Digital Communication

By

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CHAPTER 1
INTRODUCTION

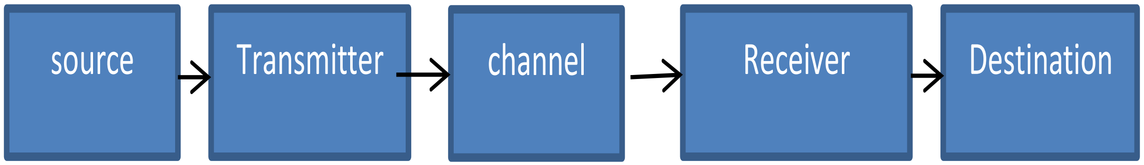
OBJECTIVES

- To understand block diagram of digital and data communication systems.
- To Compare digital with analog communication systems.
- To study the advantages of digital Comm. system

COMMUNICATION SYSTEM

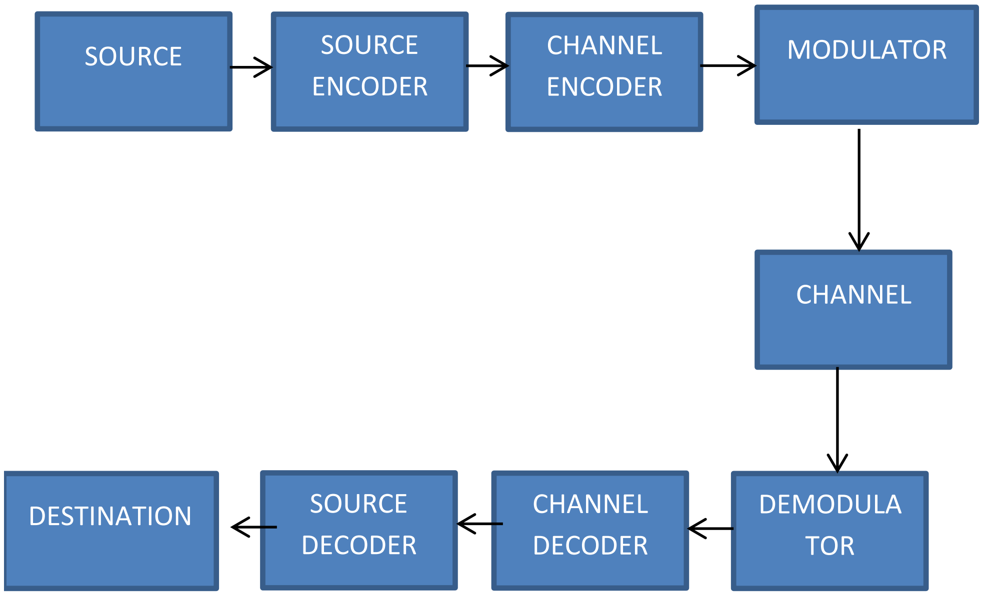
- A communication system is a system for transferring of information from a source(may be a system, a device) to a destination. An electronic communication system is transferring information using an electrical field as a mean of signal.
- A communication system may be analog or digital.

Block Digram of a communication system



Block Diagram of Digital Communication system

DIGITAL COMMUNICATION SYSTEMS.



SOURCE

The source of information can be an analog information or digital information source e.g. a speech signal or it may be produce a digital signal. If the signal is analog, it has to be converted in discrete form by sampling and quantization.

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SOURCE ENCODER

The symbols produced by the information source are given to source encoder. These are converted to digital form i.e binary sequence of 0's and 1's. Thus the source encoder assigns codewords to symbols. Some of the source encoders are Pulse Code Modulators, delta modulators etc.

CHANNEL ENCODER

The channel encoder does the coding for error correction. During the transmission of the signal, due to the noise in the channel, the signal may get altered and hence to avoid this, the channel encoder adds some redundant bits to the transmitted data. These are the error correcting bits. Channel encoder increases the reliability of the signal.

MODULATOR

The signal to be transmitted modulates a carrier. The signal is modulated ,converted to analog from the digital sequence, in order to make it travel through the channel or medium. Digital modulators map the input binary sequence of 1's and 0's to analog signal waveforms.

CHANNEL

The channel is the medium that allows the analog signal to travel from the transmitter end (source end) to the receiver end. It may be space, wires or optical cables.

DEMODULATOR

This is the first step at the receiver end. The received signal is demodulated as well as converted again from analog to digital. The signal is reconstructed here.

CHANNEL DECODER

The extra bits which were added by the channel encoders are used by the channel decoder to detect nd correct errors if any.

SOURCE DECODER

It does the reverse of encoder. It converts the binary output of the channel decoder into symbols

DESTINATION

This is the output which is produced after the whole process. example – The sound signal received.

COMPARISON OF DIGITAL WITH ANALOG COMMUNICATION SYSTEMS

Conventional radio broadcasting, TV broadcasting, pioneer telephone network used analog comm. System. Analog TV broadcasting employed AM for picture transmission and FM for sound transmission. The quality of received signal depends on the channel imperfections and noise in the channel. The channel may be space, coaxial cables, twisted pair wires. The fidelity of the signal depends on how well the waveforms are carried over the channel. It depend on the signal to noise ratio at the receiving end.

For this, in AM large power transmitters are required.
In FM, a wider frequency spectrum is required.
Thus in analog transmission, band width and power are not efficiently utilised.
Also **the advent of internet requires the transmission of speech , image, text, video to be transmitted over the common communication channel which rules out the use of analog system**.

Also in analog system, the baseband signal can have infinite values. Thus the **reproduction of analog signal at the receiving end requires to reconstruct the signal from infinite set of values.**

In analog system the information is in the shape of the signal. If the noise in the channel impairs the shape of the signal the information changes. Bit in digital comm. system, the signal is in the form of pulses. Here the shape of the pulse doesn't matter. The information depends on whether the pulse is present or not. So effect of noise is less in digital systems.

ADVANTAGES OF DIGITAL COMMUNICATION

1. The effect of distortion, noise, and interference is much less in digital signals as compared to analog signals.

2. Digital circuits are easy to design and cheaper than analog circuits.

3. The hardware implementation in digital circuits, is more flexible than analog.

4. The signal has high fidelity as the information depends on presence or absence of pulse .

5. secure communication as encryption and compression are employed in digital circuits to maintain the secrecy of the information.

6. Digital circuits are more reliable.

7. The probability of error occurrence is reduced by employing error detecting codes and error correcting codes.

8. Combining digital signals using Time Division Multiplexing *is* easier than combining analog signals using Frequency Division Multiplexing.

9. Digital signals can be stored and retrieved more conveniently than analog signals.

10. The capacity of the channel is effectively utilized by digital signals.

CHAPTER 2
SAMPLING THEOREM AND ITS BASIC CONCEPT.

OBJECTVES

To study Sampling Theorem
Introduction to PAM, PPM, PWM
To understand Quantization and error of Quantization
PCM, DPCM, their advantage and disadvantage
DELTA and ADAPTIVE DELTA Modulation concept of COMPANDING
Frequency hopping spread spectrum technique

Sampling Theorm

The sampling theorem, which is also called as **Nyquist criteria**, delivers the theory of sufficient sample rate for reproduction of the signal at the receiving end. This ensures that there is no loss of information in transmission of signal after sampling.
The sampling theorem states that a signal can be exactly reproduced if it is sampled at the rate **Fs** which is greater than twice the maximum signal frequency.

Sampling rate, Fs > 2signal frequency.

Quantization

The entire range of values of the analog signal is divided into quantization levels. Then the amplitude of each sample is taken to the nearest quantization level. This does the digitization of analog signals, rounding off of the values which are approximately equal to the analog values. The method of sampling chooses a few points on the analog signal and then these points are rounded off the value to a near stabilized value. Such a process is called as **Quantization.**

Quantization Error

Because of quantization inherent error is introduced in the signal. This is known as quantization error. It can be reduced by increasing the number of quantization levels i.e. by reducing the quantum step size.

Quantization Error

When a signal is quantized, the coded signal is an approximation of the actual amplitude value.

The difference between actual and coded value (midpoint) is referred to as the quantization error.

The more quantum levels, the smaller step will be the step size Δ which results in smaller errors. But more the levels, more are the bits required to encode the samples

Quantizing an Analog Signal

The analog-to-digital converters perform quantization to create a series of digital values equivalent of the given analog signal. The following figure represents an analog signal. This signal to get converted into digital, has to undergo sampling and quantizing.

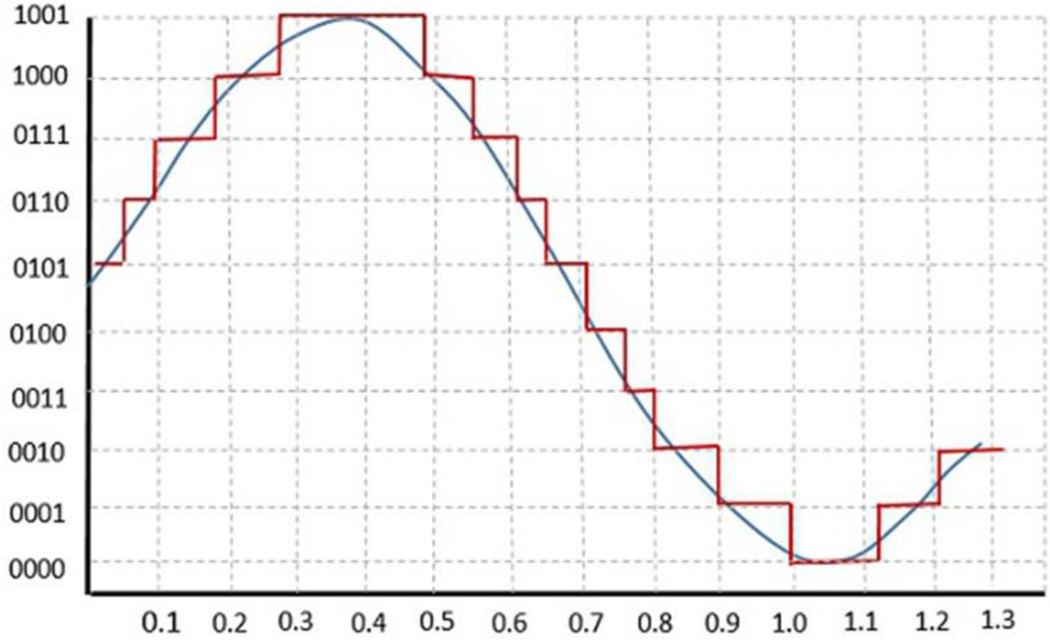
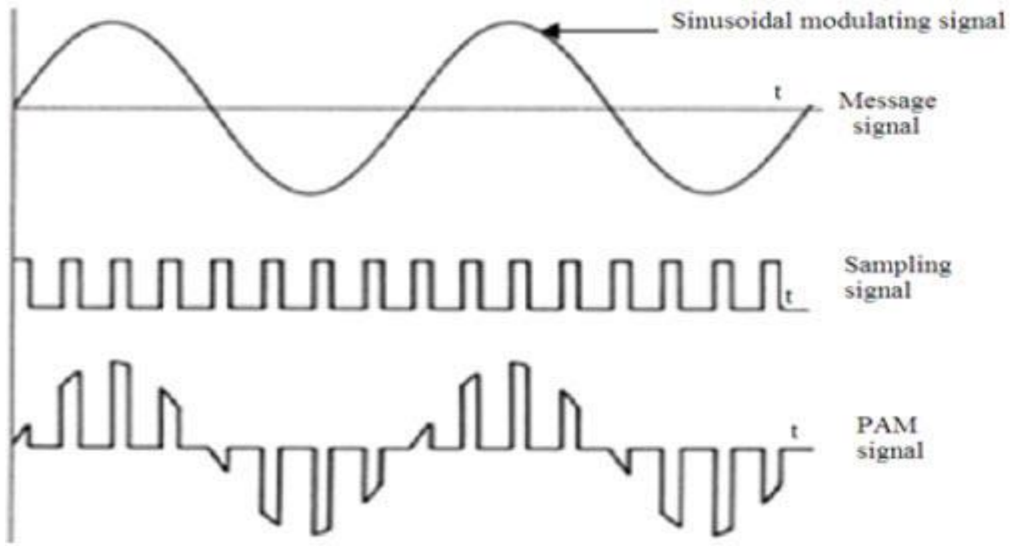


fig .quantization

PAM(Pulse Amplitude Modulation)

Pulse amplitude modulation is defined as the modulation in which the amplitude of the pulses is varied in accordance with the instantaneous value of the modulating or message signal.



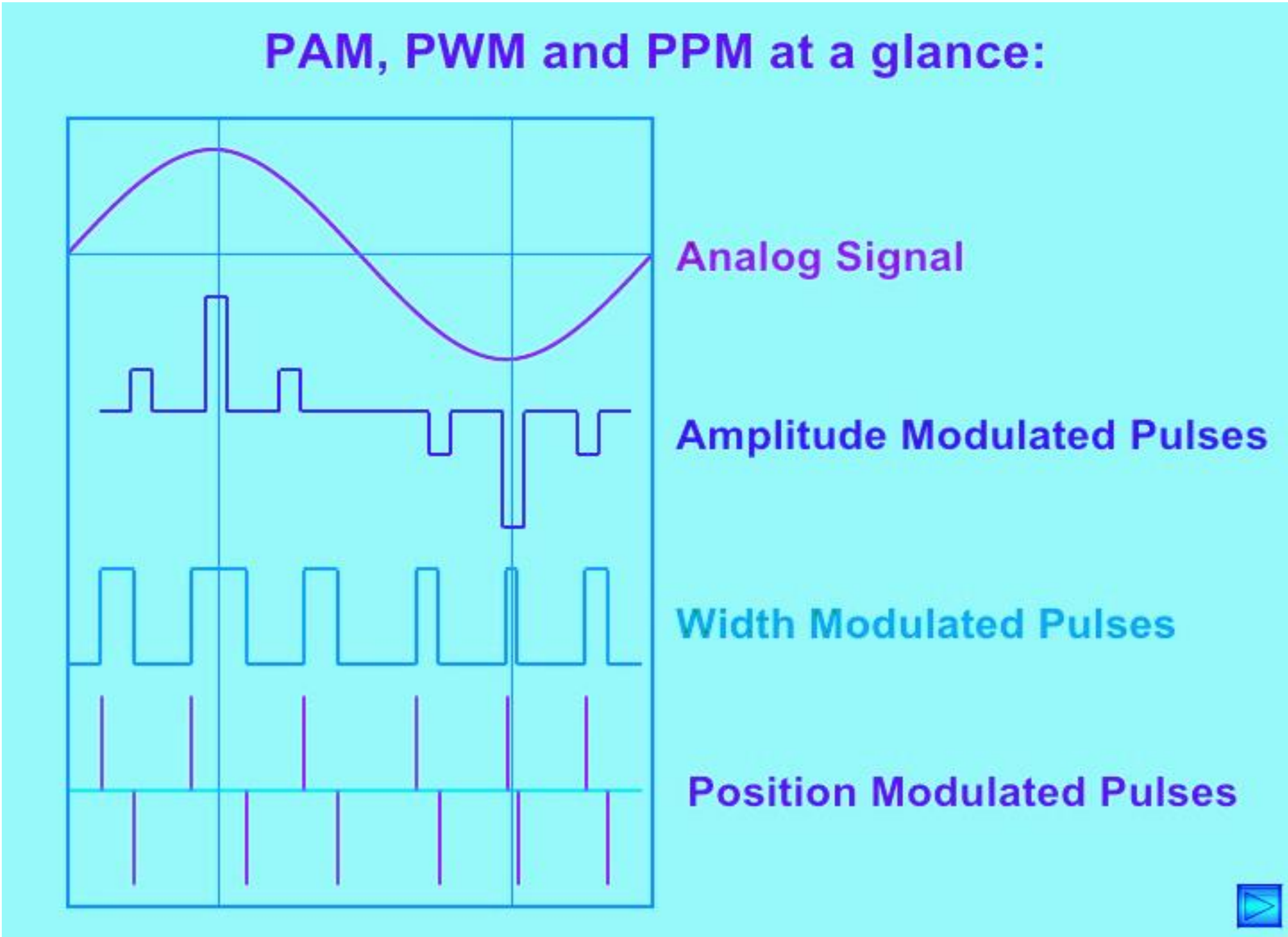
PPM(Pulse Position Modulation).

In this the amplitude and width of the pulses are kept constant, while the position of each pulse with reference to the position of a reference pulse is changed according to the instantaneous sampled value of the modulating signal.

The tramsmitter and receiver are to be kept in synchronism.

Pulse width modulation

In this the amplitude and position of the pulses are kept constant, while the width of each pulse is changed according to the instantaneous sampled value of the modulating signal.


PAM, PWM and PPM at a glance:
Analog Signal
Amplitude Modulated Pulses
Width Modulated Pulses
Position Modulated Pulses

PCM(Pulse Code Modulation)

In Pulse Code Modulation, the message signal is represented by a sequence of coded pulses. This message signal is achieved by representing the signal in discrete form in both time and amplitude.

Basic Elements of PCM

The transmitter section of a Pulse Code Modulator circuit consists of
Sampling,
Quantizing and
Encoding,

Sampling

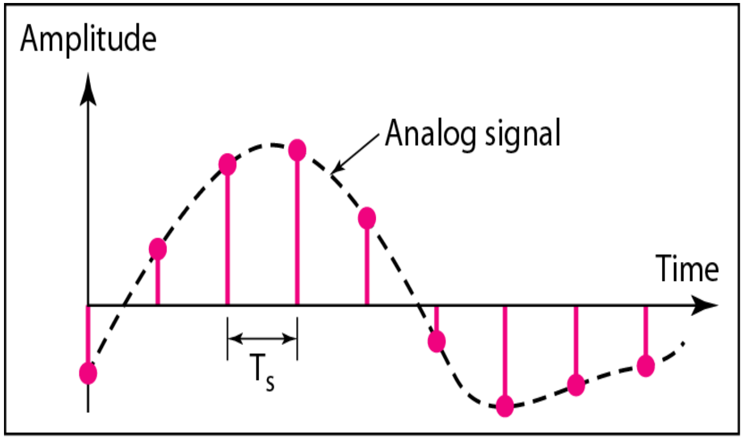
Instead of continuously transmitting the signal, Analog signal is sampled every T_S secs.

T_s is referred to as the sampling interval.

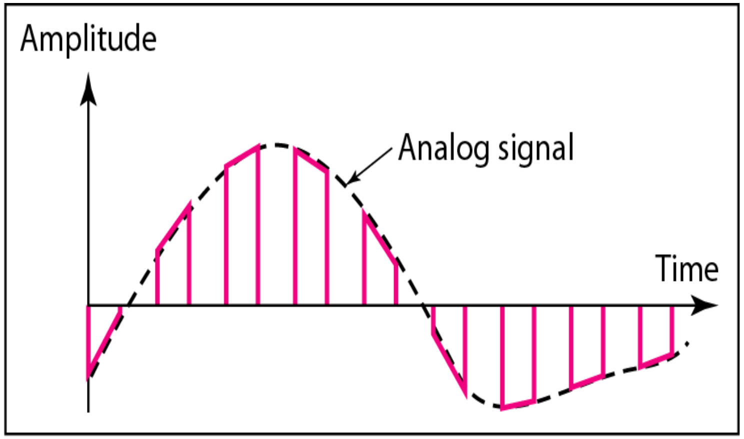
$F_s = 1/T_s$ is called the sampling rate or sampling frequency.

F_s > 2signal frequency(Nquist criteria)

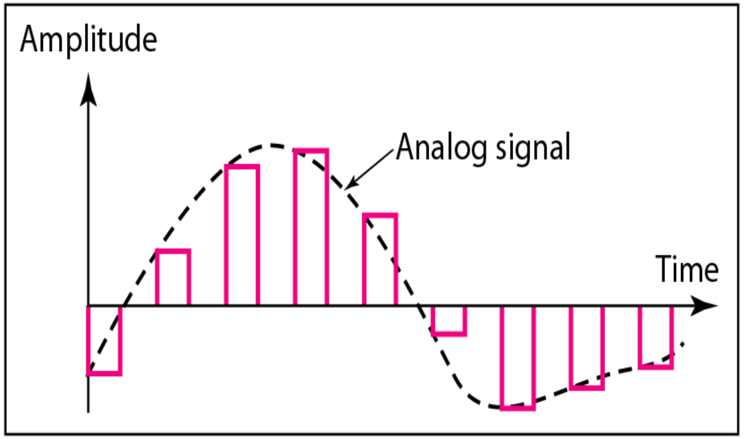
Different types of sampling in PCM



a. Ideal sampling



b. Natural sampling



c. Flat-top sampling

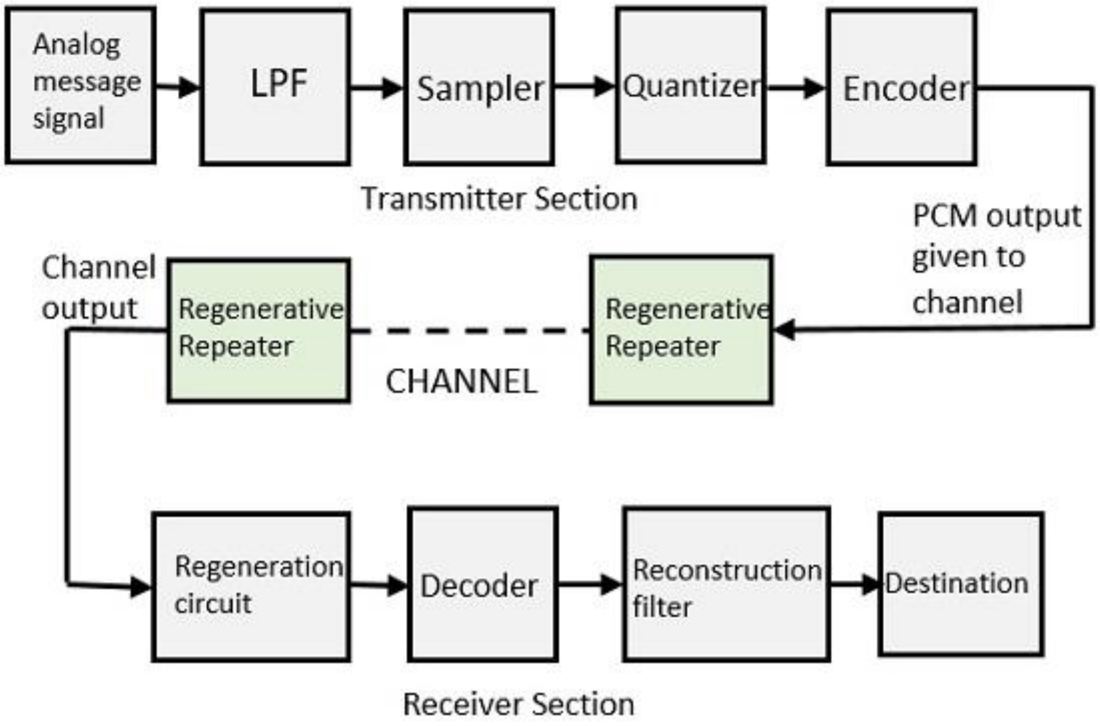
Quantization

The entire range of values of the analog signal is divided into quantization levels. Then the amplitude of each sample is taken to the nearest quantization level. This does the digitization of analog signals, rounding off of the values which are approximately equal to the analog values. The method of sampling chooses a few points on the analog signal and then these points are rounded off the value to a near stabilized value. Such a process is called as **Quantization.**

Encoding.

The quantized value is concertes to binary sequence.

Block Diagram of PCM



Low Pass Filter

This filter eliminates the high frequency components present in the input analog signal which is greater than the highest frequency of the message signal.

Sampler

The signal at the input is sampled i.e the instantaneous values of the signal is taken for the samples .The sampling theorem states that a signal can be exactly reproduced if it is sampled at the rate **Fs** which is greater than twice the maximum signal frequency

Quantizer

The value of the samples is quantized to the nearest quantum level.This compresses the data but introduces some noise known as quantization noise.

Encoder

The digitization of analog signal is done by the encoder. It designates each quantized level by a binary code.

Regenerative Repeater

Therepaeter increases the signal strength.

Decoder

The decoder circuit decodes the PCM waveform to reproduce the original signal. This circuit acts as the demodulator.

Reconstruction Filter

After the digital-to-analog conversion is done by the decoder and the regenerative circuit, a low-pass filter is employed, called as the reconstruction filter to retreive the original signal.

Companding in PCM

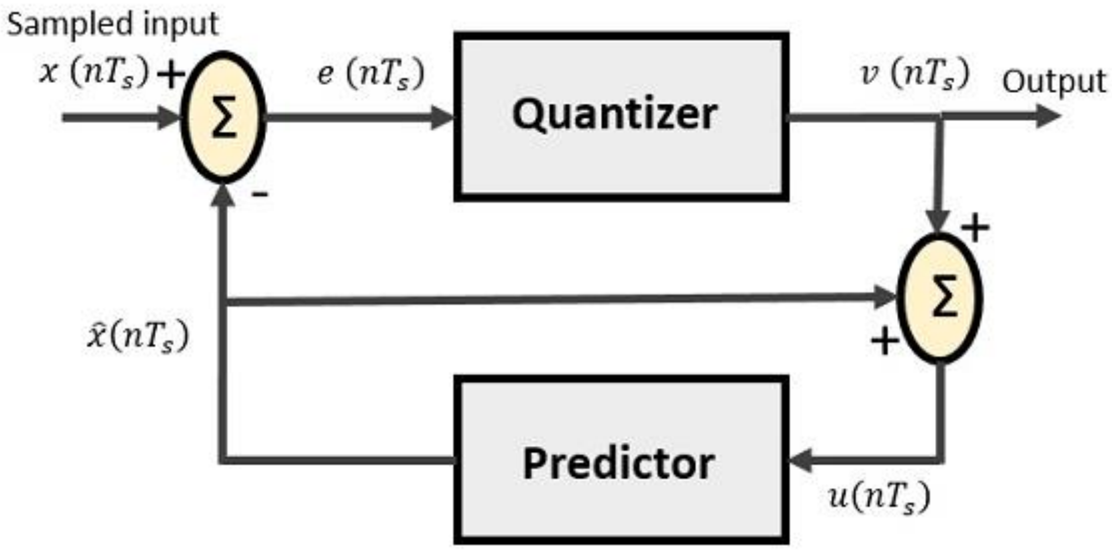
The quantization noise depends on the step size. At low signal levels, the noise can be kept low by keeping the step size small. The step size should be varied according to the signal level to keep the signal to noise ratio at the required value. The effect of adaptive step size is achieved through companding.

The signal is amplified at low signal levels and attenuated at high signal level. . At receiving end a reverse process is carried out. Thus the compression of the signal at the transmitting end and compression of he signal at the receiving end is Known as companding.

DPCM(Differential Pulse Code Modulation)

The samples of a signal are highly correlated with each other.
When these samples are encoded by standard PCM system the resulting encoded signal contains some redundant information.
If instead of transmitting the absolute value of each sample , the difference between two samples is transmitted then the no. of bits required for each sample will be reduced.
This is what is known as differential Pulse Code Modulation.

DPCM(Differential Pulse Code Modulation)



DPCM works on the principle of prediction. The value of the present sample is predicted from the past sample.

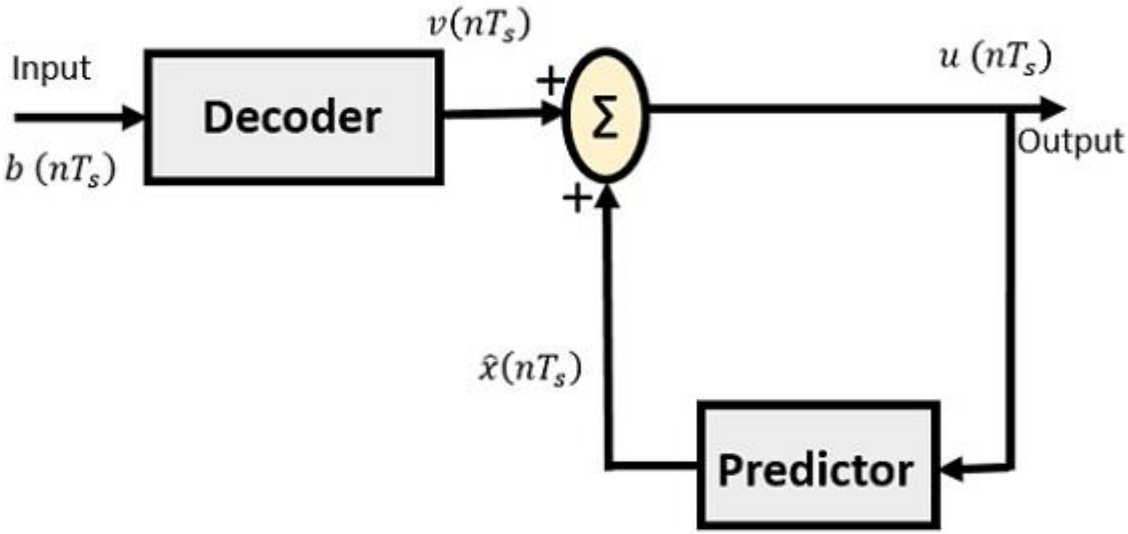
The sampled input is denoted by $x(nTs)$

the predicted sampled value is denoted by $\hat{x}(nTs)$.

The comparator finds the difference between actual sample value and predicted sample value. The predicted value is produced by predictor. The quantized output and previous predicted value is given to predictor to make the prediction more closer to actual sampled value.

DPCM Receiver

The block diagram of DPCM Receiver consists of a decoder, a predictor, and a summer circuit



Decoder reconstructs the quantized error signal from incoming binary signal.

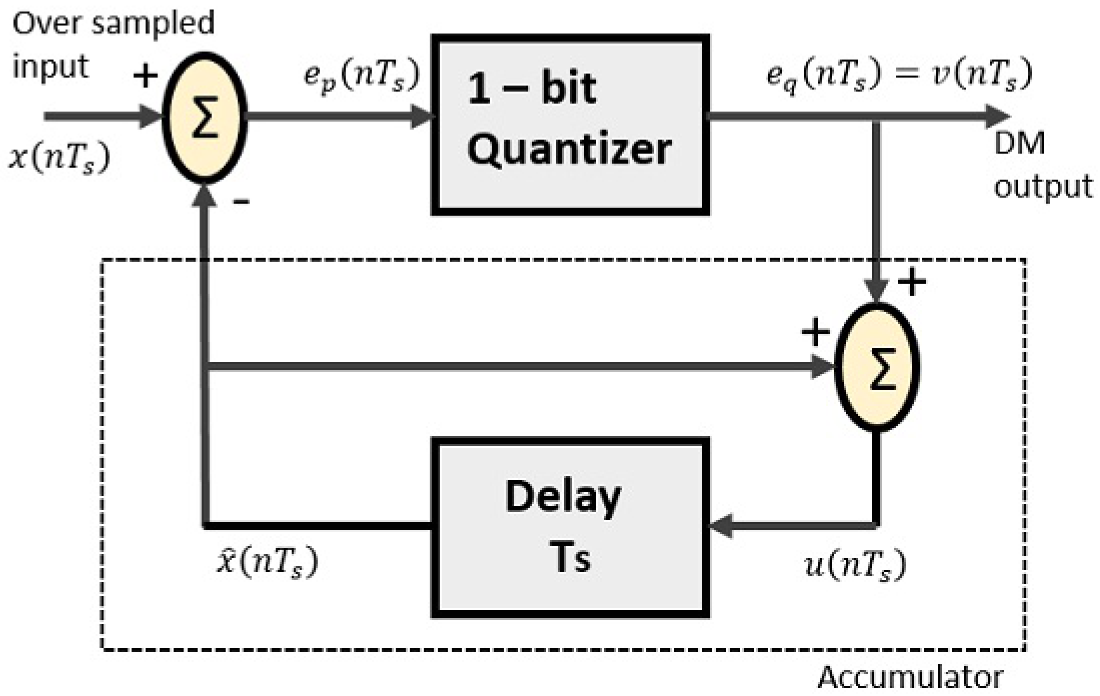
Delta Modulation

In PCM, transmission channel bandwidth is quite large. So delta modulation is preferred.
Delta modulation transmits only one bit per sample. The present sample value is compared with the previous value. Only one bit per sample is transmitted depending on whether current sample is larger or smaller than previous sample.

Advantages.
1. Transmission channel bandwidth is small.
2. The transmitter and receiver implementation is simple. ADC is not required.

Disadvantages
1. Slope overload Distortion
2. Granular Noise.

Delta Modulator: The Delta Modulator comprises of a 1-bit quantizer and a delay circuit along with two summer circuits. The predictor circuit in DPCM is replaced by a simple delay circuit in DM.



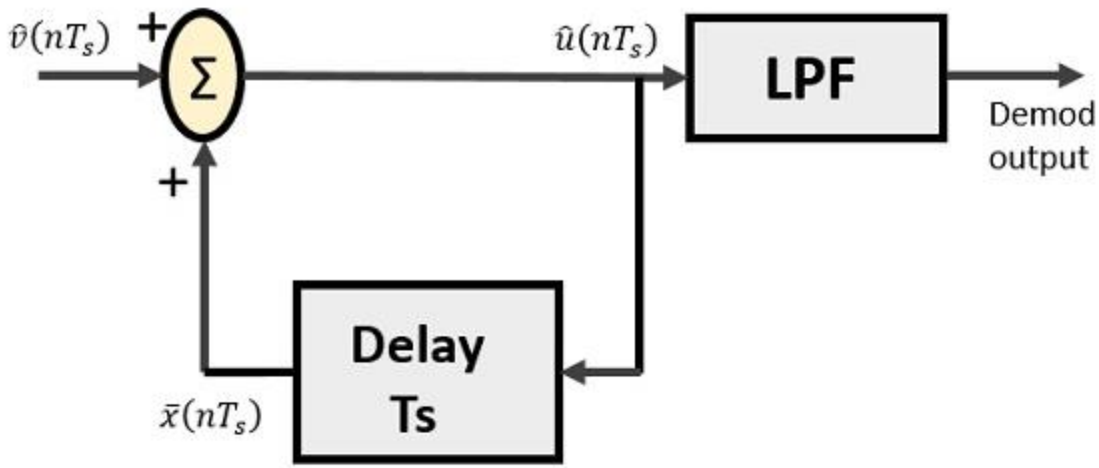
x(nTs) is over sampled input
ep(nTs) is summer output and quantizer input
eq(nTs) is quantizer output
x^(nTs) is output of delay circuit
u(nTs) is input of delay circuit

The present input of the delay unit
= The previous output of the delay unitT + the present quantizer output

$u(nTs)=u([n-1]Ts)+v(nTs)u(nTs)=u([n-1]Ts)+v(nTs)$

Delta Demodulator

The delta demodulator consists of a low pass filter, a summer, and a delay circuit. The predictor circuit is eliminated and hence no assumed input is given to the demodulator.



$\hat{v}(nTs)$ is the input sample

$\hat{u}(nTs)$ is the summer output

$\bar{x}(nTs)$ is the delayed output

A binary sequence will be given as an input to the demodulator. The stair-case approximated output is given to the LPF.

Slope OverLoad Detection.

This distortion arises due to large dynamic range of the input signal. The is a large error between the staircase approximate signal and the actual signal. This error is known as slope overload distortion.
To reduce this error the step size must be increased when the slope of the signal is high.
Delta modulation has fixed step size.

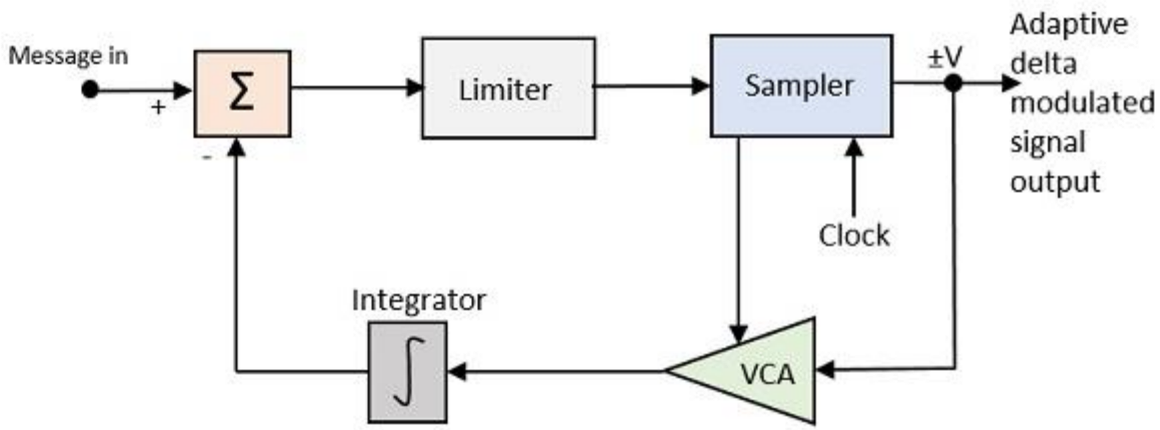
Granular Noise.

Granular Noise occurs when the step size is too large compared to small variations in the input signal. For small variations in the input signal the staircase signal changes by large amount. The solution is to make the step size small.

Adaptive Delta Modulation

To overcome the slope overload distortion and granular noise the step size should be made adaptive to the variations in the input signal. If the variations in the input signal are small the step size should be small. And input varies over a large dynamic range step size should be large. This is what done in Adaptive delta Modulation

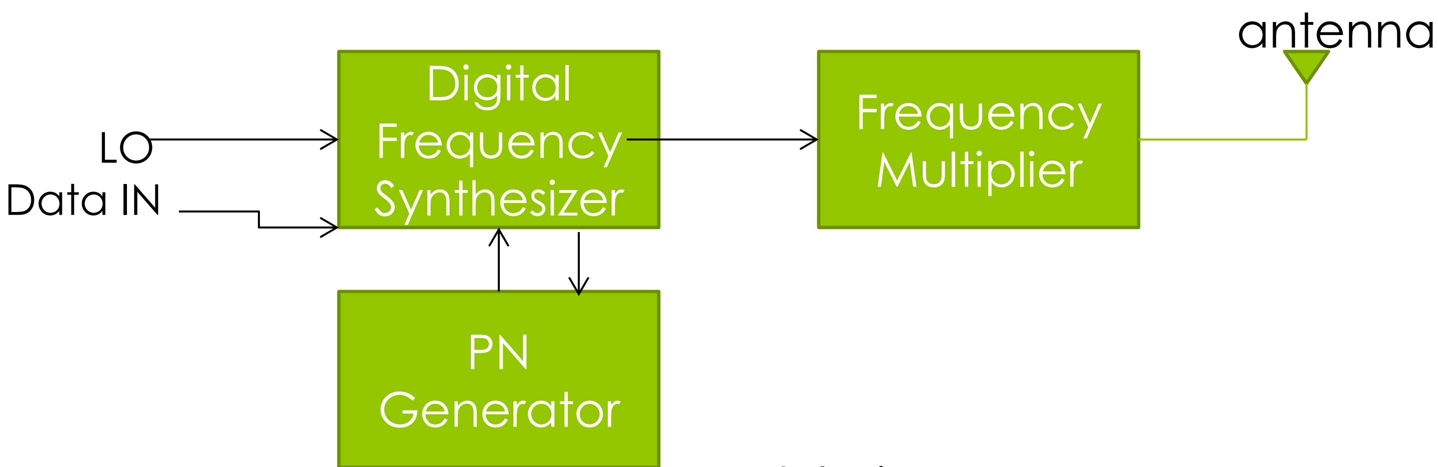
Block diagram of adaptive Delta modulator



Adaptive delta modulator

Frequency Hopping Spread Spectrum Technique

Frequency-hopping spread spectrum (**FHSS**) is a method of transmitting signals by rapidly changing the carrier frequency among many distinct frequencies occupying a large spectral band. Each bit or subinteger is transmitted on a different carrier which is selected from a range of frequencies.



Frequency Hopping Modulation

The selection of frequency is controlled by PN sequence

Chapter 3

Digital Modulation Techniques

Objectives

To understand the Basic block diagram and principle of working of the following:

Amplitude shift keying (ASK)

Frequency Shift keying (FSK)

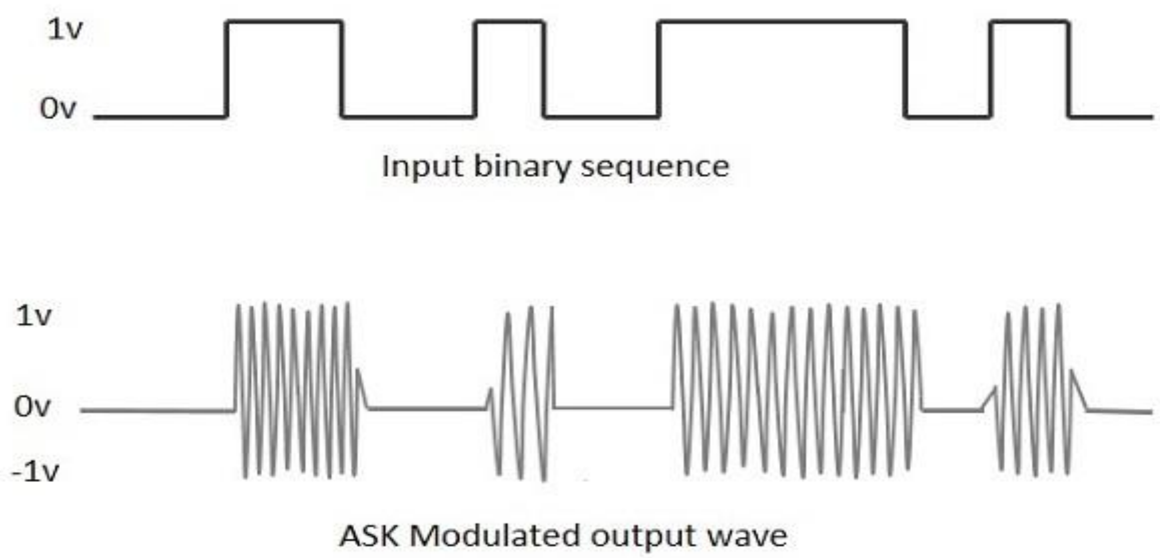
Phase shift keying (PSK)

Quadrature Phase Shift Keying(QPSK)

ASK(Amplitude shift Keying)

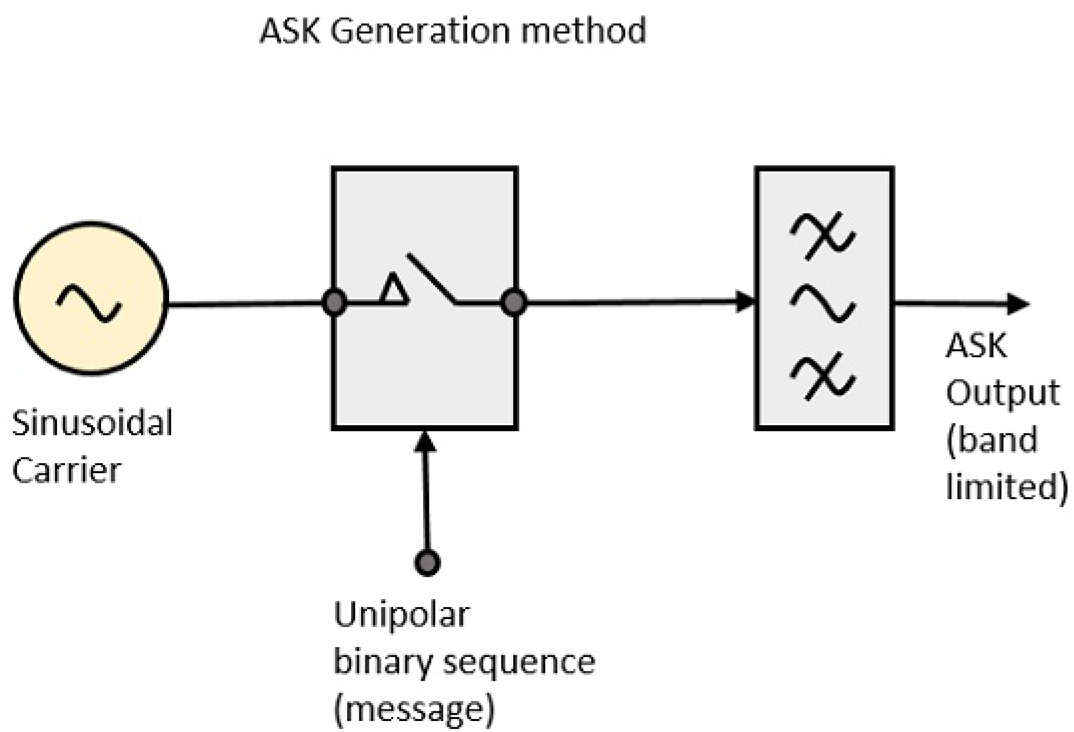
ASK is a type of Amplitude Modulation which represents the binary data in the form of variations in the amplitude of high frequency carrier signal. A ‘1’ bit may be presented by the presence of carrier and a ‘0; bit may be represented by the zero amplitude of carrier ..

ASK(Amplitude shift Keying)



The ASK modulator block diagram comprises of the carrier signal generator, the binary sequence from the message signal and the ba

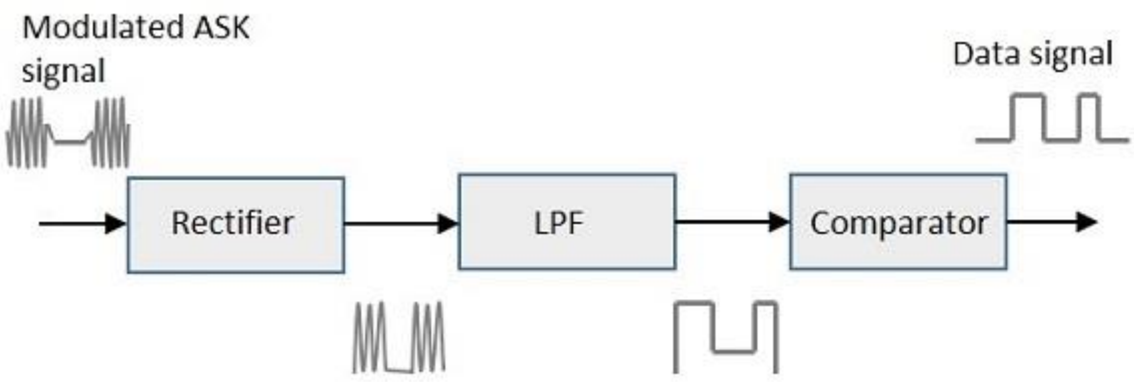
ASK Modulator



Carrier generator produces a high frequency sinusoidal carrier wave.
The message signal is a binary sequence of 0's and 1's.
This unipolar binary sequence makes the switch on or off depending on whether the input is HIGH or Low. When the switch is closed the carrier is allowed. When the input is Low ,the switch is open and carrier is not allowed.
Aband limiting filter is generally used at the output for shaping of pulses.

ASK Detector

ASK detector consists of a half-wave rectifier, a low pass filter, and a comparator. Following is the block diagram for ASK Detector



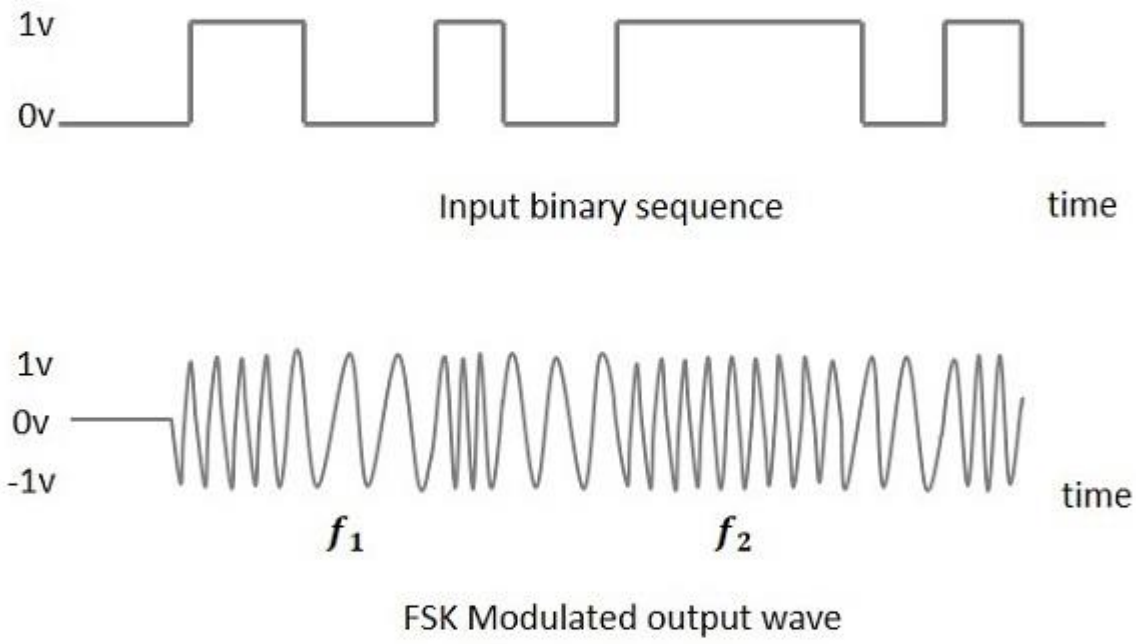
Asynchronous ASK detector

FSK(Frequency Shift Keying)

FSK is the digital modulation technique in which the frequency of the carrier signal varies according to the digital signal. FSK is a form of frequency modulation.

The output of a FSK modulator is a wave that is high in frequency for a High input and is low in frequency for a Low input.

Frequency Shift Keying



PSK(Phase shift Keying)

PSK is the digital modulation technique in which the phase of the carrier signal is changed depending on the binary input by giving the sine or cosine outputs.

Applications.

PSK technique is widely used for wireless LANs, bio-metric, contactless operations, along with RFID and Bluetooth communications.

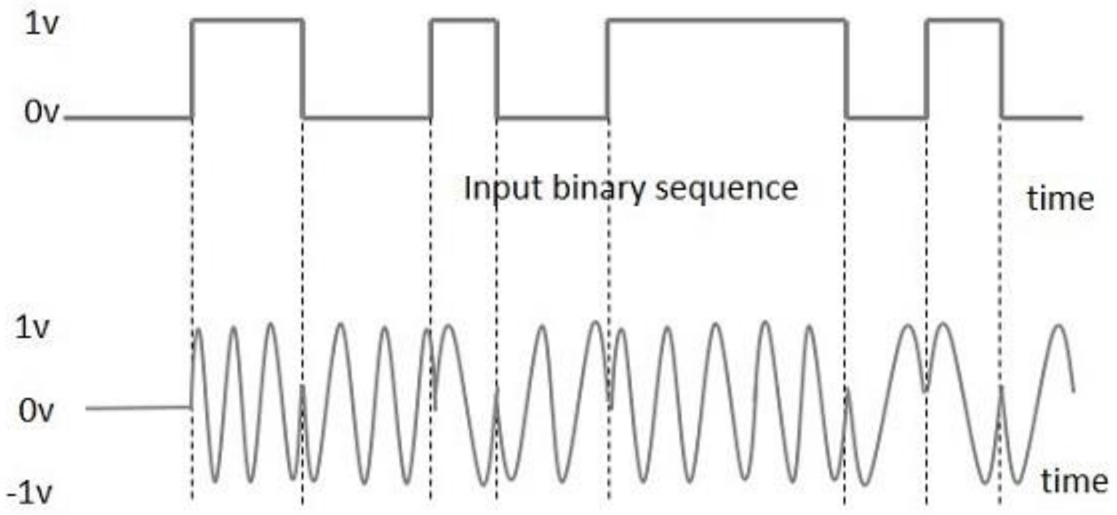
TYPES of PSK

BPSK(Binary Phase shift Keying)

As the name indicates, the output carrier signal makes one of the two phase reversals 0 degree or 180 degree depending on whether the modulating signal is HIGH or LOW.

QPSK(Quadrature Phase Shift Keying)
The output carrier signal makes one of the four phase reversals- .In QPSK two successive bits in the data are grouped together. Thus the possible combinations of two bits are 10,00,01,11.Corresponding to these four combinations, there are four phase shifts in the carrier.

BPSK(Binary Phase shift Keying)

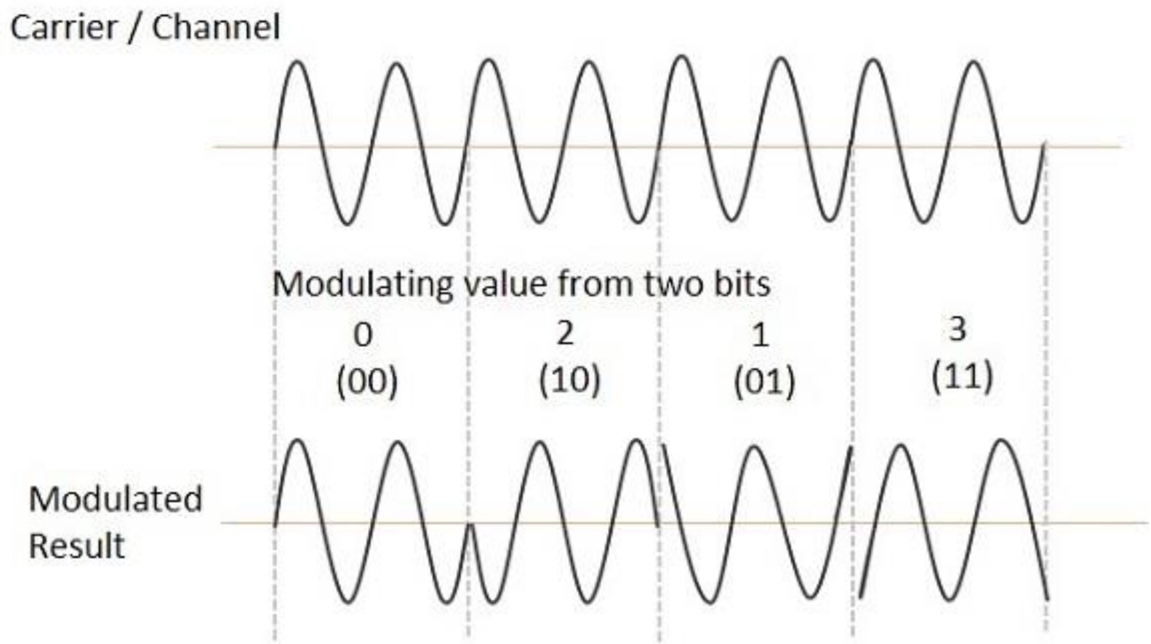


BPSK Modulated output wave

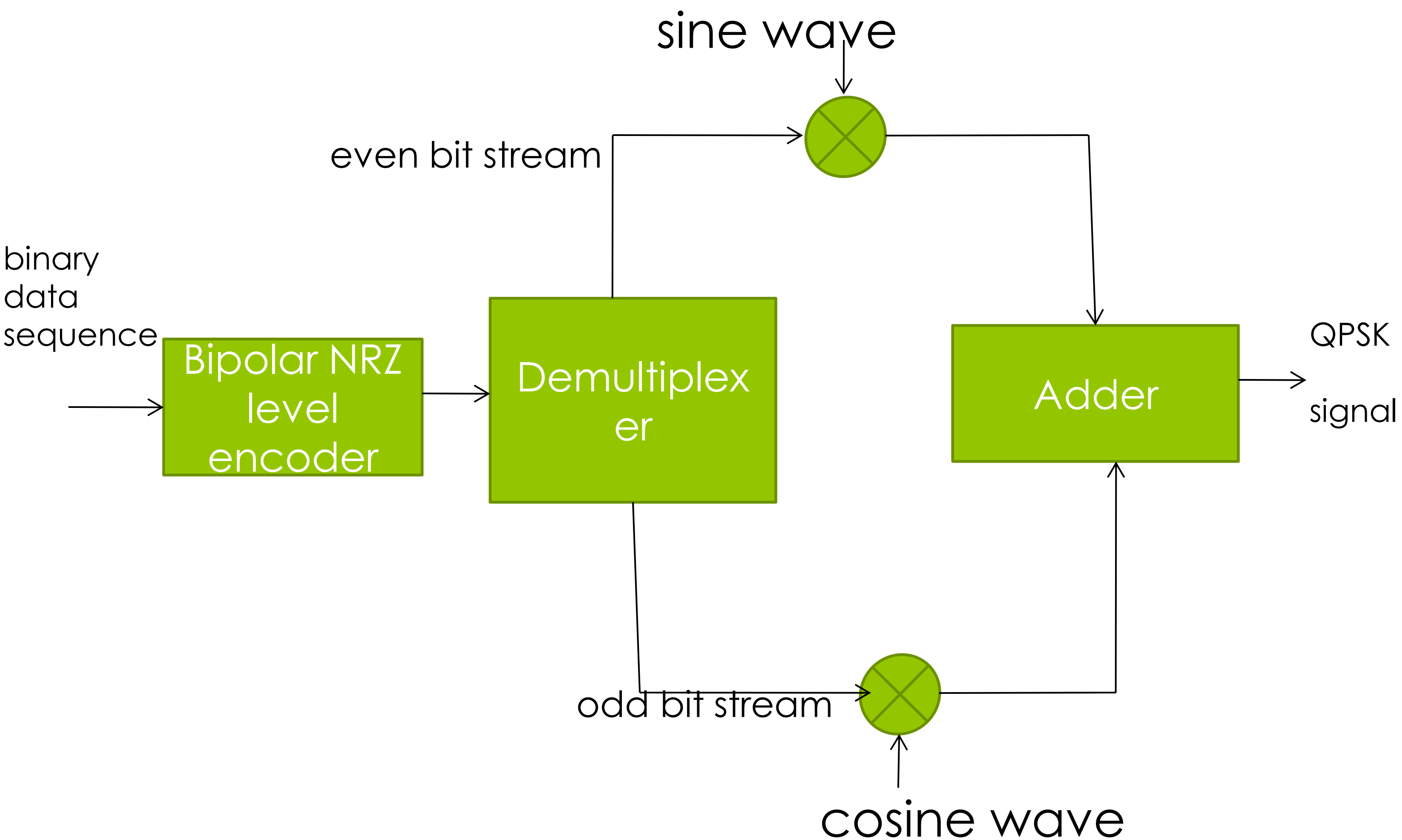
QPSK(Quadrature Phase shift Keying)

The two main resources of comm. System are transmission power and channel bandwidth.
The channel bandwidth depends on the bit rate.. If two or more bits are combined in some symbols the signalling rate can be reduced. This is what is done in QPSK. Because of grouping of the bits ,the channel bandwidth is reduced.. In QPSK, two successive bits are grouped together and represented by a particular phase reversal. Since there can be four possible combination of two bits, there are four different phase reversals of carrier.

QPSK(Quadrature Phase shift Keying)



Generation of QPSK



Generation of QPSK:

The input binary sequence is first converted to a bipolar NRZ type of signal. This signal is denoted by b(t). The de-multiplexer divides b(t) into two separate bit streams of odd numbered bits and even numbered bits..The even bit stream $b_e(t)$ modulates cos wave and odd bit stream modulates sine wave. The two carriers are known as quadrature carriers.

Chapter 4 Characteristics/working of data transmission circuits

Objectives

To study bandwidth requirements,
 To study data transmission speeds,
 To study noise, cross talk,
To study echo suppressors,
To study distortion ,equalizers

Bandwidth Requirements

The data stream is similar to a square-wave signal with rapid transitions from one voltage level to another, with the repetition rate depending on the binary representation of the data word. For instance, if an 8-bit word has the value 01010101, the resulting voltage graph would appear as a series of four square waves with each negative half-cycle equal to each positive half-cycle. If, however, the data word has the form 00001111, the voltage graph would appear as a single square wave with negative and positive half-cycles equal but longer than the first example. Figure 14-16 shows the voltage graphs for these and other binary words. It can be seen that data circuits must provide a bandwidth for the data transmissions they carry. This will be governed by the pulse rate variations and by the fact that even a single square wave occupies a frequency range because of the harmonics present.

Shannon proved that over a channel of bandwith *B* ,

the rate of information transmission, *C*,

in bits/s (binary digits per second) is given by the

equation below, where SNR is the signal-to-noise power ratio.

$C=B.\log2(1+SNR)$

Data Transmission Speeds

The rate of data transfer depends on several aspects of the transmission channel, of which signaling speed is very important. Transmission speed of a communications channel is also refered as the channel's baud rate. The baud is an important unit of signaling speed. In a system in which all pulses have equal duration, the speed in bauds is equal to the maximum rate at which signal pulses are transmitted. This should be recognized as different from information bit rate. In a system which uses only one information bit per signaling pulse, i.e., a binary system, the baud rate and the bit rate happen to be the same. In systems which encode the data in such a way that more than one information bit can be placed on each signaling pulse, the information bit rate will exceed the baud rate.

Crosstalk – Any transmission system which conveys more than one signal simultaneously can experience crosstalk, which is interference due to the reception of portions of a signal from one channel in another channel. This is common in multiplexed systems in which inadequate procedures are employed to ensure that over-modulation of the various carriers of the multiplexed groups is prevented. In modem transmission systems which convey many channels of voice and data simultaneously, the systems will become "loaded," or heavily utilized, so that the control of levels of the individual channels and the group levels becomes very important in order to preclude crosstalk. Data transmission engineers have developed specific level-setting parameters to ensure that as the circuit loading increases, crosstalk will not become a problem.

Echo suppressors – echo suppressors are used on long-distance circuits, in an effort to overcome echoes caused by circuit imbalances. This is significant characteristics of data transmission circuits because a lot of it occurs over the public switched telephone network, nationally and internationally. Although the use of echo suppressors improves voice communications, it is incompatible with characteristics of Data Transmission Circuits. Because a lot of data transmissions are both ways, or quickly alternating from one direction to the other, they require the capability of bidirectional transmission at standard levels, or at least rapid response and interrupt capability. For this type of operation to be accomplished, it is necessary to disable the echo suppressor. In fact, so-called "tone-disable" echo suppressors have been designed to accommodate the needs of data users.

Distortion:

Communication channels tend to react to signals of different frequencies within their bandpass in different ways. propagation velocity varies with frequency, thus some components of one bit position may spill over into another bit position; causing inter-symbol interference, which is a major limitation to maximum bit rate.. Since characters which have lower-frequency components propagate at a different velocity than data characters with high-frequency components, it is possible in higher-speed circuits for portions of one character to enter or remain in the time slot allocated to other characters.

Equalizers – Phase delay distortion can be reduced to acceptable levels by using equalization on the channel.

Equalizer
In digital communications, the equalizer's purpose is to reduce inter symbol interference to allow recovery of the transmitted symbols. It may be a simple linear filter or a complex algorithm.

Digital equalizer types

Linear equalizer: processes the incoming signal with a linear filter

Zero forcing equalizer

Decision feedback equalizer:

Adaptive equalizer

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