**Abstract**

The characteristics of analogue audio processors are useful and valued for musical applications. A function has been implemented that allows the user to easily and intuitively introduce and control non linear behaviours and produce an output signal that has some of the characteristics that are valued in the analogue processing domain.

**Proposal Brief**

Outlined proposal for an investment opportunity: an audio effect that can be used to shape the tone of a signal through the use of a high and low shelf digital equalization (eq). Further processing of the signal can be applied to increase the harmonic content of the signal through soft clipping. The proposed design could be incorporated as a stand alone DAW tone shaping plugin or as a saturation module in a digital synthesizer.

**Problem Description**

Using analogue equalization hardware to process a signal can be prohibitively expensive. Units take up valuable floor space in the recording studio not to mention if using several instances at the same time, multiple units need to be purchased. However, some of the associated characteristics of analogue hardware (by-products of circuit distortion, and non linear behaviour) such as warmth, colour and richness, are assets that are prized in the studio and as such hardware equalization units are still designed, purchased and highly favoured. (Robjohns, 2010)

The precise nature of DSP, whilst extremely accurate, is not always the desired tool when processing audio for musical applications. In fact, these fluctuations in accuracy are what many engineers credit as giving a sound its own unique timbre. (McIntyre, 2015)

Using a digital function that incorporates both linear and non-linear modules is one way to emulate and have greater control over the irregularities that can occur when using analogue hardware processing. Implementation of the proposed function could allow the user to shape the tone of their signal through the use of a high and low shelf eq. This function has been designed to be implemented as a broadband frequency tool as opposed to a notch or narrow band eq. The introduction of a distortion module that is based on a soft clipping function can then be used as harmonic enricher, i.e. as the signal gets driven harder, more overtones will be produced. Normalization is included in the processing to compensate for extreme amplitude variations resulting from high magnitude gain adjustments.

**Proposal brief:**

- Allow for broadband manipulation of frequency content
- Enrichment of harmonic content through controllable distortion
- Simple and intuitive parameters for ease of use
**Specification**

The application of the audio effect works to:

- Apply positive or negative gain change to a band above or below a specified cut-off frequency.
- Reduce or increase gain ratio through a variable slope parameter.
- Apply a factor of broadband exponential distortion.
- Blend between processed (wet) and unprocessed (dry) distorted signal.

**Implementation**

An infinite impulse response filter design has been implemented for this effect due to the less intensive processing power required. Finite impulse response filters can be used to approximate analogue style filters, however, the filter order needs to be very high to effectively reproduce the desired form. Due to the immense number of calculations required, computational power and processing time can be significant (Reis & McPherson, p. 69). The faster processing time offered by the IIR filter makes it suitable for applications (such as a synthesizer module) where the processing has to occur with no discernable latency.

The Matlab function `shelving.m` [Tacket, 2005, func.] implements the filter and produces an output by processing the input signal by way of adding or subtracting a filtered version of the signal with itself. The gain control is used to reduce or increase the gain of the selected frequency bandwidth around a cut-off frequency whilst a Q parameter is used to widen or narrow the transition bandwidth between affected and unaffected frequencies.

The following syntax is used for the `eq` function:

\[
[b, a] = shelving(G, fc, fs, Q, \ldots \text{type})
\]

Fig. Block diagram for shelving eq. [Reiss & Macpherson, p. 83]

The following transfer function (Dutilleux & Zolzer, p. 51) describes the implementation of the filter:

\[
H(z) = \left(1 + \frac{H_0}{2} \pm A(z)\right)
\]

where \(A(z)\) is an all pass filter which passes all frequencies but modifies phase:

\[
A(z) = \frac{z^{-1} + \frac{B}{C}}{1 + \frac{B}{C}z^{-1}}
\]

leading to the algorithm:

\[
y(n) = \frac{H_0}{2} \left(x(n) \pm y_1(n)\right) + x(n)
\]
Analogue systems have a limit to the magnitude of signal they can process. Once a signal exceeds this limit, clipping of the waveform will occur meaning that any further increase at the input will not produce a further increase in the output. (Reiss & Macpherson, p. 169) Clipping can be characterized as hard or soft, with soft being generally the more desired tone for musicians. Hard clipping, especially in the digital domain, does have its uses however it is predominantly regarded as sounding too harsh. Smooth clipping is regarded as being complimentary to musical audio and is used to add warmth, colour or grit to a signal. (Waller, 1997)

To achieve soft clipping in the digital domain, a smooth transition to the clipping level is required. For this function the output approaches clipping limit asymptotically the more the input is increased, however it never reaches it. This means that the distortion that is added to the original signal will increase gradually giving an output signal that has the musically desired characteristics. (Esquida & Valimaki, 2015)

The distortion module is implemented by an exponential function $\text{expdist.m}$ [Bendiken, Dutilleux, Zolzer and Dempwolf, 2002, func.] which takes a sum of the input signal sinusoids, applies amplification (gain) which creates a non linear series of harmonics, then combines them with the original signal via the mix control. The output signal can be described (Dutilleux & Zolzer, p. 93) as a sum of sinusoids:

$$y(n) = A_0 + A_1 \sin(2\pi f_1 T_n) + A_2 \sin(2 \cdot 2\pi f_1 T_n) + \ldots + A_N \sin(N \cdot 2\pi f_1 T_n)$$

if the input signal is a sinusoid of known amplitude and frequency according to

$$x(n) = A \sin(2\pi f_1 T n)$$

The following syntax is used for the distortion function:

$$[yb\_dist] = \text{expdist}(x\_1, \text{gain}, \ldots \text{mix})$$
Evaluation
The shelving eq produces a useable and interesting array of tonally sculpted outputs. An analysis of the frequency domain of the output signals reveal a well executed implementation with changes in gain, slope and centre frequency producing the desired spectral outcome.

Fig. Wave file (Drumbeat) with +5 gain high shelf at 10kHz

Fig. Wave file (Drumbeat) with +8 gain low shelf at 700Hz

Fig. Red: Wave file (Drumbeat) dry, Green: Wave file (Drumbeat) with low shelf and +4 Gain distortion

The distortion module applies a non linear function to the input signal in the time domain. It is however, the artefacts that occur in the frequency domain (as a product of non-linearity) that can be heard and best evaluated in a musical sense. (Reiss & Macpherson, p. 186) The output signal has increased frequency content that is applicable for the processing of a variety of musical signals. The change in timbre could be described as a smearing of the frequency spectrum that increases or decreases in intensity dependent on gain and mix values. Musicians could incorporate it in to the signal path to enrich synthesizer tones, add ‘grit’ to synthetic drums or as an acoustic instrument processor.

Method of Human Assessment
A hardware unit with a similar function to the digital implementation could be used as potential comparison source. A unit such as the TLA EQ2 valve eq and saturator could be used to process a variety of test signals, musical phrases, synth patches, drum loops and sound effects and then compared against the digitally processed versions of the same files.
The performance of the function should also be judged purely as a unique sound processor, i.e. does it adjust the timbre of a signal in a way that is pleasing, usable and interesting to a musician? A sequence of wave files could be processed at a variety of parameter settings and then subjectively judged and rated by the listener against the dry versions.

It is also worth noting that audio effects can take some time to reveal their true colours and characteristics, perhaps detailed subjective analysis could involve several musicians and producers having access to a prototype for a specified amount of time, for example 30 days. They could report on how well the effect was integrated in to their particular method of recording or playing and if the digital emulation was as valuable as analogue counterparts.

Included with this report are example wave files that demonstrate various shelf eq and distortion processing. The files names are interpreted as ‘Plus 6_600Hz_dist’: a positive gain change of 6 at 600Hz with distortion applied. The original dry wave files are also included.

**Conclusion**

The proposed audio effect meets the requirements as outlined in the problem description. The controls allow for combinations of minimal parameters to generate a wide variety of sonic manipulations. Low and high frequency shelves can be increased or reduced to provide broadband frequency spectrum changes. The output of the eq can be processed by the distortion function in order to increase the harmonic content of the signal. The simplicity of the parameters means the proposed unit would be optimized for enhancing (sweetening) a signal, equivalent to using large brush strokes but with greater finesse.

**References**


Dutilleux and Zolzer, (2002)


Robjohns, H., 2010. Analogue warmth, the sound of tubes, tape and transformers. [Article] Sound on Sound Magazine, February 2010
