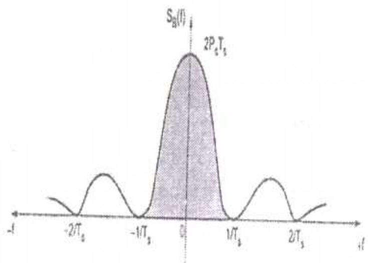
SCHEME OF VALUATION

(Scoring Indicators)

Revision :2015 Course Code: 5201

Course Title: DIGITAL COMMUNICATION

Ques. No	Scoring Indicator	Split-up Score	Total
	PART A		
I 1.	There are mainly two types of noises that affect the performance of a PCM system • Channel noise – It is due to the various disturbances on the channel. • Quantization noise – It is noise introduced by quantizer.	2x1	2
I 2.	 Fig. 2.6.7 Plot of power spectral density of QPSK signal		2
I 3.	Band Pass transmission shifts the signal to be transmitted into a higher frequency and then transmitted over a band-pass channel.		2
I 4.	If the probability of occurrence of messages M_k is P_k, then amount of information of M_k is $$I_k = \log_2 \frac{1}{P_k}$$		2
I 5.	In Packet switching, the message to be transmitted is divided and grouped into a number of fixed size unit called packets which are individually routed from the source to the destination.		2
	PART B		
II 1.	A continuous time signal can be represented in its samples and can be recovered back when sampling frequency fs is greater than or equal to the twice the highest frequency component of message signal fm. i. e. $fs \geq 2fm$. $x(t) = 10\cos(4000\pi t) + 20\cos(2000\pi t)$ $2\pi f_1 t = 4000\pi t$ ⇨ $f_1 = 2000$ $2\pi f_2 t = 2000\pi t$ ⇨ $f_2 = 1000$ therefore $f_m = f_1 = 2000$ and Nyquist rate $f_s = 2f_m = 2 \times 2000 = 4000$Hz Nyquist interval $t_s = \frac{1}{fs} = \frac{1}{4000} = 0.25$ms	2 + 4 (prob.)	6

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<td>II 2.</td>
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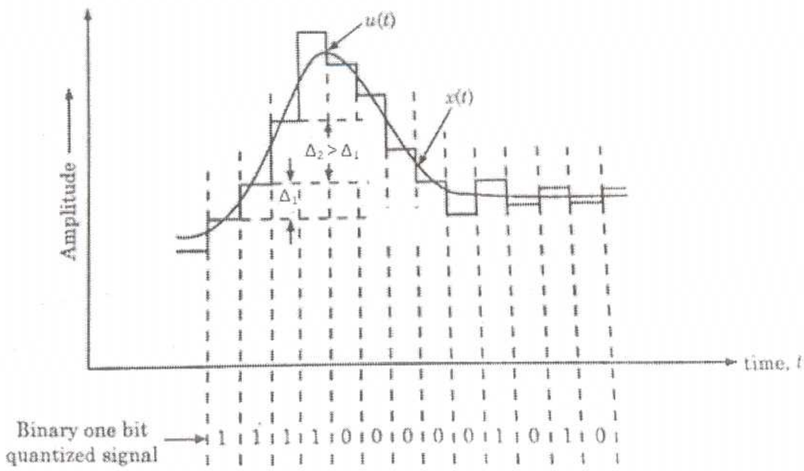
- Adaptive Delta Modulation (ADM) method was introduced to solve the granular noise and slope overload error caused during Delta modulation.
- In Adaptive Delta Modulation, the step size of the staircase signal is not fixed and changes depending upon the input signal.
- Particularly in the steep segment of the signal, the step size is increased .When the input is varying slowly, the step size is reduced.
- Fig. shows the waveforms of adaptive delta modulator and sequence of bits transmitted.



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<td>3 (diag.)
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3 (exp.)</td>
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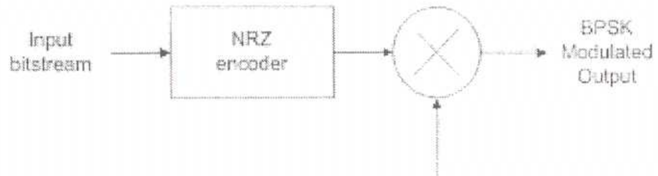

$\sqrt{2P}\, cosw_0t$

The block diagram of Binary Phase Shift Keying consists of the balance modulator which has the carrier sine wave as one input and the binary sequence as the other input. The modulation of BPSK is done using a balance modulator, which multiplies the two signals applied at the input. For a zero binary input, the phase will be 0° and for a high input, the phase reversal is of 180°.

Thus the transmitted signal is either $V_{BPSK}(t) = \sqrt{2p}\,\cos w_0 t$

or $V_{BPSK}(t) = \sqrt{2p}\,\cos(w_0 t + \pi) = -\sqrt{2p}\,\cos w_0 t$

Hence we can write BPSK signal as

$$V_{BPSK}(t) = b(t)\sqrt{2p}\,\cos w_0 t$$

Here $b(t) = \begin{cases} b(t) = +1 \text{ for binary input } 1 \\ b(t) = -1 \text{ for binary input } 0 \end{cases}$
</td>
<td>3 (diag.)
+
3 (exp.)</td>
<td>6</td>
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II 4. Entropy can be defined as the average information content per source symbol/message.

Suppose we have 'm' different messages with probability of occurrence P_1, P_2, P_3.... Suppose that, during a period of transmission "L" messages have been send. We may expect that in this L messages, we transmitted "P_1L" message of m_1, "P_2L" message of m_2 ....

∴ Total information amount in 'L' messages I_{total} = Total information amount of m_1 message + Total information amount of m_2 message +.........

Information amount of single m_1 message = $\log_2 \frac{1}{P_1}$

Total information amount of m_1 message = $P_1L \log_2 \frac{1}{P_1}$

Similarly, total information amount of m_2 message = $P_2L \log_2 \frac{1}{P_2}$

$$\therefore I_{total} = P_1L \log_2 \frac{1}{P_1} + P_2L \log_2 \frac{1}{P_2} + \ldots\ldots..$$

Average information content per message or entropy represented by "H" will be

$$H = \frac{I_{total}}{L} = \frac{1}{L}\left(P_1L \log_2 \frac{1}{P_1} + P_2L \log_2 \frac{1}{P_2} + \ldots\ldots..\right)$$

$$= P_1\log_2 \frac{1}{P_1} + P_2\log_2 \frac{1}{P_2} + \ldots\ldots..$$

$$H = \sum_{k=1}^{m} \log_2 \frac{1}{P_k}$$

The unit of entropy is bits/message.

6

II 5. It is the simplest technique for detecting error. The MSB of an 8-bits word is used as the parity bit and the remaining 7 bits are used as data bits. The parity of 8-bits transmitted word can be either even parity or odd parity.

Even parity -- Even parity means the number of 1's in the given word including the parity bit should be even.

Odd parity -- Odd parity means the number of 1's in the given word including the parity bit should be odd.

The parity bit can be set to 0 and 1 depending on the type of the parity required.

For even parity, this bit is set to 1 or 0 such that the no. of "1 bits" in the entire word is even.

P	← Data bits →
0	1001011

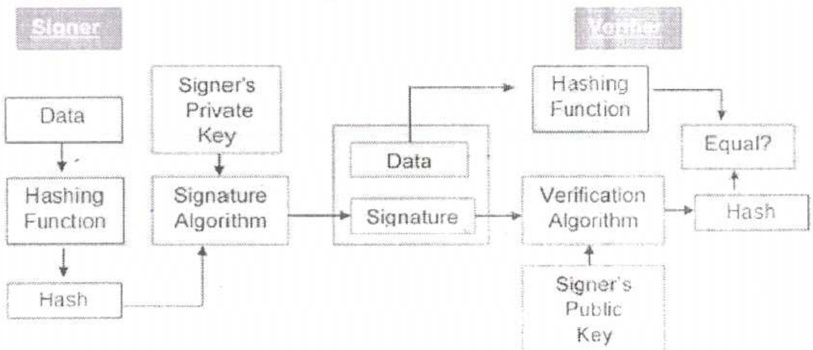
P	← Data bits →
1	1001011

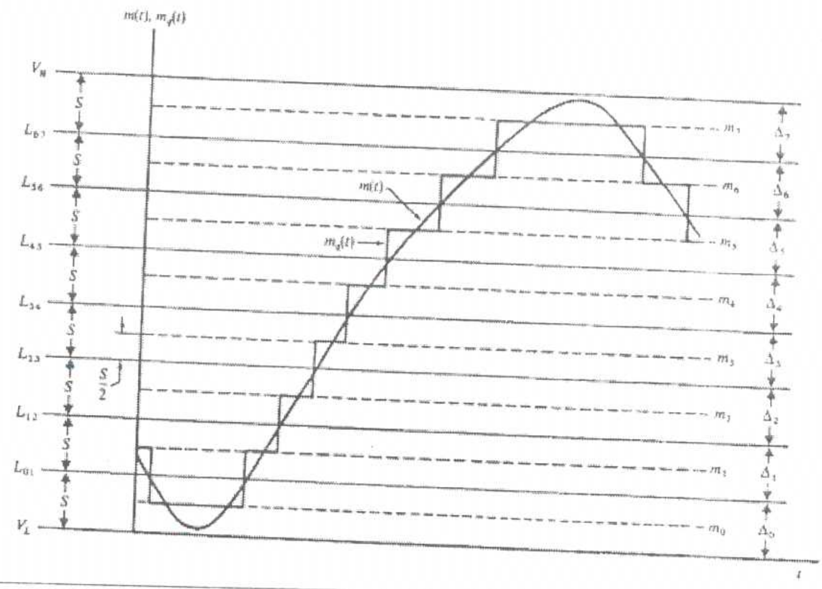
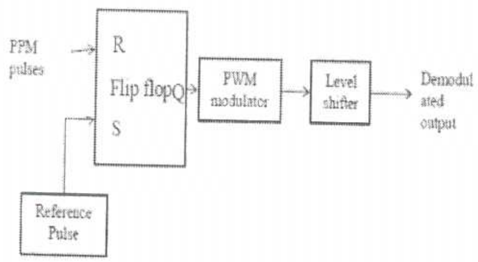
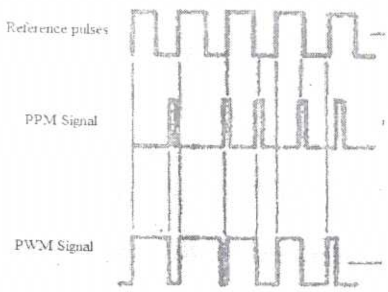
6

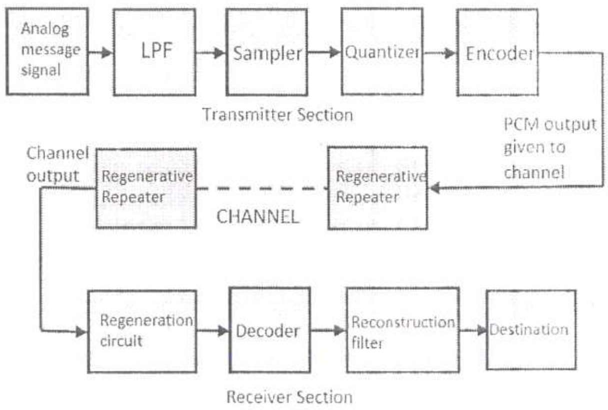
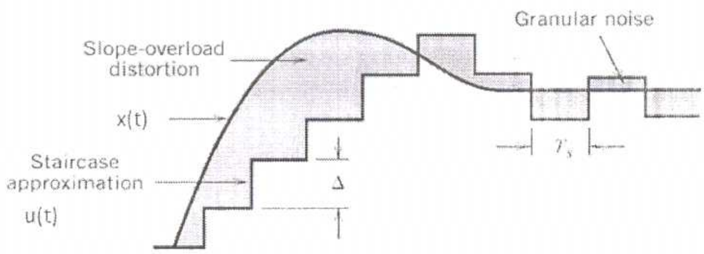
For odd parity, this bit is set to 1 or 0 such that the no. of "1 bits" in the entire word is odd.

P	← Data bits →
1	1001011

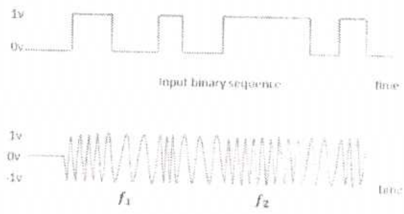
P	← Data bits →
0	1000110

	Parity checking at the receiver can detect the presence of an error if the parity of the receiver signal is different from the expected parity.		
II 6.	Transmission mode means transferring of data between two devices. There are three types of transmission mode:- • Simplex Mode • Half-Duplex Mode • Full-Duplex Mode Simplex Mode:- In Simplex mode, the communication is unidirectional, as on a one-way street. Only one of the two devices on a link can transmit, the other can only receive. Example: Radio broadcasting Half-Duplex Mode:- In half-duplex mode, each station can both transmit and receive, but not at the same time. When one device is sending, the other can only receive, and vice versa. The half-duplex mode is used in cases where there is no need for communication in both directions at the same time. Example: Walkie- talkie in which message is sent one at a time and messages are sent in both the directions. Full-Duplex Mode:- In full-duplex mode, both stations can transmit and receive simultaneously. Full-duplex mode is used when communication in both directions is required all the time. The capacity of the channel however must be divided between the two directions. Example: Telephone Network	3x2	6
II 7.	A digital signature is a mathematical technique used to validate the authenticity and integrity of a message, software or digital document. Digital signatures are the public-key primitives of message authentication. 		6
	PART C		
III (a)	The quantizing of an analog signal is done by discretizing the signal with a finite number of quantization levels. Quantization which converting a continuous-amplitude sample into a discrete-time signal. In quantisation we create a new signal called quantized signal, $m_q(t)$ which is an approximation analog signal m(t). In general entire range of signal is divided into 'M' equal intervals and the spacing between the two adjacent quantization levels is called a step-size, s. The following figure shows how an analog signal gets quantized. Here M=8. Quantized signal, $m_q(t)$ is generated as follows - If m(t) is in the range $\Delta_0 m_q(t)=m_0$, m(t) is in the range $\Delta_1 m_q(t)=m_1$ and	4(Fig.) + 5(Exp.)	9

	so on 		
III (b)	The S R flip flop is set by the reference/carrier pulses which are generated by using synchronization pulse from transmitter and reset by the PPM pulses. The resulting output is a PWM signal. This PWM signal is then demodulated using the PWM demodulator.  	3(Fig.) +3(Exp.)	6
IV. (a)	The transmitter section of a Pulse Code Modulator circuit consists of Sampling, Quantizing and Encoding. The basic operations in the receiver section are regeneration of impaired signals, decoding, and reconstruction of the quantized pulse train. The following figure is the block diagram of PCM. <u>Low Pass Filter (LPF):-</u> This filter eliminates the high frequency components present in the input analog signal which is greater than the highest frequency of the message signal, to avoid aliasing of the message signal. <u>Sampler: -</u> This is the circuit which uses the technique that helps to collect the sample data at instantaneous values of the message signal, so as to reconstruct the original signal. The sampling rate must be greater than twice the highest frequency component of the message signal, in accordance with the sampling theorem. <u>Quantizer:-</u> Quantization which converting a continuous-amplitude sample into a discrete-time signal. <u>Encoder:-</u>The digitization of analog signal is done by the encoder. It designates each quantized level by a binary code.	4(Fig.) + 5(Exp.)	9

	Regenerative Repeater: - The output of the channel has one regenerative repeater circuit to compensate the signal loss and reconstruct the signal. It also increases the strength of the signal. Decoder: - The decoder circuit decodes the pulse coded waveform to reproduce the original signal. This circuit acts as the demodulator. Reconstruction Filter : - After the decoder, a low pass filter is employed, called as the reconstruction filter to get back the original signal. 		
IV. (b)	The delta modulation has two major drawbacks as under : **1. Slope overload distortion** If the rate of rise of input signal x(t) is so high that the staircase signal cannot approximate it, the step size 'Δ' becomes too small for staircase signal u(t) to follow the step segment of x(t). Hence, there is a large error between the staircase approximated signal and the original input signal x(t). This error or noise is known as slope overload distortion. To reduce this error, the step size must be increased when slope of signal x(t) is high.  **2. Granular or idle noise** Granular or Idle noise occurs when the step size is too large compared to small variation in the input signal. Fig. shows that when the input signal is almost flat, the staircase signal u(t) keeps on oscillating by $\pm\Delta$ around the signal. The error between the input and approximated signal is called granular noise. The solution to this problem is to make the step size small.	2x3	6
V.(a)	Binary Frequency Shift Keying (BFSK) is a digital modulation technique in which the frequency of the carrier signal changes according to the input binary data. The output of a FSK modulated wave is high in frequency for a binary High ('1') input and is low in frequency for a binary Low ('0') input. The time		

domain of an FSK modulated wave is shown in fig.1 below :



FSK Modulated output wave

$\therefore$ BFSK signal can be written as $V_{BFSK}(t) = \sqrt{2p}\cos(w_0 + d(t)\Omega)\,t$

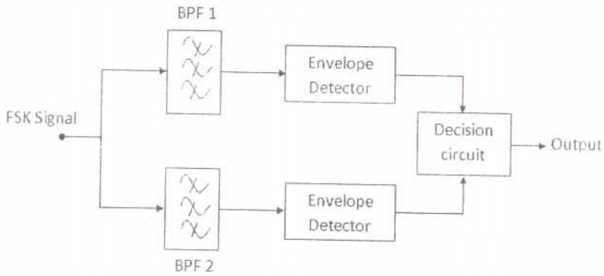
Here $d(t) = \begin{cases} d(t) = +1 \text{ for binary input } 1 \\ d(t) = -1 \text{ for binary input } 0 \end{cases}$

$\therefore$ we have two frequencies

High frequency, $w_H = w_0 + \Omega$ and Low frequency, $w_L = w_0 + \Omega$

4(Fig.) + 4(Exp.) — 8

BFSK Detector



It consists of two band pass filters, two envelope detectors, and a decision circuit. The BFSK input signal is passed through the two Band Pass Filters (BPF 1 and BPF 2) and tuned to w_H and w_L frequencies. The output from these two BPFs look like ASK signal, which is given to the envelope detector. The signal in each envelope detector which detects the BPF output signal envelope. The decision circuit chooses which output is more likely and selects it from any one of the envelope detectors. It also re-shapes the waveform to a rectangular one.

V.(b) The main concern in digital communication is power in side lobes which causes interference with adjacent channel. Sharp transitions in Binary data between "one" and "zero" states and vice versa results large side lobes. Minimum Shift Keying (MSK) is the most spectrally efficient modulation schemes available.

MSK overcome this problem by avoiding phase discontinuities. The steps in MSK modulation is

4(Exp.)+ 3(diag.) — 7

- Divide bit stream into odd and even$b_o(t)$& $b_e(t)$. These bit streams are multiplied by smoothening signals $cos\Omega t$ and $sin\Omega t$ and generate products $b_o(t)cos\Omega t$& $b_e(t)sin\Omega t$.
- These smoother signals are multiplied by carriers $cosw_0t$ and $sinw_0t$ and add these signals to form MSK signal

$$b_e(t)\sqrt{p}\,cosw_0t$$

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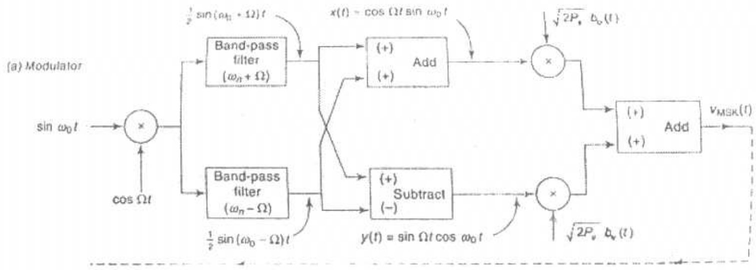
$$V_{MSK}(t) = \sqrt{2\mathrm{p}}\mathrm{b_e(t)}sin\Omega t * \mathrm{cosw_0t} + \sqrt{2\mathrm{p}}\mathrm{b_o(t)}cos\Omega t * \mathrm{sinw_0t}$$



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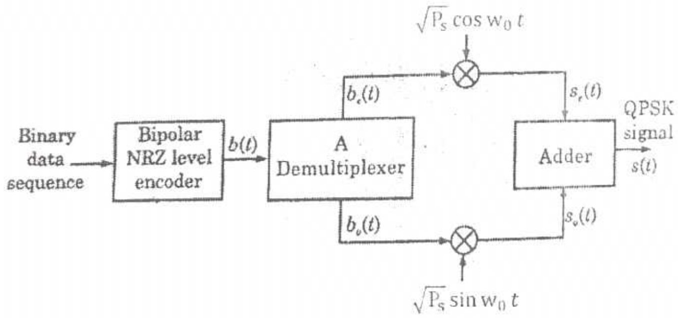
<tr><td>VI.(a)</td><td>

Quadrature phase shift keying (QPSK) is a modulation technique, and it transmits two bits per symbol. In QPSK symbol represents 00, 01, 10, or 11 and each symbol produces one of the four output phase shifts - 45°, 135°, 225°, and 315°.

A QPSK modulator can be implemented as follows. A de-multiplexer is used to separate odd and even bits from the generated information bits. The even bit is multiplied by cosine component and odd bit is multiplied by sine component. QPSK modulated signal is obtained by adding the signal from multiplier



$$s_e(t) = b_e(t)\sqrt{p}\,\cos w_0 t$$
$$s_o(t) = b_o(t)\sqrt{p}\,\sin w_0 t$$

$s(t) = b_e(t)\sqrt{p}\,\cos w_0 t + b_o(t)\sqrt{p}\,\sin w_0 t$

Advantages:

- QPSK provide very good noise immunity.
- It provides low error probability
- Bandwidth is twice efficient is compared to BPSK modulation
- For the same BER, the bandwidth required by QPSK is reduced to half as compared to BPSK

</td><td>4(Exp.)+
3(diag.)+
2(adv.)</td><td>9</td></tr>
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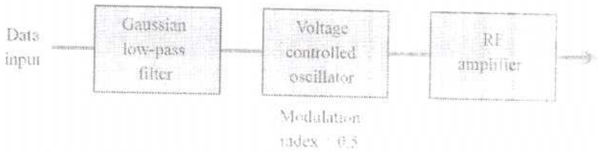
VI.(b)

GMSK is similar to standard minimum-shift keying (MSK); however, the digital data stream is first shaped with a Gaussian filter before being applied to a frequency modulator, and typically has much narrower phase shift angles than most MSK modulation systems. This has the advantage of reducing sideband power, which in turn reduces out-of-band interference between signal carriers in adjacent frequency channels.

GMSK generator is shown below



GMSK signal can be generated by filter the modulating signal using a Gaussian filter and then apply this to a frequency modulator where the modulation index is set to 0.5. This method is very simple and straightforward but it has the drawback that the modulation index must exactly equal 0.5.

3 (Fig) + 3 (exp.) — 6

VII. (a)

Shannon - Fano coding algorithm

1st step: write source symbols descending for their probability.

2nd step: We divide set of the symbols into two parts by equal probability if possible. , . For the upper set we assign the 0 digit, for lower set we assign the 1 digit.

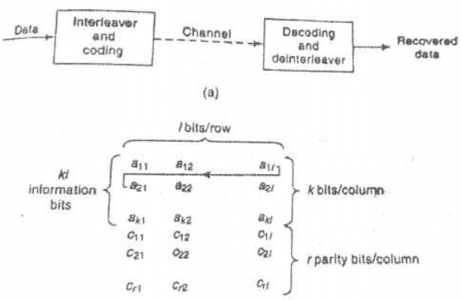
3rd step: We repeat the 2nd step for every subset, until every subset contains just one symbol.

The algorithm example

p	signal	1st step	2nd step	3rd step	code
$^1/_2$	A_1	0			0
$^1/_4$	A_2	1	0		10
$^1/_8$	A_3	1	1	0	110
$^1/_8$	A_4	1	1	1	111

4(Algo.) + 4(Eg.) — 8

VII. (b)



In block interleaving, data to be transmitted is interleaved and then error

4(Exp.) + 3(fig.) — 7

coded. At the receiver, decoding and de-interleaving will be done.

- In this kl no. of bits are entered into shift register which organized as rows with 'l 'bits per column.
- At each shift, each bit moves one position to right and right most element moves to the left most element of next row
- When data entry complete, error correction coding is applied in each column and added 'r' parity bits are added.
- When the coding is completed, entire data transmitted through channel starts from last row right most bit C_{rl} to a_{11}
- At receiver, received data is again stored in the same order as in the transmitter.
- Then error decoding will be done and discard the parity bit.
- Suppose that burst error occurs for I<l consecutive location, because of this organization only one error will appear in each column. This single error can be corrected by error correction code.

VIII. (a)

In the Hamming code, k parity bits are added to an n-bit data word, forming a new word of $n+k$ bits. The bit positions are numbered in sequence from 1 to $n+k$. Those positions numbered with powers of two are reserved for the parity bits. The remaining bits are the data bits. Here k and n are related by the following formula

$$2^k \geq k + n + 1$$

Consider, for example, the 4-bit data word 0011. We include three parity bits with this word and arrange the 7 bits as follows:

Bit position	D7	D6	D5	D4	D3	D2	D1
	0	0	1	P3	1	P2	P1

The 4 parity bits $P1$ through $P3$ are in positions 1, 2, 4 respectively. The 4 bits of the data word are in the remaining positions. Each parity bit is calculated as follows:

$P1$ _ XOR of bits (3, 5, 7)
$P2$ _ XOR of bits (3, 6, 7)
$P3$ _ XOR of bits (5, 6, 7)
P1 = 0
P2 = 1
P3 = 1

The 4-bit data word is written with the 3 parity bits as a 7-bit composite word
So code is 0011110

In the receiver, error bit position is calculated as follows.
The error of the word is checked over the same groups of bits, including their parity bits. The four check bits are evaluated as follows:

$E1$ = XOR of bits (1, 3, 5, 7)
$E2$ = XOR of bits (2, 3, 6, 7)
$E3$ = XOR of bits (4, 5, 6, 7)

5(Exp.)+ 4(Ex.) 9

Error bit position is E=E3E2E1

Eg: Assume that received data is 0011010

Bit position	D7	D6	D5	D4	D3	D2	D1
	0	0	1	1	0	1	0

$E1 = 1$; $E2 = 1$; $E3 = 0$

∴ Error bit position is 011 ie, error occurred at 3rd bit position, so complement the 3rd bit to correct the error.

∴corrected code is 0011110

VIII. (b)

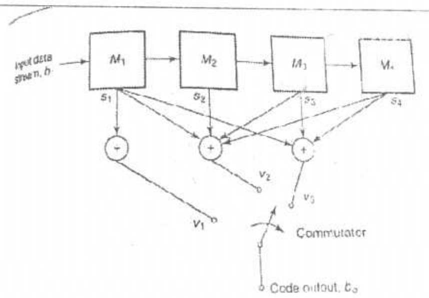


Fig. 13.22 An example of a convolutional coder.

A convolutional encoder convolutes the binary data by adding more bits. Theconvolutional encoder takes k bit at a time from incoming sequence of bit stream and computes n-bit output sequence. The computation is performed by module 2 addition (Ex-OR) operations on the current input bit and the contents of the shift registers. After performing one operation the contents of shift registers are shifted by one bit right. After last input bit fed into the M1, enough 0's are added until last input bit shift out through last shift register. At the initial all shift registers are set by zeros.

The outputs are

$v_1 = s_1$

$v_2 = s_1 + s_2 + s_3 + s_4$

$v_3 = s_1 + s_3 + s_4$

The outputs are sampled by commutator at each input bit time.

3(Diag.) + 3(Exp.) — 6

IX.(a) In large networks, there can be multiple paths from sender to receiver. The switching technique will decide the best route for data transmission.

Classification of Switching Techniques

1. Circuit Switching

- Circuit switching is a switching technique that establishes a dedicated path between sender and receiver.
- In the Circuit Switching Technique, once the connection is established then the dedicated path will remain to exist until the connection is terminated.
- A complete end-to-end path must exist before the communication takes place.
- Circuit switching is used in public telephone network. It is used for voice transmission.

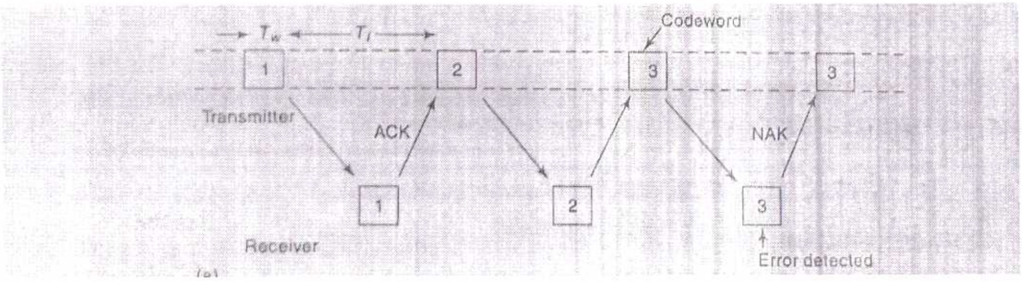
	Communication through circuit switching has 3 phases: • Circuit establishment • Data transfer • Circuit Disconnect **2. Message Switching** • Message Switching is a switching technique in which a message is transferred as a complete unit and routed through intermediate nodes at which it is stored and forwarded. • Each and every node stores the entire message and then forwards it to the next node. This type of network is known as store and forward network. • In Message Switching technique, there is no establishment of a dedicated path between the sender and receiver. • The destination address is appended to the message. Message Switching provides route through the intermediate nodes based on the information available in the message. • Message switches are programmed in such a way so that they can provide the most efficient routes.	2x4	8
IX.(b)	RSA algorithm is a public key encryption technique with is Public key is PU= $\{e, n\}$ and Private key PR = $\{d, n\}$. It is considered as the most secure way of encryption. In this plaintext as a series of numbers less than n. The encryption and decryption process is as follows for plaintext P and cyphertext C $C = P^e mod\ n$ $P = C^d mod\ n$ The keys for the RSA algorithm are generated the following way: 1. Choose two different large random prime numbers p and q 2. Calculate n = pxq 3. n is the modulus for the public key and the private keys 4. Calculate the $\emptyset(n) = (p-1)(q-1)$ 5. Select integer e as gcd($\emptyset(n)$,e) = 1. 6. Calculate d as (de) mod$\emptyset(n)$ =1 7. Public key is PU= {e,n} 8. Private key PR = {d,n}		7
X.(a)	Error control schemes that involve error detection and retransmission of lost or corrupted packets are referred to as Automatic Repeat Request (ARQ) error control. The most common ARQ retransmission schemes: – • Stop-and-Wait ARQ, • Sliding window ARQ ➢ Go-Back-N ARQ ➢ Selective Repeat ARQ. In this receiver is not correct the error, but only to detect the error. When the receiver receives a correct frame, it sent positive acknowledge (ACK) signal back to the transmitter. When the receiver receives a damaged frame, it sends a negative acknowledge NACK back to the sender and the sender must retransmit the correct frame.		

Stop-and-wait ARQ

- Transmitter sent a frame and wait for the acknowledgement signal from receiver.
- The receiver received the packet and checked the data. If there is no error, then ACK signal is send back to the transmitter. Otherwise it sends NACK back to the receiver.
- If the sender get ACK signal, it transmits the next frame in queue. If a NACK signal is received, the sender retransmits the frame again and waits for acknowledgement.

Disadvantage:

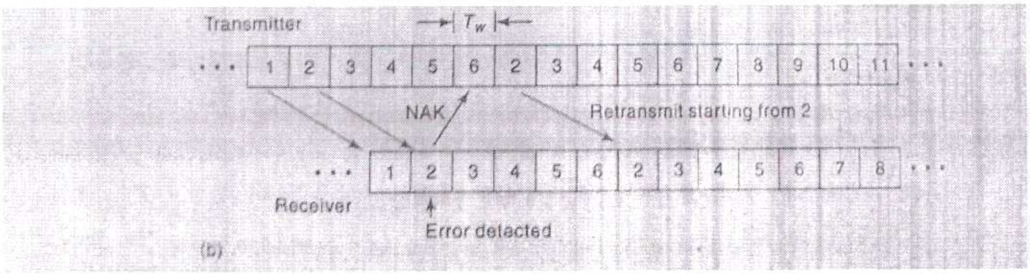
- Only one frame can be sent at a time.
- Total transmission time is large due to the waiting time for acknowledgement signal.



3x3 9

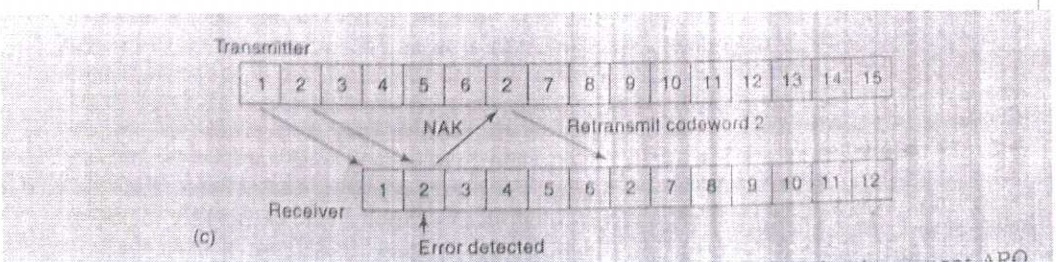
Go-Back-N ARQ

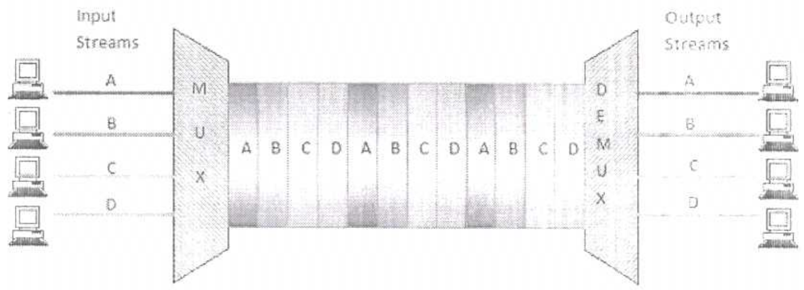
In Go-Back-N ARQ method, the sender sends multiple frames without waiting for the acknowledgement signal of the previous ones. The receiver is enabled to receive multiple frames and acknowledge them. The receiver keeps track of incoming frame's sequence number. If all frames are positively acknowledged, the sender sends next set of frames. When the sender receives a NACK, it retransmits the frame in error plus all the succeeding frames.



Selective Repeat ARQ

The selective-repetitive ARQ scheme is similar to Go Back N except that it retransmits only those frames for which NAKs are received. This is the most efficient among the ARQ schemes, but the sender must be more complex.



X.(b)	Time division multiplexing (TDM) is a communications process that transmits two or more digital signals over a common channel.In TDM, the data flow of each input stream is divided into units. Each input unit is allotted an input time slot. During transmission, one unit of each of the input streams is allotted one-time slot, periodically, in a sequence, on a rotational basis. This system is popularly called round-robin system. Consider a system having four input streams, A, B, C and D. Each of the data streams is divided into units which are allocated time slots in the round – robin manner. Hence, the time slot 1 is allotted to A, slot 2 is allotted to B, slot 3 is allotted to C, slot 4 is allotted to D, slot 5 is allocated to A again, and this goes on till the data in all the streams are transmitted. 	3(Diag.) + 3(Exp.)	6