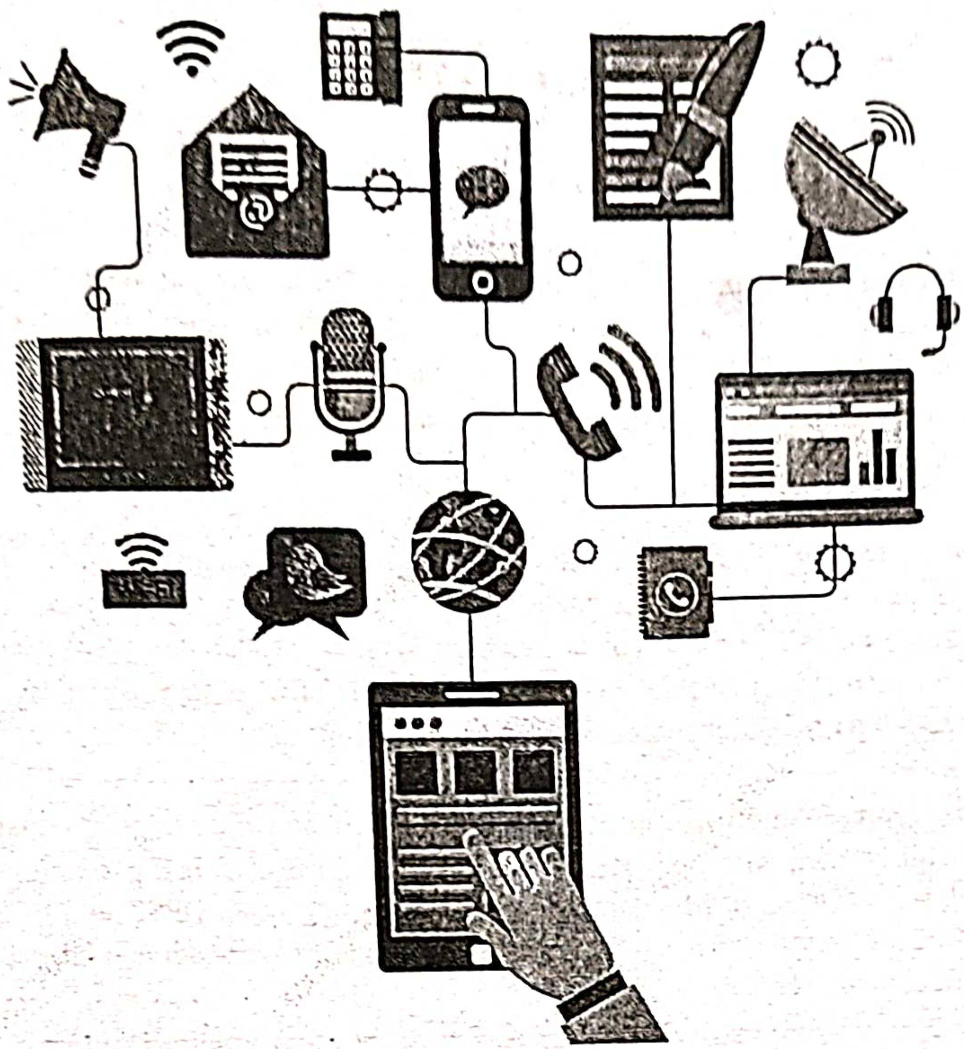


DIGITAL COMMUNICATION SYSTEMS

SYLLABUS

Chapter	Name of the Topic	Hours
Unit I	Block diagram and sub-system description of a digital communication system. Sampling of low-pass and band-pass signals, PAM, PCM, signal to quantization noise ratio analysis of linear and nonlinear quantizers, Line codes and bandwidth considerations; PCM TDM hierarchies, frame structures, frame synchronization and bit stuffing.	14
Unit II	Quantization noise analysis of DM and ADM; DPCM and ADPCM; Low bit rate coding of speech and video signals. Baseband transmission, matched filter, performance in additive Gaussian noise; Inter symbol interference (ISI), Nyquist criterion for zero ISI, sinusoidal roll-off filtering, correlative coding, equalizers and adaptive equalizers; Digital subscriber lines.	15
Unit III	Geometric representation of signals, maximum likelihood decoding; Correlation receiver, equivalence with matched filter. Generation, detection and probability of error analysis of OOK, BPSK, coherent and non-coherent FSK, QPSK and DPSK; QAM, MSK and multicarrier modulation; Comparison of bandwidth and bit rate of digital modulation schemes.	15
Unit IV	Introduction to Information and Coding Theories: Information Theory: information measures, Shannon entropy, differential entropy, mutual information, capacity theorem for point-to point channels with discrete and continuous alphabets. Coding Theory: linear block codes – definitións, properties, bounds on minimum distance (singleton, Hamming, GV, MRRW), soft versus hard decision decoding, some specific codes (Hamming, RS, Concatenated); Convolutional codes – structure, decoding (the Viterbi and BCJR algorithms); Turbo codes, LDPC codes.	16
	TOTAL	64

UNIT 01

DESCRIPTION OF A DIGITAL COMMUNICATION SYSTEM

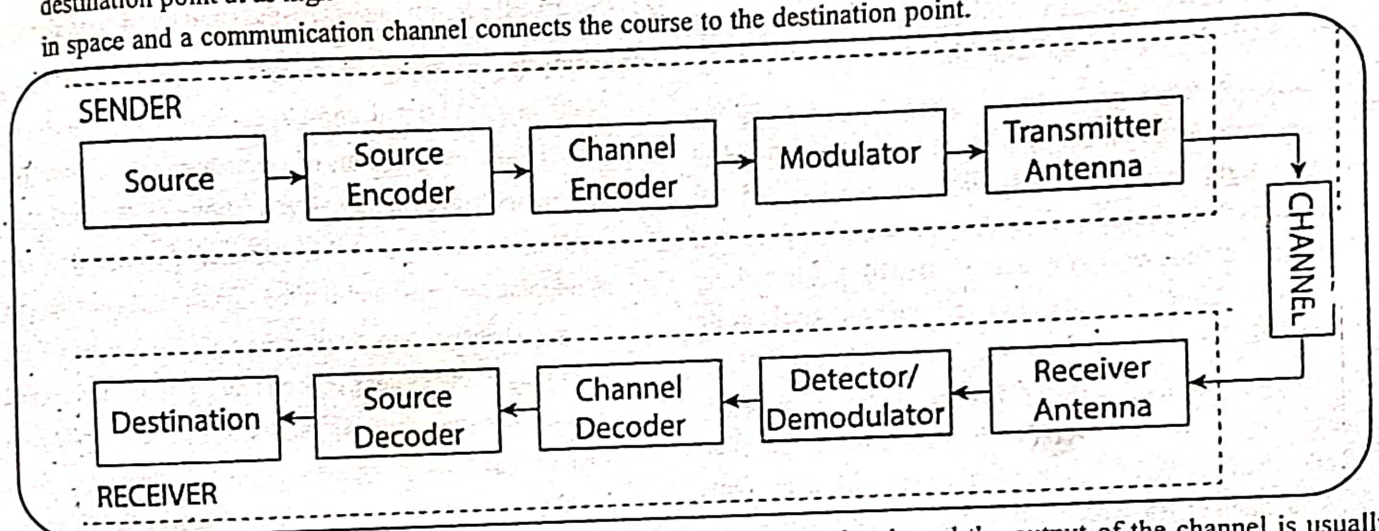
QUESTIONS AND ANSWERS

Q1. Describe digital communication system and draw its Block diagram

Or, Describe sub-system of digital communication system and write its advantage and disadvantages.

Ans. Digital Communication : In digital communication, the message signal to be transmitted is digital in nature. This means that digital communication involves the transmission of information in digital form.

Block Diagram of Digital Communication system : The below figure shows the model of a digital communication system. The overall purpose of the system is to transmit the message or sequences of symbols coming out of a source to a destination point at as high a rate and accuracy as possible. The source and the destination point are physically separated in space and a communication channel connects the course to the destination point.



The communication channel accepts electrical (i.e., electromagnetic) signals and the output of the channel is usually a smeared or distorted version of the input due to the nonideal nature of the communication channel. In addition to this, the information-bearing signal is also corrupted by unpredictable electrical signals (i.e. noise) from both man-made and natural causes. Thus, the smearing and the noise introduce errors in the conformation being transmitted and limits the rate at which information can be communicated from the source to the destination.The probability of incorrectly decoding a message symbol at the receiver is often used as a measure of the performance of a digital communication system.

- **Transmitter (sender) :**

Transducer : The diagram starts with the sender, which tra~smit the data and receiver receives the information at the end. so, the data can be audio, video, image, text or any other type. now, this information maybe analog or maybe digital. If the information signal is in electric signal then it is okay. But, if the information is not in the electrical signal form, (for example audio, video, image or text) then we need to convert them in an electric signal. However, transducer converts the non-electric signal into electric signal. So, after the first block the information is in the form of electrical signal.

Source Encoding : The main task of source encoding is to reduce redundancy so we can use bandwidth effectively. However, at this stage the digital data get compromised and there is multiple way to compress the digital data like Huffman coding and Shannon Fano coding. Moreover, for analog redundancy we go for adaptive delta modulation and Pulse-code modulation. So, after this block the signal will be digital.

Channel Encoding : It is used to provide noise immunity by adding redundancy in it. There are many techniques available for it like Block code, Cyclic code, Convolutional code and many moretechnics are available. After this block the signal will be still digital signal.

Digital Modulator and Antenna : To transmit information at longer distance we need to convert the low frequency digital signal into high frequency analog signal. Digital Modulator multiplies the digital signal we converted in last stage with high frequency carrier signal and convert it in high frequency analog signal so we can easily send it by antenna.

- **Receiver**

Digital Demodulator and Antenna : You may understood by its name that it will demodulate the signal that received by the receiver antenna. And again convert the high frequency analog signal digital signal.

Channel Decoding : Let's start from the beginning, We transmitted a signal, firstly that converted into electrical signal(digital or analog). After it we source encoded that signal and converted it into digital signal. Suppose that the source code we get of that is [010101]. After doing channel encoding we get signal channel code 011010101(suppose).

Now the digital modulator will transmit the signal but before reaching to the receiver the signal may get disturbed because of noise. So the information may not remain same, it will change. The wrong information will got at the other end. And this thing is dangers. This is why channel decoding is important at the receiver side.

Source Decoding : Source decoder will convert the digital signal into the analog signal. And that will again converted into information signal by the speaker, television or any other device. This is how our one message reach to someone within a second.

Advantages :

1. It is Simple and cheaper because of integrated circuits became smaller, speedy and chep.
2. More privacy and security through the use of encryption because, we can re-arrange digital data.

Data correction, error detection and error correction is possible

3. Flexible hardware implementation because, if hardware will change we can change the programing language.
4. Easier and sufficient multiplexing by TDMA(Time-division multiple access) & CDMA(Code-Division Multiple Access)

Disadvantages:

1. High power consumption due to multiple stages and complex circuit as we show in the Digital Communication Block Diagram.
2. Bandwidth per channel is very high.
3. Synchronization is compulsory, if we don't synchronize the data so there will be many errors in information.

Q2. Describe the Sampling of low-pass and band-pass signals.

Ans. Sampling of low-pass signals : Consider sampling a continuous real signal whose spectrum is shown in Figure. Notice that the spectrum is symmetrical about zero Hz, and the spectral amplitude is zero above +B Hz and below –B Hz, i.e., the signal is band-limited. (From a practical standpoint, the term band-limited signal merely implies that any signal energy outside the range of ±B Hz is below the sensitivity of our system.) Given that the signal is sampled at a rate of fs samples/s, we can see the spectral replication effects of sampling in Figure, showing the original spectrum in addition to an infinite number of replications, whose period of replication is fs Hz. (Although we stated in Section that frequency-domain representations of discrete-time sequences are themselves discrete, the replicated spectra in Figure are shown as continuous lines, instead of discrete dots, merely to keep the figure from looking too cluttered. We'll cover the full implications of discrete frequency spectra. Figure Spectral replications: original continuous signal spectrum; spectral replications of the sampled signal when fs/2 > B; frequency overlap and aliasing when the sampling rate is too low because fs/2 < B.

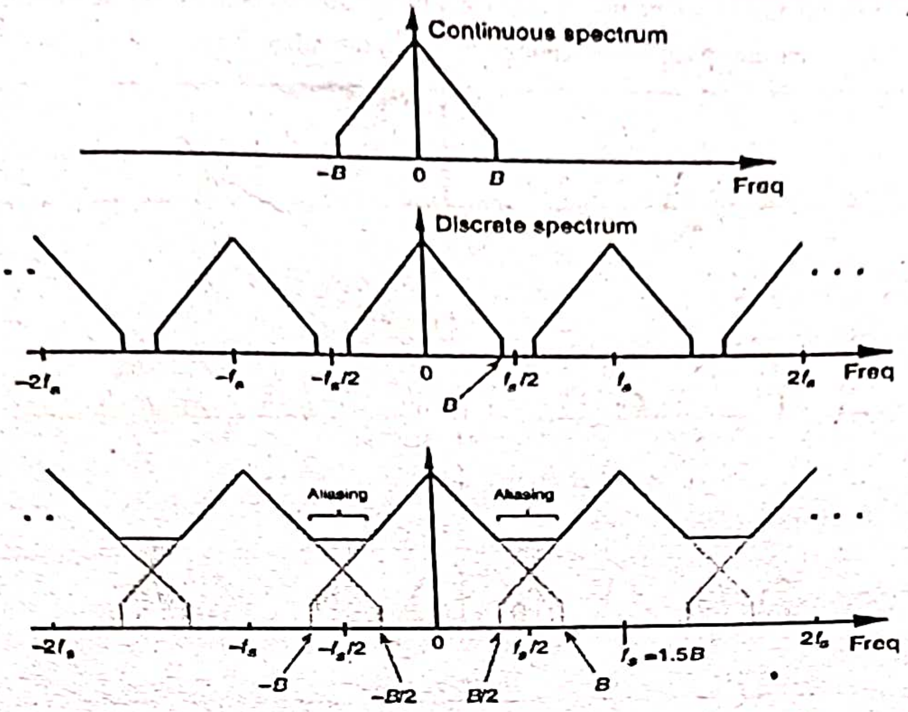
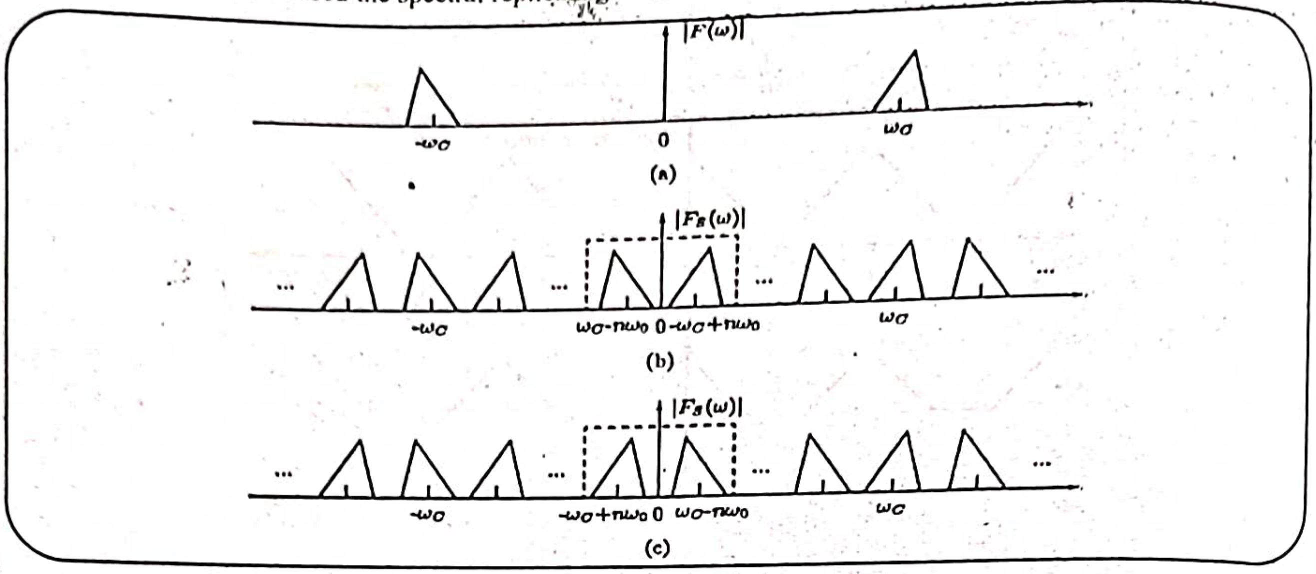


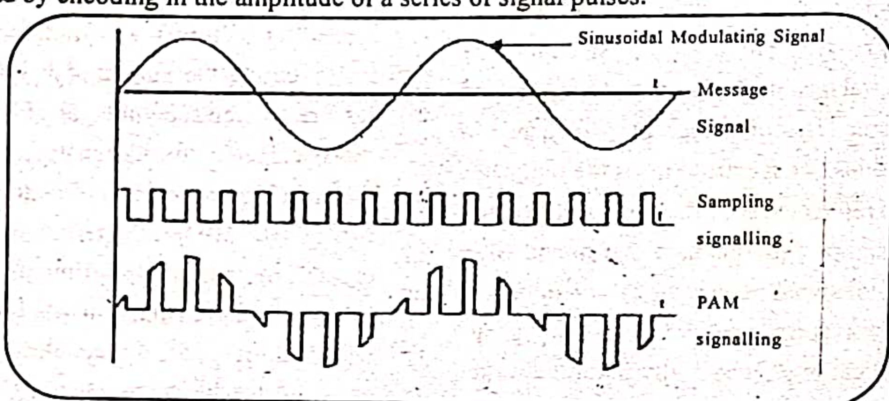
Figure is the spectrum of a continuous signal, a signal that can only exist in one of two forms. Either it's a continuous signal that can be sampled, through A/D conversion, or it is merely an abstract concept such as a mathematical expression for a signal. It cannot be represented in a digital machine in its current band-limited form. Once the signal is represented by a sequence of discrete sample values, its spectrum takes the replicated form of Figure The replicated spectra are not just figments of the mathematics; they exist and have a profound effect on subsequent digital signal processing. The replications may appear harmless, and it's natural to ask, "Why care about spectral replications? We're only interested in the frequency band within ±fs/2." Well, if we perform a frequency translation operation or induce a change in sampling rate through decimation or interpolation, the spectral replications will shift up or down right in the middle of the frequency range of interest ±fs/2 and could cause problems.

Sampling of band-pass signals : Although satisfying the majority of sampling requirements, the sampling of low-pass signals, as in Figure is not the only sampling scheme used in practice. We can use a technique known as bandpass sampling to sample a continuous bandpass signal that is centered about some frequency other than zero Hz. When a continuous input signal's bandwidth and center frequency permit us to do so, bandpass sampling not only reduces the speed requirement of A/D converters below that necessary with traditional low-pass sampling; it also reduces the amount of digital memory necessary to capture a given time interval of a continuous signal. By way of example, consider sampling the band-limited signal shown in Figure centered at fc = 20 MHz, with a bandwidth B = 5 MHz. We use the term bandpass sampling for the process of sampling continuous signals whose center frequencies have been translated up from zero Hz. What we're calling bandpass sampling goes by various other names in the literature, such as IF sampling, harmonic sampling, sub-Nyquist sampling, and under sampling. In bandpass sampling, we're more concerned with a signal's bandwidth than its highest frequency component. Note that the negative frequency portion of the signal, centered at −fc, is the mirror image of the positive frequency portion—as it must be for real signals. Our bandpass signal's highest frequency component is 22.5 MHz. Conforming to the Nyquist criterion (sampling at twice the highest frequency content of the signal) implies that the sampling frequency must be a minimum of 45 MHz. Consider the effect if the sample rate is 17.5 MHz shown in Figure. Note that the original spectral components remain located at ±fc, and spectral replications are located exactly at baseband, i.e., butting up against each other at zero Hz. Figure shows that sampling at 45 MHz was unnecessary to avoid aliasing—instead we've used the spectral replicating effects to our advantage.



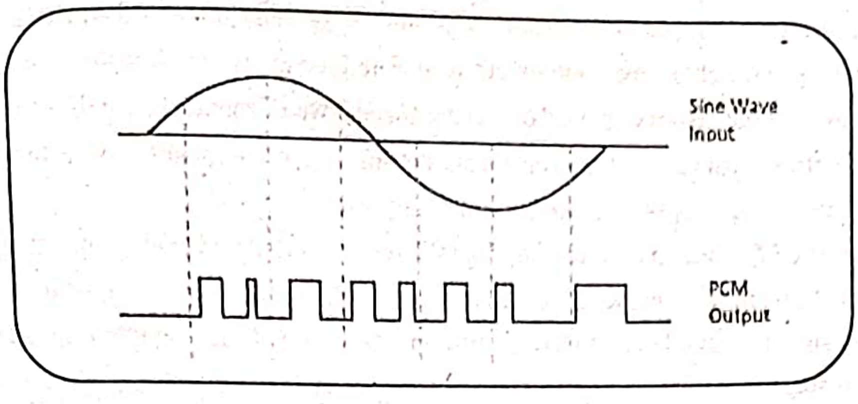
Q3. Explain Pulse Amplitude Modulation (PAM)

Ans. Pulse Amplitude Modulation : Pulse amplitude modulation is a technique in which the amplitude of each pulse is controlled by the instantaneous amplitude of the modulation signal. It is a modulation system in which the signal is sampled at regular intervals and each sample is made proportional to the amplitude of the signal at the instant of sampling. This technique transmits the data by encoding in the amplitude of a series of signal pulses.



Q4. Explain pulse code modulation (PCM).

Ans. Pulse code modulation (PCM) is a digital scheme for transmitting analog data. The signals in PCM are binary; that is, there are only two possible states, represented by logic 1 (high) and logic 0 (low). This is true no matter how complex the analog waveform happens to be. Using PCM, it is possible to digitize all forms of analog data, including full-motion video, voices, music, telemetry, and virtual reality (VR). To obtain PCM from an analog waveform at the source (transmitter end) of a communications circuit, the analog signal amplitude is sampled (measured) at regular time intervals. The sampling rate, or number of samples per second, is several times the maximum frequency of the analog waveform in cycles per second or hertz. The instantaneous amplitude of the analog signal at each sampling is rounded off to the nearest of several specific, predetermined levels. This process is called quantization. The number of levels is always a power of 2 -- for example, 8, 16, 32, or 64. These numbers can be represented by three, four, five, or six binary digits (bits) respectively. The output of a pulse code modulator is thus a series of binary numbers, each represented by some power of 2 bits. At the destination (receiver end) of the communications circuit, a pulse code demodulator converts the binary numbers back into pulses having the same quantum levels as those in the modulator. These pulses are further processed to restore the original analog waveform.



Q5. Describe Linear phase domain model approximation for mixed-signal.

Ans. This generally-adopted phase noise theory based on linear model assumption and its limit in PLL application. It begins with explaining why all types of PLL, whether be currently-adopted ones or newly-emerged ADPLL, are all non-linear system in nature. And then it explains why linear theory can be widely used in conventional PLL.

Non-linearity in mixed-signal PLL : Traditional analog PLL is a non-linear system because the phase detector's transfer function is not linear, whatever type it is. Phase detectors of traditional analog PLL are categorized in three major types:

1. Multiply phase detector
2. Digital phase detector (such as EXOR, and JK-flip flop phase detector) and
3. Phase-frequency detector (such as three-state charge-pump phase detector).

Q6. Describe Linear Approximation Model for Mixed-Signal PLL.

Ans. Non-linearity effect from quantization can be modeled as white noise of uniform probability density. Linear approximation model is structured as this white noise added to a linear model without quantization. This is similar to what has been permanently done in ADC research. But, also similar to ADC research, this linear approximation model must satisfy the following requirement -when input signal of quantizer is so big that quantization error shows irrelevance to input signal, quantization effect is equivalent to a white noise with uniform probability distribution added directly to original system; by this means, the staircase response of quantization can be modeled as "linear response" in the system.Figure demonstrates the linear phase domain model.

When $\beta=0$, it's type-I PLL ; when $\beta \neq 0$, it's type-II PLL.

Q7. Explain the term.

1. Line code.

2. Bandwidth considerations.

3. Pulse-code modulation (PCM)

4. Time-division multiplexing (TDM)

Ans. *1. Line code* : A line code is the code used for data transmission of a digital signal over a transmission line. This process of coding is chosen so as to avoid overlap and distortion of signal such as inter-symbol interference.

Properties of Line Coding : Following are the properties of line coding -

1. As the coding is done to make more bits transmit on a single signal, the bandwidth used is much reduced.
2. For a given bandwidth, the power is efficiently used.
3. The probability of error is much reduced.
4. Error detection is done and the bipolar too has a correction capability.
5. Power density is much favorable.
6. The timing content is adequate.
7. Long strings of 1s and 0s is avoided to maintain transparency.

Types of Line Coding : There are 3 types of Line Coding

1. Unipolar
2. Polar
3. Bi-polar

2. ***Bandwidth considerations*** : The term bandwidth refers to the transmission capacity of a connection and is an important factor when determining the quality and speed of a network or the internet connection.There are several different ways to measure bandwidth. Some measurements are used to calculate current data flow, while others measure maximum flow, typical flow, or what is considered to be good flow. Bandwidth is also a key concept in several other technological fields. In signal processing, for example, it is used to describe the difference between the upper and

lower frequencies in a transmission such as a radio signal and is typically measured in hertz (Hz). Bandwidth can be compared to water flowing through a pipe. Bandwidth would be the rate at which water (data) flows through the pipe (connection) under various circumstances. Instead of bits per second, we might measure gallons per minute. The amount of water that possibly can flow through the pipe represents the maximum bandwidth, while the amount of water that is currently flowing through the pipe represents the current bandwidth.

3. ***Pulse-code modulation (PCM):*** is a method used to digitally represent sampled analog signals. It is the standard form of digital audio in computers, Compact Discs, digital telephony and other digital audio applications. In a PCM stream, the amplitude of the analog signal is sampled regularly at uniform intervals, and each sample is quantized to the nearest value within a range of digital steps.

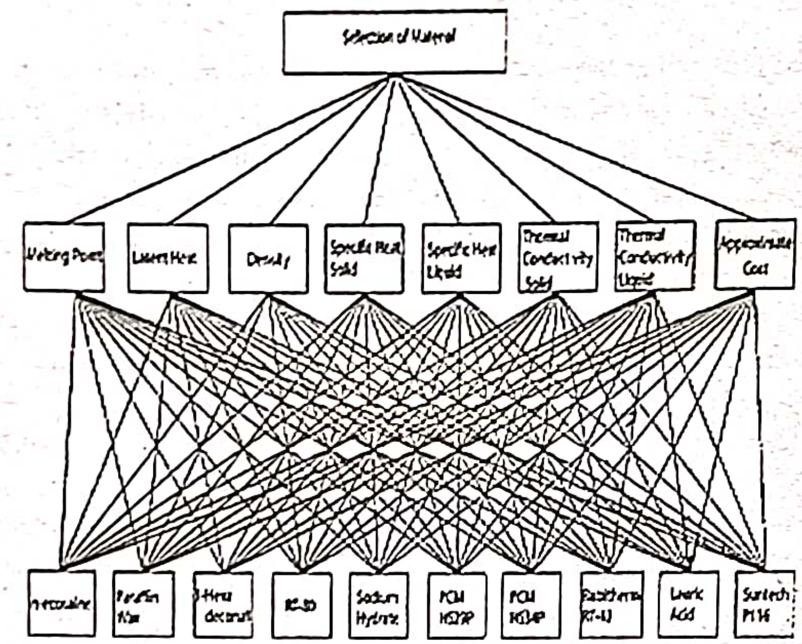


Fig. PCM hierarchy

4. ***Time-division multiplexing (TDM) :*** is a method of transmitting and receiving independent signals over a common signal path by means of synchronized switches at each end of the transmission line so that each signal appears on the line only a fraction of time in an alternating pattern. It is used when the data rate of the transmission medium exceeds that of signal to be transmitted.

- Time Division Multiplexing TDM Hierarchy :

  North America and Canada identify the digital time division multiplexing (TDM) hierarchy W as "T," as in T1, T3, and so on. Japan identifies the digital TDM hierarchy as "J," as in J1, J3, and so on. International identification of the digital TDM hierarchy is "E," as in E1, E3, and so on.

  Table 13-1 details the number of [DS0] channels and data rates for each TDM hierarchy level.

North America (Canada, Japan)			International (ITU-T)		
Designation	No. of Channels (DS0s)	Dt. Rate (Mbps)	Level	No. of Channels (DS0s)	Dt Rate (Mbps)
DS-1	24	1.544	1	30	2.048
DS-1C	48	3.152	2	120	8.448
DS-2	96	6.312	3	480	34.368
DS-3	672	44.736	4	1920	139.264
DS-4111	4032	274.176	5	7680	565.148

At the time AT&T BellLabs introduced the DS-4, the Optical Carrier (OC) was developed, which enabled the bundling of several DS-3s to be transmitted via fiber-optic cable.

Q. Explain Frame structure of broad-band wireless communication system.

Ans. The frame structure of the future broad-band wireless multimedia communication system using millimeter wave frequencies is investigated. Both portability and small power consumption of the terminal are realized by the gain of base station (BS) antenna. The frame structure in this system is constructed in terms of medium access control (MAC), user data slot allocation, and call control. In addition, tracking of mobile terminals (MTs) and wireless path selection (WPS) are considered for the sake of narrow beam width of the high gain antenna. The performances of WPS taking into account both the MT tracking and the shadowing caused by human bodies are evaluated by computer simulation. Furthermore, the effects of the overlap of covered area by-each BS are evaluated. The throughput performances of two configurations of whose overlap differs are compared. The results show that the proposed frame structure enables the site diversity configuration to become effective even for MTs.

Q9. What Does Frame Synchronization Mean?

Ans. The term frame synchronization is used in two different contexts. In the case of video, it refers to the process of synchronizing display pixel scanning to a synchronization source. In the case of telecommunication, it is the process by which incoming framed data are extracted for decoding with the help of frame alignment signals. This process is called as such because framing and synchronization must be carried out whenever a bit slip event occurs during data transmission.Frame synchronization can be defined as the process of identifying valid data from a framed data transmission. When data frames are transmitted to a receiver from the sender but get interrupted, the receiver must resynchronize. The process used for the synchronization between the sender and the receiver is known as frame synchronization.

Some of the common frame synchronization schemes are as follows:

1. Framing bit
2. Syncword framing
3. Cyclic redundancy check-based framing

The following are the four major methods of frame synchronization:

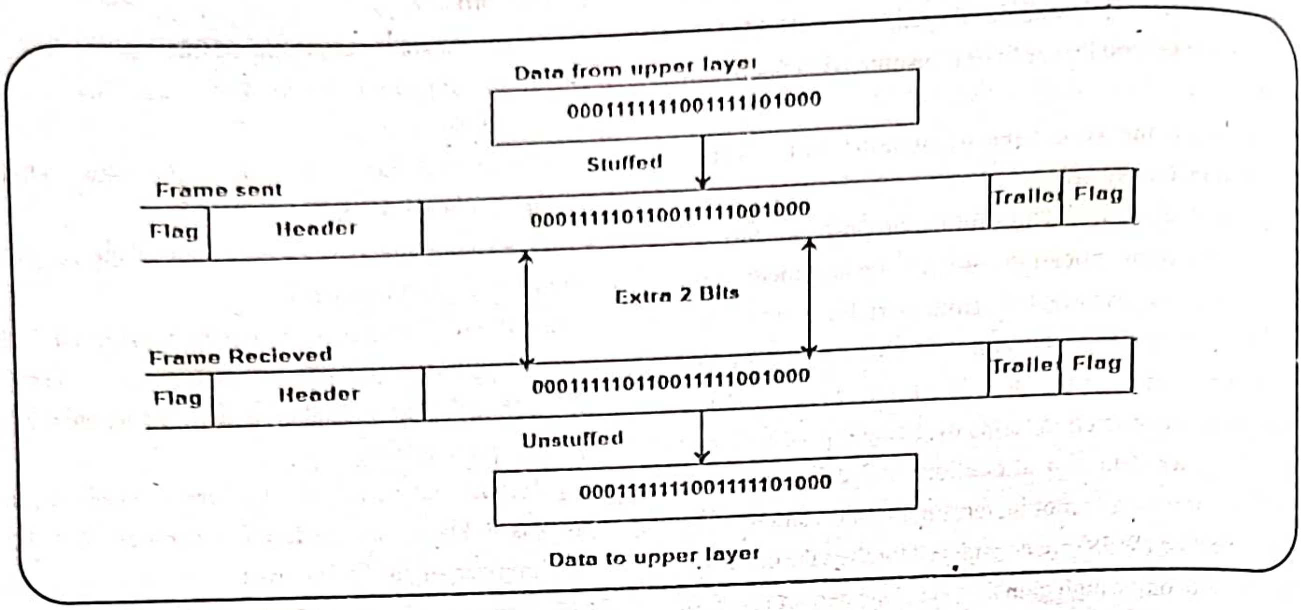
* Time based -- Uses a specific period of time between frames for the synchronization.
* Character counting -- Uses the count of the remaining characters in the frame header.
* Byte stuffing -- Uses special byte sequences like DLE (data link escape), STX (start of text) and ETX (end of text).
* Bit stuffing -- Uses special bit patterns to denote the start and the end of a frame.

The system that carries out the frame synchronization process is known as the frame synchronizer. A frame synchronizer aligns the frames of a pulse code modulation binary stream. Cross-correlation, self-referential synchronization or any similar methods can be used in the frame synchronization process. The media access control sublayer of the data link layer usually takes care of the frame synchronization process, which determines where one frame of data ends and the next one starts.

In the case of video playback, frame synchronization refers to the process of matching the timing of an incoming video source to the timing of an existing video system. The frame synchronizer used in television production matches the time base of each frame in a video to a professional video system. It also makes use of a common gunlock signal to make sure that all the equipment works with a common time base. This type of frame synchronizer is used to correct the glitches that may arise in video playbacks.

Q10. Explain bit stuffing in computer Network and its Application.

Ans. Bit Stuffing in Computer Network · Data link layer is responsible for something called Framing, which is the division of stream of bits from network layer into manageable units (called frames). Frames could be of fixed size or variable size. In variable-size framing, we need a way to define the end of the frame and the beginning of the next frame. Bit stuffing is the insertion of non information bits into data. Note that stuffed bits should not be confused with overhead bits. Overhead bits are non-data bits that are necessary for transmission (usually as part of headers, checksums etc.).



Applications of Bit Stuffing –

1. Synchronize several channels before multiplexing
2. Rate-match two single channels to each other
3. Run length limited coding

OBJECTIVE QUESTIONS AND ANSWERS

1) What are the primary features of a transmitter?
   (a) Lower clock speed (b) Lower transmitting power
   (c) Higher clock speed (d) None of these

Answer: d

2) The window gives a number of
   (a) Bytes (b) Frames
   (c) Both option a and b
   (d) None of these

Answer: c

3) A speech signal, band-limited to 8 kHz with peak to peak between +20 V to - 20 V, and the signal are sampled at Nyquist rate, and the bits 0 and 1 are transmitted using bipolar pulses. Find the minimum bandwidth for distortion-free transmission in KHz?
   (a) 98 (b) kHz
   (c) 64 kHz (d) 52 kHz
   (e) 35 kHz

Answer: b

4) Which of the given device does the data compression?
   (a) Switches (b) Circuit breakers
   (c) Microcontroller (d) Source encoder

Answer: d

5) According to Federal communications commission regulations, the maximum allowable frequency deviation is 40 kHz, for a TV signal, given that the percentage modulation of the audio portion is 50%. Find the frequency deviation of the audio signal in KHz?
   (a) 10 (b) kHz
   (c) 20 kHz (d) 30 kHz
   (e) 40 kHz

Answer: b

6) Which of the given filter has maximum flatness?
   (a) Bessel filter (b) Butterworth filter
   (c) Low pass filter (d) None of these

Answer: b

7) For an Amplitude modulation signal, the bandwidth is 20 kHz, and the highest frequency component present is 650 kHz. Find the carrier frequency used for this amplitude modulation signal?
   (a) 640 (b) kHz
   (c) 900 kHz (d) 440 kHz
   (e) 260 kHz

Answer: a

8) Space loss occurs due to a decrease in
   (a) Phase shift (b) Momentum
   (c) Electric field strength
   (d) Power

Answer: c

9) Analog data with the highest harmonic at 40 kHz generated by a sensor has been digitized using 6 level PCM. Find the rate at which digital signal generated?

(a) 300 kbps (b) 240 kbps
(c) 450 kbps (d) 600 kbps

Answer: b

10) A satellite receiver with a noise figure of 5.6 dB has a bandwidth of 24 kHz and comprises a preamplifier with a noise temperature 150 k and a gain of 40 dB. If the reference temperature is 300 k, find the equivalent noise temperature of the receiver?
(a) 450 K (b) 300 K
(c) 220 K (d) 150 K

Answer: d

11) In frequency modulation broadcast, the maximum deviation is 80 kHz, and the maximum modulating frequency is 20 kHz. In reference to Carson's rule, find the maximum required bandwidth?
(a) 300 KHz (b) 200 KHz
(c) 150 KHz (d) 80 KHz

Answer: b

12) If noise figure of a receiver is 1.8 at 20 oC, find its equivalent noise temprature?
(a) 474.9 K (b) 384.8 K
(c) 234.4 K (d) 184.6 K

Answer: c

13) If a 120 V carrier peak changes from 170 V to 50 V by a modulating signal, find the modulation factor?
(a) 0.5 (b) 1.5
(c) 2.5 (d) 3.5

Answer: a

14) RF carrier 15 kV at 1 MHz is amplitude modulated and modulation index is 0.8. Find the peak voltage of the signal?
(a) 18 kV (b) 22 kV
(c) 26 kV (d) 12 kV

Answer: d

15) A broadcast amplitude modulation radio transmitter radiates 140 kW when the modulation percentage is 75. Find the carrier power?
(a) 120 kW (b) 142 kW
(c) 109 kW (d) 172 kW

Answer: c

16) The power spectral density of a signal is
(a) Even negative and complex
(b) Odd, complex, and positive
(c) Real, odd, and negative
(d) Real, even, and non-negative

Answer: d

17) A 1200 W carrier is amplitude modulated and has a sideband power of 400 W. Find the depth of the modulation?
(a) 1.245 (b) 0.775
(c) 3.639 (d) 1.059

Answer: b

18) A modulater is a device used to
(a) Differenciates two frequencies
(b) Amplify two radio frequency signal
(c) Impress the information on to a radio frequency carrier
(d) Reduce the modulating power requirement.

Answer: d

19) If the two signals modulate the same carrier with different modulation depths of 0.4 and 0.8. Find the resulting modulation signal?
(a) 0.89 (b) 0.66
(c) 0.54 (d) 0.16

Answer: a

20) A balanced modulator is used in the generation of which of the following signal?
(a) Frequency Modulation signal
(b) DSB- SC signal
(c) ISI signal
(d) SSB -SC signal

Answer: b

21) A given AM broadcast station transmits an average carrier power output of 50kW and uses a modulation index of 0.804 for sign wave modulation. Find the maximum amplitude of the output if a 60 ohm resistive load represents the antenna?
(a) 5.88 kV (b) 7.66 kV
(c) 3.95 kV (d) 1.05 kV

Answer: c

22) Which of the given modulator is an indirect way of generating FM?
(a) Inductance FET modulator
(b) Armstrong modulator
(c) Reactance tube modulator
(d) Zener diode modulator

Answer: b.

23) In a frequency modulation system, the maximum frequency deviation is 2000, and modulating frequency is 2kHz. Find the modulation index ??
(a) 1 (b) 2
(c) 3 (d) 4

Answer: a

24) The maximum deviation allowed in a frequency modulation system is 120 kHz. If the modulating signal frequency is 20 kHz, find the bandwidth requirement as per Carson's rule?
(a) 120 kHz (b) 240 kHz
(c) 280 kHz (d) 320 kHz

Answer: c

25) In FM modulation, when the modulation index increases, the transmitted power?

(a) Half (b) Decreased
(c) Doubled (d) Unchanged

Answer: d

26) In a frequency modulated system, when the audio frequency is 600 Hz, and audio frequency voltage is 2.8 V, the frequency deviation ? is 5.6 kHz. If the audio frequency voltage is now increased to 7.4 V, find the new value of deviation?

(a) 12. 7 kHz (b) 14. 8 kHz
(c) 15. 2kHz (d) 17. 6 kHz

Answer: b

27) In phase modulation, phase deviation is proportional to which of the following.

(a) The wavelength of the message signal
(b) Message signal
(c) The amplitude of the message signal
(d) The phase shift of the message signal

Answer: b

28) A system has a receiver noise resistance of 60 ohms, and it is connected to an antenna with an output resistance of 60 ohms. Find the noise figure of the system?

(a) 1 (b) 2
(c) 3 (d) 4

Answer: b

29) Which type of noise reduction by limiters in FM receivers

(a) Jhonson noise (b) Cosmic noise
(c) Impulse noise (d) Partition noise

Answer: c

30) When a radio receiver is tuned to 600 kHz, its local oscillator provides the mixer with input at 1000 kHz. Find the frequency of other stations?

(a) 1400 kHz (b) 1600 kHz
(c) 1800 kHz (d) 2200 kHz

Answer: a

UNIT 02

QUANTIZATION NOISE ANALYSIS

QUESTIONS AND ANSWERS

Q1. Write short notes on :

1. DM
2. ADM
3. DPCM
4. ADPCM

1. DM :

* It is used to reduce the bandwidth of the signal
* It has 1-bit encoder.
* It is always considered as 1-bit DPCM.

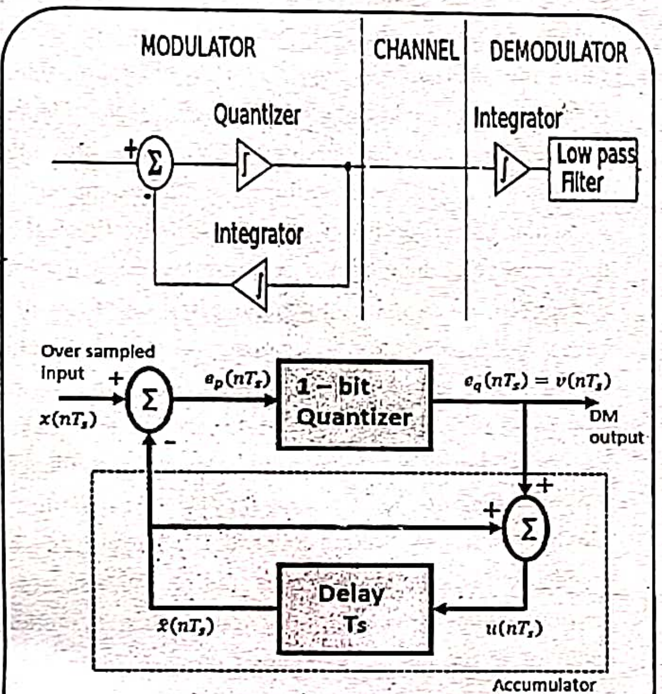
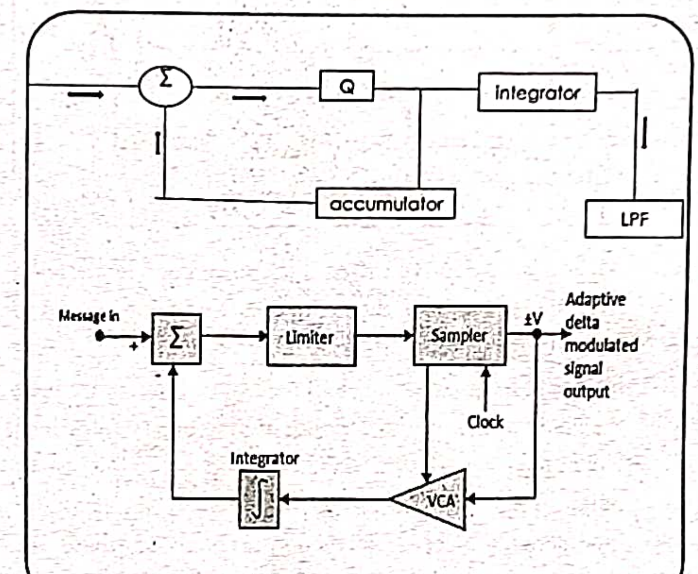


Fig. DM TRANSMITTER

2. ADM :

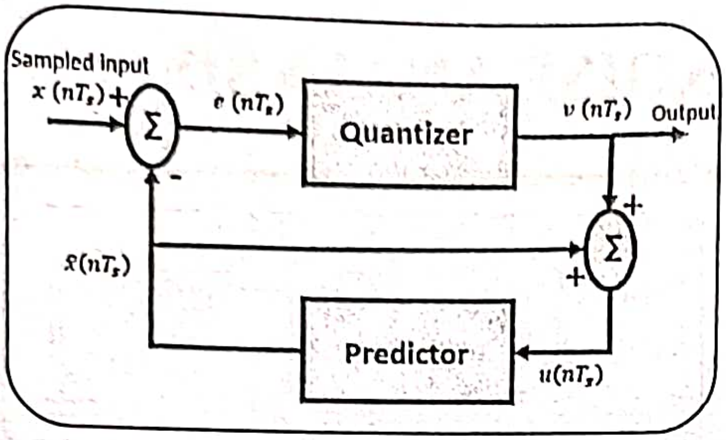
* Stands for adaptive delta modulation
* In ADM additional hardware is designed to provide variable step size .
* Thereby reducing slope overload effects without increasing the granular noise .



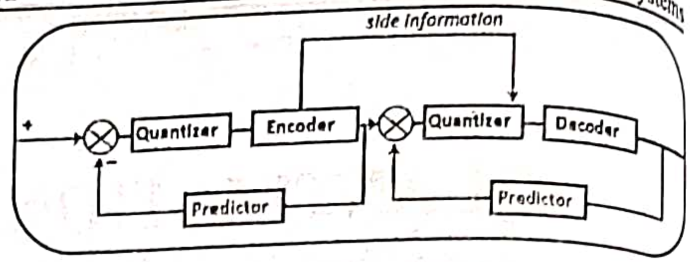
Adaptive delta modulator

3. **DPCM :** Differential pulse code modulation (DPCM) is a procedure of converting an analog into a digital signal in which an analog signal is sampled and then the difference between the actual sample value and its predicted value (predicted value is based on previous sample or samples) is quantized and then encoded forming a digital value. DPCM code words represent differences between samples unlike PCM where code words represented a sample value. Basic concept of DPCM - coding a difference, is based on the fact that most source signals show significant correlation between successive samples so encoding uses redundancy in sample values which implies lower bit rate. Realization of basic concept (described above) is based on a technique in which we have to predict current sample value based upon previous samples (or sample) and we have to encode the difference between actual value of sample and predicted value (the difference between samples can be interpreted as prediction error).Because it's necessary to predict sample value DPCM is form of predictive coding.DPCM compression depends on the prediction technique, well-conducted prediction

techniques lead to good compression rates, in other cases DPCM could mean expansion comparing to regular PCM encoding.



4. **ADPCM :** Adaptive differential pulse code modulation (ADPCM) is a method used to convert analog signals to binary signals. The technique converts the analog signals by taking frequent samples of the sound and representing the value of the sampled modulation in binary form. The technique is a variation of the digitized method known as pulse code modulation. Adaptive differential pulse code modulation is a very efficient digital coding of waveforms that was developed by Bell Labs in the 1970s for the purpose of voice coding. ADPCM was also used in the early 1990s by Interactive Multimedia Association (IMA) for the development of legacy audio codec - also referred to as ADPCM DVI, IMA ADPCM or DVI4. Some ADPCM methods are also used in VoIP communications.The concept of ADPCM is to use the past behavior of a signal to forecast it in the future. The resulting signal will represent the error of the prediction, which has no significance. Therefore, the signal must be decoded to rebuild a more meaningful original waveform.The ADPCM technique is employed to send sound signals through fiber-optic long-distance lines. This is useful especially for organizations that set up digital lines between remote sites to broadcast both voice and data. The voice signals are digitized before they are broadcasted. In the telecommunication field, the ADPCM technique is used mainly in speech compression because the method makes it possible to reduce bit flow without compromising quality. The ADPCM method can be applied to all wave forms, high-quality audio, images and other modern data.



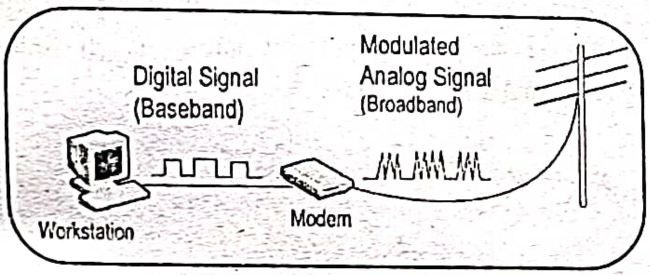
Q2. Describe Low-Bit-Rate Speech Coding and video signals.

Ans. *Low-bit-rate speech coding :* at rates below 4kb/s, is needed for both communication and voice storage applications. At such low rates, full encoding of the speech waveform is not possible; therefore, low-rate coders rely instead on parametric models to represent only the most perceptually relevant aspects of speech. While there are a number of different approaches for this modeling, all can be related to the basic linear model of speech production, where an excitation signal drives a vocal-tract filter.The basic properties of the speech signal and of human speech perception can explain the principles of parametric speech coding as applied in early vocoders. Current speech modeling approaches, such as mixed excitation linear prediction, sinusoidal coding, and waveform interpolation, use more-sophisticated versions of these same concepts. Modern techniques for encoding the model parameters, in particular using the theory of vector quantization, allow the encoding of the model information with very few bits per speech frame.Successful standardization of low-rate coders has enabled their widespread use for both military and satellite communications, at rates from 4kb/s all the way down to 600 b/s. However, the goal of toll-quality low-rate coding continues to provide a research challenge.

A New Coding Method For Low Bit-Rate Video Signals : A hybrid motion evaluation coder for low bit-rate (64 Kb/s) video sequence transmission is studied. More than one previous frames are used to estimate the temporary motion behavior in the motion evaluation process (rather than conventional motion estimation methods in which only the immediate former frame is used). The motion sequence is therefore estabished alongside the image data sequence. Through investigating and analysing the previous frame of this motion sequence, which are available to both the coder and the decoder, the moving objects can be identified from the background. The current image frame to be encoded is then divided into two types of regions: the background region which may not he transmitted at all but simply repeated with the data of the previously decoded frame, and the moving region which is encoded by compensating the motion information. The moving region is reconstructed in the normal frame sampling frequency while the background data are treated coarsely by refreshing them in a much lower frequency. The relative impact of the moving region with regard to the background region is also evaluated for assigning bits to the two separated regions. As long as this impact is significant enough, the background data may remain unchanged. The final reconstructed image frame is then the coarsely processed background superimposed by the decoded moving objects.

Q. What is Baseband Transmission?

Ans. Baseband Transmission is a signaling technology that sends digital signals over a single frequency as discrete electrical pulses. The entire bandwidth of a baseband system carries only one data signal and is generally less than the amount of bandwidth available on a broadband transmission system. The baseband signal is bidirectional so that a baseband system can both transmit and receive signals simultaneously. Baseband signals can be regenerated using repeaters in order to travel longer distances before weakening and becoming unusable because of attenuation. Baseband transmission technologies do not use modulation, but they can use time-division multiplexing (TDM) to accommodate multiple channels over a single baseband transmission line.



Baseband and Broadband , Common local area network (LAN) networking technologies such as Ethernet use baseband transmission technology. All stations on a baseband network share the same transmission medium, and they use the entire bandwidth of that medium for transmission. As a result, only one device on a baseband network can transmit at a given instant, resulting in the need for a media access control method to handle contention.

Q3. What is Matched Filter ? draw block diagram for Matched Filter.

Ans. Matched filter definition is: if a filter generates an output to maximize the output peak power ratio to mean noise power within its frequency response then it is called a matched filter. In telecommunications, it is the optimal linear filter used to increase the SNR or signal-to-noise ratio in the existence of additive stochastic noise.These types of filters are generally used in radar, where a known signal is transmitted out & the reflected signal can be compared with the transmitted signal. The best example of the matched filter is pulse compression because the impulse response can be matched with input pulse signals. In image processing, two dimensional matched filters are used to enhance the X-Ray or SNR observations

Matched Filter Block Diagram :

The block diagram of the matched filter is shown below. Consider the following diagram where g(t) is the input signal & w(t) is the white noise. These two signals are fed to the h(t) filter, which maximizes the signal-to-noise ratio(SNR) of the y(t) output.

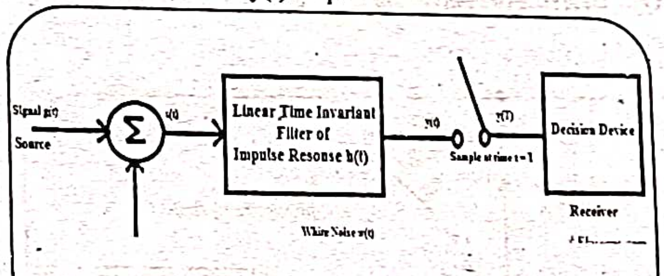


Fig. Matched Filtewhichock Diagram

The input of filter 'x(t)' includes a pulse signal 'g(t) that is corrupted through additive channel noise 'w(t)' which is shown in the following.

$$X(t) = g(t) + w(t)$$

Where 't' is an arbitrary observation interval. g(t) is the pulse signal that may signify a with a binary symbol like 0 or 1 within a digital communication system. The 'w(t)' is the sample function of a white noise procedure of zero mean & power spectral density is No/2. The source of uncertainty mainly lies in the noise.The receiver shown in the diagram should be able to receive the pulse signal g(t) with a good SNR, so the output can be given as y(t) that is sampled at t =T. So this requirement can be satisfied by optimizing the filter design to reduce the noise effects at the output of the filter in some numerical sense, and thus improve the pulse detection of the signal g(t). Since this filter is linear, then the output 'y(t)' may be simply expressed as

$$y(t) = g0(t) + n(t)$$

From the above equation, go(t) is the linear output corresponding to the pulse signal g(t) & n(t) is filtered noise. Both the go(t) and n(t) are generated by the signal & the noise components of the input x(t) correspondingly. The purpose of the matched filter is to maximise the output signal-to-noise actio.

Here, the matched filter is a type of linear filter having an impulse response h(t) in the time domain, and in frequency response, is denoted by H(f) and when signal go(t) sampled at t= T >> the average power of the filter noise. Then the maximum signal to noise ratio denoted by 'η' would be

$$\eta = |go(T)|2 / E[n^2 (t)]$$

Where,

|go(T) |2 is instantaneous power in the output signal.

E is the statistical expectation operator.

E[n^2 (t)] is the average output noise power.

Q4. Describe the performance in additive Gaussian noise.

Ans.

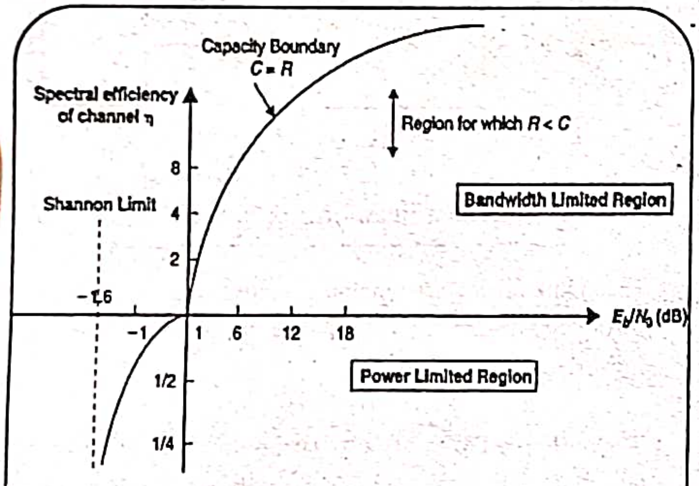


Fig. Graphical representation of Additive white Gaussian noise (AWGN)

Additive white Gaussian noise (AWGN) is a basic noise model used in information theory to mimic the effect of many random processes that occur in nature. AWGN is often used as a channel model in which the only impairment to communication is a linear addition of wideband or white noise with a constant spectral density (expressed as watts per hertz of bandwidth) and a Gaussian distribution of amplitude. The model does not account for fading, frequency selectivity, interference, nonlinearity or dispersion. However, it produces simple and tractable mathematical models which are useful for gaining insight into the underlying behavior of a system before these other phenomena are considered. The AWGN channel is a good model for many satellite and deep space communication links. It is not a good model for most terrestrial links because of multipath, terrain blocking, interference, etc. However, for terrestrial path modeling, AWGN is commonly used to simulate background noise of the channel under study, in addition to multipath, terrain blocking, interference, ground clutter and self interference that modern radio systems encounter in terrestrial operation.

Q5 . Describe Intersymbol Interference (ISI) also write its Mathematical Form.

Ans. InterSymbol Interference (ISI) is a kind of distortion that occurs when one or more symbols (pulses in digital baseband transmission) interfere with subsequent signals. This can cause noise in the signal which can cause the output to be less than ideal. ISI occurs when there is multipath propagation and/or nonlinear frequency in the channels. These causes can be reduced which can help eliminate ISI from the system to achieve an ideal output.

*** Mathematical Form :**

The following equation mathematically represents the receiver output y(t) sampled at time t_i=i*T_b (with i representing integer values):

$$y(t_i) = \mu \sum_{k=-\infty}^{\infty} a_k p(iT_b - kT_b)$$

$$= \mu a_i + \mu \sum_{\substack{k=-\infty \\ k \neq i}}^{\infty} a_k p(iT_b - kT_b)$$

The first term in the above equation, μ*a_i, is produced by the ith transmitted bit. The second term represents the residual effect or ISI of all other transmitted bits on the decoding of the ith bit.

Q6. Explain the term Nyquist Criterion for Zero-ISI

Ans. Nyquist proposed a condition for pulses p(t) to have zero-ISI when transmitted through a channel with sufficient bandwidth to allow the spectrum of all the transmitted signal to pass. Nyquist proposed that a zero–ISI pulse p(t) must satisfy the condition A pulse that satisfies the above condition at multiples of the bit period Tb will result in zero– ISI if the whole spectrum of that signal is received. The reason for which these zero–ISI pulses (also called Nyquist–criterion pulses) cause no ISI is that each of these pulses at the sampling periods is either equal to 1 at the center of pulse and zero the points other pulses are centered. In fact, there are many pulses that satisfy these conditions.

$$p(t)=\begin{cases}1 & t=0\\ 0 & t=\pm T_b,\pm 2T_b,\pm 3T_b,\ldots\end{cases}$$

For example, Any square pulse that occurs in the time period –Tb to Tb or any part of it (it must be zero at –Tb and Tb) will satisfy the above condition. Also, any triangular waveform („ Δ "function) with a width that is less than 2Tb will also satisfy the condition. A sinc function that has zeros at t = Tb, 2Tb, 3Tb, ... will also satisfy this condition. The problem with the sinc function is that it extends over a very long period of time resulting in a lot of processing to generate it. The square pulse required a lot of bandwidth to be transmitted. The triangular pulse is restricted in time but has relatively large bandwidth.

There is a set of pulses known as raised–cosine pulses that satisfy the Nyquist criterion and require slightly larger bandwidth than what a sinc pulse (which requires the minimum bandwidth ever) requires.

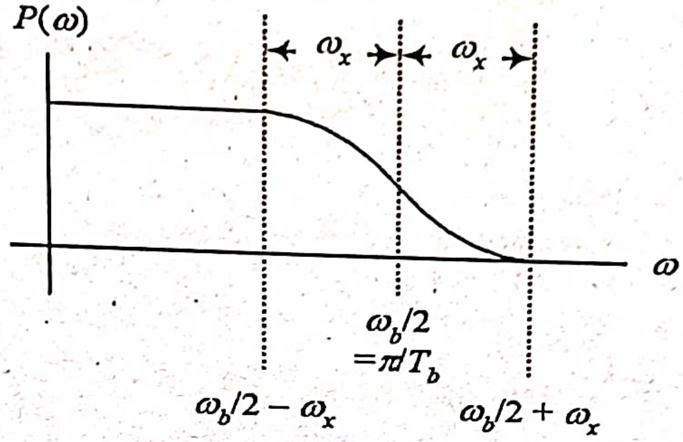
The spectrum of these pulses is given by

$$p(\omega)=\begin{cases}\frac{1}{2}\left[1-\sin\left(\frac{\pi\omega-\omega_b/2}{2\omega_x}\right)\right] & \left|\omega-\frac{\omega_b}{2}\right|<\omega_x\\ 0 & |\omega|>\frac{\omega_b}{2}+\omega_x\\ 1 & |\omega|<\frac{\omega_b}{2}+\omega_x\end{cases}$$

Where ω b is the frequency of bits in rad/s (ω b=2/Tb), and x is called the excess bandwidth and it defines how much bandwidth would be required above the minimum bandwidth that is required when using a sinc pulse. The excess bandwidth ω x for this type of pulses is restricted between.

$$0\le\omega_x\le\frac{\omega_b}{2}$$

Sketching the spectrum of these pulses we get



We can easily verify that when ω x=0, the above spectrum becomes a rect function, and therefore the pulse p(t) becomes the usual sinc function. For ω x = b/2, the spectrum is similar to a sinc function but decays (drops to zero) much faster than the sinc (it extends over 2 or 3 bit periods on each side). The expense for having a pulse that is short in time is that it requires a larger bandwidth than the sinc function (twice as much for ω x=ω b/2).

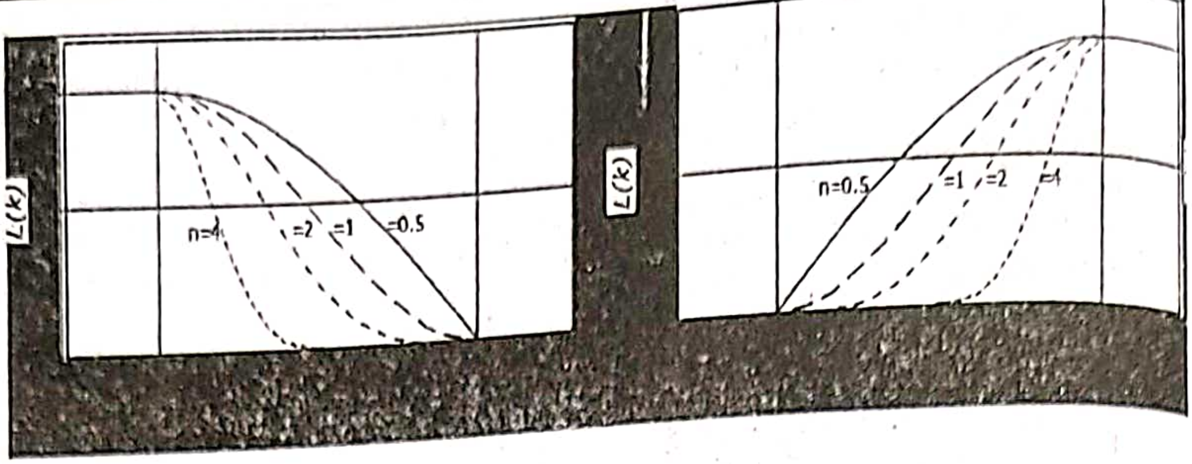
Q7. Explain Cosine Roll-off Filter.

Ans. For ease of use and interpretation, the user is prompted to enter the cutoffs in units of wavelength, starting with the small wavelength. These are internally translated to wavenumbers for processing. In the illustration below the filter is portrayed in the Fourier domain, and as a result the horizontal axis is in units of Wavenumber. Wavenumbers are the inverse of wavelengths.

$$L(k)=1,\ \text{for } k<k_0$$

$$L(k)=\cos^n\left[\frac{\pi}{2}\left(\frac{k-k_0}{k_1-k_0}\right)\right],\ \text{for } k_0\le k\le k_1$$

$$L(k)=0,\ \text{for } k>k_1$$



Parameters:

k_0 Low wavenumber, starting point of the cosine taper: k_0 = Long wavelength-1

k_1 High wavenumber, end point of the cosine taper. K_1 = Short wavelength-1

n Degree of the cosine function. The default is a degree of 2 for a cosine squared roll-off.

N Nyquist wavenumber

Q8. Describe the term Correlative coding.

Ans. The condition for zero ISI (Inter Symbol Interference) is:

$$p(nT)=\begin{cases}1, & n=0\\ 0, & n\neq 0\end{cases}$$

which states that when sampling a particular symbol (at time instant nT=0), the effect of all other symbols on the current sampled symbol is zero.

one of the practical ways to mitigate ISI is to use partial response signaling technique (otherwise called as "correlative coding"). In partial response signaling, the requirement of zero ISI condition is relaxed as a controlled amount of ISI is introduced in the transmitted signal and is counteracted in the receiver side.

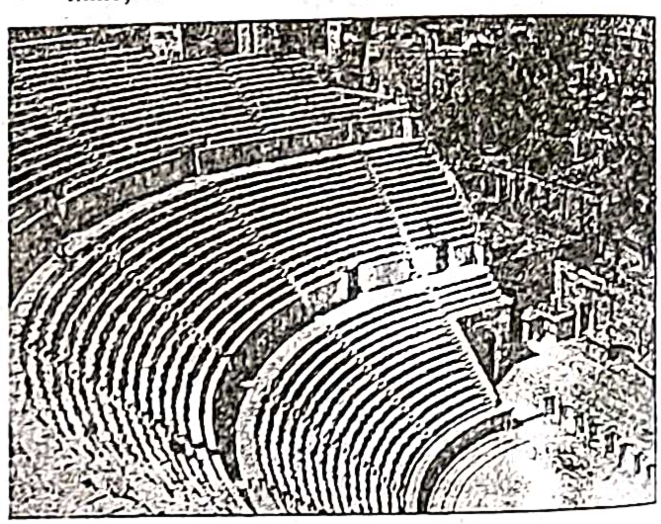
By relaxing the zero ISI condition, the above equation can be modified as,

$$p(nT)=\begin{cases}1, & n=0,1\\ 0, & \text{otherwise}\end{cases}$$

which states that the ISI is limited to two adjacent samples. Here we introduce a controlled or "deterministic" amount of ISI and hence its effect can be removed upon signal detection at the receiver.

Q. What is an Equalizer, and How does it work?

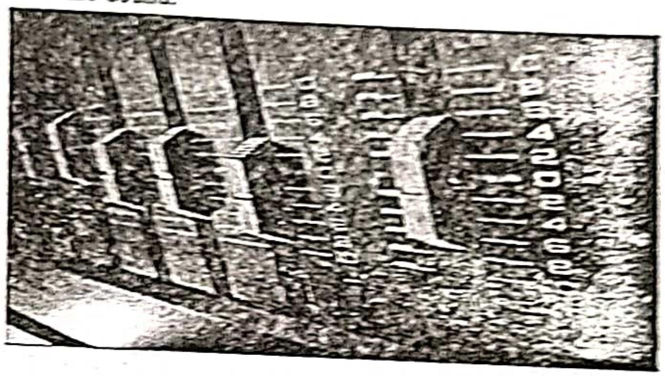
Ans. Equalizers are software or hardware filters that adjust the loudness of specific frequencies. As with all sound engineering, the basis is on the human ear. Certain frequencies are louder than others to our ears, despite having the same or even more energy behind it. Our range is around 20-20,000 Hz, and the closer we approach or exceed these boundaries, the softer things sound. Compounded by the fact that our ears, rooms, and speakers are in various shapes, sizes, and configurations, the same note from the same instrument can sound completely different, let alone a whole song! That's why ancient amphitheaters were designed with acoustic projections in mind, so voices could carry.



Equalizers were originally developed for physical venues such as movie theaters and outdoor areas, places that aren't designed with acoustics in mind, to "equalize" all of the sound frequencies. For example, some venues will respond better to bass frequencies, so the EQ can be turned down on that end to prevent feedback and turned slightly up on the higher end to even things out. In general, you equalize for the physical space, to account for the particular combination of the room and equipment.

While still used in this way for live shows and the like, everyday listeners can use EQs to not only adjust for deficiencies in their acoustics, but for more aesthetic reasons. In your car, for example, you can't really change how the sound travels much aside from speaker balance and fading. You can't move the speakers to better locations or change the layout of your seats. In this case, an EQ can be used to lessen and strengthen, or "cut" and "boost," specific ranges of frequencies.

Working of Equalized : Equalizers work in ranges, or "bands." Odds are that your car at the minimum has a dual-band EQ, meaning you can cut and boost the high and low ranges. These are also referred to as "treble" and "bass" bands, respectively. Nicer sound systems may have three, five, or even up to twelve bands. Professional music equipment uses twenty to thirty bands. The more bands you have, the more divisions you have in the wide range of human hearing. Because of this, each band controls a small range of frequencies, thus allowing more control over the sound.



Q9. Describe How Adaptive Equalization Works also draw its block diagram.

Ans. Digital communications systems are designed to transmit high-speed data through band-limited channels—for example, 6 MHz-wide downstream channels or 6.4 MHz-wide upstream channels, which are susceptible to various distortions. The presence of distortion in the channel results in something called inter-symbol interference (ISI), which can cause data transmission errors. One way to compensate for or reduce ISI is to incorporate an equalizer in the receiver or transmitter. If the channel characteristics are known and don't change over time, fixed-value equalizers can be used. However, in a typical cable network the signal path between the headend and each cable modem (or digital set-top box) is unique, so a one-size-fits-all fixed equalizer is impractical. Furthermore, distortions causing ISI can change over time, so the equalizer must somehow be adjustable to compensate for changes in channel conditions. Adaptive equalizers are most often used for this purpose.

The equalizer is a small passive circuit that has the opposite amplitude-versus-frequency response of the length of coaxial cable preceding the amp. The equalizer is in effect a broadband filter that cancels the tilted response in the operating bandwidth, resulting in a flat amplitude-versus-frequency spectrum at the second amp's internal gain stages. Adaptive Equalization

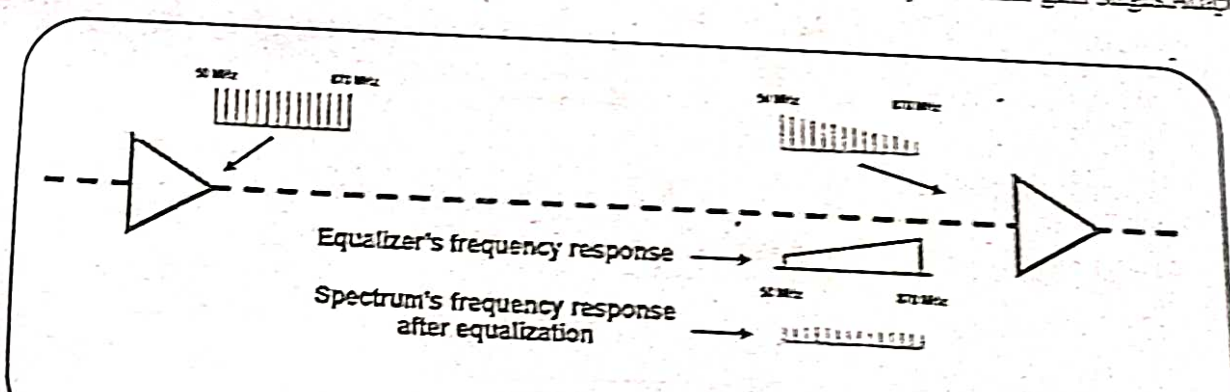
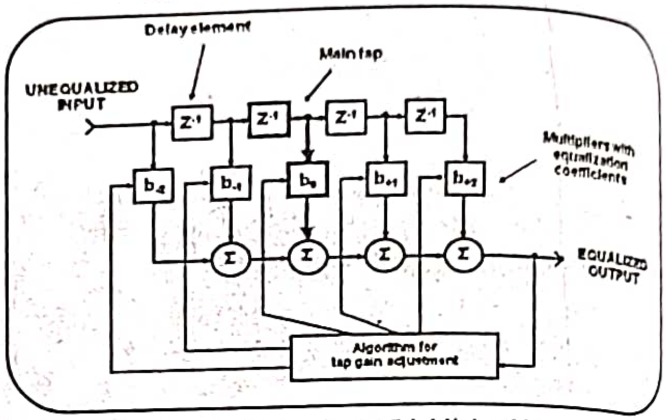


Figure 1. Equalization in the coaxial distribution network

Adaptive equalization performs a function similar to that of a cable amplifier's fixed-value plug-in equalizer. Rather than equalizing the entire downstream or upstream RF spectrum, it deals with just a single channel. Adaptive means the equalizer can change its characteristics as channel conditions change. An adaptive equalizer is a digital circuit that compensates for a digital signal's in-channel complex frequency response impairments. The cable industry has long used the term frequency response to describe amplitude (or magnitude)-versus-frequency—that is, what is seen on the display of test equipment used to sweep outside plant. True frequency response is a complex entity that has two components: amplitude-versus-frequency, and phase-versus-frequency. An adaptive equalizer can compensate for in-channel amplitude- and phase-versus-frequency impairments.

Block diagram of Adaptive equalization :

Figure illustrates a block diagram of a generic adaptive equalizer. The top row with boxes labeled Z-1 can be thought of as a tapped delay line. Each box marked Z-1 is a delay element, with the amount of time delay per "box" equal to the reciprocal of the symbol rate in a T-spaced equalizer. A delay element is often called a tap, but a tap also can be considered the combination of a delay element, the point where some of the signal is "tapped" off, and a multiplier. The boxes labeled b-2, b-1, b0, etc., are multipliers with equalization coefficients that set the gain for each tap. The algorithm adjusts the equalization coefficients that set the gain for each multiplier. The circles marked Σ are summing or combining circuits.



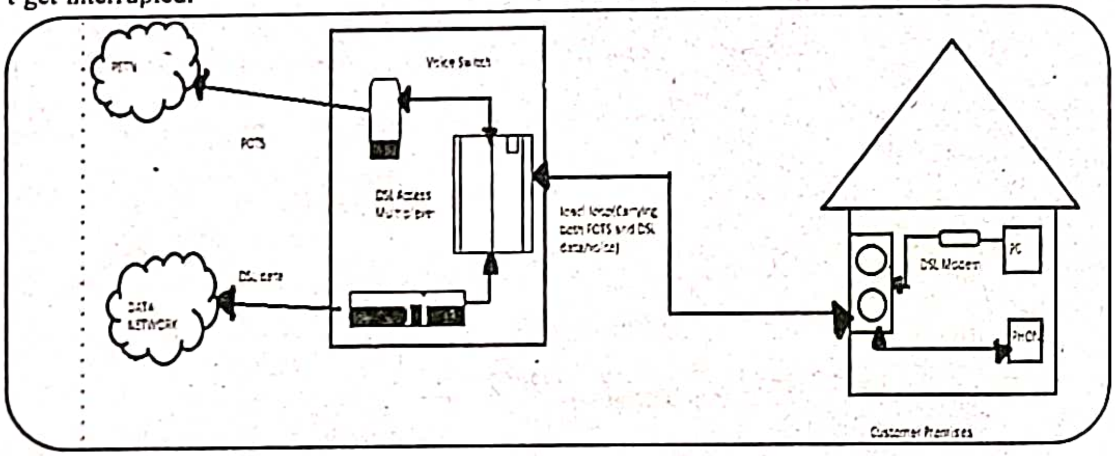
One tap is called the main tap (highlighted in red in Figure). The main tap has a gain of 1, and passes the input signal at its original amplitude. Other taps represent either the "past" or "future" relative to the main tap, and vary the amplitudes of the respective signals passing through them as required.

Q10. Explain Digital Subscriber Line (DSL), write its types and benifits.

Ans. Digital Subscriber Line (DSL, originally, digital subscriber loop) is a communication medium, which is used to transfer internet through copper wire telecommunication line. Along with cable internet, DSL is one of the most popular ways ISPs provide broadband internet access.

•Its aim is to maintain the high speed of the data being transferred.

•If we ask that how we gonna achieve such a thing i.e., both telephone and internet facility, then the answer is by using splitters or DSL filters(shown in the below diagram). Basically, the splitter is used to splits the frequency and make sure that they can't get interrupted.



Types of DSL –

1. **Symmetric DSL** – SDSL, splits the upstream and downstream frequencies evenly, providing equal speeds to both uploading

and downloading data transfer. This connection may provide 2 Mbps upstream and downstream.it is mostly preferred by small organizations.

2. Asymmetric DSL – ADSL, provides a wider frequency range for downstream transfers, which offers several times faster downstream speeds. an ADSL connection may offer 20 Mbps downstream and 1.5 Mbps upstream, it is because most users download more data than they upload.

Benefits –

- No Additional Wiring – A DSL connection makes use of your existing telephone wiring, so you will not have to pay for expensive upgrades to your phone system.
- Cost-Effective – DSL internet is a very cost-effective method and is best in connectivity
- Availability of DSL modems by the service providers.
- Users can use both telephone lines and the internet at the same time. And it is because the voice and digital signals are transferred in different frequencies.
- Users can choose between different connection speeds and pricing from various providers.

DSL Internet service only works over a limited physical distance and remains unavailable in many areas where the local telephone infrastructure does not support DSL technology. The service is not available everywhere. The connection is faster for receiving data than it is for sending data over the Internet.

OBJECTIVE QUESTIONS AND ANSWERS

1. Which modulation technique does not use past information for modulation

1. Delta modulation
2. Pulse Code modulation
3. Adaptive Differential Pulse Code Modulation
4. Adaptive Delta Modulation

Answer : 2

2. A speech signal is sampled at 8 kHz and encoded in PCM format using 8-bit/sample PCM data is transmitted through a baseband channel via 4-level PAM. Minimum Bandwidth required for transmission is

1 16 kHz 2. 8 kHz
3. 24 kHz 4. 10 kHz

Answer : 1

3. Comparing Delta Modulation (DM) with PCM systems, DM requires:

1. a lower sampling rate
2. a higher sampling rate
3. least bandwidth
4. simpler hardware

1. 12 and 4 only 2. 1,2 and 3 only
3. 2,3 and 4 only 4. 1,3 and 4 only

Answer : 3

4. Companding is used to

1. overcome quantising noise in PCM
2. protect small signals in PCM from quantising noise
3. PCM receivers to reduce impulse noise
4. increase the power content of the modulated signal

Answer : 2

5. Output SNR of a n bit PCM was found to be 30 dB, desired SNR is 42 dB. To achieve desired SNR by increasing the number of quantization levels, then new levels will be

1. 128 2. 512
3. 2018 4. 1024

Answer : 1

6. The bandwidth of a 'N' bit binary coded PCM signal for modulating a signal having bandwidth of 'f' Hz is

1. f/N Hz 2. f
3. Nf 4. N

Answer : 3

7. The temperature of a particular place varies between 14°C and 34°C. For the purpose of transmitting the temperature record of that place using PCM, the record is sampled at an appropriate sampling rate and the samples are quantized. If the error in representation of the samples due to quantization is not to exceed ± 1% of the dynamic range. what is the minimum number of quantization levels that can be used?

1 100 2. 50
3. 30 4. 15

Answer : 2

8. A signal is sampled 8 kHz and is quantized using 8 - bit uniform quantizer assuming SNR_q for a sinusoidal signal, the correct statement for PCM signal with a bit rate of R is:

1. R = 32 kbps, SNR_q = 49.8 dB
2. R = 64 kbps, SNR_q = 55.8 dB
3. R = 64 kbps. SNR_q = 49.8 dB
4. R = 32 kbps, SNR_q = 25.8 dB

Answer : 3

9. A 12-bit ADC has input signal range of ± 1 V. The signal to quantization noise ratio if a sine wave signal with 0.25 V peak voltage is given as input is:

1. 62 dB 2. 72 dB
3. 74 dB 4. 48 dB

Answer : 1

10. Discrete samples of an analog signal are uniformly quantized to PCM. If the maximum value of analog sample is to be represented within 0.1% accuracy, then the minimum number of binary digits required per sample is

1.4 2.8

3.10 4.12

Answer : 3

11. Directions: The item consists of two statements, one labeled as the 'Assertion (A)' and the other as 'Reason (R)'. You are to examine these two statements carefully and select the answers to the item using the codes given below:

Assertion (A): PCM/FM systems transmit PCM pulses by modulating a high-frequency carrier and hence occupy large bandwidth and eliminate distortion.

Reason (R): Large bandwidth ensures SNR trade-off and hence distortionless transmission is ensured.

1. Both A and R individually true and R is the correct explanation of A

2. Both A and Rare individually true but R is not the correct explanation of A

3. A is true but R is false

4. A is false but R is true

Answer : 1

12. A 13 bit PCM system performance is better than an 8 bit PCM system because:

1. Noise is lower and is proportional to reciprocal of bandwidth

2. Bandwidth is larger and detection is easier

3. Quantization noise is lesser other things being equal

4. Quantization noise is nearly zero since 2^{13} levels are available

Answer : 4

13. In a communication system, noise due to quantization error is

1. Linear and signal-dependent

2. Nonlinear and signal-dependent

3. Linear and signal independent at low frequencies only

4. Non-linear and signal-dependent allow frequencies only

Answer :

14. In a PCM system, the amplitude levels are transmitted as 7 bit code-words. The sampling is done at 10 kHz. The bandwidth of the system is

1. 10 kHz 2. 20 kHz

3. 35 kHz 4. 70 kHz

Answer : 4

15. What is the frequency and type of modulation used in an IR Remote controls for TV?

1. 108.1 kHz, FM 2. 38 kHz, PCM

3. 98 kHz, PM 4. 98.1 MHz, AM

Answer : 2

16. The amplitude of a random signal is uniformly distributed between –5V and 5V. If the signal to quantization noise ratio required in uniformly quantizing the signal is 43.5dB, the step size of the quantization is approximately –

1. 0.0333 V 2. 0.05 V

3. 0.0667 V 4. 0.10 V

Answer : 3

17. Which one of the following communication system is digital?

1. AM 2. FM

3. PAM 4. PCM

Answer : 4

18. Which is the sequence for PCM?

1. Sampling, Encoding, Quantizing

2. Quantizing, Sampling, Encoding

3. Quantizing, Encoding, Sampling

4. Sampling, Quantizing, Encoding

Answer : 4

19. A signal having a bandwidth of 5 MHz is transmitted using the Pulse code modulation (PCM) scheme as follows. The signal is sampled at a rate of 50% above the Nyquist rate and quantized into 256 levels. The binary pulse rate of the PCM signal in Mbits per second is____

Answer: 120

20. Directions: The item consists of two statements, one labelled as the Assertion (A) and the other as Reason (R)'. You are to examine these two statements carefully and select the answers to the item using the codes given below:

Assertion (A): When coding signals like speech signals a-law or p-law quantizers are used.

Reason (R): A-law and p-law quantizers occupy smaller bandwidth than uniform quantizers.

1. Both A and R individually true and R is the correct explanation of A

2. Both A and Rare individually true but R is not the correct explanation of A

3. A is true but R is false

4. A is false but R is true

Answer : 3

UNIT 03

GEOMETRIC REPRESENTATION OF SIGNALS

QUESTIONS AND ANSWERS

Q1. Illustrate the Geometric representation of Signals.

Ans. Derive Geometrical representation of signal.

combinations of two orthornormal basis functions

$\phi_1(t)$ and $\phi_2(t)$.

$\phi_1(t)$ and $\phi_2(t)$. are orthonormal if:

$$\int_0^{T_b} \phi_1(t)\phi_2(t)dt = 0 \text{ (orhogonality)},$$

$$\int_0^{T_b} \phi_1^2(t)dt = \int_0^{T_b} \phi_2^2(t)dt = 1$$

(normalized to have unit energy)

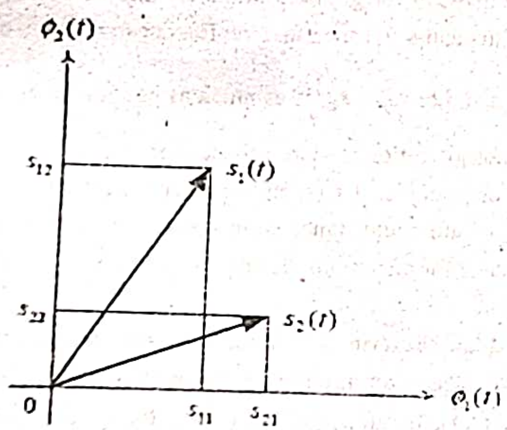
The representations are

$$s_1(t) = s_{11}\phi_1(t) + s_{12}\phi_2(t)$$

$$s_2(t) = s_{21}\phi_1(t) + s_{22}\phi_2(t)$$

where

$$s_{ij} = \int_0^{T_b} s_i(t)\phi_j(t)dt, \; i, j \in \{1,2\}.$$



$\int_0^{T_b} s_i(t)\phi_j(t)dt.$ is the projection of $s_i(t)$ onto $\phi_j(t)$.

Basis Vectors :

The set of basis vectors $\{e_1, e_2, \ldots, e_n\}$ of a space are chosen such that: Should be complete or span the vector space: any vector a can be expressed as a linear combination of these vectors.

Each basis vector should be orthogonal to all others

* Each basis vector should be normalized:
* A set of basis vectors satisfying these properties is also said to be a complete *orthonormal basis*
* In an n-dim space, we can have at most n basis vectors

Signal Space :Basic Idea: If a signal can be represented by n-tuple, then it can be treated in much the same way as a n-dim vector.

Let $f_1(t), f_2(t), \ldots, fn(t)$ be n signals

Consider a signal x(t) and suppose that If every signal can be written as above $\Rightarrow$ ~~ basisfunctions and we have a n-dim signal space

Orthonormal Basis :

Signal set {fk(t)}n is an orthogonal set if

$$\int_{-\infty}^{\infty} \phi_j(t)\,\phi_k(t)dt = \begin{cases} 0 & j \neq k \\ c_j & j = k \end{cases}$$

If $cj = 1\ \forall j \Rightarrow \{\varphi k(t)\}$ is an orthonormal set.

Consider a set of M signals (M-ary symbol) $\{si(t), i = 1, 2, \ldots, M\}$ with finite energy. That is

$$\int_{-\infty}^{\infty} s_i^2(t)dt < \infty$$

Then, we can express each of these waveforms as weighted

UNIT 03

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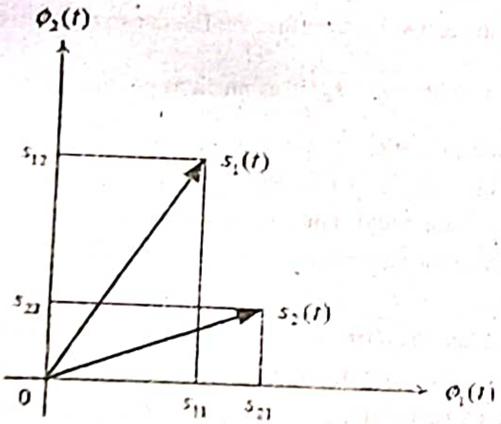

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Consider a set of M signals (M-ary symbol) $\{si(t), i = 1, 2, \ldots, M\}$ with finite energy. That is

$$\int_{-\infty}^{\infty} s_i^2(t)dt < \infty$$

Then, we can express each of these waveforms as weighted linear combination of orthonormal signals

$$s_i(t) = \sum_{j=1}^{N} s_{ij}\phi_j(t) \text{ for } i = 1,\ldots,M$$

$$\{\phi_j(t)\}_1^N$$

where $N \le M$ is the dimension of the signal space and are called the orthonormal basis functions

Let, for a convenient set of $\{\phi_j(t)\}, j = 1,2,\ldots,N$ and $0 \le t \le T$,

$$s_i(t) = \sum_{j=1}^{N} s_{ij}\varphi_j(t), \quad i = 1, 2, \ldots, M \text{ and } 0 \le t < T, \text{ such}$$

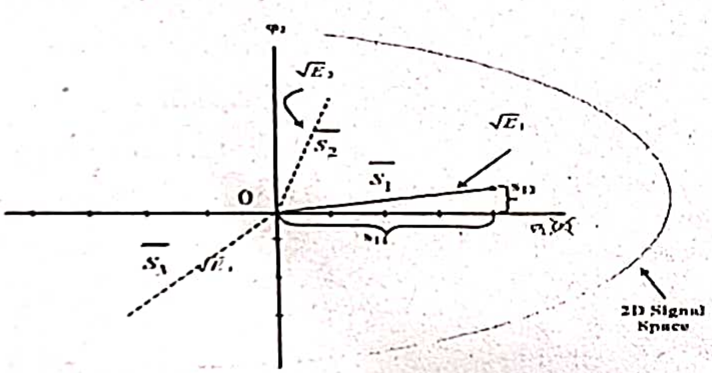
that,

$$s_{ij} = \int_0^T s_i(t)\varphi_j(t)dt$$

Now, we can represent a signal si(t) as a column vector whose elements are the scalar coefficients sij, j = 1, 2, ..., N:

$$\overline{s_i} = \begin{bmatrix} s_{i1} \\ s_{i2} \\ \vdots \\ s_{iN} \end{bmatrix}_{1\times N} ; i = 1, 2, \ldots M$$

These M energy signals or vectors can be viewed as a set of M points in an N – dimensional Euclidean space, known as the „Signal Space'. Signal Constellation is the collection of M signals points (or messages) on the signal space.



Now, the length or norm of a vector is denoted as $\|\overline{s_i}\|$.

The squared norm is the inne product of the vector.

$$\|\overline{s_i}\|^2 = (\overline{s_i}, \overline{s_i}) = \sum_{j=1}^{N} s_{ij}^2$$

The cosine of the angle between two vectors is defined as

$$\cos(\text{angle between } \overline{s_i} \,\&\, \overline{s_j}) = \frac{(\overline{s_i}, \overline{s_j})}{\|\overline{s_i}\|\|\overline{s_j}\|}$$

$\therefore$ $\overline{s_i} \,\&\, \overline{s_j}$ are orthogonal to each other if $(\overline{s_i}, \overline{s_j}) = 0$

If E_i is the energy of the i-th signal vector,

$$E_i = \int_0^T s_i^2(t)\,dt = \int_0^T \left[\sum_{j=1}^{N} s_{ij}\varphi_j(t)\right]\left[\sum_{k=1}^{N} s_{ik}\varphi_k(t)\right]dt$$

$$= \sum_{j=1}^{N}\sum_{k=1}^{N} s_{ij}s_{ik} \int_0^T \varphi_j(t)\,\varphi_k(t)\,dt$$ as $\{\varphi_i(t)\}$ forms an ortho-normal set

$$= \sum_{j=1}^{N} s_{ij}^2 = \|\overline{s_1}\|^2$$

For a pair of signals $s_i(t)$ and $s_k(t)$,

$$\|\overline{s_i} - \overline{s_k}\|^2 = \sum_{j=1}^{N}(s_{ij} - s_{kj})^2 = \int_0^T [s_i(t) - s_k(t)]^2\,dt$$

It may now be guessed intuitively that we should choose $s_i(t)$ and $s_k(t)$ such that the Euclidean distance between them, i.e. $\|\overline{s_i} - \overline{s_k}\|$ is as much as possible to ensure that their detection is more robust even in presence of noise. For example, if $s_1(t)$ and $s_2(t)$ have same energy E, (i.e. they are equidistance from the origin), then an obvious coice for maximum distance of separation is, $s_1(t) = -s_2(t)$.

Q2. Explain the term Maximum Likelihood Decoding.

Ans. Consider a set of possible codewords (valid codewords - set Y) generated by an encoder in the transmitter side. We pick one codeword out of this set (call it y) and transmit it via a Binary Symmetric Channel (BSC) with probability of error p (To know what is a BSC – click here). At the receiver side we receive the distorted version of y (call this erroneous codeword x). Maximum Likelihood Decoding chooses one codeword from Y (the list of all possible codewords) which maximizes the following probability.

P (y sent | x received)]

Meaning that the receiver computes P(y_1,x) , P(y_2,x) , P(y_3,x),\cdots,P(y_n,x). and chooses a codeword (y) which gives the maximum probability. In practice we don't know y (at the receiver) but we know x. So how to compute the probability. Maximum Likelihood Estimation (MLE) comes to our rescue. For a detailed explanation on MLE – refer here The aim of maximum likelihood estimation is to find the parameter value(s) that makes the observed data most likely. Understanding the difference between prediction and estimation is important at this point. Estimation differs from prediction in the following way … In estimation problems, likelihood of the parameters is estimated based on given data/observation vector. In prediction problems, probability is used as a measure to predict the outcome from known parameters of a model.

Q3. Describe Correlation receiver.

Ans. We have ignored the form of the signal term v(t) coming from the receiver, and have simply examined the implications of thresholding the combined signal plus noise. In practice, the shape of the signal coming from the receiver is largely under the control of the system designer, and its properties are chosen to meet such specified criteria as desirable time or range sidelobe levels, resolution, etc. The most fundamental criterion, against which all other criteria must be balanced, is the ability to detect targets, which, as we have seen, is a function of the SNR. Hence a natural priority is to design the receiver to maximize SNR.. We know that the noise power m is dependent only on the gain of the receiver, not on the shape of its impulse response function. Hence, for fixed gain, the best SNR is obtained by maximizing the response to the signal term: This is achieved by an elegant, conceptually simple and readily implemented processing scheme, known as the correlation receiver, whose frequency-domain implementation is the matched filter.

The scheme relies on a fundamental result known as the **Cauchy-Schwartz inequality.** This states that, given two functions f(s) and g(s) of finite energy, then

$$\left|\int_a^b f(s)g^*(s)ds\right|^2 \le \int_a^b |f(x)|^2\, ds \int_a^b |g(s)|^2\, dx$$

with equality if and only if

$f(s) = cg(s)$

for some constant c. Here the asterisk denotes complex conjugate, and we allow complex functions so that we can, if desired, use complex representations of real signals or....

Q4. Explain match filter and draw its block diagram. ?

Ans. Matched Filters :

- The matched filter is the optimal linear filter for maximizing the signal to noise ratio (SNR) in the presence of additive stochastic noise.
- Matched filters are commonly used in radar, in which a signal is sent out, and we measure the reflected signals, looking for something similar to what was sent out.
- Two-dimensional matched filters are commonly used in image processing, e.g., to improve SNR for X-ray pictures
- A general representation for a matched filter is illustrated in Figure.

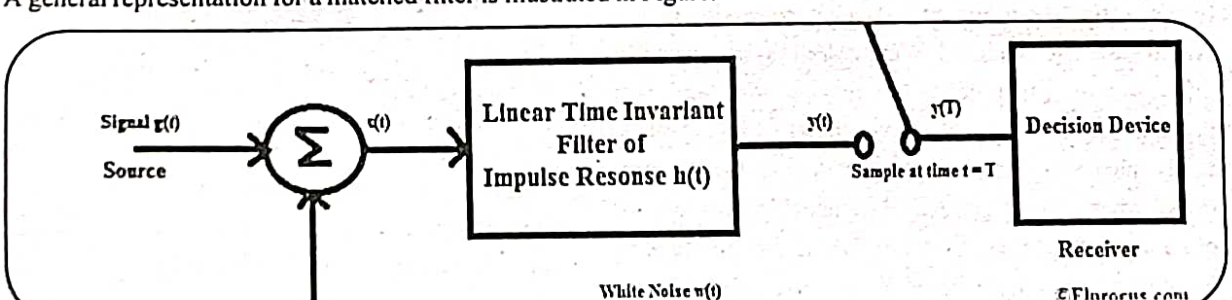


Fig. Matched filter

- The filter input x(t) consists of a pulse signal g(t) corrupted by additive channel noise w(t), as shown by,

$$x(t) = g(t) + w(t), \qquad 0 \le t \le T$$

- where T is an arbitrary observation interval. The pulse signal g(t) may represent a binary symbol 1 or 0 in a digital communication system.
- The w(t) is the sample function of a white noise process of zero mean and power spectral density No/2.
- The source of uncertainty lies in the noise w(t).
- The function of the receiver is to detect the pulse signal g(t) in an optimum manner, given the received signal x(t).
- To satisfy this requirement, we have to optimize the design of the filter so as to minimize the effects of noise at the filter output in some statistical sense, and thereby enhance the detection of the pulse signal g(t).
- Since the filter is linear, the resulting output y(t) may be expressed as

$$x(t) = g_0(t) + n(t),$$

- where go(t) and n(t) are produced by the signal and noise components of the input x(t), respectively.
- A simple way of describing the requirement that the output signal component go(t) be considerably greater than the output noise component n(t) is to have the filter make the instantaneous power in the output signal go(t), measured at time t = T, as large as possible compared with the average power of the output noise n(t). This is equivalent to maximizing the peak pulse signal-to-noise ratio,

defined as $\eta = \dfrac{|g_o(T)|^2}{E\left[n^2(t)\right]}$

Q5. Describe Average bit error rate (ABER) for uncoding OOK format.

Ans. In this section we derive the analytical expression for the ABER associated with an UWOC IM/DD system using an OOK signaling technique. For this purpose, and as a previous step, the associated conditional BER (CBER) is firstly calculated for a given electrical signal-to-noise ratio (SNR) when analyzing an AWGN channel in the ideal case of absence of turbulence (namely SNR0), assuming each transmitted symbol equally likely to be sent. In addition, the conditional bit error probabilities when the transmitted bit is "0" or "1" are assumed to be equal. Hence, and from, the CBER of IM/DD with AWGN channel using OOK is expressed as

$$P_b(e\,|\,I) = \frac{1}{2} erfc\left(\frac{i_{S0} I}{2\sqrt{2}\sigma_N}\right) = \frac{1}{2} erfc\left(\frac{SNR_0 I}{2\sqrt{2}}\right)$$

where SNR0 = i_{S0}/s_N, where $i_{S0} = aRP_t$ denotes the signal current in absence of turbulence-induced fading, with a representing the attenuation coefficient associated to the medium. Therefore, the ABER, Pb(e), can be obtained by averaging P_b(e|I) over the PDF of the irradiance, fI(I). Hence:

$$P_b = \int_0^{\infty} \frac{1}{2} erfc\left(\frac{SNR_0 I}{2\sqrt{2}}\right) f_I(I)dI.$$

In Eq., the PDF of the optical irradiance is defined according to the Weibull model, as indicated in Eq. Now, by using the integration by parts formula for solving Eq., we can obtain that

$$P_b = (P_b(e\,|\,I)F_I(I))\big|_0^{\infty} - \int_0^{\infty} \frac{d}{dI}[P_b(e\,|\,I)]F_I(I)dI.$$

Since P_b(e|8) = 0 and FI(0) = 0 (note that negative values for the optical irradiance are not allowed), then the last expression can be reduced to:

$$P_b = -\int_0^{\infty} \frac{d}{dI}\left[\frac{1}{2} erfc\left(\frac{SNR_0 . I}{2\sqrt{2}}\right)\right] F_I(I)dI$$

where the cumulative distribution function (CDF) for the irradiance, I, is directly obtained by integrating Eq. as:

$$F_I(I) = 1 - \exp\left[-\left(\frac{I}{\lambda}\right)^K\right]$$

Thus, we can employ [28, Ec. (06.27.13.0005.01)] to derive an expression for the derivative of Pb(e|I) with respect to I:

$$\frac{d}{dI}[P_b(e\,|\,I)] = -\frac{SNR_0}{2\sqrt{2\pi}} \exp\left[-\left(\frac{SNR_0 I}{2\sqrt{2}}\right)^2\right]$$

Next we introduce the last result in Eq. in order to solve the resulting integral. To this aim, a generalized Gauss-Laguerre quadrature is proposed, defined by:

$$\int_0^{\infty} x^{\beta} e^{-x} f(x)dx = \sum_{i=1}^{n} H_i f(x_i) + E_n$$

where β is a constant, x_i is the i–th zero of the Laguerre polynomial, $L_n^{\beta}(x)$, H_i is the corresponding weight coefficient and E_n is the truncation error. If the normalization of the Laguerre polynomials is chosen so that

$$L_n^{\beta} = \sum_{m=0}^{n} \binom{n+\beta}{n-m} \frac{(-x)^m}{m!}$$

then, according to, the weight coefficients are given by

$$H_i = \frac{\Gamma(n+\beta+1)x_i}{n!(n+1)^2\left[L_{n+1}^{\beta}(x_i)\right]^2}, (i = 1,2,...,n)$$

Hence, Eq.can be rewritten as

$$P_b = \frac{1}{2\sqrt{\pi}} \int_0^{\infty} x^{-\frac{1}{2}} \exp(-x)\left(1 - \exp\left[-\left(\frac{2\sqrt{2}}{\lambda SNR_0}\sqrt{x}\right)^K\right]\right)dx$$

after having used the following change of variables:

$$x = \left(\frac{SNR_0}{2\sqrt{2}}\right)^2 I^2; \quad dx = 2\left(\frac{SNR_0}{2\sqrt{2}}\right)^2 IdI.$$

Finally, we can apply Eq. to solve Eq. as:

$$P_b = \frac{1}{2\sqrt{\pi}} \sum_{i=1}^{n} H_i \left\{1 - \exp\left[-\left(\frac{2\sqrt{2}}{\lambda SNR_0}\sqrt{x_i}\right)^K\right]\right\}$$

where β = –1/2 directly obtained by comparing Eq. with Eq. so the weight coefficients, Hi, given in Eq. can be directly calculated.

As a remarking comment, we can repeat all these steps to derive a closed-from expression for the exact ABER of the UWOC system using an exponentiated Weibull distribution instead of the Weibull one shown in Eq. The analytical procedure is shown in the Appendix. For the sake of simplicity, this paper is focused on the Weibull distribution since, as we commented above, large receiving apertures are commonly employed in UWOC, which leads to an effectively reduced turbulence accurately modeled by the simplest Weibull distribution.

Q6. Illustrate Error probabilities of noncoherent and coherent FSK.

Ans. We analyze the error performance of two-hop relay networks adopting frequency shift keying (FSK) over frequency flat Rayleigh fading channels. It is assumed that relay networks consist of a source, a relay, and a destination without a direct path signal from the source to the destination and the relay adopts the amplify-and-forward protocol with a fixed gain. Firstly, considering imperfect frequency and phase synchronization, we obtain the exact error probability expressions for noncoherent and coherent binary FSK (BFSK). Secondly, assuming perfect frequency and phase synchronization, we derive a closed-form error probability approximation for coherent M-ary FSK (MFSK). The proposed methods can also be used for the error performance analysis of classical one-hop FSK systems with perfect/imperfect frequency and phase synchronization. The obtained error probability expressions will help the design of two-hop relay networks adopting FSK in determining the system parameters such as the transmission power at the source, the amplifying coefficient at the relay, and the maximum affordable frequency and phase offsets to satisfy the required error performance.

Q7. Write short notes on:

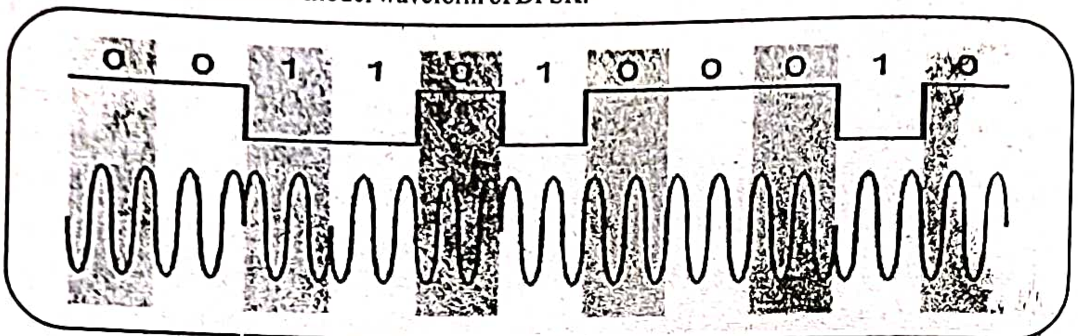
1. Understanding Quadrature Phase Shift (QPSK)

2. Differential Phase Shift Keying (DPSK)

Ans. 1. QPSK : In the world of wired electronics, analog signals exhibit continuous variations whereas digital signals assume (ideally) one of two discrete states. This distinction can be extended to systems that transmit data via electromagnetic radiation instead of electric current traveling through wires.When used for analog signals, frequency modulation and amplitude modulation lead to continuous variations in the frequency or amplitude of a carrier wave. When modulation techniques are used for digital communication, the variations applied to the carrier are restricted according to the discrete information being transmitted.Examples of common digital modulation types are OOK (on/off keying), ASK (amplitude shift keying), and FSK (frequency shift keying). These schemes cause the carrier to assume one of two possible states depending on whether the system must transmit a binary 1 or a binary 0; each discrete carrier state is referred to as a symbol.Quadrature phase shift keying (QPSK) is another modulation technique, and it's a particularly interesting one because it actually transmits two bits per symbol. In other words, a QPSK symbol doesn't represent 0 or 1—it represents 00, 01, 10, or 11.

2. **DPSK :** In Differential Phase Shift Keying DPSK the phase of the modulated signal is shifted relative to the previous signal element. No reference signal is considered here. The signal phase follows the high or low state of the previous element. This DPSK technique doesn't need a reference oscillator.

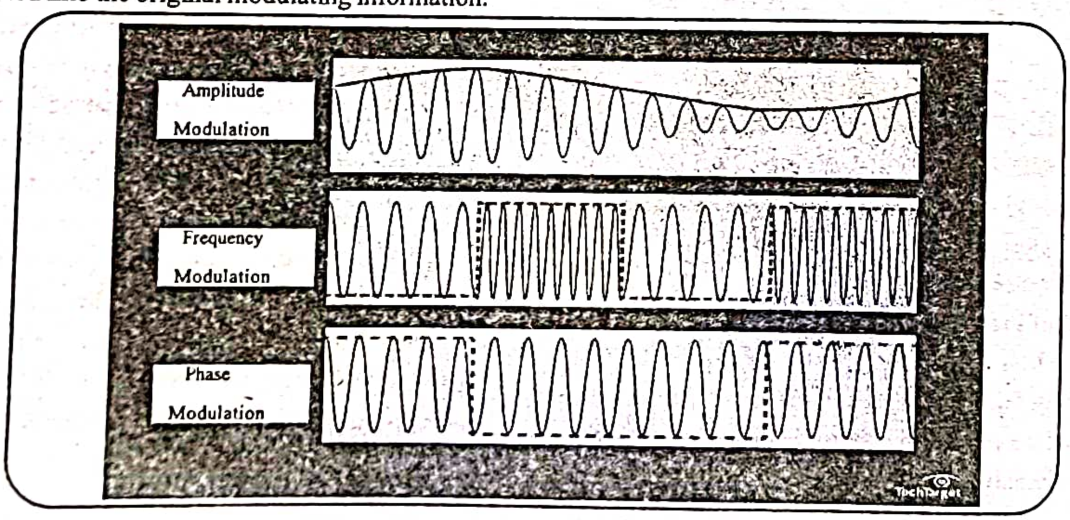
The following figure represents the model waveform of DPSK.



It is seen from the above figure that, if the data bit is Low i.e., 0, then the phase of the signal is not reversed, but continued as it was. If the data is a High i.e., 1, then the phase of the signal is reversed, as with NRZI, invert on. If we observe the above waveform, we can say that the High state represents an M in the modulating signal and the Low state represents a W in the modulating signal.

Q8. Define QAM (quadrature amplitude modulation)

Ans. QAM (quadrature amplitude modulation) is a method of combining two amplitude modulation (AM) signals into a single channel. This approach helps double its effective bandwidth. QAM is also used with pulse AM (PAM) in digital systems, like wireless applications. A QAM modulator works like a translator, helping to translate digital packets into an analog signal to transfer data seamlessly. QAM is used to achieve high levels of spectrum usage efficiency. This is accomplished by utilizing both the amplitude and phase components to provide a form of modulation. In this scenario, the QAM signal comes with two carriers. Each has the same frequency but differs in phases by 90 degrees, or one-quarter of a cycle, which is the basis for the term quadrature. One signal is called the I signal, and the other is called the Q signal. Mathematically, one of the signals can be represented with a sine wave and the other by a cosine wave. The two modulated carriers combine at the source for transmission. At the destination, the carriers separate, and the data is extracted from each. Then, the data is incorporated into the original modulating information.



Q9. Define Minimum Shift Key (MSK) Modulation and its Key features.

Ans. Minimum Shift Key Modulation is another type of digital modulation technique used to convert a digital signal into analog signals. It is also called Minimum-shift keying (MSK) or Advance Frequency Shift Keying because it is a type of continuous-phase frequency-shift keying.

Key features of Minimum Shift Key (MSK) Modulation :

- Minimum-shift keying or MSK was first developed by the Collins Radio employees Melvin L. Doelz and Earl T. Heald in the late 1950s.
- It is encoded with bits alternating between quadrature components, with the Q component delayed by half the symbol period.
- Minimum Shift Keying is the most effective digital modulation technique. It can be implemented for almost every stream of bits much easier than the Phase Shift Key, Frequency Shift Key and Amplitude Shift Key of digital modulation technique.
- The Minimum Shift Keying's concept is based on the positioning of bits such as even bits and odd bits for the given bitstream and the bit positioning frequency generating table.
- MSK is the most widely used digital modulation technology because of its ability and flexibility to handle "One(1)" and "Zero(0)" transition of binary bits.

Q10. Explain Multicarrier Modulation and its development.

Ans. Multicarrier modulation is a form of signal waveform that uses multiple normally close spaced carriers in a block to carry the information. Multicarrier modulation, MCM is a technique for transmitting data by sending the data over multiple carriers which are normally close spaced. Multicarrier modulation has several advantages including resilience to interference, resilience to narrow band fading and multipath effects. As a result, multicarrier modulation techniques are widely used for data transmission as it is able to provide an effective signal waveform which is spectrally efficient and resilient to the real world environment.

Multicarrier modulation basics : Multicarrier modulation operates by dividing the data stream to be transmitted into a number of lower data rate data streams. Each of the lower data rate streams is then used to modulate an individual carrier. When the overall transmission is received, the receiver has to then re-assembles the overall data stream from those received on the individual carriers. It is possible to use a variety of different techniques for multicarrier transmissions. Each form of MCM has its own advantages and can be sued in different applications.

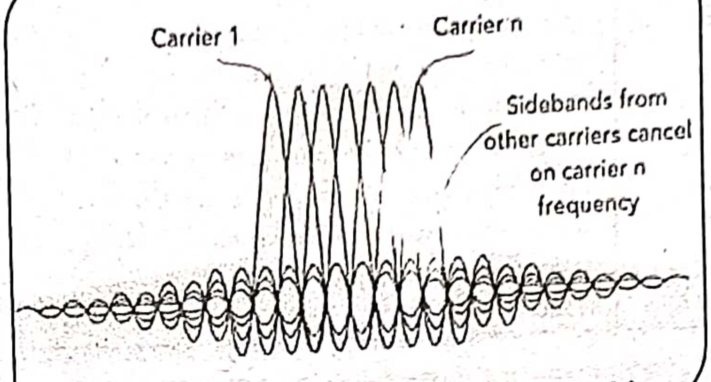


Fig. One form of multicarrier modulation is OFDM

Development of MCM : The history of multicarrier modulation can said to have been started by military users. The first MCM were military HF radio links in the late 1950s and early 1960s. Here several channels were sued to overcome the effects of fading. Originally the concept of MCM required the use of several channels that were separated from each other by the use of steep sided filters of they were close spaced. In this way, interference from the different channels could be eliminated.

Q11. Write the Difference between Bandwidth and Data Rate.

Ans. 1. ***Bandwidth :*** Bandwidth is defined as the potential of the data that is to be transferred in a specific period of time. It is the data carrying capacity of the network or transmission medium. In simple words, it is the maximum amount of data that can be transferred per second on a link. It is generally measured in bits per second(bps), Mega bits per second(Mbps) or Giga bits per second(Gbps). For example, if bandwidth is 100 Mbps, it means maximum 100 Mb data can be transferred per second on that channel.

2. ***Data Rate :*** Data Rate is defined as the amount of data transmitted during a specified time period over a network. It is the speed at which data is transferred from one device to another or between a peripheral device and the computer. It is generally measured in Mega bits per second(Mbps) or Mega bytes per second(MBps). For example, if bandwidth is 100 Mbps but data rate is 50 Mbps, it means maximum 100 Mb data can be transferred but channel is transmitting only 50 Mb data per second.

Difference between Bandwidth and Data Rate:

Sl no.	Bandwidth	Data Rate
1.	It is the potential of the data that is to be transferred in a specific period of time.	It is the amount of data transmitted during a specified time period over a network.
2.	It is the number of bits per second that a link can send or receive.	It is the speed of data transmission.
3.	Normally it is measured in bps, Mbps or Gbps.	It is normally measured in Mbps or MBps.
4.	It refers to maximum data transmission speed.	It refers to the actual data transmission speed.
5.	It is physical layer property in OSI model.	While it is common in all layers.
6.	It shows the capacity of the channel.	It shows the present speed of data transmission.
7.	It does not depend on properties of sender or receiver.	While it gets affected by sender or receiver.

OBJECTIVE QUESTIONS AND ANSWERS

1. For generation of FSK the data pattern will be
   1. RZ pattern 2. NRZ pattern
   3. Split-phase Manchester
   4. None

Answer – (2)

2. The bit rate of digital communication system is 34 M bits/sec. The Baud rate will be in QPSK modulation techniques
   1. 8.5 M bits/sec 2. 17 M bits/sec
   3. 32 M bits/sec 4. 64 M bits/sec

Answer – (2)

3. In Coherent demodulation technique of FSK signal can be affected using
   1. Correlation receiver
   2. Bandpass filters and envelope detector
   3. Matched filter
   4. Discriminator detection

Answer – (1)

4. The bit rate of a digital communication system using QPSK modulation techniques in 30 MBPS. So, The system
   1. 60 Mbps
   2. The baud rate equal to 15 Mbps
   3. The baud rate equal to 30 Mbps
   4. The baud rate equal to 7.5 Mbps

Answer – (2)

5. If the maximum instantaneous phase transition of a digital modulation techniques kept at 90°, the modulation will be organized as
   1. DPSK 2. QPSK
   3. OQPSK 4. BPSK

Answer – (2)

6. The modulation techniques employed in for telephone modems is ?
   1. QAM 2. GMSK
   3. QPSK 4. GFSK

Answer – (1)

7. BPSK signal can be demodulated by using,
   1. low pass filters 2. A band pass filter
   3. A high pass filter 4. None of these

Answer – (1)

8. In a system using in FSK, the '0' and '1' bit are represented by sine waves of 10 and 25 KHz correspondingly. These waveforms will be Orthogonal for bit interval of
   1. 45 μs 2. 200 μs
   3. 50 μs 4. 250 μs

Answer – (2)

9. If the baud rate is 400 for a QPSK signal, the rate is
   1. 200 2. 400
   3. 800 4. 1600

Answer – (3)

10. For a BPSK system, the bit error probability is given by,
    1. *erfc* () 2. *erfc* ()
    3. *erfc* () 4. *erfc* ()

Answer – (3)

11. The width of the power spectral density main love given the bandwidths of MSK signal and is given by ...... times the baseband frequency (f_b)
    1. 0.5 2. 0.75
    3. 0.25 4. 2.0

Answer – (2)

12. Which of the following gives the least probability of error?

1. In Amplitude Shift Keying
2. In Frequency Shift Keying
3. In Phase Shift Keying
4. In Differential Phase Shift Keying

Answer – (3)

13. Which of the following digital modulation techniques are employed in telephone modem?
 1. QAM 2. GMSK
 3. QPSK 4. none of these

Answer – (4)

14. Which gives maximum probability of error?
 1. ASK 2. BFSK
 3. BPSK 4. DBPSK

Answer – (1)

15. Whose bandwidth is maximum?
 1. PSK 2. ASK
 3. FSK 4. DPSK

Answer – (3)

16. Bandwidth of MSK __________ that of QPSK.
 1. higher than 2. lower than
 3. equal to 4. Both (a) and (b)

Answer – (1)

17. Equalizer is used to
 1. Increase the signal to noise ratio at the receiver
 2. Equalize the distortion introduced by channel
 3. Decrease the error probability of signal detection
 4. None of these

Answer – (2)

18. Eye-pattern is utilized for the study of
 1. Bit-error rate
 2. Error-vector magnitude
 3. The Quantization noises
 4. Inter-symbol interferences

Answer – (d)

19. The Nyquist interval for m(t) = is
 1. 0.001s 2. 0.005s
 3. 0.0025s 4. 250 μs

Answer – (c)

20. In Eye Pattern, as eye closes:
 1. ISI increase 2. ISI decrease
 3. Timing jitter increases
 4. Timing jitter decreases

Answer – (1)

21. Transversal equalizer uses tapped delay line to
 1. Reduce and SI 2. Reduce BER
 3. Increase bit rate 4. Increase bandwidths

Answer – (1)

22. AMI is another name of which process?
 1. Polar 2. Bipolar
 3. On-off 4. None of these

Answer – (2)

23. To encoding in binary, the Differential encoding utilized for
 1. The Signal transitions
 2. Signal freq.
 3. Signal's amplitude
 4. Signal's phase

Answer – (3)

24. Alternate Mark Inversion (AMI) signaling is acknowledged as
 a. The Bipolar signaling
 b. The Polar signaling
 c. The Manchester signaling
 d. The Unipolar signaling

Answer – (b)

25. Eye pattern is used to study
 1. ISI 2. Quantization noise
 3. Error rate 4. None of these

Answer – (1)

26. A scheme in that '1' is representing by a +ve. pulse for a half of symbol duration, a -ve. pulse for remaining half of the symbol and for '0' the order is inverted is identified as
 1. The NRZ unipolar 2. The NRZ polar
 3. The NRZ bipolar 4. The Manchester code

Answer – (4)

27. A line code which has zero dc element for pulse transmission of random Binary data is
 1. Unipolar-NRZ 2. Unipolar-RZ
 3. BPRZ-AMI 4. BPNRZ

Answer – (3)

28. On-off signaling is known as
 1. Bipolar signaling 2. Polar signaling
 3. Manchester signaling
 4. Unipolar signaling

Answer – (4)

29. Which is the most commonly used line coding format with best overall desirable properties?
 1. P-NRZ 2. P-RZ
 3. BP-AMI-RZ 4. UP-RZ

Answer – (3)

UNIT 04

INTRODUCTION TO INFORMATION AND CODING THEORIES

QUESTIONS AND ANSWERS

Q1. Describe Information and Coding Theories.

Ans. Information theory: Information theory is the scientific study of the quantification, storage, and communication of digital information.The field was fundamentally established by the works of Harry Nyquist and Ralph Hartley, in the 1920s, and Claude Shannon in the 1940s.The field is at the intersection of probability theory, statistics, computer science, statistical mechanics, information engineering, and electrical engineering. A key measure in information theory is entropy. Entropy quantifies the amount of uncertainty involved in the value of a random variable or the outcome of a random process. For example, identifying the outcome of a fair coin flip (with two equally likely outcomes) provides less information (lower entropy) than specifying the outcome from a roll of a die (with six equally likely outcomes). Some other important measures in information theory are mutual information, channel capacity, error exponents, and relative entropy. Important sub-fields of information theory include source coding, algorithmic complexity theory, algorithmic information theory and information-theoretic security. Applications of fundamental topics of information theory include source coding/data compression (e.g. for ZIP files), and channel coding/error detection and correction (e.g. for DSL). Its impact has been crucial to the success of the Voyager missions to deep space, the invention of the compact disc, the feasibility of mobile phones and the development of the Internet. The theory has also found applications in other areas, including statistical inference, cryptography, neurobiology, perception, linguistics, the evolution and function of molecular codes (bioinformatics), thermal physics, molecular dynamics, quantum computing, black holes, information retrieval, intelligence gathering, plagiarism detection pattern recognition, anomaly detection and even art creation.

Information theory studies the transmission, processing, extraction, and utilization of information. Abstractly, information can be thought of as the resolution of uncertainty. In the case of communication of information over a noisy channel, this abstract concept was formalized in 1948 by Claude Shannon in a paper entitled A Mathematical Theory of Communication, in which information is thought of as a set of possible messages, and the goal is to send these messages over a noisy channel, and to have the receiver reconstruct the message with low probability of error, in spite of the channel noise. Shannon's main result, the noisy-channel coding theorem showed that, in the limit of many channel uses, the rate of information that is asymptotically achievable is equal to the channel capacity, a quantity dependent merely on the statistics of the channel over which the messages are sent

Coding Theories : Coding theory is the study of the properties of codes and their respective fitness for specific applications. Codes are used for data compression, cryptography, error detection and correction, data transmission and data storage. Codes are studied by various scientific disciplines—such as information theory, electrical engineering, mathematics, linguistics, and computer science—for the purpose of designing efficient and reliable data transmission methods. This typically involves the removal of redundancy and the correction or detection of errors in the transmitted data.

There are four types of coding :

1. Data compression (or source coding).
2. Error control (or channel coding).
3. Cryptographic coding.
4. Line coding.

Data compression attempts to remove unwanted redundancy from the data from a source in order to transmit it more efficiently. For example, ZIP data compression makes data files smaller, for purposes such as to reduce Internet traffic. Data compression and error correction may be studied in combination. Error correction adds useful redundancy to the data from a source to make the transmission more robust to disturbances present on the transmission channel. The ordinary user may not be aware of many applications using error correction. A typical music compact disc (CD) uses the Reed–Solomon code to correct for scratches and dust. In this application the transmission channel is the CD itself. Cell phones also use coding techniques to correct for the fading and noise of high frequency radio transmission. Data modems, telephone transmissions, and the NASA Deep Space Network all employ channel coding techniques to get the bits through, for example the turbo code and LDPC codes.

Q2. Define Information Measures.

Ans. There are many more measures of information than are typically presented in an information theory book. Below is a list of many of them. For much more information and implementations of many of them, please see the dit documentation. In the following, alternative names for measures are given in square brackets.

There are a variety of measures directly based on Shannon's original measures, begin sums and differences of entropies:

- Entropy
- Mutual Information
- Multivariate Mutual Information [Co-Information]
- Total Correlation [Multi-Information, Integration]
- Binding Information [Dual Total Correlation]
- Residual Entropy [Erasure Entropy]

There are other measures that are not directly representable on I-diagrams, but are entropies or mutual informations of auxiliary variables:

- Gacs-Korner Common Information [Zero-Error Information]
- Wyner Common Information
- Minimal Markov Chain Information
- Minimal Functional Markov Chain Information
- Joined Minimal Sufficient Statistic
- Intrinsic Information [Intrinsically Conditional Mutual Information]
- Reduced Intrinsic Information

There are measures which are not exactly entropies and the like, but are related:

- Interaction Information
- TSE Complexity

There are a variety of measures of divergence or distance between distributions:

- Jensen-Shannon Divergence
- Relative Entropy [Kullback–Leibler Divergence]
- Cross Entropy
- Jensen-Renyi Divergence
- Jensen-Tsallis Divergence

Lastly, there are a number of alternative information measures:

- Cumulative Residual Entropy
- Extropy
- Perplexity
- Renyi Entropy
- Tsallis Entropy

Q3. Describe Shannon Entropy.

Ans. The Shannon entropy equation provides a way to estimate the average minimum number of bits needed to encode a string of symbols, based on the frequency of the symbols.

Shannon's entropy equation:

$$H(X) = -\sum_{i=0}^{N-1} p_i \log_2 p_i$$

In the Shannon entropy equation, pi is the probability of a given symbol.

To calculate $\log_2$ from another log base (e.g., log10 or loge):

$$\log_2(n) = \frac{\log_b(n)}{\log_b(2)}$$

The minimum average number of bits is per symbol is

num Bits = $\lceil H(X) \rceil$

If we have a symbol set {A,B,C,D,E} where the symbol occurance frequencies are:

A = 0.5

B = 0.2

C = 0.1

D = 0.1

E = 0.1

The average minimum number of bits needed to represent a symbol is

H(X) = –[(0.5log20.5 + 0.2log20.2 + (0.1log20.1)*3)]

H(X) = –[–0.5 + (–0.46438) + (–0.9965)]

H(X) = – [– 1.9]

H(X) = 1.9

Rounding up, we get 2 bits/per symbol. To represent a ten character string AAAAABBCDE would require 20 bits if the string were encoded optimally. Such an optimal encoding would allocate fewer bits for the frequency occuring symbols (e.g., A and B) and long bit sequences for the more infrequent symbols (C,D,E).

This example is borrowed from A Guide to Data Compression Methods by Solomon. Note that the frequence of the symbols also happens to match the frequence in the string. This will not usually be the case and it seems to me that there are two ways to apply the Shannon entropy equation :

1. The symbol set has a known frequency, which does not necessarily correspond to the frequency in the message string. For example, characters in a natural language, like english, have a particular average frequency. The number of bits per character can be calculated from this frequency set using the Shannon entropy equation. A constant number of bits per character is used for any string in the natural language.
2. Symbol frequency can be calculated for a particular message. The Shannon entropy equation can be used calculate the number of bits per symbol for that particular message.

Shannon entropy provides a lower bound for the compression that can be achieved by the data representation (coding) compression step. Shannon entropy makes no statement about the compression efficiency that can be achieved by predictive compression. Algorithmic complexity (Kolmogorov complexity) theory deals with this area. Given an infinite data set (something that only mathematicians possess), the data set can be examined for randomness. If the data set is not random, then there is some program that will generate or approximate it and the data set can, in theory, be compressed.

Note that without an infinite data set, this determination is not always possible. A finite set of digits generated for a pi expansion satisify tests for randomness. However, these digits must be pseudo-random, since they are generated from a deterministic process. Algorithmic complexity theory views a pi expansion of any number of digits as compressible to the function that generated the sequence (a relatively small number of bits).

Q4. Illustrate Differential entropy.

Ans. Differential entropy (also referred to as continuous entropy) is a concept in information theory that began as an attempt by Claude Shannon to extend the idea of (Shannon) entropy, a measure of average surprisal of a random variable, to continuous probability distributions. Unfortunately, Shannon did not derive this formula, and rather just assumed it was the correct continuous analogue of discrete entropy, but it is not.The actual continuous version of discrete entropy is the limiting density of discrete points (LDDP). Differential entropy (described here) is commonly encountered in the literature, but it is a limiting case of the LDDP, and one that loses its fundamental association with discrete entropy.

Let X be a random variable with a probability density function f whose support is a set X. The differential entropy h(X) or h(f) is defined as.

$$H(X)=E[-\log(f(X))]=-\int_{x} f(x)\log f(x)dx$$

Q5. Explain mutual information also write its properties and uses.

Ans. *Mutual Information :* The Mutual Information between two random variables measures non-linear relations between them. Besides, it indicates how much information can be obtained from a random variable by observing another random variable. It is closely linked to the concept of entropy. This is because it can also be known as the reduction of uncertainty of a random variable if another is known. Therefore, a high mutual information value indicates a large reduction of uncertainty whereas a low value indicates a small reduction. If the mutual information is zero, that means that the two random variables are independent.

Properties of Mutual Information : The main properties of the Mutual Information are the following:

* Non-negative: I(X;Y) = 0
* Symmetric: I(X;Y) = I(Y;X)
* I(X;Y) = 0 ↔ X,Y independent, because in that case P(x,y) = P(x)·P(y)

Since mutual information has only lower boundaries, sometimes it is difficult to interpret the obtained result. Looking at the equation that relates mutual information with entropy and the Venn diagram, we can see that it is possible to obtain the maximum value of the mutual information.

$$I(X;Y) = H(X) + H(Y) - (X,Y)$$

$$Imax(X;Y) = \min(H(X), H(Y))$$

$$0 \leq I(X;Y) \leq \min(H(X), H(Y))$$

$$I_{NORM}(X;Y) = \frac{I(X;Y)}{Imax(X;Y)} = \frac{I(X;Y)}{\min(H(X),H(Y))}$$

Uses : In the field of machine learning, one of its main uses is in decision trees. It is used for looking for the optimum split of the features to choose the nodes that compose the tree (also called information gain).

Another use is for feature selection. When having a big dataset with a big range of features, mutual information can help to select a subset of those features in order to discard the irrelevant ones.

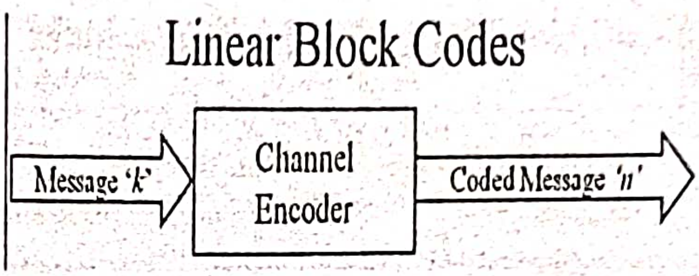
In other fields, mutual information is also widely used. For example, in telecommunications, it is used to calculate the channel capacity.

Q6. Explain Linear block coding.

Ans. The purpose of error control coding is to enable the receiver to detect or even correct the errors by introducing some redundancies in to the data to be transmitted. There are basically two mechanisms for adding redundancy:

1. Block coding
2. Convolutional coding

Block diagram Linear block codes :



The encoder generates a block of n coded bits from k information bits and we call this as (n, k) block codes. The coded bits are also called as code word symbols.

Q7. Describe Hard and Soft decision decoding.

Ans. 1. Hard decision decoding: Assume that our communication model consists of a parity encoder, communication channel (attenuates the data randomly) and a hard decision decoder The message bits "01" are applied to the parity encoder and we get "011" as the output codeword.

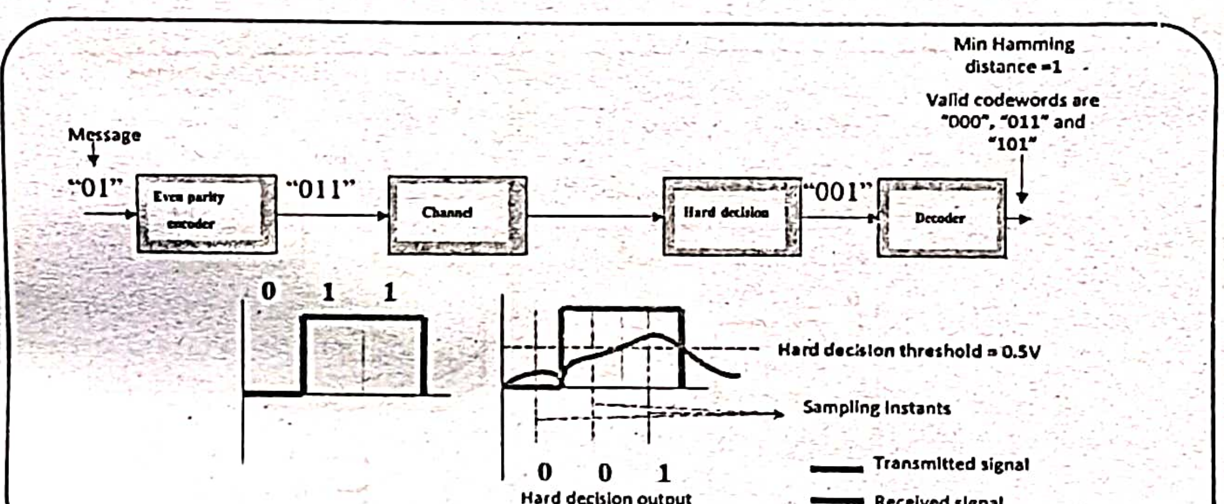


Figure 1: Hard decision decoding – a simple illustration

The output codeword "011" is transmitted through the channel. "0" is transmitted as "0 Volt and "1" as "1 Volt". The channel attenuates the signal that is being transmitted and the receiver sees a distorted waveform ("Red color waveform"). The hard decision decoder makes a decision based on the threshold voltage. In our case the threshold voltage is chosen as 0.5 Volt (midway between "0" and "1" Volt) . At each sampling instant in the receiver (as shown in the figure above) the hard decision detector determines the state of the bit to be "0" if the voltage level falls below the threshold and "1" if the voltage level is above the threshold. Therefore, the output of the hard decision block is "001". Perhaps this "001" output is not a valid codeword (compare this with the all possible codewords given in the table above) , which implies that the message bits cannot be recovered properly. The decoder compares the output of the hard decision block with the all possible codewords and computes the minimum Hamming distance for each case (as illustrated in the table below).

000	001	1
011	001	1
101	001	1
110	001	3

The decoder's job is to choose a valid codeword which has the minimum Hamming distance. In our case, the minimum Hamming distance is "1" and there are 3 valid codewords with this distance. The decoder may choose any of the three possibility and the probability of getting the correct codeword ("001" – this is what we transmitted) is always 1/3. So when the hard decision decoding is employed the probability of recovering our data (in this particular case) is 1/3. Lets see what "Soft decision decoding" offers ...

2. ***Soft Decision Decoding :*** The difference between hard and soft decision decoder is as follows :

In Hard decision decoding, the received codeword is compared with the all possible codewords and the codeword which gives the minimum Hamming distance is selected

In Soft decision decoding, the received codeword is compared with the all possible codewords and the codeword which gives the minimum Euclidean distance is selected. Thus the soft decision decoding improves the decision making process by supplying additional reliability information (calculated Euclidean distance or calculated log-likelihood ratio)

For the same encoder and channel combination lets see the effect of replacing the hard decision block with a soft decision block.

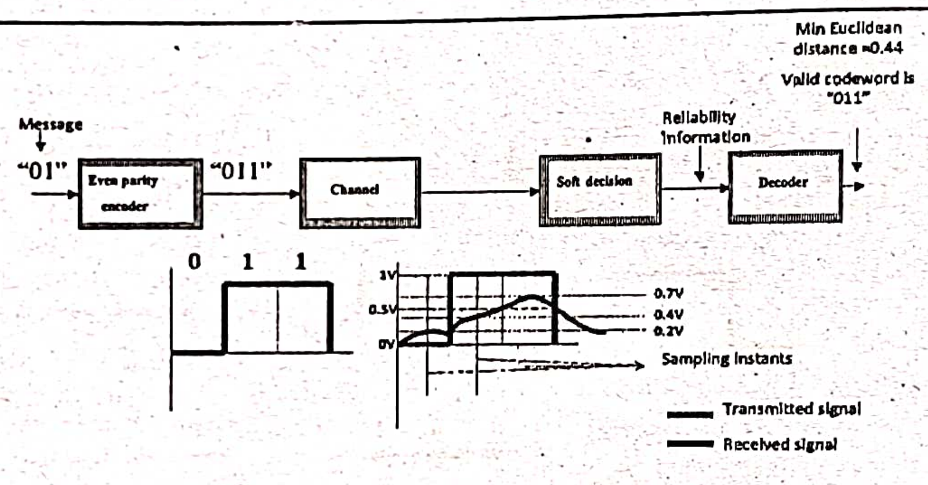


Figure 2: Soft-decision decoding – a simple illustration

Voltage levels of the received signal at each sampling instant are shown in the figure. The soft decision block calculates the Euclidean distance between the received signal and the all possible codewords.

Valid codewords	Voltage levels at each sampling instant of received waveform	Euclidean distance calculation	Euclidean distance
000 (0V 0V 0V)	0.2V 0.4V 0.7V	(0-0.2)2+(0-0.4)2+(0-0.7)2	0.69
011 (0V 1V 1V)	0.2V 0.4V 0.7V	(0-0.2)2+(1-0.4)2+(1-0.7)2	0.49
101 (1V 0V 1V)	0.2V 0.4V 0.7V	(1-0.2)2+(0-0.4)2+(1-0.7)2	0.89
110 (1V 1V 0V)	0.2V 0.4V 0.7V	(1-0.2)2+(1-0.4)2+(0-0.7)2	1.49

The minimum Euclidean distance is "0.49" corresponding to "0 1 1" codeword (which is what we transmitted). The decoder selects this codeword as the output. Even though the parity encoder cannot correct errors, the soft decision scheme helped in recovering the data in this case. This fact delineates the improvement that will be seen when this soft decision scheme is used in combination with forward error correcting (FEC) schemes like convolution codes, LDPC etc From this illustration we can understand that the soft decision decoders uses all of the information (voltage levels in this case) in the process of decision making whereas the hard decision decoders does not fully utilize the information available in the received signal (evident from calculating Hamming distance just by comparing the signal level with the threshold whereby neglecting the actual voltage levels).

Soft decision decoding scheme is often realized using Viterbi decoders. Such decoders utilize Soft Output Viterbi Algorithm (SOVA) which takes into account the apriori probabilities of the input symbols producing a soft output indicating the reliability of the decision.

Q8. Write short notes on :

1. Hamming code.

2. Reed - Solomon (RS) code.

3. Concatenated code.

Ans. 1. Hamming Code : Hamming code is a block code that is capable of detecting up to two simultaneous bit errors and correcting single-bit errors. It was developed by R.W. Hamming for error correction. In this coding method, the source encodes the message by inserting redundant bits within the message. These redundant bits are extra bits that are generated and inserted at specific positions in the message itself to enable error detection and correction. When the destination receives this message, it performs recalculations to detect errors and find the bit position that has error.

Encoding a message by Hamming Code : *The procedure used by the sender to encode the message encompasses the following steps -*

Step 1 - Calculation of the number of redundant bits.

Step 2 - Positioning the redundant bits.

Step 3 - Calculating the values of each redundant bit.

Decoding a message in Hamming Code : *Once the receiver gets an incoming message, it performs recalculations to detect errors and correct them. The steps for recalculation are -*

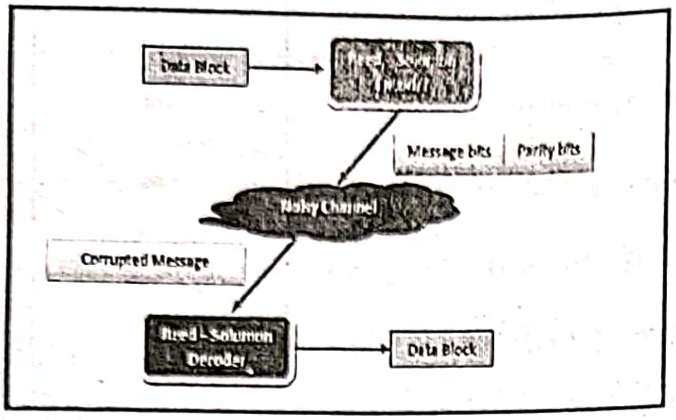
Step 1 - Calculation of the number of redundant bits.

Step 2 - Positioning the redundant bits.

Step 3 - Parity checking.

Step 4 - Error detection and correction.

2. Reed - Solomon (RS) Code : Reed - Solomon error correcting codes are one of the oldest codes that were introduced in 1960s by Irving S. Reed and Gustave Solomon. It is a subclass of non - binary BCH codes. BCH codes (Bose-Chaudhuri-Hocquenghem codes) are cyclic ECCs that are constructed using polynomials over data blocks. A Reed - Solomon encoder accepts a block of data and adds redundant bits (parity bits) before transmitting it over noisy channels. On receiving the data, a decoder corrects the error depending upon the code characteristics.



Application Areas of Reed-Solomon Codes : The prominent application areas are -

1. Storage areas like CDs, DVDs, Blu-ray Discs
2. High speed data transmission technologies such as DSL and WiMAX
3. High speed modems
4. QR Codes
5. Broadcast systems such as satellite communications
6. Storage systems such as RAID 6.

3. **Concatenated code :** Concatenated Codes is a type of error-correcting code formed by the series combinations of two or even more codes to form a complex one. By this approach, a long length code is produced that increases the randomness thereby increasing the encryption ability. This facilitates more secured data communication. Basically, the word 'concatenation' is used in reference to a serially interconnected orientation of something. Here in reference to coding, concatenation is the series combination of two different codes that forms a long stream of codes. Concatenated codes came into existence in the year 1965 and were introduced by David Forney an American Electrical Engineer, in relevance to a theoretical explanation. However, with technological advancements, the concept was improvised and became popular. Moreover, in the 1970s NASA started utilizing the concatenated codes as its standard tool for coding in space communication. As the concatenated codes are series interconnection, likewise there is another coding that includes parallel interconnection and it is known as turbo code. However, concatenated code was the first one that gained huge importance in space communication. Thus, turbo code and various other modern capacity similar coding techniques are considered as an elaboration of it.

Q9. Describe Convolutional Codes and its structure.

Ans. The encoder will be represented in many different but equivalent ways. Also, the main decoding strategy for convolutional codes, based on the Viterbi Algorithm, will be described. A firm understanding of convolutional codes is an important prerequisite to the understanding of turbo codes.

Structure : A convolutional code introduces redundant bits into the data stream through the use of linear shift registers as shown in Figure

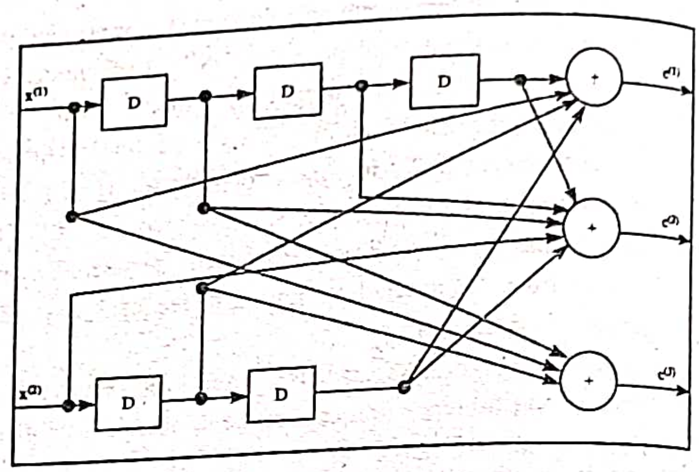


Figure: Example convolutional encoder where x(i) is an input information bit stream and c(i) is an output encoded bit stream

The information bits are input into shift registers and the output encoded bits are obtained by modulo-2 addition of the input information bits and the contents of the shift registers. The connections to the modulo-2 adders were developed heuristically with no algebraic or combinatorial foundation.

The code rate r for a convolutional code is defined as:

$$r = \frac{k}{n}$$

where k is the number of parallel input information bits and n is the number of parallel output encoded bits at one time interval. The constraint length K for a convolutional code is defined as

K m=+ 1 (2.2) where m is the maximum number of stages (memory size) in any shift register. The shift registers store the state information of the convolutional encoder and the constraint length relates the number of bits upon which the output depends. For the convolutional encoder shown in Fig. the code rate $r = 2/3$, the maximum memory size m=3, and the constraint length K=4. A convolutional code can become very complicated with various code rates and constraint lengths. As a result, a simple convolutional code will be used to describe the code properties as shown in fig.

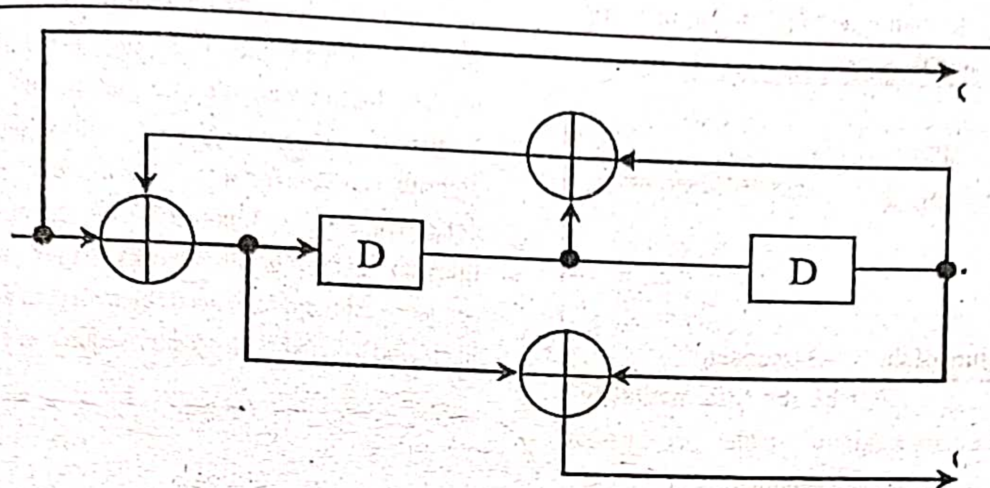


Figure : Convolutional encoder with $k=1, n=2, r=1/2, m=2$, and $K=3$.

Q10. Describe the method of decoding for the Viterbi and BCJR algorithms with the help of suitable diagrams.

Ans. 1. BCJR algorithm : The BCJR algorithm is implemented to solve the maximum a posteriori probability detection problem, which is a soft input soft output decoding algorithm with two recursions that is forward and backward both involve soft decisions invented by Bahl, Cocke, Jelnek and

Raviv. The viterbi algorithm is an algorithm, which operates on the principle of the maximum likelihood decoding. The maximun likelihood decoder, which examine received sequence and detect a valid path which has the smallest hamming distance from the received sequence. The viterbi algorithm is a soft input hard output algorithm, in which only the forward recursion involving soft decisions is possible. The BCJR algorithm is more complex than the viterbi algorithm because of backward recursions.

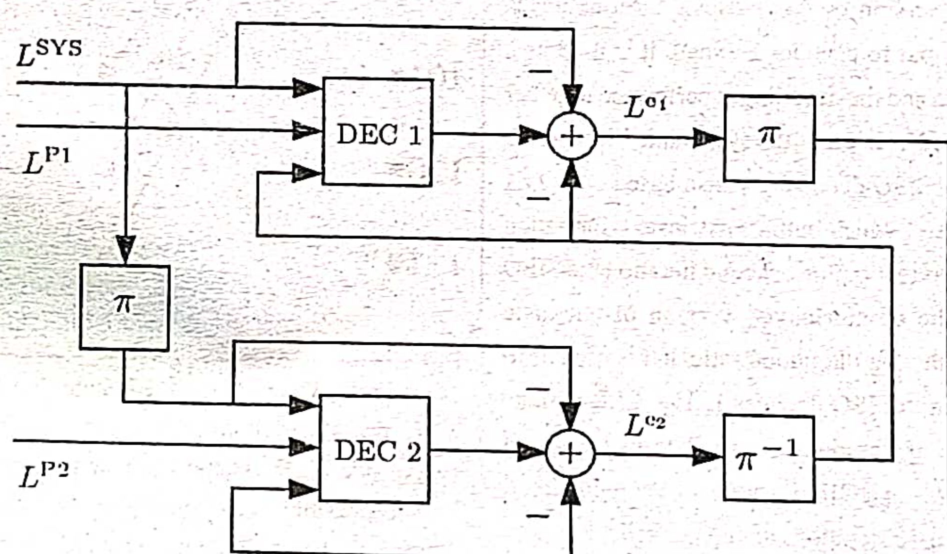


Fig : Structure of a Turbo Decoder based on either SOVA or the BCJR Algorithm.

In decoding section the received sequence is partitioned in to three, that are systematic bits, and parity check bits 1 and 2 . Here the systematic bits, parity check bits1 and a priori information, which is taken from SISO Decoder 2 is taken as the input to SISO Decoder 1 and the decoder 1 outputs extrinsic information and the log likelihood ratio as a result of estimation of a bit sequence by use of SOVA. SISO Decoder, which produces a-posteriori information by decoding a-priori information. Systematic information, parity information and a priori information are the inputs to the SISO Decoder.

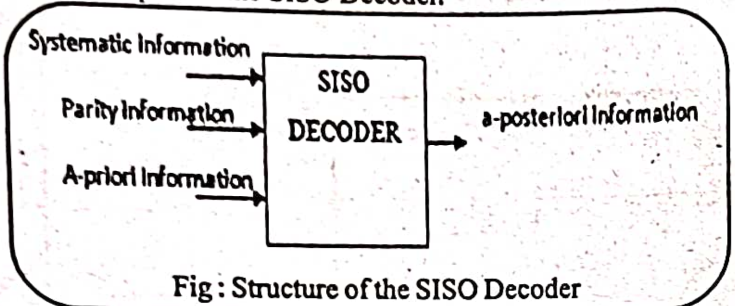


Fig : Structure of the SISO Decoder

Consider $u = (u_1, u_2 \dots u_N)$ be the information bits represented by the binary random variables. In the case of systematic encoders, one of the outputs $xs = (xs_1, xs_2 \dots xs_N)$ is similar to the information sequence u and the next is the parity information sequence output $xp = (xp_1, xp_2 \dots xp_N)$. In the MAP decoding scheme, the decoder decides whether uk =+1 or uk = -1, which depends on the sign of the log-likelihood ratio (LLR). In the case of radix-2 trellises the log domain computations of the BCJR algorithm can be separated in to three main categories that are branch metric computation, forward / backward metric computation and combination of forward and backward state metrics. The interleaved version of the extrinsic information is provided as an input to decoder 2, where it is used as a priori information and the decoding is performed together with an interleaved version of the systematic bits and the parity check bits. SISO decoder 2 – also based on SOVA like SISO decoder 1,which outputs extrinsic information and a log likelihood ratio. For a second iteration the SISO decoder takes the deinterleaved version of extrinsic information and the log likelihood ratio and is used as a-priori information in SISO decoder 1. Two LLR outputs after the number of iterations are used to make a hard decision. In the case of BCJR decoding of a convolutional turbo encoder 8 to 10 iterations are conducted.

2. ***Viterbi decording algorithm :*** The viterbi algorithm was introduced in 1967.The maximum likelihood decoding of convolutional codes can be executed by using this algorithm. This algorithm works by rejecting the less likely paths and keeping the most likely path through the trellis in each node. A hard decision on the transmitted sequence means that the path selection leaves with a single path in the Trellis. By using this algorithm the maximum likelihood sequence can be found. At the early point of the decoding process, loss of valuable information takes place due to the hard decision making. a- priori information the viterbi algorithm accepts the soft-inputs in the form of but it does not produce soft-outputs. By using the encoders Trellis diagram viterbi algorithm works as maximum likelihood sequence estimator. So that it selects a path with the highest likelihood by looking all possible sequences Trellis diagram.

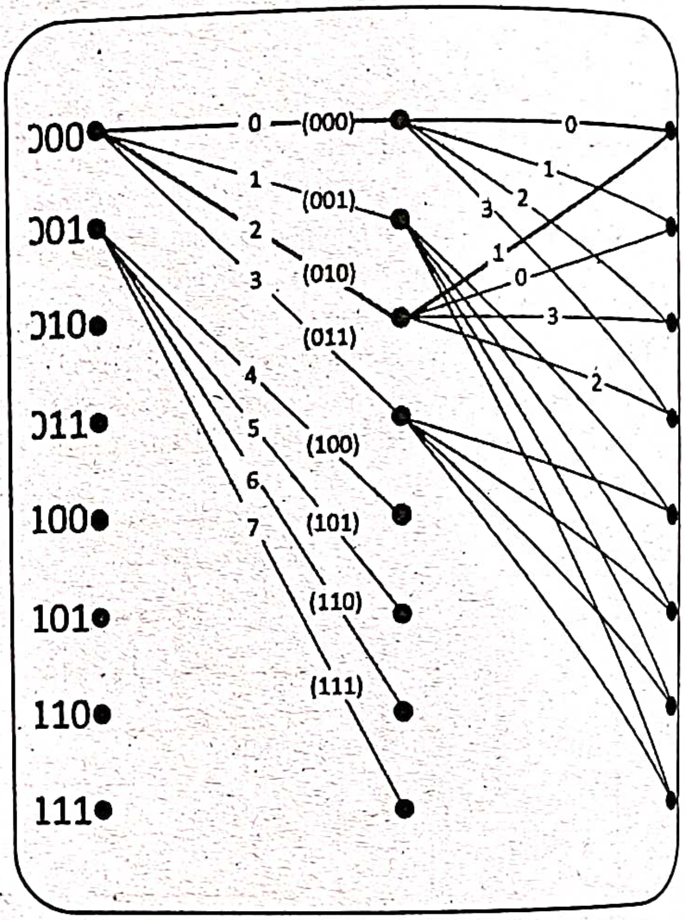


Fig : Trellis Diagram for one 8 State Constituent Encoder

It finds which path has the highest likelihood by considering the Hamming distance between incoming bits and possible transitions in the encoder (or Trellis) as a metric. The BCJR algorithm, which produces a soft estimate for each bit by considering the incoming bits as a maximum a- posteriori probability (MAP) detection problem. But the viterbi algorithm finds the most likely sequence and instead of maximizing the likelihood function for each bit it estimates several bits at once. So BCJR algorithm has the best performance than the Viterbi algorithm. Consider a constituent encoder with its trellis diagram, at which several possible paths are available. The amount of memory required to calculate the all possible paths is very large. So to reduce the amount of memory viterbi introduces paths through the Trellis diagram with smallest Hamming distance are known as survivor paths. Consider K is the constraint length of the encoder that is the encoders memory plus one (K = M+1), the Viterbi only takes 2K-1 survivor paths. The Viterbi algorithm works well on the small frames on the Trellis diagram. So for each iteration, decision of the best path is calculated. The decoding window moving forward through the branch and depends on the code in the Frame new decisions are made.

Q11. What are turbo codes? And what role of turbo codes.

Ans. Turbo codes form the basis of mobile communications in 3G and 4G networks. Invented in 1991 by Claude Berrou, and published in 1993 with Alain Glavieux and Punya Thitimajshima, they have now become a reference point in the field of information and communication technologies. As Télécom Bretagne, birthplace of these "error-correcting codes", prepares to host the 9th international symposium on turbo codes, let's take a closer look at how these codes work and the important role they play in our daily lives.

Role of turbo codes : In the digital communications sector, there are several error-correcting codes, with varying levels of complexity. Typically, repeating the same message several times in binary code is a relatively safe bet, yet it is extremely costly in terms of bandwidth and energy consumption.

Turbo codes are a much more developed way of integrating information redundancy. They are based on the transmission of the initial message in three copies. The first copy is the raw, non-encoded information. The second is modified by encoding each bit of information using an algorithm shared by the coder and decoder. Finally, another version of the message is also encoded, but after modification (specifically, a permutation). In this third case, it is no longer the original message that is encoded and then sent, but rather a transformed version. These three versions are then decoded and compared in order to find the original message.

Q12. Describe Low-Density Parity Check (LDPC) code and write its endoding and decoding.

Ans. Low - density parity check (LDPC) code is a linear error-correcting block code, suitable for error correction in large block sizes transmitted via very noisy channels. LDPC was developed by Robert G. Gallager, in his doctoral dissertation at the Massachusetts Institute of Technology in 1960. So, these codes are also known as Gallager codes.

* *Encoding by Low-Density Parity Check Codes :* A low - density parity check (LFPC) code is specified by a parity-check matrix containing mostly 0s and a low density of 1s. The rows of the matrix represent the equations and the columns represent the bits in the codeword, i.e. code symbols. A LDPC code is represented by , where is the block length, is the number of 1s in each column and is the number of 1s in each row, holding the following properties -

j is the small fixed number of 1's in each column, where j > 3

k is the small fixed number of 1's in each row, where k > j.

Example 1 - Parity Check Matrix of Hamming Code : The following parity check matrix Hamming code having n = 7, with 4 information bits followed by 3 even parity bits. The check digits are diagonally 1. The parity equations are given alongside -

Information Bits				Parity Bits			Parity Equations
x_1	x_2	x_3	x_4	x_5	x_6	x_7	
1	1	1	0	1	0	0	$x_5 = x_1 + x_2 + x_3$ or $x_1 + x_2 + x_3 + x_5 = 0$
1	1	0	1	0	1	0	$x_6 = x_1 + x_2 + x_4$ or $x_1 + x_2 + x_4 + x_6 = 0$
1	0	1	1	0	0	1	$x_7 = x_1 + x_3 + x_4$ or $x_1 + x_3 + x_4 + x_7 = 0$

Example 2 - Low - Density Parity Check Matrix : This examples illustrates an (12, 3, 4) LDPC matrix, i.e. n = 12, j = 3 and k = 4. This implies that each equation operates on 4 code symbols and each code symbol appears in 3 equations. Unlike parity check matrix of the Hamming code, this code does not have any diagonal 1s in parity bits.

x_1	x_2	x_3	x_4	x_5	x_6	x_7	x_8	x_9	x_{10}	x_{11}	x_{12}	Parity Equations
0	0	1	0	0	1	1	1	0	0	0	0	$x_3+x_6+x_7+x_8=0$
1	1	0	0	1	0	0	0	0	0	0	1	$x_1+x_2+x_5+x_{12}=0$
0	0	0	1	0	0	0	0	1	1	1	0	$x_4+x_9+x_{10}+x_{11}=0$
0	1	0	0	0	1	1	0	0	1	0	0	$x_2+x_6+x_7+x_{10}=0$
1	0	1	0	0	0	0	1	0	0	1	0	$x_1+x_3+x_8+x_{11}=0$
0	0	0	1	1	0	0	0	1	0	0	1	$x_4+x_5+x_9+x_{12}=0$
1	0	0	1	1	0	1	0	0	0	0	0	$x_1+x_4+x_5+x_7=0$
0	0	0	0	0	1	0	1	0	0	1	1	$x_6+x_8+x_{11}+x_{12}=0$
0	1	1	0	0	0	0	0	1	1	0	0	$x_2+x_3+x_9+x_{10}=0$

Decoding of LDPC Codes : There are two possible decoding techniques of LDPC codes -

1. In the first technique, the decoder does all the parity checks as per the parity equations. If any bit is contained in more than a fixed number of unsatisfied parity equations, the value of that bit is reversed. Once the new values are obtained, parity equations are recomputed using the new values. The procedure is repeated until all the parity equations are satisfied.This decoding procedure is simple and but is applicable only when the parity-check sets are small.
2. The second method performs probabilistic algorithms on LDPC graphs. The graph is a sparse bipartite graph that contains two sets of nodes, one set representing the parity equations and the other set representing the code symbols. A line connects node in first set to the second if a code symbol is present in the equation. Decoding is done by passing messages along the lines of the graph. Messages are passed from message nodes to check nodes, and from check nodes back to message nodes and their parity values are calculated.The two subclasses of these methods are belief propagation and maximum likelihood decoding. Though these decoding algorithms are complex, they yield better results than the former.

OBJECTIVE QUESTIONS AND ANSWERS

1. An event has two possible outcomes with probability $P_1 = 1/2$ and $P_2 = 1/64$. The rate of information with 16 outcomes per second is:

   1. (38/4) bits/sec 2. (38/64) bits/sec

   3. (38/2) bits/sec 4. (38/32) bits/sec

Answer : 1

2. Let (X_1, X_2) be independent random varibales. X_1 has mean 0 and variance 1, while X_2 has mean 1 and variance 4. The mutual information $I(X_1, X_2)$ between X_1 and X_2 in bits is_____

Answer : 0

3. A (7, 4) block code has a generator matrix as shown.

$$G=\begin{bmatrix} 1 & 0 & 0 & 0 & 1 & 1 & 0 \\ 0 & 1 & 0 & 0 & 0 & 1 & 1 \\ 0 & 0 & 1 & 0 & 1 & 1 & 1 \\ 0 & 0 & 0 & 1 & 1 & 0 & 1 \end{bmatrix}$$

If there is error in the 7th Bit then syndrome for the same will be

   1. 001 2. 010

   3. 100 4. 011

Answer : 1

4. If the SNR of 8 kHz white bandlimited Gaussian channel is 25 dB the channel capacity is:

   1. 2.40 kbps 2. 53.26 kbps

   3. 66.47 kbps 4. 26.84 kbps

Answer : 3

5. The channel capacity is measured in terms of:

   1. bits per channel

   2. number of input channels connected

   3. calls per channel

   4. number of output channels connected

Answer : 1

6. A binary random variable X takes the value +2 or –2. The probability $P(X=+2)=a$. The value of a (rounded off to one decimal place), for which the entropy of X is maximum, is

Answer : 0.5

7. For a White Additive Gaussian Channel, the channel bandwidth is 100 MHz, and the S/N power ratio is 40 dB, find the Channel capacity in bits/sec

   1. 1328.786×10^6 bits/sec

   2. 1248.687×10^6 bits/sec

   3. 1245.687×10^6 bits/sec

   4. 2245.687×10^6 bits/sec

Answer : 1

8. Channel capacity of a noise-free channel having m symbols is given by:

   1. m^2 2. 2^m

   3. $\text{Log}_2 m$ 4. m

Answer : 3

9. A speech signal, band-limited to 4 kHz, is sampled at 1.25 times the Nyquist rate. The speech samples assumed to

be statistically independent and uniformly distributed in the range -5 V to +5 V, are subsequently quantized in an 8-bit uniform quantizer and then transmitted over a voice-grade AWGN telephone channel. If the ratio of the transmitted signal power to channel noise power is 26 dB, the minimum channel bandwidth required to ensure reliable transmission of the signal with an arbitrarily small probability of transmission error (rounded off to wo decimal places) is_______ kHz.

Answer : 9.24–9.28

10. For a system having 16 distinct symbols, maximum entropy is obtained when probabilities are:

1. 1/8 2. 1/4
3. 1/3 4. 1116

Answer : 4

11. An analog baseband signal, band-limited to 100 Hz, is sampled at the Nyquist rate. The samples are quantized into four message symbols that occur independently with probabilities $P_1=P_4=0.125$ and $P_2=p_3$. The information rate (bits/sec) of the message source is _____

Answer: 360-363

12. A binary communication system makes use of the symbols "zero" and "one". There are channel errors. Consider the following events:

x_0: a "zero" is transmitted
x_1: a "one" is transmitted
y_0 : a "zero" is received
y_1 : a "one" is received

The following probabilities are given:

$$P(x_0)=\frac{1}{2}, P(y_0\,|\,x_0)=\frac{3}{4} \text{ and } p(y_0\,|\,x_1)=\frac{1}{2}.$$

The information in bits that you obtain when you learn which symbol has been received (while you know that a zero has been transmitted) is ______

Answer : 0.80 – 0.82

13. ______ is also called vertical redundancy check, one of the types of error detection in communications.

1. Longitudinal check
2. Sum technique
3. Parity checking
4. Cyclic check

Answer : 3

14. What is the entropy of a communication system that consists of six messages with probabilities 1/8, 1/8, 1/8, 1/8, 1/4, and 1/4 respectively?

1. 1 bits/message 2. 2.5 bits/message
3. 3 bits/message 4. 4.5 bits/message

Answer :2

15. If the probability of a message is 1/4, then the information in bits is:

1. 8bit 2. 4bit
3. 2bit 4. 1 bit

Answer : 3

16. In a high school having equal number of boy students and girl students, 75% of the students study Science and the remaining 25% students study Commerce. Commerce students are two times more likely to be a boy than are Science students. The amount of information gained in knowing that a randomly selected girl student studies Commerce (rounded off to three decimal places) is_______bits.

Answer : 3.320 – 3.325

17. An Ideal power limited communication channel with additive white Gaussian noise is having 4 kHz band width and Signal to Noise ratio of 255. The channel capacity is:

1. 8 kilo bits/see 2. 9.63 kilo bits/sec
3. 16 kilo bits/sec 4. 32 kilo bits / sec

Answer : 4

18. Discrete source S_1 has 4 equiprobable symbols while discrete source S_2 has 16 equiprobable symbols. When the entropy of these two sources is compared, the entropy of:

1. S_1 is greater than S_2
2. S_1 is less than S_2
3. S_1 is equal to S_2
4. Depends on Fate of symbols/second

Answer : 2

19. In a binary source, 0s occur three times as often as is. What is the information contained in the is?

1. 0.415 bit 2. 0.333 bit
3. 3 bit 4. 2 bit

Answer : 4

20. A source produces three symbols A, B and C with probabilities, P(A) = 1/2, P(B) = 1/4 and P(C) = 1/4. The source entropy is

1. 1/2 bit/symbol
2. 1 bit/symbol
3. 1 1/4 bits/symbol
4. 1 '1/2 bits/symbol

Answer : 4

Notes :