# ABLETON'S MIXER ANATOMY ROUTING AND SIGNAL FLOW

Ableton Lives mixer, along with any other DAW's mixer has been designed to emulate conventional hardware mixing consoles used in professional recording studios. Although hardware is still widely used, it has come to the point where software audio workstations are actually capable of a whole lot more than most hardware mixing desks. We are no longer confined to the amount of channels on our desk, or the amount of money we have to spend on hardware. With software such as Ableton, we are now able to have as many tracks as our CPU will allow. The routing capabilities throughout these tracks are virtually limitless.

With all these routing options and mix functions at our disposal, it can very quickly get messy and frustrating for the new music producer. So in this section we are going to explain the mixer, its functions, and its routing one by one. Which should allow us to easily navigate our way around Ableton's mixer.

After we have some more experience with mixing we will notice that the fundamentals are the same across all platforms, from Logic & Ableton, right the way through to a proper mixing desk. Once we understand the fundamentals, it simply becomes a case of getting familiar with the different ways of doing things in each respective program.

## WHAT IS SIGNAL FLOW

Signal flow is how the sounds are routed through our Mixing desk or DAW. We can re-route different channels in, and out of one another, as well as sending them to groups, send/return effects and mix busses and the master output.

On a conventional desk, most of the routing is done through a patch bay, which is designed to stop us having to go into the back of the mixing desk to constantly plug in equipment. Instead we already have all of the hardware wired into the patch bay, and we can simply grab some patch cables, and send signals to and from the console.

DAW’s work exactly the same as this; however, there is no need for a patch bay, as there are no physical cables. Instead we just use software plugins that are routed to one another, using Input and output dropdown menus.

When we look at the input/output section of Ableton, we can understand the basic routing of most channels.

**INPUTS**

This is where we can route any form of audio into Ableton. For example, we may want to monitor:

- Microphones
- DI's (Direct inputs from guitars etc.)
- Synthesizers
- External sources from our soundcard
- Fold-back / Cue from a live room

If we select our soundcard, we will then get a second dropdown menu. This will allow us to choose between different stereo or mono channels on our soundcard.

We can change the input routing of tracks; this is handy for if we would like one track to feed into another one. We also have a sub menu within this routing matrix to enable us to route individual sounds within a track. This is important for certain instruments such as Impulse, which may require you to make extra auxiliary channels to separate the individual sounds.

**MONITOR MODES**

Monitor modes let us choose about how we monitor the INPUT signal.

**IN** - The input signal will constantly be monitored

**AUTO** - The input signal will only be monitored if record/arm is enabled

**OFF** - Will turn off any input monitoring for the selected track

**INSERTS**

Inserts come first in the mixer chain, the audio passes through the insert on to the next part of the mixer. (Which run from left to right along the bar on the bottom of the screen.)

We get a graphical representation of the levels of the inputs and outputs of each device, which can come in handy for a quick indicator of our gain staging. We can also use grouped FX chains to perform parallel processing from the insert strip; we will go into this subject in more detail in our compression eBook of this mix series.

On a standard mixing console there are usually Insert points on the console or into the patch bay, this allows us to break the signal flow and route it into hardware such as a compressor. The output of the compressor can then go back into the patch bay, so that the signal can continue on to the mixer. Once this signal has gone through any Insert FX, it then moves on to the main mixer functions.

**SEND KNOBS**

Here we can take a copy of the signal and route it to a bus. In Ableton these busses are automatically routed to the relevant return channel that is assigned to the send.

**PAN POT**

The pan potentiometer allows us to pan the signal anywhere in the stereo field.

**MUTE BUTTON**

The mute function allows us to mute the signal or multiple signals

**SOLO BUTTON**

The solo function will mute all other tracks so only this particular track can be heard. We can also solo multiple tracks using the CMD function.

**RECORD BUTTON**

Record/arm can be used so that we can record audio into a clip slot. This button is important if we are using AUTO as an input-monitoring mode.

**VOLUME FADER**

The size of the fader can be adjusted by dragging the mixer up and down. Having this at larger sizes is useful for accuracy when making small incremental changes during mixing. We can be even more accurate by holding down the CMD function when adjusting parameters to fine-tune our volume settings.

**METERING**

We also get peak (dark green) and RMS (light Green) metering, as well as logarithmic numbering down the sides of the meter. When the meter is expanded, we also get a peak level and a second box, which allows us to type in numerical values for the output fader volume.

When a signal peaks, this peak level will stay in the peak box display till we click on the box, which will reset the meter.

**METERING AND DYNAMIC RANGE TERMINOLOGY**

Within metering, there are a few key words that we will need to fully understand.

Within a DAW we use dBFS or 'decibels full scale'. Which works **backwards** from 0dBFs.

The **noise floor** is the quietest signal that we can hear within our DAW. This is where we will usually find ground hum and tape hiss. Within a DAW this will be right down at –Inf dBFS.
Next is the **nominal level,** this is the optimal level for recording instruments into your console or DAW. On hardware and mixing consoles this is at +4dBu, which is also equivalent to 0 VU. (Analog circuitry is designed to have the best signal to noise ratio at this optimal level) however, within your DAW this is slightly different. The optimal level for recording into a DAW is between -12 and -16 dBFS although the signal to noise ratio within a DAW is not really an issue, as long as we do not clip (go over 0dBFs) then there won't be a problem.

The difference between the **noise floor** and the **nominal level** is known as the **signal to noise ratio**. Like we mentioned earlier, this is very prevalent in analog circuitry.

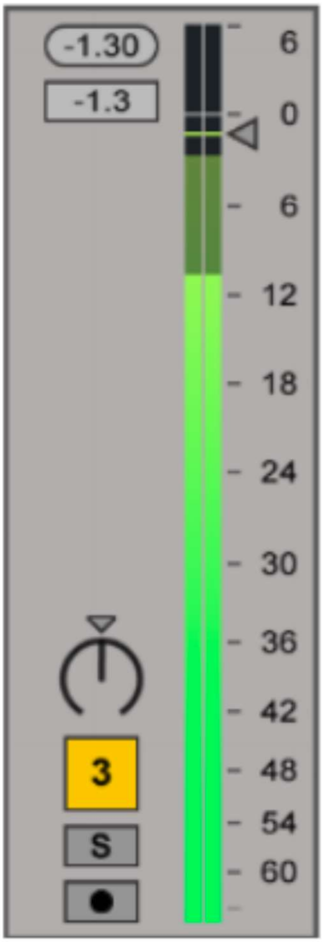
Finally we have the **maximum level**. Due to the circuitry in analog equipment, this can often be sent in hot. (Pushed hard) Into the red to give a pleasing saturated effect, as the signal is soft clipped.

In the digital realm of DAW's this is definitely not a good idea. 0dBFs is the absolute maximum your signal should ever reach within any part of your signal chain. Even occasional peaks should not exceed this limit, as digital equipment cannot accurately reproduce the waveform above this level, which will result in nasty distortion.

The difference between the **nominal level** and the **maximum level** is known as **headroom.** This is what mastering and mix engineers refer to when they say "leave some headroom in the mix for the mastering engineer to do his work."

This basically equates to not hard compressing or limiting a waveform, and leaving them ample space to work with for the mastering stage.

The difference between the **noise floor** and the **maximum level** is known as the **total dynamic range**.




As the track develops and our quietest signals start to become louder we will be decreasing our **dynamic range** in order to achieve a overall louder **RMS volume.** (Average or root mean square volume)

**PEAK METERING ANALYSIS**

Peak meters are used to tell us the maximum level of a signal. This doesn't really give us a very good indicator of how loud a signal is, but it is very useful for knowing if we are overloading our signal at any point in the chain which could cause clipping.

**RMS METERING ANALYSIS**

RMS or root-mean-square meters, use a mathematical function to work out the average loudness of a signal. This is a much better representation of the overall signal level, but cannot be relied on to see if we are clipping a signal.
The RMS volume will never be more than the

peak volume. It can only be less, or in extremely rare circumstances it can be equal to the peak volume. (In the case of a completely sustained note with no decay or release envelope.)

RMS metering is the preferred type of meter when checking the overall level of a complete mix. This is also the type of metering used for VU meters because it closely replicates how the human ear perceives sound.

**OUTPUTS**

We can also route the output of audio to certain locations, such as groups, master outputs, soundcard outputs and auxiliary tracks. With all of these different output routings, we will often get a sub menu allowing US to choose at which point in the chain we would like to send the audio.

Common options are whether we would like to send the signal Pre, or post mixer, as well as Pre, or post FX. This is very useful if we have certain effects that we may wish to bypass when sending to another channel.