Guitar Effects Generator Using DSP
Functional Requirements List and Performance Specifications

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Introduction

This project involves developing a guitar effects generator utilizing a digital signal processor. A guitar sends an audio signal to a DSP using a cable. The processor then converts the signal to digital, allowing specific filters to act upon the signal, creating and adding effects to the audio. The signal then passes through a converter to make it analog again, allowing it to output to a guitar amplifier and produce the sound. A user selects which effects are desired through an interface, creating the pathway the signal follows. The effects will be generated first on recorded guitar audio. Once these tests prove successful, the effects will be attempted on real-time guitar sounds.

Goals

The project has the following goals to keep it moving forward:

- Filter out single-coil pickup noise (approx. 60 Hz)
- All effects after the noise filter are user-defined, meaning that effects not wanted by the user shall be bypassed
- A distortion model that boosts and clips the signal at specific maximum and minimum values
- Create audio reverberation simulation
- Create digital delay and echo
- Change the signal to be an octave higher than played
- Generate an automatic volume swell that is modified by how fast it reaches maximum volume
- Produce Chorus effect to make the guitar sound like multiple guitars
- Create a whooshing sound within the signal using either delay lines (flanging) or shifting the phase of the signal (phaser)
- Develop a GUI for user control

Other effects may be added later on if time allows it. Possible future effects are acoustic guitar modeling, humbucker modeling for single-coil pickups, single-coil modeling for humbucker pickups, "tube" amplifier distortion, and auto-wah.

High-Level Block Diagram

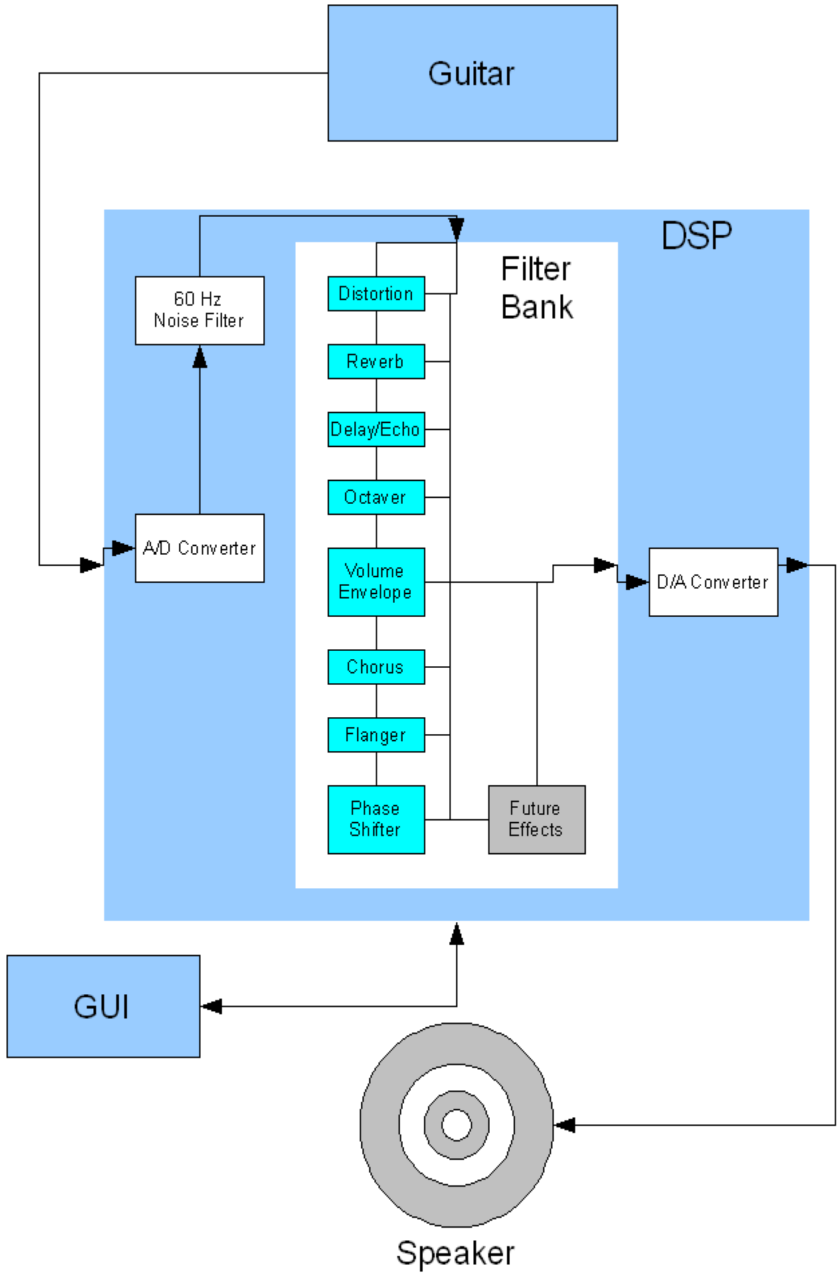

Figure 1: High-Level Block Diagram of Guitar Effects Generator

The signal comes from the guitar and goes to the DSP. It is converted to a digital signal first, then it passes through a noise filter. From the noise filter, it goes to the user-defined effects filters. After this, the modified signal is converted back to analog and is sent to the guitar amplifier for sound generation. The filters shall be controlled through a graphical user interface so that the guitar player can select which filters shall function and to what degree.

Overall DSP Requirements

The DSP system shall convert, process, and reconvert in 1 ms. This speed is for both recorded signals and real-time processing; it shall be fast enough not to cause any noticeable delay. The system shall handle all human-audible frequencies, namely the range of 20 Hz to 20 kHz. It shall filter out noise from single-coil pickups, which is at 60 Hz.

GUI Requirements

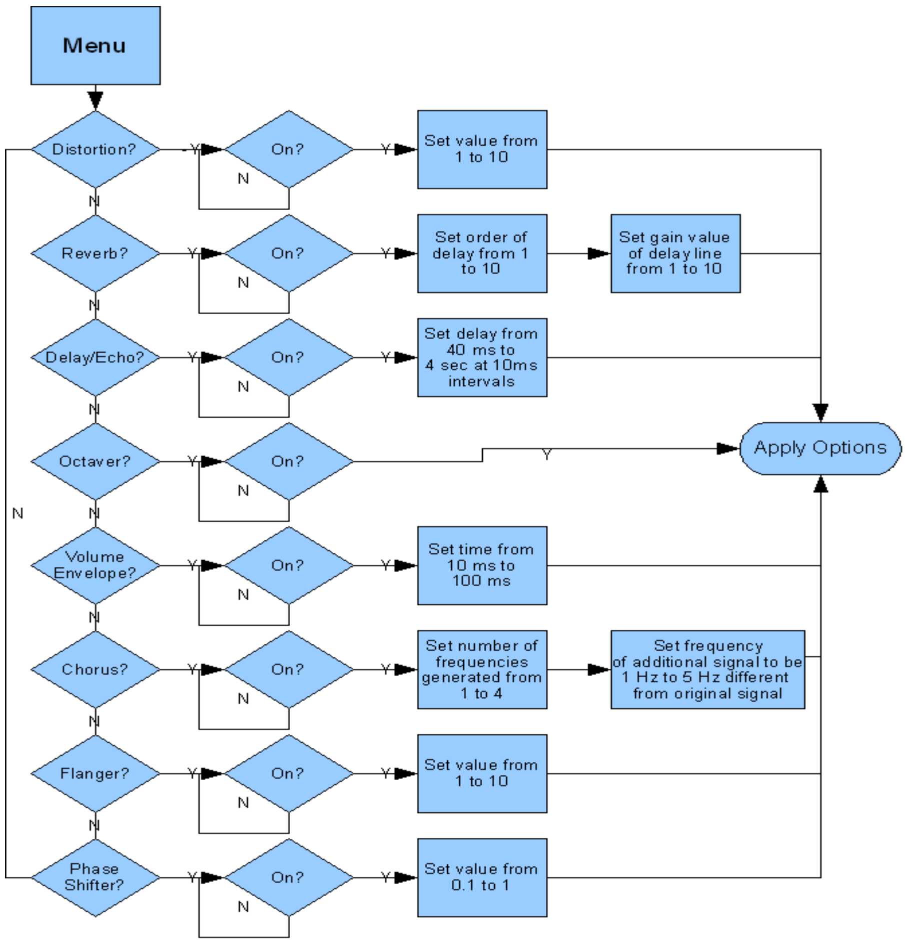

Figure 2: GUI Interface Flowchart

The GUI shall contain a drop-down menu containing each filter. The default filter in the menu shall be distortion. Each filter selection shall have different options regarding the filters and a check box to designate if the filter is on or off. Distortion shall contain a slider and text box to determine the amount of clipping that shall occur. Reverberation shall contain two value inputs – one for the delay order and one for the gain in the delay line; the inputs shall be determined by either a slider or direct input of values. Delay/Echo shall contain a slider and text box controlling the duration of the delay. Octaver shall only have the check box. Volume envelope shall contain a slider and text box controlling how fast the sound reaches full potential after the string is struck. Chorus shall contain a text box controlling the intensity of the effect. Flanger shall contain a slider and text box adjusting the delay rate of the signal. Phase shifter shall contain a slider and text box controlling the intensity of the phase change.

A/D Converter Requirements

The A/D Converter shall convert to a minimum of 16 bits. Audio signals are typically found in this resolution, although a 24-bit converter may be used if a noticeable improvement in the quality of the signal is found. It shall have a minimum sampling rate of 44.1 kHz and a maximum sampling rate of 96 kHz.

Noise Filter Requirements

The noise filter shall be set to attenuate the 60 Hz noise found in single-coil pickups. In order to allow all audio frequencies other than the noise to pass through, the noise filter shall be a notch filter with the notch at 60 Hz. The gain at the other frequencies shall be set at 1 so that the filter does not distort the incoming signal.

Distortion Filter Requirements

This filter shall allow the user to decide what the amplitude limits will be and what the gain amount shall be. The gain amount shall boost the signal, and the amplitude limits shall clip the signal, creating distortion in the signal. The values for gain shall range from 1 to 10 units, with 1 causing minimal clipping and 10 causing maximum clipping.

Reverb Filter Requirements

The magnitude response in the frequency domain shall be 1 for all frequencies. The user shall decide the delay and gain block values for the filter. Both shall be on scales of 1 to 10 units. For the delay block, the units provided shall determine the duration of the reverberation; these numbers shall be directly related to the order of the delay block (for example, 1 shall set the order to 1, and 10 shall set the order to 10). For the gain block, the units shall determine how the reverberation is sustained – the higher the gain value, the longer the sustain shall be. The values shall be related to the gain of the system by a factor of 0.1 from 1 to 9, with 10 representing 0.95 gain.
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Delay/Echo Requirements

The delay/echo filter shall determine the next occurrence of the signal played. For instance, if one note is hit, not only shall it play when it is hit, but it shall play again at the determined time. The time range the user can input shall be from 40 millisecond to 4 seconds at 10-millisecond intervals.

Octaver Requirements

The octaver filter shall act as a full-wave rectifier. This will double the frequency, causing the note to sound one octave higher. There shall only be a selection of on or off for this filter.

Volume Envelope Requirements

The volume envelope filter shall allow the signal to gradually reach full value, taking out the initial attack of the notes. The sound is similar to a note played backwards. The time for the signal to reach full value the user can input shall be a range from 10 milliseconds to 100 milliseconds at 1-millisecond intervals.

Chorus Requirements

The chorus filter shall allow the signal to generate another signal, one that is at a frequency slightly higher than the original. When combined, this shall create a multiple-guitar sound. The user shall determine how many other signals are generated, and what frequencies these signals shall be. The frequencies shall only be 0.1 to 0.5 Hz above the original signal, values chosen at every 0.01 Hz.

Flanger Requirements

The flanger filter copies the signal and delays the copy by varying values less than 20 milliseconds. The signal is then added back to the original signal, creating a audible sweeping effect. The limit of the delay shall be between 5 and 15 milliseconds. The user shall determine how fast the delay changes at a scale from 1 to 10 units. The value '1' shall represent the delay changing 1 millisecond every 2 seconds, and the value '10' represents the delay changing 1 millisecond every 200 milliseconds.

Phase Shifter Requirements

The phase shifter filter acts in a similar way to the flanger. The signal is copied, the copy is modified, and the two signals are added together. However, in this case, the phase is shifted on the copy rather than delaying the signal. This shall be created by passing the copied signal through eight cascaded all-pass filters with a feedback loop. The user shall configure the filter by choosing the depth of the notches created in the frequency response when the two signals are added. The depth is determined by a gain block at the end of the copied signal's path, with the gain of 0.1 being the minimum and a gain 1 being the maximum. The value shall be changed at 0.05 gain increments.

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