Where to set the first fader

On a regular pop mix, engineers commonly use the snare drum as the track to reference for initial loudness. The snare is probably around 75% of the loudness created by the final mix of the song. Using the meters, the level indication that the snare drum channel displays will be different depending on if you are using a peak meter or a VU. If you have a dry, transient snare, a peak meter might display a very loud level close to clipping where a VU meter might be show that the signal is low and to turn it up! Most DAW programs will have a meter that will display BOTH the peak and RMS values at the same time. The first meter to do this was a professional studio add-on called a **Dorrough Meter**. It is a good idea to have a Dorrough-style meter on the main outs of your mix to give you a good idea how loud your mix is and the level of each instrument. In Logic Pro you can use either *the Level Meter* or the *Multi Meter* (metering plug-ins) on the main output channels (usually Output 1-2). The dark blue shows peak levels and the light blue indicates average, RMS levels, thus you are more interested in the light blue portion. See page 25 for an image of the Logic Pro *Level Meter*.

Assuming you are going for a standard pop/rock level, set the RMS value of the snare to **-30dBfs**. Depending on the performance (and the mics and preamps used), the peaks of the snare might extend 20dB higher than this average level (to -10dBfs). If there is a good amount of ambience on the snare, this 'longer', greater sustaining snare will measure higher on the RMS part of the meter as well. Next, bring up the vocal on the first verse and set this level to around -20dBfs RMS. This level represents the average volume of your song during the mixing stage. If you have your monitoring system calibrated correctly, -20dBfs should be plenty loud for most balance and mix functions during a session. **If you need to hear more or less level, turn up the monitoring system, NOT the channel or output level of the DAW**. This is key and very important.

Depending on the timbre (harmonic and overtone components) of both snare and vocal sources, adjust the balance between them to taste. One should note that the loudest moments of the vocal will rise to the top of the 'average range' (around 10dB) and the softer moments may slightly dip under the average. The louder and busier the musical arrangement, the more stable the level of the voice will have to be in the mix. If the snare or voice is having a problem *sitting* within a desired dynamic range, a compressor or limiter can be used to alter the peak to RMS ratio by compressing or limiting the peaks. Thus, the track can be made louder by increasing the RMS value without causing the peaks to clip. Of course, adding a compressor or limiter is NOT a transparent operation. The tone of the drum or voice will change (it HAS to as you are altering the dynamic relationship of the fundamental and the overtones!). Sometimes this will make it sound better and sometimes it will make the sound worse. If the vocal was recorded *correctly* (a very subjective comment), you should have a relatively stable level but certain words or phrases may extend out of your 'average' range. This is fine for now, we are just setting initial levels and we are making sure we don't start with the levels too high. After the tracking, it is easier to make a mix louder and less dynamic than to make the mix softer and more dynamic. Once the song is completed and mixed, the space headroom can be used to raise the overall level of the song. This is typically done during the mastering process where the levels of multiple songs are adjusted in concert with another. The headroom above -20dBfs is like currency, you spend it carefully and wisely. Re-read that last sentence and understand the ramifications. You will spend it, but knowing where and when is a professional decision.

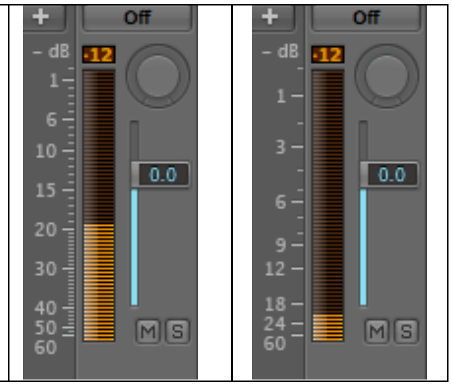
Once you have a few basic levels set, you can turn down the monitoring volume a bit (if you find the average of 86dBspl too loud) and bring the other instruments into your mix. This is just a rough setting and you want to get an initial indication of how all the elements are going to 'play' with each other. If you already know where steophonically these instruments and additional voices will be placed, go ahead and pan them there. Just keep in mind that the Pan Law of the mixer (or DAW) can cause a 3dB increase or decrease of the sound source based on where it is in the stereo field. More on this later.

Before we get off the practical aspect of meters and level setting, let's remind ourselves one more time that humans are least sensitive to bass frequencies. While getting the average volume up to 86dB gets us into the most even range of the Fletcher Munson Curves, the low end is still not heard to humans as well. The issues is that meters don't know this and don't care. A -10dBfs sine wave at 20Hz registers -10dBfs and a -10dBfs sine wave at 4kHz will also register -10dBfs on the meter. I would hope you know which one your ears would find significantly louder than the other! Meters show all frequencies equally and are not complete analogs to the way that we hear sound. If your snare drum has a lot of information in the presence frequency range and is long enough in duration, it will be louder than a snare with less 2-7kHz but with more 180Hz.

In closing, using the meters of a DAW to start your mix project will give you an accurate and repeatable target for your levels. Setting the average volume of your snare drum at -30dBfs to start with and putting the average level for your entire initial mix at -20dBfs (all elements in) is a practice. Remember, don't use the channel or output DAW channels to hear the mix louder. Calibrate your monitoring system and turn up the monitoring volume knob when you want more overall level.

Relating Meters and Alignment Levels

The previously mentioned EBU and SMTPE analog to digital alignment level standards were issued decades ago when digital equipment started showing up in professional studios (SMPTE wasn't truly established until much later than the EBU specification). Since then, a loudness war has been raging which has been pushing up the average level of a song so that it 'plays louder' compared to the next song on the radio or shuffle iPod. While much of this is issue is more sociological and not audio engineering related, one problem is that there was no easy to read average level marking. In fact, there is NOTHING on a standard DAW peak meter to indicate a desired average level. At the bottom of the meter is –60dBfs or infinity and at the top is 0dBfs. There is no 'average' indication at all. For decades engineers had a 0 mark on professional PPMs or VUs as a reference to put their signal on, under, or over. On a digital meter, <u>all levels</u> read under 0dBfs and it gets a little confusing *visually* to gauge where your levels actually are. This is especially true as most DAWs will allow you to view the dB numbers on the meter in a sectional format (similar to most analog meters) or as an exponential curve where the numbers are more spaced out closer to the 0dBfs mark. The later gives greater resolution at the top but much less resolution at the bottom. So in this scenario, not only is there no clear indication for average, the overall look of the meter can change!



The two metering styles in Logic Pro. On the left, the sectional (linear) layout, and on the right the Exponential version. Both are showing a level of -19dBfs and sound the same volume. Visually, they are very different and can lead to confusion. This level in 24 bit is still over 20dB better than 0dB full scale in 16 bit.

In the above graphic, both are showing -20dBfs which is the average level for a wide dynamic mix. On the linear meter (left) it is about 50% up the meter, on the exponential version, it is only about 15% up. Big difference! No wonder one would push up the fader; -20dBfs appears like a unusually low level on that meter. On the classic VU meter, the 0dB mark was always in the same place; it was consistent and color coded. Note however that the standard channel meters in Logic Pro are peak meters exclusively and not supposed to be average meters like a VU.

The K-System

The audio recording and mastering engineer (and author) **Bob Katz** devised a digital metering system which brought back the visual ease of the PPM and VU meters and combined it with Dorrough-style simultaneous peak and RMS displays. This metering system was dubbed the **K-System**. The unique concept he added was to have three basic average level standards related to various program material: **K-20, K-14 and K-12**. The K-20 system means that the meter will display a digital level of -20dBfs as 0 on the meter (sometimes called 0 VU). This is the starting range of the average level of the signal and extends up to -16dB (a range of 4dB). This range shows as ‘in the yellow’ (the target average volume) with signals below that as ‘in the green’ and signals above that ‘in the red’. The **K-20** standard will mean that the average level of the music is -20dB below the maximum volume of the track. This gives 20dB above the average for peaks and transient information. The **K-14** gives the range from -14dBfs to -10dBfs as an ‘in the yellow’ indication of average signal level with 14dB above for peaks. Finally **K-12** has the average starting at -12dBfs with only 12dB above for peaks. The design and concept of this moving 0vu indication is that based on the style of music and the format of distribution, the average level can, will, and arguably <u>should</u> change. If your audience listens in a relatively quiet environment, a 20dB dynamic range will work great. However if the target audience is driving a car, or wearing earbuds on a subway, the average level of the music needs to be higher to compete with the external noises. For wide dynamic material like classical and ‘art’ popular music, the K-20 system will allow 20dB above the average level for peaks. The K-14 is for standard popular music allows for a 14dB dynamic range above average and finally the K-12 system designed for broadcasting and loud popular music mixes. Katz’s system isn’t something that one needs to buy or install; they are simply published guidelines to follow to mix by – or if one creates a metering plug-