Guitar Effects Generator Using DSP
Senior Project Proposal

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Project Summary

This project deals with the creation of sound effects through manipulation of an audio signal from a guitar. The signal is processed through a digital signal processor that contains a number of digital filters to modify the signal to include the desired effects. The altered signal is then sent to a guitar amplifier as audio. There are a total of eight guitar effects that can be processed: Distortion, Reverberation, Delay/Echo, Octaver, Volume Envelope, Chorus, Flanger, and Phase Shifter. These effects are controlled by a graphical user interface, allowing the user to select which effects are active and to what degree.

Detailed Description

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:

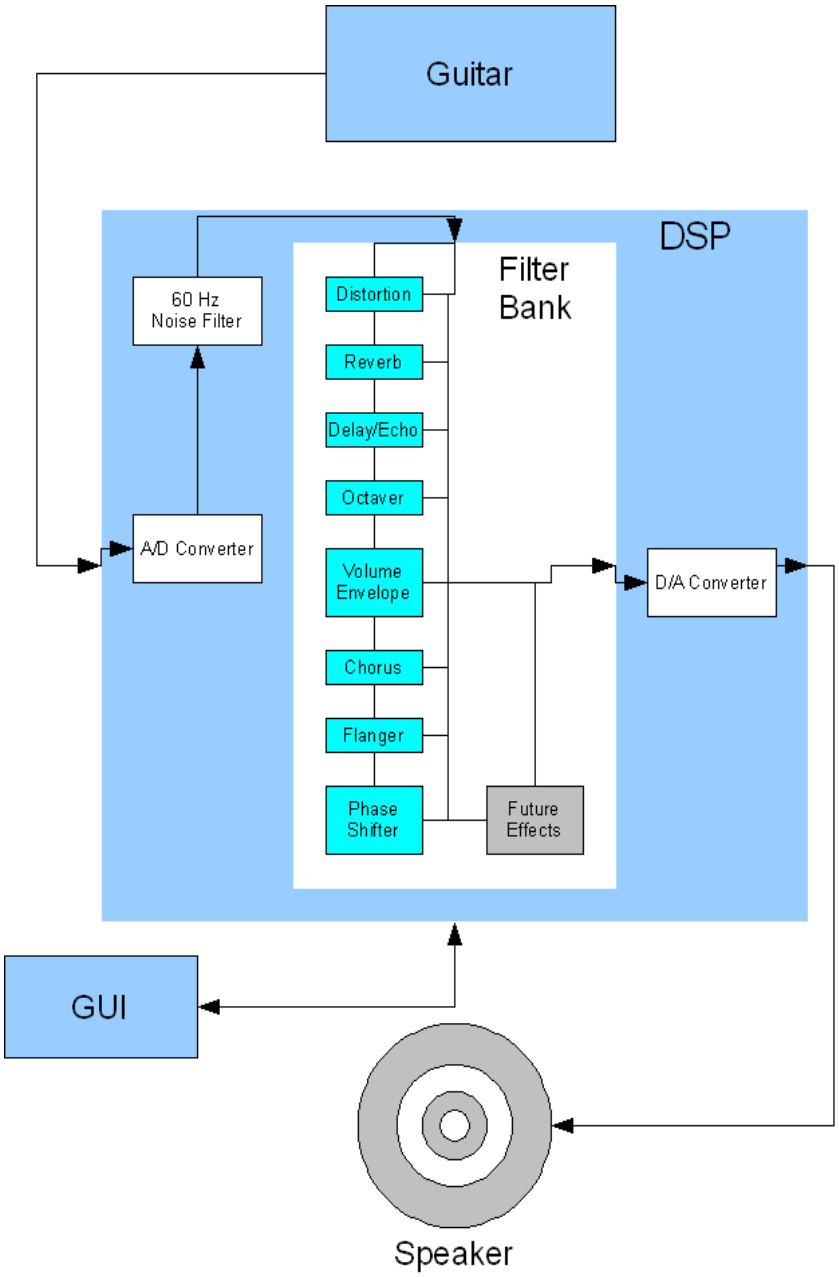

Figure 1: High-level Block Diagram of Guitar Effects Generator Project

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. Figure 1 shows how the project works graphically.

Guitar Descriptions and Specifications:

Two guitars shall act as the primary signal generators. One is a Squier Stratocaster, and the other is a Squier Telecaster Custom. The Stratocaster contains three single-coil pickups; these pickups present a lean and clear sound, but they produce a 60-Hz noise signal along with the audio signal. The Telecaster contains two humbucker pickups; essentially two single-coils with opposite polarities to cancel the hum, the humbucker outputs a warmer and broader sound. Figures 2 and 3 are the specifications of both guitars.

Model Name	Strat® (Rosewood)
Model Number	031-0600-(Color#)
Series	Affinity Series
Colors	(506) Black, (525) Metallic Red, (595) Metallic Blue, (Polyurethane Finish)
Body	Alder
Neck	Maple
Fingerboard	Rosewood, 9.5” Radius (241 mm)
No. of Frets	21 Medium Jumbo
Pickups	3 Single-Coil Pickups
Controls	Master Volume, Tone 1. (Neck Pickup), Tone 2. (Middle Pickup)
Pickup Switching	5-Position Blade: Position 1. Bridge Pickup Position 2. Bridge and Middle Pickup Position 3. Middle Pickup Position 4. Middle and Neck Pickup Position 5. Neck Pickup
Bridge	Synchronous Tremolo
Machine Heads	Standard Die-Cast Tuners
Hardware	Chrome
Pickguard	1-Ply White
Scale Length	25.5” (648 mm)
Width at Nut	1.61” (41 mm)
Unique Features	Large Headstock ‘60s Style Headstock, White Plastic Parts, Black Silkscreen Logo, Dot Position Inlays
Strings	Fender Super 250L, (.009 to .042) Nickel Plated Steel p/n 073-0250-003
Accessories	None
Introduced	1/2001
Notice	Product Prices, Features, Specifications and Availability Are Subject To Change Without Notice

Figure 2: Squier by Fender Stratocaster Affinity Series Specifications[1]

Model Name	Vintage Modified Telecaster® Custom
Model Number	032-7502-(Color#)
Series	Vintage Modified Series
Colors	(506) Black (Polyester Finish)
Body	Agathis
Neck	Maple, C-Shape, (Gloss Polyurethane Finish)
Fingerboard	Maple, 7.25” Radius (184 mm)
No. of Frets	22 Medium Jumbo Frets
Pickups	2 Chrome Covered Humbucking Pickups (Neck/Bridge)
Controls	Volume 1. (Neck Pickup), Tone 1. (Neck Pickup), Volume 2. (Bridge Pickup), Tone 2. (Bridge Pickup)
Pickup Switching	3-Position Toggle: Position 1. Bridge Pickup Position 2. Bridge and Neck Pickups Position 3. Neck Pickup
Bridge	6-Saddle Strings-Thru-Body Tele Bridge
Machine Heads	Standard Die-Cast Tuners
Hardware	Chrome
Pickguard	3-Ply Black
Scale Length	25.5” (648 mm)
Width at Nut	1.650” (42 mm)
Unique Features	Dot Inlays
Strings	Fender Super 250L, Nickel Plated Steel, (.009 to .042), p/n 073-0250-003
Accessories	None
Introduced	7/2003
Notice	Product Prices, Features, Specifications and Availability Are Subject To Change Without Notice

Figure 3: Squier by Fender Telecaster Custom Specifications[2]

DSP Description and Specifications:

The DSP contains converters so that the signal can become digitized for effects processing and then return to analog form for audio representation – an A/D converter and a D/A converter. A noise filter follows the A/D converter, eliminating the noise inherent in single-coil pickups. The effects filters are connected to allow certain effects to be utilized when selected and other effects to be bypassed when not chosen. The number of effects planned are eight, but time may allow for more effects filters to be designed. The effects filters lead to the D/A converter.

The DSP system shall convert, process, and reconvert in the inverse of the sampling frequency. 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.

Guitar Amplifier Description and Specifications:

A Fender Frontman 15R Guitar Amplifier shall be used to output the audio. It has a reverberation potentiometer on it, but this will be set to 0 so that the designed reverberation effects can be tested. It also has a distortion channel, but this will be ignored to test out the designed filter. Figure 4 contains the specifications of the amplifier.

Model Name	Frontman™ 15R
Model Number	023-1501-000
Series	Frontman Series
Type	Solid State
Output	15 watts into 8 ohms
Ohms	8 ohms
Speakers	1-8" Fender® Special Design Speaker, p/n 0025421000
Channels	Dual Selectable Channels (Normal and Drive)
Features	Reverb, 3-Band EQ, Headphone Jack, Auxilliary Input for CD, Tape or Drum Machine, Closed Back, Blackface Styling with Silver Grille Cloth
Controls	Volume Normal Channel, Gain, Drive Select Switch, Volume Drive Channel, Treble, Mid, Bass, Reverb
Covering	Black Textured Vinyl with Silver Grille Cloth
Weight	15 lbs. (6.80 kg)
Dimensions	Height: 12.5" (31.8 cm), Width: 13.25" (33.65 cm), Depth: 7.25" (18.41 cm)
Power Handling	N/A
Tube Complement	N/A
Cover	None
Accessories	None
Footswitch	None
Introduced	1/2004
Notice	Product Prices, Features, Specifications and Availability Are Subject To Change Without Notice

Figure 4: Fender Frontman 15R Guitar Amplifier Specifications[3]

GUI Specifications:

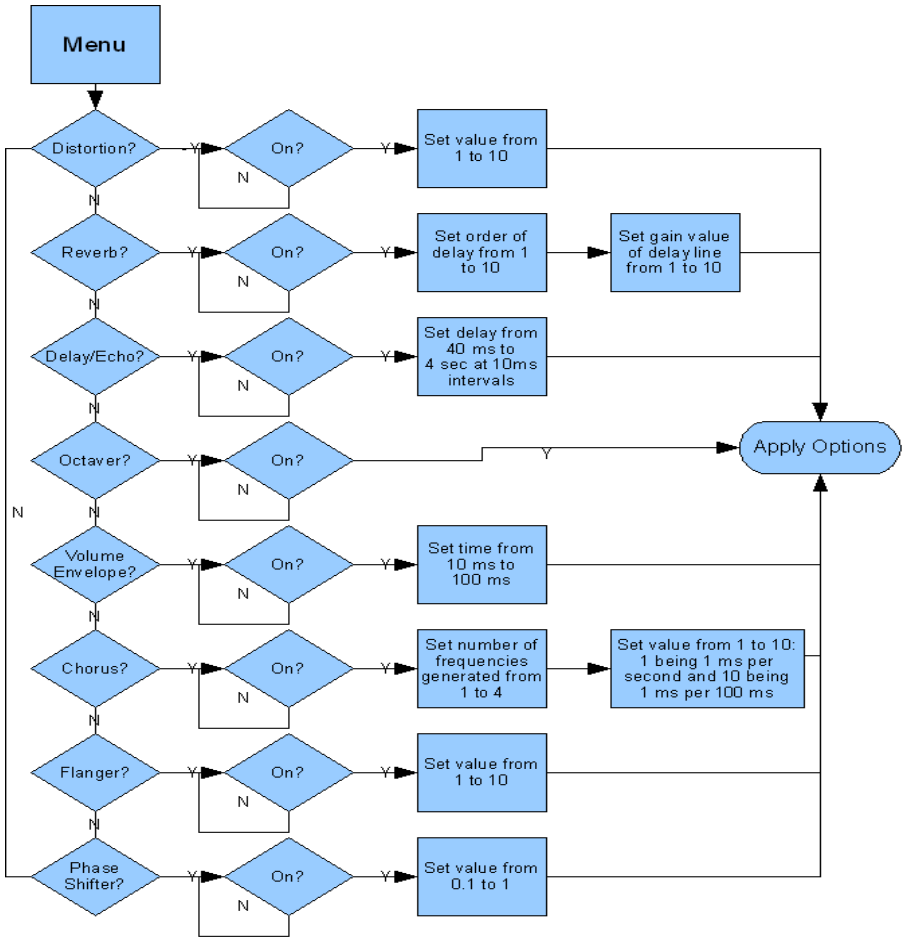

Figure 5: Flowchart of GUI Interface

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 drop-down box containing how many times the signal shall be copied and a slider for each copied signal determining how much the delay rate varies.. 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. Figure 5 acts as an overall software flowchart of the GUI and a flowchart of the menu's functionality.

A/D and D/A Converter Requirements:

The A/D and D/A converters shall sample either at 44,100 samples per second or 48,000 samples per second. The typical audio conversion to digital occurs with these two sampling rates because they are greater than twice the highest audible frequency, which is around 20 kHz. Typically, audio is converted to a 16-bit digital representation, so this shall be the complexity of the digital signal.

Effects Specifications:

Each effects filter has its own set of requirements and specifications. All shall function as all-pass filters aside from the noise filter. The individual effects' specifications are as follows:

- Noise Filter: 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: 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.
- Reverberation: 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
- Delay/Echo: 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: 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: 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: The chorus filter shall allow the signal to generate another signal, one that is delayed variably from 30 to 40 milliseconds. When combined, this shall create a multiple-guitar sound. The user shall determine how many other signals are generated, and how fast the the delay varies for each signal. A maximum of 4 copies shall be made, and the range of variation shall be from 1 to 10 units, with 1 representing a 1-millisecond shift every second and 10 representing a shift every 100 milliseconds.

- Flanger: 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: 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.

These effects shall be connected in a chain, but the user shall have the option to select which effects shall be used. To allow this, the filters shall be in parallel and in series with simple switch indicators determining whether the effect is bypassed.

Schedule

The preliminary schedule for the project consists of winter break and spring semester. Fall semester consisted of research of the topic and understanding how analysis of audio waves works in MATLAB and Simulink. The schedule is divided between the two partners.

Week and Date	Alex Czubak	Gorav Raheja
Week 0: Winter Break 2007/2008	Research of all-pass filter design, clipping signals, frequency change	Research of GUI design and programming
Week 1: January 24, 2008	Design and test of Noise Filter, Aid in GUI design	Begin programming of GUI
Week 2: January 31, 2008	Design and Test Reverberation Filter	Program GUI
Week 3: February,7,2008	Design and Test Distortion Filter	Program GUI
Week 4: February 14, 2008	Design and Test Octaver Filter	Program GUI, Aid in Octaver Filter Design
Week 5: February 21, 2008	Design Volume Envelope Filter	Program GUI, Aid in Volume Envelope Filter
Week 6: February 28, 2008	Continue Design of Volume Envelope and Test, Help with GUI	Program GUI
Week 7: March 6, 2008	Finish Testing Volume Envelope, Design Flanger Filter	Program GUI
Week 8: March 13, 2008	Test Flanger, Begin design of Phase Shifter Filter	Debug GUI, Look into design schemes for the Chorus Filter
Week 9: Spring Break	Research Phase Shifter Design	Research Chorus Design
Week 10: March 27, 2008	Design and Test Phase Shifter	Design Chorus Filter
Week 11: April 3, 2008	Design Delay/Echo Filter	Design and Test Chorus Filter
Week 12: April 10, 2008	Design and test Delay/Echo Filter, Begin Real-time audio resting	Begin Real-time audio testing
Week 13: April 17, 2008	Real-time implementation	Debug GUI, Real-time audio testing
Week 14: April 24, 2008	Real-time implementation, Work on presentation and paper	Work on presentation and paper
Week 15: May 1, 2008	Presentation	Presentation

Figure 6: Preliminary Schedule for Project and Project Division

The schedule is subject to change, but this is the current outlook of the coming weeks.

Annotated References

[1] "http://www.squierguitars.com - Strat (Rosewood)." Squier Guitars by Fender. 10 Dec. 2007 <http://www.squierguitars.com/products/view_specs.php?full_partno=0310600&name=Strat %26reg%3B+%28Rosewood%29>.

[2] "http://www.squierguitars.com - Vintage Modified Telecaster Custom." Squier Guitars by Fender. 10 Dec. 2007 <http://www.squierguitars.com/products/view_specs.php?full_partno=0327502&name=Vintage+Modified+Telecaster%26reg%3B+Custom>.

[3] "http://www.fender.com - Frontman 15R." Fender.com. 10 Dec. 2007 <http://www.fender.com/products//view_specs.php?full_partno=0231501000&name=Frontman %26trade%3B+15R>.

Other References

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Karjalainen, Matti, Henri Penttinen and Vesa Valimaki. IEEE. "Acoustic Sound from the Electric Guitar Using DSP Techniques." 2000: Helsinki, Finland.

Fernandez-Cid, Pablo and Javier Casajus-Quiros. IEEE. "Multiband Approach to Digital Audio FX." 2000: Madrid, Spain.

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Keen, R.G. Guitar Effects FAQ. May 20, 2000. <http://www.geofex.com/effxfaq/fxfaq.html>

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Equipment Used

- MATLAB Software w/ Simulink and Signal Processing Toolbox
- Texas Instruments DSP board
- Computer used in Lab
- Fender Frontman 15R Guitar Amplifier
- Squier Stratocaster Affinity Series Electric Guitar
- Squier Telecaster Custom Series Electric Guitar