ELECTRICAL ENGINEERING

QUALS QUESTIONS

2010

HTTP://EE.STANFORD.EDU/PHD/QUALS

2010 PhD Quals Questions
J. S. Harris

1. I have several LEDs here of different colors. Can you first briefly tell me how an LED works and what is different about the devices that produce different colors of light?

2. Can you draw the Current-Voltage and Light-Current (L-I) curves for me for say two different color LEDs and identify which is which?

3. Can you sketch the characteristic spectral distribution for one of the LEDs? Why can you have photons with $\lambda < E_g$, but not greater than E_g?

4. There is a lot of effort today to produce solid state lighting. How can I produce a White LED?

5. I now have a red laser. Can it be made of the same materials as a red LED? What is different about the laser compared to the LED?

6. Can you draw the I-V and L-I curves and spectral distribution for the laser? How do they differ from the LED and why?

7. If I now change the temperature of the LED and laser, how do the spectral characteristics change and why?

8. I have a polarizing filter and when I put it in front of the LED, it has virtually no effect, but when I put it in front of the laser, and rotate it, it changes from virtually zero transmission to unchanged transmission. What does this tell you about the fundamental recombination processes occurring in the laser vs. the LED?

9. If stimulated emission and absorption are reciprocal processes, can I use a LED or laser as a solar cell? What might I do differently to optimize the solar cell vs. a laser or LED?

EE QUALIFYING EXAM
JANUARY 2010

This is a problem about discrete signals (vectors) of length N.

Let I be a subset of $\{0, 1, \ldots, N-1\}$ and let I' be the complementary subset. (For example, I could be the set of even numbers in $\{0, 1, \ldots, N-1\}$ and I' would then be the set of odd numbers.)

Let $\mathbb{B}^I$ be the set of signals whose spectrum is supported on I, i.e.,

$$\underline{f} \in \mathbb{B}^I \iff \underline{\mathcal{F}}\, \underline{f}[m] = 0 \quad \text{if } m \in I'.$$

Here $\underline{\mathcal{F}}$ is the discrete Fourier transform.

- What is the set of signals that are *orthogonal* to $\mathbb{B}^I$, i.e., what is the orthogonal complement to $\mathbb{B}^I$?

Let $\underline{h}$ be the signal defined by

$$\underline{\mathcal{F}}\, \underline{h}[m] = \begin{cases} 1, & m \in I \\ 0, & m \in I' \end{cases}$$

- Show that the orthogonal projection onto $\mathbb{B}^I$ is given by

$$K\underline{f} = \underline{h} * \underline{f}.$$

- What is the orthogonal projection onto the orthogonal complement of $\mathbb{B}^I$?

Solutions

For the first question, two signals $\underline{f}$ and $\underline{g}$ are orthogonal if their inner product, $\underline{f} \cdot \underline{g}$ is 0. By Parseval's theorem

$$\underline{f} \cdot \underline{g} = \frac{1}{N}(\underline{\mathcal{F}}\, \underline{f} \cdot \underline{\mathcal{F}}\, \underline{g}).$$

If $\underline{f} \in \mathbb{B}^I$ then

$$\begin{aligned} \underline{\mathcal{F}}\, \underline{f} \cdot \underline{\mathcal{F}}\, \underline{g} &= \sum_{n=0}^{N-1} \underline{\mathcal{F}}\, \underline{f}[n] \overline{\underline{\mathcal{F}}\, \underline{g}[n]} \\ &= \sum_{n \in I} \underline{\mathcal{F}}\, \underline{f}[n] \overline{\underline{\mathcal{F}}\, \underline{g}[n]} \end{aligned}$$

since $\underline{\mathcal{F}}\,\underline{\mathrm{f}}[n] = 0$ if $n \in I'$. This will be 0 *for all* $\underline{\mathrm{f}} \in \mathbb{B}^I$ if and only if $\underline{\mathcal{F}}\underline{\mathrm{g}}[n] = 0$ for all $n \in I$. This says that $\underline{\mathrm{g}}$ must be in $\mathbb{B}^{I'}$. Sybolically,

$$(\mathbb{B}^I)^\perp = \mathbb{B}^{I'}.$$

For the second question, to show that $K\underline{\mathrm{f}} = \underline{\mathrm{h}} * \underline{\mathrm{f}}$ defines the orthogonal projection onto $\mathbb{B}^I$ we have to do several things. First, if $\underline{\mathrm{f}}$ is any signal we have to show that $\underline{\mathrm{h}} * \underline{\mathrm{f}} \in \mathbb{B}^I$. For this, for any m the convolution theorem and the definition of $\underline{\mathrm{h}}$ gives

$$\begin{aligned}\underline{\mathcal{F}}(\underline{\mathrm{h}} * \underline{\mathrm{f}})[m] &= (\underline{\mathcal{F}}\,\underline{\mathrm{h}}[m])(\underline{\mathcal{F}}\,\underline{\mathrm{f}}[m]) \\ &= \begin{cases} \underline{\mathcal{F}}\,\underline{\mathrm{f}}[m], & m \in I \\ 0, & m \in I' \end{cases}\end{aligned}$$

Thus $\underline{\mathrm{f}} * \underline{\mathrm{h}}$ is supported on I, i.e., $\underline{\mathrm{h}} * \underline{\mathrm{f}} \in \mathbb{B}^I$.

Second, if $\underline{\mathrm{f}}$ is already in $\mathbb{B}^I$ we should have $\underline{\mathrm{h}} * \underline{\mathrm{f}} = \underline{\mathrm{f}}$. But if $\underline{\mathrm{f}} \in \mathbb{B}^I$ then already $\underline{\mathcal{F}}\,\underline{\mathrm{f}}[m] = 0$ for $m \in I$ and so by the definition of $\underline{\mathrm{h}}$,

$$(\underline{\mathcal{F}}\,\underline{\mathrm{h}})(\underline{\mathcal{F}}\,\underline{\mathrm{f}}) = \underline{\mathrm{f}}.$$

Taking the inverse DFT gives

$$\underline{\mathrm{h}} * \underline{\mathrm{f}} = \underline{\mathrm{f}}.$$

As an aside, another way to do this part of the problem is to observe that

$$(\underline{\mathcal{F}}\,\underline{\mathrm{h}})(\underline{\mathcal{F}}\,\underline{\mathrm{h}}) = \underline{\mathcal{F}}\,\underline{\mathrm{h}},$$

hence

$$\underline{\mathrm{h}} * \underline{\mathrm{h}} = \underline{\mathrm{h}}.$$

Thus for any signal $\underline{\mathrm{f}}$,

$$K^2\underline{\mathrm{f}} = K(K(\underline{\mathrm{f}})) = \underline{\mathrm{h}} * (\underline{\mathrm{h}} * \underline{\mathrm{f}}) = (\underline{\mathrm{h}} * \underline{\mathrm{h}}) * \underline{\mathrm{f}} = \underline{\mathrm{h}} * \underline{\mathrm{f}} = K\underline{\mathrm{f}},$$

that is

$$K^2 = K,$$

which is the definition of a projection.

Why is this an orthogonal projection? If $\underline{\mathrm{f}}$ is in $\mathcal{B}^{I'}$, the orthogonal complement of $\mathbb{B}^I$, then $\underline{\mathcal{F}}\,\underline{\mathrm{f}}[m] = 0$ for $m \in I$, hence, by definition of $\underline{\mathrm{h}}$,

$$(\underline{\mathcal{F}}\,\underline{\mathrm{h}})(\underline{\mathcal{F}}\,\underline{\mathrm{f}}) = 0,$$

whence

$$K\underline{\mathrm{f}} = \underline{\mathrm{h}} * \underline{\mathrm{f}} = 0.$$

Finally, for the last question, the orthogonal projection onto $\mathbb{B}^{I'}$ is given by

$$K' = I - K.$$

With

$$(I-K)\underline{\mathrm{f}} = \underline{\mathrm{f}} - K\underline{\mathrm{f}} = \underline{\mathrm{f}} - \underline{\mathrm{h}} * \underline{\mathrm{f}}$$

we can also write

$$K'\underline{\mathrm{f}} = \underline{\mathrm{f}} - \underline{\mathrm{h}} * \underline{\mathrm{f}}.$$

or as a convloution

$$K\underline{\mathrm{f}} = \underline{\mathrm{h}}' * \underline{\mathrm{f}},$$

where $\underline{\mathrm{h}}'$ is given by

$$\underline{\mathcal{F}\mathrm{h}}'[m] = \begin{cases} 1, & m \in I' \\ 0, & m \in I \end{cases}$$

Problem 1. Suppose that X is a geometric random variable, with probability of success p. Let t, s be positive integers with $t > s$. Explain why $P(X > t|X > s) = P(X > t - s)$.

Problem 2. For what values of N does the following statement hold?

"If $X_1, \ldots, X_N$ are jointly Gaussian random variables such that $\mathrm{Var}(\sum_i X_i) = \sum_i \mathrm{Var}(X_i)$, then $X_1, \ldots, X_N$ are independent."

Problem 3. Suppose that $X_1, X_2, \ldots$ are a sequence of random variables on the nonnegative integers, such that $E[X_n] \to \infty$ as $n \to \infty$. Does it follow that $P(X_n = 0) \to 0$ as $n \to \infty$?

Now suppose that in addition, $\mathrm{Var}(X_n) = E[X_n]$. Does it follow that $P(X_n = 0) \to 0$ as $n \to \infty$?

S. Mitra

2010 EE Qualifying Exam Questions

1. What is the difference between a latch and a flip-flop?

2. Helped student draw a gate-level MUX and a simple D-latch out of that MUX.
OUT = OR(A, B), A = AND (D, CLK), B = AND (OUT, E), E = NOT(CLK)
Does the above circuit work correctly? Why not?

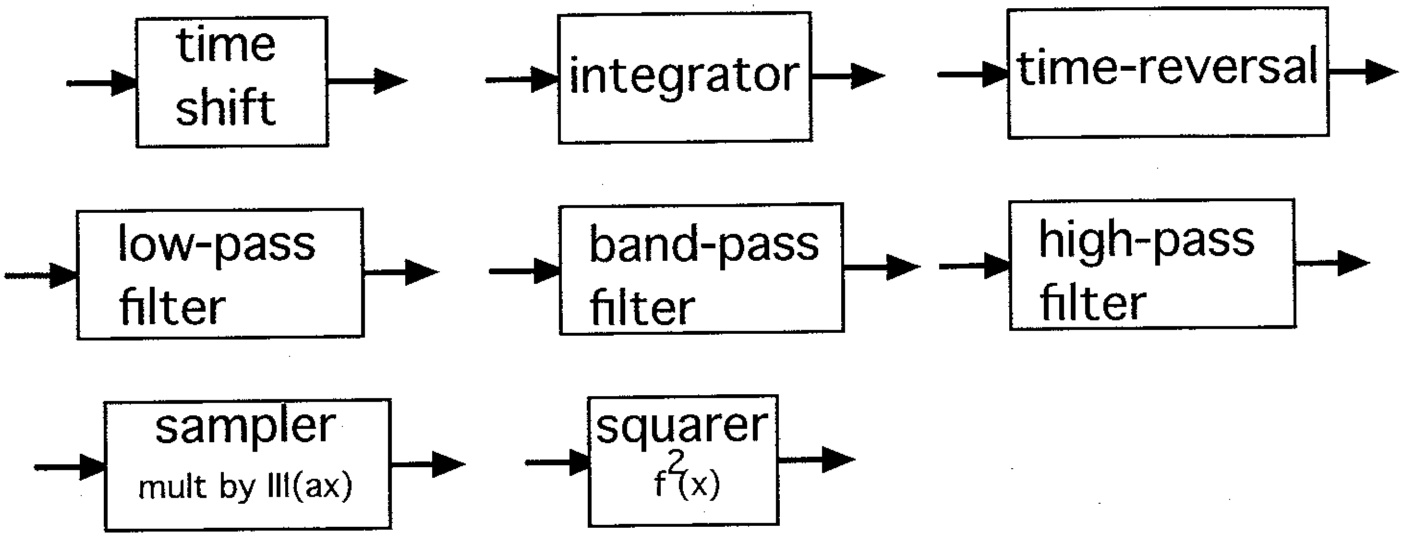
3. How will you fix it?

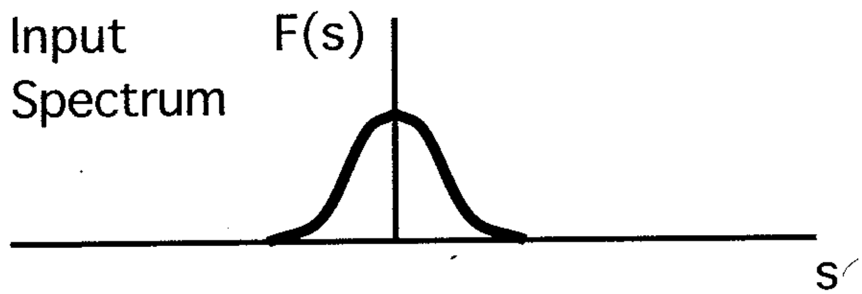
4. Suppose that you are given a flip-flop-based FSM. I claim that you can automatically convert the circuit into one where there are no flip-flops / clk or any sequential element (i.e., replace each flip-flop by a wire) without changing the combinational logic function. How can you do that? Assume that for each gate there is a single delay value associated with that gate.

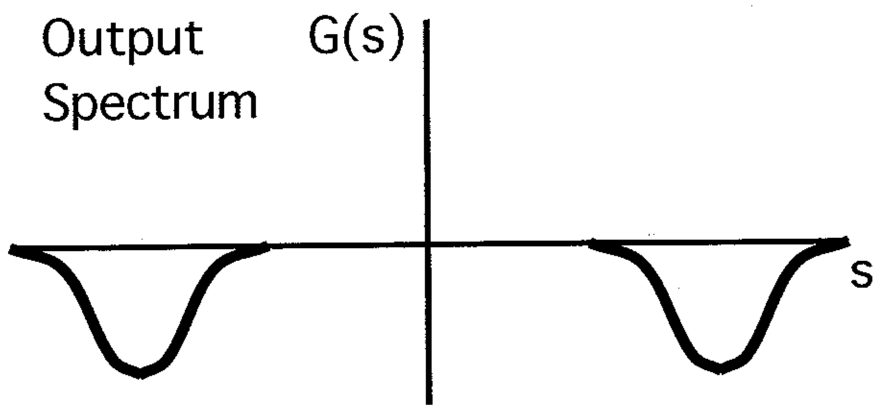
5. You can have trillions of paths if you plan on path delay balancing. What's a scalable way to do this?

Nishimura 2010

① Selecting the appropriate building block(s) from those below, design a system to achieve the output spectrum.







(2)

Let f(x) represent an unknown distribution of mass for an object along x. You are given its Fourier transform F(s).

Is it possible to determine the object's center of mass from F(s) without taking the inverse Fourier transform? Explain.

(3)

Dr. T says the center of mass can be determined from just a single value of $F(s)$, call it $F(s_o)$, if $f(x)$ is symmetric about its center of mass.

Do you agree with Dr. T? Explain.

Answers Nishimura 2010

1) Sampler + bandpass filter gets you close.
The time shift plays a role in achieving the inverted spectrum but it requires some finesse.

2) Yes. The center of mass is $\int x\, f(x)\, dx$ which relates nicely to $F(s)$.

3) It would be best to agree and disagree with Dr. T.
Dr. T. is partially correct but additional constraints on s_0 and $F(s_0)$ are needed to avoid potential ambiguities in determining the center of mass.

2009-2010 PhD Qualifying Examination

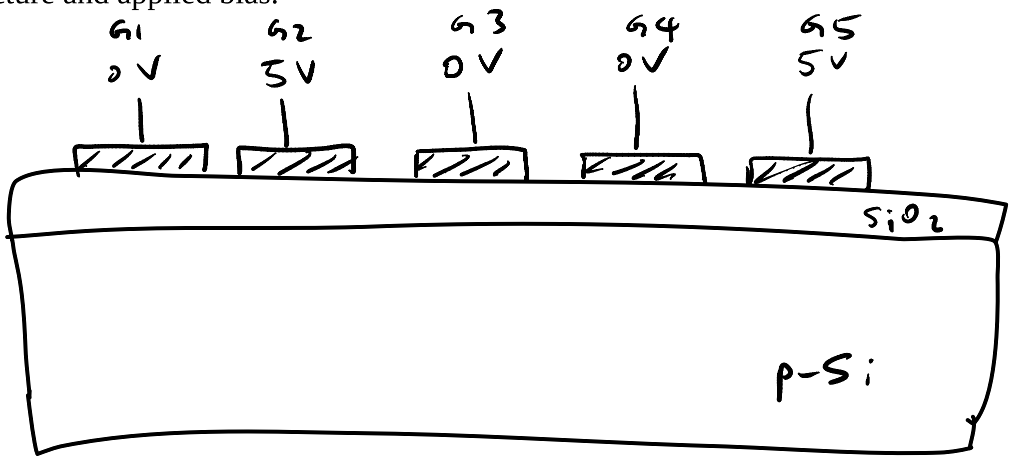
Professor Yoshio Nishi

1. When two different metals with different work functions are coming closer, what would happen before and after those two metals contact to each other? Draw a band diagram, and explain current-voltage characteristics across the two metals.
2. If one of the metals is replaced by a semiconductor, draw a band diagram and explain current-voltage characteristics across the metal-semiconductor..
3. If there are energetically localized electronic states between the metal and the semiconductor, how the band diagram could change?

2010 Qual Exam Questions

Prof. H.-S. Philip Wong

1. Do you know how a Charge-Coupled Device (CCD) works?
2. Draw the energy band diagram along the SiO2/Si interface for the following device structure and applied bias.



3. Assume some electrons are collected under G2. Now, we want to move the electrons to the right under G3. What biases would you apply to the gates G1 to G5?
4. What are the forces acting on the electrons that move the charges from G2 to G3?
5. If we want to move the electrons faster, what would you do? You can change anything you want, including (but not limited to) applied biases, device structure, doping, and the materials of the device.
6. If there is an n+ doping in the p-Si in between the gates, how does the band diagram look like and how does it affect the charge transfer?

ELECTRICAL ENGINEERING

QUALS QUESTIONS

2010

HTTP://EE.STANFORD.EDU/PHD/QUALS

EE Quals 2010 (Hardware)

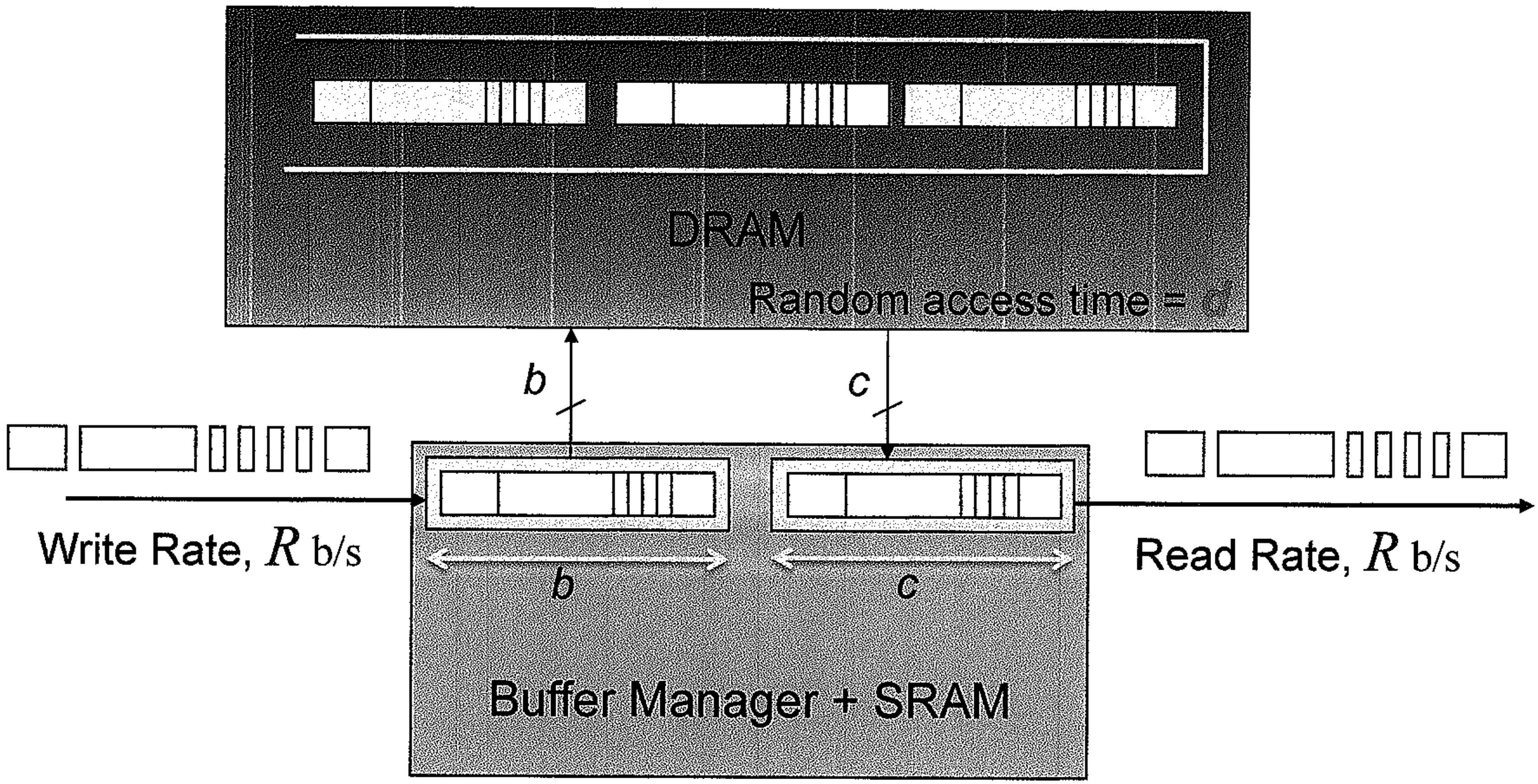
Nick McKeown

Question 1

(a) Why do we have a cache in a computer?

(b) I have an SRAM with random access time s, and a DRAM with random access time d. The probability of finding an entry in the SRAM is p. If I assume $d = 100s$, what value of p do I need so that the expected lookup time is twice as long as the random access time of the SRAM?

Single FIFO Queue



Question 2

Consider a cache for a FIFO queue in a network switch, router or network interface card, built from the same SRAM and DRAM.

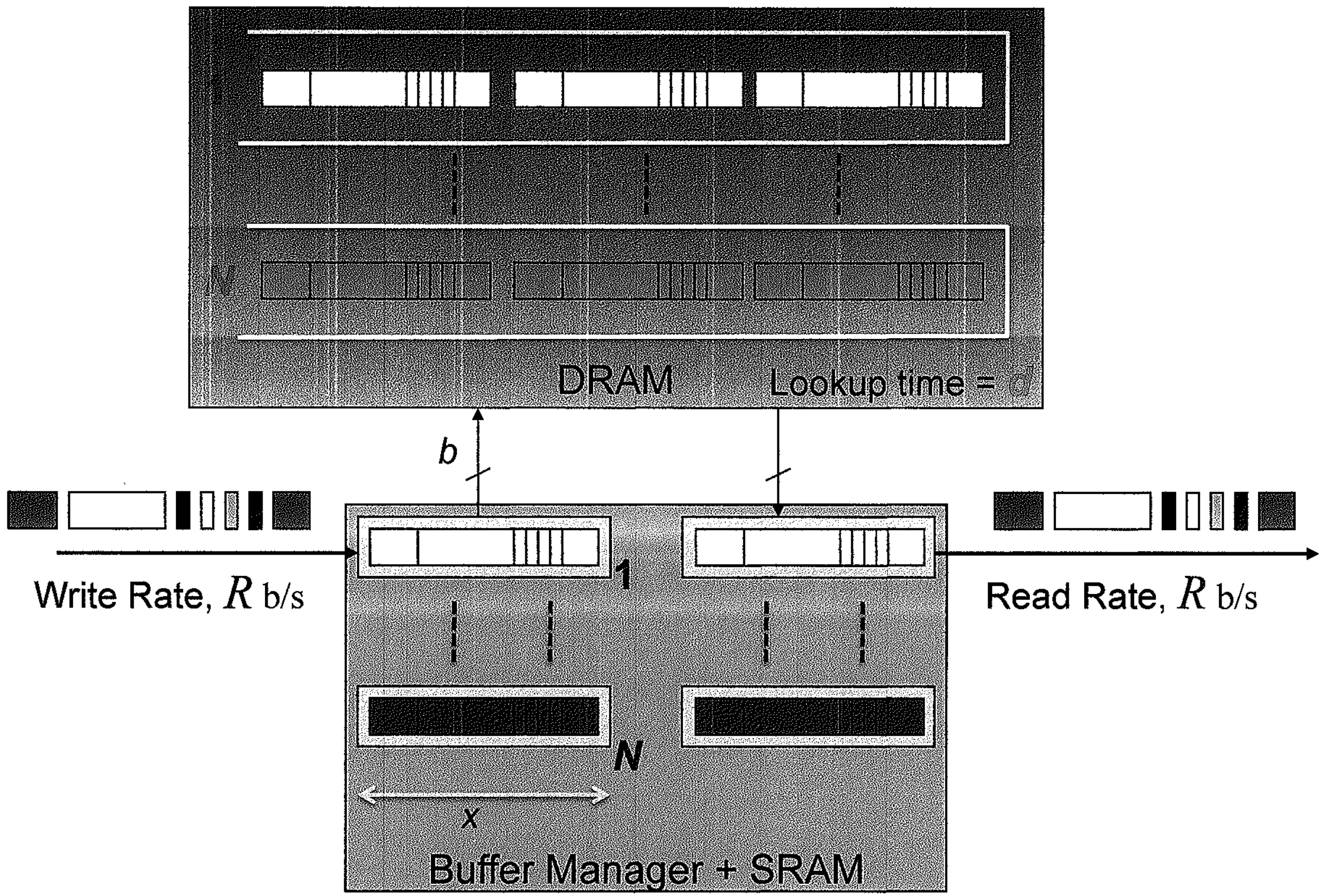
(a) Explain how it works.

(b) How large does the block size, b, need to be so that it won't overflow?

(c) How about c, so that the "head cache" won't underflow?

(d) What problems do we run into if we want to build a cache for N FIFO queues, instead of just 1?

Multiple FIFO Queues



Question 3

We want to figure out how large x needs to be, so the SRAM will never overflow.

(a) Is $x = b$ big enough? Explain.

(b) How can we figure out how large the SRAM needs to be?

2009-2010 PhD Qualifying Examination

Professor Yoshio Nishi

1. When two different metals with different work functions are coming closer, what would happen before and after those two metals contact to each other? Draw a band diagram, and explain current-voltage characteristics across the two metals.
2. If one of the metals is replaced by a semiconductor, draw a band diagram and explain current-voltage characteristics across the metal-semiconductor..
3. If there are energetically localized electronic states between the metal and the semiconductor, how the band diagram could change?

2010 PhD Quals Questions
J. S. Harris

1. I have several LEDs here of different colors. Can you first briefly tell me how an LED works and what is different about the devices that produce different colors of light?

2. Can you draw the Current-Voltage and Light-Current (L-I) curves for me for say two different color LEDs and identify which is which?

3. Can you sketch the characteristic spectral distribution for one of the LEDs? Why can you have photons with $\lambda < E_g$, but not greater than E_g?

4. There is a lot of effort today to produce solid state lighting. How can I produce a White LED?

5. I now have a red laser. Can it be made of the same materials as a red LED? What is different about the laser compared to the LED?

6. Can you draw the I-V and L-I curves and spectral distribution for the laser? How do they differ from the LED and why?

7. If I now change the temperature of the LED and laser, how do the spectral characteristics change and why?

8. I have a polarizing filter and when I put it in front of the LED, it has virtually no effect, but when I put it in front of the laser, and rotate it, it changes from virtually zero transmission to unchanged transmission. What does this tell you about the fundamental recombination processes occurring in the laser vs. the LED?

9. If stimulated emission and absorption are reciprocal processes, can I use a LED or laser as a solar cell? What might I do differently to optimize the solar cell vs. a laser or LED?

05400D

EE QUALIFYING EXAM
JANUARY 2010

This is a problem about discrete signals (vectors) of length N.

Let I be a subset of $\{0, 1, \ldots, N-1\}$ and let I' be the complementary subset. (For example, I could be the set of even numbers in $\{0, 1, \ldots, N-1\}$ and I' would then be the set of odd numbers.)

Let $\mathbb{B}^I$ be the set of signals whose spectrum is supported on I, i.e.,

$$\underline{f} \in \mathbb{B}^I \iff \underline{\mathcal{F}}\,\underline{f}[m] = 0 \quad \text{if } m \in I'.$$

Here $\underline{\mathcal{F}}$ is the discrete Fourier transform.

- What is the set of signals that are *orthogonal* to $\mathbb{B}^I$, i.e., what is the orthogonal complement to $\mathbb{B}^I$?

Let $\underline{h}$ be the signal defined by

$$\underline{\mathcal{F}}\,\underline{h}[m] = \begin{cases} 1, & m \in I \\ 0, & m \in I' \end{cases}$$

- Show that the orthogonal projection onto $\mathbb{B}^I$ is given by

$$K\underline{f} = \underline{h} * \underline{f}.$$

- What is the orthogonal projection onto the orthogonal complement of $\mathbb{B}^I$?

Solutions

For the first question, two signals $\underline{f}$ and $\underline{g}$ are orthogonal if their inner product, $\underline{f} \cdot \underline{g}$ is 0. By Parseval's theorem

$$\underline{f} \cdot \underline{g} = \frac{1}{N}(\underline{\mathcal{F}}\,\underline{f} \cdot \underline{\mathcal{F}}\,\underline{g}).$$

If $\underline{f} \in \mathbb{B}^I$ then

$$\begin{aligned}\underline{\mathcal{F}}\,\underline{f} \cdot \underline{\mathcal{F}}\,\underline{g} &= \sum_{n=0}^{N-1} \underline{\mathcal{F}}\,\underline{f}[n]\overline{\underline{\mathcal{F}}\,\underline{g}[n]} \\ &= \sum_{n \in I} \underline{\mathcal{F}}\,\underline{f}[n]\overline{\underline{\mathcal{F}}\,\underline{g}[n]}\end{aligned}$$

since $\mathcal{F}\underline{\mathrm{f}}[n] = 0$ if $n \in I'$. This will be 0 *for all* $\underline{\mathrm{f}} \in \mathbb{B}^I$ if and only if $\mathcal{F}\underline{\mathrm{g}}[n] = 0$ for all $n \in I$. This says that $\underline{\mathrm{g}}$ must be in $\mathbb{B}^{I'}$. Sybolically,

$$(\mathbb{B}^I)^\perp = \mathbb{B}^{I'}.$$

For the second question, to show that $K\underline{\mathrm{f}} = \underline{\mathrm{h}} * \underline{\mathrm{f}}$ defines the orthogonal projection onto $\mathbb{B}^I$ we have to do several things. First, if $\underline{\mathrm{f}}$ is any signal we have to show that $\underline{\mathrm{h}} * \underline{\mathrm{f}} \in \mathbb{B}^I$. For this, for any m the convolution theorem and the definition of $\underline{\mathrm{h}}$ gives

$$\begin{aligned}\mathcal{F}(\underline{\mathrm{h}} * \underline{\mathrm{f}})[m] &= (\mathcal{F}\underline{\mathrm{h}}[m])(\mathcal{F}\underline{\mathrm{f}}[m]) \\ &= \begin{cases} \mathcal{F}\underline{\mathrm{f}}[m], & m \in I \\ 0, & m \in I' \end{cases}\end{aligned}$$

Thus $\underline{\mathrm{f}} * \underline{\mathrm{h}}$ is supported on I, i.e., $\underline{\mathrm{h}} * \underline{\mathrm{f}} \in \mathbb{B}^I$.

Second, if $\underline{\mathrm{f}}$ is already in $\mathbb{B}^I$ we should have $\underline{\mathrm{h}} * \underline{\mathrm{f}} = \underline{\mathrm{f}}$. But if $\underline{\mathrm{f}} \in \mathbb{B}^I$ then already $\mathcal{F}\underline{\mathrm{f}}[m] = 0$ for $m \in I$ and so by the definition of $\underline{\mathrm{h}}$,

$$(\mathcal{F}\underline{\mathrm{h}})(\mathcal{F}\underline{\mathrm{f}}) = \underline{\mathrm{f}}.$$

Taking the inverse DFT gives

$$\underline{\mathrm{h}} * \underline{\mathrm{f}} = \underline{\mathrm{f}}.$$

As an aside, another way to do this part of the problem is to observe that

$$(\mathcal{F}\underline{\mathrm{h}})(\mathcal{F}\underline{\mathrm{h}}) = \mathcal{F}\underline{\mathrm{h}},$$

hence

$$\underline{\mathrm{h}} * \underline{\mathrm{h}} = \underline{\mathrm{h}}.$$

Thus for any signal $\underline{\mathrm{f}}$,

$$K^2\underline{\mathrm{f}} = K(K(\underline{\mathrm{f}})) = \underline{\mathrm{h}} * (\underline{\mathrm{h}} * \underline{\mathrm{f}}) = (\underline{\mathrm{h}} * \underline{\mathrm{h}}) * \underline{\mathrm{f}} = \underline{\mathrm{h}} * \underline{\mathrm{f}} = K\underline{\mathrm{f}},$$

that is

$$K^2 = K,$$

which is the definition of a projection.

Why is this an orthogonal projection? If $\underline{\mathrm{f}}$ is in $\mathcal{B}^{I'}$, the orthogonal complement of $\mathbb{B}^I$, then $\mathcal{F}\underline{\mathrm{f}}[m] = 0$ for $m \in I$, hence, by definition of $\underline{\mathrm{h}}$,

$$(\mathcal{F}\underline{\mathrm{h}})(\mathcal{F}\underline{\mathrm{f}}) = 0,$$

whence

$$K\underline{\mathrm{f}} = \underline{\mathrm{h}} * \underline{\mathrm{f}} = 0.$$

056000

Finally, for the last question, the orthogonal projection onto $\mathbb{B}^{I'}$ is given by

$$K' = I - K.$$

With

$$(I-K)\underline{\mathrm{f}} = \underline{\mathrm{f}} - K\underline{\mathrm{f}} = \underline{\mathrm{f}} - \underline{\mathrm{h}} * \underline{\mathrm{f}}$$

we can also write

$$K'\underline{\mathrm{f}} = \underline{\mathrm{f}} - \underline{\mathrm{h}} * \underline{\mathrm{f}}.$$

or as a convloution

$$K\underline{\mathrm{f}} = \underline{\mathrm{h}}' * \underline{\mathrm{f}},$$

where $\underline{\mathrm{h}}'$ is given by

$$\underline{\mathcal{F}\mathrm{h}}'[m] = \begin{cases} 1, & m \in I' \\ 0, & m \in I \end{cases}$$

JOHAR[illegible]

Problem 1. Suppose that X is a geometric random variable, with probability of success p. Let t, s be positive integers with $t > s$. Explain why $P(X > t|X > s) = P(X > t - s)$.

Problem 2. For what values of N does the following statement hold?
"If $X_1, \ldots, X_N$ are jointly Gaussian random variables such that $\mathrm{Var}(\sum_i X_i) = \sum_i \mathrm{Var}(X_i)$, then $X_1, \ldots, X_N$ are independent."

Problem 3. Suppose that $X_1, X_2, \ldots$ are a sequence of random variables on the nonnegative integers, such that $E[X_n] \to \infty$ as $n \to \infty$. Does it follow that $P(X_n = 0) \to 0$ as $n \to \infty$?

Now suppose that in addition, $\mathrm{Var}(X_n) = E[X_n]$. Does it follow that $P(X_n = 0) \to 0$ as $n \to \infty$?

S. Mitra

2010 EE Qualifying Exam Questions

1. What is the difference between a latch and a flip-flop?

2. Helped student draw a gate-level MUX and a simple D-latch out of that MUX.
OUT = OR(A, B), A = AND (D, CLK), B = AND (OUT, E), E = NOT(CLK)
Does the above circuit work correctly? Why not?

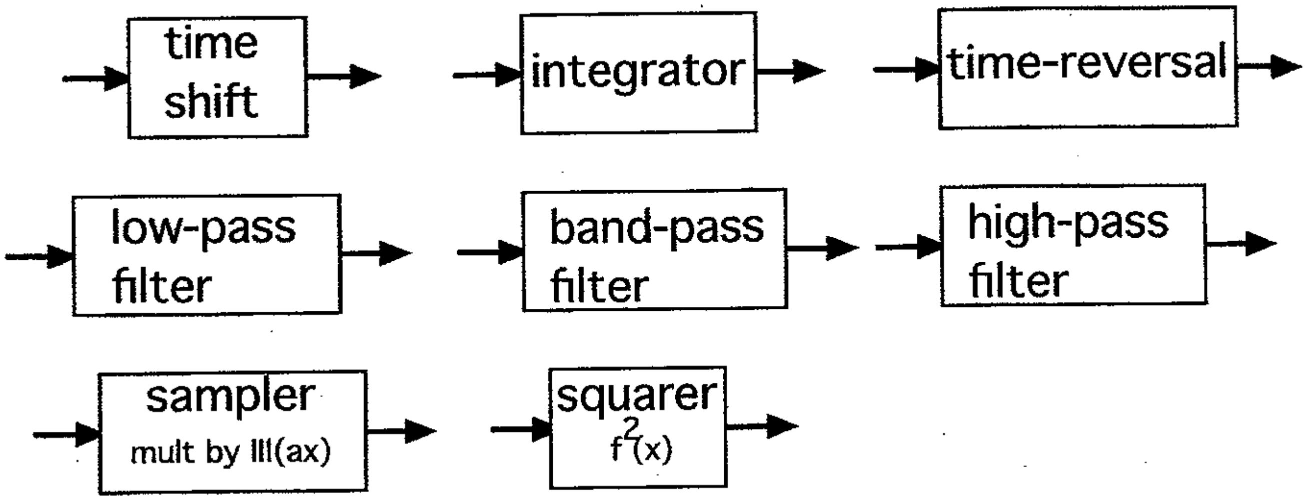
3. How will you fix it?

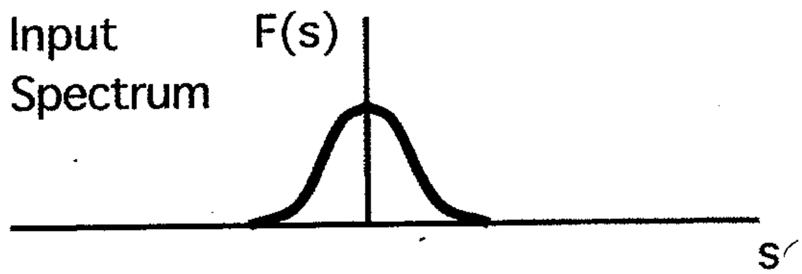
4. Suppose that you are given a flip-flop-based FSM. I claim that you can automatically convert the circuit into one where there are no flip-flops / clk or any sequential element (i.e., replace each flip-flop by a wire) without changing the combinational logic function. How can you do that? Assume that for each gate there is a single delay value associated with that gate.

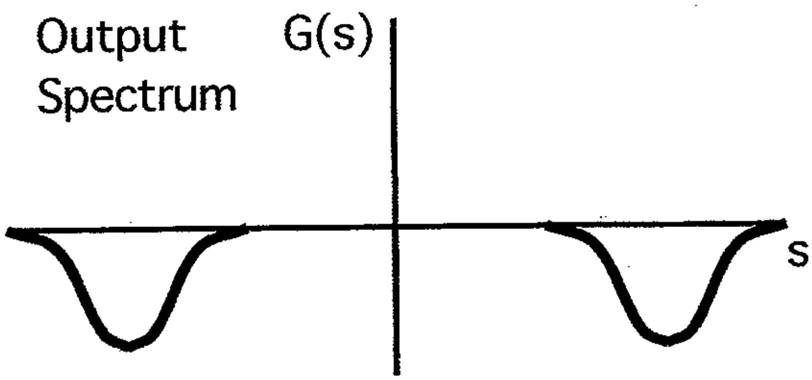
5. You can have trillions of paths if you plan on path delay balancing. What's a scalable way to do this?

Nishimura 2010

(1) Selecting the appropriate building block(s) from those below, design a system to achieve the output spectrum.







(2)

Let f(x) represent an unknown distribution of mass for an object along x. You are given its Fourier transform F(s).

Is it possible to determine the object's center of mass from F(s) without taking the inverse Fourier transform? Explain.

(3)

Dr. T says the center of mass can be determined from just a single value of $F(s)$, call it $F(s_o)$, if $f(x)$ is symmetric about its center of mass.

Do you agree with Dr. T? Explain.

Answers Nishimura 2010

1) Sampler + bandpass filter gets you close.
The time shift plays a role in achieving the inverted spectrum but it requires some finesse.

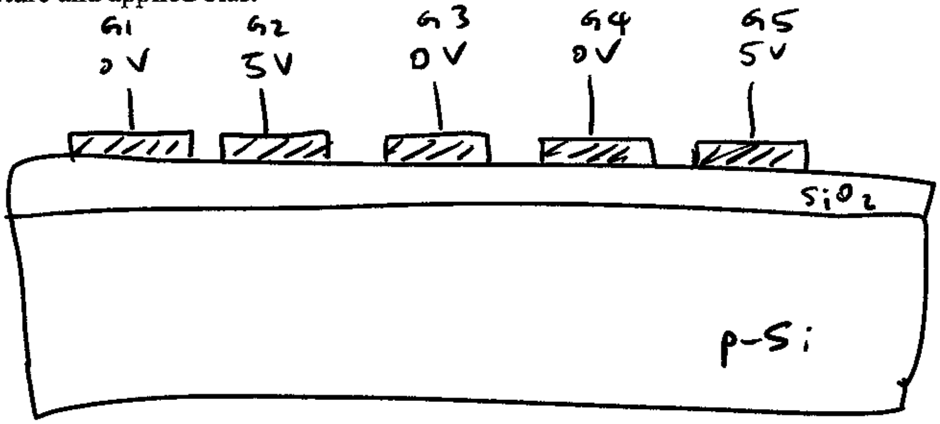
2) Yes. The center of mass is $\int x\, f(x)\, dx$ which relates nicely to $F(s)$.

3) It would be best to agree and disagree with Dr. T.
Dr T. is partially correct but additional constraints on s_0 and $F(s_0)$ are needed to avoid potential ambiguities in determining the center of mass.

2010 Qual Exam Questions

Prof. H.-S. Philip Wong

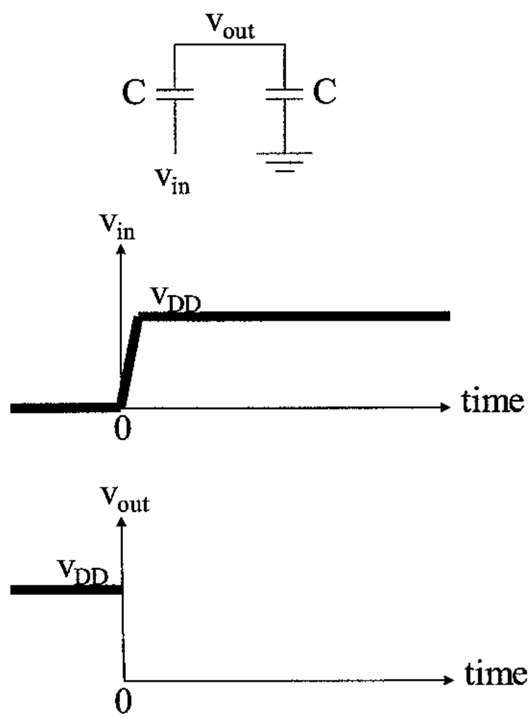
1. Do you know how a Charge-Coupled Device (CCD) works?
2. Draw the energy band diagram along the SiO2/Si interface for the following device structure and applied bias.



3. Assume some electrons are collected under G2. Now, we want to move the electrons to the right under G3. What biases would you apply to the gates G1 to G5?
4. What are the forces acting on the electrons that move the charges from G2 to G3?
5. If we want to move the electrons faster, what would you do? You can change anything you want, including (but not limited to) applied biases, device structure, doping, and the materials of the device.
6. If there is an n+ doping in the p-Si in between the gates, how does the band diagram look like and how does it affect the charge transfer?

2010 Qualifying Exam
Simon Wong

1. Sketch V_{out}.
   Ans: final $V_{out} = 1.5\ V_{DD}$
2. What is initial total stored energy ? What is final total stored energy ?
   Ans: initial $E = CV_{DD}^2$; final $E = 1.25CV_{DD}^2$
3. If C can be varied, what is the maximum final V_{out} ?
   Ans: $C_{left} >> C_{right}$; maximum final $V_{out} = 2\ V_{DD}$
4. Modify the circuit to achieve final $V_{out} = 3\ V_{DD}$.
   Many possible solutions including the following :
   $C_{left} >> C_{right}$; insert a diode between the 2 C, drive the bottom plate of the right C to V_{DD}.



MENG

1. You have access to a cheap analog low-pass filter which has a fairly flat unit gain response from $\Omega=0$ to $\Omega=60$ *KHz*, and a stopband attenuation of *100 dB* for $\Omega>80$ *KHz*. With this analog anti-aliasing filter you are asked to design a 16-bit A/D converter system with a baseband sampling rate at *60 KHz*.

a) If you were to use the above analog filter for the design of an oversampling A/D converter system, what is the **minimum** sampling frequency that you should choose?

b) With the sampling rate that you picked in part (a), draw a block diagram of the A/D converter system from the analog input signal to the final sampled output sequence at *60 KHz*. This block diagram needs to include the specification (passband bandwidth, stopband attenuation, gain, etc.) of any filter that would be needed in the system.

c) Is your answer in part (b) the most hardware-efficient solution? If not, use another sampling rate to design this A/D converter system with less amount of computation. Draw a block diagram to describe this system.

a): The minimum sampling rate is 30Khz + 80Khz = 110 Khz

b): The block diagram consists of the above anti-aliasing filter, a sampler at 110Khz, followed by an up-converter by a factor of 6, a digital filter, and a down-converter by a factor of 11. The digital filter specs are: passband, $\pi/11$; stopband attenuation: 100dB, gain: 6.

c): 120Khz.

MENG

2. Indicate whether the following statements are true or false. If the statement is true, give a brief justification. If the statement is false, give a simple counter example or a clear reason.

   a) If a real-coefficient digital filter has a zero-phase frequency response, then it must be a non-causal filter.

   True.

   b) All periodic continuous-time signals will remain periodic after sampling.

   False. The sampled signal is periodic only if the ratio of the period of the original signal and the sampling period is a rational number.

   c) If a z-transform doesn't have a region of convergence on the Z-plane, then its time-domain sequence doesn't exist.

   False. Many time-domain sequences don't have a ROC in their Z-transform.

   d) The sum of the impulse responses of two minimum-phase filters is always the impulse response of another minimum-phase filter.

   False. The sum of the impulses responses will have a transfer function that is the sum of the transfer functions of the two original impulse responses. The poles of this transfer function will remain the same. The zeros of this transfer function, however, will not be the same and may move out of the unit circle.

   e) If N samples of the discrete-time Fourier transform of a discrete-time sequence $h(n)$ are taken at $2\pi k/N$, where $k=0,...,N-1$, then this set of samples represents the N-point discrete-Fourier transform of $h(n)$.

   False. The N samples of the DTFT of $h(n)$ are the DFT of the "aliased" $h(n)$.

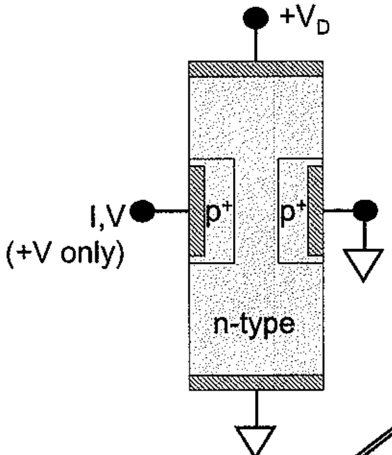
Fabian Pease

For what was the most recent Nobel Prize in Physics awarded?

a) How does the image sensor in your digital camera work? (if you don't know, invent one)
b) Why, do you think, the CCD won the prize when other devices (Solar cell, DRAM, SRAM, Flash memory, CPU-chip) did not?
c) Why, do you think, has the CMOS sensor largely replaced the CCD in digital cameras?

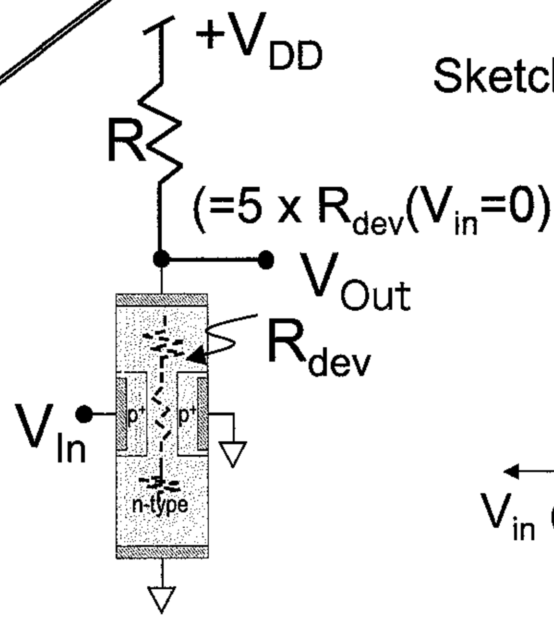
DUTTON

Part A)

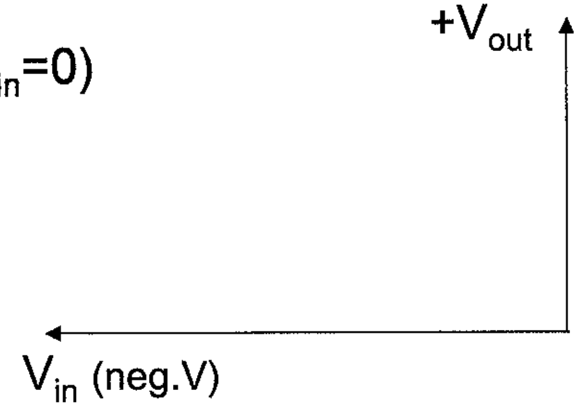


Sketch the I, +V for a fixed $V_D>0$

Part B)



Sketch the V_{out} vs. V_{in} ($V_{in}<0$)

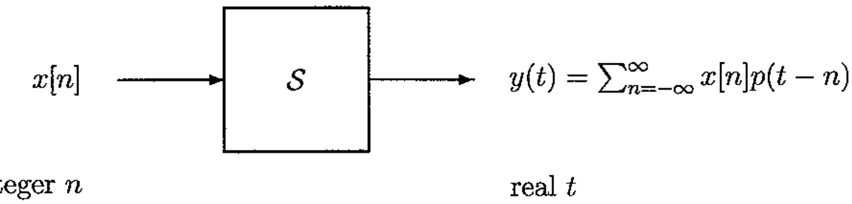


After discussion of Parts A) and B) there were follow-on questions (not included here) dealing with different terminal connections and bias conditions for Part A) and how to “optimized” the device performance in Part B)

$\boxed{\text{January 2010}}$

The questions are colored red.
Solutions to R.M. Gray's 2010 qualifying exam problem.

The following system is useful as a model in pulse amplitude modulation (PAM) systems and digital-to-analog (D/A) converters:



where $p(t)$ is a real-valued continuous-time signal satisfying

$$\int_{-\infty}^{\infty} p(t)p(t-n)dt = \delta_n = \begin{cases} 1 & n = 0 \\ 0 & \text{all nonzero integers} \end{cases} \tag{1}$$

Define the discrete-time Fourier transform (DTFT) of a signal $x[n]$ by

$$X(f) = \sum_{n=-\infty}^{\infty} x[n]e^{-j2\pi fn}; -\frac{1}{2} \leq f \leq \frac{1}{2}$$

and the continuous-time Fourier transform (CTFT) of a signal $y(t)$

$$Y(f) = \int_{-\infty}^{\infty} y(t)e^{-j2\pi ft}dt; -\infty < f < \infty,$$

where $j = \sqrt{-1}$.

First Question:
Find a *simple* relationship between $Y(f)$ and $X(f)$.

Solution This was intended as a straightforward start using standard Fourier proof techniques — substitute (plug in) the definition of the signal to the definition of the transform $Y(f)$ asked for, interchange the order of summation and integral, and then simplify.

$$\begin{aligned} Y(f) &= \int_{-\infty}^{\infty} y(t)e^{-j2\pi ft}dt = \int_{-\infty}^{\infty} \left[\sum_{n=-\infty}^{\infty} x[n]p(t-n)\right] e^{-j2\pi ft}dt \\ &= \sum_{n=-\infty}^{\infty} x[n] \int_{-\infty}^{\infty} p(t-n)e^{-j2\pi ft}dt = \sum_{n=-\infty}^{\infty} x[n]P(f)e^{-j2\pi fn} \end{aligned}$$

where the last step is the usual Fourier shift theorem for CT signals (or just change variables in the integral). Thus

$$Y(f) = X(f)P(f).$$

A tricky point here is that $Y(f)$ should be defined for all real f, but $X(f)$ was defined only for f in $[-1/2, 1/2]$. The formula makes sense, however, if we take $X(f)$ to be the periodic extension, that is, just use the sum in the DTFT definition for all real f.
Many people complicated the problem by trying to convert the DT signal into a CT signal using impulse trains. This way leads to the answer, but it makes things much more complicated. Some people observed correctly that the left hand side resembles a convolution and tried to quote the convolution theorem, but here the "convolution" is discrete time while the output signal is continuous time, so the usual convolution theorems (for DT or CT) do not directly apply.

Recall that $p(t)$ is real and

$$\begin{aligned}\int_{-\infty}^{\infty} p(t)p(t-n)dt &= \delta_n \qquad (1)\\ P(f) &= \int_{-\infty}^{\infty} p(t)e^{-j2\pi ft}dt\end{aligned}$$

Second Question:
Find a *simple* expression for

$$\int_{-\infty}^{\infty} |P(f)|^2 e^{j2\pi fn} df$$

Solution There are *many* ways to do this problem.
The most straightforward approach is the standard Fourier proof method of substitution and interchanging order of integration.

$$\begin{aligned}\int_{-\infty}^{\infty} |P(f)|^2 e^{j2\pi fn} df &= \int_{-\infty}^{\infty} P(f)\left[\int_{-\infty}^{\infty} p(t)e^{-j2\pi ft}dt\right]^* e^{j2\pi fn} df\\ &= \int_{-\infty}^{\infty} p(t)\left[\int_{-\infty}^{\infty} P(f)e^{j2\pi f(t+n)}df\right] dt\\ &= \int_{-\infty}^{\infty} p(t)p(t+n)dt\end{aligned}$$

where we have used the Fourier inversion formula to recover p from P. From (1) the answer is

$$\int_{-\infty}^{\infty} |P(f)|^2 e^{j2\pi fn} df = \delta_n$$

Some people got bogged down by substituting the time domain intergral for both occurrences of $P(f)$, which is messier because of the triple integration. I tried to warn people who took a path that was likely to get tangled in details.
A shortcut to the answer is to recognize the integral as the continuous-time inverse Fourier transform of $|P(f)|^2$ evaluated at time n and that $|P(f)|^2$ is the transform of the CT autocorrelation of p,

$$r_p(\tau) = \int_{-\infty}^{\infty} p(t)p(t-\tau)dt$$

(from the correlation theorem for continuous time Fourier transforms), which when evaluated at an integer time yields the Kronecker delta δ_n from (1).
Equivalently, the integral asked for is the integral of the product of $P(f)$ and $P^*(f)e^{j2\pi fn} = \left(P(f)e^{-j2\pi fn}\right)^*$. From the generalized Parseval's theorem this is the integral in the time domain of the product of the inverse Fourier transforms of these signals, which are $p(t)$ and $p^*(t-n) = p(t-n)$, which from (1) is the Kronecker delta δ_n.
Several people tried another short cut that does not work. They correctly recognized (1) as an autocorrelation and reasoned that therefore if they transformed both sides the left hand side should be $|P(f)|^2$ (the transform of a correlation) and the right hand side should be 1 (the transform of a delta function), thus $|P(f)|^2 = 1$ for "all" f. But the correlation is a continuous time correlation, while the equation is a discrete time relation – the delta function is a Kronecker

delta (discrete time), not a Dirac delta (continuous time). Thus to transform both sides of (1) the appropriate transform is the DTFT and not CTFT, and one can not infer from (1) that the transform of the left hand side is a constant. One can not use the DT autocorrelation result (or Parseval's theorem) since a sum, not an integral, would be needed on the left.

Recall that

$$\begin{aligned}\int_{-\infty}^{\infty} p(t)p(t-n)dt &= \delta_n & (1)\\ P(f) &= \int_{-\infty}^{\infty} p(t)e^{-j2\pi ft}dt\end{aligned}$$

Third Question:
Find a waveform $p(t)$ which has the desired property (1) and has the additional property that $P(t)$ *also* satisfies property (1); that is,

$$\int_{-\infty}^{\infty} P(t)P(t-n)dt = \delta_n \tag{2}$$

Solution Here are two possible approaches:

1. Guess and show it works. There are not many signals p which satisfy (1), so it is easy to see if they also satisfy (2) if you either know or can find the CTFT.

2. Look at the formulas the signals must satisfy and find a solution.

First Approach: A simple signal which satisfies (1) is the box or rect function or square pulse

$$p_0(t) = \begin{cases} 1 & |t| \le \frac{1}{2} \\ 0 & \text{otherwise.} \end{cases}$$

Its Fourier transform is easily found (or recalled from memory) as

$$\begin{aligned} P_0(f) &= \int_{-1/2}^{1/2} e^{-j2\pi ft}dt = \left.\frac{e^{-j2\pi ft}}{-j2\pi ft}\right|_{-1/2}^{1/2} \\ &= \frac{e^{-j\pi f}}{-j\pi f} - \frac{e^{j\pi f}}{-j\pi f} = \frac{\sin(\pi f)}{\pi f} \end{aligned}$$

which is the sinc function, $P_0(f) = \text{sinc}(f)$. But these are also orthogonal as in (1) from Parseval's theorem and the modulation theorem:

$$\int_{-\infty}^{\infty} \text{sinc}(f-n)\,\text{sinc}(f)df = \int_{-\infty}^{\infty} p_0(t)e^{-j2\pi tn}p_0(t)dt = \int_{-1/2}^{1/2} e^{-j2\pi tn}dt = \delta_n.$$

Second Approach: Combining the Second Question with (2) yields

$$\int_{-\infty}^{\infty} |P(f)|^2 e^{j2\pi fn}df = \int_{-\infty}^{\infty} P(f)P(f-n)df = \delta_n$$

so $P(f)$ must satisfy

$$\int_{-\infty}^{\infty} P(f)\left[P(f)e^{j2\pi fn} - P(f-n)\right]df = 0.$$

A sufficient condition is that the bracketed term is itself 0 for all integer n. It is 0 for $n = 0$ and it will be 0 for all nozero integers if we pick

$$P(f) = P_1(f) = \begin{cases} 1 & |f| \leq \frac{1}{2} \\ 0 & \text{otherwise} \end{cases}.$$

Similar to the first approach, taking the (inverse) Fourier transform yields $p_1(t) = \text{sinc}(t)$. Thus both the box function $p_0(t)$ and the sinc function $p_1(t) = \text{sinc}(t)$ have the desired properties. The signal $p(t)$ is time limited, but occupies an infinite bandwidth in the time domain (although most of the energy is between the first zeros of the sinc function), while the second has infinite extent in time, but is bandlimited to $[-1/2, 1/2]$.

Lam, Monica

1. How do you parallelize matrix multiplication?

2. How do you parallelize a SOR (Successive Over-Relaxation) algorithm?

Mark Horowitz

I have a clock in my office that uses an oscillating wand with LEDs on it that it uses to display the time. It contains a microcontroller that runs the clock.

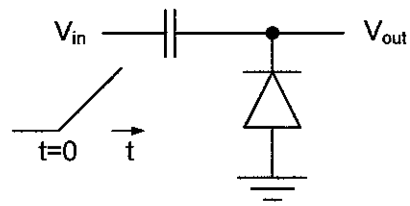
1. How would you use the controller to control the freq and amplitude of the wand oscillator.

2. How do you sync the timing of the display to the position of the wand (where does the sensor need to be place).

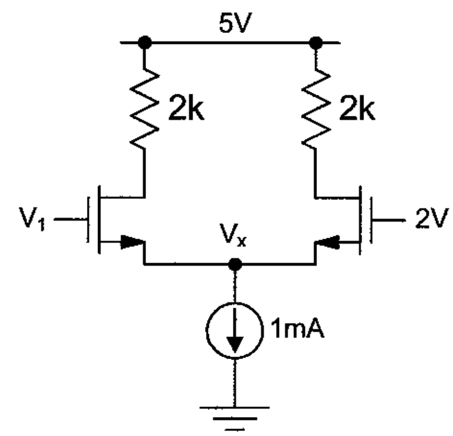
3. Please make the timing of the dots on the display all the same size.

Name:	Stanford EE Quals 2010 Murmann

1. The circuit below is driven by a voltage ramp that increases linearly with time. Sketch (qualitatively) V_{out} (t), assuming $V_{out}(0) = 0V$.



2. For the circuit below, sketch the voltage V_x as a function of V_1, ranging from 0...5V. The MOSFETs obey the ideal square law equation with $\mu C_{ox}W/L = 4mA/V^2$, and $V_t = 1V$.

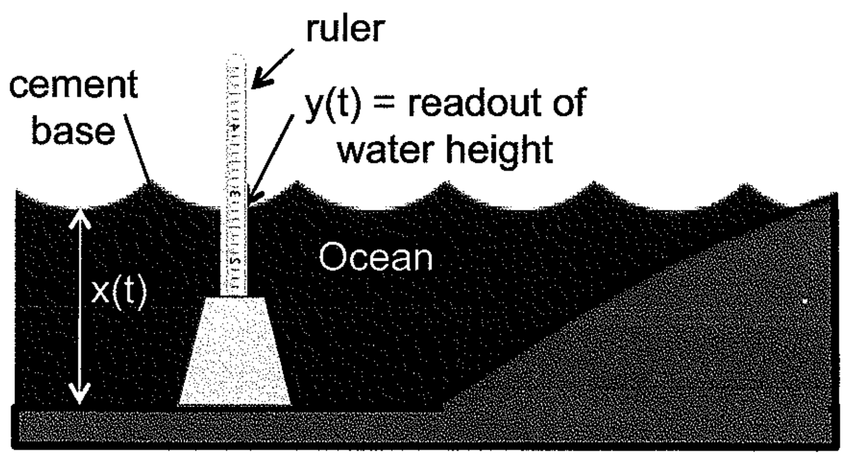


2010 EE PhD Quals

Prof. Daniel Spielman

Your goal is to build a system to measure the tidal fluctuations of the ocean.

Consider the following device:



1. Is this device linear and time-invariant? If so, sketch the frequency response.

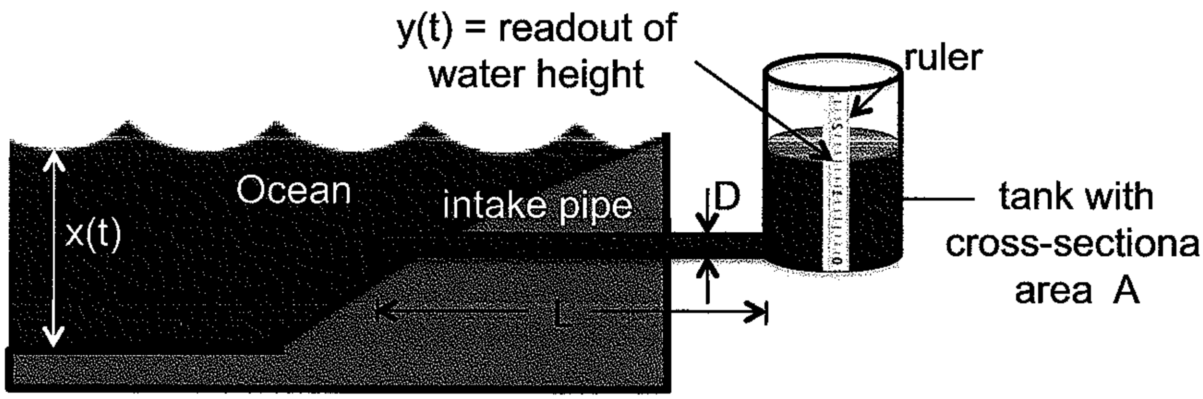
2. What are the advantages and disadvantages of the design?

3. Can you design a better device?

2010 EE PhD Quals

Prof. Daniel Spielman

Consider the following alternative system:



4. Is this device linear and time-invariant? If so, sketch the frequency response.

5. Qualitatively describe the effects of:

    a) increasing A (cross-sectional area of the tank).

    b) increasing L (length of the intake pipe).

    c) increasing D (diameter of the intake pipe).

2010 EE PhD Quals: Solutions

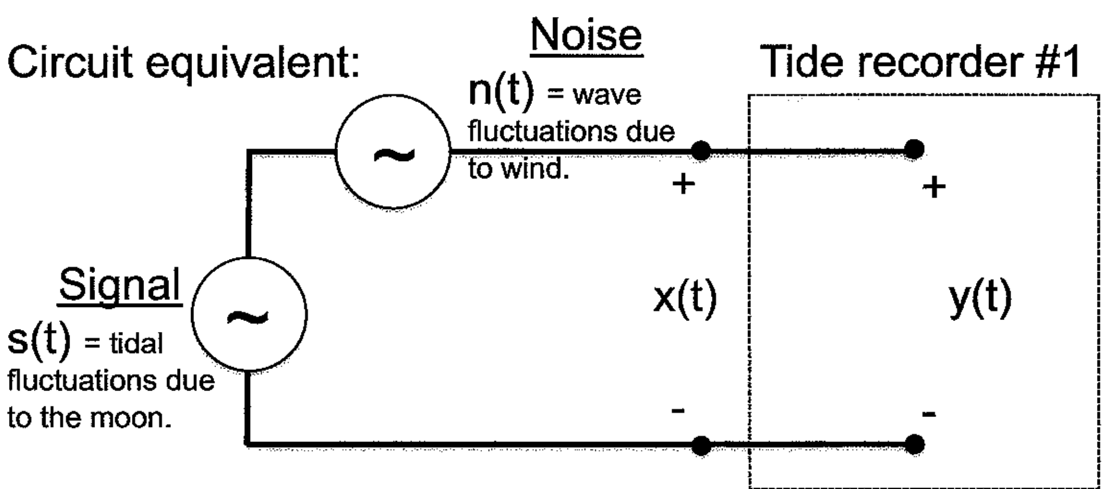
Prof. Daniel Spielman

1. For this system, $y(t) = x(t)$, hence the device is linear and time-invariant. The frequency response is constant for all frequencies.

2. Advantages: inexpensive, easy to build, durable.

   Disadvantages: probably need a boat to actually make the measurements. More importantly, the data will be <u>noisy</u>. The measurements are subject to unwanted variations from surface wave fluctuations. An improved device would filter out this unwanted high frequency noise. Note: tidal fluctuations are on the order of 10^{-5} Hz, while wave action has a dominant component around 10^{-1} Hz.

Circuit equivalent:

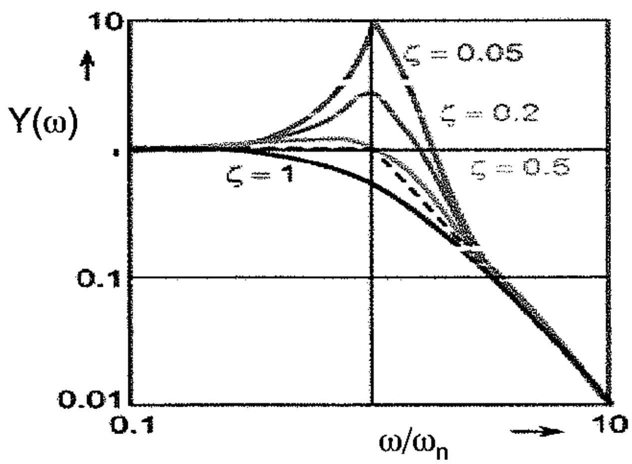


3. There are lots of choices for better measurement systems which can incorporate the desired <u>low-pass filtering</u>. I like the device shown in Question 4.

2010 EE PhD Quals: Solutions

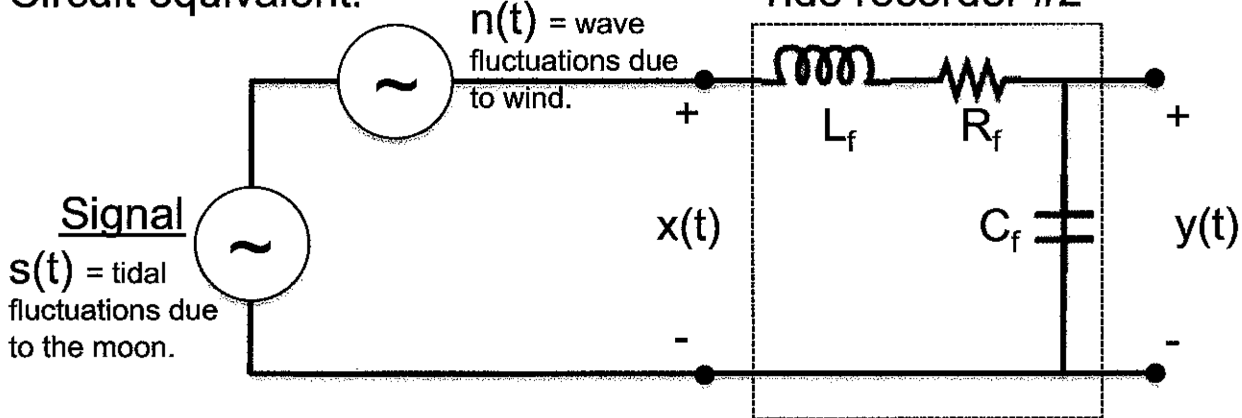
Prof. Daniel Spielman

4. Device #2 is a low pass filter (equivalent to an LRC circuit). The frequency response would look something like (depending on the particular choices of A, L, and D):



In choosing A, L, and D, we want to avoid an underdamped system with a natural frequency in the range of ocean surface waves, typically on of the order of 0.1 Hz (ω_n = 1/sqrt(L_fC_f) where L_f is the inertance [fluid analog of electrical inductance] and C_f is the fluid capacitance) . The damping ratio, ζ, equals (R_f/2)sqrt(C_f/L_f), where R_f is the fluid resistance. We want surface wave fluctuations >> ω_n and tidal variations << ω_n.

Circuit equivalent:



5a. Increasing A increases the fluid capacitance, thereby decreasing the natural frequency ω_n.

5b. Increasing L increases the fluid inertance (thereby decreasing the natural frequency ω_n) and increases the fluid resistance (thereby increasing the damping ζ).

5c. Increasing D decreases the fluid inertance (thereby increasing the natural frequency ω_n) and decreases the fluid resistance (thereby decreasing the damping ζ).

WEISSMAN

Quals Question

Let the random variable X have CDF F. Suppose that X_i iid$\sim X$.

(a) What is the CDF of $\max_{1\leq i\leq n} X_i$?

(b) What is the CDF of $\min_{1\leq i\leq n} X_i$?

Suppose now that $X_{i,j}$ iid$\sim X$

(c) What is the CDF of $f(n,m) = \max_{1\leq i\leq n} \min_{1\leq j\leq m} X_{i,j}$?

(d) Suppose $X \sim$ exponential(λ). Let $Y_m = f\left(e^{\beta m}, m\right)$, for some parameter $\beta > 0$. What does Y_m converge to as $m \to \infty$? In what sense?

(e) Repeat the previous part for the general case $X \sim F$ (can assume that F is continuous and strictly increasing).

Qualifying Exam Questions

January 11 – 15, 2010

Yoshi Yamamoto

What are thermal noise and quantum noise in physical systems? You can choose any one of the following systems and describe the origins of those fluctuations.

1. **Simple macroscopic conductor**
2. **Mesoscopic conductor under ballistic regime**
3. **pn junction under either forward or reverse bias**
4. **Tunnel junction**
5. **Laser/maser**
6. **Parametric oscillator**
7. **Mechanical oscillator**
8. **Bose-Einstein condensation**

2009-2010 EE Ph.D. Qualifying Exam
Question area: *Engineering Physics*
Examiner: *Jelena Vuckovic*

---

1. Assume that there is a lossless oscillating system whose physical property a oscillates over time with angular frequency ω. For example, you can assume that this system represents electromagnetic field in a resonator. (However, you can also think about a mass on a spring, or another oscillating system of your choice, if it is simpler for you.)

a) Write the equation that describes the behavior of a over time. What is its solution?

b) If the system is not lossless, how does the property a vary in time? How do you have to modify the equation from the part (a) to account for losses?

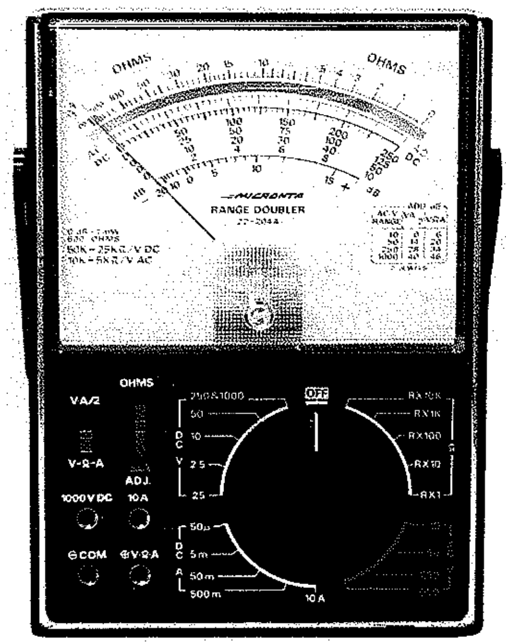
c) Now assume that such a lossless oscillator is coupled to another, identical lossless oscillator. Would the system still oscillate harmonically? If yes, at what frequency?

d) Could you write equations which describe the behavior of such a coupled system?

Ph.D. Quals Question
January 11-15, 2010
A.C. Fraser-Smith
Department of Electrical Engineering
Stanford University

Multimeter

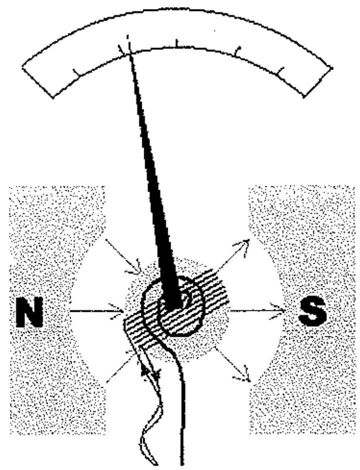
The figure below shows the analog multimeter that was shown to each student. Two leads were also on hand – sometimes inserted in the two sockets on the bottom left (in picture) – sometimes not, but terminating in two pointed probes. The multimeter was not plugged in to a power socket (it was quite clearly not intended to be plugged in) and it had no obvious power source. There was no easily removable panel on the back for a battery; in fact, the back was attached by two not very obvious screws. The students were told that the multimeter was about 40 years old and still worked perfectly, enabling its user to measure DC current, DC voltage and resistance. There were also a few switch settings for measuring AC voltage, but the student was told not to bother about AC. **First question**: Look inside the multimeter in the region where the pointer pivots and explain what electromagnetic principles and mechanical tricks are involved in making it work. One hint was given: Start with its measurement of current.



Simple analog multimeter

The answer to this question usually involved two steps: (i) identification of the key components of the device, and then, closely related to this first step, (ii) an explanation for how these components worked to give a measurement.

A number of students immediately recognized that they were dealing with a galvanometer; some even mentioned a D'Arsonval galvanometer, which was encouraging, but which did not necessarily lead to a correct explanation for how it worked! Although the complete mechanism surrounding the pivot of the pointer was not particularly easy to see, all its important features could be seen. With a little prodding from the examiner, if necessary, the student usually – but not always – ended up sketching out something approximating the mechanism shown below:



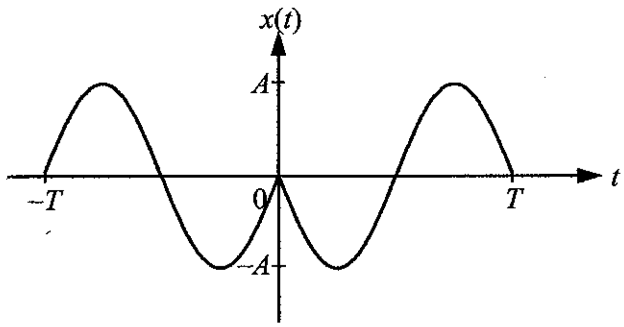
Sketch of a galvanometer mechanism. The red wires carry the current to be measured; it passes through a coil wound around the (steel) cylinder holding the pointer. The green object is a restoring spring. N and S indicate the two poles of a magnet.

Following identification of the components, the examiner looked for some discussion of the force on wires or coils carrying current in a magnetic field. He then looked for some discussion of the curved shape of the faces of the magnet, which combined with the pivoting steel (!) cylinder holding the pointer, would lead to a uniform magnetic field surrounding the coil, independent of its angular position. In other words, the torque on the coil would be dependent only on the strength of the current passing through the coil. The restraining spring would balance this torque and lead to the current reading.

The student was then given a **second question**: If the operation of the meter depended on current passing through the coil, how did it measure resistance? Around this time, as a hint, the examiner would switch the multimeter to a resistance setting and demonstrate how the pointer would move toward its zero setting (on the right side of the dial) when the two probes were touched. The knurled knob labeled "Ohms" in the figure above enabled the pointer to be moved exactly to its zero setting. At this time the better students would carefully, and sometimes not so carefully, inspect the multimeter and declare with authority that there had to be a battery inside, since some source of energy was required to move the pointer, even though there was no obvious way of inserting such battery. The examiner would confirm this fact and then again ask how resistance was measured. The answer: the preparatory step, touching the probes together and adjusting the resistance measurement to zero, established a full-scale deflection for a known internal resistance. When the unknown resistance is placed in series in the circuit the deflection is less than full scale due to the reduced current flowing and the calibrated scale can indicate the resistance.

Stanford University, Department of Electrical Engineering
Qualifying Examination, Winter 2009-10
Professor Joseph M. Kahn

A continuous-time signal $x(t)$, $-\infty < t < \infty$, has a Fourier transform $X(j\omega)$, $-\infty < \omega < \infty$.



Without computing $X(j\omega)$, answer the following:

a. What is $X(j0)$?

b. What is $\int_{-\infty}^{\infty} X(j\omega) e^{j\frac{\omega T}{4}} d\omega$?

c. What is $\int_{-\infty}^{\infty} |X(j\omega)|^2 d\omega$?

d. By what power of $|\omega|$ does $|X(j\omega)|$ decrease as $|\omega| \to \infty$?

Answer

a. $X(j0) = \int_{-\infty}^{\infty} x(t) e^{-j0t} dt = 0$.

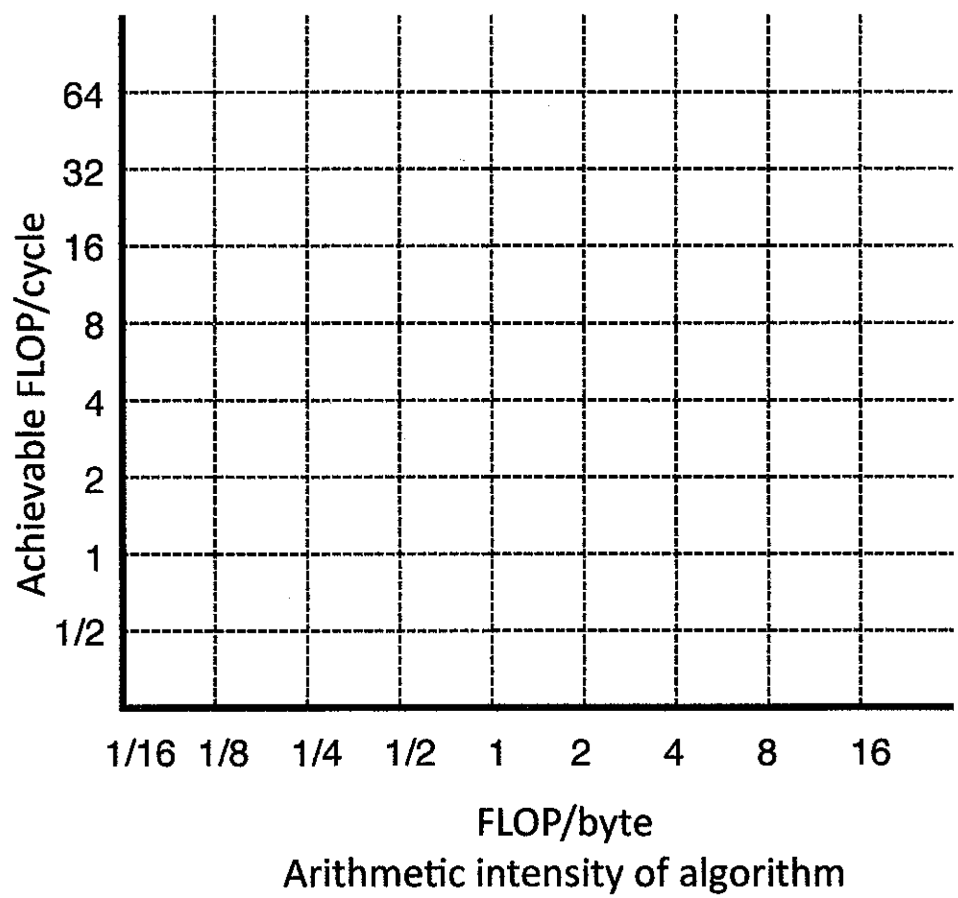
b. $\int_{-\infty}^{\infty} X(j\omega) e^{j\frac{\omega T}{4}} d\omega = 2\pi x\left(\frac{T}{4}\right) = -2\pi A$.

c. $\int_{-\infty}^{\infty} |X(j\omega)|^2 d\omega = 2\pi \int_{-\infty}^{\infty} |x(t)|^2 dt = 2\pi \cdot 2 \cdot \frac{A^2 T}{2} = 2\pi A^2 T$.

d. Since $x(t)$ and $\frac{dx}{dt}$ do not have any impulses (delta functions), but $\frac{d^2x}{dt^2}$ has impulses, $|X(j\omega)| \propto |\omega|^{-2}$ as $|\omega| \to \infty$.

OLUKOTUN

Performance Bounds

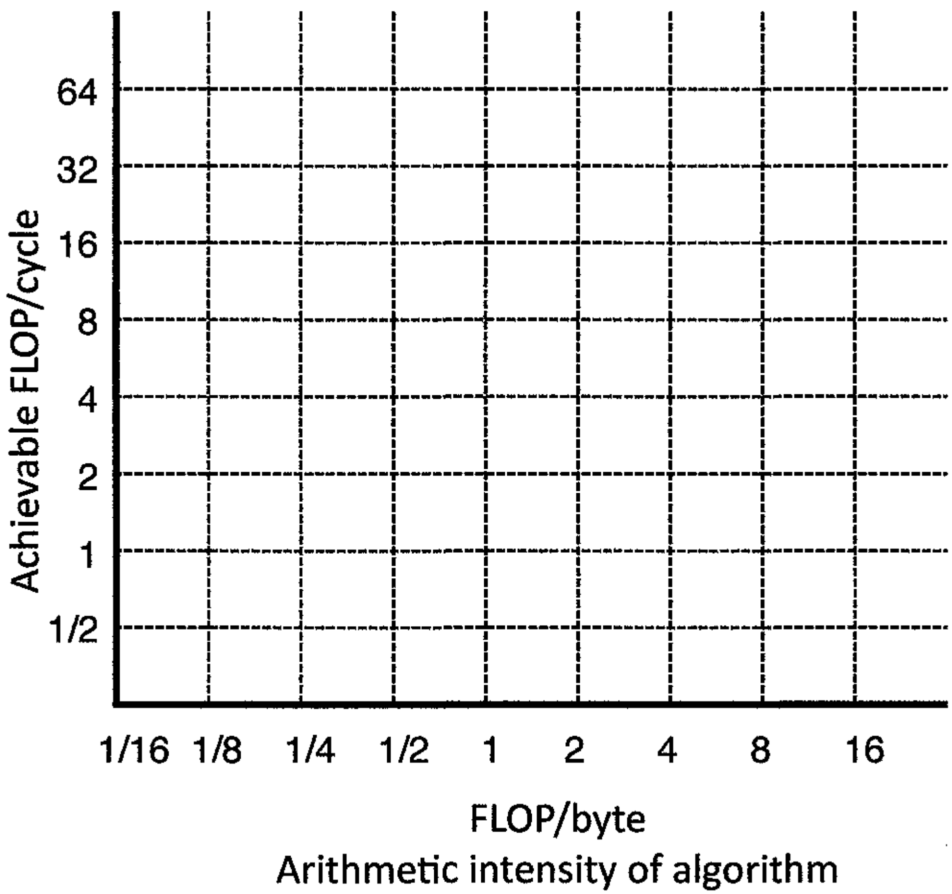


Given processor with:

1. 8 fully pipelined FP units
2. 8 byte/cycle memory BW
3. 4 byte floats

Plot the performance bounds on the graph

Performance Bounds



Given a processor with:

1. 8 fully pipelined FP units
2. 8 byte/cycle memory BW
3. 4 byte floats

What's the performance bound on the SAXPY loop below?

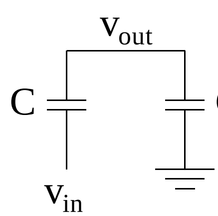
1. X and Y are in main memory

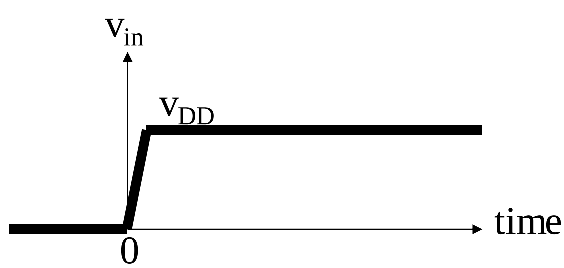
2. C loop:
```
for (i=0; i < 100,000; i++)
    Y(i) = a*X(i) + Y(i);
```

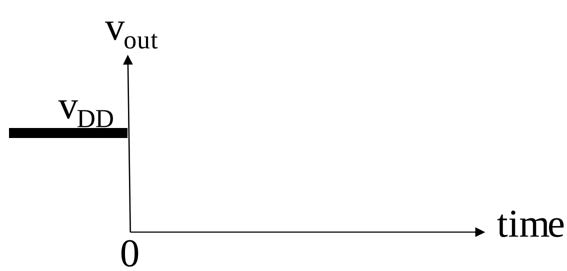
```
foo:    LF      F2, 0 (R1) // load X(i)
        MULTF   F4, F2, F0 // multiply a*X(i)
        LF      F6, 0 (R2) // load Y(i)
        ADDF    F6, F4, F6 // add a*X(i) + Y(i)
        SF      0 (R2), F6 // store Y(i)
        ADDI    R1, R1, #4 // increment X index
        ADDI    R2, R2, #4 // increment Y index
        SGTI    R3, R1, #100000// test if done
        BEQZ    R3, foo    // loop if not done
```

2010 Qualifying Exam
Simon Wong

1. Sketch V_{out}.
   Ans: final $V_{out} = 1.5\ V_{DD}$
2. What is initial total stored energy ? What is final total stored energy ?
   Ans: initial $E = CV_{DD}^2$; final $E = 1.25CV_{DD}^2$
3. If C can be varied, what is the maximum final V_{out} ?
   Ans: $C_{left} >> C_{right}$; maximum final $V_{out} = 2\ V_{DD}$
4. Modify the circuit to achieve final $V_{out} = 3\ V_{DD}$.
   Many possible solutions including the following :
   $C_{left} >> C_{right}$; insert a diode between the 2 C, drive the bottom plate of the right C to V_{DD}.







1. You have access to a cheap analog low-pass filter which has a fairly flat unit gain response from $\Omega=0$ to $\Omega=60\ KHz$, and a stopband attenuation of $100\ dB$ for $\Omega>80\ KHz$. With this analog anti-aliasing filter you are asked to design a 16-bit A/D converter system with a baseband sampling rate at $60\ KHz$.

a) If you were to use the above analog filter for the design of an oversampling A/D converter system, what is the **minimum** sampling frequency that you should choose?

b) With the sampling rate that you picked in part (a), draw a block diagram of the A/D converter system from the analog input signal to the final sampled output sequence at $60\ KHz$. This block diagram needs to include the specification (passband bandwidth, stopband attenuation, gain, etc.) of any filter that would be needed in the system.

c) Is your answer in part (b) the most hardware-efficient solution? If not, use another sampling rate to design this A/D converter system with less amount of computation. Draw a block diagram to describe this system.

a): The minimum sampling rate is 30Khz + 80Khz = 110 Khz

b): The block diagram consists of the above anti-aliasing filter, a sampler at 110Khz, followed by an up-converter by a factor of 6, a digital filter, and a down-converter by a factor of 11. The digital filter specs are: passband, $\pi/11$; stopband attenuation: 100dB, gain: 6.

c): 120Khz.

2. Indicate whether the following statements are true or false. If the statement is true, give a brief justification. If the statement is false, give a simple counter example or a clear reason.

a) If a real-coefficient digital filter has a zero-phase frequency response, then it must be a non-causal filter.

True.

b) All periodic continuous-time signals will remain periodic after sampling.

False. The sampled signal is periodic only if the ratio of the period of the original signal and the sampling period is a rational number.

c) If a z-transform doesn't have a region of convergence on the Z-plane, then its time-domain sequence doesn't exist.

False. Many time-domain sequences don't have a ROC in their Z-transform.

d) The sum of the impulse responses of two minimum-phase filters is always the impulse response of another minimum-phase filter.

False. The sum of the impulses responses will have a transfer function that is the sum of the transfer functions of the two original impulse responses. The poles of this transfer function will remain the same. The zeros of this transfer function, however, will not be the same and may move out of the unit circle.

e) If N samples of the discrete-time Fourier transform of a discrete-time sequence $h(n)$ are taken at $2\pi k/N$, where $k=0,\ldots,N-1$, then this set of samples represents the N-point discrete-Fourier transform of $h(n)$.

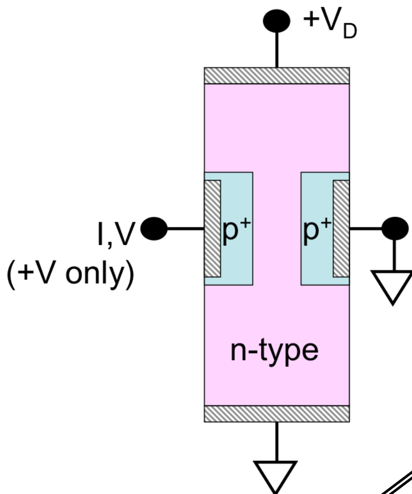
False. The N samples of the DTFT of $h(n)$ are the DFT of the "aliased" $h(n)$.

Fabian Pease

For what was the most recent Nobel Prize in Physics awarded?

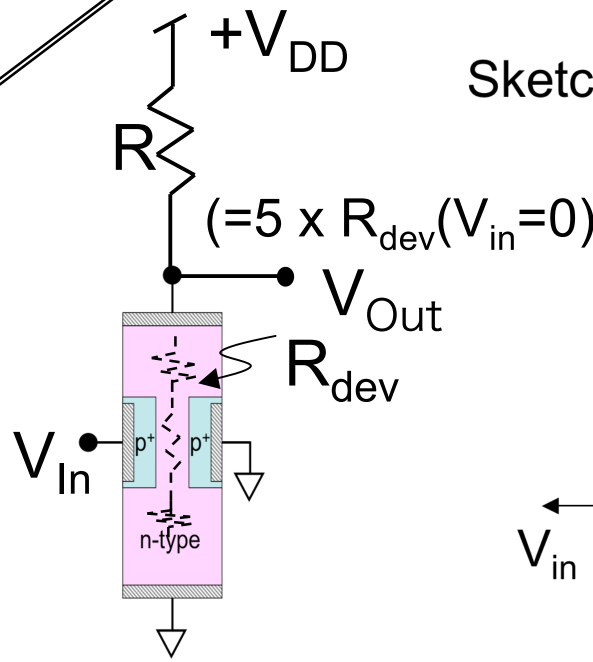
a) How does the image sensor in your digital camera work? (if you don't know, invent one)
b) Why, do you think, the CCD won the prize when other devices (Solar cell, DRAM, SRAM, Flash memory, CPU-chip) did not?
c) Why, do you think, has the CMOS sensor largely replaced the CCD in digital cameras?

Part A)

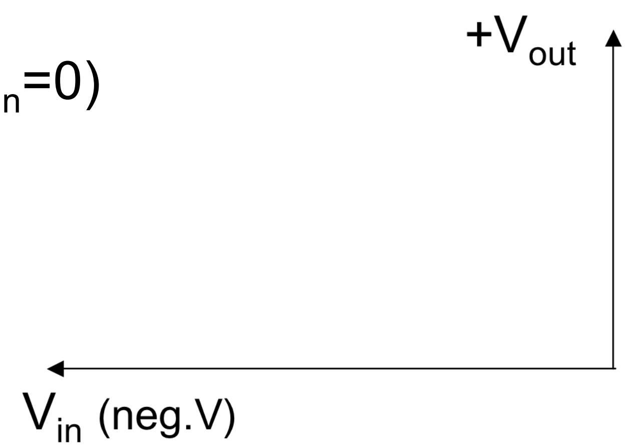


Sketch the I, +V for a fixed $V_D>0$

Part B)



Sketch the V_{out} vs. V_{in} ($V_{in}<0$)



After discussion of Parts A) and B) there were follow-on questions (not included here) dealing with different terminal connections and bias conditions for Part A) and how to “optimized” the device performance in Part B)

$\boxed{\text{January 2010}}$

The questions are colored red.
Solutions to R.M. Gray's 2010 qualifying exam problem.

The following system is useful as a model in pulse amplitude modulation (PAM) systems and digital-to-analog (D/A) converters:

$$x[n] \longrightarrow \boxed{\mathcal{S}} \longrightarrow y(t) = \sum_{n=-\infty}^{\infty} x[n]p(t-n)$$

integer n real t

where $p(t)$ is a real-valued continuous-time signal satisfying

$$\int_{-\infty}^{\infty} p(t)p(t-n)dt = \delta_n = \begin{cases} 1 & n = 0 \\ 0 & \text{all nonzero integers} \end{cases} \tag{1}$$

Define the discrete-time Fourier transform (DTFT) of a signal $x[n]$ by

$$X(f) = \sum_{n=-\infty}^{\infty} x[n]e^{-j2\pi fn}; -\frac{1}{2} \leq f \leq \frac{1}{2}$$

and the continuous-time Fourier transform (CTFT) of a signal $y(t)$

$$Y(f) = \int_{-\infty}^{\infty} y(t)e^{-j2\pi ft}dt; -\infty < f < \infty,$$

where $j = \sqrt{-1}$.

First Question:
Find a *simple* relationship between $Y(f)$ and $X(f)$.

Solution This was intended as a straightforward start using standard Fourier proof techniques — substitute (plug in) the definition of the signal to the definition of the transform $Y(f)$ asked for, interchange the order of summation and integral, and then simplify.

$$\begin{aligned} Y(f) &= \int_{-\infty}^{\infty} y(t)e^{-j2\pi ft}dt = \int_{-\infty}^{\infty}\left[\sum_{n=-\infty}^{\infty} x[n]p(t-n)\right]e^{-j2\pi ft}dt \\ &= \sum_{n=-\infty}^{\infty} x[n]\int_{-\infty}^{\infty} p(t-n)e^{-j2\pi ft}dt = \sum_{n=-\infty}^{\infty} x[n]P(f)e^{-j2\pi fn} \end{aligned}$$

where the last step is the usual Fourier shift theorem for CT signals (or just change variables in the integral). Thus

$$Y(f) = X(f)P(f).$$

A tricky point here is that $Y(f)$ should be defined for all real f, but $X(f)$ was defined only for f in $[-1/2, 1/2]$. The formula makes sense, however, if we take $X(f)$ to be the periodic extension, that is, just use the sum in the DTFT definition for all real f.

Many people complicated the problem by trying to convert the DT signal into a CT signal using impulse trains. This way leads to the answer, but it makes things much more complicated. Some people observed correctly that the left hand side resembles a convolution and tried to quote the convolution theorem, but here the "convolution" is discrete time while the output signal is continuous time, so the usual convolution theorems (for DT or CT) do not directly apply.

Recall that $p(t)$ is real and

$$\begin{aligned}\int_{-\infty}^{\infty} p(t)p(t-n)dt &= \delta_n \qquad (1)\\ P(f) &= \int_{-\infty}^{\infty} p(t)e^{-j2\pi ft}dt\end{aligned}$$

Second Question:
Find a *simple* expression for

$$\int_{-\infty}^{\infty} |P(f)|^2 e^{j2\pi fn}df$$

Solution There are *many* ways to do this problem.
The most straightforward approach is the standard Fourier proof method of substitution and interchanging order of integration.

$$\begin{aligned}\int_{-\infty}^{\infty} |P(f)|^2 e^{j2\pi fn}df &= \int_{-\infty}^{\infty} P(f)\left[\int_{-\infty}^{\infty} p(t)e^{-j2\pi ft}dt\right]^* e^{j2\pi fn}df\\ &= \int_{-\infty}^{\infty} p(t)\left[\int_{-\infty}^{\infty} P(f)e^{j2\pi f(t+n)}df\right]dt\\ &= \int_{-\infty}^{\infty} p(t)p(t+n)dt\end{aligned}$$

where we have used the Fourier inversion formula to recover p from P. From (1) the answer is

$$\int_{-\infty}^{\infty} |P(f)|^2 e^{j2\pi fn}df = \delta_n$$

Some people got bogged down by substituting the time domain intergral for both occurrences of $P(f)$, which is messier because of the triple integration. I tried to warn people who took a path that was likely to get tangled in details.
A shortcut to the answer is to recognize the integral as the continuous-time inverse Fourier transform of $|P(f)|^2$ evaluated at time n and that $|P(f)|^2$ is the transform of the CT autocorrelation of p,

$$r_p(\tau) = \int_{-\infty}^{\infty} p(t)p(t-\tau)dt$$

(from the correlation theorem for continuous time Fourier transforms), which when evaluated at an integer time yields the Kronecker delta δ_n from (1).
Equivalently, the integral asked for is the integral of the product of $P(f)$ and $P^*(f)e^{j2\pi fn} = \left(P(f)e^{-j2\pi fn}\right)^*$. From the generalized Parseval's theorem this is the integral in the time domain of the product of the inverse Fourier transforms of these signals, which are $p(t)$ and $p^*(t-n) = p(t-n)$, which from (1) is the Kronecker delta δ_n.
Several people tried another short cut that does not work. They correctly recognized (1) as an autocorrelation and reasoned that therefore if they transformed both sides the left hand side should be $|P(f)|^2$ (the transform of a correlation) and the right hand side should be 1 (the transform of a delta function), thus $|P(f)|^2 = 1$ for "all" f. But the correlation is a continuous time correlation, while the equation is a discrete time relation – the delta function is a Kronecker

delta (discrete time), not a Dirac delta (continuous time). Thus to transform both sides of (1) the appropriate transform is the DTFT and not CTFT, and one can not infer from (1) that the transform of the left hand side is a constant. One can not use the DT autocorrelation result (or Parseval's theorem) since a sum, not an integral, would be needed on the left.

Recall that

$$\begin{aligned}\int_{-\infty}^{\infty} p(t)p(t-n)dt &= \delta_n \quad (1)\\ P(f) &= \int_{-\infty}^{\infty} p(t)e^{-j2\pi ft}dt\end{aligned}$$

Third Question:
Find a waveform $p(t)$ which has the desired property (1) and has the additional property that $P(t)$ *also* satisfies property (1); that is,

$$\int_{-\infty}^{\infty} P(t)P(t-n)dt = \delta_n \quad (2)$$

Solution Here are two possible approaches:

1. Guess and show it works. There are not many signals p which satisfy (1), so it is easy to see if they also satisfy (2) if you either know or can find the CTFT.

2. Look at the formulas the signals must satisfy and find a solution.

First Approach: A simple signal which satisfies (1) is the box or rect function or square pulse

$$p_0(t) = \begin{cases} 1 & |t| \le \frac{1}{2} \\ 0 & \text{otherwise.} \end{cases}$$

Its Fourier transform is easily found (or recalled from memory) as

$$\begin{aligned} P_0(f) &= \int_{-1/2}^{1/2} e^{-j2\pi ft}dt = \left.\frac{e^{-j2\pi ft}}{-j2\pi ft}\right|_{-1/2}^{1/2} \\ &= \frac{e^{-j\pi f}}{-j\pi f} - \frac{e^{j\pi f}}{-j\pi f} = \frac{\sin(\pi f)}{\pi f} \end{aligned}$$

which is the sinc function, $P_0(f) = \text{sinc}(f)$. But these are also orthogonal as in (1) from Parseval's theorem and the modulation theorem:

$$\int_{-\infty}^{\infty} \text{sinc}(f-n)\,\text{sinc}(f)df = \int_{-\infty}^{\infty} p_0(t)e^{-j2\pi tn}p_0(t)dt = \int_{-1/2}^{1/2} e^{-j2\pi tn}dt = \delta_n.$$

Second Approach: Combining the Second Question with (2) yields

$$\int_{-\infty}^{\infty} |P(f)|^2 e^{j2\pi fn}df = \int_{-\infty}^{\infty} P(f)P(f-n)df = \delta_n$$

so $P(f)$ must satisfy

$$\int_{-\infty}^{\infty} P(f)\left[P(f)e^{j2\pi fn} - P(f-n)\right]df = 0.$$

A sufficient condition is that the bracketed term is itself 0 for all integer n. It is 0 for $n = 0$ and it will be 0 for all nozero integers if we pick

$$P(f) = P_1(f) = \begin{cases} 1 & |f| \leq \frac{1}{2} \\ 0 & \text{otherwise} \end{cases}.$$

Similar to the first approach, taking the (inverse) Fourier transform yields $p_1(t) = \text{sinc}(t)$.
Thus both the box function $p_0(t)$ and the sinc function $p_1(t) = \text{sinc}(t)$ have the desired properties. The signal $p(t)$ is time limited, but occupies an infinite bandwidth in the time domain (although most of the energy is between the first zeros of the sinc function), while the second has infinite extent in time, but is bandlimited to $[-1/2, 1/2]$.

Mark Horowitz

I have a clock in my office that uses an oscillating wand with LEDs on it that it uses to display the time. It contains a microcontroller that runs the clock.

1. How would you use the controller to control the freq and amplitude of the wand oscillator.

2. How do you sync the timing of the display to the position of the wand (where does the sensor need to be place).

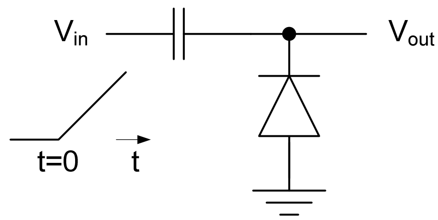
3. Please make the timing of the dots on the display all the same size.

1. How do you parallelize matrix multiplication?

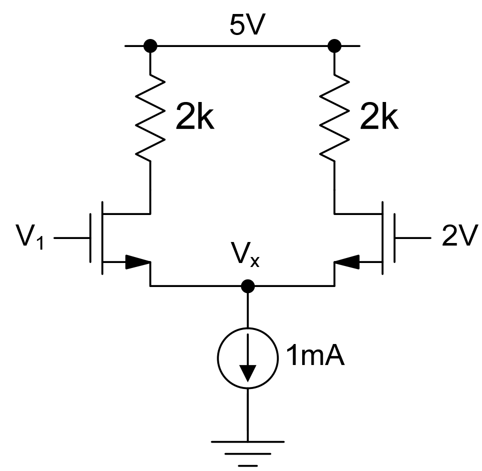
2. How do you parallelize a SOR (Successive Over-Relaxation) algorithm?

Name:	Stanford EE Quals 2010 Murmann

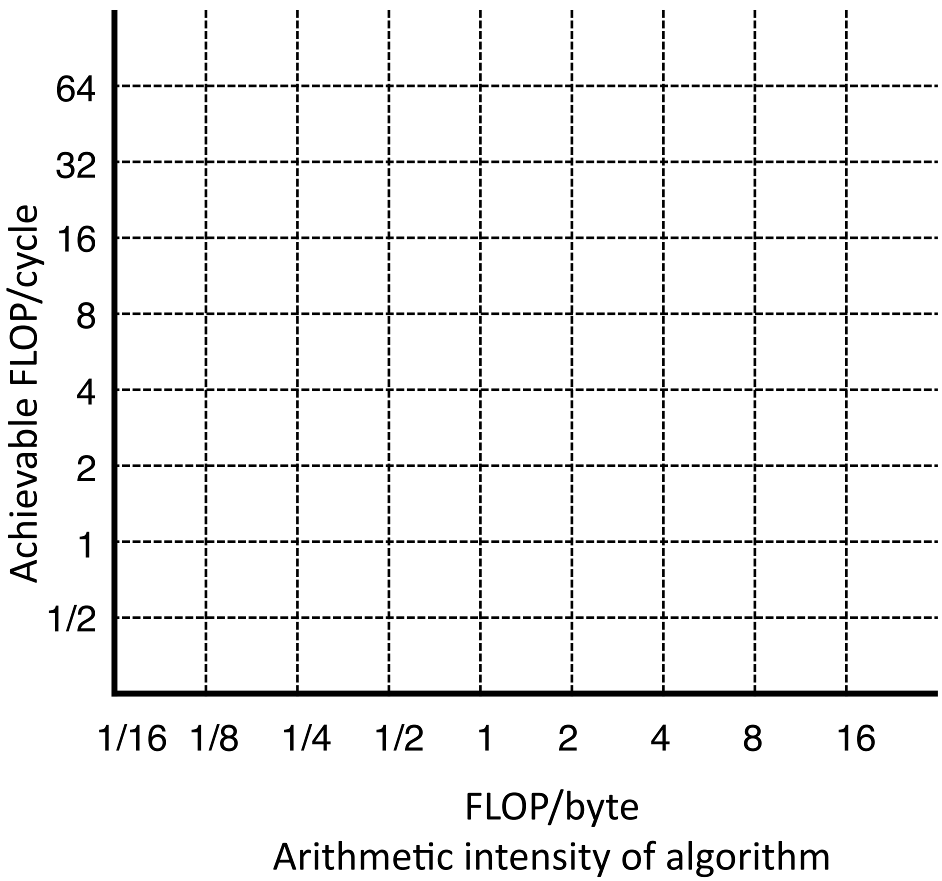
1. The circuit below is driven by a voltage ramp that increases linearly with time. Sketch (qualitatively) V_{out} (t), assuming $V_{out}(0)$ = 0V.



2. For the circuit below, sketch the voltage V_x as a function of V_1, ranging from 0…5V. The MOSFETs obey the ideal square law equation with $\mu C_{ox}W/L = 4mA/V^2$, and $V_t = 1V$.



Performance Bounds

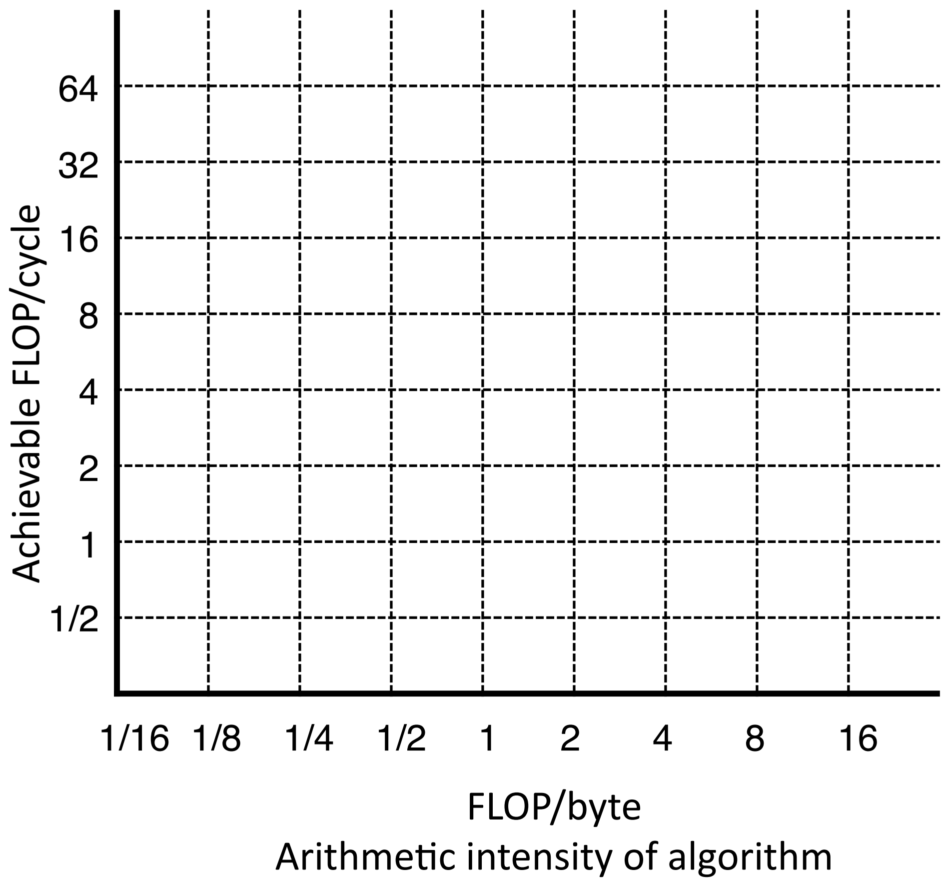


Given processor with:

1. 8 fully pipelined FP units
2. 8 byte/cycle memory BW
3. 4 byte floats

Plot the performance bounds on the graph

Performance Bounds



Given a processor with:

1. 8 fully pipelined FP units
2. 8 byte/cycle memory BW
3. 4 byte floats

What's the performance bound on the SAXPY loop below?

1. X and Y are in main memory

2. C loop:

```
for (i=0; i < 100,000; i++)
      Y(i) = a*X(i) + Y(i);
```

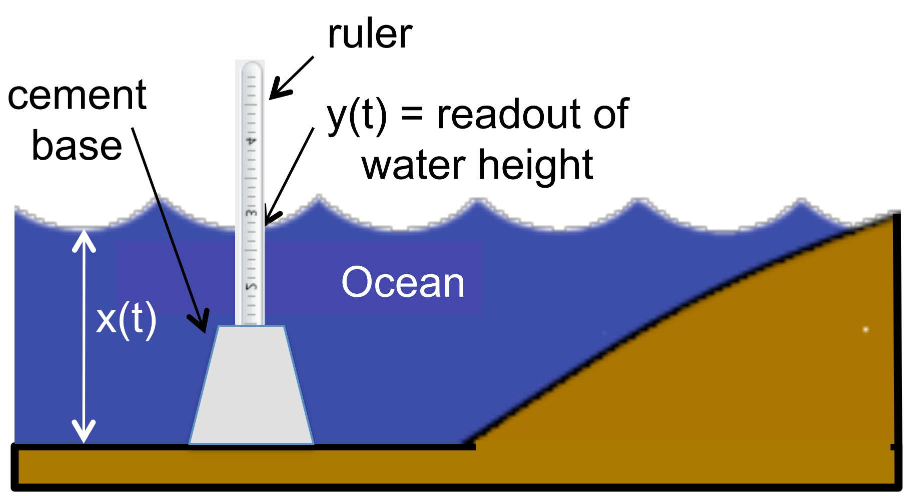
```
foo:    LF      F2, 0 (R1) // load X(i)
        MULTF   F4, F2, F0 // multiply a*X(i)
        LF      F6, 0 (R2) // load Y(i)
        ADDF    F6, F4, F6 // add a*X(i) + Y(i)
        SF      0 (R2), F6 // store Y(i)
        ADDI    R1, R1, #4 // increment X index
        ADDI    R2, R2, #4 // increment Y index
        SGTI    R3, R1, #100000// test if done
        BEQZ    R3, foo    // loop if not done
```

2010 EE PhD Quals

Prof. Daniel Spielman

Your goal is to build a system to measure the tidal fluctuations of the ocean.

Consider the following device:



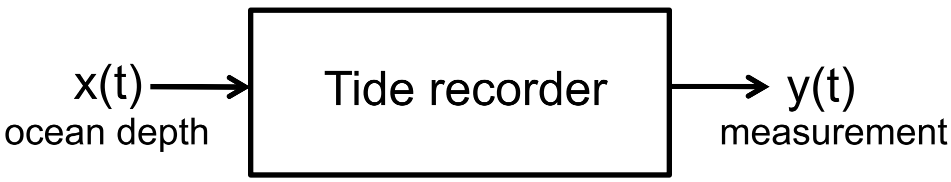
1. Is this device linear and time-invariant? If so, sketch the frequency response.

2. What are the advantages and disadvantages of the design?

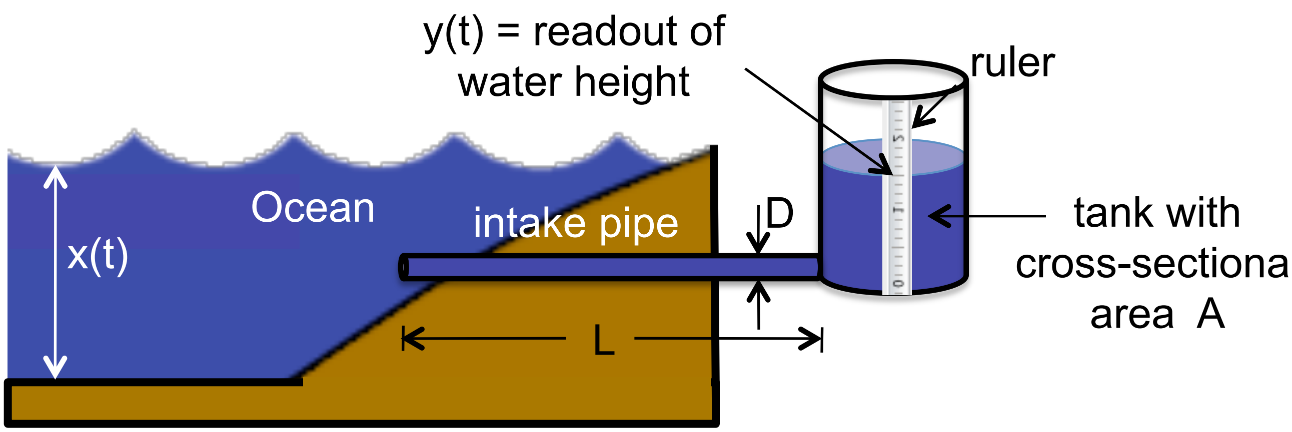
3. Can you design a better device?

2010 EE PhD Quals

Prof. Daniel Spielman



Consider the following alternative system:



4. Is this device linear and time-invariant? If so, sketch the frequency response.

5. Qualitatively describe the effects of:

   a) increasing A (cross-sectional area of the tank).

   b) increasing L (length of the intake pipe).

   c) increasing D (diameter of the intake pipe).

2010 EE PhD Quals: Solutions

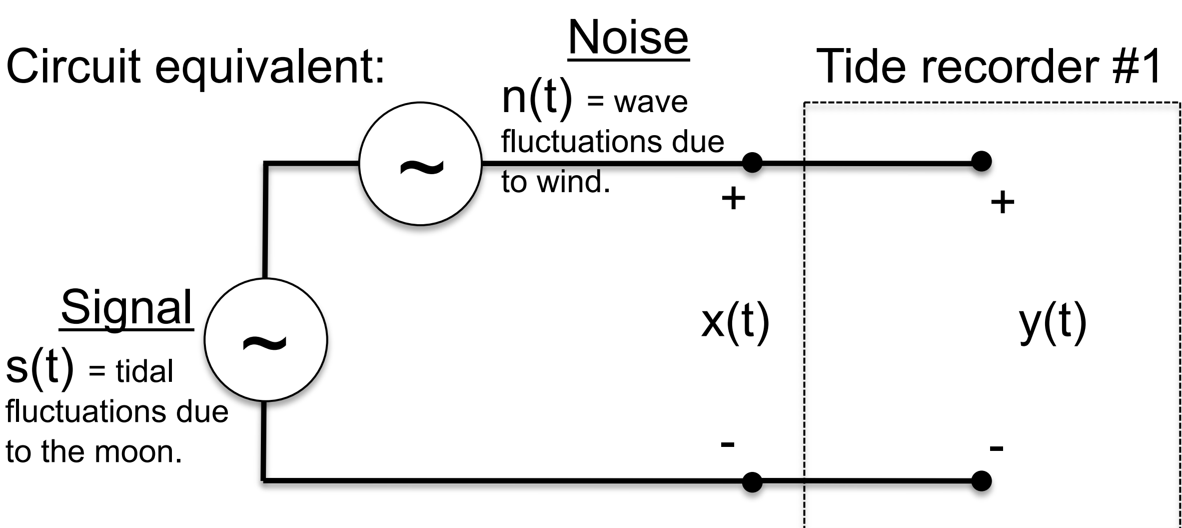
Prof. Daniel Spielman

1. For this system, y(t) = x(t), hence the device is linear and time-invariant. The frequency response is constant for all frequencies.

2. Advantages: inexpensive, easy to build, durable.

   Disadvantages: probably need a boat to actually make the measurements. More importantly, the data will be <u>noisy</u>. The measurements are subject to unwanted variations from surface wave fluctuations. An improved device would filter out this unwanted high frequency noise. Note: tidal fluctuations are on the order of 10^{-5} Hz, while wave action has a dominant component around 10^{-1} Hz.

Circuit equivalent:

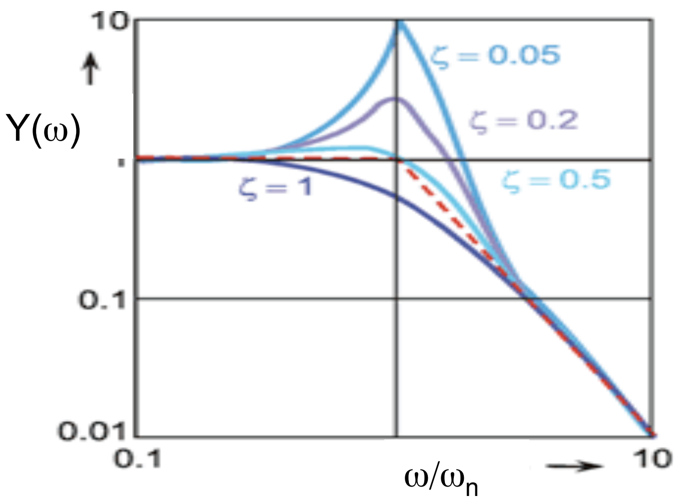


3. There are lots of choices for better measurement systems which can incorporate the desired <u>low-pass filtering</u>. I like the device shown in Question 4.

2010 EE PhD Quals: Solutions

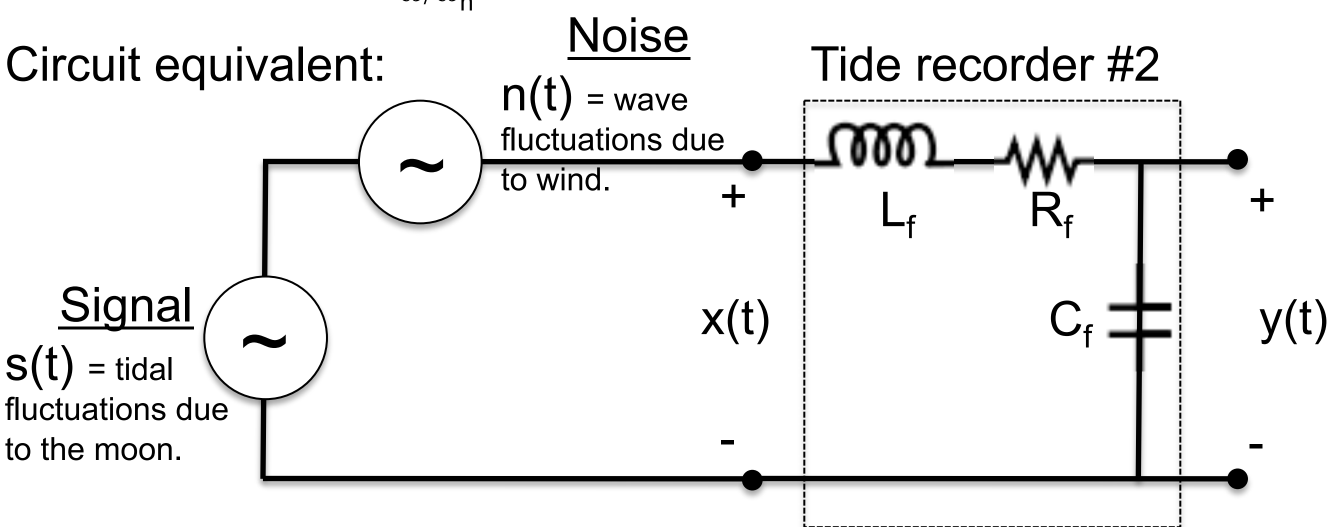
Prof. Daniel Spielman

4. Device #2 is a low pass filter (equivalent to an LRC circuit). The frequency response would look something like (depending on the particular choices of A, L, and D):



In choosing A, L, and D, we want to avoid an underdamped system with a natural frequency in the range of ocean surface waves, typically on of the order of 0.1 Hz ($\omega_n = 1/\text{sqrt}(L_f C_f)$ where L_f is the inertance [fluid analog of electrical inductance] and C_f is the fluid capacitance) . The damping ratio, ζ, equals $(R_f/2)\text{sqrt}(C_f/L_f)$, where R_f is the fluid resistance. We want surface wave fluctuations >> ω_n and tidal variations << ω_n.

Circuit equivalent:



5a. Increasing A increases the fluid capacitance, thereby decreasing the natural frequency ω_n.

5b. Increasing L increases the fluid inertance (thereby decreasing the natural frequency ω_n) and increases the fluid resistance (thereby increasing the damping ζ).

5c. Increasing D decreases the fluid inertance (thereby increasing the natural frequency ω_n) and decreases the fluid resistance (thereby decreasing the damping ζ).

Quals Question

Let the random variable X have CDF F. Suppose that X_i iid$\sim X$.

(a) What is the CDF of $\max_{1\le i\le n} X_i$?

(b) What is the CDF of $\min_{1\le i\le n} X_i$?

Suppose now that $X_{i,j}$ iid$\sim X$

(c) What is the CDF of $f(n,m) = \max_{1\le i\le n}\min_{1\le j\le m} X_{i,j}$?

(d) Suppose $X \sim$ exponential(λ). Let $Y_m = f\left(e^{\beta m}, m\right)$, for some parameter $\beta > 0$. What does Y_m converge to as $m \to \infty$? In what sense?

(e) Repeat the previous part for the general case $X \sim F$ (can assume that F is continuous and strictly increasing).

Qualifying Exam Questions

January 11 – 15, 2010

Yoshi Yamamoto

What are thermal noise and quantum noise in physical systems? You can choose any one of the following systems and describe the origins of those fluctuations.

1. **Simple macroscopic conductor**
2. **Mesoscopic conductor under ballistic regime**
3. **pn junction under either forward or reverse bias**
4. **Tunnel junction**
5. **Laser/maser**
6. **Parametric oscillator**
7. **Mechanical oscillator**
8. **Bose-Einstein condensation**

1. Assume that there is a lossless oscillating system whose physical property a oscillates over time with angular frequency ω. For example, you can assume that this system represents electromagnetic field in a resonator. (However, you can also think about a mass on a spring, or another oscillating system of your choice, if it is simpler for you.)

a) Write the equation that describes the behavior of a over time. What is its solution?

b) If the system is not lossless, how does the property a vary in time? How do you have to modify the equation from the part (a) to account for losses?

c) Now assume that such a lossless oscillator is coupled to another, identical lossless oscillator. Would the system still oscillate harmonically? If yes, at what frequency?

d) Could you write equations which describe the behavior of such a coupled system?

Ph.D. Quals Question
January 11-15, 2010
A.C. Fraser-Smith
Department of Electrical Engineering
Stanford University

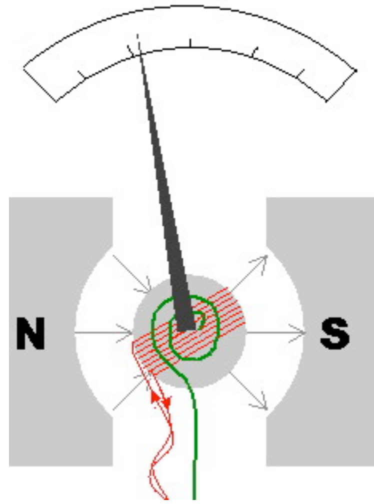
Multimeter

The figure below shows the analog multimeter that was shown to each student. Two leads were also on hand – sometimes inserted in the two sockets on the bottom left (in picture) – sometimes not, but terminating in two pointed probes. The multimeter was not plugged in to a power socket (it was quite clearly not intended to be plugged in) and it had no obvious power source. There was no easily removable panel on the back for a battery; in fact, the back was attached by two not very obvious screws. The students were told that the multimeter was about 40 years old and still worked perfectly, enabling its user to measure DC current, DC voltage and resistance. There were also a few switch settings for measuring AC voltage, but the student was told not to bother about AC. **First question**: Look inside the multimeter in the region where the pointer pivots and explain what electromagnetic principles and mechanical tricks are involved in making it work. One hint was given: Start with its measurement of current.

Simple analog multimeter

The answer to this question usually involved two steps: (i) identification of the key components of the device, and then, closely related to this first step, (ii) an explanation for how these components worked to give a measurement.

A number of students immediately recognized that they were dealing with a galvanometer; some even mentioned a D'Arsonval galvanometer, which was encouraging, but which did not necessarily lead to a correct explanation for how it worked! Although the complete mechanism surrounding the pivot of the pointer was not particularly easy to see, all its important features could be seen. With a little prodding from the examiner, if necessary, the student usually – but not always – ended up sketching out something approximating the mechanism shown below:



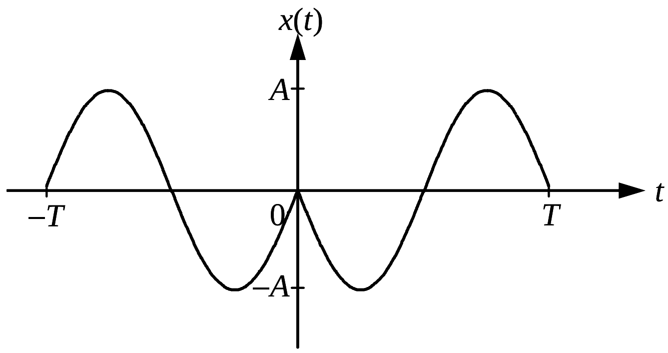
Sketch of a galvanometer mechanism. The red wires carry the current to be measured; it passes through a coil wound around the (steel) cylinder holding the pointer. The green object is a restoring spring. N and S indicate the two poles of a magnet.

Following identification of the components, the examiner looked for some discussion of the force on wires or coils carrying current in a magnetic field. He then looked for some discussion of the curved shape of the faces of the magnet, which combined with the pivoting steel (!) cylinder holding the pointer, would lead to a uniform magnetic field surrounding the coil, independent of its angular position. In other words, the torque on the coil would be dependent only on the strength of the current passing through the coil. The restraining spring would balance this torque and lead to the current reading.

The student was then given a **second question**: If the operation of the meter depended on current passing through the coil, how did it measure resistance? Around this time, as a hint, the examiner would switch the multimeter to a resistance setting and demonstrate how the pointer would move toward its zero setting (on the right side of the dial) when the two probes were touched. The knurled knob labeled “Ohms” in the figure above enabled the pointer to be moved exactly to its zero setting. At this time the better students would carefully, and sometimes not so carefully, inspect the multimeter and declare with authority that there had to be a battery inside, since some source of energy was required to move the pointer, even though there was no obvious way of inserting such battery. The examiner would confirm this fact and then again ask how resistance was measured. The answer: the preparatory step, touching the probes together and adjusting the resistance measurement to zero, established a full-scale deflection for a known internal resistance. When the unknown resistance is placed in series in the circuit the deflection is less than full scale due to the reduced current flowing and the calibrated scale can indicate the resistance.

Stanford University, Department of Electrical Engineering
Qualifying Examination, Winter 2009-10
Professor Joseph M. Kahn

A continuous-time signal $x(t)$, $-\infty < t < \infty$, has a Fourier transform $X(j\omega)$, $-\infty < \omega < \infty$.



Without computing $X(j\omega)$, answer the following:

a. What is $X(j0)$?

b. What is $\int_{-\infty}^{\infty} X(j\omega) e^{j\frac{\omega T}{4}} d\omega$?

c. What is $\int_{-\infty}^{\infty} |X(j\omega)|^2 d\omega$?

d. By what power of $|\omega|$ does $|X(j\omega)|$ decrease as $|\omega| \to \infty$?

Answer

a. $X(j0) = \int_{-\infty}^{\infty} x(t) e^{-j0t} dt = 0$.

b. $\int_{-\infty}^{\infty} X(j\omega) e^{j\frac{\omega T}{4}} d\omega = 2\pi x\left(\frac{T}{4}\right) = -2\pi A$.

c. $\int_{-\infty}^{\infty} \left|X(j\omega)\right|^2 d\omega = 2\pi \int_{-\infty}^{\infty} \left|x(t)\right|^2 dt = 2\pi \cdot 2 \cdot \frac{A^2 T}{2} = 2\pi A^2 T$.

d. Since $x(t)$ and $\frac{dx}{dt}$ do not have any impulses (delta functions), but $\frac{d^2 x}{dt^2}$ has impulses, $\left|X(j\omega)\right| \propto \left|\omega\right|^{-2}$ as $\left|\omega\right| \to \infty$.

EE Quals 2010 (Hardware)

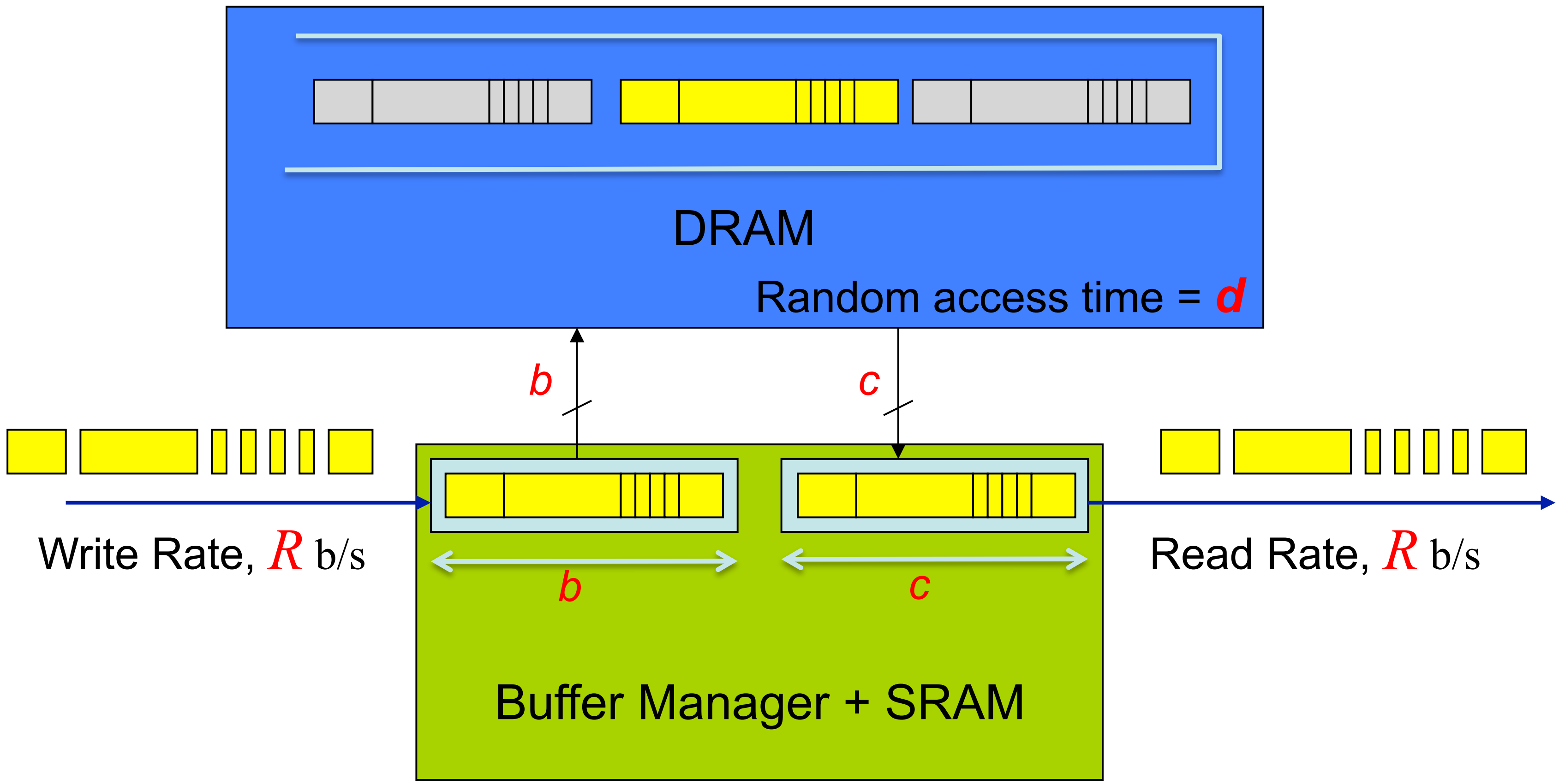
Nick McKeown

Question 1

(a) Why do we have a cache in a computer?

(b) I have an SRAM with random access time s, and a DRAM with random access time d. The probability of finding an entry in the SRAM is p. If I assume $d = 100s$, what value of p do I need so that the expected lookup time is twice as long as the random access time of the SRAM?

Single FIFO Queue



Question 2

Consider a cache for a FIFO queue in a network switch, router or network interface card, built from the same SRAM and DRAM.

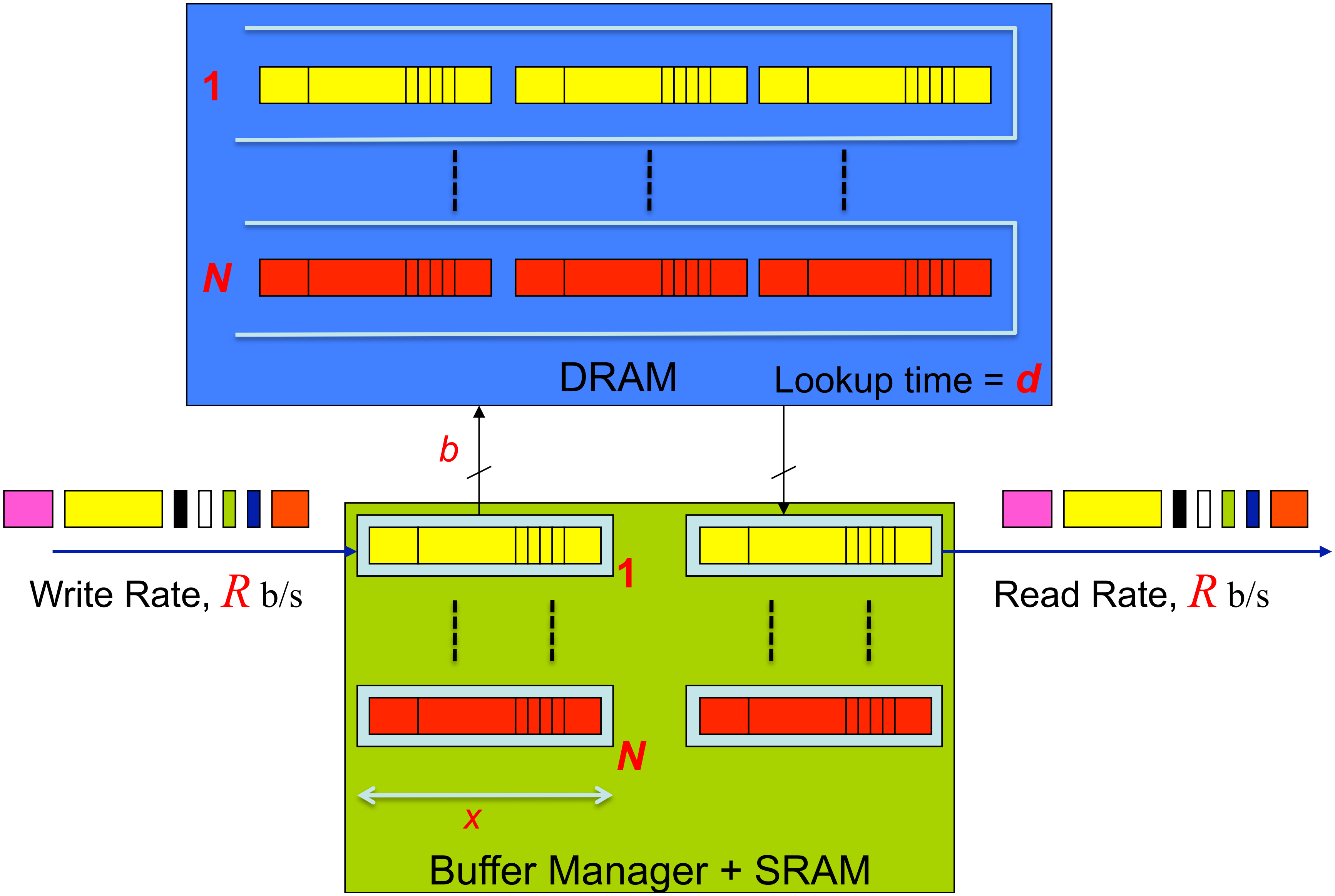
(a) Explain how it works.

(b) How large does the block size, *b*, need to be so that it won't overflow?

(c) How about *c*, so that the "head cache" won't underflow?

(d) What problems do we run into if we want to build a cache for *N* FIFO queues, instead of just 1?

Multiple FIFO Queues



Question 3

We want to figure out how large x needs to be, so the SRAM will never overflow.

(a) Is $x = b$ big enough? Explain.

(b) How can we figure out how large the SRAM needs to be?