The Bright Side of the Dark Side of DSP: Audio Effects using GNU Radio

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GNU Radio Conference 2018
Henderson, NV, U.S.A.

Deconstructing the Title

The Bright Side of the **Dark Side of DSP**:
Audio Effects using GNU Radio

- Signal processing phenomena that RF and communications engineers try to **avoid**, or work around...
- … that we **embrace,** in order to build…
- ... some cool **guitar effects** in GNU Radio.

Presentation Outline

Overview: Audio DSP

Filter Resonance

- Wah-Wah Guitar Effect
- Demo

Non Linearity

- Distortion Guitar Effect
- Demo

Takeaways

Audio Spectrum: Frequency

Frequency Range: 20 Hz to 20 kHz

Human Perception Bands

1. Sub-bass (20 - 60 Hz): *Deep bass that is “felt”*
2. Bass (60 - 250 Hz): *Determines thickness of sound*
3. Low Midrange (250 - 500 Hz): *Low order harmonics of instruments*
4. Midrange (500 Hz - 2 kHz): *Vocals*
5. Upper Midrange (2 - 4 kHz): *Determines the timbre of the audio*
6. Presence (4 - 6 kHz): *Conveys the distance of sound*
7. Brilliance (6 - 20 kHz): *Conveys sparkle and air of a sound*

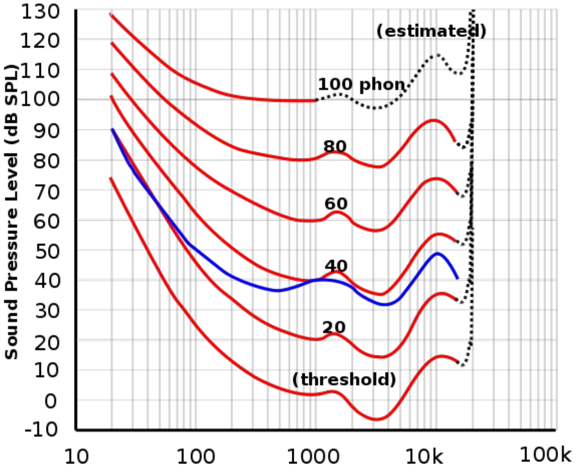
Human perception of frequency is logarithmic

https://www.headphonezone.in/pages/what-are-the-different-components-of-the-audio-spectrum

Audio Spectrum: Amplitude

- The “amplitude” of sound is defined in terms of sound pressure level.
- dB SPL (decibels of sound pressure level) is the unit loudness. *dB SPL is often abbreviated as dB.*

- Human perception of loudness (volume) is logarithmic
- Human threshold of hearing is assumed to be 0 dBSPL (typically 20 uPa)
- Volume Ranges
  - Whisper: 15-25 dB
  - Background noise: about 35 dB
  - Normal home or office background: 40-60 dB
  - Normal speaking voice: 65-70 dB
  - Live Rock music: 120 dB+
  - Pain Threshold: 130 dB
  - Jet aircraft: 140-180 dB



Equal-loudness contours (red) (from ISO 226:2003 revisi
Original ISO standard shown (blue) for 40-phons

https://en.wikipedia.org/wiki/Sound_pressure
https://www.lifewire.com/what-are-decibels-db-1846876

Is Audio DSP Really Different from RF DSP?

Fundamentally, NO. There are some differences though.

- Sample rates are much lower than typical RF applications
  - 44.1 kHz, 48 kHz, 96 kHz, 192 kHz (studio quality)
- Human perception drives filter design
- Frequency axis on spectrum plots is often logarithmic
  - So are cutoff frequencies on filters
- It's hard for us to "hear" phase except in the stereo sense
  - IIR filters for the win!
  - It is typical to do audio DSP in the spectral domain

Resonance

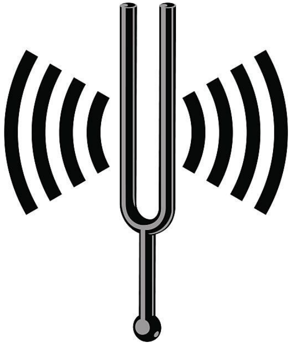
Wah-Wah Effect

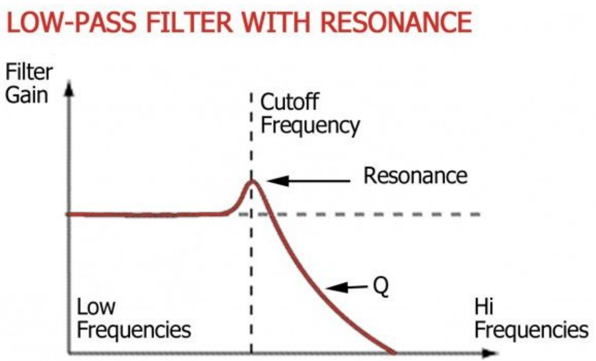
Resonance

Resonance is the tendency of a physical system to oscillate at a high amplitude at certain frequencies.

- The term actually originates from acoustics

- Resonance is not inherently bad, however, it can cause filters and control systems to go unstable.
- A filter can “ring” indefinitely when stimulated with an input at the resonant frequency
- Resonance is usually minimized (damped) when designing a filter or control loop





https://pedals.thedelimagazine.com/production-advice-know-your-filters-low-pass-band-pass-high-pass-resonance/

The Wah-Wah Effect

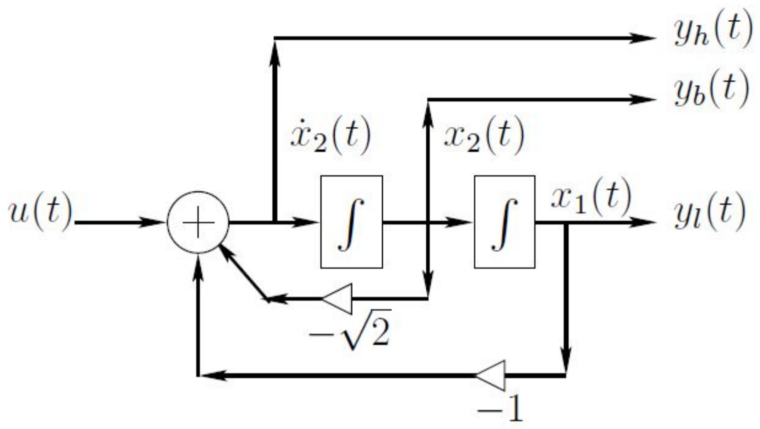
A guitar effect that alters the tone of the guitar signal to create a distinctive sound, mimicking the human voice saying the onomatopoeic name "wah-wah".

The effect sweeps the cutoff frequency of a resonant digital State-Variable Filter (SVF) to create the sound, a spectral glide, know as the wah effect.

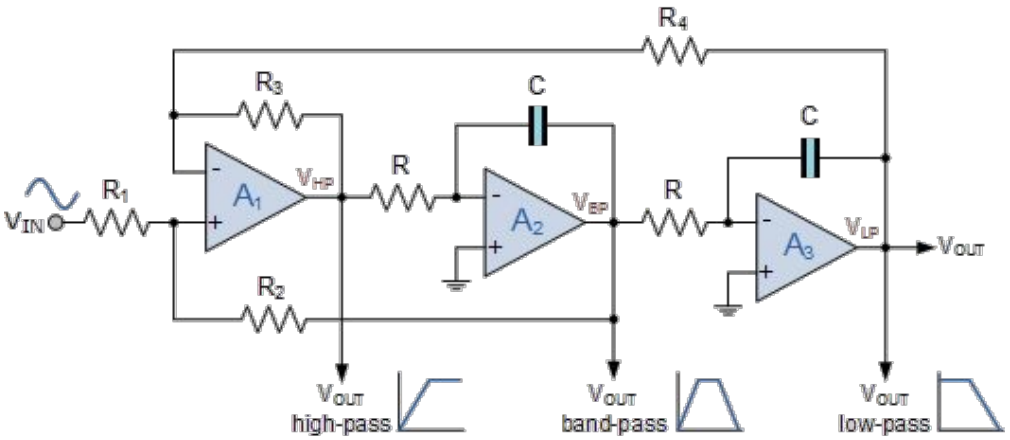
Recognizable as the quintessential Jimi Hendrix sound

State Variable Filter (SVF)

A filter comprised of multiple series integrators, each feeding back and summing.



Example: 2-pole Butterworth LP filter

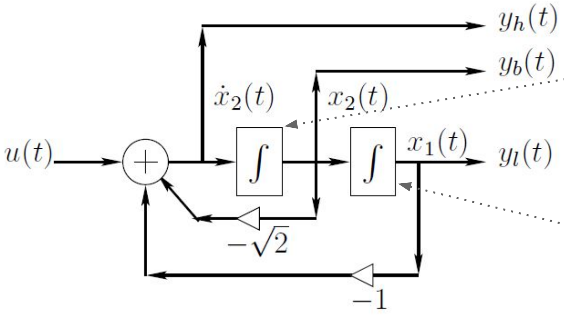


Hardware Implementation of an SVF

https://ccrma.stanford.edu/~jos/svf/

Digital State Variable Filter

Digitize the SVF using the Forward and Backward Euler transform

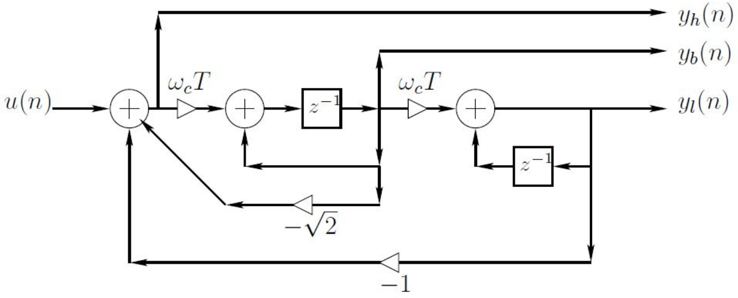


Forward Euler

$$\tilde{s} \leftarrow \frac{z-1}{T} = \frac{1-z^{-1}}{T\,z^{-1}}$$

Backward Euler

$$\tilde{s} \leftarrow \frac{1-z^{-1}}{T}$$



$$H_l(s) = \frac{1}{\tilde{s}^2 + \frac{1}{Q}\tilde{s} + 1}$$

Damping

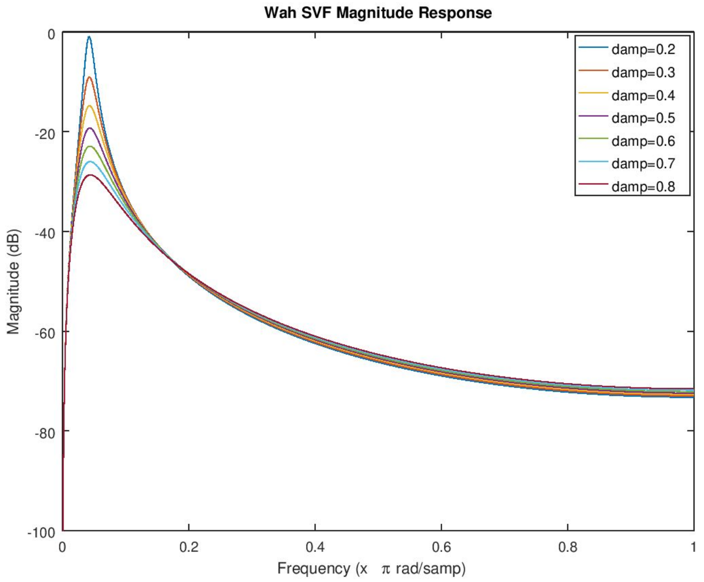
$$Q \triangleq \frac{\omega_c}{2\alpha} \qquad \tilde{s} \triangleq \frac{s}{\omega_c}$$

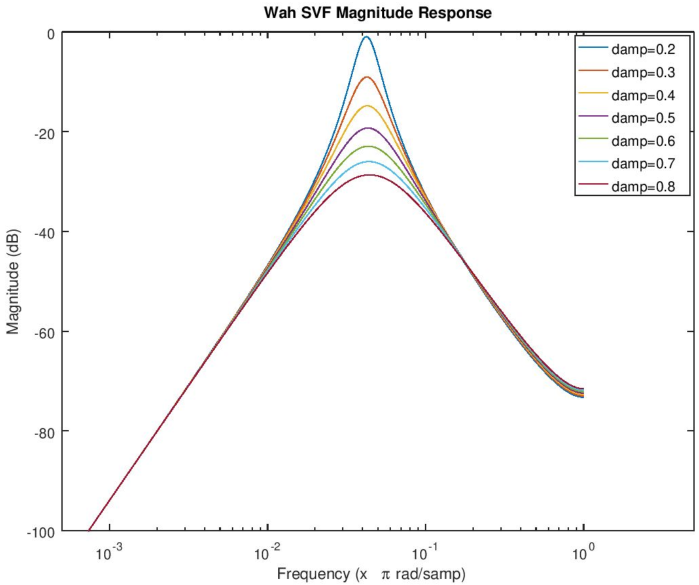
Cutoff Frequency

$$\begin{aligned}
y_h(n) &= u(n) - y_l(n) - Q y_b(n)\\
y_b(n) &= \omega_c T y_h(n) + y_b(n)\\
y_l(n) &= \omega_c T y_b(n) + y_l(n);
\end{aligned}$$

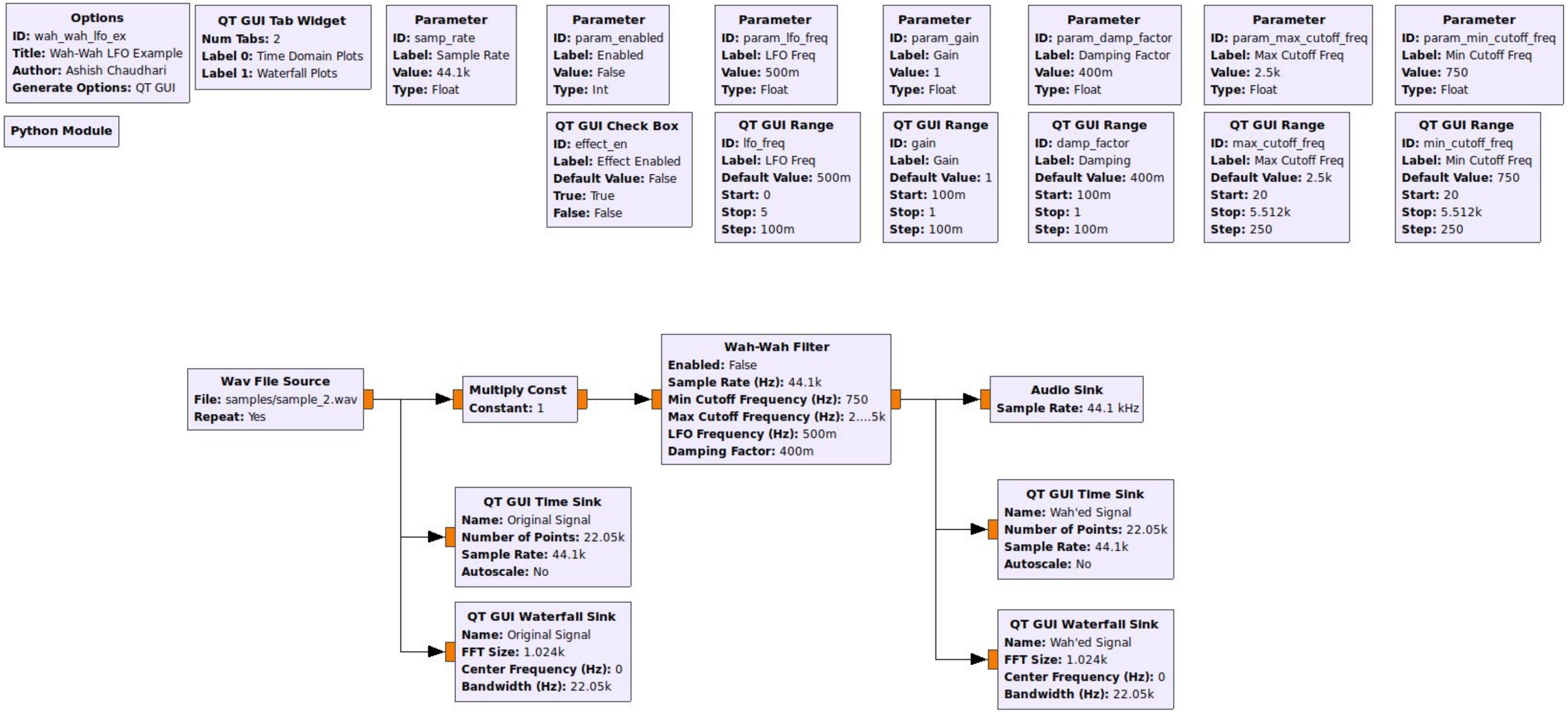
The Wah SVF

- Effect = A linear combination of the bandpass and lowpass output of the SVF
- Vary the cutoff frequency (ω_C) over time





Wah SVF: DEMO



Options
ID: wah_wah_lfo_ex
Title: Wah-Wah LFO Example
Author: Ashish Chaudhari
Generate Options: QT GUI
Python Module
QT GUI Tab Widget
Num Tabs: 2
Label 0: Time Domain Plots
Label 1: Waterfall Plots
Parameter
ID: samp_rate
Label: Sample Rate
Value: 44.1k
Type: Float
Parameter
ID: param_enabled
Label: Enabled
Value: False
Type: Int
QT GUI Check Box
ID: effect_en
Label: Effect Enabled
Default Value: False
True: True
False: False
Parameter
ID: param_lfo_freq
Label: LFO Freq
Value: 500m
Type: Float
QT GUI Range
ID: lfo_freq
Label: LFO Freq
Default Value: 500m
Start: 0
Stop: 5
Step: 100m
Parameter
ID: param_gain
Label: Gain
Value: 1
Type: Float
QT GUI Range
ID: gain
Label: Gain
Default Value: 1
Start: 100m
Stop: 1
Step: 100m
Parameter
ID: param_damp_factor
Label: Damping Factor
Value: 400m
Type: Float
QT GUI Range
ID: damp_factor
Label: Damping
Default Value: 400m
Start: 100m
Stop: 1
Step: 100m
Parameter
ID: param_max_cutoff_freq
Label: Max Cutoff Freq
Value: 2.5k
Type: Float
QT GUI Range
ID: max_cutoff_freq
Label: Max Cutoff Freq
Default Value: 2.5k
Start: 20
Stop: 5.512k
Step: 250
Parameter
ID: param_min_cutoff_freq
Label: Min Cutoff Freq
Value: 750
Type: Float
QT GUI Range
ID: min_cutoff_freq
Label: Min Cutoff Freq
Default Value: 750
Start: 20
Stop: 5.512k
Step: 250
Wav File Source
File: samples/sample_2.wav
Repeat: Yes
Multiply Const
Constant: 1
Wah-Wah Filter
Enabled: False
Sample Rate (Hz): 44.1k
Min Cutoff Frequency (Hz): 750
Max Cutoff Frequency (Hz): 2....5k
LFO Frequency (Hz): 500m
Damping Factor: 400m
Audio Sink
Sample Rate: 44.1 kHz
QT GUI Time Sink
Name: Original Signal
Number of Points: 22.05k
Sample Rate: 44.1k
Autoscale: No
QT GUI Waterfall Sink
Name: Original Signal
FFT Size: 1.024k
Center Frequency (Hz): 0
Bandwidth (Hz): 22.05k
QT GUI Time Sink
Name: Wah'ed Signal
Number of Points: 22.05k
Sample Rate: 44.1k
Autoscale: No
QT GUI Waterfall Sink
Name: Wah'ed Signal
FFT Size: 1.024k
Center Frequency (Hz): 0
Bandwidth (Hz): 22.05k

Non-Linearity

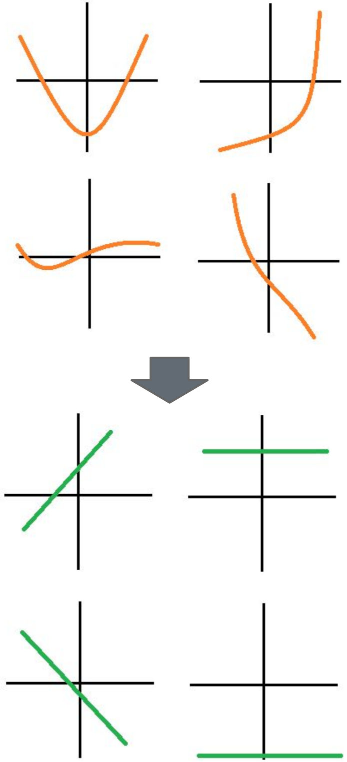
Distortion

Do we care about non-linearity?

***Yes**, all the time!*

- Power Amplifiers
  - PAs are driven outside their linear range for higher efficiency
  - Sophisticated methods like digital predistortion (DPD) to make a non-linear PA linear
- RF/Comms Design
  - Calibration to remove physical nonlinear effects
  - We prefer FIR filters because phase is linear
- ...

Generally we go through a lot of effort to make nonlinear things linear



Distortion Effect

Distortion (a.k.a. overdrive) is an effect where the gain of an instrument is increased to push amplitudes into a non-linear region of operation to produce a "fuzzy" or "gritty" tone.

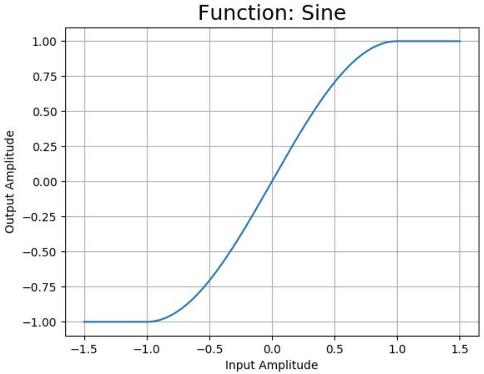
Stages of Distortion

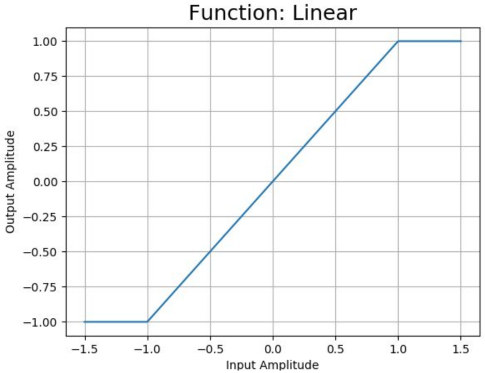
1. Boost the input signal (apply gain)
2. Apply a non-linearity to the signal, primarily to drive it into clipping

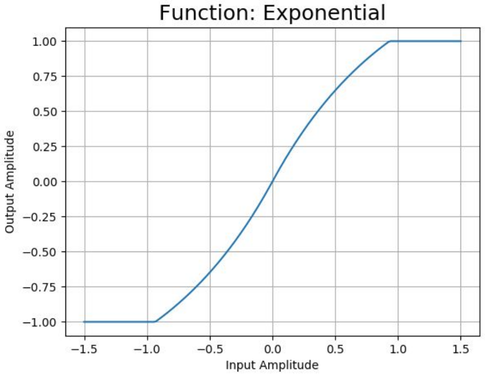
Typically used with an electric guitar

Prevalent in gernes like rock, metal, blues

Clipping Functions

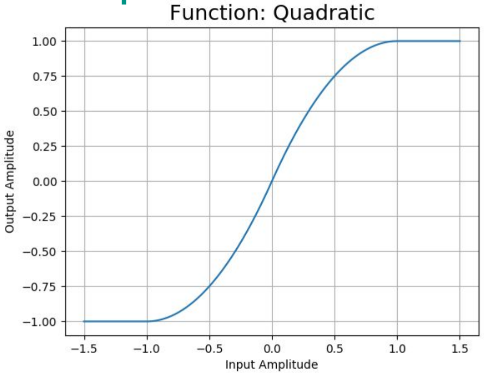


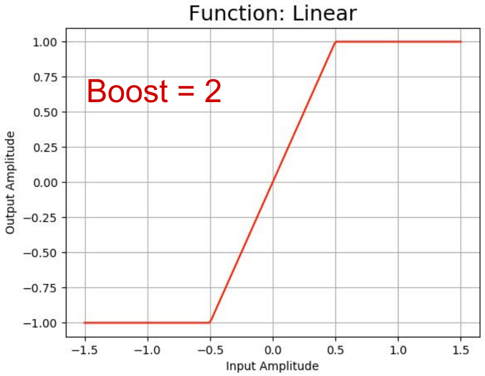


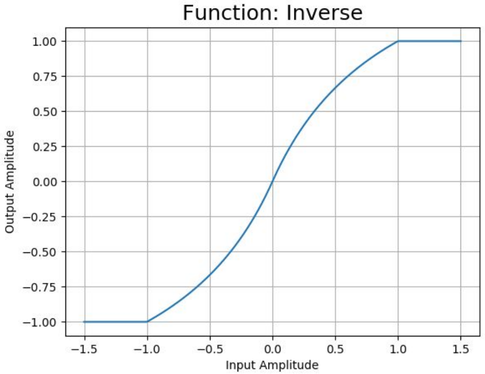


Soft Clip

Hard Clip

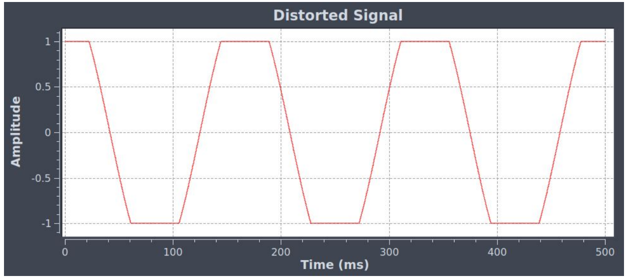


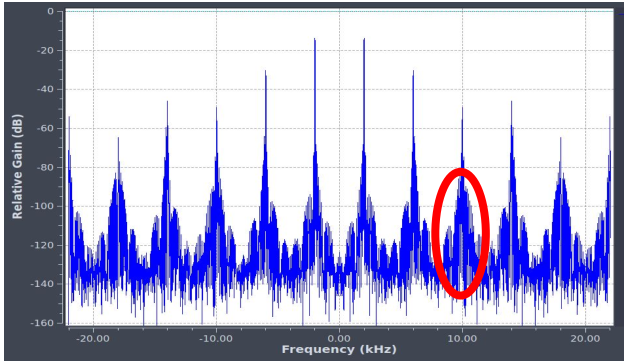


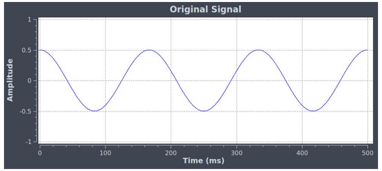


Clipping Function Evaluation

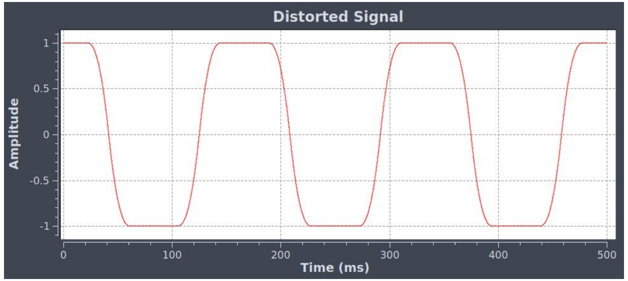
Hard Clipping

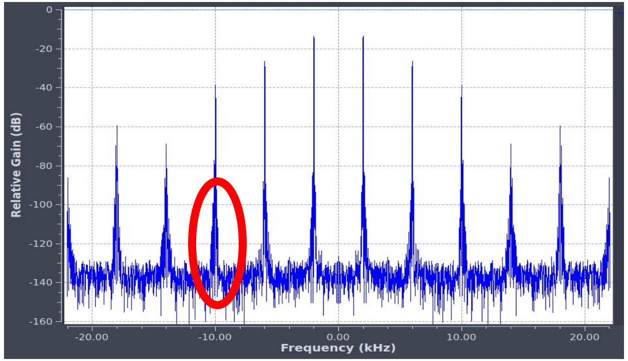






Soft Clipping

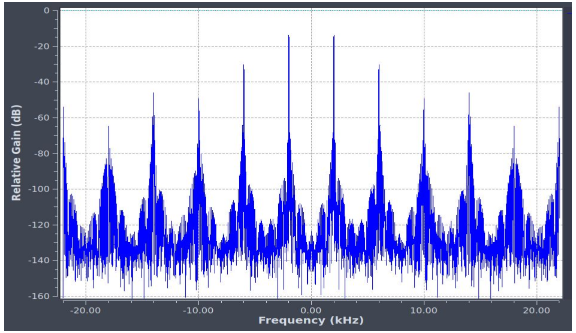


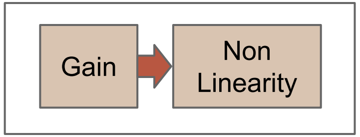


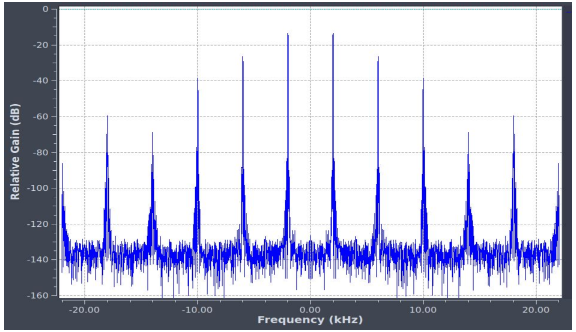
Oversampling + Clipping

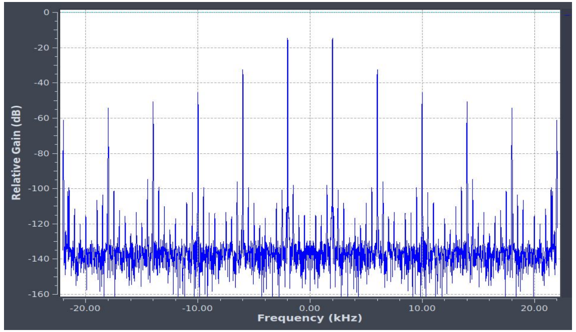
Hard Clipping

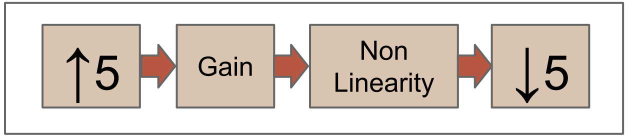
Soft Clipping

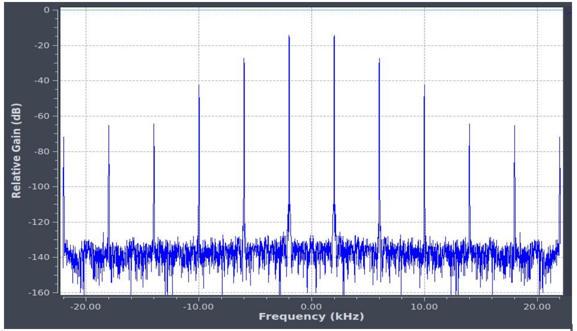








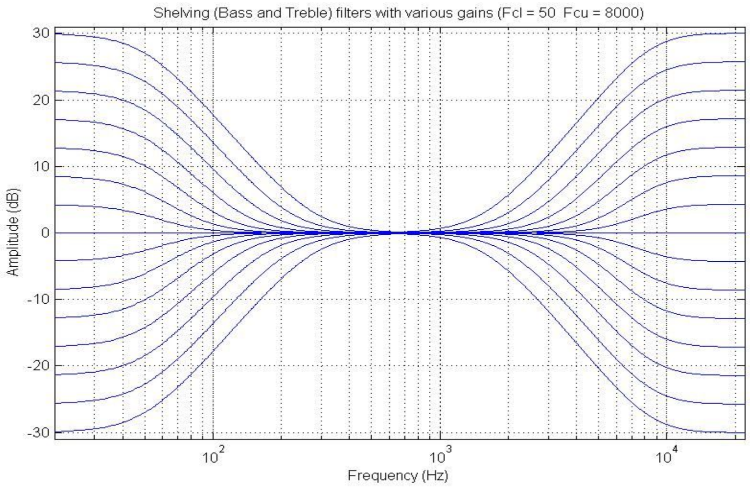




Post Filtering

Distortion can be post-filtered using shelving equalization filters to add “character”

While high and low pass filters are useful for removing unwanted signal above or below a set frequency, shelving filters can be used to reduce or increase signals above or below a set frequency.

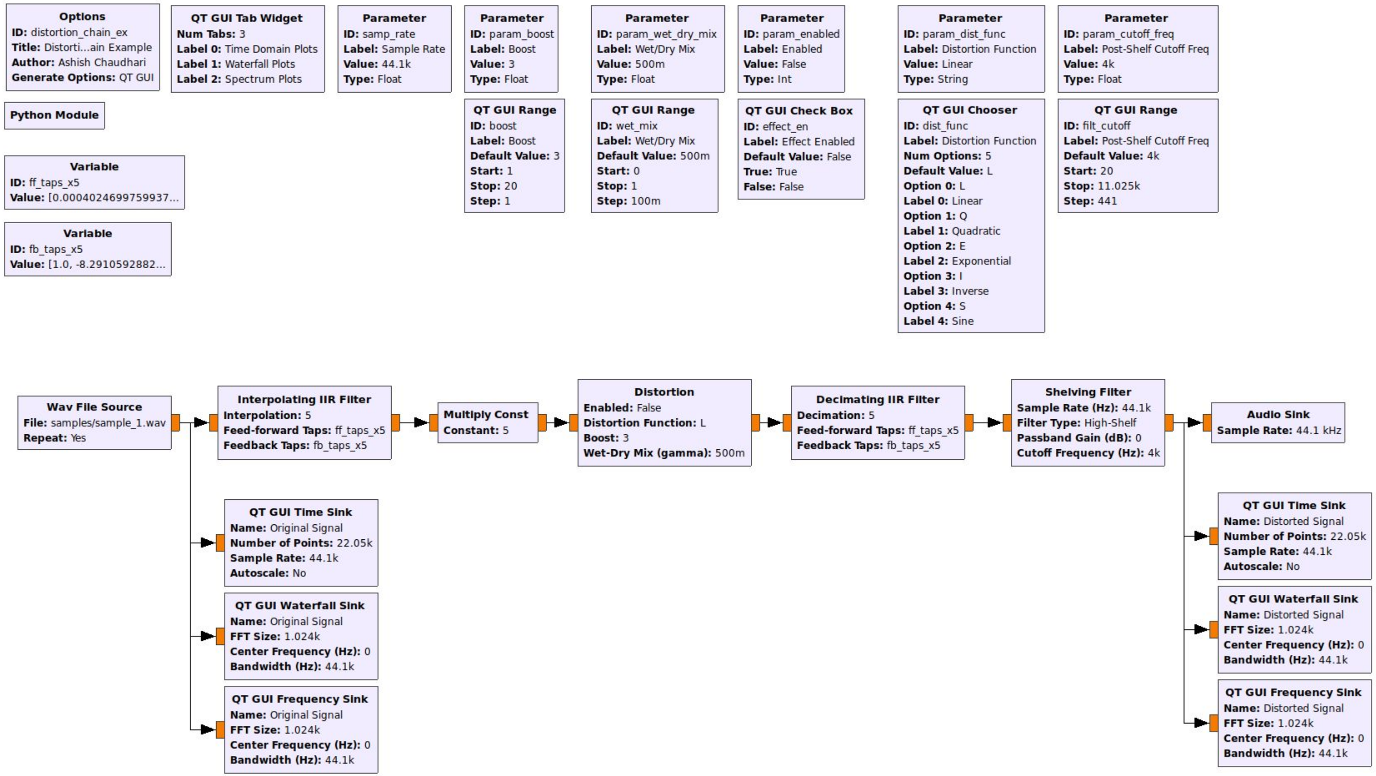


Distortion Block Diagram

Distortion Chain Components

1. Interpolating IIR Filter: Upsample and anti-imaging IIR filter
2. Distortion: Boost and apply non-linearity
3. Decimating IIR Filter: Anti-aliasing IIR filter and downsample
4. Shelving Filter: Attenuate frequencies higher than the upper midrange

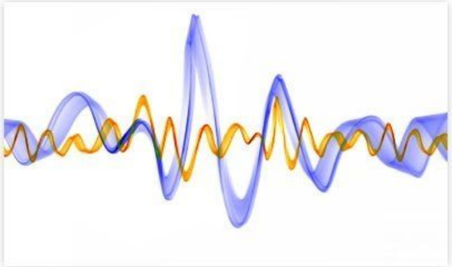
Distortion: DEMO



Takeaways

Educational Value

- Easy to perceive a DSP algorithm when you hear it
  - That's how I learned my DSP (my background is in computer architecture, not DSP or Comms)
- The effects I presented are not complicated but they cover several fundamental concepts
  - Easy to present in an educational setting
- Implementing them in GNU Radio was extremely easy
  - No need for additional hardware to "graduate" from simulation. Every computer has a sound card.
  - Started with Python, moved to C++ because Python is not real-time (even for 44.1 kHz)
- Focus on the DSP, not the plumbing or data movement.



Thank You!

gr-guitar
https://github.com/achaudhari/gr-guitar