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Delay and Release System by utilized Lorenz Attractor

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Abstract—In this paper, we propose a chaotic delay and release system which can change the tone of a sound with time. The chaotic system uses characteristics of chaos waveform and one-pole lowpass filters, and produces a sound with vibrato which is similar to live sound. The output of chaotic system is simulated by using the Csound.

1. Introduction

The chaotic delay and release system is applied to the sound which is converted into electric signals such as electronic musical instruments and electric musical instruments, and the system invents a near sound more naturally. The system is a kind of sound effecter which changes a tone. In general, the sound effector shows a device that gives the effect of the sound. It is inserted between the device that invents the sound and the speaker, and an original sound is processed[1].

The sounds by stringed instruments such as piano change harmonic sound with time. The changes of the sound volumes are based on the decrease of the vibration with time[2], and this phenomenon is simulated by using a envelope which controls the sound volumes.

The sound is a bright tone at the beginning of waveform which is outputted. The overtone element of the sound has a lot of elements at the beginning of waveform, and attenuates as fast as a high overtone element, and disappears[3,4].

In this proposed system, the filter is utilized in order to express the variation of the sound tone. The sound which is outputed from the oscillator originally has a high overtone element. The attenuation of the overtone element is expressed by lowering the cutoff frequency of the filter which is set high freqency, and being cut from one with a high overtone element gradually. The sound is brought close to the sound of real by lowering the cutoff frequency in chaos. The system proposed method is simulated by using the Csound.

The time change of the overtone element and the filter are shown in Fig. 1. The overtone element attenuates with time, and the inclination of the overtone element grows to attenuate as fast as the high frequency side.

The chaos waveform which is utilized in this proposed system obeys a principle of Lorenz Attractor which has many frequences. The Lorenz Attractor is one of the nonlinear equations that shows a chaos demeanour. The sound of the natural world has chaos waveform which has characteristicis felt warm to us.

The Csound is a programming language that treats the sound, and a text file of two kinds of special forms, orchestra file and a score file, is used as an input[5]. The orchestra file describes the characters of musical instruments, and the score file describes the aging parameter of the score[6]. The Csound executes the instruction group that exists in these files, and outputs the sound. The advantages of the Csound are a lot of the modules and the height of the extendibility by the user[7,8].

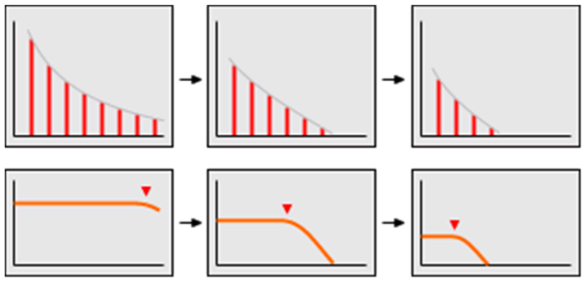

Figure 1: Time change and filter of overtone element

2. Algorithm

2.1. Lorenz Attractor

The Lorenz Attractor is defined by

$$dx/dt = \sigma(y - x), \quad (1)$$
$$dy/dt = -y - xz + \gamma x, \quad (2)$$
$$dz/dt = xy - \beta z, \quad (3)$$

where

$$\sigma = 10, \gamma = 28, \beta = 8/3. \quad (4)$$

The Lorenz Attractor enters the state of continuous chaos under the condition of Eq. (4). There are three point characteristic of Lorenz Attractor. First, the result is greatly different in the difference of a little initial value (initial value sensitive). Second, the orbit doesn't return to the same point (non-periodicity). Third, there is no order, but the orbit settles down in the inside of set.

The output of Lorenz Attractor is assumed to be "k1" which is defined by

$$k1 = x + y + z. \tag{5}$$

2.2. Algorithm of delay and release system

2.2.1. Input Sound

The frequency spectrum of a piano sound at the beginning of waveform which is outputted is shown in Fig. 2.

The frequency spectrum of a input sound created with the oscillator is shown in Fig. 3. To evaluate a output sound which is utilized the proposed system, the input sound modeled and produced the frequency spectrum of a piano sound.

Several oscillators are used for making the input sound. Each oscillator send a different respectively frequency by the volume of the proper quantity. The method of adding the oscillator is shown in Fig. 4.

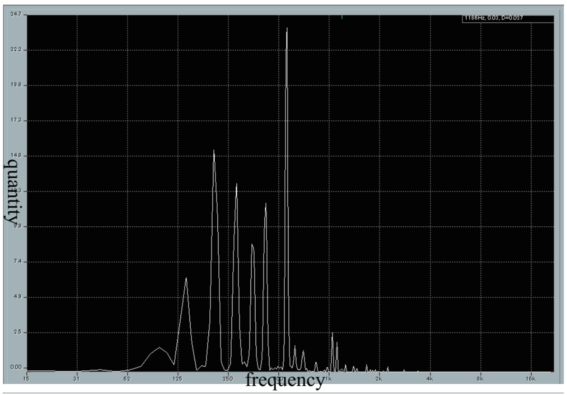

Figure 2: The frequency spectrum of a piano sound

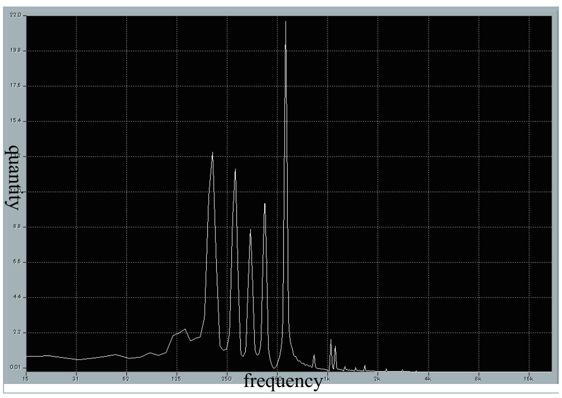

Figure 3: The frequency spectrum of the input sound

2.2.2. Filter

The block diagram of the proposed system is shown in Fig. 5. In "EXPON", a numerical value is changed with the time change, and the output is "kcut". The numerical value is changed from 700hertz to 0hertz in the simulation.

"TONE" gives the effect of the low-pass filter to the shape of waves of a input sound from the oscillator. The shape of waves which is outputed from "a1" of "OSCIL" is attenuated from the high frequency by the numeric change that depends on "EXPON" and "Lorenz Attractor". The shape of waves is outputed from "out".

In the simulation, the input sounds are compared by using the filter by "EXPON" or by "EXPON" and "Lorenz Attractor".

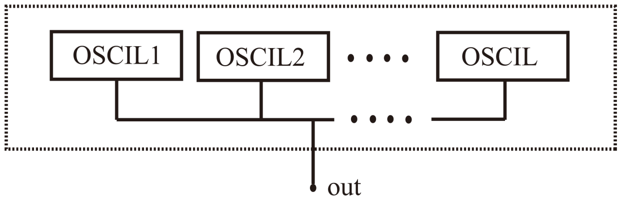

Figure 4: Block diagram of input sound

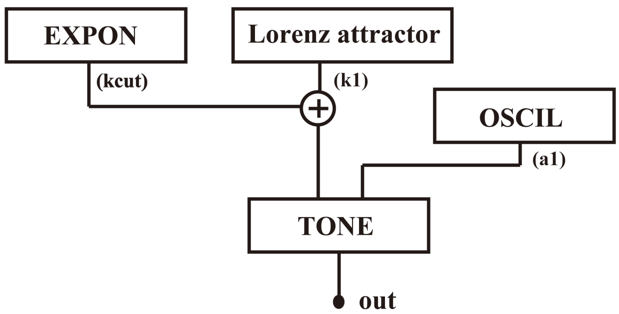

Figure 5: Block diagram of The Delay and Release System

3. Simulation results

3.1. Attenuation of volume

The passage of the time of a piano sound and the shape of waves of the volume are shown in Fig. 6. It is shown that the volume is attenuate with the passage of time.

The shape of waves of the time change and the volume which are outputed from the oscillator is shown in Fig. 7. The change of the volume is few.

The shape of waves of sound which is attenuated with the passage of time by "EXPON" is shown in Fig. 8.

The shape of waves of sound which is attenuated with the passage of time by "EXPON" and "Lorenz Attractor" is shown in Fig. 9. To make differences of Fig. 8 and Fig.

9 brought close, the shape of waves of Fig. 8 and Fig. 9 can be brought close to a piano sound by adding not only the filter but also a specific envelope to the input sound.

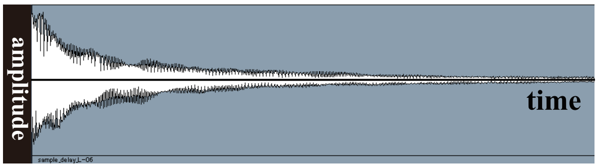

Figure 6: The shape of waves of piano sound

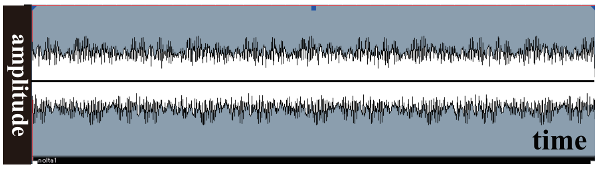

Figure 7: The shape of waves of input sound

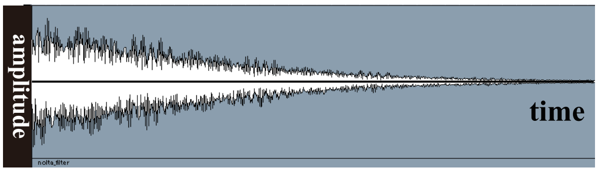

Figure 8: The shape of waves of sound by ”EXPON”

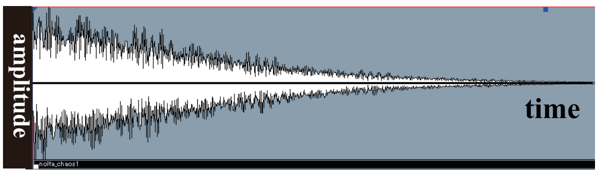

Figure 9: The shape of waves of sound by ”EXPON” and ”Lorentz Attractor”

3.2. Time change of frequency

The volume of the high and low frequency of a piano sound on the time change are shown in Fig. 10. At the beginning of waveform which is outputted, the volume of the high frequency is large, however it attenuates at once. The volume of low frequency attenuates slowly.

The volume of the high and low frequency on the time change when only the volume of the sound is attenuated with the passage of time is shown in Fig. 11. The high frequency and low frequency attenuate similarly respectively. When Fig. 10 and Fig. 11 are compared, it is shown that the time change of each frequency is greatly different.

On the time change, the volume of high and low frequency which is attenuated with passage of time by ”EXPON” is shown in Fig. 12. On the time change, the volume of high and low frequency which is attenuated with the passage of time by ”EXPON” and ”Lorentz Attractor” is shown in Fig. 13.

The ratio corresponding to the passage of the time of each high frequency and each low frequency in Fig. 10 and Fig. 12 and Fig. 13 is shown in Fig. 14. It is shown that the attenuation method using ”Lorentz Attractor” approaches a piano sound more than the attenuation method using only ”EXPON”.

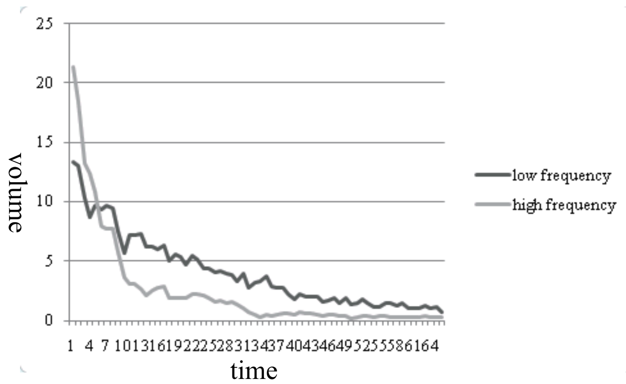

Figure 10: The high and low frequency of a piano sound

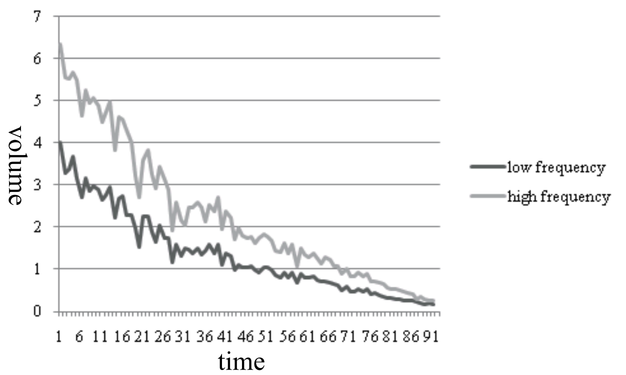

Figure 11: The high and low frequency of attenuated sound averagely

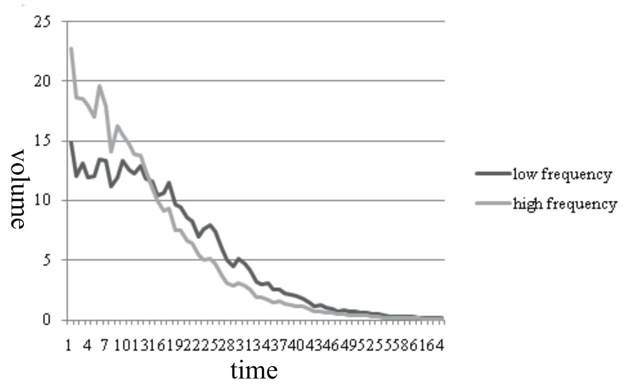

Figure 12: The high and low frequency of sound by "EX-PON"

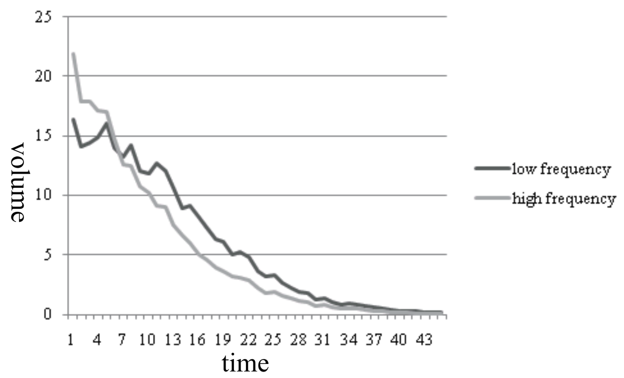

Figure 13: The high and low frequency of sound by "EX-PON" and "Lorentz Attractor"

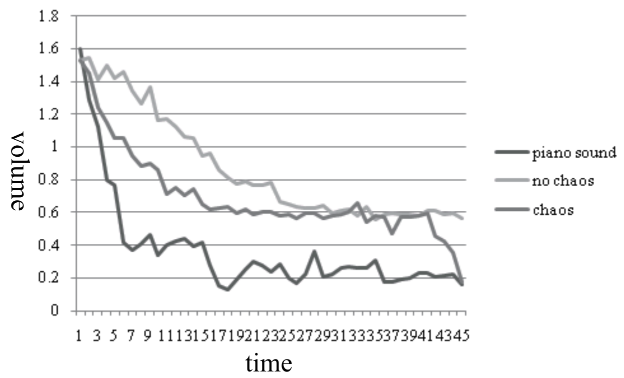

Figure 14: the ratio corresponding by Fig.10, Fig. 12, and Fig. 13

4. Conclusion

The proposed method is to bring the attenuation change in the frequency of a digital sound using several oscillators close to the attenuation change in the frequency of a piano sound that is the sample sound. In the ratio of high frequency (540hertz) and low frequency (205hertz), the method of moving the filter in chaos has approached the sample sound which is a piano sound more than the method of the average movement which do not use chaos. However, the change in the volume was different from the sample sound in the proposed system.

As future works, the system that executes the proposed method and the method of changing the envelope of the volume at the same time will be researched.

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