



The MaximalSound Algorithm

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This document provides a complete overview of the MaximalSound algorithm. It's organized according to the processing order.

1. Analyze
2. Harmonic Enhancement
3. Crossover
4. De-Expander
5. Limiter
6. Format Conversion
7. Links

1 Analyze

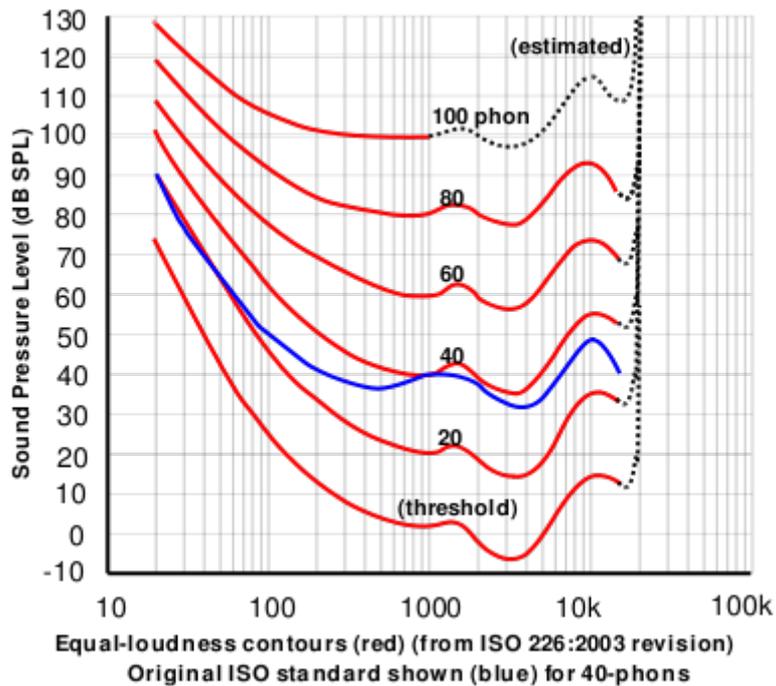
Because the MaximalSound service is a fully automated pre-mastering process, the first step of the treatment is analyze of the whole file. This allows to optimize settings of the downstream processing. Unfortunately, this also prevents to run the algorithm in real time.

2 Harmonic Enhancement

The harmonic enhancer, this old invention of the Aphex company, has the advantage on an equalizer to not apply a fixed amplification on the top of the audio spectrum. This technique allows to naturally clarify the sound without fatiguing constant high frequencies boost. In the MaximalSound algorithm the even and odd harmonics are generated separately depending on whether the attack or the body of the note are processed. This adds some vacuum tube character to your sound. Note also that all lossy audio formats such as mp3 always benefit from such harmonic enhancement.

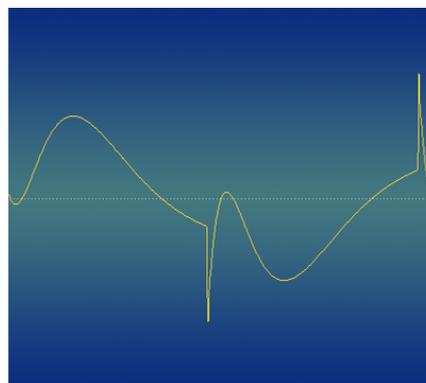
3 Crossover

A 32 bands crossover filter splits the signal to be processed by 32 de-expanders before being rebuilt in broad-band by summing all the bands individually processed. This produces a dynamic equalizer where each part of the audio spectrum is magnified in the low levels according to a psycho acoustic model standard.



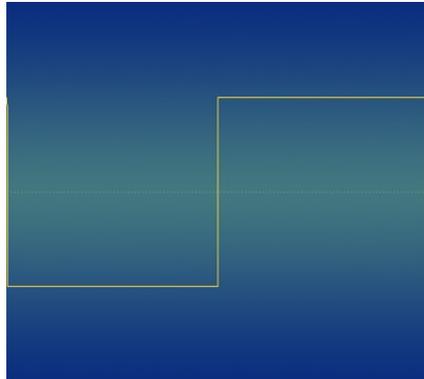
Because this model was established using pure and constant tones, an adaptation has been done to make it suitable for the variable complex timbers of music. This particular adjustment is purely empirical. It only relies on thousands hours of attentive listening.

Despite an IIR (infinite impulse response) filter solution has been chosen, the maximal possible phase error is reduced to a 5° margin on the complete processing. This provides a perfect reconstruction of a square wave, by preserving the time relation between frequencies on the whole audio spectrum. The FIR (finite impulse response) filter topology has the theoretical advantage to feature both linear phase and constant group delay across the whole audio bandwidth. But it has been rejected after listening tests, due to harsh artifacts when the de-expander treatment was applied.

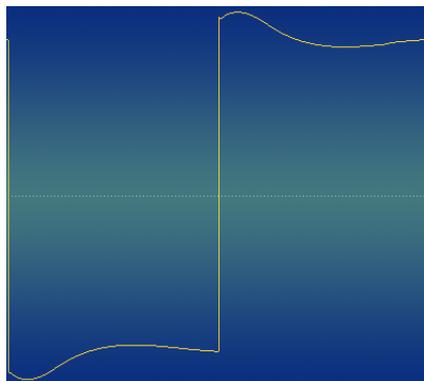


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The previous picture displays a square signal reconstructed after passing thru a 3 poles IIR crossover. The peak and RMS values of the original square signal are drastically modified. The human hearing isn't very sensitive to these changes, but all downstream processors will have to deal with these changes.



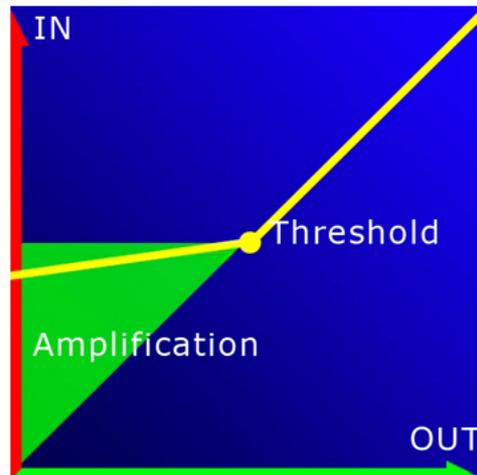
The picture above displays a square signal reconstructed after passing thru the MaximalSound 32 bands crossover (no other treatment applied). The downstream processors won't be affected by any peak and RMS values modification. This point is very important when dealing with dynamics.



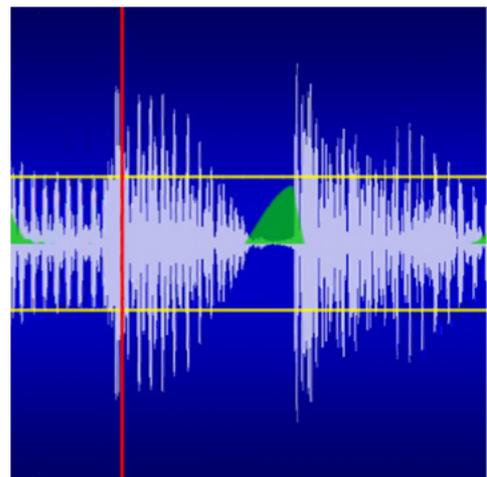
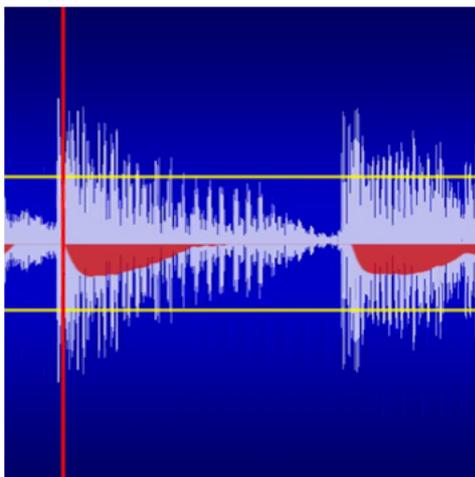
This is a picture of a full scale square signal passing thru the complete MaximalSound processing (harmonic enhancer, crossover, multi-band de-expander, limiter).

4 De-Expander

A de-expander can also be called reverse-expander (negative ratio) or upward compressor. A de-expander amplifies the signal below a threshold in contrast to an expander which attenuates the signal below this threshold. This choice is fundamentally different than techniques related to compression. The signal is amplified when it is weak and not planed when it becomes too high (above threshold).



The difference may seem subtle, but it is significant when the attack time of treatment is taken into account . A compressor will act always too late (attack time) on the peak values, forcing the limiter to affect the transients which are so valuable to the perception of space. The de-expander just amplifies signal after the attack time, avoiding all unnecessary limiting.



This technique can't be compared with parallel compression since two different parts of the signal are processed. Note that no "look-ahead" is applied. This technique is very effective to prevent any threshold violation and produce a very smooth sound, but it usually kills the sound impact, and it may become very unnatural for ears.

5 Limiter

Unlike the analog domain, digital doesn't support any overload, even briefly. As a consequence, our limiter design is focused on the audio fidelity, ignoring bad DAC which can't stand few dB of inter sample peak (ISP) in the analog domain. The limiter has to handle situations where the summation of all processed bands produces a signal exceeding the allowed maximum value (0 dBFS) in the digital domain. The limiter is an essential part of treatment since at this stage the signal is full-band and any defect may lead to tonal unbalance by favoring or attenuating certain parts of the audio spectrum. In limiters the attack time is zero or negative (look ahead) to prevent any violation of the limit value. The art is to find an optimum release time, in order to preserve the frequency content and the perception of the dynamics of the original signal regardless of the complexity of this signal.

6 Format Conversions

- Mono or multi-channels are converted to stereo
- Sampling frequencies inferior to 44100 Hz are converted to 44100 Hz
- Sampling frequencies superior to 96000 Hz are converted to 96000 Hz
- Lossy **mp3, ogg, mpga, wma, mp4a** audio formats come back as mp3 320kbps, and wav (CD - 16 bits @ 44100Hz)
 - Lossless audio codecs are best suited for mastering
- **aif, flac, wav**. Lossless audio formats come back as mp3 320kbps, wav (CD - 16 bits @ 44100 Hz), high definition (HD - 24 bits @ 96000 Hz), and "same format" if different from the 3 restitution formats mentioned before
 - Note that a 8 channels 32 bits @ 192 KHz file comes back in stereo 32 bits @ 96 KHz format even if labeled "same format"
 - Note that the wav file format (uncompressed data) must be used if non-standard sample rates are required (e.g. 47952 Hz, 48048 Hz)

7 Links

Video

- [MaximalSound At Work](#)
- [The Phase Matters](#)
- [Multiband De-expander](#)
- [Compressor vs De-expander](#)
- [Dynamic Processors by Transfer Curves](#)
- [Digital Overload](#)
- [All Limiters aren't Equal](#)

Paper

- [Good Practices For Mastering](#)