



# *A Cortex M3 Guitar Tuner*

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# Guitar Tuners

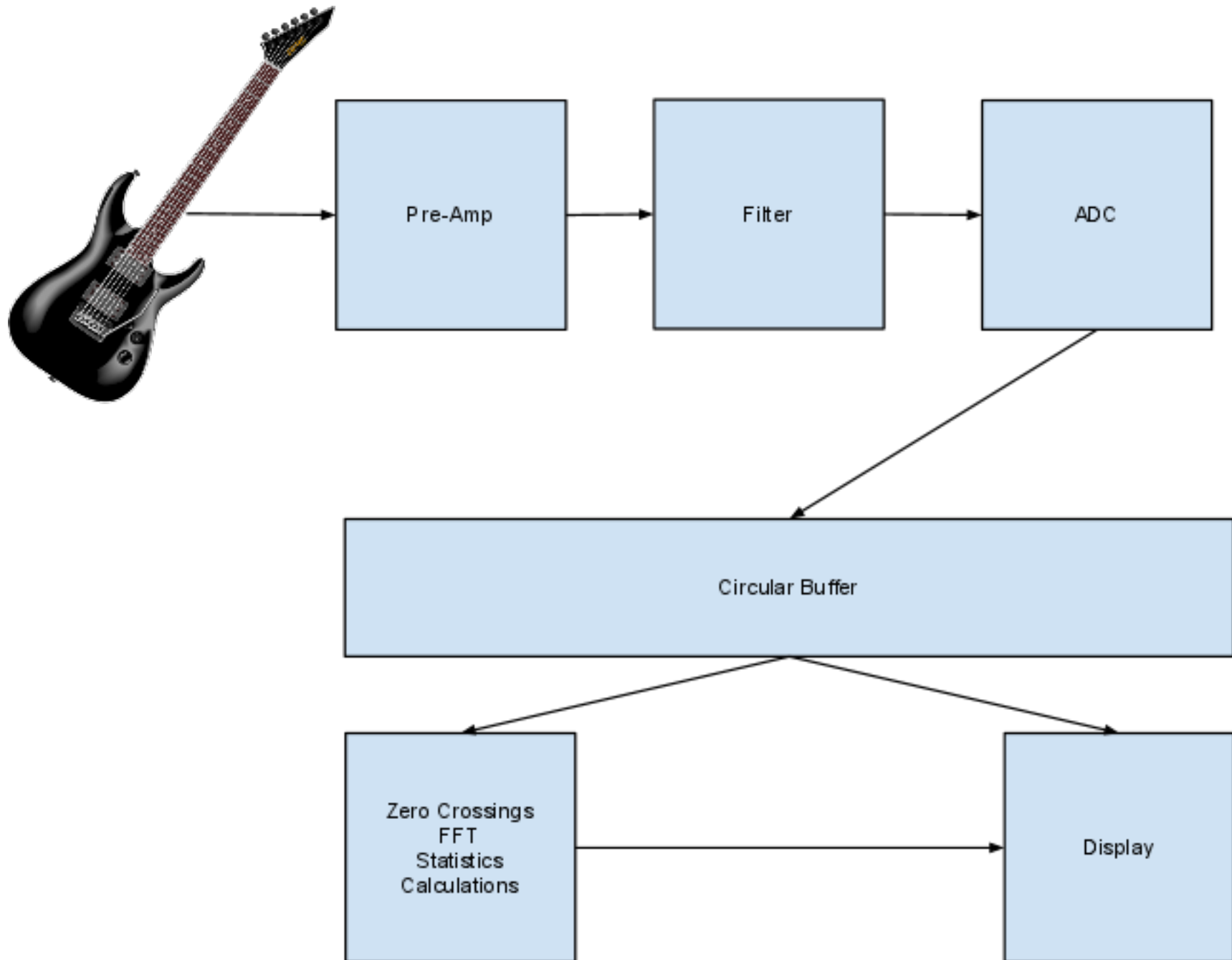
- Guitars need to be tuned often
- Involves increasing and decreasing tension in strings
- Can plug into a tuner which tells if notes are flat or sharp
- Need to respond quickly to a plucked note



# Requirements

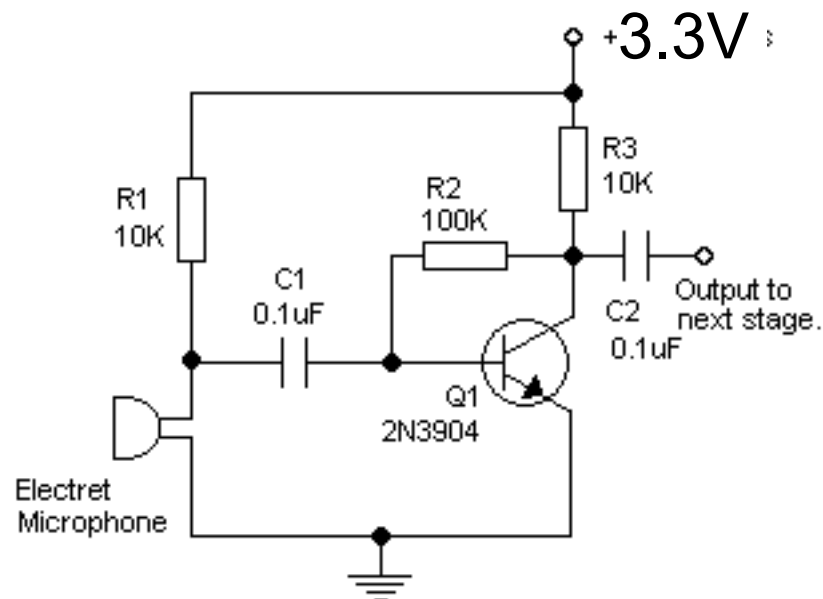
- Inputs: Guitar signal, Buttons
- Output: OLED Display
- Functions: Tuner, Waveform display, FFT display
- Performance: Fast enough to tune guitar, sampling freq. high enough for all guitar strings.
- Multi-rate System
  - ADC Sampling - Hard Deadline
  - Display results on OLED - Soft Deadline
- Respond to user input of buttons
  - Asynchronous
- Amplify guitar signal and digitize
  - Must be done in hardware before sampling
- Detect if a guitar string is in tune
  - Calculations must be done between displaying results

# Guitar Tuner Architecture



# Guitar->Preamp

- Vibrating strings induce voltage on magnetic coils (pickups)
- Voltage is plugged into an amplifier
- Voltage [1]
  - 100mV - 1V RMS
- Fundamental String Frequencies [2]
  - 82 Hz, 110 Hz, 147 Hz, 196 Hz, 247 Hz, 330 Hz
- Need to amplify signal before going into ADC!
- Preamp boosts signal such that it can be digitized by ADC
- Used this circuit [3]:



## SIMPLE AUDIO PREAMP

This easy circuit provides good gain to weak audio signals. Use it in front of an RF oscillator to make an RF transmitter that is very sensitive to sound.

# Filter

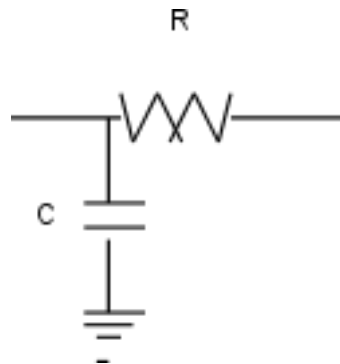
Needed to filter for interference on audio line

Since we only need 330 Hz, can design filter for this.

I had  $R = 22\text{k}\Omega$  and  $C = 0.001\text{ }\mu\text{F}$

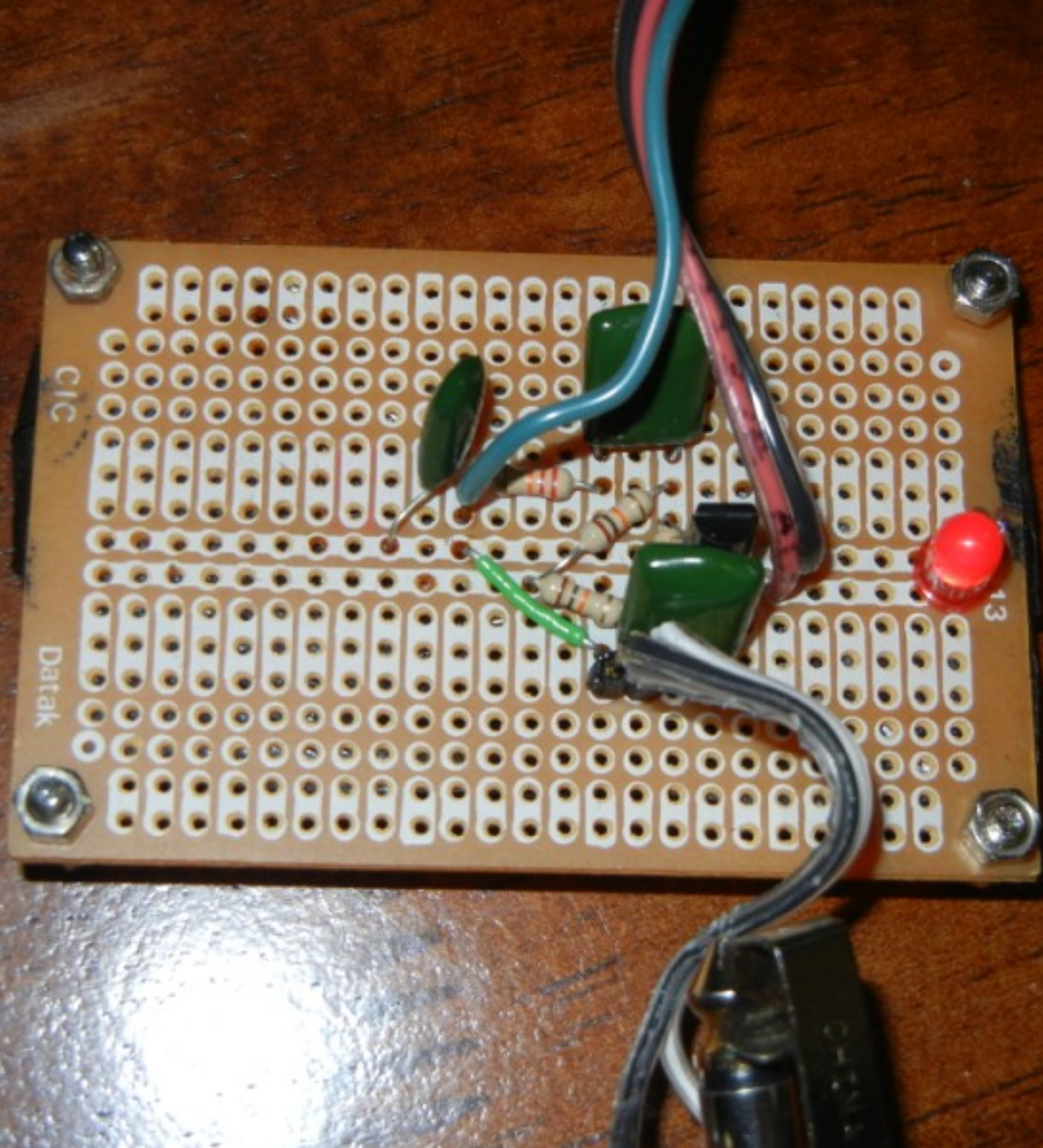
Thus,  $f_c = 8376\text{Hz}$ , should be good enough.

$$f_c = \frac{1}{2\pi RC} \text{ Hz} \text{ <--Low-Pass Butterworth Filter Equation [7]}$$



Add filter to output of pre-amp stage.





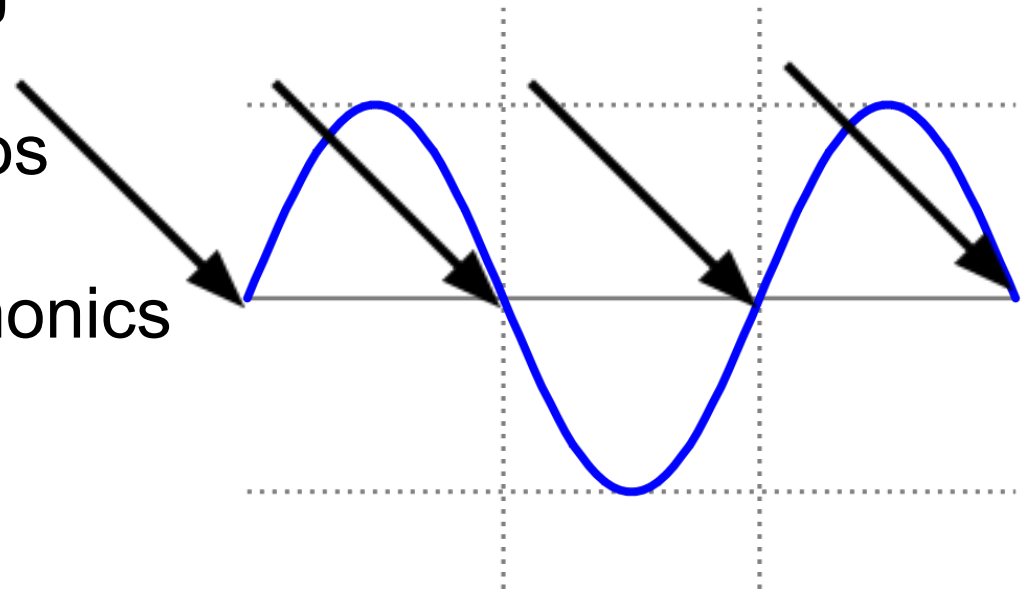
# ADC

- Can input voltages from 0 to 3V
- Outputs a 10-bit value from 0x000 to 0x3FF [4]
- Nyquist Theorem [5]
  - Need to sample at twice the highest frequency
  - Highest frequency = 330Hz
- Oversampling
  - Sampling isn't that accurate, thus sample many times and get an average
  - I've set this for 8x oversampling
- Timer
  - We use a timer to drive the ADC
  - ADC Interrupts when 8 samples have been averaged and ready to read
- Thus we set our timer for at least  $330 * 2 * 8 = 5280$  Hz



# Tuning Calculation

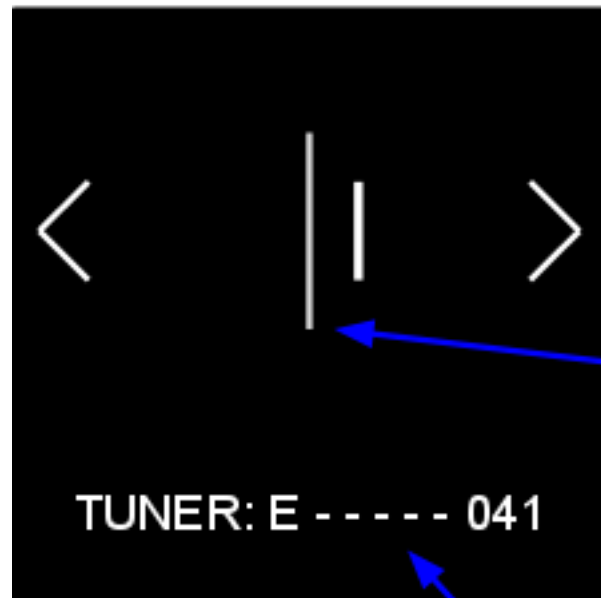
- Multiple ways to handle this
- I chose to count 'zero-crossings', whenever a transition from previous to next signal crosses 'zero'
- Zero is defined as the rolling average of the signal since we range from 0x000 to 0x3fff, and because there is some bias in the signal
- Use a reference signal to determine the number of zero crossings needed for a string
- Roughly
  - higher pitch == more zeros
  - lower pitch == less zeros
  - Need to account for harmonics
  - Filtering would help



# FFT

- Transforms the time domain signal into a frequency domain signal.
- Needed fixed point arithmetic and no trig functions, therefore a lookup table is needed.
- Tried to write this in C myself and failed miserably.
- Found an efficient 128 point function written in cortex m3 assembly.
- 128 point not good enough for tuning, can't see enough variation in the maximum bin.

# Display - Tuner Mode

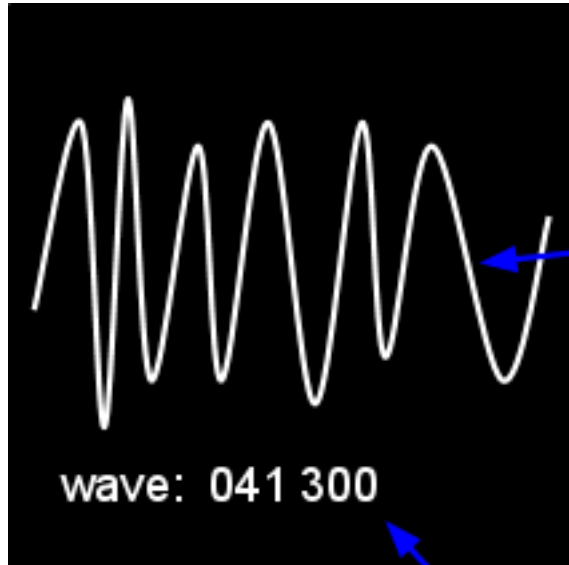


Arrows  
indicate if pitch  
is higher or  
lower

Center line is 'in-  
tune', thicker line is  
the current reading.  
Want to get these  
matching.

Need to select the current  
string (E,a,d,g,b,e) with Left  
and Right buttons. The  
number shows the zero  
crossings.

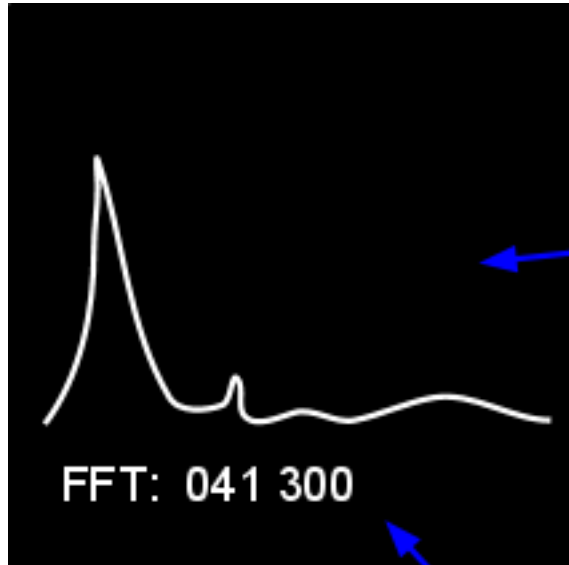
# Display - Waveform Mode



This display shows the contents of the circular buffer at a particular interval (FPS rate)

shows zero crossings and difference between max and min values for a period of time (used for threshold calculations)

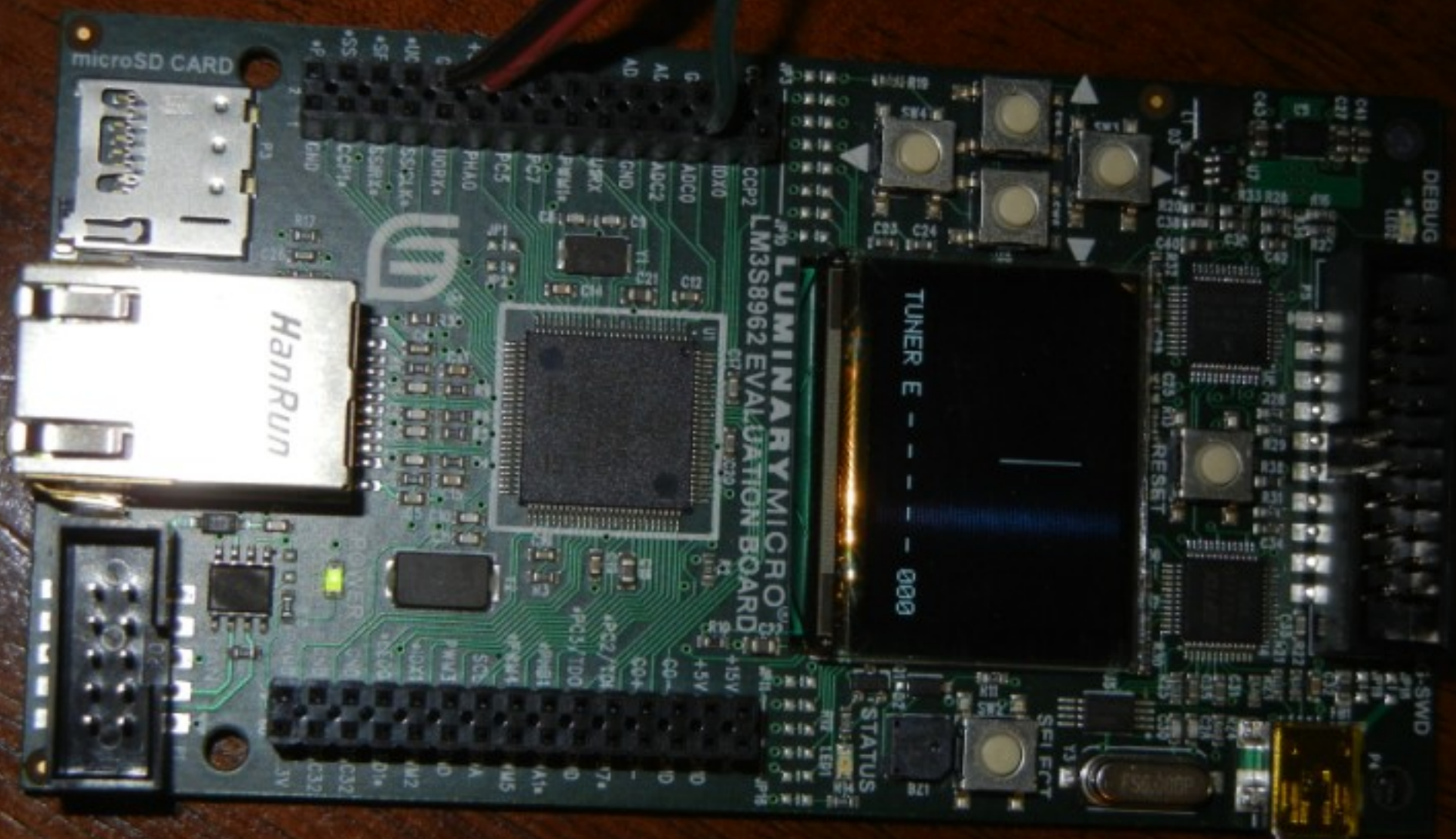
# Display - FFT Mode



This display shows  
the contents of the  
FFT transformation.

shows a count and  
bin with maximum  
value (not the 0th bin)





# Demo

# If I had more time...

- Better pre-amp
  - Need more gain and make it less noisy
- Filtering per string (Digital)
- Better resolution of tuner, faster response
- Better UI / User Feedback

# References

- [1] <http://buildyourguitar.com/resources/lemme/>
- [2] <http://ffden-2.phys.uaf.edu/211.web.stuff/billington/main.htm>
- [3] <http://www.reconnsworld.com/forum/read.php?9,10>
- [4] TI Stellaris Documentation (spms001f.pdf)
- [5] [http://en.wikipedia.org/wiki/Nyquist%E2%80%93Shannon\\_sampling\\_theorem](http://en.wikipedia.org/wiki/Nyquist%E2%80%93Shannon_sampling_theorem)
- [6] <http://www.precisionstrobe.com/apps/guitarharm/guitarharm.html>
- [7] [http://www.electronics-tutorials.ws/filter/filter\\_8.html](http://www.electronics-tutorials.ws/filter/filter_8.html)

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