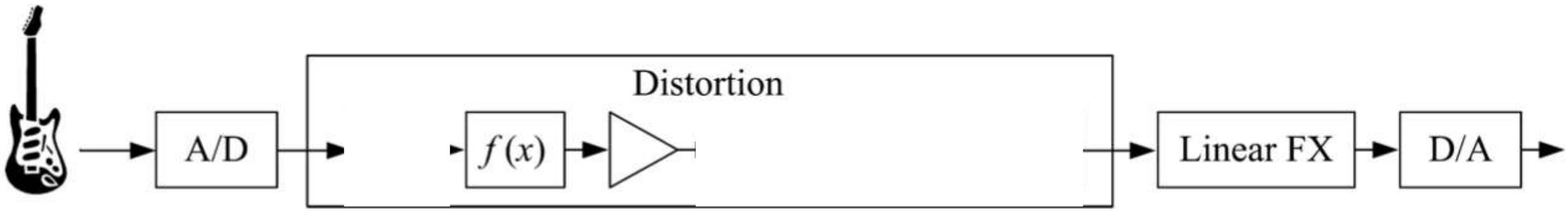
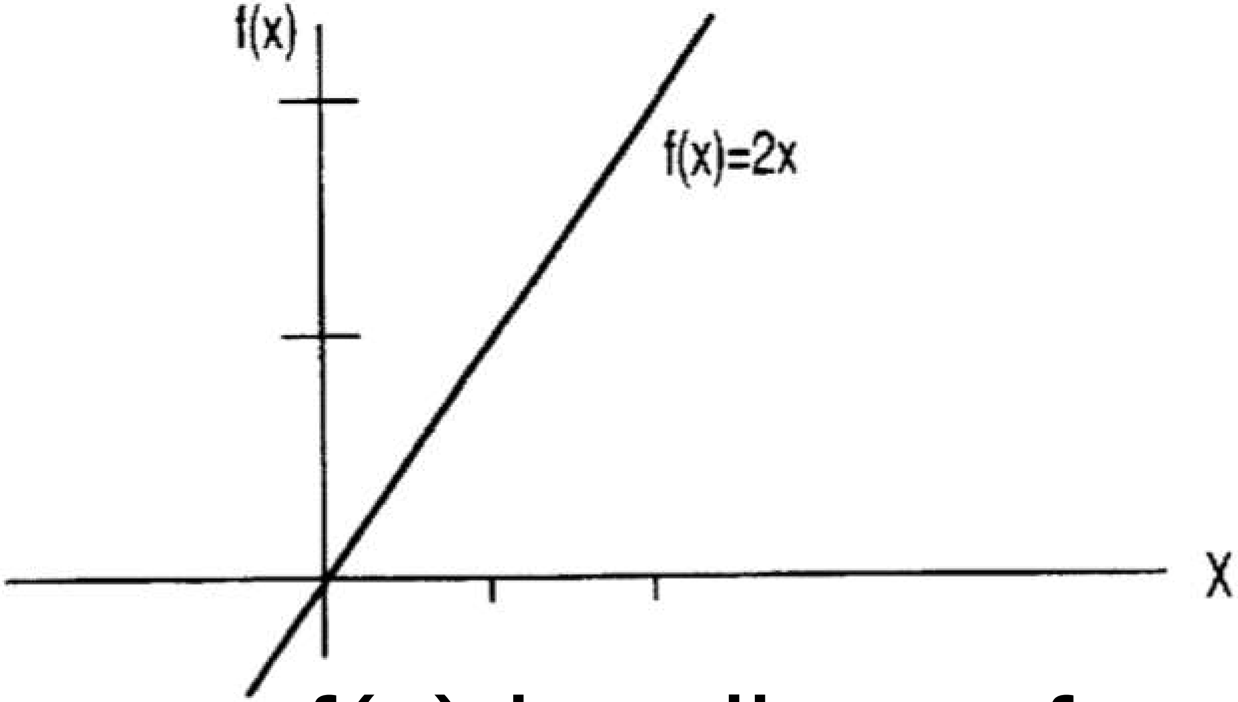
Distortion at large



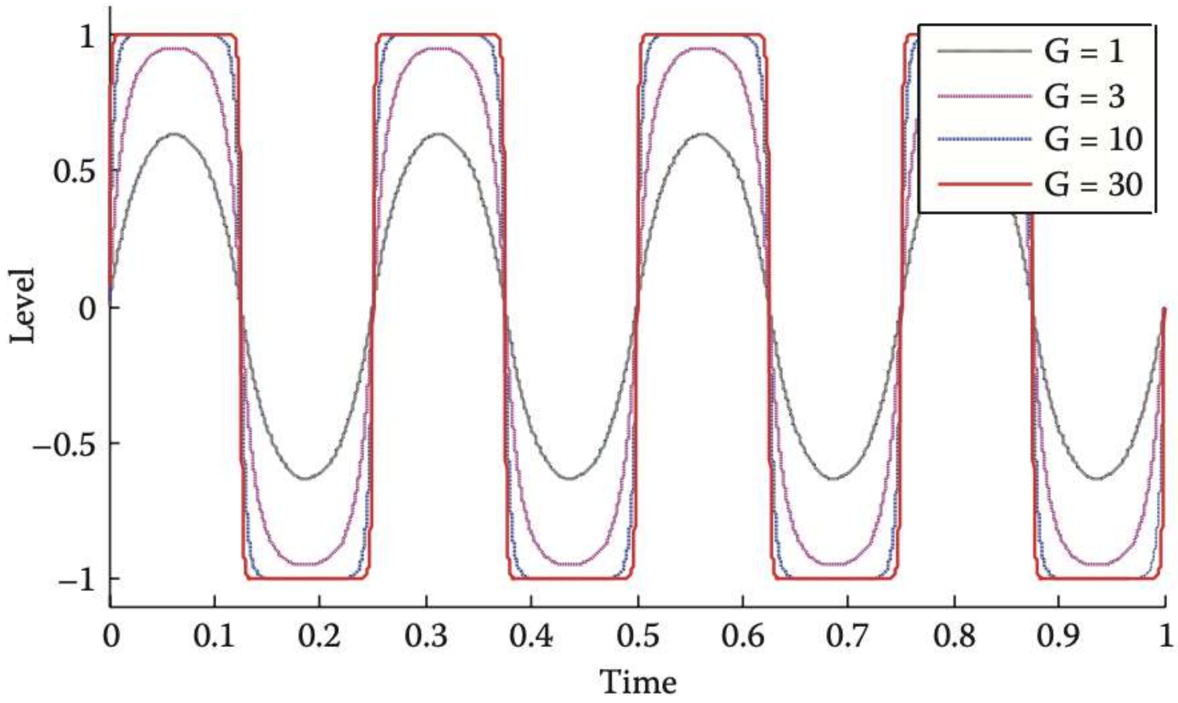
f(x) is called the transfert function

f(x)=Gx where G is Gain



In this case, f(x) is a linear function

f(x)=Gx



A saturation / clip occurs...

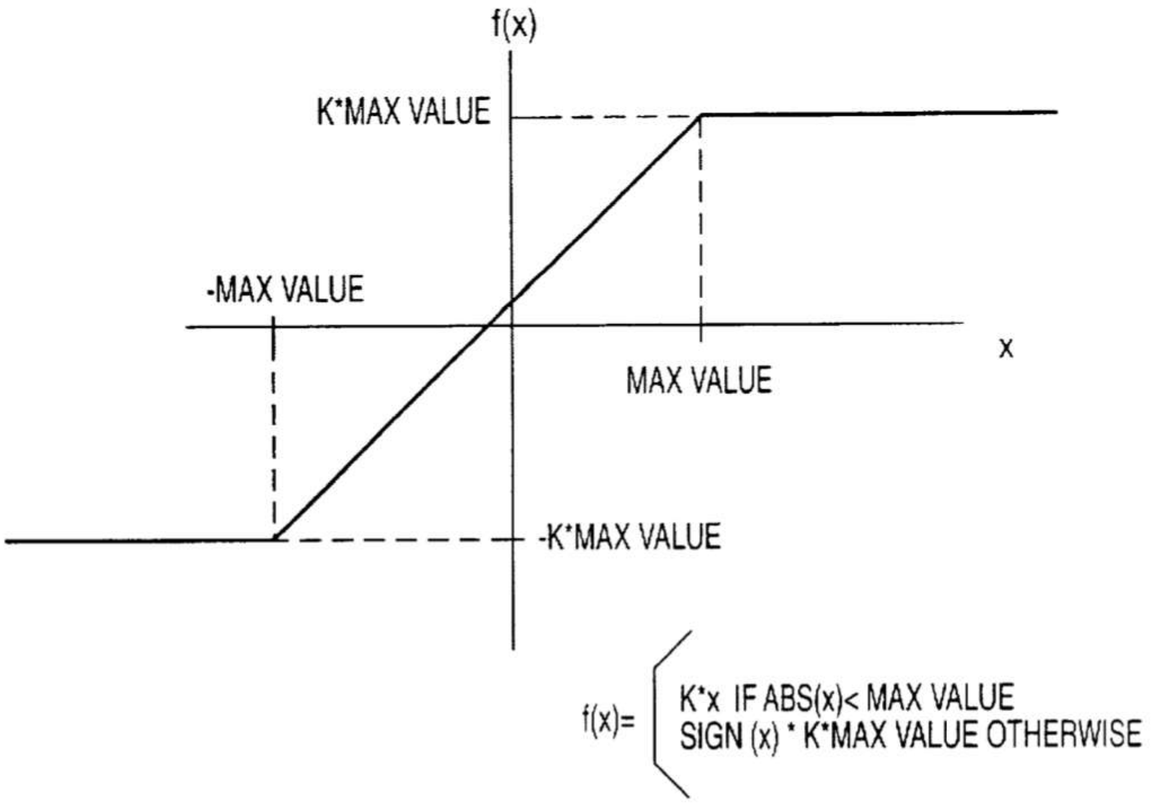
f(x)=x

A saturation is called clipping if it is defined like:

$$f(x) = \begin{cases} a & for & x \leq a \\ x & for & a < x < b \\ b & for & x \geq b \end{cases}$$

so, it is perfectly linear in the range $[a, b]$, and clips (cuts) the signal if it is out of the range.

Simple clip prediction...



Simple clip prediction...

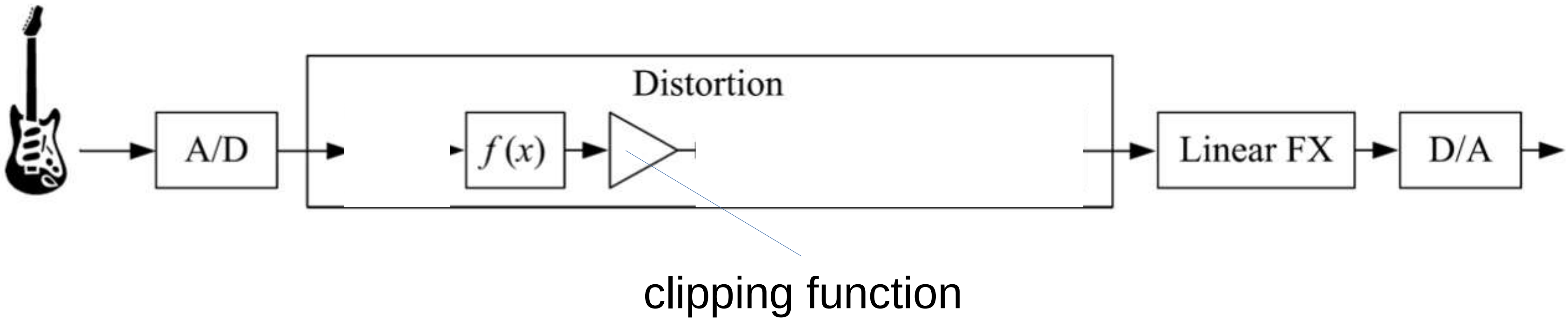
Other clipping functions...

$$Output = Alpha1 \times f(x), \text{ if } x > 0$$
$$Output = Alpha2 \times f(x), \text{ if } x \leqslant 0$$
$$x = Input \times \left(\frac{1}{Alpha1}\right), \text{ if } Input > 0$$
$$x = Input \times \left(\frac{1}{Alpha2}\right), \text{ if } Input \leqslant 0$$
$$f(x) = -\frac{2}{3}, x \leqslant -1;\quad x - \frac{x^3}{3}, -1 < x < 1;\quad \frac{2}{3}, x \geqslant 1$$

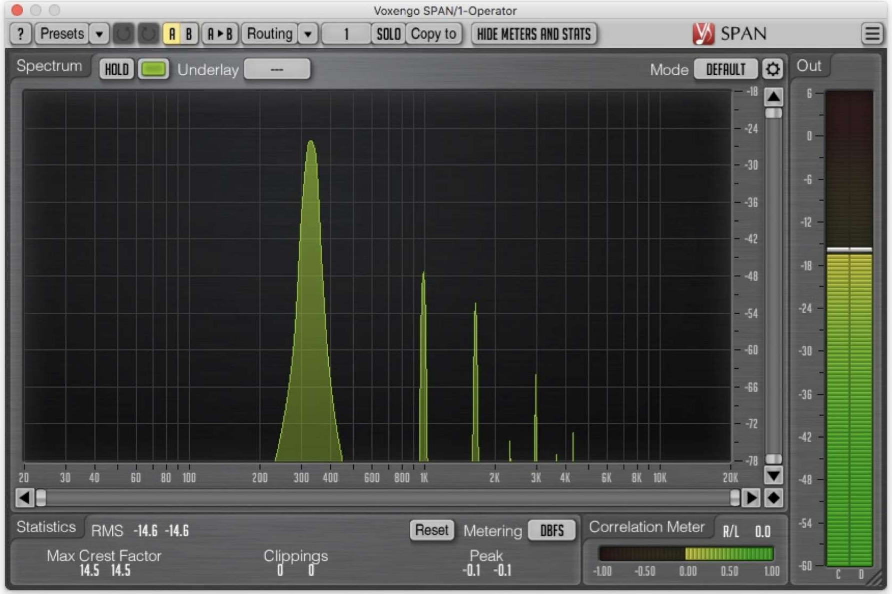
$$\text{smootherstep}(x) = S_2(x) = \begin{cases} 0 & x \le 0 \\ 6x^5 - 15x^4 + 10x^3 & 0 \le x \le 1 \\ 1 & 1 \le x \end{cases}$$

$$\begin{aligned}
\mathrm{S}_0(x) &= x, \\
\mathrm{S}_1(x) &= -2x^3 + 3x^2, \\
\mathrm{S}_2(x) &= 6x^5 - 15x^4 + 10x^3, \\
\mathrm{S}_3(x) &= -20x^7 + 70x^6 - 84x^5 + 35x^4, \\
\mathrm{S}_4(x) &= 70x^9 - 315x^8 + 540x^7 - 420x^6 + 126x^5, \\
\mathrm{S}_5(x) &= -252x^{11} + 1386x^{10} - 3080x^9 + 3465x^8 - 1980x^7 + 462x^6, \\
\mathrm{S}_6(x) &= 924x^{13} - 6006x^{12} + 16380x^{11} - 24024x^{10} + 20020x^9 - 9009x^8 + 1716x^7.
\end{aligned}$$

Distortion at large

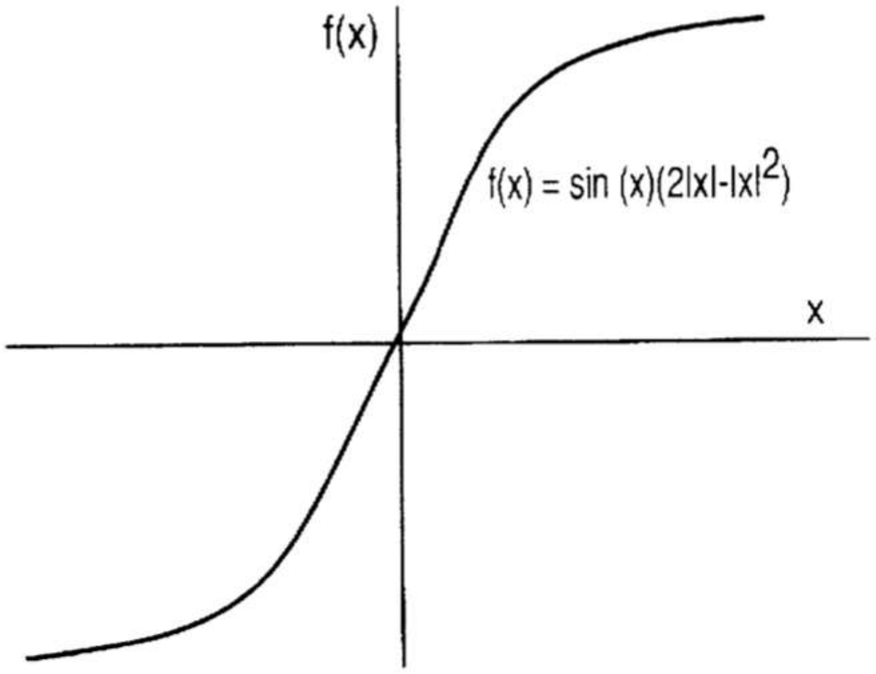


A famous simple « disto » :-)

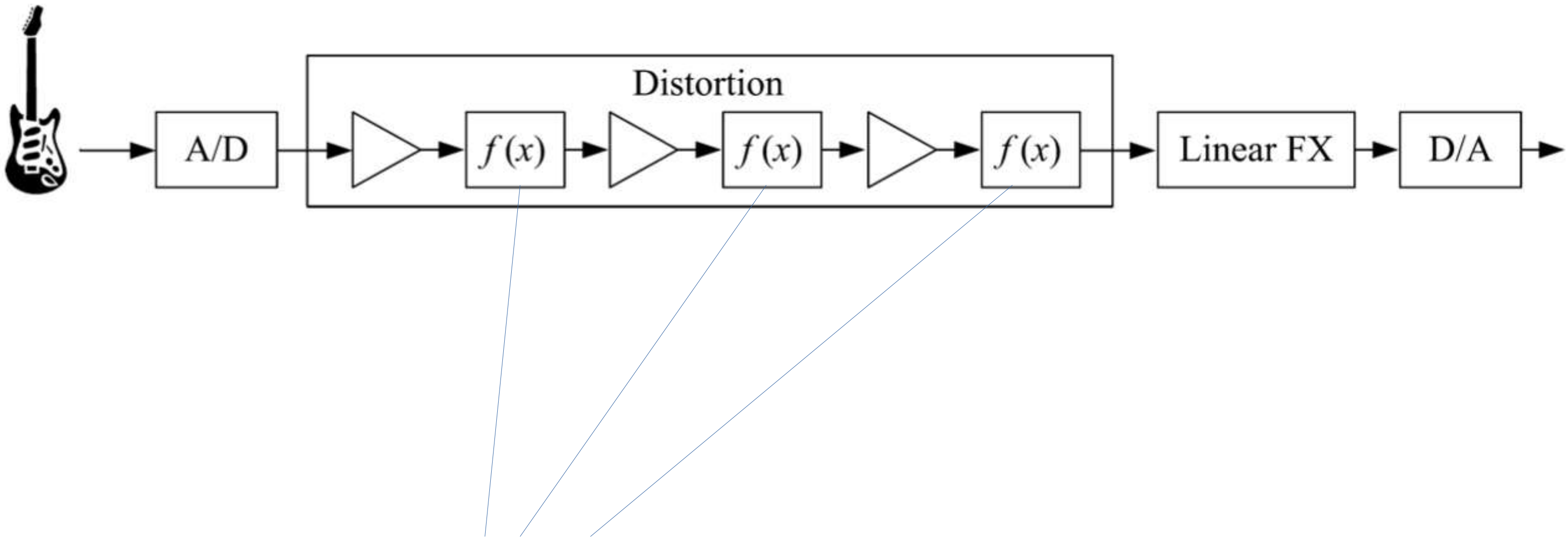


$f(x) = \tanh(x)$

More advanced disto :-)



More advanced disto :-)



$$f(x) = (|2x| - x^2)\ \mathrm{sign}(x)$$

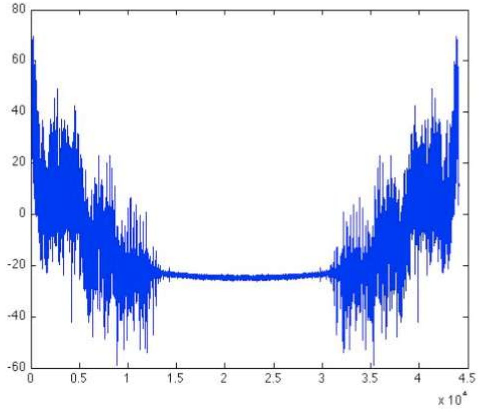


Figure 3: FFT of the clean input signal

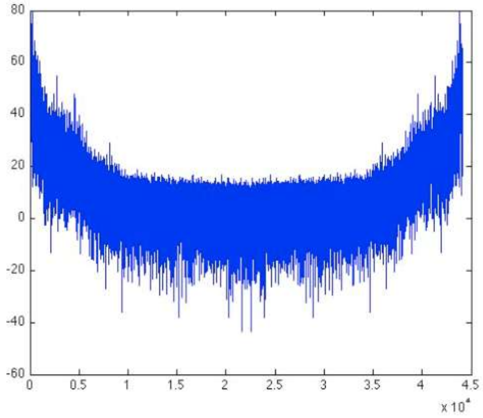
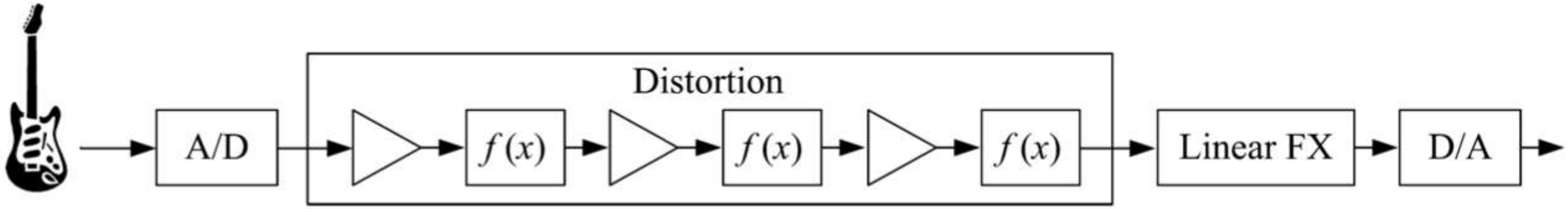


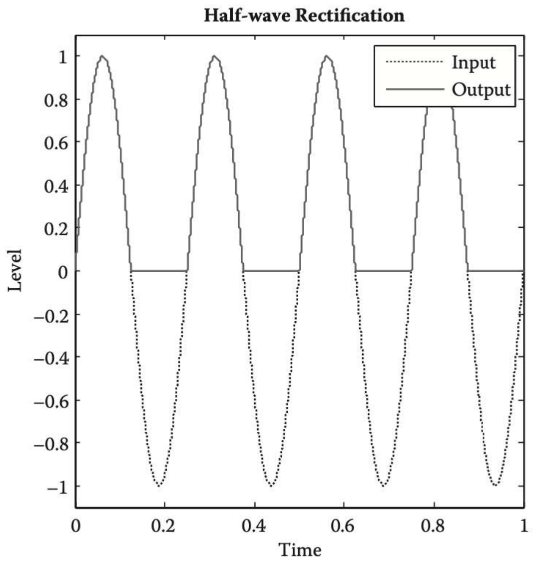
Figure 4: FFT of the distorted Signal

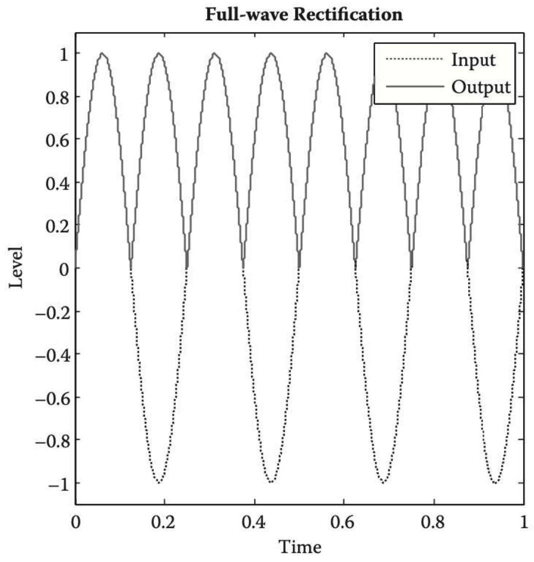
Tube Vacuum !



Other distortion techniques…

Half Wave Clipper





But f(x) is not linear ????

$$f(x) = (|2x| - x^2)\ \mathrm{sign}(x)$$

Sound unwanted artefacts occurs !!!

But f(x) is not linear ????

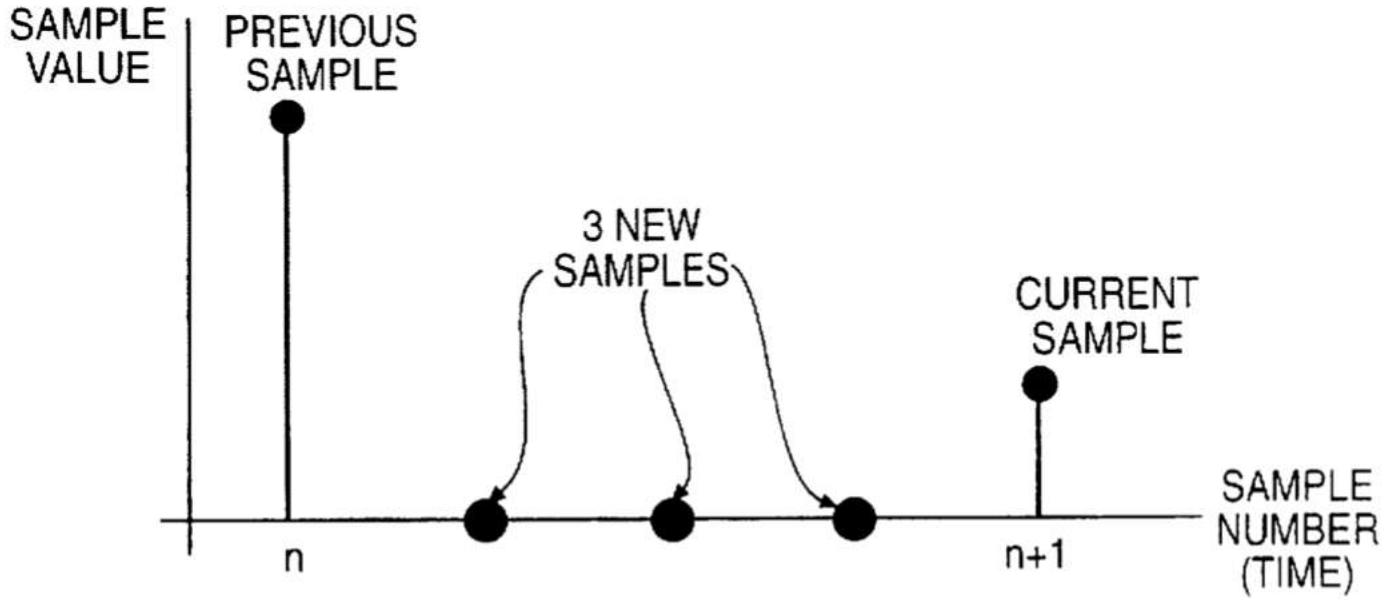
$$f(x) = (|2x| - x^2)\ \mathrm{sign}(x)$$

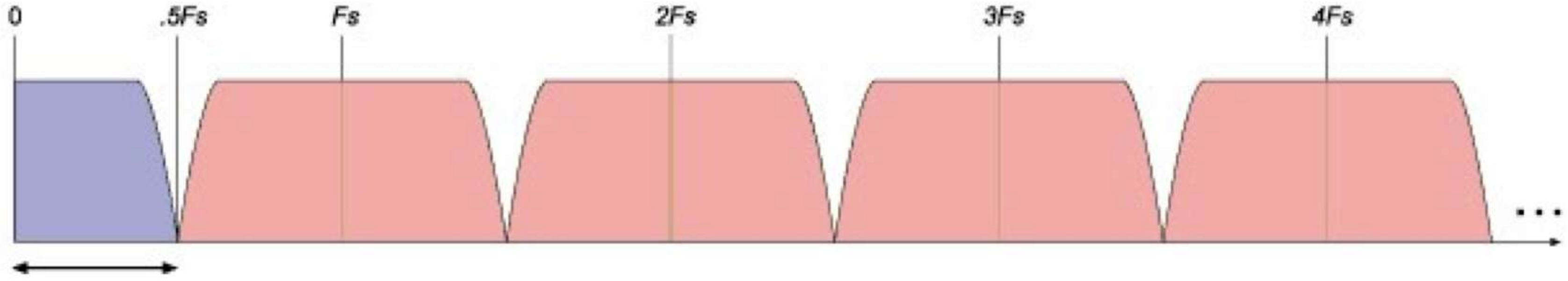
Sound unwanted artefacts occurs !!!

Due to the fact that we don’t have enough data to process correctly figures (the buffer is too small, but that’s not the only reason).

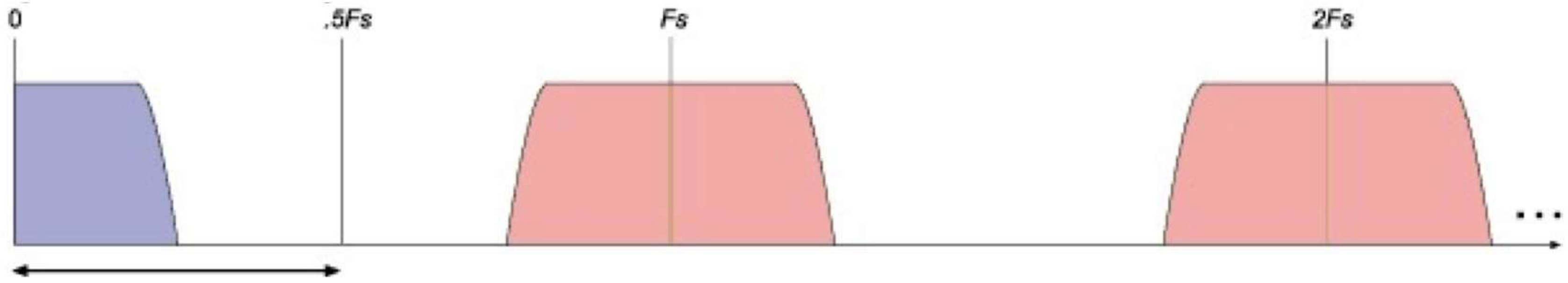
The non linear function creates VALUES that will be consider as « overflow » by the computer (I do it simple), the overflow data will be interpreted as « low » values adding « noize » at low frequency … We need to increase the buffer size but also to rearrange the non-linear function to gather together real numbers and gather in another zone zeros or very low values (which can be consider as bad noize).

We need to create a buffer with more DATA





The goal of oversampling is to give that purple region (the signal bandwidth) more room before bumping into $f_s/2$, thus giving us more headroom to run our processes before we cross into the mirrored region. For example, oversampling by a factor of 2 should double f_s, and thereby double $f_s/2$ (image credit again goes to the EarLevel Engineering blog):



Oversampling is adding spaces and filtering !

1. Insert zeros between input samples
2. Lowpass filter at what is now $f_s/4$
3. Run your nonlinear process
4. Repeat the lowpass filter from step 2
5. First order hold into the output buffer

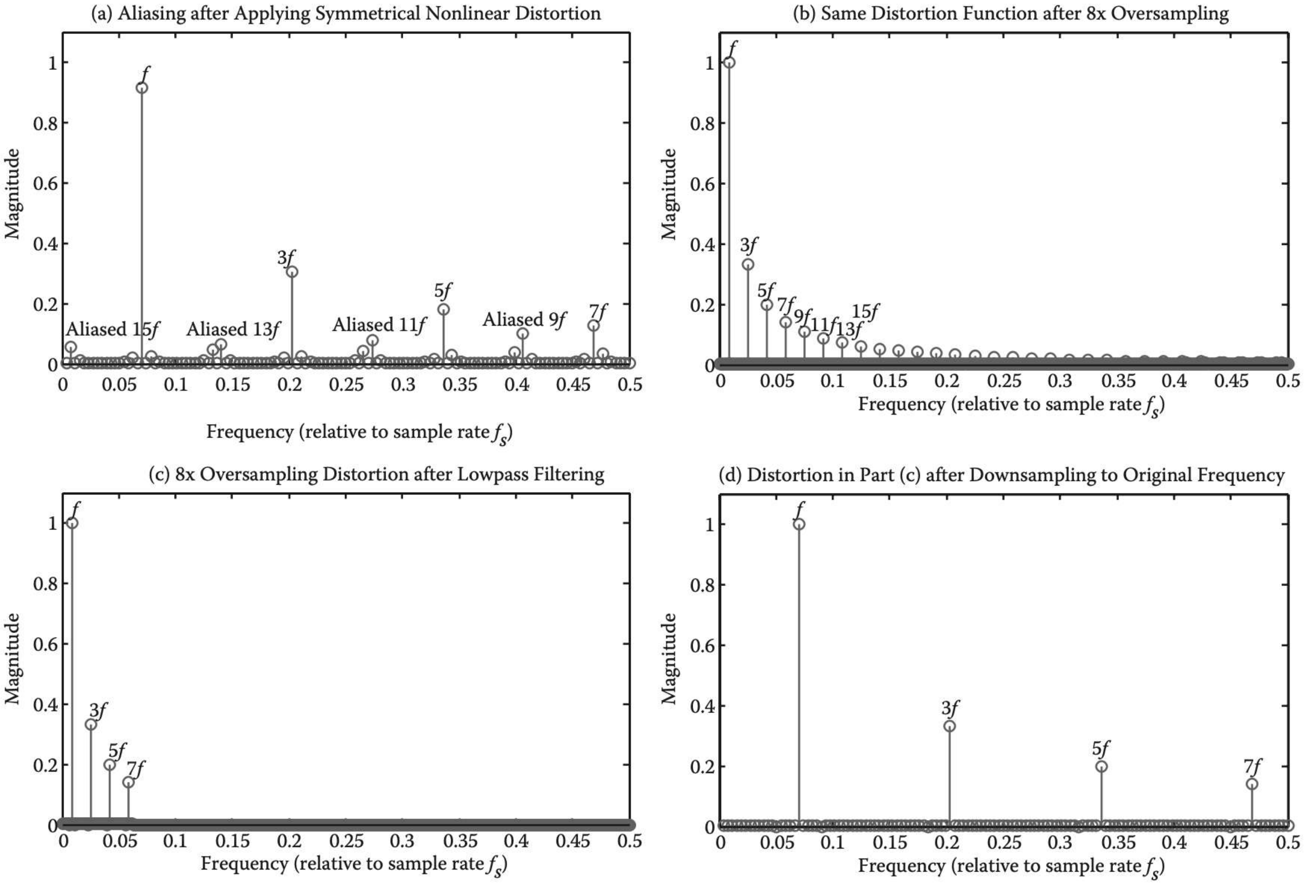


FIGURE 7.9

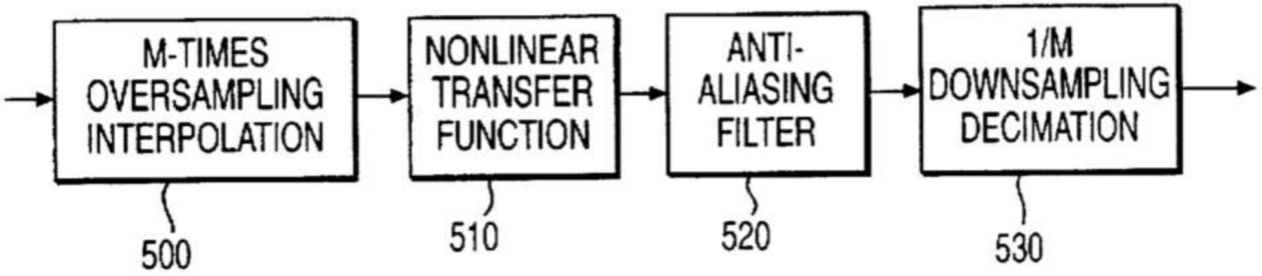
Output spectrum with aliasing due to distortion (a), and the output spectrum after oversampling (b), low-pass filtering (c), and downsampling (d).

My vision for distortion sounds

1. Use a soft clipping characteristic curve that rounds the corners of the waveform as it approaches the clipping level.

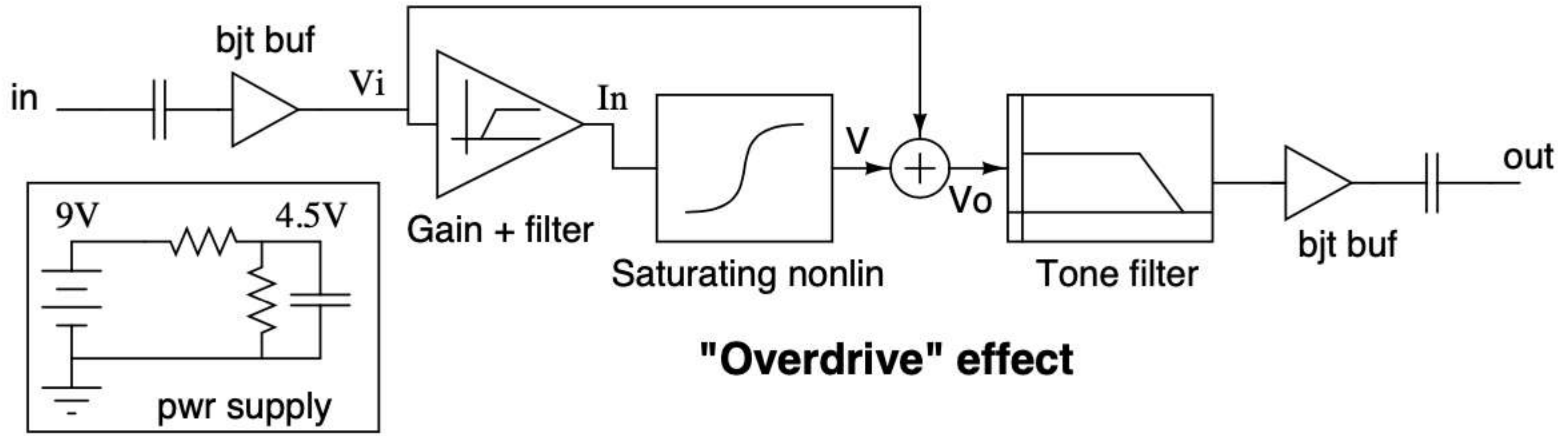
2. Choose the curve to be at least mildly asymmetrical, which will pro- duce even and odd harmonics. For example, the top and bottom half- waves in Equation (7.4) could use a different input gain.

3. Use oversampling to control nonharmonic products from aliasing. If the sound is still too harsh, consider adding a gentle low-pass filter before or after the nonlinear function.



Others distortion schema …

Ibanez



bjt buf
in
Vi
In
V
Vo
out
9V
4.5V
Gain + filter
Saturating nonlin
Tone filter
bjt buf
pwr supply
"Overdrive" effect

BOSS DS-1

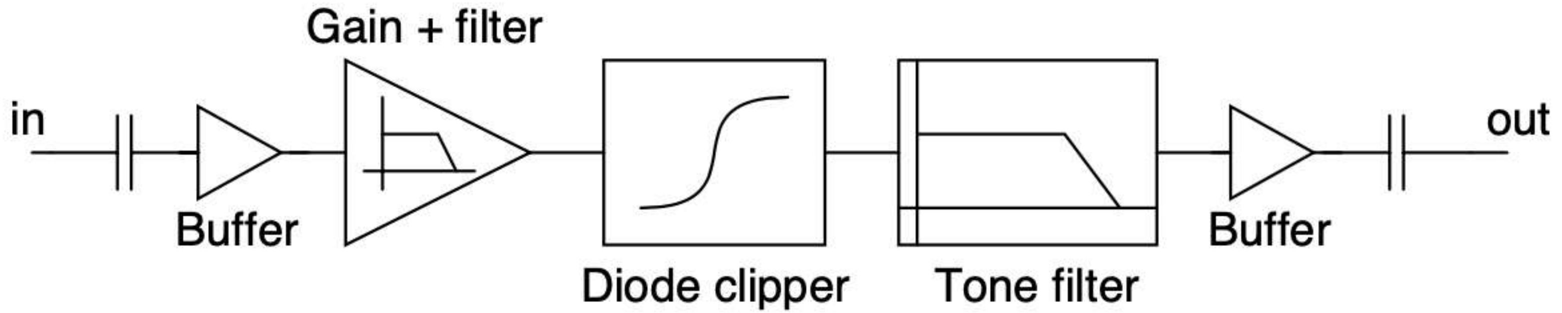
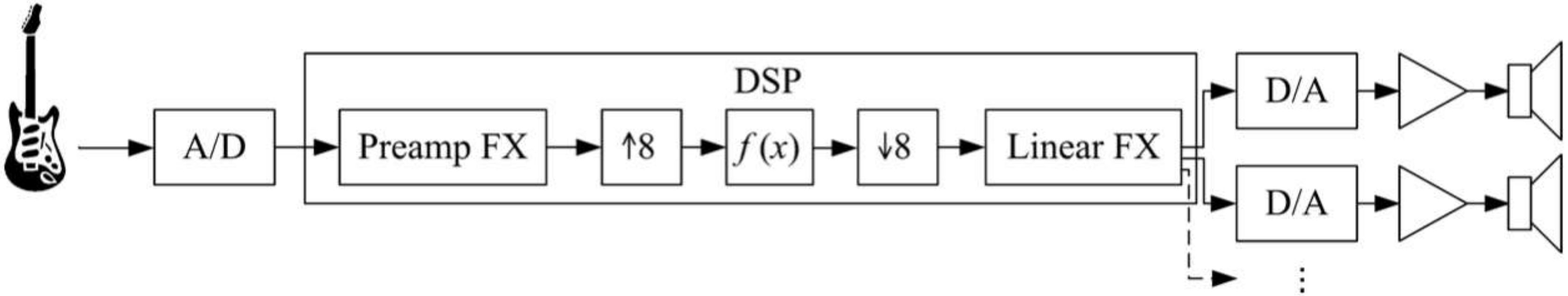
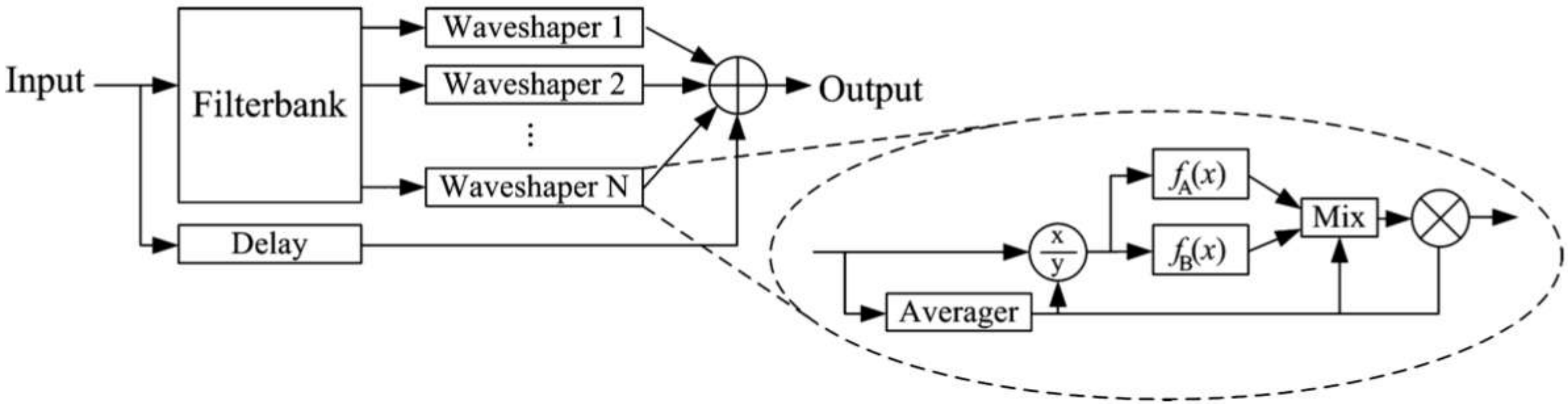


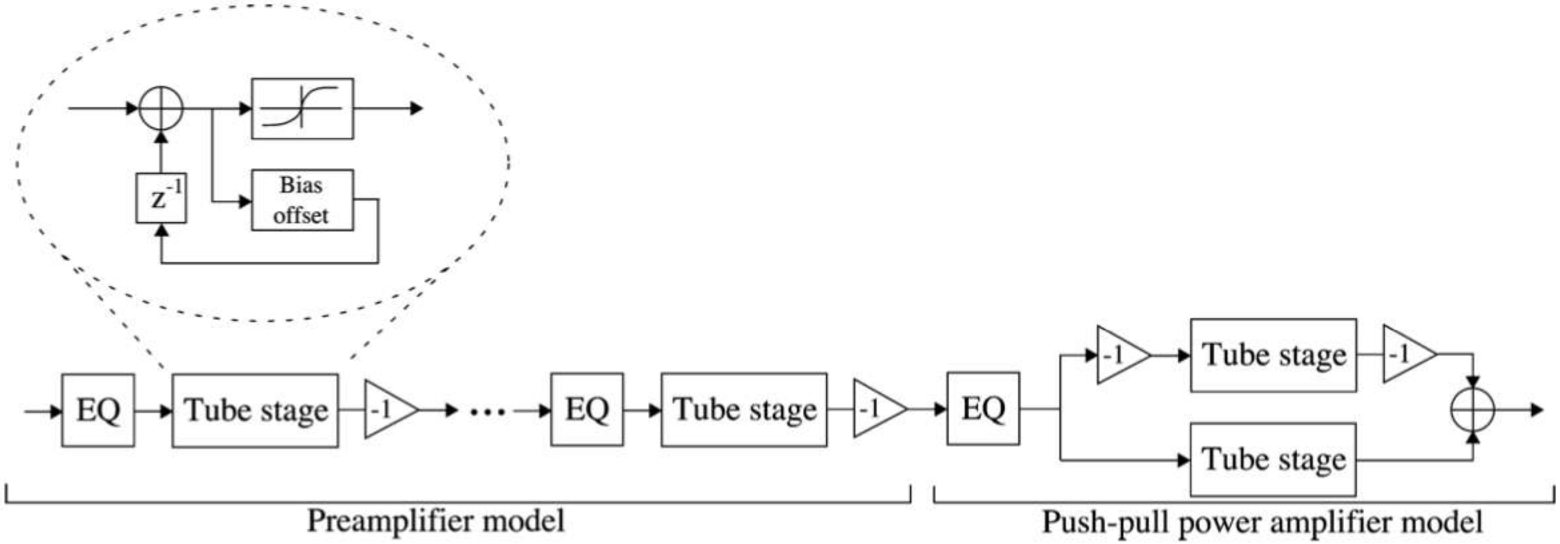
Figure 3.2: Partitioning scheme and block diagram for the Boss DS-1 circuit.



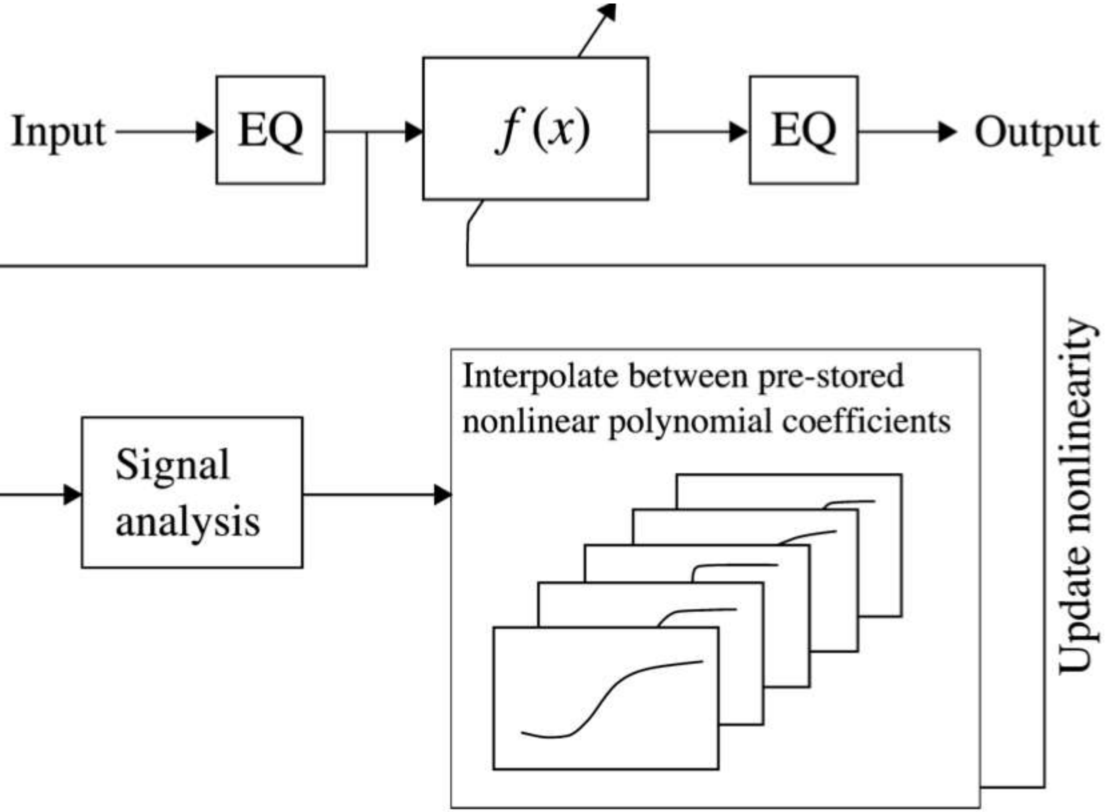
DSP
A/D
Preamp FX
↑8
f (x)
↓8
Linear FX
D/A
D/A



Input
Filterbank
Waveshaper 1
Waveshaper 2
Waveshaper N
Delay
Output
x
y
$f_A(x)$
$f_B(x)$
Mix
Averager



Bias
offset
z-1
EQ
Tube stage
-1
EQ
Tube stage
-1
EQ
-1
Tube stage
-1
Tube stage
Preamplifier model
Push-pull power amplifier model


Input
EQ
f(x)
EQ
Output
Signal
analysis
Interpolate between pre-stored
nonlinear polynomial coefficients
Update nonlinearity

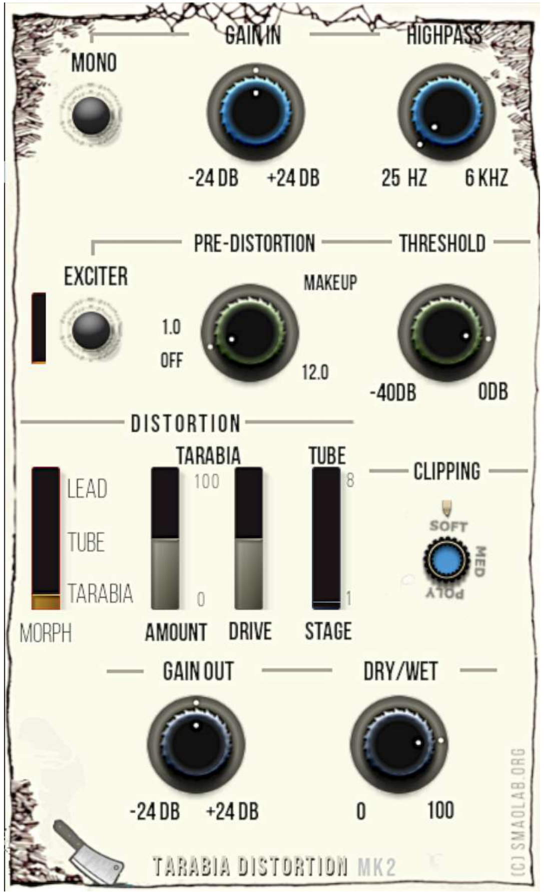
My distortion
(well in dec.2018)

Code example :
https://github.com/rstephane/MusicPlugin/tree/master/Tarabia

GAIN
HihgPassFilter
Exciter (Low Comp)
PRE-COMPRESOR

Oversamplingx4
   DISTORTION
   CLIPPING
Downsamplingx4

GAIN OUT
DRY/WET



Version in march 2022 !

References

- A perceptual approach on clipping and saturation by Stefania Barbati
- A Review of Digital Techniques for Modeling Vacuum-Tube Guitar Amplifiers by Jyri Pakarinen and David T. Yeh
- DIGITAL IMPLEMENTATION OF MUSICAL DISTORTION CIRCUITS BY ANALYSIS AND SIMULATION, STANFORD UNIVERSITY, David Te-Mao Yeh
- LAB REPORT 1: DISTORTION GUITAR EFFECT, Koosha Ahmadi 311176976, Digital Audio Systems, DESC9115, Semester 1 2012 Graduate Program in Audio and Acoustics, Faculty of Architecture, Design and Planning, The University of Sydney.
- https://www.nickwritesablog.com/introduction-to-oversampling-for-alias-reduction/
- **IEEE SSCS, Santa Clara Valley Chapter March 22, 2012, The Evolution of Oversampling Analog-to-Digital Converters Bruce A. Wooley, Stanford University.**
- **IMPROVING ADC RESOLUTION BY OVERSAMPLING AND AVERAGING (Silicon Labs)**
- Audio Effects Theory, Implementation and Application, Joshua D. Reiss , Andrew P. McPherson
- USPATENT (ask google ;-)) : US4797902, US5524074, US5789689, US8275477
- **OLD Dplug Code example** : https://github.com/rstephane/MusicPlugin/tree/master/Tarabia