

LAB REPORT 1: DISTORTION

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ABSTRACT

From a musical perspective, the characteristic of an electric guitar, or a simple synth wave without any modulation may not be considered as desirably musical. The desire of seeking for variation in tone motivates musicians and engineers to apply effects on their instrument sounds. Distortion, being one of the most popular effects on rock music and hardcore dance music, is examined in this report.

1. INTRODUCTION

This report is concerned with the particular Matlab distortion code used for this lab session. The theory of function and the equation will be explained in details in this paper.

The code `expdist.m` used in this report was obtained from DAFX: Digital Audio Effects programmed by the author Udo Z'olzer, and modified by the writer of this report.

The sound file used in this project is name as `bass.wav`, programmed by the author of this report.

2. NONLINEAR SYSTEM/FUNCTION

The disproportional relation between the input and the output defines this distortion function as a nonlinear processor. (See Figure 1.1)

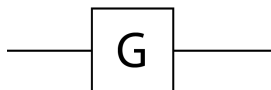


Figure 1.1: G block varies with input level in a nonlinear system such as distortion

This function does not carry the nature of Homogeneity that a linear system does, since the amplitude of the input signal may varies after being processed by the function [1]. (See Figure 1.2 and 1.3).

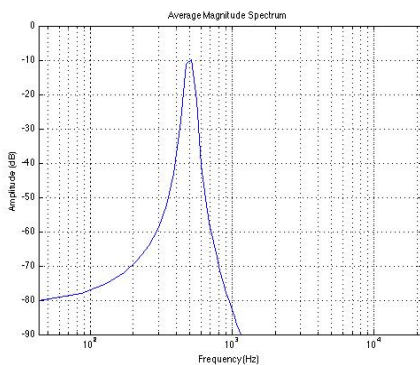


Figure 1.2: The amplitude of the input signal sits at -10db

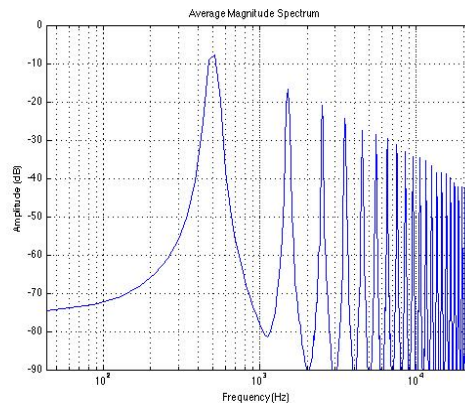


Figure 1.3: The amplitude of the output signal exceeds -10db after being processed by the function

3. DISTORTION FUNCTION – CHARACTERISTIC

This function (distortion) applied additional harmonic to the signal during the process. Compare the frequency content shown in Figure 1.2 and Figure 1.3, there are numbers of additional frequencies added after the signal being distorted. Those appeared after the fundamental frequency are referred as harmonics. Note that the major sonic characteristic, which the function produces, has more to due with the harmonic distortion added rather than the increased amplitude.

4. DISTORTION FUNCTION – THEORY

The simulation of clipping is applied in this function for the purpose of distortion. It simulates a clipping function during the process. The approach is given by the following equation.

$$f(x) = \begin{cases} 2x, & 0 \leq x \leq 1/3 \\ \frac{3-(2-3x)^2}{3}, & 1/3 \leq x \leq 2/3 \\ 1, & 2/3 \leq x \leq 1 \end{cases} \quad (1)$$

When the input signal is being driven to the threshold of 1/3 of the system, it gets doubled while the system remains in its linear region. The function is not as affective when being operated in linear mode since the major part of the sonic signature is affected by its nonlinearities. Furthermore, a form of compression is produced between the input values of 1/3 up to 2/3, which can be described as $\frac{3-(2-3x)^2}{3}$ in Equation (1). Once the input values exceed the 2/3 threshold region, the output will be set one [2]. A simple demonstration of the concept of clipping is shown in Figure 2.2

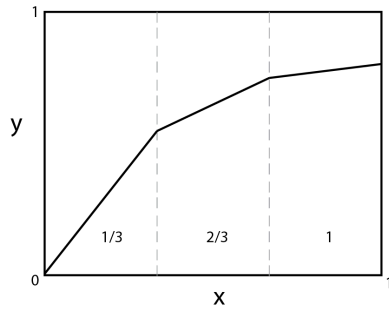


Figure 2.2: an example of the relation between the input and output of the function when being clipped

Furthermore, the odd order harmonics (e.g. the third and the fifth...) produced by the nonlinear clipping starts to fade out corresponding to the compression that is reached in the lower threshold, shown in the middle term in Equation (1).

In addition, higher harmonics with higher levels will be added to the signal when a harder clipping is performed without a soft compression (see Figure 2.2 for demonstration of a hard clipping). This function however does not provide options for users regarding on choosing soft or hard clipping. As mentioned previously, the function is a distortion that simulates a clipping function. Instead, the control of the amount of distortion of this function determines the hardness or softness of the clipping.

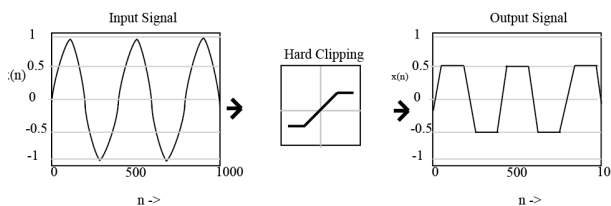


Figure 2.2: Input Signal through Hard Clipping

This function is capable of producing a “harder” and “harsher” sound characteristic especially when the function are being fed by two or more different frequencies simultaneously, the reason is because this function introduces odd harmonics to the signal, as well as a range even-order harmonics as shown in Figure 1.3, giving the signal a broader range of harmonic content.

As the users increase the amount of distortion when using this function, the positive part of the waveform will be clipped. If the positive clipping level is not proportional to the negative clipping value, asymmetrical clipping will be performed [3], shown in the figure below.

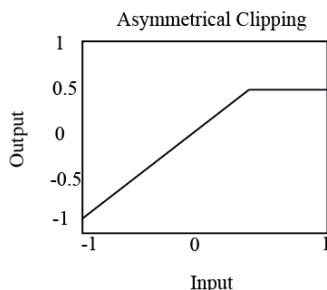


Figure 2.2: Example of asymmetrical clipping that the function applies

5. OUTCOME

This function in fact, is not constructed with the structure of a clipping function, however, the added harmonic content that the function produced is cause by the theory of a clipping distortion. This simulation allows users to simply apply the desired amount of distortion to a signal when using this function, without worrying about the all parameters of a clipping distortion function.

A synth bass loop named *bass.wav* (programmed in Logic Pro) was used in this lab report to demonstrate the characteristic and purpose of this distinction function. As shown in the diagram below:

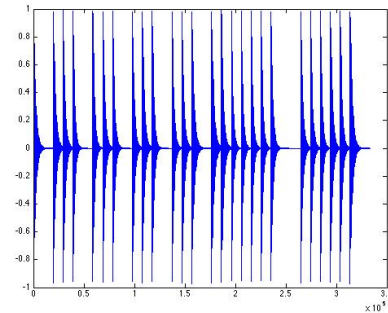


Figure 3.1: Input Signal: *bass.wav*

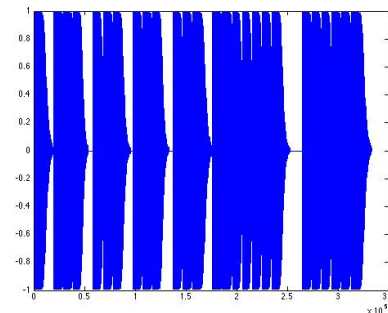


Figure 3.2: Output Signal: *bassoutput.wav*

The distorted signal processed by this function contains more harmonic content for each note. The result of the extra frequencies affects the musicality of the loop. The tone and timbre of each note are more defined in the distorted version. In comparison to the output, the original input loop may not even have enough harmonic content for the listeners to determine which note are being played through out the loop.

6. BIBLIOGRAPY

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7. REFERENCES

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- [3] J.O. Smith, *Physical Audio Signal Processing: for Virtual Musical Instruments and Digital Audio Effects*, W3K Publishing, web based, December.2010, pp.336 – 340