# Musical Instrument Tuner 

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#### Abstract

: Most musicians use special devices to tune their instruments. Tuning devices can be as simple as tuning forks which vibrate at specific frequencies, or more user-friendly devices such as analog strobe tuners or digital tuners. This project prototypes a digital tuner consisting of a keypad, microcontroller, microphone, and LED display. The user selects the note they wish to tune by pressing a key on the keypad. This note letter is displayed for the user's convenience. The user then plays the note into the microphone. The microcontroller samples the note played and performs a Fast Fourier Transform (FFT) on it. The microcontroller sends the primary frequency of the note to an FPGA, where glue logic determines the sharpness or flatness of the note. The result is indicated on a row of LEDs. This digital tuner performs as expected within limitations of the implementation.


## Introduction

## Background:

We have constructed a working tuner for use in musical applications. Standard musical convention has defined certain frequencies as "notes". Sharp notes sound at a slightly higher frequency than the standard. Flat notes are slightly lower in frequency. Musicians use audio tuners to tune their instruments properly. Our tuner gives a visual indication of the flatness or sharpness of a played note.

## Specifications:

Our tuner is configured to tune the 6 standard strings of a guitar (E, A, D, G, B, high e). It uses a microphone to obtain an input note. The user chooses the note that they wish to tune to by using a keypad for input. The tuner uses a standard seven-segment display to show the note tested, a series of 7 LEDs to indicate note quality, and another LED to show if the note played is out of range.


Figure 1: LED Display

## Method:

Our tuner uses a microphone to convert the input note into an analog voltage. This analog voltage is then read by the $68 \mathrm{HC11}$ and converted to an array of digital values. The $68 \mathrm{HC11}$ EVB takes the FFT of this data array and determines the strongest frequency component of the input note. The 68 HC 11 passes this frequency over a parallel port to the FPGA. This frequency is compared against a fixed frequency that was determined by the note that the user had chosen. The difference in frequencies is calculated; and the result is used to turn on the appropriate LED.

The EVB is used for A/D conversion and FFT computation. The FPGA utility board is used for input and output interfacing and the frequency comparison.

## Materials:

This project uses:

- 68 HC 11 EVB
- Utility board with FPGA
- Protoboard
- 4X4 Matrix keypad
- Seven segment display
- LEDs
- Microphone
- Amplifier


## Functional Diagram:



Figure 2: Block Diagram

## New Hardware

## Microphone:

The tuner uses a microphone element to acquire an input note. Interfacing the microphone is simple but worth mentioning here in case any future students wish to use a microphone input. Our microphone interface consists of two stages. The first is the microphone element itself, and the second is an amplifier circuit whose output is the input to the tuner system.

Our microphone is a balanced, three-terminal device. Pin 1 takes power, pin 3 is grounded, and the output signal is on pin 2. We found that a range of DC voltages can be used to power the microphone. We power it with 15 VDC through a circuit with two series resistors and a shunt capacitor to ground between them. The resistors are $1 \mathrm{k} \Omega$ each, and the capacitor is $100 \mathrm{uF} / 25 \mathrm{~V}$ (electrolytic). We have chosen these component values per recommendation from the microphone element manufacturer. The output from pin 2 has a DC level of about 0.3 V and swings on the order of a few millivolts.

Because of the small output, we use an inverting op-amp amplifier to boost the microphone output before sending it to the rest of the tuner system. Because we desire to control the tuner input's DC offset, we use a $0.1 u \mathrm{~F}$ capacitor at the microphone output to block its 0.3 V offset. This signal then goes to an inverting amplifier constructed with a $411 \mathrm{op-amp}$ and two resistors which control the gain. Our resistor values are $750 \Omega$ (at the negative input) and $910 \mathrm{k} \Omega$ (in the feedback path). We have arrived at these values experimentally, as they provide an output sufficient for the tuner. +15 and -15 VDC power the op-amp, and its output feeds to the rest of the tuner system.

Future microphone element users can handle the microphone output however they choose to; the important details to remember are that the output has a small DC offset and has a typical amplitude of only a few millivolts. Also, experimentation may reveal that the power input to the microphone can be handled with different component values or even a different circuit.

## Schematics

Show and describe schematics of anything on your breadboard, including interfacing of the $\mathrm{HC11}$ and FPGA board. If you constructed a printed circuit board instead, document the function, schematics, and layout of your board.


## Microcontroller Design

The microcontroller subsystem of the tuner accepts the user input note, calculates its frequency spectrum using a FFT, determines the note's strongest frequency, and sends a representation of this frequency to the FPGA over parallel port B.

## A/D Conversion of an Input Signal:

The microcontroller input is a signal on channel PE7. This input signal is converted to the digital domain with A/D reference voltages of +5 V and 0 V . Our A/D conversion routine takes 256 samples at a rate of about 1.8 kHz . The value of each sample is written to a byte in memory beginning at \$DD00. This data block, then, occupies addresses $\$ \mathrm{DD} 00$ to $\$ \mathrm{DDFF}$. The A/D subroutine contains a loop that writes an A/D result to memory 256 times. At the beginning of each time through the loop (labeled cklp), the data block pointer is incremented, and once it equals $\$ \mathrm{DE} 00$, the loop breaks and the $\mathrm{A} / \mathrm{D}$ is complete. Within this overall loop, eight 128 clock cycle loops slow down the effective $\mathrm{A} / \mathrm{D}$ sampling rate. The maximum $\mathrm{A} / \mathrm{D}$ sampling rate with a 2 MHz E clock on the 68 HC 11 is 62.5 kHz . Reading only one result after a completion of four conversions reduces the effective sampling rate to 15.625 kHz . Performing A/D conversions without saving results can further slow this rate. The loops labeled wait1 - wait7 begin with a write to the $\mathrm{A} / \mathrm{D}$ control register (at $\$ 1030$, thus beginning an $\mathrm{A} / \mathrm{D}$ conversion sequence) and break only after four conversions are complete (indicated by CCF in ADCTL being set). The final "wait" loop also uses 128 clock cycles by checking the CCF flag, and then the routine proceeds to read an A/D conversion result from the first result register (ADR1 at \$1031). This value is then written to the data block. Dividing 15.625 kHz by 8 in addition to extra clock cycles from other instructions in the A/D loop gives us an effective sampling rate of approximately 1.8 kHz .

## Calculate the $\mathbf{2 5 6}$ Point FFT:

To implement the FFT, we adapt code (found on the Internet) written by Ron Williams at Ohio University. The FFT has three main components. The code that we used actually added a fourth component to further speed up the process. The first step in the FFT is bit reversal; this step reorganizes data so that subsequent calculations can be performed using simple looping structures. The second step in the FFT is the first pass, this function takes advantage of the fact that in the first pass of the FFT there are no imaginary components, therefore all the components are simply multiplied by 1,0 , or -1 . The final pass, is the brute force calculation of the FFT, but by performing the previous steps, this step is considerably sped up. Finally, the code that we used utilized another shortcut; instead of calculating sine and cosines in assembly (a lengthy and complicated process utilizing floating point arithmetic), it used a sine lookup table.

The bit reversal rearranges the order of the data; it will become apparent in later steps why this process is helpful. This process is best explained with diagrams. The data that is located at address A is swapped with the data at address B . The relation between address A and address B is that the most significant bit of A is equal to the least significant bit of $B$. A smaller scale example is shown below.


Figure 3: Bit reversal

Our data set has more elements, but the principle is the same. The data in \$DD00 and \$DDFF remain in the same locations, but for instance, data in \$DD41 is swapped with the data in \$DD82, etc. This is done at label revbit. Label rev1 uses a loop to reverse the bits of address one; then it saves the data from address two (the bit reversed address one). Next it stores the value in address one to address two. Finally it stores the saved data to address one. It continues in this fashion until the entire 256 -element array has been rearranged.

The next step is to sum together the first pass components. In an 8-bit transform the first pass will be only in multiples of 180 degrees, and differences of 4 cells apart. It turns out that for any size array that is a multiple of 2 ; the bit reversal puts the necessary multiples next to each other. So the paired values are added and subtracted together. The example is continued below.


Figure 4: Calculations after bit reversal
This is done at label fpss. This is a well-commented section. To clarify data one and data two are summed and stored in address one, and data one and data two are subtracted and stored in address two. Then the algorithm moves on to the next pair. This process is repeated until the entire array has been completed.

After this step, the process must continue as a standard DFT, but again the reordering of the bits is crucial, because it makes the looping process considerably easier. The next step would pair in twos, the pass after that would pair off in fours, and the final pass would be the first and eighth cell. Unfortunately, due to the fact that the angles are not "nice", there are going to be cosines and imaginary sine products. This calculation is helped considerably by the sine lookup table, which is a table of fractional values of 128 to approximate the sine/cosine function. This function begins at label four to initialize variables, but each successive pass loops back to label npass. Essentially, npass uses a subroutine smul to multiply the data by their respective angle offset cosine and sine. These are then summed and subtracted in the usual method and then stored in their respective data locations. Each time that npass is done, the data is re-scaled if necessary to ensure that there is no overflow.

Unfortunately, implementation of the FFT in assembly is considerably difficult. The algorithm for addressing can only be done for sample cases less than or equal to 256 (the maximum number of addresses expressible in a single byte of information). Furthermore since each data address can only hold an integer value from 0 to 256 which must be converted to two's complement, the resolution of the data is poor. The code utilized a scaling function to ensure that there would be no overflow in data.

Finally, all of these issues combined in an unforeseen bug in the program. The DC offset (the first address in memory) was very large in our sample data (it required a large offset to protect the ADC circuits in the HC6811). Unfortunately, this left us in a difficult position. If we included the DC offset in the scaling function, then the other harmonics would be scaled out rapidly (the DC offset was much stronger than the fundamental frequency of the note). And yet, the DC offset is critical to the FFT function. The solution
that we came to was to zero out the DC offset position throughout the calculation. Unfortunately, this made our implementation of the FFT inaccurate, however it was accurate enough to distinguish characteristic frequencies. While this is not the most satisfying solution, it did work.

## Absolute Value of FFT Results:

After the FFT routine stores its data in the data block, we convert it out of two's compliment form by using an absolute value subroutine. The subroutine loops 256 times until it has looked at every value in the data block. Within the loop (labeled all256), the value at the data pointer is AND masked with \# $\$ 80$ to determine whether it is positive or negative. If it is already positive, the data pointer is incremented and the all256 loop rerun. If the value is negative, its two's compliment is obtained by flipping its bits with the COMB instruction and adding one. After rewriting the original value's absolute value to the memory block, the all256 reruns.

This subroutine is not necessary for the program to continue and find the strongest frequency component; however, it allows us to look at a meaningful plot of the FFT result in a graphing program such as Excel.

## Find the Strongest Frequency Component:

The 68 HC 11 program uses another subroutine to find the strongest frequency component of the stored FFT. This routine examines only the first 127 data points after the DC component because the 256-point FFT is symmetric. This routine uses a modified version of David Honeycutt's program from Lab 5 which finds the largest of 5 numbers stored in consecutive memory locations. Our routine finds the largest of 256 values written to consecutive bytes of memory beginning at \$DD00, and when it does, saves the value's memory location to \$DDF0. Thus, the strongest frequency position of the FFT relative to DC is saved at \$DF01.

## Send Strongest Frequency to the FPGA Using Parallel Port B:

After finding the strongest frequency position, the 68 HC 11 program writes this position to parallel port B (at $\$ 1004$ ) to send the value to the FPGA board. After sending the data, the program resets the stack pointer to $\$ 004 \mathrm{~A}$ and loops back to obtain another 256 samples and process them. We reset the stack pointer because somewhere in the FFT routine, it increments by 2 . Resetting it after each time through the FFT loop ensures the program can run forever.

## FPGA Design

Our FPGA system handles user input on a keypad, input from the 68 HC 11 parallel port, and formats output for display LEDs. The hardware in the FPGA is similar to that generated for Lab 4. For the tuner, the FPGA performs subtraction and additional encoding.

## Display and Frequency Encoding (poll_kb):

Whereas in Lab 4 the keypad encoding process converted an 8-bit key press to a 4-bit value for display decoding, our FPGA also encodes a valid key press into another 8-bit value. This value is a representation of the standard note frequency to which the frequency position from the 68 HC 11 is compared.

We determined the frequency values encoded by key presses experimentally by running our FFT routine on signals with frequencies set by a function generator to standard note frequencies. They expectedly came out to be close to what we had calculated assuming a 7 Hz spacing between FFT data points. The note frequencies our tuner tunes to and their associated values in hardware are summarized here:

| Note | Frequency $(\mathrm{Hz})$ | Representation (hex) | Representation (decimal) |
| :--- | :--- | :--- | :--- |
| E | 164.81 | 18 | 23 |
| A | 220 | 20 | 31 |
| D | 293.66 | 2 B | 42 |
| G | 392 | 39 | 56 |
| B | 493.88 | 48 | 71 |
| e | 659.26 | 5 F | 94 |

Table 1: Note Frequencies

## Seven Segment Decoder Module (seven_seg_display):

Keypad encodings are modified to display 6 note names on the 7 -segment display corresponding to 6 keys, and to display dashes for the other keys (a sign of an invalid key press). The keys on the standard E157 keypad display the following:

| Key on keypad | Letter printed on display |
| :--- | :--- |
| A | A |
| 0 | G |
| B | b |
| D | d |
| E | E (for low E string on a guitar) |
| F | (for high E string on a guitar) |
| All rest | - |

Table 2: Key presses

## Compare Module and Output Decoding (compare):

Our FPGA system uses a separate module for comparison and output formatting. The compare module takes the value encoded by a key press as one input, and the value sent from the 68 HC 11 as a second input. A subtraction is performed, and the result is decoded into an 8-bit output value:

| Subtraction result (decimal) | Subtraction result (hex) | Decoded output (hex) |
| :--- | :--- | :--- |
| 1 | 01 | 10 |
| 2 | 02 | 20 |
| 3 | 03 | 40 |
| -1 | FF | 04 |
| -2 | FE | 02 |
| -3 | FD | 01 |
| 0 | 00 | 08 |
| Anything else | Anything else | 80 |

## Table 3: Comparison and output

The output is assigned to the pins connected to the 8 utility board LEDs. When the tuner is running, only one LED is lit at a time, such that the leftmost LED indicates a note out of range, and the rest show how close to "in tune" an input note is.

## Actual 7-segment Display:

While the tuner only uses one side of the 7-segment display to show a note name, we left the multiplexed display hardware in the FPGA for simplicity's sake. To only display on one side, we simply multiplex the same input key press twice and, externally, remove power to one side of the display.

## Results

## Tuner Response to Various Inputs:

Our project was successful, however the final product still has some problems. We found that our tuner could successfully tune all 6 notes that we had intended to tune to. The software implementation of the design runs an infinite loop, outputting a result approximately every 500 milliseconds. Unfortunately, the microphone posed a significant problem in analog electronics. The amplifier and microphone implementation was designed experimentally. Due to these analog difficulties, the tuner had difficulty tuning an actual guitar. We believe this is in part due to the complex waveform, as well as rapid drop off in strength of the guitar's note. We tried to use a speaker to output a continuous amplitude note, and the tuner's performance was slightly better, although still not completely reliable. If a signal generator is directly hooked up to the A/D converter, then performance is near perfect.

## Changes From the Original Proposal:

Our initial proposal suggested the usage of 7 LEDs. We thought that the 7 LEDs could give an accurate idea of the sharpness/flatness of the note. An improvement was made later when we added an eighth LED to show whether or not the frequency that was being played was out of range of the tuner's ability. Furthermore, for aesthetic appeal, we used different colors for the LEDs. The green center LED indicates "in tune", the six red LEDs indicate degree of flatness/sharpness. The out of range LED is yellow.

## Future Recommendations:

Future recommendations for this project are related to increasing the resolution of the tuner. This can be done in two ways. The first would be to increase the number of points in the FFT (therefore also increasing the number of $\mathrm{A} / \mathrm{D}$ samples). If this is done properly, higher resolution is possible; unfortunately, this approach requires a new method of addressing. The FFT algorithm that we used can only handle 256 different pieces of data because the bit reversal operation depends on byte long addresses. A simpler approach to this dilemma is to further slow down the A/D conversion. This would end up giving a smaller range of tuning ability, but it would give better resolution to the frequency peaks. Another possible improvement would be an automatic gain control for the microphone. Unfortunately, neither of us knows how to implement a proper AGC circuit. Finally, it would not take long to make an auto-tuning device; this could be implemented using a comparison loop for all of the valid frequencies. Only the frequency that the played note is actually closest to would be used for tuning calculations. Finally, our project can tune to any note within its frequency range, so modifying this tuner to tune a piano or trumpet would not be terribly difficult.

## Comical Side Note:

Once we got the guitar back to a commercial tuner, we realized that the guitar was more out of tune than it had been before we had used our tuner on it.

## References

[1] R. Lord, "Fast Fourier for the 6800." Byte, February 1979, pp. 108-119.
[2] ftp://ftp.stanford.edu/class/ee281/WWW97/routines/ffthc11.asm for FFT code

## Parts List

Components used in the tuner include:

| Part | Source | Vendor Part \# | Price |
| :--- | :--- | :--- | :--- |
| Microphone Element | MarVac on Holt Ave. | $10-84$ | $\$ 2.79$ |
| 411 Op-Amp | Electronics Lab |  |  |
| Table 4: Parts List |  |  |  |
|  |  |  |  |

## Appendices

## 68HC11 ASM code listing:

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*     - tuner.asm
*     - A 68HC11 program to continuously sample an
* input waveform, obtain its fourier spectrum,
* and output frequency information on parallel
* port B.
*     - By Rohit Mishra and Mark Wang
* for E157, Harvey Mudd College
*     - 12/9/99
* 
*     - One loop of the program runs in several stages.
* First, an input is acquired from an $A / D$ channel
* on PE7. 256 1-byte samples are taken and stored
* memory locations \$DD00 - \$DDFF.
* 
* Secondly, a 256 point FFT is performed on the
* sample space. The FFT algorithm was implemented
* by Ron Williams of Ohio University; his comments
* appear below.
* 
* After the FFT has been calculated, the location
* in memory of the strongest frequency is found.
* Its position relative to the DC component of the
* FFT is determined and written to parallel port B.
* 
* Before looping back to acquire another input
* sample, the stack pointer is reset to a
* specific value so as to ensure safe infinite
* looping.
*     - fast fourier transform for the 68HC11
* originally written by:
* Ron Williams
* Department of Chemistry
* Ohio University
* Athens, OH 45701
* 
*     - Excerpts of Ron Williams' comments:
* This is a modification of the 6800 FFT presented by:
* Richard Lord
* Byte Magazine, pp. 108-119
* February 1979
* 
* My version is written in ROMable code for the HC11.
* It uses a sine look-up table for speed and can only
* transform 256 8-bit data points. The program
* assumes that the address of the real data is pushed
* on the stack prior to the call and that a 256 byte
* imaginary buffer is at data+256 therefore you must
* declare a 512 byte data array in the calling routine
* and load the lower 256 bytes with data. The FFT
* will zero out the imaginary portion. Also note that
* the FFT uses memory in the stack RAM for its dynamic



| 0168 | c071 6a 14 |  |  | dec | tmp, X | decrement \# cells |
| :---: | :---: | :---: | :---: | :---: | :---: | :---: |
| 0169 | c073 26 e8 |  |  | bne | fpss | go back if not done |
| 0170 |  |  | * |  |  |  |
| 0171 |  |  | * now | E FFT | oper for | passes 2 thru N |
| 0172 |  |  | * |  |  |  |
| 0173 | c075 8640 |  | four | ldaa | \#64 | \# of cells is now 64 |
| 0174 | c077 a7 04 |  |  | staa | celnm, X | store |
| 0175 | c079 a7 08 |  |  | staa | delta, x | so is delta |
| 0176 | c07b 8602 |  |  | ldaa | \#02 | number of pairs is 2 |
| 0177 | c07d a7 06 |  |  | staa | pairnm, |  |
| 0178 | c07f a7 07 |  |  | staa | celdis, | X so is distance betw |
| 0179 | c081 bd c1 | 4d | npass | jsr | scale | check for over-range |
| 0180 |  |  |  |  |  |  |
| 0181 | c084 8600 |  | ldaa | \#\$00 | * st | re 0 to DC and \$40 |
| 0182 | c086 b7 dd | 00 |  | staa | \$dd00 | * (our bug fix for |
| 0183 | c089 b7 dd | 40 | staa | \$dd40 |  | roblems in Williams' |
| 0184 |  |  | * |  |  | mplementation) |
| 0185 | c08c a6 04 |  |  | ldaa | celnm, X | get current cell \# |
| 0186 | c08e a7 05 |  |  | staa | celct, X | store at cell counte |
| 0187 | c090 1a ee | 02 |  | ldy | real, X |  |
| 0188 | c093 1a ef | 0 e |  | sty | reall, X | get copy of data |
| 0189 | c096 18 ce | c1 b4 | ncell | ldy | \#sintab | get address of sines |
| 0190 | c09a 1a ef | 0c |  | sty | sinpt, X | save copy |
| 0191 | c09d a6 06 |  |  | ldaa | pairnm, | $X$ get current pairnm |
| 0192 | c09f 36 |  | np1 | psha |  | save pair counter |
| 0193 | c0a0 18 a6 | 00 |  | ldaa | 0, Y | get cosine |
| 0194 | c0a3 18 e6 | 40 |  | ldab | 64,Y | get sine |
| 0195 | c0a6 a7 0a |  |  | staa | cosa, X | save copy |
| 0196 | c0a8 e7 0b |  |  | stab | sina, X | ditto |
| 0197 | c0aa 1a ee | 0 e |  | ldy | reall, X | point to top of data |
| 0198 | c0ad e6 07 |  |  | ldab | celdis, | $X$ get current offset |
| 0199 | c0af 183 a |  |  | aby |  | add to Y for current |
| 0200 | c0b1 1a ef | 10 |  | sty | real2, X | copy it |
| 0201 | c0b4 18 a6 | 00 |  | ldaa | $0, Y$ | get data point rn |
| 0202 | c0b7 36 |  |  | psha |  | copy it |
| 0203 | c0b8 e6 0a |  |  | ldab | cosa, X | get cosine |
| 0204 | c0ba bd c1 | 92 |  | jsr | smul | $r n * \cos (\mathrm{a})$ |
| 0205 | c0bd a7 12 |  |  | staa | treal, X |  |
| 0206 | c0bf 32 |  |  | pula |  | get copy of rn |
| 0207 | c0c0 e6 0b |  |  | ldab | sina, X | get sin (a) |
| 0208 | c0c2 bd c1 | 92 |  | jsr | smul | $r n * \sin (\mathrm{a})$ |
| 0209 | c0c5 a7 13 |  |  | staa | timag, X | store imaginary tmp |
| 0210 | c0c7 1808 |  |  | iny |  |  |
| 0211 | c0c9 18 a6 | ff |  | ldaa | \$FF, Y | get imaginary data |
| 0212 | c0cc 36 |  |  | psha |  | save it |
| 0213 | c0cd e6 0b |  |  | ldab | sina, X | get $\sin (\mathrm{a})$ |
| 0214 | c0cf bd c1 | 92 |  | jsr | smul | in*sin(a) |
| 0215 | c0d2 ab 12 |  |  | adda | treal, X |  |
| 0216 | c0d4 a7 12 |  |  | staa | treal, X | tr=rn*cos + in*sin |
| 0217 | c0d6 32 |  |  | pula |  | get data back |
| 0218 | c0d7 e6 0a |  |  | ldab | cosa, X | get cosine |
| 0219 | c0d9 bd c1 | 92 |  | jsr | smul | in*cos(a) |
| 0220 | c0dc a0 13 |  |  | suba | timag, X | ti=in* $\cos -r n * \sin$ |
| 0221 | c0de a7 13 |  |  | staa | timag, X |  |
| 0222 | c0e0 1a ee | 0 e |  | ldy | reall, X |  |
| 0223 | c0e3 18 a6 | 00 |  | ldaa | 00, Y | get rm |
| 0224 | c0e6 16 |  |  | tab |  | save a copy |




0339 c1b4 7f 7f 7f 7f 7e 7e 7e 7d 7d 7c
0340 c1be 7b 7a 7a 797876 75747371
0341 c1c8 70 6f 6d 6b 6a 68 66646260
0342 c1d2 5e 5c 5a 585553 51 4e 4c 49
0343 c1dc 474441 3f 3c 39 $3633312 e$
0344 c1e6 2b 282522 1f 1c 19161310
0345 c1f0 0c 09060300 fd fa f7 f4 f0
0346 c1fa ed ea e7 e4 e1 de db d8 d5 d2
0347 c204 cf cd ca c7 c4 c1 bf bc b9 b7
0348 c20e b4 b2 af ad ab a8 a6 a4 a2 a0
0349 c218 9e 9c 9a 989695 $9391908 f$
0350 c222 8d 8c 8b 8a 8887 86868584
0351 c22c 838382828281 81818181
$\begin{array}{llllllll}0352 & c 236 & 81 & 81 & 82 & 82 & 82 & 83\end{array}$ 83848586
0353 c240 868788 8a 8b 8c 8d 8f 9091
0354 c24a $939596989 a 9 c$ 9e a0 a2 a4
0355 c254 a6 a8 ab ad af b2 b4 b7 b9 bc
0356 c25e bf c1 c4 c7 ca cd cf d2 d5 d8
0357 c268 db de e1 e4 e7 ea ed f0 f4 f7
0358 c272 fa fd 00030609 0c 101316
0359 c27c 19 1c 1f 222528 2b 2e 3133
0360 c286 $36393 c 3 f 4144$ 47494 c 4 e
0361 c290 51535558 5a 5c 5e 606264
0362 c29a 6668 6a 6b 6d 6f $\begin{array}{llll}70 & 71 & 73 & 74\end{array}$
0363 c2a4 $757678797 a 7 a$ 7b 7c 7d 7d
0364 c2ae 7e 7e 7e 7f 7f 7f 0365
0366
0367
0368
0369
0370
fcb 127, 127, 127, 127, 126, 126, 126, 125, 125, 124
fcb 123, 122, 122, 121, 120, 118, 117, 116, 115, 113
fcb 112, 111, 109, 107, 106, 104, 102, 100, 98, 96
fcb 94, 92, 90, 88, 85, 83, 81, 78, 76, 73
fcb 71, 68, 65, 63, 60, 57, 54, 51, 49, 46
fcb $43,40,37,34,31,28,25,22,19,16$
fcb $12, \quad 9,6,3,3,-3,-6,-9,-12,-16$
fcb $-19,-22,-25,-28,-31,-34,-37,-40,-43,-46$
fcb $-49,-51,-54,-57,-60,-63,-65,-68,-71,-73$
fcb $-76,-78,-81,-83,-85,-88,-90,-92,-94,-96$
fcb $-98,-100,-102,-104,-106,-107,-109,-111,-112,-113$
fcb $-115,-116,-117,-118,-120,-121,-122,-122,-123,-124$
fcb $-125,-125,-126,-126,-126,-127,-127,-127,-127,-127$
fcb $-127,-127,-126,-126,-126,-125,-125,-124,-123,-122$
fcb $-122,-121,-120,-118,-117,-116,-115,-113,-112,-111$
fcb -109,-107,-106,-104,-102,-100, -98, -96, -94, -92
fcb $-90,-88,-85,-83,-81,-78,-76,-73,-71,-68$
fcb $-65,-63,-60,-57,-54,-51,-49,-46,-43,-40$
fcb $-37,-34,-31,-28,-25,-22,-19,-16,-12,-9$
fcb $-6,-3,0,3,6,12,16,19,22$
fcb $25,28,31,34,37,40,43,46,49,51$
fcb 54, 57, 60, 63, 65, 68, 71, 73, 76, 78
fcb 81, 83, 85, 88, 90, 92, 94, 96, 98, 100
fcb 102, 104, 106, 107, 109, 111, 112, 113, 115, 116
fcb $117,118,120,121,122,122,123,124,125,125$
fcb $126,126,126,127,127,127$
**************************************

* A/D conversion routine for obtaining
* 256 samples on channel PE7. This
* routine samples at a rate of about
* 1.8 kHz .





## FPGA Verilog Modules:

See attached.

