PANDIAN SARASWATHI YADAV ENGINEERING COLLEGE

DEPARTMENT OF ELECTRONICS & COMMUNICATION ENGG.

EC6501 DIGITAL COMMUNICATION

Branch: ECE **Semester: V**

Part A: Question & Answers

UNIT I SAMPLING & QUANTIZATION

1. What do you understand by the term aliasing

If the sampling frequency fs is less than 2W or condition is not satisfied (fs>2W), then an error produced it is called aliasing or folded over error.

2. What is the slope overload error? how it can be minimized?

Here the slope of the staircase approximation [m q(t)] is behind the slope of the sampled signal[m(t)].Here step size of m q(t) is too small. This m q(t) does not reach the m(t).so difference between m(t) and m q(t) is called slope overload error.

Elimination: To reduce this noise, the step size will be increased.

3. What are the drawbacks of PAM signal?

The amplitude of PAM signals changes according to the amplitude of modulating signal. Therefore like AM, the effect of additive noise is maximum in PAM. The added noise cannot be removed easily.

Due to the changes in amplitude of PAM pulses, the transmitted power is not constant.

The transmission bandwidth required for a PAM signal is too large as compared to the maximum frequency content x(t)

4. Advantages and Disadvantages of Digital Communication?

Advantages:

Immunity for Noise and Interference

It is possible to store the signal and process it further.

Communication can be kept “private” and “secured through the use of encryption.

Techniques such as data compression and image enhancement can be used.

Disadvantages:

Transmission bandwidth is increased to great extent.

System complex city is increased.

5. Define Quantization Error? [Or] Quantization noise?

The difference between the message signal x(t) and quantized signal **x q(t**) is called as Quantization error

or Quantization noise.
e = x q (t) - x(t)
where, **x(t)** –message signal and **x q(t**) **-** quantized signal

6. Why Quantization is required? (Or) Why do we need equalization in base band pulse transmission?

If we do not use the quantizer in the PCM transmitter, then we will have to convert each and every sampled value in to digital signal. So, It increases the large number of bits (i.e) Bit rate and Bandwidth requirement.
If we use quantizer means all the sampled values will be finally approximated into 256 distinct voltage levels. This reduces the number of bits per word .so this reduces bit rate and bandwidth requirement.

7. What is the Comparison between natural PAM and Flat top sampling? (MAY2010)

S.No	**Parameter**	**Natural PAM**	**Flat top PAM**
1.	Sampling rate	Satisfies Nyquist criteria	Satisfies Nyquist criteria
2.	Effect of Noise	Moderate	Moderate
3.	Circuit arrangement	Uses a chopper	Uses sample and hold circuit

8. Define Nyquist rate.
Let the signal be bandlimited to „W‟ Hz. Then Nyquist rate is given as,
Nyquist rate = 2W samples/sec
Aliasing will not take place if sampling rate is greater than Nyquist rate.

9. What is meant by aliasing effect?
Aliasing .effect .takes .place .when .sampling .frequency .is .less .than .Nyquist .rate. Under .such .condition, .the .spectrum .of .the .sampled .signal .overlaps .with .itself. Hence higher frequencies take the form of lower frequencies. This interference of the frequency components is called as aliasing effect.

10. Define PWM.
PWM is basically pulse width modulation. Width of the pulse changes according to amplitude of the modulating signal. It also referred as pulse duration modulation or PDM.

11. How the message can be recovered from PAM?
The .message .can .be .recovered .from .PAM .by .passing .the .PAM .signal .through reconstruction filter integrates amplitude of PAM pulses. Amplitude reconstruction signal is done to remove amplitude discontinuities due to pulses.

12. Write an expression for bandwidth of binary PCM with N messages each with a maximum frequency of *fm* Hz.
If „v‟ number of bits are used to code each input sample, then bandwidth of PCM is given as,
BT ≥ N.v.*fm*

Here v. *fm* is the bandwidth required by one message.

13. How is PDM wave converted into PPM message?
The PDM is signal is clock .signal to monostable multivibrator. The .multivibraor triggers on falling edge. Hence a PPM pulse of fixed width is produced after falling edge of PDM pulse. PDM represents the input signal amplitude in the form of width of the pulse. A PPM pulse is produced after the width of PDM pulse. In other words, the position of the PPM pulse depends upon input signal amplitude.

Part B

1. Explain the process of quantisation, encoding and decoding in PCM? In what way DPCM is better than PCM? (16)
Explain in adaptive pulse code modulation
Explain in typical power spectrum for speech signals
Auto correlation function for speech signals

2. Explain a uniform quantization process (8).
Explain in Linear quantization
And robust quantization

3. Explain the impulse sampling process and explain how to reconstruct the signal.(16)

Explain the block diagram of sampling
Explain low pass sampling
Draw that Discrete signal in time domain

4.Explain the aliasing effect and method to overcome it.(16)

Explain the aliasing effect and aliasing effect
Define distortion
Method of tackle aliasing

5. Explain the Time Division Multiplexing process in detail.(8)

Introduction in TDM system
Explain the block diagram

UNIT II WAVEFORM CODING

Part A

1. State Sampling theorem.

A band limited signal of finite energy, which has no frequency components higher than W Hz, may be completely recovered from the knowledge of its samples taken at the rate of 2W samples per second.

2. Mention the merits of DPCM.

1 .Bandwidth requirement of DPCM is less compared to PCM.
2. .Quantization error is reduced because of prediction filter
3. .Numbers .of bits used to represent .one sample .value are also reduced
compared to PCM.

3. What is the main difference in DPCM and DM?

DM encodes the input sample by one bit. It sends the information about + δ or -δ, ie step rise or fall. DPCM can have more than one bit of encoding the sample. It sends the information .about difference .between .actual sample value and the predicted sample value.

4. .Advantages and Disadvantages of “Delta Modulation”?

Advantages:

One bit is transmitted per sample. So speed will be high.
Because of single bit per sample Bandwidth is small
Less complicated transmitter and receiver circuit when compared to PCM circuits.

Disadvantages:

Two distortions are produced (i.e) Slope overload distortion and Granular noise.
Slope overload distortion will be much higher than that of PCM.

5. What are the two limitation of delta modulation?

Two two limitations of delta modulation (i.e.)

Slope overload distortion
Granular noise.

6. Application of PCM?

Telephone systems
Space communication systems

7. Mention the use of adaptive quantizer in adaptive digital waveform co ding schemes.

Adaptive quantizer changes its .step .size .according variance of .the .input signal.Hence quantization error is significantly reduced due to the adaptive quantization. ADPCM uses adaptive quantization. The bit rate of such schemes is reduced due to adaptive quantization.

8. What do u understand from adaptive coding?
In adaptive coding, the quantization step size and prediction filter coefficients are changed as per properties of input signal. This reduces the quantization error and number of bits to represent the sample value. Adaptive coding is used for speech coding at low bits rates.

9. What is meant by quantization?
While converting the signal value from analog to digital, quantization is performed. The analog value is assigned to nearest digital value. This is called quantization. The . quantized .value .is .then .converted .into equivalent .binary .value. .The quantization levels are fixed depending upon the number of bits. Quantization is performed in every Analog to Digital Conversion.

10. The signal to quantization noise ratio in a PCM system depends on what criteria?

The signal to quantisation noise ratio in PCM is given as, $(S/N)db \leq (4.8+6v)$Db Here v is the number of bits used to represent samples in PCM. Hence signal to quantization noise ratio in PCM depends upon the number of bits or quantization level.

11. What is meant by adaptive delta modulation?
In adaptive delta modulation, the step size is adjusted as per the slope of the input signal. Step size is made high if slope of the input signal is high. This avoids slope overload distortion.

12. What is the advantage of delta modulation over pulse modulation schemes?
Delta modulation encodes one bit per samples. Hence signalling rate is reduced DM

13. What should be the minimum bandwidth required to transmit a PCM channel?
The minimum transmission bandwidth in PCM is given as,
$BT = vW$
Here v is the number of bits used to represent one pulse.
W is the maximum signal frequency.

14. What is the advantage of delta modulation over PCM?
Delta .modulation .uses .one .bit .to .encode .on sample. .Hence .bit .rate .of .delta modulation is low compared to PCM.

15. What are the two limitations of delta modulation?
1 Slope of overload distortion.
2. Granular noise.

16. How does Granular noise occurs?
It occurs due to large step size and very small amplitude variation in the input signal.

17. What are the advantages of the Delta modulation?

1 Delta modulation transmits only one bit for one sample. Thus the signalling rate and transmission channel bandwidth is quite small for delta modulation.
2. .The .transmitter .and .receiver .implementation .is .very .much .simple .for .delta modulation. There is no analog to digital converter involved in delta modulation.

Part B

1. Draw the block diagram and explain the process of a PCM system in detail. (16)

Introduction in encoding technique for analog sources
Draw the block diagram
Explain the PCM in each block
Explain the noise consideration in PCM system

2. Compare the principles of Delta and adaptive delta modulation systems. (16)

Principle of delta modulation
Operation of DM transmitter
SNR calculation of DM
Explain in Adaptive Delta Modulation

3. Draw the block diagram of differential PCM and explain the function performed by each block. (16)

Draw the block diagram
Explain in Adaptive Differential pulse code modulation in each block

4. Explain the Linear Predictive coding in detail. (16)

Explain the concept
Define the speech synthesizer
Explain the block diagram of Transmitter and Receiver

5. Enumerate the principles of ADPCM. (16)

Explain in adaptive quantisation with forward estimation
Explain in adaptive quantisation with backward estimation
Explain Adaptive prediction with forward estimation
Explain Adaptive prediction with backward estimation

UNIT III BASEBAND TRANSMISSION

Part A

1. What is intersymbol interference in baseband binary PAM systems?

In .baseband binary PAM, symbols are transmitted one after another. These symbols are separated by sufficient time durations. The transmitter, channel and receiver acts as a filter to this baseband data. Because .of .the .filtering characteristics, transmitted PAM pulses are spread in time.

2. What is correlative coding?

Correlative level coding is used to transmit a baseband signal with the signalling rate of 2Bo over the channel of bandwidth Bo. This is made physically possible by allowing ISI in the transmitted in controlled manner.

This ISI is known to receiver. .The .correlative .coding .is .implemented .by .duobinary .signalling .and modified duobinary signalling.

3. **Define Duobinary baseband PAM system**

Duobinary encoding reduces the maximum frequency of the baseband signal. The word „duo‟ means to double the transmission capacity of the binary system. Let the PAM signal ak represents kth bit. Then the encoder the new waveform as

$C_k = a_k + a_{k-1}$

Thus two successive bits are added to get encoded value of the kth bit. Hence Ck becomes a correlated signal even though a_k is not correlated. This introduces intersymbol interference in the controlled manner to reduce the bandwidth.

4. What are eye pattern?

Eye pattern is used to study the effect of ISI in baseband transmission.

1) Width of .eye .opening defines the .interval over .which the .received wave can be sampled without error from ISI.

2.) The sensitivity of the system to timing error is determined by the rate of closure of the eye as the sampling time is varied.

3.) Height of the eye opening at sampling time is called margin over noise

5. How is eye pattern obtained on the CRO?

Eye pattern can be obtained on CRO by applying the signal to one of the input channels and given an external trigger of $1/T_b$ Hz. This makes one sweep of beam equal to T_b seconds.

6. Why do you need adaptive equalization in a switched telephone network.

In switched telephone network the distortion depends upon

1) Transmission characteristics of individual links.

2) Number of links in connection.

Hence fixed pair of transmit and receive filters will not serve the equalization problem. The transmission characteristics keep on changing. Therefore adaptive equalization is used.

7 .What are the necessity of adaptive equalization?

Ans. Most of the channels are made up of individual links in switched telephone network, the distortion induced depends upon

1) transmission characteristics of individual links
2) number of links in connection

8. Define the principle of adaptive equalization?

Ans. The filters adapt themselves to the dispersive effects of the channel that is .the .cofficients .of .the .filters .are .changed .contineously according to .the received .data. .The .filter .cofficients .are .changed .in .such .a .way .that .the distortion in the data is reduced

9. Define duobinary encoding?

Ans. Duobinary encoding reduces the maximum frequency of the base band signal the “word duo” means to the double transmission capacity of the binary system

10. Write a note on correlative level coding?

Correlative level coding .allows the signal scaling rate of 2Bo in the channel of bandwidth Bo this is made physically possible by allowing ISI in the transmitted signal in controlled manner this ISI is known to the receiver

11. Define the term ISI?

Ans. The presence of outputs due to other bits interference with the output of required bit . this effect is called inter symbol interference (ISI)

12. Write the performance of data transmission system using eye pattern technique?

Ans. The width of the eye opening defines .the interval over which the received wave can can be sampled without error from inter symbol interference . The sensitivity of the system to timing error is determined by the rate of closure of the eye as the sampling time is varied.

13. What is the necessity of equalization?

Ans. When the signal is passed through the channel distortion is introduced in terms of 1) amplitude 2) delay this distortion creates problem of ISI. The detection of the signal also become difficult this distraction can be compensated with the help of equalizer.

14. What is raised cosine spectrum?

In the raised cosine spectrum, the frequency response P(f) decreases towards zero gradually That is there is no abrupt transition).

15. What is nyquist Bandwidth?
The B.i0s called nyquist bandwidth. .The nyquist bandwidth is the minimum transmission bandwidth for zero ISI.

Part B

1. Derive the expression for bit error probability due to a matched filter(16)
Explain in Receiving Filter
Expalin in Correlator type Receiving Filter
Explain in Matched filter type receiving filter

2. Discuss on signal design for ISI illumination (16)
Explain in Equalising
Explain in Signal and system design for ISI illumination

3.Obtain an expression for nyquist criterion for distortion less base band transmission for zero intersymbol interference. (16)
Nyquist pulse shaping of first type
Nyquist method of second type
Generation of partial responce signal

4.Write briefly about eye pattern and adaptive equalization for data transmission.(8)

Implementation of equalising filter
Eyepattern and Eye diagram analysis

5.What you understand by intersymbol interference (ISI)? Discuss in detail the nyquist criterion for minimizing (ISI).(16)
Explain in Equalising filter
Explain in Signal and system design for ISI illumination

UNIT IV DIGITAL MODULATION SCHEME

Part A

1. Mention the need of optimum transmitting and receiving filter in ba s eband data transmission.

When binary data is transmitted over the baseband channel, noise interfaces with it. Because .of this noise interference, .errors .are introduced .in signal detection. .Optimum .filter performs .two functions .while receiving .the noisy signal:

1) Optimum filter integrates the signal during the bit interval and checks the output at the .time instant where signal to noise ratio is maximum

2) Transfer function of the optimum filter is selected so as to maximise signal to noise ratio.

3) Optimum filter minimizes the probability of error.

2. Define ASK.

In .ASK, carrier .is .switched on .when binary .1 .is .to .be .transmitted and .it .is switched off when binary D is to be transmitted ASK is also called on-off keying.

3. What is meant by DPSK?

In DPSK, the input sequence is modified. Let input sequence be d(t) and output sequence be b(t). Sequence b(t) changes level at the beginning of each intervalin which d(t)=1 and it does not changes level when d(t)=0. When b(t) changes level, phase of the carrier is changed. And as stated above, b(t) changes t=its level only when d(t) =1. This means phase of the carrier is changed only if d(t)=1. Hence the technique is called Differential PSK.

4. Explain coherent detection?

In coherent detection, the local carrier generated at the receiver is phase locked with the carrier at the transmitter. The detection is done by correlating received noisy .signal .and .locally .generated .carrier. .The .coherent .detection .is .a synchronous detection.

5. What is the difference between PSK and FSK?

In PSK, phase of the carrier is switched according to input bit sequence. In FSK frequency of the carrier is switched according to input bit sequence. FSK needs double of the bandwidth of PSK.

6. What is meant by coherent ASK?

In coherent .ASK, correlation .receiver .is used to detect .the signal. .Locally generated carrier is correlated with incoming ASK signal. The locally generated carrier is in exact phase with the transmitted carrier. Coherent ASK is also called as synchronous ASK.

7 . What is the major advantage of coherent PSK over coherent ASK?

ASK .is .on-off .signalling, .where .as .the .modulated .carrier .is .continuously transmitted in PSK. Hence peak power requirement is more ASK, whereas it is reduced in case of PSK.

8. Explain the model of bandpass digital data transmission system?
The bandpass digital data transmission system consists of source, encoder and modulator in the transmitter. Similarly receiver, decoder and destination form the transmitter.

9. What is baseband signal receiver?
A baseband signal receiver increases the signal to noise ratio at the instant of sampling. This reduces the probability of error. The baseband signal receiver is also called optimum receiver.

10. What is matched filter?
The matched filter is a baseband signal receiver, which works in presence of white Gaussian noise. The impulse response of .the matched response of .the matched filter is matched to the shape pf the input signal.

11. What is the value of maximum signal to noise ratio of the matched filter? When it becomes maximum?
Maximum signal to noise ratio is the ratio of energy to psd of white noise. i.e.,
$\rho max = E/ (N0/2)$
This maximum value occurs at the end of bit duration i.e. Tb

12. What is correlator ?
Correlator is the coherent receiver. It correlates the received noisy signal .f(t) with .the .locally .generated replica .of .the .unknown .signal .x(t). .It‟ s output .is denoted as r(t).

13. On what factor, the error probability of matched filter depends.
Error probability is given as
$Pe = 1/2erfc\sqrt{E/N}$.
This equation shows that error probability depends only on energy but not on shape of the signal.

14. Bring out the difference between coherent & non coherent binary modulation scheme.
a. Coherent detection:
In this method the local carrier generated at the receiver is phase locked with the carrier at the transmitter.Hence it is called synchronous detection
b. Non coherent detection:
In this method, the receiver carrier need not be phase locked with transmitter carrier. Hence it is called envelope detection.

15. Write the expression for bit error rate for coherent binary FSK. Bit error rate for coherent binary FSK is given as,
$Pe = 1/2erfc\sqrt{0.6E/N}$.

16. Highlight the major difference between a QPSK & MSK signal.
MSK signal have continuous phase in all the cases, where as QPSK has phase shift of π/2 or π.

17. What is the error probability of MSK & DPSK?
Error probability of MSK: Pe = 1/2erfc√E/N.
Error probability of DPSK: Pe = 1/2e -Eb/No

18. In minimum shift keying what is the relation between the signal frequencies & bit rate.
Let the bit rate be f b and the frequency of carrier be f. The h0 igher and lower MSK signal frequencies are given as,
fH . = f.+0f./4 b
fL . = f.0f./4b

19. List the advantages of Passband transmission
a. Long distance.
b. Analog channels can be used for transmission.
c. Multiplexing techniques can be used for bandwidth conservation.
d. Transmission can be done by using wireless channel also.

20. List the requirements of Passband transmission.
b. Maximum data transmission rate.
c. Minimum probability of symbol error.d.Minimum transmitted power.

Part B

1.Derive the bit error probability due to coherent ASK,PSK & FSK systems. Compare the. Performance (16)
Explain the generation and detection of ASK,PSK & FSK signal,and explin in each block
Explain spectral characteristics of FSK,ASK,PSK signal.
Compare the error performance ASK,PSK & FSK systems

2.Discuss Noncoherent detection method of BFSK Signalling (16)
Representation of FSK signal
Define signal space diagram
spectral characteristics of FSK
Noncoherent detection of FSK signal

3.Draw the block diagram of MSK transmitter and explain the function of each block.(16)

Explain the Minimum Shift keying
Explain signal space diagram of MSK
spectral characteristics of MSK

4.With necessary equations and signal space diagram, explain briefly about FSK system.(16)

Define signal space diagram
spectral characteristics of FSK
derive the error probability of FSK signal

5.Draw the block diagram of MSK transmitter and explain the function of each blockwith the constellation diagram. (16)

Explain signal space diagram of MSK
spectral characteristics of MSK
explain in each block –coherent MSK receiver

UNIT V ERROR CONTROL CODING

Part A

1.What is hamming distance?

The hamming .distance .between .two code vectors .is equal to the number .of elements in which they differ. For example, let the two code words be,

X = (101) and Y= (110)

These .two .code .words .differ .in .second .and .third .bits. .Therefore .the .hammin distance between X and Y is two.

2. Define code efficiency.

The code efficiency is the ratio of message bits in a block to the transmitted bits for that block by the encoder

i.e., Code efficiency= (k/n)

k=message bits

n=transmitted bits.

3. What is meant by systematic and non-systematic codes?

In a Systematic block code, message bits appear first and then check bits. In them non-systematic .code, .message .and .check .bits .cannot .be .identified .in .the .code vector.

4. What is meant by linear code?

A code is linear if modulo-2 sum of any two code vectors produces another code vector. This means any code vector can be expressed as linear combination of other code vectors.

5. What are the error detection and correction capabilities of hamming codes ?

The minimum distance (dmin) of hamming codes is „3‟ . .Hence it can be used to detect double errors or

correct single errors. Hamming codes are basically linear block codes with dmin =3.

6. What is meant by cyclic codes?

Cyclic codes are the subclasses of linear block codes. They have the property that a cyclic shift of one codeword produces another code word.

7 . How syndrome is calculated in Hamming codes and cyclic codes?

In hamming codes the syndrome is calculated as, S=YH.T

Here Y is the received and

H.T.is the e transpose of parity check matrix

8. What is BCH code?

BCH .codes .are .most .extensive .and .powerful .error .correcting .cyclic .codes. .The decoding of BCH codes is comparatively simpler. For any positive integer „m‟ and „t‟ (where t<2 m-1)there exists a BCH code with following parameters:
Block length: n= 2 m-1
Number of parity check bits : n-k<=mt
Minimum distance: d min>=2t+1

9. What is RS code?
These are non binary BCH codes. The encoder for RS code operates on multiple bits simultaneously. The (n,k) RS code takes the groups of m- bit symbols of incoming binary data stream. It takes such „k‟ number of symbols in one block. Then the encoder acts (n – k) redundant symbols to form the code word of „n‟symbols RS code has:
Block Length : n=2.-1msymbols Message
size: K symbols
Parity check size: n-k= 2t symbols Minimum
distance: dmin=2t+a symbols

10. What is difference between block codes and convolutional codes?
Block codes takes‟ k‟ .number of bits simultaneously form „n‟ -bit .code vector. This code vector is also called block. Convolutional code takes one message bits at a time and generates two or more encoded bits. Thus convolutional codes generate a string of encoded bits for input message string.

11. Define constraint length in convolutional code?
Constraint length is the number of shift over which the single message bit influence the encoder output. It is expressed in terms of message bits.

12. Define free distance and coding gain.
Free distance is the minimum distance between code vectors. It is also equal to minimum weight of the code vectors. Coding gain is used as a basis of comparison for different coding methods. To achieve the same bit error rate the coding gain is defined as,
A= (E.b /N.)oencoded (E.b /N.)ocoded
For convolutional coding, the coding gain is given as,
A = rd./f2
Here „r‟ is the code rate And „df is the free distance.

13. What is convolution code?
Fixed number of input bits is stored in the shift register & they are combined with the help of mod 2 adders. This operation is equivalent to binary convolution coding.

14. What is meant by syndrome of linear block code?
The non zero output of the produce YH.is calleTd syndrome & it is used to detect error in y. Syndrome is denoted by S & given as,

S=YH T

15. What are the advantages of convolutional codes?

Advantages:

1 .The decoding delay is small in convolutional codes since they operate o smaller blocks of data.

2. .The storage hardware required by convolutional decoder is less since the block sizes are smaller.

Disadvantages:

1 .Convolutional codes are difficult to analyze since their analysis is complex.

2. .Convolutional codes are not developed much as compared to block codes.

16. Define sates of encoder?

The constraint length of the given convolutional encoder is K=2. Its rate is . means for single message bit .input, two bits .x1 and x2 .are encoded at the output. .„S 1‟ represents the input message bit and S.stores the2 previous message bit. .Since only one previous message bit is stored, .this encoder can have states depending upon this stored message bit. .Let‟ s represent,

S.2= 0 and S.2= 1 state „b‟ state „a‟

17. Compare between code tree and trellis diagram?

1. Code tree indicates flow of the coded signal along the nodes of the tree
Trellis diagram indicates transitions from current to next states

2. Code tree is lengthy way of representing coding process Code
trellis diagram is shorter or compact way of representing coding process

18. Write the futures of BCH Codes?

BCH codes are most extensive and powerful error correcting cyclic codes. The decoding of BCH codes is comparatively simpler.

The decoding schemes of BCH codes can be implemented on digital computer. Because of software implementation of decoding schemes they a re quite flexible compared to hardware implementation of other schemes.

19. What is Golay codes?

Golay code is the (23,12) cyclic code whose generating polynomial is,

G(p) = p.+1p1.+p.9+p.+7p.+6p+1 5

This code has minimum distance of d min= 7. This code can correct upto 3 errors. But Golay code cannot be generalized to other combinations of n and k.

20. Define constraint length in convolutional codes?

Constraint length is the number of shifts over which the single message bit can influence the encoder output. .This expressed in terms of message bits.

Part B

1. Discuss the convolution decoder algorithm in detail for the encoder of constraint length 3 and Code rate.1/2 (16)

Explain in convolutional encoder
Calculate the constrain length
Structural properties of convolutional encoder

2. Explain how encoding is done by convolutional codes with a suitable example (16)

Explain in convolutional encoder
Calculate the constrain length
Structural properties of convolutional encoder

3. Describe a decoding procedure for linear block code.(16)

Expalin in classification of codes
Define in block codes
Systematic codes

4. Briefly explain the viterbi-decoding algorithm. (16)

Expalin Viterbi algorithm
And draw backs