Good Practices for Mastering

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This document provides some useful information for preparing your files to our pre-mastering treatment.

1 Peak Level Regarding the Bit Definition

The number of bits determines the description resolution for each sample.
A 16 bits signal is described according to 65 536 (2 ^ 16) values. This allows a theoretical 96 dB signal-to-noise ratio.
A 24 bits, the signal is described next 16 777 216 (2 ^ 24) values. This allows a theoretical 144 dB signal-to-noise ratio.
The best analog preamplifiers exhibit a 129 dB signal-to-noise ratio. Since they are placed before any A/D converter, no captured signal can exceed this value even using a perfect 24 bits converter. This 129 dB value roughly corresponds to a 21 bits conversion.

The above signal-to-noise ratio values have some consequences about the optimum levels for each bit definition.
In 16 bits, the A/D conversion has inferior noise specifications than the upstream analog preamplier. So peaking between -6 and -3 dB is a reasonable compromise between signal-to-noise ratio preservation and hard clipping prevention when mixing/bouncing/rendering.
For 24 bits, the signal-to-noise ratio of the analog section determines the noise level. The peaking level can be as low as -18 dB without any signal-to-noise ratio degradation.

2 Loudness Management

Usual meters in the DAW are only peak meters. They are useful to display instant level values, but they don't reflect the perceived track level. The peak meters on the master section must be used to respect the suggested peak values in the previous section in order to prevent any clipping while preserving a correct signal-to-noise ratio.

Depending of the style of music, a loud level may be desirable. To achieve this goal, mixing is the key point. No compressor, limiter, EQ or clipper must be inserted on the master section. A straight master section is the best way to obtain great results when mastering. If needed, all previous FX can be inserted on different stems before the final summing of the master section. This way, their actions don't affect the complete mix, but a particular part of the mix which is far less dangerous for the overall dynamic and tonal balance of the file provided for mastering.

The files mastered with MaximalSound never reach the 0 dBFS maximal value to prevent any inter-sample clipping in the D/A converters during playback. This is also a good point for downstream lossy encoding.
If a post fader processing is used on the master section, exporting/rendering/bouncing/mixing down to a 32 bits float audio format, is also a safe way to avoid any possible clipping.

3 Track Relative Loudness

If the relative level between different tracks must be preserved, a single file containing all the pieces to be processed must be created.
When each track is processed individually, the analysis before processing doesn't take account of the different track levels. Then the optimization is set to the maximum for every track.

It is possible to mix the source sound with the sound processed by MaximalSound to reduce the final average level of a track.

- The first step is to delete the first 2353 samples from the processed file that correspond to the latency introduced by the processing.
- The second step is to import on two different tracks both source file and shortened processed file into your favorite audio editing software.
- In step three the source file will be normalized.
- Finally a new rendering will be calculated by mixing the two tracks according to the values provided by the table below in order never to exceed the limit of 0 dBFS.

Master dB	Source dB
0,00	-inf
-1,00	-19,27
-2,00	-13,74
-3,00	-10,69
-4,00	-8,66
-5,00	-7,18
-6,00	-6,04
-7,00	-5,14
-8,00	-4,41
-9,00	-3,81
-10,00	-3,30

4 Sample Rate Conversion

Using a high sampling rate (88,2 KHz or 96 KHz) is interesting to reach the best audio quality in the digital domain. Most mixing treatment of the DAW including plug-ins have better performances especially about aliasing which is an usual problem in the digital world. But the high definition formats are still rare for diffusion. 44,1 KHz and 16 bit is the current standard for music delivery. That means that a sampling rate conversion must be done. In order to preserve a high-definition production chain, the mastering has to be applied before down-sampling or bit conversion.
Note that the MaximalSound processing provides enough safety margin on the peak level to perform any sampling rate conversion without clipping.

5 Bit Depth Conversion and Dithering

When required, the bit depth conversion must be applied at the very last step. Dithering and noise shaping are better than a simple bit truncation, especially in very low levels encountered in classical and acoustical musics.