Module – 2

DIGITAL TRANSMISSION

Digital to Digital Conversion

• Data can be analog or digital, so can be the signal that represents it.

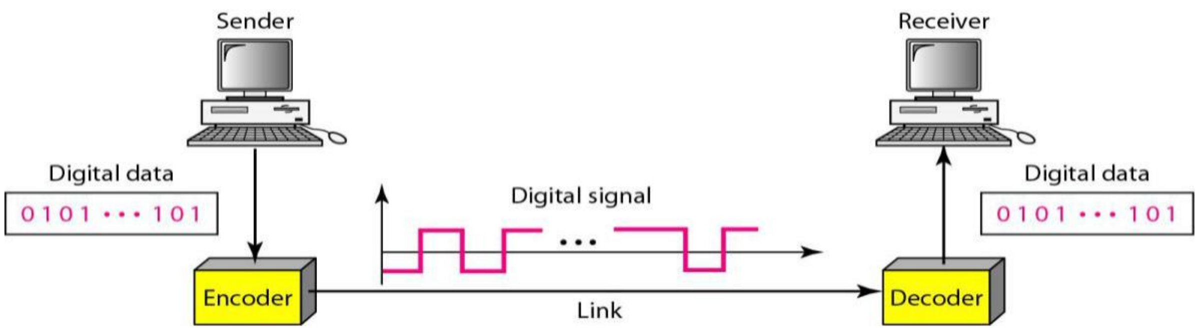
• Signal encoding is the conversion from analog/digital data to analog/digital signal.

• The possible encodings are: 1) Digital data to digital signal 2) Digital data to analog signal 3) Analog data to digital signal 4) Analog data to analog signal

Line Coding

Line coding is the process of converting digital data to digital signals.

Line coding converts a sequence of bits to a digital signal. At the sender, digital data are encoded into a digital signal; at the receiver, the digital data are recreated by decoding the digital signal.



Characteristics

Different characteristics of digital signal are

1) Signal Element Vs Data Element
2) Data Rate Vs Signal Rate
3) Bandwidth
4) Baseline Wandering
5) DC Components
6) Built-in Error Detection
7) Self-synchronization
8) Immunity to Noise and Interference
9) Complexity

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1) Data Element vs. Signal Element

Data Element	Signal Element
A data-element is the smallest entity that can represent a piece of information (Figure 4.2).	A signal-element is shortest unit (timewise) of a digital-signal.
A data-element is the bit.	A signal-element carries data-elements.
Data-elements are being carried.	Signal-elements are the carriers.

- Ratio r is defined as number of data-elements carried by each signal-element.

2) Data Rate vs. Signal Rate

Data Rate	Signal Rate
The data-rate defines the number of data-elements (bits) sent in 1 sec.	The signal-rate is the number of signal-elements sent in 1 sec.
The unit is bits per second (bps).	The unit is the baud.
The data-rate is sometimes called the bit-rate.	The signal-rate is sometimes called the pulse rate, the modulation rate, or the baud rate
Goal in data-communications: increase the data-rate.	Goal in data-communications: decrease the signal-rate.
Increasing the data-rate increases the speed of transmission.	Decreasing the signal-rate decreases the bandwidth requirement.

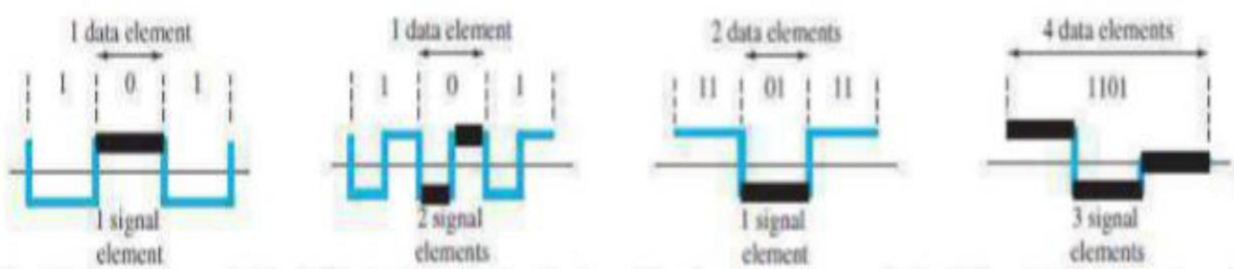
- The relationship between data-rate and signal-rate is given by

$$S_{ave} = c \times N \times (1/r) \quad \text{baud}$$

where N = data-rate (in bps)
c = case factor, which varies for each case S = number of signal-elements and
r = previously defined factor.

- This relationship depends on
  → value of r.
  → data pattern.

(If we have a data pattern of all 1s or all 0s, the signal-rate may be different from a data pattern of alternating 0s and 1s).



a. One data element per one signal element ($r = 1$) b. One data element per two signal elements ($r = \frac{1}{2}$) c. Two data elements per one signal element ($r = 2$) d. Four data elements per three signal elements ($r = \frac{4}{3}$)

Figure 4.2 *Signal element versus data element*

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3) <u>Bandwidth</u>

Digital signal that carries information is non-periodic. The bandwidth of a non-periodic signal is continuous with an infinite range. However, most digital-signals we encounter in real life have a bandwidth with finite values. The effective bandwidth is finite.The baud rate, not the bit-rate, determines the required bandwidth for a digital-signal.More changes in the signal mean injecting more frequencies into the signal. (Frequency means change and change means frequency.) The bandwidth refers to range of frequencies used for transmitting a signal. Relationship b/w baud rate (signal-rate) and the bandwidth (range of frequencies) is given as $B_{min} = c \times N \times (1 / r)$

where N = data-rate (in bps)

c = case factor, which varies for each case

r = previously defined factor

Bmin = minimum bandwidth

4) <u>Baseline Wandering</u>

While decoding, the receiver calculates a running-average of the received signal-power. This average is called the baseline.The incoming signal-power is estimated against this baseline to determine the value of the data-element.A long string of 0s or 1s can cause a drift in the baseline (baseline wandering). Thus, make it difficult for the receiver to decode correctly.A good line-coding scheme needs to prevent baseline wandering.

5) DC Components.

When the voltage-level in a digital-signal is constant for a while, the spectrum creates very low frequencies. These frequencies around zero are called DC (direct-

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current) components. DC components present problems for a system that cannot pass low frequencies. For example: Telephone line cannot pass frequencies below 200 Hz. For Telephone systems, we need a scheme with no DC component.

6) **Built-in Error Detection**

Built-in error-detecting capability has to be provided to detect the errors that occurred during transmission.

7) Self Synchronization

To correctly interpret the signals received from the sender, the receiver's bit intervals must correspond exactly to the sender's bit intervals.If the receiver clock is faster or slower, the bit intervals are not matched and the receiver might misinterpret the signals.

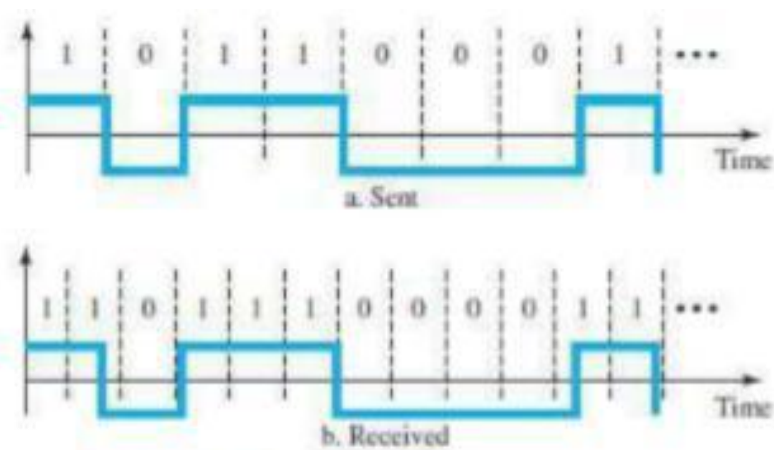


Figure 4.3 *Effect of lack of synchronization*

As shown in figure 4.3, we have a situation where the receiver has shorter bit duration.

The sender sends 10110001, while the receiver receives 110111000011.

A self-synchronizing digital-signal includes timing-information in the data being transmitted.

- This can be achieved if there are transitions in the signal that alert the receiver to the beginning, middle, or end of the pulse.

- If the receiver's clock is out-of-synchronization, these points can reset the clock.

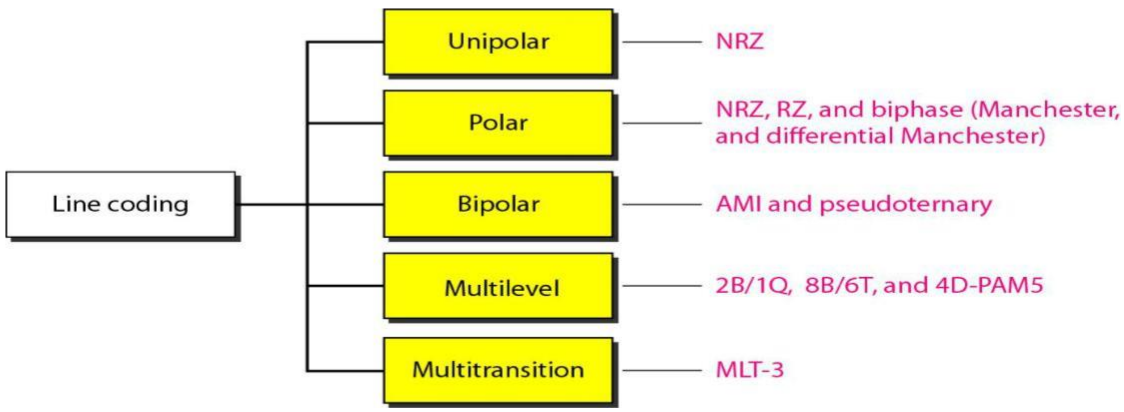
8) Immunity to Noise & Interference

The code should be immune to noise and other interferences.

9) Complexity

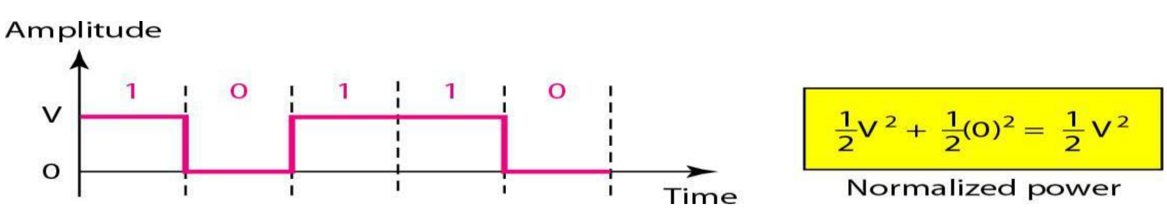
A complex scheme is more costly to implement than a simple one.For ex: A scheme that uses 4 signal-levels is more difficult to interpret than one that uses only 2 levels.

Line Coding Schemes



Unipolar Scheme

In a unipolar scheme, all the signal levels are on one side of the time axis, either above or below. Traditionally, a unipolar scheme was designed as a non-return-to-zero (NRZ) scheme in which the positive voltage defines bit 1 and the zero voltage defines bit 0. It is called NRZ because the signal does not return to zero at the middle of the bit.



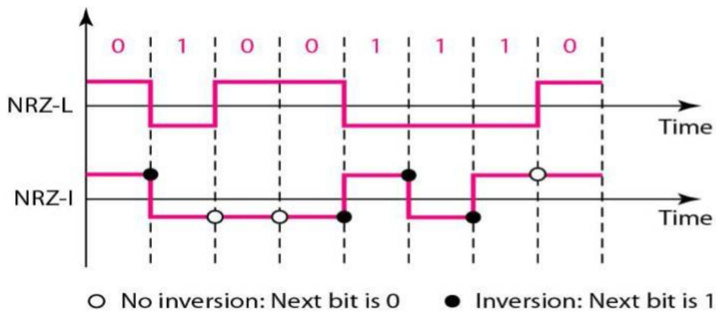
Polar Schemes

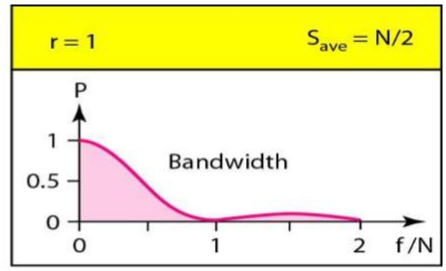
=In polar schemes, the voltages are on the both sides of the time axis. For example, the voltage level for 0 can be positive and the voltage level for I can be negative.

Non-Return-to-Zero (NRZ): In polar NRZ encoding, we use two levels of voltage amplitude. We can have two versions of polar NRZ: NRZ-Land NRZ-I

In **NRZ-L** the level of the voltage determines the value of the bit.

In **NRZ-I** the inversion or the lack of inversion determines the value of the bit.



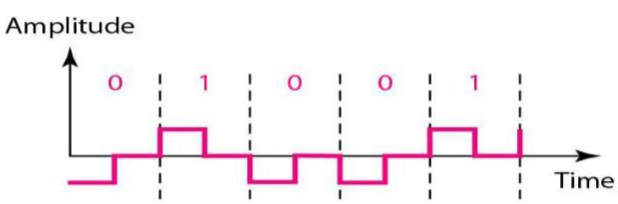


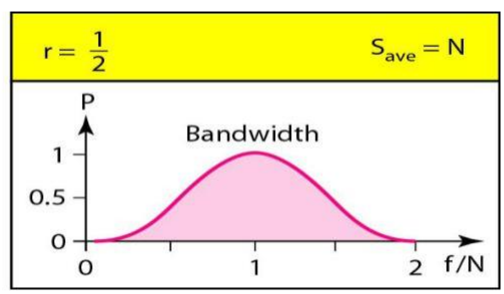
Return to Zero (RZ)*:*

The main problem with NRZ encoding occurs when the sender and receiver clocks are not synchronized. The receiver does not know when one bit has ended and the next bit is starting. One solution is the return-to-zero (RZ) scheme, which uses three values: positive, negative, and zero.

In RZ, the signal changes not between bits but during the bit. In the below Figure we see that the signal goes to 0 in the middle of each bit. It remains there until the beginning of the next bit.

The main disadvantage of RZ encoding is that it requires two signal changes to encode a bit and therefore occupies greater bandwidth.



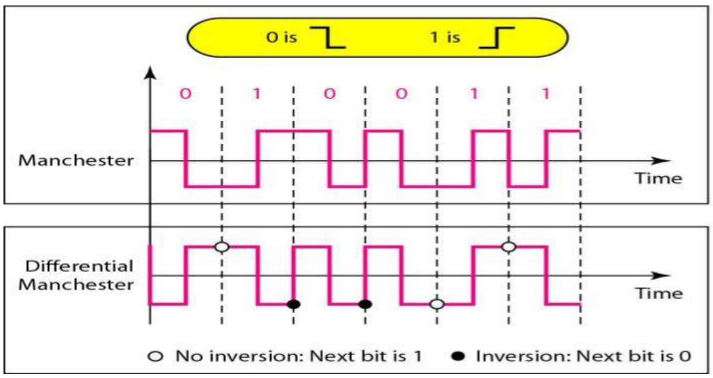


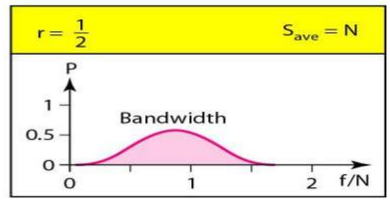
Manchester and Differential Manchester*:*

The idea of RZ (transition at the middle of the bit) and the idea of NRZ-L are combined into the Manchester scheme.

In Manchester encoding, the duration of the bit is divided into two halves. The voltage remains at one level during the first half and moves to the other level in the second half. The transition at the middle of the bit provides synchronization.

Differential Manchester on the other hand, combines the ideas of RZ and NRZ-I. There is always a transition at the middle of the bit, but the bit values are determined at the beginning of the bit. If the next bit is 0, there is a transition; if the next bit is 1, there is none.

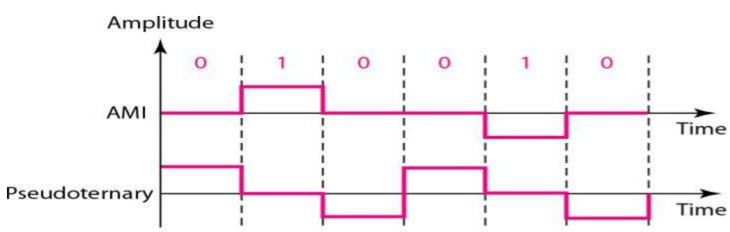


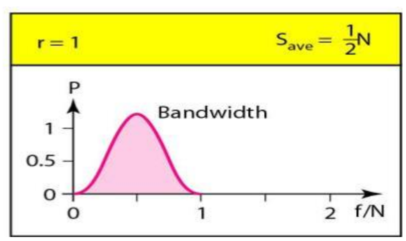


Bipolar Schemes

In bipolar encoding (sometimes called *multilevel binary),* there are three voltage levels: positive, negative, and zero. The voltage level for one data element is at zero, while the voltage level for the other element alternates between positive and negative.

AMI and Pseudoternary: A common bipolar encoding scheme is called bipolar alternate mark inversion (AMI). In the term alternate mark inversion, the word mark comes from telegraphy and means 1. So AMI means alternate I inversion. A neutral zero voltage represents binary O. Binary Is are represented by alternating positive and negative voltages. A variation of AMI encoding is called pseudoternary in which the 1 bit is encoded as a zero voltage and the 0 bit is encoded as alternating positive and negative voltages.





Physical Layer- 2

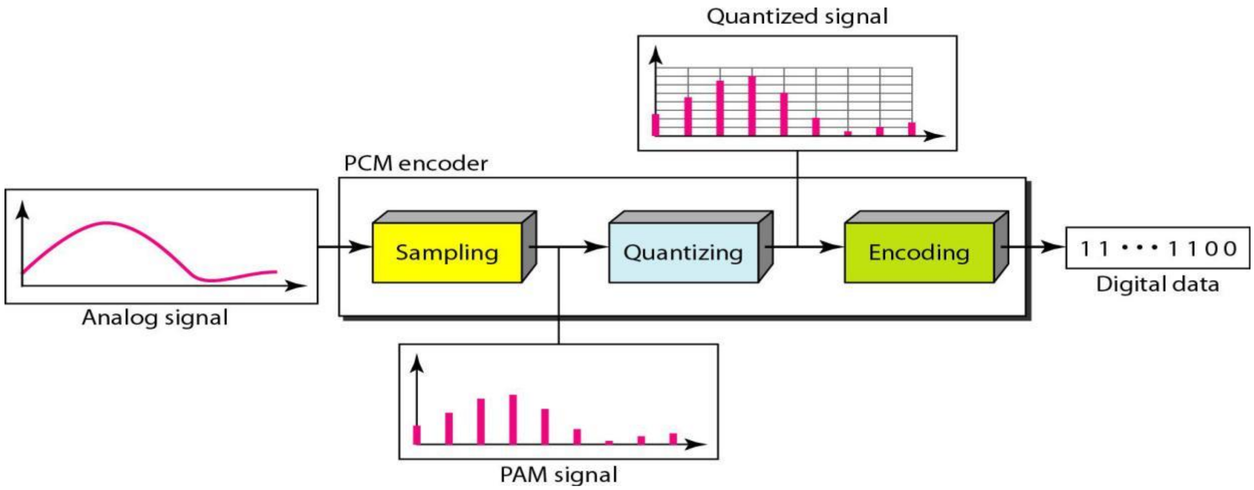
Analog-To-Digital Conversion

- The tendency today is to change analog signal to digital data.The two techniques are PCM(Pulse Code Modulation) and DM(Delta Modulation)

Pulse Code Modulation (PCM)

The most common technique to change an analog signal to digital data (digitization) is called pulse code modulation (PCM). A PCM encoder has three processes.

1. The analog signal is sampled.
2. The sampled signal is quantized.
3. The quantized values are encoded as streams of bits.



Sampling:

The first step in PCM is sampling. The analog signal is sampled every *Ts* s, where *Ts* is the sample interval or period.

The inverse of the sampling interval is called the sampling rate or sampling frequency and denoted by *fs*, where $fs = 1/Ts$.

There are three sampling methods-ideal, natural, and flat-top.

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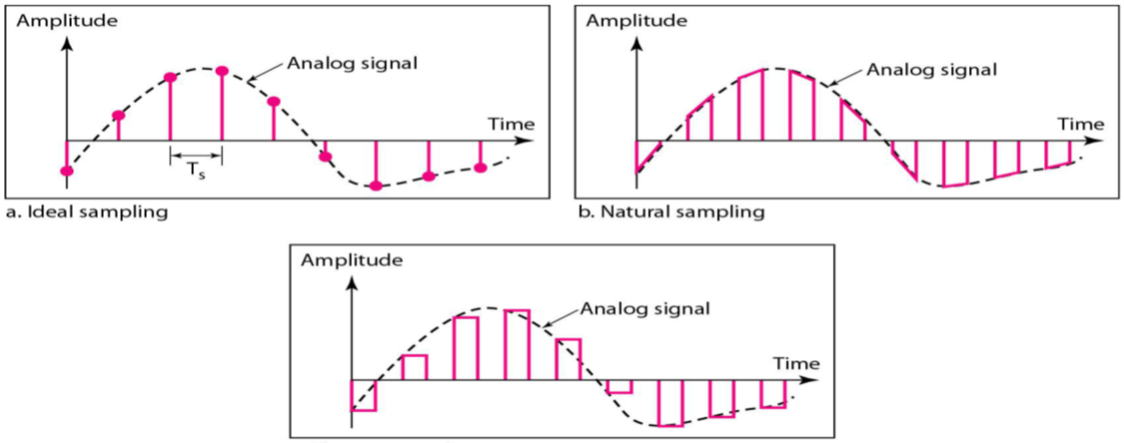


a. Ideal sampling

b. Natural sampling

c. Flat-top sampling

In ideal sampling, pulses from the analog signal are sampled.

In natural sampling, a high-speed switch is turned on for only the small period of time when the sampling occurs. The result is a sequence of samples that retains the shape of the analog signal.

The most common sampling method, called **sample and hold**, however, creates flat-top samples by using a circuit.

The sampling process is sometimes referred to as **pulse amplitude modulation** (PAM).

Sampling Rate: According to the Nyquist theorem, to reproduce the original analog signal, one necessary condition is that the sampling rate be at least twice the highest frequency in the original signal.

Quantization

The result of sampling is a series of pulses with amplitude values between the maximum and minimum amplitudes of the signal.

The set of amplitudes can be infinite with nonintegral values between the two limits.

These values cannot be used in the encoding process.

The following are the steps in quantization:

1. We assume that the original analog signal has instantaneous amplitudes between *V*min and *V*max.

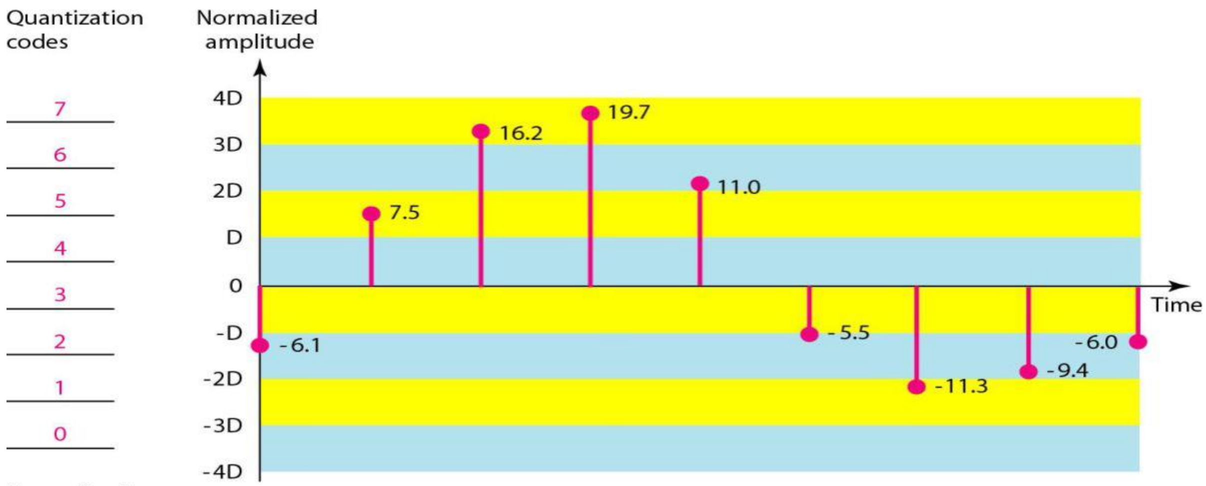
2. We divide the range into *L* zones, each of height (delta).

 = (Vmax-Vmin)/L

3. We assign quantized values of 0 to *L* - 1 to the midpoint of each zone.

4. We approximate the value of the sample amplitude to the quantized values.

As a simple example, assume that we have a sampled signal and the sample amplitudes are between -20 and +20 V. We decide to have eight levels (L = 8). This means that =5 V.



Normalized PAM values	-1.22	1.50	3.24	3.94	2.20	-1.10	-2.26	-1.88	-1.20
Normalized quantized values	-1.50	1.50	3.50	3.50	2.50	-1.50	-2.50	-1.50	-1.50
Normalized error	-0.38	0	+0.26	-0.44	+0.30	-0.40	-0.24	+0.38	-0.30
Quantization code	2	5	7	7	6	2	1	2	2
Encoded words	010	101	111	111	110	010	001	010	010

Quantization Levels: In the previous example, we showed eight quantization levels. The choice of L, the number of levels, depends on the range of the amplitudes of the analog signal and how accurately we need to recover the signal. If the amplitude of a signal fluctuates between two values only, we need only two levels; if the signal, like voice, has many amplitude values, we need more quantization levels. In audio digitizing, L is normally chosen to be 256; in video it is normally thousands. Choosing lower values of L increases the quantization error if there is a lot of fluctuation in the signal.

Quantization Error: One important issue is the error created in the quantization process. Quantization is an approximation process. The input values to the quantizer are the real values; the output values are the approximated values. The output values are chosen to be the middle

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value in the zone. If the input value is also at the middle of the zone, there is no quantization error; otherwise, there is an error.

The quantization error changes the signal-to-noise ratio of the signal, which in turn reduces the upper limit capacity according to Shannon.

It can be proven that the contribution of the quantization error to the SNR_{dB} of the signal depends on the number of quantization levels L, or the bits per sample nb' as shown in the following formula:

$$SNR_{dB} = 6.02 n_b + 1.76 \text{ dB}$$

Encoding:

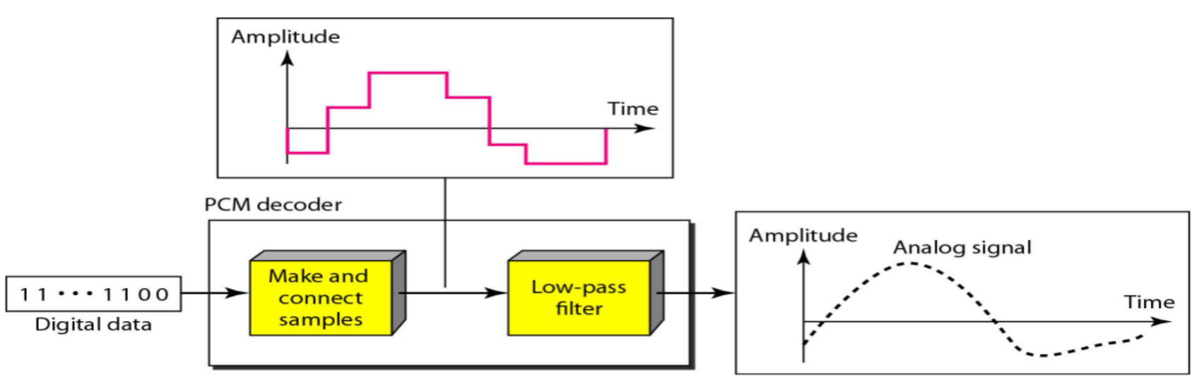
The last step in PCM is encoding. After each sample is quantized and the number of bits per sample is decided, each sample can be changed to an n_b-bit code word.

A quantization code of 2 is encoded as 010; 5 is encoded as 101; and so on. Note that the number of bits for each sample is determined from the number of quantization levels. If the number of quantization levels is L, the number of bits is $n_b = \log_2 L$. In our example L is 8 and n_b is therefore 3. The bit rate can be found from the formula

$$\text{Bit rate} = \text{sampling rate} \times \text{number of bits per sample} = f_s \times n_b$$

Original SIgnal Recovery

- It requires PCM decoder.Decoder first usescircuitry to code words into pulse that holds the amplitude untill the next pulse.
- After the stair case signal is completed,it is passed through a low pass filterto smooth the stair case signal into analog signal.The below figure shows the components of PCM decoder.



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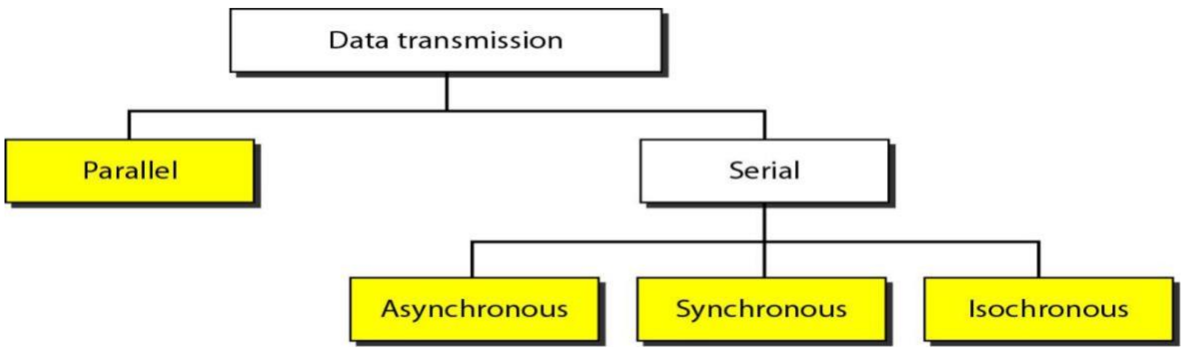
Transmission Modes

The transmission of binary data across a link can be accomplished in either parallel or serial mode.

In parallel mode, multiple bits are sent with each clock tick.

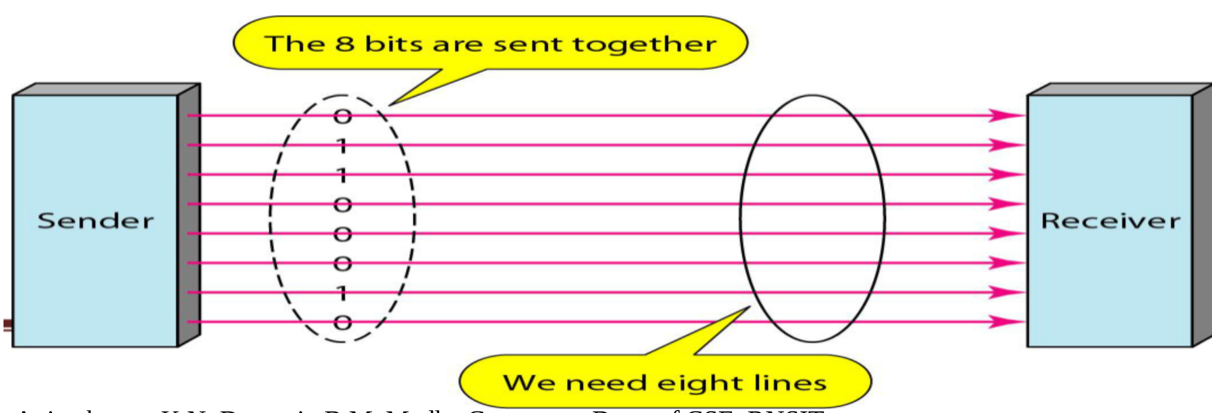
In serial mode, 1 bit is sent with each clock tick. While there is only one way to send parallel data, there are three subclasses of serial transmission: asynchronous, synchronous, and isochronous.



Parallel Transmission

Binary data, consisting of 1s and 0s, may be organized into groups of n bits each. Computers produce and consume data in groups of bits much as we conceive of and use spoken language in the form of words rather than letters. By grouping, we can send data n bits at a time instead of 1. This is called parallel transmission.

The mechanism for parallel transmission is a conceptually simple one: Use *n* wires to send *n* bits at one time. That way each bit has its own wire, and all *n* bits of one group can be transmitted with each clock tick from one device to another. Figure shows how parallel transmission works for *n* =8. Typically, the eight wires are bundled in a cable with a connector at each end.



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The advantage of parallel transmission is speed. All else being equal, parallel transmission can increase the transfer speed by a factor of n over serial transmission.

But there is a significant disadvantage: cost. Parallel transmission requires n communication lines (wires in the example) just to transmit the data stream. Because this is expensive, parallel transmission is usually limited to short distances.

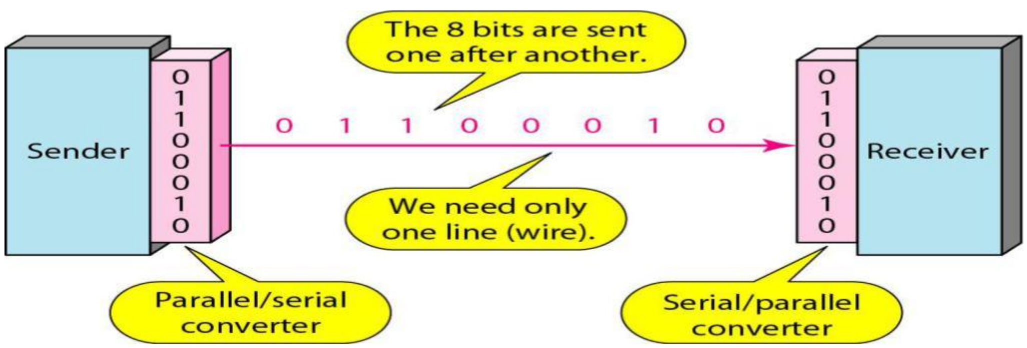
Serial Transmission

In serial transmission one bit follows another, so we need only one communication channel rather than n to transmit data between two communicating devices.

The advantage of serial over parallel transmission is that with only one communication channel, serial transmission reduces the cost of transmission over parallel by roughly a factor of *n*.

Since communication within devices is parallel, conversion devices are required at the interface between the sender and the line (parallel-to-serial) and between the line and the receiver (serial-to-parallel).

Serial transmission occurs in one of three ways: asynchronous, synchronous, and isochronous.



Asynchronous Transmission

Asynchronous transmission is so named because the timing of a signal is unimportant.

Instead, information is received and translated by agreed upon patterns.

As long as those patterns are followed, the receiving device can retrieve the information without regard to the rhythm in which it is sent.

Patterns are based on grouping the bit stream into bytes. Each group, usually 8 bits, is sent along the link as a unit. The sending system handles each group independently, relaying it to the link whenever ready, without regard to a timer.

Without synchronization, the receiver cannot use timing to predict when the next group will arrive.

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To alert the receiver to the arrival of a new group, therefore, an extra bit is added to the beginning of each byte. This bit, usually a 0, is called the start bit.

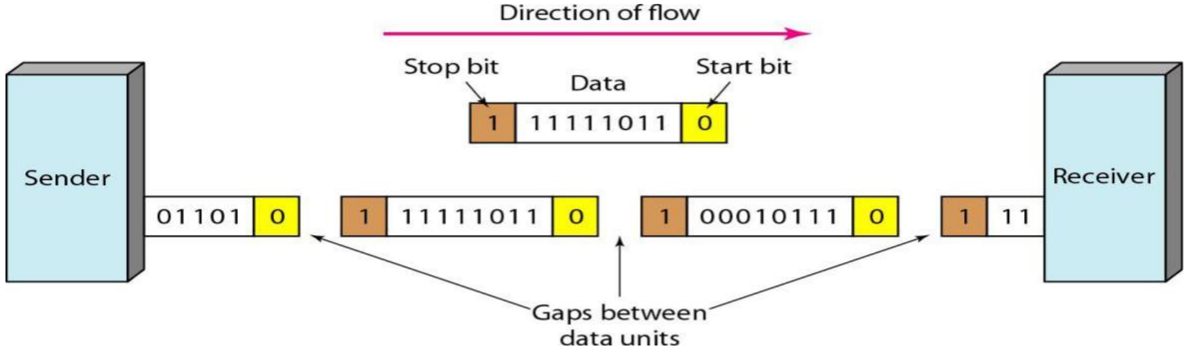
To let the receiver know that the byte is finished, 1 or more additional bits are appended to the end of the byte. These bits, usually 1 s, are called stop bits.

By this method, each byte is increased in size to at least 10 bits, of which 8 bits is information and 2 bits or more are signals to the receiver.

In addition, the transmission of each byte may then be followed by a gap of varying duration. This gap can be represented either by an idle channel or by a stream of additional stop bits.

This mechanism is called *asynchronous* because, at the byte level, the sender and receiver do not have to be synchronized. But within each byte, the receiver must still be synchronized with the incoming bit stream.

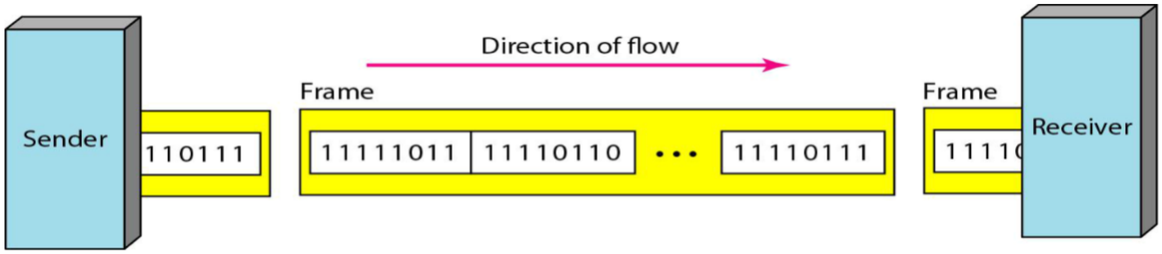
The addition of stop and start bits and the insertion of gaps into the bit stream make asynchronous transmission slower than forms of transmission that can operate without the addition of control information. But it is cheap and effective.



Synchronous Transmission

In synchronous transmission, the bit stream is combined into longer "frames," which may contain multiple bytes.

Each byte, however, is introduced onto the transmission link without a gap between it and the next one. It is left to the receiver to separate the bit stream into bytes for decoding purposes.



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The advantage of synchronous transmission is speed. With no extra bits or gaps to introduce at the sending end and remove at the receiving end, and, by extension, with fewer bits to move across the link, synchronous transmission is faster than asynchronous transmission. Although there is no gap between characters in synchronous serial transmission, there may be uneven gaps between frames.

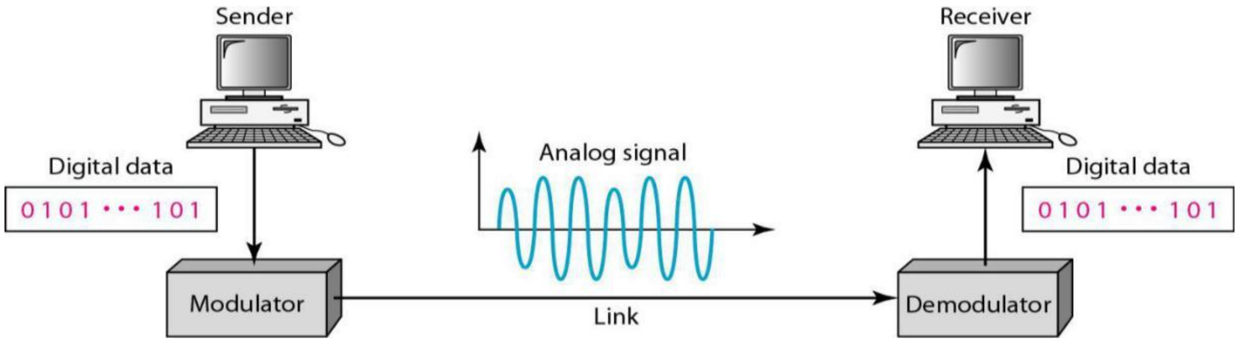
Isochronous

In real-time audio and video, in which uneven delays between frames are not acceptable, synchronous transmission fails. For example, TV images are broadcast at the rate of 30 images per second; they must be viewed at the same rate. If each image is sent by using one or more frames, there should be no delays between frames. For this type of application, synchronization between characters is not enough; the entire stream of bits must be synchronized.

The isochronous transmission guarantees that the data arrive at a fixed rate.

Digital-To-Analog Conversion

Digital-to-analog conversion is the process of changing one of the characteristics of an analog signal based on the information in digital data.

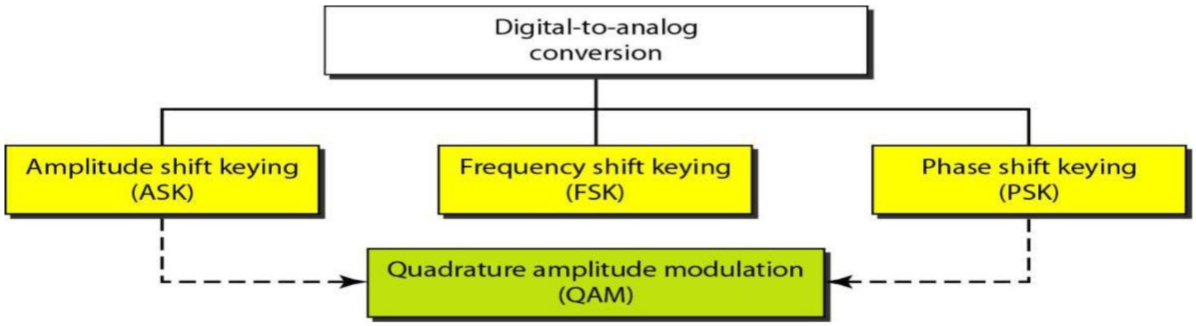


Sine wave is defined by three characteristics: amplitude, frequency, and phase.

When we vary anyone of these characteristics, we create a different version of that wave. So, by changing one characteristic of a simple electric signal, we can use it to represent digital data.

Any of the three characteristics can be altered in this way, giving us at least three mechanisms for modulating digital data into an analog signal: amplitude shift keying (ASK), frequency shift keying (FSK), and phase shift keying (PSK).

In addition, there is a fourth (and better) mechanism that combines changing both the amplitude and phase, called quadrature amplitude modulation (QAM). QAM is the most efficient of these options and is the mechanism commonly used today.



Aspects of Digital-to-Analog Conversion

Data Element versus Signal Element: Data element is the smallest piece of information to be exchanged, that is the bit. Signal element is the smallest unit of a signal that is constant.

Data Rate Versus Signal Rate: The relationship between them is

$$S = N \times 1/r \text{ baud}$$

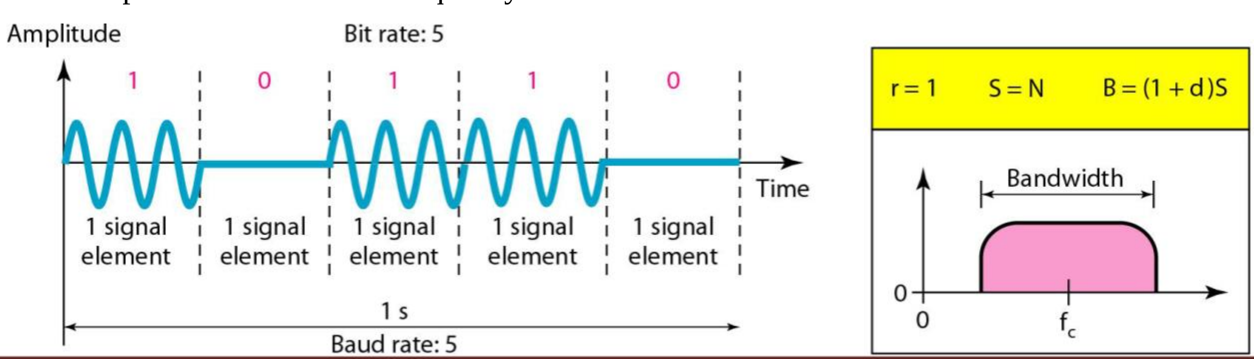
Where, N is the data rate (bps), r is the number of data elements carried in one signal element. $r = \log_2 L$, where L is the type of signal element.

Amplitude Shift Keying

In amplitude shift keying, the amplitude of the carrier signal is varied to create signal elements. Both frequency and phase remain constant while the amplitude changes.

Binary ASK (BASK)

Although we can have several levels (kinds) of signal elements, each with a different amplitude, ASK is normally implemented using only two levels. This is referred to as binary amplitude shift keying or on-off keying (OOK). The peak amplitude of one signal level is 0; the other is the same as the amplitude of the carrier frequency.



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Bandwidth of ASK is

$$B = (1 + d) \times S$$

Where S is the signal rate and the B is the bandwidth. The d depends on the modulation and filtering process. The value of d is between 0 and 1.

Multilevel ASK

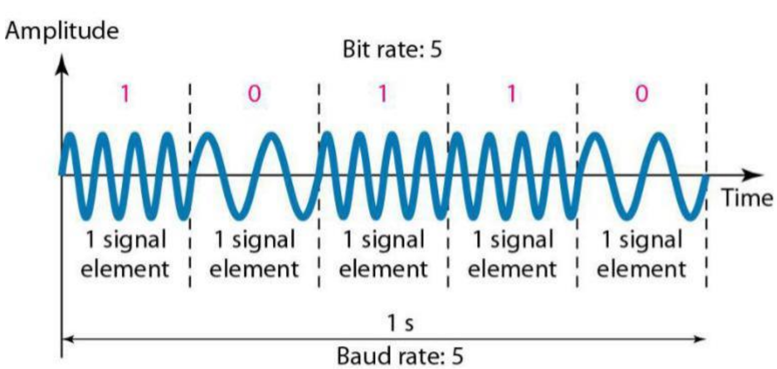
We can have multilevel ASK in which there are more than two levels. We can use 4,8, 16, or more different amplitudes for the signal and modulate the data using 2, 3, 4, or more bits at a time.

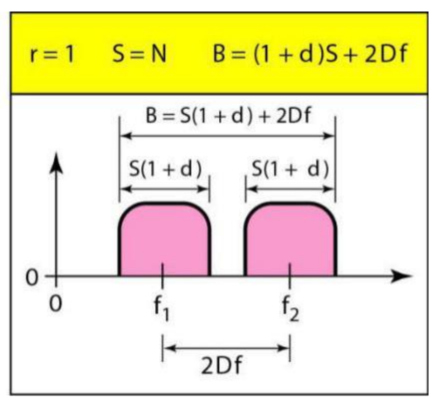
Frequency Shift Keying

In frequency shift keying, the frequency of the carrier signal is varied to represent data. The frequency of the modulated signal is constant for the duration of one signal element, but changes for the next signal element if the data element changes. Both peak amplitude and phase remain constant for all signal elements.

Binary FSK (BFSK)

One way to think about binary FSK (or BFSK) is to consider two carrier frequencies. In Figure, we have selected two carrier frequencies,f} and12. We use the first carrier if the data element is 0; we use the second if the data element is 1. However, note that this is an unrealistic example used only for demonstration purposes. Normally the carrier frequencies are very high, and the difference between them is very small.





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Bandwidth for BFSK:

We can think of FSK as two ASK signals, each with its own carrier frequency (*f1 or f2*) If the difference between the two frequencies is $2\Delta f$, then the required bandwidth is

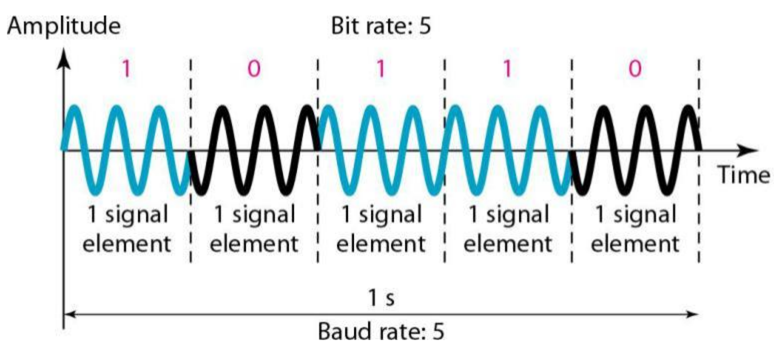
$$B=(1+d) \times S+2\Delta f$$

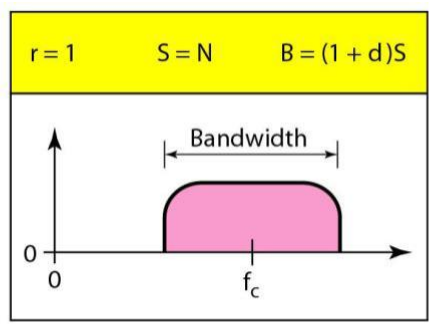
Phase Shift Keying

In phase shift keying, the phase of the carrier is varied to represent two or more different signal elements. Both peak amplitude and frequency remain constant as the phase changes.

Binary PSK (BPSK)

The simplest PSK is binary PSK, in which we have only two signal elements, one with a phase of 0°, and the other with a phase of 180°. Below Figure gives a conceptual view of PSK. Binary PSK is as simple as binary ASK with one big advantage-it is less susceptible to noise. In ASK, the criterion for bit detection is the amplitude of the signal; in PSK, it is the phase. Noise can change the amplitude easier than it can change the phase. In other words, PSK is less susceptible to noise than ASK. PSK is superior to FSK because we do not need two carrier signals.





Bandwidth:

$$B = (1 + d) \times S$$

Where S is the signal rate and the B is the bandwidth. The d depends on the modulation and filtering process. The value of d is between 0 and 1.

Quadrature PSK (QPSK):

The simplicity of BPSK enticed designers to use 2 bits at a time in each signal element, thereby decreasing the baud rate and eventually the required bandwidth. The scheme is called quadrature PSK or QPSK because it uses two separate BPSK modulations; one is in-phase, the other

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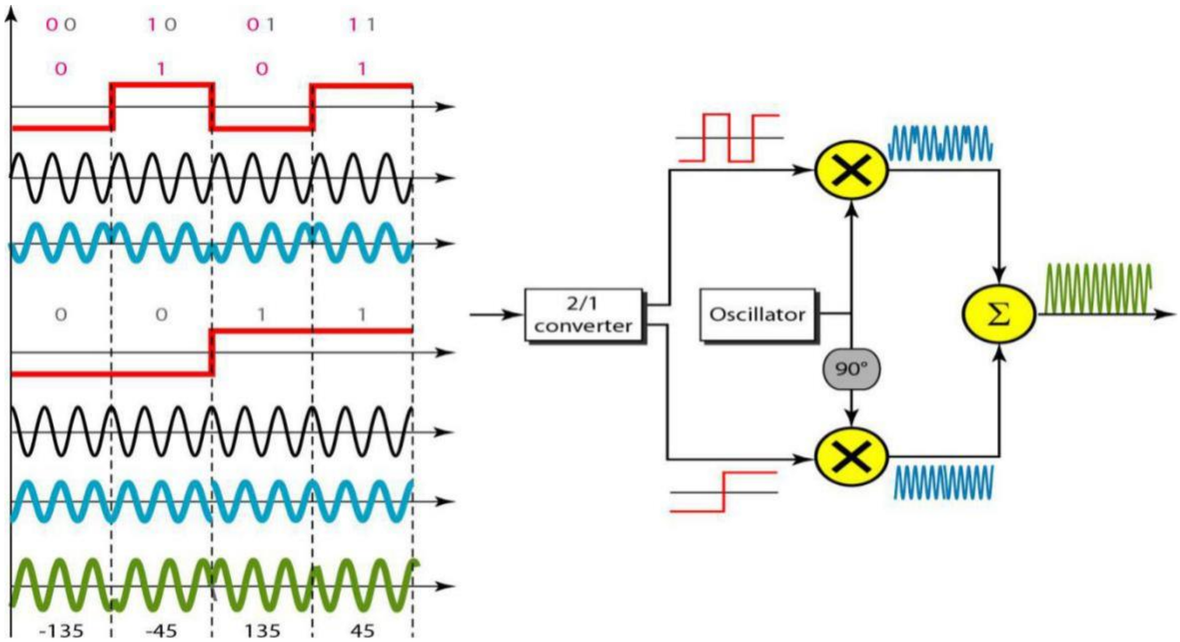
quadrature (out-of-phase). The incoming bits are first passed through a serial-to-parallel conversion that sends one bit to one modulator and the next bit to the other modulator. If the duration of each bit in the incoming signal is T, the duration of each bit sent to the corresponding BPSK signal is 2T. This means that the bit to each BPSK signal has one-half the frequency of the original signal.



Quadrature Amplitude Modulation

Quadrature amplitude modulation is a combination of ASK and PSK. The idea of using two carriers, one in-phase and the other quadrature, with different amplitude levels for each carrier is the concept behind quadrature amplitude modulation (QAM).

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