Computer Music - Languages and Systems

Group 5: PLAS

Assignment 5: Distortion Effect

Homework 2 Report

Students

Alberto Doimo
Paolo Ostan
Stefano Donà
Lorenzo Curcio

POLITECNICO
MILANO 1863

Description of the problem

This homework assignment aims at the creation of a distortion effect plugin using the *JUCE framework*.

Distortion is a form of audio signal processing that is achieved by applying a non-linear transformation to the signal. Different types of distortion are associated with different response curves, and depending on the characteristics of these, the resulting harmonic content undergoes specific variations. In order to develop the plugin, it has been crucial to determine what the ending result would have to sound like. After some tests with different characteristic curves the group decided to select one main curve that controls the main distortion and another one called DESTROY. Furthermore, a graphical user interface was obviously needed, in order to allow the user to have control over the parameters that define how the transformation of the signal is applied.

Approach to the problem

Given the task assigned to the group, the first main focus was to test some suitable distortion curves in order to understand dynamic and harmonic response of each one. After some preliminar tests the group agreed on the implementation of two functions: one characterized by a more standard curve (*regular mode*), and one where the resulting curve undergoes a more complex distortion function (*destroy mode*). In order to allow a deep processing customization the plugin has been divided in two sections, one for the left channel and one for the right one. Furthermore, the user may decide to apply a mid-side encoding instead. The channel-strip contains a filtering section, applied before the distortion, a distortion section and a tone filtering section that darken the distorted signal. Moreover, the whole process stands in the middle of two oversampling blocks in order to avoid aliasing effects. All of the previous elements are represented in fig.1 and described in the following sections.

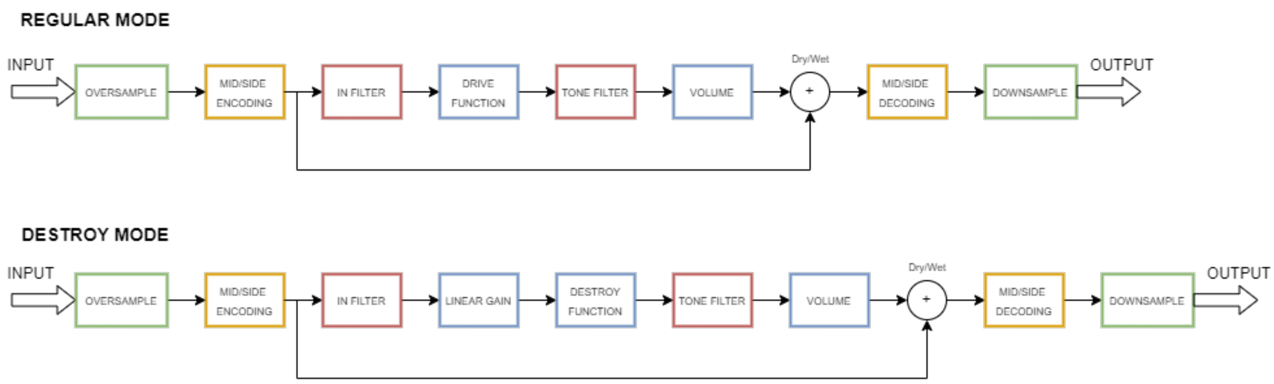

Figure 1: Final processing block diagram

GUI

The user interface is divided into three parts to help the user having the best possible experience while using the plugin and its features. There are two bands dedicated to the two channels, as previously described. In each section starting from the left there is an *ON* button to activate the distortion effect, a small section dedicated to input filter, with three different filtering options (low-pass, band-pass, high-pass) and two knobs that control the frequency and the Q factor. Another knob controls the value of the gain parameter of the distortion curve function, and it is followed by tone filter, volume and wet/dry controls. On the right side a double meter shows the input signal, for both the L/R and M/S possibilities, and the stereo output signal. The channel encoding can be selected with the specific buttons. The *DESTROY* button and the *DESTRUCTION* knob allow to control the second distortion function.

The final appearance of the plugin can be seen in fig.2, while the destroy mode can be seen in fig.3.

Figure 2: Normal Mode

Figure 3: Destroy Mode

Distortion Curve(s)

As said, two distortion curves were implemented. The first of the two is a variation of an odd exponential curve, which was modified in order to be asymmetric in respect to the origin. The considered curve is therefore a combination of two exponential functions, as seen in fig. 4, where the gain g ranges from 0 to 1.

$$f(x) = \begin{cases} -0.5 \cdot (1 - e^{(g+1)\cdot 3\cdot x}) & -1 \le x \le 0 \\ 1 - e^{-(g+1)\cdot 5\cdot x} & 0 < x \le 1 \end{cases}$$

The final curve that is used in the plugin, however, is a linear combination of the resulting signal and the original one, weighted using the value of g. This decision was motivated by the fact that the resulting effect could never become a soft-clipping type distortion. With the following combination, when g=0, the signal is left untouched, and by increasing g, the final curve better approximates the curve f(x).

$$z(x) = gf(x) + (1-g)x$$

A "destroy" mode was also implemented, whose distortion curve is the following:

$$f(x) = sin((1+g)\pi x) + 0.2 \cdot sin(g^2 \pi x)$$

It's reported in fig. 5. and fig. 6 with different values of g. To avoid absolute values greater than one, all such values are finally flattened to 1 or -1.

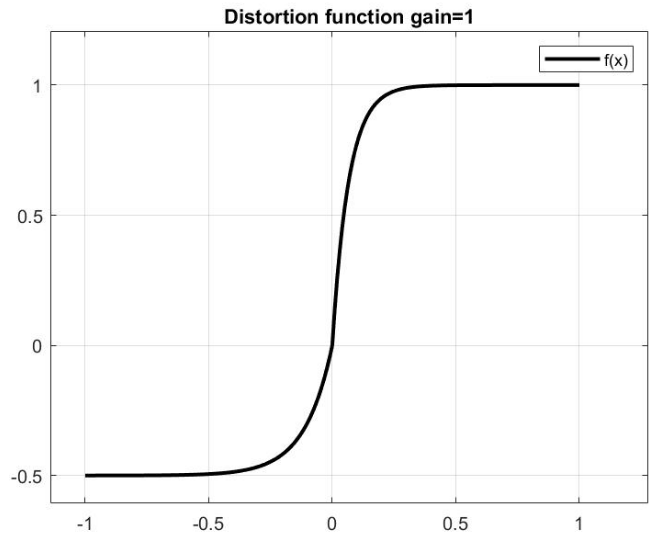

Figure 4: First distortion curve (Gain g = 1)

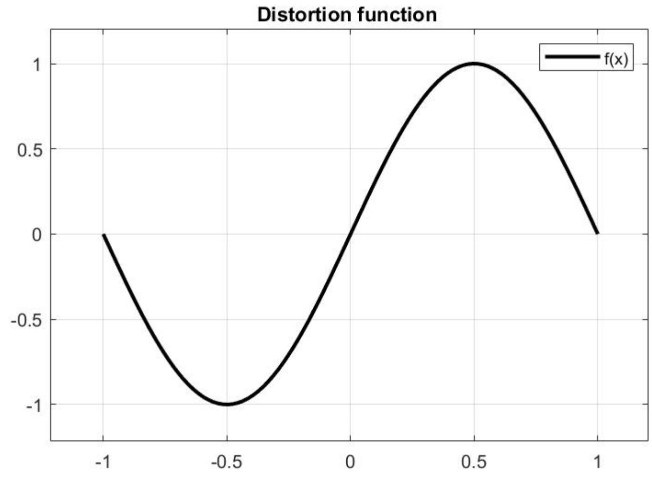

Figure 5: Destroy mode distortion curve (Gain g = 0)

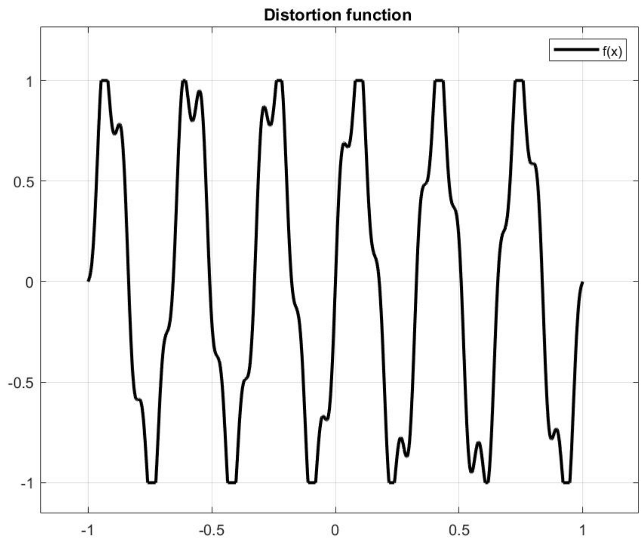

Figure 6: Destroy mode distortion curve (Gain g = 5)

Stereo Processing

As anticipated, the plugin allows for a dual-channel processing approach. The user has the possibility to decide whether to operate on the left and right channels separately, or to have a mid-side separation instead. Mid-Side encoding is a particular technique that allows the user to manipulate the stereo field in a different way. The main concept is to divide the audio signal in two component: the **MID** which describes what left and right channels have in common, and the **SIDE** which describes how left and right channels differ. To do so, the **mid** signal is selected as the sum of the left and right channels, while the **side** one is the subtraction of the two. Both **mid** and **side** are scaled by a factor $\frac{\sqrt{2}}{2}$ in order to compensate for the amplitude after the summing process and keep the behaviour of the plug-in consistent. This allows to maintain a coherent working point between LR and MS processing in the distortion function.

$$\begin{cases} M = (L+R) \cdot \frac{\sqrt{2}}{2} \\ \\ S = (L-R) \cdot \frac{\sqrt{2}}{2} \end{cases}$$

After all the distortion and filtering processes have been applied, the decoding block takes place:

$$\begin{cases} (M+S) = \frac{\sqrt{2}}{2} \cdot [(L+R) + (L-R)] = \sqrt{2} \cdot L \\ \\ (M-S) = \frac{\sqrt{2}}{2} \cdot [(L+R) - (L-R)] = \sqrt{2} \cdot R \end{cases}$$

$$\begin{cases} L = (M+S) \cdot \frac{1}{\sqrt{2}} \\ \\ R = (M-S) \cdot \frac{1}{\sqrt{2}} \end{cases}$$

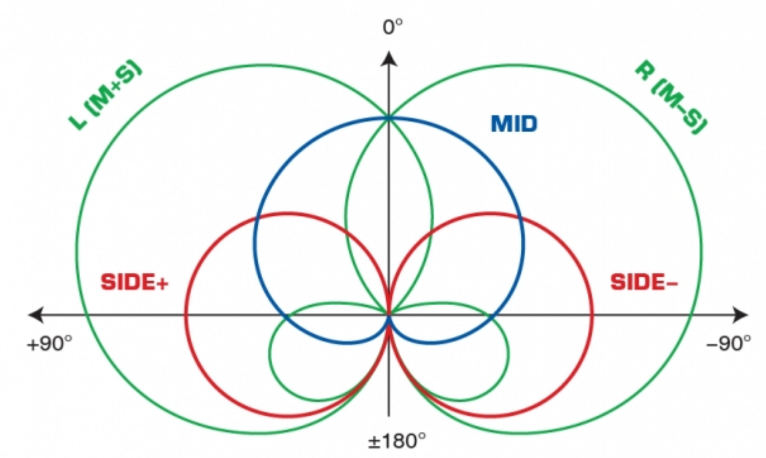

Figure 7: Mid/Side encoding

In both cases, the distortion parameters, the filters and the dry/wet parameter can be set independently among the two streams. To support a smooth workflow, the possibility of linking the two sections together was included. Doing this the user is able to easily process both the channels without having to toggle each one separately.

Over-sampling

To avoid problems related to aliasing, the signal undergoes an over-sampling stage, in particular the sampling frequency is increased by a factor of 4 starting from the DAW sample rate. As a matter of fact, when the audio signal is subjected to a distortion process, additional harmonic components are created. However these harmonics might cross the Nyquist's frequency, producing an aliasing effect. As shown in fig. 8, assuming 48kHz as sample rate, these higher frequencies will be mirrored with respect to Nyquist's frequency and create undesirable harmonic content in the audible range. A common technique to avoid this aliasing effects is to apply an over-sampling to the original signal in order to increase the Nyquist's frequency, so that the mirrored frequencies become great enough to be inaudible. After that, once all the process in the plug-in has been accomplished, a low-pass filter to the original Nyquist's frequency is applied in order to clean-up the harmonic content and a down-sampling to the original sample rate occurs.

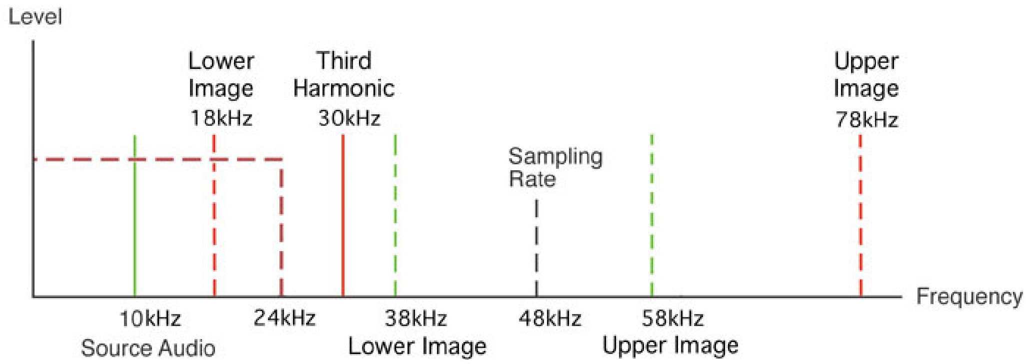

Figure 8: Aliasing effects due to distortion

Filtering

For each of the two channels, the user has the possibility to apply a pre-processing filter of one of three types: low-pass, band-pass, high-pass. This allows to choose which frequency range of the original signal's spectrum will be distorted. For example if a boost at low frequencies is needed, the input low-pass filter can be engaged. For these filters the cutoff frequency as well as the Q-factor can be modified in ranges that go respectively from 10 Hz to 16 kHz and from 0.1 to 2. Furthermore, a post-processing low-pass filter is present, whose cutoff frequency can be adjusted using the Tone knob in a range between 100 Hz and 15 kHz. For this purpose the group choose to use a particular type of filters called *State Variable Filter TPT* that aims, through some given parameters, to emulate the behaviour of an analog filter.

Conclusions

The subject matter allowed the experimentation of different non-linear distortion systems, using JUCE Framework and also other testing environments (like Matlab and Google Colab) for this purpose. The result is a distortion plugin that features many different processing possibilities as well as two different distortion systems.

The harmonic content created by the distortion is pleasant and very versatile, it can be used as pre-amp for vocals, bass saturator or drum crusher. Also experimenting in destroy mode with the destruction e the gain can produce a charming harmonic result.

The plug-in has been tested successfully in several Digital Audio Workstations like Garage Band, Reaper and Ableton Live. Unfortunately in DAWs like FL Studio, due to a different buffer management the plug-in may present some noise.