EAR EXCITING EXCITER

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Final Review for Digital Audio Systems, DESC9115, 2016
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Abstract

This aim of this review is to inspect the specifications and implementations of the 'Ear Exciting Exciter' by Aural Technologies. Comparing vocals which may need 'excitation' to the resulting output of the developed exciter.

1. Introduction

An ongoing problem since the invention of the microphone is the loss of high frequency content. The Aphex 'Aural Exciter' invented a hardware processor, which was able to create high frequency harmonics through harmonic distortion. This invention proved to be successful as many units were sold, and the demand for a more 'real' like recordings was high (Aphex manual pg 3).

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The 'Ear Exciting Exciter' is a digital audio effect based on a similar algorithm to the Aphex design. The Ear Exciting Exciter was designed as a 'psychoacoustic' processor, with the main goal of increasing speech intelligibility and introducing brightness to recordings with either nil or low amount of high frequency content, whilst not requiring much processing power.

2. Problem

There is an ongoing debate that analogue systems provide better tonal qualities than digital systems. Though analogue systems has its limitations, such as, requiring more processing power, taking up space, generating heat, and not being cost efficient. The need for digital systems that can emulate the tonal qualities of an analogue device without those limitations is in high demand.

The digital audio effect designed, must not take up too much processing power, not have any audible artefacts (which can be quite common in digital signal processing (DSP)), be an affordable option. And as a bonus, be able to emulate some of the tonal qualities of its analogue counterpart in order for it to be a desired product.

The earlier mentioned problem regarding the loss of high frequency content during recordings, which reduces speech intelligibility must be taken into consideration. To add more brightness to the signal, harmonic distortion, compression, or the combination of both can be introduced. Introducing these high frequency components will reduce the chances of low frequency parts masking the signal, thus increasing speech intelligibility.

3. Specification

The Ear Exciting Exciter uses a Chebyshev Filter algorithm for its high-pass filter, the Chebyshev was chosen over the Butterworth design to create a steeper roll-off, ensuring an accurate cut-off frequency, whilst only attenuating 0.05 dB in equiripple. The user has the option to select the cut-off frequency to guarantee desired results.

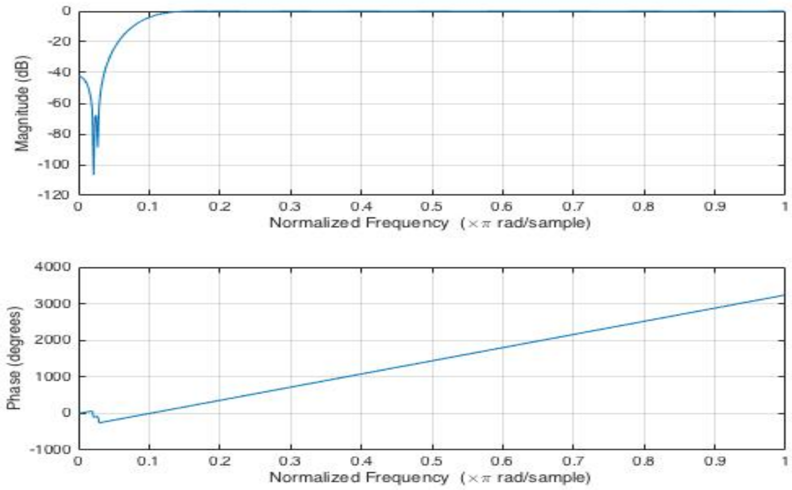

Figure 1. *High-pass filter frequency response with a cut-off frequency of 2kHz.*

The unit includes a compressor function with gain, threshold, ratio, attack and release time as its variable input functions. The gain input occurs before the RMS of the signal is taken, based on part of the design of the Urei 1177Ln (Urei 1176 manual pg 55).

The harmoniser function includes a user changeable option for either half-wave or full-wave rectification, replicating the Aural Exciter's design and creating even and odd order harmonics.

4. Implementation

The exciter follows a similar signal flow to the Aphex 'Aural Exciter' as depicted in figures 1 & 2. The input signal goes to the forward path, the side-path first includes a high-pass filter, the output of the high-pass filter is then split into the harmoniser system and the compressor system, the resulting outputs are then multiplied and attenuated to compensate for the added gain, the signal is then summed with the original input signal.

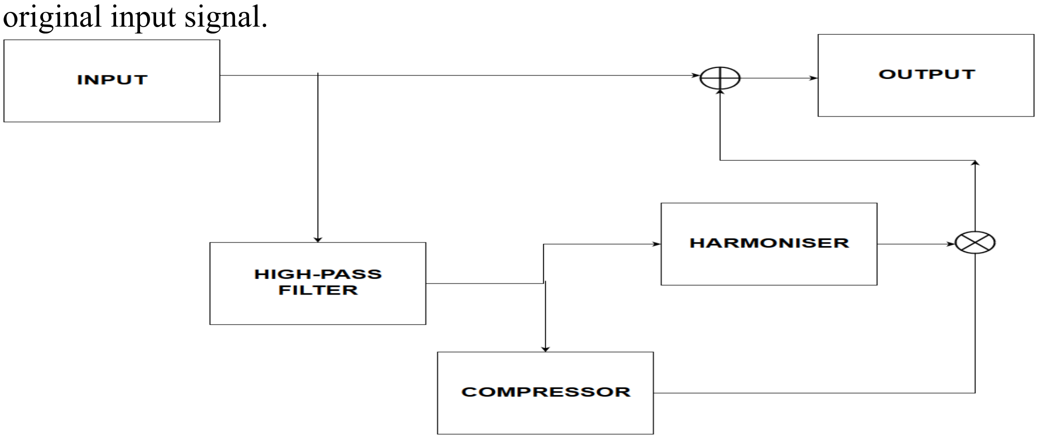

Figure 1. *Signal flow of the Ear Exciting Exciter*

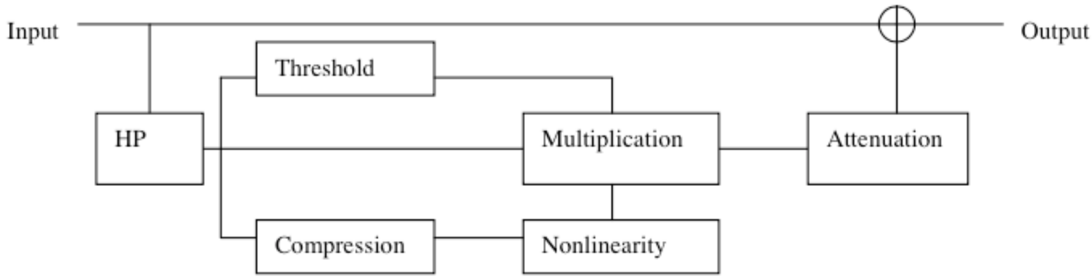

Figure 2. *Simplified signal flow of Aphex Aural Exciter (figure Chalupper pg. 11).*

The implemented Chebyshev type 1 filter, is a Finite Impulse Response (FIR) version. The formula used can be seen in Equation 1. A 40^{th} order filter has been included. The order of N = 40 has been tested to ensure the right amount of roll off and pass-band equiripple.

$$H(z) = \sum_{n=0}^{N} b_n z^{-n}$$

Equation 1. *Implemented FIR filter formula, where N = 40*

The compressor function is based on code by M.Holters. Some minor changes have been included to fit well with the exciter. Compressors use characteristic curves to define ratio and slopes, they are CR > 1 and 0 < CS < 1 (Zolzer pg 110).
In Equation 2, gain is calculated. Where CR is the corresponding ratio, CS is the characteristic curve

$$g(n) = \min\left(1, \left(\frac{x_{\mathrm{RMS}}(n)}{ct^2}\right)^{-CS/2}, \left(\frac{x_{\mathrm{RMS}}(n)}{et^2}\right)^{-ES/2}\right),$$

Equation 2. *Compressor and expander formula, only the compressor formula is implemented in the Ear Exciting Exciter. Gain calculated, RMS value is squared. Where ct is compression threshold and et is expander threshold* (figure Zolzer pg 111).

The added code to this compressor function is an additional pre-gain, where the signal is multiplied by a user specified value (y = x * g), the signal is then attenuated by the compressor's threshold if the gain exceeds the predefined limit.

In this exciter, a harmoniser function using earlier mentioned half and full-wave rectification is used. Taking the input signal and outputting the absolute value of the signal, is one way of introducing harmonic distortion. Harmonic distortion is essential for adding new harmonics to a signal. Whilst there are a few ways of producing harmonic distortion, a DSP version of full-wave and half-wave rectification has been implemented in the Ear Exciting Exciter, emulating the 'Aural Exciter's' analogue design.

Equation 3 and 4 shows the formula for both half and full-wave rectification in the digital domain.

$$fwr = |x_n|$$

Equation 3. *Full-wave rectification (fwr) is simply the absolute value of the input signal.*

$$hwr = \frac{x_n + |x_n|}{2}$$

Equation 4. *Half-wave rectification (hwr), the input signal is summed with the absolute value of the input, then divided by 2, ensuring all negative values = 0.*

5. Evaluation

Audio examples of the exciter being used with variable inputs have been included. The test signal contains a vocal, which is lacking high frequencies and includes slight masking on certain words. In tests 1-4 only 2 parameters are being changed, whilst the other others remain a set value:

```
fc = 5000;    %Cutoff in Hertz
wc=2*fc/fs;   %FIR inputs
N=40;          % Filter Order
type = 'high';
NC = N + 1; %Chebyshev inputs
NO = N - 10;
% Harmoniser inputs:
%hBP is the bypass for the harmoniser set at 1 to bypass
hBP = 0;

maxlevel = -6; % Max level for normalisation

% Compressor inputs:
 gain = 3;      %Pregain
    CT = -7;        %Threshold in dB
    CS = 1.6;       %Slope (Ratio)
    AT = 0.01;      %Attack Time
    RT = 0.003;     %Release Time
harmtype = input('\n Would you like to use half-wave or full-wave
(rectification)?\n\n >','s');
```

Figure 3. *MATLAB code, showing the user defined input arguments*

The audio results of each of tests can be audibly compared to the original signal (Vocal.wav). Audio results of test 1 (figure 3), can be referred to ‘5kHzFullWave.wav’. When comparing spectrally, it can be observed that there is an increase and an addition of upper harmonic content as shown in figure 4.

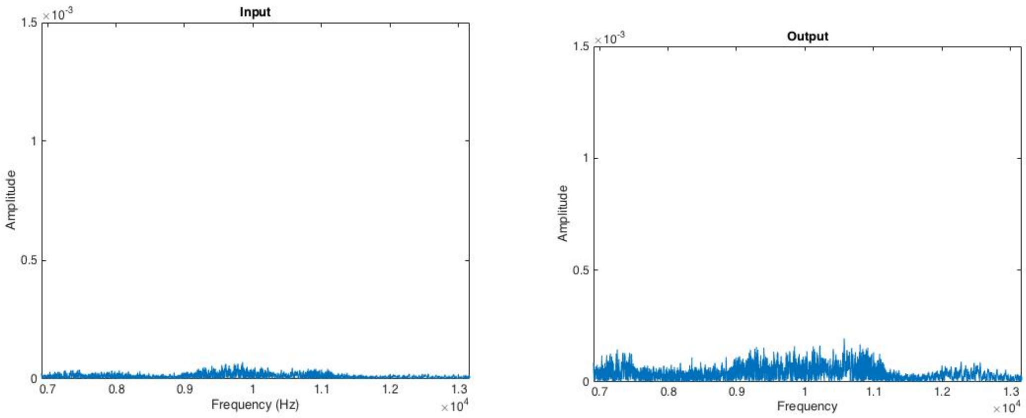

Figure 4. *Comparison of Vocal input and Vocal Output with a 5kHz cut-off using full-wave rectification*

In test 2, the cut-off frequency is changed to 2kHz and half-wave rectification is used. Refer to audio file ''2kHzHalfWave.wav' for a listening example. Again there is a noticeable addition and increase of upper harmonics as shown in figure 5.

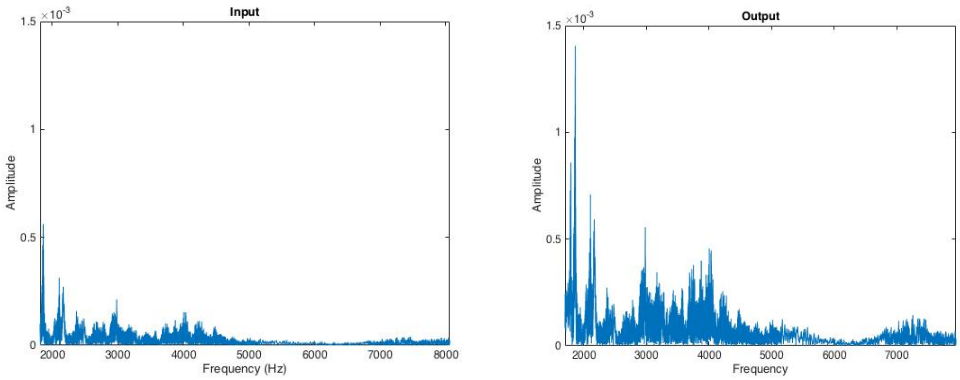

Figure 5. *Test 2, A cut-off frequency of 2kHz using half-wave rectification is used.*

Tests 3-4 can be compared audibly. Refer to '2kHzFullWave.wav' for test 3 and '5kHzHalfWave.wav' for test 4.

If the user wants to increase the high frequency content by a large amount, digital artefacts can be heard. Refer to audio example 'ExcessiveGain.wav' for test 5, where the input arguments are the same as figure 3, with the change of gain from 3 to 7.

After audibly comparing tests 1-4 to the original signal, it can be concluded that the resulting output has increased and added new upper harmonics, thus increasing speech intelligibility. However, if the inputs of each of the functions are overvalued, the output may result in unwanted distortion.

It can be concluded that the Ear Exciting Exciter works as intended. Further implementations would include a Graphic User Interface (GUI), which can also be used to restrict user defined inputs, so that unwanted sound will not be easily achieved by the user.

[1] Chalupper, J. (2000). AURAL EXCITER AND LOUDNESS MAXIMIZER: WHAT'S PSYCHOACOUSTIC ABOUT "PSYCHOACOUSTIC PROCESSORS" ? *Audio Engineering Society Convention 109*. Retrieved March 17, 2016.
[2] Zölzer, U. (2011). *DAFX: Digital audio effects*. Chichester: Wiley.
[3] Aphex: Aural Exciter Type III Model 250, Operating Guide and Service Manual
[4] Urei 1176ln Operating Guide and Service Manual