Studio cables and types

Section 1: Why the need for high, good quality cables?

Broken cables or noisy cables can lead to poor audio quality and lead to problems. Therefore, utilising good quality cables will make it less susceptible to issues and you can ensure that you are working with quality sound.

Section 2: What are the types of cables?

The common types of cables that we will be utilising in home-based music production studios can be summarised into the following along with some common properties:

1. TS cable/ instrument cable
    - Has a quarter inch jack
    - Has two segments, a tip and a sleeve
    - Is a single conductor cable, the signal is sent along the single conductor
    - Outer sleeve acts as a braided shield, to prevent noise from getting into the cable
    - It is really susceptible to picking up noise, and hence as short of this cable should be used where possible
    - Commonly used to connect instruments to output
2. TRS cable
    - Has a quarter inch jack
    - Has three segments, the tip, the ring and the sleeve
    - Is a two conductor cable with a shield
    - Frequently used to connect headphones, or as a balanced configuration where a single signal is passed across it but is done in a way that can cancel noise
    - Also susceptible to picking up noise
3. XLR cable (standard mic cable)
    - Has three segments, the tip, the ring and the sleeve
    - Not used for stereo signals, balanced configuration only

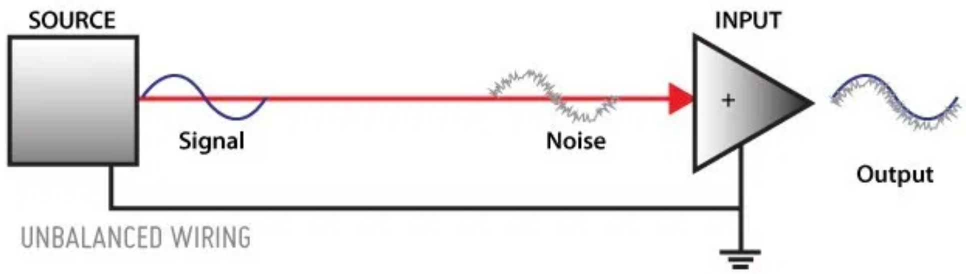
Section 3: Balanced vs unbalanced cables

As you might have read previously, some cables are susceptible to noise.

Balanced cables are relatively immune to noise from interference such as radio frequencies, electronic equipment, etc. Which is why they are the standard for pro audio. They have 3 wires, signal (+), signal (-), and ground.

On the contrary, unbalanced cables are relatively susceptible to noise from interference such as radio frequencies, electronic equipment, etc. They have 2 wires, signal, and ground.

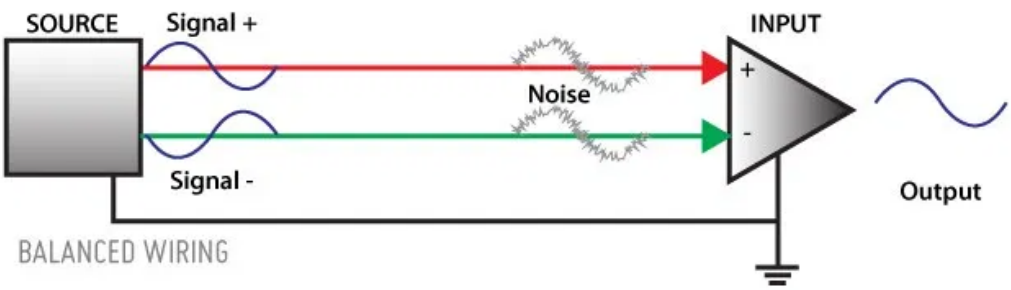
So how does their configuration lead to the difference in noise susceptibility? Let's look at the following diagram to visualise.



Unbalanced wiring uses just two conductors and is susceptible to picking up noise.

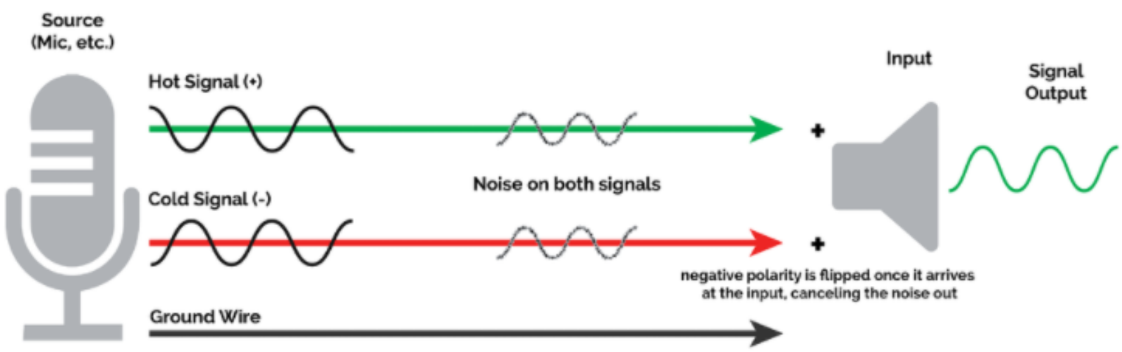
As the signal flows from the source to our input/output, it picks up noise along the way, which will be added to the overall output sound.

Meanwhile, balanced cables carry two copies of the signals, with one of them reversed. The same noise is then picked up by both the cables.



Balanced wiring uses two signal conductors plus a ground, allowing noise picked up along the way to be canceled through polarity inversion.

At the end before the input/output, the negative signal is then flipped. The noise thus cancels itself out. Therefore, we will be left with the original source signal with no background noise.



Section 4: What cable should I use?

Balanced type of cable (e.g. XLR) should be used if there is long runs needed, since it will help to reject the noise along the cable. The cables used should also depend on the jacks on the end of the gear. Conversion of cable types might be needed, covered below.

Section 5: Cable type conversions

Conversion can be done through a direct box. It allows us to go from a unbalanced cable to a balanced cable.

One example of usage is to plug in a short unbalanced cable and do the extension via another balanced cable.

You can also use the direct box for secondary output.

Section 6: Other notable types of cables

These are some other cable types that you might encounter!

- 8 inch stereo cable used for headphones
- RCA cable designed for cable, function just like a TS cable. Usually used to connect consumer grade equipment.

  ! Might need to convert -10 to +4

- MIDI cables
- Toslink cables
- SPDIF cables.

Section 7: References

The following resources are referenced here. Feel free to read more from their sites.

https://www.boxcast.com/blog/balanced-vs.-unbalanced-audio-whats-the-difference

https://www.aviom.com/blog/balanced-vs-unbalanced/#:~:text=An%20unbalanced%20cable%20consists%20of,wire%20and%20a%20ground%20wire.

Summary on DAWs channel and signal flow

Section 1 The Mixer layout

The picture above shows the mixer layout. This mixer is from Logic Pro X, so it may differ for some other DAW, but the features are likely to stay largely the same.

The mixer contains many channel strips, and more will be created when you create new instrument/MIDI or drum tracks, or when buses are created. A channel strip is an individual strip as shown above, specific to a certain track or bus.

To look at the workflow, we will start off with the Input section. Though the strip is laid out from top to bottom, that is not the workflow. This will be explained more in the following sections.

Section 2 : Main signal flow

The main signal flow starts off from the input section, where audio comes in. The signal then goes through the audio FX one by one from top to bottom. Note that the order matters, the signal passes through the audio FX right at the top first before moving on to the next one and so forth. Some audio FX include EQs, limiters, compressors, delays etc.

After going through the audio FX section, it heads down to the sends section. You can choose to send your partial mix to a bus for further mixing, into buses, or for parallel affects. For sends, you can choose the amount of signal that you want to send out, and sends are usually used for intermediaries or to buses. You can create as many sends as you like, though it gets complicated when you have too many routes.

Once you are done with the sends, the signal moves on to the output. Output will always output to somewhere, stereo out as the default. As a contrast to sends, output will always be full signal. That is the key distinction that must be made when using output.

We shall then skip all the way to the bottom for now, at the volume fader. The volume fader is the one with the slider. Adjusting it will affect the playback volume. This does not mean that it will affect the sends (KIV). For now, just understanding that it is used as a volume control. At the dB above the slider, it shows the volume your slider is at, followed by the peak volume for each strip. This can help you monitor your volume levels and ensure that it does not peak into the red.

Above the dB is the pan section. This is very intuitive to understand, it basically pans to the left or to the right (think listening to earphones, there is left/right audio). You just have to turn the knob to pan the sound. Once again, it does not mean it will affect the sends (KIV).

Section 3 : Sends settings

We had a lot of KIV in the previous section, especially about sends. I shall address them now. For one, we said previously that you can pick the percentage of signal to send over. Additionally, for sends, you can click on the particular send, and select post pan, post fader, or pre fader. This is why we had to keep in view that the sends might not always be affect. This summary is the easiest way to visualize it:

Post pan – input → Audio FX → Volume Fader → Pan → Send
Post fader – input → Audio FX → Volume Fader → Send
Pre fader – input → Audio FX → Send

This is why it is important to configure the settings correctly before sending, if not it may be difficult to diagnose what led to the sounds being off.

Section 4: Display settings

The other settings do not affect the flow of the strip. It either ends at the send or output.

Let's look at the items below output. The Group simply groups the tracks into a singular group, to assist in editing. You can then select the same settings for all those in the same group. However, unless you are doing mega mixes, you generally do not require this utility.

Next is the automation. This displays the automation data that has been assigned to your tracks if any, but does not affect your signal flow.

The next one is simply the instrument logo. You can select any logo to represent your instrument as you deem fit by pressing it. Again, this does not change the instrument, only the representative logo.

Skipping to the bottom, there is M and S. These are simply Mute and Solo buttons to allow you to hear your mix. Mute means that that particular strip will not be sounded, solo means that that particular strip will be the only one sounded along with other solos if any.

This covers everything from input down. After you are done, you would want to send it to stereo out, do some mixing and send it to master.

Just to go a bit more in depth into buses, think of your strips as passengers. They then board the same bus to a common destination. So, your strips are sent to a common location to be mixed together to reduce CPU usage, and because they can be mixed as a group.

Now, for the top part, the setting till MIDI FX, they are before your input, so they do not affect your signal flow at all, similar to the logos. These are just display data.

Section 5 : Others

You might also see a BnC at stereo. This is the bounce setting where you render the signal to somewhere else.

Music Modulation

Section 1: Key terminology explained

In this section, I will cover what are the key terms and how they tie in to each other, so that we can explore the functionality further on. We start off with having an input signal. Delay is how long the signal is delayed till output. Signal is inputted. After the delay, the signal is outputted. Dry means that only the original signal will be outputted. Wet means that only the delayed signal will be outputted. A mixture of dry/wet means that a mixture for the two signals will be outputted. Feedback is basically how many times the delay block is repeated one after another. A low level of feedback means that the repeats will fade away quickly with decreasing volume. A high level of feedback means that the repeats will fade away slowly with decreasing volume.

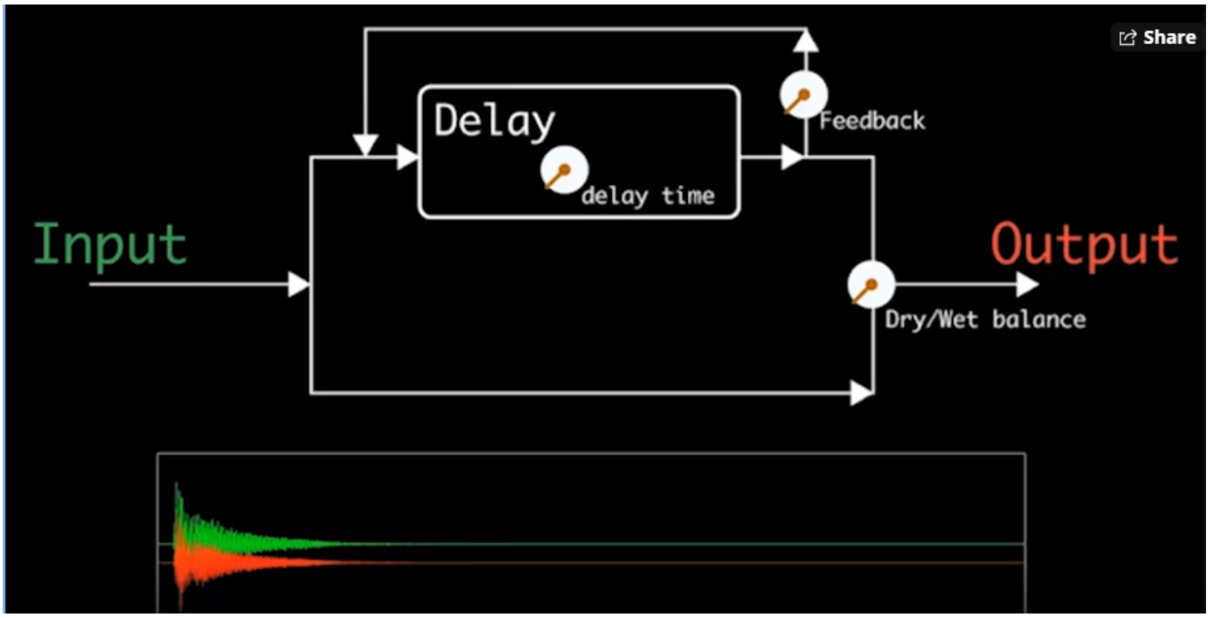


Fig. 1 Full Dry

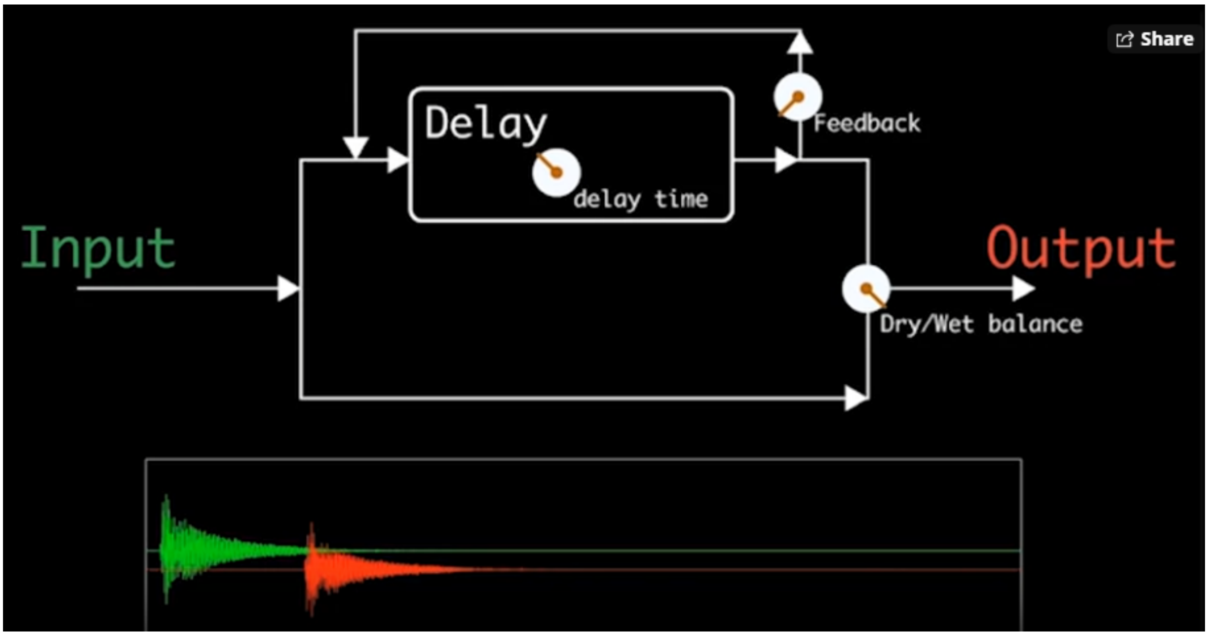


Fig 2. Full wet

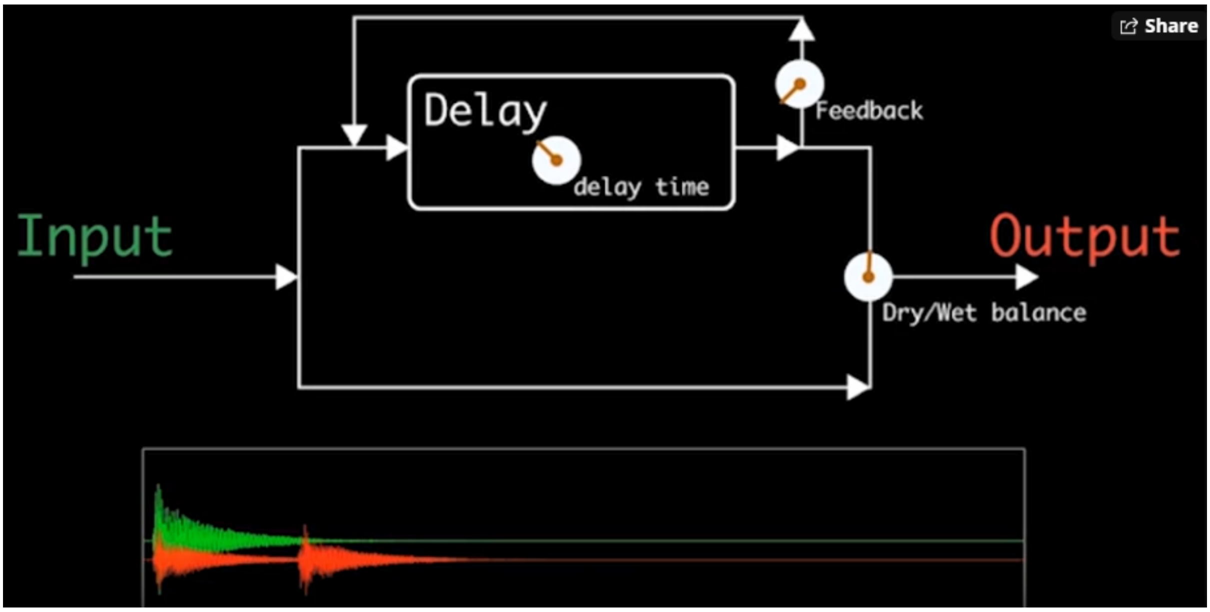


Fig 3. Mixture of Dry and Wet

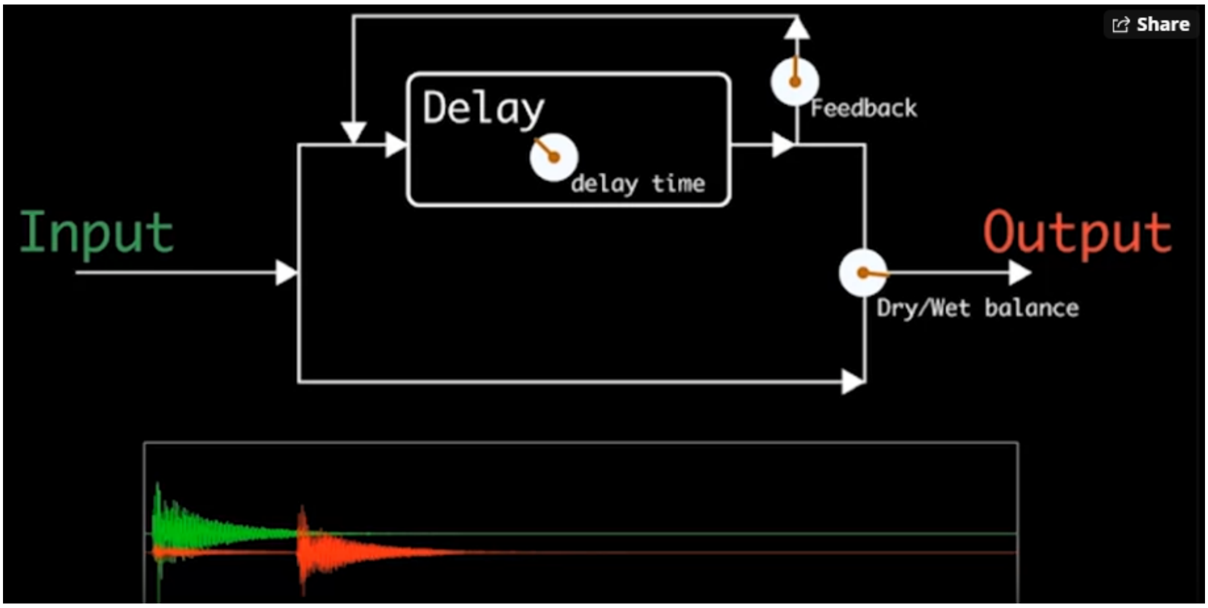


Fig 4. Low Feedback

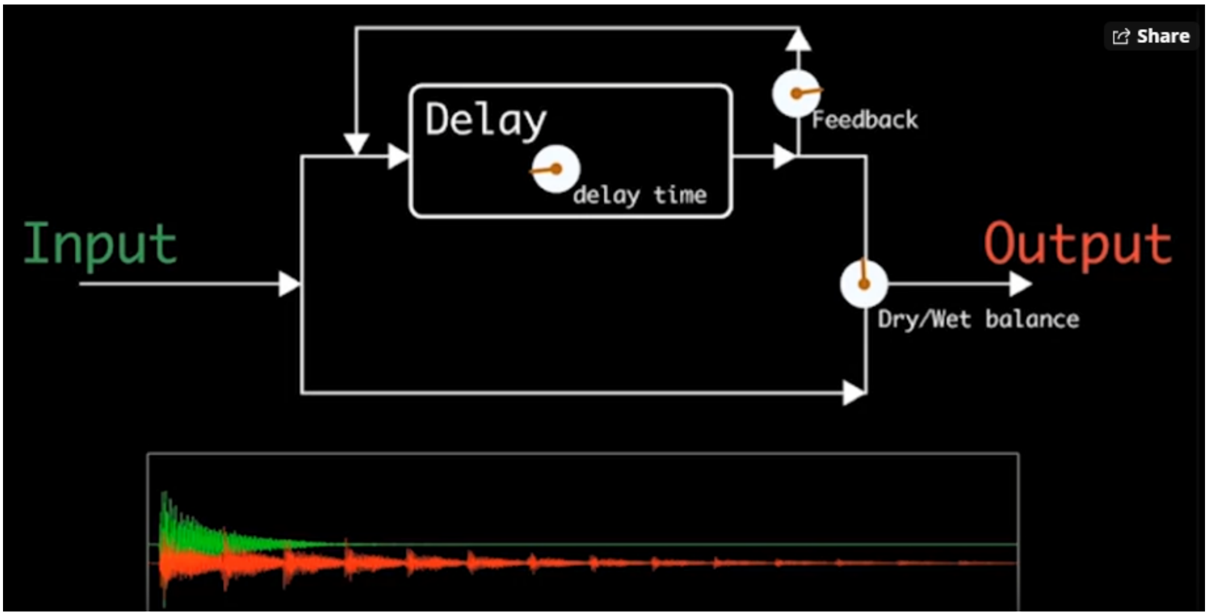


Fig 5. High Feedback

LFO stands for low-frequency oscillator, and it is used as a modulator. It is typically between 0 – 20 Hz, below the human hearing range. It is great for creating things like vibrato.

Section 2: Flanger, Phaser and chorus

In this section, two of the three will be explained.

Flanger is a comb filter in motion. A flanger is created by mixing two slightly delayed signals together. The delay is determined by an LFO. As such, the harmonics will sweep up and down ther frequency spectrum.

On the contrary, phasers can be deemed as a series of deep notches across the spectrum. It can be noted that these notches are not evenly spaced the same way that the comb filter was.

Section 3: Flangers and Phasers uses

Flangers can be great as transitional effects and momentary ear candy to keep the listener's attention. On the other hand, the flanger can sound more extreme than a phaser, so phasers can be used when more nuance is needed. However, the utility of these are very variable and dependable on the specific contexts.

References:

https://www.izotope.com/en/learn/understanding-chorus-flangers-and-phasers-in-audio-production.html