# Real-Time Audio Spectrum Analyzer – Final Report

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#### **Abstract**

The goal of this project is to build a real-time audio spectrum analyzer. The system performs Fourier analysis on a real-time audio signal from a microphone using the FPGA and displays the spectrum of the audio signal on an LCD. The microcontroller is responsible for sampling data from the microphone, sending and receiving data from the FPGA, and updating the spectrum on the LCD.

#### Introduction

Growing up using applications such as Windows Media Player and QuickTime, there would be various graphs on the side providing information about the audio signal being captured by the application. Captivated by these curious moving lines as young children, we would play different sounds with the computer and make noises to see how the graphs changed. As we entered college and began taking engineering courses, we learnt that these curious moving lines were in fact outputs of Fourier analysis, which helps to determine the frequency composition of a given signal. For audio signals, this allows us to perform tasks such as visualizing different pitches present and their intensities, performing filtering to remove noise from a signal, and audio recognition. Ignited by this interest, we decided to recreate the frequency versus amplitude graphs we saw as children for this project, where we will build a real-time audio spectrum analyzer.

Audio data is first sampled from the electret microphone through the microcontroller's analog-to-digital converter (ADC), with the sampling frequency controlled by a timer on the microcontroller. The audio data is stored in the microcontroller's memory from the ADC using direct memory access (DMA) until all the samples are collected. The microcontroller then sends the audio data through SPI to the FPGA, which implements a FFT hardware accelerator. After the FPGA calculates the FFT, the output is sent through SPI back to the microcontroller. The microcontroller takes the FFT output and converts it to a pixel array, which is sent through SPI to the graphic LCD to display the frequency versus amplitude graph on the display. The system block diagram is shown in Figure 1.

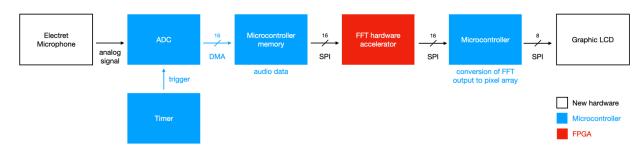


Figure 1: System block diagram

# **Schematics**

The schematic diagram for the breadboarded circuit is shown in Figure 2, and the pin assignments for the FPGA are shown in Table 1.

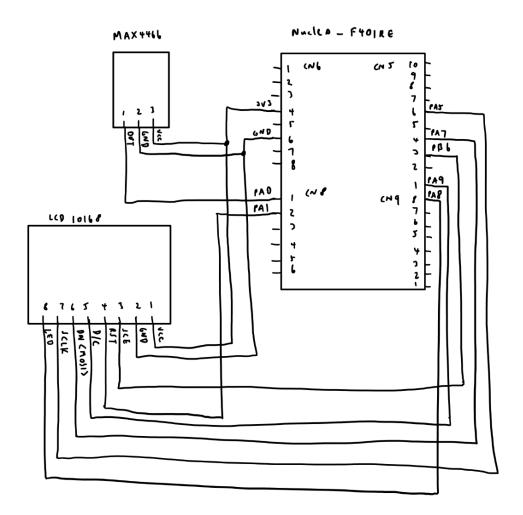


Figure 2: Schematic of breadboard circuit

Node Name	Direction Location	
CLK	Input	PIN_H6
DONE	Output	PIN_H5
LOAD	Input	PIN_K11
SCK	Input	PIN_H4
SDI	Input	PIN_J1
SDO	Output	PIN_J2

Table 1: Pin Assignments for the FPGA

#### **New Hardware**

There are two pieces of new hardware used in the system: the electret microphone and the graphic LCD.

## **Electret Microphone**

The breakout (as shown in Figure 3) consists of a MAX4466 amplifier with an electret microphone, and the gain can be adjusted using a small trimmer pot on the back of the breakout.

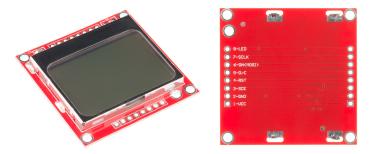


Figure 3: Electret microphone with amplifier (ADA 1063) [4]

The electret microphone is used to capture (analog) audio signals as the input to the system, which is sent to the microcontroller for further processing. Interfacing with the microphone was performed using the microcontroller's ADC to convert the analog signal from the microphone to a digital signal that can be processed by the rest of the system.

## **Graphic LCD**

The 48 by 84 pixel graphic LCD, which comes from an old Nokia 5110, is mounted on a board and with a PCD8544 controller. Figures 4 and 5 show the graphic LCD and the pinouts.



Figures 4-5: Graphic LCD with pinouts on the back of the board (LCD 10168) [6]

The LCD is used to display the spectrum of the input audio signal as a frequency versus amplitude graph. The dominant frequency in the audio signal is also displayed on the side of the graph (as shown in Figures 16-19).

The microcontroller (controller device) communicates with the LCD (peripheral device) through SPI. The D/C pin (pin 5 in Figure 5) allows the microcontroller to toggle between communicating data and commands. To begin interfacing with the LCD, the first step is to reset the LCD using the RST pin (pin 4 in Figure 5) and then configure the settings by sending commands to the LCD, including choosing the instruction set, the addressing mode and the display mode. For this system, vertical addressing was selected, as the main purpose of the LCD is to display the frequency versus amplitude graph as a series of bars, thus it is more logical to populate the pixel array by column. The microcontroller then sends data to the LCD 8 bits at a time, with each bit representing one pixel being turned on or off. It is possible to configure specific 8-bit blocks of pixels, however this was not needed for this system as for each new FFT output, the entire display is updated.

# **Microcontroller Design**

The microcontroller in the system acts as a coordinating device — responsible for sampling data from the microphone, sending and receiving data from the FPGA, and updating the spectrum on the LCD — as shown in blue in Figure 1.

The first step for the microcontroller is to set up the initial configurations, including for the GPIO pins, the SPI, the direct memory access (DMA), the analog-to-digital converter (ADC) and the graphic LCD. After that, the microcontroller enters an infinite loop that repeatedly executes a series of functions to sample and store the audio data, send the audio data to the FPGA, receive the FFT output from the FPGA, convert the FFT output to the pixel array, and send the pixel array to the graphic LCD.

To sample the audio data from the microphone, the microcontroller's ADC is used. The sampling frequency is controlled by one of the peripheral timers. To collect the audio data samples, an ADC conversion is performed every time the timer's counter reaches the desired threshold. The ADC is configured to be in DMA mode, so once an ADC conversion is completed, the data in the ADC's data register is stored in the specified location (the audioData array). For this system, the sampling frequency was chosen to be 3200 Hz (but this can be easily changed) and 32 samples are collected, due to the design of the FFT (see the "FPGA Design" section). Therefore, the system can display audio signals up to 1600 Hz (the Nyquist frequency is  $\frac{3200}{2} = 1600 \text{ Hz}$ ) with a bin width of 100 Hz (each bin is  $\frac{1}{32/3200} = 100 \text{ Hz}$  wide).

The next step in the infinite loop for the microcontroller is to send and receive data from the FPGA, which performs the FFT. The microcontroller and FPGA communicate via SPI, as per the transmission interface specified by the FFT SPI module (see the "FPGA Design" section). The LOAD pin is used to communicate when data for the FFT is being between the microcontroller and the FPGA, and the DONE pin is used to communicate when the FFT is complete and the data is ready to be sent from the FPGA to the microcontroller. The output from the FFT is a series of complex numbers in Q15 (further explained in the "FPGA Design" section). Therefore, before further processing the microcontroller converts the real and imaginary components of each complex number into doubles and then computes the magnitudes of each of these complex numbers to determine the amplitudes at each frequency.

Lastly, the microcontroller also creates the pixel array, which is sent through SPI to the graphic LCD. As discussed in the "New Hardware" section, vertical addressing is used for this system (i.e. the pixels are sent to the LCD by column). The function first converts the FFT output to the frequency versus amplitude bar graph. Thus for each frequency, the pixel array is populated based on the amplitude corresponding to the frequency, normalized by a fixed amplitude determined empirically (and can be easily changed). The second part of the function is responsible for displaying the dominant frequency as text on the right side of the bar graph. This is done by storing the digits 0-9 and characters 'H' and 'z' as arrays of pixels, identifying the dominant frequency (that with the highest amplitude) from the FFT output, and then for each digit or character, adding each column of pixels iteratively to the pixel array. Finally, the rest of

the pixel array is populated with 0s to overwrite the pixels from the previous array loaded onto the LCD.

A bare-metal design was used for this system, performing the SPI communication with the FPGA and LCD in series (instead of in parallel). This resulted in reduced complexity, as interrupts do not have to be used. Through testing, we also found that even with this design, there was no perceivable lag between the microphone capturing the audio signal and the output on the display, thus further justifying this design choice. Additionally, in the implementation of the microcontroller operations, each function was kept modular, which allowed for smoother unit testing of the system to identify parts of the system that needed to be debugged. For example, to unit test the conversion of the FFT output to the pixel array and the SPI communication with the LCD (especially before the FFT hardware accelerator was complete), test FFT output was used and the operation of this module was verified by comparing the output on the display to the expected output generated by a Python program.

# **FPGA Design**

The FFT hardware accelerator consists of two modules: the "FFT core" which is responsible for performing the FFT on loaded data and the "SPI module" which handles the SPI communication with the microcontroller. The top-level block diagram of the 32-point FFT module is shown in Figure 6.

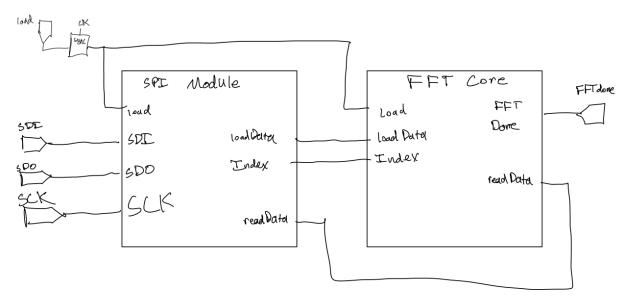


Figure 6: Top-level block diagram of FFT module

#### FFT Core

At its root, the FFT is a series of "butterfly" operations. The individual butterfly operation is described by the following pair of equations:

$$A' = A + \omega^k * B \tag{1}$$

$$B' = A + \omega^k * B \tag{2}$$

Where  $\omega^k$  is the corresponding "root of unity". A schematic of an 8-point FFT is shown in Figure 7. For an 8-point FFT, there are 12 butterfly operations that need to be performed  $(\frac{N}{2}\log_2 N)$ . It is useful to identify the specific butterfly operation in terms of its i and j index as identified in Figure 7.<sup>1</sup> Importantly, each butterfly only depends on the results from the previous level. This means that one can compute each butterfly sequentially if the outputs are stored. Moreover, the order that one performs the butterfly operations at each level does not matter as long the results are stored and completed before the butterfly of the next level.

Given this insight, a relatively straightforward way to implement the FFT in hardware is to have a butterfly module that performs the butterfly operation, a read-write data structure to store the inputs/outputs to each butterfly operation, a read-only data structure that stores the "roots of

<sup>&</sup>lt;sup>1</sup> For a more complete description and derivation of the FFT, see [2].

unity", and an "address generation unit" that generates the addresses to retrieve and store values from/in the data structures. This implementation of the FFT follows this general implementation, as shown in the top-level schematic in Figure 8.

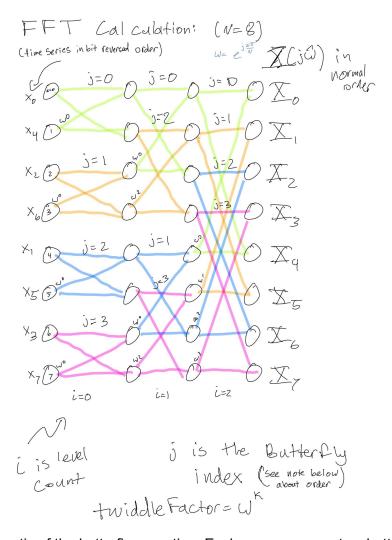


Figure 7: Schematic of the butterfly operation. Each cross represents a butterfly operation.

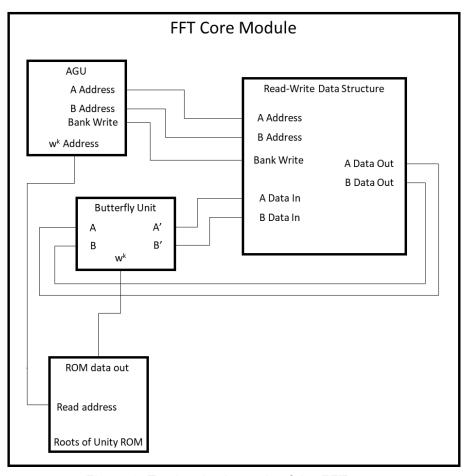


Figure 8: Top-level schematic of the FFT core

## **Butterfly Unit**

The operation of the butterfly unit is specified by (1) and (2) above. This arithmetic was specified combinationally. The resulting block diagram is shown in Figure 9. There are a couple of subtleties worth noting regarding the implementation of the butterfly unit. Firstly, A, B, and w<sup>k</sup> are generally complex. This means that each input must have a real and imaginary part, and multiplication requires four individual multipliers. Secondly, all of the signals are represented using Q15, similar to Slade in [1]. Q15 is a desirable number format because the product of two Q15 numbers is between -1 and 1. Furthermore, it is easy to represent the product as a Q15 number by simply shifting out the least significant bits of the product.

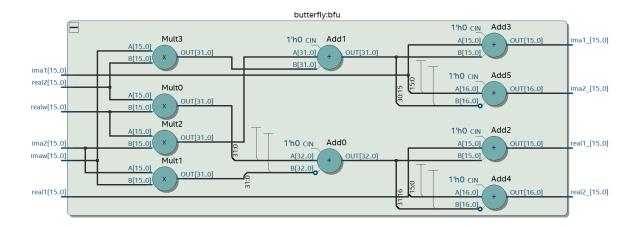


Figure 9: Schematic of butterfly unit

#### Read-Write Data Structure

Figure 10 shows a schematic of the memory structure. Because memory cannot be read from and written to on the same clock cycle, there are two memory banks and the module alternates reading from one and writing to the other at every FFT level. Theoretically, each memory bank consists of two (one for real/imaginary data each) dual-port RAM blocks. However, the current implementation uses registers instead of RAM, as the read address line of the RAM is registered, which would add a single clock cycle delay between the read addresses and the data out delay. Figure 11 shows the block diagram of a memory bank.

The time-series data is loaded into data bank 1 in bit reversed order, then the transformed data will end up in data bank zero in normal order (address 0 corresponds to X(0), etc.). Data is loaded into data bank 1 by asserting the "load" signal, and on the positive clock edge, the data on the Load Data lines will be loaded into the address at the bit reverse of the index line.

# Roots of Unity ROM

The roots of unity ROM is a 16 x 32 bit read-only memory that contains all of the roots of unity  $(w^0 \text{ through } w^{15})$  in Q15 format. The most significant bits correspond to the real part and the least significant bits correspond to the imaginary value. The address is the power of the root of unity (e.g.  $w^0$  will be stored in address zero).

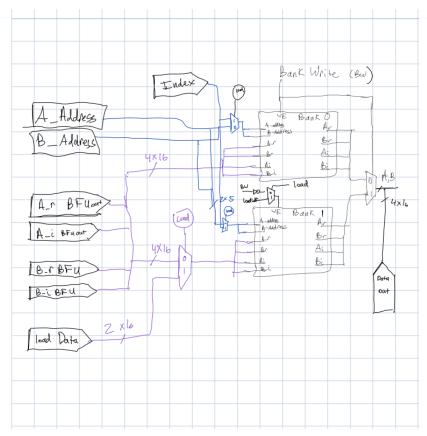


Figure 10: Schematic of the memory structure

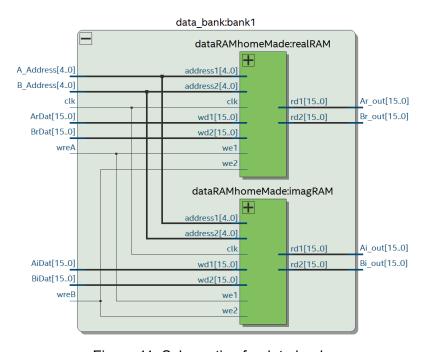


Figure 11: Schematic of a data bank

#### Address Generation Unit

The general strategy for address generation is outlined by Cohen in [3]. In short, if we do the butterfly operations in the order described by Cohen, there is a relatively simple relationship between the level (i) and *j* index and the addresses for A and B:

A\_address = Rotate<sub>5</sub>(
$$2*j$$
, i)  
B\_address = Rotate<sub>5</sub>( $2*j+1$ , i)

Furthermore, the correct root of unity address for butterfly (i,j) is found by zeroing out the lowest (4 - i) bits of j :

$$K = j \& [5b111111 << (5 - i - 1)]$$

We can keep track of i and j for the 32-point FFT with a single 7-bit counter. The lowest 4 bits are the j counter and the top 3 bits are the i counter. Figure 12 shows a schematic of the address generation unit.

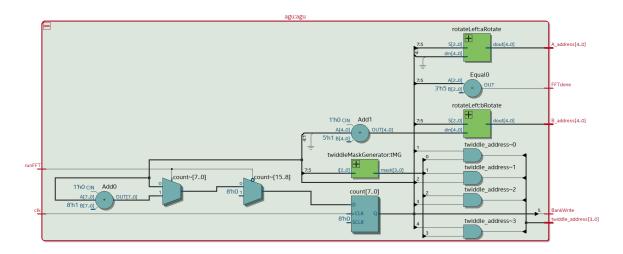


Figure 12: Schematic of the address generation unit. The twiddleMaskGenerator and the RotateLeft modules are combinational logic blocks.

#### FFT Core Function

To validate the functionality of the FFT core, a 32-point FFT was performed on the same signal (an impulse at n = 3) using Matlab's FFT function and using a ModelSim simulation of the FFT Core module. Figure 13 shows the results of these simulations. There is good agreement between Matlab FFT and the transform generated by the FFT Core Module.

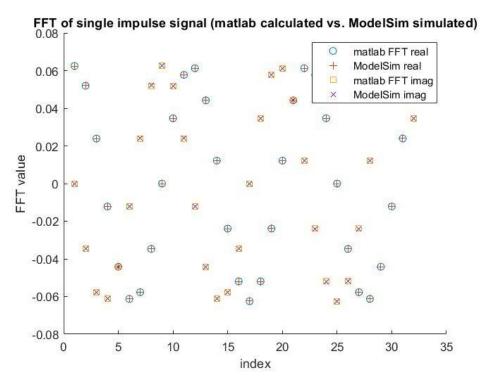


Figure 13: Comparison of FFT of test signal using Matlab and ModelSim

### FFT Core Module Interface

The inputs and outputs of the FFT core module are:

- Inputs:
  - Clk system clock that drives all synchronous modules
  - 2x loadData (16 bit) busses (1 r, 1 i)
  - 1x Index (5'bit)
  - Load control signal driven high to load/read data from FFTcore
  - runFFT control signal driven high to run FFT after load/read is completed
- Outputs:
  - 2x readData (16 bit) busses (1 r, 1 i,)
  - FFTdone signal goes high when FFT has finished

The FFT can either be in a done state or a running state. When the FFT is in a done state, the results of the previous transform can be read out simultaneously with the writing in of new data values. To do this, load must be asserted. When load is asserted, data is loaded into the data bank on the positive clock edge; the data on the Load Data lines will be loaded into the

addresses on the Load Addresses lines. Simultaneously, the data at the address on the read line will appear on the data lines to be readData lines. Importantly, there are two read/load addresses that are loaded/read simultaneously. This means that the data must be loaded in and read out in pairs. To have the FFT core run after loading is complete, drive load low and runFFT high. RunFFT must be driven high continuously until the FFTdone signal is asserted.

#### SPI Module

To load data into the FFT Core module, the SPI module must drive the index and loadData lines. To read data from the FFT core, the SPI module must be able to take in the read data. Figures 14 and 15 show block diagrams of the SPI module, which is built around a 32-bit shift register. The shift register shifts on the negative clock edge, and the SDI is strobed on the rising edge. This corresponds to a clock polarity of 0 and a clock phase of 0. The control signal generator is responsible for capturing the index, loading the read data into the shift register, and then capturing the load data after the read data is shifted out, as shown in Figure 15. The control signal generator is built around a counter, which keeps track of the number of negative clock edges so that the SPI module can correctly interpret the SPI communication.

The SPI module specifies the following 40-bit transmission protocol:

- Microcontroller sends: {5'b index, 3'b0, MSB real data, LSB real data, MSB imaginary data, LSB imaginary data}
- Microcontroller receives: {8'b X, MSB real data, LSB real data, MSB imaginary data, LSB imaginary data}

Sending the index first allows time for the data being read out of the FFT core to arrive on the Read Data line before it needs to be loaded into the shift register. Care was taken to ensure that the index and load data signals are properly synchronized so they do not cause metastability problems. For the index, this was achieved by having an "async" capture register that is clocked by sck and a "sync" index register that is clocked by clk. The enable for the synchronized clk index register only goes high after the "async" register has been settled.

To achieve synchronization for the Load Data line, the enable for the Load Data capture register is only high for one clock cycle when a synchronized count signal reads 40. However, this strategy only works if the frequency of sck is less than  $\frac{1}{2}$  of the frequency of clk. Because we used the 12 MHz oscillator on the FPGA as the system clock for the FFT module, this means the maximum sck frequency would be 6MHz. In practice, the sck frequency is set significantly lower at  $\sim$ 1.3 MHz.

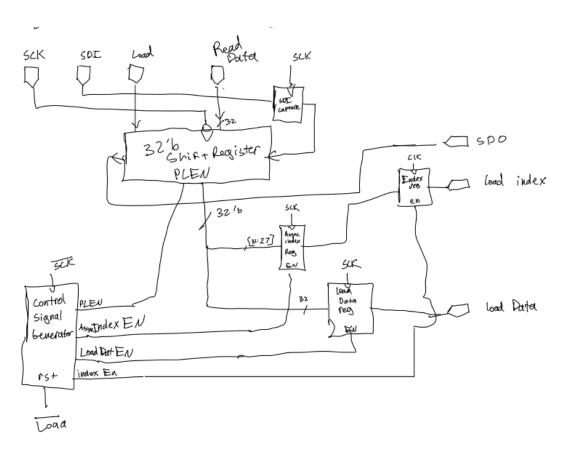


Figure 14: Block diagram of FFT SPI module

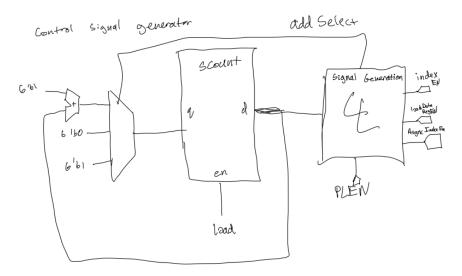


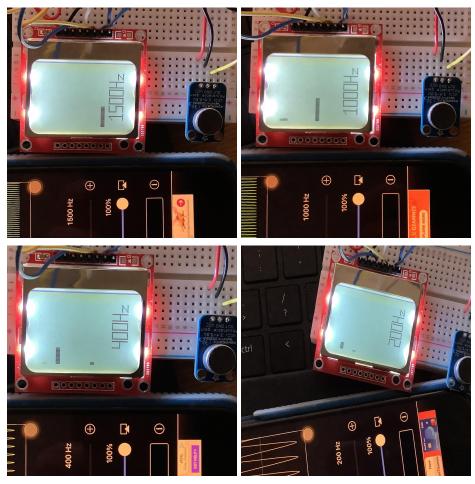
Figure 15: Block diagram of FFT SPI module control signal generator

## Results

The system was able to successfully meet all the specifications outlined in the project proposal:

- Sample audio from a microphone
- Perform Fourier analysis on the audio signal using the FPGA
- Display the spectrum of the audio signal on a graphic LCD

This behavior was verified through extensive testing, as shown in Figures 16-19 and videos linked below. We played different notes in succession, and observed that the frequency versus amplitude graph and dominant frequency displayed changed with the music, with no perceptible lag between the audio signal and the display. The graph also clearly reflects different sound volumes, such that when the music is played more softly, the amplitudes are correspondingly smaller on the graph. Additionally, we also played various pure tones, and observed that there was a clear dominant frequency (more than one in the case where the frequency did not fit exactly into one of the 100 Hz bins) that corresponded to the frequency of the tone.



Figures 16-19: The spectrum analyzer in operation

## Video Links

- Playing "Drop the Game" (<u>link</u>)
- Playing a series of pure tones (link)

#### References

- [1] The Fast Fourier Transform in Hardware: A Tutorial Based on an FPGA Implementation (https://web.mit.edu/6.111/www/f2017/handouts/FFTtutorial121102.pdf)
- [2] W. H. Press, S. A. Teukolsky, W. T. Vetterling, and B. P. Flannery, "Fast Fourier Transform," in Numerical Recipes in C, Cambridge Univ. Pr., 1992.
- [3] D. Cohen, "Simplified control of FFT hardware," in IEEE Transactions on Acoustics, Speech, and Signal Processing, vol. 24, no. 6, pp. 577-579, December 1976, doi: 10.1109/TASSP.1976.1162854.
- [4] Electret Microphone Amplifier MAX4466 with Adjustable Gain (https://www.adafruit.com/product/1063)
- [5] MAX4466 Low-Noise Microphone Amp Datasheet (https://cdn-shop.adafruit.com/datasheets/MAX4465-MAX4469.pdf)
- [6] Graphic LCD 84x48 Nokia 5110 (https://www.sparkfun.com/products/10168)
- [7] Graphic LCD 84x48 Nokia 5110 Datasheet (<a href="https://www.sparkfun.com/datasheets/LCD/Monochrome/Nokia5110.pdf">https://www.sparkfun.com/datasheets/LCD/Monochrome/Nokia5110.pdf</a>)

## **Bill of Materials**

Item	Vendor	Part Number	Quantity	Unit Price	Total Price
MAX1000	Trenz Electronic	TEI0001-03-0 8-C8	1	\$26.66	\$26.66
Nucleo-F401RE	Mouser		1	\$13.83	\$13.83
Electret Microphone Amplifier - MAX4466 with Adjustable Gain	Adafruit	ADA1063	<b>2</b> <sup>2</sup>	(1) \$6.95 (1) \$9.68	\$16.63
Graphic LCD 84x48 - Nokia 5110	SparkFun	LCD-10168	1	\$7.95	\$7.95
	\$65.07 + shipping				

 $^{2}$  Two microphones were purchased due to extenuating circumstances. Only one microphone is used in the system.

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# **Appendix A: FPGA Code**

# FFTcore and top level module:

```
timescale 1ns/1ns
module FFTcoreTestbench();
      logic runFFT;
      logic FFTdone;
      logic rst;
      logic [4:0] loadAddress, readAddress;
  FFTcore dut(clk, runFFT, load, loadAddress, readAddress, loadData r, loadData i,
FFTdone, readData r, readData i);
          rst = 1'b1;
          runFFT = 1'b0;
          clk = 1'b0; #5;
          runFFT = 1'b1;
           rst = 1'b0;
```

```
load = 1'b1;
         clk = 1'b0; #5;
         runFFT = 1'b0;
        clk = 1'b0; #5;
        clk = 1'b1; #5;
                imaginaryData[j] <= 16'h0000;</pre>
         i = 1'b0;
         finished = 1'b0;
  always @(posedge clk) begin
     if (i < 32 && ~FFTdone) begin
            readAddress = i[4:0];
readAddress[3], readAddress[4]);
            loadData r = realData[readAddress];
            loadData i = imaginaryData[readAddress];
```

```
else if (i == 32 && ~FFTdone) begin
          runFFT = 1'b1;
          load = 1'b0;
else if (FFTdone & ~finished) begin
          runFFT = 1'b0;
          load = 1'b1;
         i = 1'b0;
          finished = 1'b1;
         readAddress = i[4:0];
          realData[readAddress] = readData r;
 else if (finished && i < 32) begin
          readAddress = i[4:0];
      $stop();
```

```
output logic FFTdone);
  logic loadDone;
   logic [4:0] loadAddress, readAddress;
   logic loadHalfSync, loadSync;
  always ff @(posedge clk) begin
           loadHalfSync <= load;</pre>
           loadSync <= loadHalfSync;</pre>
   FFTcore core(clk, loadSync, loadAddress, readAddress, loadData_r, loadData_i,
FFTdone, readData r, readData i);
   onePoint spi spi(clk, sck, sdi, loadSync, readData, sdo, readAddress, loadData);
readAddress[3], readAddress[4]};
  assign loadData r = loadData[31:16];
```

```
loadAddress location
                                 [4:0] loadAddress, readAddress,
                                         FFTdone,
  logic BankWrite;
  logic [3:0] twiddle address;
imag1bank0, real2bank0, imag2bank0;
imag2_);
  agu agu(clk, runFFT, A address, B address, twiddle address, FFTdone, BankWrite);
  assign wreBOA = (~BankWrite && ~load);
  logic [4:0] bank0Address 1, bank0Address 2, bank1Address 1, bank1Address 2;
```

```
assign bank1Address 2 = B address;
  assign real1 = BankWrite ? real1bank0 : real1bank1;
  assign real2 = BankWrite ? real2bank0 : real2bank1;
  assign imag1 = BankWrite ? imag1bank0 : imag1bank1;
  data bank bank0(clk, wreB0A, wreB0B, bank0Address 1, bank0Address 2, real1 ,
imag1_, real2_, imag2_, real1bank0, imag1bank0, real2bank0, imag2bank0);
bank1datal i, bank1data2 r, bank1data2 i, real1bank1, imag1bank1, real2bank1,
imag2bank1);
  assign readData i = imag1bank0;
      if (load) isDone <= 1'b0;
      else if(FFTdone) isDone <= 1;</pre>
```

```
module twiddle_rom(input logic
assign realw = rom[address][31:16]; // TODO: can this be in hex?
endmodule
module data bank(input logic
                                  clk, wreA, wreB,
  logic clkEn = 1'b1;
Ar out, Br out);
  dataRAMhomeMade imagRAM(A Address, B Address, clk, AiDat, BiDat, wreA, wreB,
Ai out, Bi out);
endmodule
module dataRAMhomeMade(input logic [4:0] address1, address2,
      logic [15:0] RAM[31:0];
      always ff @(posedge clk)
```

```
if (we1) RAM[address1] <= wd1;
    if (we2) RAM[address2] <= wd2;
    end
endmodule</pre>
```

```
module butterflyTestBench();
ima2 ;
ima2 );
          real1 =16'b0;
          ima1 =16'b0;
          ima2 =16'b0;
          imaw = 16'b0;
          realw =16'h7fff;
           real1 =16'b0000000101000111;
           ima1 =16'b0000001010001111;
           ima2 =16'hfeb8;
endmodule
```

```
module butterfly(input logic signed [15:0] real1, ima1, real2, ima2,
bits in 16-bit container (5 bits for bit growth)
  logic signed [31:0] mult1, mult2, mult3, mult4;
  assign mult2 = ima2*imaw;
  assign mult3 = realw*ima2;
  assign mult4 = imaw*real2;
```

```
rst = 1'b1; #5;
           clk = 1'b0; #5;
           clk = 1'b1; #5;
   always @(posedge clk) begin
       if(~runFFT)
                   rst = 1'b0;
                   runFFT = 1'b1;
                  rst = 1'b1;
     if(FFTdone)
       $stop();
endmodule
module agu(input logic clk, runFFT,
          output logic FFTdone, BankWrite);
```

```
if (!runFFT)
                              count <= 8'b0;
rotateLeft aRotate(arotIn, i, A address);
rotateLeft bRotate(brotIn, i, B address);
twiddleMaskGenerator tMG(i, twiddleMask);
logic justStarted;
logic wasRunFFT;
    wasRunFFT <= runFFT;</pre>
    justStarted <= runFFT & ~wasRunFFT;</pre>
always ff @(posedge clk) begin
    if (justStarted) finished <= 1'b0;</pre>
```

```
assign FFTdone = finished;
endmodule
module rotateLeft(input logic [4:0] din,
           3'b000 : dout <= din;
           3'b010 : dout <= {din[2:0], din[4:3]};
           3'b110 : dout <= {din[3:0], din[4]};
endmodule
module twiddleMaskGenerator(input logic [2:0] i,
```

```
case(i)
    3'b000 : mask <= 4'b0000;
    3'b001 : mask <= 4'b1000;
    3'b010 : mask <= 4'b1100;
    3'b011 : mask <= 4'b1110;
    3'b100 : mask <= 4'b1111;
    default: mask <= 4'b0000;
endcase
endmodule</pre>
```

### SPI module:

```
timescale 1ns/1ns
 logic clk, sck, cs, MISO, MOSI, MISOcapture;
 logic [31:0] memBank0[31:0], memBank1[31:0];
```

```
cs = 1'b1; #5;
cs = 1'b0; #5;
cs = 1'b1;
clk = 1'b0; #5;
            MISOcapture = MISO;
            sck = 1'b0;
            master_buffer = {master_buffer[38:0], MISOcapture};
```

```
cs = 1'b0;
   $stop();
logic sdi_strobed, notCaptured;
logic [31:0] shift register;
```

```
always ff @(posedge sck)
         if(sCount == 7) shift_register <= readData;</pre>
  always ff @(posedge sck)
     if(\simcs) index <= 5'b0;
     else if(sCountSync > 5) index <= asyncIndex;</pre>
  assign sdo = shift register[31];
  always ff @(posedge clk)
      if (sCountSync == 40 && notCaptured) begin
            loadData <= shift register;</pre>
            only sampled once
```

```
else if (sCountSync == 2) notCaptured <= 1'b1; // Once sCount passes 2,
notCaptured resets high
    else if (~cs) loadData <= 32'h0;

// sck falling edge counter
always_ff @(negedge sck or negedge cs) begin
    if (!cs)
        sCount <= 6'b0;
    else if (sCount == 40) begin // this allows for consecutive read/write
transmissions without driving cs low.
        sCount <= 6'b000001;
        end
    else
        sCount <= sCount + 6'b1;
end

// scount synchronizer for use with registers clocked with clk.
logic [5:0] sCountHalfSync, sCountSync;
always_ff @(posedge clk) begin
        sCountHalfSync <= sCount;
        sCountSync <= sCountHalfSync;
end
endmodule</pre>
```

## **Top Level Testbench:**

```
logic [4:0] index;
        load = 1'b0;#2;
        load = 1'b1;
           clk = 1'b0; #5;
           clk = 1'b1; #5;
```

```
sck = 1'b1;
            MISOcapture = sdo;
            #42;
            sck = 1'b0;
            sdiBuffer = {sdiBuffer[38:0], MISOcapture};
load = 1'b0;
            MISOcapture = sdo;
            #42;
            sdiBuffer = {sdiBuffer[38:0], MISOcapture};
            #42;
```

```
readData_i[j[4:0]] = sdiBuffer[15:0];
end

// Stop.
// At this point, misoED should have same contents as dataBank0
// and dataBank1 should have same contents as dataBank0

$stop();
end

always @ (posedge clk) begin
    if (FFTdone) load = 1'b1;
end
endmodule
```

## twiddle\_rom.txt

```
7fff0000
7d891859
764130fb
6a6d471c
5a825a82
471c6a6d
30fb7641
18f97d89
00007fff
e7077d89
cf057641
b8e46a6d
a57e5a82
9593471c
89bf30fb
82771859
```

## **Appendix B: Microcontroller Code**

```
#include <math.h>
#include "STM32F401RE.h"
#define LOAD PIN
#define DONE PIN
#define SCE LCD PIN
#define ADC PIN
#define LCD RESET PIN
#define LCD DC PIN
#define LCD BL PIN
#define LCD COLS PER FREQ 4
#define LCD NUM COLS
#define LCD NUM ROWS
#define DISPLAY RAM SIZE 4032
#define LCD CHAR COLS
#define LCD_CHAR_ROWS
#define LCD CHAR SIZE
#define AUDIO DATA SIZE
#define FFT DATA SIZE
#define SAMPLING FREQ
#define FUNDAMENTAL FREQ
```

```
char char4[LCD_CHAR_SIZE] = {0, 0, 0, 0, 0, 0,
```

```
void initLCD();
void getAudioData(uint16 t *audioData);
void performFFT(uint16 t *fftInput, uint16 t *fftOutputReal, uint16 t *fftOutputImag);
void displayFFT(double *fftOutput);
void clearDisplay();
int displayDominantFrequency(char *displayRAM, int index, int dominantFreq);
int appendCharColumn(char *displayRAM, int index, char *charArray, int
charStartIndex);
double q15toDouble(uint16 t q15val);
double getMagnitude(double real, double imag);
  configureFlash();
  configureClock();
  RCC->AHB1ENR.GPIOAEN = 1;
  RCC->AHB1ENR.GPIOBEN = 1;
  RCC->APB2ENR.ADC1EN = 1;
   RCC->APB2ENR.SPI1EN = 1;
  RCC->APB1ENR.TIM2EN = 1;
  RCC->APB1ENR.TIM5EN = 1;
  pinMode (GPIOA, ADC PIN, GPIO ANALOG);
  pinMode (GPIOB, LOAD PIN, GPIO OUTPUT);
  pinMode(GPIOA, DONE PIN, GPIO INPUT);
  configureADC();
   initDMA2((uint32 t) &(ADC1->DR), (uint32 t) &audioData, AUDIO DATA SIZE); //
```

```
Memory: array buffer
  spiInit(0b101, 0, 0);
  initTIM5();
  digitalWrite(GPIOB, SCE LCD PIN, 1); // active low
  initLCD();
  clearDisplay();
  uint16_t fftInput[AUDIO_DATA_SIZE];
  uint16 t fftOutputReal[FFT DATA SIZE];
      getAudioData(audioData);
      delay micros(1000);
      performFFT(audioData, fftOutputReal, fftOutputImag);
```

```
fftOutputMagnitudes[i] = 0;
              fftOutputMagnitudes[i] = getMagnitude(
                  q15toDouble(fftOutputReal[i]),
                  g15toDouble(fftOutputImag[i])
      displayFFT(fftOutputMagnitudes);
void initLCD() {
  pinMode (GPIOA, LCD RESET PIN, GPIO OUTPUT);
  pinMode(GPIOA, LCD_BL_PIN, GPIO_OUTPUT);
  digitalWrite(GPIOA, LCD BL PIN, 1);
  digitalWrite(GPIOA, LCD RESET PIN, 0);
  digitalWrite(GPIOA, LCD RESET PIN, 1);
  digitalWrite(GPIOA, LCD DC PIN, 0);
  digitalWrite(GPIOB, SCE_LCD_PIN, 0);
  spiSendReceive(0x22); // Set PD = 0 (chip is active),
  digitalWrite(GPIOB, SCE LCD PIN, 1);
```

```
digitalWrite(GPIOB, SCE_LCD_PIN, 0);
  spiSendReceive(0x0C); // Set DE = 10 (normal mode)
  digitalWrite(GPIOB, SCE LCD PIN, 1);
  digitalWrite(GPIOA, LCD DC PIN, 1);
     while(!(TIM2->SR & 1));
void performFFT(uint16 t *fftInput, uint16 t *fftOutputReal, uint16_t *fftOutputImag)
  digitalWrite(GPIOB, LOAD PIN, 1);
  digitalWrite(GPIOB, LOAD PIN, 1);
  delay micros(1000);
     spiSendReceive(i << 3);</pre>
     while(SPI1->SR.BSY);
     while (SPI1->SR.BSY);
     while (SPI1->SR.BSY);
```

```
spiSendReceive(0);
      while (SPI1->SR.BSY);
      spiSendReceive(0);
      while (SPI1->SR.BSY);
  delay micros(1000);
  digitalWrite(GPIOB, LOAD PIN, 0);
be read out
  while(!digitalRead(GPIOB, DONE PIN));
  delay micros(1000);
  digitalWrite(GPIOB, LOAD PIN, 1);
      spiSendReceive(i << 3);</pre>
      while (SPI1->SR.BSY);
      fftOutputReal[i] = spiSendReceive(i);  // Receive {MSB r}
      fftOutputReal[i] = fftOutputReal[i] << 8;</pre>
      while(SPI1->SR.BSY);
      fftOutputReal[i] |= spiSendReceive(i); // Receive {LSB r}
      while (SPI1->SR.BSY);
      fftOutputImag[i] = spiSendReceive(i);  // Receive {MSB i}
      while (SPI1->SR.BSY);
      fftOutputImag[i] |= spiSendReceive(i); // Receive {LSB i}
      while (SPI1->SR.BSY);
  digitalWrite(GPIOB, LOAD PIN, 0);
void displayFFT(double *fftOutputMagnitudes) {
```

```
int dominantFreq = 0;
   if (fftOutputMagnitudes[i] > maxAmplitude) {
       maxAmplitude = fftOutputMagnitudes[i];
index = displayDominantFrequency(displayRAM, index, dominantFreq);
```

```
displayRAM[i+4] << 4 | displayRAM[i+5] << 5 |</pre>
       spiSendReceive(pixels);
       digitalWrite(GPIOB, SCE LCD PIN, 1);
void clearDisplay() {
      spiSendReceive(0b0);
int displayDominantFrequency(char *displayRAM, int index, int dominantFreq) {
  char digitsReversed[4];
      digitsReversed[0] = 0;
```

```
digitsReversed[i] = dominantFreq % 10;
      dominantFreq /= 10;
       index = appendCharColumn(displayRAM, index, charZ, col * LCD CHAR ROWS);
      index = appendCharColumn(displayRAM, index, charH, col * LCD CHAR ROWS);
                   index = appendCharColumn(displayRAM, index, char0, col *
LCD CHAR ROWS);
                   index = appendCharColumn(displayRAM, index, char1, col *
LCD CHAR ROWS);
                   index = appendCharColumn(displayRAM, index, char2, col *
LCD CHAR ROWS);
                   index = appendCharColumn(displayRAM, index, char3, col *
LCD_CHAR_ROWS);
                   index = appendCharColumn(displayRAM, index, char4, col *
LCD CHAR ROWS);
                   index = appendCharColumn(displayRAM, index, char5, col *
LCD CHAR ROWS);
                   index = appendCharColumn(displayRAM, index, char6, col *
LCD CHAR ROWS);
```

```
index = appendCharColumn(displayRAM, index, char7, col *
LCD CHAR ROWS);
                   index = appendCharColumn(displayRAM, index, char8, col *
LCD CHAR ROWS);
                   index = appendCharColumn(displayRAM, index, char9, col *
LCD CHAR ROWS);
          displayRAM[index] = 0;
int appendCharColumn(char *displayRAM, int index, char *charArray, int charStartIndex)
       displayRAM[index] = charArray[charStartIndex + j];
```

```
// Convert Q15 to double
double q15toDouble (uint16_t q15val) {
    uint16_t mask = 1 << 15;
    double doubleValue = 0;
    double difference;

if (mask == (mask & q15val)) {
        doubleValue = -1;
    }

for (int i = 0; i < 15; ++i) {
        mask = 1 << i;
        if (mask == (mask & q15val)) {
                difference = 15-i;
                doubleValue += pow(0.5, difference);
        }
    }

return doubleValue;
}

// Calculate magnitude of vector from real and imaginary components
double getMagnitude(double real, double imag) {
    return sqrt(pow(real, 2) + pow(imag, 2));
}</pre>
```

```
#include "STM32F401RE DMA.h"
void initDMA2(uint32 t peripheralAddress, uint32 t memoryAddress, uint16 t numData) {
  DMA2 -> SOCR.PL = Ob10;
  DMA2->SOCR.MSIZE = 0b01;
  DMA2->SOCR.PSIZE = 0b01;
  DMA2 -> SOCR.MINC = 1;
  DMA2->SOCR.CIRC = 1;
  DMA2 -> SOCR.EN = 1;
```

```
// STM32F401RE_FLASH.c
// Source code for FLASH functions
```

```
#include "STM32F401RE_FLASH.h"

void configureFlash() {
   FLASH->ACR.LATENCY = 2; // Set to 0 waitstates
   FLASH->ACR.PRFTEN = 1; // Turn on the ART
}
```

```
#include "STM32F401RE SPI.h"
#include "STM32F401RE RCC.h"
#include "STM32F401RE GPIO.h"
void spiInit(uint32 t br, uint32 t cpol, uint32 t cpha) {
  pinMode(GPIOA, 5, GPIO_ALT);
  pinMode(GPIOA, 7, GPIO ALT);
  GPIOA->OSPEEDR |= (0b11 << 2*5);</pre>
   GPIOA->AFRL |= (0b101 << 4*5) | (0b101 << 4*6) | (0b101 << 4*7);
   SPI1->CR1.CPOL = cpol; // Set the polarity
   SPI1->CR1.CPHA = cpha; // Set the phase
   SPI1->CR1.SSM = 0;
```

```
SPI1->CR1.MSTR = 1;
  SPI1->CR1.SPE = 1;
uint8 t spiSendReceive(uint8 t send) {
  SPI1->DR.DR = send;
  while(!(SPI1->SR.RXNE));
uint16 t spiSendReceive16(uint16 t send) {
  SPI1->CR1.SPE = 1;
  SPI1->DR.DR = send;
  SPI1->CR1.SPE = 0;
```

```
TIM2 -> EGR \mid = 1;
void initTIM5(){
TIM5->PSC = (psc div - 1);
TIM5->DIER.UIE = 1;
TIM5->EGR |= 1;
TIM5->CR1 |= 1; // Set CEN = 1
void delay_millis(uint32_t ms) {
TIM5->EGR |= 1;  // Force update
TIM5->CNT = 0; // Reset count
while(!(TIM5->SR & 1)); // Wait for UIF to go high
void delay_micros(uint32_t us) {
TIM5->EGR |= 1; // Force update
TIM5->SR &= \sim (0x1); // Clear UIF
```