Muleteer Math — About

Dear audio enthusiast,

I'm glad to introduce the Muleteer DSP — a rapidity maintainer module. This DSP designed to compensate transducer excessive mass, thus freeing the system from associated non-linear distortions. In a first place, it's a speaker's moving subsystem, which is usually sum of diaphragm and a voice coil masses. Much of this overweight effectively fought by negative feedback of amplifier's voltage source. Still unclear why engineers used linear devices to control load with evidently reactive character. Thru the years these devices took impressive gain in linearity, shamefully this wasn't helped to solve even half of a task. Clearly, this approach is incapable to solve this problem completely, since perfect solution lays through indefinitely short feedback roundtrip and infinitely fast amplifier slew rate.

Muleteer DSP takes the challenge to supply right amounts of energy just at the right instants of time, overcoming critical drawbacks of modern solutions. Thanks, it has all the information about ongoing evolutions. Joining this data with persistent mass of transducers, demanding governance, sports a nice opportunity to counteract with reactive system by absolutely complementary organized value, avoiding employment any exceptional steps. So, at least mathematically, mentioned problem, gets a perfect solution and you — (at least theoretically) perfect implementation.

Download: foo_dsp_muleteer.zip

The final notes:

1. Activate "Rapidity Maintainer" from available DSP processors list.

2. Guess your system overweight. The modern systems have this value about 0.5-1.5. So, try to stay close with these values. Only happy owners of valve based systems can go beyond 2.0.

3. Since Muleteer operates on a quite finer level than one considered "enough" by engineers failed to use Claude Shannon

findings[1], it's highly recommended (as essential condition of accurate work) to up-sample standard definition material to at least a double rate. Simply insert resampler to the DSP chain before Muleteer's module. Try to avoid any kind of dithering features, such as noise shaping and e.t.c., especially after the muleteer's work, since it's outcome, strictly, is not PCM anymore.

4. Fine-tune module to desired withstand value.
5. Enjoy.

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1 Shannon-Kotelnikov sampling theorem states: «Band-limited analog signal can be perfectly reconstructed from an infinite sequence of samples if the sampling rate exceeds 2B samples per second». Note that critical condition: sequence of samples should has INFINITE duration. This statement backed by very simple example. Let sine wave with frequency F at 2F sample-rate have [1,-1,1,-1,1,-1,1] sequence. So, if we sample this sine phase-shifted by pi()/4 radian, we'll get [0.7,-0.7,0.7,-0.7,0.7,-0.7,0.7]. But we'll get same results if we sample previous sine attenuated by 0.7. This evidence demonstrates impossibility to discriminate amplitude and phase at frequency F on live (finite) signal. For Compact Disc this means unavoidable decoding errors, since nobody nor want, nor able play by infinite notes. On the other hand if sample-rate equals 4F, sampled signal can be reconstructed by as little as two samples (equivalent to the only sample at 2F), since first derivative can be calculated as per 2F sample with perfect alignment. This way, on a 2F sample-rate, bandwidth 0F to 1/2F is error free. And bandwidth 1/2F to F has harmonic error rate: E'f(t) = f/Fs*t if i doin' math right… ;-)