Specialized Subjects

9:00～11:30, Tuesday, August 23, 2011

Instructions

1. Do not open this booklet before permission is given.

2. This booklet contains 5 problems. The number of pages is seven excluding this cover sheet and blank pages. If you find missing or badly printed pages, ask the attendant for exchange.

3. Answer three problems. You can select any three out of the five. Your answer to each problem should be written on a separate sheet. You may use the reverse side of the sheet if necessary.

4. Fill the top parts of all your three answer sheets as instructed below. Before submitting your answer sheets, make sure that the top parts are correctly filled.

専 門 科 目

第 問

↑
Write the Problem No.

受験番号				

↑
Write the Examinee No.

5. The three answer sheets must be submitted at the end of the examination, even if they are blank ones.

6. You must answer either in Japanese or in English.

7. This booklet and the preparation sheet must be returned at the end of the examination.

8. This English translation is informal but provided for the convenience of applicants. Japanese version is the formal one.

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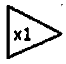
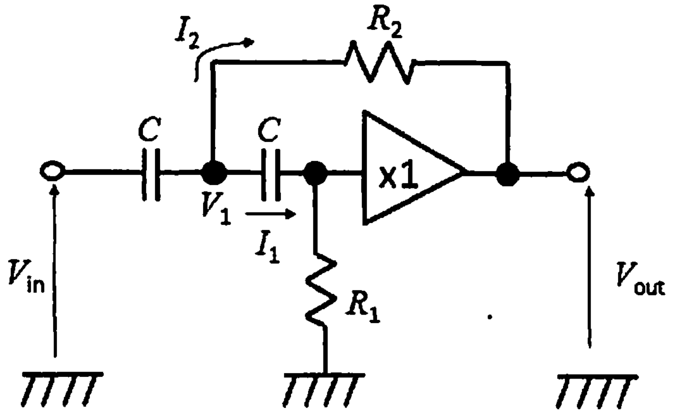
Problem 1

In the figure below, we assume 

represents an ideal voltage amplifier with voltage gain 1, input impedance ∞, and output impedance 0.



(1) We want to obtain the voltage transfer function $H(s)= V_{out}(s)/ V_{in}(s)$. For this purpose, formulate circuit equations established among voltages V_{in}, V_1, V_{out} and currents I_1, I_2 in the complex frequency domain s (or Laplace transform domain).

(2) Using the result of the above question, obtain the voltage transfer function $H(s)= V_{out}(s)/ V_{in}(s)$.

(3) When a unit step voltage u(t) is applied to the input voltage V_{in} of the figure, show the relations among C, R_1, and R_2 when the output voltage V_{out} becomes oscillatory, critical and non-oscillatory. Refer to the Laplace transform table shown below.

(4) Obtain the time waveform of the output voltage V_{out} when a unit step voltage is applied to the input V_{in} and C=0.001[F], R_1=100[kΩ], R_2=50[Ω].

Laplace Transform Table

f(t)	F(s)
u(t) : unit step function	1/s
e^{-at}	1/(s+a)
$\sin\omega t$	$\omega/(s^2+\omega^2)$
$\cos\omega t$	$s/(s^2+\omega^2)$
te^{-at}	$1/(s+a)^2$
$e^{-at}\sin\omega t$	$\omega/\{(s+a)^2+\omega^2\}$
$e^{-at}\cos\omega t$	$(s+a)/\{(s+a)^2+\omega^2\}$
$(e^{-at}-e^{-bt})/(b-a)$	$1/\{(s+a)(s+b)\}$
$e^{-at}(1-at)$	$s/(s+a)^2$

Problem 2

Answer the following questions on data dependency and instruction-level parallelism.

(1) Data dependencies that influence instruction-level parallelism are categorized into four types, i.e., flow, anti, input, and output dependencies. Explain these four types of dependencies.

(2) Indicate all the types of dependencies that cause precedence constraint among the four types in (1).

(3) Data dependencies are also categorized into true ones and false ones. True dependency represents data transfer between instructions, whereas false one doesn't. Answer if each of the dependencies in (2) is true one or false one.

The code below is assembly code of a processor. In this code, L1 to L4 indicate labels. And, r0 to r2 indicate registers, and the register in the left-hand side of each assignment operator "=" indicates destination and the registers in the right-hand side indicate sources. L1:ld is a load instruction, which accesses the main memory with the contents of r0 as the target address. L2:sla and L3:sla are shift-left-arithmetic instructions.

```
L1: ld   r1 = *r0;
L2: sla  r2 = r1 << 1;
L3: sla  r1 = r1 << 2;
L4: add  r1 = r1 + r2;
```

(4) Answer all the pairs of instructions with flow dependencies in the code. Also answer all the pairs of instructions with anti dependencies.

(5) Answer a method to solve false dependencies.

Then, explain how the instruction-level parallelism of the code above is improved when the method is applied.

Problem 3

Answer the following questions on database management systems.

(1) Discuss the differences between file systems and database management systems.

(2) Describe the features of SQL language in a relational database system.

(3) Consider a database for a book lending system in a library. The database comprises of the following three relational tables.

Users (UserID, Name, Address, Phone_Number),

Books (BookID, Title, Author, Publisher),

Lending_Status (UserID, BookID, Lent_Date).

Here, users have been assigned their user IDs through the registration process. The relational table Users manages the name, the address, and the phone number for each user. The relational table Books is for managing information of the books in the library, each of which is identified by a unique number BookID. In addition, the Books table contains other kinds of information such as book title, author name, and publisher name. Here, we assume that each book has one author. The relational table Lending_Status manages when and which book has been lent to whom.

Describe an SQL statement for enumerating the name and the phone number of users who borrowed books published by publisher "A" no less than 10 days before. Here, we assume that the subtraction operation on date-type data is available.

(4) Represent the answer of (3) in relational algebra. Discuss the relationship between the order of executing operations and the time required to execute the query.

(5) We want to derive a table (popular author list) which lists authors according to the total lending count in descending order. How should we change the schema? Describe an SQL statement for deriving that table.

Problem 4

Consider the digital transmission with the radio signal $s(t)$ shown in the equation (4.1).

$$s(t) = a_m(t)\cos[2\pi\{f_c + f_m(t)\}t + \varphi_m(t)] \qquad (4.1)$$

where, f_c is the carrier frequency.

(1) Answer the name of the digital transmission, where the radio signal $s(t)$ is given by

$$s(t) = A\cos[2\pi f_c t + \varphi_m(t)] \quad A: \text{constant value} \qquad (4.2)$$

(2) Equation (4.2) can be expressed by the equation (4.3),

$$s(t) = \mathrm{Re}[z(t)e^{j\,2\pi f_c t}] \quad \text{where } z(t) = Ae^{j\,2\pi\varphi_m(t)} \qquad (4.3)$$

using the real part (Re) of complex function. The equation (4.3) can be expressed as vector notation, shown in Fig. 1. Hereafter, the point (●) in Fig.1 is called the signal point.

Consider the scheme shown in the equation (4.4), where four signal points are assigned to four couples of bits, "00", "10", "11", "01", as follows.

$$\left.\begin{array}{lll} \mathrm{Re}[z(t)] = A/\sqrt{2} & \mathrm{Im}[z(t)] = A/\sqrt{2} & (\text{for "00"}) \\ \mathrm{Re}[z(t)] = -A/\sqrt{2} & \mathrm{Im}[z(t)] = A/\sqrt{2} & (\text{for "10"}) \\ \mathrm{Re}[z(t)] = -A/\sqrt{2} & \mathrm{Im}[z(t)] = -A/\sqrt{2} & (\text{for "11"}) \\ \mathrm{Re}[z(t)] = A/\sqrt{2} & \mathrm{Im}[z(t)] = -A/\sqrt{2} & (\text{for "01"}) \end{array}\right\} (4.4)$$

Explain the scheme shown in equation (4.4), within 100 words.

(3) Consider two kinds of schemes which assign four bits to each signal point.

In the scheme 1, shown in Fig. 2, 16 signal points are equally placed on a circle with radius A. The mean square of the amplitude is A^2 and the distance of neighboring two signal points is $2A\sin(\pi/16)$ ($\fallingdotseq 0.39A$) .

In the scheme 2, shown in Fig. 3, 16 signal points are differently placed from scheme 1. Explain the scheme 2, using the equation (4.1).

(4) Obtain mean square of the amplitude for scheme 2. In the calculation, suppose that the probability of each signal point is identical.

(5) Explain superiority of scheme 2 compared to scheme 1.

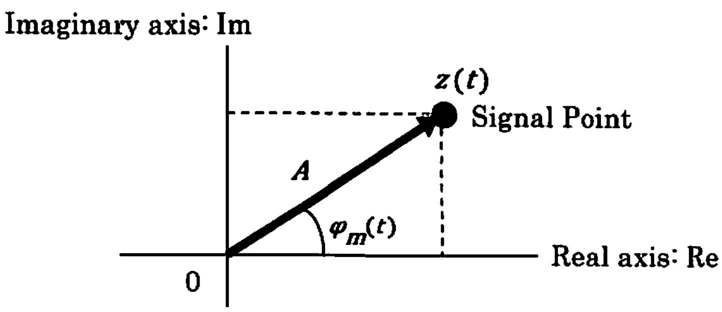


Figure 1

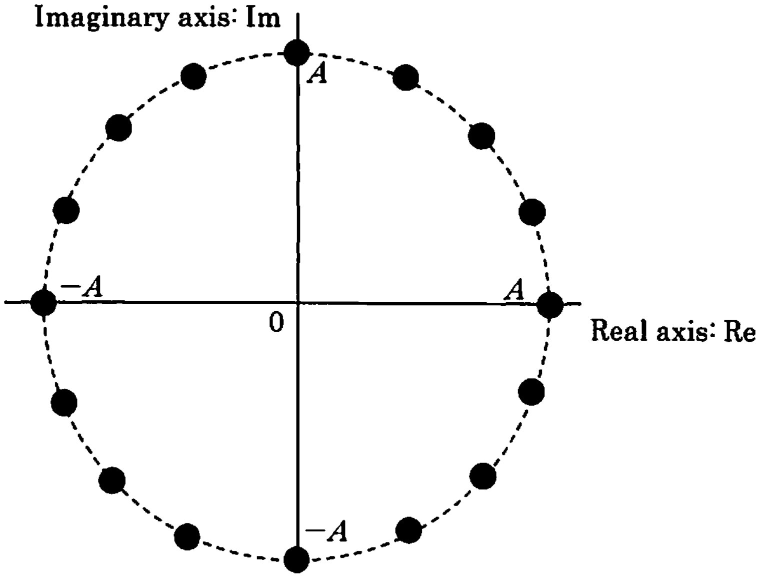


Figure 2

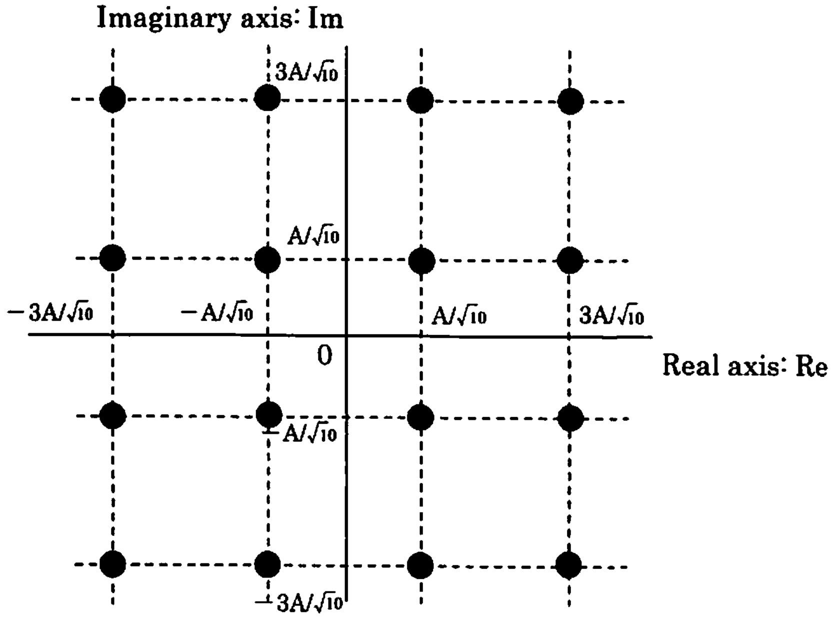


Figure 3

Problem 5

Figure 1 describes a process of generation of source signal $g(t)$, its transformation by a time-invariant filter, whose impulse response is $h(t)$ and frequency characteristics are $H(\omega)$, observation of intermediate signal $s(t)$, addition of noise $n(t)$, and reception of output signal $o(t)$.

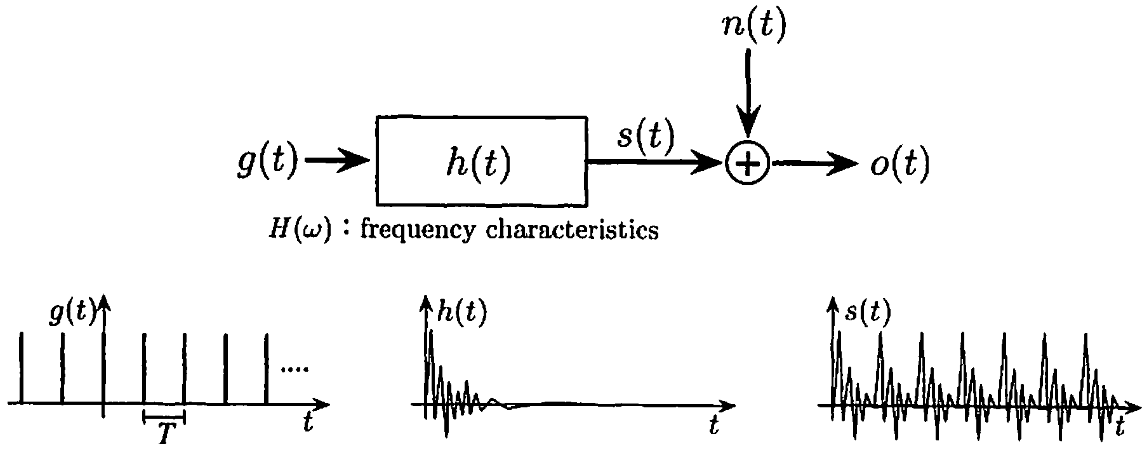


Figure 1

The source signal $g(t)$ is a sequence of impulses. By convoluting $g(t)$ with $h(t)$, $s(t)$ is obtained. With the noise $n(t)$ added to $s(t)$, $o(t)$ is received.

$$\begin{aligned} o(t) &= s(t) + n(t) \\ &= g(t) \otimes h(t) + n(t) \quad (\otimes \text{ is an operator of convolution.}) \end{aligned}$$

Answer the following questions. You can refer to the Fourier Transform table below.

$f(t)$	$\leftrightarrow$	$F(\omega)$
$\delta(t)$		1
1		$2\pi\delta(\omega)$
$f(t-t_0)$		$F(\omega)e^{-j\omega t_0}$
$e^{j\omega_0 t}f(t)$		$F(\omega-\omega_0)$

(1) Spectrums (Fourier Transforms) of $g(t)$, $h(t)$, $n(t)$, and $o(t)$ are denoted as $G(\omega)$, $H(\omega)$, $N(\omega)$, and $O(\omega)$, respectively. Show $O(\omega)$ mathematically by using $G(\omega)$, $H(\omega)$, and $N(\omega)$.

(2) $g(t)$ is a sequence of impulses and it can be represented as $g(t) = \sum_{k=-\infty}^{\infty} \delta(t-kT)$, where T is the sampling period. It is known that $G(\omega)$, $g(t)$'s Fourier Transform, also becomes a sequence of impulses. Explain this fact mathematically. You can use the fact that $g(t) = \frac{1}{T}\sum_{k=-\infty}^{\infty} e^{jk\omega_0 t}$ $(\omega_0 = \frac{2\pi}{T})$.

(3) $o(t)$ was received with no noise ($n(t) = 0.0$). Show mathematically the logarithmic amplitude characteristics of $O(\omega)$, $\log|O(\omega)|$, by using $G(\omega)$ and $H(\omega)$.

(4) AD conversion of $o(t)$ was done with no noise and discrete Fourier transform was performed over N samples around $t = t_0$. Figure 2 shows the resulting logarithmic spectrum $\log|O(\omega)|$. Considering the properties of $G(\omega)$ explained in (2) and those of $\log|O(\omega)|$ shown in (3), draw roughly $\log|H(\omega)|$ on the figure of $\log|O(\omega)|$ and explain why it is like that.

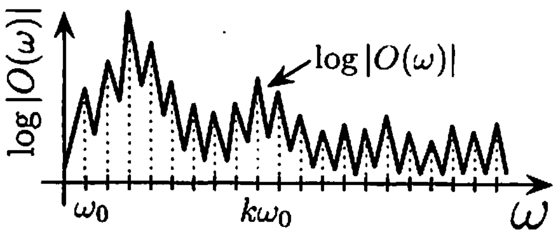


Figure 2

Through questions (1) to (4), the filter was treated as time-invariant. In the following question, it is treated as time-variant and its frequency characteristics at time t are denoted as $H(\omega, t)$. Here, $S(\omega)$ and $O(\omega)$ should also be treated as time-variant and they are denoted as $S(\omega, t)$ and $O(\omega, t)$.

(5) We want to estimate $|S(\omega, t_0)|$ from $|O(\omega, t_0)|$, which is the amplitude characteristics of discrete Fourier transform of N samples around $t = t_0$. With noise ($n(t) \neq 0.0$), however, this task is generally difficult. Even in this case, by assuming the following properties, it is possible to estimate spectrum $|S(\omega, t_0)|^2$ from $|O(\omega, t_0)|^2$. Fill in the following blanks.

- $|S(\omega, t)|$ and $|N(\omega, t)|$ are independent of each other.
- $|N(\omega, t)|$ is stationary and independent of t. We can measure the noise only again as needed.

Fourier transforms around $t = t_0$ are denoted as $O(\omega, t)$, $S(\omega, t)$, and $N(\omega, t)$. From $o(t) = s(t) + n(t)$, the following equation is obtained.

$$|O(\omega, t)|^2 = \left\{ \boxed{\text{(5-1)}} \right\}^2 + 2 \times \boxed{\text{(5-2)}} + \left\{ \boxed{\text{(5-3)}} \right\}^2$$

We perform the temporal expectation operation over the above equation. Due to independency between $|S(\omega, t)|$ and $|N(\omega, t)|$, the expected value of $\boxed{\text{(5-4)}}$ will be close to zero. By obtaining multiple and consecutive $|O(\omega, t)|^2$ around $t = t_0$ and calculating its expected value, we can estimate the expected value of $|S(\omega, t)|^2$ around $t = t_0$ as $\boxed{\text{(5-5)}}$.

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