Faust Model of the Echoplex Tape Delay

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***Abstract*—The Echoplex is a popular tape delay effect widely used in the 60's and 70's, which had an important role in characterizing the sound of those years. In this work a virtual model of the device is presented, based on the previous work by Arnardottir et al. The algorithm is made with the Faust programming language. Firstly, an overview of the main characteristics of the physical device is provided, then it will be described the proposed implementation. Finally, conclusions will be drawn, with an evaluation of the model based on a listening test.**

I. Introduction

Delay effects have been fundamental tools for music production and recording for decades. From a theoretical point of view, this type of effect involves recording the input signal and playing it back after a desired period, summed with the original signal; the delayed signal is then often played back multiple times using a feedback line. In order to achieve the task, multiple techniques were developed over time, each one making use of different kind memory units: from magnetic tapes to digital ring-buffers. Each implementation of the delay effect add a different *character* to the sound, depending on the nature of the medium where the signal is recorded and played back. Nowadays most of music production is carried out digitally, therefore there is an increasing demand of digital effects. While able to emulate most of their analog counterparts, digital effects are usually considered as less *warm* than analog ones. For this reason, faithful digital emulation of analog effects are widely requested.

Tape delays make use of magnetic tape as a medium to record the delayed signal. Tape of fixed length is kept in movement by a capstan, a record head records the input signal, a playback head reads it and an erase head clears the tape getting it ready to record new signal. The delay is achieved by keeping a distance between the record and playback head. There are two kinds of tape delay: one varies the delay by changing the speed of the tape (*speed-type*), the Roland Space Echo is an example [1]; the other one allows to move the record or playback head, the delay then changes in relation of the varying distance between the heads (*length-type*). The Echoplex belongs to the latter family. In general, tape delays are still appreciated nowadays for the imperfections produced by analog components: for instance, saturation, unintended alterations in frequency response, time-variation of the delay length, frequency-band limitations given by the tape. This work aims to develop an implementation in the Faust programming language of a delay dsp which includes some of the characteristics of the Echoplex, based on the on the study conducted by Arnardottir et al. [2].

II. Characteristics

As stated above, the Echoplex is a length-type delay. Playback and erase heads are fixed at each side of the Delay Handle moving section, the record head writes to the moving tape which has previously been erased by the erase head, the playback head then reads what has been written to the tape. Moving the Delay Handle changes the position of the record head, and therefore the distance between the latter and the playback heads, this way the delay length is varied. Fig. 1 shows the internal mechanism of the Echoplex. Input signal is delayed and fed back to the input, the amount of feedback is controlled by the Repeats knob. which has a range that goes from 0 to 2. Saturation in the tape and circuitry limit the signal amplitude when feedback > 1, in this case the unit self-oscillates and moving the Delay Handle will produce interesting sound effects. There are several other factors that make the Echoplex a peculiar effect, which will be described in the following sections.

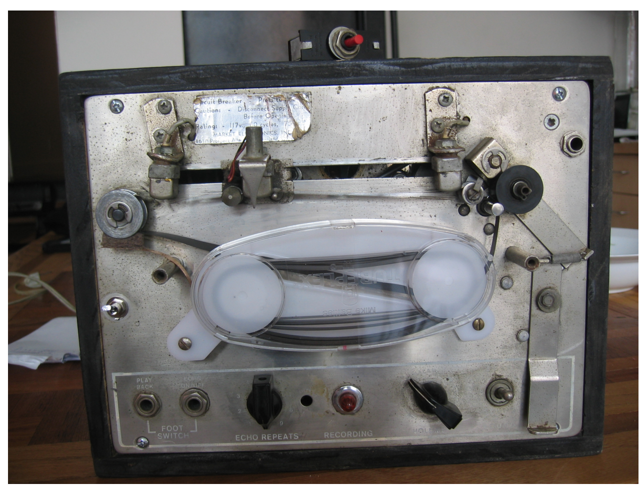

Fig. 1. Maestro Echoplex Ep-2 tape mechanism

A. Continuous delay-time changes

In digital delays the process of varying the delay length involves discretely moving the read pointer in a ring buffer. In contrast, in the case of tape delays this operation happens in a continuous way, leading to interesting sound features. When the record head is moved towards the direction of the tape, i.e. when the delay length is reduced, the delay signal is doppler-shifted towards upper frequencies, while vice-versa, when the Delay Handle is moved in the opposite direction, the delayed sound is shifted to lower frequencies. This can be proven with an analogy with the classic doppler effect, described by eq. 1.

$$f = (\frac{c + v_r}{c + v_s}) f_0 \tag{1}$$

Where f_0 is the recorded frequency; f is the reproduced frequency; c is the speed of the wave, in this case the speed of the tape, which has a nominal value of $8ips$ [3]; v_r is the speed of the receiver, in this case the playback head, so $v_r = 0$; v_s is the speed of the source, therefore the speed of the moving Delay Handle, and it is positive when moving away from the playback head, negative when moving towards it.

There are also extreme cases to consider: if the Delay Handle is moved very fast away from the playback head, than the tape-bias signal can be doppler-shifted into the audio band. On the contrary, if the delay control is moved towards the playback head faster than the tape speed, then when it eventually stops (because the track where it can move ends) a high-bandwidth transient is recorded, resulting in a "sonic boom".

B. Fluctuating Time Delay

One of the appreciated features of tape delays is the *character* of the delay sound: a large part of this characteristic is due to the fluctuating time delay resulting from imperfections of the hardware. Measurements run in Arnardottir's work [2] on the stability of the time delay value evidence three main spurious frequencies at 2.5, 5 and 26 Hz, which correspond to irregularities in the pinch wheel and the capstan rotation frequency. Moreover, the spectra of the delay presents the scalloped structure of a comb filter, with a slight slope towards the high frequencies indicating a low-pass filter. Analysis report that the spectral nulls occur at frequencies which are odd integer multiples of the record head-playback head distance, as such, they are probably due to mechanical disturbances propagating along the unit. For images related to the analysis see [2].

III. Implementation

The model has been implemented using Faust: a high-level, domain specific functional programming language for developing digital signal processing algorithms, which can be exported and integrated into different software. In Faust data is treated as a signal stream and processed using mathematical functions. Different signals can be routed together using composition operation in a block-diagram fashion. Code written in Faust can be both compiled for being used inside multiple applications, or used directly inside its IDE. Given that Faust does not provide access to the sample-by-sample processing of the signal, more effort has been put in modeling the *fluctuating time delay* part of the unit, leaving accurate emulation of the *Continuous delay-time change* for future C++ developments.

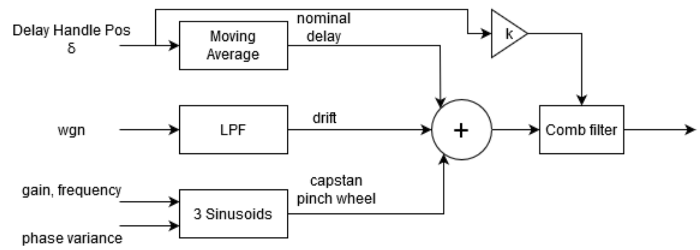

Fig. 2. Delay generation signal flow

The block-diagram of the delay signal flow is represented in Fig. 2, whose parts will be described in the following sections.

A. Delay Handle

The delay handle is implemented with a slider which controls the delay value in milliseconds, ranging between 1 and 1000 ms, the value is then converted in samples with the sample rate getter function provided by Faust: *SR*. In order to emulate the continuous delay-time change, the slider values are smoothed with a moving average filter, a linear smoothing function. The filter has been implemented as described in [4], using a rectangular window of 500 samples, chosen empirically. The filtered value represents the nominal delay of the filter.

B. Capstan and Pinch Wheel

Not having a real device to test, all the details on the characteristics of the fluctuation signals had to be obtained from the analysis run in Arnardottir's paper [2], which unfortunately does not provide much detail on the measured quantities a part from graphs which show only part of the measures. Capstan and pinch wheel components are modeled using a sum of three sinusoids which modulate the nominal delay: as mentioned above, these signals have frequencies of 2.5, 5 and 26 Hz. From the graphs it is possible to roughly extrapolate the amplitudes of the sinusoids: respectively $0.75, 0.15$ and 0.15 ms. Since the two lower frequency signals seem to contribute to the modulation only at high values of the nominal delay, the amplitude of the first two sinusoids was made to be directly proportional to the record head-playback head separation, i.e. the nominal delay value. Phases were roughly set according to the graphs: respectively $(3/4)\pi$, 0. $(3/2)\pi$. Then low-passed white noise was added to the phases to introduce the observed irregularities in the fluctuations. Since these are smooth, noise was low passed with a cutoff frequency directly proportional to the sinusoid frequency, while the noise amplitude is inversely proportional to it. The low-pass filters used are the elliptic filters available in Faust. Again, since Arnardottir do not provide much detail on these features, noise amplitude and frequency band was chosen mostly empirically.

C. White Noise

The low frequency drift was implemented using white noise, filtered with an elliptic low-pass filter. Since this noise is due to

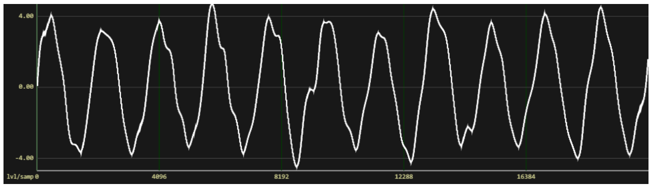

Fig. 3. Delay signal for minimum head-to-head separation, y-axis values are in samples

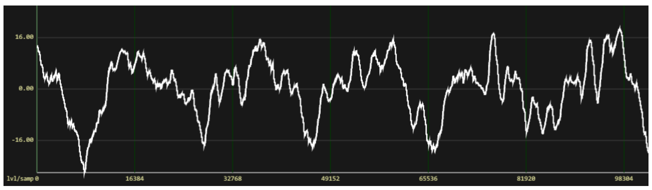

Fig. 4. Delay signal for maximum head-to-head separation

variations in the tape speed, amplitude is directly proportional to the head separation, i.e. the nominal delay. At maximum delay the amplitude is $0.75\,ms$, the cutoff frequency of the filter is fixed at $70\,Hz$. Values were chosen empirically.

D. Comb Filter

Comb filter was implemented in a feed-forward fashion by summing the signal with a version of itself multiplied by a constant $b < 0$ and delayed by a certain value. This technique was chosen instead of the feedback one because the frequency response measured in [2] presents convex notches. Filter is implemented as indicated in [4], difference equation and transfer function of the filter are reported below (2, 3).

$$y[n] = x[n] + bx[n - M] \tag{2}$$

$$H(z) = 1 + bz^{-M} \tag{3}$$

As previously mentioned, spectral nulls occur at odd integer multiples of the record head-playback head distance. The frequency location of the nulls in the filter spectra is given by the relation $f = nf_s/2M$, where f_s is the sampling rate of the signal to be filtered, M is the delay value and $n = 1, 3, 5, \ldots$ is an odd integer. Graphs indicate that for the minimum and maximum head-to-head distance the first null is located at, respectively 7.5 and $2\,Hz$. Filter delay is then set accordingly. Fig 3, 4 show the delay signal for minimum and maximum head-to-head separation with Delay Handle values subtracted: delay fluctuates at $0.2ms$ at minimum distance and at $0.7ms$ at maximum distance. The character is very similar to the one measured in [2].

E. Tape Saturation

Tape saturation and limiting were implemented using Faust limiter_1176 function: a virtual analog model of the 1176 Peak Limiter, popular studio compressor. This gives the nonlinear saturation typical of analog compressors while limiting the bandwidth, so that when the feedback value is greater that one the signal does not clip. Then a smooth low-pass filter is applied to simulate the limited bandwidth of the tape.

IV. Conclusions and Future Work

The aim of this work was to implement a Faust model of a part the Echoplex tape delay based on Arnardottir's work. Evaluating the quality of the final results is difficult, mainly because a proper assessment would need a physical device to test. Nevertheless, a listening test clearly indicates that the model is quite noisier than the original, and that delay fluctuations are more intense. However, when self-oscillating the unit presents behaviours similar to the original one. To summarize, the algorithm presented does an adequate job, despite several difficulties were encountered: among others, the lack of detail in the information given by the paper, and the impossibility of measuring a real device. Other difficulties originated from Faust structure: the IDE does not provide advanced debugging tools, making it almost impossible to deeply monitor the algorithm, and the documentation on built in functions is poor. Moreover Faust language itself, not being based on a sample-by-sample processing as classical programming mechanisms, posed a challenge in the implementation of the algorithm, especially since most literature on the subject is based on C-like programming languages. Nevertheless, Faust resulted being a useful tool: its high-level nature, the intuitiveness of its signal routing and the possibility of exporting for different platforms speeded up the development of classic dsp routines. However, the fact that the compiled code is put in just one file caused the C++ compiler's parser to stack-overflow, meaning that a JUCE implementation would need to manually separate portions of generated code into multiple files. Future work could involve improving the exported C++ algorithm using Faust C++ libraries, this way a more accurate model of the delay-time changes and tape saturation could be done. Moreover, proper measurement on a physical unit could surely improve the accuracy of the modeled parts.

References

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