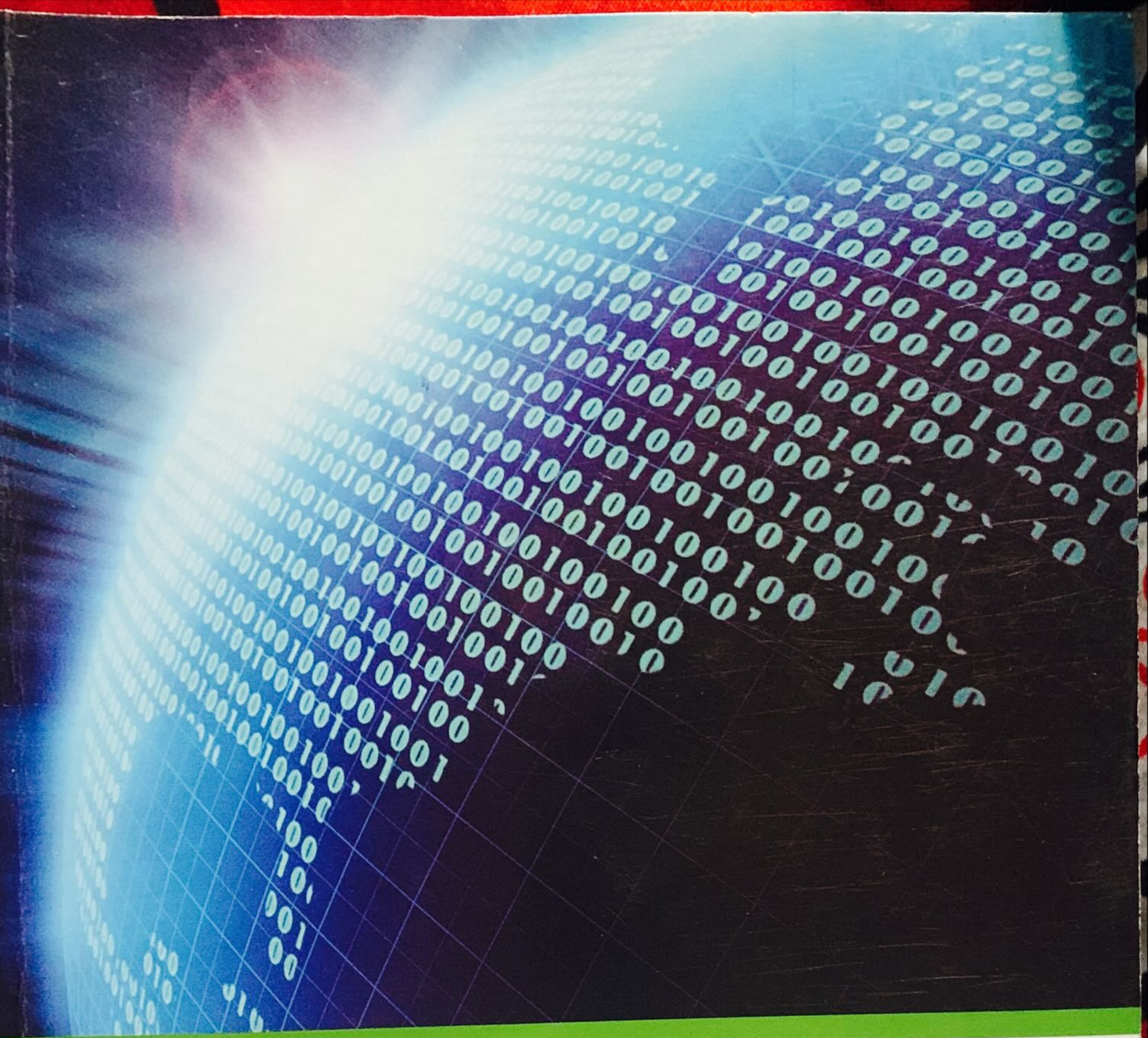

DIGITAL COMMUNICATION

for Bachelor's Degree in Engineering

Prof. Dr. Dinesh Sharma
Er. Suramya Sharma

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Preface

This book focuses on the study of ***Basic Principles of Digital Communications***. There is no other book by a Nepalese author which caters to all the topic as per the revised syllabus prescribed by Institute of Engineering (**IOE**) and Pokhara University (**PU**). The text emphasizes the basic aspects of information theory, discrete modulation techniques, and coding theory at a level appropriate for students of bachelor in electronics and communication with the assumption that the students have knowledge of signals and systems.

This book is divided into eight chapters. Every chapter starts with the brief introduction of that chapter. Chapter 1 provides the basic introduction to digital communications based on individual blocks of Digital Communication System (DCS). Chapter 2 deals with Sampling and Reconstruction focusing of Sampling Theorem and its proof, reconstruction, and practical considerations of sampling and reconstruction. Chapter 3 discusses analog and digital pulse modulation techniques. Initially, the chapter focuses on analog pulse modulation techniques such as Pulse Amplitude Modulation (PAM), Pulse Width Modulation (PWM) and Pulse Position Modulation (PPM), and later focusing on digital pulse modulation techniques such as Pulse Code Modulation (PCM), Differential Pulse Code Modulation (DPCM) and Delta Modulation (DM). Chapter 4 provides a tutorial of Information Theory and Channel Capacity. Later this chapter focuses on pulse shaping and line coding.

Chapter 5 gives the brief overview of digital continuous wave modulation techniques such as amplitude (ASK) and angle (FSK, PSK) modulations. Background on theories of probability and random process is provided in Chapter 6, which are considered second tools required in study of any communication system. Chapter 7 focuses on noise performance analysis techniques used in digital and analog modulation and there comparison. Finally, the principles and key practical aspects of error control coding are given in Chapter 8.

The authors have taken every effort to ensure the accuracy in the book. However, the book may contain some error. The authors would be very grateful to the readers for bringing any such errors into their notice. Any suggestions for further improvement would by highly appreciable.

Prof Dr. Dinesh Sharma

Er. Suramya Sharma

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Chapter 1 INTRODUCTION

Introduction

Communication is the study of the fundamental concepts and principles related to transferring information from one place to another. In simple words communication is the process of establishing connection or links between two points for information exchange. Information exchange involves transmission, reception and processing of information between locations.

Various intermediate steps are involved while transmitting information from the source to destination. Each step is carried out using specific equipment to perform certain transformation to the information. These individual equipment which are used for communication purpose, are called communication equipment. Different communication equipment when assembled together forms a communication system.

Basically, communication system is of analog or digital type. In analog communication system, the information bearing signal is continuously varying in both amplitude and time, and it is directly used to modify some characteristic of a sinusoidal carrier wave, such as amplitude, phase, or frequency. Whereas in digital communication system, the information bearing signal is processed so that it can be represented by a sequence of discrete messages. This book deals with the study of digital communications.

1.1 Signal

A signal is a function of one or more independent variables which contains some information. In electrical sense, the signal can be voltage or current which is the function of time as an independent variable. In our daily life, we come across several signals such as radio signal, T.V. signal, computer signal etc.

Major types of signal encountered in the communication systems are:

Analog Signals

The continuous variation of dependent variable (i.e., voltage or current) with respect to a continuous variation of independent variable (usually the time), is called analog or continuous time signal. In other words an analog signal is defined for every value of time and they take on continuous values in a given interval of time.

For example.

$$x(t) = 50 \sin(100\pi t) \tag{1.1}$$

is an analog or continuous time signal, because for every instance of time 't' the value of x(t) can be defined.

Discrete Signals

If we have infinite continuous variation of dependent variable with respect to discrete values of independent variable, then it is called discrete signal. Thus, discrete signals are defined for discrete valve of time so they can be referred as discrete time signal. It doesn't mean that dependent variable don't have any valve at the times other than defined by discrete valve of independent variable, time. It may or may not have valves at those instant.

For example,

$$x[n] = 50 \sin(100\pi n) \tag{1.2}$$

where, n= 0, 1, 2, . . .

This signal is defined, i.e., the value of x[n] is known, only for the given values of 'n'. Here the value of x[n] could be anything between -∞ to +∞.

Digital Signal

If we limit the infinite continuous variation in a discrete signal by allowing only finite fixed variations of dependent variables with respect to discrete values of independent variable, then signal is called discrete time; discrete valued signal. In general language it is called Digital signal.

For example,

$$x[n] = \begin{cases} 1 & for\ n = 0 \\ 0 & for\ n = 2 \end{cases} \tag{1.3}$$

So we can summarize the signal as follows:

Analog signal	**Continuous dependent variable**	**Continuous independent variable**
Discrete signal	Continuous dependent variable	Discrete independent variable
Digital signal	Discrete dependent variable	Discrete independent variable

1.2 Elements of Digital Communication System

The purpose of a communication system is to transmit an information bearing signal, from a source to destination, located at some distance away. The intermediate steps involved while transmitting information from the source to destination in digital communication system is described by the block diagram shown in Fig.1.1.

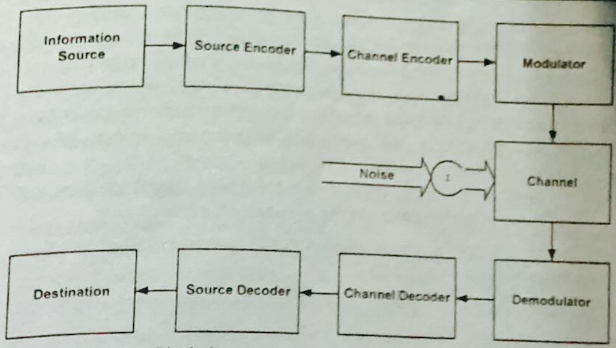

Fig.1.1: Block diagram of digital communication system.

In the overall functional block diagram of digital communication system shown in Fig 1.1, three basic signal processing operations are, source coding / decoding, channel coding / decoding, and modulation / demodulation.

The various elements of the block are as follows.

1. **Information source**

   Based on the nature of electrical signal produced at the output of information source, it is classified into two categories .i.e., analog information source and discrete information source.

   The output of analog information source is analog electrical signal, which is a continuous function of time. A speech signal, a television signal, and a signal representing atmospheric temperature or pressure at some location are the examples of analog signals produced by analog information source.

   In digital communication system, the information source produces a message signal which is an analog but discrete time signal. The output of discrete information sources such as a teletype or the numerical output of a computer consists of a sequence of discrete symbols or letters. An analog information source may be transformed into a discrete information source with the help of quantization process. Discrete information sources are characterized by the following parameters:

   i. Source symbols: These are letters, digits or special characters available from the information source.

   ii. Symbol rate: It is the rate at which the information source generates source symbols. It is generally expressed in symbols/sec.

   iii. Source symbol probabilities: Each source symbol from the source has independent occurrence rate in the sequence. As an example in normal English text, letters A, E, I, O, U etc., occur frequently in the sequence. Hence probability of the occurrence of each source symbol can become one of the important properties useful in digital communication.

iv. Probabilistic dependence of symbols in sequences: The information carrying capacity of each source symbol is different in a particular sequence. This parameter, called the source entropy, defines average information content of the symbols. The entropy of a source describes the average information content per symbol in long and statistically independent sequence of symbols in the messages. Entropy may be defined in terms of bits per symbol. Bit is the abbreviation for a binary digit (or sometime the binary unit to represent the measure of information).

This means that the source information rate is the product of symbol rate and source entropy i.e.,

Information rate = Symbol rate × Source entropy

(bits/sec) (symbols/sec) (bits/symbols)

Thus, the information rate represents minimum average data rate required to transmit information from source to the destination. The detail description about entropy and information rate will be dealt in upcoming chapters.

2. **Source encoder/decoder**

The sequence of symbols from the information source acts as an input to the source encoder. In digital communication, these symbols are in the form of discrete signals. These symbols are converted into binary sequence of 1's and 0's by assigning code words (i.e., combination of 1's and 0's) as shown in Fig.1.2. For each distinct symbol, there is a unique code word.

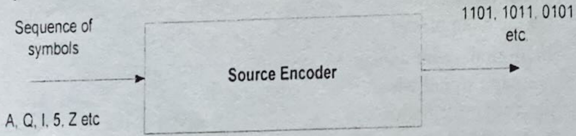

Fig.1.2. Operation of Source encoder

For example the output of an English teletype is a sequence of symbols containing total 32 different symbols. To assign code words to each symbol we need at least 5 bit code i.e., 2^5=32 symbols. Now if the symbol rate is 10 symbols/sec then the data rate at the output of the source encoder will be 10 · 5=50 bits/sec.

The above coding technique is an example of a Fixed Length Coding (FLC), in which each symbol is represented by fixed number of bits in a code word. FLC is efficient only if the input symbols occur with equal probability and are statistically independent. This is not the case in English text, probability of occurrence of symbols a, e, i, o, u is much higher than symbols q, v, or z.

Secondly, the symbol occurrence sequence is not statistically independent. It is most probable that symbol 'u' will occur after symbol 'q' and consonant in most cases are followed by a vowel symbol. This property is exploited in generation of a source code by assigning short code words to frequent source symbols, and long code words to rare source symbols. Such a source code is known as Variable Length Code (VLC). The examples of variable length codes are Huffman coding and Shannon-Fano coding which will be discussed later in this book.

An optimum source encoder is that which has the output data rate R_{out} nearly or equal to input symbol rate R_{in}. The efficiency of source encoder is therefore measures in terms of ratio of output data rate to minimum theoretically achievable data rate.

Source decoder is used to perform the reverse operation of to the source encoder. It converts the binary output of the channel decoder into sequence of symbols with minimum error (due to bit errors) and maximum efficiency.

3. **Channel encoder/decoder**

Channel encoder is used to enhance reliability & efficiency of high-speed digital signal transmission. In general the output of source encoder, in the binary form, can be directly fed to modulator. However, in doing so, loss or improper detection of any of the information bearing bit in the receiver side may distort the complete word or even the complete message. The channel encoder therefore adds some error control bits to the bit stream of source encoder output. These error control bits (redundancy), that does not carry any information make possible for the receiver to detect and, in most cases, correct, some of the errors in message bearing bits.

Usually in each block of '*k*' information bearing bits, '*r*' error control bits are added. This type of coding method is called block coding. In other method, called convolution method, information bit and error control bits are continuously interleaved. All channel encoding methods require storage & processing of the information.

Basic parameters of channel encoder are, method of coding, rate or efficiency of the coder, error control capabilities, complexity etc.

Channel decoder recovers information bearing bit streams from coded bit streams with minimum error and maximum efficiency. Complexity of the decoder and the time delay are the two basic design criteria of the channel decoder.

4. **Channel modulator/demodulator**

Channel modulator is intended to convent bit streams from channel encoder to electrical waveform suitable for transmission over communication channel. The modulator operates by keying shifts in the amplitude, frequency, or phase of a sinusoidal carrier wave by the pulses from channel encoder output resulting in digital modulation technique referred to as amplitude-shift keying (ASK), frequency-shift keying (FSK), or phase-shift keying (PSK) respectively. Proper design of channel modulator can effectively minimize effects of noise, increase matching of signal-characteristic (frequency, power and bandwidth) with channel characteristics and provide multiple data communication over the same physical channel. (multiplexing).

Channel demodulator converts received electrical signal into sequence of bits (pulses) with minimum error & maximum efficiency.

5. **Communication Channel**

The communication channel is a physical media (cable for wire communication or free space for radio communication) with parameters that limit the speed or rate of data communication. Communication channel has finite frequency bandwidth, i.e., the range of frequencies for which the channel does not introduce substantial attenuation. The signal power is attenuated as it travels along the channel and noise is introduced by the system blocks. Because of power attenuation and noise, the received signal is distorted.

Shannon, Hartley & Nyquist played significant role in characterizing channel in terms of its capacity to transmit signal with minimum acceptable signal to noise ratio (SNR) or equivalent Bit Error Rate (BER).

The Shannon-Hartley channel capacity theorem state that for given channel bandwidth (B) and required (at the input of the receiver) level of signal-to-noise ratio (SNR), the maximum speed of data transmission (or the channel capacity (C)) is limited i.e., for given B & SNR only C can maintain error free communication.

$$C = B \log (1+SNR) \tag{1.4}$$

From users point of view the complete digital communication system comprises of Data Terminal Equipment (Teletype, computer, data-logger etc) and modulator-demodulator (MODEM) units.

1.3 Noise

Unwanted or undesirable electrical signal which are introduced adds up with a message signal during the transmission are called noise. Noise corrupts a desired message signal and degrades the performance of communication system. In general, noise is unpredictable (random) in nature. Detection of the message signal at the receiver depends upon the amount of noise accompanied by the message during the process of communication. The amount of noise power present in the received signal decides the minimum power level of the desired message signal at the transmitter.

There are various sources of random noise and are broadly classified as external noise, and internal noise.

a. **External Noise**

External noise is generated by outside the device or circuit. Three primary types of external noise are atmospheric, extraterrestrial, and industrial or manmade.

(i) **Atmospheric Noise**

Atmospheric noise are the electrical disturbance within the earth's atmosphere which produces strange sounds like sputtering and crackling in shortwave receivers. Most of noise are from other radio signals which propagate over the earth's atmosphere in the same way as ordinary radio waves of same frequencies.

Atmospheric noise caused by lightning and thunderstorms (local or distant) is commonly called static electricity noise. The magnitude of noise energy and frequency are inversely related resulting in less noise amplitude at higher frequencies. Thus, atmospheric noise interferes less in television reception than that of radio.

(ii) **Extraterrestrial Noise**

Noise originating from outside the earth's atmosphere is known as Extraterrestrial noise or deep space noise. It originates from the Milky Way, other galaxies, and the sun.

(iii) **Industrial Noise (Man-made noise)**

Electrical noise produced by sources such as automobiles and aircraft ignition, electrical motors and switch gears, leakage from high voltage lines, fluorescent lights, and numerous other heavy electrical machines is known as industrial or man-made noise. Such noises are produced by the arc discharge taking place during operation of these machines. Man-made noise is most intensive in industrial and densely populated areas. Man-made noise in such areas far exceeds all other sources of noise in the frequency range extending from about 1 MHz to 600 MHz.

b. **Internal noise**

Noise generated within a device or circuit is known as Internal noise. Active or passive devices found in the receiver mainly contribute to internal noise. Such noise is generally random, thus described and observed statistically, since it is distributed randomly over the entire radio spectrum. Random noise power is proportional to the bandwidth over which it is measured. Various types of internal noise are discussed below.

(i) **Thermal noise (Johnson Noise)**

Thermal noise is linked with the rapid and random movement of charge carriers (usually the electrons) within a conductor at equilibrium. The thermal agitation happens regardless of any applied voltage. Free electrons within an electrical conductor possess kinetic energy as a result of which, heat is exchanged between conductor and its surroundings. The atoms and molecules of all substances vibrate constantly in a minute motion. As they vibrate, they send electromagnetic waves of all frequencies. Thermal noise also known as Johnson noise names after its discoverer Johnson, who proved that the thermal noise power is proportional to the product of bandwidth and temperature. Mathematically, the noise power P_n is

$$Pn = kTB \quad (1.5)$$

Where, P_n is the noise power in watt.

T is the absolute temperature in Kelvin.

B is the bandwidth in Hz.

K is the Boltzmann's constany (1.38×10^{-23} joule/kelvin).

(ii) **Shot noise**

Shot noise is always associated with direct current flow. Current flow is not continuous, at some (supposedly small, presumed microscopic) level, currents vary in unpredictable ways. This occurs because the charge particles (i.e., electrons and/or holes) do not cross the potential barrier simultaneously, but in a random distribution pattern. This gives rise to a random component of current which gets superimposed on the steady base junctions charge particles resulting in shot noise.

Shot noise also has a flat spectrum similar to thermal noise except in the range of microwave frequency.

(iii) **Burst Noise**

Burst noise is a low-frequency noise, observed in bipolar transistor. The name "burst" because the noise appears as a series of bursts. It produces a popping sound and hence called popcorn noise. The source of this noise is not clearly understood, but spectral density is known to increase as the frequency decreases.

(iv) **TransitTime Noise**

Transit Time noise is observed in transistors. Transit time is the duration of time that current carrier (hole or electrons) take to move from the input to the output. Even though the distances involved are minimal, the time it takes for the current carriers to move even a short distance is finite. The problem is more pronounced when the frequency of operation is high. The transit time shows random characteristics and is directly proportional to the frequency of operation.

(v) **Partition Noise**

Partition noise occurs due to random fluctuation whenever current has to divide between two or more paths. In case of partition noise, a diode would be less noisy than a transistor (all other factors being equal). It is for this reason that the inputs of microwave receivers are often directly fed to the diode mixers.

(vi) **Flicker Noise**

Flicker noise or modulation noise appears in transistors operating at low audio frequencies. Flicker noise is proportional to the emitter current and junction temperature. As flicker noise is inversely proportional to frequency, it may be neglected at frequencies above about 500 Hz.

1.4 Interference

Interference typically refers to the addition of unwanted, usually manmade signals which alters, modifies, or disrupts a useful signal as it travels along a channel between a source and a receiver. Most common interference are manmade and is usually signal from various broadcasting and communication systems.
Most common type of interference encountered in digital communication is Intersymbol Interference (ISI). ISI is a form of distortion of a signal in which one symbol interferes with subsequent symbols. This is an unwanted phenomenon as the previous symbols have similar effect as noise, thus making the communication less reliable. ISI is usually caused by multipath propagation or the inherent non-linear frequency response of a channel causing successive symbols to "blur" together.
The presence of ISI in the system introduces errors in the decision device at the receiver output. Therefore, in the design of the transmitting and receiving filters, the objective is to minimize the effects of ISI, and thereby deliver the digital data to its destination with the smallest error rate possible. Ways to fight intersymbol interference include adaptive equalization and error correcting codes.

1.5 Distortion

Distortion refers to the deviation in any parameter (like amplitude, frequency, shape) of a signal from that of an ideal signal. When a piece of equipment changes the shape of this sine wave, or any signal, then distortion is created. Simple sine wave consists of only one frequency. Distortion generates additional frequencies that were not present in the original signal. Two parameters that are of particular interest in communication system regarding distortion are the internal power dissipation that reduce the size of the output signal, and energy storage that alter the shape of the output. Distortions can be categorized into two types, linear and non-linear distortion.

1. **Linear distortion**

The term "linear distortion" means any change to the signal that does not change the shape of the individual sine wave components of a signal. This could be an irregularity in the frequency response, or it could be a change in the relative phase (timing) of the various frequencies. It can make the individual sine wave components bigger or smaller, or shift them in time, but it doesn't change them. Linear distortion includes any amplitude or delay distortion associated with a linear transmission system. Amplitude and delay distortion is easily described in the frequency domain as

(i) **Amplitude Distortion**

When different frequency components of the input signal are amplified or attenuated by different amounts, the output signal consists of distortions, known as amplitude distortions i.e., when the amplification or attenuation of the signal is not constant over the useful range of frequencies. Amplitude distortion, which occurs when

$$|H(f)| \neq |K| \tag{1.6}$$

Where H(f) is the transfer function of the system.

(ii) **Phase or Delay Distortion**

If the phase of the output signal is different from the phase of input signal then such distortion is known as phase distortion. Phase distortion leads to delay in the transmission of the signal. Hence, it is known as delay distortion. If different amounts of phase shifts occur at different frequencies of an output signal then it becomes necessary to compensate for such phase distortions. Whereas if same amount of phase shift occurs at all frequencies then such phase distortion can be ignored. Delay distortion occurs when

$$\arg H(f) \neq -2\pi t_d f \pm 180° \tag{1.7}$$

Where H(f) is the transfer function of the system, t_d is the finite delay time and m is an integer.

Distortionless transmission means that the output signal has the same shape as the input. More precisely, given an input signal x(t), we say that the output is undistorted if it differs from the input only by a multiplying constant and a finite time delay.

Analytically, we have distortionless transmission if

$$y(t) = Kx(t - t_d) \tag{1.8}$$

Where K is multiplying constant and t_d is the finite delay time and are constants.

Thus the output spectrum is

$$Y(f) = F[y(t)] = Ke^{-jwt_d} X(f) \tag{1.9}$$

Now by definition of transfer function,

$$Y(f) = H(f)\, X(f) \tag{1.10}$$

$$H(f) = Ke^{-jwt_d} \tag{1.11}$$

In words, a system giving distortionless transmission must have constant amplitude response and negative linear phase shift, so

$$|H(f)| = |K| \tag{1.12}$$

$$\arg\ H(f) = -2\pi t_d f \pm m180° \tag{1.13}$$

These criteria for distortionless transmissions are theoretical formulation. In practical application the transmission system always produce some amount of signal distortion.

2. **Nonlinear distortion**

When the sine wave components of the signal are changed by nonlinear elements then the distortion is nonlinear. A system having nonlinear elements cannot be described by a transfer function. Instead, the instantaneous values of input and output are related by a curve or function y(t) = T[x(t)], commonly called the transfer characteristic. In general, the input and output characteristics of a device/system can be approximated using the following equation:

$$y(t) = a_1x(t) + a_2x^2(t) + a_3x^3(t) + \ldots \tag{1.14}$$

The first component *x(t)* of the right hand side of the above equation produces the exact replica of the input signal scaled by some constant factor a_1. The remaining other terms produce non-linear distortion.

The nonlinear distortion appears as harmonics of the input wave. Generation of harmonics of the fundamental frequency of the input signal due to nonlinearity of the device can be evaluated as the ratio of amplitude of nth harmonic component and the amplitude of the fundamental frequency expressed in percentage as

$$\%n^{th} \text{ harmonic distrotion} = \%D_n \frac{|A_n|}{|A_1|} \times 100\% \tag{1.15}$$

Total harmonic distortion (THD) is expressed as

$$\%TDH = \sqrt{D_2^2 + D_3^2 + D_4^2 + \ldots + \ldots} \times 100\% \tag{1.16}$$

Higher-order harmonics are treated similarly. However, their effect is usually much less, and many can be removed entirely by filtering.

If the input is a sum of two cosine waves, say $\cos w_1t + \cos w_2t$, , the output will include all the harmonics of f_1, and f_2, plus crossproduct terms which yield $f_2 - f_1$, $f_2 + f_1$, $f_2 - 2f_1$, etc. These sum and difference of frequencies are designated as intermodulation distortion. Generalizing the intermodulation effect, if $x(t) = x_1(t) + x_2(t)$, then y(t) contains the components like $x_1^2(t)$, $x_2^2(t)$, and crossproduct $x_1(t)x_2(t)$. The frequency component of $x_1(t)x_2(t)$ may overlap with frequency components of $x_1(t)$ and $x_2(t)$ causing intermodulation distortion (or cross talk) and is of great concern in case of multiplexed transmission of number of message over common channel.

1.6 Advantages and Disadvantages of Digital Communication

Following are the advantages of digital communication

i. **Cheaper:** Digital communication system are simpler and cheaper than analog communication system because of the advancement made in the IC technology.

ii. **Accommodation of other services:** Since all signals in a digital system have the same format, other traffic sources (video, data) can be intermixed in a single transmission medium without mutual interference.

iii. **Performance monitorability:** The quality of digital signal is easier to monitor by observing the signal patterns or by introducing redundancy in form of parity bits. Analog systems cannot be monitored or tested for quality while in service.

iv. **Ease of signalling:** Call set-up establishment and other control information such as (on hook/off hook) are digital in nature and can be integrated easily into the digital system. In analog systems, this information needs a special attention.

v. **Encryption:** Using data encryption, only permitted receiver may be allowed to detect the transmitted data. This property is of its most importance in military applications.

vi. **Signal regeneration:** In analog systems, the attenuation in the signal is corrected by employing amplifiers, which not only boost the signal's power but the noise's too. In digital systems, the attenuated signals are regenerated by reading the binary digits and retransmitting a replica of the original signal while removing any noise, distortion, or interference.

vii. **Noise tolerence:** Since the transmitted signal is digital in nature, large amount of noise may be tolerated.

viii. **Error detection:** Since in digital communication, channel coding is used, therefore the error maybe detected and corrected in the receivers.

Disadvantage

i. **A/D and D/A conversion:** Because of the analog nature of voice, conversion between analog and digital forms is required. The electronics required to perform this task add additional cost to the digital systems.

ii. **Need for time synchronizatio:** A clock is needed whenever a digital signal is transmitted for time reference. The synchronization problem is more obvious when links and switches are interconnected to form the network.

iii. **Incompatibility with existing analog equipment:** The full advantages of digital systems are not realized unless the networks are all digital

Previous Exam Questions

1. State and Explain merits of Digital Communication System (DCS).
2. Compare Digital and Analog Communication Systems.
3. Draw functional block diagram of DCS and explain the function of each block
4. Why is Digital Communication preferred over analog? Explain with examples.
5. Differentiate between Noise and Interference. What are the limitations posed by them in Communication System?
6. Write short Notes on Distortion.

✡✡✡

Chapter 2

SAMPLI

Introduction

In any communication system, the
may be in digital or analog form.
digital source (i.e., digital comput
translation for transmission through
signal is analog in nature, as in sp
converted into digital form befo
system. The process of transform
known as sampling and is don
sampling process the signal is
encoding.

In sampling process, the continu
time signal by measuring the sign
instant of time or sampling rate
signal to be reconstructed from
sampling theorem provides the
rate for a given signal.

2.1 Sampling Theorem

In general, the sampling theo
reproduced from an appropria
interval of time". This theorem
samples of analog signal by e
words suitable for digital commu

In broad sense the sampling theo

1. A finite energy, strictly
frequencies higher than
samples (values) of the si
seconds apart (for transmit

2. A finite energy, strictly
frequencies higher than W
its samples (values) taken