DAW METERING: TIPS AND TRAPS

Where Clipping Really Occurs

When it comes to metering and headroom configuration in DAWs, things can really get complicated. RTW's experts provide more insight into potential issues.
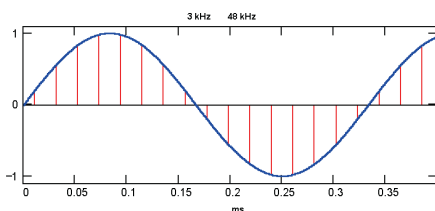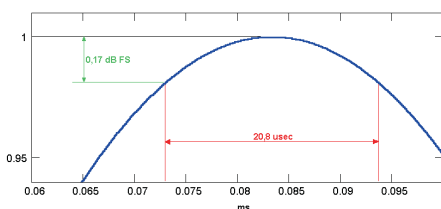

Figure 1a: Sampling a 3-KHz signal

Figure 1b: It is the sample rate that determines the width of a single sample—and thus the accuracy of the digital representation.
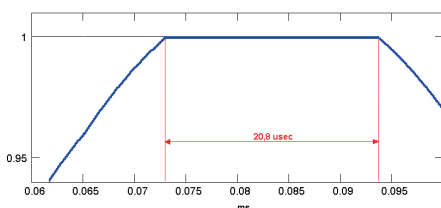

Figure 2: A normalized recording truncated at the top.

Since 2012, the audio-engineering departments of many TV broadcasters have gradually switched from Quasi Peak Program Metering (QPPM, a legacy standard from the analog era) to loudness normalization as per EBU R128 or other international recommendation. In addition to the European Broadcasting Union (EBU) as the umbrella organization of broadcasters in Europe, the International Telecommunication Union (ITU), the American Advanced Television Systems Committee (ATSC), the Congress of the United States who issued the Commercial Advertisement Loudness Mitigation (CALM) Act, and various other organizations deal with standardizing measuring systems and loudness adjustment.

What is more, loudness metering has been supplemented by a new unit of level: True Peak (dBTP).

Loudness

Unlike traditional peak metering using QPPM instruments that allow for evaluating audio only by its electric properties, loudness metering considers the subjective auditory sensation. The term "loudness" should not be confused with "volume" since volume refers to the sonic pressure. The purpose of loudness metering is to use metering techniques for eliminating abrupt changes in loudness between channels or programs (for example, from a feature to a commercial) rather than levelling the monitoring volumes of the users' TV sets

What it Means to the Users

Let's talk about the positive aspects first: There is a global standard that clearly defines the metering methods and is the ultimate basis for all loudness metering: ITU 1770/1771. And here comes the downer: The target and tolerance specifications depend on the broadcasting platform. Therefore, the German ARD has specifications that differ from the BBC's ones or from those Disney applies in the U.S. Therefore, each producer needs to adhere to the target-level specifications applicable to his or her employee before delivery.

Let's consider the specifications applicable to German TV broadcasters. First of all, these differ with regard to whether the material at hand is short (for example, a commercial or a trailer) or not. The target loudness of any material is −23 LUFS; however, additional criteria apply to short productions such as commercials and the like: In addition to the overall loudness, specifications require metering and evaluating the maximum momentary loudness (400-ms integration time) and the maximum short-term loudness (3-s integration time); these values must not exceed specific limits either.

LUFS (Loudness Unit relative to Full Scale) is an absolute unit representing the highest digital level in a 16-bit or 24-bit word; the corresponding relative unit is referred to as "LU". Therefore, similar to dB, LU always goes with the reference: 0 LU relative to −23 LUFS. In addition, some countries use LKFS (Loudness, K-weighted, relative to Full Scale). There is no difference between LUFS and LKFS—it's just another name, so −23 LUFS is the same as −23 LKFS.

We want to further explain the term "full scale" as it is relevant for the discussion of true peak, too: Since you just cannot achieve a higher level than with all bits set, the full-scale (FS) level cannot be exceeded. 24-bit provides more dyna-

mic range than 16-bit; this is because the analog signal is represented by one from a total of 224 discrete values (compared to 216 values). Expressed in dB (which is probably more intuitive for the majority of readers), this means that the theoretical signal-to-noise ratio of an n-bit system is appropriately $n \times 6.02 + 1.76$ dB. (For 16-bit, this is 98 dB; for 24-bit, it is 146 dB.)

Analog or Digital?

When performing conversions, many users are not sure about the relation between digital and analog level units. There is no single correct answer here. In the end of the day, each user individually selects the analog level that corresponds to 0 dBFS. The specification for German broadcasters is −9 dBFS = +6 dBu, meaning that the maximum analog level is +15 dBu. Therefore, if a digital level of −9 dBFS is applied to an analog PPM at the broadcasting center, the meter will show 0 dB due to its calibration (+6 dBu = 0 dB). This short discussion clearly shows that there is no such thing as a standardized analog reference; it becomes obvious why loudness metering has been specified for the digital domain only—because a clearly defined maximum exists only here: 0 dB Full Scale.

Dynamic Range is Key

A loudness target of l = −23 LUFS is specified for productions in Europe. With the 1000-Hz test frequency, the K-filter curve crosses the origin, so the filter is not active. This means that the following can be defined for that filter-curve point and a stereo signal: A loudness of −23 LUFS is achieved when each channel of a stereo signal comprises a 1000-Hz signal at a level of −23 dBFS (ITU 1170-3).

Now if you look at the overall dynamic range of a 24-bit signal at a level of 146 dB, setting the operating level to −23 dBFS (−23 LUFS/0 LU) is almost perfect.
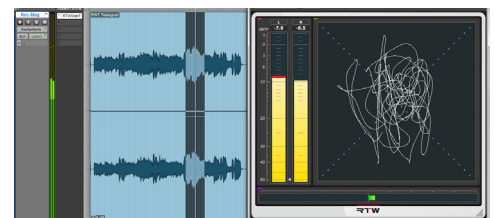

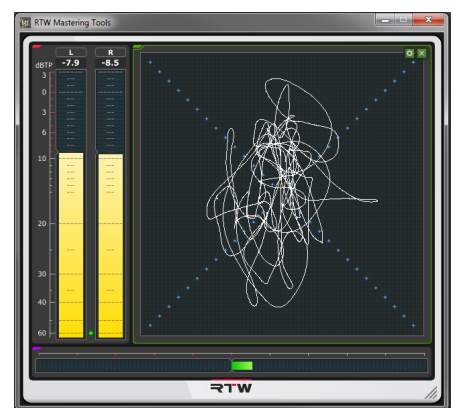Figure 3: Level-meter and vectorscope plug-in inside the DAW, "overdriven"

Figure 4: Level-meter and vectorscope plug-in outside the DAW, "overdriven"

There is sufficient headroom, and the difference to the lower threshold (typically defined by the noise floor) is large enough, too. So from a technical perspective, there is a large dynamic range available. This means that at this time, you can ignore any level limits while adjusting the level for target loudness.

True Peak

True-peak metering is part of the ITU recommendation, too. It replaces traditional QPPM metering and allows for higher precision in detecting digital clipping. The approach is particularly important for applications operating right below the technical clipping level (for example, in CD mastering) in order to prevent overloading D/A converters of CD, MP3, or other players. Similarly, specifications exist for any other type of production that a maximum level of −1 dBTP must not be exceeded.

Digital audio consists of equally long slices or samples, with their numerical values each representing actual signal voltages. At a sample rate of 44.1 kHz, the sample length is 22.68 μs, meaning that a new value is sampled every 22.68 μs. Between any two samples, however, an analog signal can achieve an infinite number of possible signal voltages at lengths inclining towards infinitely short. The shorter the signal, the higher the frequency ($f = 1/t$).

Digital audio typically has a finite level. This means that signals exceeding the full-scale level are just truncated, resulting in undesirable distortion. Therefore, normalizing the sample from figure 1 to full scale would result in the waveform depicted in figure 2. True-peak metering was introduced to avoid this. With true peak, the signal is sampled with at least four times the nominal sample rate—in our case, every 5.5 μs. As this still does not ensure safe recording, an upper limit of −1 dBTP was additionally specified.

Metering in the DAW

Normally, all of the above conditions don't apply to signals within DAWs. Almost all DAWs use floating-point arithmetic; simply put, this creates an almost unlimited range of values. In DAWs, the meters often show clipping that is neither audible nor visible in a vectorscope.

Notice the numeral shown above the meter in figure 3: It reads +10 dBFS, which is theoretically impossible in the digital domain! The vectorscope does not show any clipping—but the output signal is totally distorted. Figure 4 shows a measurement of the same signal in the digital stream using a stand-alone program external to the DAW. This one shows what we actually hear. So what's the reason of this phenomenon? It is actually the above-mentioned floating-point operation. Inside the DAW, the value range provided for the audio is unlimited; however, as soon as the DAW outputs the signal, the final 16 bit or 24-bit width applies.

Which means that DAW meters are far from being reliable for absolute output level reading and using a vectorscope for clipping checks is unsuitable. Things are different when using an external program for metering and generating vectorscopes from the output signal: As such a program applies the finite 24-bit range, it will show all artifacts any converter would produce while playing the signal.
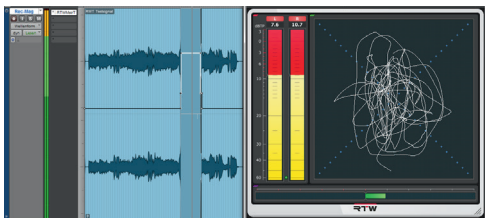
Michael Kahsnitz
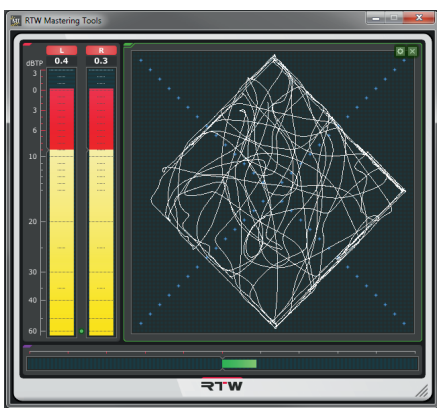

Figure 5: Level-meter and vectorscope plug-in inside the DAW

Figure 6: Level-meter and vectorscope plug-in outside the DAW