Q.2 a. Distinguish between source coding and channel coding, how Huffman codes are generated, give example?

Ans: Source coding is required to represent the information in a particular coding format. The objective is to reduce or eliminate redundancy, so that the decoder simply performs the inverse mapping and give the original data output. Huffman code is one of such codes. Channel coding helps in minimizing the effect of channel noise. This can be done by introducing redundancy. Channel coding can effectively be designed if channel characteristics are known.
Huffman code is a source code whose average word length is decided by the entropy of the memory less source. Huffman coding algorithm depends on:

1. The probability of source symbols. Lowest probability is assigned 0 and 1.
2. Symbols are listed in order of decreasing probability.
3. The lowest probability are added and to create new symbols.
4. The probabilities added are placed at the heighest point of the same probability.
5. The code is thus generated.

b. Describe signal processing operations in Digital Communication

Ans: **Page No 4, Article 1.2 of Textbook I**

Q.3 a. What is the difference between low pass sampling and band pass sampling. How reconstruction takes place?

Ans: In a low pass sampling the frequency/rate of sampling is = or > $2f_{max}$ because the LPF has one cotoff frequency. Once the signal modulates the carrier the filter has cutoffs –W and +W, thus the sampling becomes Qadrature sampling. (The student is expected to give explanation of these processes).

Page- 140- 143, Article 4.1 and 4.2 of Textbook by 'Simon Haykin'.

b. What is the need of sample and hold circuit, how does it help in quantization.

Ans: When sampling is carried out, the amplitude of the signal at that instant is converted into equivalent code bits. This process requires an ADC (PCM, DPCM, ADPCM etc) which have their own conversion time, hence the sampled amplitude should be kept at that value till conversion is over. This requires to hold the sampled amplitude. Hence the need of sample and hold circuit.

To quantize therefore sample and holding is requires.

Page- 160, Article 4.5 of Textbook by 'Simon Haykin'.

c. Two signals of 1KHz and 1.5KHz are to be transmitted over a common channel as TDM signals. What is the minimum sampling rate required.

Ans: Assuming that the same sample and hold is used to keep the transmission rate same the minimum sampling frequency will be twice the higher frequency that is 3KHz.

Q.4 a. What are the various noise that effect the performance of a digital system? Show that in a uniform quantizer the noise variance grows as square of step size. How this problem is taken care of?

Ans: Two types of noises that effect the performance are Quantization noise and channel noise. If Δ is the step size distinguishing the two quntizing levels then the decision of a level being interpreted as one code or next code if $-\Delta/2< q<\Delta/2$. The probability density function will be $f_o(q) =1/\Delta$ between $-\Delta/2< q<\Delta/2$ and 0 elsewhere

Therefore variance

$\sigma^2 = \int q2\, f_o(q)dq$ which will yield

$\sigma^2 = \Delta^2/12$

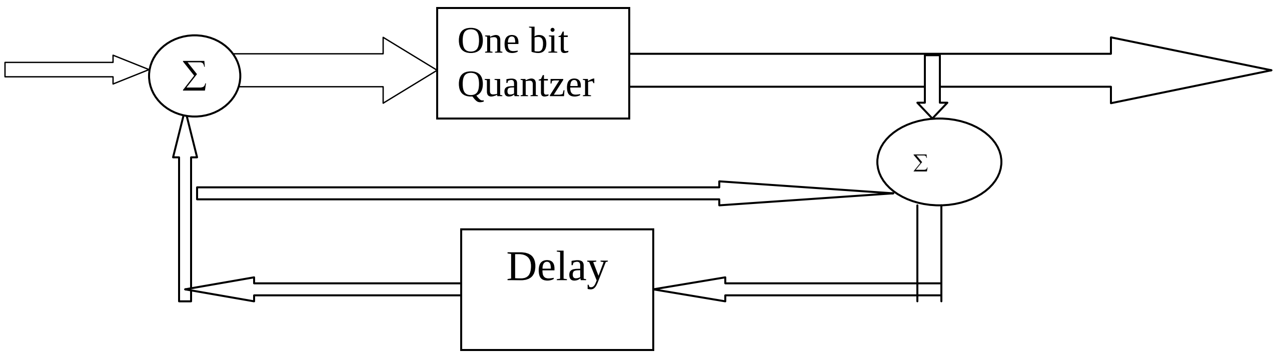
Page - 191/195 of Textbook 'Simon Haykin'.

b. Explain the process of Delta modulation. What are the various errors that occurs in this type of waveform coding?

Ans: DM provides a staircase approximation to the oversampled input base band signal. DM is a single bit quantizer which indicates δ change between the present and past sample value. It may be positive going or negative going.

The two types of distortion that happen are slope overload and granular noise.

Page - 205 of Textbook by 'Simon Haykin'



Q.5 **a**. What is the Nyquist criterion for distortion less base band transmission? If a 8 bit PCM voice data is to be transmitted in the TDM mode calculate the BW requirement when a cosine filter with roll off of 0.6 is used. Assume the frame period to be 125 microsec.

Ans: Each sample is 8bit and in TDM there are 24 channels
$Rb=(N_n +1)$ fs = (24* 8+1)

There are 24x8+1 bits which are transmitted in 125 microsec hence each bit= 0.647 microsec.

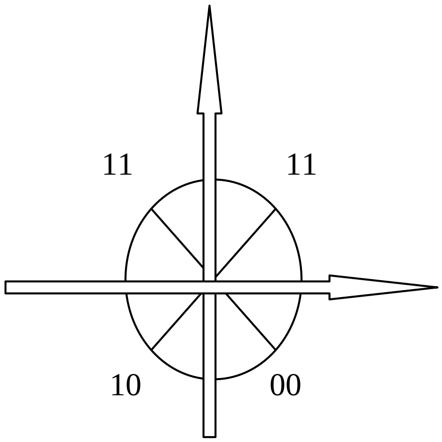
BW = $(1+\ )R_b/2 = 1.6 \text{ x}(1/2\text{x}T_b) = 1.6 \text{ x } (1/2\text{x}0.647)\text{-}=1.23$ MHz
If BPSK is used then BW= 2.46MHz

b. What is Inter symbol interference? Explain its effects and methods to reduce it.

Ans: **Page No 243 of Textbook I**

Q.6 **a**. How many message points does a QPSK represent, draw the signal space characteristic of a QPSK.

Ans: In QPSK there is four signal points represented by four phases. Each signal point represents a two-bit symbol. On a signal space diagram a unit circle represents the amplitude as there is no change in the amplitude of symbols but phases are represented by 45, 135, 225 and 315 degrees.



Page no. 252 of Textbook by 'Simon Haykin'

b. Find the probability of error if E_b/N_o requirement is 8dB in the case of BPSK and QPSK?

Ans: Derivation of BER.
8db gives a ratio E_b/No. of 6.309
For BPSK P_e=(1/2)erfs $\sqrt{\ }(E_b/No)$ =
From error function Table P_e= 2.51x10-4
For QPSK P_e= erfs $\sqrt{\ }(E_b/No)$ = 4.1x10-4

Page no. 289 of Textbook by 'Simon Haykin'

Q.7 **a**. What is a maximum- likelihood detector, explain its operation using a phasor diagram.

Ans: When a signal is received at the transmitter its characteristics change during the propagation due to various effects of the channel, thus it is difficult to identify a symbol. The average probability of symbol error during decision making P_e(mi, X)= P(mi, $X_{not\ sent}$) = 1-P(mi,X_{sent}) where X is an observation vector which needs to map $m^{\sim} = m_i$ the transmitted message.
The idea is to find that how close is $m^{\sim} = m_i$. this decision rule is refered to as maximum posteriori probability and given by
$P(m_{i,} X_{sent}) > P(m_{k,} X_{sent})$ for all k i
K= 1,2,….M

If probability of occurrence of symbol m_k is p_x and f_x (X/m_k) is the likely hood function and since we need to do comparison we need to maximize the ratio of probability of occurrence to joint probability $f_x(x)$

p_k $f_x(x/mi)/f_x(X/m_k)$ should tend to maximum for k=i
$f_x(x/m_k)$ is always negative.
In other words the received and transmitted symbol

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b. What are the properties of a matched filter, explain each one of them?

Ans: There are four properties:

- The output signal of a matched filter is proportional to the energy spectral density.
- The output signal of a matched filter is proportional to shifted version of auto correlation function.
- The output signal of a matched filter depends on the signal energy to noise spectral density of white noise.
- The matching must happen both in phase (which gives desired peak at time T) and amplitude, which gives optimum value of SNR.

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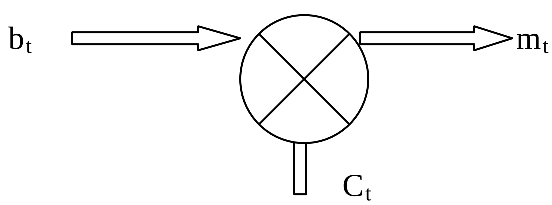
Q.8 a. What are the advantages of spread spectrum modulation? Show that the effect of interference is minimized in this technique.

Ans: Spread spectrum helps in improving the spectral efficiency, reduces noise density, natural security of information etc.



$M(t)= b(t)C(t)$

Received signal $r(t)=m(t)+i(t)= b(t)C(t) + i(t)$

Demodulated signal $z(t)= r(t)C(t) = c(t)b(t)C(t) + I(t)) = b(t) + c(t)i(t)$

The second term is a wide band signal while first tem narrow band which can be easily separated, hence interference is minimized.

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Q.9 **a**. What is the need of bit stuffing in multiplexer hierarchy?

Ans: Bit stuffing helps in synchronization as in the hierarchy rates of transmission may vary at different multiplexing points (Required to write the diagram of hierarchy)

Page no. 220 of Textbook by 'Simon Haykin'

b. What is meant by TDMA, how does it work?

Ans: TDMA is one of the multiple access techniques used in multiuser communication. TDMA stands for Time Division multiple access. In this every user or device/client is given a time slot for transmission of its information. This time slot can be sequential or otherwise. TDMA is essentially used in digital transmission or packet transmission.

TDMA has a frame structure that consists of overheads which gives information regarding synchronization, data rate, source, and destination, type of information etc. in the beginning of every data packet. The data packets are received by receiver in sequence.

c. Explain the use of spread spectrum in CDMA.

Ans: Code division multiple access is a application of SS where the SS signals from various users are transmitted with different PNs or random carrier hoppings to the same destination/ resource. This helps in spectrum reuse and improves the spectrum efficiency apart from the other SS advantages.

<u>TEXTBOOK</u>

Digital Communications, Wiley Student Edition, Simon Haykin