Meters – what they tell you (and what they don't)

In 1999, the classic VU meter celebrated its 60th birthday. It served the industry well, and when properly interpreted, it's still a useful tool for indicating loudness.

We've previously addressed the 0 VU (nominal level) and peak (clip) points on the console meters, but there's a lot of useful information in between. Today's digital recording processes have brought us around to taking a hard look at the inadequacies of both the traditional VU meter and the peak meters for level management.

What's a VU?

The Volume Unit meter was originally designed to help broadcast engineers keep the overall program level consistent between speech and music. The classic VU meter has well-defined mechanical rise-and-fall response rates for the pointer, rates that are actually a little too fast to accurately represent perceived loudness but fast enough to show some movement between syllables. Basically, the VU meter was designed so the pointer movement looks like you think it should when responding to speech, making it easy to tell at a glance when something wasn't working. Engineers learned that brief excursions to the +3 dB top end of the meter scale rarely caused distortion, nor did these brief peaks result in an significant change in perceived loudness.

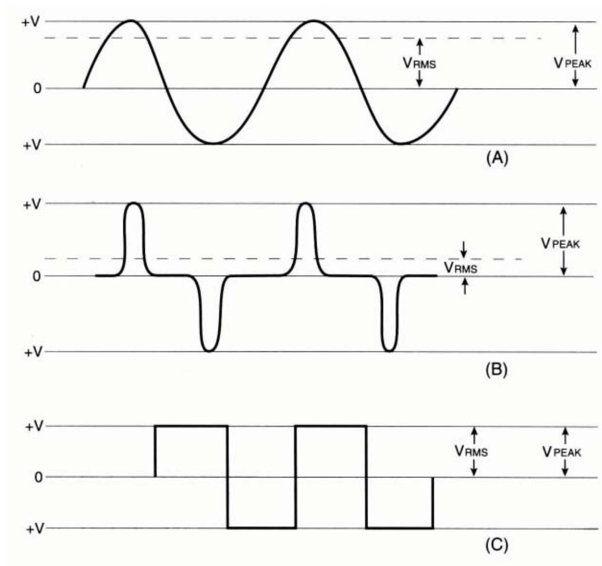
RMS vs. Peak Audio Levels

Once sound is converted to electricity, we can represent the audio level by measuring electrical voltage. The audio signal alternates rapidly between positive and negative values of voltage. A symmetrical signal such as a sine wave - a single-pitched note with no overtones or distortion - spends as much time on the positive side of 0 volts as the negative. The numerical average of the positive and negative voltages over many cycles is zero - not a very useful number when what we want to know is how loud a sound that voltage represents.

A method of measurement that corresponds fairly closely with how we perceive loudness is called the RMS (root mean square) average, a mathematical expression of the amount of energy in the waveform.

The RMS value is calculated by squaring the voltage at each point on the waveform (remember, the square of a negative number is a positive number), summing the squares, dividing by the number of points measured (the more points, the more accurate the calculation), then taking the square root of that quotient.

For a sine wave, the RMS voltage is 0.707 times the peak voltage. You've probably heard of this numerical relationship (it's one-half the square root of 2) between peak and RMS before. Understand, though, that it holds true only for a pure sine wave (waveform A in the following illustration).



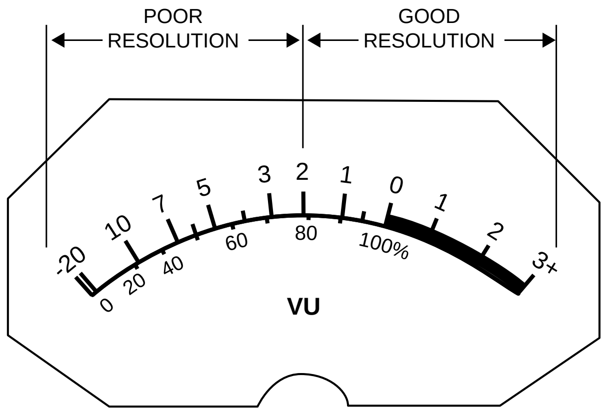
Look at the string of narrow pulses (Waveform B) in the figure. The voltage spends much more time at zero than at some positive or negative voltage. Even though the peak voltage of this pulse waveform is the same as the sine wave above, the RMS value of the pulse waveform is lower. It will indicate lower on a VU meter, and it sounds quieter than a sine wave with the same peak amplitude. This is why a drum may be at the correct volume in a mix but appear to read low on the VU meter.

Since a square wave (C) spends all its time either at peak positive or peak negative level, its RMS and peak amplitudes are equal.

The Classic VU meter

Today we use a VU meter primarily to indicate an impending overload condition, but that was never part of its original function. But is the VU meter a good headroom indicator? Only if you're working with program material that's already consistent in level and doesn't require a generous allowance for surprise peaks. Just look at the meter scale.

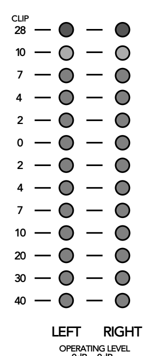
While the meter scale has a range of 23 dB, fully half of the scale (the top half) represents only 6 dB.



VU METER

This is excellent resolution when reading steady tones, but not very useful when working with a recording medium that's capable of handling a dynamic range of better than 90 dB. There's virtually no usable resolution below -10 VU, so if we take it on faith that we have at least 10 dB of headroom above 0 VU even though we can't see it on the meter, we have an indication of only about a 13 dB headroom range.

An LED meter such as you'll find on most Mackie mixers provides a wider measurement range – four calibrated points above 0 VU plus a Clip indicator. However, since you don't know what the signal level is when it's in the cracks between the LEDs, resolution isn't very good above +4 or below –10 VU.



Why compress such a wide range onto a scale with this little resolution? Practicality. Loudness is a logarithmic function. It takes more than twice the signal voltage for something to appear twice as loud, so the indications that you can see still represent loudness fairly well.

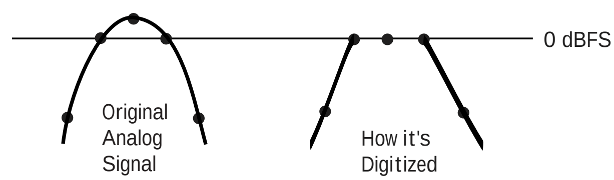
Digital Metering

In the analog world, we count on the designers to provide a reasonable amount of headroom above 0 VU. An analog tape recorder generally doesn't reach unacceptable levels of distortion until the level exceeds +10 VU.

In an analog system, we're not so concerned about how many dB above the nominal level we're running as long as the system gain structure is set properly so that everything peaks out at about the same point.

Digital metering is different, however, because there is no headroom above 0. The digitized value of the analog signal can never exceed the number represented by all the bits turned on in the analog-to-digital converter.

We don't call the 0 point on a digital meter 0 VU, it's called 0 dBFS (dB relative to full scale), and zero is all the way at the top of the meter, not in the middle like it is on an analog meter. Any input voltage that exceeds the calibrated 0 dBFS level will be digitized as 0 dBFS, leaving a nice flat-topped waveform looking quite unlike the original.



The digital level on the meter never exceeds 0 dBFS, but clearly there's something wrong here, and we need to be warned about it.

The conventional way of determining a potential digital overload condition is by counting the number of consecutive samples at 0 dBFS, and turning on the "Over" indicator when that number exceeds a preset threshold. A common standard for an "Over" indication in a 16-bit system is three consecutive full-scale samples. This is reasonable since we can assume that the first sample of a such a string occurred while the waveform was on the way up, and the last occurred on the way down, therefore it must have tried to go over the 0 dBFS level at some time in between.

Three consecutive 0 dBFS samples for an Over is pretty stringent. It could mean that only one sample tried to go over the limit (the ones on either side being correctly digitized to full scale). At 44.1 kHz, one sample is too brief to be detected audibly by most listeners on most program material. With 24-bit systems, where we can allow a little more headroom, many meters indicate a single full-scale sample as an "over." There's no standard, so you have to use your judgement and your ears, and if you're lucky, you'll find something in the manual about how "overs" are indicated..

Making the Meters Agree

It's nice, when the console's VU meter is reading 0, that the VU meter on the recorder connected to it also reads 0. Getting them to agree is akin to setting up the gain structure of a system.

In an analog system, it's usually as simple as setting the Record Level control on the recorder. With a steady tone set so that the console's meter reads 0, adjust the recorder's input level control until its meter reads 0 and you're there. (That is, assuming that your recorder has an input level control.) Today many don't, and they assume that you'll provide the recorder with the proper signal level. It's possible that you'll never be able to make the meters agree, but hopefully they'll not deviate by more than a few dB.

Calibrating Digital Meters

Since it's expected that you'll have frequent peaks above 0 VU on the console's meter, you don't want to set the recorder's input gain so that 0 VU on the analog console's meter corresponds to 0 (full scale) on the digital recorder's meter. If you do, you'll be running the recorder into digital clipping much of the time.

You do have one pair of points that can correspond, however – the clipping point on the console and the clipping point (0 dBFS) on the digital meter. If your recorder has an input gain control, you can set the console with a test tone so that it's just barely below clipping level, then adjust the recorder's input gain control so that its meter just hits 0 dBFS. That way, both the console and recorder will reach their clipping points together.

Generally recorders with fixed input gain are designed so that a signal at the recorder's nominal input level (usually +4 dBu or –10 dBV) gives a reading somewhere between –20 and –12 dBFS. If the recorder's nominal level is –20 dBFS, this means that it will reach 0 dBFS at an input level of +24 dBu. Since some Mackie consoles can put out +28 dBu before clipping, the recorder can max out before the console does. However, if you're feeding the recorder from an unbalanced mixer (+22 dBu maximum), you won't be able to reach peak level on the recorder before the console clips.

If you follow the console's meters and mix at around 0 VU (which is a good idea) you'll probably find that the recorder's meters run comfortably below full scale. The recorder manufacturer has allowed himself a generous amount of headroom so that it won't be his product that's clipping. Understand this, and don't worry about it. The meters are just a guide.

It's a popular belief that if a digital recorder's meters aren't hitting full scale all the time, you're losing resolution. This is a fact of life. A record level that never gets above -20 dBFS is probably too conservative, but if the level averages around –10 dBFS with peaks getting close to 0, that's just fine.

Computer Workstation Metering

Something to be aware of with the on-screen metering provided by digital audio workstation (DAW) programs is that redrawing the meter on the screen is usually a low priority task. Some DAW programs don't even bother to try – the meters only work when not actually recording, allowing you to set levels during a rehearsal, but not monitor the level during recording.

On-screen meters that function while recording often lag far enough behind when the A/D conversion takes place to allow you to get into trouble if you're working close to the limit. While it's difficult to drive a nominal +4 dBu digital input all the way to clipping from your Mackie mixer, many computer sound cards are designed to operate at a nominal input level of –10 dBV. If you connect a –10 dBV input to a +4 dBu output, you can easily drive it to clipping. Beware!

If your recording software provides record level controls, it's a good idea to find a setting experimentally that allows you to set the proper recording level by watching the mixer's meter, and then leaving the software record gain setting fixed at that point. Check your sound card's manual though. Some recommend setting the input gain to maximum since they attenuate following A/D conversion, causing a loss of resolution.

Hint: When using this method to calibrate your computer software to the analog mixer's meters, use live sources or some of your own recordings. Commercial CDs are usually heavily compressed and don't have peaks (which the digital meters will indicate) that exceed the average level (which the analog meters indicate) by more than a few dB. Real life, which is what you'll be recording, isn't so neatly packaged.

Practice Safe Level Setting

Consider keeping your peaks a few dB below 0 dBFS. It won't hurt the signal-to-noise ratio enough to worry about, and if your recording will be tweaked by a mastering engineer, it'll leave a little breathing room for digital level adjustment or equalization.

Be aware, when looking at the waveform display on a DAW recording, that anything lower than a couple of dB below full scale looks visually pretty feeble. It's just how they scale the graphics.

Just like you shouldn't worry if you aren't turning on all the meter lights, don't worry if your waveforms look like they're at too low a level. If it sounds OK, it IS OK.