COSC 4P98 Project Report

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Description

For the COSC 4P98 project, I chose to work on creating a guitar amplifier simulation VST. Originally, I had created a one parameter distortion VST for the third assignment.

The majority of the concepts/ ideas I ended up getting rid of and starting from scratch due to a lack of interesting or useful distortion sound. Adding random samples to a wav results in some form of distortion, but mainly just becomes white noise and does not offer much more interesting or useful sound.

Initially I began with three built in sliders, a slider for pre-clipping, for clipping value, and for post volume. I moved the VST over to the surrounddelay example to connect these parameters to a GUI. Once I had achieved this, I was able to start customizing the interface, adding additional parameters, and other DSP effects with subsequent parameter control.

Since the concept behind the VST was to create a guitar amplifier and distortion VST, I created a GUI and VST to emulate that of what would be found on a standard guitar amplifier. The main left most section on the GUI represents an emulation of a standard overdrive/ distortion and as such can create a distortion similar to an amplifier's distortion.

The next section is the equalizer section, which has three parameters controls. These controls represent low, middle and treble frequencies allowing the user to cut or boost said frequency. The last section is additional distortion techniques that create distortion, but in a less traditional guitar amplifier way. The first parameter controls the level of bit quantization/truncation. The second parameter is a frequency distortion based on playback speed and can distortion a random range of frequencies. The third parameter is the percentage of the amount/value clipping during the distortion.

Lastly, the GUI is a edited image of a Peavey 6505 guitar amplifier head. This was done in order to create a GUI for the user, but also give a bit of an authentic feel to the VST's source of inspiration.

Background Information and Implementation

Regarding the extent of the background of this audio plugin, the main purpose is to provide user control of distortion and achieve sounds similar to that of a guitar amplifier. As such, the main concepts of DSP in this plugin deal with distortion and equalization.

Distortion

The main concept behind the distortion method is the idea that once the source signal reaches a certain threshold, the signal is then truncated and set to the value of the threshold/ limit. This effectively creates a hard clipped square out of the original wave, cutting off at the truncation point.

Since a sound wave has both positive and negative wave signal value, the signal must be clipped at both the positive and negative sides. As such similar to previous year's distortion projects, I have parameterized the negative and positive truncation values for which the signal is clipped at. This allows the user to alter the signal clipping effect in many different ways.

Equalization

An equalizer is a common component on a guitar amplifier, providing control to manipulate the source signal frequency response. As such, I felt it was necessary to include at least a three band equalizer, a low frequency band, mid frequency band, and a high band. Essentially the high band can be cut to create a low pass filter, or the low band can be cut to create a high pass filter. Alternatively you can cut out the mid frequencies, which is common among high gain amplifier tone/ timbre.

To provide user a better control of the output signal sound, a three band equalization was added. This was based heavily off a three band equalization tutorial. The eq works by having a struct (EQSTATE) hold information on the eq values, frequencies poles (low, mid, high) as well as gain values which are controlled by the three eq sliders in the GUI. An object of type EQSTATE is created and initialized with the low frequency of 880Hz and high frequency of 5000Hz. The low and high poles are calculated by the equation

```
"2 * sin(M_PI * ((double)frequency / (double)SampleRate));"
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and are set when eqstate is initialized. During this point, the slider values are mapped to initializing the low, mid and treble gain values of the struct. If all the values are at 100%, the equalization function will return the original value as there is nothing to additionally compute. Otherwise the lowpass, highpass, midrange frequency poles are computed based on the changed slider values. The history buffer of the struct is updated with the current sample. The function finally returns the combined low, mid and high values at the updated frequency computed.

Additional Distortion

Lastly, the final two sliders control the less conventional types of distortion. These two are bit quantization, and frequency distortion.

Quantization

The idea behind quantization is to reduce the number of bits representing the same audio signal. As such, the lower the number of bits the low in sound quality will be and the more distorted and noisy it will become.

Frequency Distortion

The frequency distortion results in a distortion of random frequencies when the slider is adjusted. This is done by having a global wave int value that begins at -10 and is incremented each time the frequency distortion function is called. This happens until the was is less than equal to 20, at that point the wave int will be reset to -20. The local wav value is calculated to be

wav = (wav + (20*playback));

which is the value the wav should increment at, times the percent of the value assigned by the parameter.

Truncation (Distortion)

On its own, this particular slider does not affect the signal. This slider works in conjunction with the distortion side of the VST, and affects the output clipped signal. As mentioned with distortion in the traditional methods, the signal is clipped when it reaches or exceeds the clipping value.

Generally the sample at that point is set to equal the clipping value to achieve a light distortion, however the truncation value affects the percentage of the truncation on the signal. For example, if the clipping value was at a value of four and the truncation value was set to 50%, the resulting sample value would be two at each point the signal "clips". This can create a unique digital style of distortion when set to very low numbers, as it essentially creates a signal with a large range of values, depending on the clipping value of course.

User Manuel:

To use the VST plugin, it must be loaded into a VST host Digital Audio Workstation. In this example I will be using Reaper to demonstrate the VST.



To achieve a basic overdrive/ distortion using the sample provided, set the preclip value to 50%, and the negative and positive clip values also to 50% as shown.



The lower the clipping values, the greater the amount of clipping occurs. The volume fader acts as a global fader for the entire VST, so if you have a preclip value of 100% with a low clip values (2-3%) you can boost the signal with the volume slider should need be. For example, for a heavier sounding distortion try a preclip of 100%, clipping faders at 2% and volume at 80%.

Sample Mix Settings

To mix achieve the same mix as Sample_Mix_Final.mp3, follow each step on the tracks provided in "Samples\Sample Track\Dry_Tracks". For both guitar, set the preclip values to 100%, both clipping values to 3% and the volume to 75%. Since the scope of the VST was to simulate a guitar amplifier and not a cabinet, included in the "Samples\Sample Track\Impulse_Response_Loader" is a cabinet IR loader VST with an included impulse response "57B C_00.wav". Use the impulse response loader on the same guitar track, loading the impulse response provided setting the mix to 100% and the gain to -8.8dB. Apply this to the second guitar track, and pan one guitar left and the other right. Also apply these settings to the bass track for desired distortion and include the " drum_track.mp3" done in EZDrummer.

By default the Equalization, Quantize, frequency and truncate are set to by pass in their default positions. As such returning the faders to their default position will result in a bypass of the calculation (resulting in the original signal).

Equalization

Low Pass filter -

In order to create a low pass filter, simply move the slider for the high/ treble frequencies to zero. This will only allow the middle and low frequencies to be passed through.



Mid Boost- For boosting the mid values frequencies, the EQ is set to 40% low, 88% mid, and 40% high



Quantize

To get the quantization effect move the slider down, roughly around 30-40% for audible distortion.



For the second sample included, the quantization has been moved to 18% resulting in only a slight amount of sound.

Frequency

Due to the randomness of this effect, there are several samples to demonstrate the distortion is can create.

This first style of distortion creates a mid-heavy crunchy chiptune style of sound when set to around 4% on the frequency bar.



Frequency sample 2 is the exact same setting except with the frequency fader set to 43%. At this setting, the frequency function creates a grainy high-end sound reminiscent of low quality recordings.

Frequency sample 3 is at same settings but with frequency fader at 68%

Frequency sample 4 is at same settings but with frequency fader at 96%

Truncation

As stated previously the truncation function works by multiplying the percentage of the fader value by the value truncated(clipped). In order to demonstrate this, the fader value is set to 0% which means each time the signal is equal to or greater than the clipping value, it is set to clip value*truncation (in this case zero). Also the preclip is set to 100% and both clip values are set to 60%.



Bibliography

3-band Equalizer

http://learnvst.wordpress.com/ http://learnvst.wordpress.com/old-site/weq-plugin-project/ http://en.wikipedia.org/wiki/Equalization_%28audio%29

Surround Delay VST 2.x SDK Sample

https://www.cosc.brocku.ca/Offerings/4P98/lectures/VST_examples/adelay/

VST SDK 2.4/ VST GUI

http://www.cosc.brocku.ca/Offerings/4P98/vst/

Distortion (Concept)

http://en.wikipedia.org/wiki/Distortion_%28music%29

Quantization

http://en.wikipedia.org/wiki/Audio_bit_depth

keFIR Impulse Response loader (for cabinet simulation in mix) Toontrack EZDrummer - Drums in Sample_Mix_Final.mp3

GUI image (Modified Peavey 6506+ head)

http://peavey.com/