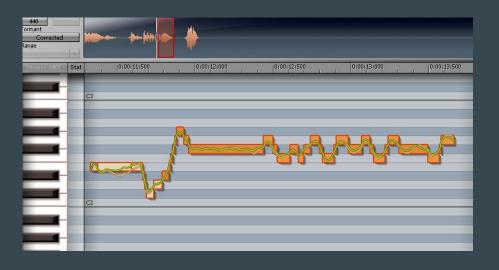
# **FPGAAAutotune**

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Elaine Ng + Kika Arias

### What *is* Autotune?





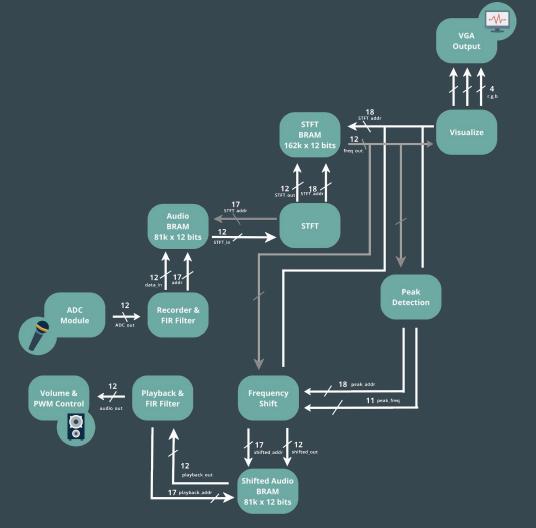
## **Methods for Autotune**

Time Domain Method (TD-PSOLA)	Frequency Domain Method
<ol> <li>Divide signal into chunks</li> <li>Change frequency by shifting chunks:         <ul> <li>a. closer together = higher freq</li> <li>b. farther apart = lower freq</li> </ul> </li> <li>Reconstruct signal by adding chunks</li> </ol>	<ol> <li>Find STFT</li> <li>Find frequency corresponding to each note</li> <li>Shift by convolving original signal with two deltas at correct frequency</li> </ol>

## We're doing this in the FREQUENCY DOMAIN

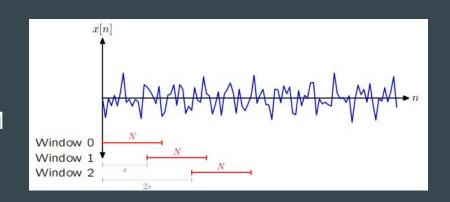
- Most pitch correction implementations (including actual Auto-Tune) uses TD-PSOLA because it's simpler and computationally less intensive...
- So...why are we not doing this?

### **Overview**



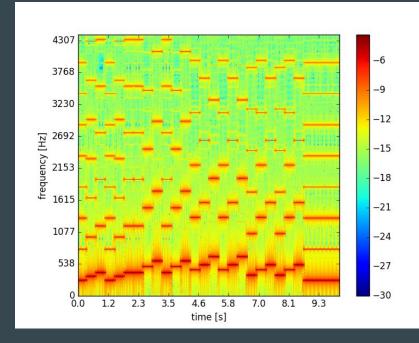
## **Short Time Fourier Transform (STFT)**

- 1. Shift in 1024 samples
- 2. Apply Hann window
- 3. Calculate FFT using FFT Core
- 4. Calculate squared magnitude
- 5. Store squared magnitude in STFT BRAM
- 6. Go back a few addresses in audio BRAM
- 7. Shift in new 1024 samples
- 8. Repeat (step 2-6)



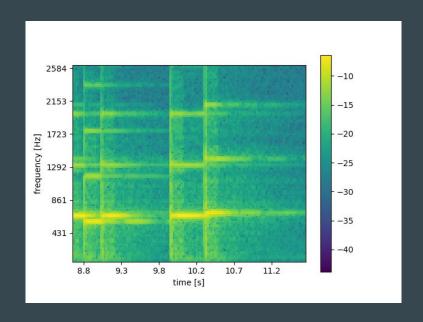
### **Visualization**

- Get the squared magnitudes from FFT BRAM
- Map squared magnitude to colors (RGB)
- 3. Plot a spectrogram for the whole signal like →



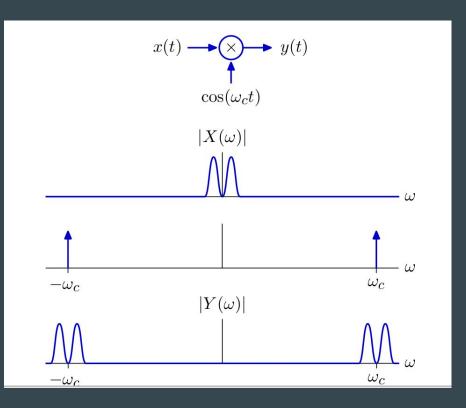
#### **Peak Detection**

- 1. Find the location of the notes
- Find the freq that best represents that chunk
- 3. Use binary search on LUT to find closest "natural" frequency



## Pitch Shifting

- Construct a big filter that is: two deltas at desired frequency for each harmonic
- 2. Convolve filter with actual signal!
- 3. Take the IFFT and output it



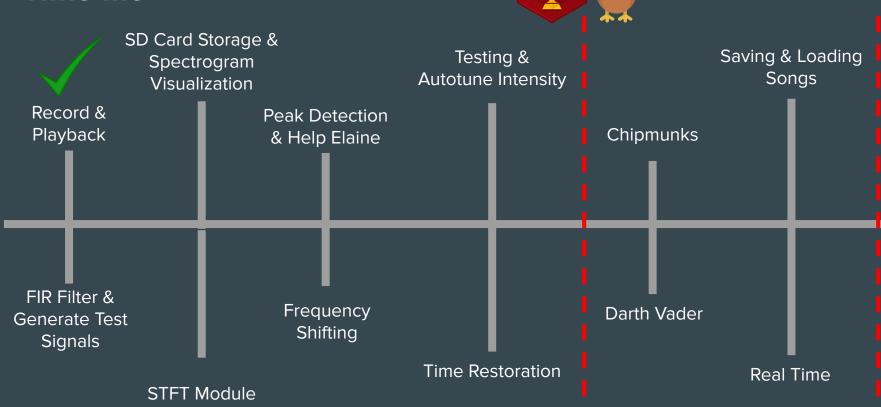
## MVP +



#### Foreseeable Issues

- Memory (STFT, peak detection, pitch shifting, actual audio)
- Tuning sampling rates
- Tuning window sizes
- Timing (efficiency, modules synced properly)
- Restoring original durations

#### Timeline



## Questions?