Program : **B.Tech**

Subject Name: **Digital Systems**

Subject Code: **CS-304**

Semester: **3rd**

Unit 5

Syllabus:

Introduction to Digital Communication: Nyquist sampling theorem, time division multiplexing, PCM, quantization error, introduction to BPSK & BFSK modulation schemes, Shannon's theorem for channel capacity.

5.1 Introduction to Digital Communication

In digital communication, the information is first converted in to the digital form and then transmitted. First the signal is converted in to electrical form using an input transducer. Then the analog signal obtained is quantized as well as sampled to get the digital signal. The sampling rate and the number of quantization levels thus determine the quality of the digital signal. More the number of samples per seconds as well as more the quantization levels will be there, better the quality of the digital signal. The block diagram of a digital communication system is explained in figure 5.1.1.

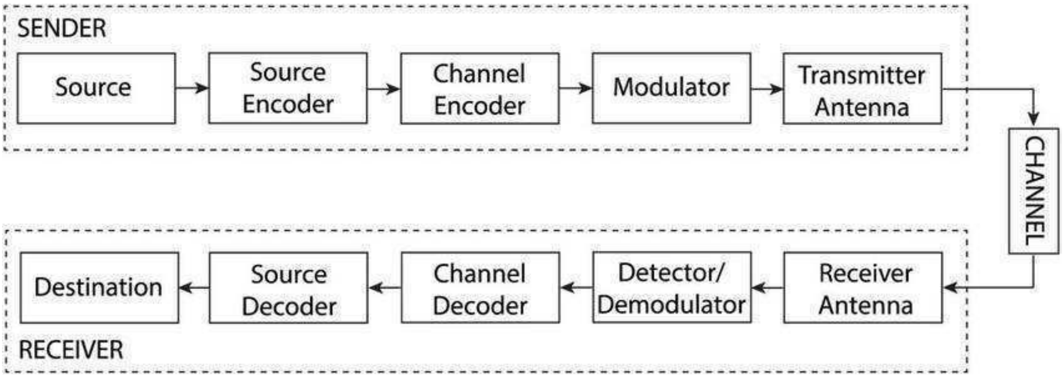


Figure 5.1.1 Digital Communication System

The information generated by the source is coded by a source coder. The aim of the source coding is to represent the given information in minimum number of bits per sample. Then channel coder allows the maximum utilization of the channel, through which the information is to be transmitted. Then by selecting a suitable modulation technique the data is modulated and transmitted using the transmitter antenna or through the media.

At the receiver the received signal is first sent to the demodulator, which demodulates the signal and then data is decoded by the channel decoder and the source decoder. Now the information received is in its digital form which can be extracted after conversion in to its original form.

5.2 Sampling

Sampling is defined as, "The process of measuring the instantaneous values of continuous-time signal in a discrete form."

Sample is a piece of data taken from the whole data which is continuous in the time domain. When a source generates an analog signal and if that has to be digitized, having 1s and 0s i.e., High or Low, the signal has to be discretized in time. This discretization of analog signal is called as Sampling. The following figure indicates a continuous-time signal x (t) and a sampled signal x_s (t). When x (t) is multiplied by a periodic impulse train, the sampled signal x_s (t) is obtained.

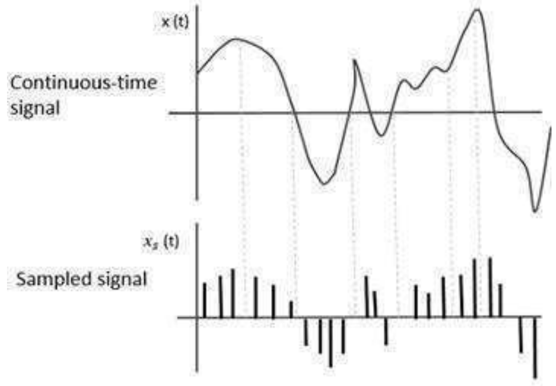


Figure 5.2.1 Sampling

Sampling Rate

To discretize the signals, the gap between the samples should be fixed. That gap can be termed as a sampling period Ts.

$$Sampling\ Frequency = 1/Ts = fs$$

Where,

$$Ts = \ sampling\ time$$
$$fs = \ sampling\ frequency\ or\ the\ sampling\ rate$$

Sampling frequency is the reciprocal of the sampling period. This sampling frequency can be simply called as Sampling rate. The sampling rate denotes the number of samples taken per second, or for a finite set of values.

Nyquist Rate

For an analog signal to be reconstructed from the digitized signal, the sampling rate should be highly considered. The rate of sampling should be such that the data in the message signal should neither be lost nor it should get over-lapped. Hence, a rate was fixed for this, called as Nyquist rate.

Suppose that a signal is band-limited with no frequency components higher than W Hertz. That means, W is the highest frequency. For such a signal, for effective reproduction of the original signal, the sampling rate should be twice the highest frequency.

Which means, $fS = 2W$

Where,

$$f_S = \ sampling\ rate$$
$$W = \ highest\ frequency$$

This rate of sampling is called as Nyquist rate. A theorem called, Sampling Theorem, was stated on the theory of this Nyquist rate.

5.2.1 Nyquist Sampling Theorem

The sampling theorem, i.e. Nyquist theorem, states that, "a signal can be recovered exactly from its samples if the sampling rate f_s is greater or equals to the twice the maximum frequency W."

$$f_s \geq 2W$$

Proof: Consider a continuous time signal x(t). The spectrum of x(t) is a band limited to f_m Hz i.e. the spectrum of x(t) is zero for $|\omega|>\omega_m$.

Sampling of input signal x(t) can be obtained by multiplying x(t) with an impulse train δ(t) of period T_s. The output of multiplier is a discrete signal called sampled signal which is represented with y(t) in the following diagrams:

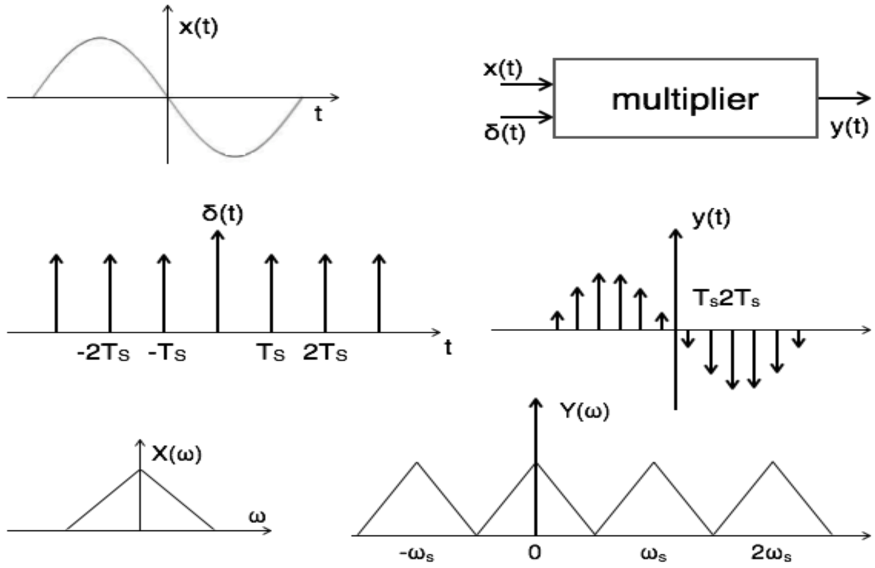


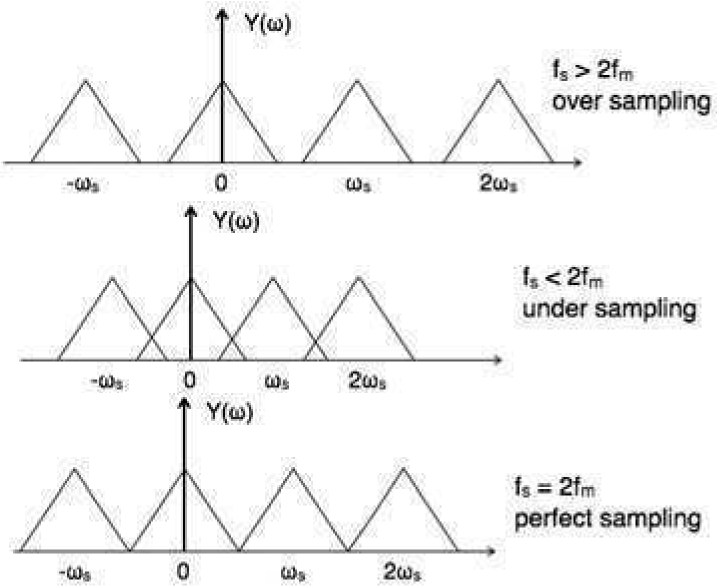
Figure 5.2.2 Sampling of signal x(t)

Here, you can observe that the sampled signal takes the period of impulse. Possibility of sampled frequency spectrum with different conditions is given by the following diagrams:

(a) Over Sampling

(b) Under Sampling

(c) Perfect sampling



Therefore for proper reproduction of the signal from its samples, the sampling frequency must be atleast twice of the maximum frequency of the message signal.

5.2.2 Signals Sampling Techniques

There are three types of sampling techniques:

- Impulse sampling.
- Natural sampling.
- Flat Top sampling.

(a) Impulse Sampling

Impulse sampling can be performed by multiplying input signal x(t) with impulse train $\Sigma \infty n = -\infty \delta(t - nT)$ of period 'T'. Here, the amplitude of impulse changes with respect to amplitude of input signal x(t). The output of sampler is given by

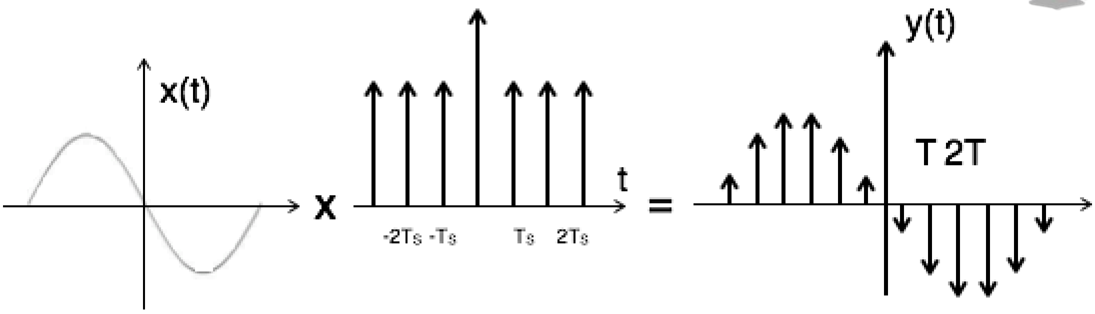


Figure 5.2.2.1 Impulse Sampling

This is called ideal sampling or impulse sampling. You cannot use this practically because pulse width cannot be zero and the generation of impulse train is not possible practically.

(b) Natural Sampling

Natural sampling is similar to impulse sampling, except the impulse train is replaced by pulse train of period T. i.e. you multiply input signal x(t) to pulse train $\sum_{n=-\infty}^{\infty} P(t-nT)$ as shown below

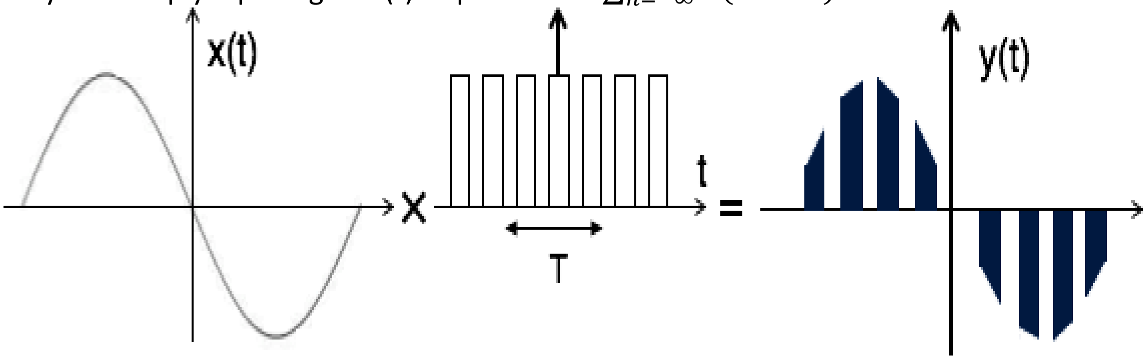


Figure 5.2.2.2 Natural Sampling

(c) Flat Top Sampling

During transmission, noise is introduced at top of the transmission pulse which can be easily removed if the pulse is in the form of flat top. Here, the top of the samples are flat i.e. they have constant amplitude. Hence, it is called as flat top sampling or practical sampling. Flat top sampling makes use of sample and hold circuit.

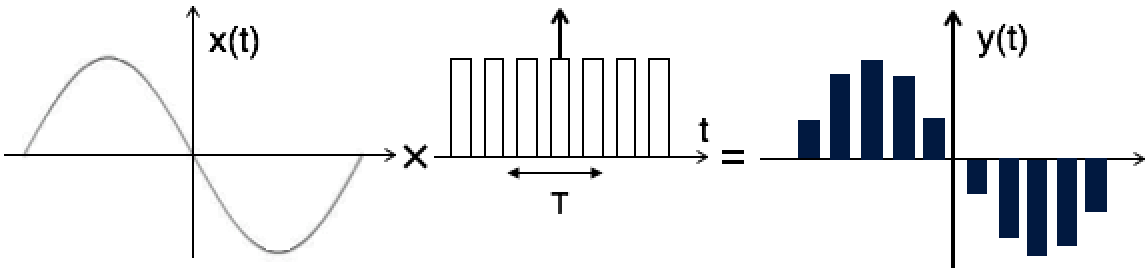


Figure 5.2.2.3 Flat Top Sampling

5.3 Time Division Multiplexing

The sampling theorem allows us to multiplex the samples. That is, the transmission of the message samples engages the communication channel for only a fraction of the sampling interval, and for the rest of the interval transmitting the samples of the other signals. We thereby obtain a time-division multiplex (TDM) system, which enables the joint utilization of a common communication channel by a number of independent message sources without mutual interference among them. The concept of TDM is illustrated by the block diagram shown in Fig. 5.3.1. Each input message signal is first restricted in bandwidth by a low-pass anti-aliasing filter to remove the frequencies that are nonessential to an adequate signal representation. The low-pass filter outputs are then applied to a commutator, which is usually, implemented using electronic switching circuitry. The function of the commutator is twofold: (1) to take a narrow sample of each of the N input messages at a rate that is slightly higher than Nyquist rate 2W, where W is the cutoff frequency of the anti-aliasing filter, and (2) to sequentially interleave these N samples inside the sampling interval Indeed, this latter function is the essence of the time-division multiplexing operation.

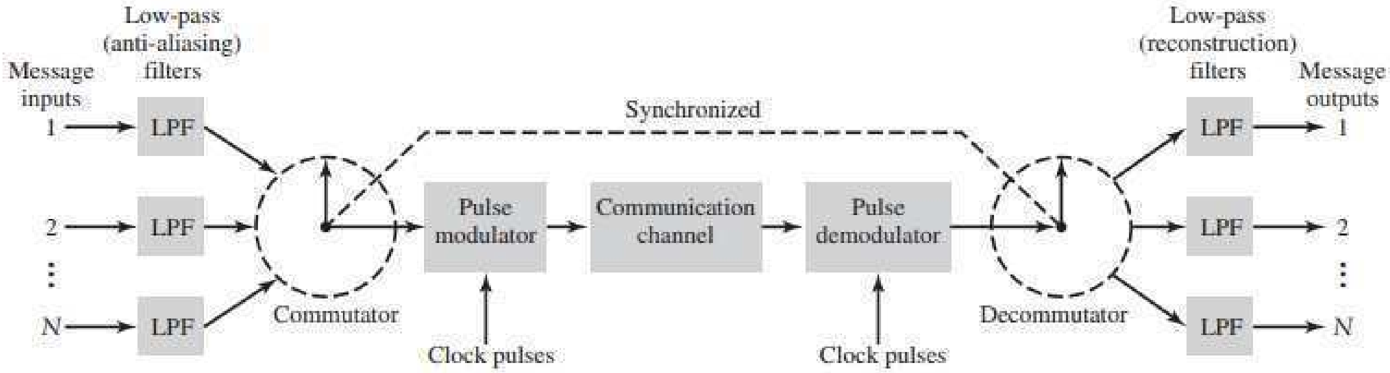


Figure 5.3.1 Time Division Multiplexing

Following the commutation process, the multiplexed signal is applied to a pulse modulator, the purpose of which is to transform the multiplexed signal into a form suitable for transmission over the common channel. It is clear that the use of TDM introduces a bandwidth expansion factor N, because the scheme must squeeze N samples derived from N independent message sources into a time slot equal to one sampling interval. At the receiving end of the system, the received signal is applied to a pulse demodulator, which performs the reverse operation of the pulse modulator. The narrow samples are distributed to the appropriate low-pass reconstruction filters by means of a decommutator, which operates in synchronism with the commutator in the transmitter. This synchronization is essential for a satisfactory operation of the system. Thus the samples are separated at the receiver section.

5.4 Pulse Code Modulation:

A signal which is to be quantized before transmission is sampled as well. The quantization is used to reduce the effect of noise and the sampling allows us to do the time division multiplexing. The combined operation of sampling and quantization generate a quantized PAM waveform i.e. a train of pulses whose amplitude is restricted to a number of discrete levels.

Rather than transmitting the sampled values itself, we may represent each quantization level by a code number and transmit the code number. Most frequently the code number is converted in to binary equivalent before transmission. Then the digits of the binary representation of the code are transmitted as pulses. This system of transmission is called binary **Pulse Code Modulation**. The whole process can be understood by the figure 5.4.1.

PCM Transmitter:

Basic Blocks:

1. Anti aliasing Filter, 2. Sampler, 3. Quantizer, 4. Encoder

An anti-aliasing filter is basically a filter used to ensure that the input signal to sampler is free from the unwanted frequency components. For most of the applications these are low-pass filters. It removes the frequency components of the signal which are above the cutoff frequency of the filter. The cutoff frequency of the filter is chosen such it is very close to the highest frequency component of the signal.

Sampler unit samples the input signal and these samples are then fed to the Quantizer which outputs the quantized values for each of the samples. The quantizer output is fed to an encoder which generates the binary code for every sample.

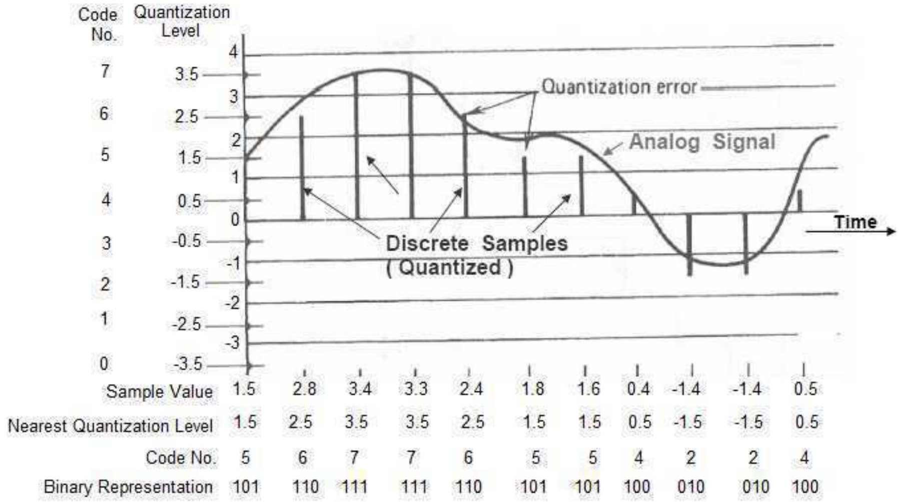


Figure 5.4.1 Pulse Code Modulation

The quantizer and encoder together are called as analog to digital converter.

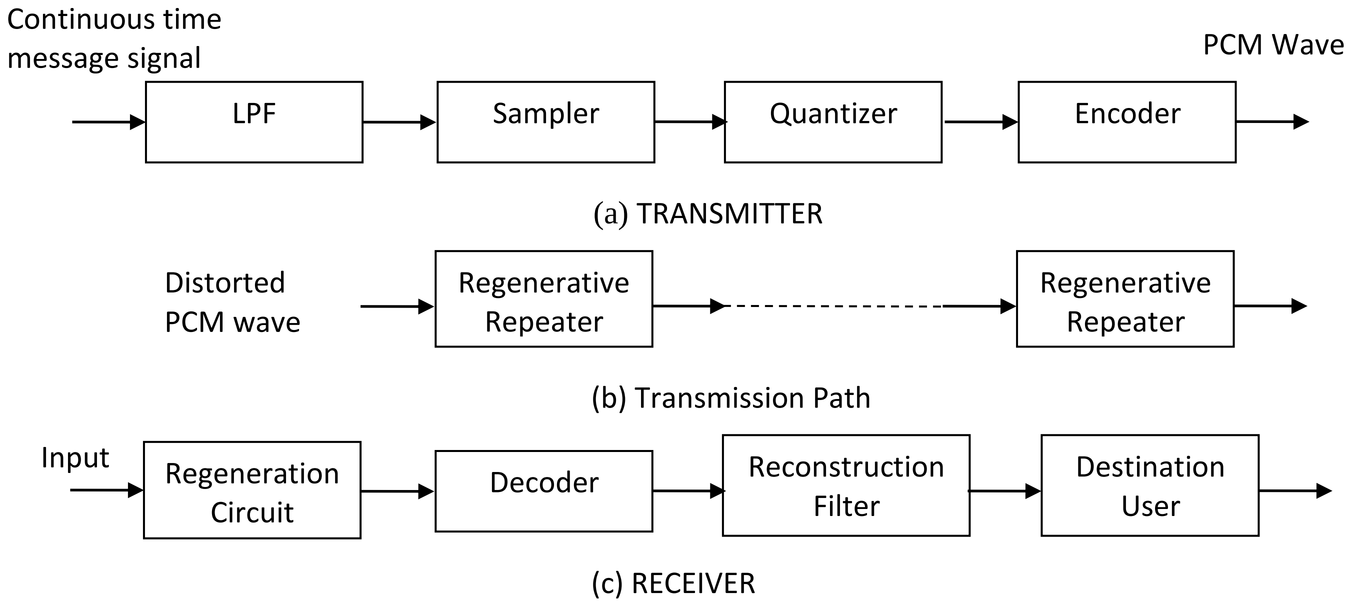


Figure 5.4.2 PCM System Basic Block Diagram

Advantages of Pulse Code Modulation:

- Pulse code modulation will have low noise addition and data loss is also very low.
- Pulse code modulation is used in music play back CD's and also used in DVD for data storing whose sampling rate is bit higher.
- Pulse code modulation can be used in storing the data.
- PCM can encode the data also.
- Multiplexing of signals can also be done using pulse code modulation. Multiplexing is nothing for adding the different signals and transmitting the signal at same time.
- Pulse code modulation permits the use of pulse regeneration.

Disadvantages:

- Pulse code modulation requires large bandwidth
- Specialized circuitry is required for transmitting and also for quantizing the samples at same quantized levels.
- We can do encoding using pulse code modulation but we need to have complex and special circuitry.
- Pulse code modulation receivers are cost effective when we compared to other modulation receivers.

- Developing pulse code modulation is bit complicated and checking the transmission quality is also difficult and takes more time.
- Channel bandwidth should be more for digital encoding.
- PCM systems are complicated when compared to analog modulation methods and other systems.
- Decoding also needs special equipment's and they are also too complex.

Applications of Pulse Code Modulation (PCM):

- Pulse code modulation is used in telecommunication systems, air traffic control systems etc.
- Pulse code modulation is used in compressing the data that is why it is used in storing data in optical disks like DVD, CDs etc. PCM is even used in the database management systems.
- Pulse code modulation is used in mobile phones, normal telephones etc.
- Remote controlled cars, planes, trains use pulse code modulations.

5.5 Quantization Error

Quantization: In the process of quantization we create a new signal $m_q(t)$, which is an approximation to $m(t)$. The quantized signal $m_q(t)$, has the great merit that it is separable from the additive noise.
The operation of quantization is represented in figure 2.7.1. Here we have a signal m(t), whose amplitude varies in the range from V_H to V_L as shown in the figure.
We have divided the total range in to M equal intervals each of size S, called the step size and given by

$$S = \Delta = \frac{(V_H - V_L)}{M}$$

In our example M=8. In the centre of each of this step we located quantization levels $m0, m1, m2, \dots m7$. The $m_q(t)$ is generated in the following manner-
Whenever the signal $m(t)$ is in the range Δ_0, the signal $m_q(t)$ maintains a constant level m_0, whenever the signal $m(t)$ is in the range Δ_1, the signal $m_q(t)$ maintains a constant level m_1 and so on. Hence the signal $m_q(t)$ will found all times to one of the levels $m0, m1, m2, \dots m7$. The transition in $m_q(t)$ from $m0$ to $m1$ is made abruptly when $m(t)$ passes the transition level L_{01}, which is mid way between m_0 and m_1 and so on.

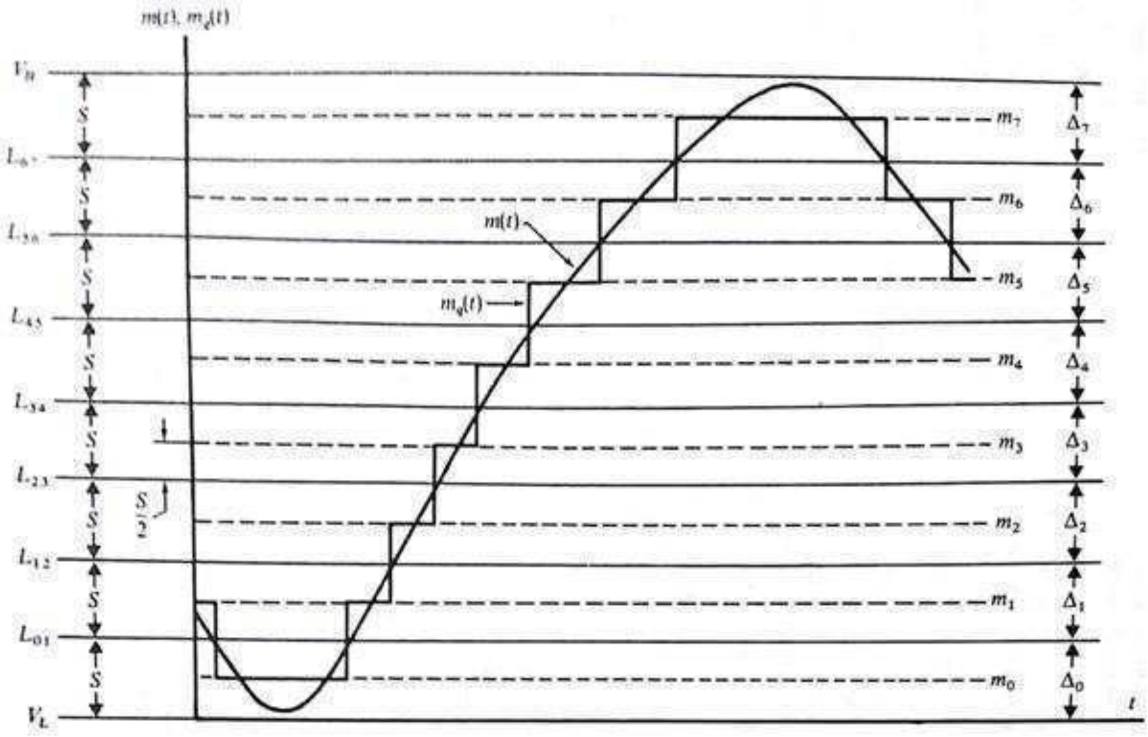


Figure 5.5.1 Quantization of signals

Using quantization of signals, the effect of noise can be reduced significantly. The difference between $m(t)$ and $m_q(t)$ can be regarded as noise and is called quantization noise.

$$quantization\ noise = m(t) - m_q(t)$$

Also the quantized signal and original signal differs from one another in a ransom manner. This difference or error due to quantization process is called quantization error and is given by

$$e = m(t) - m_k$$

when $m(t)$ happens to be close to quantization level m_k, quantizer output will be m_k.

Quantization Error:

For any system, during its functioning, there is always a difference in the values of its input and output. The processing of the system results in an error, which is the difference of those values.

The difference between an input value and its quantized value is called a Quantization Error. A Quantizer is a logarithmic function that performs Quantization (rounding off the value). An analog-to-digital converter (ADC) works as a quantizer.

The following figure illustrates an example for a quantization error, indicating the difference between the original signal and the quantized signal.

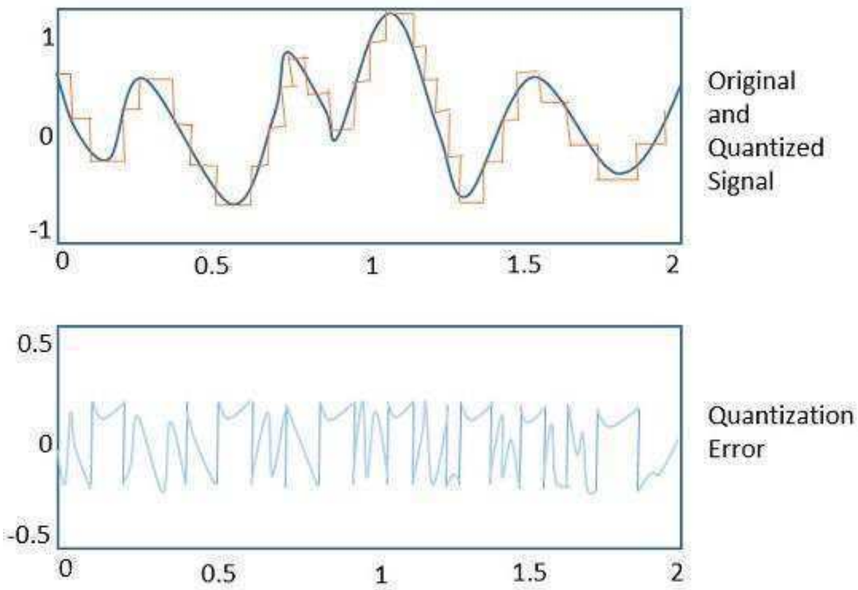


Figure 5.5.2 Quantization Error

Quantization Noise:

It is a type of quantization error, which usually occurs in analog audio signal, while quantizing it to digital. For example, in music, the signals keep changing continuously, where regularity is not found in errors. Such errors create a wideband noise called as Quantization Noise.

5.6 Digital Modulation

It provides more information capacity, high data security, quicker system availability with great quality communication. Hence, digital modulation techniques have a greater demand, for their capacity to convey larger amounts of data than analog modulation techniques.

There are many types of digital modulation techniques and also their combinations, depending upon the need. Of them all, we will discuss the prominent ones.

ASK – Amplitude Shift Keying

The amplitude of the resultant output depends upon the input data whether it should be a zero level or a variation of positive and negative, depending upon the carrier frequency.

FSK – Frequency Shift Keying

The frequency of the output signal will be either high or low, depending upon the input data applied.

PSK – Phase Shift Keying

The phase of the output signal gets shifted depending upon the input. These are mainly of two types, namely Binary Phase Shift Keying (BPSK) and Quadrature Phase Shift Keying (QPSK), according to the number of phase shifts. The other one is Differential Phase Shift Keying (DPSK) which changes the phase according to the previous value.

Amplitude Shift Keying (ASK) is a type of Amplitude Modulation which represents the binary data in the form of variations in the amplitude of a signal.
Any modulated signal has a high frequency carrier. The binary signal when ASK modulated, gives a zero value for Low input while it gives the carrier output for High input.
The following figure represents ASK modulated waveform along with its input.

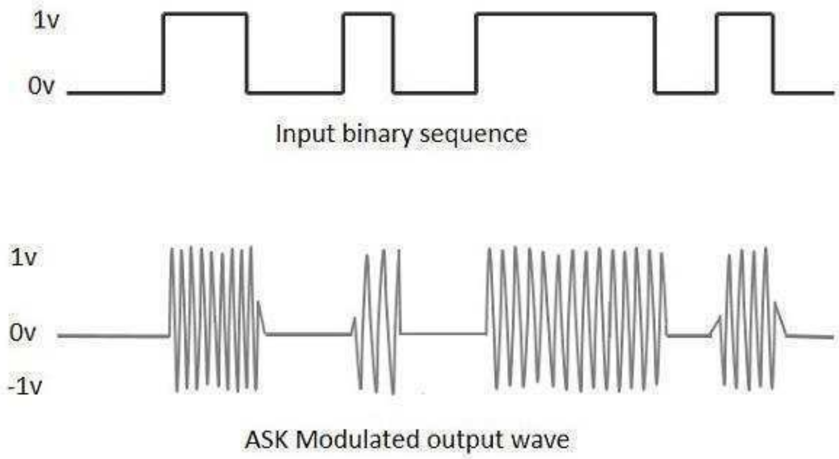


Figure 5.6.1 ASK Modulation

To find the process of obtaining this ASK modulated wave, let us learn about the working of the ASK modulator.

ASK Modulator
The ASK modulator block diagram comprises of the carrier signal generator, the binary sequence from the message signal and the band-limited filter. Following is the block diagram of the ASK Modulator.

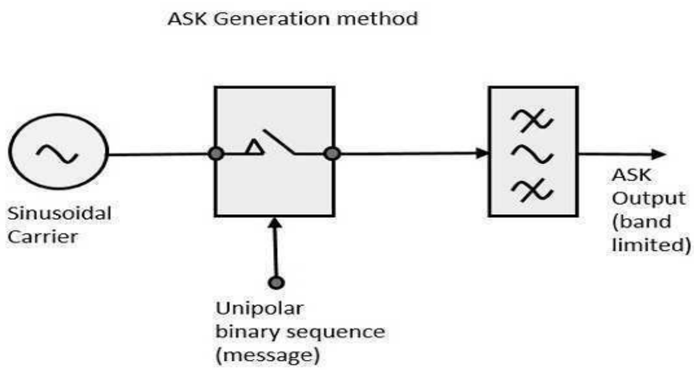


Figure 5.6.2 ASK Modulator

The carrier generator sends a continuous high-frequency carrier. The binary sequence from the message signal makes the unipolar input to be either High or Low. The high signal closes the switch, allowing a carrier wave. Hence, the output will be the carrier signal at high input. When there is low input, the switch opens, allowing no voltage to appear. Hence, the output will be low.
The band-limiting filter, shapes the pulse depending upon the amplitude and phase characteristics of the band-limiting filter or the pulse-shaping filter.

ASK Demodulator
There are two types of ASK Demodulation techniques. They are –

- Asynchronous ASK Demodulation/detection

- Synchronous ASK Demodulation/detection

The clock frequency at the transmitter when matches with the clock frequency at the receiver, it is known as a Synchronous method, as the frequency gets synchronized. Otherwise, it is known as Asynchronous.

Asynchronous ASK Demodulator

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

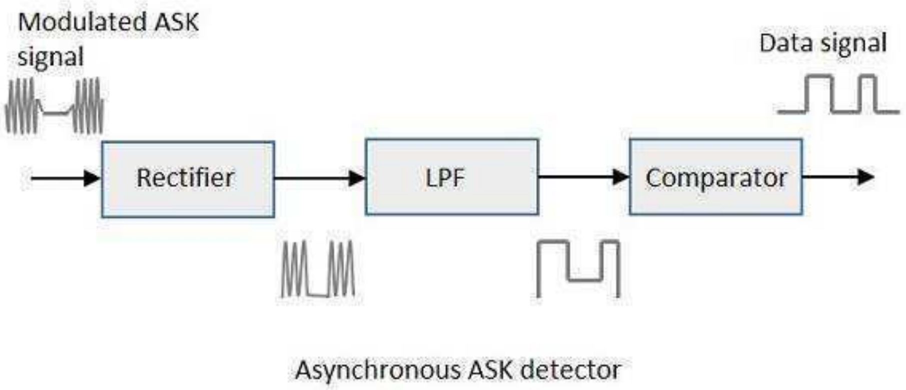


Figure 5.6.3 ASK Demodulator

The modulated ASK signal is given to the half-wave rectifier, which delivers a positive half output. The low pass filter suppresses the higher frequencies and gives an envelope detected output from which the comparator delivers a digital output.

Synchronous ASK Demodulator

Synchronous ASK detector consists of a Square law detector, low pass filter, a comparator, and a voltage limiter. Following is the block diagram for the same.

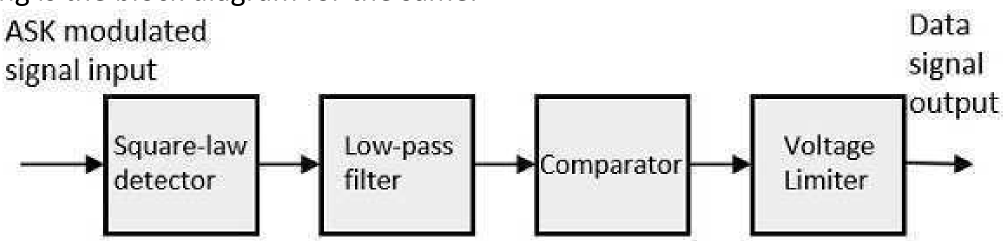


Figure 5.6.4 Synchronous ASK Demodulator

The ASK modulated input signal is given to the Square law detector. A square law detector is one whose output voltage is proportional to the square of the amplitude modulated input voltage. The low pass filter minimizes the higher frequencies. The comparator and the voltage limiter help to get a clean digital output.

5.7 Frequency Shift Keying (FSK)

FSK is the digital modulation technique in which the frequency of the carrier signal varies according to the digital signal changes. FSK is a scheme of frequency modulation. The output of a FSK modulated wave is high in frequency for a binary High input and is low in frequency for a binary Low input. The binary 1s and 0s are called Mark and Space frequencies. The following image is the diagrammatic representation of FSK modulated waveform along with its input.

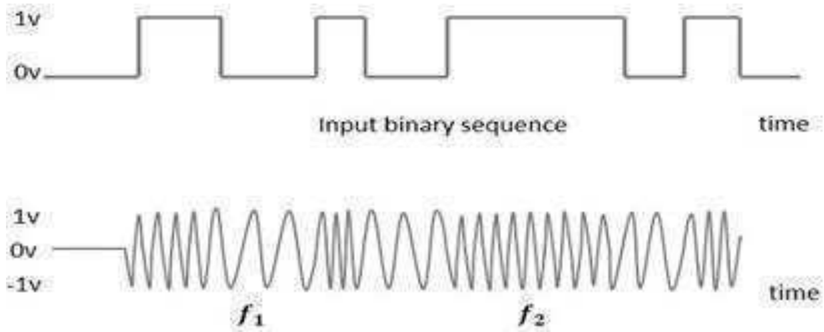


Figure 5.7.1 Frequency Shift Keying (FSK)

To find the process of obtaining this FSK modulated wave, let us know about the working of a FSK modulator.

FSK Modulator

The FSK modulator block diagram comprises of two oscillators with a clock and the input binary sequence. Following is its block diagram.

FSK Transmitter

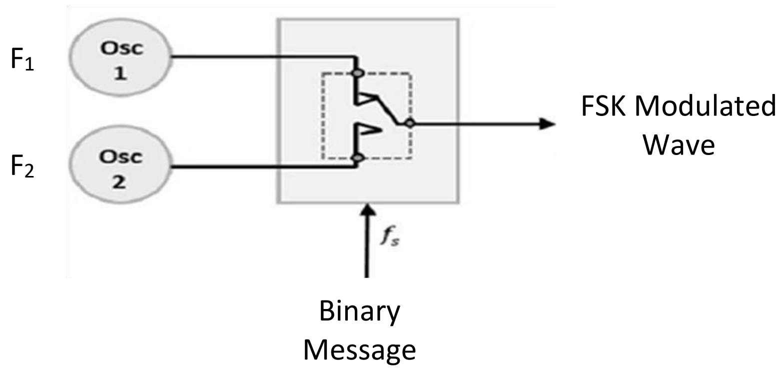


Figure 5.7.2 FSK Modulator

The two oscillators, producing a higher and a lower frequency signals, are connected to a switch along with an internal clock. To avoid the abrupt phase discontinuities of the output waveform during the transmission of the message, a clock is applied to both the oscillators, internally. The binary input sequence is applied to the transmitter so as to choose the frequencies according to the binary input.

FSK Demodulator

There are different methods for demodulating a FSK wave. The main methods of FSK detection are asynchronous detector and synchronous detector. The synchronous detector is a coherent one, while asynchronous detector is a non-coherent one.

Asynchronous FSK Detector

The block diagram of Asynchronous FSK detector consists of two band pass filters, two envelope detectors, and a decision circuit. Following is the diagrammatic representation.

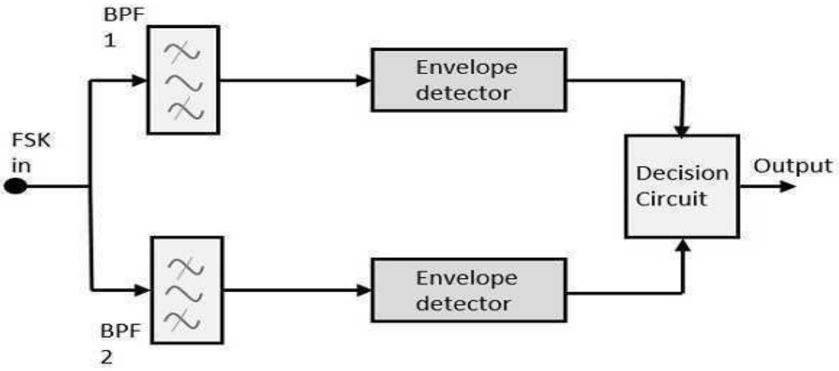


Figure 5.7.3 Asynchronous FSK Detector

The FSK signal is passed through the two Band Pass Filters (BPFs), tuned to Space and Mark frequencies. The output from these two BPFs look like ASK signal, which is given to the envelope detector. The signal in each envelope detector is modulated asynchronously.

The decision circuit chooses which output is more likely and selects it from any one of the envelope detectors. It also re-shapes the waveform to a rectangular one.

Synchronous FSK Detector

The block diagram of Synchronous FSK detector consists of two mixers with local oscillator circuits, two band pass filters and a decision circuit. Following is the diagrammatic representation.

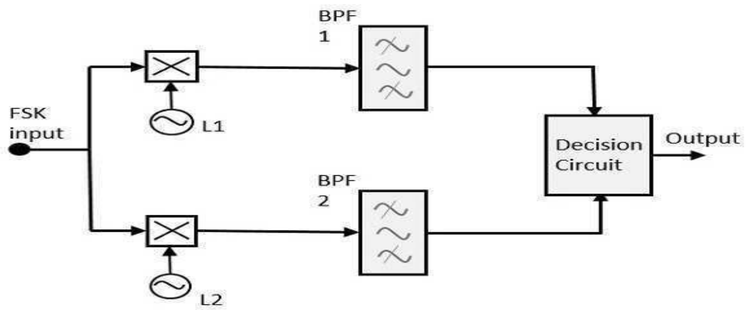


Figure 5.7.4 Synchronous FSK Detector

The FSK signal input is given to the two mixers with local oscillator circuits. These two are connected to two band pass filters. These combinations act as demodulators and the decision circuit chooses which output is more likely and selects it from any one of the detectors. The two signals have a minimum frequency separation.

For both of the demodulators, the bandwidth of each of them depends on their bit rate. This synchronous demodulator is a bit complex than asynchronous type demodulators.

5.8 Phase Shift Keying (PSK)

PSK is the digital modulation technique in which the phase of the carrier signal is changed by varying the sine and cosine inputs at a particular time. PSK technique is widely used for wireless LANs, bio-metric, contactless operations, along with RFID and Bluetooth communications.

PSK is of two types, depending upon the phases the signal gets shifted. They are –

Binary Phase Shift Keying (BPSK)

This is also called as 2-phase PSK or Phase Reversal Keying. In this technique, the sine wave carrier takes two phase reversals such as 0° and 180°.

BPSK is basically a Double Side Band Suppressed Carrier (DSBSC) modulation scheme, for message being the digital information.

BPSK Modulator

The block diagram of Binary Phase Shift Keying consists of the balance modulator which has the carrier sine wave as one input and the binary sequence as the other input. Following is the diagrammatic representation.

BPSK Modulator

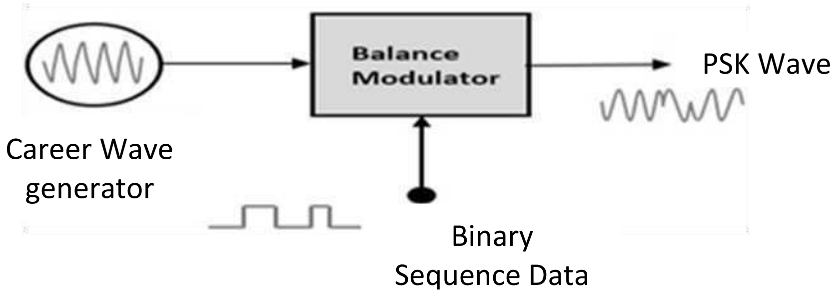


Figure 5.8.1 BPSK Modulator

The modulation of BPSK is done using a balance modulator, which multiplies the two signals applied at the input. For a zero binary input, the phase will be 0° and for a high input, the phase reversal is of 180°. Following is the diagrammatic representation of BPSK Modulated output wave along with its given input.

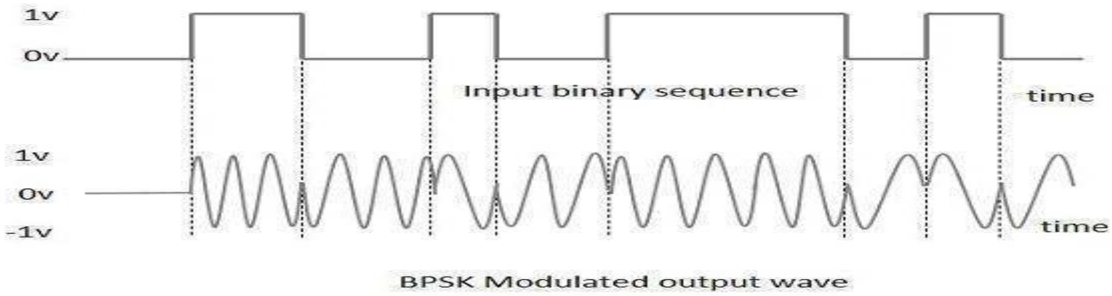


Figure 5.8.2 BPSK Modulated Waveform

The output sine wave of the modulator will be the direct input carrier or the inverted (180° phase shifted) input carrier, which is a function of the data signal.

BPSK Demodulator

The block diagram of BPSK demodulator consists of a mixer with local oscillator circuit, a band pass filter, a two-input detector circuit. The diagram is as follows.

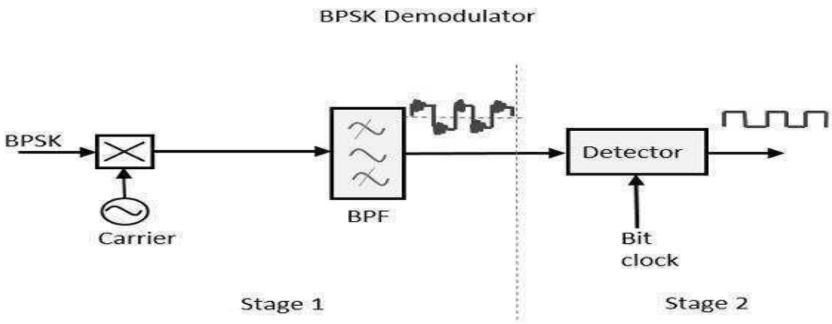


Figure 5.8.3 BPSK Demodulator

By recovering the band-limited message signal, with the help of the mixer circuit and the band pass filter, the first stage of demodulation gets completed. The base band signal which is band limited is obtained and this signal is used to regenerate the binary message bit stream.

In the next stage of demodulation, the bit clock rate is needed at the detector circuit to produce the original binary message signal. If the bit rate is a sub-multiple of the carrier frequency, then the bit clock regeneration is simplified. To make the circuit easily understandable, a decision-making circuit may also be inserted at the 2^{nd} stage of detection.

The Quadrature Phase Shift Keying (QPSK) is a variation of BPSK, and it is also a Double Side Band Suppressed Carrier (DSBSC) modulation scheme, which sends two bits of digital information at a time, called as bigits.

Instead of the conversion of digital bits into a series of digital stream, it converts them into bit pairs. This decreases the data bit rate to half, which allows space for the other users.

5.9 Shannon's Theorem for Channel Capacity:

Channel Capacity

We have so far discussed mutual information. The maximum average mutual information, in an instant of a signaling interval, when transmitted by a discrete memory less channel, the probabilities of the rate of maximum reliable transmission of data, can be understood as the channel capacity.
It is denoted by C and is measured in bits per channel use.

Shannon Limit

Shannon-Hartley equation relates the maximum capacity (transmission bit rate) that can be achieved over a given channel with certain noise characteristics and bandwidth. For an AWGN the maximum capacity is given by

$$C = B\log_2\left(1+\frac{S}{N}\right)$$

Here CC is the maximum capacity of the channel in bits/second otherwise called Shannon's capacity limit for the given channel, BB is the bandwidth of the channel in Hertz, SS is the signal power in Watts and NN is the noise power, also in Watts. The ratio S/N is called Signal to Noise Ratio (SNR). It can be ascertained that the maximum rate at which we can transmit the information without any error, is limited by the bandwidth, the signal level, and the noise level. It tells how many bits can be transmitted per second without errors over a channel of bandwidth B Hz, when the signal power is limited to S Watt and is exposed to Gaussian White Noise of additive nature.

