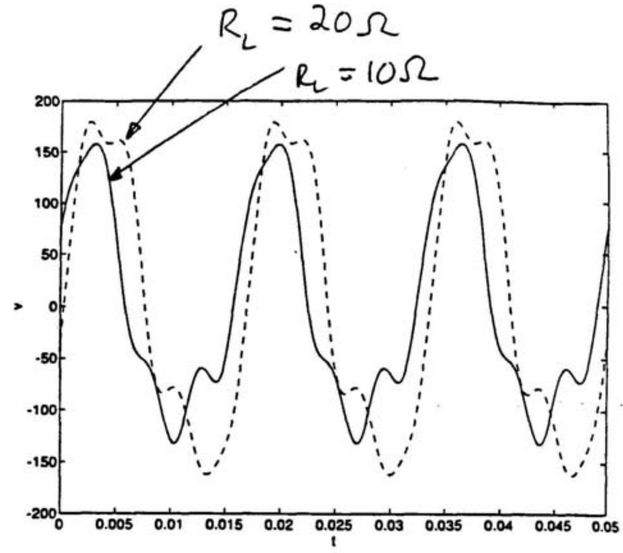
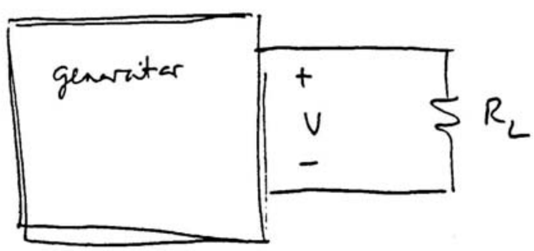
Electrical Engineering

Quals Questions

1994

Prof. Stenhen P. Boyd
January 1994





Can you predict what v will be for a 5Ω load?

Discuss

1994 Electrical Engineering Department PhD Qualifying Exam
Solid State Questions by Prof. C. Chang-Hasnain leave

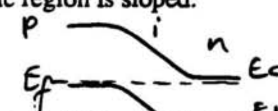
1. For a typical p-n diode with an abrupt doping profile of $N_A = 1 \times 10^{15}$ cm^{-3} for the p-side and $N_D = 1 \times 10^{16}$ cm^{-3} for the n-side, sketch
(a) the space charge distribution
(b) electric field distribution
(c) the energy band diagram indicating the conduction band edge E_C and the valence band edge E_V
(d) discuss what changes if the n-doping density is increased.

2. For a p-i-n diode with the same doping densities as above, discuss the difference between a p-i-n and a p-n diodes in
(a) space charge distribution
(b) electric field distribution and
(c) the energy band diagram.
(d) Discuss, as the thickness of the i-region increased, what happens to (a), (b) and (c).

3. (a) Draw the energy band diagram of a p-i-n-i-p-n$^+$ junction (assume some reasonable dimensions and doping densities).
(b) Explain what happens as the thickness of the middle n layer varies.
(c) Explain what key features one should expect from the I-V curve of this device.

Answers:
1. a,b,c can be found in any text book.
(d) As n doping is increased, the built in voltage is increased and the n-side depletion width is decreased. From the area under the E field, one should easily see that E_{max} should increase and p-side depletion width should increase. This problem does not require putting down equations, but it
does require good visual understanding of the three diagrams.

2. (a) and (b) are straight forward from prob. 1.
(c) There is a constant E field drop accross the intrinsic region and thus the band diagram of the intrnsic region is sloped.



(d) As i region thickness is increased, the depletion width on n and p sides both decrease and E_{max} is decreased. Again this can be seen directly from the figures.

3. This device is basically a pnpn thyristor. As the forward bias is increased, the reverse biased (middle junction) takes all the voltage and very little current flows through the entire device. As the voltage still increased, the reverse biased junction breaks down and there is a large increase in current flow. The large current flow changes the reverse-biased junction to forward-biased and the band diagram now looks like three serially connected forward junctions.

David

1994 EE Qualifier Questions Prof. Dill CS

Consider the context-free grammar:

```
expr -> expr + expr
expr -> expr * expr
expr -> ( expr )
expr -> NUM
```

Q: What's the first problem you would run into using YACC or another LALR(1) (or LR(1)) parser for this grammar?

A: It's ambiguous

Q: How would the problem be manifested?

A: shift-reduce conflict (e.g., on expr + expr ^ + 3, do we reduce expr + expr or shift +?)

Q: Assume the lexer provides the correct value for NUM ($1 in last production). Write actions next to the productions to compute the proper numerical value for the expression. You may assume that the parser implements the correct precedence.

A:

```
expr -> expr + expr  {$$ = $1 + $3;}
expr -> expr * expr  {$$ = $1 * $3;}
expr -> ( expr )     {$$ = $2;}
expr -> NUM          {$$ = $1;}
```

Q: Ok, now suppose we add the ability to bind a single variable, which is always "x".

```
expr -> letx = expr in expr
expr -> expr + expr
expr -> expr * expr
expr -> ( expr )
expr -> NUM
expr -> x
```

so 1 + (letx = 10 in x + x) + 3 = 24.

What YACC actions would be required to compute the value correctly? You may use one global variable, of type "integer", if you wish.

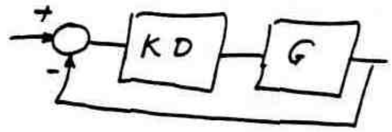
A: Let's use the variable "xval". The tricky bit is to save old xval on the value stack and restore later, in case of nested letx's.

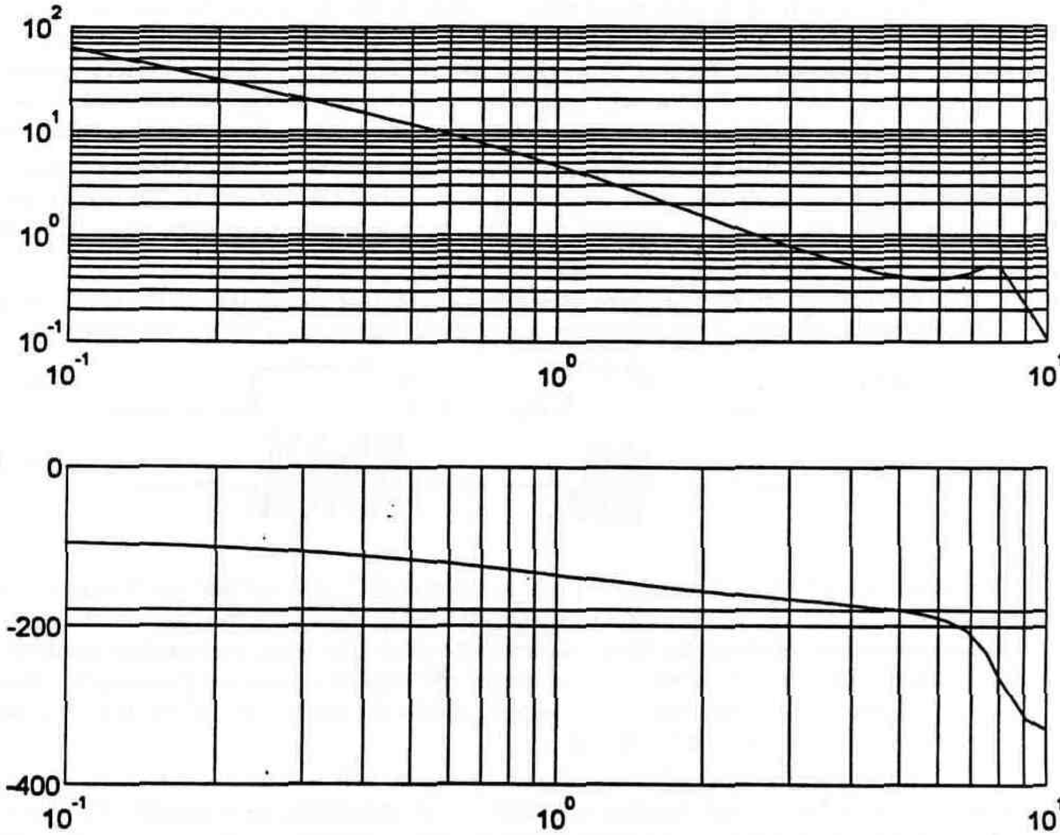
```
expr -> letx = expr in {$$ = xval; $$ = $3;} expr {xval = $5; $$ = $6;}
expr -> expr + expr      {$$ = $1 + $3;}
expr -> expr *  expr     {$$ = $1 * $3;}
expr -> ( expr )         {$$ = $2;}
expr -> NUM              same
expr -> x                {$$ = xval; }
```

Q: Ok, now do it without using any additional variables.

A: This is tricky. You can save and restore the value of "x", but it is hard to access when you need it in a different production. You can use $0 (which reaches below the current production in the stack), though. This requires copying the value whenever you have an expression that is not at the beginning of the right-hand side of a production.

```
expr -> letx = expr in {$$ = $3;} expr {$$ = $6;}
expr -> expr + {$$ = $0;} expr  {$$ = $1 + $4;}
expr -> expr * {$$ = $0;} expr  {$$ = $1 * $4;}
expr -> ( {$$ = $0;} expr )     {$$ = $3;}
expr -> NUM                {$$ = $1; }
expr -> x                  {$$ = $0; }
```


+
-
KD
G


10^2
10^1
10^0
10^-1
10^-1
10^0
10^1
0
-200
-400
10^-1
10^0
10^1

OFFICE MEMORANDUM ◊ STAR LABORATORY

February 3, 1994

To: Gene Franklin/Diane Shankle

From: Tony Fraser-Smith

Subject: Ph.D. Quals Question, 1994

Electromagnetics

$\beta = \sqrt{\frac{\omega\mu\sigma}{2}}$

$v_{ph} = \frac{\omega}{\beta} = \sqrt{\frac{2\omega}{\mu\sigma}}$

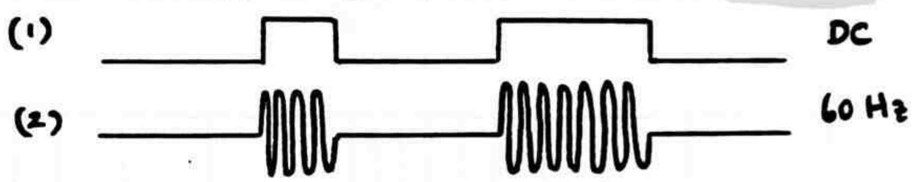
$v_g = \frac{d\omega}{d\beta} = \sqrt{\frac{8\omega}{\mu\sigma}}$

Following a brief review of the student's previous work on the propagation of electromagnetic fields in conducting media at low frequencies, in which it is established that both the attenuation and velocity (actually both the phase and group velocities) increase with increasing frequency, the following question is asked:

Question: You have gone down a deep mine and an accident has occurred in which a large amount of the overlying rock has filled up the elevator shaft and cut the telephone link with the surface. You want to send some kind of signal to the surface to let the rescuers know you are still alive. You look around and find (1) a considerable length of telephone wire strung along the side of the tunnel in which you are located, (2) a large battery (powering an emergency light), and (3) a light bulb that is still lit, indicating that some 60 Hz power is still available. What is the best way to send a signal to the surface with this equipment?

Answer: The wire first has to be arranged into some kind of antenna. Obviously doubling the wire back on itself and then connecting it to either the battery or the 60 Hz power will be ineffective, but a loop antenna or a straight dipole antenna configuration (the latter preferably with earthed ends) should be adequate.

The two ends of the wire can then be connected to either the battery or the 60 Hz supply. The student is advised to try generating a dot/dash pattern as is used in Morse code:



Concentrating first on the frequency content of the "DC" and "60-Hz" dot/dash patterns, we notice that the DC dot converts to a $\boxed{\text{sinc}}$ pattern centered on a frequency of 0 Hz in the frequency domain, whereas the 60-Hz dot converts to an equivalent sinc pattern centered on 60 Hz. Knowing that the attenuation increases with frequency, and not knowing the depth, or the conductivity, of the conducting material above the tunnel, we decide that it is safer to work with the DC dot/dash pattern.

Concentrating now on the DC dot/dash pattern, we first notice that the sharp edges are associated with the highest frequencies, which will be attenuated most rapidly. The dot and the dash will be converted to rounded pulses as they propagate upwards toward the earth's surface. In addition, since the dash is longer and thus has lower frequencies associated with it, it will probably reach the surface with a somewhat larger amplitude than the dot (assuming the dot and dash initial amplitudes are equal). $\boxed{\text{Dispersion}}$ also needs to be taken into account. The dot and the dash share some frequencies, which will reach the earth's surface at the same time. However, the dash has a lower average frequency content than the dot, so its shape will become distorted in such a way that the dot, which is assumed to be following the dash, will tend to catch up with it as time progresses. To keep the dot and dash clearly separate, their spacing needs to be increased.

>From hector@db.stanford.edu Fri Feb 4 18:37:56 1994
Return-Path: <hector@db.stanford.edu>
-0800
Received: by Coke.Stanford.EDU (5.57/25-DB-eef) id AA10328; Fri, 4 Feb 94 18:37:50 -0800
Date: Fri, 4 Feb 94 18:37:50 -0800
From: Hector Garcia-Molina <hector@db.stanford.edu> CS
Message-Id: <9402050237.AA10328@Coke.Stanford.EDU>
To: gray@isl.stanford.edu
Subject: My Qual Question
Cc: eileen@shasta.stanford.edu, hector@db.stanford.edu
Status: R

Here is a short description of my Qual question.
Sorry I am late, I was busy finishing a big NSF Digital Libraries proposal...

hector

Hector Garcia-Molina
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Stanford, California 94305-2140 USA
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FAX: (415) 725-2588
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Consider a calendar management application running on a distributed computing system with four computers, A, B, C, and D. Each computer has a copy of the calendar file.

[?]e computers can become disconnected (e.g., they are mobile the network may partition). To avoid having more than one group of computers updating the calendar file, we can use the following scheme:
Computer A gets 2 votes; the rest one vote each.
If a group has a majority of votes (in this case 3 out of 5), then it can update the file. For example, A and B have a total of 3 votes, so they can perform an update. If B and C from a group, they only have 2 votes, so they cannot update.

This scheme can also be represented by listing the set of *minimum* groups that can update the calendar file. For the above case, we have
S = { {A,B}, {A, C}, {A,D}, {B,C,D} }

PART I:
Find a vote assignment that is equivalent to the set of minimal groups, for this five computer system:
S = { {A,B}, {A,C,D}, {A,C,E}, {A,D,E}, {B,C,D}, {B,C,E}, {B,D,E} }

A	B	C	D	E
2	2	1	1	1

PART II:
Consider the following two vote assignments:
ASSIGNMENT 1: votes(A) = 1; votes(B) = 1; votes(C) = 1; votes(D) = 1
ASSIGNMENT 2: votes(A) = 2; votes(B) = 1; votes(C) = 1; votes(D) = 1
Describe how the behavior of these assignments differs.
Which one maximizes the probability that there exists an operational group (with a majority)?

$$\frac{C_4^3 + C_4^4}{C_4^1 + \cdots + C_4^4} = \frac{4+1}{2^4-1} = \frac{5}{15}$$

$$\frac{C_3^1 + C_4^3 + C_4^4}{15} = \frac{7}{15}$$

A B
A C
A D
B C D

1994 Quals Questions
J. S. Harris Devices
James

1. The original transistor was a bipolar transistor and this was the dominant transistor technology until the the mid '60s when the MOSFET became the dominant technology. What were the advantages of the MOSFET and its associated technology?

A. Lower power, simpler fabrication technology, natural device isolation, higher integration levels, high input impedance and no gate leakage, charge storage, complementary architecture.

2. What areas has the bipolar transistor remained an important technology?

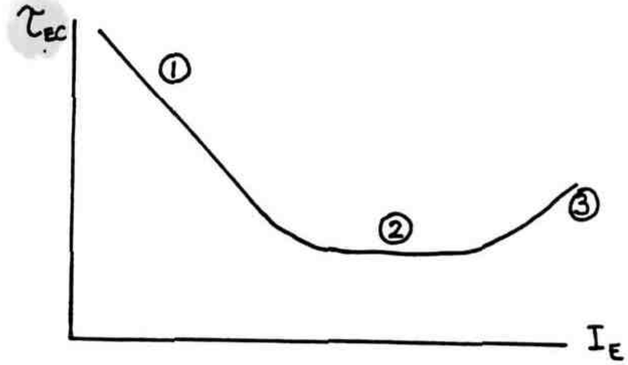
A. Analog and A/D circuits, microwave and high speed digital circuits, ? ← BiCMOS circuits, high power devices.

3. Why has the bipolar remained faster?

A. It has been much easier to control the base width to very small dimensions by ion implantation and epitaxy than the source drain spacing for MOSFETs by lithography. As devices are scaled to very small dimensions, parasitic capacitances no longer scale and the high current drive capability and lower voltage operation of bipolar transistors enables them to operate at higher speeds.

4. The high speed performance of a bipolar transistor varies with emitter ? current as follows: How is this related to the physics of the device operation?

not know answers.



To: "Diane J. Shankle" <shankle@sierra.stanford.edu>
Subject: Re: Quals Questions
In-Reply-To: Your message of Mon, 14 Feb 94 16:02:10 -0800.
<CMM.0.90.0.761270530.shankle@Sierra.Stanford.EDU>
Date: Tue, 15 Feb 94 10:21:42 PST
From: Mark Horowitz <horowitz@chroma.Stanford.EDU>

Here is my quals question:

Area: Digital Design

Question:

You need to design a priority encoder. This is a block that has n inputs (where n is around 32) and n outputs. One only output is allowed to be high at anytime, and should correspond to the highest priority input that is asserted. Assume that the highest priority input is connected to the MSB of the encoder, and the lowest priority input is connected to the LSB.

Examples of a 4 bit priority encoder:

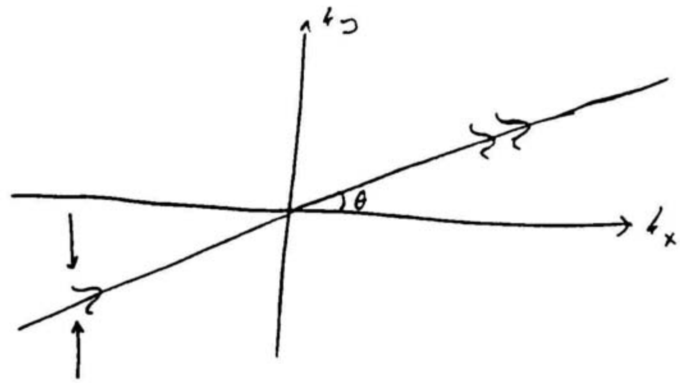
In	1111	0101	0011	0000
Out	1000	0100	0010	0000

How would you design this circuit?

Answer:

There is no right answer. I am more interested in the process of getting a solution. Most often people come up with a single gate for each output. The fanin of this gate grows with the number of inputs, so this is not a great answer. There is also a ripple-carry like solution which is very dense, but slow. Good students go through both of these, and then synthesize a solution is like a ripple-carry, but is faster using some block based approach.

R>



a sinc profile

line still has infinite length here (in F. domain

has sinc profile cross-sections parallel to

k_y (and k_x) axis.

physics

Prof. Leonid Kazovsky

Ph. D. Quals. January 1994.

You have constructed a new laser shown in Fig. 1 below. When the voltage at the modulator input is 0 V, the laser emits a single-frequency optical signal at the wavelength 1.5 μm.

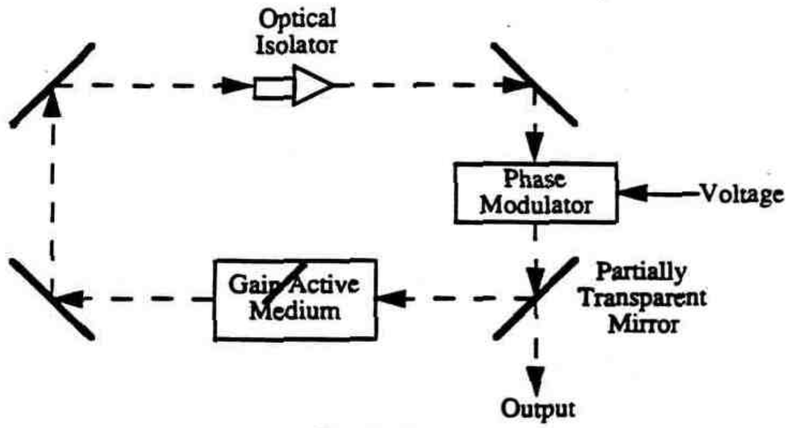


Fig.1. Laser.

Now, you apply to the modulator a cosine voltage at the frequency 1 MHz, and construct a measurement experiment shown in Fig. 2.

Sketch the optical spectrum (output of the optical spectrum analyzer in Fig. 2) and the electrical spectrum (output of the electrical spectrum analyzer in Fig. 2) of the laser, assuming that all the components in Figs. 1 and 2 are ideal (i. .e., the modulator changes the light phase only and has zero voltage-independent loss, the active medium bandwidth is much larger than 1 MHz, etc).

Will the two spectra depend on the amplitude of the voltage applied? If yes, how?

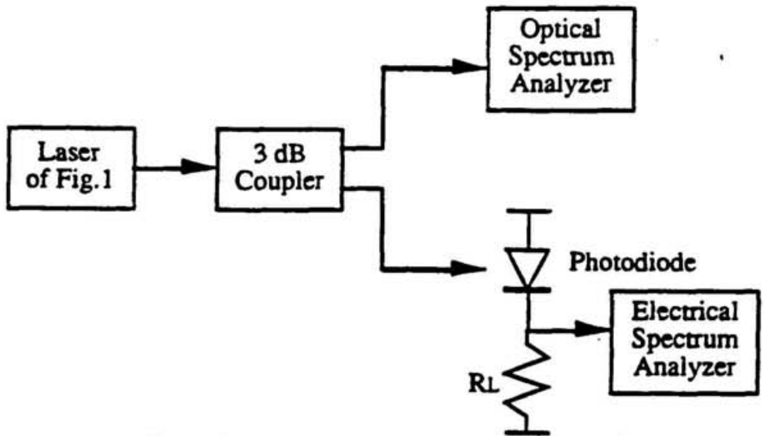


Fig. 2. Measurement Experiment.

From: "Butrus T. Khuri-Yakub" <khuri-ya@sierra.Stanford.EDU>
To: "Diane J. Shankle" <shankle@sierra.Stanford.EDU>
Subject: Re: LATE QUALS QUESTIONS
In-Reply-To: Your message of Tue, 22 Feb 94 14:55:48 PST
Message-ID: <CMM.0.90.0.761962420.khuri-ya@Sierra.Stanford.EDU>

Diane:

The following was my quals question:

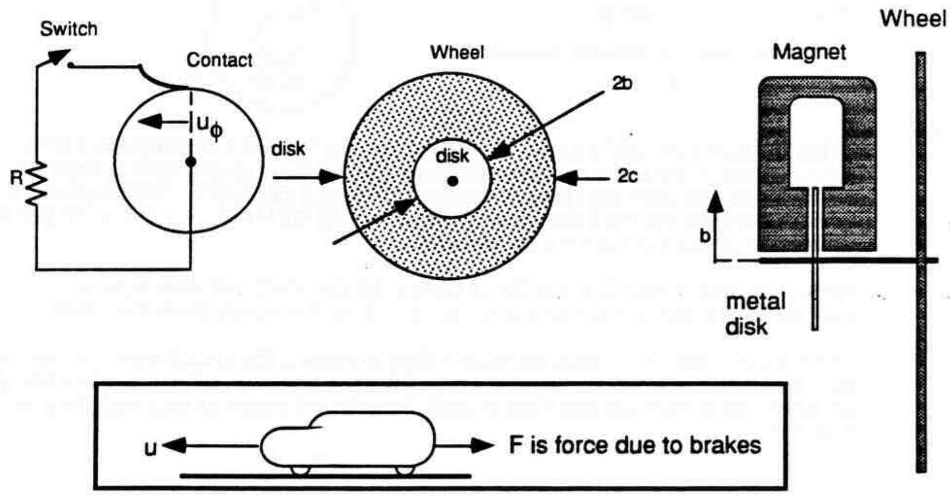
Find the equivalent circuit of a quarter wave section of a transmission line terminated with a low impedance load.

Khuri-Yakub physics

Rough- typed by Kino himself

Ph.D Qualifying Exam. Question by Prof. G. S. Kino

Induced Potentials by a Magnetic Field, Forces and Conservation of Power.



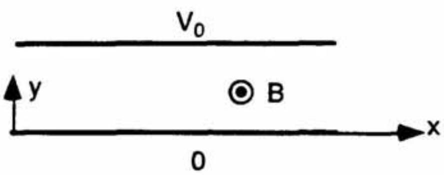
A magnetic brake rather than a standard frictional brake is to be used on a car. This consists of a rotating metal disk is placed between the pole pieces of a magnet, which provides a uniform magnetic field B bcross the disk. As the disk rotates, a voltage is induced across it which causes a current to flow in an external circuit when the brake switch is thrown. The current in the external circuit can be adjusted, and hence the braking force changed by varying the resistor R . Take the angular velocity of the wheel to be ω , the velocity of the car to be u, and assume that the radius of the disk is b.

Determine the power dissipated by the braking force F against the direction of movement of the car, and use conservation of power to find the braking force.

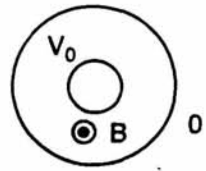
The student is expected to understand the expressions for mechanical power and electrical power and know how to deal with a rotating disk in a magnetic field.

Alternative Question by Prof. G. S. Kino

PLANAR MAGNETRON



MODEL OF VAC-ION PUMP (CIRCULAR MAGNETRON)



When the magnetic field is small, an electron leaving the lower electrode with zero velocity moves straight to the upper electrode. When the magnetic field is large the electron may not reach the upper electrode of the planar magnetron. Why in physical terms? Work out the condition for an electron leaving the lower electrode of the planar magnetron to just not reach the top electrode.

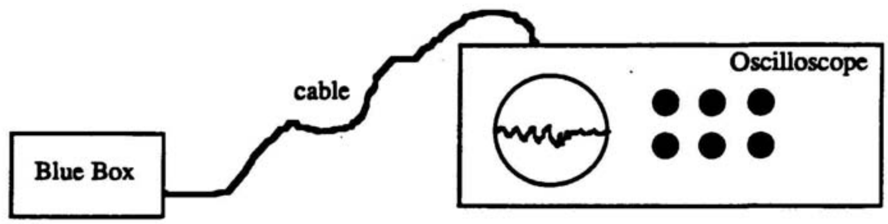
Need to write down the Lorentz Eq. of motion, be able to do this in cartesian coordinates, be able to spot the integration of $\ddot{x}$ *and integrate (with some hints).*

Then, if time, talk about what happens if there is a gas in the system (circular version shown). I state that this works as a pump if the electrodes absorb gas atoms when they hit them. What happens, how does it work, how does the current vary with the gas pressure?

G. Kovacs' QUALS QUESTION JAN 1994

Circuits

The student was asked to sit down in front of an oscilloscope displaying a signal coming from a blue plastic box in front of it. A brief, but carefully-prepared set of instructions was read, instructing the student that they were to focus on the blue box, that playing with the oscilloscope controls was allowed but would only waste time (they were preset correctly) and that without "smashing, disassembling or setting fire to" the blue box, they should tell me as much as possible about its contents.



Time was allowed for the student to study the situation and if they had not touched the box within a couple of minutes, they were prompted to do so. Along the way, they were asked to describe the waveform seen on the oscilloscope when the box was not being moved (noise) and whether or not they could tell if the circuit within the blue box was active or passive based only on the signal displayed (the approach to answering the latter question was more important than the answer -> active).

Once the student had played with the device, they were asked to try to describe its contents and, if needed, were given hints to describe the quantity the box might be measuring (it contained an accelerometer). The student's approach to the problem was most important.

Once the student had established that it was an accelerometer of some type, they were asked to determine its sensitivity in volts per "g" (unit of gravity). Various solutions were possible, such as observing that the sensor had a DC response, so turning it upside down causes a variation of a known acceleration, etc., etc.

The student was then asked to estimate the noise level and give some examples of uses of accelerometers "in the real world."

Finally, the student was asked to think about a "generic" device to measure acceleration and explain what basic elements it would require (i.e. a mass to be accelerated, a restoring force of some kind and a transducer to measure the force or position and convert it into an electrical signal.

1994 EE PhD quals question

Marc Levoy CS

My overall strategy was to a) guide the student through (hopefully) familiar terrain into a dilemma, then see b) how creative they were in suggesting ways out of it, and c) whether they realized that the dilemma had no solution as posed.

Q. Sketch the rendering pipeline of a modern graphics workstation.

A. geometry → transform → clip → project → shade → rasterize → Z-compare → image

(Many variations were accepted.)

Q. Polygons rendered using such a pipeline usually have "jaggies".
How might you modify the pipeline to eliminate these artifacts?

A. 1. Supersampling and averaging down, or
2. Analytic antialiasing (convolve the polygon with a filter and then sample)

(Hints provided as necessary.)

Q. How would you extend method 2 to handle a scene that contained overlapping polygons?

A. Convolve each polygon, sample it, and digitally composite its color into the image, i.e.
(α p + (1- α) i) where α = convolved shape, p = polygon's color, i = image pixel color

Q. Any problems with this approach?

A. It often produces incorrect results in pixels where two edges cross.

(Most students get this far without much thinking.)

Q. How might you solve this problem within the context of the given pipeline?

A. You can't. You have discarded the subpixel geometry of the first polygon rendered, without which you cannot determine how much is obscured by the second polygon. You must fundamentally alter the pipeline. Reasonable strategies include:

- solve for polygon visibility before rendering, producing a set of visible polygon fragments whose contributions you then add instead of compositing, or
- reintroduce the discarded subpixel geometry using a bitmask (aka an A-buffer), which is a special case of...
- use supersampling throughout and hope you aren't rendering picket fences.

(I was more interested in how well students reasoned through the dilemma they found themselves in than on whether or not they produced this particular list of solutions.)

Electromagnetics

QUALS ORALS (Jan. 1994)
Prof. Bruce Lusignan
Communications Satellite Planning Center

Question No. 1

An Astronaut is 1000 km away from his space station. I want you to design a radio system for him to communicate with the space station, only voice is needed. First what are the major system components and what alternatives are there for each? Then go through each system recommending and explaining your choice of alternatives.

Answer: Typical parameters are antennas at both ends, transmitter, receiver amplifier, modulator-demodulator, possibly voice encoder, error-correction coder.

Question No. 2:

On the antennas what might be good choices (I give some guidance as the "customer")?

Answer: For the Astronaut the need to move freely without worry might lead to the choice of an omni, zero dB gain, antenna. The space station can point the antenna towards the astronaut but has a size limit (I suggest 2M diameter).

Question No. 3:

What is an appropriate choice of radio frequency given the above choices of antenna characteristics, say between 100MHz and 10GHz?

Answer: The link equation and antenna gain-size relationship are written down, inspected, and led to the conclusion that choice of frequency does not change transmitter power requirement. ? Because one side is omni-direction, so P_R is independent of wavelength.

Question No. 4:

How about difficulty of pointing the Space Station 2M antenna?

Answer: The 100 MHz frequency gives wider beam width easier pointing. Write down beam width equation. $\alpha_x = \frac{\lambda}{D_x}$ $\alpha_y = \frac{\lambda}{D_y}$

Question No. 5:

On another parameter.. what might you choose to encode the voice on the radio carrier?

Answer: Alternatives include FM, AM, SSB in analog form. PCM or voice encoder can digitize the voice. QPSK might be typical digital modulation scheme, error coding might be used to allow greater noise. The choice should be made to minimize power required, the bandwidth, B, multiplied by the carrier to Noise C/N. Only PCM can be easily rejected. Detailed analysis is needed for remaining choices to compare power required as a function of the minimum voice quality required.

why ?

pre emphasis, companding. psophometric weighting ⟹ base band processing

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Communications Satellite Planning Center

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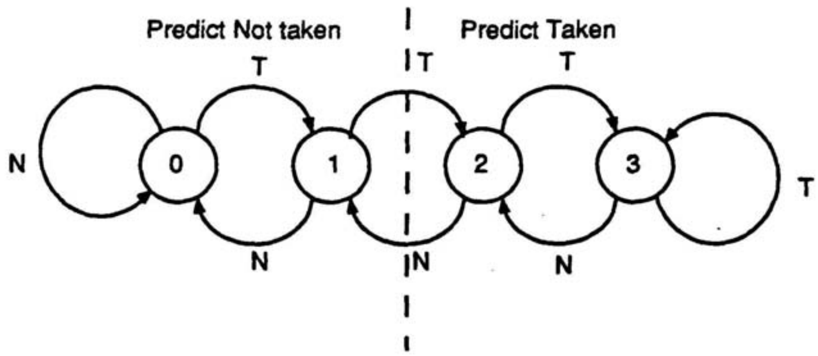
2-bit branch prediction counter



Question:

The above counter requires 2 bits per branch in a branch history table. Suppose you may keep as much state (bits) as you want, devise a general scheme to improve branch prediction accuracy. Use the code sequence and branch history trace below to devlop your answer. Show specifically how your scheme would be an improvement over the 2-bit counter prediction accuracy for branch b3.

```
b1:   if (aa == 2)
             aa = 0;

b2:   if (bb == 2)
             bb = 0;

b3:   if (aa != bb)
       {
             .....
       }
```

Time →

b1:	T	N	N	N	N	T	T	N	T	T	T	T
b2:	N	N	T	T	N	T	T	T	T	T	N	N
b3:	T	T	N	T	T	N	N	T	N	N	N	N

Answer:

There are many possible answers. The best approach is to realize that the behavior of branch b3 is correlated with the behaviors of branches b1 and b2. And that in general branch behavior may be corelated with previous branches. Thus, by keeping track of whether the last n branches were taken or not taken and by associating a 2 -bit counter with each combination of branch outcomes we may significantly increase branch prediction accuracy. Really good answers showed how the branch history trace for b3 can be broken into four cases using the branch histories of b1 and b2. The best answers showed how you would implement this scheme in a branch history table.

Received: from localhost by Sierra.Stanford.EDU (8.6.4/25-eef) id IAA15967; Thu, 17 Mar 1994 08:41:28 -0800
Full-Name: Shuye Huan
Date: Thu, 17 Mar 94 8:41:28 PST
From: Shuye Huan <shuye@Sierra.Stanford.EDU>
To: shankle
Subject: pantell's Ph.D quals
Message-ID: <CMM.0.90.0.763922488.shuye@Sierra.Stanford.EDU>

Professor Richard H. Pantell

1994 Quals Questions and Answers

1. If I wanted a broad-band, high reflectivity mirror, what material parameter would I specify?

   High conductivity.

   How to design a mirror with high reflectivity for only a picosecond?

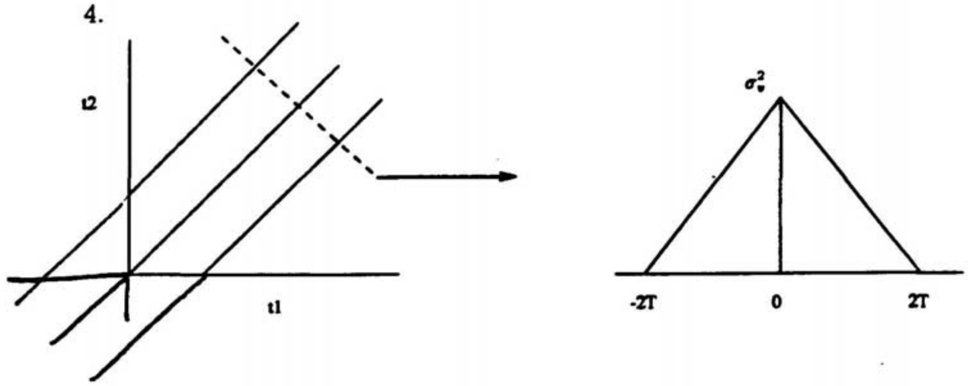
   Use a semiconductor. Generate free carriers with a picosecond laser.

2. Two coherent waves overlap at a junction. Write input and output electric fields. What are input and output average powers? Is there conservation of power? If not, why not?

   Power conservation can be maintained only by having a portion of the input power reflected. The fraction of power reflected depends upon the phases of the incident waves.

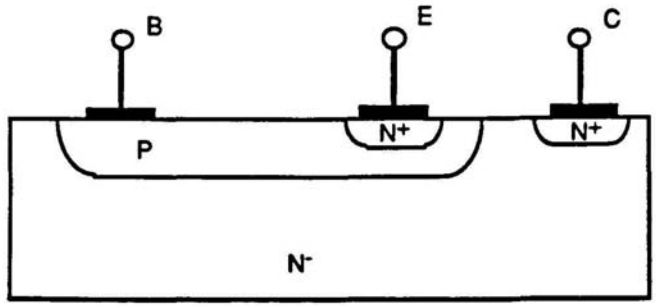
R>

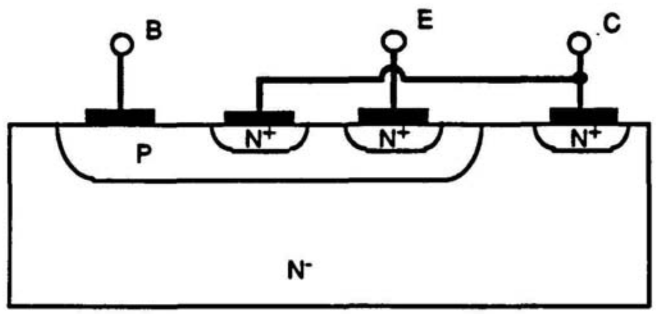
4.



5. No

1994 EE QUALS - PLUMMER





The first part of the exam (top drawing) was simply to make sure that each student understood the basics of bipolar transistor operation. I asked for a brief description of how the device operates (at the level of electrons and holes), and asked for a sketch of the standard I-V characteristics. This part of the exam took only the first 2 minutes or so, and was not weighted very heavily in the score.

The main part of the exam dealt with the bottom drawing in which an extra N^+ region has been added. I asked what effect this would have on the BJT terminal characteristics and for an explanation of how the dew device would function. Basically, the extra N^+ region acts as a JFET and will pinch off the base voltage at high V_{CE}, stopping the bipolar transistor from operating. The I-V characteristics look "normal" for low V_{CE}, and collapse down to 0 for higher V_{CE}.

There were a number of follow-up questions including asking for an equivalent circuit for the new device and asking what difference it would make if a current source (I_B) drove the base rather than V_{BE}. Finally, for those that got this far, I asked if the student could think of any interesting applications for this device.

In grading the exam, I was primarily looking for how the student reasoned about things he/she had not seen before. Generally I offered as much help as was needed to figure out what the new device structure would do and the grade I gave was influenced by how much help I had to give.

Message 91 (2799 chars)
Return-Path: <pratt@CS.Stanford.EDU>
Received: from Coraki.Stanford.EDU by Sierra.Stanford.EDU with SMTP (8.6.4/25-eef) id MAA25797; Wed, 23 Feb 19
Full-Name:
Received: by Coraki.Stanford.EDU (4.1/25-CORAKI-eef) id AA25109; Wed, 23 Feb 94 12:42:36 PST
Date: Wed, 23 Feb 94 12:42:36 PST
From: Vaughan Pratt <pratt@CS.Stanford.EDU>
Message-Id: <9402232042.AA25109@Coraki.Stanford.EDU>
To: shankle@sierra.Stanford.EDU
Subject: EE qual question
Cc: pratt@CS.Stanford.EDU, siroker@CS.Stanford.EDU

1. Does nondeterminism increase the class of languages accepted by finite-state automata? If yes, give an example showing this, if not prove that no such example exists.

Solution. No it does not. Any nondeterministic automaton A=(Q,S,d,q0,F) can be simulated by a deterministic machine B=(2^Q,S,d',{q0},F') whose states are the subsets of Q. B has the same alphabet S as A. B's transition function d' is defined so that for each subset R of Q and symbol a of S, d'(R,a) is the set of those states of Q reachable in A by an a transition from some state in R. The initial state of B is the singleton {q0} where q0 is the initial state of A. The final states of B are those subsets of Q containing a final state of A.

B is deterministic by construction. Any string accepted by B does so via a path from {q0} to an element of F' that can be seen to exist in A as a path from q0 to an element of F. Conversely every path of the latter kind can be found as a path of the former kind, establishing L(A)=L(B).

2. Define ambiguity for context-free grammars. Give an example of an ambiguous grammar. Does every ambiguous context-free grammar have an equivalent unambigous context-free grammar? If so, prove it, if not, give an example of such a grammar.

Solution. A context-free grammar is ambiguous when it generates some string by two different leftmost derivations, equivalently when it has two parse trees. S->SS, S->a is such a grammar, since aaa can be generated by either of the leftmost derivations

```
S => SS => aS => aSS => aaS => aaa
S => SS => SSS => aSS => aaS => aaa
```

This grammar is equivalent to the unambiguous grammar S->aS, S->a. Not every grammar has such an equivalent unambiguous grammar, for example any grammar generating the set of strings of the form a^i b^j c^k such that either i=j or j=k, evidently a context-free language. This language is inherently ambiguous because for every grammar G there exists sufficiently large N that a^N b^N c^N must, by the pumping lemma, have a phrase in the ab part covering more than half of b, and a phrase in the bc part also covering more than half of b. These phrases overlap without properly nesting and hence cannot belong to the same parse tree. Hence this string has at least two distinct parse trees.

Message 49 (1031 chars)
Return-Path: <siegman>
Received: from localhost by Sierra.Stanford.EDU (8.6.4/25-eef) id PAA27894; Fri, 25 Feb 1994 15:47:28 -0800
Full-Name: Anthony E. Siegman
Date: Fri, 25 Feb 94 15:47:28 PST
From: "Anthony E. Siegman" <siegman@sierra.Stanford.EDU>
To: "Diane J. Shankle" <shankle@sierra.Stanford.EDU>
Subject: Re: LATE QUALS QUESTIONS
In-Reply-To: Your message of Fri, 25 Feb 94 13:57:43 PST
Cc: siegman
Message-ID: <CMM.0.90.0.762220048.siegman@Sierra.Stanford.EDU>

Siegman quals question:

How high can a battery lift itself? That is, suppose a battery drives an ideal motor or hoist which lifts the battery only, with perfect efficiency -- how high can that battery lift itself?

I'd like you to tell me both (a) the basic physics needed to answer this question, and (b) based on your practical experience with some real battery, like maybe a flashlight or car battery, I'd like a numerical value, even if it's a rather rough estimate, of what this characteristic height might be for a real battery.

Received: by nova.Stanford.EDU (5.57/Ultrix3.0-C); Thu, 17 Mar 94 10:39:33 -0800Date: Thu, 17 Mar 94 10:39:33
To: shankle@Sierra.Stanford.EDU
Subject: Quals questions
Cc: len@nova.stanford.edu

Electromagnetics

Qualifying Exam '93-94

Question 1: Consider a regular lattice of conducting spheres. The lattice is infinite in x-y, but finite in z. If this lattice is immersed in an initially uniform, static, z-directed electric field, what is the field configuration in within the lattice.

Answers should properly take into account the boundary conditions for electric fields, the symmetries of the problem, assumptions articulated by the examinee, etc. The ability to visualize and describe the fundamental properties of the fields in important. A sketch of the field configuration and a discussion of the three dimensional field arrangement is sufficient.

Is this static or time variant field?

Follow-up question: Is the average field strength within the lattice different from the field strength in the absence of the lattice.

Answers should consider the nature of the electric field source and the induced polarization of the spheres.

Question 2: What electromagnetic waves can propagate in a uniform "forrest" of thin conducting wires? The wires are of infinite extent in z, and uniformly spaced in x-y.

Answers should consider the boundary conditions for EM waves, rudimentary propagation modes in the vicinity of conductors, and the periodic nature of the medium. A knowledge of how the boundary conditions for EM fields control solutions to the wave equation, and the ability to apply this knowledge to deduce the character of solutions in new situations is important.

Quantitative analysis is not required for either question, but the fundamental nature(s) of the solutions should be articulated.

S. Leonard Tyler len@nova.stanford.edu (415) 723 3535

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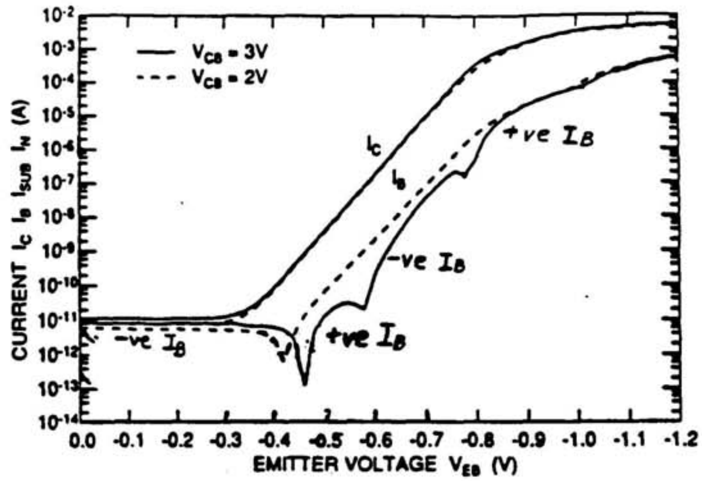
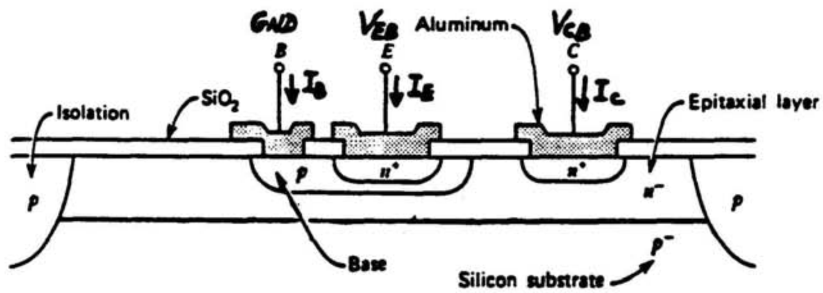
94 Qualifying Examination
Simon Wong

circuits





This problem is not intended to test how much device physics a student knows. Instead, it is meant to determine how the student approaches the problem and arrives at the solution when he or she is given sufficient hints. Indeed, all students tested need some hints.

The collector and base currents of a NPN bipolar transistor (fabricated in CIS) under forward active bias are shown.

(1) For $V_{CB} = 2V$, why is the base current negative at low V_{BE} ($V_{BE} < 0.4$ V) ?
Hint : $I_C = -I_B$ at low V_{BE}
Expected Solution : Negative base current means that holes are flowing out of the base instead of into the base. This implies that excess holes have to be generated inside the transistor. The only place where carriers can be generated is in a reverse biased P-N junction (i.e., base-collector junction). Since $I_C = -I_B$, the negative base current must be the leakage current through the reverse biased base-collector junction, which is mostly due to electron-hole pair generation in the base-collector depletion region.

(2) For $V_{CB} = 3V$, why is the base current negative even for $V_{BE} = 0.6$ to 0.8 V ?
Hint : The negative base current is not observed at lower V_{CB} (e.g., 2V). The negative base current tracks the collector current.
Expected Solution : Similar to the first part, negative base current means that excess holes are generated in the reverse biased base-collector junction. Since the negative base current tracks the collector current and only occurs at higher V_{CB}, it is related to the collector current (electrons) going through a larger potential drop (high electric field). The highly accelerated electrons can cause impact ionization and generate excess electron-hole pairs.

John M. Cioffi

QUALS' 94

1. A loan is made at annual interest rate r for a sum of S dollars. Payments begin at the end of the first month after which the loan was made. Call the remaining balance on this loan y_k where k is the time at the k^{th} monthly payment (# of payments = k). Assume payments are fixed at P/month.

a). Write a difference equation for the monthly balance.

b). Find the Z-transform of the balance, $Y(z)$.

i). What is the region of convergence?
ii). Is $Y(z)$ stable?

c). What are two dominant modes of y_k?

d). Solve for y_k using 2 initial conditions.

e). Suppose you borrow \$1,000,000 at 5% interest, fixed over 30 yrs. Monthy payments. What is the monthy payment?

Solution '94

1.

1 a) $y_{k+1} = y_k \underbrace{(1 + r/12)}_{g} - P$ $\quad P =$ payment

3 b). $z Y(z) = g Y(z) - P z^{-1}/1 - z^{-1}$

$$Y(z) = \frac{-P z^{-1}}{(z-g)(1-z^{-1})} = \frac{-P}{(z-g)(z-1)}$$

i). $|z| > g$

ii). no

2 c). g^k and constant

2 d). $y_k = A g^k + B$ $\qquad y_0 = S$

$y_1 = S \cdot g - P$

$A + B = S$

$A \cdot g + B = S \cdot g - P \rightarrow Ag + S - A = S \cdot g - P$

$A(g-1) = S(g-1) - P$

$y_k = \left(S - \frac{P \cdot 12}{r}\right)(1 + r/12)^k + \frac{P \cdot 12}{r}$ $\qquad A = S - P/g-1$

$B = + P/g-1$

2 e). $y_{30 \cdot 12} = 0$ $\qquad K = 30 \cdot 12$

$$\left(S - \frac{P \cdot 12}{r}\right)(1 + r/12)^K = -\frac{P \cdot 12}{r}$$

$$S (1 + r/12)^K = \frac{12P}{r}\left[(1 + r/12)^K - 1\right]$$

$$P = \frac{Sr}{12} \cdot \frac{1}{1 - (1 + r/12)^{-K}}$$

$$= \frac{10^6 \cdot (.05)}{12} \cdot \frac{1}{1 - (1 + .05/12)^{-360}}$$

.1 $5368.22

1994 Ph.D. Qualifying Exam Question

D. C. Cox signal

* We will discuss a phase shift keyed digital radio signal. I will give you a description of the signal.

* The signal can be represented either as a complex exponential function $e^{-j\omega t}$ or as a cosine function. Choose which way you want to represent the signal.

\- About half of the students choose each one.

* I gave the representation chosen:

$s(t) = \mathrm{Re}\ \{e^{-j[\omega_0 t + \theta(t)]}\}$

or

$s(t) = \cos[\omega_0 t + \theta(t)]$

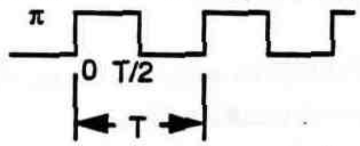
and

$\theta(t)$ is a square-wave-like periodic function that alternates between 0 and π with period T.

$\theta(t)$ is symmetrical with half periods T/2.

$\theta(t)$ is periodic over all time.

\- The student would write s(t) on the board and sketch

$\theta(t) =$ 

\- I then gave the information $\omega_0 >> \frac{1}{T}$

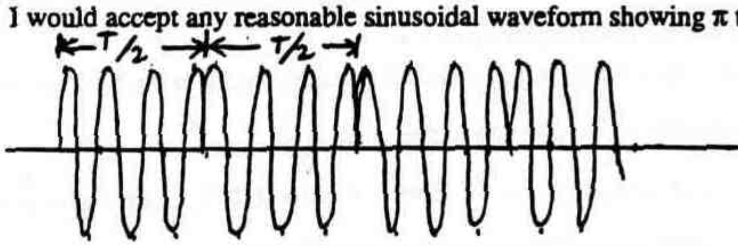
(no one seemed bothered by my mixing ω_0 and $\frac{1}{T}$, but it doesn't really matter.)

I stressed ω_0 was much much greater than $\frac{1}{T}$

1. The first question was to roughly sketch the waveform of s(t) in time.

* I would accept any reasonable sinusoidal waveform showing π transitions at intervals of T/2.

e.g.,



* Most (maybe 80%) sketched an acceptable representation. Some noted directly from the equation that the π phase shift "inverted" the waveform; some expanded $e^{-j(\omega_0 t + \theta(t))}$ to

$e^{-j\omega_0 t}\, e^{-j\theta(t)}$ or

$\cos[\omega_0 t + \theta(t)]$ to

$\cos \omega_0 t \cos \theta(t) - \sin \omega_0 t \sin \theta(t)$,

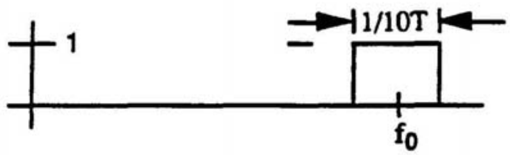
noted that the cosine of 0 and π corresponded to + 1 and -1, and, also, that $\sin \theta(t)$ was always 0. A very few didn't know where or how to start, and needed help in sketching the waveform.

2. For the second question, I stated that we now wanted to pass the signal s(t) through a bandpass filter. I described the filter as:

- being centered at w_0 or f_0
- being a non-realizable filter having a rectangular passband with total bandwidth of $\frac{1}{10T}$
- having a gain of unity

I ask for a sketch of the filter, accepting any reasonable version, and adding the 2π to the width if they used ω_0, although it doesn't really matter.

e.g.,



Some drew the negative frequency side too.

* I then ask what the filter output would be with s(t) as the input and ask them to find it any way they wanted to, e.g., using frequency domain or time domain analysis.

* There are many ways to proceed (many more than I considered at the outset).

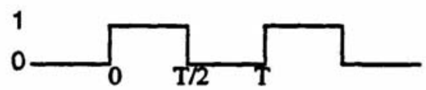
- A reasoning approach:
  - Note from the equation and/or the sketch that $\theta(t)$ is equivalent to multiplying $e^{-j\omega_0 t}$ or $\cos \omega_0 t$ by a square wave alternating between + 1 and -1 with period T.
  - Note that multiplication in time is convolution in the frequency domain.
  - Note that the $\omega_0 t$ term is a δ function at f_0 or ω_0.
  - Note that a periodic square wave has a Fourier series, i.e. is δ functions, with the fundamental, i.e. the first frequency at 1/T.
  - Note that a "dc" term for the square wave is translated to f_0 or ω_0 by convolution, and the fundamental is translated to $f_0 \pm \frac{1}{T}$ or $\omega_0 \pm \frac{2\pi}{T}$.
  - A crucial step is to recognize that the filter is narrow, 1/10T, compared to $\frac{1}{T}$ so only the "dc" component of the square wave would pass.

- Another crucial step is to recognize that the "dc" component of a symmetrical ± 1 periodic square wave has 0 "dc" component.
- Therefore, the filter output is 0!

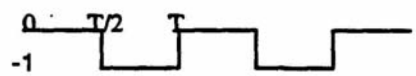
• more mathematically inclined students would expand $e^{-j(\omega_0 t + \theta(t))}$ or $\cos[\omega_0 t + \theta(t)]$ as noted in 1. above and conclude that multiplication by the ± 1 square wave resulted.

• the number of approaches taken to get the frequency spectrum of the square wave was astonishing.

- Some computed Fourier series coefficients without looking first to see which were relevant to the filter.
- Some computed a Fourier transform. When done correctly, this resulted in the needed discrete spectrum, but there were many more wrong calculations made than correct ones, and too many students did not arrive at the conclusion or recognize that it would be a discrete line spectrum, i.e. of δ functions. Some times the hint that $\theta(t)$ was periodic helped them proceed correctly, but many times it did not. The hardest approach taken was to break the square wave into a 0 to +1 wave



and a T/2 time shifted 0 to -1 wave



and then compute the discrete spectra of the two waves from the Fourier integral. The two spectra were then added to arrive at he overall spectrum. Only <u>one</u> of the students who chose this approach succeeded in keeping signs and constants straight and arriving the correct spectrum with the dc components of the two waves being equal and of opposite sign, resulting in a 0 "dc" component.

- A few proceeded with a time domain approach by converting the bandpass filter to a low-pass equivalent and attempting convolution with the square wave. Only <u>one</u> reasoned his way to concluding that the filter was averaging several periods of the square wave to result in a 0 output.
- One student looked at the sketched waveform of s(t), looked at the filter sketch, did no calculations or writing and in about a minute calmly stated that the output

would be 0 because all the frequency components would be outside the passband. She had remarkable insight or reasoning !?! This was a sharp contrast to the many students who either did not note that the filter would not pass the fundamental at $\pm \frac{1}{T}$, or did not note that the square wave had a 0 "dc" value.

- Significantly fewer than half arrived at the 0 output for the filter. I gave some credit for proceeding in a logical way, and for thinking through steps of the problem. I reduced the score for hints, depending on how obscure (or obvious) the point was that needed the hint. Hints were usually in the form of a question, or a "what if you....."

3. For the few students who finished the filter output question, the next question was as follows:
If you want to obtain the frequency ω_0 or f_0 from s(t) what can you do? The linear filter obviously didn't give you an output at ω_0.

* There are several possible answers to this question. As examples:

- The simplest approach to see from the information of the problem is to square s(t). The $\pm$ 1 square wave becomes all +1, and the ω_0 term becomes a "dc" term and a $+2\ \omega_0$ term, e.g., $\cos^2 \omega_0 t = \frac{1}{2} + \frac{1}{2} \cos 2\ \omega_0 t$. The $\frac{1}{2} \cos 2\ \omega_0 t$ can then be bandpass filtered, and divided in frequency to yield a $\cos \omega_0 t$ signal.

- Another way if you know T, is to generate a square wave with period T. Multiply the s(t) by the square wave, pass the result through a bandpass filter, and detect (rectify) the filter output. The detected output can be used to control the phase of the square wave in a feedback loop to track $\theta(t)$. There are several variations of this "remodulator" approach.

* The remaining question, for the several who arrived at this point, was to discuss any disadvantages resulting from the method they proposed.
A few examples:

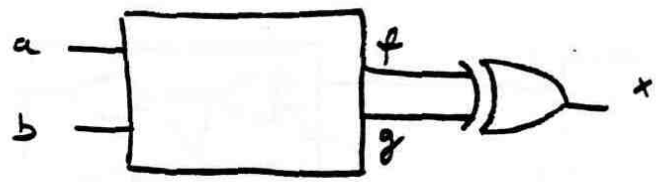
  - Squaring and dividing by two results in a π phase ambiguity for the recovered ω_0.
  - The remodulator has an acquisition time associated with aligning the square wave phase with the phase of $\theta(t)$.
  - The nonlinear operation, squaring, has a noise penalty in low signal-to-noise ratio conditions.

PH.D. QUALIFYING EXAMINATION 1994 / SOLUTIONS

Topic: Logic design and Boolean algebra

NOTE: We are dealing with combinational circuits only.

1) An engineer claims correctly that he can replace the XOR gate in the circuit below by a NAND gate.
Can you give an example of a circuit that can fit in the box?

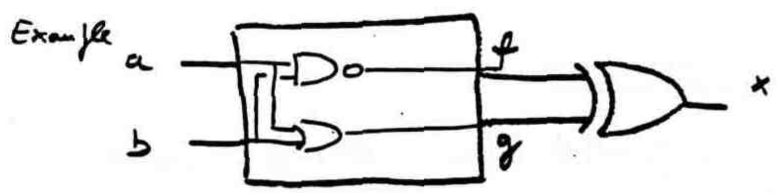


Compare XOR to NAND

a	b	XOR	NAND
0	0	0	1
0	1	1	1
1	0	1	1
1	1	0	0

XOR differs from NAND on input pattern 00

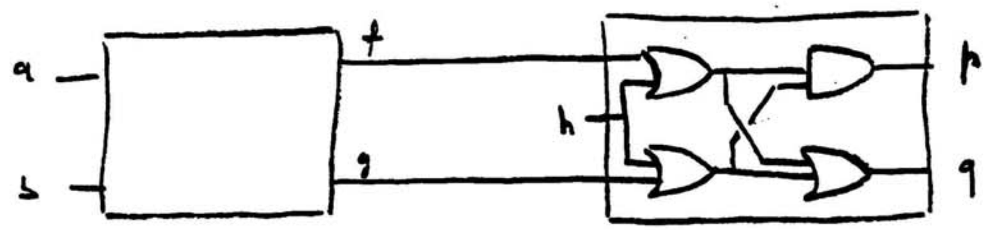
The network in the box should not yield 00



2) Is the circuit in the box unique?
How can you characterize the class of circuits that fit the box?
Hint: consider functions f(a,b) and g(a,b).

$$f + g = 1$$

3) The circuit in the box is driving another circuit as shown below.
What are the possible output patterns in terms of variables (p,q)?



$$p = (f+h)(g+h) = fg + h$$
$$q = (f+h)+(g+h) = f+g+h$$

Since $f+g = 1 \Longrightarrow q = 1$

p can be 1, 0

Output vectors are $\begin{bmatrix}1\\0\end{bmatrix}$ $\begin{bmatrix}0\\1\end{bmatrix}$

4) Assume that f and g can take any value,
what are the possible output vectors?

Vector $\begin{bmatrix}0\\0\end{bmatrix}$ is possible when all inputs are 0

Vector $\begin{bmatrix}1\\0\end{bmatrix}$ is NOT possible because

$$fg + h = 1 \Longrightarrow f+g+h = 1$$

Robert dutton

-- using template mhl.format --
Date: Mon, 31 Jan 1994 11:50:21 PST
To: gray@isl.Stanford.EDU
cc: dutton@gloworm.Stanford.EDU, group@gloworm.Stanford.EDU

From: dutton@gloworm.Stanford.EDU@isl.Stanford.EDU
Subject: Re: Qualifying Exam Questions

Replied: Mon, 31 Jan 1994 12:01:04 -0800
Replied: "dutton%gloworm.Stanford.EDU@isl.Stanford.EDU "
Return-Path: dutton@gloworm.Stanford.EDU
Return-Path: <dutton@gloworm.Stanford.EDU>
Full-Name:
In-reply-to: Your message of "Mon, 31 Jan 94 11:10:36 PST."
<199401311910.LAA22345@isl.Stanford.EDU>

Hi Bob:
Here's an e-mail version of my quals question. As I've said before, I strongly object to giving an expected answer etc.
Best Regards,
Bob Dutton

circuits

```
   _________________________________
  |\                                \
  | \                                \
  |  \                                \
Xi|-> \________________________________\->Xo
   \   |                                |
    \  |                                |
     \ |                                |
      \|________________________________|
       <-----------1 cm---------------->
```

Consider the following "black box" which has a total length of 1cm. An incoming signal Xi goes in at one end and Xo, the output signal which is INVERTED (ie if you use voltages as the variable, it has VL->VH at input and then output goes VH->VL), appears at some time later. You are free to put whatever you want in the box (and if you're and OEIC person you could do it with photons as well).

Question 1: How long does it take to get the signal through the box and what are the fundamental limits on minimizing that time.

(Hints: What is the absolute fastest it could be without in inversion? What are the limits on how aggressively you can scale the time for the inversion? What is the role and effect of the wire?)

Question 2: What are the energy requirements on getting the signal through the box and again, what are the fundamental limits.

(Hints: What is the dynamic power dissipated? What is the role of static power if any? If you tried to minimize power, what would be the limits?)

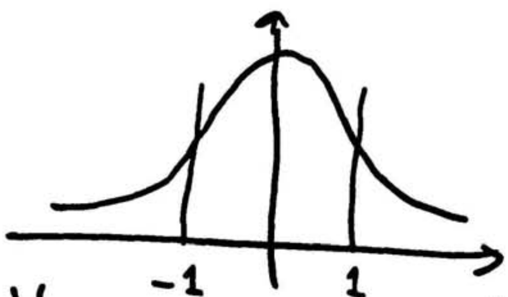
PS--Most people chose to use some inverter circuit, either bipolar or MOS, and the rest of the 1cm was some kind of a wire. When people went the wire route, I told them that the total capacitance of that wire was 0.5pF. A very few people, of their own choice, went for photons. In this case the challenge was what to do about the inversion of the signal.

Problem 3

Let X1, X2, ..., X16 be a sequence of independent identically distributed random variables each with zero mean and unit variance. What is the probability that the absolute value of their sum exceeds 4 ?

Cannot find an exact soln.
By Central limit thm.

$$Z_n = \frac{1}{\sqrt{n}} \sum_{i=1}^{n} X_i \xrightarrow{\mathcal{D}} \mathcal{N}(0,1)$$



$$P\left\{ \left| \sum_{i=1}^{16} X_i \right| > 4 \right\} = P\left\{ \left| \frac{1}{4} \sum_{i=1}^{16} X_i \right| > 1 \right\}$$

$$\approx 0.32$$

Problem 2

You decide to sell your car. The offers you receive $X_1, X_2, \ldots$ are independent exponentially distributed random variables, all with the same mean of A dollars. You decide to accept the first offer that exceeds B dollars. Let N be the index of the first offer that exceeds B. What is the expected selling price of your car, i.e. what is $E(X_N)$?

$$E(X_N) = E\,E(X_N \mid N = n)$$

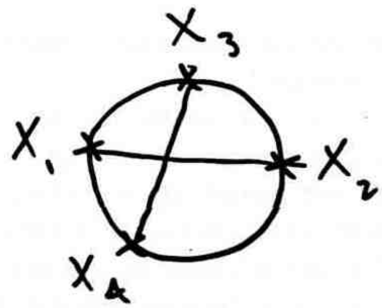
$$= E(X_1 \mid X_1 > B)$$

$$= A + B$$

Quals Questions 1994

Problem 1

Let X_1, X_2, X_3, X_4 be four points chosen independently and at random on a circle. What is the probabilty that the cords X_1 to X_2 and X_3 to X_4 intersect ?



There are $3!$ ways of ordering three points (one can be fixed WLOG), two lead to intersection

$$P = \frac{1}{3}.$$

Orals questions. Gene Franklin 1994

Consider the block diagram shown and the corresponding Bode plot.

1) What is the gain margin of this design?
2) What is the system error if the input is a unit amplitude sinusoid at 0.2 rad/s?
3) How would you change the compensation to reduce this error to 0.01 without changing the gain margin?
4) Give a transfer function for your compensation.
5) Give a circuit realization of your compensation.
6) No one got this far; I'd have thought up something if they did! viz. What is the steady-state error of this system to a constant/ramp/parabolic input? How would you change the phase margin to 60 degrees? What is the closed loop bandwith and overshoot expected of the system with your compensation? etc.

1994 Electrical Engineering Qualifying Examination Questions

John Gill

Signal

A *packet* or *frame* of data to be transmitted over a network consists of 8-bit bytes followed by a single end of frame (EOF) byte consisting of all ones ($11111111_2 = FF_{16}$).

byte #1	byte #2	...	byte #n	11111111
←		n data bytes	→	← EOF →

Question 1: Because arbitrary binary data can be transmitted over this network, there is a possibility that the EOF byte may occur within the data frame, thereby misleading the receiver into thinking that the frame has ended. Suppose that n data bytes are generated at random (all bits are independent and 0 or 1 occur with equal probability). Find the probability that the EOF byte occurs within the data.

Answers:

1. The probability that any particular byte equals the EOF byte is $p = 2^{-8} = 1/256$. The probability that any particular byte is *not* the EOF byte is $1 - p = 255/256$. The probability that all n bytes are *not* the EOF byte is $(1-p)^n$ because the bytes are independent. Finally, the probability that at least one of the n bytes equals the EOF byte is $1 - (1-p)^n$.

2. Using the binomial probability distribution, the probability of at least one occurrence of the EOF byte is given by

$$\sum_{i=1}^{n} \binom{n}{i} p^i (1-p)^{n-i} = 1 - \binom{n}{0} p^0 (1-p)^{n-0} = 1 - (1-p)^n .$$

3. If there is at least one occurrence of EOF, then the first occurrence is in either byte 1 or byte 2 ... or byte n. The probability that the *first* occurrence is in byte i is $(1-p)^{i-1}p$, which is the probability that the first $i-1$ bytes are not EOF but byte i does match. Therefore the probability of at least one occurrence of EOF is

$$\sum_{i=1}^{n} p(1-p)^{i-1} = p \sum_{i=0}^{n-1} (1-p)^i = p \cdot \frac{1-(1-p)^n}{1-(1-p)} = 1 - (1-p)^n .$$

Question 2: What happens to the above probability as n ranges from 1 to ∞?

Answers: The probability that the EOF byte occurs approaches 1 as n gets large.

Question 3: Suppose that data bytes are generated at random until the EOF byte is produced. What is the average number of *bytes* generated?

Answers: Let X be the random variable that counts the number of bytes generated, up to and including the EOF byte. Then X has possible values $1, 2, 3, \ldots$, and the probability that $X = i$ is given by

$$\Pr(X = i) = q^{i-1}p \qquad \text{where } p = \frac{1}{256},\ q = \frac{255}{256}.$$

The expected value of X is defined by

$$E[X] \;=\; \sum_{i=1}^{\infty} i \Pr(X = i) \;=\; \sum_{i=1}^{\infty} iq^{i-1}p.$$

The expected value can be determined by several methods:

1. Memory. The expected value of a geometric random variable with success probability p is $1/p$. In this case, $1/p = 256$.

2. Calculus. The series for the expected value is the derivative of a simpler series:
$$\sum_{i=1}^{\infty} iq^{i-1}p = p\sum_{i=1}^{\infty} iq^{i-1} = p\frac{d}{dq}\sum_{i=1}^{\infty} q^i = p\frac{d}{dq}\frac{q}{1-q} = p\frac{1}{(1-q)^2} = \frac{p}{p^2} = \frac{1}{p}.$$

3. Recursion. With probabiltity p, the value of X is 1. Otherwise, with probability q, the first trial is wasted, the conditional expectation is now 1 plus the original expectation. In other words, $E[X]$ satisfies the equation
$$E[X] = p \cdot 1 + (1-p)(1 + E[X]) = 1 + (1-p)E[X],$$
which is easily solved to obtain $E[X] = 1/p = 256$.

Question 4: Suppose now that we relax the requirement that the EOF byte must be byte-aligned. In other words, we generate data one bit at a time until the EOF pattern (8 consecutive ones) appears. What is the average number of *bits* generated up to and including the last EOF bit?

Answers: Let X be the random variable that counts the number of bits generated. The probability distribution for X is easy to determine for small values. For example,

$$\begin{aligned} \Pr(X = 8) &= \Pr 11111111 = 2^{-8} \\ \Pr(X = 9) &= \Pr 011111111 = 2^{-9} \\ \Pr(X = 10) &= \Pr x011111111 = 2^{-9} \end{aligned}$$

But the general formula is rather complex:

$$\begin{aligned} \Pr(X = i) &= \Pr(\text{first } i-9 \text{ bits do } \textit{not} \text{ contain 8 consecutive ones}) \times \\ &\quad\ \Pr(\text{last 9 bits are } 01111111) \end{aligned}$$

The expected value will have to be determined by some other method. Here are several solutions.

John Gill

1. Let X_1 be the expected number of bits until the first 1 is generated. Obviously, $E[X_1] = 2$, since X_1 has a geometric probability distribution. Let X_2 be the expected number bits until two consecutive ones are generated. Once the first one occurs, the next bit is a one with probability 1/2; otherwise the next bit is zero, which causes the search for two consecutive ones to start over. This leads to the following formula for $E[X_2]$ in terms of $E[X_1]$:

$$E[X_2] = E[X_1] + \frac{1}{2}\big(1 + (1 + E[X_2])\big) \quad \Rightarrow \quad E[X_2] = 2E[X_1] + 2 = 6\,.$$

   Similarly, $E[X_3] = 2E[X_2] + 2 = 14$. The sequence $\{E[X_i]\}$ for $i = 1, \ldots, 8$ is $\{2, 6, 14, 30, 62, 126, 254, 510\}$. (Obviously, $E[X_i] = 2^i - 2$.) Thus $E[X_8] = 510$.

2. If the first bit is 0, then the conditional expected value of X_8 is $E[X_8] + 1$, since the first bit is wasted and the experiment has returned to its initial state. Similarly, if the first two bits are 10, then conditional expected value of X_8 is $E[X_8] + 2$. Continuing in this way we obtain the following table:

Initial bits	Probability	Conditional expectation
0	2^{-1}	$E[X_8] + 1$
10	2^{-2}	$E[X_8] + 2$
110	2^{-3}	$E[X_8] + 3$
1110	2^{-4}	$E[X_8] + 4$
11110	2^{-5}	$E[X_8] + 5$
111110	2^{-6}	$E[X_8] + 6$
1111110	2^{-7}	$E[X_8] + 7$
11111110	2^{-8}	$E[X_8] + 8$
11111111	2^{-8}	8

   The unconditional expected value of X_8 is obtained by averaging the conditional expectations in the above table. This leads to a formula for $E[X_8]$:

$$E[X_8] = 8 \cdot 2^{-8} + \sum_{i=1}^{8} 2^{-i}(E[X_8] + i) = \frac{255}{256}E[X_8] + \frac{510}{256}\,.$$

   Therefore $E[X_8] = 256 \cdot \dfrac{510}{256} = 510$ bits.

3. Each time a zero is generated, the experiment ends if the next 8 bits are all ones. This occurs with probability $p = 2^{-8}$. Since the trials separated by zeroes are independent, the expected number of trials is $1/p = 2^8$. That is, the expected number of zeroes seen before 8 consecutive ones is 256. In fact, the first trial does not require an initial zero, so the average number of zeroes is 255. Since zeroes and ones are equally likely, the average number of ones is also 255. Therefore the expected number of bits generated is 510.

There were some variations to the following questions, not all of which are listed. The derivations listed below were not the only possible ones.

1. A block diagram is shown showing a signal $e^{j\omega t}$; $t \in (-\infty, \infty)$ as input to a box (system) with a question mark in it. The output to the box is $y_\omega(t)$; $t \in (-\infty, \infty)$. The first question is when is knowing this enough to describe the output for an arbitrary input?

   Answer:

   If you know the output to complex exponentials and you know it for all real ω, then you can find the output for any input signal that has a Fourier representation if the system is linear. In particular, if the input can be represented as a Fourier series $x(t) = \sum_{k=-\infty}^{\infty} c_k e^{j2\pi \frac{k}{T} t}$, then the output will be

$$y(t) = \sum_{k=-\infty}^{\infty} c_k y_{2\pi k/T}(t).$$

   If the input can be expressed as a Fourier integral: $x(t) = \int_{-\infty}^{\infty} X(f) e^{j2\pi ft}\, df$, where $X(f)$ is the Fourier transform of $x(t)$: $X(f) = \int_{-\infty}^{\infty} x(t) e^{-j2\pi ft}\, dt$, then the output is

$$y(t) = \int_{-\infty}^{\infty} y_{2\pi f}(t) X(f)\, df.$$

   An important point here is that time invariance is NOT needed to describe the output in terms of the given information.

   If one assumes that the system is both linear and time invariant, then the above formulas hold true with

$$y_\omega(t) = H(\omega/2\pi) e^{j\omega t},$$

   where $H(f)$ is the Fourier transform of the impulse response. Alternatively, in this case you can write the output transform as $Y(f) = H(f)X(f)$ and infer $H(f)$ from the above relation. This follows immediately because complex exponentials are eigenfunctions of LTI systems. If the system is time varying, this representation is not true in general, but the answer to the question is still "yes" provided the input can be written as a linear combination of complex exponentials.

   If the system is nonlinear, then the answer is "no" in general, you cannot describe the output for an arbitrary input knowing only how the system effects complex exponetials.

2. A diagram is shown with a signal $x(t)$ as input to a box with output $x(t)\cos(2\pi t)$.

   (a) Is the system linear or time invariant?

   Answer: The system is linear (can easily prove from basic definitions). It is not time invariant (shifting the input does not simply shift the output).

   (b) Suppose the input has period 1 and has a Fourier series with Fourier coefficients, say, c_k. What are the Fourier coefficients in the Fourier series for the output in terms of the c_k?

   Answer: The output also has period 1 and its Fourier coefficients are

$$\begin{aligned} b_k &= \int_0^1 [x(t)\cos(2\pi t)]e^{-j2\pi kt}\,dt = \int_0^1 [x(t)\frac{1}{2}(e^{2\pi t} + e^{-j2\pi t})]e^{-j2\pi kt}\,dt \\ &= \frac{1}{2}\int_0^1 e^{j2\pi t}e^{-j2\pi kt}\,dt + \frac{1}{2}\int_0^1 e^{-j2\pi t}e^{-j2\pi kt}\,dt \\ &= \frac{1}{2}(c_{k-1} + c_{k+1}). \end{aligned}$$

3. A diagram is shown of a system with an input signal $x(t)$ producing an output signal of $y(t) = e^{j\gamma x(t)}$.

   (a) Is the system linear? Time invariant?

   Answer: The system is not linear because exponentiation is a nonlinear operation. The system is time invariant because shifting the input simply shifts the output.

   (b) Suppose that the input has period 1 and has a Fourier series. What can you say about the output?

   Answer: The output also has period 1 and therefore also has a Fourier series.

   (c) Write an expression for a Fourier coefficient of the output.

   Answer: $b_k = \int_0^1 e^{j2\pi x(t)}e^{-j2\pi kt}\,dt$.

   (d) If the input is a cosine or sine, do you know what kind of integral this is?

   Answer: A Bessel function. A point here is that even when the system is nonlinear, you can sometimes find Fourier representations of the output for specific inputs. Just not in general.

   (e) Suppose now that you are told that $x(t)$ is Gaussian with mean 0 and variance 1. What is $E[y(t)]$? $E[y^2(t)]$?

   Answer: $E[y(t)] = E[e^{j\gamma x(t)}]$. This is recognizable as the characteristic function of a Gaussian variable with argument γ and hence is simply $e^{-\frac{\gamma^2}{2}}$. For the second moment, just replace γ in the answer by 2γ.

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Message 55 (3896 chars)
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id AA20071; Tue, 1 Feb 94 14:04:39 -0800
Message-Id: <9402012204.AA20071@pepper.Stanford.EDU>
To: shankle@sierra.stanford.edu
Cc: ag@pepper.Stanford.EDU
Subject: EE Quals
Date: Tue, 01 Feb 94 14:04:38 PST

Hi Diane! I got a email request that we are supposed to get you the questions and answers for the EE quals. Here they are for me. Let me know if you need anything more. -- Anoop.

EE Quals 1994: Questions and Answers

Anoop Gupta leave

1. Why has the design of memory systems become increasingly more important over the last 10-15 years?

Ans. The reason is that processor speeds have been growing much faster than memory speeds. (The rates are about 54% year for processors in contrast to about 7% for DRAMS). The increase in performance of processors comes from faster clock rates, deeper pipelines and superscalar designs. This has resulted in need for both lower latency and much higher-bandwidth access to caches.

2. What is the rate at which DRAMS are increasing in size?

Ans. 4x every 3 years.

3. HP designs workstations using a single large external cache. In contrast, DEC, SGI, SUN have been designing workstations with small on-chip caches, a large external cache, and then the main memory. What are the advantages/disadvantages of the two approaches?

Ans. The advantage of the HP approach is that the miss rate to the first-level cache is much smaller as compared to the alternatives. The miss latency is also smaller, as there is no second-level cache that consumes cycles. The key disadvantage is that it is very difficult to build processors with a very high clock rate. The approach has worked well upto 100MHz or so, but the approach does not seem feasible for the 200MHz/250MHz designs that DEC-alpha is targetting, for example.

4. Why are single cycle writes tough to do in a cache? Suggest a way of doing them in write-through caches.

Ans. Single cycle writes are tough to do because it is necessary to know the outcome of the tag comparison before we go and write data into the data RAMS. (In contrast, for reads, the data can be fetched in parallel with the tags.)

t is feasible to achieve single cycle writes in write-through caches as ollows. For a multiword cache line we have sub-block valid bits. When the rocessor supplies the data and the address, we write the data in the ppropriate word of the cache line, set the subblock valid bit, and in arallel do a tag compare. If the tag matches (the case of write-hit), we re done with processing the write. If the tag does not match, then we have write-miss, so we are allowed to take multiple cycles. In this case we go nd overwrite the tag with the tag of the address just written, and reset ll the remaining subblock valid bits.

. What is meant by a virtually-indexed physically-tagged cache? Why would processor use such a cache for its on-chip cache?

s. In a virtually-indexed physically-tagged cache, we index into the che using the virtual address, while the tag that is stored in the cache a physical-address tag. The reason for using such a cache is to avoid iting for the VA-PA translation before indexing into the cache. The VA-PA anslation happens in parallel with the data and tag fetch from the cache, o by the time the tag comes out the translation has completed and the ysical tag can be compared.

ing physical tag helps over using virtual tags because: (i) we don't have flush the cache on context switches and (ii) the physical tags are aller than the virtual tags, thus reducing space required for the on-chip che.

1994 Ph.D. Qualifying Examination Questions

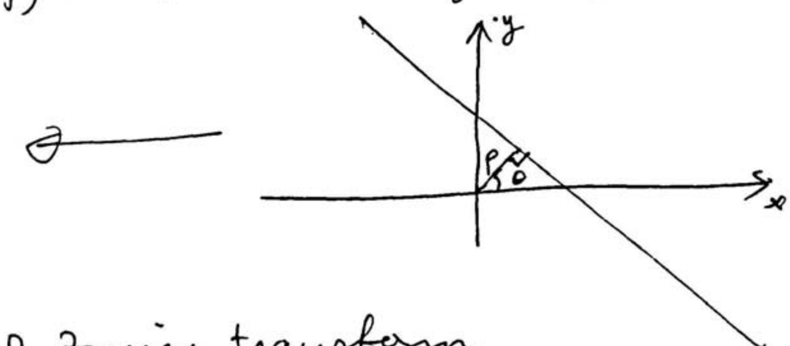
– T. Kailath

There were two questions. The second system is shown below. The first is on the attached sheet.

Ques. 2.

~~i)~~ Given a straight line in the x-y plane described by

$$f(x,y) = \delta(\rho - x\cos\theta - y\sin\theta)$$



find its 2D Fourier transform

$$F(u,v) = \iint_{-\infty}^{\infty} f(x,y)\, e^{-j2\pi(ux+vy)}\, dx\, dy$$

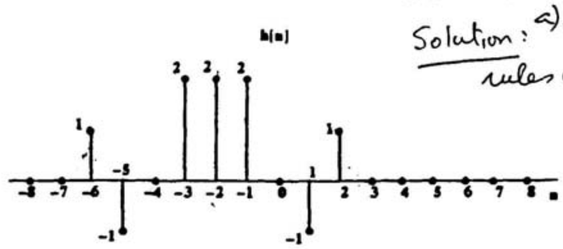
Solution: By using the Projection-Slice Theorem or by (longer)

~~ii) What happens if the line has finite extent?~~

direct computation, the answer is

$$\frac{1}{|\cos\theta|} e^{-j2\pi\rho u/\cos\theta}\, \delta(v - u\tan\theta).$$

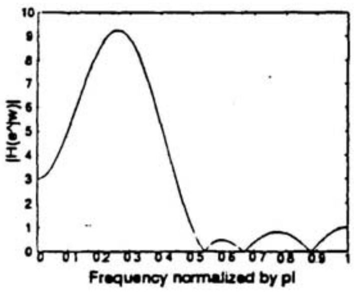
1. The nonzero values of the impulse response of a system $h[n]$, are given below.



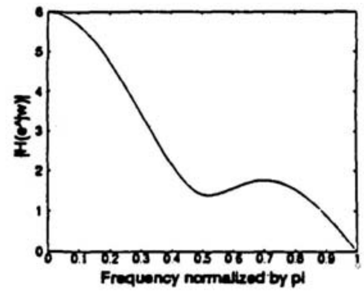
Solution: a) $H(0) = 6$ rules out A
$H(\pi) = 2$ rules out C.

b) group delay $= \frac{d\phi(\omega)}{d\omega} = -2$.

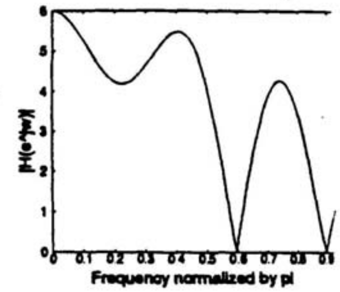
(a) One of the plots below, labeled A through C, is the Fourier Transform magnitude of $h[$ shown above. Which of these is the correct plot? Explain your reasons for choosing th plot over the others.



A

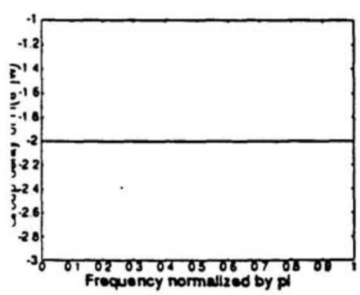


B

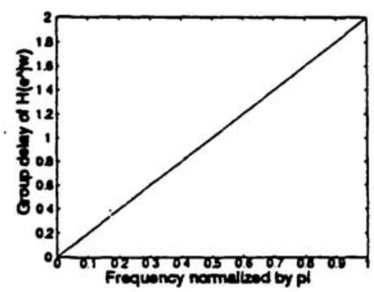


C

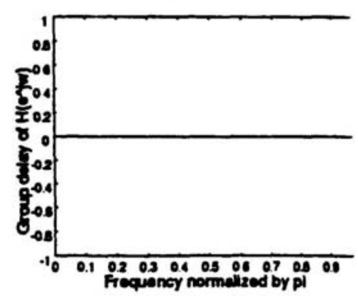
(b) One of the plots below, labeled D through F, is the group delay of the Fourier Transfor of $h[n]$ shown above. Which of these is the correct plot? Again, explain your reasons f choosing this plot over the others.



D



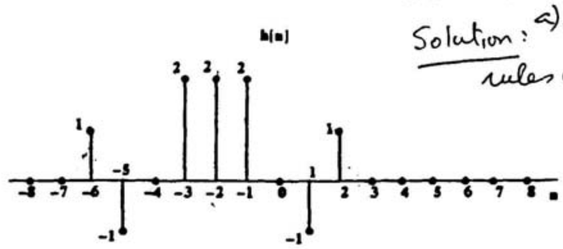
E



F

$$H(\omega) = \sum_{-\infty}^{\infty} h(n) e^{-j\omega n} \quad , -\pi \leq \omega \leq \pi .$$

1. The nonzero values of the impulse response of a system $h[n]$, are given below.



Solution: a) $H(0) = 6$ rules out A

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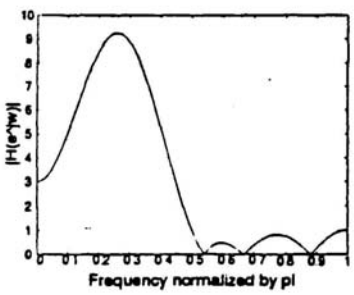
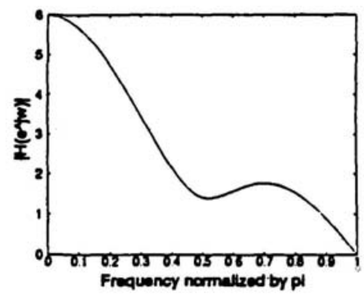
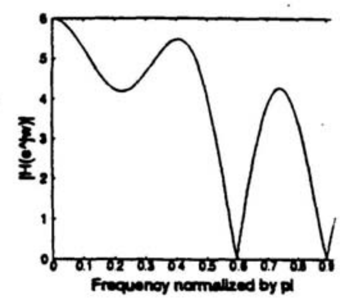


A



B



C

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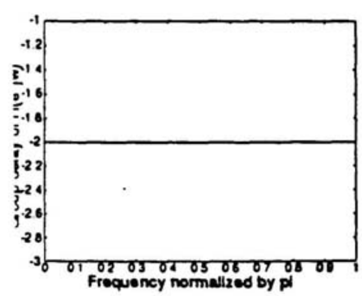
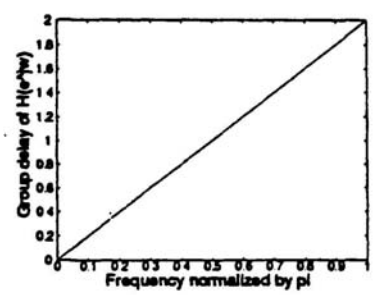
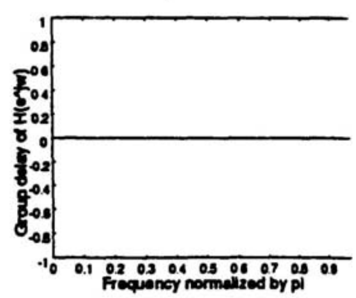


D



E



F

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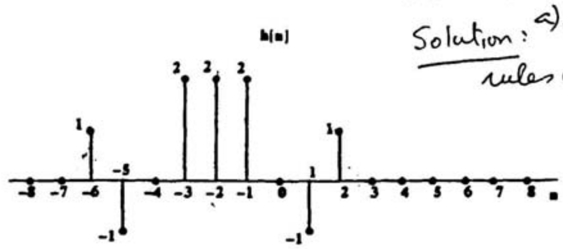

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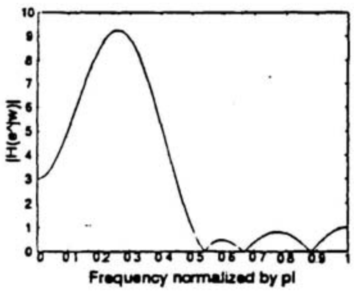
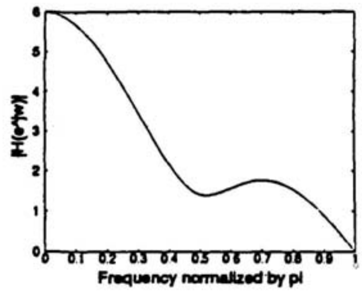
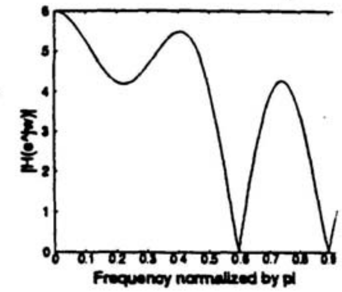


A



B



C

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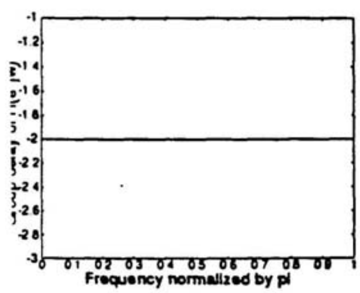
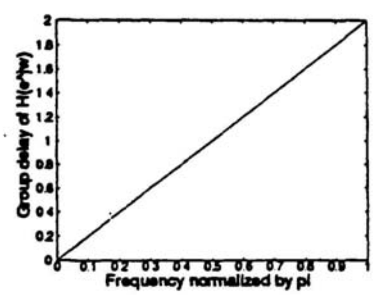
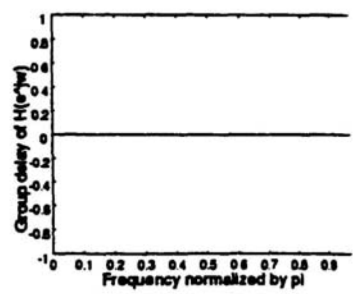


D



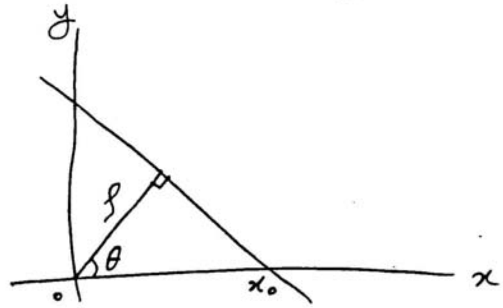
E



F

$$H(\omega) = \sum_{-\infty}^{\infty} h(n) e^{-j\omega n} \quad , -\pi \leq \omega \leq \pi .$$

* F.T. of an Ideal Straight Line



$x_0 = \frac{\rho}{\cos\theta}$

$$f(x,y) = \delta(\rho - x\cos\theta - y\sin\theta)$$

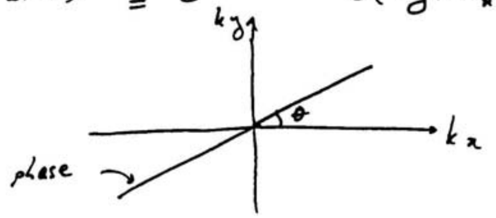
$$\mathcal{F}\{f(x,y)\} = \int_y \int_x \delta(\rho - x\cos\theta - y\sin\theta)\, e^{-j2\pi(k_x x + k_y y)}\, dx\, dy$$

integrate over x: (δ nonzero at $x = \frac{\rho}{\cos\theta} - y\tan\theta$)

$$(*) = \int_y e^{-j2\pi k_x\left(\frac{\rho}{\cos\theta} - y\tan\theta\right)} \cdot e^{-j2\pi k_y y}\, dy$$

$$= e^{-j\frac{2\pi\rho k_x}{\cos\theta}} \int_y e^{-j2\pi(k_y - k_x\tan\theta)y}\, dy$$

$$= e^{-j\frac{2\pi\rho k_x}{\cos\theta}} \cdot \delta(k_y - k_x\tan\theta) = e^{-j2\pi x_0 k_x}\, \delta(k_y - k_x\tan\theta)$$



* If there are 2 parallel lines with angle θ and offsets x_{0_1}, x_{0_2} (or distances from origin $\rho_1, \ldots$)

$\Rightarrow$ The difference will be that an interference pattern appears in the F. domain on the line.

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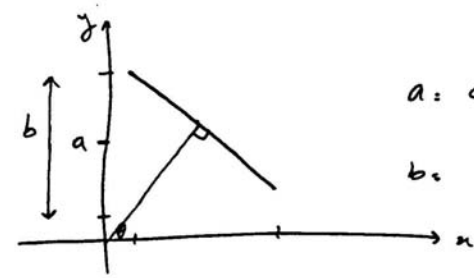
* More realistically, assume the line has finite length:



a: center point y-axis
b: length along y-

$$f(x,y) = \delta(\rho - x\cos\theta - y\sin\theta)\, \Pi\left(\frac{y-a}{b}\right)$$

where $\Pi(x) = \begin{cases} 1 & \frac{1}{2} < x \\ 0 & \text{other} \end{cases}$

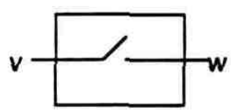
$$\mathcal{F}\{f(x,y)\} = \left[e^{-j2\pi x_0 k_x}\,\delta(k_y - k_x\tan\theta)\right] * \left[e^{-j2\pi a k_y}\,\text{Sinc}(bk_y)\,\delta(k_x)\right]$$

$$= e^{-j2\pi x_0 k_x} \cdot e^{-j2\pi a(k_y - k_x\tan\theta)}\,\text{Sinc}\left[b(k_y - k_x\tan\theta)\right]$$

Who ?

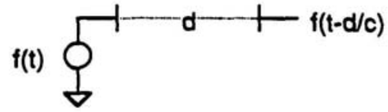
FOR THE FOLLOWING SYSTEMS, INDICATE WHETHER THE SYSTEM IS: a. LINEAR
b. TIME-INVARIANT

1.) $w = dv/dt$
2.) $w = v^2$
3.) $w = v\cos\omega t$
4.) $w = vt$
5.) $w = vt^2$
6.) $w = \mathcal{F}\{v\} = \int v(t)e^{-j\omega t}dt$
7.) $w = v^2 t$
8.) $w = v(t)rect\frac{t}{T}$
9.) $w = v(t) * rect\frac{t}{T}$

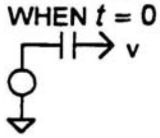


10.) Switch is closed at t=T
11.) Switch is closed at v=V

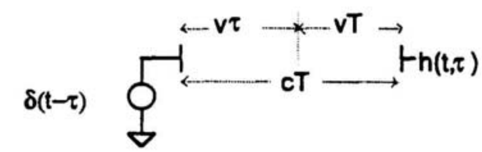
A SIGNAL IS SENT FROM A SENDING STATION TO A RECEIVING STATION WITH THE SIGNAL MOVING AT A VELOCITY c. FOR A STATIONARY RECEIVER THE SIGNAL IS DELAYED BY d/c.



THE RECEIVING STATION MOVES AT A VELOCITY v, STARTING AT d=0 WHEN $t = 0$, WHERE c>v.



A DELTA FUNCTION IS APPLIED AT $t = \tau$. THE IMPULSE TRAVELS TO THE RECEIVER IN TIME T.



FIND THE IMPULSE RESPONSE $h(t, \tau)$.

IS THIS SYSTEM TIME INVARIANT?

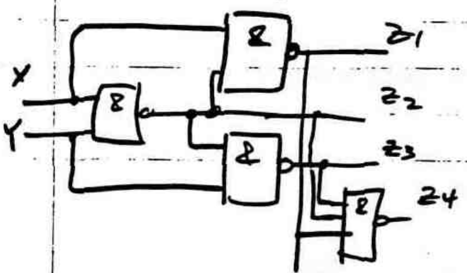
FIND THE RESPONSE TO AN INPUT $\cos \omega t$.

2-94 Quals

EJ McCluskey

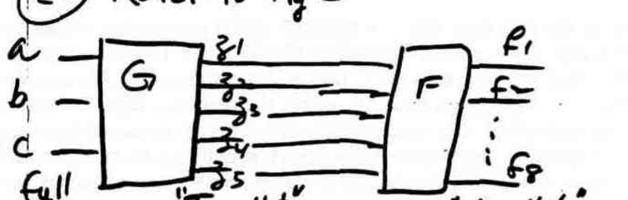
Computer Architecture

(1) Warmup - Circuit in Fig 1. is standard library (ASIC) function - Can you identify it



ANS
Decoder

(2.) Refer to Fig 2



full binary

"Two Hot" code

"One Hot"

a b c	"Two Hot" code	"One Hot"
0 0 0	11000	10000000
0 0 1	10100	01000000
0 1 0	⋮	⋮
0 1 1		00000001
1 0 0		
1 0 1		
1 1 0		
1 1 1		

Part (i) design an efficient circuit for F (converts from 2-hot to 1-hot codes) $\boxed{\text{Ans: } f_1 = z_1 z_2, \cdots}$

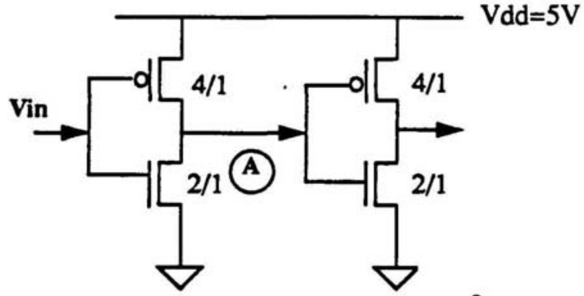
(ii) design an efficient circuit for G:

ANS $z_1 = c'$, $z_2 = a'b' + bc$, $z_3 = a'b$, $z_4 = ab + ac$, $z_5 = ab' + b'c$

(3) Draw counters for the following moduli: 4, 3, 5, 127 $\boxed{\text{Ans: See Text}}$

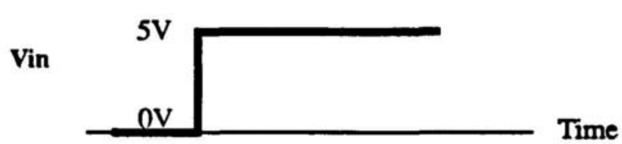
Teresa Meng

1. Delay and Output Stage



Process transconductance: $k_N' = 20\ \mu A/V^2$ and $k_P' = 10\ \mu A/V^2$.

Input waveform:



The first question was on the gate capacitance in standard CMOS technology. Amazing enough most students' answers were off by at least three orders of magnitude. The second question was to calculate the gate delay from the input to node A. Here is just one example. Assume that the gate capacitance was about 10 fF, and that the transistor was in the saturation region to simply calculation. The answer was around 0.2 ns. The third question was to design an output stage that aims at minimizing the delay of driving a capacitance 1000 times bigger than gate capacitance. The students were asked to give an estimate on the number of stages needed, and the total delay, etc. The last question was on the effect of inductance on ground noise.

Teresa Meng

2. Input Offset Voltage

For the circuit below, assume that the only offset contribution is from a mismatch in the W/L of the input transistors. Neglect transistor output resistance.

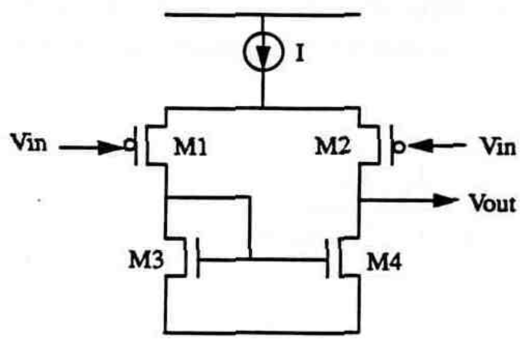


Table 1: Parameters affecting the input offset voltage

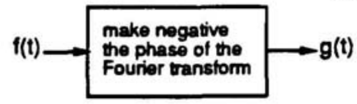
Parameters	Increase	Decrease	Does not Change
W1, W2	✓		
L1, L2		✓	
W3, W4			✓
L3, L4			✓
Current I		✓	
Oxide thickness t_{ox}		✓	

The question was to change the design parameters listed in the above table so that the input offset voltage as introduced by the mismatch in the input transistors' W/L ratio will be minimized. The students were asked to give reasons for their choice of actions. (This problem is actually much simpler than most students thought.)

Nishimura

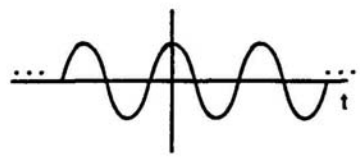
system

Problem 1

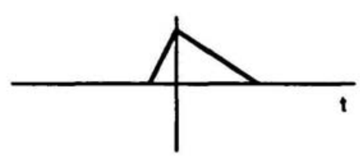


Input signal $f(t)$ and output signal $g(t)$ differ only in the phase of their Fourier transforms. The phases are the negative with respect to each other. Sketch the output for the following inputs:

a)



b)

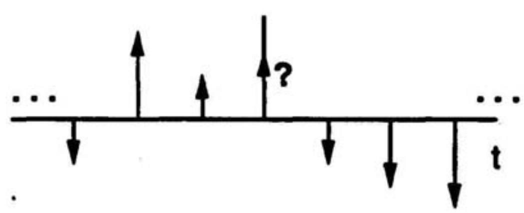


Is this system causal?

Is this system linear?

Is this system time-invariant?

Problem 2



A signal $f(t)$, bandlimited to ± 1 kHz, is sampled at the rates given below.

If the $t = 0$ sample is lost, is it possible to restore this sample point? If yes, describe how. If not possible to restore, why not?

a) Sampling rate = 5 kHz

b) Sampling rate = 2.5 kHz

c) Sampling rate = 1.25 kHz

Nishimura

Solutions

Problem 1

Multiplying the phase of the Fourier transform by -1 amounts to conjugating the spectrum; hence,

$$G(s) = F^*(s) \, .$$

Therefore

$$g(t) = f^*(-t) \, .$$

If the input is real-valued, the output is a time-reversed version. The system is non-causal, non-linear, and time variant. If the input is constrained to be real valued, the system behaves linearly.

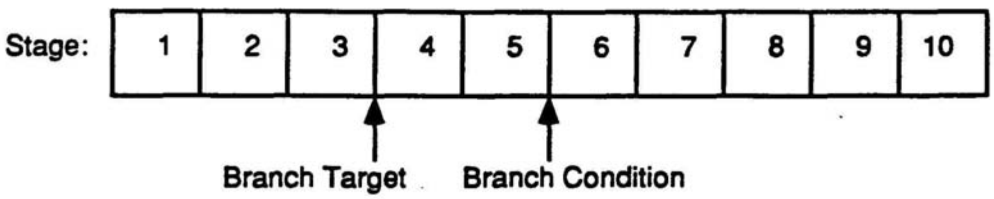
Problem 2

(a) Yes, one can throw away every other sample and still exceed the Nyquist frequency.

(b) Yes, model the "defective" sampling by $f(t)[1/T\,\mathrm{comb}(t/T) - \delta(t)]$. The transform of this quantity is the original replicated spectrum shifted vertically based on the value $f(0)$. Therefore, $f(0)$ can be determined by evaluating the spectrum in the gaps between replication islands since the values there should be zero. A time domain approach is possible too.

(c) No, not possible because of aliasing.

Kunle Olukotun
January 1994 Quals Questions



Question:

For this pipeline what percentage of branches must be taken so that the "always taken" and "always not taken" schemes have the same performance

Answer:

To solve this problem we need to equate the branch penalties for each prediction scheme. If t is the percentage of taken branches, then we get the following equation:

Always taken		Always not taken
$2t + 4(1 - t)$	$=$	$4t$

The rest is algebra

$$t = 66.7\ \%$$

PROF. A. PAULRAJ

Let $v(t)$ be a continuous-time zero mean stationary Gaussian random process with variance σ_v^2. Let $x(t)$ be a sample and held process obtained from $v(t)$.

$$x(t) = \sum_{n=-\infty}^{n=\infty} v(nT)h(t-nT)$$

where T is the sampling period and $h(.)$ is the holding pulse

$$h(t) = \begin{cases} 1 & \text{for } 0 \leq t < T \\ 0 & \text{otherwise} \end{cases}$$

If T is sufficiently large to render adjacent time samples $v(nT)$ statistically independent

1. Sketch $x(t)$
2. Is $x(t)$ stationary
3. Is $x(t)$ Gaussian
4. Derive $\mathbf{K}_x(t_1, t_2)$, the autocovariance of $x(t)$
5. Plot $\mathbf{K}_x(t_1, t_2)$ on a (t_1, t_2) plane
6. Is $x(t)$ cyclostationary

Consider now a randomized version of $x(t)$, say $y(t)$, where the periodic sampling clock is random and uniformly jittered within one clock period.

$$y(t) = \sum_{n=-\infty}^{n=\infty} v(nT+\alpha)h(t-nT-\alpha)$$

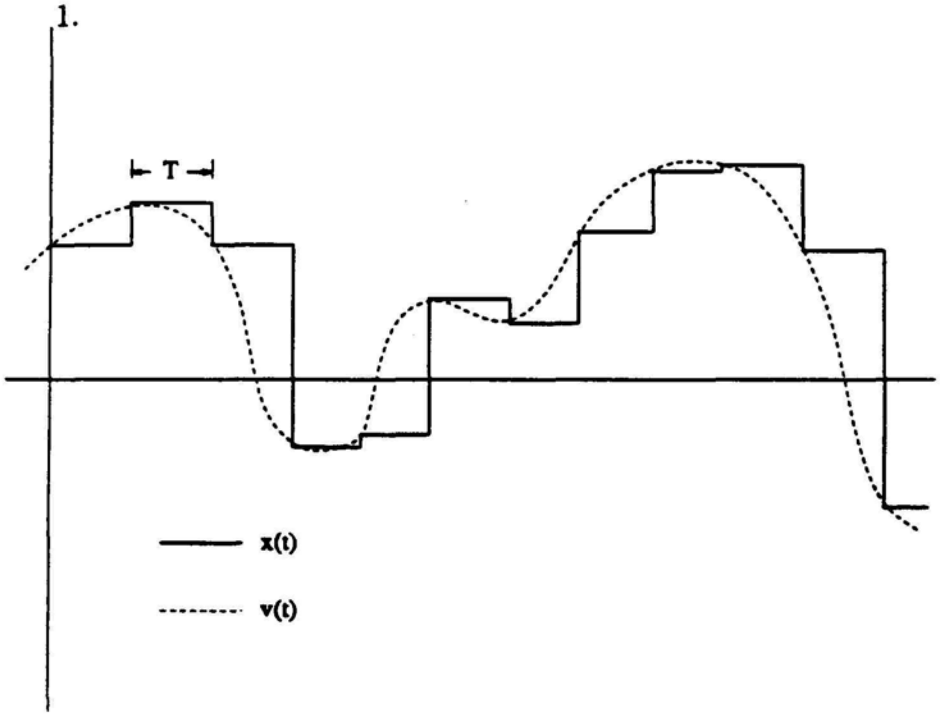
$$p_\alpha(u) = \begin{cases} \frac{1}{T} & \text{for } -\frac{T}{2} \leq u < \frac{T}{2} \\ 0 & \text{otherwise} \end{cases}$$

1. Sketch $y(t)$
2. Is $y(t)$ stationary
3. Derive $\mathbf{K}_y(t_1, t_2)$, the autocovariance of $y(t)$
4. Plot $\mathbf{K}_y(t_1, t_2)$ on a (t_1, t_2) plane
5. Is $y(t)$ cyclostationary.

Prof. Arogyaswami J. Paulraj Feb. 2, 1994

Problem I

1.



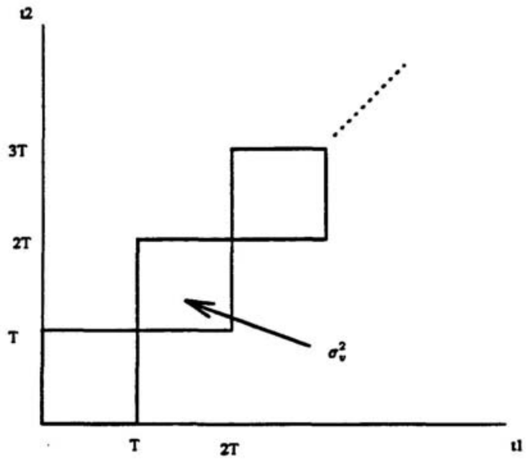
2. No

3. Yes

4.

$$\mathbf{K}_x(t1,t2) = \begin{cases} \sigma_v^2 & \text{if } nT \le t_1, t_2 < (n+1)T \text{ for any } n \\ 0 & \text{otherwise} \end{cases}$$

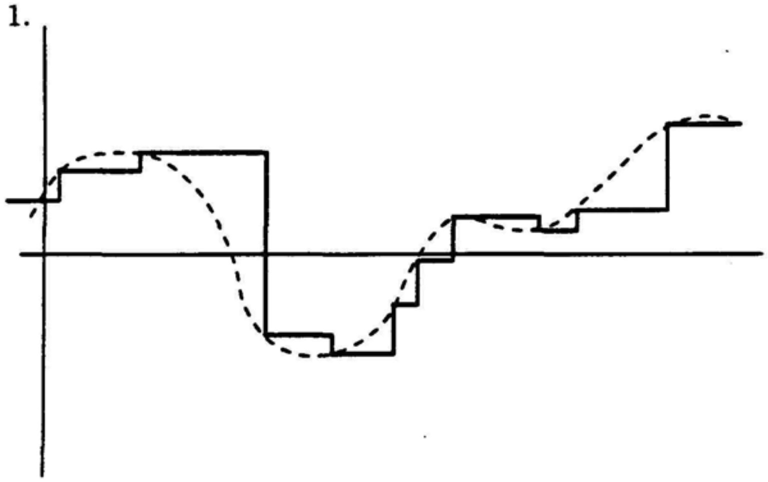
5.



6. Yes

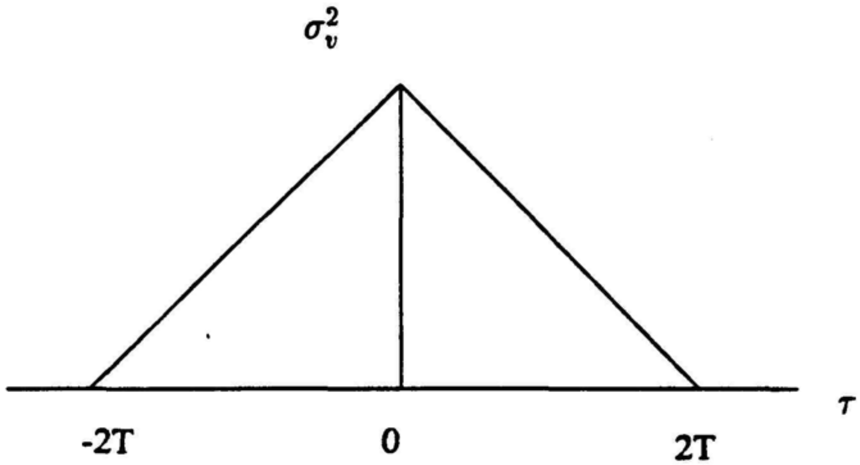
Problem II

1.



2. Yes

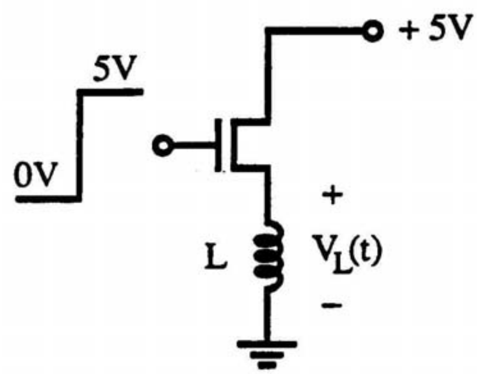
3.



$$\begin{aligned} \mathbf{K}_x(t1, t2) &= \mathbf{K}_x(t1 - t2) \\ &= \mathbf{K}_x(\tau) \end{aligned}$$

Qualifying Exam 1994
Bruce Wooley circuits

For the following circuit the input is stepped form 0V to 5V at $t = 0$. The transistor is an NMOS device with a threshold of $V_T = +1V$.



What happens to the voltage across the inductor for $t > 0$, assuming $V_L(t) = 0$ for $t < 0$?

The question evolves in a fashion that depends on the student's background and ability to first describe the circuit's behavior qualitatively.