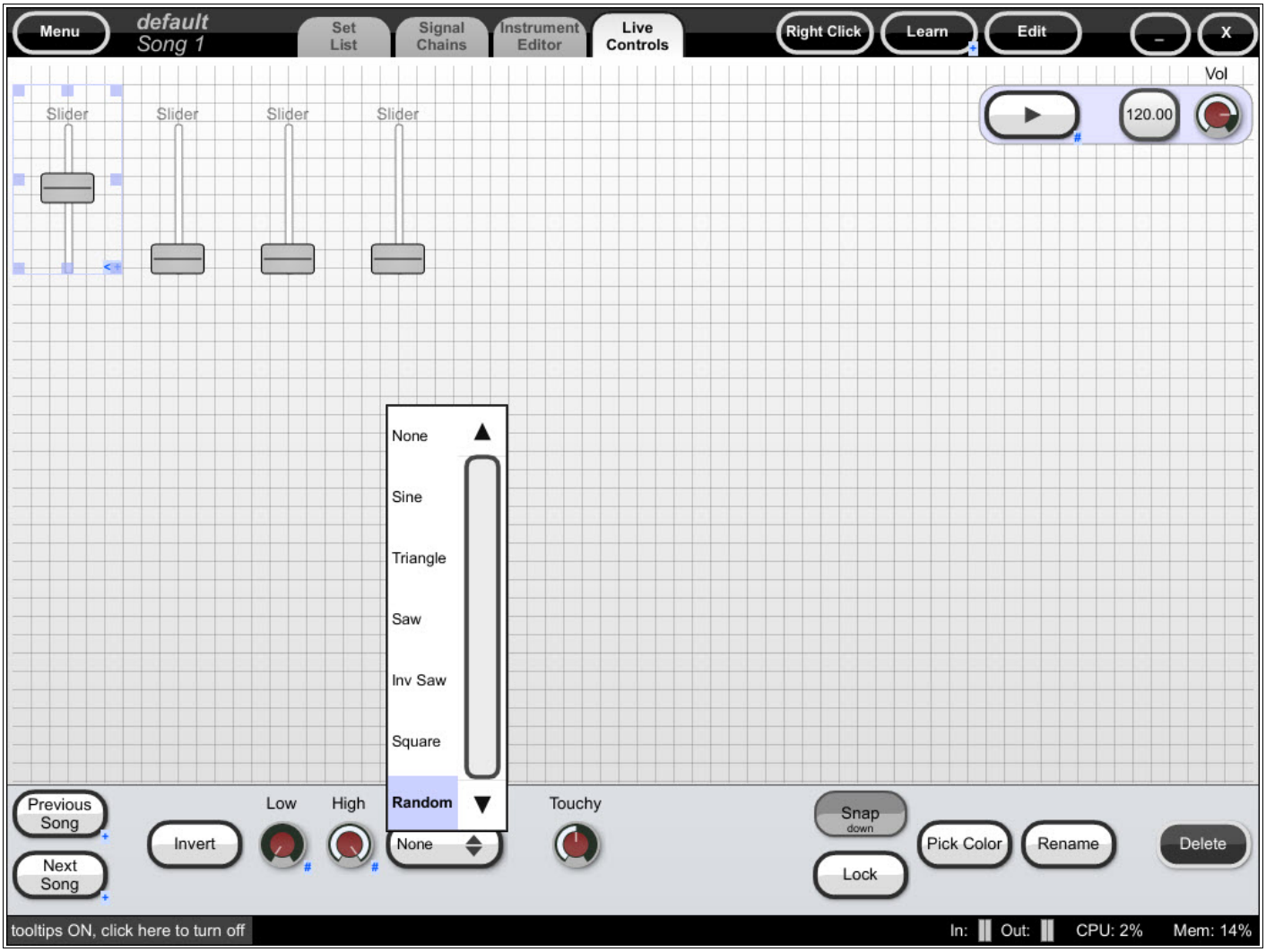
Go ahead and learn the other three Vertical Sliders to the controls of the Envelope section of Wusikstation.

Using Oscillators with Knobs/Sliders

Knobs and sliders can have an oscillator added to them. Refer to Chapter 3 for the types of oscillators used in Riff. Also, there is a definition of the term "oscillator" in Appendix A.

First off, setup a knob in your Live Controls. For this tutorial, I am going to use the same Live Controls made in the last tutorial with Wusikstation.

Click on the knob, then select the "Oscillator Type". Pick the type of osciallater type you would like to use. Once you do this the oscillator will take control over the knob.



- You can sync the oscillator to the BPM of the current song by clicking on the [Use BPM] button.
- Also, you can sync to the sequencer host by using the [Use Seq] button.

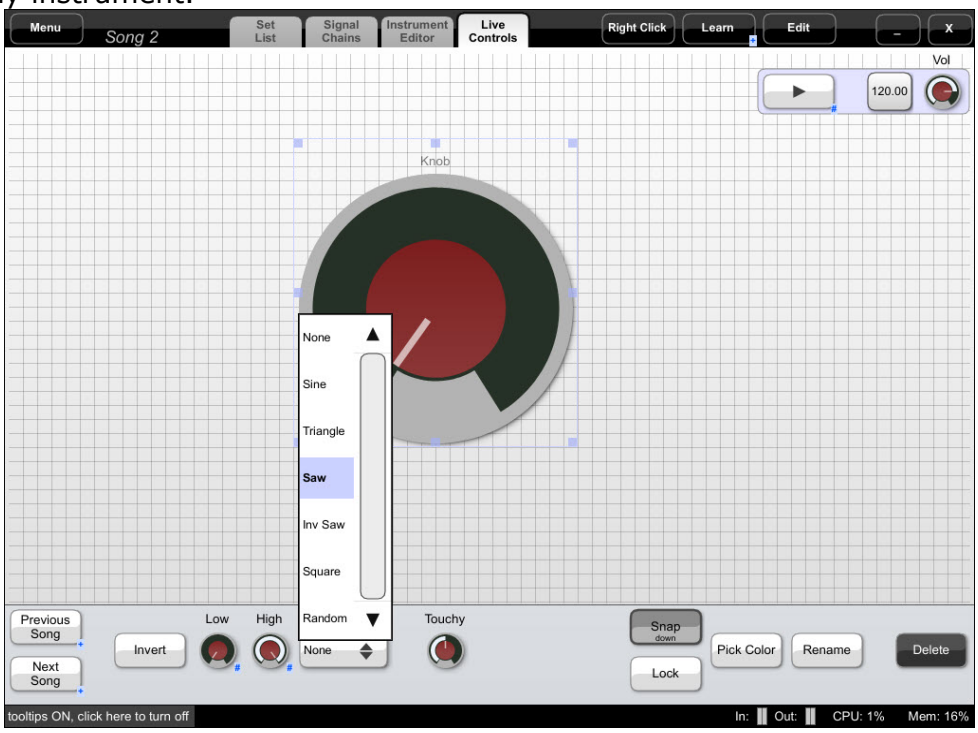
You can latch the oscillators to start with the play button of the host and any other sequencer that has been synced to the host. When you hit play the oscillators start oscillating along with a sequencer (if you have put it to sync with the host). *This only applies to the selected live control.*

Mapping Live Controls to the Lower Control Section

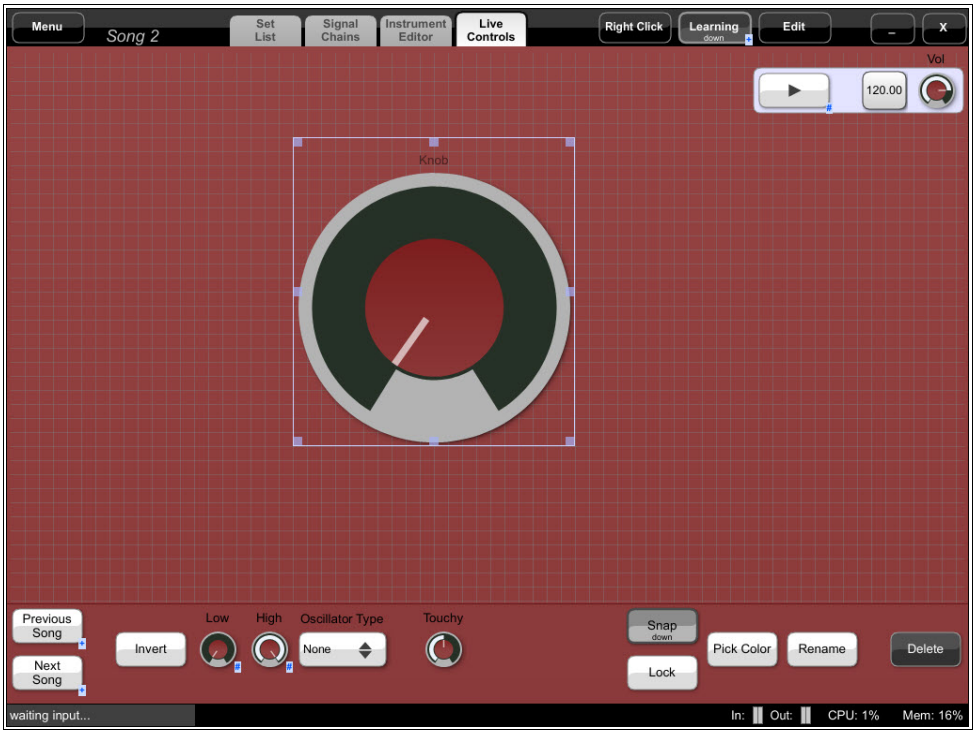
If you have an oscillator latched to a Live Control knob, you can have that oscillator control the volume or pan control of a signal chain. The following tutorial will show you how to MIDI learn a Live Control Oscillator to the volume knob of a signal chain.

Create a new Song and add any instrument.

1. Go to the [Live Controls] tab and add a new knob. Select the new knob and add an oscillator.



2. Select the [Learn] button, then select the Knob.



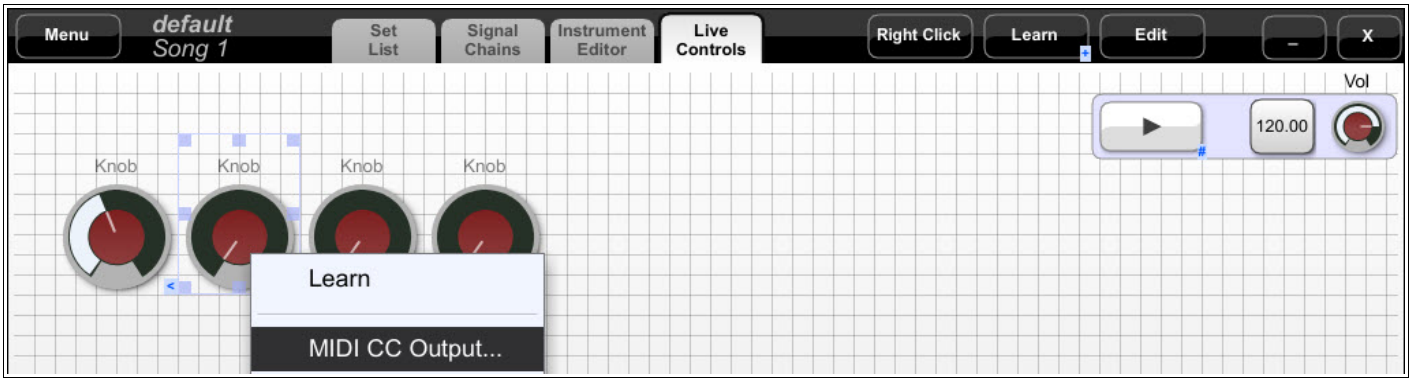
3. Select the [Signal Chains] tab and select the volume knob in the Lower Controls.

Map a Live Control to an External MIDI CC

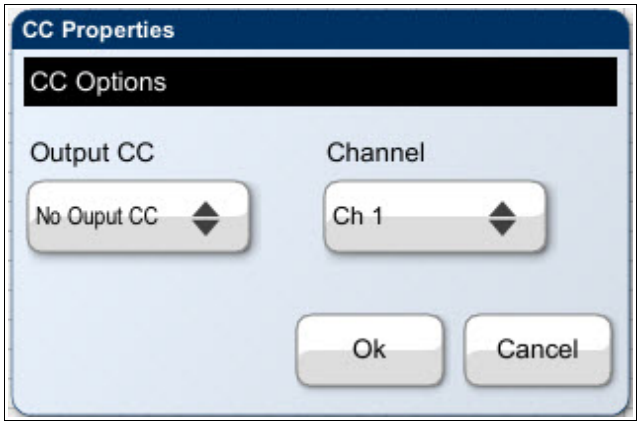
Live Controls can also be assigned to control a parameter from another application which offers controls that respond to MIDI CC messages. Such as track faders, transport controls, etc. First you will need to know what the CC of the other program is using. This information can usually be found in the particular products documentation, or you can Google search for a MIDI implementation chart.

Once you know the MIDI CC, it's very simple to assign a Live Control to that particular item.

Right-click on the Live Control and Select [MIDI CC Output....].

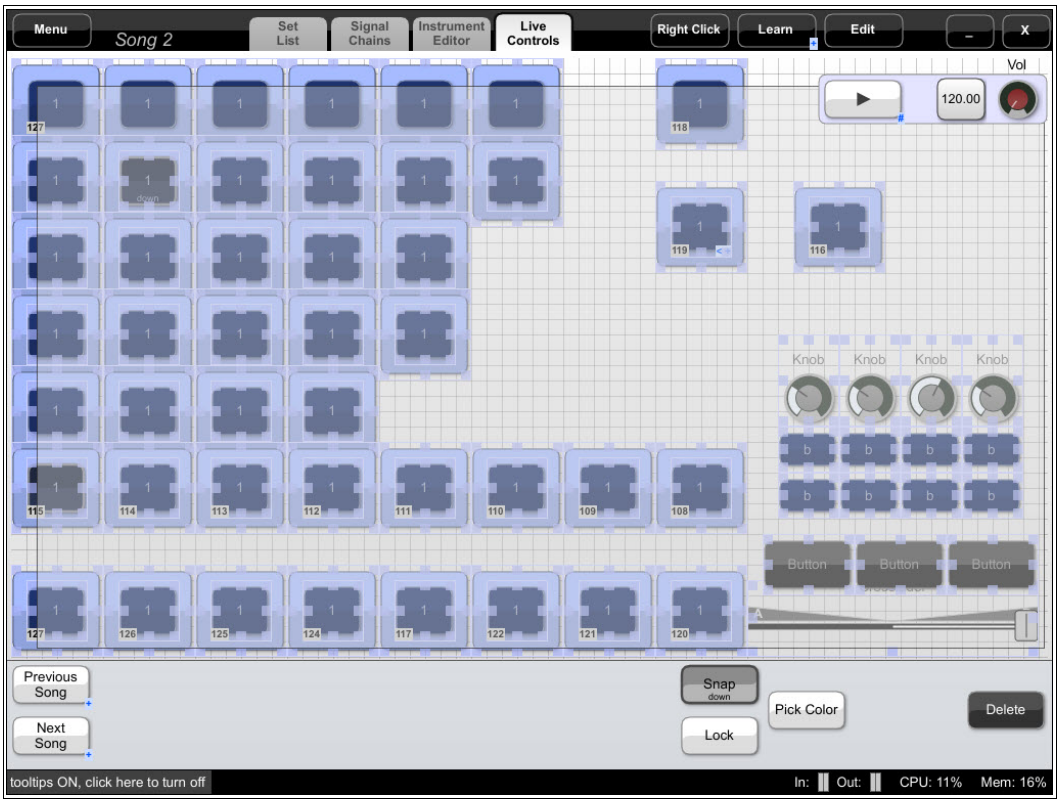


Once you choose the *MIDI CC Output....* The CC Properties window will pop up. This is where you assign the output CC and the MIDI Channel the target control is residing.



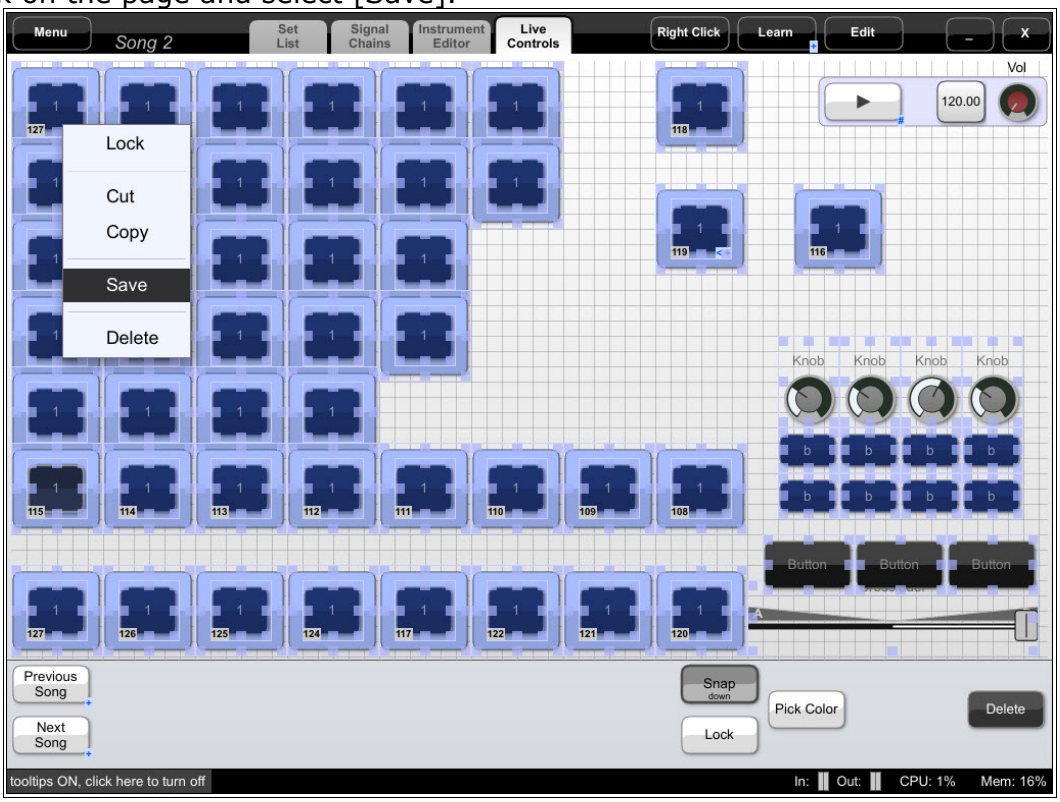
Saving Live Controls as a Template

You can save a Live Control or a whole Live Control set. To this, just left-click drag to select all the elements of the Live Controls you want to save as a template.



1. Select all the items to be saved.

2. Right-click on the page and select [Save].



3. Choose a name for the Live Controls that are being saved.



- *File* - This is where you type in the name for the Live Control being saved.
- *Description* - Give the Live Control a brief description.
- *Save ALL Live Controls* - Place a checkmark here to save all the Live Controls on the current page.
- *Save Selected Live Controls* - Select this option if you only want to save the currently selected Live Controls.

Using Multi-timbral Instruments / Sharing Instruments

This section of the tutorial will focus on using Multi-timbral instruments in Riff, but will also give you some insight on using the "Shared Instrument feature". We will you show you two ways to use a multi-timbral instrument.

A multi-timbral instrument is a VSTi that can use up to 16 MIDI channels at once. Purity is an instrument that falls into this category, and is also preinstalled on an Open Labs production station.

If you are using an Instrument like Purity, which is multi-timbral and a sample player, the benefits are crucial.

Using the "Currently Loaded Instrument" option allows you to use one instance of an instrument in multiple songs. This will save you tons of system performance and allow you to achieve far more complex instrument arrangements.

Anytime you are using big libraries it's a good idea to set an instrument to shared.

Shared instruments are not limited to an instrument having to be multi-timbral. You can set any instrument to be shared. Some normal instruments like True Pianos can be shared. True Pianos is known to use more system resources than other similar instruments. Instruments that sound better generally use more resources.

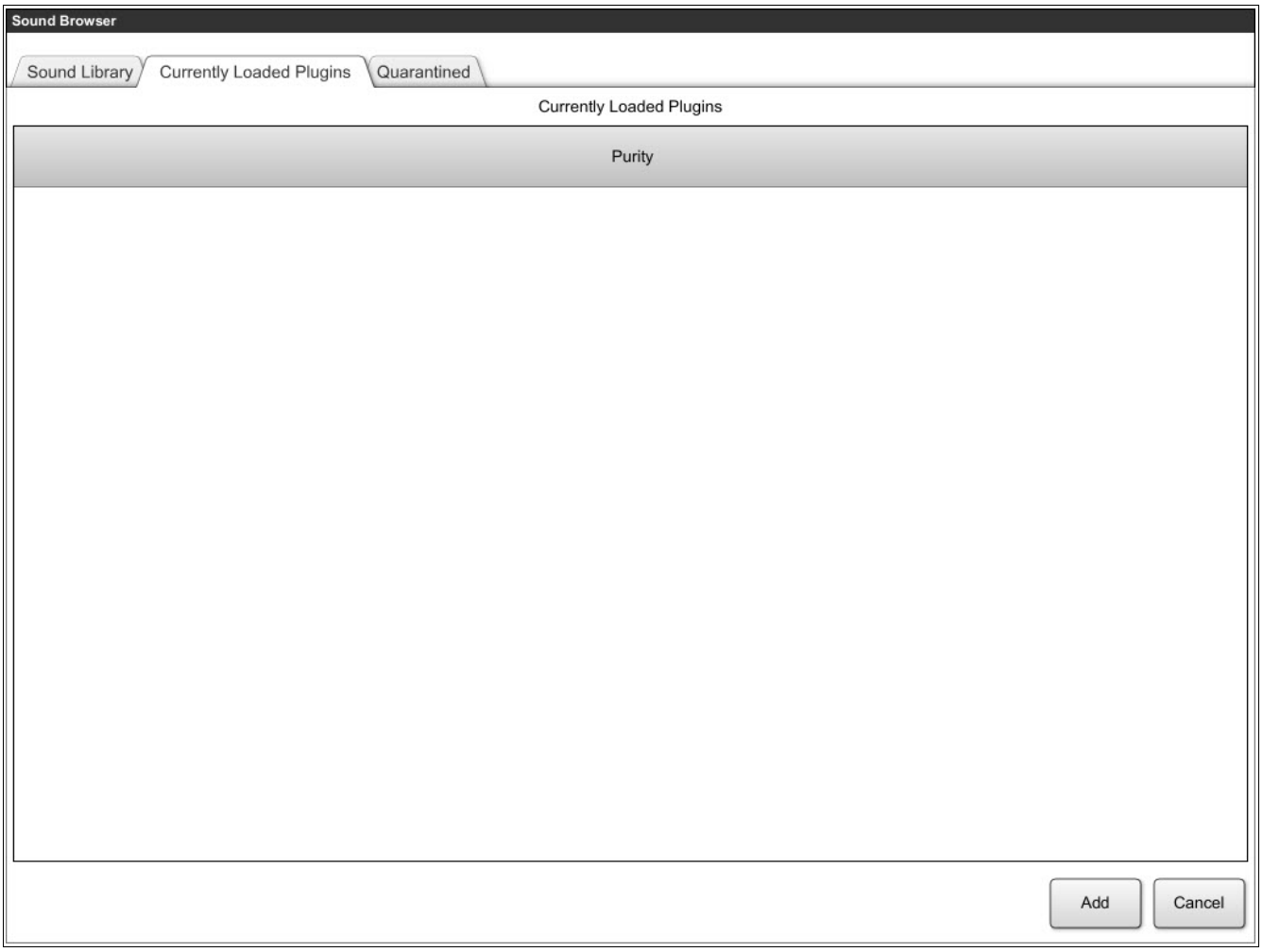
Sharing Instruments

To help save system resources, Riff has a great solution. This solution is called sharing instruments. For instance if you are planning to use the instrument "Purity" on more than one Song, normally you would have to load a new instance of Purity on each Song.

Every time you add a new instance of that instrument you use more system resources.

Riff allows you to use a "Currently Loaded Instrument" multiple times. Hence, this instrument becomes a "Shared" instrument.

To Share an instrument, first create a new Song, and add Purity to it. After doing that, go to a new grid of the Setlist, select [Add], the sound browser appears. However, this time instead of adding a new instance of Purity, select the [Currently Loaded Plugins] tab.

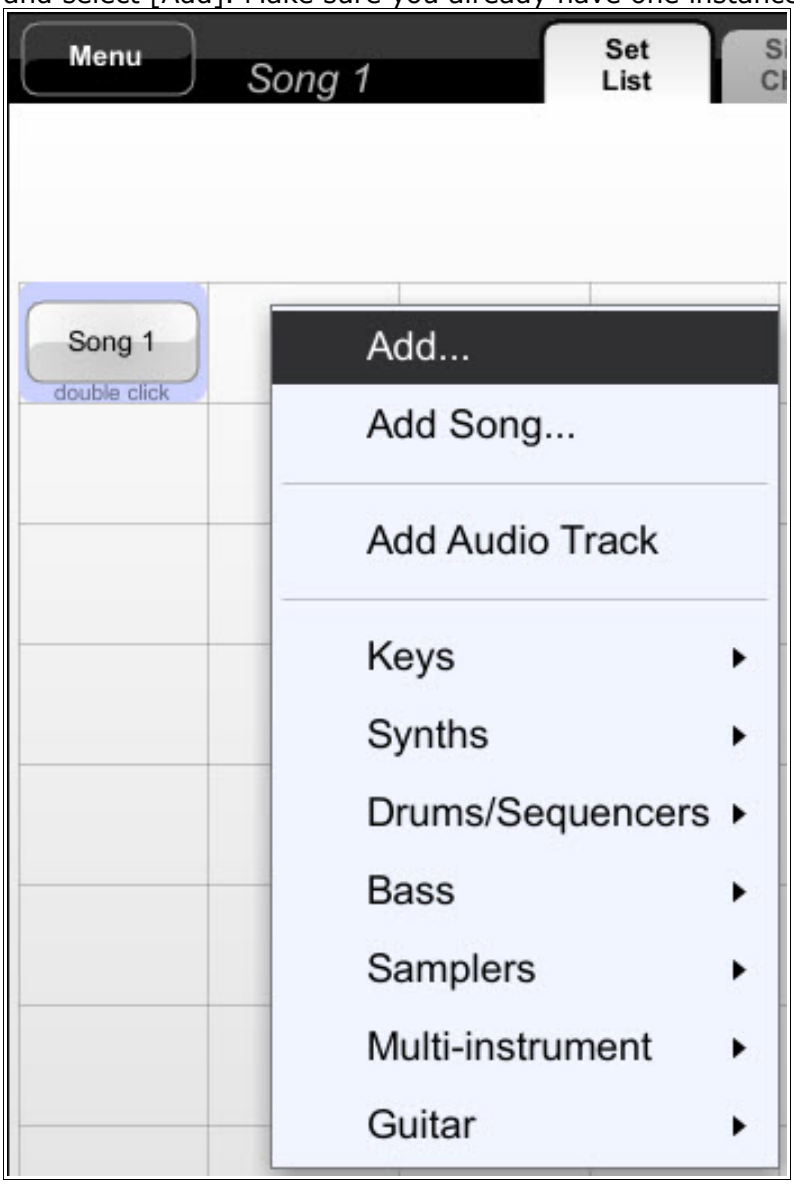


Here you will see your list of Currently Loaded Plugins. Instead of loading a new instance of an instrument, select the currently loaded instance of Purity and either press the [Add] button or double left-click the instrument. This is how you share an instrument in Riff.

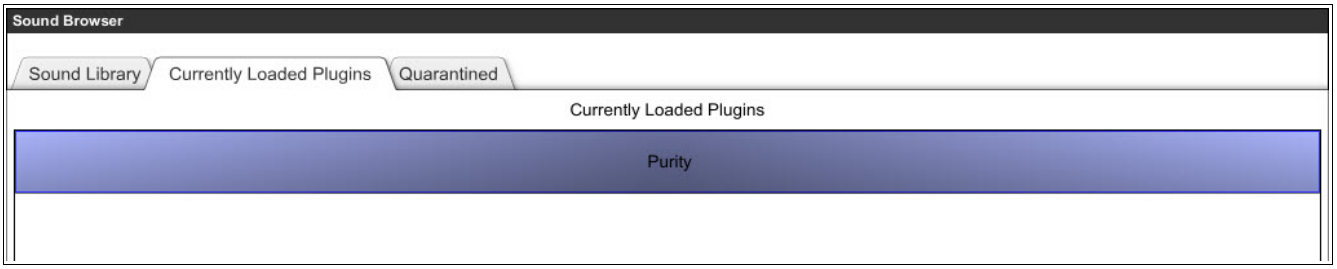
The next part tutorial will show you how to use this shared instance of Purity as a multitimbral instrument.

Song Grid and the Multi-timbral Instrument

1. Click a grid space other than the one that says "Song 1", then click on the grid to create a new Song and select [Add]. Make sure you already have one instance of Purity on the Setlist.

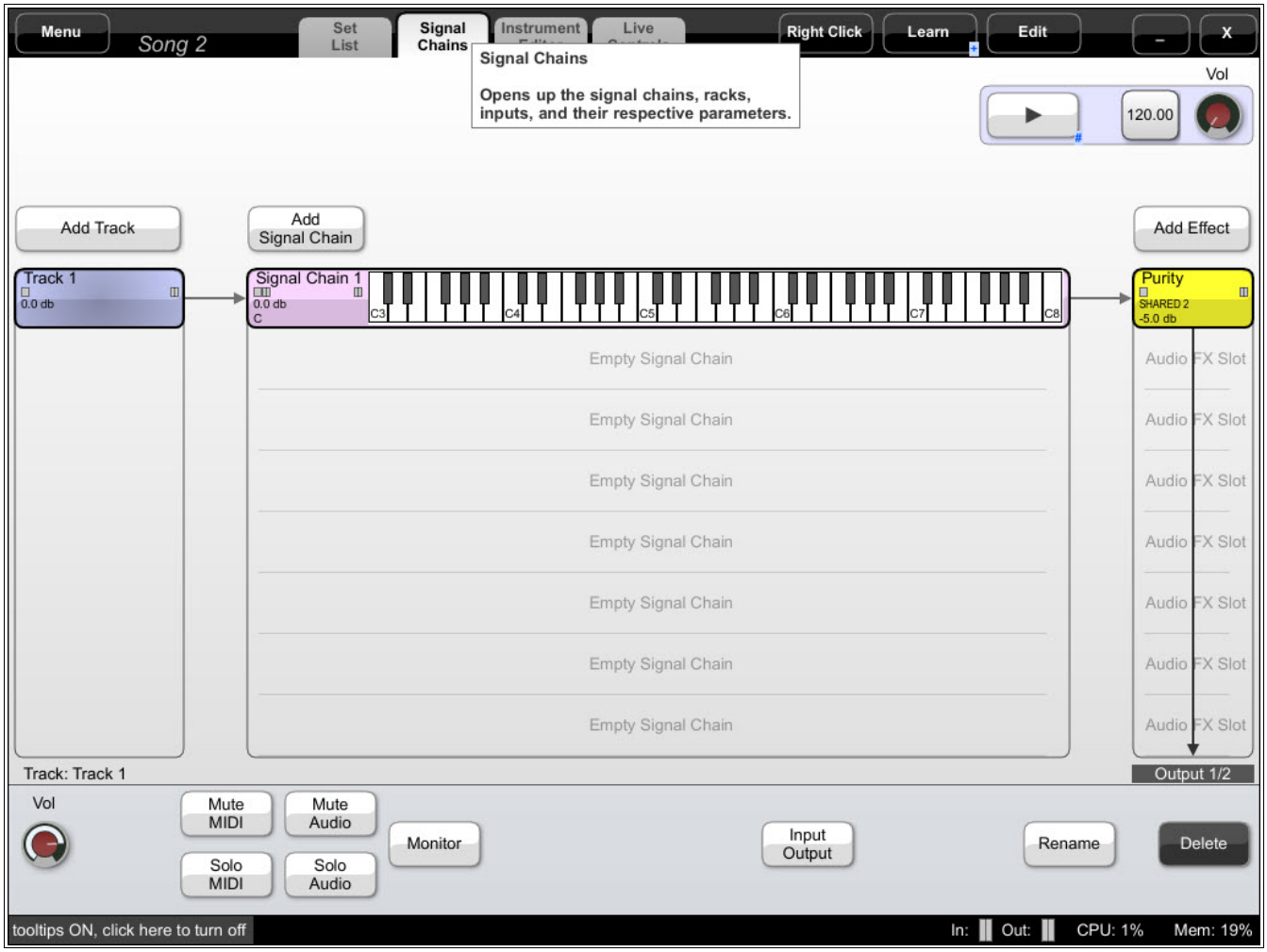


2. Select the [Currently Loaded Plugins] tab and add the currently loaded (shared) version of "Purity".



After [Add] has been pressed, a new Song is created with this "shared" instance of Purity as the instrument.

3. Go to the [Signal Chains] tab. Look at the Stream Processor, Purity is now yellow and the word "SHARED" is present in the tab. This confrms this instrument is indeed shared.



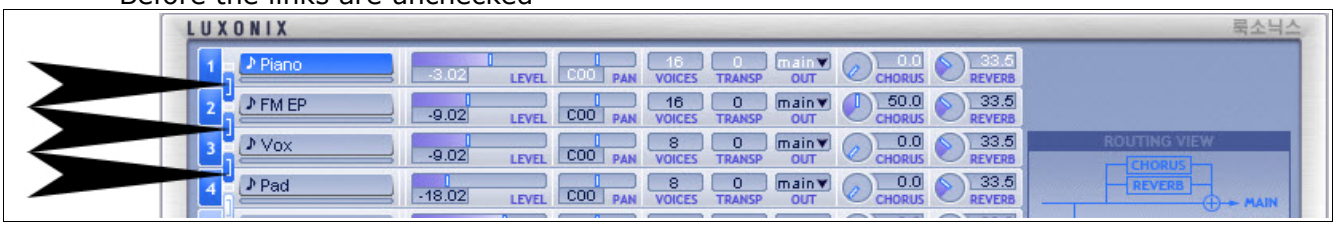
Time to set up Purity for multi-timbral use. Purity is just the example for this tutorial, however many multi-timbral instruments work similar.

The goal here is to have one instance of Purity running, utilizing its multi-timbral interface having sounds loaded on separate Songs.

For this tutorial, only four Purity MIDI channels and sounds will be used.

Go back to the Setlist page, select "Song 1" then select the [Instrument Editor] tab, click on the channel link icons in Purity. If this is not done, each pair of MIDI channels will remain linked together.

Before the links are unchecked

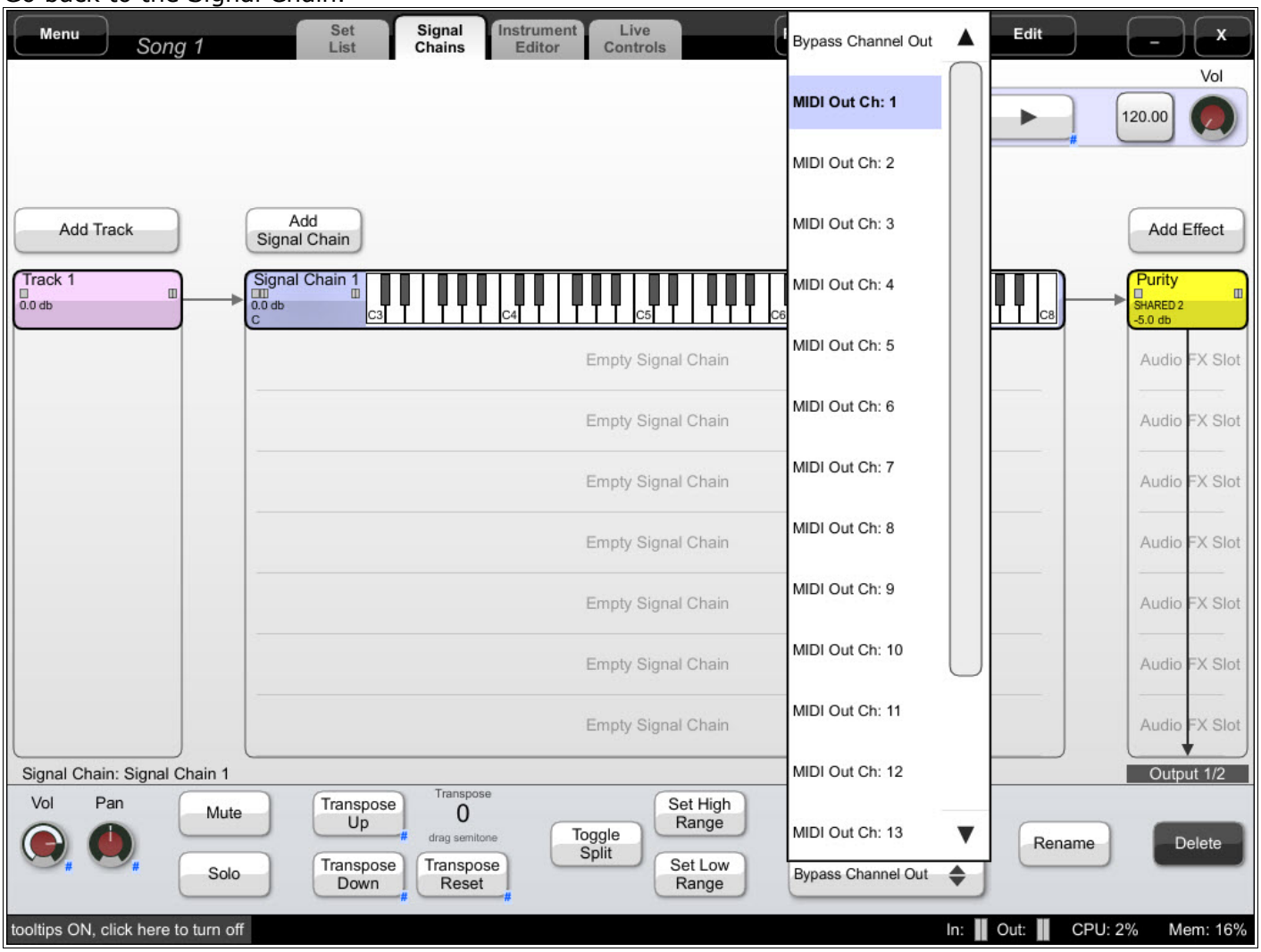


After the links have been unchecked

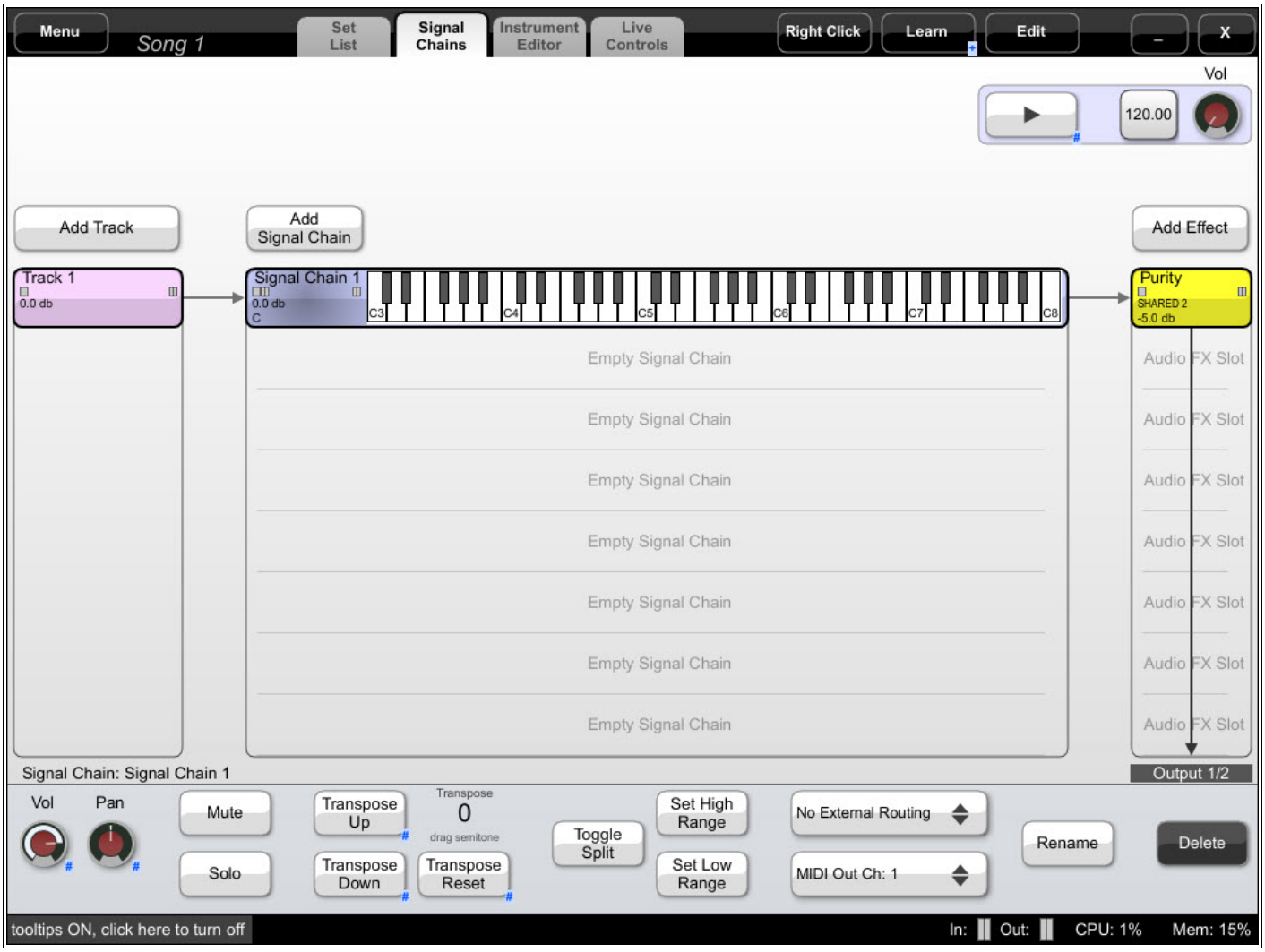


Other VSTi's will have dfferent sets of rules, however the basic principles will relate. Purity can use 16 instruments at once over a spread of 16 MIDI channels. Select sounds for the one through four slots of Purity. These channels will also represent the corresponding MIDI channels.

Go back to the Signal Chain.



Song 1 should only use MIDI Channel 1. On the Signal Chain page click the [Bypass Channel Out] button in the lower controls and select MIDI channel 1. (See the prevous image)

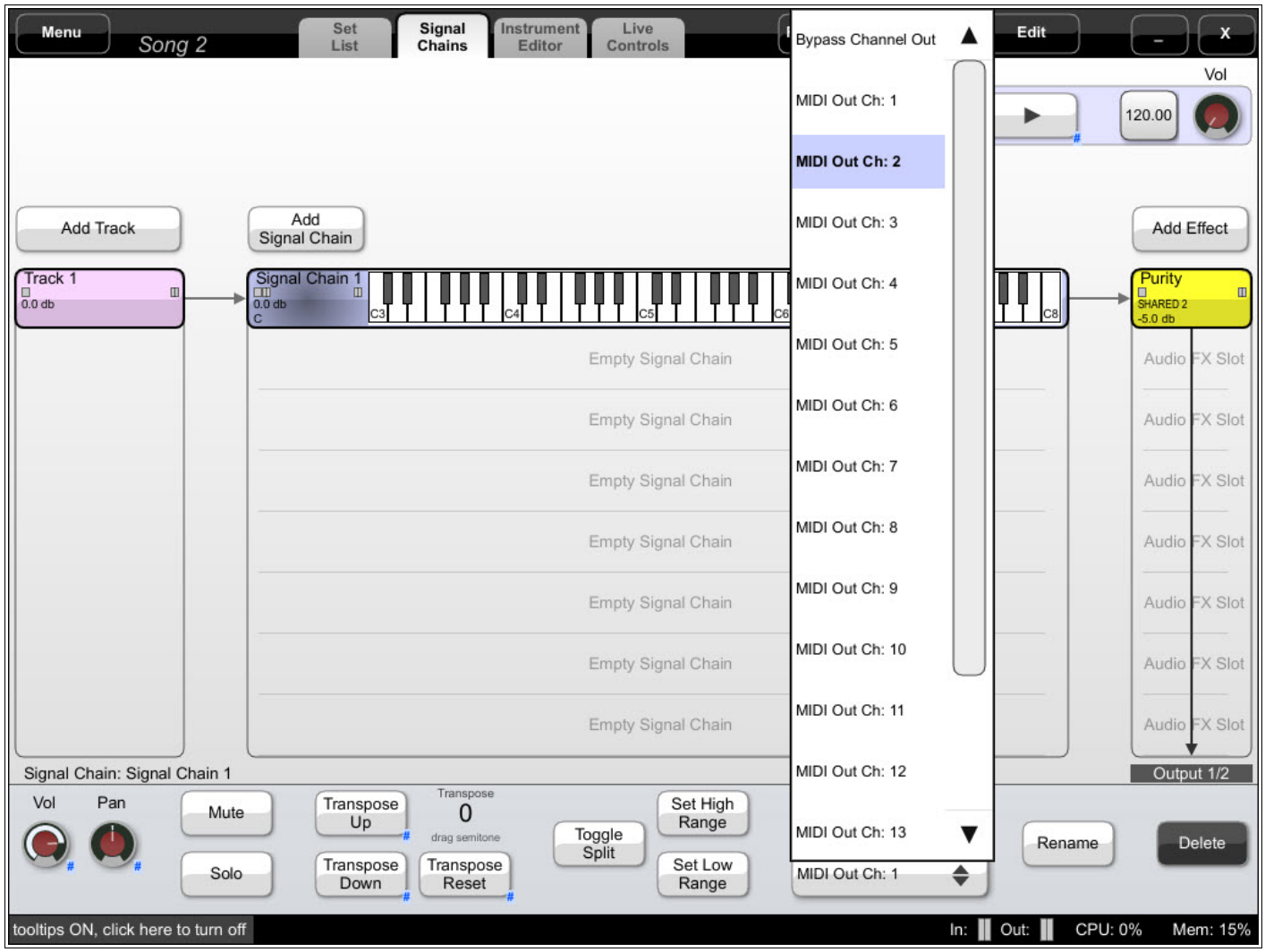


You can see this signal chain is using MIDI channel 1. And will only trigger the sound in MIDI channel 1 of Purity.

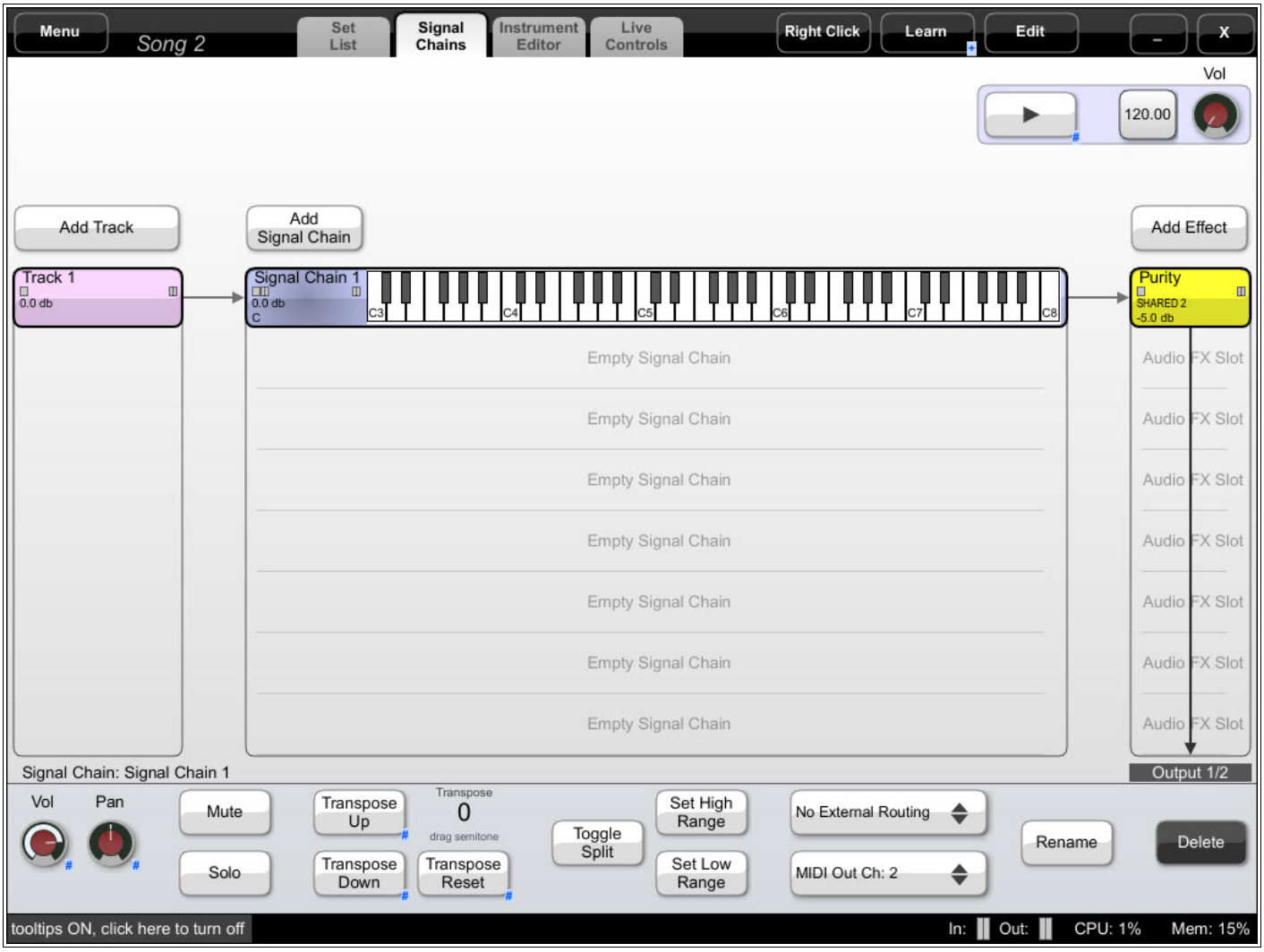
Now go to Song 2 on the setlist, and select the [Signal Chains] tab.

Go to the Signal chain of Song 2. Remember this Song is using a "Currently Loaded" shared version of Purity.

We want this Song and Signal Chain to only use MIDI Channel 2. On the Signal Chain page click the [Bypass Channel Out] button in the lower controls and select MIDI channel 2.



You can see this signal chain is using MIDI channel 2.



Using a multi-timbral instrument this way saves system performance and memory.

Now go back to the Setlist and [Add] another Song (Song 3), as before use a "Currently Loaded" version of Purity. Go to the [Signal Chains] tab of this Song and set it to MIDI channel 3. This will trigger whatever sound you selected for the third slot of Purity.

For the fourth Song, go back to the Setlist and [Add] another Song (Song 4), as before use a "Currently Loaded" version of Purity. Go to the [Signal Chains] tab of this Song and set it to MIDI channel 4. This will trigger whatever sound you selected for the fourth slot of Purity.

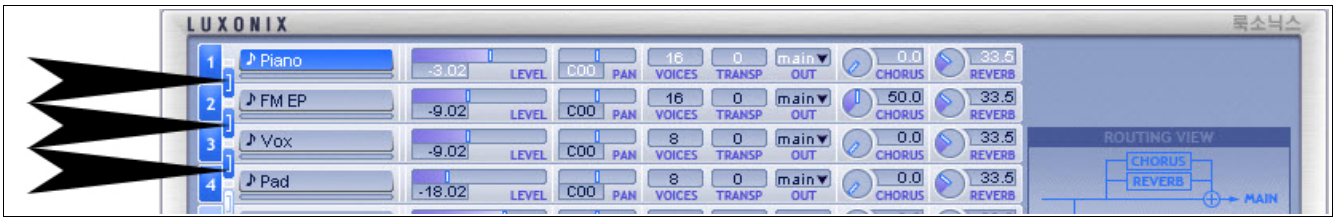
Now, when you switch Songs on the Setlist, the selected Song will trigger the adjacent channels in Purity or whatever multi-timbral instrument you are using.

Creating Keyboard Splits with Multi-timbral Instruments

For this tutorial, Purity will be used. Create a Song and add Purity as the instrument.

Select the [Instrument Editor] tab then click on the channel link icons in Purity. If this is not done, each pair of MIDI channels will remained linked together.

Before the links are unchecked



After the links have been unchecked



Other VSTi's will have dfferent sets of rules, however the basic principles will relate. Purity can use 16 instruments at once over a spread of 16 MIDI channels. Go ahead and select sounds for the one through four slots of Purity. These channels represent MIDI channels.

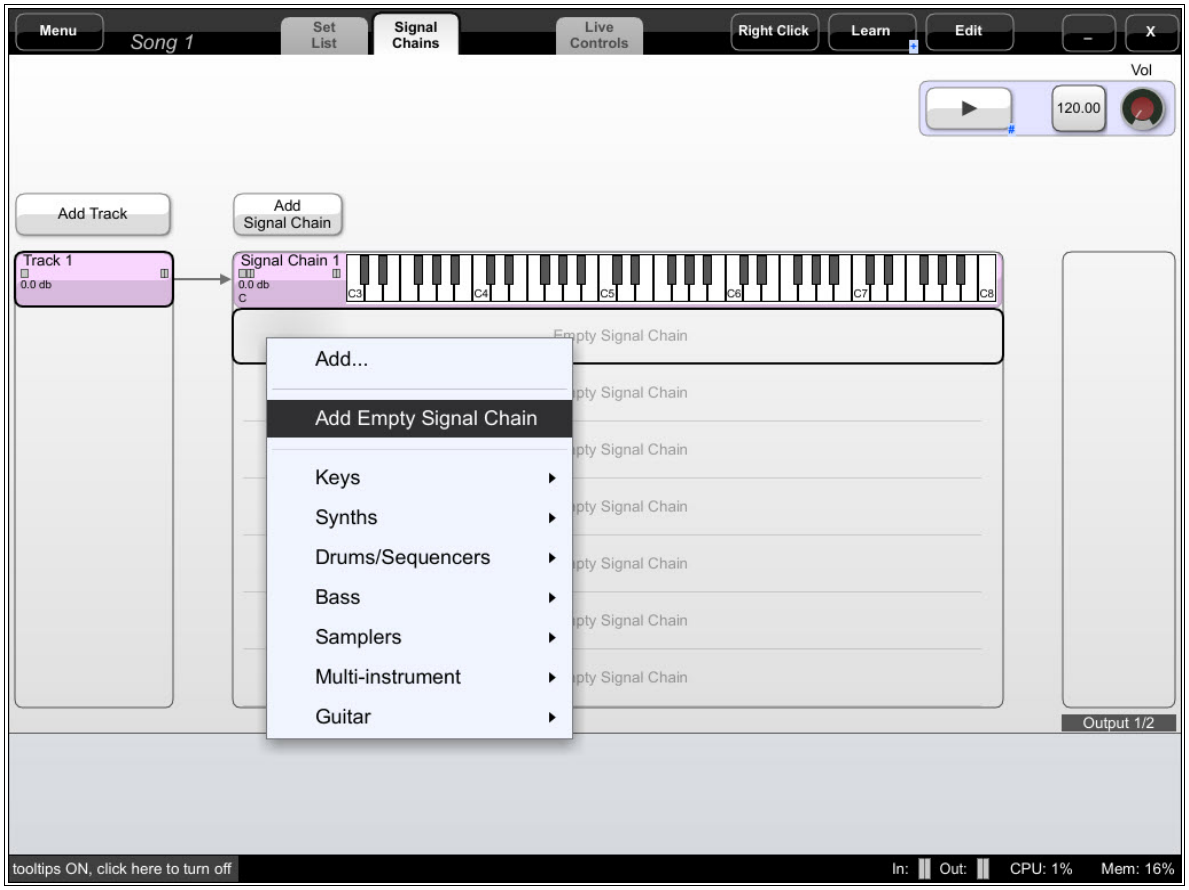
For this tutorial the instruments in the first four MIDI channels of Purity are

1. Piano MIDI channel 1
2. FM - EP MIDI channel 2
3. Vox MIDI channel 3
4. Pad MIDI channel 4

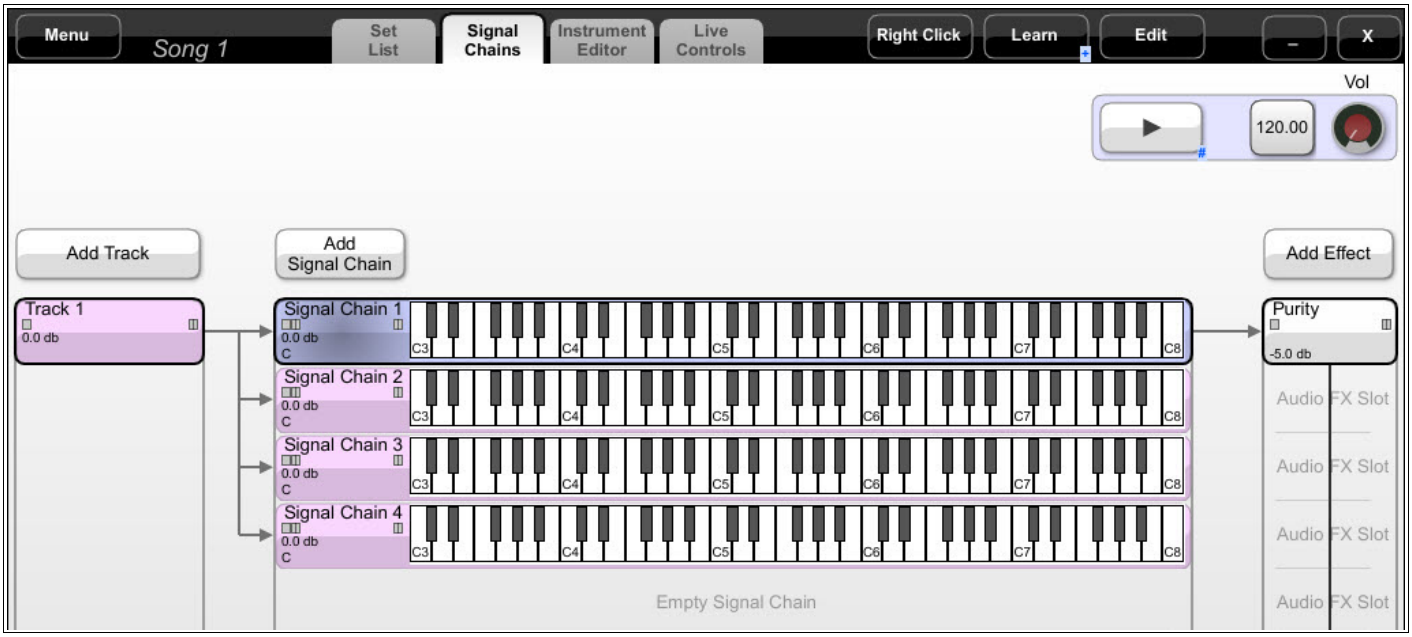
We will create splits in the signal chain to match these sounds and MIDI channels.

Go back to the Signal Chain and add three "Empty Signal Chains". Refer to the instructions and images on the next page.

To add an empty signal chain, click in the first empty signal chain slot. From the menu select [Add Empty Signal Chain]. Add three empty signal chains below "Signal Chain 1".

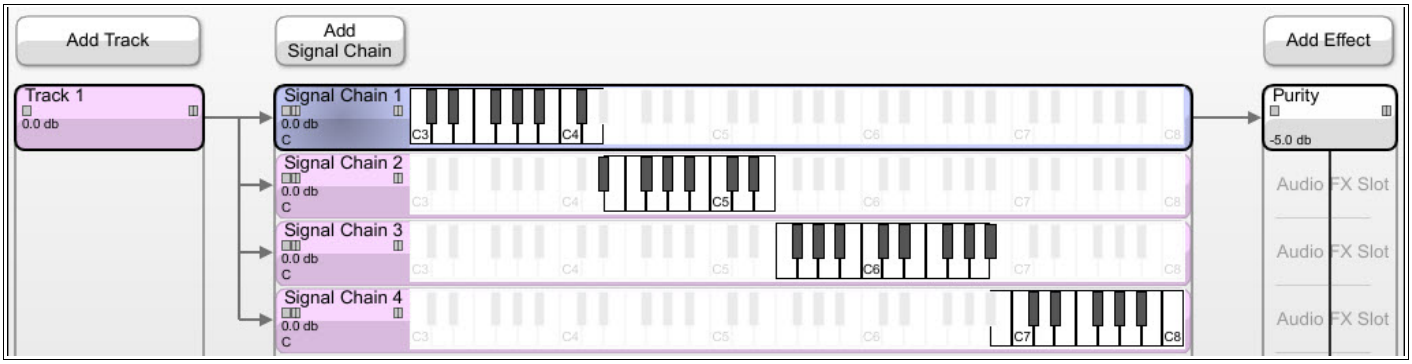


As you can see in the following image, there are three additional signal chains stemming from Signal chain 1.



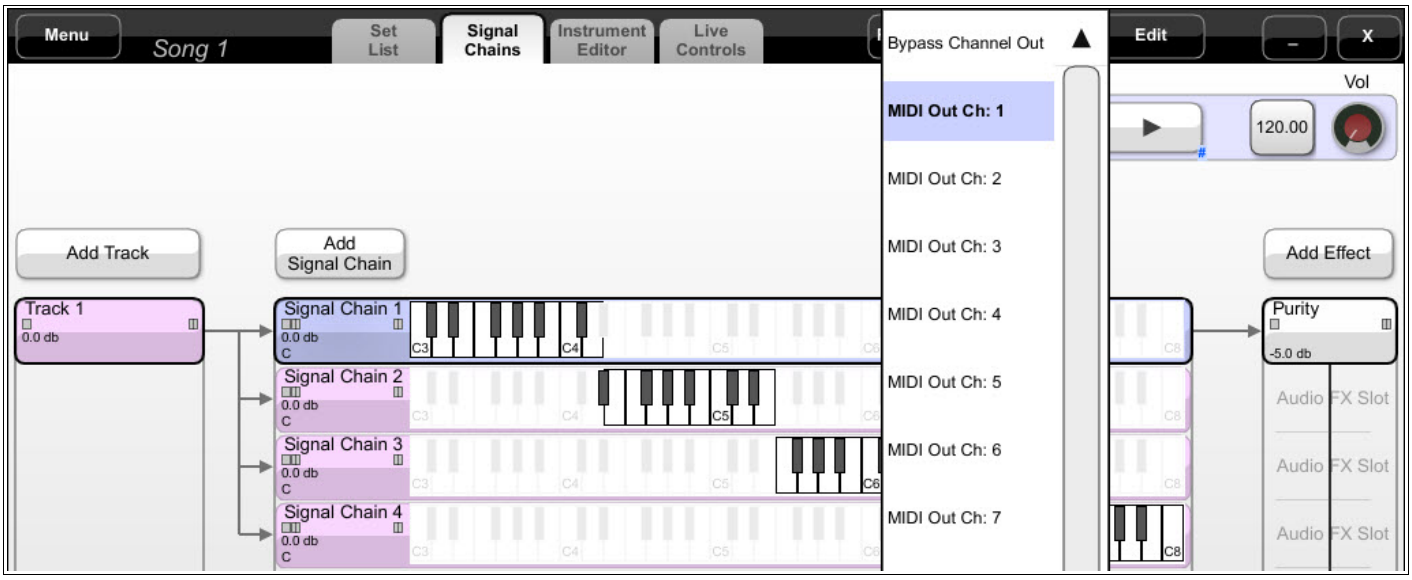
Next, the Signal Chains need to be routed to the proper MIDI channels. And the splits set up for the four chains.

Go ahead and set an array of splits similiar to this. On a smaller controller, you will need to make your splits closer together, or just use the transpose buttons to reach those octaves.



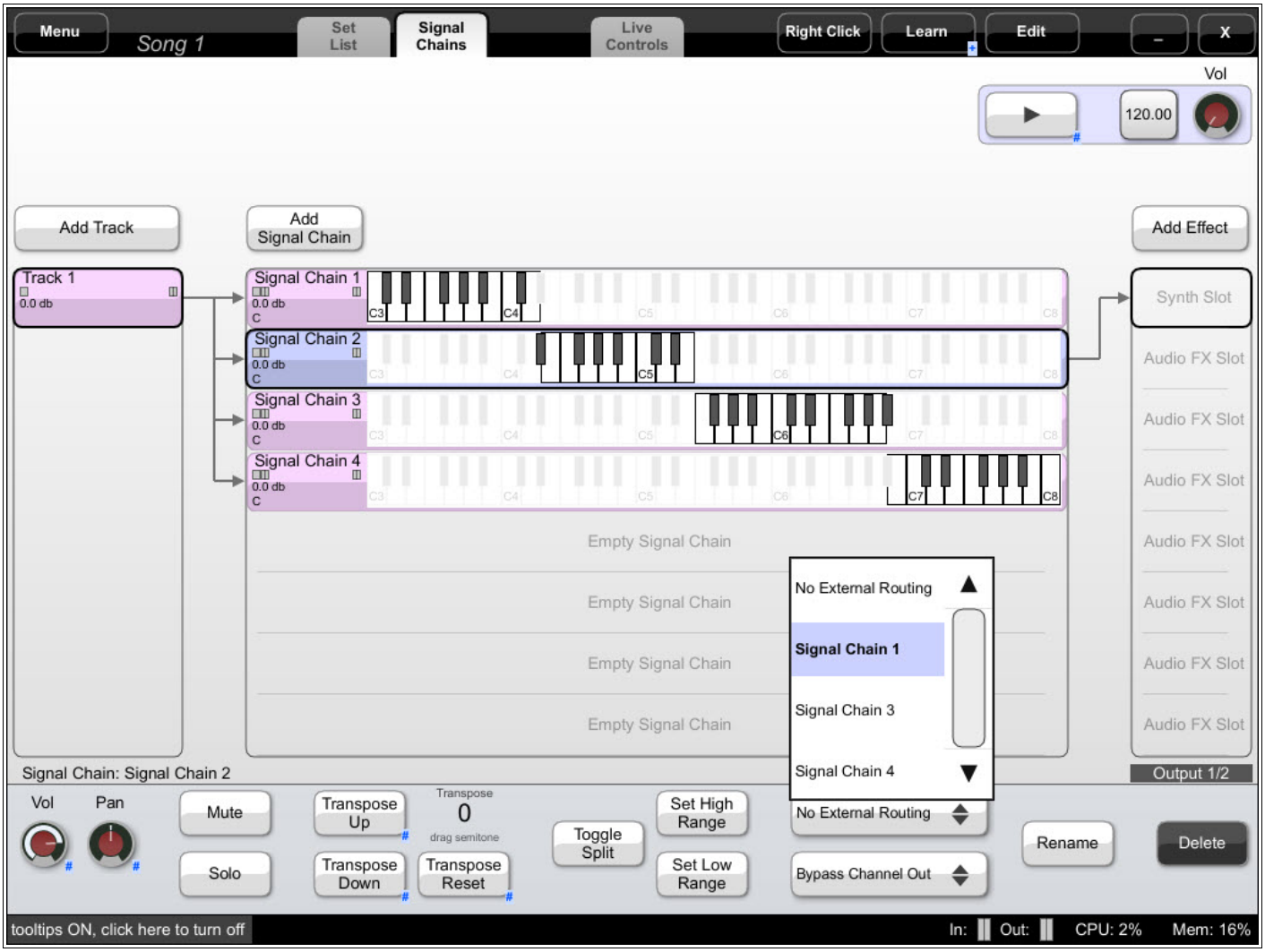
Each split needs to be set to an individual MIDI channel. Signal chains two, three, and four need to be routed to the first signal chain. Since two through four are empty signal chains they need to be routed to Signal Chain 1, this gives them access to the sounds in Purity.

Click [Bypass Channel Out] in the lower controls section and set Signal Chain 1 to MIDI out Ch: 1.

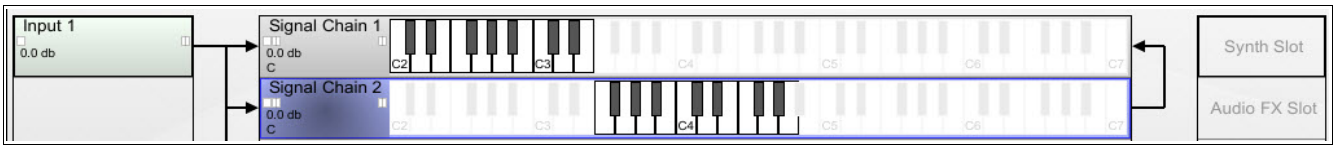


Next, signal chains two, three, and four, need to be routed through Signal Chain 1 which contains the instance of Purity.

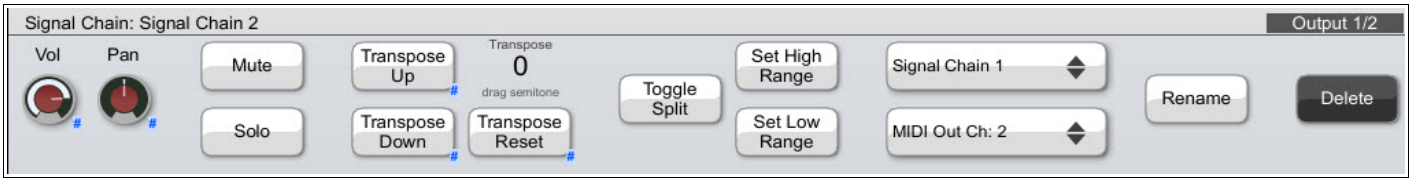
Select Signal Chain 2. Click [No External Routing] in the lower controls section and select "Signal Chain 1". And while you are there, go ahead and set the channel out to MIDI channel 2.



Once the routing has been assigned, you will see that Signal Chain 2 is routed directly through Signal Chain 1.



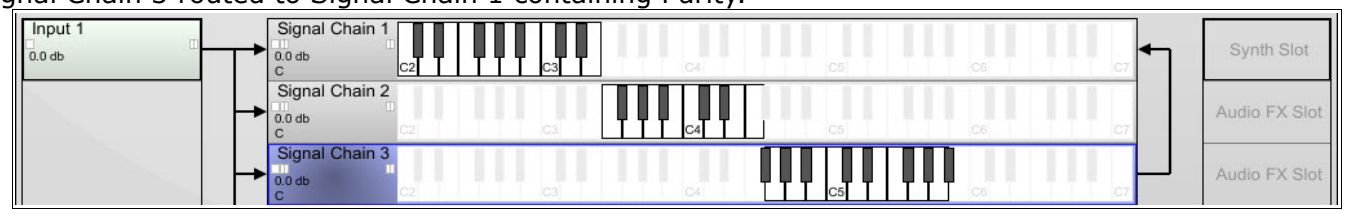
In the lower controls section of Signal Chain 2 you will also see that the routing and MIDI channel out options are properly set.



Go ahead and route signal chains three and four to Signal Chain 1, and MIDI outs to channels three and four respectively.

Follow the same procedure that was used to set Signal Chain 2.

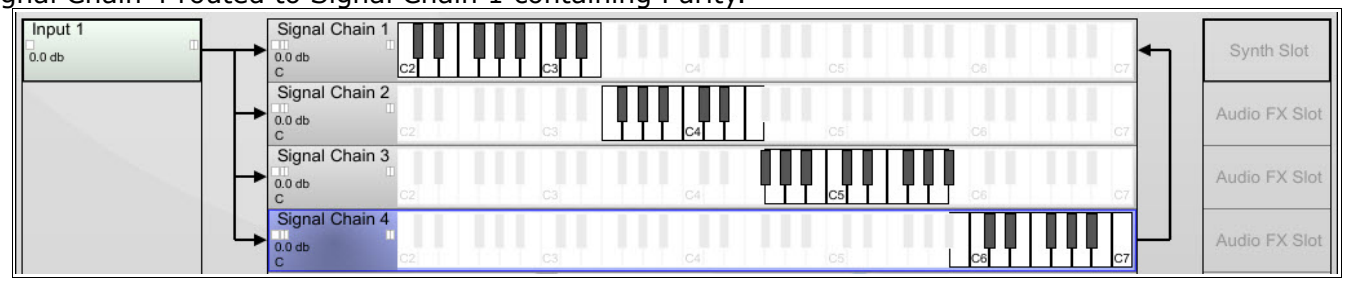
Signal Chain 3 routed to Signal Chain 1 containing Purity.



Signal Chain 3 lower controls, MIDI channel out set to channel three.



Signal Chain 4 routed to Signal Chain 1 containing Purity.



Signal Chain 4 lower controls, MIDI channel out set to channel four.



This will complete this multi-timbral setup, each split should be triggering the patch that you have in the corresponding channel of Purity.

Advanced Multi-timbral Instruments / Signal Chain

Now that you have a good idea of how to use multi-timbral instruments, it is time to learn an advanced way to setup Riff to work with your multi-timbral instruments.

As you can see in the Signal Chain, there are only eight slots . Plus, the signal chains are made for layering sounds, and creating keyboard splits.

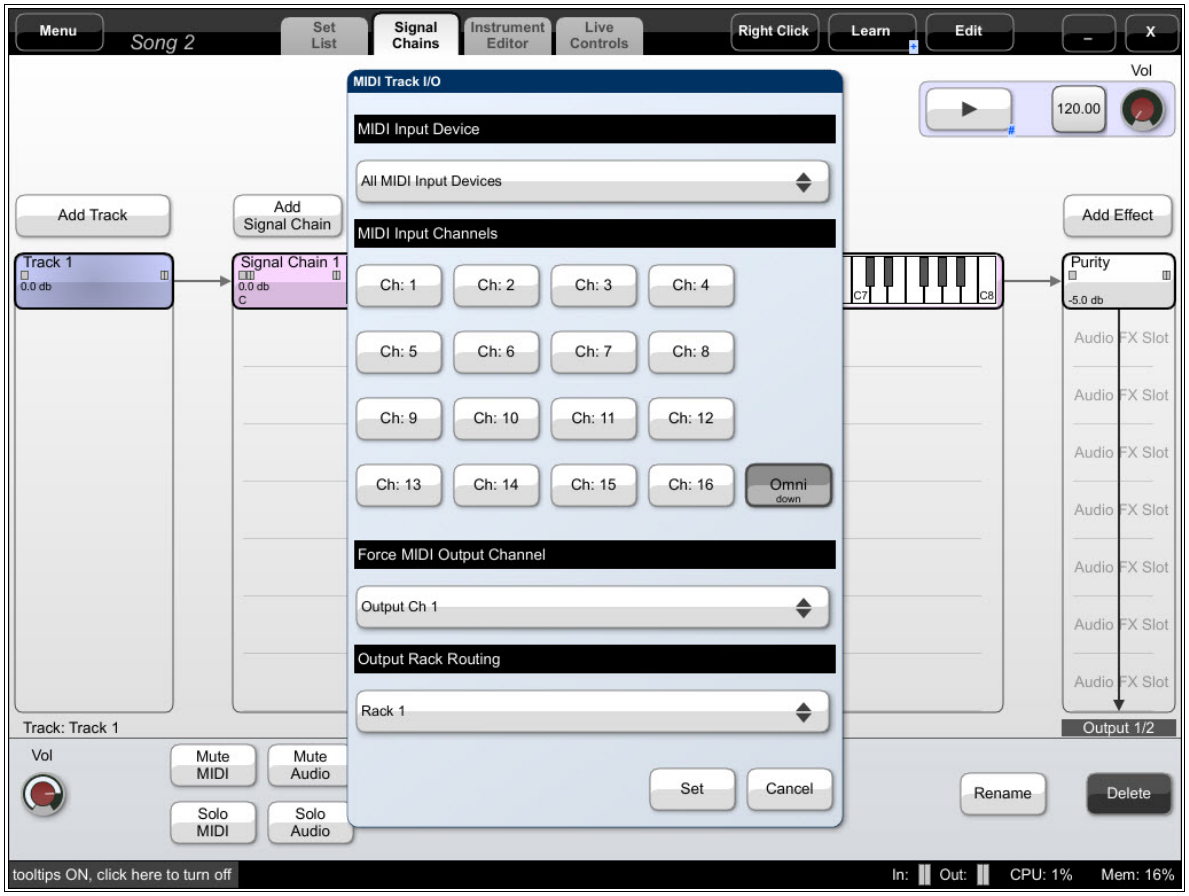
Let's say you wanted to use all 16 MIDI channels of a multi-timbral instrument in one Song, as opposed to setting up 16 songs each using a different MIDI channel.

The earlier tutorial explained how to do it that way. However, it is good to know the different ways to setup instruments in Riff.

Setup a new Song, and again, add Purity to the Song.

Remember to unlink the MIDI channels in Purity. Go head and assign sounds to some channels.

Go to the signal chain, select the "Input" for Signal Chain 1. This Input needs to only use MIDI channel 1. The input can be forced to use a particular MIDI channel by pressing the [Input Output] button located in the lower controls section. Do this and the "MIDI Input I/O" option box appears. Set "Force MIDI Output Channel" to "Output Ch 1".
Then press [Set].



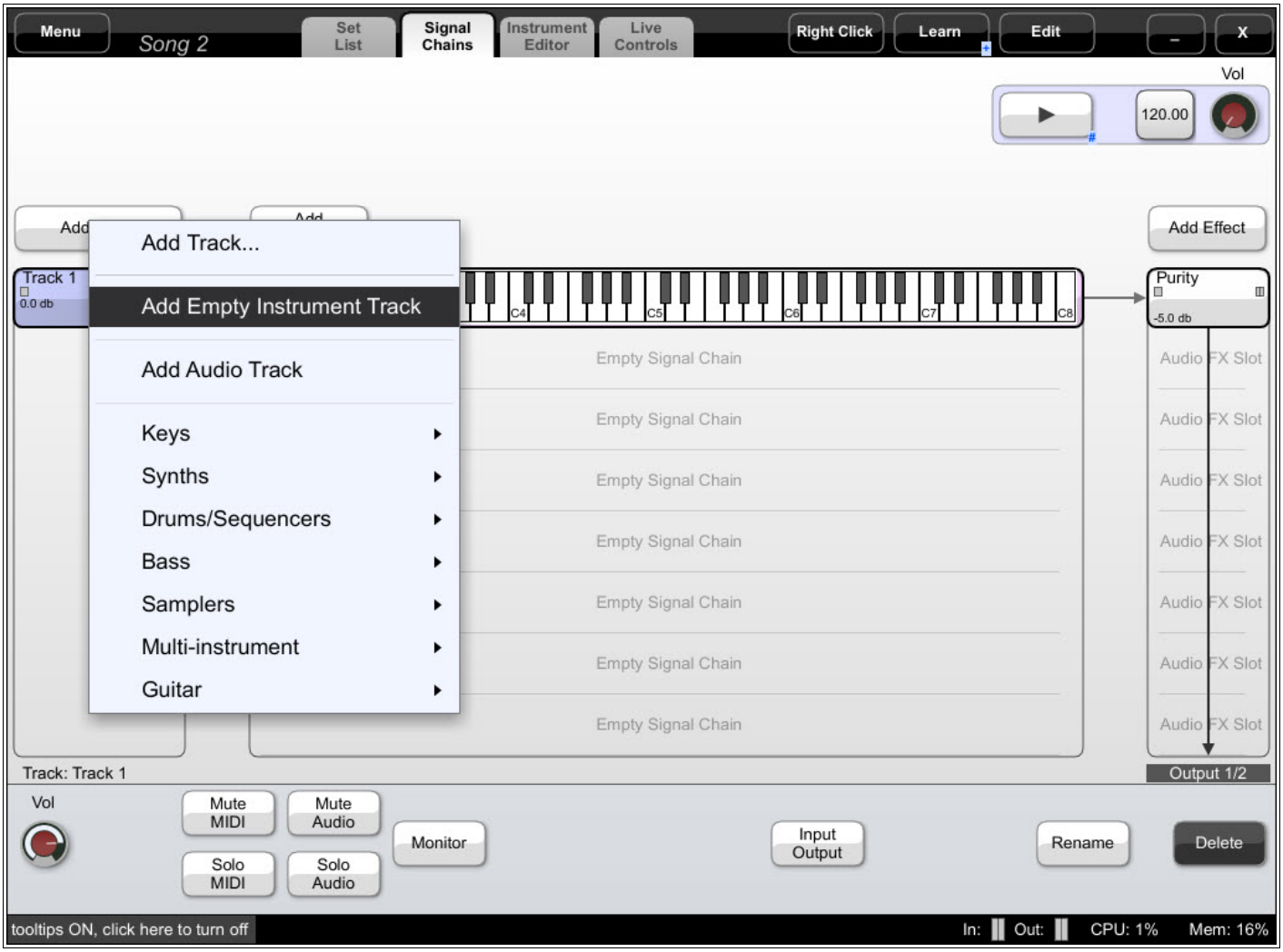
This forces the input to only use MIDI channel 1.

Since you can add virtually unlimted inputs to the Input Column, it is the ideal section to setup a multi-timbral instrument in a song.

Every additional input we add to this song will be setup to use the instance of Purity in the first input slot. Again this will save lots of system resources.

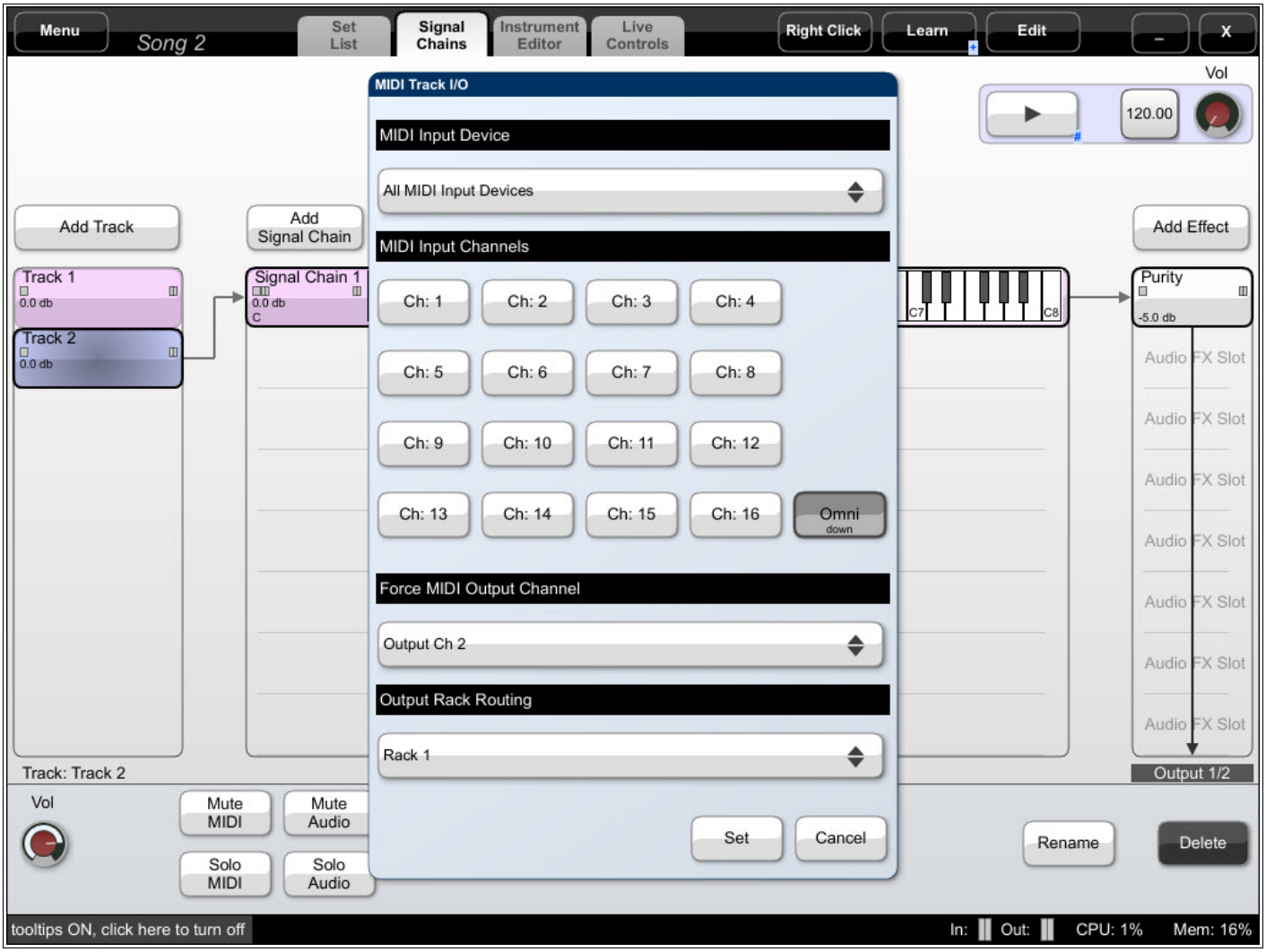
Now it is time to add some inputs to use for accessing the MIDI channels used in the first inputs instance of Purity.

Click the [Add Input] button and select [Add Empty Instrument Input].

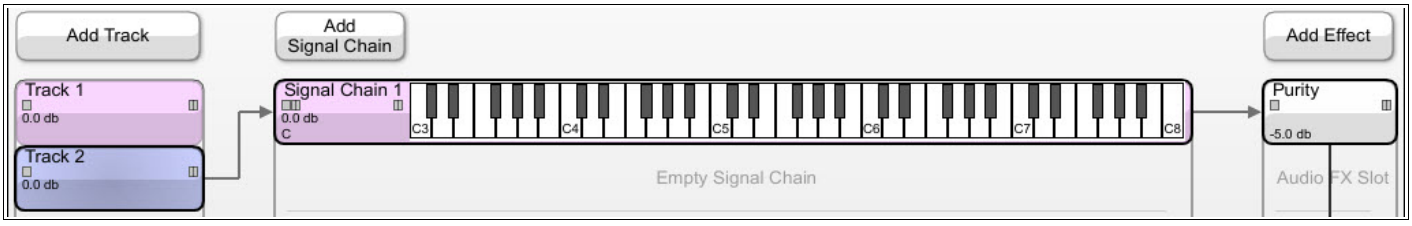


With Input 2 selected, click the [Input Output] button located in the lower controls section. To allow this input to access the MIDI channel 2 of the Purity instance in Input 1, select the following options.

Under "Output Rack Routing", select Rack 1, and under "Force MIDI Output Channel", select "Output Ch 2". Then press [Set].

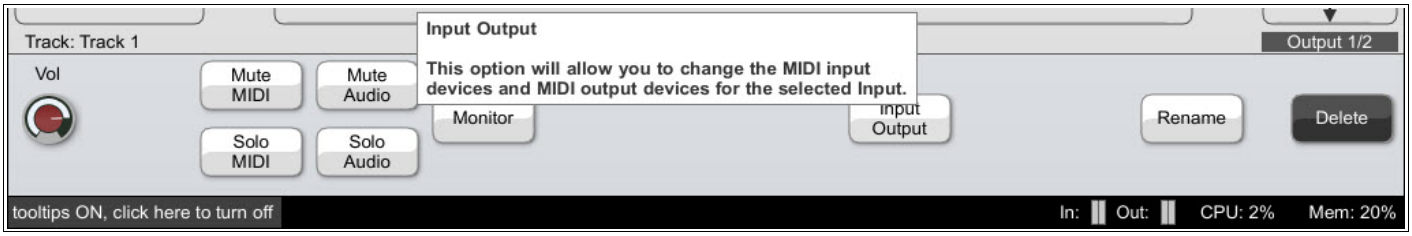


The following image shows that Input 2 is routed to the Signal Chain 1 instance of Purity. Now, whatever sound you add to MIDI channel 2 of Purity will be triggered by Input 2.

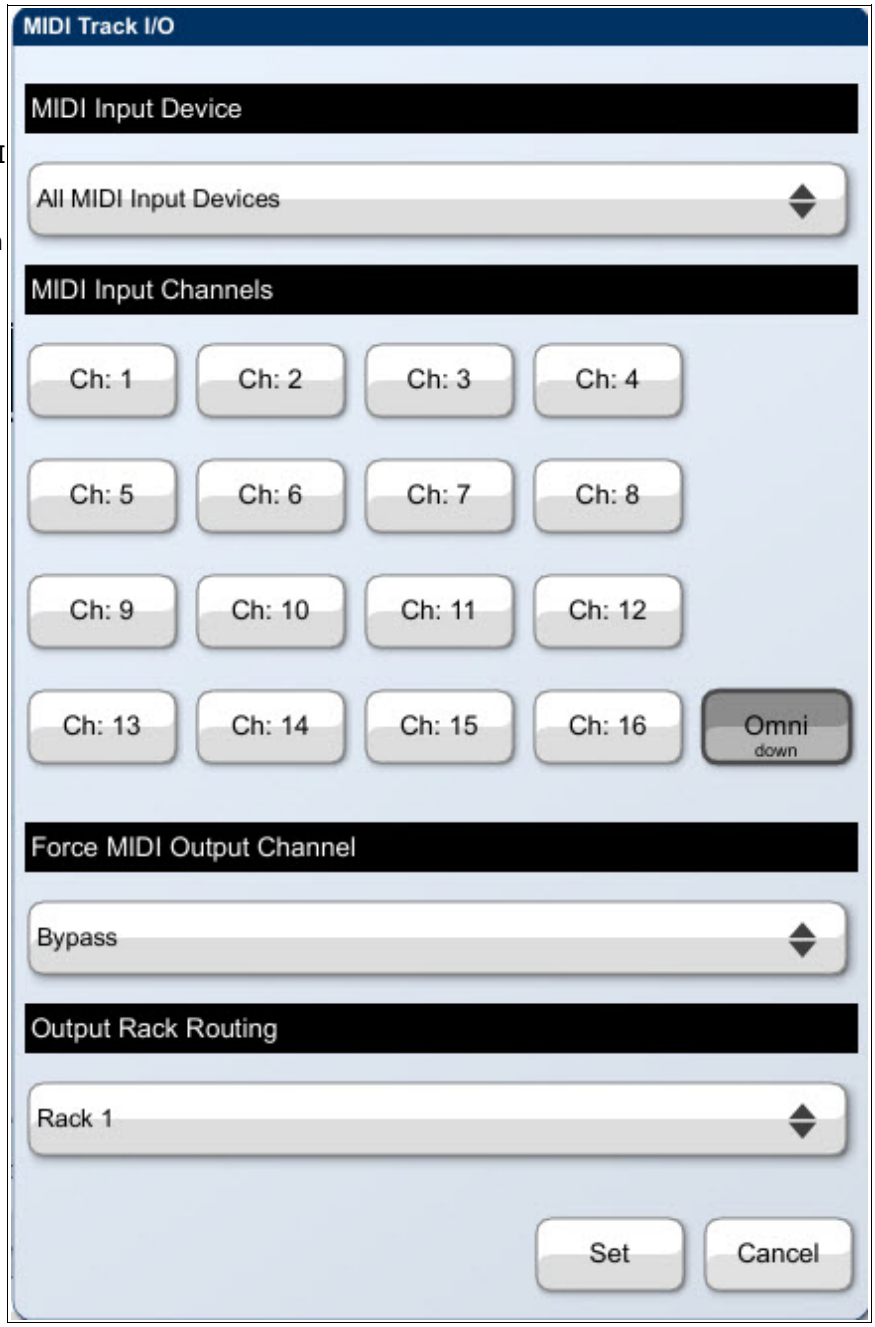


From here you should be able to add more Inputs and experiment with the settings.

MIDI Input I/O Options Window



Pressing the [Input Output] button in the lower controls section brings up the "MIDI Input I/O" window.



MIDI Track I/O:
Various options for per Input MIDI routing.

- *MIDI Input Devices* - Whatever MIDI input devices you enabled in the main MIDI setup options for Riff will be present here. You can also set an MIDI-in for a connected external MIDI controller to play the sound in the selected input.
- *MIDI Input Channels* - Choose the MIDI Input Channel for the selected Input.
- *Force MIDI Output Channel* - This option enables the selected input to send MIDI out on a particular MIDI channel.

Output Rack Routing:
Allows you to pick which Rack the selected Input should route to.

Using Different Preset Sounds with a Shared Instrument

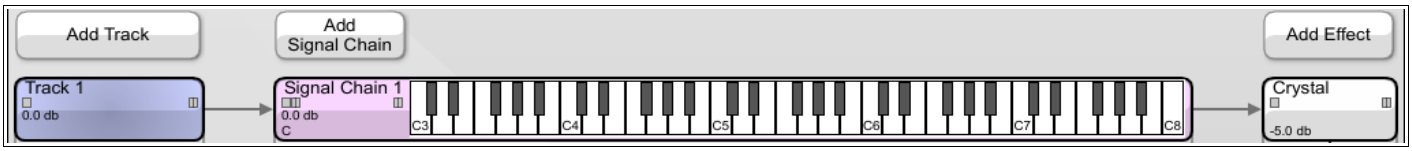
If you are sharing an instrument over multiple Songs, you will more than likely want to use different presets in each song.

For instance, if you are using the patch "-Atmospehere-" from the plug-in Crystal in Song one, but in Song two you also want to have Crystal using a different patch. There is no need to add a brand new instance of Crystal.

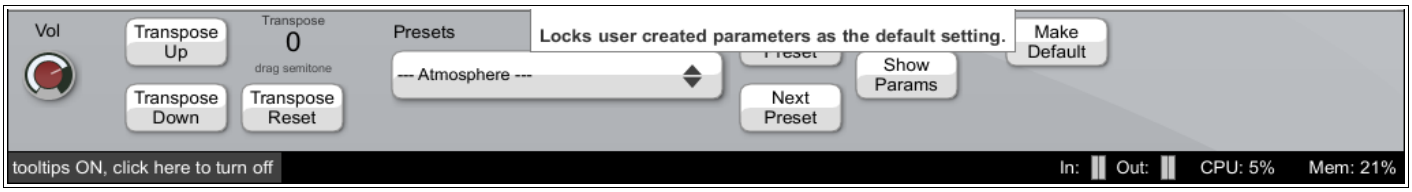
This next tutorial will show you how to do this.

First, start a new song and add the instrument Crystal. Jump over to the Signal chain.

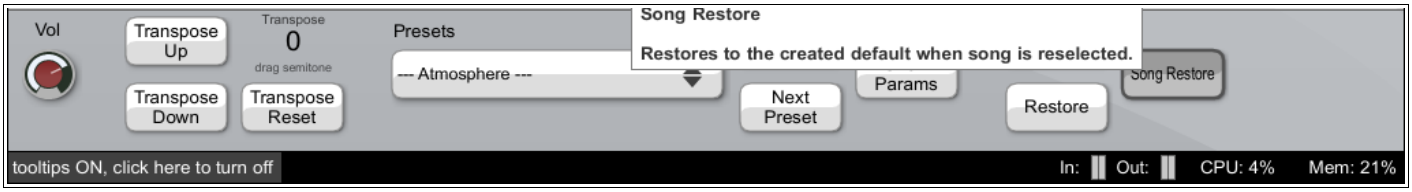
This is what a Song looks like that has not been shared.



Select the Instrument Editor of Song one. Make sure you are on the patch "-Atmospehere-". Next, press [Make Default] in the lower controls section of the Instrument Editor.



Then press the Song Restore button.



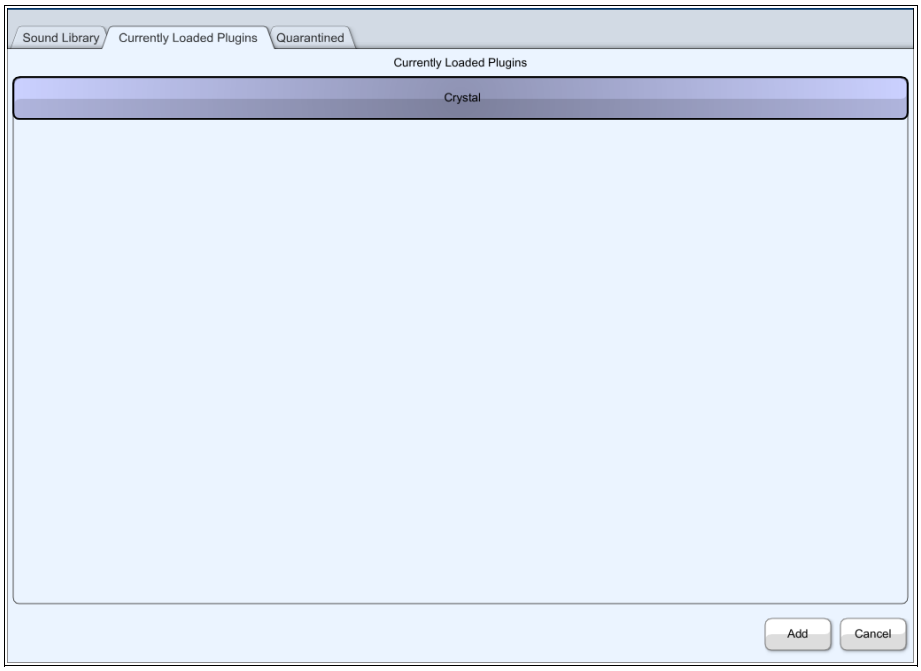
Doing this locks Song one to the preset "-Atmospehere-".

Now, select a spot directly beside the first Song on your Setlist and click "Add".

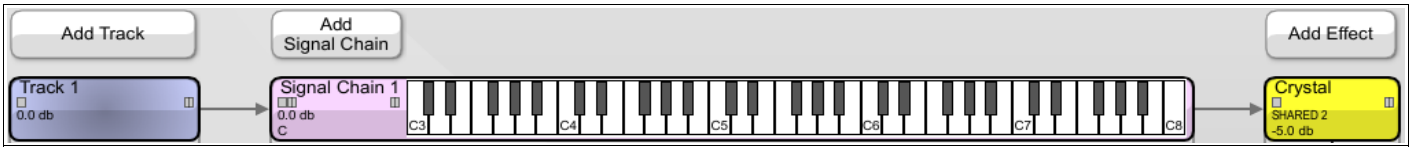
Once you reach the "Sound Browser" click the [Currently Loaded Plugins] tab. Here you will see your list of loaded instruments.

Instead of loading a new instance of Crystal, select the currently loaded Crystal and either press the [Add] button or double left-click the instrument.

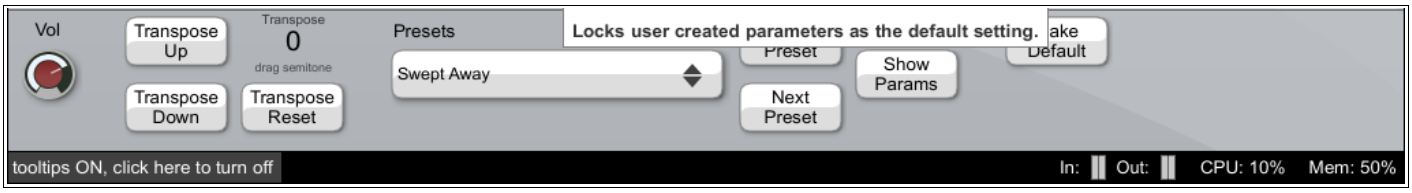
This is how you share an instrument in Riff.



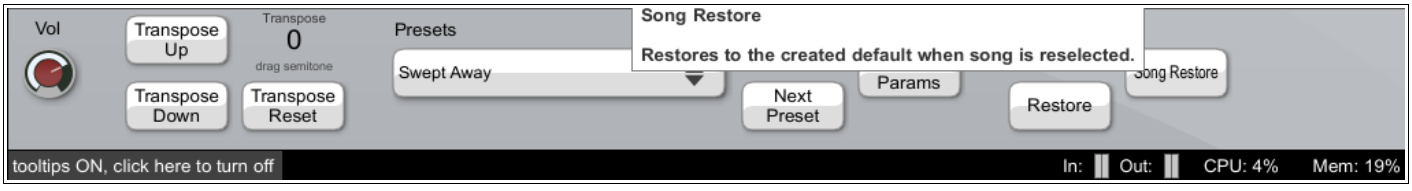
The next picture shows you what an Instrument looks like after "Shared" has been enabled. The Crystal icon in the Stream Processor is now yellow. This represents an instrument is shared.



Go ahead and jump over to the Instrument Editor of the second song and set Crystal to the patch "Swept Away". Next, press [Make Default] in the lower controls section of the Instrument Editor.



Then press the Song Restore button.



Doing this locks Song two to the preset Swept Away.

Once you reselect Song one, it's locked in patch will recall. Song two will behave the same way.

That is all you have to do to have multiple Songs share the same instrument, while still being able to have a different sound from that instrument in each Song.

Here is a quick recap of this section.

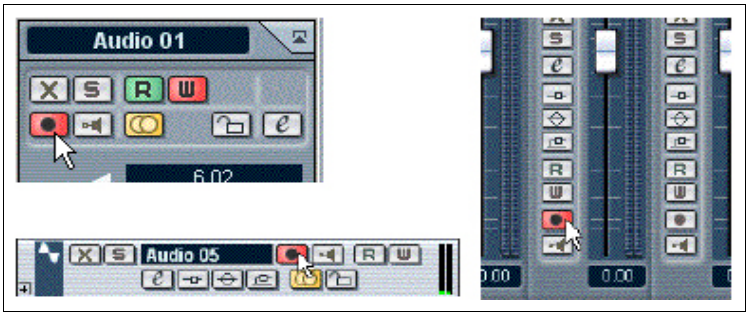
1. Create a Song with any VSTi.
2. Go into the Instrument Editor and find a patch you like, then hit [Make default], then hit [Song restore].
3. Add a new Song, once you reach the "Sound Browser" click the [Currently Loaded Plugins] tab. Here you will see your list of loaded instruments. Instead of loading a new instance of an instrument, select the instrument that is to be shared, and either press the [Add] button or double left-click the instrument.
4. Go into the Instrument Editor of the new Song, find a new preset, Then hit the [Make Default] button. After you hit the [Make Default] press the [Song Restore] button in this song.
5. Now you can change songs while getting different sounds from the same plug-in.

5

Chapter 5 – SoundSlate as a Hardware DAW - Cubase LE

SoundSlate ships with Cubase LE, the following documents were extracted from the manual. You can find the manual in its entire form at: C:\Program Files\Steinberg\Cubase\Help
Cubase LE can record on a single track or on several tracks (audio and/or MIDI) simultaneously. To make a track ready for recording, click the Record Enable button for the track in the Track list, in the Inspector or in the Mixer. When activated, the button(s) turn red, indicating record ready mode.

Record enabling a track



Record Enable in the Inspector, Track list and Mixer.

- If the option "Enable Record on Selected Track" is activated in the Preferences dialog (Editing page), tracks are automatically record enabled when you select them in the Track list.
- You can have a maximum of 48 audio tracks in a project. However, the exact number of audio tracks you can record simultaneously depends on your computer CPU and hard disk performance. Furthermore, it would be pointless to record more audio tracks than you have audio inputs, since this would only result in duplicate tracks and audio files.

Manually activating recording:

You activate recording by clicking the Record button on the Transport panel or toolbar, or by using the corresponding key command (by default [*] on the numeric keypad).
Recording can be activated from Stop mode (from the current cursor position or from the left locator) or during playback:

- If you activate recording from Stop mode, and the option "Start Record at Left Locator" is activated on the Transport menu, recording will start from the left locator. The preroll setting or the metronome count-in will be applied (see About Preroll and Postroll).
- If you activate recording from Stop mode, and the "Start Record at Left Locator" is deactivated, recording will start from the current project cursor position.
- If you activate recording during playback, Cubase LE will immediately enter Record mode and start recording at the current project cursor position. This is known as "manual punch in".

Automatically activating recording:

Cubase LE can automatically switch from playback to recording at a given position. This is known as "automatic punch in". A typical use for this would be if you need to replace a section of a recording, and want to listen to the previously recorded audio up to the recording start position.

1. Set the left locator to the position at which you want recording to start.
2. Activate the Punch In button on the Transport panel. Punch In activated.
3. Activate playback from some position before the left locator.

When the project cursor reaches the left locator, recording is automatically activated.

Stopping recording

Again, this can be done automatically or manually:

- If you click the Stop button on the Transport panel (or use the corresponding key command, by default [0] on the numeric keypad), recording is deactivated and Cubase LE goes to Stop mode.
- If you click the Record button or use the key command for recording, by default [*], recording is deactivated but playback continues. This is known as "manual punch out".
- If the Punch Out button is activated on the Transport panel, recording will be deactivated when the project cursor reaches the right locator. This is known as "automatic punch out". By combining this with automatic punch in, you can set up a specific section to record - again very useful if you want to replace a certain part of a recording.



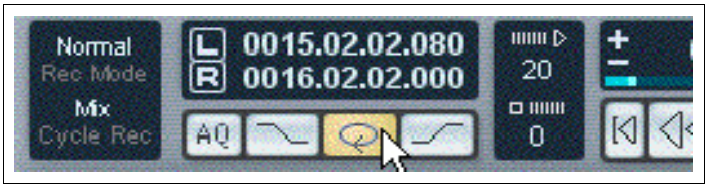
Punch In and Out activated.

Cycle Recording

Cubase LE can record and play back in a cycle - a loop. You specify where the cycle starts and ends by setting the left and right locators. When cycle is active, the selected section is seamlessly repeated until you hit Stop or deactivate cycle mode.

- To activate cycle mode, click the cycle button on the Transport panel.

If you now activate Play, the section between the left and right locator is repeated indefinitely until you stop.



Cycle activated.

- To record in Cycle mode, you can start recording from the left locator, from before the locators or from within the cycle, from Stop mode or during playback. As soon as the project cursor reaches the right locator, it will jump back to the left locator and continue recording a new lap.

Activating and selecting VST inputs

Cubase LE allows you to use audio hardware with multiple inputs and route different inputs to different audio channels. To activate inputs, first open the VST Inputs window on the Devices menu.

Note that active inputs use processing power! Make it a habit to only activate audio inputs that you actually intend to use.



The VST Inputs window is divided into three columns:

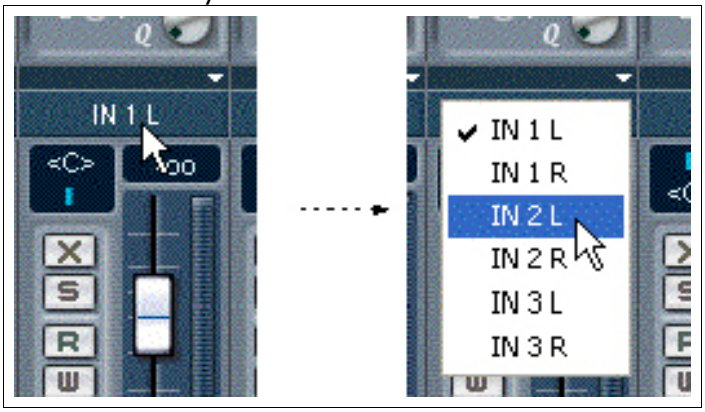
- The left column contains the available physical input ports on the audio hardware.
- The middle column indicates which input pairs are activated. You turn input pairs on or off by clicking the buttons in this column.
- The right column shows the names that will be used for each input throughout the program. You can rename an input by clicking in this column and typing a new name.

If you deactivate an input pair that is currently used (selected as input source for one or several audio channels), you will be asked whether you want Cubase LE to remap these inputs (change the input source selection for the channels in question).

Routing activated VST inputs to a channel

Selecting an input source for a track's corresponding channel is done in the Mixer. Proceed as follows:

1. Open the Mixer from the Devices menu.
2. Locate the mixer channel strip for the audio track on which you plan to record.
3. Pull down the input pop-up for the channel strip and select the input to which the signal source you want to record is connected.



Selecting a recording file format

The format of recorded files is set in the Project Setup dialog on the Project menu. There are three settings: sample rate, record format (bit depth) and record file type. While the sample rate is set once and for all when you start working on a new project, the bit depth and file type can be changed at any time.

Record format (bit depth)
The available options are 16 bit and 24 bit. Use the following guidelines:

- Normally, select record format according to the bit depth delivered by your audio hardware.

For example, if your audio hardware has 20 bit A/D converters (inputs), you may want to record at 24 bit resolution, to capture the full bit depth. On the other hand, if your hardware has 16 bit inputs, it's pointless to record with a higher bit depth - this will only make the audio files larger, with no difference in audio quality.

- The higher the bit depth, the larger the files and the more strain is put on the disk system.

If this is an issue, you may want to lower the record format setting.

Record file type

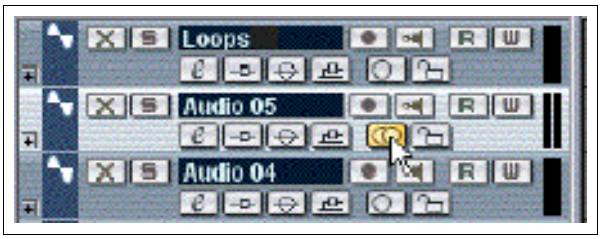
The Record File Type setting determines which type of files will be created when you record:

File Type	Description
Wave File	Wave files have the extension ".wav" and are the most common file format on the PC platform.
AIFF File	Audio Interchange File Format, a standard defined by Apple Computer Inc. AIFF files have the extension ".aif" and are used on most computer platforms.

Setting up tracks for mono or stereo recording

One of the initial decisions you have to make before you start recording audio is whether the recording should be stereo or mono. This is determined by the stereo/mono status of the audio track selected for recording:

- To set a track to mono or stereo, click the Stereo/Mono button in the Track list or in the Inspector. A lit stereo button indicates a stereo track, while a dark mono button indicates a mono track.



The Stereo/Mono button.

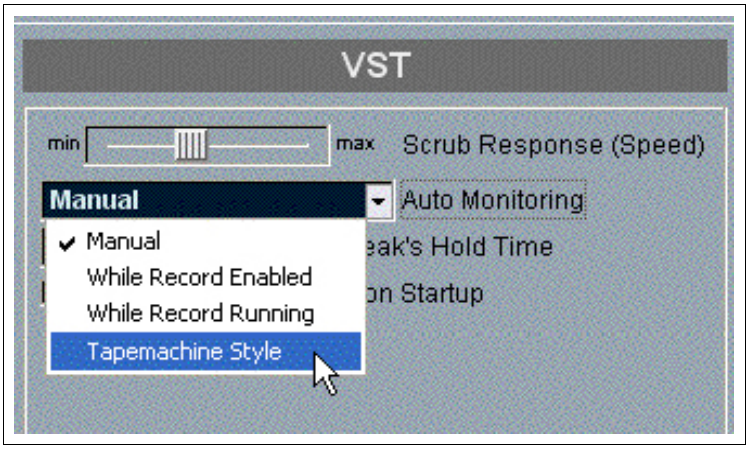
Monitoring

In this context, "monitoring" means listening to the input signal during recording. There are three fundamentally different ways to do this: via Cubase LE, externally (by listening to the signal before it reaches Cubase LE) or by using ASIO Direct Monitoring (which is a combination of both of the other methods - see below).

Monitoring via Cubase LE

If you monitor via Cubase LE, the input signal is mixed in with the audio playback. The advantage of this is that you can adjust the monitoring level and panning in the Mixer, and add effects and EQ to the monitor signal just as during playback. The disadvantage of monitoring via Cubase LE is that the monitored signal will be delayed according to the latency value (which depends on your audio hardware and drivers). Therefore, monitoring via Cubase LE requires an audio hardware configuration with a low latency value.

- *Manual*. This option allows you to turn input monitoring on or off by clicking the Monitor button in the Inspector, the Track list or in the Mixer.
- *While Record Enabled*. With this option you will hear the audio source connected to the channel input whenever the track is record enabled.
- *While Record Running*. This option switches to input monitoring only during recording.
- *Tapemachine Style.* This option emulates standard tapemachine behavior: input monitoring in stop mode and during recording, but not during playback.



External monitoring

External monitoring (listening to the input signal before it goes into Cubase LE) requires some sort of external mixer for mixing the audio playback with the input signal. This can be a stand-alone physical mixer or a mixer application for your audio hardware, if this has a mode in which the input audio is sent back out again (usually called "Thru", "Direct Thru" or similar).

When using external monitoring, you cannot control the level of the monitor signal from within Cubase LE, or add VST effects or EQ to the monitor signal. The latency value of the audio hardware configuration does not affect the monitor signal in this mode.

- If you want to use external monitoring, you need to make sure that monitoring via Cubase LE isn't activated as well. Select the "Manual" monitoring mode in the Preferences dialog (VST page) and simply don't activate the Monitor buttons.

ASIO Direct Monitoring

If your audio hardware is ASIO 2.0 compatible, it may support ASIO Direct Monitoring. In this mode, the actual monitoring is done in the audio hardware, by sending the input signal back out again. However, monitoring is controlled from Cubase LE. This means that the audio hardware's direct monitoring feature can be turned on or off automatically by Cubase LE, just as when using internal monitoring.

- To activate ASIO Direct Monitoring, open the Device Setup dialog on the Devices menu and use the Direct Monitoring checkbox on the Setup tab for the VST Multitrack device. If the checkbox is grayed out, this means that your audio hardware (or its current driver) doesn't support ASIO Direct Monitoring. Consult the audio hardware manufacturer for details.
- When ASIO Direct Monitoring is activated, you can select a monitoring mode in the Preferences dialog (VST page), as when monitoring via Cubase LE (see above).
- Depending on the audio hardware, it may also be possible to adjust monitoring level and panning from the Mixer.
-

Consult the documentation of the audio hardware if in doubt.

- VST effects and EQ cannot be applied to the monitor signal in this mode, since the monitor signal doesn't pass through Cubase LE.
- Depending on the audio hardware, there may be special restrictions as to which audio outputs can be used for direct monitoring. For details on the routing of the audio hardware, see its documentation.
- The latency value of the audio hardware configuration does not affect the monitor signal when using ASIO Direct Monitoring.

Setting input levels

When monitoring a channel signal source in stop mode, the meters show the level at the input selected for the audio channel. So if the connected signal source is sounding you should see activity in both the Track list meter and in the mixer strip for that channel.
Note that it is not possible to set input gain with the mixer fader!

Check the levels coming in to Cubase LE, and if necessary, adjust the input level in one of the following ways:

- Adjust the output level of the sound source or external mixer.
- Use the audio hardware's own application program to set the input levels, if this possibility is provided. See the documentation of the audio hardware.
- If your audio hardware supports the ASIO Control Panel function, it may be possible to make input level settings. To open the ASIO Control Panel, open the Device Setup dialog on the Devices menu and click the Control Panel button on the Setup tab for the VST Multitrack device.
-

Input levels should be as high as possible, without exceeding 0dB.

Recording

Recording is done using any of the general recording methods. After you finish recording, an audio file has been created in the Audio folder within the projects folder. In the Pool, an audio clip is then created for the audio file, and an audio event that plays the whole clip appears on the recording track. Finally, a waveform image is calculated for the audio event. If the recording was very long, this may take a while.

- If the option "Create Images During Record" is activated in the Preferences dialog (Audio page), the waveform image will be calculated and displayed during the actual recording process. This should only be activated if you have a fairly powerful computer system, since the real-time calculation uses some extra processing power.

Undoing recording

If you decide that you don't like what you just recorded, you can delete it by selecting Undo from the Edit menu. The following will happen:

- The event(s) you just created will be removed from the Project window.
- The audio clip(s) in the Pool will be moved to the Trash folder.
- The recorded audio file(s) will not be removed from the hard disk. However, since their corresponding clips are moved to the Trash folder, you can delete the files by opening the Pool and selecting "Empty Trash" from the Pool menu.

About overlap and audio tracks

If you record again where something has already been recorded, you get a new event that overlaps the previous one(s). When you play back, only the events that are actually visible are played back.

One audio track can only play back one audio event at a time.
For more advance recording information you can find the manual in its entire form at: C:\Program Files\Steinberg\Cubase\Help

Or visit Steinberg’s official forums. http://www.steingberg.de/20_1.html

5.1

Chapter 5.1 – Steinberg Cubase LE and the Emu Digital Audio System

Steinberg Cubase LE is a 24-bit/96kHz multi-track audio/MIDI recorder with high-quality effects, automation, virtual instruments (VSTi) and many other professional features.

The following step-by-step tutorials are designed to get you recording as quickly as possible. After you have finished the tutorial we encourage you to read the Cubase LE pdf manual in order to learn about the many features of this powerful program. We have provided a PatchMix DSP Session, as well as a special Cubase LE Project to go along with the following tutorials.

Setting up Cubase LE

Follow these instructions carefully to ensure that Cubase LE runs smoothly the first time. Cubase LE will remember these settings, so you'll only have to do this once.

1. Open Cubase LE from the Start menu or by double-clicking the desktop icon. An ASIO multimedia driver test dialog box will pop up to ask if you want to run the ASIO test. Choose No, because you won't be using the driver anyway.
2. Select Device Setup… from the Devices menu.
3. Select VST Multitrack from the list and the following screen appears.

4. In the Setup section of the dialog box, select E-MU ASIO as the ASIO Driver. A pop up dialog box asks, "Do you want to switch the ASIO driver?" Select **Switch**.
5. Click the **Control Panel button**. The pop-up dialog box shown at right appears.
6. Set the **ASIO Buffer Latency** to 10 ms or less. A low latency setting is important to assure fast MIDI response time and to minimize delay when monitoring through Cubase. Click OK.
7. Make sure the Direct Monitoring box is Unchecked.
8. Close the Device Setup screen by selecting **OK**.
9. Close **Cubase LE** by clicking the button. (Close the application even if you're going to continue with the tutorials, in order to "lock-in" your Preference settings.)

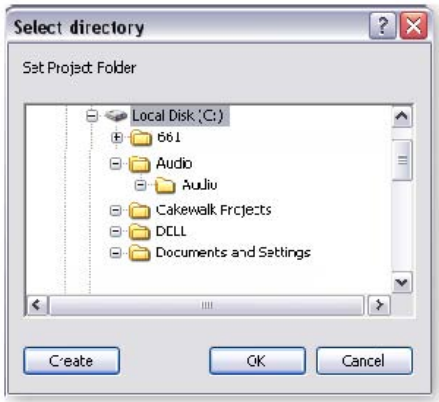


Basic Multitrack Recording

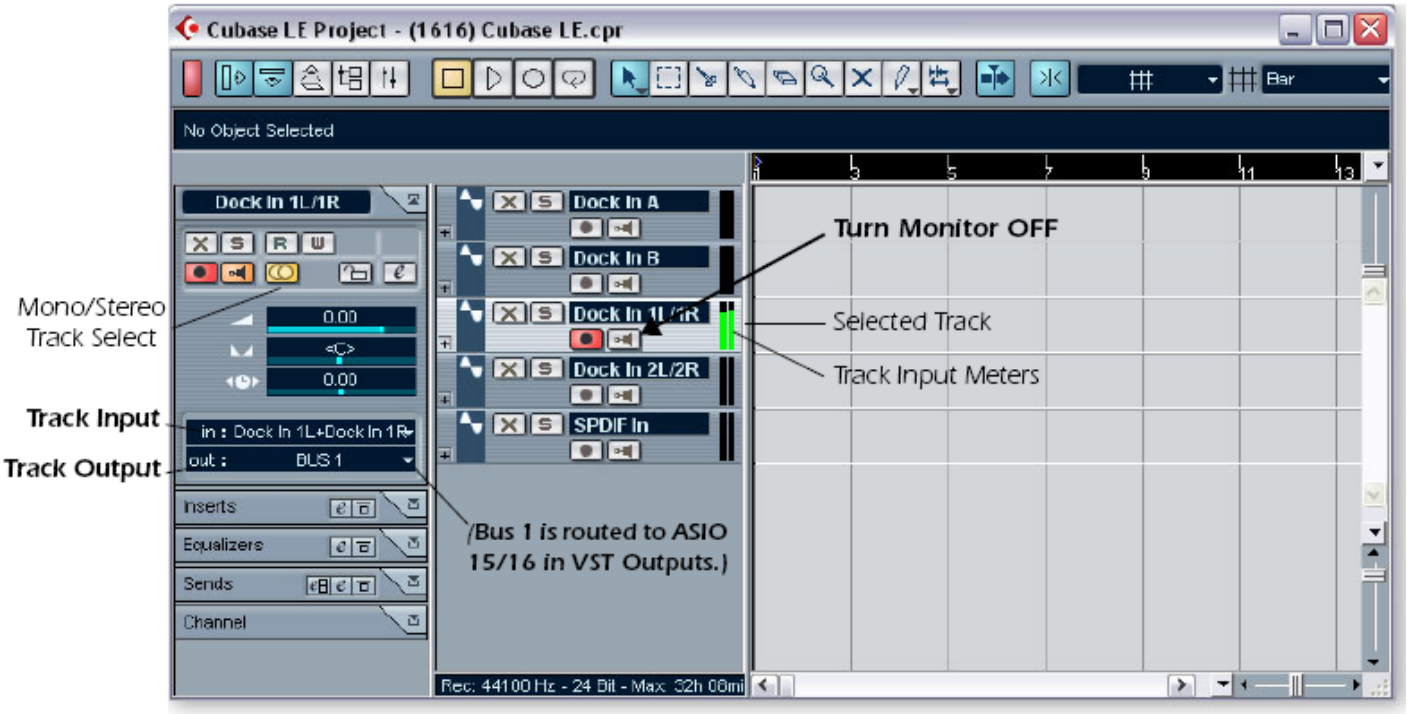
In this tutorial, we'll assume you're using a single input or pair of inputs. For more advanced recording, refer to the Cubase LE manual.

Select the Cubase LE Recording Template

1. Choose (1616) **Cubase LE** or (1212) **Cubase LE** from the **Start -> Programs** menu. The path is: (Program Files\Creative Professional\E-MU PatchMix DSP\VST\Recording Templates). Cubase LE will launch and load the Project.
2. A pop-up dialog box will appear, asking if you to **Select Directory**. Choose a where you wish to store your audio files, then click **OK**.
3. The Cubase LE Project window shown below appears. Five empty tracks are ready to record. Each track is already connected to an input or pair of inputs in PatchMix.
4. **Select the Track** that matches the inputs you've connected to the Digital Audio System by clicking on it. Note that the selected track brightens as you select it. In the example below, we are using Line Inputs 1L/1R.
5. **Turn the Monitor button OFF** for the track. You will be monitoring the input through PatchMix DSP which has zero-latency.
6. If you want to record in mono, set the track to mono by clicking the Mono/Stereo Selector button. Use the **Left** input of the stereo pair for mono recording.



The Cubase LE Project Window



7. You should see activity on the Track Input Meter when feeding a signal into the Digital Audio System. If not, open PatchMix to verify meter activity on the input channel (strip). Make sure the PatchMix input channel (strip) matches the Cubase LE Track Input.

8. If your input signal is either too weak or too strong, adjust your input source or the Digital Audio System gain controls.
   If you are using the front panel Mic/Line inputs, adjust the input gain knobs as follows. If you can't get a proper level from the line inputs, open the PatchMix DSP, Session Settings by clicking the O button above the TV screen, then select I/O. If your input signal is too strong, use +4. If your input signal is too weak, use -10.

Record a Track

9. Press **Record** on the Cubase LE Transport control and start playing.



10. Press **Stop** when you're finished recording the first track.
11. Press the **Return-to-Zero** button on the Cubase Transport.
12. Press **Play** on the Cubase Transport to play back your track. The audio from Cubase comes back into PatchMix on the **ASIO Out 15/16 Strip**.

Record Another Track

13. Press the **Return to Zero** button on the Cubase transport.
14. Add another track by selecting **Add Track**, from the Project menu. Select **Audio**.
15. A new audio track is added. Notice that the red **Record Enable** button has been automatically turned on in the new track. (If you wish to record multiple tracks simultaneously, simply turn Record Enable on for each track you wish to record.)
16. Again, set up the track to be either **mono or stereo**, then choose the **Track Input** routing. The input can be the same one you used for the first track or you can choose a different input.

Save Your Project

17. Choose **Save As...** from the Cubase LE file menu to save your project. Choose a name and location that will make the project easy to find later.

To record into Cubase LE, you must have a PatchMix session containing these 4 basic components:

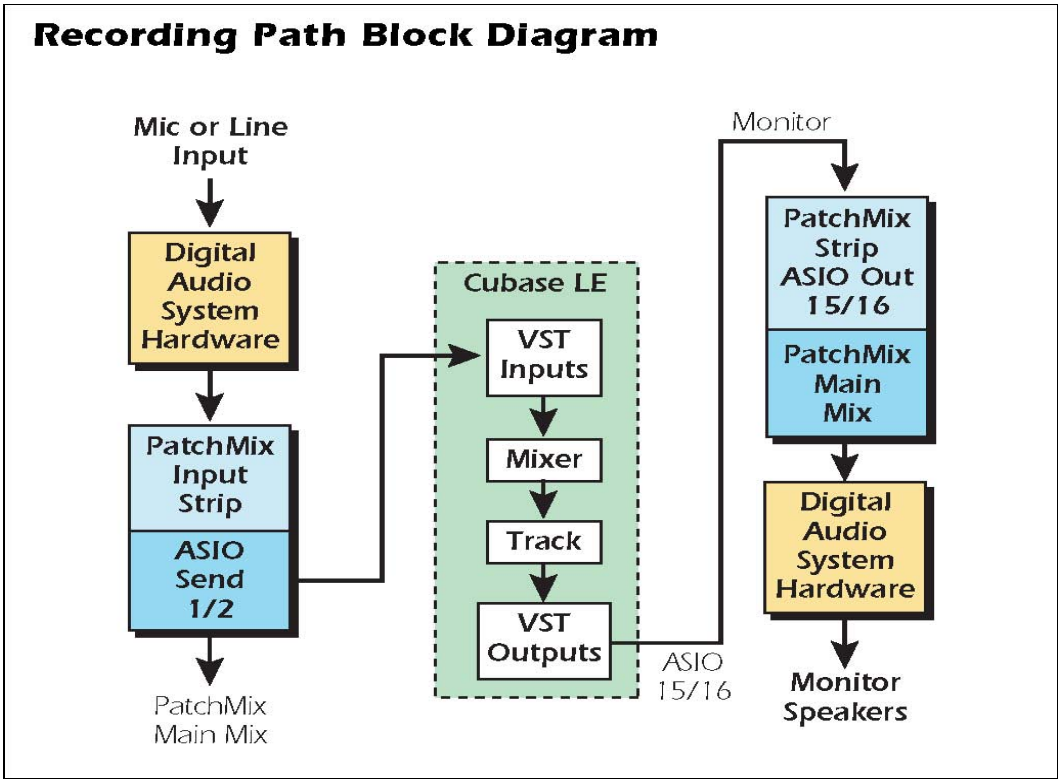
- a **physical input strip** for your input source.
- an **ASIO Send** on the physical input strip (so that Cubase can receive your input).
- an **ASIO output strip** for Cubase to output its audio back through PatchMix.
- an **output assigned** to any physical output (to your monitor speakers).

TIP: *A quiick way to record additional tracks using the same input is to simply drag the Part (audio region) you just recorded down to another location in the Cubase Arrange window, then hit Record again and go.*

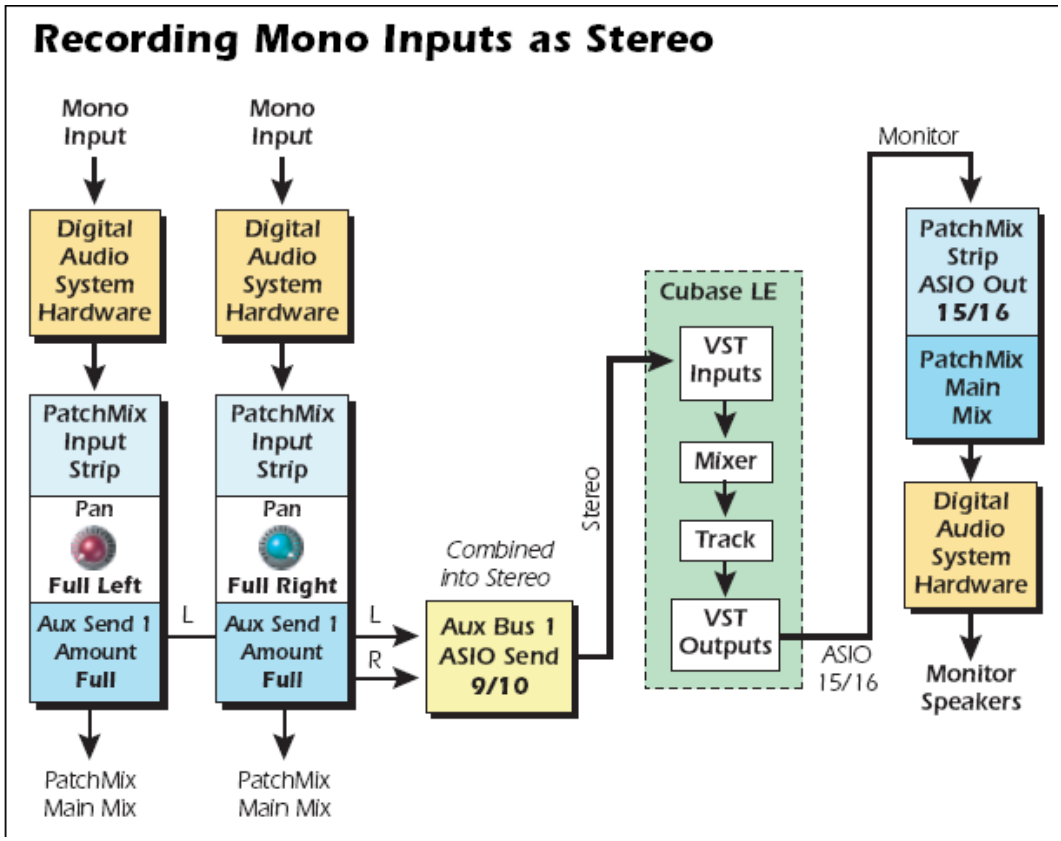
Signal Routing Diagram

These diagrams may be helpful to visualize the signal routing path.

Example A.

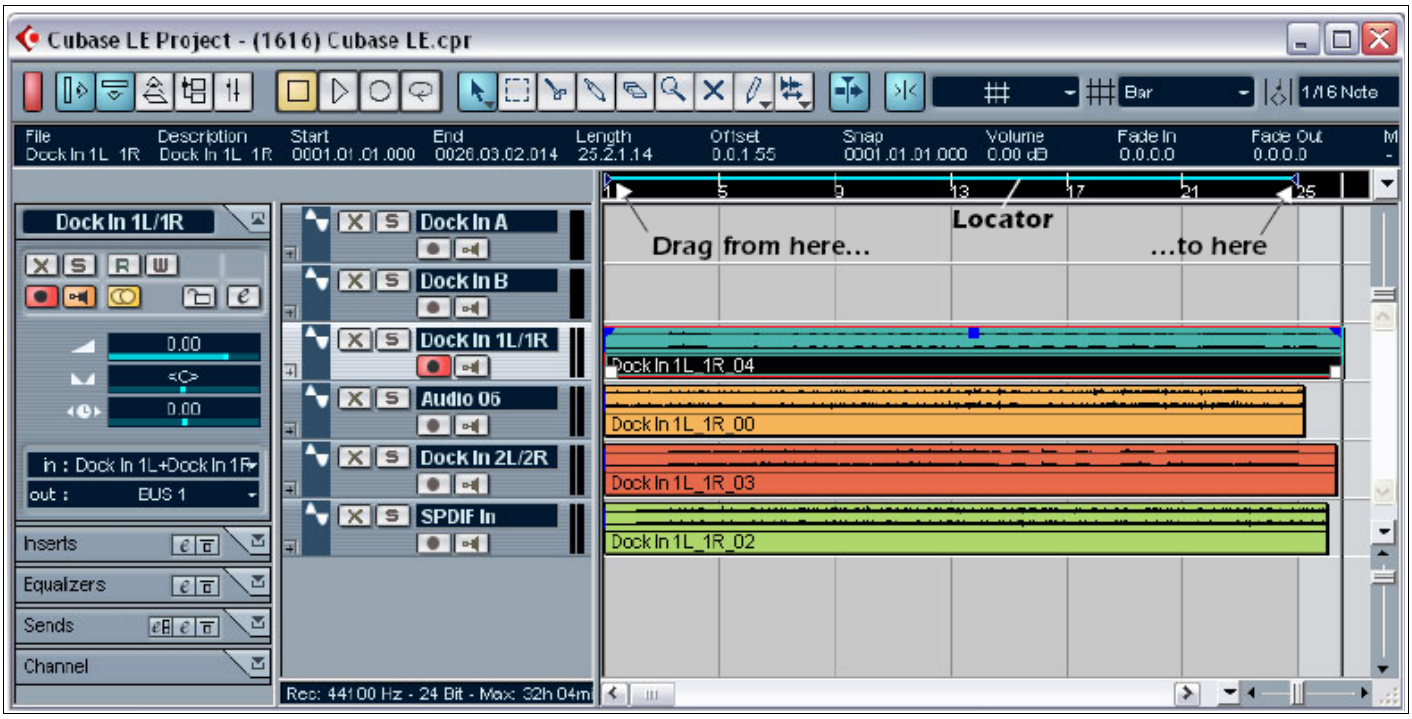


Example B.



Export Audio Mixdown to Stereo

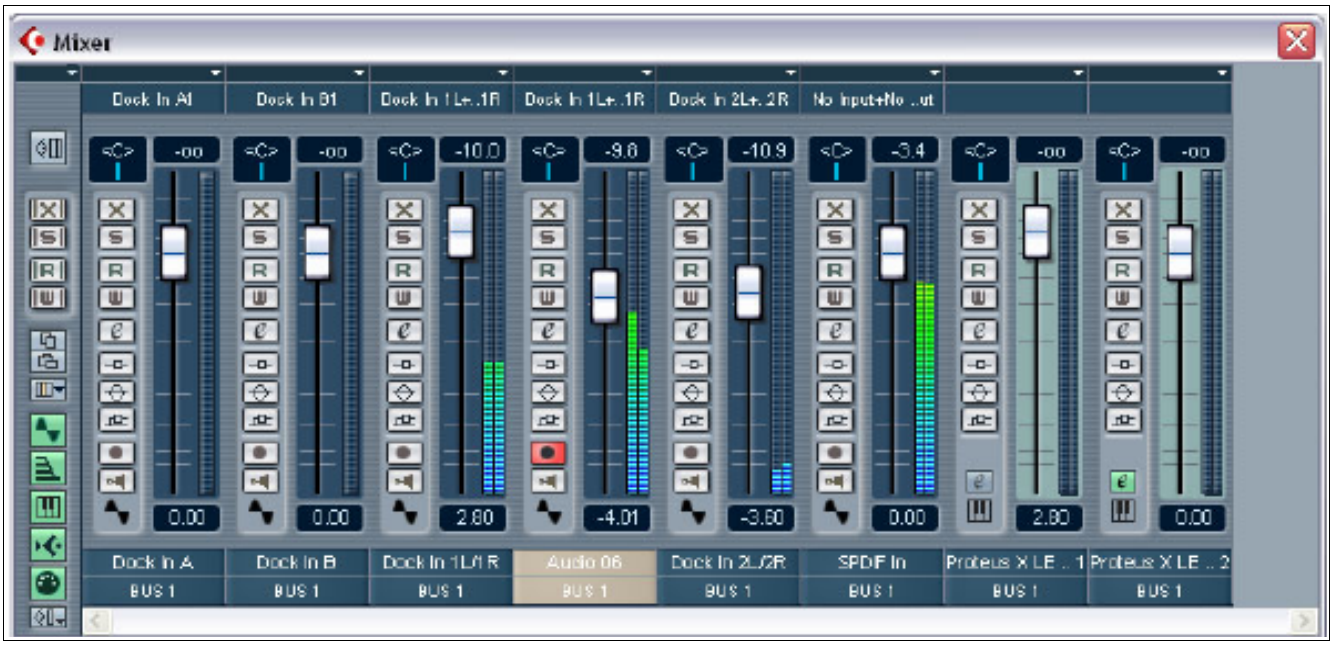
At some point, you'll want to mix down your multitrack project to stereo. Cubase LE has a special function (Audio Mixdown) that makes this job a breeze. The Cubase LE Project window below shows a short song with 4 stereo tracks recorded.



1. Drag the Right Locator (little blue triangle) to the right. Keep dragging until the blue Locator Bar encompasses the entire song or section you want to mix down. See above.

Mix Your Song

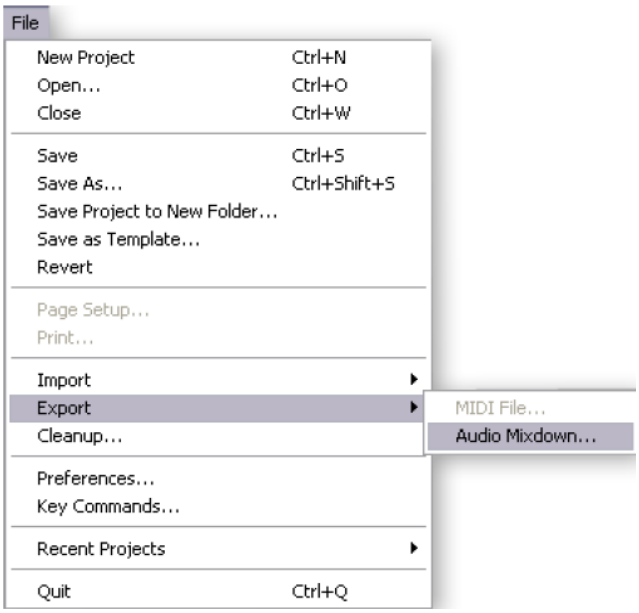
2. From the Cubase LE Project window, select **Mixer** from the **Devices** menu (or press F3). The Cubase LE Track Mixer appears.



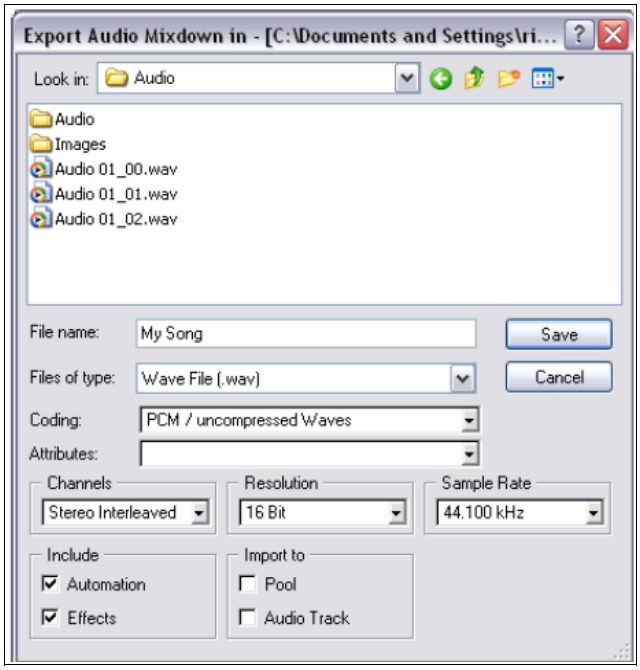
3. Play your song and use the Faders to set the levels of each part. *(Note: These moves can be automated in Cubase LE. See the online documentation for details).*

Export Audio

4. From the Cubase LE Project window, select **Export** from the **File** menu and choose **Audio Mixdown**.



5. The following screen appears. First choose a location for your file.



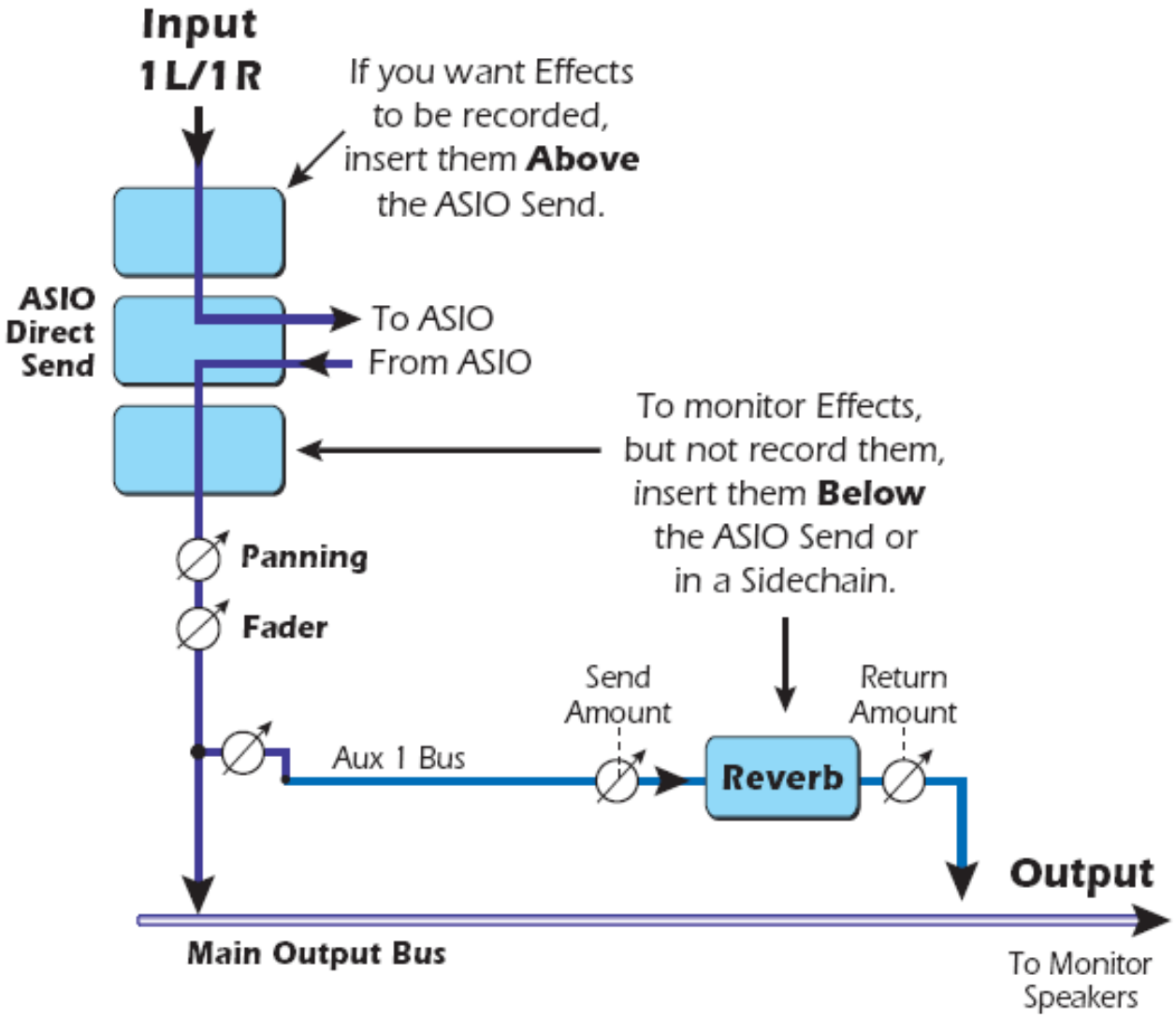
6. Next choose the file type. Cubase LE can export as: **WAVE** (.wav), **AIFF** (.aif), **MPEG Layer 3** (.mp3), **RealAudio G2** (.rm), or **Windows Media Audio** (.wma). WAVE is a good choice for a master file since it is an uncompressed format. "Stereo Interleaved" creates a single stereo file.
7. Select the desired **Sample Rate** and bit **Resolution**.
8. Press **Save** to process your Audio Mixdown.

Recording and Monitoring with Effects

The Order of Effects

PatchMix DSP allows you to record your tracks without effects (dry) and monitor with effects enabled (wet). It works like this: If the effect is inserted BEFORE the ASIO send in the signal path, it will get recorded; if the effect is inserted AFTER the ASIO send, it will not be recorded. Recording dry allows you to hear your performance with effect (to get the proper feel), but gives you the flexibility to add or modify effects later during mixdown. This way if you don't like the way the effect sounds, you can change or modify the effect without having to perform the part again.

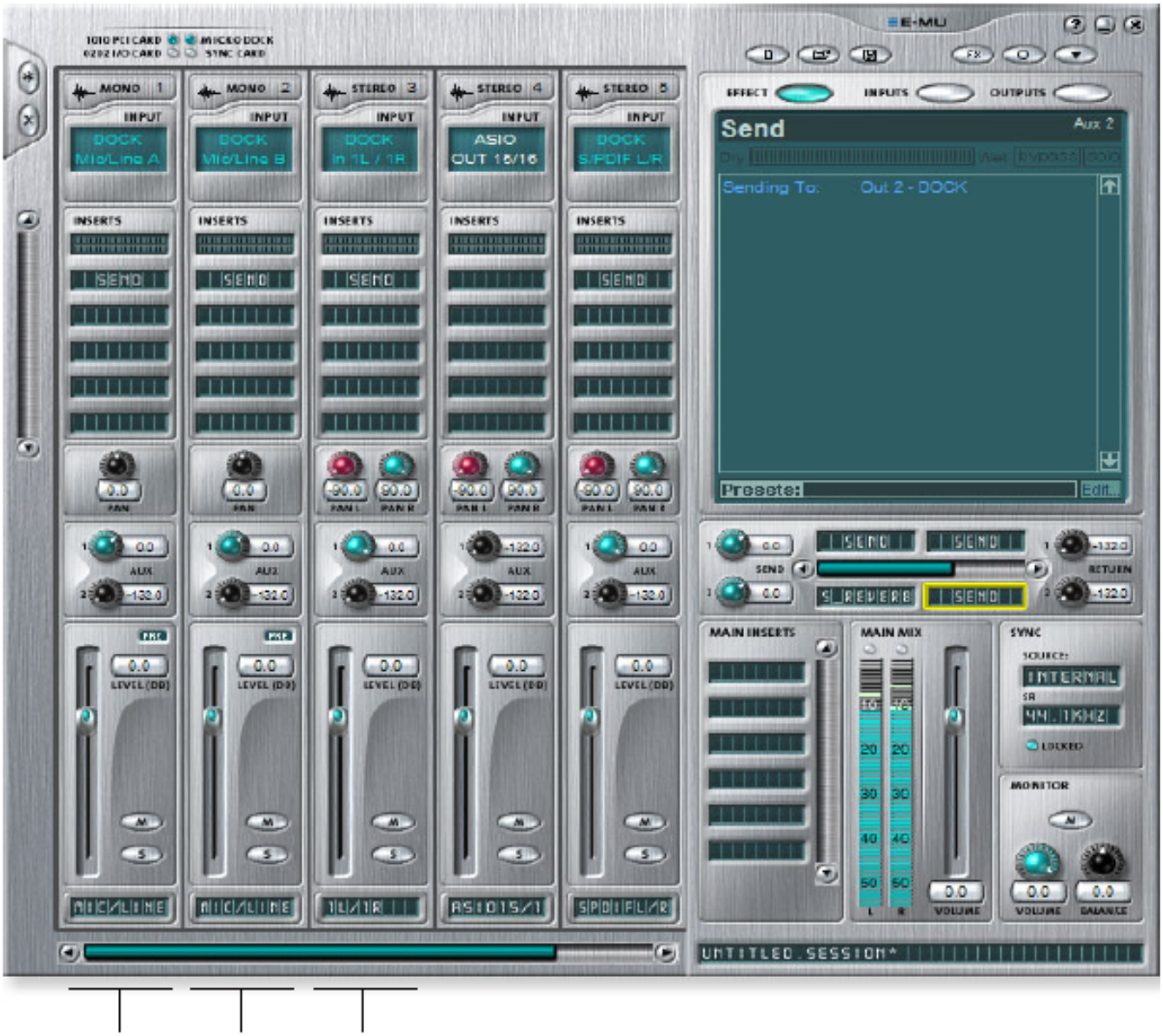
The basic session setup for this is shown below.



Creating a Headphone Submix

Note: Although the SoundSlate only has 2 analog outputs, you can stll create a headphone mix using these instructions and send it to the S/PDIF or ADAT digital outputs.

A headphone submix allows you to create a special mix for the musician while recording. In the example below, a headphone submix has been created from the first three strips in the mixer using the Aux 2 bus. The three strips are muted to prevent the audio from going straight through to the main mix—the submix send amounts now act as the level faders. A reverb effect has been inserted into Aux bus 2 before the Send to the headphone out.



These three strips are being submixed to Aux Bus 2 which feeds output 2L/2R. The strips are all muted and the output of Aux 2 is turned all the way down to block the audio from the main output.

To Create a Headphone Submix:

1. Create Pre-fader strips for the inputs you want to feed the headphone submix. (Otherwise the Mute button will mute the Aux Send.)
2. Turn up the Aux Sends (for this example we'll use Aux 2) on the strips feeding the headphone submix and mute the channel outputs as shown above.
3. Turn the Input Send Level Up on Aux Bus 2 (0 dB or higher).
4. Turn the Output Return Level Down on Aux Bus 2 (-132 dB).
5. Right-click on one of the Aux 2 insert locations and choose "Insert Send (Output to ASIO/WAVE or Physical Out)" from the popup list.
6. Choose the desired Send Output from the list. Choose a pair of outputs for your headphone submix such as "DOCK Out 2L/2R".
7. Click the Output button at the top of the TV screen to view the output assignments.
8. Disconnect the submix from the main or monitor mix if connected.

About PatchMix DSP

PatchMix = Patchbay + Mixer

It's a simple concept. PatchMix is a patchbay for your computer, where any physical input or output (analog, S/PDIF, ADAT) can be connected to any HOST input or output (ASIO, WDM, WAVE). Strips = Inputs and Insert Sends = Outputs. That's the "Patch" part of the name. Now add a simple mixer with two auxiliary sends and a bunch of effects. This lets you hear zero-latency effects while recording, a feat that no plug-in effect can match. It's a super-flexible patchbay first and a mixer second—PatchMix.

Too Much Power?

We included the PatchMix DSP mixer and powered effects because they allow you to do things that you can't do with any other sound card. However…please do not be intimidated by PatchMix, because you DON'T HAVE TO USE IT unless you want to. The Product Default Session connects every input and output for you. Once this session is selected, you can forget about PatchMix and use the Digital Audio System like any other sound card without the hardware effects and flexible routing features. PatchMix was never meant to replace the mixer of your recording application. It's forte is allowing you to connect your hardware inputs and outputs to your software applications in a number of useful ways. You may not need this additional power right away. Use PatchMix in the background if you prefer, and come back to it whenever you're ready.

IMPORTANT -> Activating ASIO
It is important to understand that ASIO inputs do not appear in Cubase or other host applications until they are CREATED in PatchMix DSP. This allows you to dynamically assign the 32 available ASIO channels according to your specific needs.

An Example:

1. Create a new strip from a physical source such as S/PDIF L/R.
2. Next, right-click on the new strip's insert chain and select the Insert Send option.
3. Choose an available host ASIO input.
4. This S/PDIF L/R input should now be available to record in your ASIO application.

ASIO Channels are STEREO!
ASIO channels are stereo, period. Even when you are recording a mono source such as a microphone, the ASIO stream is stereo (with the mono signal on both channels). To record a mono input, you simply set up a sequencer track as a mono track and use either the left or right ASIO channel from PatchMix as the input for the track.

6

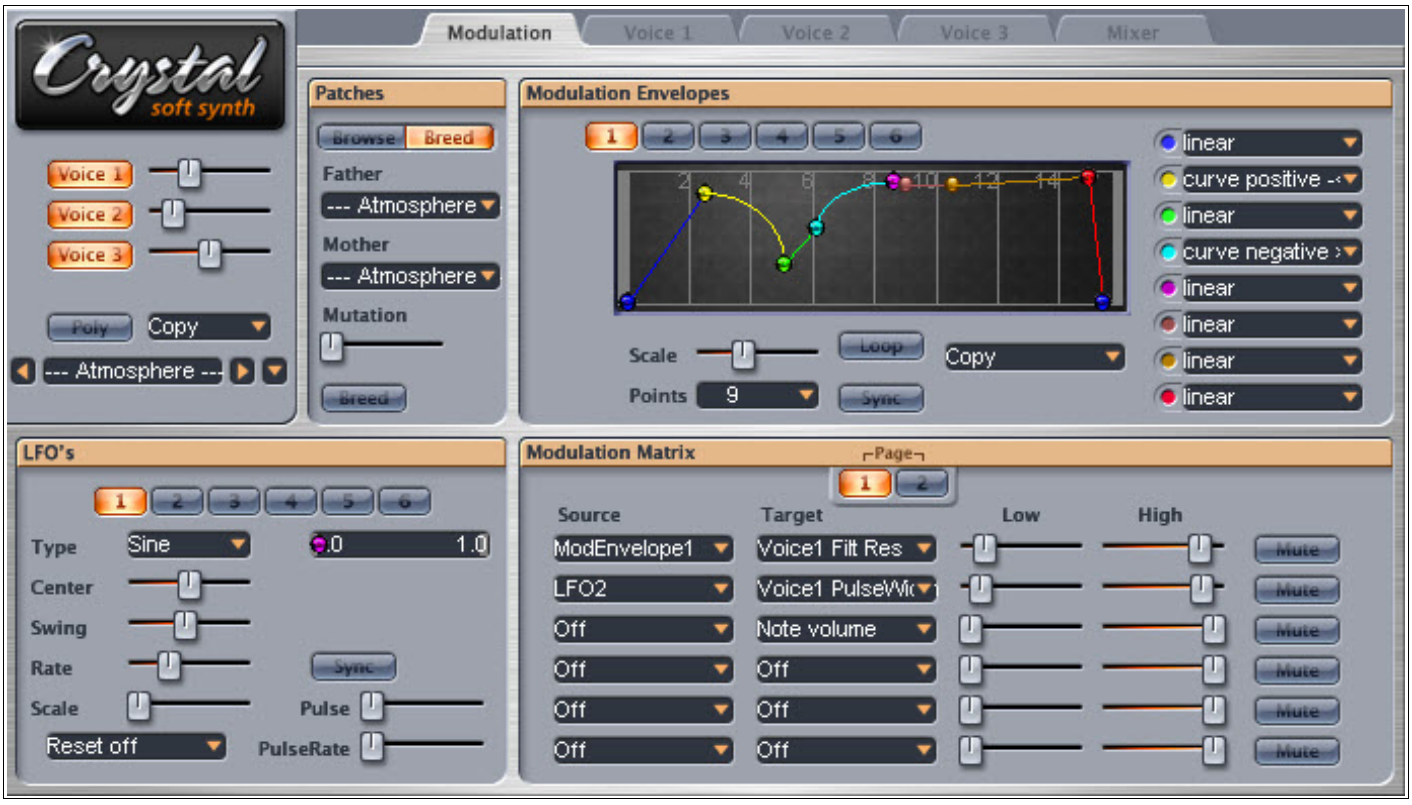
Chapter 6 – VST Instruments

The VST instruments described in this guide can be found and used inside of Open Labs Riff.

VST Instruments

Crystal

Crystal is a semi-modular software synthesizer featuring both subtractive synthesis and frequency modulation (FM) synthesis. It offers abundant modulation control with over 90 parameters which may be modulated, multi-stage envelopes with graphical editors, extensive tempo sync of envelopes/lforates/delay times, built-in effects for chorus/flanging/comb filtering/echoes, and a band splitter for effects processing by frequency band. Not only that, Crystal has wave sequencing, granular synthesis, program morphing, MIDI learn, and imports soundfonts. The musical applications for Crystal are literally boundless.



Dimension LE

Is a Multi-instrument sample player. Dimension LE combines real instruments with advanced synthesis, giving you endless sound possibilities. Left-click on "Empty Progam" to load a sound.



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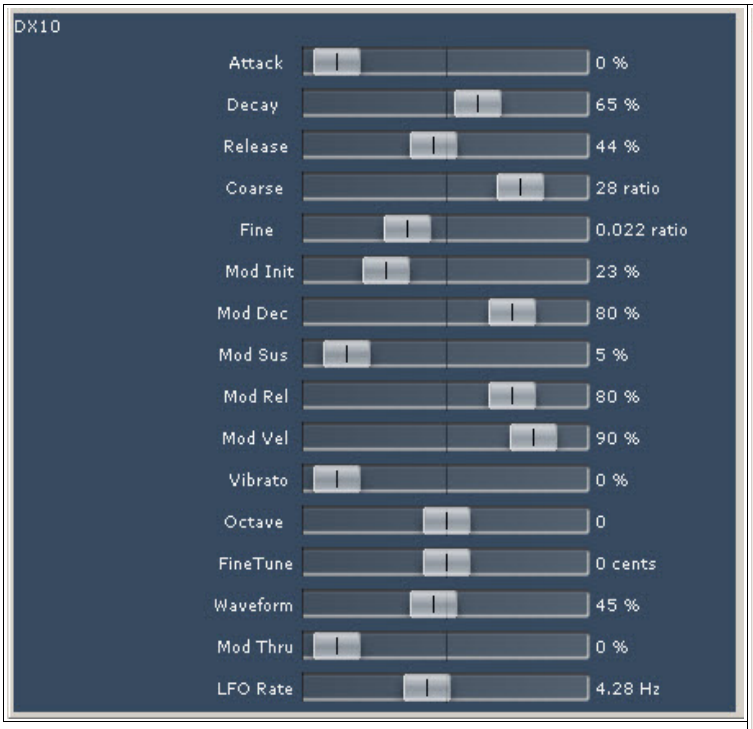
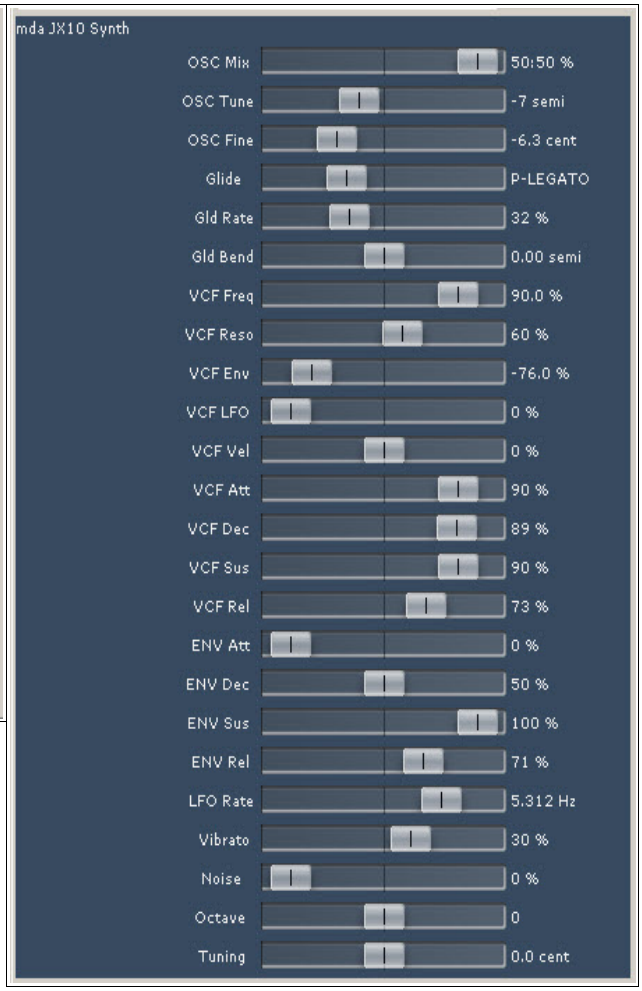
mdaDX10 / mdaJX10

The DX10 is a FM synth that recreates the classic FM sound but with a much simpler and more intuitive interface than any DX synth, or any of their software imitations. The JX10 is a simple 2-Oscillator Analog Synthesizer.





Plugsound Free

Plugsound is a sample-playback virtual instrument powered by the powerful UVI-Engine. Plugsound provides you with essential and useful synthesis tools:
- Two resonant filters, one Multimode, and one switchable between HPF and LPF called the UVI filter.
- Amplitude and Filter envelopes (ADSR type). The Multimode Filter has an envelope amount knob.
- One LFO that can be used for tremolo, auto pan etc. This is in addition to the integrated LFO that provides the modulation you expect when using a Mod wheel
- Mono Mode with a glide time knob. This provides the most realistic mono-legato you've heard on a virtual synth.
- Reverb module with 3 knobs to set the time, the high damping and the reverb' mix
- MIDI control of every knob and slider on the interface via standard MIDI continuous controllers



Purity

Is a next generation digital musical instrument workstation and PCM sound module software. PURITY is designed to realize and surpass the hardware musical instrument workstation perfectly on computer-based music production environments offering higher sound quality and more usability.

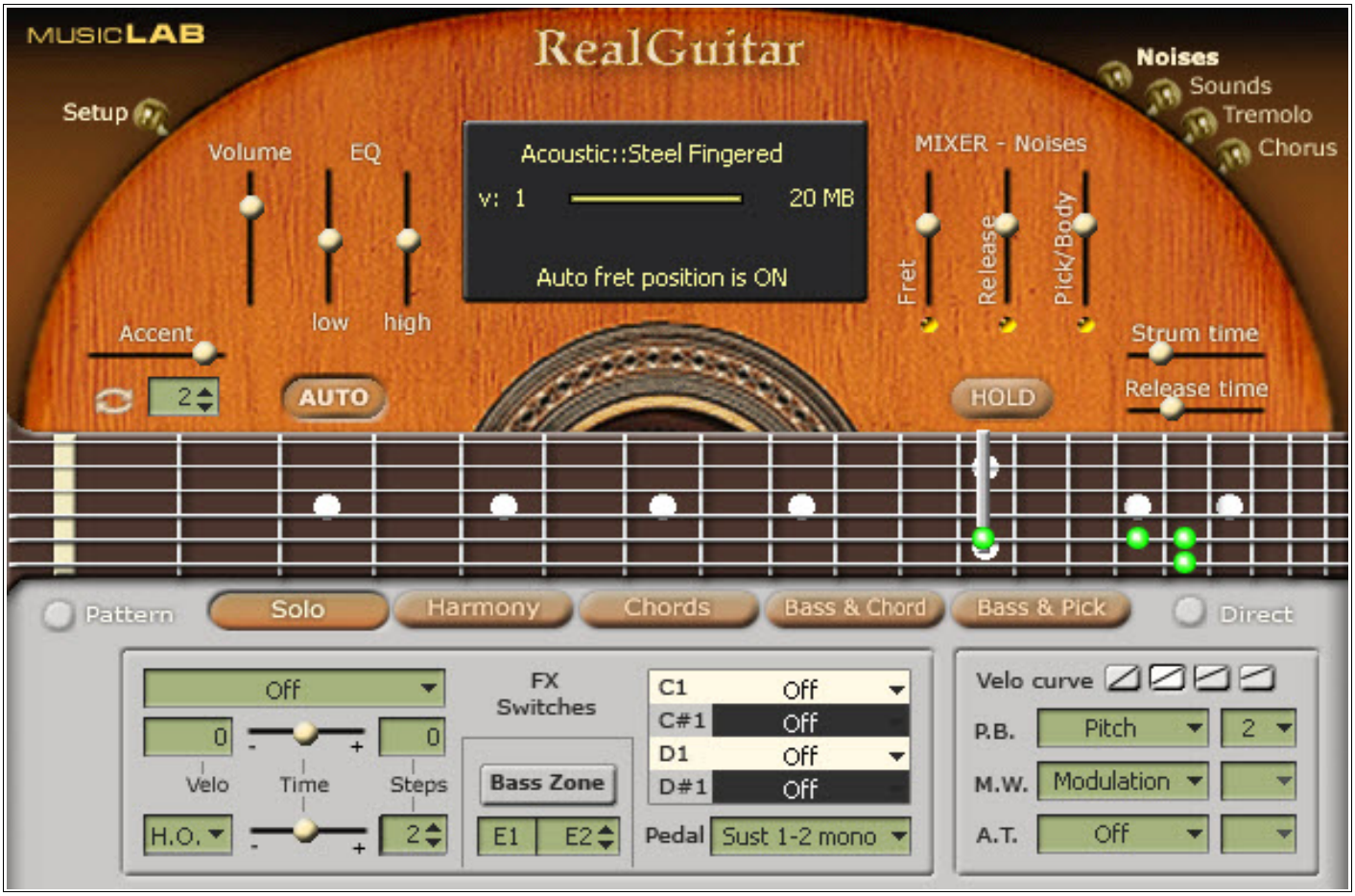
PURITY is based on well-sampled PCM wave data including a lot of sound of hardware workstations, sound modules, drum machines, vintage analog synths and even modern digital synths. Moreover, PURITY comes with over 1,300 ready-to-use sound presets that are most popular in each era of the electronic music.



RealGuitar2SE Demo

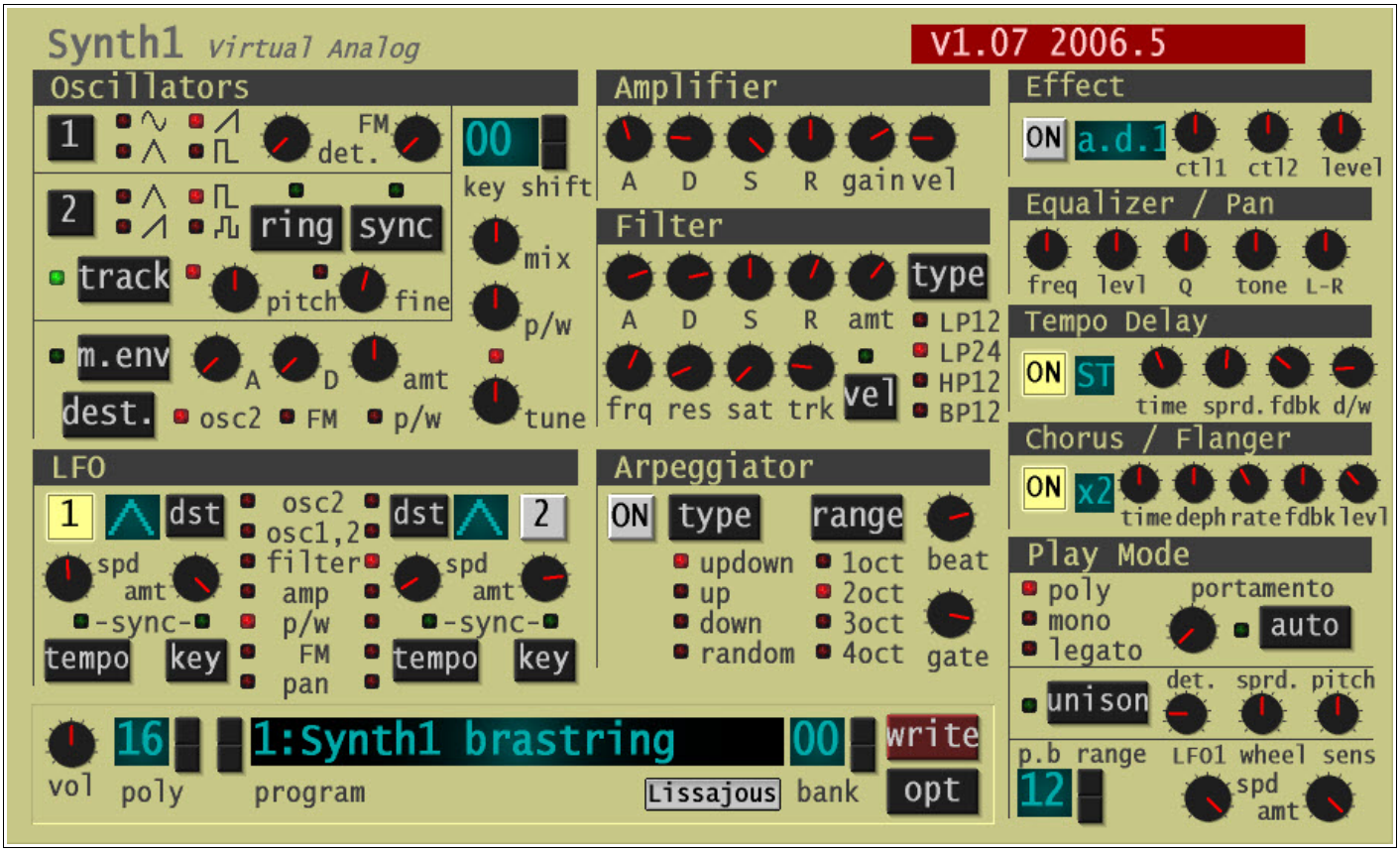
The true power of RealGuitar lies in 3 brand new technologies incorporated in this innovative guitar performance tool: "New multi-channel layering technology", incorporating a custom library of specially recorded samples taken from every fret of all 6 strings of a real guitar. "The unique Floating Fret Position technology", which imitates change of fret position of a guitarist's hand on the neck. This gives you the possibility of playing on up to 104 guitar frets using just 43 keys of a standard keyboard! "Our original Guitar Touch technology" letting you easily imitate basic guitar techniques (strumming, plucking, sliding, bending, muting, etc..), using standard MIDI keyboard and MIDI controllers, such as Pitch Bender, Modulation Wheel, Sustain Pedal, Aftertouch.

Homepage: http://www.musiclab.com



Synth1

- 2 oscillators, with FM, Ring modulation and Sync.
- 4 Filters (LP12/24/HP12/BP12) with Distortion.
- 2 LFOs.
- EQ, Tempo delay, Stereo chorus, Flanger.
- Mono/Poly/Legato modes.
- Arpeggiator
- 16 voice polyphonic.
- 128 presets.



Wusikstation

Sample Player - with leads, bass, pads, electronic.



7

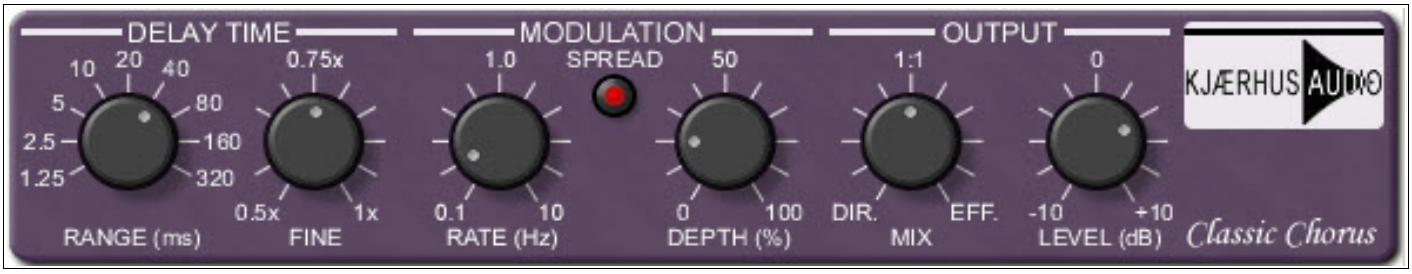
Chapter 7 - VST Effects

VST Effects - Delay/Modulation/Reverb

Classic Chorus

Classic Chorus is a great sounding and easy to use classical Chorus VST plugin. Nice Plugin for adding Depth, Fatness and Stereo perspective to almost any Electrical Instruments or Vocal.

- Make an old mono Rhodes track come alive with the factory preset "Fat on Rhodes", or make your vocal tracks sound better by creating a stereo doubling effect using the preset "Vocal Overdub". Many more great presets are available for fattening up your instruments!
- Use it for a good vibrato effect.



Classic Delay

The Classic Delay emulates three of the most commonly used delays / echo sounds types: Tape, Analog, and Digital, but without the introduction of noise. A quality control is included where the sound of the echo machine can be adjusted to give the preferred amount of "imperfection". The delay can be synchronised to the host, and the feedback can be reversed to give a ping-pong effect.

- The tape delay type gives you the sound of echo being degraded slightly each time it is fed-back and re-recorded. You can adjust the quality control for the amount of flutter and distortion introduced.
- The analog delay type has a very "dark" sound due to the low sampling rate and sharp filters used to save analog memory. The "darkness" of the sound can be adjusted via the quality control, which emulates the changing of the sample rate.
- The digital delay can be adjusted from bright and clean on the highest quality setting, to a "darker" and more noisy / distorted sound if the quality setting is set low. This enables you to emulate the sound of older digital delays with less bit depth and a lower sample rate.



Classic Filter

The Classic Auto-Filter is an analog modeled four-pole filter with resonance control. It can be used for creative effects like auto-wah-wah and filter sweeps, as well as a fixed filter for equalizing purposes. The filter is selectable between low-pass, high-pass, band-pass and band-reject (notch). The filter cut-off frequency can be modulated from the envelope follower and the LFO.



- Modulated four pole filter with resonance
- Syncable LFO with sine, random-hold and stereo spread
- Envelope follower with adjustable attack and release
- Presets
- Supports sampling rates on and above 44100 Hz
- Full VST automation

Classic Flanger

Take a trip back to the mid 80's the Classic Flanger. It will give you that subtle or massive swirling effect that is used mainly for guitar, but can be used for many other instruments as well. Due to a tightly controlled feedback path, it is possible to boost the flanging effect to maximum without getting unwanted side effects.

- When feedback is set at a high level, the output can become very dynamic. In that case, just use our Classic Compressor or Classic Master Limiter after the Classic Flanger to control the dynamics.
- Don't settle with the flanger only on your guitar. We have had great results using the Classic Flanger on everything from strings to drums. Try it on all your tracks to see what it can do for you. You might be surprised!



Classic Phaser

The Classic Phaser emulates analog phasers from the 70's and 80's. These vintage phasers got their characteristic sound from the analog phase shifting filters used, unlike the delay lines used in most modern digital phasers. To get a deeper and fatter phasing effect, more filter stages were cascaded. The Classic Phaser works just like an analog phaser, and with selectable filter stages from 4 to 12, it can resemble anything from phaser pedals to expensive studio phasers from that time period.

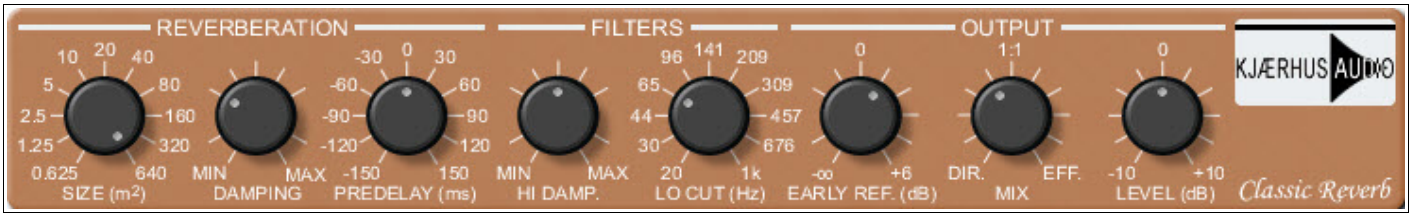
- The phasing effect is most distinct when the mix is set to 1:1.
- Use feedback to emphasize the phasing effect, but be careful; high amounts of feedback can give very dynamic signal peaks; a limiter can be used to tame these peaks.
- Use "Spread" on stereo tracks to make the sound wider.



Classic Reverb

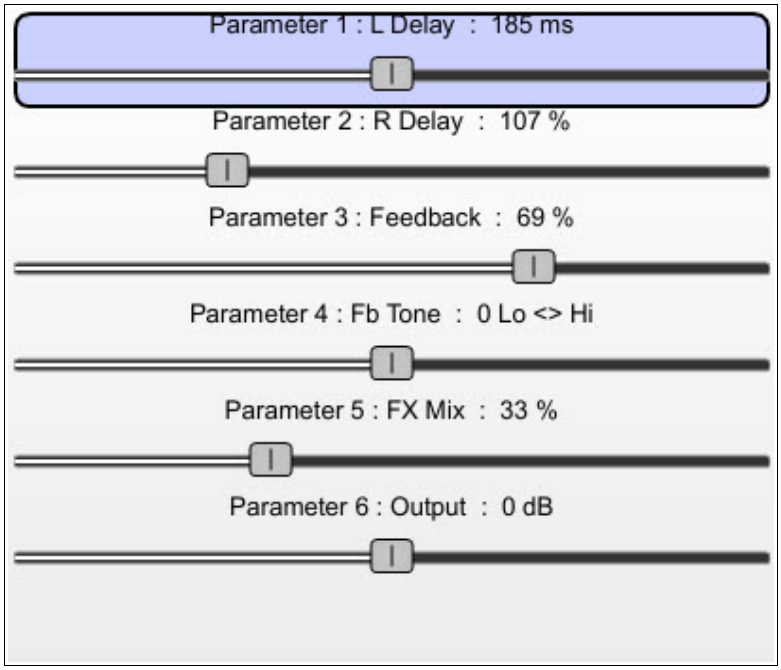
Classic Reverb is a nice and smooth reverb that does a good job on almost any instrument. It can be adjusted to sound like most any acoustic space, ranging from a small shower to a large concert hall. With the "Hi-Damping" control, it is possible to change the sound from bright and sibilant to a more natural / warmer sound.

- Due to the smooth reverb tail, this reverb is very well suited for drums.
- Use the "Lo-Cut" filter to remove rumble from drums and bass guitar.



mdaDelay

A simple stereo delay with feedback tone control.



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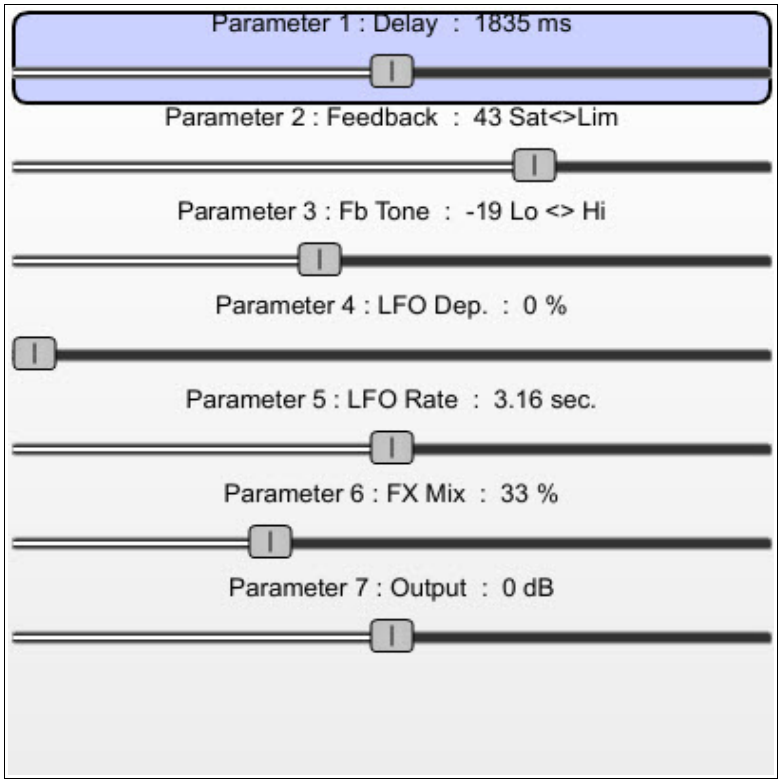
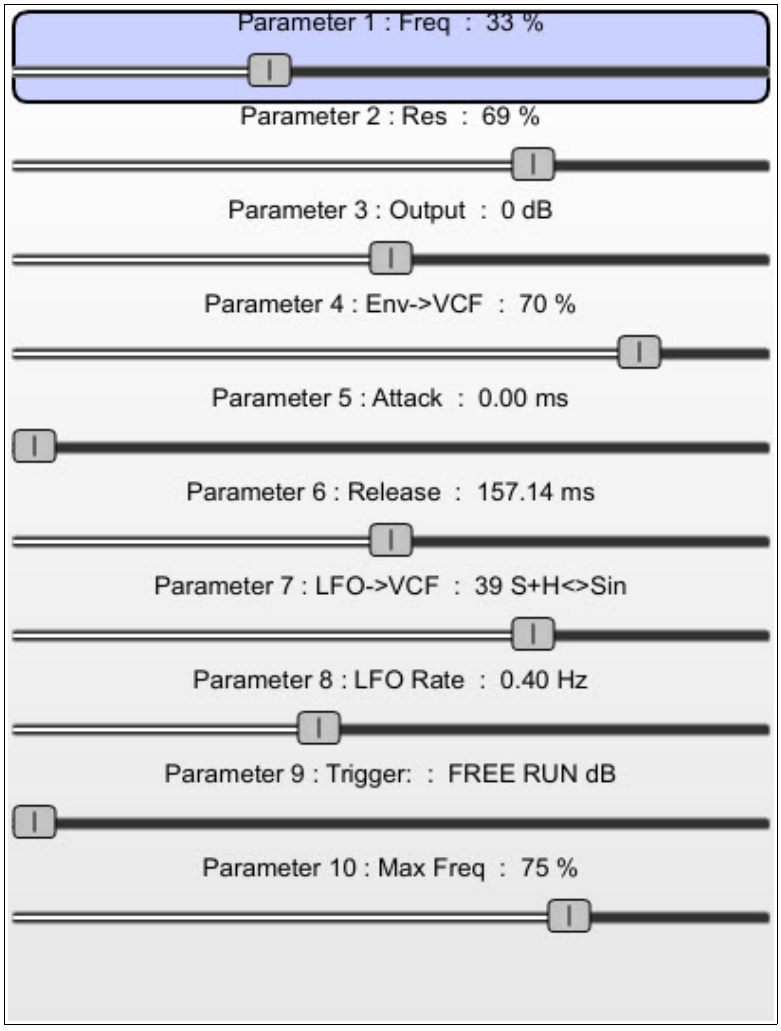
mdaDubDelay

A delay with feedback saturation and time/pitch modulation.



mdaRezFilter

Resonant filter with LFO and envelope follower.



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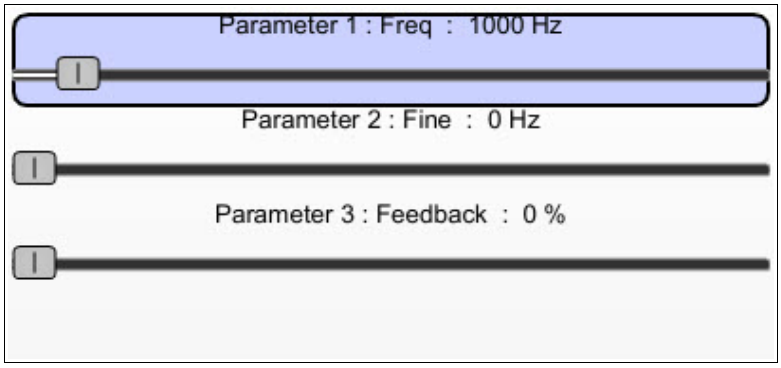
mdaRingMod

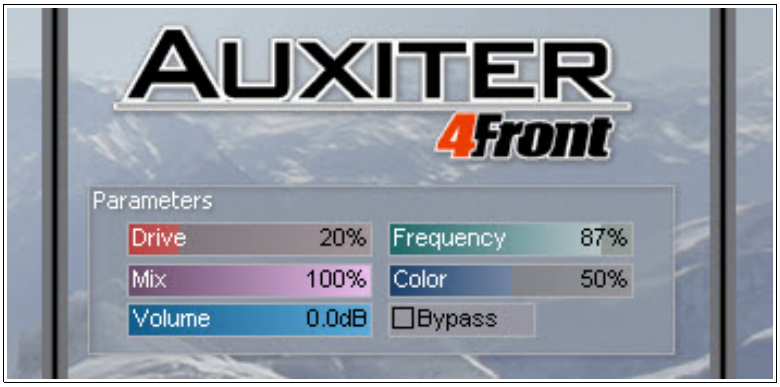
Simple ring modulator.



VST Effects - Dynamics Processing

4FRONT AUXITER

This is a high frequency refresher exciter-type plugin. It brings clarity to the high end by regenerating high frequency harmonics.

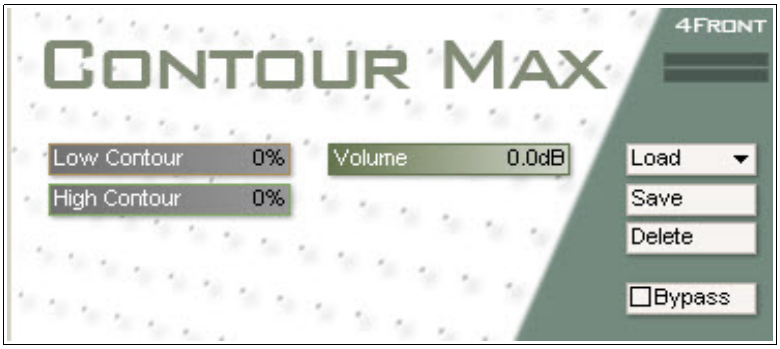


Primary design goals:
- Vocal enhancer
- Per/track enhancer
- Master track restoration/refreshment
- Old tracks restoration

Usage:
First, try to figure the right Frequency slider value for your track, then control the Harmonic, Drive and Mix levels to get the best enhancement.

4FRONT CONTOUR MAX

The plugin is designed to change the basic frequency contour of a sound track, or the overall frequency balance of a master.



Primary design goals:
- Master processing
- Per-track processing for balancing low and high end
- Track restoration

The plugin contains multiple presets for various track types.

4FRONT SAND BRUSH

The plugin completely regenerates high frequencies, basing on the existing partials.

The plugin can be used to apply on hihats and other percussive sounds with high frequency content, and also can be used on other tracks to add "sand".



Depending on the settings, plugin can be also used to emulate the vintage vinyl sound.

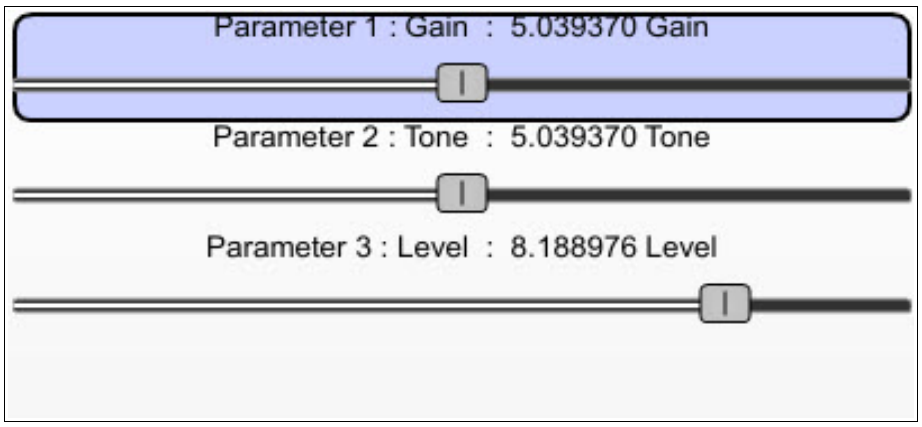
Primary design goals:
- Sand for vocal and solo tracks
- Bring life back to sampled strings
- Track restoration (including MP3)
- Vintage effect (to create vintage sound)

The plugin contains multiple presets for various track types.

VST Effects - EQ / Signal Processing

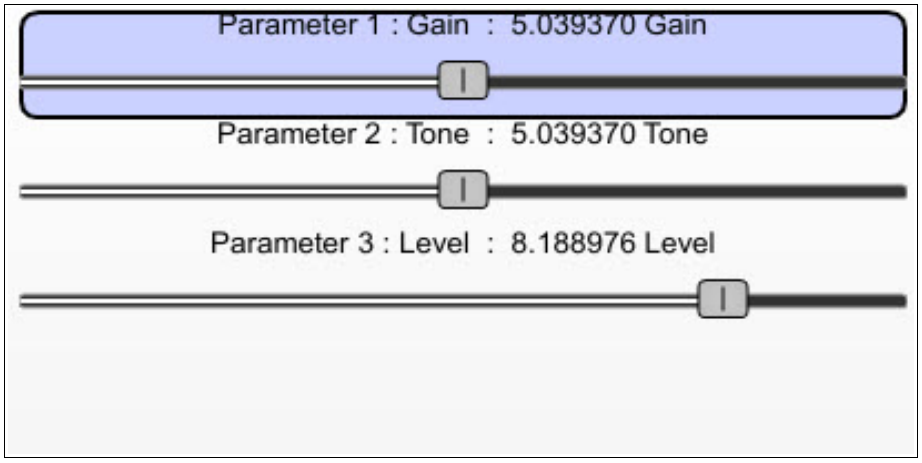
Boss DS-1

An effect that is modeled after the Boss DS-1 distortion pedal. Great for guitars and basses!



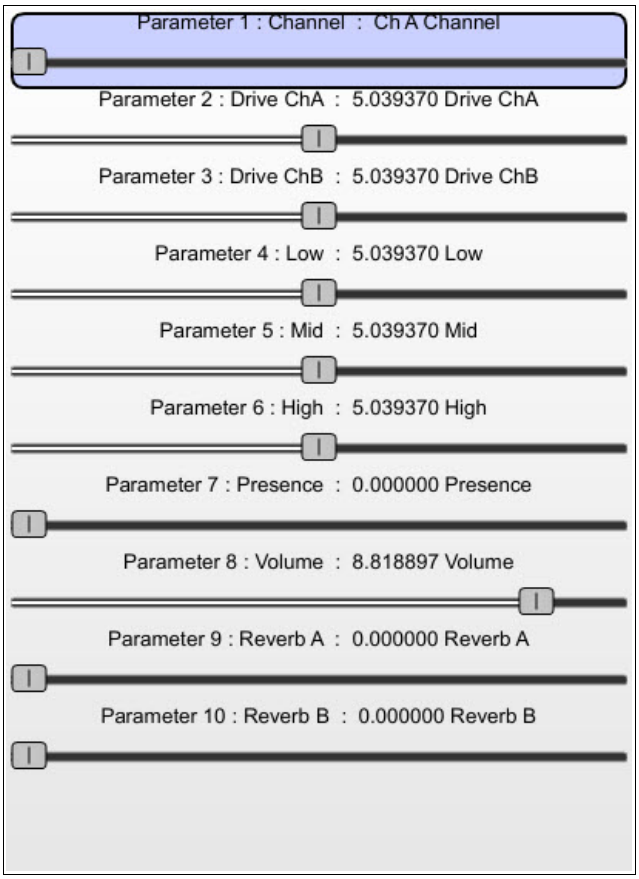
Boss SD-1

An effect that is modeled after the Boss SD-1 super overdrive pedal. Great for guitars and basses!



JCM900

An effect that is modeled after the Marshall JCM900 guitar amplifier.



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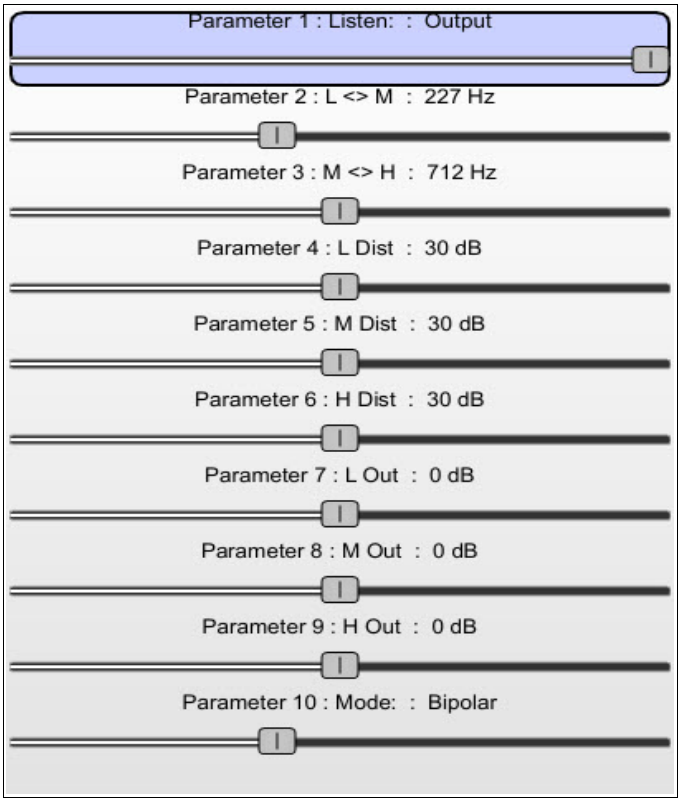
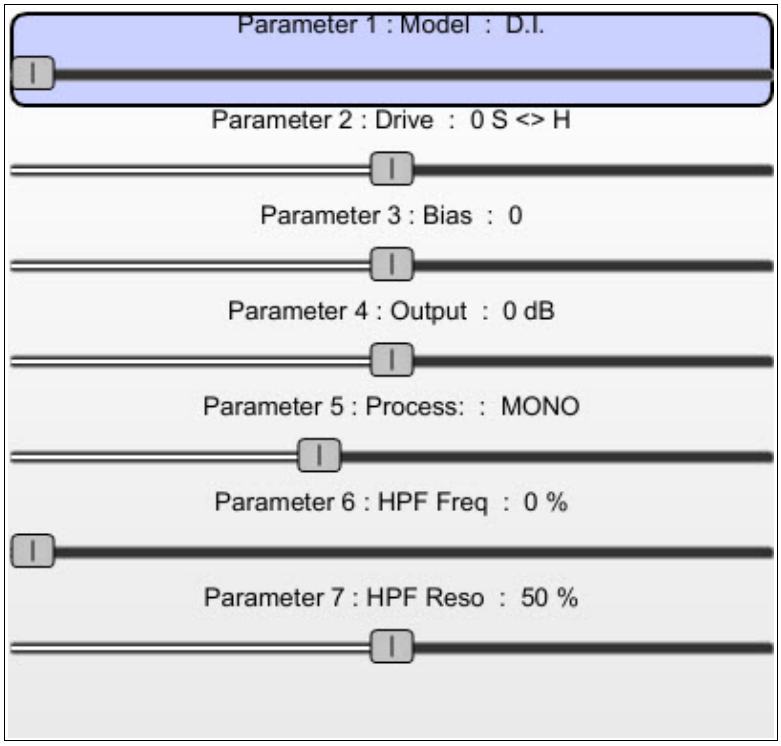
mdaBandisto

This is a multi-band distortion.



mdaCombo

And amp and speaker simulator.



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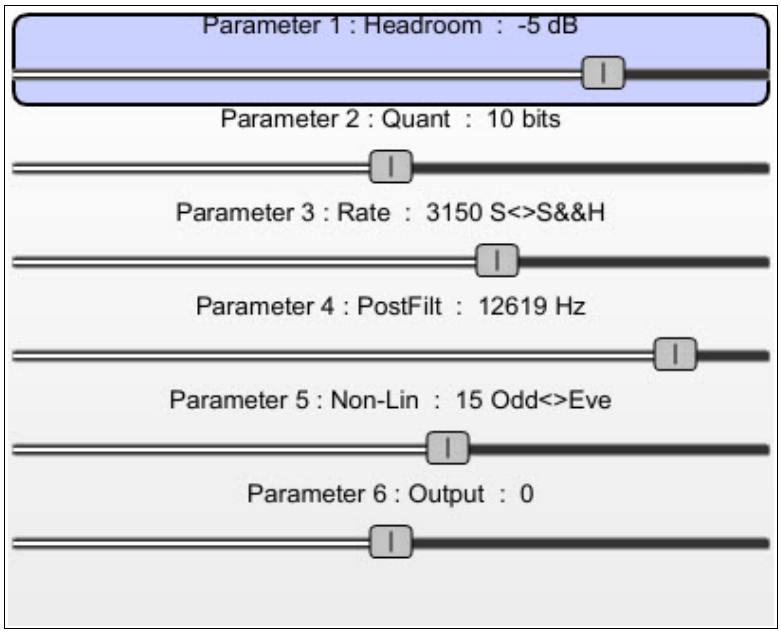
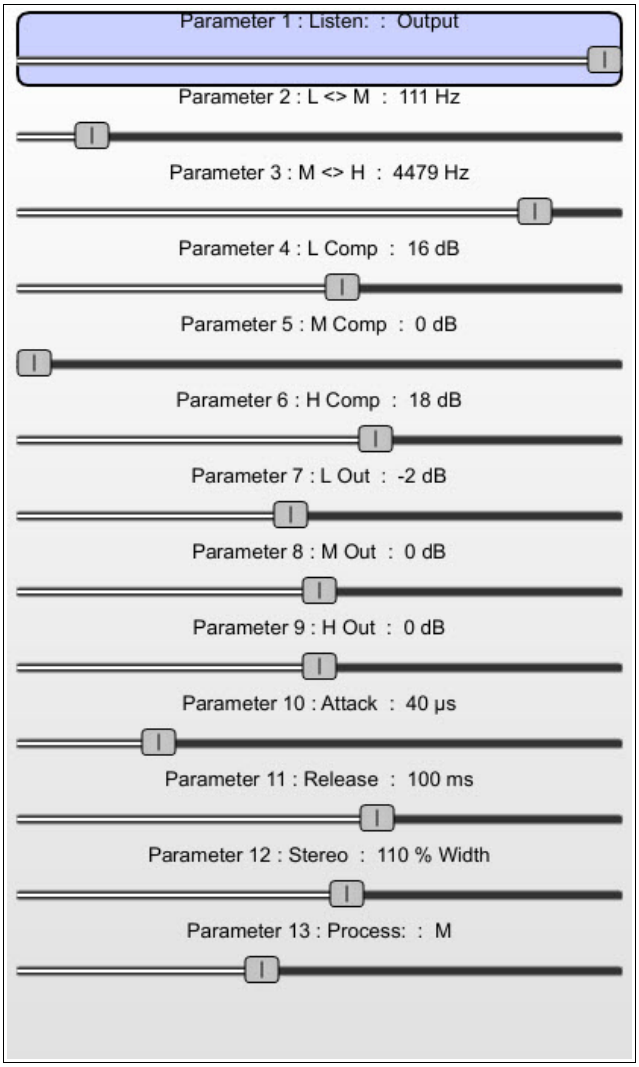
mdaDegrade

Sample quality reduction. Bit crusher.



mdaMultiband

Multi-band compressor with M-S processing modes.



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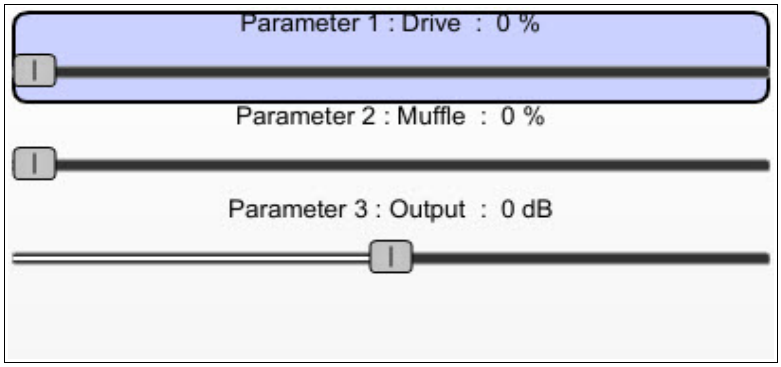
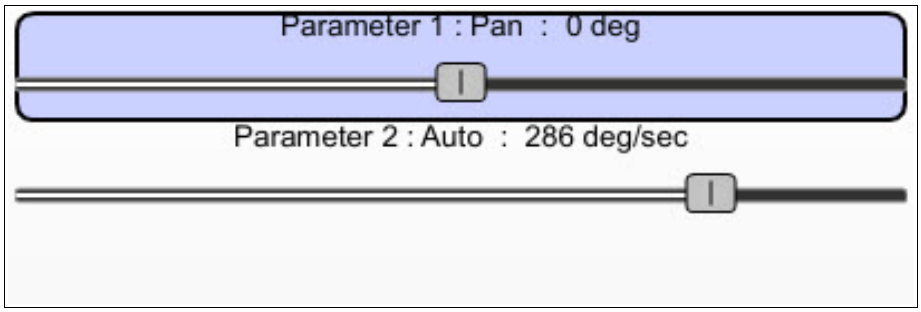
mdaOverdrive

A soft distortion effect.



mdaRoundPan

Autopan that pans the signal in a circle around your head, if you sit in the right place in front of the speakers!



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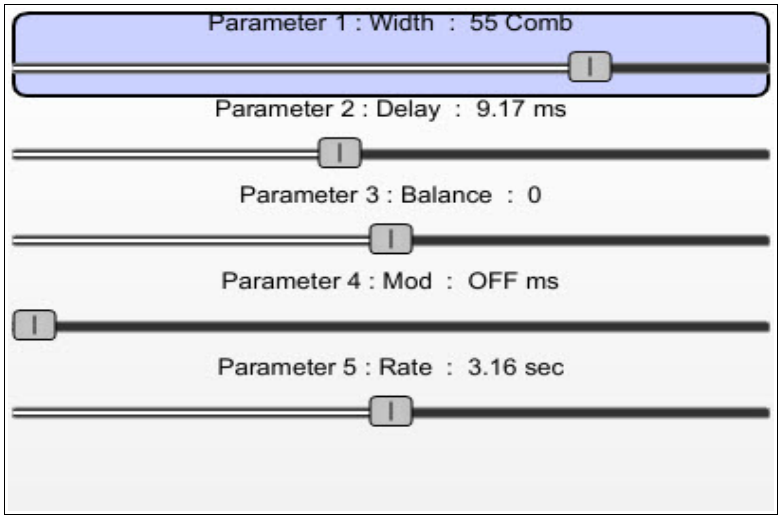
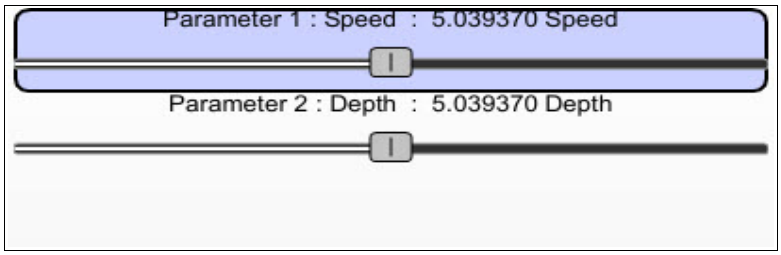
mdaStereo

Synthesizes a mono signal into stereo, using comb filtering or "Haas effect" delay.



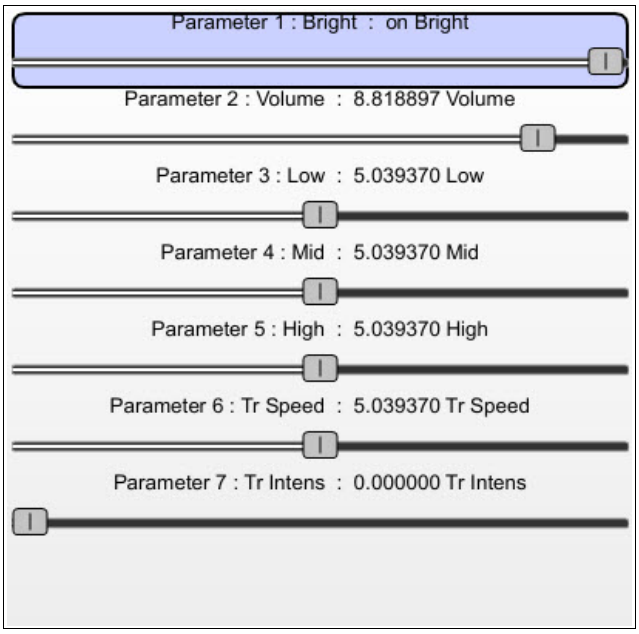
OB PS-1

Simple phaser stompbox style effect modeled after the Oberheim PS-1.



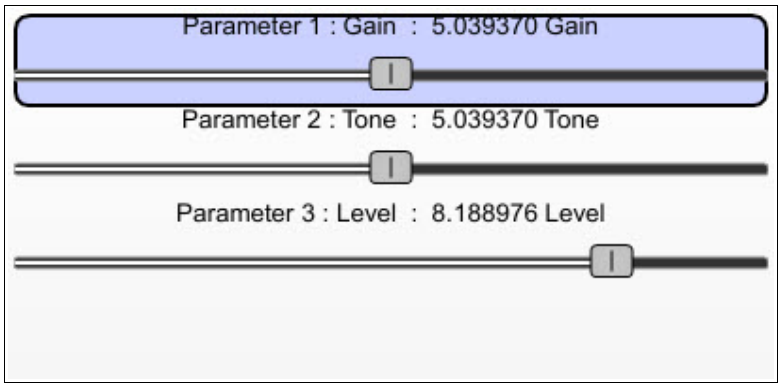
Rednef Twin

Based on the Fender Twin 1969 guitar amplifier.



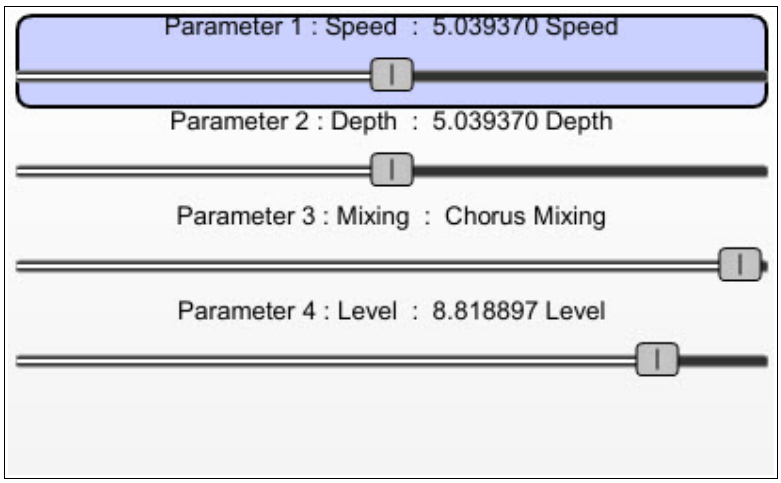
Tube Screamer

Stompbox style effect that is modeled after a overdrive stompbox.



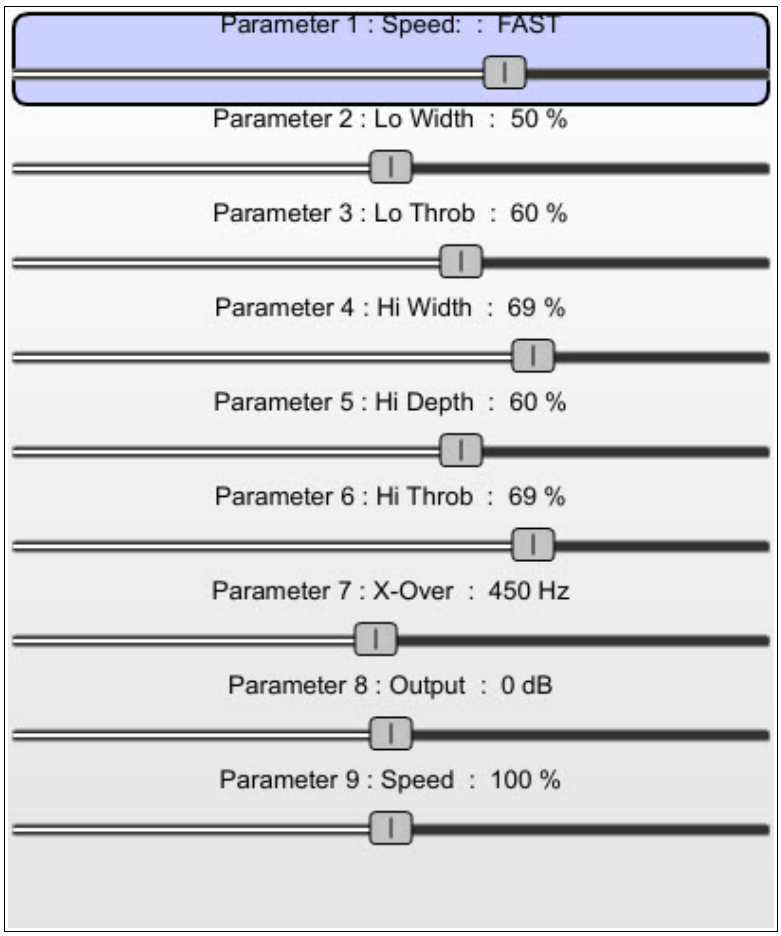
Univibe

A modulation stompbox modeled after the Univox Univibe.



mdaLeslie

Rotary speaker simulator.



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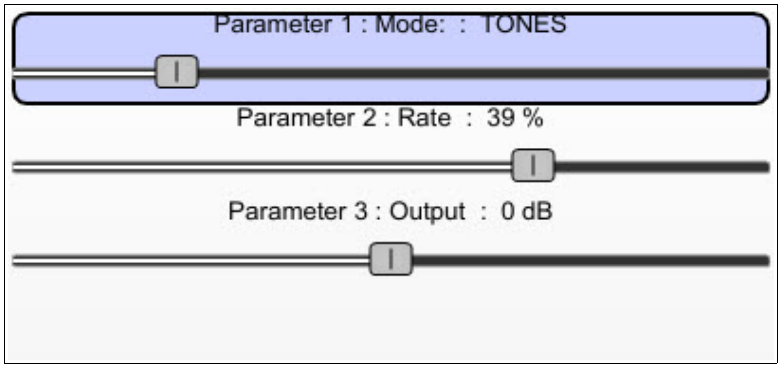
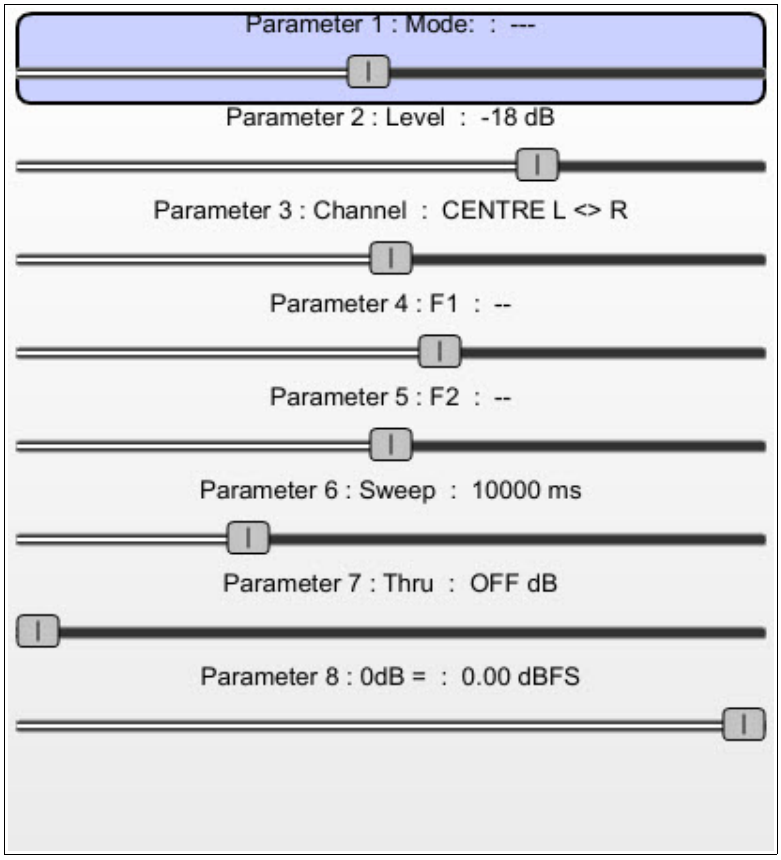
mdaShepard

Continuously rising/falling tone generator.



mdaTestTone

Signal generator with pink and white noise, impulses and swept sines.



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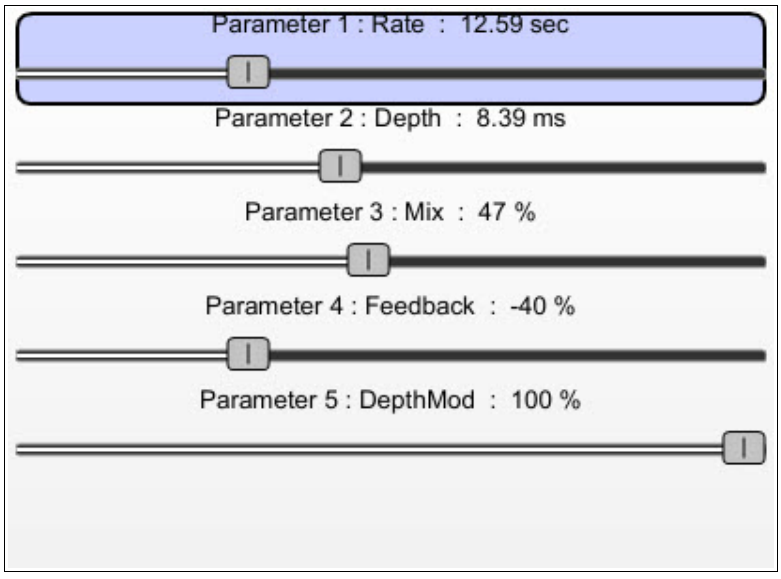
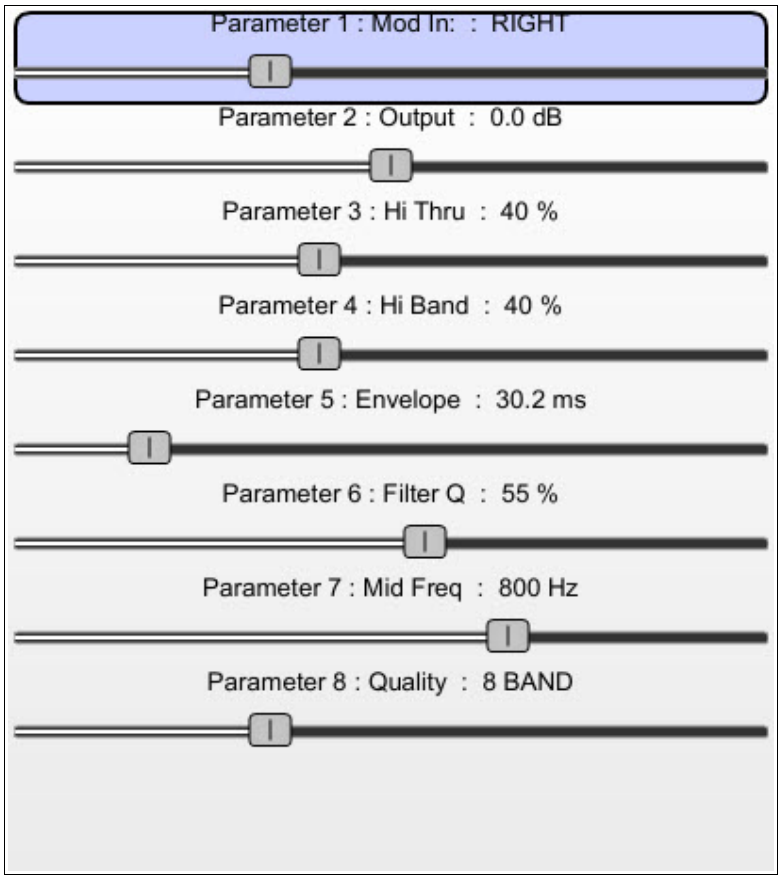
mdaThruZero

Classic tape-flanging simulation - the sound disappears as playback positions of the two "tape machines" cross each other.



mdaVocoder

Switchable 8 or 16 band vocoder with separate "VocInput" plug-in for adding a pitch-tracked oscillator to input vocals.



Appendix A Pro Audio Primer

The following glossary will help you to understand some of the more common terms found in the digital audio workstation world. Also, be sure to check out some of the following online resources:

- Open Labs User Forums – http://forum.openlabs.com
- Modern Recording Techniques – http://www.modrec.com
- Wikipedia – http://en.wikipedia.org/wiki/Professional_audio
- Wikiaudio – http://www.wikiaudio.org
- Audio Engineering Society – http://www.aes.org
- Recording.org – http://recording.org/index.php

Open Labs Pro Audio Primer

ASIO
(Audio Stream Input/Output), developed by Steinberg, is a cross-platform, multi-channel audio transfer protocol that is being adopted by many of the manufacturers of audio/MIDI sequencing applications. It allows software to have access to the multi-channel capabilities of a wide range of powerful sound cards. ASIO expands on the basic capabilities of a standard computer sound card, most of which can only provide stereo (two-channel) audio input and output. The ASIO specification defines the interface that manufacturers of professional audio sound cards must use to create an ASIO driver for their hardware. This driver allows the host audio/MIDI application to "see" all of the inputs and outputs available on the sound card. The user can then assign these I/O ports as needed for recording or playback when using an ASIO-compatible software program. This allows the users to record more tracks simultaneously than the previous limitation of two channels imposed by a standard sound card.

ATTACK
The initial transient or first part of the envelope of a signal. The beginning of a note. Also, see *Release.*

AUDIO
Of, pertaining to, or employed in the transmission, reception, or reproduction of sound.

AUDIO EFFECT
A special type of plug-in that accepts an audio stream and produces a new audio stream. Examples include compressors, distortion and reverb.

AUDIO EFFECT CHAIN
A collection of plug-ins that start with an audio input, then goes through a chain of optional audio effects before going out to an audio bus.

BALANCED LINE
A cable having two conductors and a ground connection and often surrounded by a shield. These lines are often used in professional settings to reduce or eliminate induced noise and interference from external electromagnetic sources.

BANK
A collection of Presets that are loaded into memory for easy access and low load time.

BITRATE
In digital multimedia, bit rate represents the amount of information, or detail, that is stored per unit of time of a recording. The bit rate depends on several factors:

1. The samples may use different numbers of bits.
2. The original material may be sampled at different frequencies.
3. The data may be encoded by different schemes.
4. The information may be digitally compressed by different algorithms or to different degrees.

Generally, choices are made about the above factors in order to achieve the desired trade-off between minimizing the bit rate and maximizing the quality of the material when it is played.

If lossy data compression is used on audio or visual data, differences from the original signal will be introduced; if the compression is substantial, or lossy data is decompressed and recompressed, this may become noticeable in the form of compression artifacts. Whether these affect the perceived quality, and if so how much, depends on the compression scheme, encoder power, the characteristics of the input data, the listener's perceptions, the listener's familiarity with artifacts, and the listening or viewing environment.
Experts and audiophiles may detect artifacts in many cases in which the average listener would not. Some musicians enjoy the distinct artifacts of low bit rate (sub-FM quality) encoding and there is a growing scene of net labels distributing stylized low bit music.

The bit rates in this section are approximately the minimum that the average listener in a typical listening or viewing environment, when using the best available compression, would perceive as not significantly worse than the reference standard:

- Audio (MP3)
    - 4 - kbit/s - minimum necessary for recognizable speech (using special-purpose speech codecs)
    - 8 - kbit/s - telephone quality
    - 32 - kbit/s - MW (AM) quality
    - 96 - kbit/s - FM quality
    - 128 - kbit/s - Typical "acceptable" music quality
    - 256 - 320 kbit/s - Near audio CD quality
- Video (MPEG2)
    - 16 kbit/s - videophone quality (minimum necessary for a consumer-acceptable "talking head" picture)
    - 128 - 384 kbit/s - business-oriented videoconferencing system quality
    - 1 Mbit/s - VHS quality
    - 5 Mbit/s - DVD quality
    - 15 Mbit/s - HDTV quality

CARDIOID MICROPHONE
A common mic pickup pattern designed to attenuate signals arriving 180 degrees off-axis, while fully picking up those sounds that arrive at the front (on-axis).

CONDENSOR MICROPHONE
A microphone that operates on an electrostatic principle rather than on the electromagnetic principle used in dynamic and ribbon mics. These microphones require phantom power, or a +48 volt charge.

CONTROL
A single GUI item that performs a task. Examples include list boxes, right click menus and ok buttons.

DC OFFSET
A DC Offset is an offset in the baseline, or zero mark of the waveform. A sample wave not centered on the zero baseline in the Waveform Display is said to have a DC Offset. This is usually caused by a calibration problem in the audio hardware. It is desirable to remove this offset after

recording because undesirable pops can occur when editing a file with a large DC offset. To remove DC offset, a number of options are available ranging from applying a Low-Pass Filter to the waveform to using correction features built into most Audio Editors such as Wavelab or Audition. Dedicated DC Offset plug-ins in several formats such as VST and Direct X are also available.

DECIBEL (dB)
A unit of audio measurement of sound-pressure level (SPL), signal level, and changes or differences in signal level. The decibel is a logarithmic (log) mathematical function that reduces large numeric values into smaller, more manageable numbers. Decibel is calculated as being 10 times the log of the ratio of two powers and 20 times the log of the ratio of two voltages.

DESTRUCTIVE EDITING
When the audio data recorded on a hard disk is altered and rewritten to disk in such a way that it can't be recovered in its original form**.**

DIALOG
A window that is not full screen. In this host all Dialog boxes will be modal (meaning they will be the foremost window and nothing else can be done until they are closed).

DYNAMIC MICROPHONE
A microphone that operates by electromagnetic induction to generate an output signal. When an electrically conductive coil of wire or ribbon is made to cut across the flux lines of a magnetic field, a current of specific magnitude and direction is generated within that coil or ribbon.

EFFECTS SEND
An auxiliary send feeding an effects device.

EQUALIZER
A frequency-dependent amplifier that enables a recording or mix engineer to control the relative amplitude of various frequencies in the audible bandwidth. Put another way, the equalizer lets you exercise tonal control over the harmonic content or timbre of a recorded sound.

EXPANDER
A device that increases the dynamic range of a signal.

FADER
A linear attenuation device or linear volume control.

FILE TYPES

- ***WAVE*** - **WAV (or WAVE)**, short for Waveform audio format, is a Microsoft and IBM audio file format standard for storing audio on PCs. It is a variant of the RIFF bit stream format method for storing data in "chunks", and thus also close to the IFF and the AIFF format used on Macintosh computers. Both WAVs and AIFFs are compatible with Windows and Macintosh operating systems. It takes into account some differences of the Intel CPU such as little-endian byte order. The RIFF format acts as a "wrapper" for various audio compression codecs. It is the main format used on Windows systems for raw audio.
- ***MIDI*** - ***Musical Instrument Digital Interface***, or ***MIDI***, is an industry-standard electronic communications protocol that defines each musical note in an electronic musical instrument such as a synthesizer, precisely and concisely, allowing electronic musical instruments and computers to exchange data, or "talk", with each other. MIDI does not transmit audio - it simply transmits digital information about a music performance.
- ***MP3*** - **MPEG-1 Audio Layer 3**, more commonly referred to as **MP3**, is a popular digital audio encoding and lossy compression format, designed to greatly reduce the amount of data required to represent audio, yet still sound like a faithful reproduction of the original

uncompressed audio to most listeners. It was invented by a team of European engineers who worked in the framework of the EUREKA 147 DAB digital radio research program and finally standardized by ISO/IEC in 1991

- ***OGG* - Ogg** is a patent-free, fully open multimedia bit stream container format designed for efficient streaming and file compression (storage).The name "Ogg" refers to the file format which includes a number of separate independent open source codecs for audio, video and text (e.g. subtitles). Files ending in the .ogg extension may be of any Ogg media file type, and because the format is free, Ogg's various codecs have been incorporated into a number of different free and commercial media players.

FREQUENCY
The rate at which an acoustic generator, electrical signal, or vibrating mass repeats a cycle of positive+ and negative- going amplitude. The number of cycles that occurs over the period of one second is measured in hertz (Hz). Often, the perceived range of human hearing is from 20 Hz to 18,000 Hz.

GAIN
Amount of amplification in dB.

GUI
A graphical user interface (or GUI, often pronounced "gooey") is a method of interacting with a computer through a metaphor of direct manipulation of graphical images and Virtual Live Controls in addition to text. GUIs display visual elements such as icons, windows and other gadgets. The precursor to GUIs was invented by researchers at the Stanford Research Institute (led by Doug Engelbart) with the development and use of text-based hyper links manipulated with a mouse for the On-Line System. The concept of hyper links was further refined and extended to graphics by researchers at Xerox PARC, who went beyond text-based hyper links and used GUIs as the primary interface for the Xerox Alto computer. Most modern general-purpose GUIs are derived from this system. For this reason some people call this class of interface a PARC User Interface (PUI) (note that PUI is also an acronym for perceptual user interface). The PUI consists of graphical Virtual Live Controls (often provided by Virtual Live Control toolkit libraries) such as windows, menus, radio buttons, check boxes and icons, and employs a pointing device (such as a mouse, trackball or touchscreen) in addition to a keyboard. Those aspects of PUIs can be emphasized by using the alternative acronym WIMP, which stands for Windows, Icons, Menus and Pointing device.

The GUI familiar to most of us today is either the Windows or the Macintosh operating systems and their applications originated at the Xerox Palo Alto Research Laboratory in the late 1970s. Apple used it in their first Macintosh computers. Later, Microsoft built on many of Apple's ideas in their first version of the Windows operating system for IBM-compatible PCs.

HARD DISK RECORDER
A system that uses a computer hard disk to record, edit, and reproduce digital data.

INSTRUMENT
A collection of plug-ins that starts with a MIDI input, then goes through optional MIDI effects, into a soft synth, and then into optional audio effects before going out to an audio bus. An instrument is set up anytime a user wants to play a soft synth.

LATENCY
Latency is the amount of time an application allocates to prepare a buffer full of audio data for playback. Lower latency settings add processing time because of the need to refill the smaller data buffers more often. You may need to increase your applications latency under the following conditions:

- You use lots of real-time effects, and you hear dropouts. Check the CPU meter for high

readings; try increasing the latency.
- Your sound card does not function well at lower latency. Some sound cards just do not function well at lower latency settings.
- Even if your application reports no problems, if you hear dropouts try increasing the mixing latency.

LIMITER
A device used to keep signal peaks from exceeding a certain level in order to prevent the clipping or distortion of amplifier signals, recorded signals on tape or disc, broadcast transmission signals, and so on.

LINE LEVEL
A signal level that is referenced to either 4 dBm (pro) or -10 dBV (semi-pro/consumer). Devices other than mics, speakers, and power-amplifier outputs operate at these levels.

MASTERING
The processing and transferring of a final, sequenced audio tape to a medium for duplication.

MFUSION
Open Labs mFusion software unifies multiple MIDI interfaces, MIDI keyboards, and MIDI controllers into a single MIDI control panel for assigning MIDI channels, providing keyboard splits, assigning MIDI controllers, and setting other parameters for incoming MIDI data. In mFusion this is all called mapping. Keyboards and other controls may be mapped from one type of MIDI data to another, or mapped to other functions such as keystroke shortcuts, application launching, or even changing the mapping of another MIDI controller. All of the MIDI data mapped by mFusion is unified into a "virtual" MIDI port so that any music application may receive data from all attached MIDI controllers through one single MIDI port. The individual MIDI ports from each device remain available, however, for maximum flexibility.

MIDI
Musical Instrument Digital Interface, or MIDI, is an industry-standard electronic communications protocol that defines each musical note in an electronic musical instrument such as a synthesizer, precisely and concisely, allowing electronic musical instruments and computers to exchange data, or "talk", with each other. MIDI does not transmit audio - it simply transmits digital information about a music performance.

MIDI CC
Short for "MIDI Continuous Controller", a specific kind of MIDI message. The modulation wheel of your synth, for example, sends out "MIDI Continuous Controller 1", or CC#1. Likewise, Volume is CC#7, GLIDE - Portamento Time is CC#5.

MIDI CHANNEL
A discrete portion of the complete MIDI signal that can include as many as 16 channels. Each channel carries independent messages; individual instruments tune in to these, just as a television tunes in to one of many channels. Instruments that can respond to multiple channels at once are called multi-timbral.

MIDI EFFECT
A special type of plug-in that accepts an MIDI stream and produces a new MIDI stream. Examples include arpeggiators, note repeat, chorders and MIDI echo.

MIDI HOST
A MIDI Host is an application that allows you to use virtual effects and instruments (plug-ins). They can be basic programs used to audition plug-in effects and instruments right up to a complex multitrack recording environment for audio and MIDI.

MIDI SEQUENCING
In the field of electronic music, a sequencer was originally any device that recorded and played back a sequence of control information for an electronic musical instrument. Nowadays, the term almost always refers to the feature of recording software which allows the user to record, play back and edit MIDI data. This is distinct from the software features which record audio data. Early analog music sequencers used control voltage/trigger interface, but were replaced by digital hardware- or software-based MIDI sequencers, which play back MIDI events and MIDI control information at a specified number of beats per minute.

MONOPHONIC
Of or noting a system of sound recording, reproduction, or musical instrument using only a single channel.

Multi-timbral
The capability of an electronic musical instrument to respond to and output multiple voice patches at one time.

MULTITRACK RECORDING
Multitrack recording ('multitracking' or just 'tracking' for short) is a method of sound recording that allows for the separate recording of multiple sound sources to create a cohesive whole. This is the most common method of recording popular music.
Multitracking can be achieved with hard disk based systems like an Open Labs production station, often employing a computer and multitrack audio recording software. Multitrack recording devices vary in their specifications, such as the number of simultaneous tracks available for recording at any one time; in the case of tape based systems this is limited by, among other factors, the physical size of the tape employed. For computer based systems the trend is towards unlimited numbers of record/playback tracks, although issues such as memory and CPU available will in fact limit this from machine to machine. It has to be noted that on computer based systems, the number of simultaneously available recording tracks is limited by the sound card discrete analogue or digital inputs.
When recording, audio engineers can select which track (or tracks) on the device will be used for each instrument.

At any given point on the tape, any of the tracks on the recording device can be recorded or playing back, so that an artist is able to record onto track 2 and, simultaneously, listen to track 1, allowing him to sing or to play an accompaniment to the performance already recorded on track 1. He might then record on track 3 while listening to track 2. All three performances can then be played back in perfect synchrony, as if they had originally been played and recorded together. This can be repeated until all of the available tracks have been used, or in fact, reused.
When recording is completed, the many tracks are "mixed down" through a mixing console to a two-track stereo recorder in a format which can then be duplicated and distributed. Most of the records, CDs and cassettes commercially available in a music store are recordings that were originally recorded on multiple tracks, and then mixed down to stereo. These stereo mixes can in turn be recorded onto two tracks of a four-track recorder, allowing additional sound to be layered on the remaining tracks.

NOISE GATE
A device that acts as an infinite expander, allowing a signal above the selected threshold to be passed through to the output at unity gain and without dynamic processing. When the input signal falls below this threshold level, the device effectively shuts down the signal by applying full attenuation to the output.

NON-DESTRUCTIVE EDITING
Editing a hard-disk sound file by moving pointers, without altering in any way the digital audio data originally recorded to disk.

NORMALIZING
A specialized gain-related process that makes the best use of a digital system's dynamic range by automatically determining the amount of gain required to increase the level of the highest amplitude signal to its full-scale amplitude value, and then increasing the level of the selected region or entire file by this gain ratio.

OCTAVE
The most basic musical interval, and the second harmonic of the natural scale, which represents a doubling of frequency.
The diatonic scales in the western harmonic system have 8 notes to the octave, hence the name. There are 12 semi-tones to an octave and in some middle-eastern scales, 24 quarter-tones.

OMNI-DIRECTIONAL MICROPHONE
A mic that outputs signals received from all incident angles at the same (or nearly equal) level.

OSCILLATOR
In modern musical terms, an oscillator is a component that originates a signal. The period of an oscillator is the time it takes to oscillate, or repeat. The period is also referred to as frequency or pitch. The classic analog synthesizers had regular periodic oscillator shapes such as Sine, Triangle, Pulse, Ramp, and Saw. They also sometimes had a special non-periodic oscillator called noise. Although not technically an oscillator because it does not repeat a pattern, the noise oscillator generates semi-random values within a certain frequency.

PHANTOM POWER
Power for a condenser mic that comes directly from the console through balanced mic cables by supplying a positive DC supply voltage of 48V (usually) to both conductors (pins 2 & 3) with respect to pin 1. This voltage is distributed through identical value resistors so that no differential exists between the two leads; therefore, the voltage is electrically invisible to the alternating audio signal. The DC circuit is completed by connecting the negative side of the supply to the cable's grounding shield.

PLUG-IN
Plug-ins are audio processing modules. They are usually of the VST or DX format. They are usually third party external DLLs that will usually take an input stream and produce output streams. In the case of music software this generally takes the form of an instrument, such as a synth or sampler, or an effect processor, such as a reverb or compressor.

POLAR PATTERN
A polar graph of the sensitivity of a microphone at all angles of sound incidence relative to the sensitivity on-axis.

POLYPHONIC
The capability of an electronic musical instrument to output multiple notes at one time.

PRESET
A collection of Instruments and/or Audio Effect chains with unique settings for each plugin in the preset.

PRESET GROUP
User defined group of Presets contained within a Bank.

PUNCH-IN PUNCH-OUT
The entering into and out of record mode on a track that contains existing program material for the purpose of correcting or erasing an unwanted segment.

QUANTIZATION
The amplitude component of the digital sampling process. In an A/D converter, the process of generating a binary number (made of 1s and Os) that represents the voltage of the analog waveform at the instant it is measured or sampled.

RELEASE
The final portion of a note's envelope, which falls from the sustain signal level to silence.

SAMPLE RATE
Level of accuracy with which audio data is stored. The higher the better the sound quality, but the more costly in computer time and memory.
In digital audio, common sampling rates are:

- 8,000 Hz - telephone, adequate for human speech
- 22,050 Hz - radio
- 32,000 Hz - miniDV digital video camcorder
- 44,100 Hz - audio CD, also most commonly used with MPEG-1 audio (VCD, SVCD, MP3)
- 47,250 Hz - world's first commercial PCM sound recorder by Nippon Columbia (Denon)
- 48,000 Hz - digital sound used for digital TV, DVD, DAT, films and professional audio
- 50,000 Hz - first commercial digital audio recorders from the late 70's from 3M and Soundstream
- 50,400 Hz - sampling rate used by the Mitsubishi X-80 digital audio recorder
- 96,000 or 192,400 Hz - DVD-Audio, some LPCM DVD tracks, BD-ROM (Blu-ray Disc) audio tracks, and HD-DVD (High-Definition DVD) audio tracks
- 2.8224 MHz - SACD, 1-bit sigma-delta modulation process known as Direct Stream Digital, co-developed by Sony and Philips

SEQUENCING
MIDI Sequencer - In the field of electronic music, a sequencer was originally any device that recorded and played back a sequence of control information for an electronic musical instrument. Nowadays, the term almost always refers to the feature of recording software which allows the user to record, play back and edit MIDI data. This is distinct from the software features which record audio data. Early analog music sequencers used control voltage/trigger interface, but were replaced by digital hardware- or software-based MIDI sequencers, which play back MIDI events and MIDI control information at a specified number of beats per minute.

SMPTE TIME CODE/MTC MIDI TIME CODE
Time codes contains binary coded decimal hour:minute:second:frame identification and 32 bits for use by users. There are also drop-frame and color framing flags and three extra 'binary group flag' bits used for defining the use of the user bits. The formats of other forms SMPTE timecodes are derived from that of the longitudinal timecode.

Time code can have any of a number of frame rates: common ones are:

- 24 frame/s (film)
- 25 frame/s (PAL color television)
- 29.97 (30/1.001) frame/s (NTSC color television)
- 30 frame/s (American black-and-white television) (almost obsolete)

In general, SMPTE timecode frame rate information is implicit, known from the rate of arrival of the timecode from the medium, or other metadata encoded in the medium. The interpretation of several bits, including the "color framing" and "drop frame" bits, depends on the underlying data rate. In particular, the drop frame bit is only valid for a nominal frame rate of 30 frame/s: see below for details. SMPTE/MTC is a position and timing reference that indicates the current location in the project and how quickly the project should be playing. Time code labels the position in a

project in hours, minutes, seconds, and frames. The speed of playback is indicated by a frame rate. Time code is recorded onto tape using a device called a time code generator. The process of recording a time code signal onto a track is called striping. Normally, the start of a tape stripe has a particular time, expressed in hours, minutes, seconds, and frames. For example, the tape stripe might start at 00:00:00:00, 01:00:00:00, or any other time. The material recorded on the tape usually starts anywhere from 10 seconds to several minutes after the start of the time code. Sometimes, the tape stripe starts at a time like 00:59:50:00, and the material starts 10 seconds later, at 01:00:00:00.

SOFT SYNTH
A soft synth, also known as a software synthesizer or virtual instrument is a computer program for digital audio generation. Computer software which can create sounds or music can accomplish the same tasks as dedicated hardware. Softsynths can be cheaper and more portable than dedicated hardware, and easier to interface with other music software such as MIDI sequencers.

SOLO
A monitor function that lets the engineer hear a single instrument or group of instruments without affecting the studio's headphone monitor mix, recorded tracks, or mixdown signal.

STREAMING
A system for transmitting audio and/or data through a transmission media (such as the web) in real-time.

SUSTAIN
In a traditional envelope, the level at which an enveloped setting remains after the Attack and Decay stages until the key is released.

SYNC
The locking of relative transport or playback speeds of various devices to allow them to work together as a single, integrated system.

TRANSPOSE
To write or perform (a composition) in a key other than the original or given key.

UNBALANCED LINE
A cable having only one conductor plus a surrounding shield, in which the shield is at ground potential. The conductor and the shield carry the signal.

VST/VSTi
Virtual Studio Technology: Its acronym VST refers to an interface standard for connecting audio synthesizer and effect plug-ins to audio editors and hard-disk recording systems. VST and similar technologies allow the replacement of traditional recording studio hardware with software counterparts. A VST instrument (or VSTi) is a type of VST plug-in that is generally used to synthesize sound or play-back sampled audio. Types of VST instruments include (among others) virtual synthesizer devices and sampler devices. VST instruments can be played in real-time when used with an appropriate software and hardware configuration.

VSTi LINK
The connection between a plug-in parameter and a physical hardware MIDI control (or a virtual soft control). An example of a link would be when a user moves a hardware MIDI knob and the volume output of a soft synth changes.

WAVEFORM

A usually graphic representation of the shape of a wave that indicates its characteristics (as frequency and amplitude). When talking about synthesizers, the term generally refers to the most common types of waveforms, such as the sine, triangle, sawtooth, and pulse waves. It can also refer to digital waveforms which are rendered from pre-recorded samples.

WORD CLOCK
A word clock is a device used to create a regular pulse used to synchronize other devices, such as digital audio tape machines and compact disc players, which interconnect via digital audio. S/PDIF, AES/EBU, ADAT and other formats use a word clock. A word clock neither produces nor uses timecode; it is used entirely to keep a perfectly-timed and constant bitrate to avoid data errors.

Appendix B – Internet

Connecting an Open Labs system to the Internet

All Open Labs systems can connect to the internet via the embedded ethernet port. Usually, this is just as simple as plugging in an active ethernet cable from your modem to the Open Labs system.

The advantages of having an internet connection are, you can schedule remote support sessions with Open Labs tech support. Remote Support is support involving an OL technician "taking control" of your Open Labs system over the internet and giving you the best possible support available. Remote Support will be detailed later in this guide.

If you are completely new to using the internet this guide will help you get your Open Labs system connected to the net.

Finding an Internet Service Provider

You must find an internet service provider (ISP) to connect to the internet. Time Warner, and AT&T are two, but there are many more. Usually, ISP's are located in certain areas, and the area you are located in determines what type of ISP you can use.

These are the internet access technologies that will work with Open Labs systems.

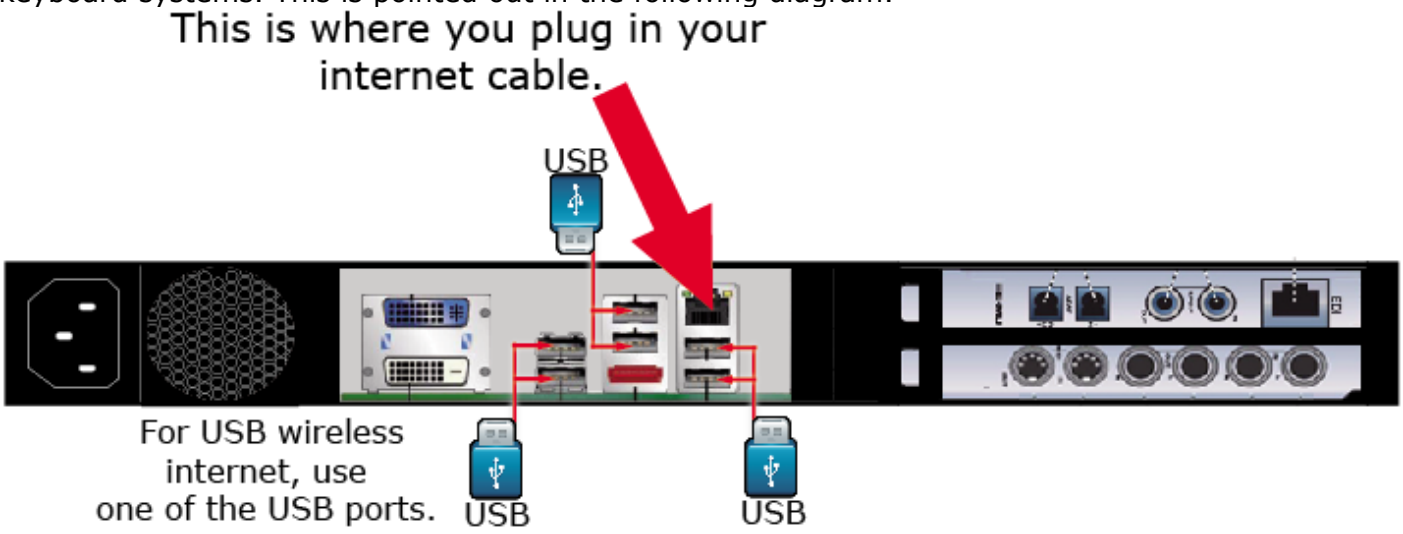
- ***Cable modems:*** The best performing and most affordable option available to customers, most cable TV providers offer broadband Internet access.
- ***DSL:*** An excellent choice for businesses, DSL typically offers better reliability than cable modems. However, DSL tends to be more expensive than cable modems for similar levels of service.
- ***Dial-up:*** (Requires a telephone modem, not included with Open Labs systems)The slowest method of connecting to the Internet, dial-up enables you to connect to the Internet using your existing phone lines. Dial-up is convenient because it is available to any location with a phone. However, slow performance makes using the Internet frustrating.
- ***Satellite:*** Satellite broadband services provide high-speed Internet access to any location with a clear view of the sky (currently available in North America and certain other locations). Satellite services may be the only broadband option for people living in rural areas. The cost of satellite services is significantly higher than other services. While you can transfer large files quickly with satellite, browsing the Web or playing online games can seem slower than with dial-up because of the delay caused by sending signals to and from satellites.

If you are already paying for cable television, or satellite tv, contact your provider, they may have internet plans available.

To connect to a wireless connection, you will need to purchase a wireless USB adapter. Also, your modem will need to be wireless compatible. Consult your ISP to see if they offer modems offereing wireless access points.

Location of the Ethernet (internet) port

Regardless of the generation, the ethernet port is located in the same place on Open Labs keyboard systems. This is pointed out in the following diagram.

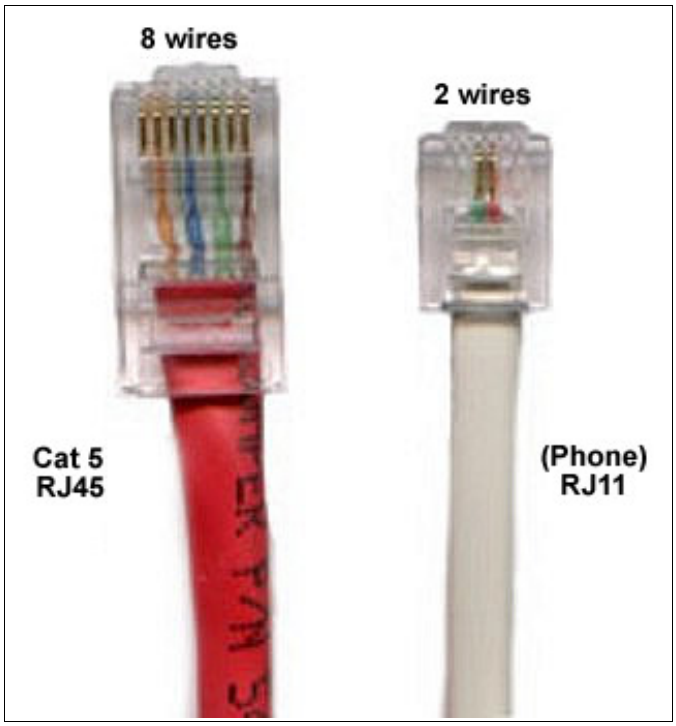


The next image shows you where to connect the cables or devices for internet access using the DBeat system.

What is an Ethernet Cable?

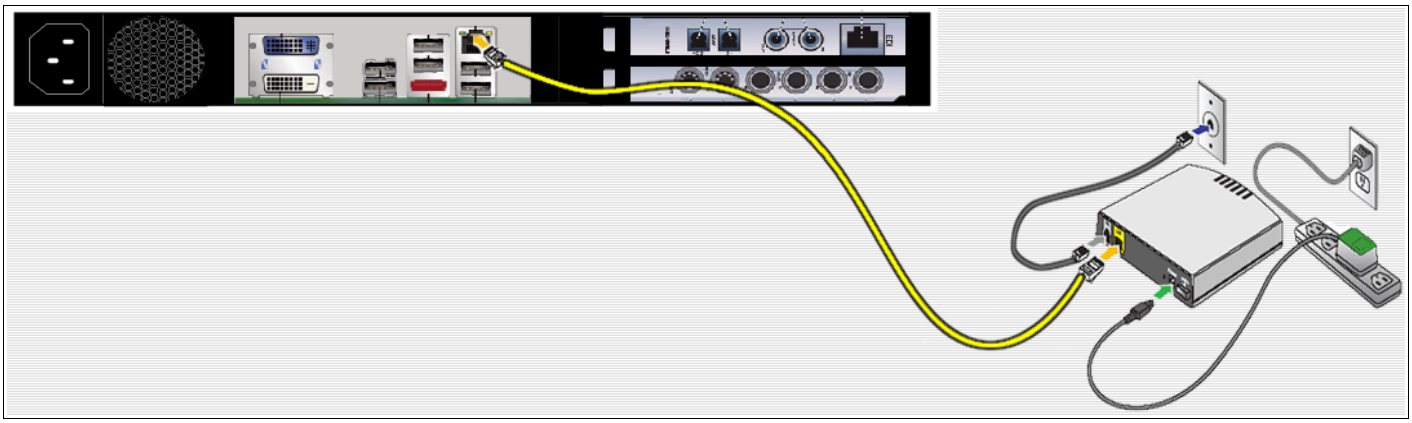
The cable that you need to connect from your Open Labs system to a broadband modem is called an *ethernet cable*. Ethernet cables are technically known as *Cat 5 RJ45.*

An ethernet cable resembles a standard telephone cable. Telephone cables only have two wires, whereas the ethernet cable has eight. The image to the right shows you the difference.



DSL / Cable Modem Connection Diagram

The following diagram will show you how to connect your Open Labs system to your modem.

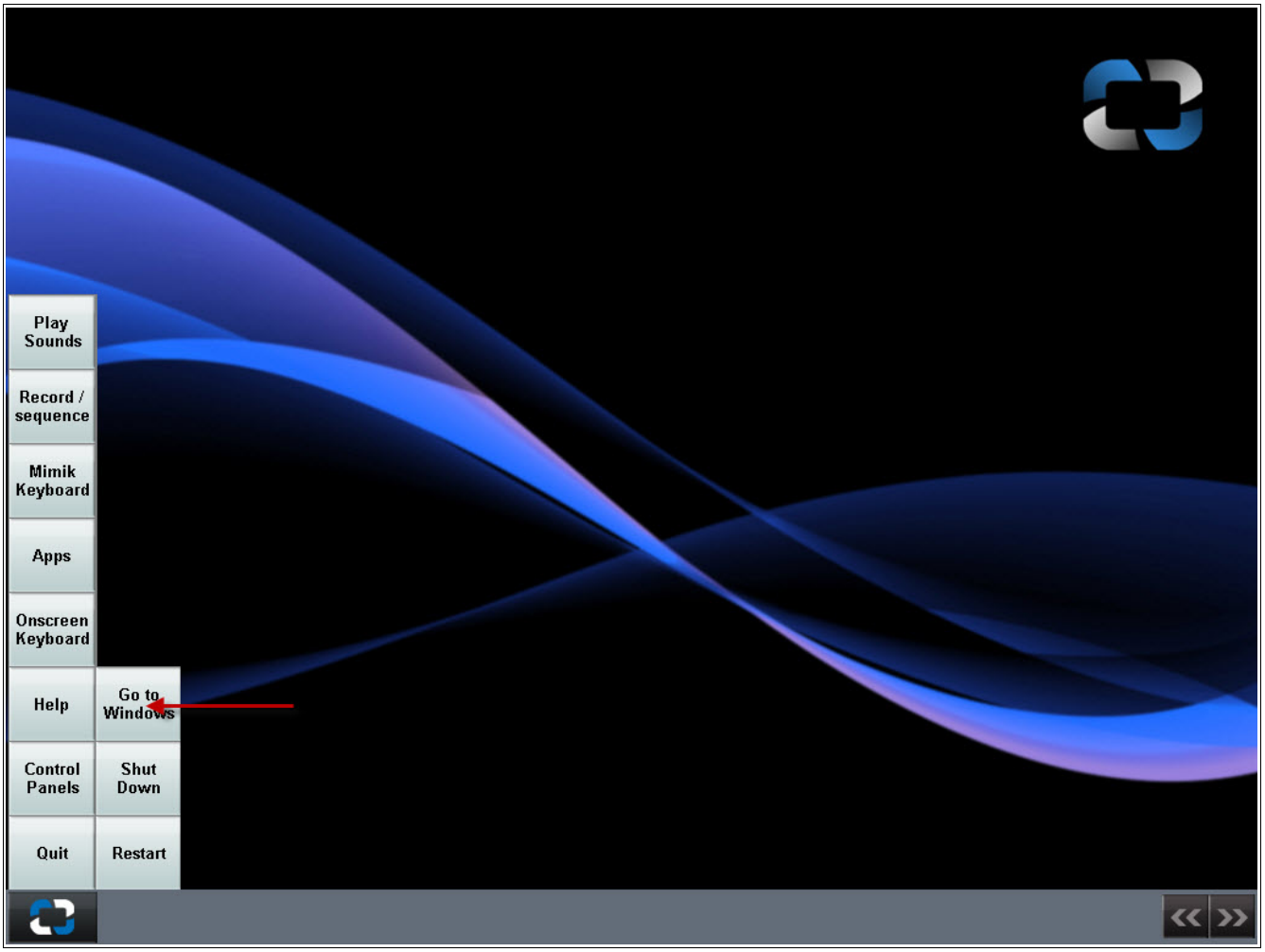


First, make sure your Open Labs system is turned off.

1. Plug an ethernet cable from the Open Labs system to the modem.
2. Connect your modem to either a telephone line (DSL), or to a coaxial jack (cable modem).
3. Connect the signal from either the DSL and Coax line into your modem.
4. Connect the power of the modem to a power source. *It is advised to use a surge protector power strip if you have one.*
5. Power up the modem, and wait for all the lights to come on. Refer to the manual of your modem to determine which lights show you that the modem is receiving a live net signal.
6. Once that is done. Power up the Open Labs system.

Your system should be connected to the internet now. Go ahead and open up a web browser.

Open a Web Browser to Test the Internet Connection
Exit the Open Labs shell. Press the [Quit] button, then select [Go to Windows].



Try to connect to the Open Labs home page using the web browser Firefox.

Press the Start Menu in the lower left and select; "Internet - Mozilla Firefox".

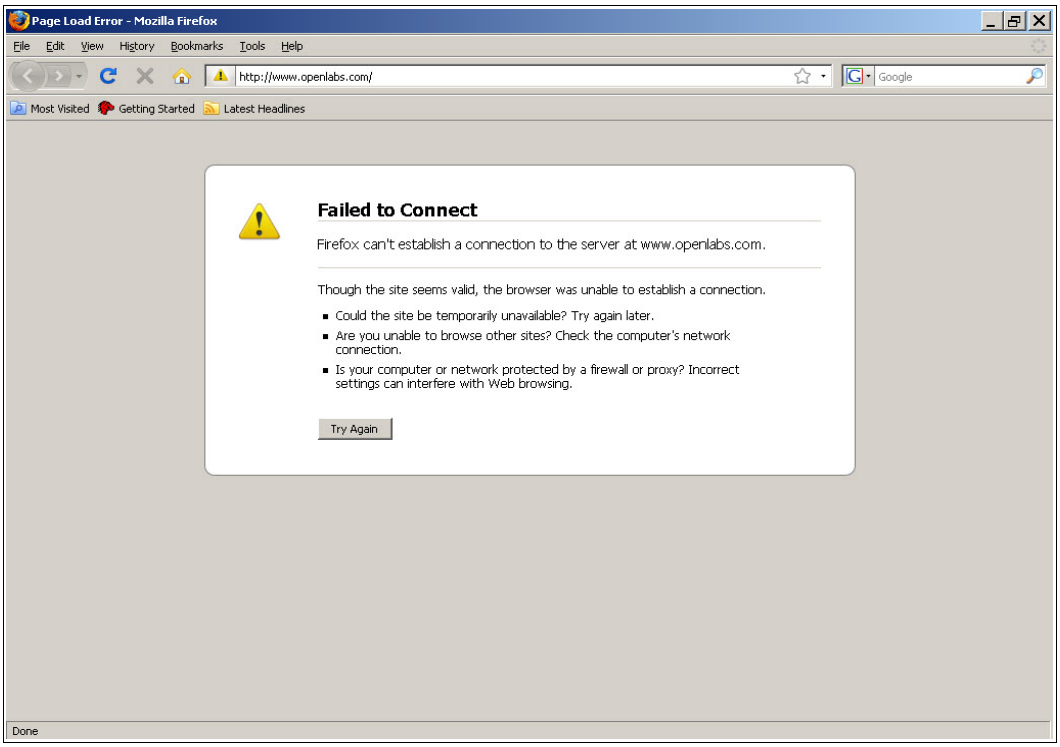


You will know you are connected to the net when you see the Open Labs home page.

This image shows Mozilla Firefox open with www.openlabs.com as the web page.



If you see the following page, then it means the internet connection is not working.



Troubleshooting a Connection

Because there are a few different things to check, troubleshooting an internet connection can be tricky. If you have an active internet connection, most issues can be resolved just by checking a few options.

First off, let's get you familiar with the various levels involved with the internet connection.

The levels are:

- Active internet connection - This means you have an active subscription with an ISP.
- Modem - Provided by your ISP
- Open Labs system - Connected to your modem.

To locate and resolve an issue requires you to figure out which level the issue resides.

If your active internet connection is working properly, and all status lights are on, but you are still unable to reach the internet:

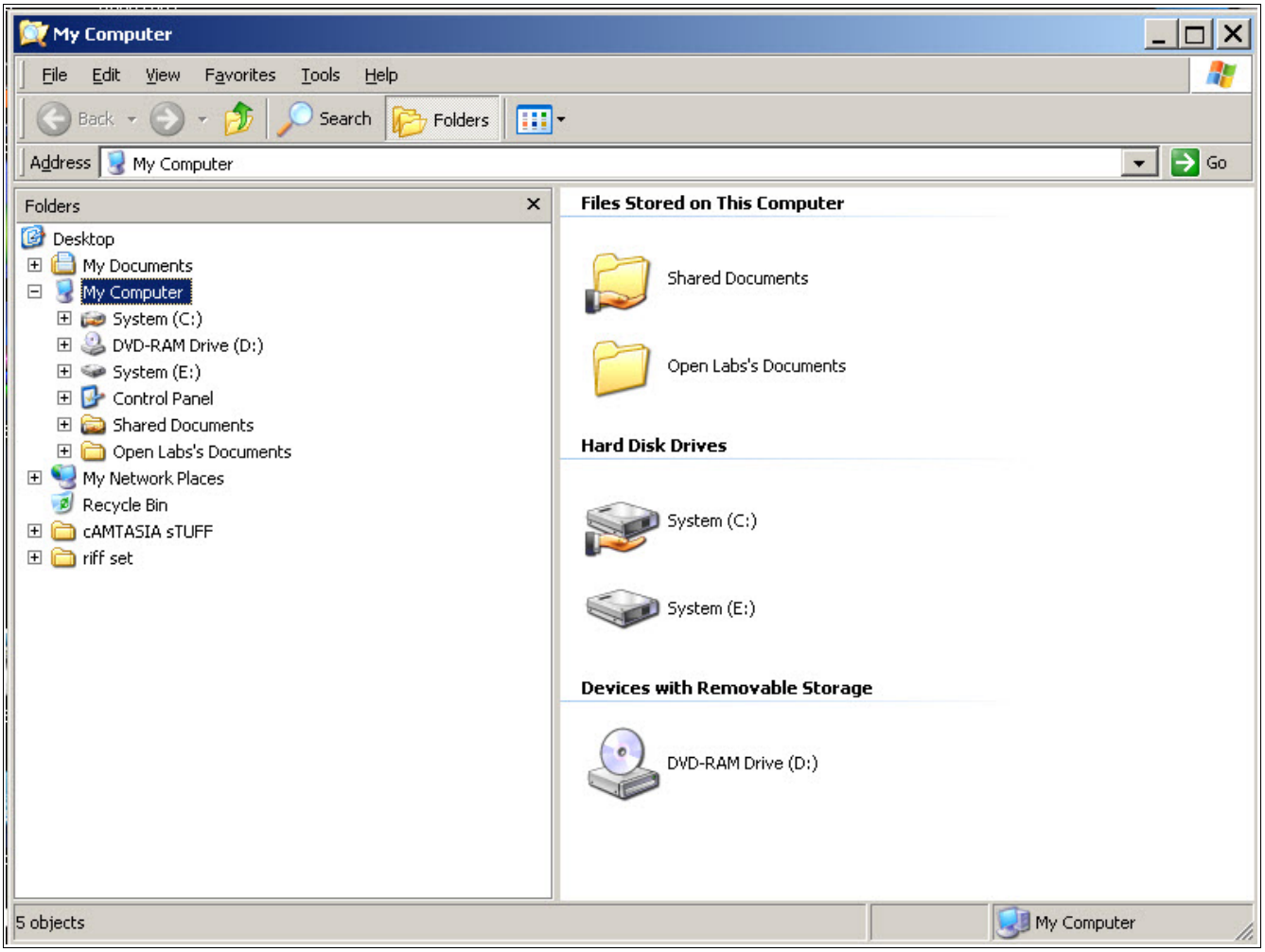
Check the network connections first.

On a QWERTY keyboard press [Windows key + E].

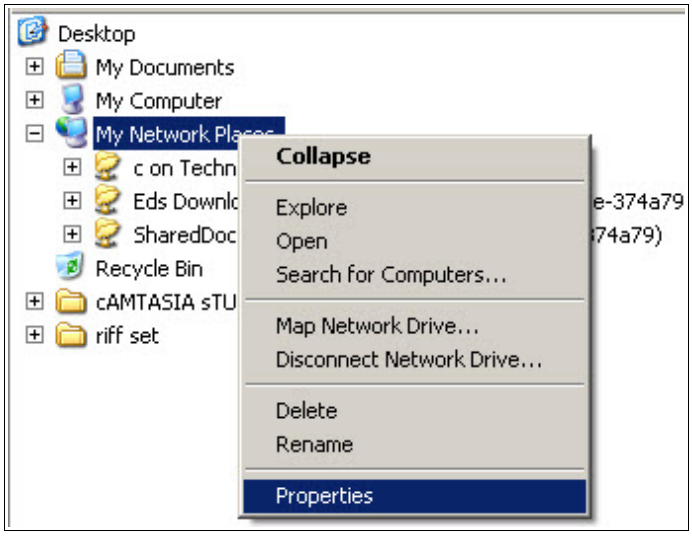


This opens Windows explorer. As shown on the following page.

Windows Explorer View

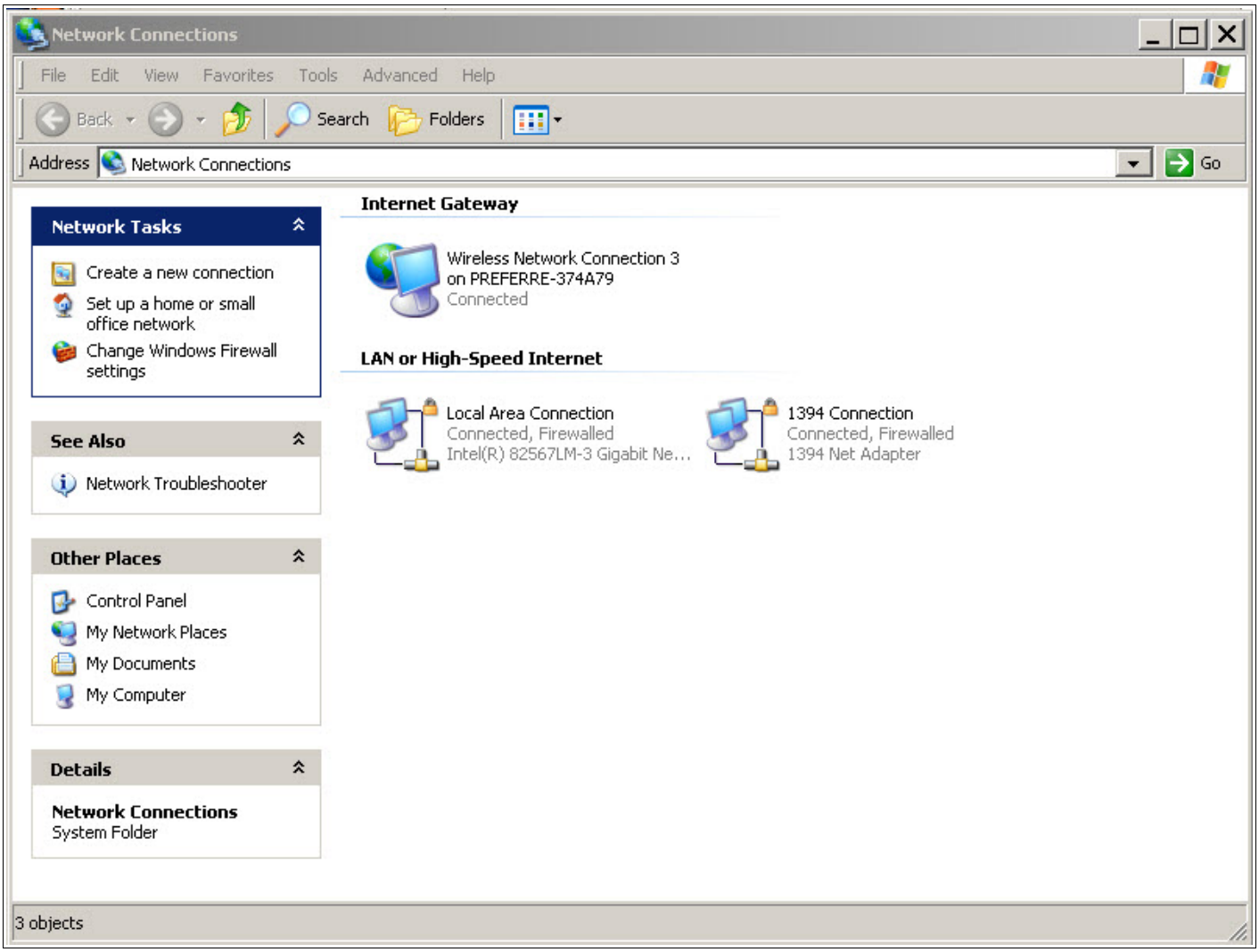


Select "My Network Places", right-click it and select properties.



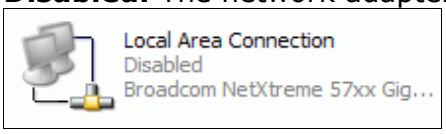
This will launch the Network Connections window. *As shown on the following page.*

Understanding the Network Connections Window

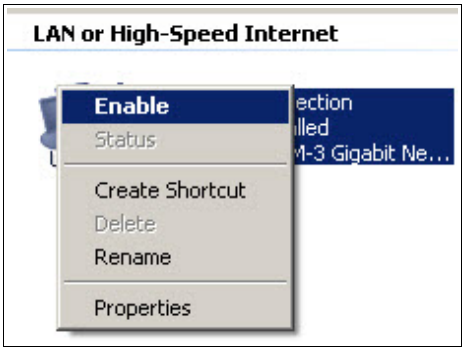


The following examples explains the icons from the Network Connections window.

Disabled: The network adapter has been manually disabled.



To Re-enable the network adapter, right-click it and click **Enable.**

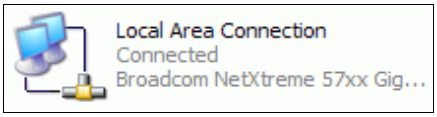


Not connected: The Open Labs system is unable to connect to the wireless network.



For wireless connections, please read the next section of this guide under **Wireless Connections.**

Connected: Your Open Labs systems is properly configured and connected to your modem or wireless network.



Windows XP says your network adapter is connected, but you can't reach the Internet

If you reached this section, your Open Labs system is probably connected to your home network properly, but there is a failure with your router, modem, or Internet connection. In this case:

1. First, verify that you can't reach the Internet. Start Mozilla Firefox, and attempt to connect to the following Web sites: www.openlabs..com, www.google.com, and www.microsoft.com. If any of the Web sites open correctly, your Internet connection is functioning properly. If all of the Web sites fail to open, continue following these steps.
2. Unplug your modem. If you are unsure which device might be your modem, it is the device that is connected directly to your phone line (if you have DSL) or cable connection (if you have a cable modem). Wait one minute, and then plug your modem back in again.
3. If you have a router connected to your modem, unplug it. If you are unsure which device might be your router, look for the device that is connected directly to your modem. Wait one minute, and then plug your router back in again.
4. Restart your Open Labs system.
5. If you have more than one computer on your network, turn the other computers off. Also, close any network applications that you might have open, including file sharing and peer-to-peer applications. These applications can consume so much bandwidth that they might prevent you from being able to browse the Web. Repeat step 1 to retest your Internet connection. If the connection works now, the problem is that one of your applications is consuming too much of your Internet bandwidth or attempting to connect to too many different computers. Close the application, or configure it to use less bandwidth or fewer connections.
6. If you have previously configured your Open Labs system with a static IP address (for example, to connect to your office network), you probably need an automatic IP address at home. Fortunately, you can configure Windows XP to use both by following the steps in Automatic configuration for multiple networks.
7. If you are using a router, unplug the network cable that connects your modem to your router, and connect your Open Labs system directly to your modem. Then restart your system. If your system connects properly after restarting, the problem is with your router. Contact your router manufacturer for support.
8. If possible, connect a different computer to your modem, and attempt to access the Internet. If the computer is able to access the Internet, you might have a problem with your Web browser. Contact Open Labs for support.
9. Your modem or your Internet connection has failed. To resolve the problem, contact your ISP technical support. To speed up the troubleshooting process, describe the steps you have already taken, including restarting your modem, router, and computer, and connecting your computer directly to your modem.

After following these steps, you should have identified the problem as your router, modem, Internet connection, or Web browser, and then either reconfigured or repaired the failed component.

Unplugged: The Open Labs systems cannot detect a connection to your modem.

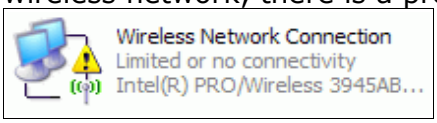


If Windows indicates that your network adapter is unplugged, it could be because of several different problems. Follow these steps to troubleshoot the problem. (You can stop following the steps if your network connection starts to work.)

1. Verify that both ends of the network cable are properly connected.
2. If the cable is properly connected, verify that your modem and router are plugged in and turned on.
3. Replace the network cable with a new cable. You might have a faulty network cable.
4. The network adapter on your computer might have failed. If possible, connect a different computer to the same network cable. If the connection works, the problem is with your network adapter. Contact Open Labs technical support. If the other computer also fails to connect and you have already tried a new network cable, your network hardware (the device you connect the network cable to) has failed. Contact technical support for the network hardware, or replace it.

After following these steps, you should have identified the problem as your computer's network adapter, the network cable, or your network equipment, and then either replaced or repaired the failed device.

Limited or no connectivity: The Open Labs system is properly connected to the modem or wireless network, there is a problem between the modem and the internet.



It could be due to several different problems, including a failed Internet connection, a misconfigured router, or a misconfigured network adapter. Follow these steps to troubleshoot the problem. (You can stop following the steps if your network connection starts to work.)

1. In the Network Connections window, right-click your network adapter, and then click Repair.
2. Unplug your modem. If you are unsure which device might be your modem, it is the device that is connected directly to your phone line (if you have DSL) or cable connection (if you have a cable modem). Wait one minute, and then plug your modem back in again.
3. If you have a router connected to your modem, unplug it. If you are unsure which device might be your router, look for the device that has at least two network cables: one that connects to your modem; and another that connects to your computer. You may also have a wireless router, which would be connected to your modem and would have one or more antennas. Wait one minute, and then plug your router back in again.
4. Restart the Open Labs system.
5. If your network adapter still shows "Limited or no connectivity" and you have customized your router's configuration, verify that the router has DHCP (Dynamic Host Configuration Protocol) enabled. Enable DHCP, and then restart your computer. DHCP automatically assigns an Internet Protocol (IP) address to your computer, which uniquely identifies your computer on your network. For instructions on how to enable DHCP for your router, refer to your router's documentation.
6. If you are using a router, unplug the network cable that connects your modem to your router, and connect your computer directly to your modem. Then restart your computer. If your computer connects properly after restarting, the problem is with your router. Contact your router manufacturer for support.
7. If your network adapter still indicates "Limited or no connectivity" when your computer is

connected directly to your modem, contact your Internet service provider (ISP) for support. To speed up the troubleshooting process, describe the steps you have already taken, including restarting your modem, router, and computer, and connecting your computer directly to your modem. The "Limited or no connectivity" error message indicates that your computer is properly connected to your home network; however, the ISP's DHCP server is not assigning it an IP address. This problem could be caused by a failed DHCP server on the ISP's network, a failed modem, or a problem with your Internet connection. Alternatively, the ISP may require you to use a "static" IP address, which only its support staff can help you configure.

After following these steps, you should have identified the problem as your router, your modem, or your Internet connection, and then either reconfigured or repaired the failed component.

Internet Connection Issues

The information in this section pertains to both wireless and wired internet connections.

What is an IP?

IP stands for Internet Protocol, and is used for communicating from a source to a destination. Most connection issues can be traced back to this subject.

When you connect to your internet service provider, your Open Labs system obtains an IP address from one of your ISPs servers. IP addresses sometimes need to be "renewed".

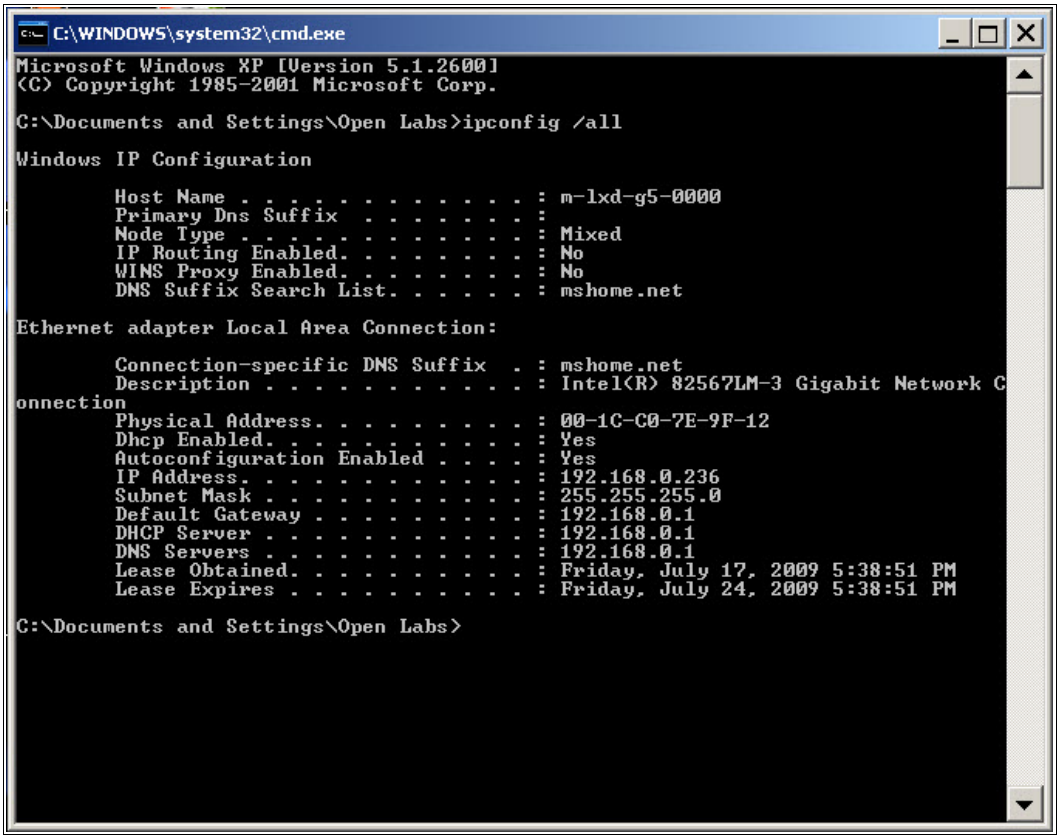
Release and Renew an IP

Simple way:

1. Press **[Windows key + E]** to open **Windows Explorer**.
2. Right-click on **Network Places** and select **Properties**.
3. Right-click the **Local Area Connection** or **Wireless** icon under **LAN or High-Speed Connection**, select **Disable** to release your IP address.
4. After it has been disabled, right-click it again and select **Enable** to renew your IP address.

Advanced:

1. Exit the Open Labs shell, then press the **Start Menu** button.
2. Select **Run**. Type in **cmd**. This will open the **Command Prompt**.
3. Type **ipconfig /all** to see the current status of the Open Labs system's IP address(es).



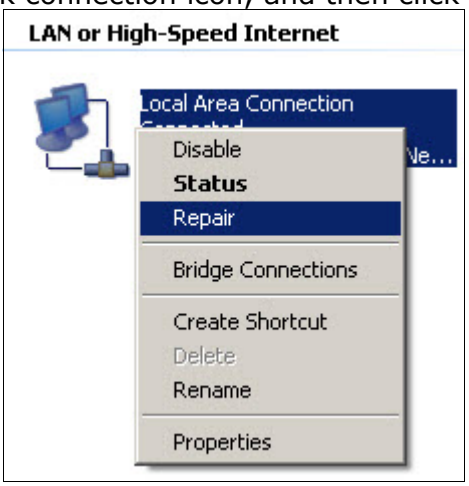
4. If the system is holding a current IP address, type **ipconfig /release** to release the current IP address.
5. Type **ipconfig /renew** to obtain a new IP.

If you go from location to location using networks, you may need to do this to get online.

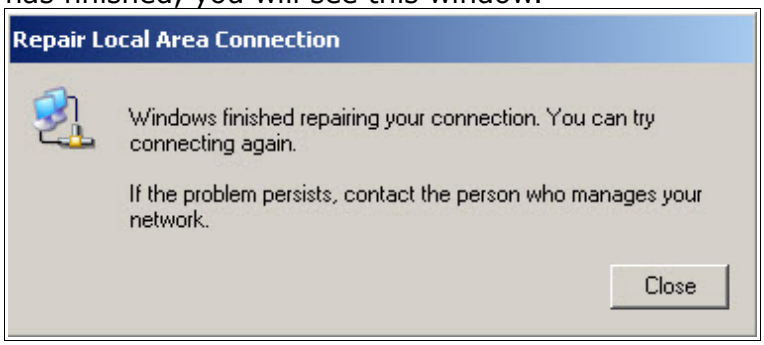
Repair a Connection

Sometimes your Network connection will be connected, yet it still will not let you online. Using the "Repair" may get you back online. This command basically "flushes" old information.

1. Right-click on your network connection icon, and then click **Repair**.



2. Once the repair has finished, you will see this window.



The repair option will not be available if no ethernet cable is plugged in, or if the connection is disabled.

Wireless Connection
If you want to use wireless internet, you will need an internet service, wireless modem, and a USB wireless adapter.

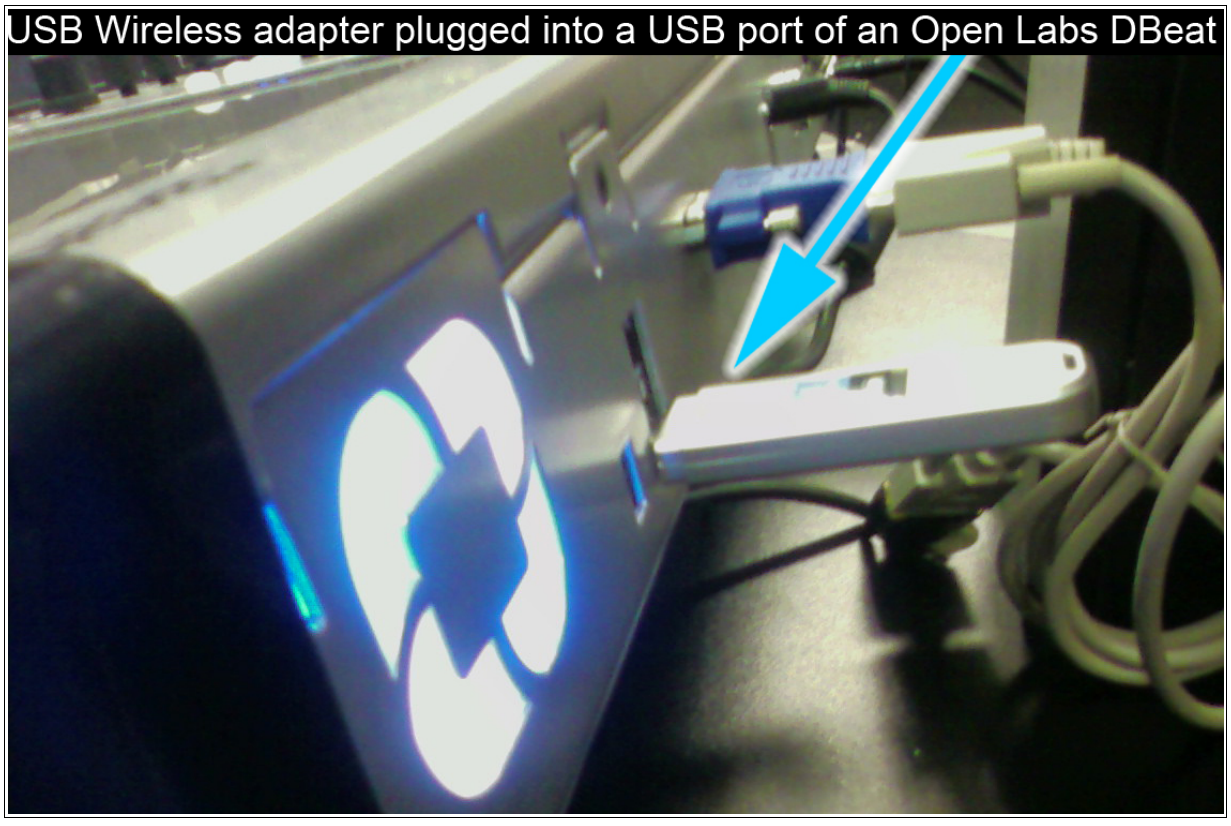
Wireless modem



USB wireless adapter



Once the wireless software is installed, plug in your USB wireless adapter. As shown in the following image.



Installing and Configuring the Wireless Connection
If you have an active internet connection, wireless modem, and a wireless USB adapter you have everything that is required to connect your Open Labs system to a wireless connection.

The first thing you need to do is setup your modem for wireless. Please contact your ISP, they will

help you setup up that device. They will help you enable the wireless portion, and help you create a "Wireless Key". The key is a password that will allow only you to gain access to your wireless internet. Otherwise, anyone with a wireless device in range of your network can connect and use your internet.

Once that is done, you can move on.

The second thing that needs be done is the installation of the USB wireless device drivers to the Open Labs system. Never plug the USB wireless in first. It will be unable to function without the manufacturers drivers.

Place the manufacturers installation disc into the optical (CD/DVD) drive and close the tray.

The software will probably auto-run, consult the users manual for that product for instructions on installation.

Follow the on-screen instructions.

Once the software is installed there should be a screen prompt that says "*Please connect the USB wireless device now*".

Pick a USB port on your Open Labs system, and plug the USB adapter in. Consult your devices documentation for instructions regarding connecting to your wireless network.

Allow Windows to Control the Wireless Connection
Most wireless adapters come with its own software to maintain the internet connection, however these applications run in the background and can intefere with system performance.

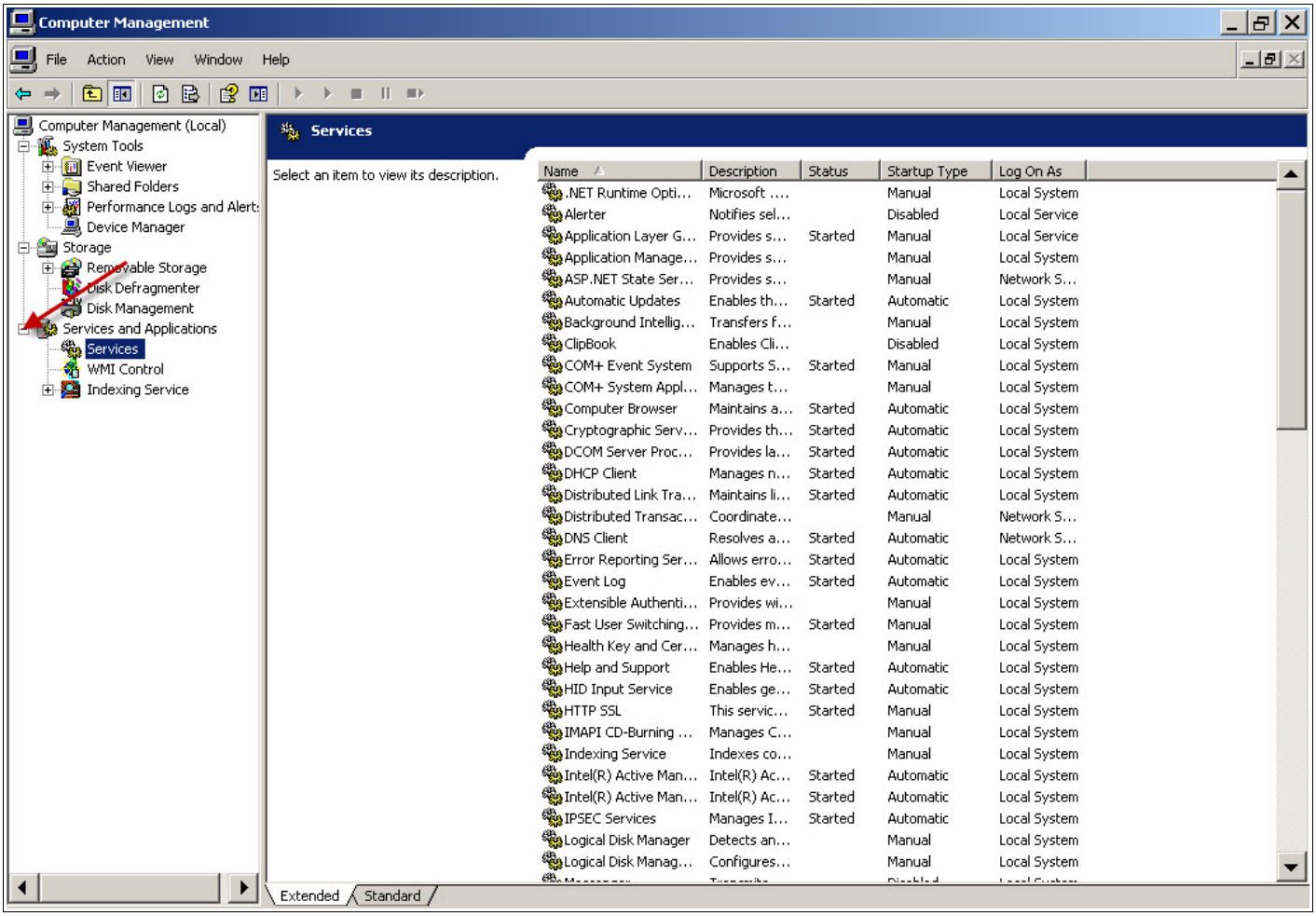
Windows can maintain the wireless connection from Network Connections. First, you need to check and make sure "*Wireless Zero Configuration*" utility is set to start.

Right-click "*My Computer*" and select "*Manage*".

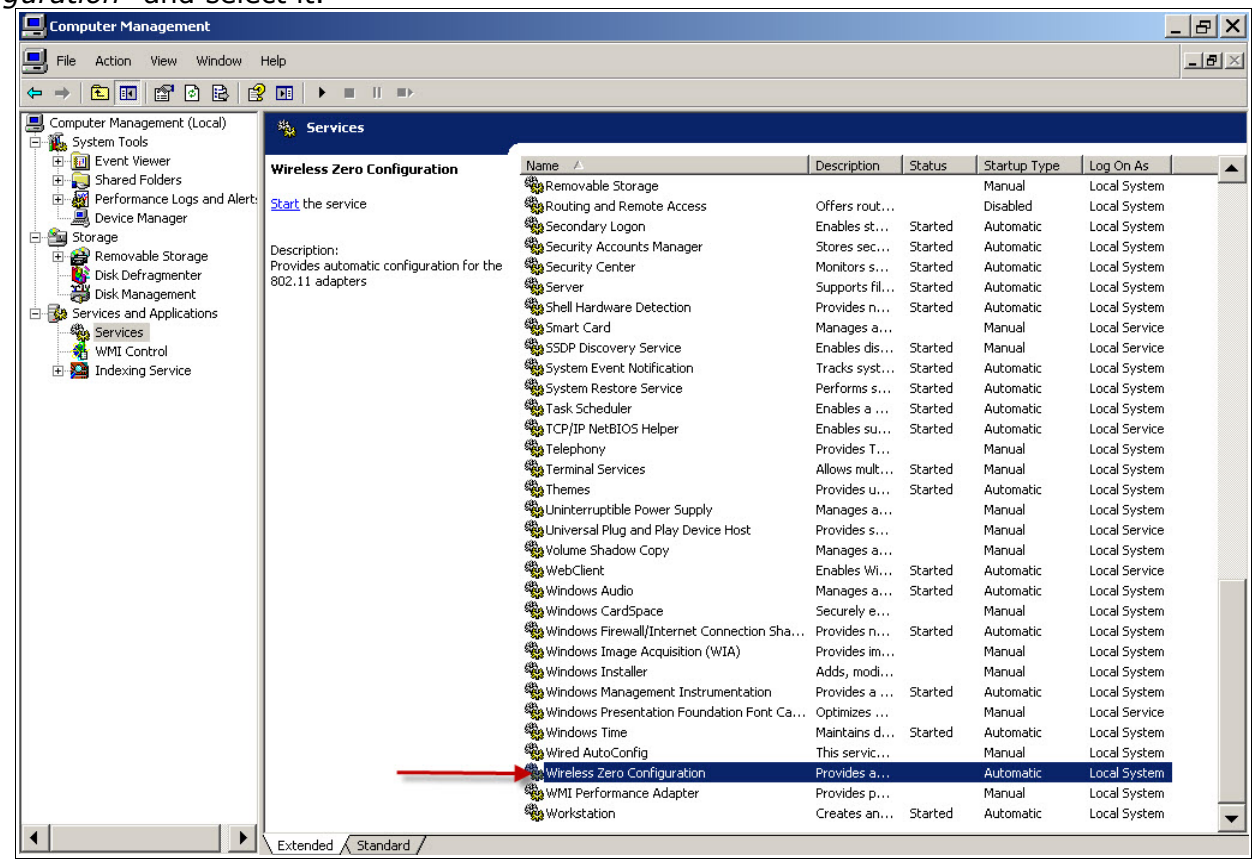


This will open up the "*Computer Management*" screen.

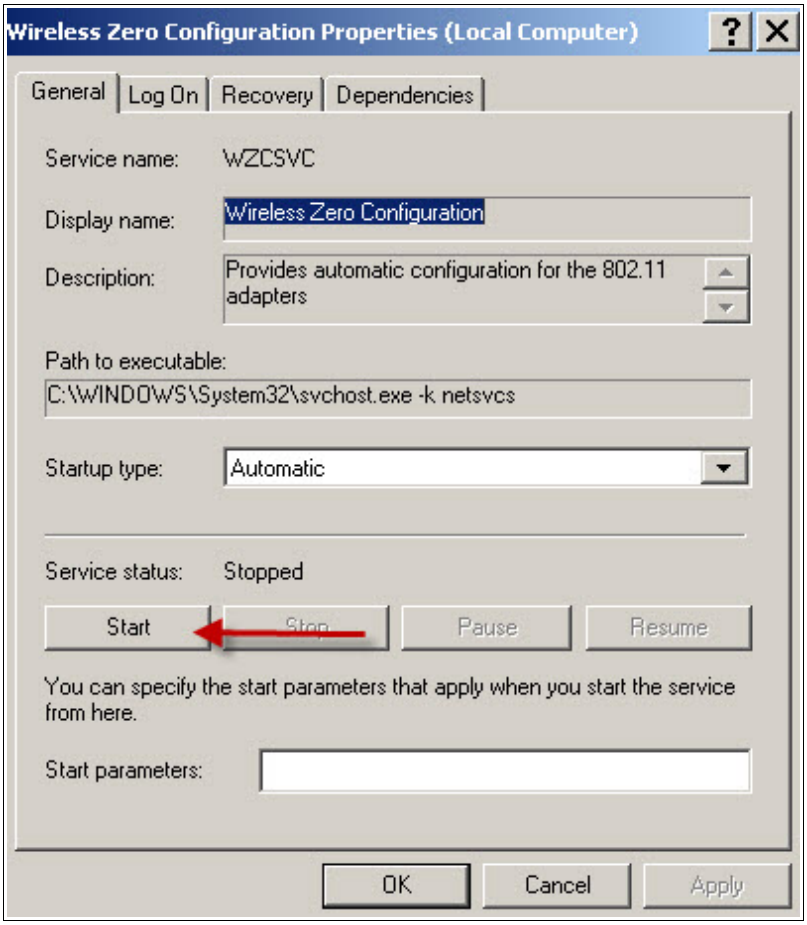
Under the Computer Management window, in the left column click on the "+" symbol beside "*Services and Applications*" this will expand that option. Select "*Services*".



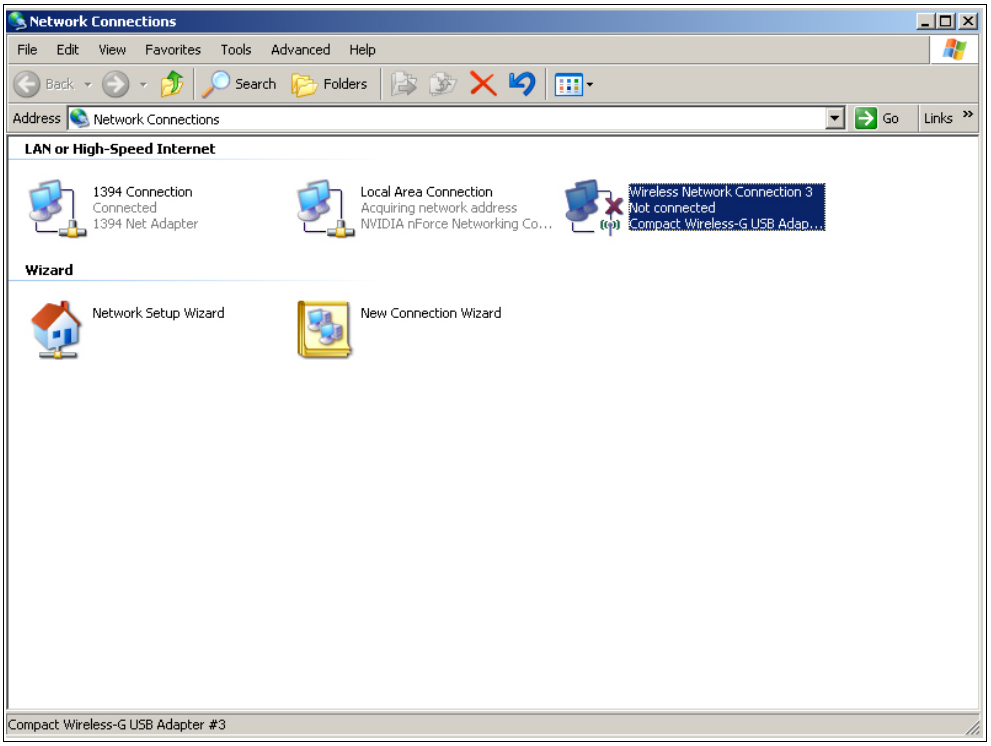
In the window on the right side of the "*Services*" box, scroll down the "*Wireless Zero Configuration*" and select it.



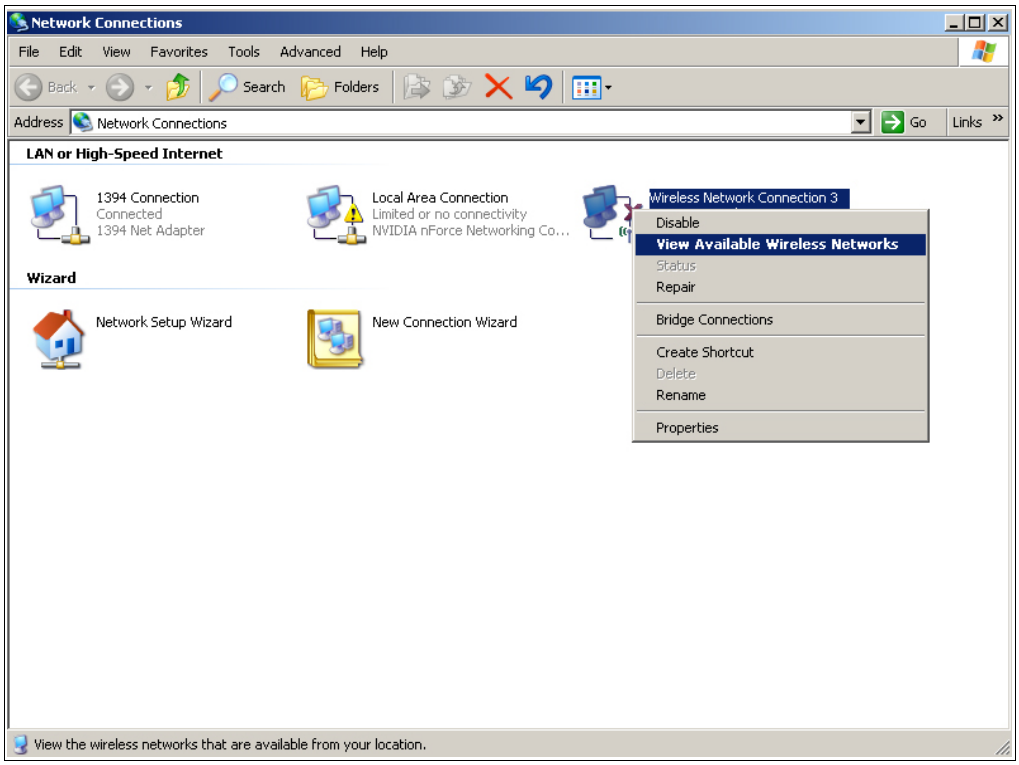
Press the [START] button on this options page. After the service starts, press [APPLY] then [OK].



Go back to the Network Connections window. You will see your Wireless Network Connection icon says "Not connected". It is now time to connect to your wireless network.



Right-click the wireless network connecton icon, and select "*View Available Wireless Network*" icon.

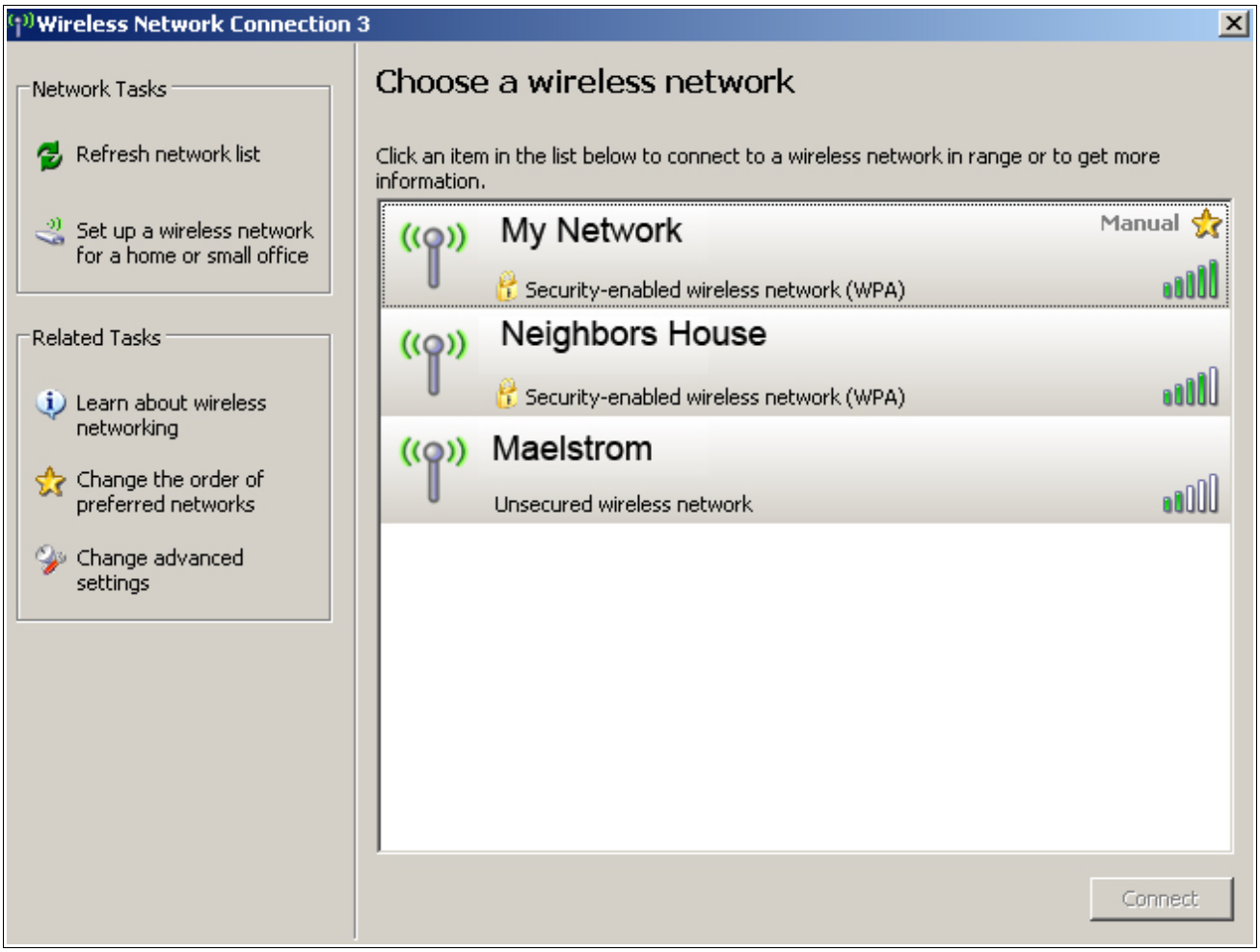


Connecting to your Network

When the window opens, you should see all networks that are in range of your wireless adapter.

When you set up the network in your modems options. You should see the name of the network you created at the top of this list. It will be at the top because it is the closest network.

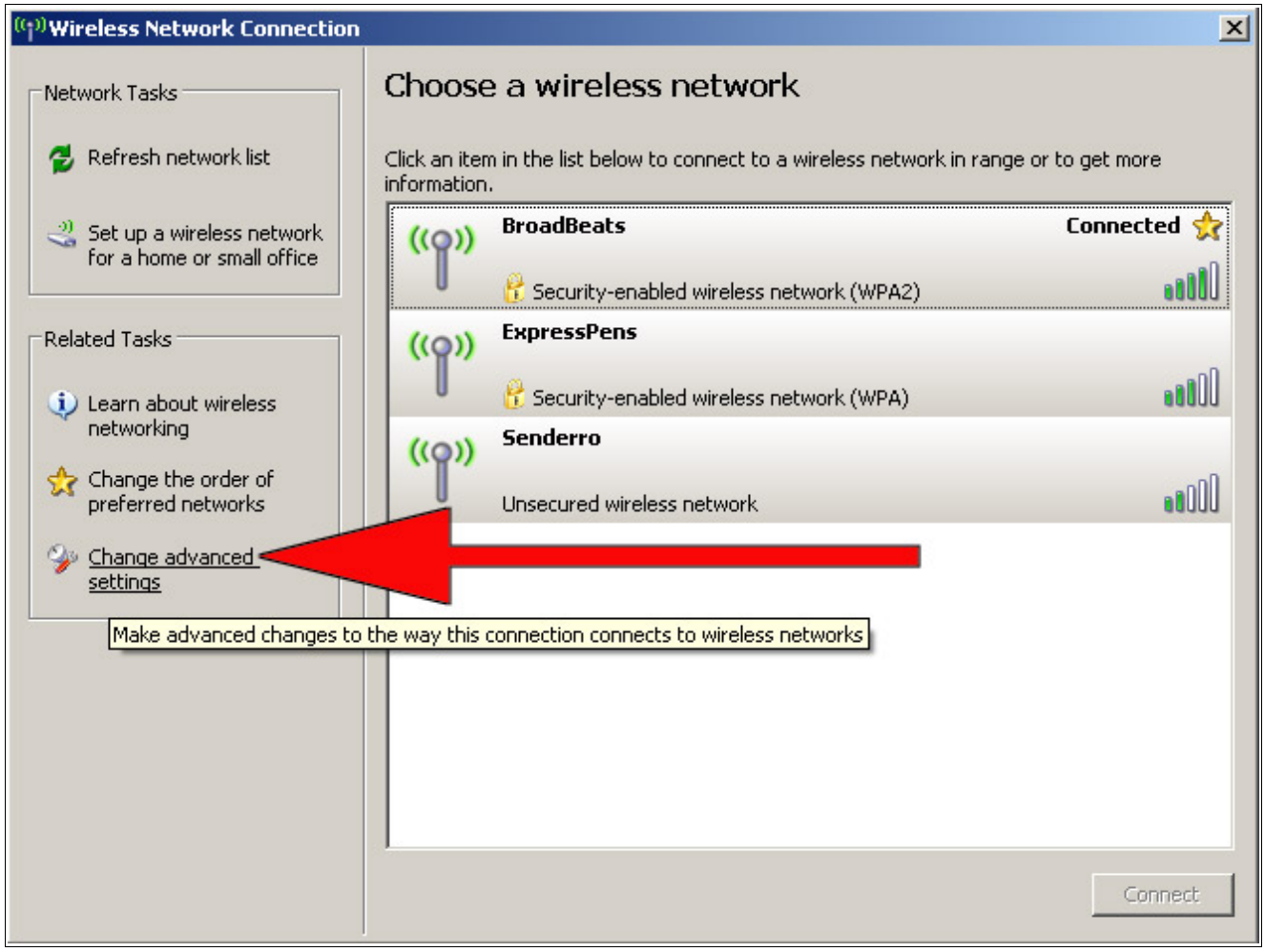
If you do not see any networks, press the "Refresh network list button" under Network Tasks.



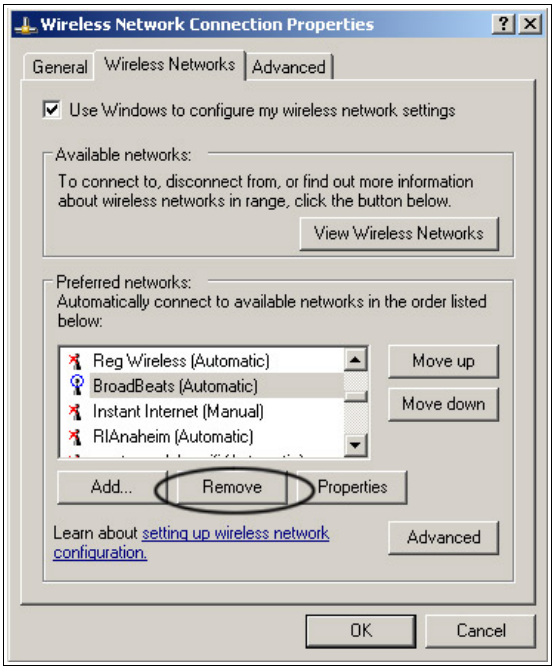
Select your network, then press the [Connect] button at the bottom right of the screen. A window should popup asking you for your network key. Type it in. This should connect you to your wireless network and allow your Open Labs system to access to the internet.

Troubleshooting a Wireless Connection

In the Wireless Network Connection window, select "Change advanced settings". Which allows you to make advanced changes to the way this connection connects to wireless networks.



Now, select your network from the **Preferred networks** window. Then press the [Remove] button. That network will be deleted.



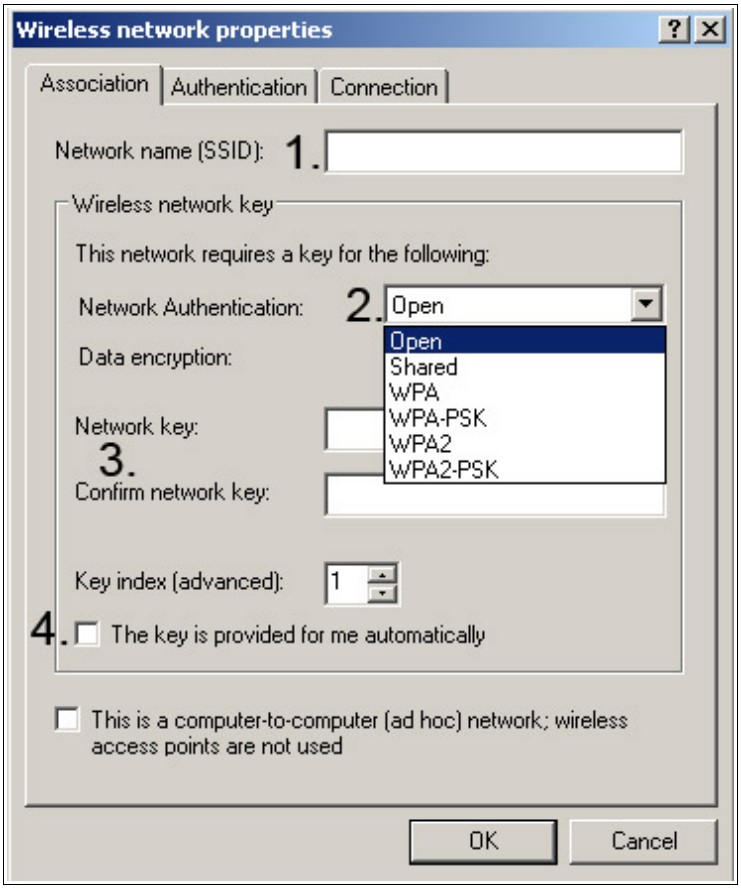
To manually add your wireless network to the preferred networks list, do the following:

1. Type in the exact name of the network
2. Select which type of authentication. Most basics networks use WPA, or WPA2.
3. Enter your network key. You will enter it once then enter it again to confirm it.
4. If the network is unsecured or a key is provided automatically you can place a check mark here.

Once you enter all your information you can have your Open Labs system automatically connect to this network when you are in range of its signal.

In this same window, click on the **Connection** tab and place a check mark beside "*Connect when this network is in range*".

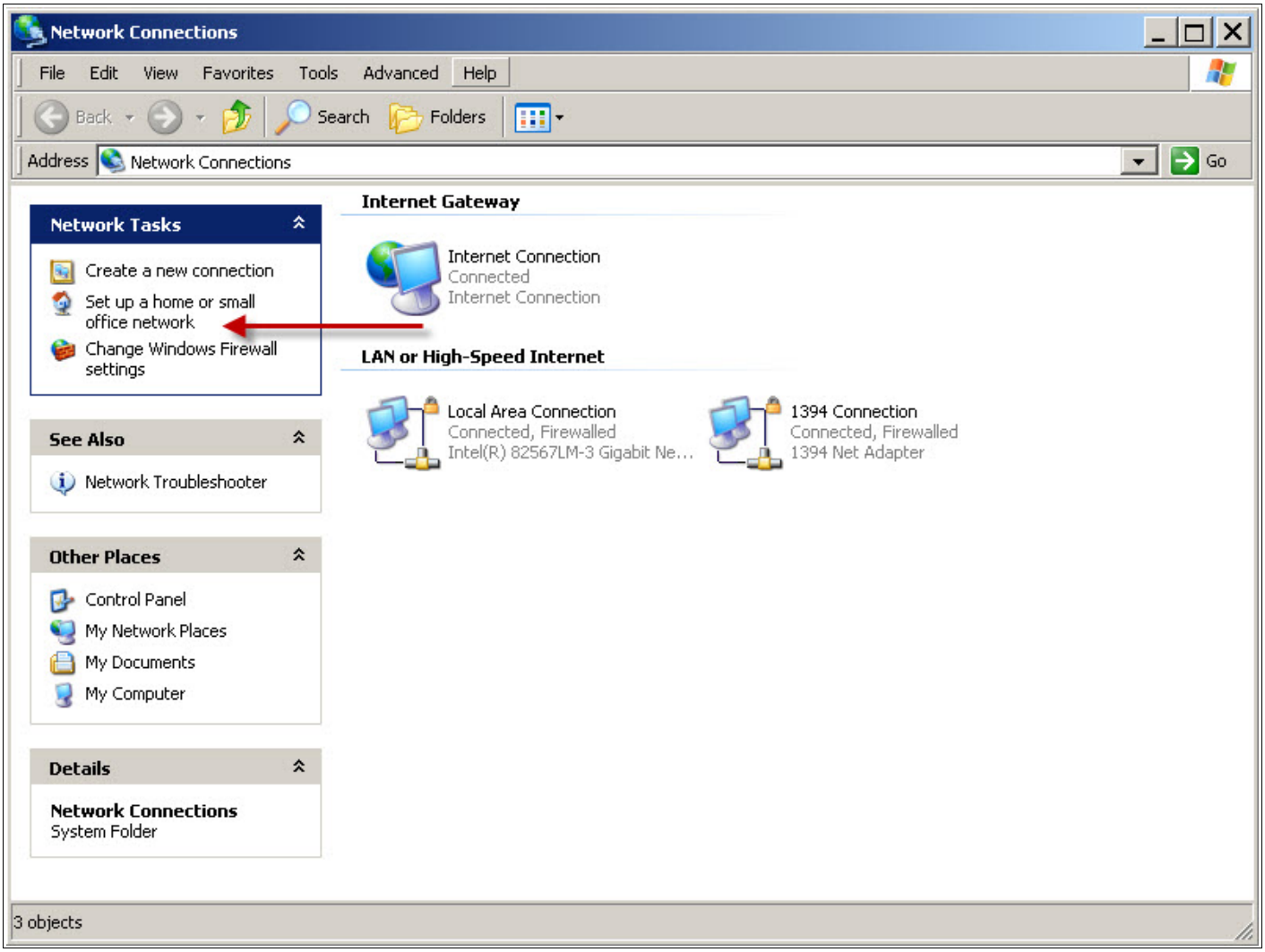
The network information will be stored on your system, usually you will not have to reconfigure this particular network.



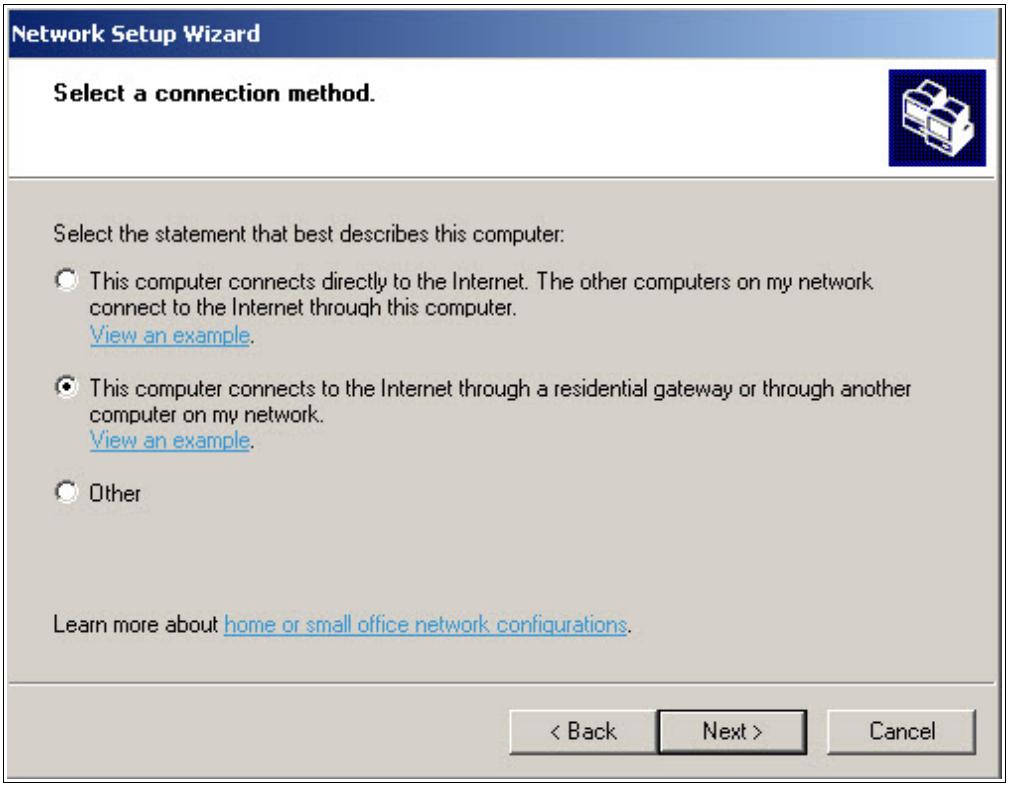
Setup a home or small office network

If you still are experiencing connection problems it could be due to your network not being properly configured. This section will show you how to run the "*Setup a home or small office network*" wizard. You can run this wizard to determine how your Open Labs system connects to the internet.

1. Press **[Windows key + E]** to Open **Windows Explorer**.
2. Right-click on **Network Places** and select **Properties.**
3. Under **Network Tasks**, select **Set up a home or small office network**.



You can just select "next" on the first couple of pages until you reach the "Select a connection method" page. This page allows you to pick how you connect to the net.



One of the first two options will be the connection type you will use.

1. This computer connects directly to the Internet. The other computers on my network connect to the Internet thought this computer.

   Use this option if there is no "master" computer or router present. This option also allows your internet connection to be shared by other computers. (Requires additional hardware for sharing).

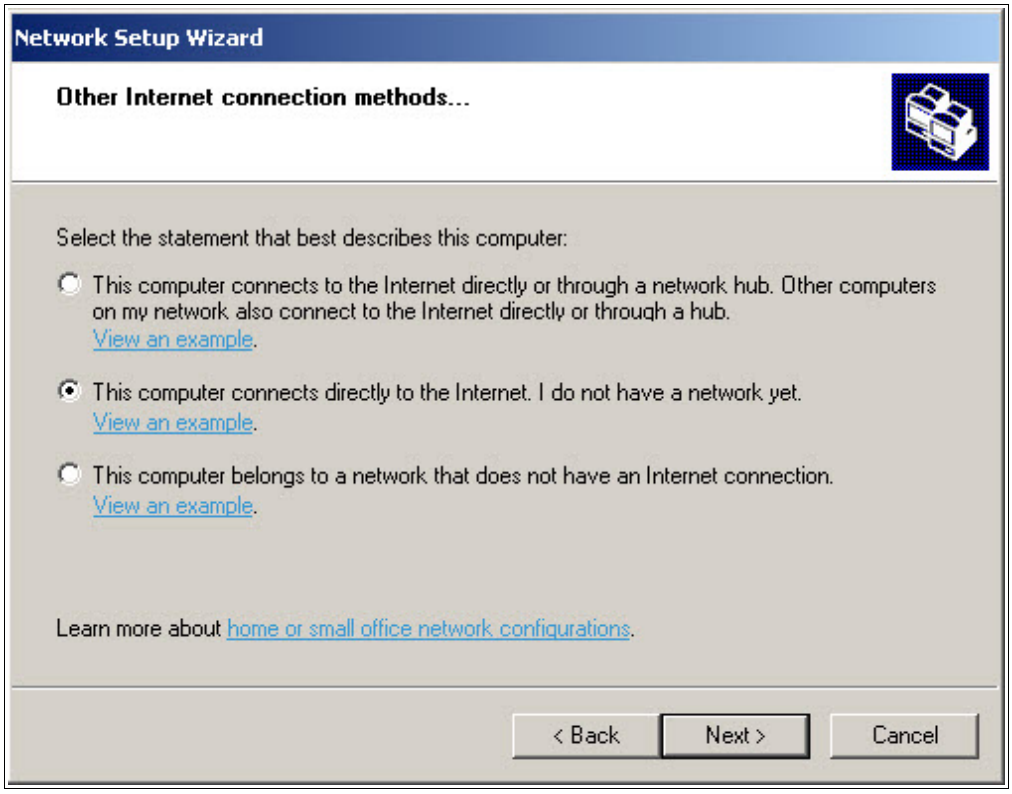
2. This computer connects to the Internet through a residential gateway or though another computer on my network. Sharing is not enabled with this option.

   Use this option if there is a "master" computer or router present. The master computer controls the internet gateway.

3. Other - Use Other if you have no network setup at all and want to connect directly to the internet.

If you have no network and just want to connect directly to the internet, choose "Other" now and press [Next].

Once you reach the "Other Internet connection methods...." window, select the second radio button; "This computer connects directly to the Internet. I do not have a network yet".



1. This computer connects to the Internet directly or though a network hub. Other computers on my network also connect to the Ineternet directly or through a hub.

   If you are using a router to connect multiple systems to the same internet connection, then choose this option.

2. This compter connects directly to the Internet. I do not have a network yet.

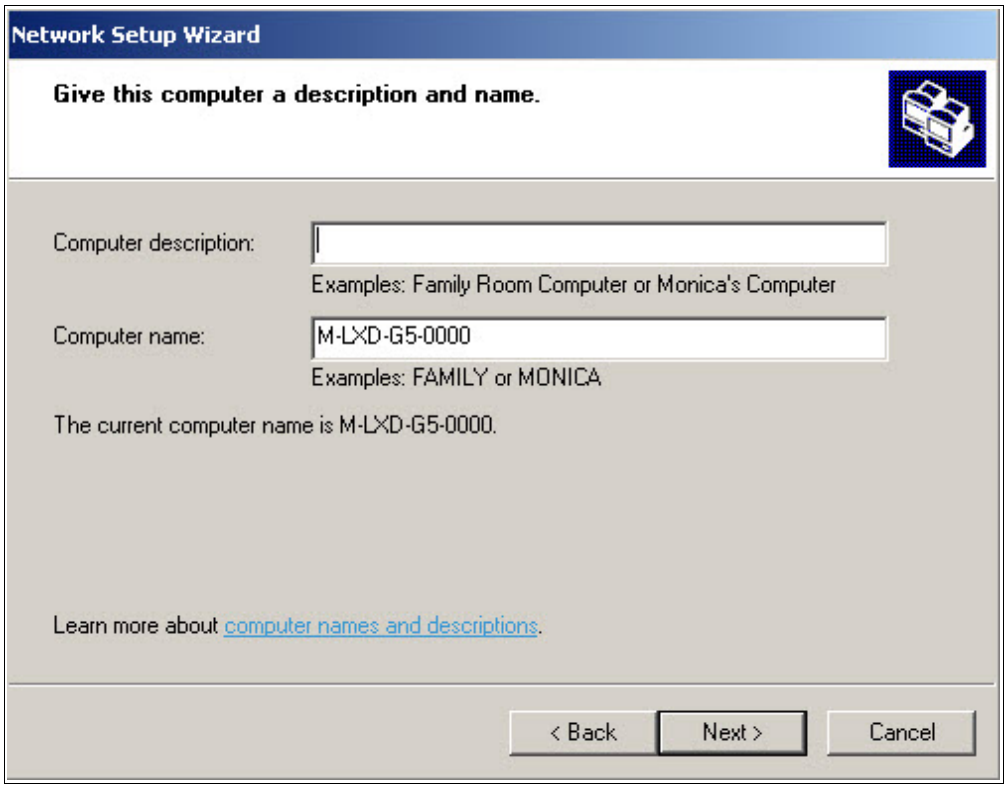
   Use this option if you have no network and want to connect one system to the internet.

3. This computer belongs to a network that does not have an internet connection.

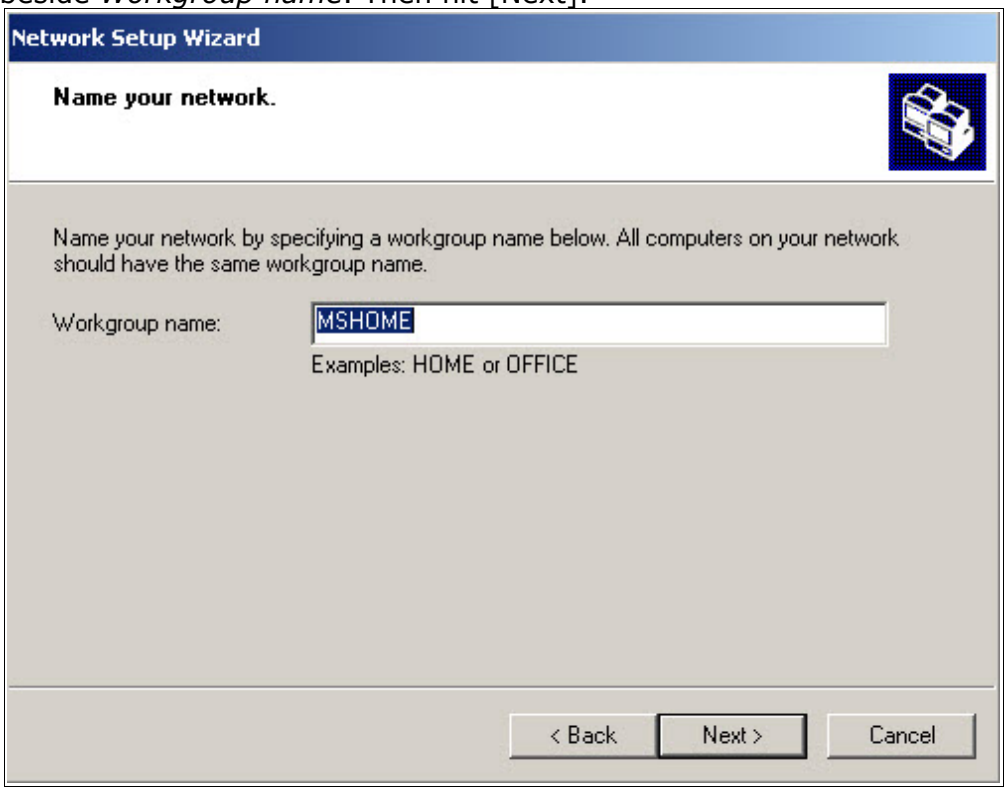
   This option speaks for itself.

Select and option, and press [Next].

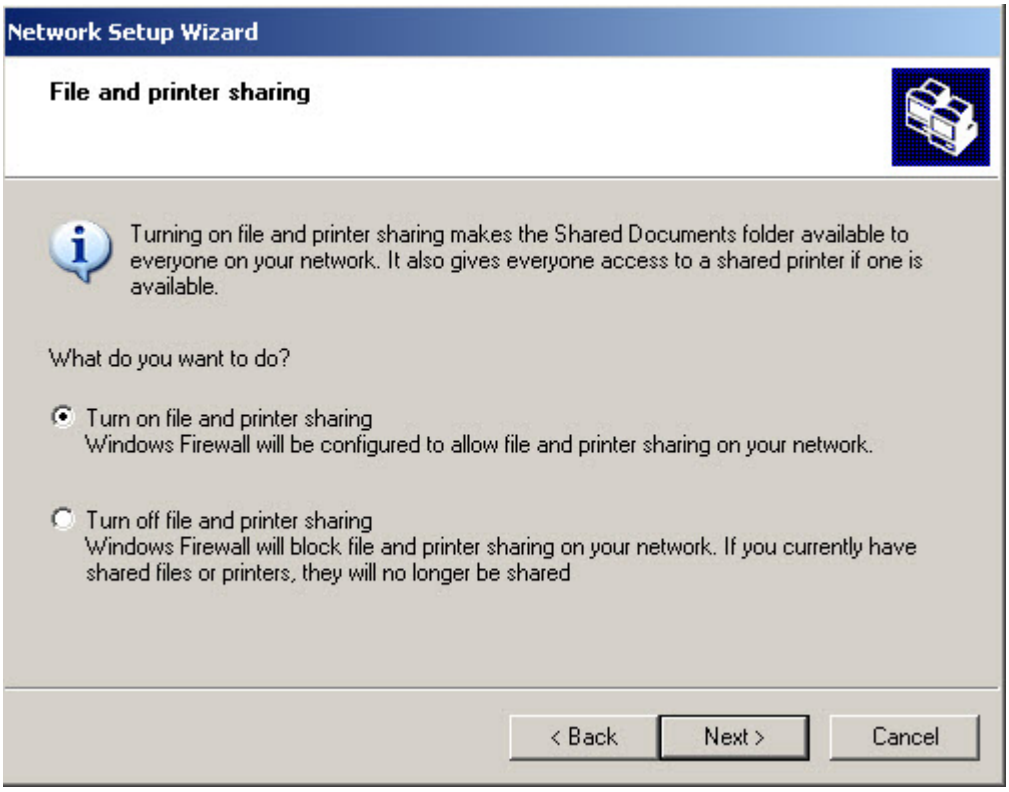
You can use this screen to give your Open Labs system a descriptive name. The "Computer Name" field should be saved for the default name of your Open Labs system, which will have the model number and the serial number. Hit [Next].



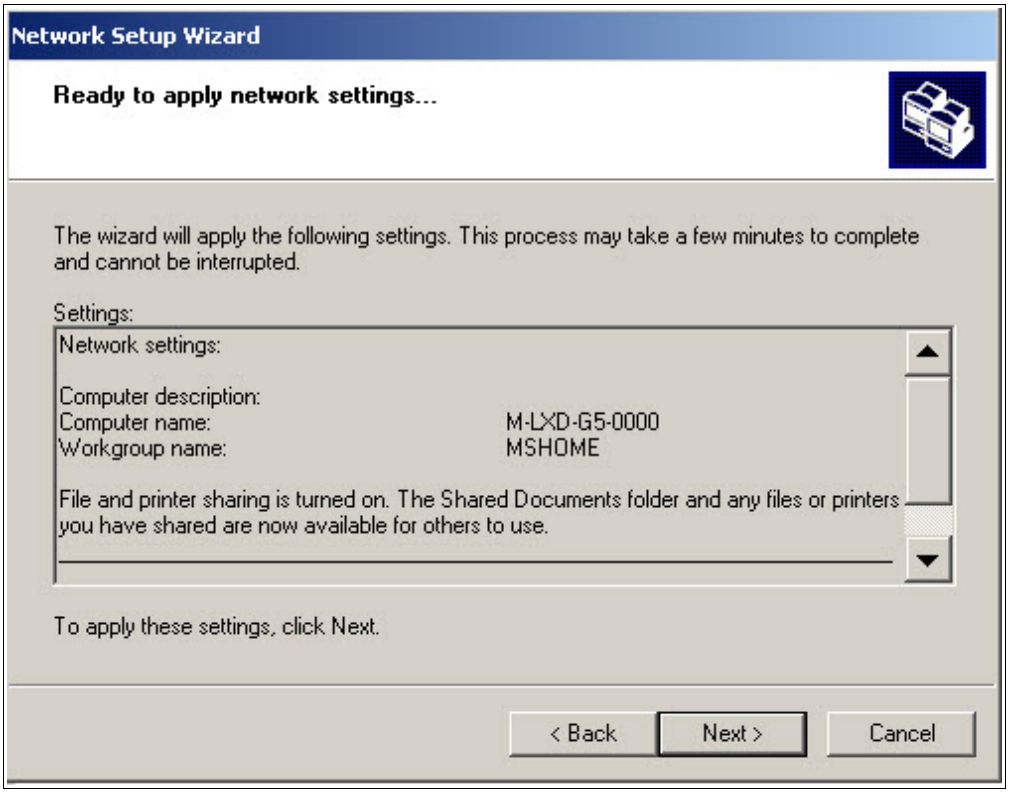
On the "Name your network" page, unless you are a part of a specific network, you can just type in "MSHOME" beside *Workgroup name*. Then hit [Next].



If you have multiple systems on your network, you can turn on "File and printer sharing". The following page allows that. Otherwise just turn off file and printer sharing.

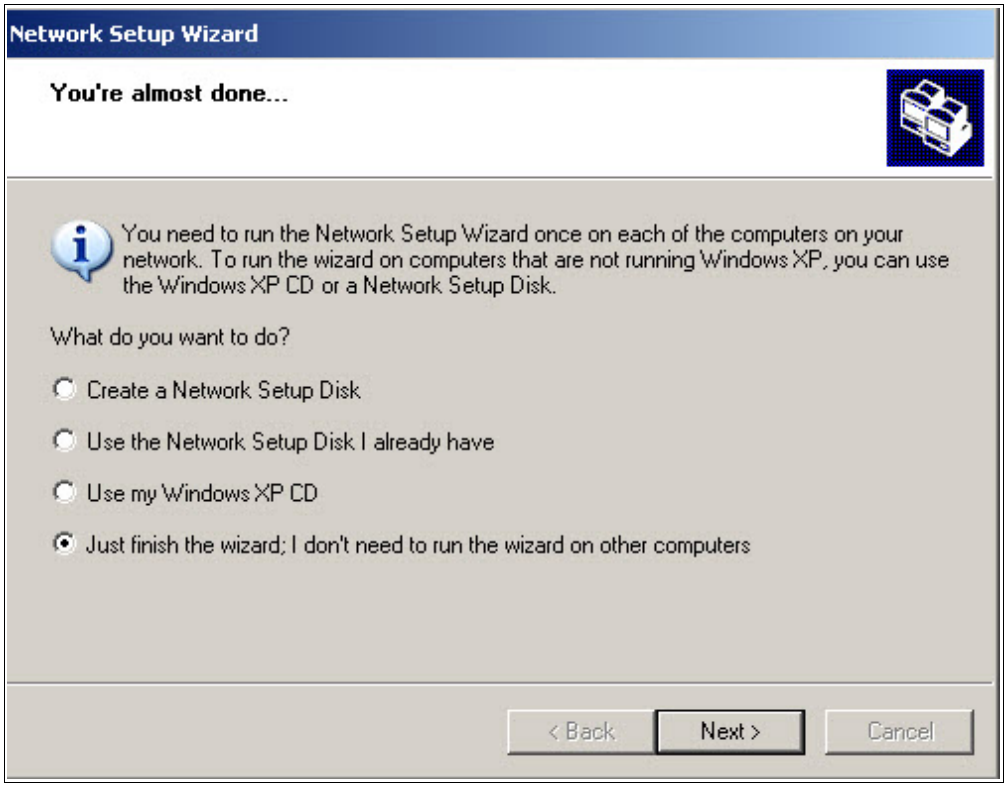


Press [Next]. The "Connection method" you chose will determine what you will see in the following window. It is an information window that describes all the options you chose.



Press [Next].

This is the last page, you can just select "Just finish the wizard; I don't need to run the wizard on other computers".



Hit [Next] and that is it.

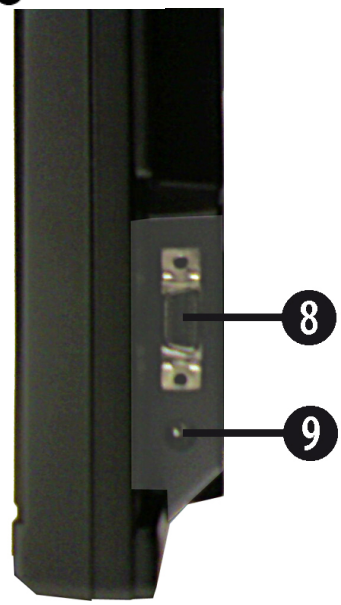
Appendix C – Optional Touchscreen

If you have purchased the optional touchscreen kit, this section will get you familiar with the screen.

Touchscreen Features



1. Used to move the cursor down in OSD (On Screen Display.
2. Used to move the cursor up in OSD mode.
3. **< -** Program selection key or osed to select function in OSD mode.
4. M/E key - Menu Key - Used to start OSD mode.
5. **>** - Program selection key or used to select function in OSD mode.
6. VGA, AV switch.
7. Power Switch
8. SKS InRemput
9. DC 12V Input



Cable Identification



DC12V Adapter



VGA & Touchscreen Cable - Connects the USB & VGA wires to the SoundSlate & and the aux cable.



The remote lets you control the screen parameters from your chair.

APPENDIX D – MimiK Users Guide



What is MimiK?

MimiK is a program designed to sample your MIDI-capable keyboards and hardware. It can be set up to automatically record all the sounds on your favorite hardware for use on your Open Labs production station. The basic algorithm behind MimiK is that it will send MIDI messages for every note on the target keyboard you want to sample.

At the same time it will record each note played back by the keyboard. It has the ability to capture different note lengths, velocities, note ranges, note off samples and much more. After it has recorded/sampled all the notes of a preset, it will store a special file that can be read by a SFZ sample player. Each sample is stored separately as a WAV or OGG file on the MimiK machine.

MimiK Suggested Usage

Due to the amount of time it takes to correctly mimic one sound well, we recommend sampling each preset one at a time. You can mimic batches of presets or entire banks, but the time and disk space required can be inhibiting.

Straightforward sounds like pianos, drums, and basses have a pretty simple architectural structure. This makes it very easy to mimic an entire bank at once. Performance banks and custom patches may have nuances that really need to be dialed in, therefore increasing the time and disc space required. There are a number of ways to reproduce a custom or performance patch.

For instance, some performance patches are made up of multiple sounds. Instead of MimiKing the performance patch, MimiKeach sound that is in that performance patch. Then blend them all back together in your host. This will allow you to really dial a sound in.

It is also a good thing to write down key settings of the original sound effects settings, ADSHR and envelope settings.

What is a Preset and a Bank?

In this documentation the terms "preset" and "bank" will be used frequently. A preset is one complete unique sound. A bank is a collection of presets.

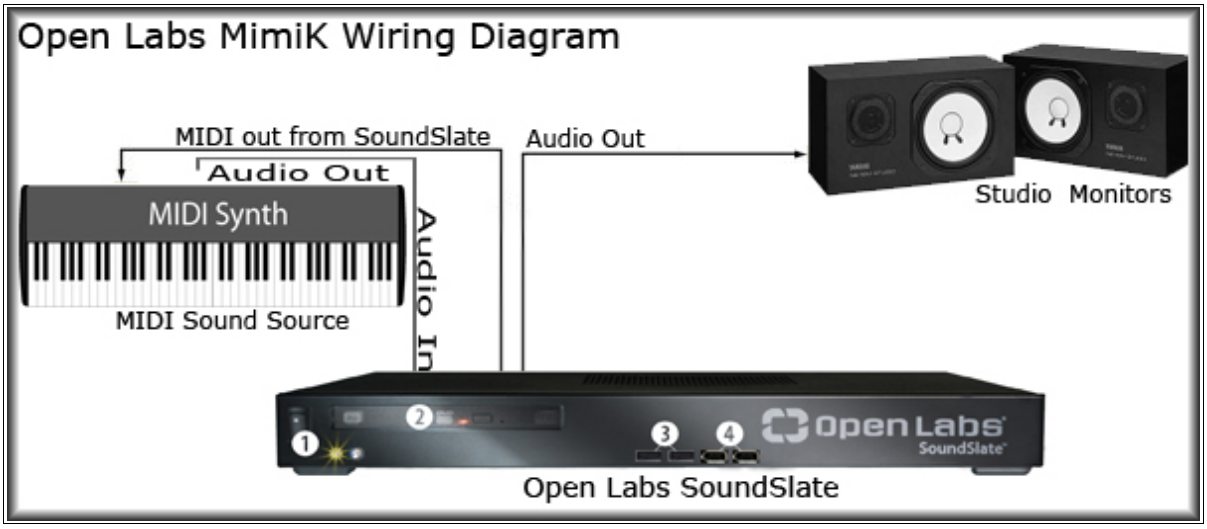
SFZ Format

MimiK uses the open SFZ format for storing sample libraries. The SFZ format is very simple. Every preset captured by MimiK will have a matching SFZ file (.sfz). This SFZ file is a human readable text file that can be opened and edited in Notepad or Microsoft Word. The SFZ file is a list of all the samples that are in the library. Each sample is either a WAV or OGG file. The SFZ file also contains information about each sample, such as what note the sample is and things of that nature. SFZ files can be opened up with the free SFZ sample player Dimension LE.

How to Physically Set Up MimiK

Wiring Diagram

The following image shows you a basic wiring diagram.



Turn on the keyboard you want to mimic. You will need a standard MIDI cable and one or two ¼-inch audio cables (One audio cable if you want to record in mono, two for stereo).

Now plug in one end of a standard MIDI cable from the "MIDI out" port of your Open Labs production station. The port is clearly marked on the back of the unit. Plug the other end of the MIDI cable into the "MIDI in" port of the keyboard or module you want to mimic.

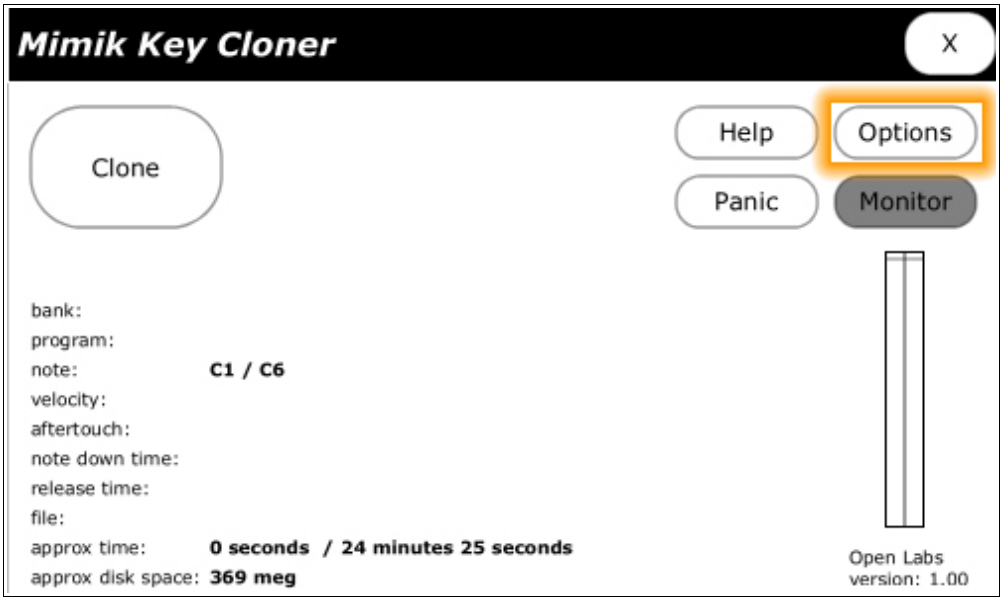
Next, obtain one or two ¼-inch audio patch cables. Plug one end into the audio output(s) of the keyboard you want to mimic. Then plug in the other end to audio input one (and two if you are recording a stereo keyboard) of the Open Labs production station.

Now we need to test if cables have been correctly hooked up.

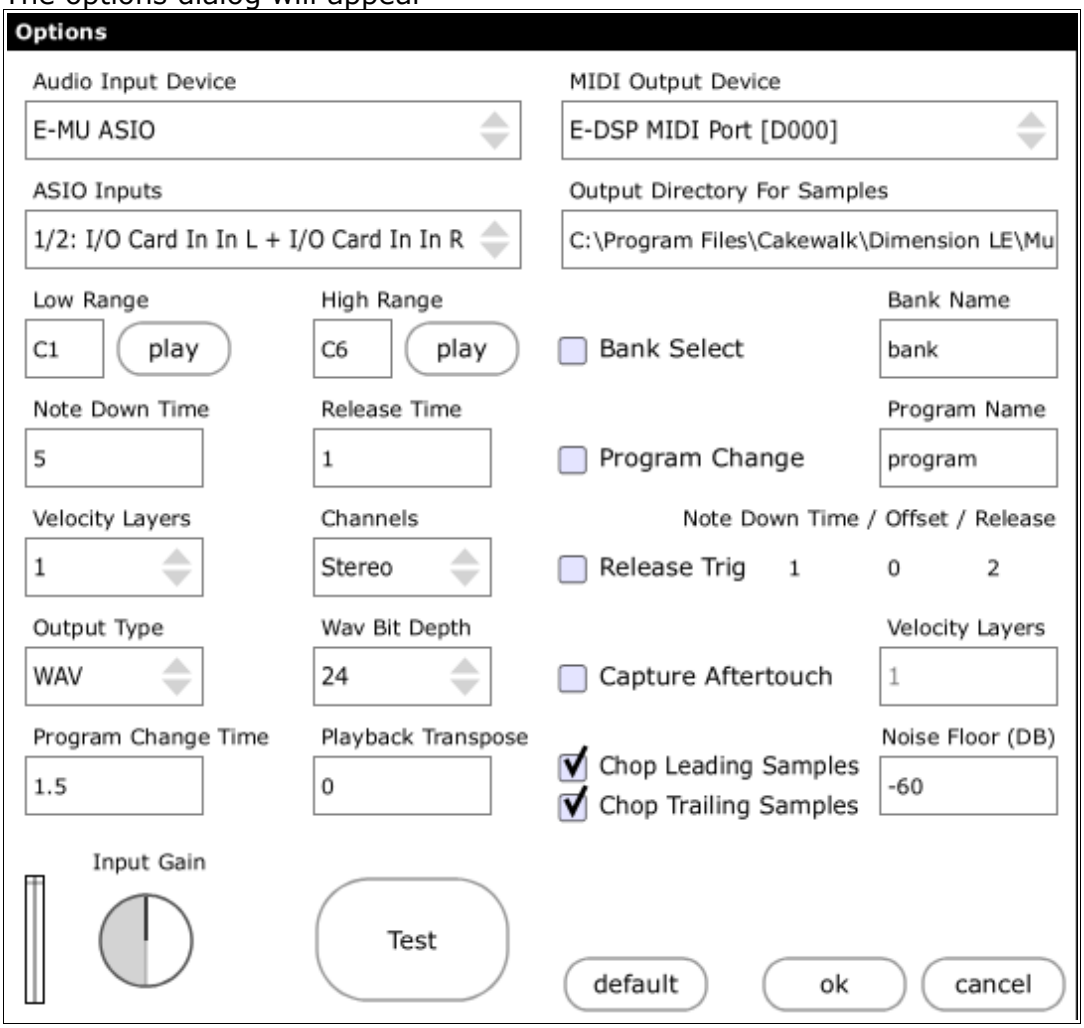
Make sure the Open Labs production station is hooked up to a set of speakers or that you have a pair of headphones plugged into the headphone jack. You will use these to monitor the sounds coming from the keyboard you are cloning.

Setting up MimiK

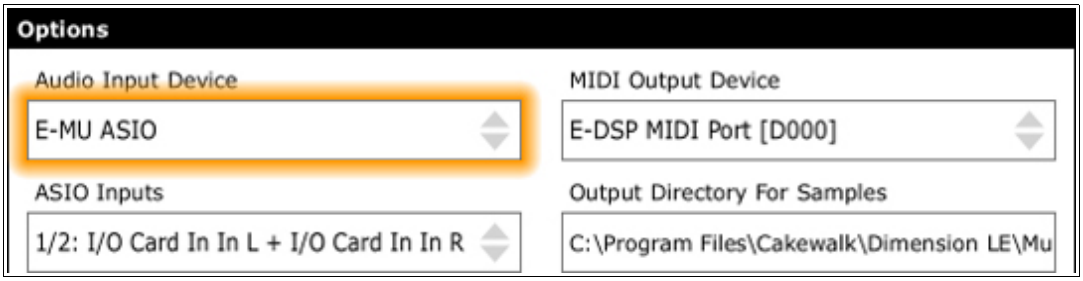
If the Open Labs production station is not turned on, do so now. Shut down all other programs that are running. Other programs may interfere with the recording process. Launch MimiK. The main window should appear. Click on the Options button.



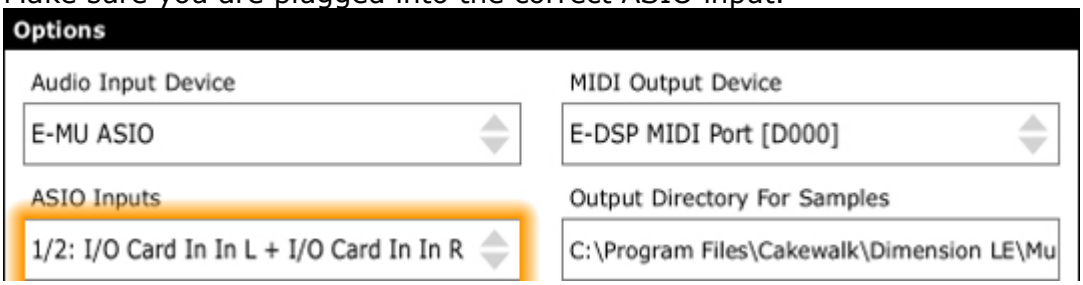
The options dialog will appear



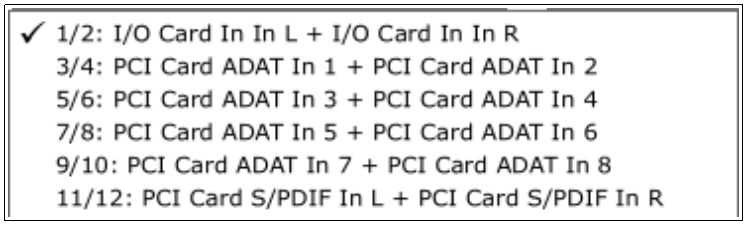
Make sure that the correct Audio Input Device is chosen.



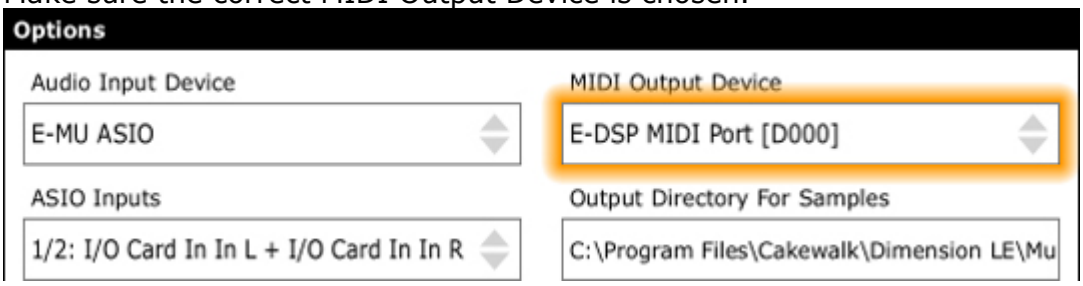
Make sure you are plugged into the correct ASIO input.



Here are the available ASIO inputs for the SoundSlate.

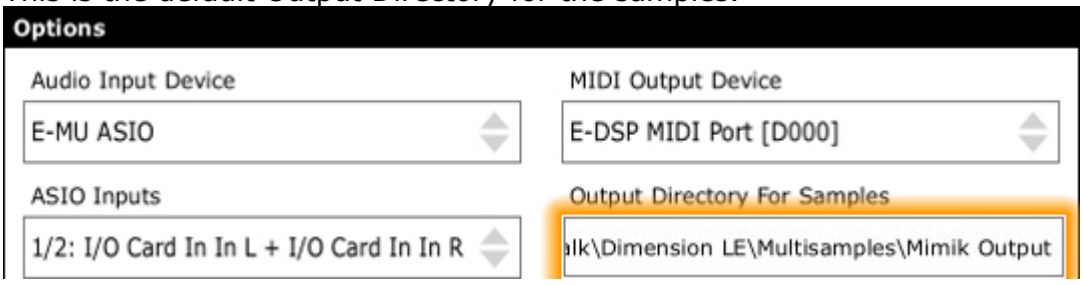


Make sure the correct MIDI Output Device is chosen.

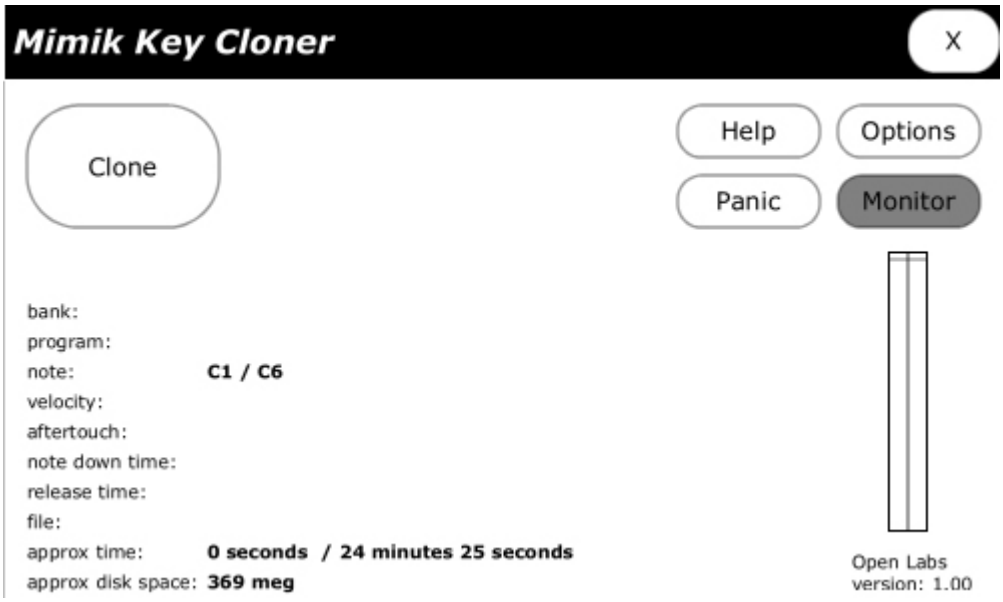


For most users it will be the PreSonus Firebox MIDI Out. This will be the default MIDI device driver for your installed MIDI output device, but it can be any MIDI output device you have installed.

This is the default Output Directory for the samples.



The default save path is "*C:\Program Files\Cakewalk\Dimension LE\Multisamples\MImik Output*".



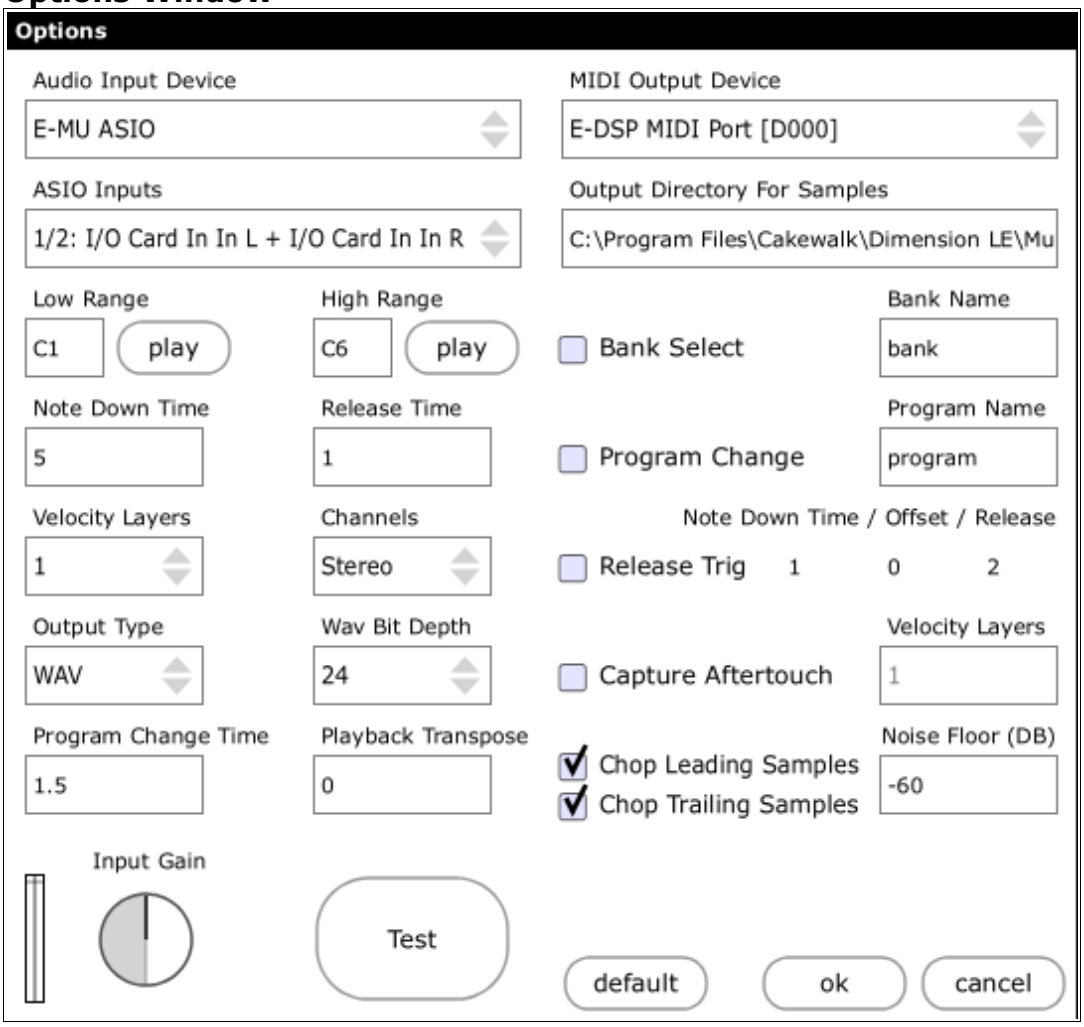
Options: Launches the Options dialog window.

Panic Button: Sometimes keyboards drop note off MIDI messages. You'll know when that happens because you'll get a stuck note. The Panic button sends a "note all off" message to the keyboard. This should get rid of any stuck notes.

Monitor Button: By default this is on. When highlighted, all audio input will be routed to the audio outputs of the Open Labs production station. This is useful when you want to hear the progress of the MimiK process.

MimiK Button: Starts the MimiK process. The MimiK process basically consists of sending MIDI notes for each key on the keyboard for a specified period of time and recording the output. This process can be paused or stopped at any time.

Options Window



Defaults Button: Resets all the options to the original defaults.

Program Change Time: Time (in seconds) that a program change will take. If the current program is being sampled then this is irrelevant. Some keyboards have notoriously long program/preset change times.

ASIO Inputs Device: Exposes which ASIO audio input device is used.

ASIO Inputs: Exposes which inputs on the ASIO audio input device are used to record.

MIDI Output Device: Exposes which MIDI output device is used.

Output Directory: Defines the output directory that all the samples are stored in. To pick another output directory, click on this text box to bring up a directory picker dialog.

Low Range and High Range: Defines what MIDI messages will be sent from MimiK to the keyboard. These notes are inclusive. You may either drag up or down to raise or lower each range. Or you may type in a valid note. Click on the play button beside each range to send a test MIDI message to the keyboard.

Note Down Time: Defines the time in seconds that each velocity layer of each note is sampled. The longer the note down time the better. The trade off is in how much disk space is used and how long it takes to MimiK a preset. You will need to experiment with each preset to see what works best.

Release Time: Defines the time that MimiK will continue recording output from the keyboard after a Note Off MIDI message is sent. This is useful for capturing the decay/release of a sound after a note has been released. Some sounds have long decays (sounds with lots of reverb) so you may want to adjust depending upon the preset.

Velocity Layers: Choose how many velocity layers to MimiK. If "1" velocity layer is chosen, then each note will be sampled once as if it was hit at full strength (a velocity of 127). If "2" velocity layers are chosen then two velocities are recorded (63 and 127), and so on. Keep in mind that most keyboards don't actually playback that many different sounds per velocity. Volume attenuation is used, but the actual sound played back is usually the same. We recommend using 1, 2, or 4 velocity layers for most keyboards.

Channels: You can decide whether you want to capture a mono or stereo sound. If you choose to capture a mono sound, the left input will be used. This can cut the amount of disk space in half. This option makes the most sense when cloning keyboards only capable of creating mono output.

Output Type: You can decide whether you want to capture a mono or stereo sound. If you choose to capture a mono sound, the left input will be used. This can cut the amount of disk space in half. This option makes the most sense when Cloning keyboards only capable of creating mono output.

WAV Bit Depth: Defines how many bits are used for each sample (CDs are 16-bit). The higher the better. The higher the bitrate, the more disk space used.

Quality: When the OGG/Vorbis format is used, the quality of the output files can be edited. The higher the quality, the better sounding the sample will sound. Keep in mind that higher quality settings will also create larger files.

Quality 0 is roughly equivalent to 64kbps on average, 5 is roughly 160kbps, and 10 gives about 400kbps. Most people seeking very-near CD-quality audio encode at a quality of 5 or, for lossless stereo coupling, 6. The default setting is quality 3, which at approximately 110kbps gives a smaller file size and significantly better fidelity than .mp3 compression at 128kbps.

The default setting of 3 seems to work for many sounds.

As always, if you need CD-quality sound use the WAV output type.

Test Button: The Test button is extremely useful. It will behave exactly like the "MimiK" button on the main window, except that nothing is recorded to disk. Therefore you can hear a preview of what MimiK will actually do once you hit the "MimiK" button. The test can be canceled at any time.
Playback Transpose: This value is In octaves. It specifies the amount of transposing that happens when the samples are played back. Really this just shifts what keys will trigger the samples. It will have no effect on the actual sound. Take for example, you sample a few of the lower octaves from a keyboard (C0 - C2). For some reason you want to play them back with your right hand. This option will let you shift the keys that will play back the samples, up a few octaves if necessary (maybe C5-C7). Experiment to see.

VU Meter: Audio input VU meter. It mirrors the VU on the main window.
Input Gain: Some keyboards can be a bit soft or loud. Adjusting this will allow for keyboards with different levels of output.

Bank Select: When this is NOT selected, you can type in the name of the Bank you will be recording It will be left up to you to manually (on the keyboard) pick the bank you want to sample from.

If Bank Select is selected, you will be allowed to select an inclusive range of banks that MimiK will attempt to capture. Warning: many keyboards do not support the General MIDI Bank Select command, so running this may or not work. It is recommended that you test if your Keyboard even has more than one bank. If it does, then check if it even supports Bank Select via MIDI. Due to the amount of time it takes to sample one preset we recommend not using the Bank Select option even if your keyboard supports it. Sampling one bank at a time is what we recommend over attempting to sample multiple banks. Regardless the option is there.

Program Change: When this is NOT selected you can type in the name of the preset you will be recording. It will be left up to you to manually (on the keyboard) pick the preset you want to sample from. If Program Change is selected then you will be allowed to select an inclusive range of programs that MimiK will attempt to capture.

Release Trig: ***This setting will record a separate sample!!!***

Short for Release Trigger.

Some keyboards trigger sound/samples upon the MIDI note off message. Enabling this option will enable MimiK to capture just the sounds made after the MIDI note off is sent. Sounds captured this way can then be played back whenever a note off is sent for a particular sample.

The Note Down Time parameter is the time that a note will be played. During this time period no sound is actually recorded! The recording will begin once the note is let up (MIDI note off message is sent).

The Offset parameter is any additional time to NOT record after the MIDI note off message is sent. This is useful for tweaking and isolating a specific part of the sound.

The Release parameter is how long to record after the note off is sent plus the Offset time. This is the only time actual recording takes place within the Release Trig event.

Capture Aftertouch: If your keyboard supports it, aftertouch can be captured separately from regular note down events. To keep things simple it sends an aftertouch message immediately after the note down is sent. All the other parameters (note down time, release time) apply. Different aftertouch layers can be captured. If "1" layer is selected, then an aftertouch value of 127 is sent and recorded. If "2" layers are used, then an aftertouch value of 63 and 127 are sent and recorded, and so on.

Chop Leading Samples: Anything below the noise floor is chopped off at the beginning of the sample. There is a slight delay in the time that the MIDI information is sent by the Open Labs production station and when the keyboard output sound is recorded by MimiK. This delay can be noticeable, so this option is useful in removing it.

Chop Trailing Samples: Anything below the noise floor is chopped off from the end of the sample. This can be a disk space saver for sounds that end before the note down and release time expire.

Noise Floor: Floating point noise floor for Chop Leading Samples and Chop Trailing Samples. Anything below this noise floor will be cut out of the samples.

Click on the Play button under the Low Range label. This will cause MimiK to send a MIDI note to the keyboard. This should in turn cause the keyboard play a note back to the Open Labs production station. If you do NOT hear a note played back from the Open Labs production station or see the VU meter light up in the options screen then:

1. Move the Low Range note around (up or down). Your keyboard may only play back from a small note range. Dragging up or down on the control will move the note range up and down. Do the same for the High Range.
2. Recheck all your cables. Check for possible bad cables.
3. Make sure the keyboard you are trying to mimic has its MIDI input enabled (read the keyboard manual). If you plan on MimiKing multiple sounds from multple banks, make sure your MIDI keyboard or module has MIDI bank select enabled, and MIDI program change selected.

If you hear a note and see the VU meter lighting up when you hit the play button, then you are ready to pick what you want to sample from your keyboard.

How to Playback a Sound after using MimiK

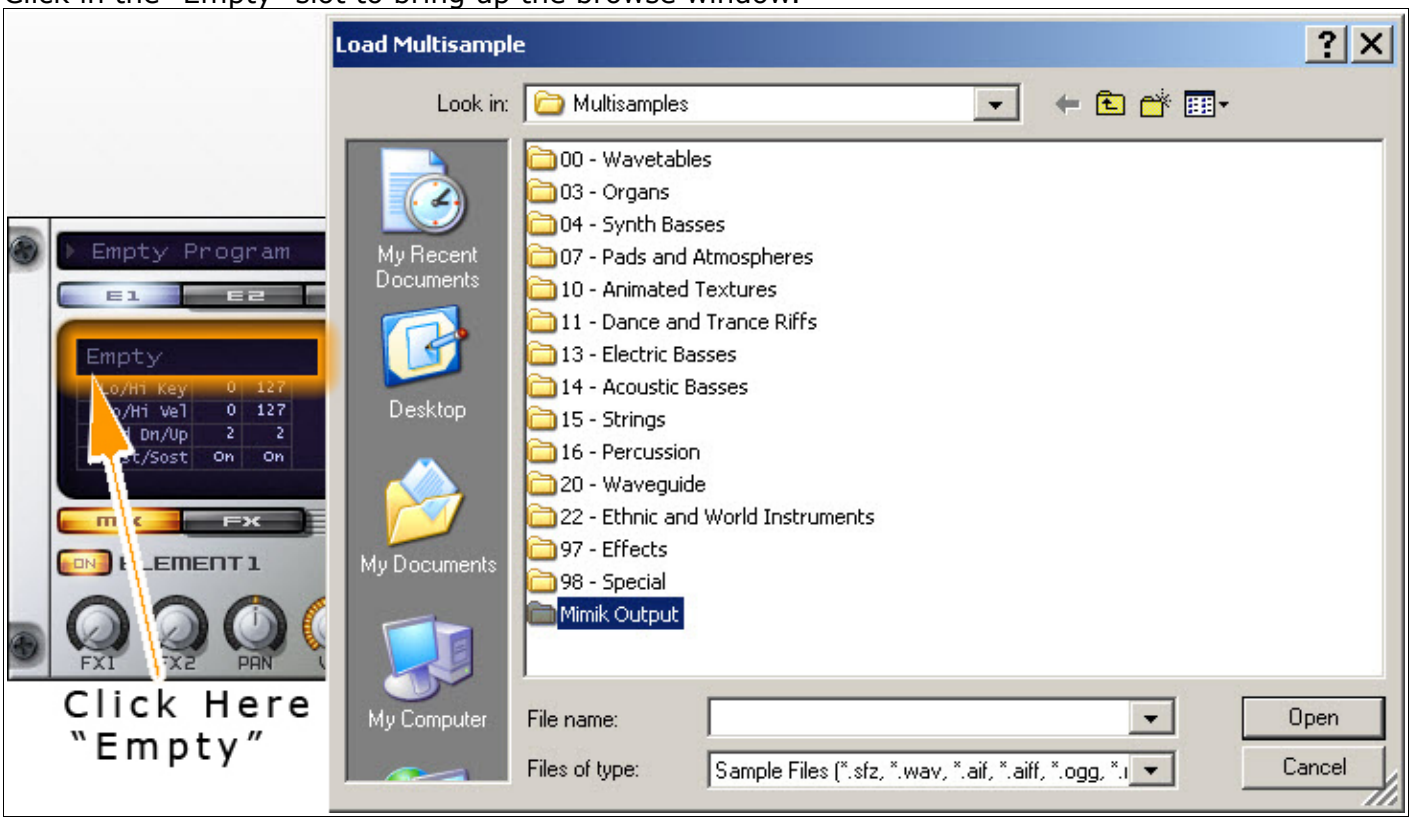
To play back sounds that have been made in Mimik, Open Labs has provided you with the sample player Dimension LE.

Load Dimension LE into either Riff or Reaper.



Once you load up an instance of Dimension LE, you will need to browse to the folder where the default Mimik folder is located. If you setup Mimik to output to another location just select that folder instead.

Click in the "Empty" slot to bring up the browse window.



Select the folder "Mimik Output" and explore the folder until you find the "*.sfz*" file. Select the "*.sfz*" file and press the [Open] button.

The test file made for this lesson was called "Grand Piano". You can see "*Grand Piano.sfz*" loaded into Dimension LE.



This sound is ready to be played.

To learn more about Dimension LE, please read the Dimension users guide located on the SoundSlate.