

**LABORATORY EXPERIMENTS UNIFYING CONCEPTS IN THE
COMMUNICATIONS, DIGITAL SIGNAL PROCESSING (DSP) AND
VERY LARGE SCALE INTEGRATION (VLSI) COURSES**

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Abstract - The hallmark of the Rowan College of Engineering undergraduate program is to provide effective laboratory based instruction that illustrates important scientific concepts. This paper presents the results of an effort by the Department of Electrical and Computer Engineering at Rowan University to configure a novel method of teaching the junior level Communications (COMM), Digital Signal Processing (DSP) and Very Large Scale Integration (VLSI) courses under a common laboratory framework. These three courses are taken concurrently during the spring semester of the junior year. The described interdisciplinary experiments cut across individual course boundaries and integrate hands-on experience and software simulation. Software is integrated with the experiments through MATLAB and SIMULINK, C/C++ and Mentor Graphics.

Introduction

This project is an effort by the Department of Electrical and Computer Engineering at Rowan University to configure a novel method of teaching the junior level Communications (COMM), Digital Signal Processing (DSP) and Very Large Scale Integration (VLSI) courses under a common framework. These three courses are taken concurrently during the spring semester of the junior year. The effort is aimed at establishing a “proof of concept”.

There has been a historical division and separation of the fields of Communications, DSP and VLSI in electrical engineering education. This separation has crept up to the very high professional circles in both industry and university. Engineers specialized in one area find it hard to collaborate with their colleagues and separate cliques within the department start to form. This

type of segregation is no longer acceptable as we must provide an integrative experience at the undergraduate level.

Rowan University began as a teacher education institution. It then evolved into a comprehensive state college and now into a university. The School of Engineering is a recent expansion for the college; a major gift in 1992 from the Rowan Foundation was the catalyst for adding engineering. Our new engineering programs seek to use innovative methods of teaching and learning to better prepare students for entry into a rapidly changing and highly competitive marketplace. Key program features include: (1) an analytical and hands-on balance created through collaborative laboratory and lecture material; (2) an emphasis on teamwork as the necessary framework for solving complex problems; (3) incorporation of appropriate technologies throughout the curricula; and (4) creation of continuous opportunities for technical communication. To best meet these objectives, our programs include a multidisciplinary *engineering clinic* every semester [1][2][3]. Sharing many features in common with the model for medical training, the clinic provides an atmosphere of faculty mentoring in a hands-on, laboratory setting. In addition to the clinic, specialized courses are taught to deliver a well blended combination of theoretical and practical skills. This project is in accordance with the aims of our new programs and strives to meet the requirements of industry in hiring engineers who can move across rather artificial course boundaries with great ease.

The common educational framework we envision will enable the students to better comprehend the conceptual relationships of COMM, DSP and VLSI. Implementation of our ideas is facilitated by the fact that the three courses are run in the same semester. Each course has three hours of lectures and three hours of laboratory per week. The illustrative laboratory experiences enforce the conceptual relationships. This philosophy is further motivated by the need to promote the two main learning styles that students have [4]. Most students, instructors and curricula are sequential in that the process functions with partial understanding, there is steady progress, and details are emphasized [4]. Global learners need the big and overall picture for proper comprehension and progress in leaps despite being slow initially [4]. The present implementation of our curriculum caters extremely well to the majority of students that adopt sequential learning. However, stressing the common framework and thereby the big picture that envelopes COMM, DSP and VLSI accomplishes the very important need of addressing the

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minority of global learners who would otherwise be weeded out and be a serious loss to society [4].

Goals and Objectives

We want to accomplish the following:

1. Expose students to the different aspects and conceptual relationships among COMM, DSP and VLSI. This will be achieved by (1) laboratory experiences that cut across the course boundaries and (2) tuning the lecture material to stress how the fundamentals (including the mathematical framework) envelope the material in each course.
2. Perform hardware based work that will involve VLSI architectures. The VLSI digital circuit designs are done using complementary metal oxide semiconductor (CMOS) technology [5].
3. Integrate software simulation with hands-on laboratory work using MATLAB, its associated SIMULINK package, C++ programming and Mentor Graphics all of which we have at Rowan.
4. Expand student teamwork experience by making group laboratory projects an integral part of the course structure.
5. Continue to improve written and oral communication skills of our students.

The proposed educational material development aims to cut across traditional course boundaries and embodies cross-platform, interdisciplinary knowledge necessary for today's students. The main focus is to first write a laboratory manual which describes the interdisciplinary laboratory experiences and includes the relevant theory and conceptual relationships linking COMM, DSP and VLSI. A long term goal is to use the laboratory manual to write a laboratory oriented textbook. Examples of laboratory based textbooks in the DSP area include those involving experiments with hardware boards [6][7]. However, the theoretical background is rather brief in stressing the conceptual issues that tie DSP with COMM and VLSI. There is one textbook that shows concepts linking DSP and COMM [8]. It is a traditional book describing the fundamentals, mathematical background and applications but which has no laboratory component. Here is a description of 3 experiments that will make up a part of the laboratory manual.

Experiment 1 – Pulse Code Modulation (PCM)

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Objective

To gain a thorough understanding of Pulse Code Modulation (PCM) and about its hardware and software implementation.

Theory

Pulse Code Modulation [9][10] is essentially analog-to-digital conversion of a special type in which the information contained in the instantaneous samples of an analog signal is represented by digital words. If each of the digital words used by the system contains n bits, then there are 2^n unique code words that are possible. Each of these code words represents a certain amplitude level and the grouping of all of these code words is called a codebook. Unfortunately, because an analog signal can be any one of an infinite number of amplitudes, it is not possible for a digital word to represent these exact values. Instead, the system chooses the digital word that represents the amplitude closest to the actual sample value used. This is called quantization and it is the fundamental principle underlying PCM.

PCM is very popular in communication circuitry and bandwidth limited devices because of the following:

- The number of bits that need to be transmitted through a medium can be greatly reduced.
- It uses relatively inexpensive circuitry.
- PCM signals that are derived from any type of analog source (audio, video, etc.) can be merged with data signals and be transmitted over a high-speed digital communication system.
- The noise performance of a digital system can be superior to that of an analog system.

Despite all of these advantages PCM has one noticeable disadvantage and that is that the original signal will never be recovered once it is quantized. In most systems, this loss of precision has no effect on the performance of the system.

There are 3 steps that are involved in creating a PCM signal. These include sampling, quantizing and encoding. A diagram of this process follows:

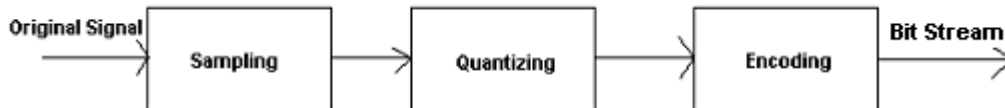


Figure 1 How to Create a PCM Signal

The first step in creating a PCM signal is to sample the original signal. In order to successfully sample a signal, the sampling frequency must be 2 times greater than the maximum frequency present in the original signal. This is also known as the Nyquist sampling theorem [11][12]. Next, the signal must be approximated so it matches up with one of the codebook values. This is done by comparing the sampled value to the values in the codebook and finding the closest entry in the codebook. This is the process that introduces error into the PCM signal and it leads to the final step, which is encoding. The encoding process outputs a digital word (or bit stream) that will be transmitted to a receiver. The other part of PCM occurs at the receiver end of the system. At the receiving end, the digital word or bit stream is decoded to appropriate codebook value. The sequence of codebook values is the quantized discrete signal analog reconstruction (digital to analog or D/A) is performed. An example of this process follows:

4 bit code word	Sample Value
0000	-3.2
0001	-2.75
0010	-2.3
0011	-1.85
0100	-1.4
0101	-0.95
0110	-0.5
0111	-0.05
1000	0.4
1001	0.85
1010	1.3
1011	1.75
1100	2.2
1101	2.65
1110	3.1
1111	3.55

Table 1 Example of a codebook

For the values -1.45 , 0 and 1.60 , give the codebook value and digital word that would result from the following values. Also, determine the quantization error which is the absolute difference between the original value and the quantized value. It is easily noticeable that the closest value to -1.45 (see Table 1) is -1.4 and the bit stream is 0100 . The error is simply $0.05V$. The closest codebook value to 0 is -0.05 . The digital code word that corresponds to this value is 0111 and the error is again 0.05 . The closest code book value to 1.60 is 1.75 and the corresponding digital word is 1011 . The error in choosing this value is now 0.15 .

General Laboratory Description

Now that you have a general understanding of what PCM is, you are now going to simulate a PCM system and build the components that are necessary to run a PCM system. First, you will use C/C++ to simulate a PCM system. This lab does not require building anything and should show you that a software implementation of PCM is relatively simple.

The final 3 parts of this lab are designed to teach you how to build the 3 main parts of a PCM system. First, you will design a block of memory. Memory is used quite extensively throughout the PCM system. First, it is used so that the bits coming out of the A/D converter can be saved so that they can be compared to a codebook value. Second, it is used to store all of the codebook values and their respective digital words. Finally, it is used to remember which values have been eliminated when the comparison stage has begun. Although you are only going to be required to make 1 block of memory, realize that several blocks of memory need to be used to successfully implement a PCM system.

In the next part of this lab, you will need to implement a 4-bit full subtractor. This will be used to measure the distance between the signal value and each codebook value. These results will be stored in a block of memory and be used for the final phase of this PCM system, which is the comparison phase. In the comparison phase, each of these subtracted values has an individual bit of memory allocated for it. All of these bits start with a logic level of "1". After each individual comparison, the value that is further away receives a reset pulse so it is dropped out of the comparison competition. This comparison phase continues until there is only 1 codebook value with a bit of logic '1'. The system then knows that this is the appropriate value that should be sent out and the respective digital word is transmitted.

Part 1 – Software Implementation

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Objective

To achieve a software implementation of a PCM system and observe how the number of bits used affects quantization error and hence, signal integrity.

Instructions

1. A speech signal sampled at 8 kHz will be supplied to you. Use MATLAB to convert this signal to zero mean. Record the true standard deviation of the signal and call this the variable G . Divide the signal by $2.5G$ and use this scaled signal for step 2.
2. Quantize the scaled signal using codebooks of size 4, 8 and 16 (2, 3 and 4 bits) that have been optimized for a Laplacian input. These codebooks are used since it has been shown that speech has a distribution function that is approximately Laplacian [10]. To perform this step, write a C/C++ program. The codebooks will be supplied to you and can be found in [10].
3. Multiply the quantized signals by $2.5G$ using MATLAB and calculate the quantization error (using the norm function) for each of the 3 codebooks. Compare the error for the 3 codebooks and comment on your results. Listen to the 3 quantized signals (using the soundsc function) and comment on how perceptual quality is affected by the number of bits used.

Part 2 – Memory

Objective

To design and implement in VLSI a simple memory circuit to store values in a codebook.

Theory

Memory is one of the most important building blocks of a computer, the other two being Input/Output and the Central Processing Unit. Memory is used to store values and instructions that are being processed, or waiting to be processed. However, the memory within the CPU itself is not like other memory. It is instead a collection of registers that store values and addresses of outside memory.

This section of the lab deals with the memory needed to store the codebook values, input streams, and calculated values. This memory does not need to be anything as extravagant as SDRAM, or even SRAM. These types of memory would require extensive address generation, storage and manipulation. That is beyond the scope of this lab. Instead, concentration will be

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turned to registers to store the various values needed by the PCM system to properly carry out its function.

Registers are comprised of a very basic building block of all digital systems – the D-Flip Flop. A D-Flip Flop allows a value to be stored within a circuit, effectively acting as a “memory” circuit. It is made up of smaller components, with a D-Latch as the Master latch, and an RS- or D-Latch acting as the slave. This configuration allows values to be stored within the circuit until it is “loaded” otherwise by the clock. The truth table for a D-Flip Flop is:

D	CLK	Q_{n+1}	Q_{n+1}
		1	*
0	0	1	0
0	1	0	1
1	0	0	1
1	1	1	0

Table 2 Truth table for a D flip-flop

What this truth table says is that when the clock pulse is high, the NEXT state of Q will be that of the current D input. The output will follow the input as long as it is high. However, when the clock is low, there will be no change in the next state of Q.

The D-Flip Flop is the essential component that comprises registers. One D-Flip Flop represents a 1-bit register. However, all of the values that are to be stored in the registers are required to be 4-bit. Also, since there are at least 16 codebook values, there needs to be a method for selecting the correct register of the 16 used to store the values. The select must not only allow for a read select, but also a write select.

To build a 4-bit register is a simple task. It is simply 4 1-bit registers with all of their clocks tied together into one master clock. That way, when that register set is selected, all of the D-Flip Flops will either write or read all at the same time.

Now that all of the clocks are tied together, there comes the task of selecting between the registers. For example, if there were 16 registers, you would need $n = \log_2 16$ select bits, which makes $n = 4$. Also, because of the nature of the register, loading and/or reading occur only when the clock is high. Therefore, a circuit is required that will allow a load of that particular register set when the clock is high. An example circuit is shown below.

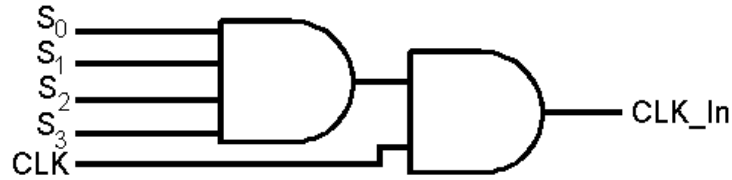


Figure 2 Circuit for CLK_In

When all 4 select bits are high, AND the Master Clock is also high, the CLK_In output will be high. The CLK_In output acts as the clock for the registers. And thus, only when that particular register is selected, AND the clock is high, will the register load the D values.

Now, the problem arises of reading from the correct register. Since all 16 outputs will always be outputting a value, the outputs that are not desired must be “blocked” from reaching the read lines. Again, the desired register for reading has 4 bits, the same 4 as that of the selection circuit. However, the clock will not be used in this instance. Instead, a “Read” bit is added, which performs the same function. Now, the problem arises of allowing only the selected register to output, but keep ALL of the others blocked. An easy way to accomplish this is to use a “Tri-State” buffer. The truth table for a Tri-State buffer is:

X	Hi-Z	Out
x	0	--
1	1	1
0	1	0

Table 3 Truth table for Tri-State buffer

Basically, when the Hi-Z (High Impedance) state is 0, no matter what the X input is, there will be no output. The input is effectively blocked from passing. When the High Z state goes high, then the input is allowed to pass through, thus allowing the selection of the desired register for reading. An example of such a circuit is shown below:

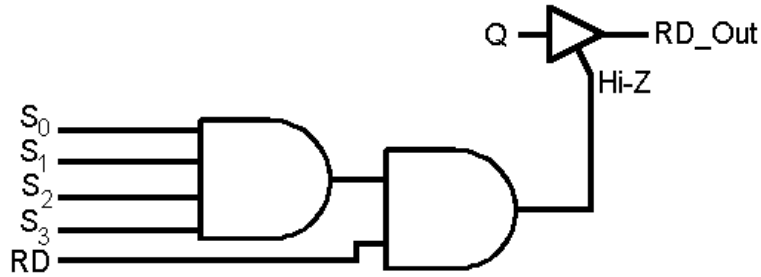


Figure 3 Circuit for RD_out

Now that we have a circuit for selecting the output of the register we require, as well as a circuit to load it, we can put all of these sections together to effectively build the 4-bit register with load and read selects. The following circuit is the finished product:

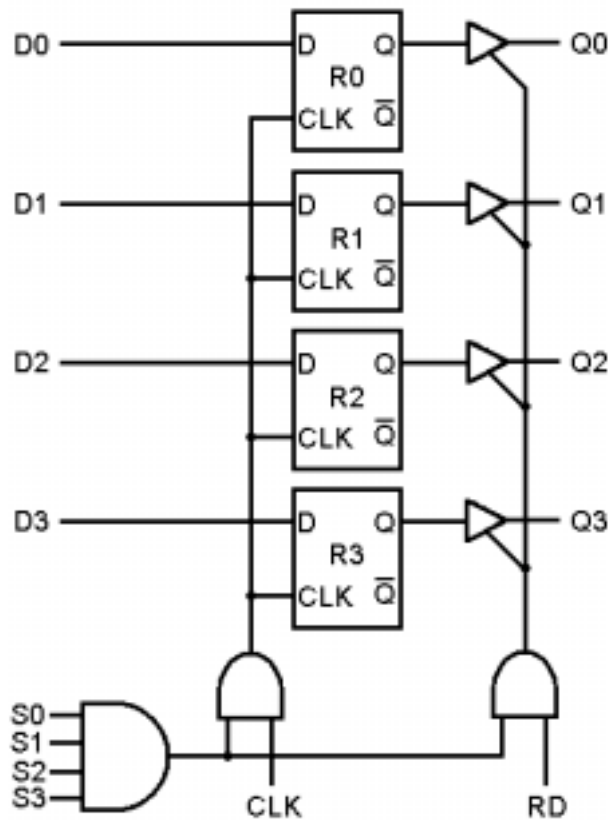


Figure 4 Circuit for 4 bit register

Procedure

1. Design the basic D-Flip Flop circuit in Design Architect, using a D-Latch and another latch circuit. Be sure that the output follows the input only when the Clock pulse is high,

and retains its value otherwise. Keep in mind the use of real estate on the DSP chip (e.g. minimize the number of FETs).

2. Design both select circuits in Design Architect. Ensure that the tri-state buffer works correctly for all inputs. Again, keep in mind real estate.
3. Completely simulate all circuit designs in Quicksim and Accusim. Show that the circuits work completely.
4. Construct each circuit in IC Station. Try to keep the size of each block to a minimum. Perform Back Annotation and Parasitic Extraction on all segments.
5. Assemble all components together into the final design of a 4-bit Register with Read and Write select. Re-simulate in Accusim and show that the circuit works.

Questions

1. What type of latch did you use in your final D-F/F design? Why did you choose that particular latch? How efficient is your choice in terms of FETs used?
2. When simulating your final design, did you encounter any difficulties in reading or writing? What conflicts, if any, did you need to consider when simulating?
3. Why did we simply label each select bit S_0 , S_1 , S_2 , and S_3 instead of specifying the select bits to act for a specific register? What advantage does this allow?
4. Think about the overall picture for the PCM DSP chip. How many FETs do you foresee it requiring in order to store ALL values that may need to be stored? What does this value tell you about how and why memory is delegated within a computer (e.g. Why main memory is external of the CPU)?

Part 3 – Four Bit Binary Full Subtractor

Objective

To create a circuit that performs binary subtraction of 2 4-bit binary numbers and outputs the absolute value of the resulting 4-bit number.

Theory

In order to find the difference between two 4-bit binary numbers, it is necessary to use a 4-Bit Binary Full Subtractor. This can be done quite easily using 4, 1-Bit Binary Full Adders. A Full Adder works exactly like binary addition. Just like in binary addition, if $1 + 1$ is performed, the result is 0 with a carry bit of 1. This bit is carried over to the next place, and is then counted

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in the next bit addition. The initial carry bit is often called C_n , while the carry bit resulting from an addition is called C_{n+1} . The truth table for a Full Adder is:

A_n	B_n	C_n	S_n	C_{n+1}
0	0	0	0	0
0	0	1	1	0
0	1	0	1	0
0	1	1	0	1
1	0	0	1	0
1	0	1	0	1
1	1	0	0	1
1	1	1	1	1

Table 3 Truth table for Full Adder

A schematic for a 1-Bit Binary Full Adder is:

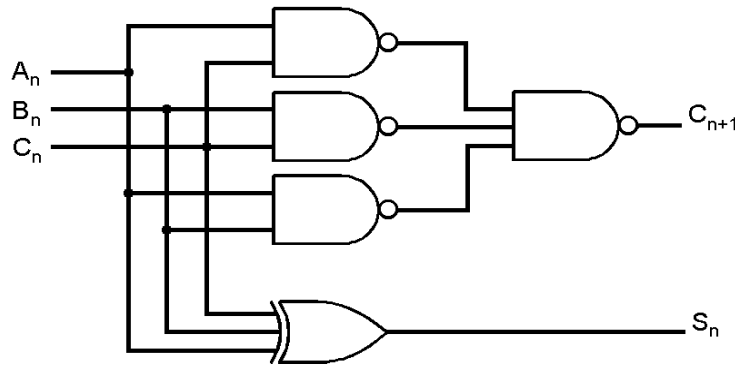


Figure 5 Circuit for 1 bit binary full adder

To perform addition on a 4-bit binary number, it is necessary to have 4 1-bit Full Adders. The C_{n+1} bit from the previous adder is fed into the C_n input of the next adder. This supplies the proper carry from the previous bit addition. The first carry bit is tied to 0, since in performing addition there is no previous carry.

Performing subtraction between two binary numbers can be accomplished through the use of Full Adders. Since binary numbers can only be added together, it is necessary to invert the incoming B_n bits. By inverting these bits, 1's Complement addition occurs. However, in order for the 1's Complement addition to work correctly, it is necessary to let $C_n = 1$. For example,

$$\begin{array}{r}
 12 \Rightarrow 1100 \Rightarrow 1100 \\
 - 4 \quad - 0100 \quad + 1011_1 \\
 \hline
 8 \qquad \qquad \qquad 11000
 \end{array}$$

If the Most Significant Bit (MSB), in this case C_{n+1} of the fourth Full Adder, is dropped, the resulting binary number 1000 is the correct answer, 8. The schematic for a 4-bit binary Full Subtractor is:

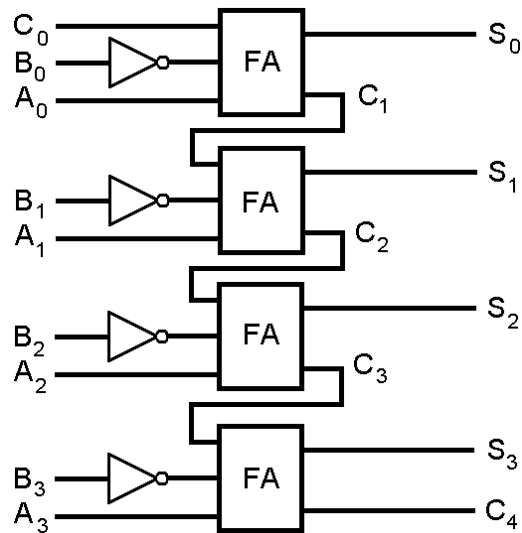


Figure 6 Circuit for 4 bit binary full subtractor

This binary subtraction works only in cases where $A_n \geq B_n$. But what if B_n is greater than A_n ?

$$\begin{array}{r}
 3 \Rightarrow 0011 \Rightarrow 0011 \\
 - 8 \quad - 1000 \quad + 0111_1 \\
 \hline
 - 5 \qquad \qquad \qquad 01011
 \end{array}$$

As can be observed, when the 1's Complement addition occurs, the resulting answer is 1011, or 11. This is most definitely not $|-5|$ (since we are interested in the difference of two binary numbers, the sign is unimportant – hence, the absolute value of the resulting number). This problem can be corrected by performing subtraction using 2's Complement addition. If the 2's Complement of the subtrahend is added to the minuend, then the 2's Complement of the resulting binary number will yield the correct answer.

Subtraction using 2's Complement addition can be implemented using a 4-bit Full Adder with the inputs of B_n inverted and the first carry $C_n = 1$. The output of the Full Adder is then sent through a "correctional" circuit, or a circuit that outputs the correct binary number if $A_n < B_n$ and does nothing if $A_n \geq B_n$. This circuit can be easily implemented using another 4-bit Full Adder.

Procedure

1. Create a 1-bit binary Full Adder in Design Architect. Be sure to generate a symbol, as you will use this block later. Simulate in Quicksim II and Accusim II, and ensure that the outputs match those of the truth table for the Full Adder above. Wire this circuit up in IC Station. Back Annotate and perform parasitic extraction.
2. Using the 1-bit binary Full Adder, create a 4-bit binary Full Subtractor using 1's Complement. Check to make sure that the Subtractor works for $A_n \geq B_n$.
3. Design the "Correctional" circuit to allow for 4-bit subtraction when $A_n < B_n$. This circuit should allow for the 4-bit output to be positive (just drop the carry bit – it is useless in this case). HINT: This circuit can be easily made using a 4-bit Full Adder!
4. Test the completed 4-bit binary Full Subtractor. Simulate the following 4-bit numbers, using Quicksim II to test the outputs:

1001 – 0110
1100 – 1001
0011 – 0101
1111 – 1111
0000 – 1111

5. Wire up the 4-bit binary Full Subtractor in IC Station.

Just a reminder. When designing logic circuits for implementation in CMOS, keep in mind the problem of "real estate" on the actual IC. Designs should use the fewest number of FETs as possible to conserve board space and allow for more circuits to be implemented.

Questions

1. Is your design an efficient design?
2. What types of problems did you encounter when designing the correction circuit?

Part 4 – Comparator Circuit

Objective

To build a comparator circuit with NAND gates and a comparator with Reset Pulses.

Theory

Now that we have the results from our subtractor, we need to determine which of the code book values is the closest to the inputted signal. As a result, it is necessary to compare the outputted differences to one another. This is going to be accomplished by comparing 2 of the differences at once and keeping the closer value active while turning off the other code book value. However, the goal of this lab is to produce a simple 2 input comparator.

We are going to create this comparator by using 4 NAND gates. We do this by feeding the input of both code book values, with most significant bit first, into a NAND gate. This output is fed into 2 NAND gates with the other input to each of the NAND gate is the original signal of each code book value respectively. Next, we feed this output into a final NAND gate with the output of the first NAND gate. The output of this final gate will be logic '0' if the inputs have the same logic value and an output of logic '1' if the input have different logic values. A graph of this setup follows:

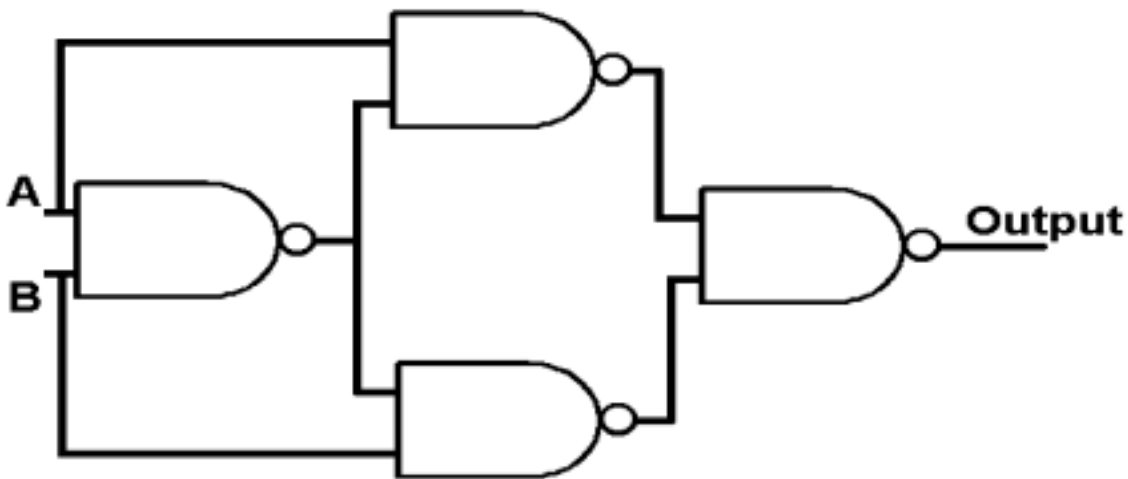


Figure 6 XOR Circuit for with NAND gates

Although this circuit can determine if two bits are equivalent, it can not determine which bit code book value is closer to the inputted signal. As a result, it is necessary to determine a scheme that will choose the closest code book value. We do this by using a “tournament” format. In the first round of this “tournament” we compare the differences of the first 2 code book values and pass the smaller value to the next round. We do this for each subsequent pair of differences until all

of the differences have been compared. This is done for all of the subsequent rounds until the closest value is determined. A diagram of this application follows:

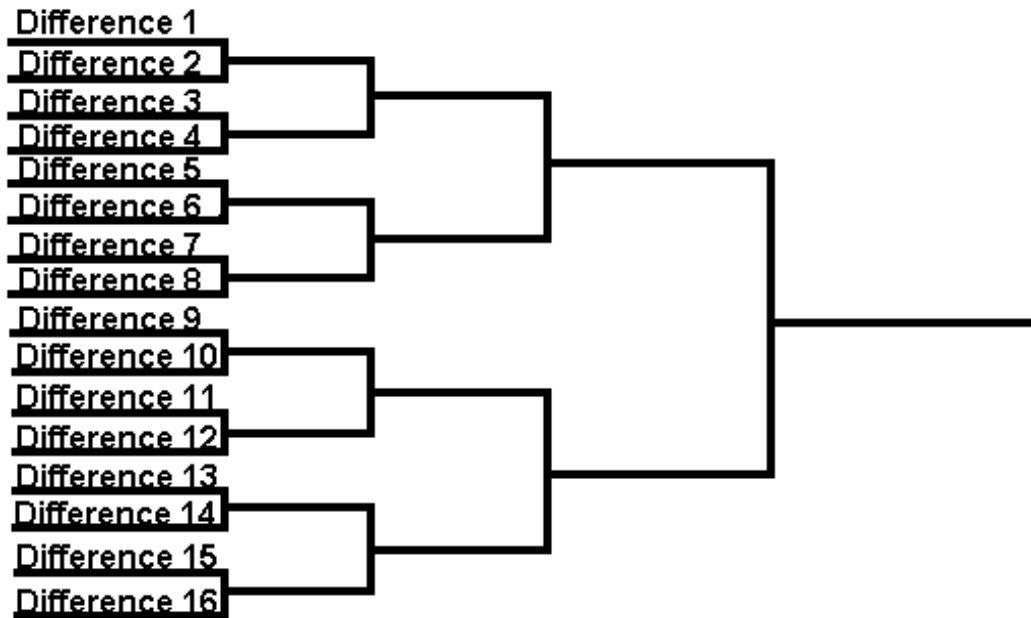


Figure 7 Configuring of differences - Lowest Values Advance to the next round

Now that the basic algorithm of choosing the closest code book value has been designed, the next question is how to turn off the losers of each round and how do we keep track of the winners. In order to accomplish this, the memory circuits designed previously will do this.

Each of the inputted code book values will be connected to one j/k flip-flop. As soon as a new signal is brought in from the A/D converter, all of these flip-flops will become active and given a logic value of 1. When one of the differences lose a round, a pulse will be sent to the reset pin of the appropriate flip-flop, which will turn that code book value off. The end result of this procedure will be that only one flip-flop will be left on at the end of this tournament filtration and that will be the code book value that will be sent out by the PCM system.

The final problem of this setup is to determine which of the values should be turned off in each round. Currently, we have an XOR gate that will determine if 2 bits are similar or if 2 bits are different, however, it does not tell us which difference has the 1 or the 0. The code book difference with the 1 is further away then the code book difference with the 0 and as a result needs to be turned off. So when the compared bits are different, an output of 1 must be present

on the output of the input bits with the logic 1 signal and a 0 for all other input values. A truth table follows:

A	B	Output A	Output B
0	0	0	0
0	1	0	1
1	0	1	0
1	1	0	0

Table 4 Truth table for Reset Pulse

Knowing the truth table for our function allows us to design a circuit for the logic operation. The following circuit performs this operations:

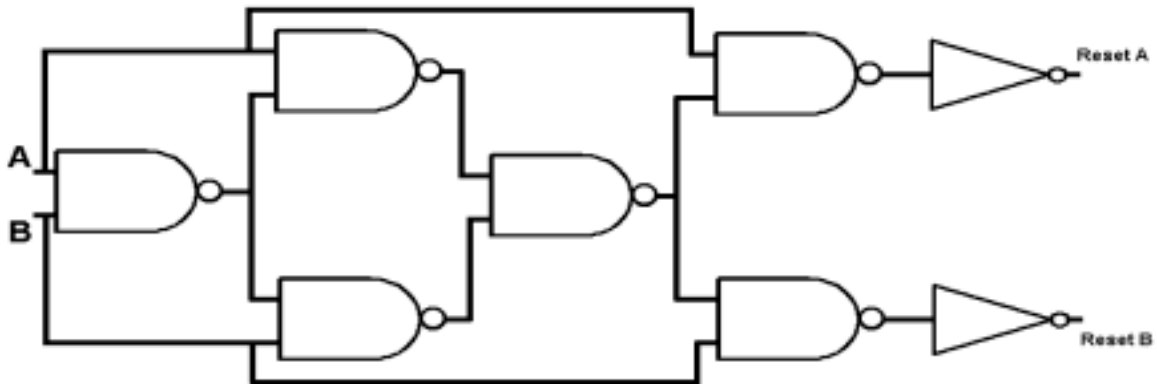


Figure 8 Comparator With Reset Keys

In this lab, we are only going to build one of these 2 input gates, however in an actual PCM circuit, there would be several of these comparators as well as memory circuits.

Procedure

1. Build a comparator or an XOR gate using NAND gates in Mentor Graphics' Design Architect. Print out your design.
2. Create the truth table for this gate.

3. Simulate this circuit in quicksim and accusim and confirm the validity of your circuit. Print out all of your simulation results.

*******DO NOT BEGIN LAYOUT AT THIS TIME!!!**

4. Reopen Design Architect and add the appropriate NAND gates and inverters to create the comparator with reset keys. Print out your design.
5. Simulate this circuit in quicksim and accusim and confirm your results with the above truth table. Print out your simulation results.
6. Open ic station and layout your circuit. Make sure it passes DRC check and LVS. Print out your LVS report and your layout design.
7. Perform a parasitic extraction of your circuit.

Questions and Conclusions

1. Verify that the given XOR circuit with reset keys gives the appropriate truth table. **Hint:** Evaluate the output of every NAND gate and INV for all possible inputs.
2. What are the tradeoffs of block designing vs. MOSFET design?
3. What would happen if the code book differences in a given round were exactly the same?
4. What could you do to rectify this situation?

Experiment 2 – Central Limit Theorem

Objective

To understand and demonstrate the notions of probability, random variable, the probability density function and the Central Limit Theorem. The materials for this experiment include a Plinko machine and 100 wooden balls.

Theory

In communication systems today, signals are almost always corrupted with noise after being sent across a channel. Given a received signal, it is important to determine which one of M signals was actually transmitted. This is done by calculating the probability of error and choosing the best candidate of M possible signals as the one that minimizes the probability of error. In this laboratory, we introduce the concept of probability, random variable, the probability density function and the Central Limit Theorem [9][16].

In order to determine the probability of a certain event happening, engineers set up random experiments. For an experiment to be random, it must satisfy three requirements: (1) It must be repeatable under identical conditions, (2) On any trial conducted, the outcome must be unpredictable and (3) For a large number of trials, the outcomes exhibit statistical regularity, which means a definite pattern of outcomes emerges. After all of these experiments are run, the relative frequency of an outcome of a random event is calculated. This is done by calculating the number of times an event A occurs divided by the number of experiments conducted. The formula for the relative frequency of an event A is [9]

$$F_A = \frac{n_A}{n}$$

where n_A is the number of times event A occurs in n trials. A property closely related to the relative frequency of a random event is the probability of a random event. The probability of the random event A is the value of F_A as the number of trials, n , approaches infinity. This is denoted as $P(A)$. or each type of experiment run, there will be m outcomes. For example if we drop a square box, it will have to land on one of its six sides. Events or outcomes A, B, C, D, E and F represent the landing on the sides of the box. The set of all possible outcomes is called the sample space (denoted by S) and $P(S) = 1$. Instead of saving all of these values as outcomes, a random variable is used. A random variable, $X()$, is a **function** that takes each outcome of S and assigns a real number value to that outcome [17]. A sample table displays the events (or outcomes), probabilities and random variables associated with the dropping of the box. Note that the random variable takes on discrete values.

Event	Random Variable X()	Probability of Event P(x)
A	-2.0	0.09
B	-1.5	0.25
C	-3.0	0.10
D	0.0	0.21
E	3.0	0.30
F	5.0	0.05
		Total = 1.00

Table 5 – Events, Random Variables and Probabilities

Graphing the above result ($P(x)$ versus $X()$) gives the discrete probability density function (PDF) as shown below.

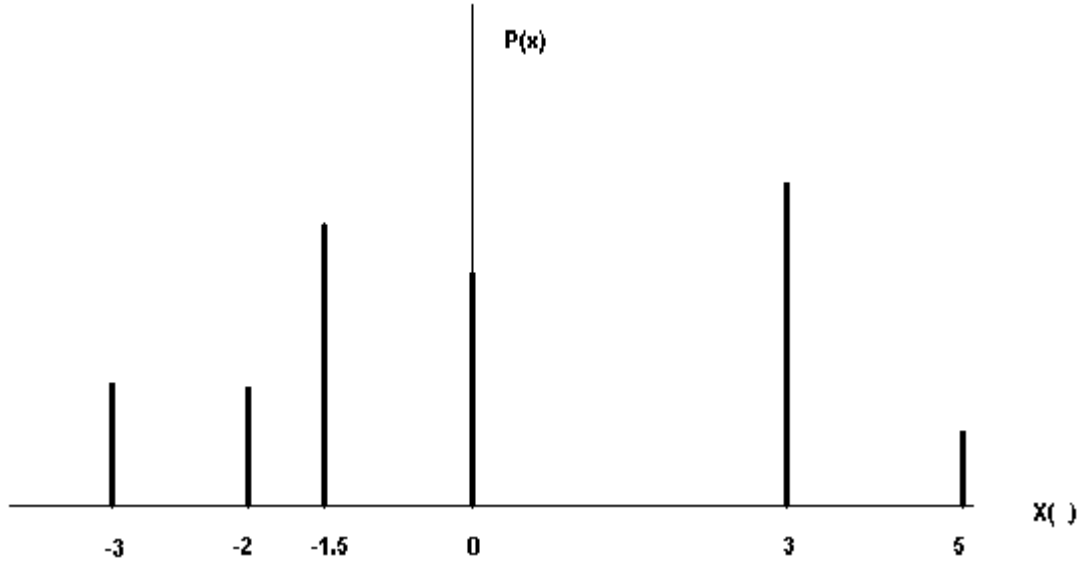


Figure 9 – Discrete Probability Density Function

The Cumulative Distribution Function (CDF) $F(x)$ simply adds up the probabilities of the above graph as x increases. The CDF is a monotonically increasing function whose value at $x = a$ is found by $F(a) = P(x \leq a)$

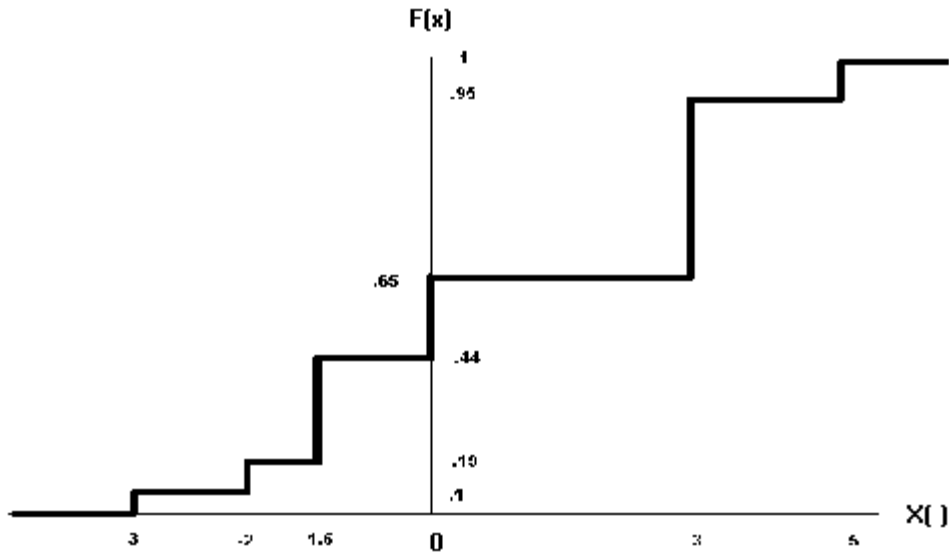


Figure 10 – Cumulative Distribution Function

A continuous Probability Density Function (PDF) $P(x)$ shows the case when $X()$ is a continuous random variable. The probability of the random variable falling into a certain range of values (between a and b) is given by the following:

$$P(a \leq x \leq b) = F(b) - F(a) = \int_a^b p(x) dx$$

A commonly used PDF model is the Gaussian or Normal PDF. The equation is given by

$$P(x) = \frac{1}{\sqrt{2\pi} \sigma} e^{-(x-m)^2 / 2\sigma^2}$$

where m is the mean and σ^2 is the variance. The mean and variance are calculated as

$$m = \int x p(x) dx$$

$$\sigma^2 = \int x^2 p(x) dx - m^2$$

If X_i ($i = 1$ to n) are random variables with the same PDF, they are said to be identically distributed. If in addition, the X_i are independent, they are said to be independent and identically distributed (iid) [9]. If X_i ($i = 1$ to n) are iid random variables with common mean m and common variance σ^2 , the sum of these random variables has a PDF approximated by a Gaussian PDF with mean nm and variance $n\sigma^2$. This is known as the Central Limit Theorem which can be explained intuitively as follows. Suppose an experiment is carried out in which a coin is tossed 50 times and the random variable is the number of heads [16]. Four hundred repetitions of this experiment generate 400 iid random variables. A graph with the horizontal axis being the number of heads (value of the random variable) and with the vertical axis being the number of occurrences (over the 400 repetitions) is a histogram. Normalizing this histogram to unit area leads to a plot that resembles a Gaussian PDF. The resemblance to a Gaussian PDF becomes more apparent as the number of repetitions of the experiment increases. This is one of the applications of the Central Limit Theorem. The Central Limit Theorem allows us to model many practical phenomena using the Normal Distribution.

Procedure

1. Place the Plinko Machine in the upright position with the funnel on top.
2. Place 10 wooden balls into the funnel. If the balls get stuck shake the machine gently until they begin to move again.
3. Observe what compartments the balls come to rest in and drop another 10 balls. Record the number of balls in each compartment.
4. Continue dropping in 10 balls at a time and record your measurements when you have dropped 40, 60, 80 and 100 balls.
5. Place cup below bottom hole. Pull out the bottom drawer to empty the machine. Repeat the experiment again.

Conclusions and Questions

1. How did your outputs compare to the Central Limit Theorem at each step in the experiment, e.g. at 20, 40, 60, 80, and 100 balls?
2. What could be some reasons why a Normal Distribution would not result in this experiment?
3. Did all of the 17 containers at the bottom of the Plinko machine contain balls? Which ones didn't and why didn't they have any balls?
4. Were your results from your first trial similar to your second trial? Is this expected and why?
5. Record any other conclusions you may have reached.

Experiment 3 – Design and Layout of a Hamming Error Correcting Code

Objective

1. To create a circuit specifically designed to encode and decode a 4-bit data word using the Hamming error correcting code algorithm.
2. To layout the resulting circuit in Mentor Graphics.

Theory

The Hamming error correcting code algorithm [9][14] must be understood before these objectives can be achieved. The fundamental principle embraced by Hamming codes is parity. The Hamming weight of a word is defined as the number of binary '1' bits in the data stream. For example, the code word 11100111 has a Hamming weight of 6 because there are 6 '1' bits

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present. The Hamming distance between two code words, d , is defined as the number of bits that differ between the two codewords. For example, given the code words 11001 and 01010, this has a Hamming distance of 3 because the first, fourth and fifth bits differ from the first code word to the second.

The purpose of the Hamming codes is to check for errors. It is also possible for a Hamming decoder to not only detect but also correct these errors if certain conditions are present. For s errors to be detected and t errors to be corrected, it is required that $d \geq s + t + 1$. Also, if there are no greater than t errors, then these errors can be both detected and corrected if $d \geq 2t + 1$. A general code word can be written in the general form:

$$i_1 i_2 i_3 \dots i_k p_1 p_2 p_3 \dots p_r$$

in which i are the information bits, p are the parity bits, k are the number of information bits and r are the number of parity bits. We now define the general form of a Hamming Encoder. In a (n, k) block code, $n = k + r$.

The arrangement of parity bits in a code word varies slightly. A block code, which has all of the information bits at the beginning of the word and all of the parity bits at the end, is called a systematic block code. A block code that has parity bits interleaved between the information bits is commonly called an equivalent code.

The smallest Hamming encoder possible is a block code having a Hamming Distance of 3. This occurs because $d \geq 2t + 1$. If $t = 1$, then a single error can be detected and corrected. It also worthwhile to point out that only certain Hamming codes are allowed. These allowable codes are given by the following formula:

$$(n, k) = (2^m - 1, 2^m - 1 - m)$$

For a (7,4) Hamming code, the encoder reads in four information bits (I1, I2, I3 and I4) and outputs three parity bits (P1, P2 and P3) as given by

$$P3 = I4 \oplus I2 \oplus I1$$

$$P2 = I4 \oplus I3 \oplus I1$$

$$P1 = I4 \oplus I3 \oplus I2$$

The three parity bits are packaged in with the four information bits and then sent over a channel after which the bits are presented to the the decoder. We now rewrite these bits keeping the information bits first. I4, I3, I2, and I1 will be recognized as R7, R6, R5 and R4 respectively and P3, P2 and P1 will be recognized as R3, R2 and R1, respectively in the decoder example.

The decoder's job is to receive all seven bits (four information and three parity) and send out a message that tells you if there is any error or not. If all elements are zero, the codeword was received correctly. If s contains non-zero elements, the bit in error can be determined by analyzing which parity checks have failed, as long as the error involves only a single bit for this specific example of a (7,4) Hamming Encoder. The equations are:

$$S3=R7 \oplus R5 \oplus R4 \oplus R3$$

$$S2=R7 \oplus R6 \oplus R4 \oplus R2$$

$$S1=R7 \oplus R6 \oplus R5 \oplus R1$$

Assignment

Using Mentor Graphics, design the (7,4) Hamming encoder/decoder as two separate circuits. Use the least amount of board space and fewest FETS possible. Note the dimensions of your circuits. In your simulation you should show, with hand calculations and your circuit output, that your Encoder circuit generates the correct code and that the Decoder can detect an error which you introduce intentionally. The design should be constructed economically. You must perform parasitic extraction, backannotation and re-simulation with all parasitics.

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Steven H. Chin is the Associate Dean of Engineering at Rowan University. He has both research, laboratory development and teaching experience in Communications, Information Theory, Digital Signal Processing, Image Processing and Neural Networks. At Rowan, he is the faculty coordinator for ABET accreditation activities. He joined Rowan after spending 8 years at the Catholic University of America.