Q 2 (a) A discrete Memory less source has four symbols S_0, S_1, S_2, S_3 with probabilities of 0.4, 0.3, 0.2 and 0.1 respectively.
i) Calculate the amount of information in each symbol
ii) Calculate entropy of the source

Answer

i) Amount of information:

$I(S_0) = \log_2[1/P(S_0)] = \log_2(1/0.4) = 1.32$

$I(S_1) = \log_2[1/P(S_1)] = \log_2(1/0.3) = 1.73$

$I(S_2) = \log_2[1/P(S_2)] = \log_2(1/0.2) = 2.32$

$I(S_3) = \log_2[1/P(S_3)] = \log_2(1/0.1) = 3.32$

$$H(X) = -\sum_{i=1}^{4} P(Xi) log2[P(Xi)]$$

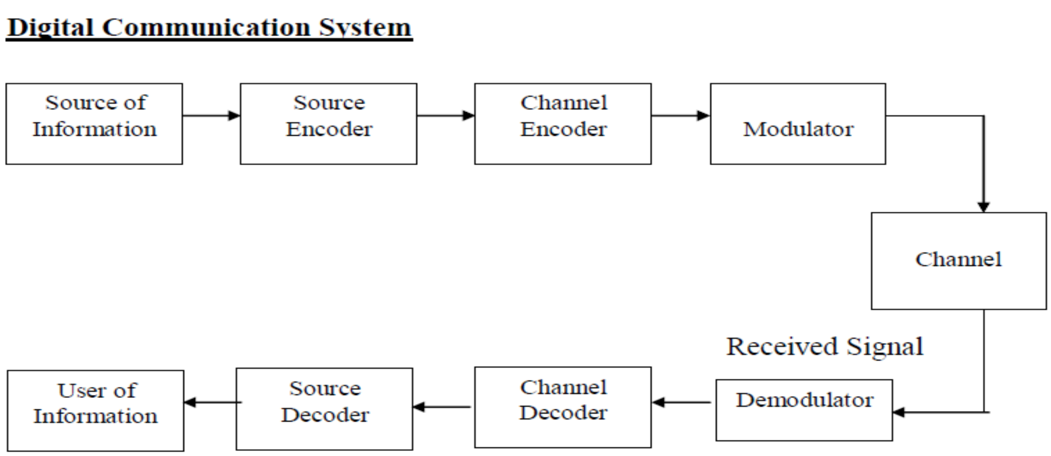
$H(X) = -0.4\log_2 0.4 - 0.3\log_2 0.3 - 0.2\log_2 0.2 - 0.1\log_2 0.1$

$= 1.85$ b/symbol.

Q 2 (b) Draw and explain the block diagram of a digital communication system.

Answer

The block diagram of the digital communication system is shown below



The figure shows the functional elements of a digital communication system. Source of Information:
1. Analog Information Sources.
2. Digital Information Sources.

Analog Information Sources → Microphone actuated by a speech, TV Camera scanning a scene, continuous amplitude signals.

Digital Information Sources → These are teletype or the numerical output of computer which consists of a sequence of discrete symbols or letters. An Analog information is transformed into a discrete information through the process of sampling and quantizing.

SOURCE ENCODER / DECODER: The Source encoder (or Source coder) converts the input i.e. symbol sequence into a binary sequence of 0‟s and 1‟s by assigning code words to the symbols in the input sequence. For eg. :-If a source set is having hundred symbols, then the number of bits used to represent each symbol will be 7 because 27=128 unique combinations are available. The important parameters of a source encoder are block size, code word lengths, average data rate and the efficiency of the coder (i.e. actual output data rate compared to the minimum achievable rate) At the receiver, the source decoder converts the binary output of the channel decoder into a symbol sequence. The decoder for a system using fixed – length code words is quite simple, but the decoder for a system using variable – length code words will be very complex. Aim of the source coding is to remove the redundancy in the transmitting information, so that bandwidth required for transmission is minimized. Based on the probability of the symbol code word is assigned. Higher the probability, shorter is the codeword. Ex: Huffman coding.

CHANNEL ENCODER / DECODER: Error control is accomplished by the channel coding operation that consists of systematically adding extra bits to the output of the source coder. These extra bits do not convey any information but helps the receiver to detect and / or correct some of the errors in the information bearing bits. There are two methods of channel coding:
1. Block Coding: The encoder takes a block of „k‟ information bits from the source encoder and adds „r‟ error control bits, where „r‟ is dependent on „k‟ and error control capabilities desired.
2. Convolution Coding: The information bearing message stream is encoded in a continuous fashion by continuously interleaving information bits and error control bits.

The Channel decoder recovers the information bearing bits from the coded binary stream. Error detection and possible correction is also performed by the channel decoder. The important parameters of coder / decoder are: Method of coding, efficiency, error control capabilities and complexity of the circuit.

MODULATOR: The Modulator converts the input bit stream into an electrical waveform suitable for transmission over the communication channel. Modulator can be effectively used to minimize the effects of channel noise, to match the frequency spectrum of transmitted signal with channel characteristics, to provide the capability to multiplex many signals.

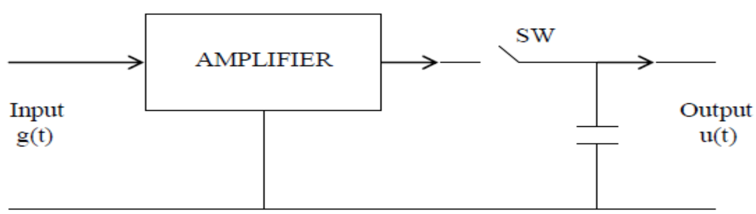
DEMODULATOR: The extraction of the message from the information bearing waveform produced by the modulation is accomplished by the demodulator. The output of the demodulator is bit stream. The important parameter is the method of demodulation.

CHANNEL: The Channel provides the electrical connection between the source and destination. The different channels are: Pair of wires, Coaxial cable, Optical fibre, Radio channel, Satellite channel or combination of any of these. The communication channels have only finite Bandwidth, non-ideal frequency response, the signal often suffers amplitude and phase distortion as it travels over the channel. Also, the signal power decreases due to the attenuation of the channel. The signal is corrupted by unwanted, unpredictable electrical signals referred to as noise. The important parameters of the channel are Signal to Noise power Ratio (SNR), usable bandwidth, amplitude and phase response and the statistical properties of noise.

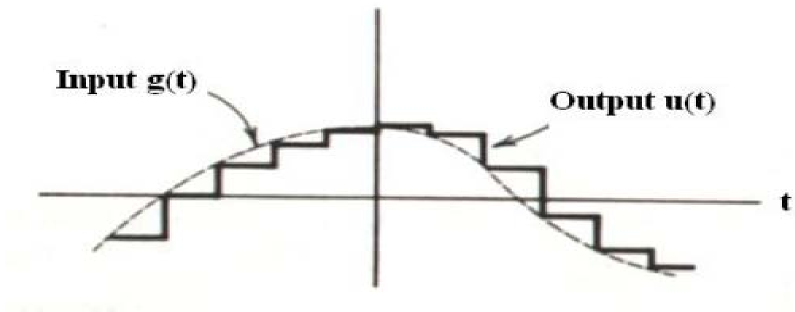
Q 3 (a) Explain how Sample and Hold circuit is used for signal recovery.

Answer

In both the natural sampling and flat-top sampling methods, the spectrum of the signals are scaled by the ratio τ/Ts, where τ is the pulse duration and Ts is the sampling period. Since this ratio is very small, the signal power at the output of the reconstruction filter is correspondingly small. To overcome this problem a sample-and-hold circuit is used.



a) Sample and Hold Circuit



b) Idealized output waveform of the circuit

The Sample-and-Hold circuit consists of an amplifier of unity gain and low output impedance, a switch and a capacitor; it is assumed that the load impedance is large. The switch is timed to close only for the small duration of each sampling pulse, during which time the capacitor charges up to a voltage level equal to that of the input sample. When the switch is open , the capacitor retains the voltage level until the next closure of the switch. Thus the sample-and-hold circuit produces an output waveform that represents a staircase interpolation of the original analog signal.

Q 3 (b) State and prove sampling theorem for low pass signal and bandpass signals.

Answer

Statement:- "If a band –limited signal g(t) contains no frequency components for |f| > W, then it is completely described by instantaneous values g(kTs) uniformly spaced in time with period Ts ≤ 1/2W. If the sampling rate, f_s is equal to the Nyquist rate or greater ($f_s \geq$ 2W), the signal g(t) can be exactly reconstructed.

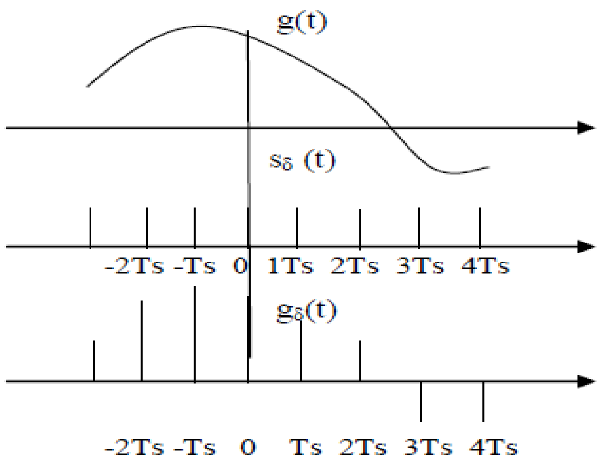


Fig : Sampling process

Proof:

Part - I If a signal x(t) does not contain any frequency component beyond W Hz, then the signal is completely described by its instantaneous uniform samples with sampling interval (or period) of Ts < 1/(2W) sec.

Part – II The signal x(t) can be accurately reconstructed (recovered) from the set of uniform instantaneous samples by passing the samples sequentially through an ideal (brick-wall) lowpass filter with bandwidth B, where $W \le B < fs - W$ and $fs = 1/(Ts)$.
If x(t) represents a continuous-time signal, the equivalent set of instantaneous uniform samples {x(nTs)} may be represented as,

$$\{x(nTs)\} \equiv xs(t) = \Sigma\, x(t).\delta(t- nTs) \text{ ---------- } 1.1$$

where $x(nTs) = x(t)|_{t=nTs}$, δ(t) is a unit pulse singularity function and n is an integer.The continuous-time signal x(t) is multiplied by an (ideal) impulse train to obtain {x(nTs)} and can be rewritten as,

$$xs(t) = x(t).\Sigma\, \delta(t- nTs) \text{ -------------} 1.2$$

Now, let X(f) denote the Fourier Transform F(T) of x(t), i.e.

$$X(f) = \int_{-\infty}^{+\infty} x(t).\exp(-j2\pi ft)dt \qquad \text{-------------} 1.3$$

Now, from the theory of Fourier Transform, we know that the F.T of Σ δ(t- nTs), the impulse train in time domain, is an impulse train in frequency domain:

$$F\{\Sigma\, \delta(t- nTs)\} = (1/Ts).\Sigma\, \delta(f- n/Ts) = fs.\Sigma\, \delta(f- nfs) \text{ -----} 1.4$$

If Xs(f) denotes the Fourier transform of the energy signal xs(t), we can write using Eq. (1.2.4) and the convolution property:

$$\begin{aligned} Xs(f) &= X(f)^* F\{\Sigma\, \delta(t- nTs)\} \\ &= X(f)^*[fs.\Sigma\, \delta(f- nfs)] \\ &= fs.X(f)^*\Sigma\, \delta(f- nfs) \text{--------------} 1.5 \end{aligned}$$

This equation, when interpreted appropriately, gives an intuitive proof to Nyquist"s theorems as stated above and also helps to appreciate their practical implications. Let us note that while writing Eq.(1.5), we assumed that x(t) is an energy signal so that its Fourier transform exists.
With this setting, if we assume that x(t) has no appreciable frequency component greater than W Hz and if fs > 2W, then Eq.(1.5) implies that Xs(f), the Fourier Transform of the sampled signal xs(t) consists of infinite number of replicas of X(f), centered at discrete frequencies n.fs, -∞ < n < ∞ and scaled by a constant fs= 1/Ts

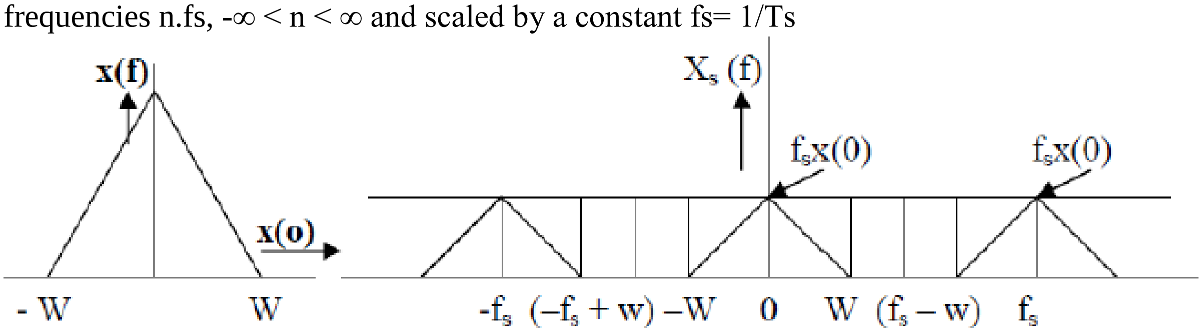


Fig. 1

Fig. 1 indicates that the bandwidth of this instantaneously sampled wave xs(t) is infinite while the spectrum of x(t) appears in a periodic manner, centered at discrete frequency values n.fs.

Part – I of the sampling theorem is about the condition fs > 2.W i.e. (fs – W) > W and (– fs + W) < – W. As seen from Fig. 1, when this condition is satisfied, the spectra of xs(t), centered at f = 0 and f = ± fs do not overlap and hence, the spectrum of x(t) is present in xs(t) without any distortion.

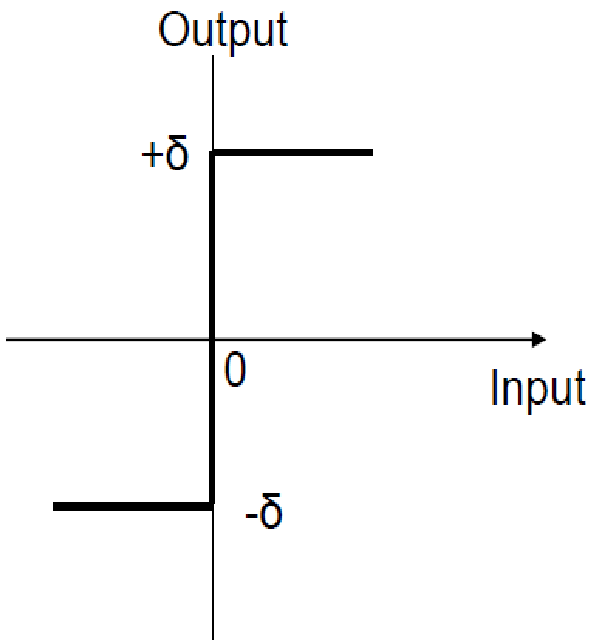
This implies that xs(t), the appropriately sampled version of x(t), contains all information about x(t) and thus represents x(t).

The second part suggests a method of recovering x(t) from its sampled version xs(t) by using an ideal lowpass filter. As indicated by dotted lines in Fig. 1, an ideal lowpass filter (with brick-wall type response) with a bandwidth W ≤ B < (fs – W), when fed with xs(t), will allow the portion of Xs(f), centered at f = 0 and will reject all its replicas at f = n fs, for n ≠ 0. This implies that the shape of the continuous time signal xs(t), will be retained at the output of the ideal filter.

Q4 (a) Explain Delta Modulation (DM) in detail with the help of neat block diagram. Also discuss its advantages and disadvantages?

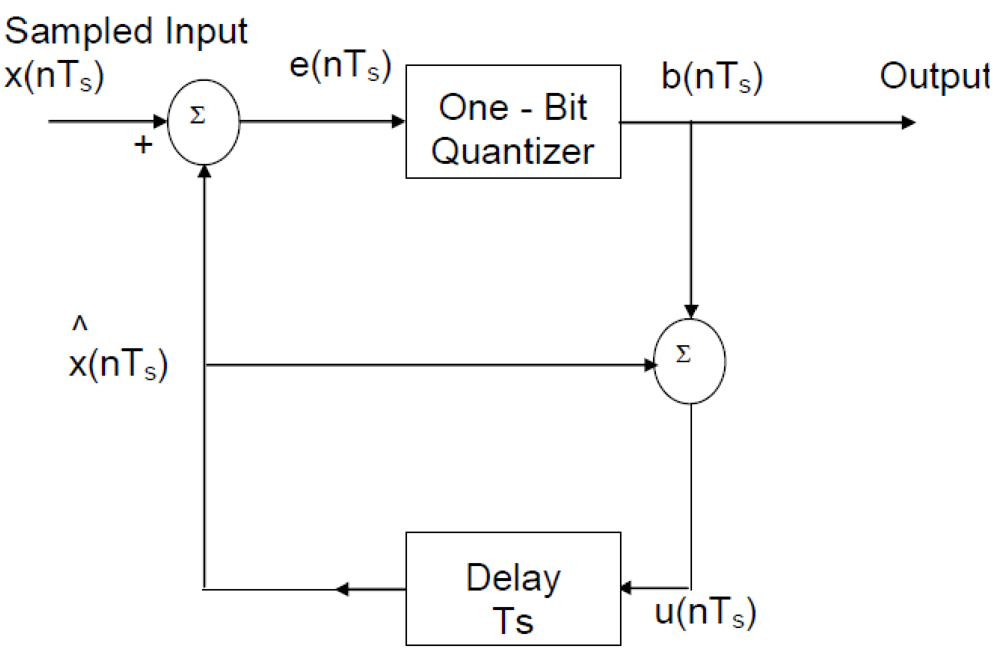
Answer

Delta Modulation is a special case of DPCM. In DPCM scheme if the base band signal is sampled at a rate much higher than the Nyquist rate purposely to increase the correlation between adjacent samples of the signal, so as to permit the use of a simple quantizing strategy for constructing the encoded signal, Delta modulation (DM) is precisely such as scheme. Delta Modulation is the one-bit (or two-level) versions of DPCM. DM provides a staircase approximation to the over sampled version of an input base band signal. The difference between the input and the approximation is quantized into only two levels, namely, ±δ corresponding to positive and negative differences, respectively, Thus, if the approximation falls below the signal at any sampling epoch, it is increased by δ. Provided that the signal does not change too rapidly from sample to sample, we find that the stair case approximation remains within ±δ of the input signal. The symbol δ denotes the absolute value of the two representation levels of the one-bit quantizer used in the DM.



Input-Output characteristics of the delta modulator.

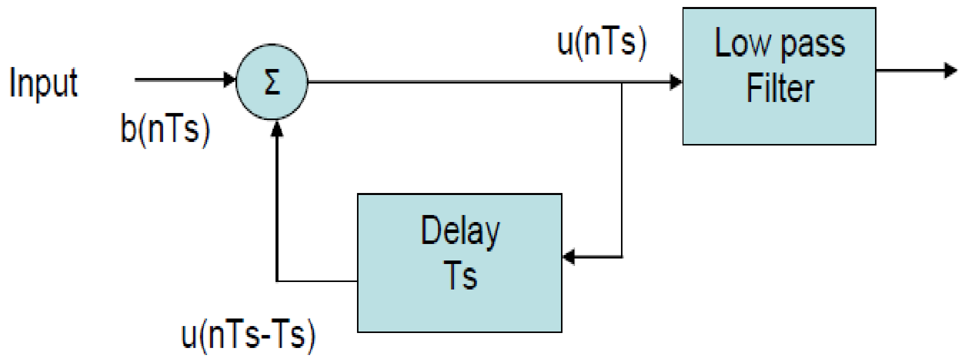
Let the input signal be x(t) and the staircase approximation to it is u(t).



Block diagram for Transmitter of a DM system

In the receiver the stair case approximation u(t) is reconstructed by passing the incoming sequence of positive and negative pulses through an accumulator in a manner similar to that used in the transmitter. The out-of –band quantization noise in the high frequency

staircase waveform u(t) is rejected by passing it through a low-pass filter with a band-width equal to the original signal bandwidth. Delta modulation offers two unique features:
1. No need for Word Framing because of one-bit code word.
2. Simple design for both Transmitter and Receiver



Block diagram for Receiver of a DM system

Q4 (b) A PCM signal uses a uniform Quantizer followed by a 7 bit binary encoder. The bit rate of the system is equal to 50 X 10^6 bits/sec.
ii) What is the maximum message bandwidth for which system operates satisfactory?
iii) Calculate the output signal to quantization noise ratio when the full load sinusoidal modulating wave of frequency 1 MHz is applied to the input.

Answer

i) Let us assume the message bandwidth is W. Sampling frequency is equal to

$$fs \geq 2W$$

The signalling rate is given as

$$r \geq n.fs$$
$$7 \text{ X } 2W$$
$$r = 50 \text{ X } 10^6 = 14 \text{ W}$$
$$W \leq 3.57 \text{ MHz}$$

The maximum bandwidth is 3.57 MHz

ii) The output S/N ratio is equal to

$$(S/N)_{dB} = 1.8 + 6n$$
$$= 1.8 + 6 \text{ X}7$$
$$= 43.8 \text{ dB}$$

Q 5 (a) Explain Inter Symbol Interference (ISI) in PCM system. How eye pattern technique is used to study ISI.

Answer

Inter symbol Interference

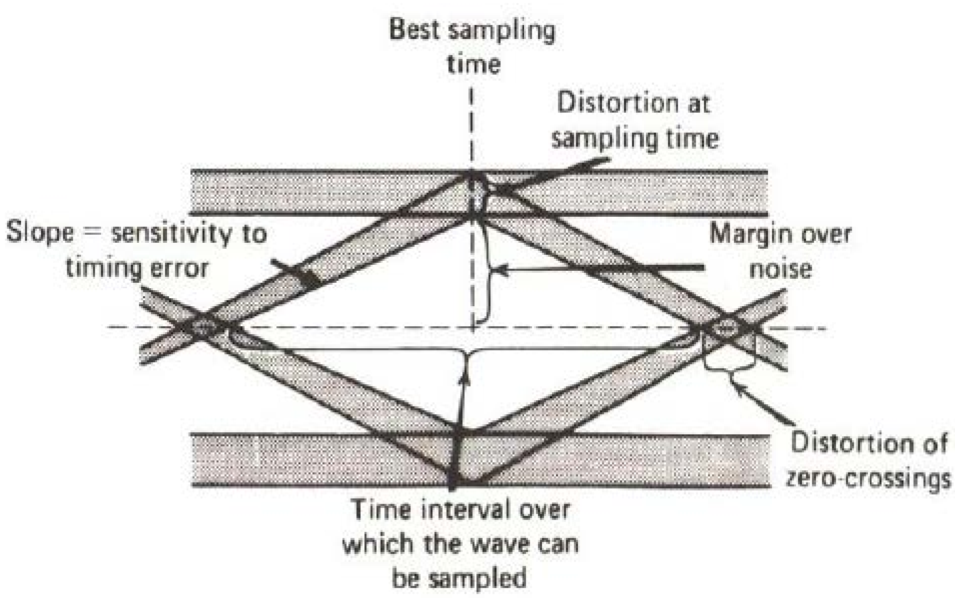
Generally, digital data is represented by electrical pulse, communication channel is always band limited. Such a channel disperses or spreads a pulse carrying digitized samples passing through it. When the channel bandwidth is greater than bandwidth of pulse, spreading of pulse is very less. But when channel bandwidth is close to signal bandwidth, i.e. if we transmit digital data which demands more bandwidth which exceeds channel bandwidth, spreading will occur and cause signal pulses to overlap. This overlapping is called **Inter Symbol Interference**. In short it is called ISI. Similar to interference caused by other sources, ISI causes degradations of signal if left uncontrolled. This problem of ISI exists strongly in Telephone channels like coaxial cables and optical fibers.

EYE PATTERN

The quality of digital transmission systems are evaluated using the bit error rate. Degradation of quality occurs in each process modulation, transmission, and detection. The eye pattern is experimental method that contains all the information concerning the degradation of quality. Therefore, careful analysis of the eye pattern is important in analyzing the degradation mechanism.

• Eye patterns can be observed using an oscilloscope. The received wave is applied to the vertical deflection plates of an oscilloscope and the sawtooth wave at a rate equal to transmitted symbol rate is applied to the horizontal deflection plates, resulting display is eye pattern as it resembles human eye.

• The interior region of eye pattern is called eye opening



• The width of the eye opening defines the time interval over which the received wave can be sampled without error from ISI

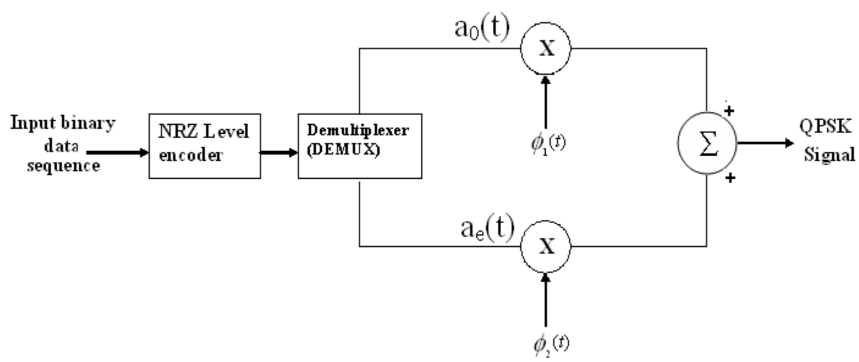
• The optimum sampling time corresponds to the maximum eye opening
• The height of the eye opening at a specified sampling time is a measure of the margin over channel noise.

The sensitivity of the system to timing error is determined by the rate of closure of the eye as the sampling time is varied. Any non linear transmission distortion would reveal itself in an asymmetric or squinted eye. When the effected of ISI is excessive, traces from the upper portion of the eye pattern cross traces from lower portion with the result that the eye is completely closed.

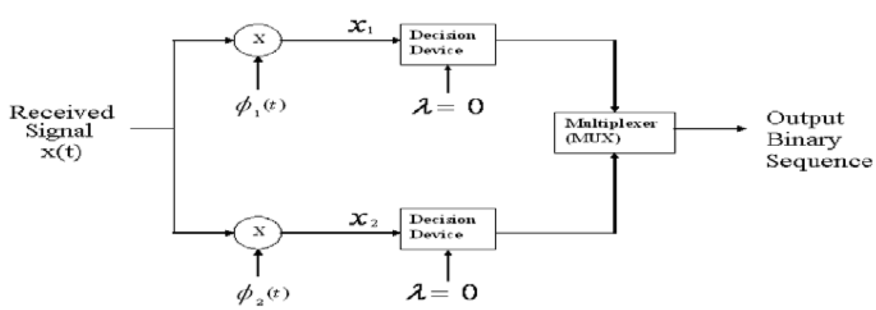
Q 6 (a) Draw and explain the block diagram of QPSK.

Answer

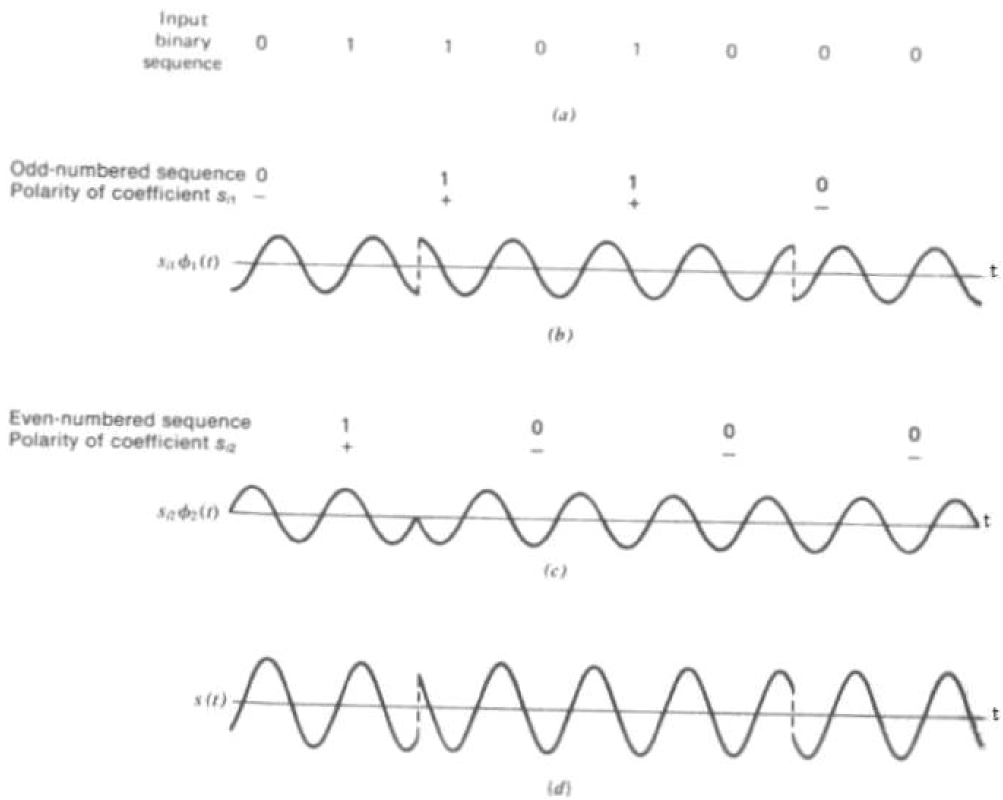
In a sense, QPSK is an expanded version from binary PSK where in a symbol consists of two bits and two orthonormal basis functions are used. A group of two bits is often called a „dibit". So, four dibits are possible. Each symbol carries same energy. Let, E: Energy per Symbol and T: SymbolDuration = 2.* Tb, where Tb: duration of 1 bit.



QPSK transmitter



QPSK receiver



In QPSK system the information carried by the transmitted signal is contained in the phase.

QPSK Receiver:- The QPSK receiver consists of a pair of correlators with a common input and supplied with a locally generated pair of coherent reference signals $\phi1(t)$ & $\phi2(t)$as shown in fig(b).The correlator outputs x1 and x2 produced in response to the received signal x(t) are each compared with a threshold value of zero. The in-phase channel output : If x1 > 0 a decision is made in favour of symbol 1 x1 < 0 a decision is made in favour of symbol 0 Similarly quadrature channel output: If x2 >0 a decision is made in favour of symbol 1 and x2 <0 a decision is made in favour of symbol 0 Finally these two binary sequences at the in phase and quadrature channel outputs are combined in a multiplexer (Parallel to Serial) to reproduce the original binary sequence

Q 6 (b) For an FSK system, the following data are observed. Transmitted binary data rate = 2.5 X 10^6 bits/sec, power spectra density of noise = 10^{-20} watts/Hz. Amplitude of received signal = 1µV. Determine the average probability of symbol of symbol error assuming coherent detection [erfc(2.23)=1.84 X 10^{-3}]

Answer

The error probability of FSK is given by

$$Pe = \frac{1}{2}\,\text{erfc}\sqrt{\left(\frac{E}{2N0}\right)}$$

$$E = A^2T/2$$
$$= (1X10^{-6})^2 \text{ X } 1/(5X10^6)$$
$$= 1/5 \text{ X } 10^{-18}$$

Also psd (N0/2) = 10^{-20}
N0= 2 X 10^{-20}

$$Pe = \frac{1}{2}\,\text{erfc}(\sqrt{5}$$
$$= 0.92 \text{ X } 10^{-3}$$

Q 7 (a) Explain geometric interpretation of signals.

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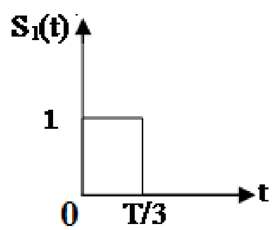
Q 7 (b) Obtain the orthonormal basis functions for the signal $S_1(t)$ shown below.



Answer
The basis function is given by $Ø_1(t)$

$$Ø_1(t) = S_1(t) / \sqrt{E}$$
$$E = \int_0^T s_1^2(t)\,dt$$
$$E = \int_0^{T/3} 1^2\,(t)\,dt$$
$$= T/3$$
$$Ø_1(t) = \begin{cases} \sqrt{\frac{3}{T}} & for\ 0 \le t \le \frac{T}{3} \\ 0 \end{cases}$$

Q 7 (c) Why do we go for Gram-Schmidit Orthogonalization procedure?

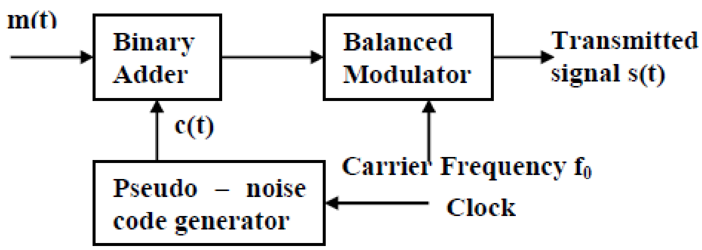
Answer

Consider a message signal m. The task of transforming an incoming message mi=1,2,…..M, into a modulated wave si(t) may be divided into separate discrete time & continuous time operations. The justification for this separation lies in the Gram-Schmidt orthogonalization procedure which permits the representation of any set of M energy signals, {si(t)}, as linear combinations of N orthonormal basis functions.

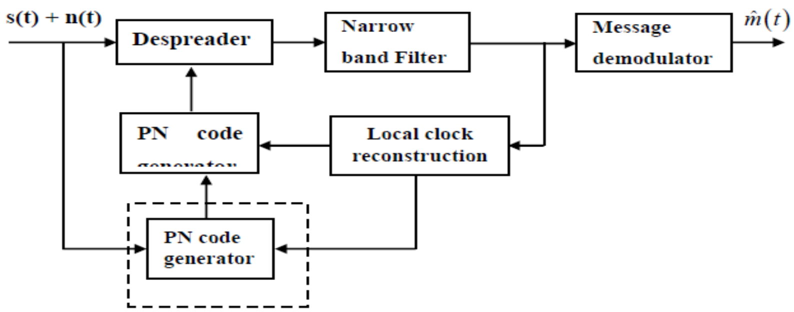
Q 8 (a) Draw the block diagram of transmitter and receiver section of direct sequence spread spectrum. Also list the advantages of DS-SS.

Answer

Block diagram of DSSS Transmitter



Block Diagram of DSSS Receiver



Advantages of Spread Spectrum

a) Reduced interference: In SS systems, interference from undesired sources is considerably reduced due to the processing gain of the system.

b) Low susceptibility to multi-path fading: Because of its inherent frequency diversity properties, a spread spectrum system offers resistance to degradation in signal quality due to multi-path fading. This is particularly beneficial for designing mobile communication systems.

c) Co-existence of multiple systems: With proper design of pseudo-random sequences, multiple spread spectrum systems can co-exist.

d) Immunity to jamming: An important feature of spread spectrum is its ability to withstand strong interference, sometimes generated by an enemy to block the communication link. This is one reason for extensive use of the concepts of spectrum spreading in military communications.

Text Book

Digital Communications, Wiley Student Edition, Simon Haykin