

# SSL 2+ MKII User Guide

SSL [support.solidstatellogic.com/hc/en-gb/articles/20859917333789-SSL-2-MKII-User-Guide](https://support.solidstatellogic.com/hc/en-gb/articles/20859917333789-SSL-2-MKII-User-Guide)



## Introduction to SSL 2+ MKII

Congratulations on purchasing your SSL 2+ MKII USB audio interface. A whole world of recording, writing and production awaits you! We know you're probably keen to get up and running, so this User Guide is set out to be as informative and useful as possible. It should provide you with a solid reference for how to get the best out of your SSL 2+ MKII. If you get stuck, don't worry; our website's support section is full of useful resources to get you going again.

## What is SSL 2+ MKII?

SSL 2+ MKII is a USB-powered audio interface that enables you to get studio-quality audio into and out of your computer with minimal fuss and maximum creativity. On Mac, it's class-compliant - this means that you don't need to install any software audio drivers. On Windows, you'll need to install our SSL USB Audio ASIO/WDM driver, which you'll find on our [website](#) - see the Quick-Start section of this guide for more information on getting up and running. Once you've done this, you'll be ready to start connecting your microphones and musical instruments. The signals from these inputs will be sent into your favourite music creation software / DAW (Digital Audio Workstation). The outputs from the tracks in your DAW session (or indeed your favourite media player) can be sent out of the monitor and headphone outputs, so you can hear your creations in all their glory, with stunning clarity.

## Features

- **2 x SSL-designed microphone preamps** with unrivalled EIN performance and huge gain range for a USB-powered device.

- Switchable **Mic/Line**, **+48V** phantom power & **high-pass filter** per input
- **LINE** input bypasses pre-amp stage - ideal for connecting the output of an external preamp
- Auto-detect **Instrument (DI)** input per input
- **Per channel Legacy 4K switches** - analogue colour enhancement for any input source, inspired by the 4000-series console
- **2 x professional-grade, independent headphone outputs** with separate volume controls & plenty of power
- **32-bit / 192 kHz AD/DA Converters** - capture and hear all the detail of your creations
- **Easy-to-use Monitor Mix Control** for critical low-latency monitoring tasks
- **4 x balanced outputs**, with stunning dynamic range. The outputs are DC-coupled, making them suitable for controlling CV input instruments & FX
- Stereo **Loopback** virtual input for podcasting, content creation and streaming
- **SSL Production Pack Software Bundle**: including SSL Native Vocalstrip 2 and Drumstrip DAW plug-ins, plus much more!
- **USB 2.0, bus-powered audio interface** for Mac/PC - no power supply required
- **MIDI 5-Pin DIN Inputs & Outputs**
- **K-Lock** Slot for securing your SSL 2+

## SSL 2 MK II vs SSL 2+ MK II

---

Which one is right for you, the SSL 2 MKII or the SSL 2+ MKII? The table below will help you to compare and contrast the differences between SSL 2 MKII and SSL 2+ MKII. Both have 2 input channels for recording and balanced monitor outputs for connecting to your speakers. The SSL 2+ MKII gives you 'that little bit more', with 2 additional balanced outputs (outputs 3&4) and 2 x independent high-powered outputs, with their own volume controls. SSL 2+ also features traditional MIDI input and MIDI outputs, for connecting to drum modules or keyboards.

FEATURE	SSL 2 MKII	SSL 2+ MKII
Best Suited For	Individuals	Collaborators
Mic/Line/Instrument Inputs	2	2
Legacy 4K Switches	Yes	Yes
Input High Pass Filters	Yes	Yes
Balanced L & R Monitor Outputs	Yes	Yes
Additional Balanced Outputs	-	Yes x 2 (4 Total)
Headphone Outputs	2 (same mix & levels)	2 (independent mixes & levels)

Low Latency Monitor Mix Control	Yes	Yes
MIDI I/O	-	Yes
Stereo Loopback	Yes	Yes
SSL Production Pack Software	Yes	Yes
DC-Coupled Outputs	Yes	Yes
USB Bus-Powered	Yes	Yes

## Get-Started

---

### Unpacking

---

The unit has been carefully packed and inside the box, you will find the following items:

- SSL 2+ MKII
- Safety Guide
- 1.5m 'C' to 'C' USB Cable
- 'C' to 'A' USB adapter

### USB Cables & Power

---

Please use the provided USB 'C' to 'C' Cable to connect the SSL 2+ MKII to your computer. The connector on the rear of SSL 2 MKII is a 'C' type. The type of USB port you have available on your computer will determine if you should use the included 'C' to 'A' adapter. Newer computers may have 'C' ports, whereas older computers may have 'A' ports. As this is a USB 2.0-compliant device, it will make no difference to the performance if you require the additional adapter to connect to your system. SSL 2+ MKII is powered entirely from the computer's USB-bus power and therefore requires no external power supply. When the unit is receiving power correctly, the green USB LED will light a steady green colour. For best stability and performance, we recommend using one of the included USB cable. Long USB cables (especially 3m and above) should be avoided as they tend to suffer from inconsistent performance and are unable to provide steady and reliable power to the unit.

### USB Hubs

---

Wherever possible, it is best to connect SSL 2+ MKII directly to a spare USB port on your computer. This will give you the stability of an uninterrupted supply of USB power. However, if you do need to connect via a USB 2.0 compliant hub, then it is recommended that you choose one of high enough quality to provide reliable performance - not all USB hubs were created equally. With SSL 2+ MKII , we've pushed the limits of audio

performance on a USB bus-powered interface and as such, some low-cost self-powered hubs might not always be up to the task. Usefully, you can check out our FAQs at [solidstatellogic.com/support](http://solidstatellogic.com/support) to see which hubs we've successfully used and found to be reliable with SSL 2+ MKII.

## System Requirements

---

Mac and Windows operating systems and hardware are constantly changing. Please search for 'SSL 2+ MKII Compatibility' in our online FAQs to see if your system is currently supported.

## Registering Your SSL 2+ MKII

---

Registering your SSL USB audio interface will grant you access to an array of exclusive software from us and other industry-leading software companies - we call this incredible bundle the '**SSL Production Pack**'



To register your product, head to [www.solidstatellogic.com/get-started](http://www.solidstatellogic.com/get-started) and follow the on-screen instructions. During the registration process, you'll need to input the serial number of your unit. This can be found on the label on the base of your unit.

**XX-XXXXXX-XXXXXXXXXXXXX**  
**SERIAL NUMBER**

*Please note: the actual serial number begins with the letters 'SP2'*

Once you have completed registration, all of your software content will be available in your logged-in user area. You can return to this area at any time by logging back into your SSL account at [www.solidstatellogic.com/login](http://www.solidstatellogic.com/login) should you wish to download the software another time.

## What is the SSL Production Pack?

---

The [SSL Production Pack](#) is an exclusive software bundle from SSL and other third-party companies. To find out more please visit the SSL 2+ MKII product pages on the website.

## Quick-Start / Installation

---

1. Connect your SSL USB audio interface to your computer using the included USB cable, with or without the included adapter.

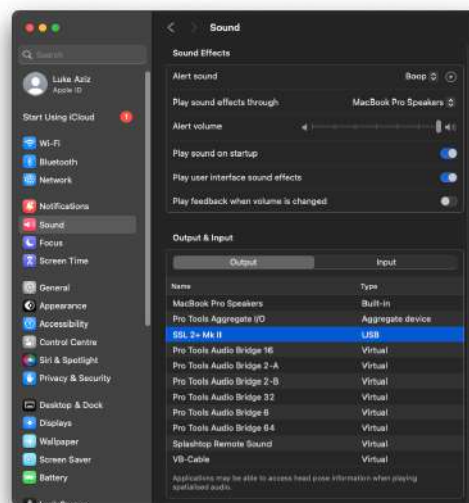


## Apple Mac Installation

---



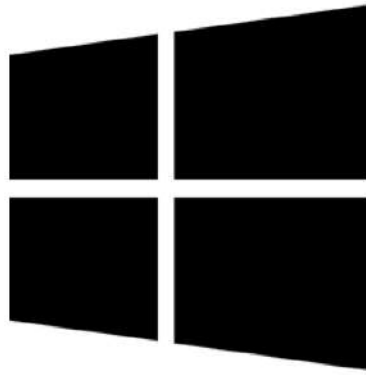
2. Go to 'System Preferences' then 'Sound' and select 'SSL 2+ MKII' as the input and output device (drivers are not required for operation on Mac)



3. Open up your favourite media player to begin listening to music or open up your DAW to begin creating music.

## Windows Installation

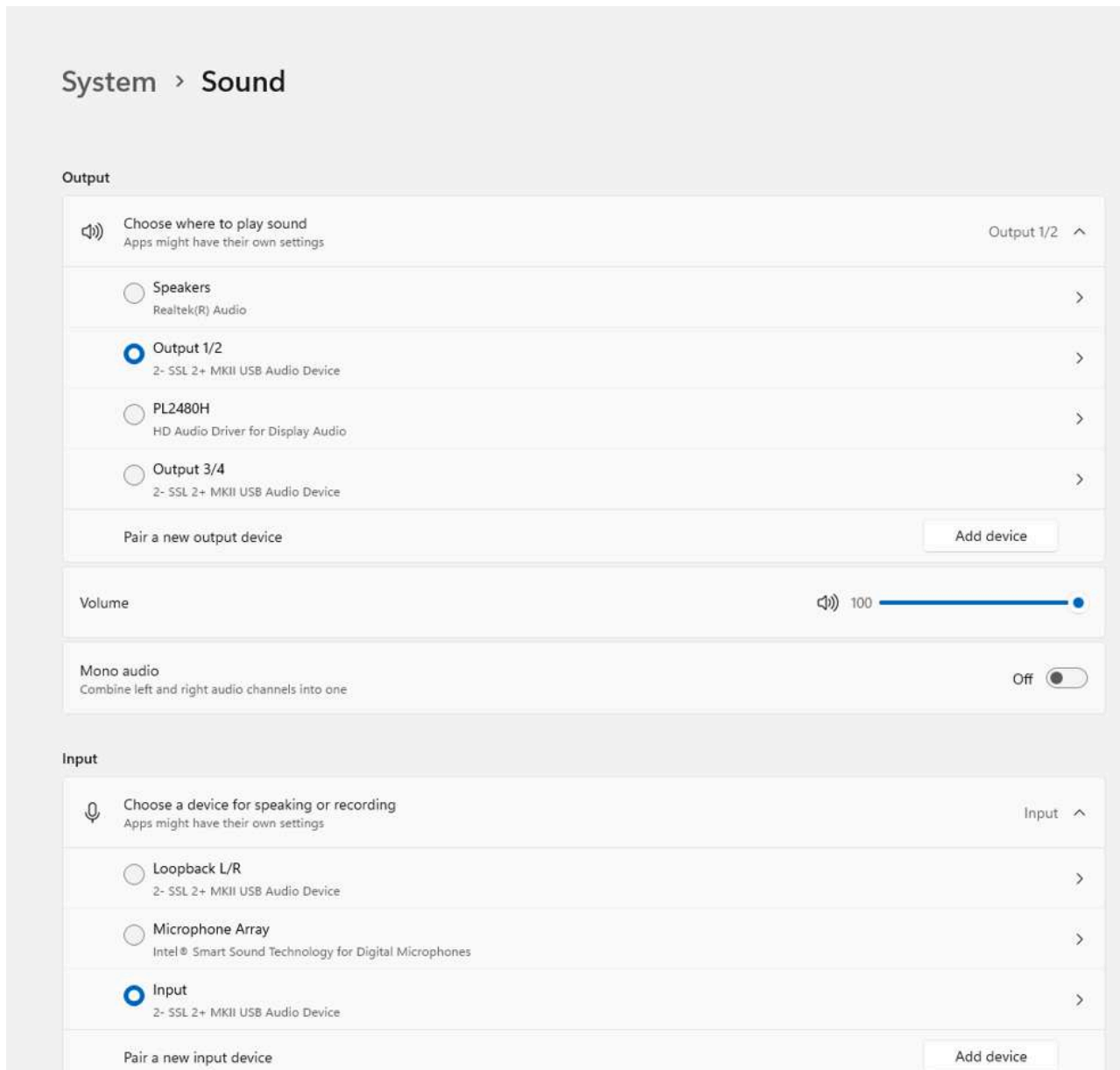
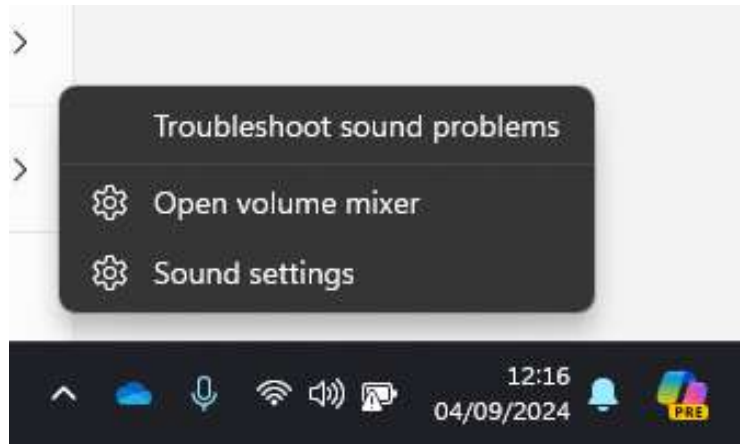
---



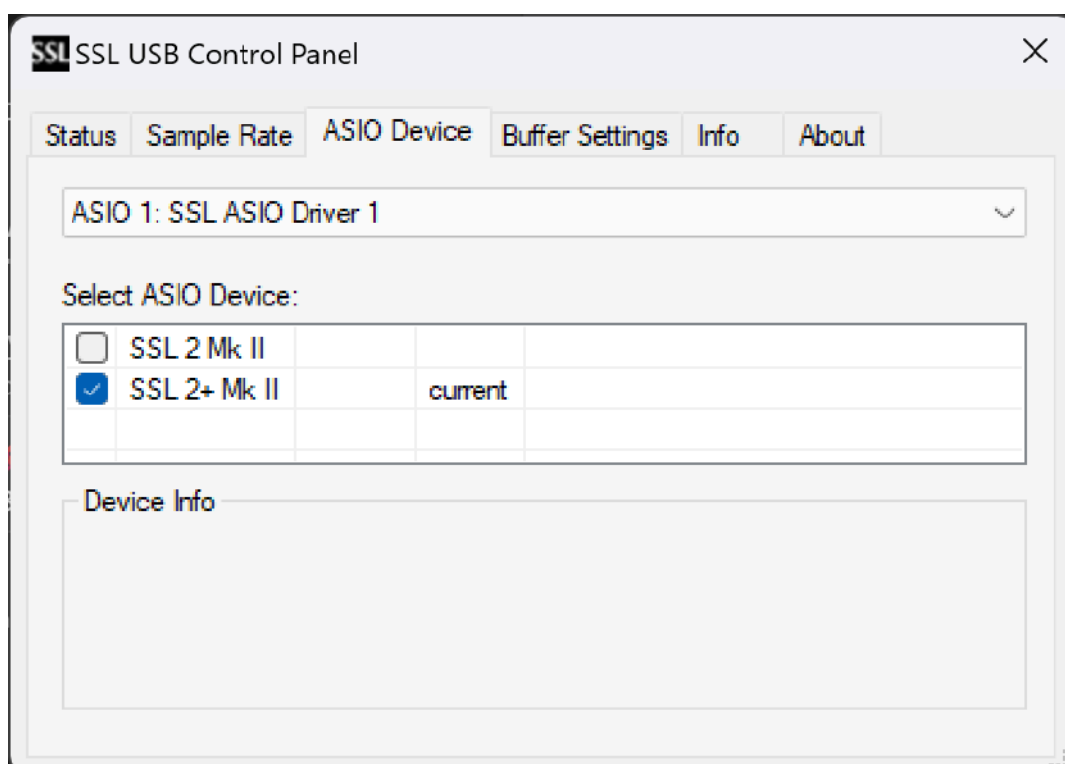
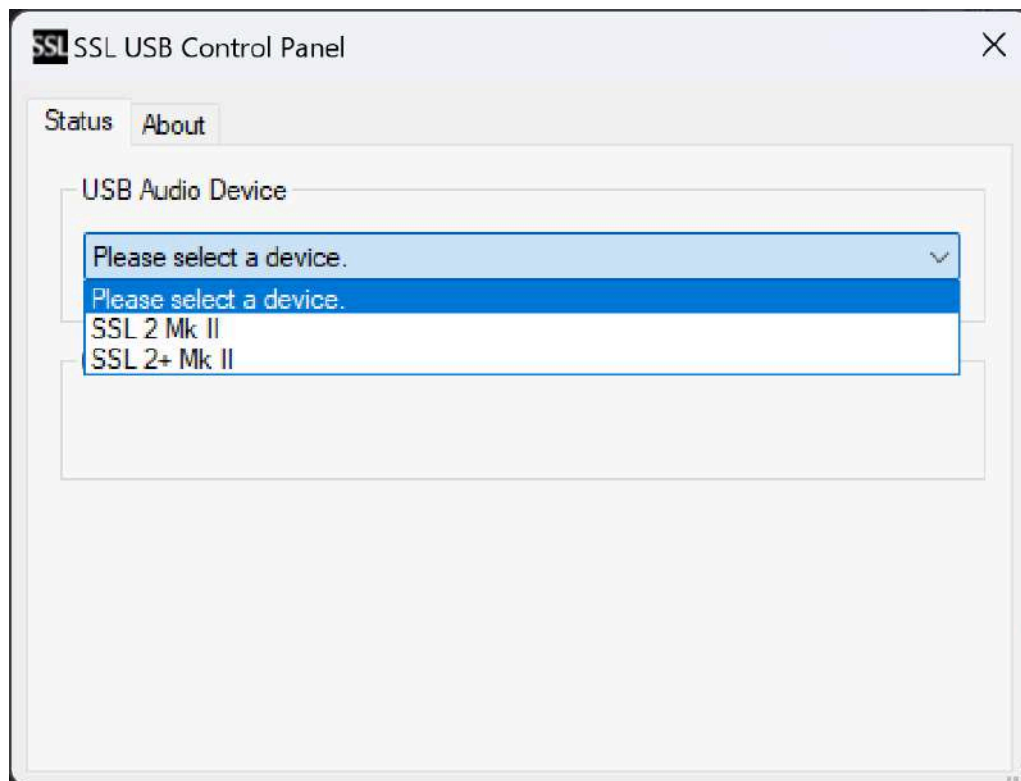
2. Download and install the SSL USB ASIO/WDM audio driver for your SSL 2+ MKII. Go to the following web address: [www.solidstatellogic.com/support/downloads](http://www.solidstatellogic.com/support/downloads)



3. Go to 'Control Panel' then 'Sound Settings' and select 'SSL 2+ MKII USB' as the default device on both the 'Playback' and 'Recording' tabs

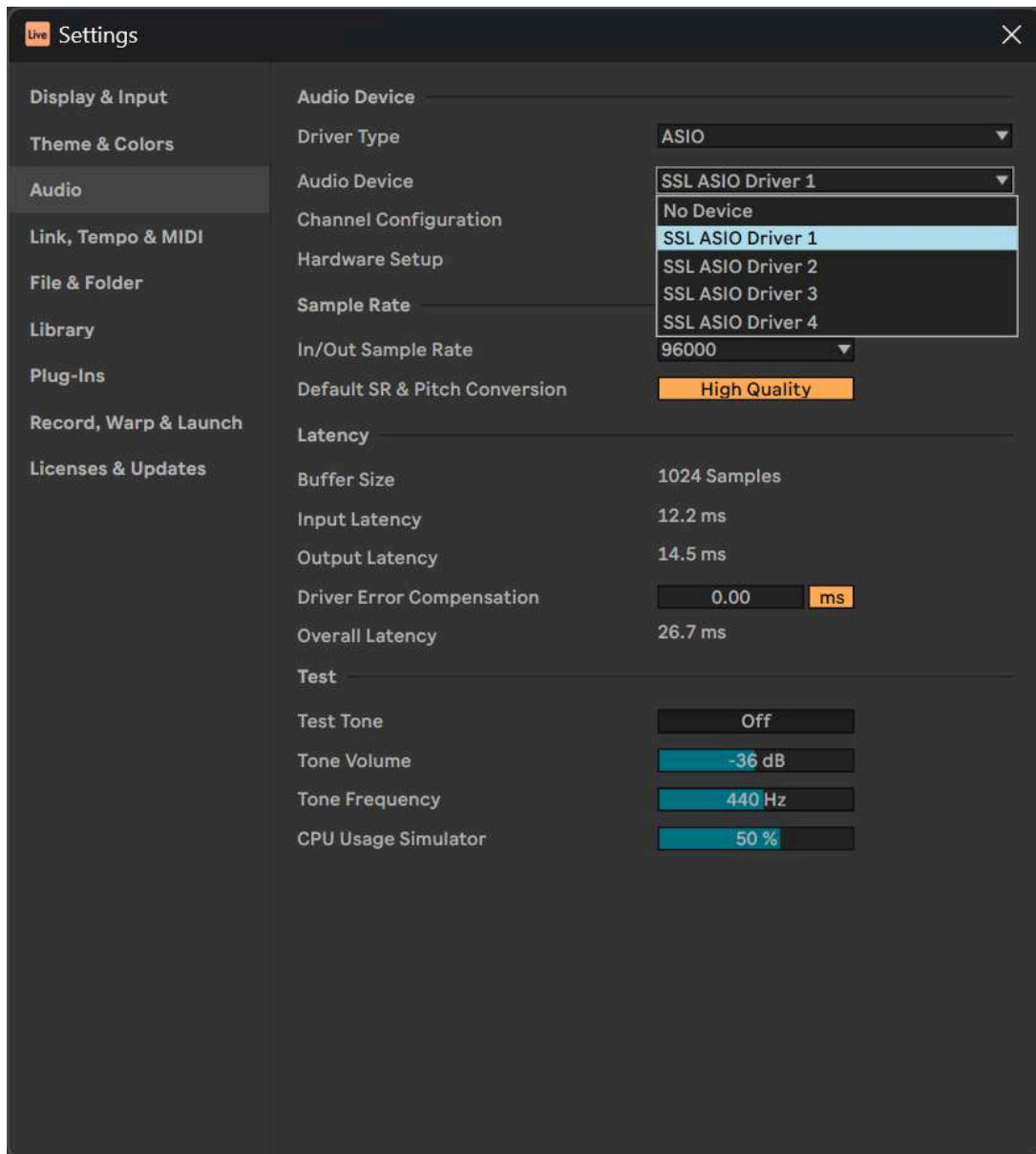


4. Go Into the SSL USB Control Panel & Select your SSL Interface & Assign the ASIO Driver (1-4)



5. Go to your DAW's Audio preferences panel and select the correct ASIO Driver for the interface you are using.





The SSL USB ASIO/WDM driver supports multiple ASIO instances. This means that you can have multiple ASIO applications working with multiple SSL USB devices. For example, SSL 2 MKII working with Pro Tools and SSL 12 working with Ableton Live. Meaning that the driver can be used in a Multi Client environment.

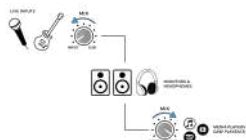
Even if you do not plan to use multiple ASIO devices, there have been some changes in how the driver presents to the DAW and as such, you need to follow the below steps to have your SSL USB audio device work with your DAW – you need to link your desired SSL device to one of the 4 ASIO Driver instances in the control panel and then choose that same Driver (SSL ASIO Driver X) in your DAW.

For more information on this process, please visit the [SSL Windows ASIO Driver setup Page](#)

## Can't Hear Anything?

---

If you have followed the Quick-Start steps but are still not hearing any playback from your media player or DAW, check the position of the MIX control. In the left-most position, you will hear only the inputs you have connected. In the right-most position, you will hear the USB playback from your media player/DAW.



*In your DAW, ensure that 'SSL 2+ MKII' is selected as your audio device in the audio preferences or playback engine settings. Don't know how? Please see below....*

## Selecting SSL 2+ MKII As Your DAW's Audio Device

---

If you have followed the Quick-Start / Installation section then you are ready to open up your favourite DAW and start creating. You can use any DAW that supports Core Audio on Mac or ASIO/WDM on Windows.

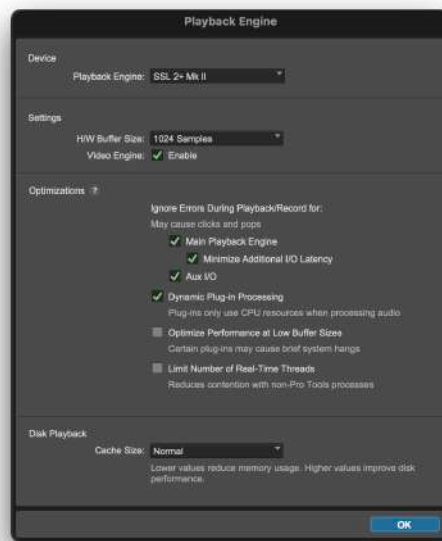
No matter which DAW you are using, you need to ensure that SSL 2+ MKII is selected as your audio device in the audio preferences/ playback settings. Below are examples in Pro Tools and Ableton Live Lite. If you are unsure, please refer to your DAW's User Guide to see where these options can be found.

## Pro Tools Setup

---

Open Pro Tools and go to the 'Setup' menu and choose 'Playback Engine...'. Make sure that SSL 2+ MKII is selected as the 'Playback Engine' and that 'Default Output' is Output 1-2 because these are the outputs that will be connected to your monitors.

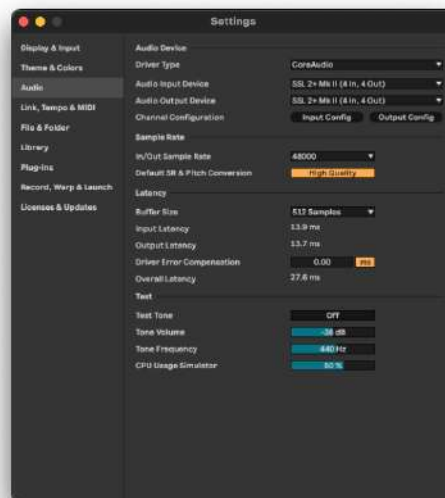
*Note: On Windows, ensure that 'Playback Engine' is set to 'SSL 2+ MKII ASIO' for the best possible performance.*



## Ableton Live Lite Setup

Open up Live Lite and locate the 'Preferences' panel. Make sure that SSL 2+ MKII is selected as the 'Audio Input Device' and 'Audio Output Device' as shown below.

*Note: On Windows, ensure that the Driver Type is set to 'ASIO' for the best possible performance.*



## Front Panel Controls

---



## Input Channels

---

This section describes the controls for Channel 1. The controls for Channel 2 are exactly the same.

### +48V

This switch enables phantom power on the combo XLR connector, which will be sent down the XLR microphone cable to the microphone. Phantom power is required when using Condenser or Active Ribbon microphones. **BE AWARE! Dynamic & Passive Ribbon** microphones **DO NOT** require phantom power to operate and can damage some microphones if improperly engaged.

### LINE

This switch changes the source of the channel input to be from the balanced Line input. Connect line-level sources (such as keyboards and synth modules) using a TRS Jack cable into an input on the rear panel. The **LINE** input bypasses the pre-amp, instead sending the signal only through a clean line input stage with a limited amount of trim, making it ideal to connect the output of an external preamp to if you so wish. When operating in **LINE** mode, the **GAIN** control provides up to 27 dB of clean gain.

### HI-PASS FILTER

This switch engages the Hi-Pass Filter with a cut off frequency at 75Hz with a 18dB/Octave slope. This is ideal for removing unwanted low-end frequencies from an input signal and cleaning up unnecessary rumble. This is suitable for sources such as Vocals or Guitars.

### LED METERING

5 LEDs show the level at which your signal is being recorded into the computer. It is good practice to aim for the '-20' mark (the third green meter point) when recording. Occasionally going into '-10' is fine. If your signal is hitting '0' (top red LED), that means it

is clipping, so you'll need to lower the GAIN control or output from your instrument. Scale markings are in dBFS.

## **GAIN**

This control adjusts the pre-amp gain applied to your microphone, line-level source or instrument. Adjust this control so that your source is lighting all 3 green LEDs most of the time whilst you are singing/playing your instrument. This will give you a healthy recording level into the computer. Note that when in **LINE** mode, the gain range is 27 dB (instead of 64 dB for Mic/Instrument).

## **LEGACY 4K - ANALOGUE ENHANCEMENT EFFECT**

Engaging this switch allows you to add some extra analogue 'magic' to your input when you need it. It injects a combination of high-frequency EQ-boost, together with some finely tuned harmonic distortion to help enhance sounds. We have found it to be particularly pleasant on sources such as vocals and acoustic guitar.

This enhancement effect is created completely in the analogue domain and is inspired by the kind of extra character the legendary SSL 4000-series console (often referred to as '4K') could add to a recording. The 4K was renowned for many things, including a distinctive 'forward', yet musical-sounding EQ, as well as its ability to impart a certain analogue 'mojo'. You will find that most sources become more exciting when the 4K switch is engaged!

*'4K' is the abbreviation given to any SSL 4000-series console. 4000-series consoles were manufactured between 1978 and 2003 and are widely regarded as one of the most iconic large-format mixing consoles in history, due to their sound, flexibility and comprehensive automation features. Many 4K consoles are still in use today by the world's leading mix engineers.*

## **Monitoring Section**

---

This section describes the controls found in the monitoring section. These controls affect what you hear through your monitor speakers and the headphone output.



### MIX (Top-Right Control)

This control directly affects what you hear coming out of your monitors and headphones. When the control is set to the left-most position labelled **INPUT**, you will hear only the sources you have connected to Channel 1 and Channel 2 directly, **without latency**.

If you are recording a stereo input source (e.g. a stereo keyboard or synth) using Channels 1 and 2, press the **STEREO** switch so that you hear it in stereo. If you are only recording using one Channel (e.g. a vocal recording), make sure that **STEREO** is not pressed, otherwise, you will hear the vocal in one ear!

When the **MIX** control is set to the right-most position labelled **USB**, you will hear only the audio output from your computer's USB stream e.g. music playing from your media player (e.g. iTunes/Spotify/Windows Media Player) or the outputs of your DAW tracks (Pro Tools, Live, etc).

Positioning the control anywhere in between **INPUT** and **USB** will give you a variable blend of the two options. This can be really useful when you need to record with no audible latency. Please refer to the How-To / Application Examples section for more information on using this feature.

### GREEN USB LED

Illuminates solid green to indicate that the unit is successfully receiving power over USB.

## MONITOR Level (Large Black Control)

This control directly affects the level sent out of **OUTPUTS 1** (Left) and **2** (Right) to your monitors. Turn the knob to make the volume louder. Please note the **MONITOR LEVEL** goes to 11 because it's one louder...

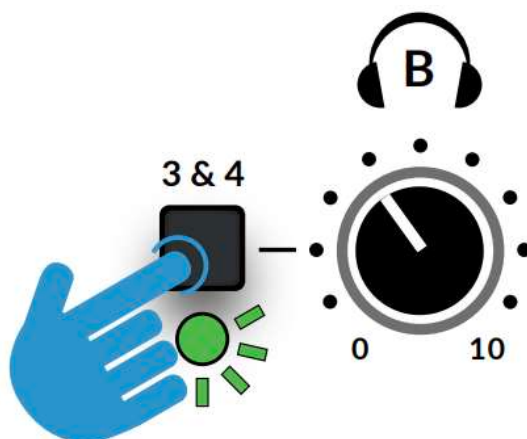
## HEADPHONE OUTPUTS

**PHONES A & B** allow for two sets of headphones to be connected, both of which can be configured to allow independent mixes for artists and engineers. Their output levels are set by the **PHONES A** and **PHONES B** controls on the front panel.

### 3&4 Button

Next to the Headphones B control, is a button labelled **3&4**. When unselected, **Headphones B** will receive the same mix as **Headphones A** (DAW Outputs 1-2). Engaging the **3&4** button instead sources **Headphones B** from DAW Outputs 3-4, allowing for the creation of an independent mix (perhaps for the artist). You would use aux sends in the DAW routed to outputs 3-4 to create this independent mix.

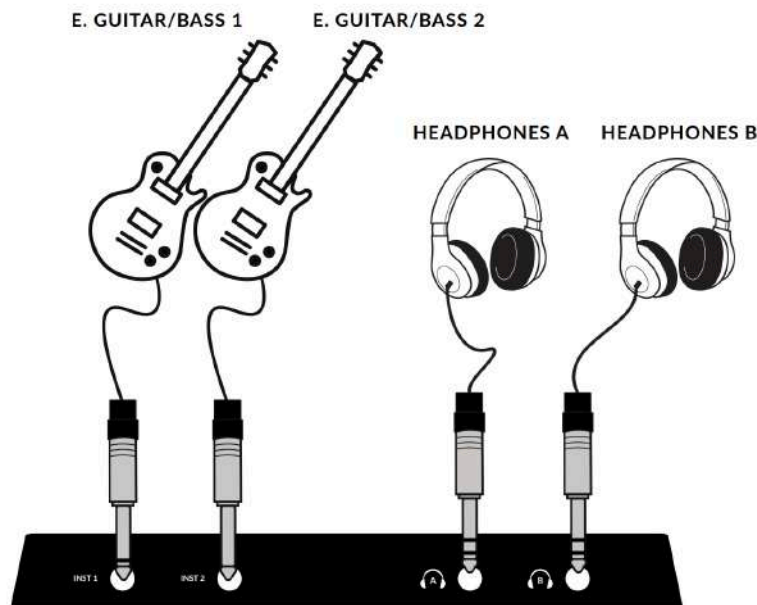
By default, the **Headphones B** Output with **3&4** engaged will not respect the **MIX** control e.g. Only DAW Outputs 3-4 are sent to **Headphones B**. Pressing and holding **3&4** until the LED starts to flash will allow **Headphones B** to respect the **MIX** control, allowing the artist to benefit from a mix of the low-latency input signals (Inputs 1-2), together with a custom headphone mix (3&4). You can toggle between the two modes whenever you like.



## Front Panel Connections

---

This section describes the 1/4" Jack Connections found at the front of the Interface. These connections allow for direct instrument inputs & headphone outputs.



### INST 1 & 2: 1/4" Input Jacks

2 x Hi-Z (DI) 1/4" input jacks for connecting Instrument sources such as electric guitar or bass. Plugging into the **INST** jack will automatically select it, overriding Mic/Line selection on the channel.

### PHONES A & B: 1/4" Output Jacks

2 x independent Headphones outputs, with individual level controls and ability for **PHONES B** to source outputs 1-2 or 3-4.

## Rear Panel Connections



### INPUTS 1 & 2 : Combo XLR / 1/4" Jack Input Sockets

This is where you connect your mic/line input sources (microphones, keyboards etc) to the unit. Once connected, your inputs are controlled using front panel Channel 1 and Channel 2 controls respectively. The combo XLR / 1/4" Jack socket contains an XLR and a 1/4" Jack in one connector (the Jack socket is the hole in the middle). If you are connecting a microphone, then use an XLR cable. If you want to connect a Line level Input such as or keyboard/synth, then use a Jack cable (TS or TRS Jacks). For connecting an instrument directly (bass guitar/guitar), use the **INST 1 & 2** jack connections at the front (not the Combo XLR/Jack socket on the rear), which automatically apply an appropriate instrument impedance (1 MΩ).

*Please note that the line-level input can only be accessed via the rear panel Combo jack socket, not the XLR). If you have a line-level devices that outputs on XLR, please use a XLR to jack adapter.*



## BALANCED LINE OUTPUTS 1 - 4 : 1/4" TRS Jack Output Sockets

Outputs 1 & 2 are primarily to be used for your main monitors and the physical volume is controlled by the Monitor Knob on the front of the Interface. Outputs 3 & 4 can be used for miscellaneous tasks such as feeding external headphone mixers/amps or sending signals to external effects units.

All outputs are also **DC-coupled** and able to send +/-5v signal to allow CV control to Semi & Modular Synths, Eurorack and CV-enabled outboard FX.

*Please Note:* More information is available in the **CV Control via Ableton® Live CV Tools** section in this User Guide.

When using outputs 1-2 for CV output, remember the Monitor Control Knob is still affecting the signal. Some experimentation on finding the best level for your connected CV controlled synth/FX unit may be required.

## USB 2.0 Port : 'C' Type Connector

Connect this to a USB port on your computer, using one of the two cables provided in the box.

## MIDI IN & OUT

The MIDI (DIN) IN & OUT allow the SSL 2+ MKII to be used as a MIDI interface. MIDI IN will receive MIDI signals from keyboards or controllers & MIDI OUT allows MIDI information to be sent out to trigger Synths, Drum machines or any MIDI controllable equipment you have available.

## Kensington Security Slot

The K slot can be used with a Kensington Lock to secure your SSL 2+ MK II.

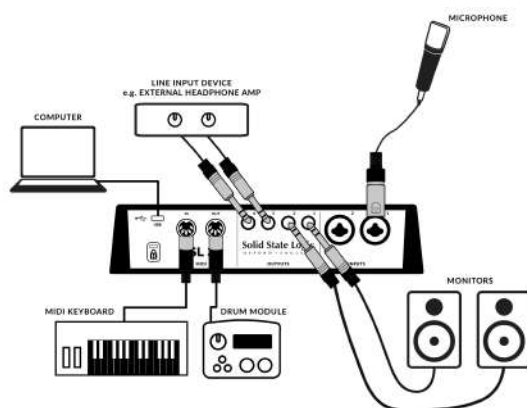
## How-To / Application Examples

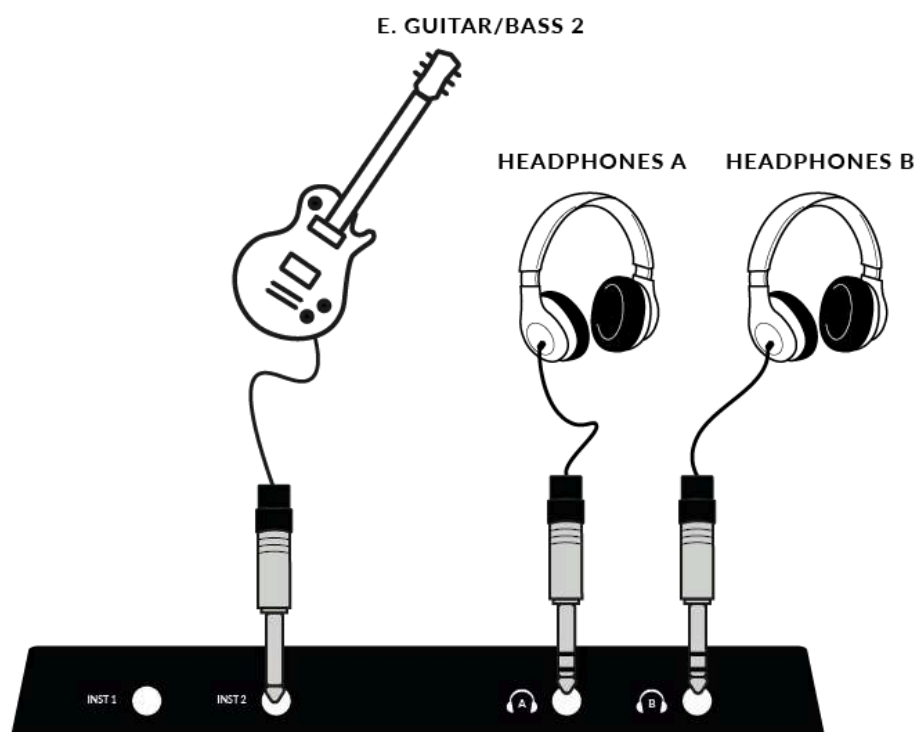
---

### Connections Overview

---

The diagram below illustrates where the various elements of your studio connect to SSL 2+ MKII on the rear panel.





This diagram shows the following:

- A microphone plugged into **INPUT 1**, using an XLR cable
- An Electric Guitar / Bass plugged into **INST 2**, using an TS cable
- Monitor speakers plugged into **OUTPUT 1** (Left) and **OUTPUT 2** (Right), using TRS jack cables (balanced cables)
- An external Line Input Device being plugged from **OUTPUTS 3 & 4**
- A MIDI enabled keyboard connected to the MIDI input
- A MIDI enabled Drum Machine connected to the MIDI output
- A computer connected to the USB 2.0, 'C' Type port using one of the provided cables
- A Pair of Headphones connected to **HEADPHONES A & B**

## Selecting Your Input and Setting Levels

### Dynamic & Passive Ribbon Microphones

Plug your microphone into **INPUT 1** or **INPUT 2** on the rear panel using an XLR cable.

1. On the front panel, make sure that Neither **+48V** or **LINE** are pressed down.
2. Whilst singing or playing your instrument that has been mic'd up, turn the **GAIN** control up until you consistently get 3 green lights on the meter.

This represents a healthy signal level. It's OK to light up the **amber LED** (-10) occasionally but make sure you don't hit the top **red LED**. If you do, you'll need to turn the **GAIN** control down again to stop clipping. Engage the **High Pass Filter** switch to remove unwanted subsonic rumble, if you need it.

3. Push the **LEGACY 4K** switch to add some extra analogue character to your input, if you need it.



---

## Condenser & Active Ribbon Microphones

---

Condenser & Active Ribbon microphones require phantom power (**+48V**) to work. If you're using a condenser or Active Ribbon microphone, you'll need to engage the **+48V** switch. **LINE** should remain unpressed. You'll notice the top red LEDs blink whilst phantom power is applied. The audio will be muted for a few seconds. Once phantom power has been engaged, proceed with steps 2 and 3 like before.



---

## Keyboards and Other Line-Level Sources

---

- Plug your keyboard/line-level source into **INPUT 1** or **INPUT 2** on the rear panel using a jack cable.
- Follow Steps 2, 3 & 4 on the previous page to set your levels for recording.



---

## Electric Guitars and Basses (Hi-Impedance Sources)

---



- Plug your guitar/bass into **INST 1** or **INST 2** on the lower front panel using a jack cable.
- Follow Steps 2 and 3 on the previous page to set your levels for recording.

---

## Monitoring Your Inputs

---

Once you have selected the correct input source and have a healthy 3 green LEDs of signal coming in, you're ready to monitor your incoming source.

1. First, ensure that the **MIX** control is rotated towards the side labelled **INPUT**.
2. Secondly, turn up the **PHONES** control to listen on headphones. If you want to listen through your monitor speakers, turn up the **MONITOR LEVEL** control.



**CAUTION!** If you are using a microphone, and monitoring the **INPUT** be careful about turning the **MONITOR LEVEL** control up because this can cause a feedback loop if the microphone is close to your speakers. Either keep the monitor control at a low-level or monitor through headphones.

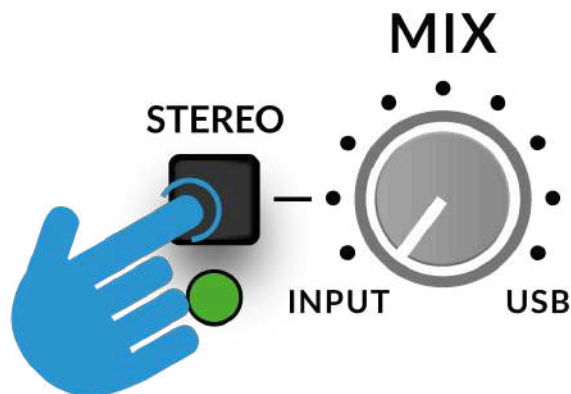
## Monitoring Your DAW

If you wish to blend the playback of your DAW with your input for low latency monitoring you can use the Mix control to blend the Input signal and DAW playback.

1. First, ensure that the **DAW INPUT** channel is muted to avoid doubling the signal in your headphones.
2. Secondly, turn the **MIX** control to listen to a balance of the signals, finding a suitable level for each for comfortable levels.

1/2  
0dB  
-20dB

## When To Use The STEREO Switch



If you are recording a single source (a single microphone into one channel) or two independent sources (such as a microphone on the first channel and a guitar on the second channel), leave the **STEREO** switch unpressed, so that you hear the sources in the middle of the stereo image. However, when you are recording a stereo source such as the left and right sides of a keyboard (coming into channels 1 and 2 respectively), then pressing the **STEREO** switch will allow you to monitor the keyboard in true stereo, with **CHANNEL 1** being sent to the left side and **CHANNEL 2** being sent to the right side.

## Using The 3&4 Button

---

Engaging the **3&4** button changes the source for **Headphones B** from **Outputs 1&2** to DAW Outputs 3-4, allowing for the creation of an independent mix (perhaps for the artist). You would use aux sends in the DAW routed to outputs 3-4 to create this independent mix.



By default, the **Headphones B** Output with **3&4** engaged will not respect the **MIX** control e.g. Only DAW Outputs 3-4 are sent to **Headphones B**. Pressing and holding **3&4** until the LED starts to flash will allow **Headphones B** to respect the **MIX** control, allowing the artist to benefit from a mix of the low-latency input signals (Inputs 1-2), together with a custom headphone mix (3&4). You can toggle between the two modes whenever you like.

## Setting Up Your DAW To Record

---

Now that you have chosen your input(s), set the levels and can monitor them, it's time to record into the DAW. The following image is taken from a Pro Tools session but the same steps will apply to any DAW. Please consult your DAW's User Guide for its operations. If you have not already done so, please ensure that SSL 2+ MKII is the selected Audio Device in your DAW's audio setup.



## Setting Up Your DAW Tracks

---

- Set up new audio track(s) in your DAWs.
- Set the appropriate input on your DAW track(s): Input 1 = Channel 1, Input 2 = Channel 2.
- Record Arm the tracks you are recording.
- You are ready to hit record and do a take.

## Low Latency - Using The Mix Control

---

### What is Latency in relation to recording sound?

---

Latency is the time it takes for a signal to pass through a system and then be played out again. In the case of recording, latency can cause the performer significant issues as it results in them hearing a slightly delayed version of their voice or instrument, sometime after they played or sung a note, which can be very off-putting when trying to record.

The main purpose of the **MIX** control is to provide you with a way of hearing your inputs before they pass into the computer, with what we describe as 'low-latency'. It is, in fact, so low (under 1ms) that you will not hear any perceivable latency when playing your instrument or singing into the microphone.

### How To Use The Mix Control When Recording & Playing Back

---

Often when recording, you'll need a way of balancing the input (microphone/instrument) against the tracks playing back from the DAW session.

Use the **MIX** control to balance how much of your 'live' input you are hearing with low-latency in the monitors/headphones, against how much of the DAW tracks you have to perform against. Setting this correctly will help enable either yourself or the performer to produce a good take. To put it simply, turn the knob to the left to hear 'more me' and to the right for 'more backing track'.

1.2  
4000

## Hearing Double?

When using the **MIX** to monitor the live input, you'll need to mute the DAW tracks you are recording onto, so that you do not hear the signal twice. When you want to listen back to what you have just recorded, you'll need to un-mute the track you have recorded onto, to hear your take.



Record Tracks Muted

## DAW Buffer Size

From time to time, you may need to alter the Buffer Size setting in your DAW. Buffer Size is the amount of samples stored/buffered, before being processed. The bigger the Buffer Size, the more time the DAW has to process the incoming audio, the smaller the Buffer Size, the less time the DAW has to process the incoming audio.

Generally speaking, higher buffer sizes (256 samples and above) are preferable when you have been working on a song for some time and have built up several tracks, often with processing plug-ins on them. You'll know when you need to increase the buffer size because your DAW will start producing playback error messages and is unable to playback, or it will play back audio with unexpected pops and clicks.

*Lower buffer sizes* (16, 32 and 64 samples) are preferable when you want to record and monitor processed audio back from the DAW with as little latency as possible. For instance, you want to plug an electric guitar directly into your SSL 2+ MKII, put it through a guitar amp simulator plug-in (like Native Instruments Guitar Rig Player) and then monitor that 'affected' sound whilst you record, instead of just listening to the 'dry' input signal.

## **Sample Rate**

---

### **What is meant by Sample Rate?**

---

All musical signals coming into and out of your SSL 2+ MKII USB audio interface need to be converted between analogue and digital. Sample rate is a measure of how many 'snapshots' are taken to build a digital 'picture' of an analogue source being captured into the computer, or deconstruct a digital picture of an audio track to play back out of your monitors or headphones.

The most common sample rate that your DAW will default to is 44.1 kHz, which means that the analogue signal is being sampled 44,100 times per second. SSL 2 MKII supports all major sample rates including 44.1 kHz, 48 kHz, 88.2 kHz, 96 kHz, 176.4 kHz and 192 kHz.

### **Do I need to change the Sample Rate?**

---

The pros and cons of using higher sample rates are beyond the scope of this User Guide but in general, the most common sample rates of 44.1 kHz and 48 kHz are still what many people choose to produce music at, so this is the best place to start.

One reason to consider increasing the sample rate you work at (e.g. to 96 kHz) is that it will lower the overall latency introduced by your system, which could be handy if you need to monitor guitar amp simulator plug-ins or lots of virtual instruments through your DAW. However, the trade-off of recording at higher sample rates is that it requires more data to be recorded onto the computer, so this results in much more hard-drive space being taken up by the Audio Files folder of your project.

### **How do I change the Sample Rate?**

---

You do this in your DAW. Some DAW's allow you to change the sample rate after you have created a session - Ableton Live Lite for instance allows this. Some require you to set the sample rate at the point at which you create the session, like Pro Tools.



## SSL USB Control Panel (Windows Only)

---

If you're working on Windows and have installed the USB Audio Driver required to make the unit operational, you will have noticed that as part of the installation, the SSL USB Control Panel will be installed onto your computer. This Control Panel will report details such as what Sample Rate and Buffer Size your SSL 2+ MKII is running at. Please note that both Sample Rate and Buffer size will be taken control of by your DAW when it is opened.

## Safe Mode

---

One aspect you can control from the SSL USB Control Panel is the tickbox for Safe Mode on the 'Buffer Settings' tab. Safe mode defaults to ticked but can be unticked. Unticking Safe Mode will reduce the overall Output Latency of the device, which may be useful if you are looking to achieve the lowest possible roundtrip latency in your recording. However, unticking this may cause unexpected audio clicks/pops if your system is under strain.



## SSL 2+ MKII DC-Coupled Outputs

---

The SSL 2+ MKII Interface allows the user to send out a **DC** Signal from any output on the interface. This allows **CV-enabled** equipment to receive the signal to control parameters.

## What is CV?

---

CV is an abbreviation of "**C**ontrol **V**oltage"; an analogue method of controlling synthesizers, drum machines and other similar equipment.

## What are CV Tools?

---

[CV Tools](#) is a free pack of CV-enabled instruments, synchronisation tools, and modulation utilities that enable users to seamlessly integrate Ableton Live with various devices in the Eurorack format or Modular Synthesisers & Analog effects units.

## Setting Up Ableton Live CV Tools

---



- Open your **Ableton Live** session
- First set up a new **Audio Track** that you'll use to send out the CV Signal.

- Then insert onto the Audio track a **CV Utilities Plug-In** from the packs menu.
- Once the CV Utility Plug-In is Open, set the **CV To** to your designated **Output**. In this example, we've set this to **Output 3/4** from the **SSL 2+ MKII**.
- Set up a second Audio track with the input signal from the Effect/Instrument and record arm to monitor the input back into Ableton Live.
- Now using the **CV Value** knob on the CV Control channel, you can automate the **CV** signal sent out of Ableton to your External Instrument/FX unit. This can then be mapped to a MIDI controller to control in realtime, record the Automation into your session, or like here assign the CV to an LFO.
- Now you can record the audio back into your Ableton Session, or other DAW you may be using to record your Audio back onto your system.
- Please note that multiple CV Utility plugs can be set up when using the SSL 2+ MKII as **EVERY PHYSICAL OUTPUT** is able to send DC signal for CV Control. Therefore you can use up to 8 CV control signals at any one time using CV Tools and an **SSL 2+ MKII**

## Requirements for CV Tools

---

- Live 10 Suite (version 10.1 or later)
- Live 10 Standard + Max for Live (version 10.1 or later)
- A DC-coupled audio interface (for CV hardware integration) such as the **SSL 2+ MKII**
- Some understanding of [Ableton Live Packs](#)
- Some understanding of [how to use CV-enabled hardware with Live](#)

## Best Practices & Safety

---

- **Never** send **CV** directly to your speakers (direct voltage can cause damage to your speakers).
- The CV Instrument device is only capable of calibrating oscillators that use bipolar voltage (+/-5V) for 1v/oct. tuning. However, some digital oscillator modules exclusively use unipolar signals (+5V or above) for tuning. As a result, CV Tools will be incompatible with these modules. If you are unsure whether this applies to the modules in your system, please consult the user manual for the device.
- Remember - Eurorack signals are up to 5x louder than line-level audio! Before connecting your modular system to a digital audio interface, be sure to reduce the signal down to line-level using a dedicated output module.

## Specifications

---

### Audio Performance Specifications

---

Unless specified otherwise, default test configuration:

Sample Rate: 48kHz, Bandwidth: 20 Hz to 20 kHz

Measurement device output impedance: 40  $\Omega$  (20  $\Omega$  unbalanced)

Measurement device input impedance: 200 k $\Omega$  (100 k $\Omega$  unbalanced)  
Unless otherwise quoted all figures have a tolerance of  $\pm 0.5$ dB or 5%

### **Microphone Inputs**

Frequency Response:  $\pm 0.1$  dB  
Dynamic Range (A-Weighted): 116.5 dB  
THD+N (@ 1kHz): -100 dB / < 0.001 % @ -8 dBFS  
EIN (A-Weighted, 150  $\Omega$  termination): -130.5 dBu  
Maximum Input Level: +9.7 dBu  
Gain Range: 64 dB  
Input Impedance: 1.2 k $\Omega$

### **Line Inputs**

Frequency Response:  $\pm 0.05$  dB  
Dynamic Range (A-Weighted): 117 dB  
THD+N (@ 1kHz): -104 dB / < 0.0007 % @ -1 dBFS  
Maximum Input Level: +24 dBu  
Gain Range: 27dB  
Input Impedance: 14 k $\Omega$

### **Instrument Inputs**

Frequency Response:  $\pm 0.05$  dB  
Dynamic Range (A-Weighted): 116 dB  
THD+N (@ 1kHz): -99 dB / < 0.001 % @ -8 dBFS  
Maximum Input Level: +15 dBu  
Gain Range: 64 dB  
Input Impedance: 1 M $\Omega$

### **Balanced Outputs**

Frequency Response:  $\pm 0.03$  dB  
Dynamic Range (A-Weighted): 120 dB  
THD+N (@ 1kHz): -108 dB / < 0.0004%  
Maximum Output Level: +14.5 dBu  
Output Impedance: 150  $\Omega$

### **Headphone Outputs**

Frequency Response:  $\pm 0.05$  dB  
Dynamic Range: 119.5 dB  
THD+N (@ 1kHz): -106 dB / < 0.0005% @ -8 dBFS  
Maximum Output: Level +13 dBu  
Output Impedance: <1  $\Omega$

## Digital Audio

Supported Sample Rates: 44.1 kHz, 48 kHz, 88.2 kHz, 96 kHz, 176.4 kHz, 192 kHz

Clock Source Internal

USB 2.0

Low-Latency Monitor Mix Input to Output: < 1ms

Roundtrip Latency at 96 kHz: Windows 10, Reaper: < 3.65 ms (Safe Mode Off)

Mac OS, Reaper: < 5.8 ms

## Physical Specifications

---

### Analogue Inputs 1&2

Connectors XLR: "Combo" for Microphone/Line/Instrument on rear panel

Input Gain Control: Via front panel

Microphone/Line Switching: Via front panel switches

Instrument Switching: Automatic upon jack connecting

Phantom Power: Via front panel switches

Legacy 4K Analogue Enhancement: Via front panel switches

### Analogue Outputs

Connectors: 1/4" (6.35 mm) TRS jacks: on rear panel

Stereo Headphone Output 1/4" (6.35 mm) TRS jack: on rear panel

Monitor Outputs L/R Level Control: Via front panel

Monitor Mix Input - USB Blend: Via front panel

Monitor Mix - Stereo Input: Via front panel

Headphones Level Control: Via front panel

### Rear Panel Miscellaneous

USB 1 x USB 2.0, 'C' Type Connector

Kensington Security Slot 1 x K-Slot

### Front Panel LEDs

Input Metering Per Channel - 3 x green, 1 x amber, 1 x red

Status LEDs: +48V red, LINE green, HPF green, STEREO green, 3&4 green

Legacy 4K Analogue Enhancement Per Channel - 1 x red

USB Power 1 x green

### Weight & Dimensions

Width x Depth x Height 234 mm x 159 mm x 70 mm (including knob heights)

Weight 900 g

Box Dimensions 277 mm x 198 mm x 104 mm

Boxed Weight 1.22 kg

## **Troubleshooting & FAQs**

---

Frequently Asked Questions and additional support contacts can be found on the Solid State Logic Website at: [www.solidstatellogic.com/support](http://www.solidstatellogic.com/support)