Digital Guitar Effects in MATLAB

Signal Processing Implementation & Spectrogram Analysis

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1. Introduction

This project demonstrates a set of audio effects implemented entirely in MATLAB using signal processing fundamentals. Rather than relying on external plugins or toolboxes, the effects were created through direct manipulation of discrete-time signals, emulating classic guitar effects such as delay, reverb, chorus, flanger, and distortion.

The input is a raw waveform of a clean electric guitar recording. Each effect is applied digitally, and its result is visualized through time-frequency analysis using spectrograms. This report provides a technical summary of each effect along with its audible and visual characteristics. The input guitar signal used in this project is an isolated solo section from an original song by my band, *Buluşalım Rüyalarda- Siyah Orkide.*

This solo was performed, recorded, and processed by the author.

Audio Samples: Audio files for each effect are available here: Google Drive

2. Methodology

Each effect was implemented in MATLAB using time-domain operations. The following sections outline the core logic behind each effect, accompanied by observations of their impact on the signal's spectrogram.

3. Delay

Effect description: A delayed copy of the signal is added to the original, creating an echo-like result. The delay time is adjustable in samples.

Core equation:

$$y[n] = x[n] + a_0 \cdot x[n - D]$$

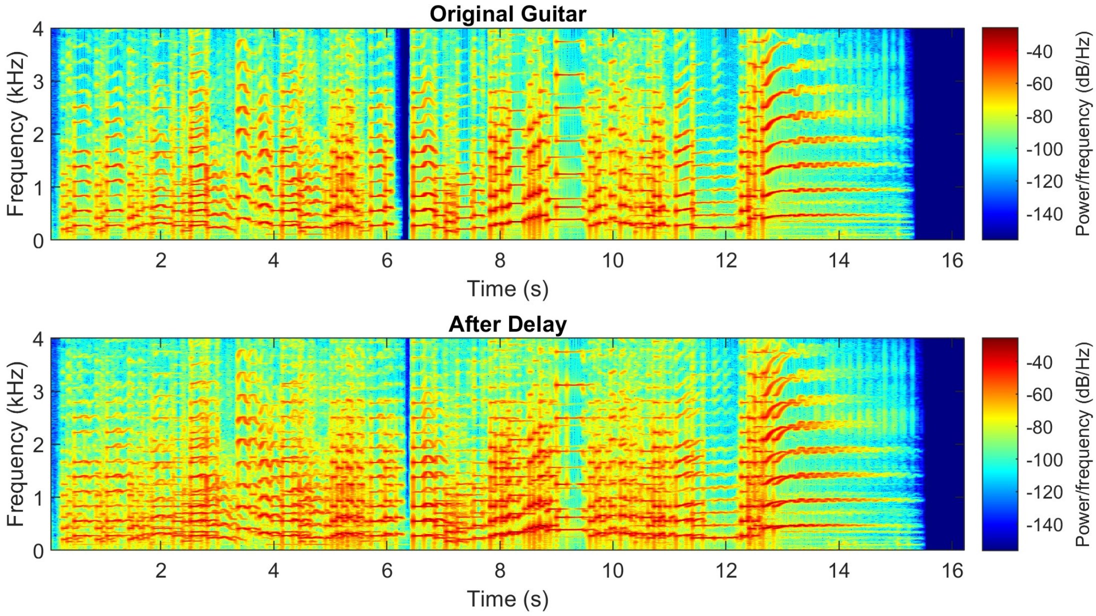

Figure 1: Spectrogram of the guitar signal with delay effect

Spectrogram Observation – Delay

Repeating diagonal traces appear at regular intervals, indicating temporal echoes. Frequency content remains unchanged, but delayed energy bands are visible.

4. Reverb

Effect description: Multiple delayed and attenuated copies of the input are layered to simulate spatial reflections, resulting in a more ambient and sustained sound.

Core equation:

$$y[n] = x[n] + a_1 \cdot (x[n - D] + a_0 \cdot x[n - 2D])$$

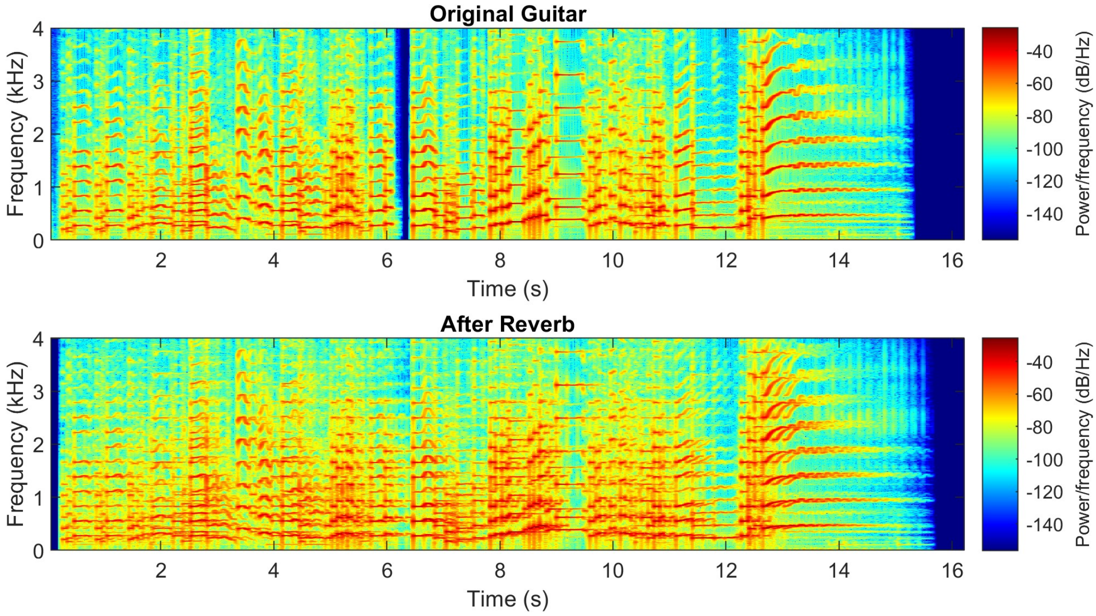

Figure 2: Spectrogram of the guitar signal with reverb effect

Spectrogram Observation – Reverb

Frequency components become smeared over time — vertical edges in the original become blurred. Sustained harmonics and a cloud of energy appear in the higher frequencies.

5. Chorus & Octaver

Effect description: This effect mixes the original signal with pitch-shifted versions, creating a layered, richer tone. Semitone shifts are applied using a time-scaling technique.

Core concept:

$$\alpha = 2^{\delta/12}, \quad y[n] = x[n] + x[\alpha n]$$

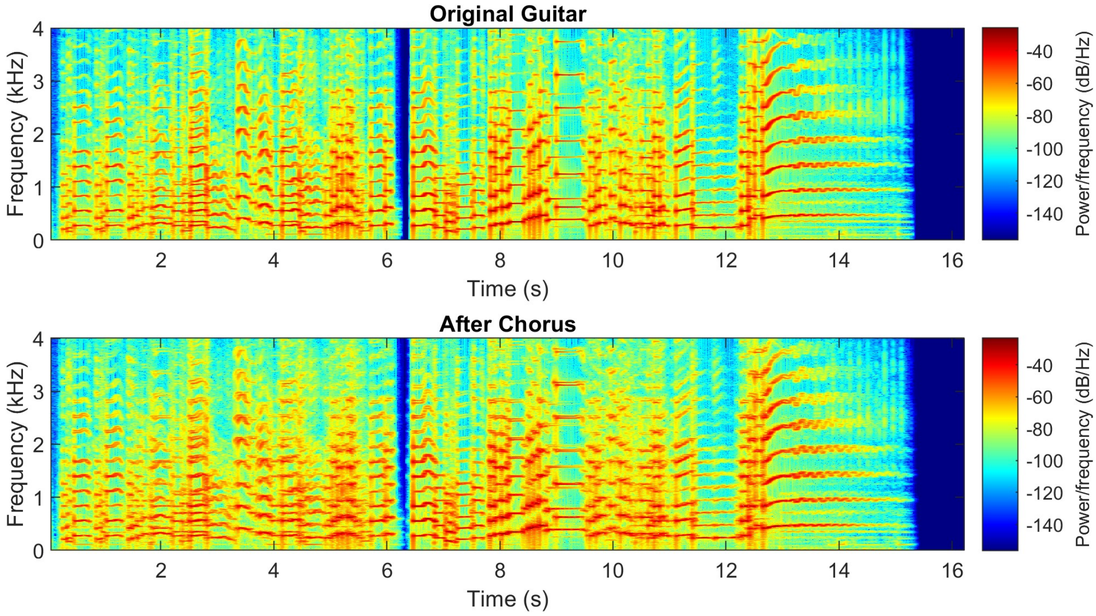

Figure 3: Spectrogram of the guitar signal with chorus effect

Spectrogram Observation – Chorus

Additional frequency bands appear slightly above or below the original harmonics. The result is a thicker, modulated frequency distribution.

6. Flanger

Effect description: A short delay is modulated with a low-frequency oscillator (LFO), creating phase cancellations and comb-filtering effects.

Core equation:

$$y[n] = x[n] + x[n - D(t)] + g \cdot y[n - D(t)]$$

Where:

$$D(t) = D_{\max} \cdot (1 + d \cdot \sin(2\pi f_{\mathrm{LFO}} t))$$

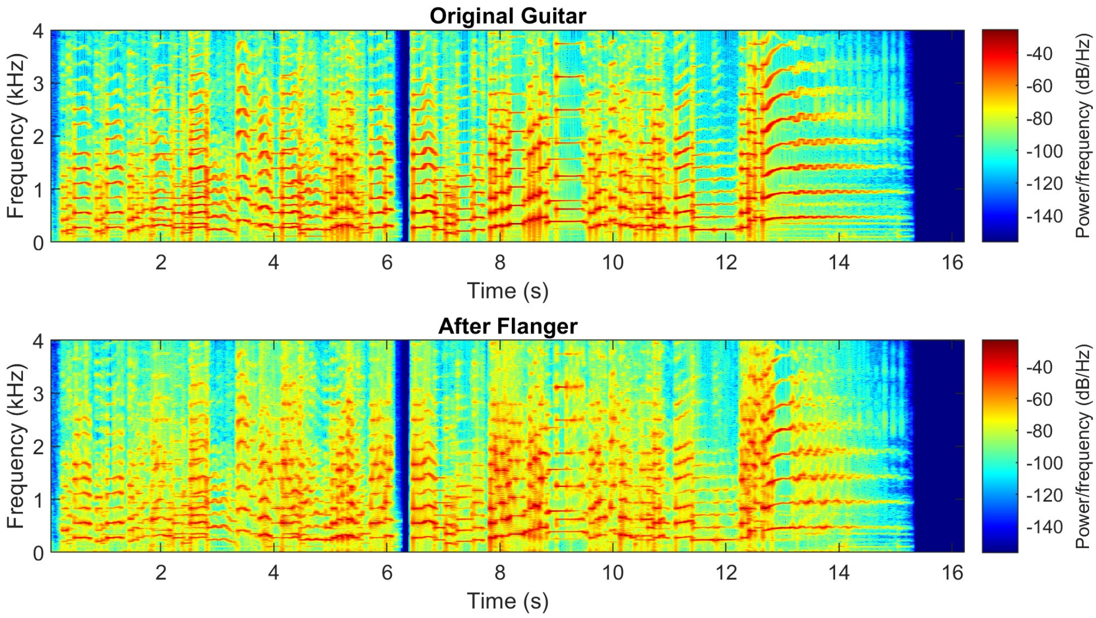

Figure 4: Spectrogram of the guitar signal with flanger effect

Spectrogram Observation – Flanger

Repeating notches and sweeping wave-like interference patterns occur in the frequency domain, caused by LFO-modulated comb filtering.

7. Distortion

Effect description: Clipping the waveform introduces additional harmonics, resulting in a more aggressive tone.

Core equation (simplified):

$$y[n] = G \cdot x[n] \quad \text{with clipping}$$

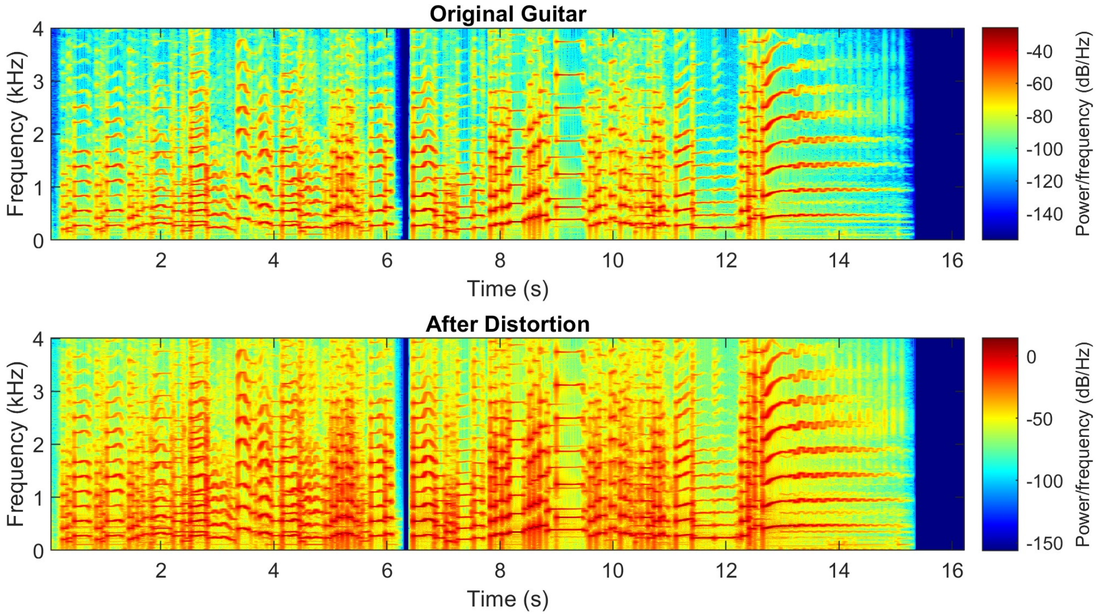

Figure 5: Spectrogram of the guitar signal with distortion effect

Spectrogram Observation – Distortion

High-frequency harmonics become more pronounced. New frequency content appears above the original, resulting in a denser and brighter spectrogram.

8. Conclusion

All effects were implemented using sample-based processing in MATLAB. Spectrogram analysis revealed unique and characteristic frequency patterns for each effect. These visual tools not only confirm expected outcomes but also provide insights into how signal processing transforms audio perceptually and spectrally.

Further developments could include real-time implementations or porting these algorithms into VST plugins or embedded DSP platforms.

Appendix: Equations and Abbreviations

Abbreviations and Mathematical Symbols	
$x[n]$	Input signal at time step n
$y[n]$	Output signal at time step n
Δt	Sampling period
G	Filter gain or amplification factor
D	Fixed delay (in samples)
$D(t)$	Time-varying delay (modulated by LFO)
$D_{\max}$	Maximum delay time
d	Modulation depth (LFO amplitude factor)
f_{LFO}	Frequency of low-frequency oscillator (Hz)
$a_0,\ a_1$	Attenuation coefficients (feedback/echo gain)
g	Feedback gain in flanger
δ	Semitone shift amount in pitch shifting
α	Time-scaling factor for pitch shifting: $\alpha = 2^{\delta/12}$