台灣聯合大學系統107學年度碩士班招生考試試題

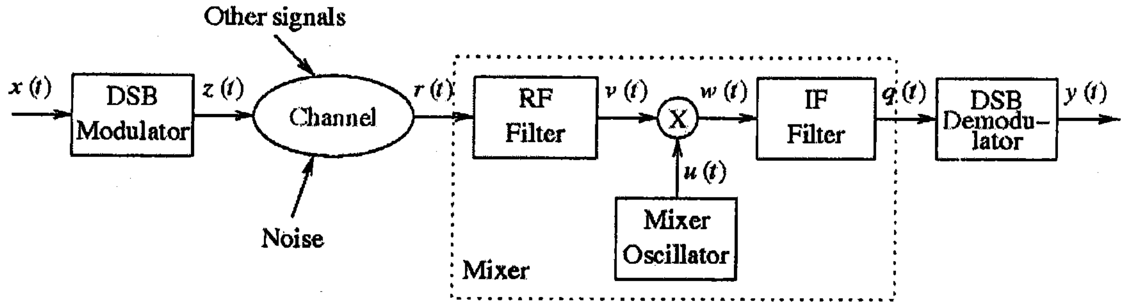
類組：電機類　科目：通訊系統(通訊原理)(300E)　　共 3 頁 第 1 頁

※請在答案卷內作答

參考用

Problems:

1. (Total = 30%) Consider the following transmission system, where DSB stands for "double sideband," RF stands for "radio frequency" or "radiation frequency," and IF stands for "intermediate frequency":



(a) (5%) Let the frequency spectrum of the continuous-time signal $x(t)$ be given by $X(f)=\frac{1}{2000}\Pi\left(\frac{f}{2000}\right)$ where $\Pi(f)$ is the "unit pulse function," i.e., $\Pi(f)=1$ for $|f|\leq 0.5$ and $\Pi(f)=0$ otherwise. Find $x(t)$ and express it in terms of the sinusoidal function and t. Note: Do *not* express it *directly* in terms of the sinc function.

(b) (6%) Following part (a), the DSB modulator output is given by $z(t)=A_c[1+ax(t)]\cos(f_c t)$ where f_c = 800 kHz, a = 1, and A_c = 2. Sketch the frequency spectrum $Z(f)$ of $z(t)$. Label your plot clearly so that one can reconstruct $z(t)$ exactly from your plot of $Z(f)$ with no uncertainty.

(c) (5%) Following part (b), let the channel and the RF filter be such that $v(t)$ = $z(t)$. Let the mixer oscillator output $u(t)=\cos(f_m t)$ where f_m = 900 kHz. Sketch the frequency spectrum $W(f)$ of $w(t)$. Label it fully as in the case for $Z(f)$ in part (b).

(d) (3%) Following part (c), consider the IF filter. Specify a suitable passband for it.

(e) (5%) Following part (d), let the DSB demodulator be an envelope detector consisting of an ideal diode, a capacitor, and a resistor. Sketch the envelope detector; indicate clearly where its input and output are. *In addition*, give a proper upper bound and a proper lower bound for the *RC* constant for the signal considered in this problem.

(f) (6%) Consider the envelope detector in your answer to part (e). Prove mathematically whether it (from $q(t)$ to $y(t)$) is a linear system.

2. (Total = 15%) Consider the 16-QAM signaling scheme where each modulated signal can be expressed by

$$s_i(t)=\sqrt{\tfrac{2}{T_s}}\cdot\left(A_i\cos 2\pi f_c t+B_i\sin 2\pi f_c t\right),\quad \text{for } 0\leq t<T_s, 1\leq i\leq 16,$$

where A_i and B_i take values on $\{-3a,-a,+a,+3a\}$ with equal probability, T_s is the symbol interval, and $f_c=\frac{1}{T_s}$ is the carrier frequency. The signal is transmitted over an AWGN channel with the signal model $y(t)=s(t)+n(t)$ where $s(t)\in\{s_1(t),\cdots,s_{16}(t)\}$ and $n(t)$ is AWGN independent of $s(t)$ with power spectral density $N_0/2$.

(a) (3%) Find the transmission bit rate of the system and express the average symbol energy in terms of a.

(b) (3%) In digital communications, it is common to convert the continuous waveforms to

equivalent discrete vectors with appropriate choice of basis functions in the vector space in which the inner product of any two waveforms $f(t)$ and $g(t)$ is defined as $\langle f(t), g(t)\rangle \triangleq \int_0^{T_s} f(t)g(t)dt$.
Suppose we choose $\phi_1(t) = \sqrt{2/T_s}\cos 2\pi f_c t$ as one basis function in the signal space of 16-QAM. Please find the other basis function $\phi_2(t)$ that is orthonormal to $\phi_1(t)$. Sketch the signal constellation using this basis.

(c) (4%) Suppose the 4-bit sequence 0000 is assigned to the symbol point $(-3a, -3a)$ and the 4-bit sequence 0001 is assigned to the symbol point $(-3a, -a)$. Please specify the 4-bit sequences to the remaining 14 symbol points using *Gray coding* on the constellation plot in Part (b).

(d) (5%) The optimum receiver for the communication system with 16-QAM consists of two parallel branches: the in-phase branch $Y_1 \triangleq \langle y(t), \phi_1(t)\rangle$ and the quadrature branch $Y_2 \triangleq \langle y(t), \phi_2(t)\rangle$, where the inner product is defined in Part (b). Please explain why the decoding of the optimum receiver can be carried out by considering these two branches *separately*.

3. (Total = 20%) Alice is sending a binary message X (X=+1 or X=−1 equally likely) to her new boyfriend Bob through an AWGN communication channel. However, Alice's ex-boyfriend Chuck is so jealous that he is jamming their communications by sending a binary interference Z (Z=+A or Z=−A equally likely) over the same frequency band. The signal Y that Bob is receiving can be modeled as $Y = X + Z + N$,
where N is AWGN with zero mean and variance σ^2, and X, Z and N are statistically independent. As Chuck is very upset, he is sending Z with very large A which is much greater than 1 and σ. Assume that Bob knows the value of A, but he doesn't know which one ($+A$ or $-A$) was sent.
(You may need this: the Q-function is defined as $Q(x) = \frac{1}{\sqrt{2\pi}}\int_x^\infty e^{-y^2/2}dy$.)

(a) (5%) Bob first employs a simple decision strategy by approximating $Z + N$ as Gaussian with zero mean and variance $A^2 + \sigma^2$. What is the optimum decoding strategy, in the sense of minimum probability of decision error, under the Gaussian approximation of $Z + N$? Please find the corresponding probability of error decision. (Express your answer in terms of the Q-function.)

(b) (5%) Actually the decoding strategy in Part (a) doesn't provide a good decoding quality. So, instead of treating Z+N as Gaussian, Bob tries the following new decoding strategy:

$$\begin{cases} \widehat{X} = +1, & \text{if } Y - A \geq 0 \text{ or } 0 \leq Y + A < A, \\ \widehat{X} = -1, & \text{if } Y + A < 0 \text{ or } -A < Y - A < 0, \end{cases}$$

where $\widehat{X}$ denotes the decoded result of X. Please find the probability of error decision for this new decoding strategy. (Express your answer in terms of the Q-function.)

(c) (5%) Please interpret the decoding strategy in Part (b). Explain why the decoding strategy in Part (b) performs better than that in Part (a).

(d) (5%) What is the best decoding strategy, in the sense of minimum probability of error decision, that Bob can come up with? Please provide details of your derivations.

4. (Total = 15%) A communication system is described by a discrete model as $y_n = x_n * h_n + w_n$ where n is the time index, y_n is the received sequence, x_n is the transmitted symbol

sequence, h_n is a length-L sequence representing the channel's impulse response, $*$ means convolution, and w_n is AWGN.

(a) (5%) The purpose of a zero-forcing equalizer is to achieve the zero-ISI condition by operating on the received sequence without considering the AWGN. Please formulate the math equations that may lead to the design of a linear filter as a zero-forcing equalizer.

(b) (5%) The purpose of an MMSE equalizer is to minimize the Mean Squared Error (MSE) between the actual transmit symbol and the equalized symbol. Please give a mathematical formulation for the MSE of a linear equalizer.

(c) (5%) Assume the channel's impulse response to be $h_n = [1, -0.5]$. Find the linear zero-forcing equalizer for it. Please express the equalizer in its impulse response.

5. (Total = 20%) A speech signal is properly sampled with a sampling rate of 8 kHz. Then each sample is quantized with $q = 2^n$ uniform quantization levels in which n is the word length.

(a) (5%) Determine the minimum n such that the quantization noise is within $\pm 0.25\%$ of the peak-to-peak full-scale value.

(b) (5%) Following part (a), the quantized speech signal (in binary bits) is to be transmitted over a QPSK system. Determine the minimum symbol rate of the QPSK system such that the quantized speech data can be transmitted in real time.

(c) (5%) Assume that the quantized speech data first goes through a compression operation such that the data rate is only 20% of the original rate. Does the compressed data stream have lower entropy per data bit than the original data stream has? You need to give detailed explanations.

(d) (5%) Furthermore, a rate 2/3 convolutional code is added to enhance the reliability of the transmission. Now determine the minimum symbol rate of the QPSK system to support the real-time transmission of the encoded and compressed speech data.