## MODULE III

## SIGNAL ENCODING TECHNIQUES

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For digital signaling, a data source g(t), which may be either digital or analog, is encoded into a digital signal x(t).The actual form of x(t) depends on the encoding technique and is chosen to optimize use of the transmission medium.

The basis for **analog signaling** is a continuous constant-frequency signal known as the **carrier signal. Modulation** is the process of encoding source data onto a carrier signal with frequency All modulation techniques involve operation on one or more of the three fundamental frequency domain parameters: amplitude, frequency, and phase.

The input signal m(t) may be analog or digital and is called the **modulating signal** or **baseband signal**. The result of modulating the carrier signal is called the **modulated signal s(t).** It is a bandlimited (bandpass) signal. The location of the bandwidth on the spectrum is related to $f_c$ and is often centered on $f_{c.}$

Both analog and digital information can be encoded as either analog or digital signals.

1. Digital data, Digital signal
2. Digital data, Analog signal
3. Analog data, Digital signal
4. Analog data, Analog signal

## DIGITAL DATA, DIGITAL SIGNAL

- The simplest form of digital encoding of digital data is to assign one voltage level to binary one and another to binary zero.
- A digital signal is a sequence of discrete, discontinuous voltage pulses.
- Each pulse is a signal element.
- Binary data are transmitted by encoding each data bit into signal elements.
- Binary 1 is represented by a lower voltage level and binary 0 by a higher voltage level.
- If the signal elements are all positive or negative, then the signal is **unipolar**.

- In **polar** signaling, one logic state is represented by a positive voltage level, and the other by a negative voltage level.
- The **data signaling rate**, or **data rate**, of a signal is the rate, in bits per second, that data are transmitted.
- The duration or length of a bit is the amount of time it takes for the transmitter to emit the bit; for a data rate R, the bit duration is 1/R.
- The **modulation rate**, in contrast, is the rate at which the signal level is changed.
- The terms mark and space refer to the binary digits 1 and 0, respectively.

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## 1.NONRETURN TO ZERO (NRZ)

- To transmit digital signals is to use two different voltage levels for the two binary digits.
- The voltage level is constant during a bit interval; there is no transition (no return to a zero voltage level).
- For example, the absence of voltage can be used to represent binary 0, with a constant positive voltage used to represent binary 1.
- A negative voltage represents one binary value and a positive voltage represents the other.
- This code is known as **Nonreturn to Zero-Level.**

- **NRZ-L** is the code used to generate or interpret digital data by terminals and other devices.
- If a different code is to be used for transmission, it is generated from an NRZ-L signal by the transmission system [Figure: NRZ-L is g(t) and the encoded signal is x(t)].

- A variation of NRZ is known as **NRZI (Nonreturn to Zero, invert on ones).**
- As with NRZ-L, NRZI maintains a constant voltage pulse for the duration of a bit time.
- The data themselves are encoded as the presence or absence of a signal transition at the beginning of the bit time.
- A transition (low to high or high to low) at the beginning of a bit time denotes a binary 1 for that bit time; no transition indicates a binary 0.

- **NRZI** is an example of **differential encoding**. In differential encoding, the information to be transmitted is represented in terms of the changes between successive signal elements rather than the signal elements themselves. The encoding of the current bit is determined as follows: If the current bit is a binary 0, then the current bit is encoded with the same signal as the preceding bit; if the current bit is a binary 1, then the current bit is encoded with a different signal than the preceding bit.

**Advantages of differential encoding**

- It may be more reliable to detect a transition in the presence of noise than to compare a value to a threshold.
- With a complex transmission layout, it is easy to lose the sense of the polarity of the signal.

**Advantages of NRZ Codes**

- Easiest to engineer
- Make efficient use of bandwidth.
- Because of their simplicity and relatively low frequency response characteristics, NRZ codes are commonly used for digital magnetic recording.

**Disadvantages of NRZ Codes**

- The presence of a dc component
- Lack of synchronization capability.
- Unattractive for signal transmission applications.

**Average signal rate:**

NRZ-L and NRZ-I both have an average signal rate of **N/2 Bd**.

**Example:** A system is using NRZ-I to transfer 10-Mbps data. What are the average signal rate and minimum bandwidth?

**Solution:** The average signal rate is S =N/2 =500 kbaud.
The minimum bandwidth for this average baudrate is $B_{min}$ = S = 500 kHz.

## 2.MULTILEVEL BINARY

- **Multilevel binary** addresses some of the deficiencies of the NRZ codes.
- These codes use more than two signal levels.
- Two examples of this scheme are **bipolar-AMI (alternate mark inversion)** and **pseudoternary**.
- In the case of the **bipolar-AMI scheme**, a binary 0 is represented by no line signal, and a binary 1 is represented by a positive or negative pulse. The binary 1 pulses must alternate in polarity.
- **Bandwidth B=N/2** (for AMI)

**Advantages of bipolar-AMI:**

1. There will be no loss of synchronization if a long string of 1s occurs. Each 1 introduces a transition, and the receiver can resynchronize on that transition. A long string of 0s would still be a problem.
2. Because the 1 signals alternate in voltage from positive to negative, there is no net dc component.
3. The bandwidth of the resulting signal is considerably less than the bandwidth for NRZ (Figure).
4. Finally, the pulse alternation property provides a simple means of error detection. Any isolated error, whether it deletes a pulse or adds a pulse, causes a violation of this property.

The **advantages of bipolar-AMI** also apply to **pseudoternary**. In this case, it is the binary 1 that is represented by the absence of a line signal, and the binary 0 by alternating positive and negative pulses.

**Disadvantages:**

- A long string of 0s in the case of AMI or 1s in the case of pseudoternary still presents a problem.
- **Solution**: To insert additional bits that force transitions. This technique is used in ISDN (integrated services digital network) for relatively low data rate transmission. Of course, at a high data rate, this scheme is expensive, because it results in an increase in an already high signal transmission rate. To deal with this problem at high data rates, a technique that involves **scrambling** the data is used.

## 3.BIPHASE

- Biphase overcomes the limitations of NRZ codes.
- Two of these techniques, **Manchester** and **differential Manchester**, are in common use.
- In the **Manchester code**, there is a transition at the middle of each bit period. The midbit transition serves as a clocking mechanism and also as data: a low-to-hig transition represents a 1, and a high-to-low transition represents a 0.
- In **differential Manchester**, the midbit transition is used only to provide clocking. The encoding of a 0 is represented by the presence of a transition at the beginning of a bit period, and a 1 is represented by the absence of a transition at the beginning of a bit period.
- Differential Manchester has the added **advantage** of employing differential encoding.
- All of the **biphase** techniques require at least one transition per bit time and may have as many as two transitions. Thus, the **maximum modulation rate** is twice that for NRZ; this means that the bandwidth required is correspondingly greater.

**Advantages:**

1. **Synchronization**: Because there is a predictable transition during each bit time, the receiver can synchronize on that transition. For this reason, the biphase codes are known as self-clocking codes.
2. **No dc component**: Biphase codes have no dc component
3. **Error detection:** The absence of an expected transition can be used to detect errors. Noise on the line would have to invert both the signal before and after the expected transition to cause an undetected error.

## DIGITAL DATA, ANALOG SIGNAL

Modem converts digital data to an analog signal so that it can be transmitted over an analog line. Involves altering one or more characteristics of a carrier frequency to represent binary data.

The basic techniques are

1. Amplitude shift keying (ASK)
2. Frequency shift keying (FSK)
3. Phase shift keying (PSK).

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## 1.AMPLITUDE SHIFT KEYING (ASK)

- The two binary values are represented by two different amplitudes of the carrier frequency.
- One of the amplitudes is zero; that is, one binary digit is represented by the presence, at constant amplitude of the carrier, the other by the absence of the carrier.
- The resulting transmitted signal for one bit time is

$$\textbf{ASK} \qquad s(t) = \begin{cases} A\cos(2\pi f_c t) & \text{binary 1} \\ 0 & \text{binary 0} \end{cases}$$

where the carrier signal is A $\cos(2\pi f_c t)$.

- ASK is susceptible to sudden gain changes and is a rather inefficient modulation technique.
- On voice-grade lines, it is typically used only up to 1200 bps.
- The ASK technique is used to transmit digital data over optical fiber.
- For LED (light-emitting diode) transmitters, equation is valid. That is, one signal element is represented by a light pulse while the other signal element is represented by the absence of light.

**Bandwidth: B = (1+d) x S**

B- Bandwidth,
S – Signal rate,
d - depends on the modulation and filtering process. The value of d is between 0 and 1.

**Example:** We have an available bandwidth of 100 kHz which spans from 200 to 300 kHz. What are the carrier frequency and the bit rate if we modulated our data by using ASK with d =1?
**Solution**: The middle of the bandwidth is located at 250 kHz. This means that our carrier frequency can be at $f_c$ =250 kHz. We can use the formula for bandwidth to find the bit rate (with d =1 and r =1).

$$\mathbf{B = (l + d) \times S = 2 \times N \times (1/r) = 2 \times N = 100\ kHz \rightarrow N = 50\ kbps}$$

## 2.FREQUENCY SHIFT KEYING (FSK)

**Binary FSK (BFSK):**

- The most common form of FSK is **Binary FSK (BFSK),** in which the two binary values are represented by two different frequencies near the carrier frequency (Figure b).
- The resulting transmitted signal for one bit time is

$$\textbf{BFSK} \qquad s(t) = \begin{cases} A\cos(2\pi f_1 t) & \text{binary 1} \\ A\cos(2\pi f_2 t) & \text{binary 0} \end{cases}$$

where $f_1$ and $f_2$ are typically offset from the carrier frequency $f_c$ by equal but opposite amounts.

- BFSK is less susceptible to error than ASK.
- On voice-grade lines, it is typically used up to 1200 bps.
- It is also commonly used for high-frequency (3 to 30 MHz) radio transmission.
- It can also be used at even higher frequencies on local area networks that use coaxial cable.

We can think of FSK as two ASK signals, each with its own carrier frequency $f_1$ or $f_2$ If the difference between the two frequencies is , then the required **bandwidth** is

**Bandwidth: B = (1+d) x S + 2**

**Example:** We have an available bandwidth of 100 kHz which spans from 200 to 300 kHz. What should be the carrier frequency and the bit rate if we modulated our data by using FSK with d =1?
**Solution:** This problem is similar to above example, but we are modulating by using FSK. The midpoint of the band is at 250 kHz. We choose 2 to be 50 kHz; this means

**B = (1 + d) x S + 2 = 100 -> 2S =50 kHz -> S = 25 kbaud ->N= 25 kbps**

**Multiple FSK (MFSK):**

- A signal that is more bandwidth efficient, but also more susceptible to error, is **multiple FSK (MFSK),** in which more than two frequencies are used.
- In this case each signaling element represents more than one bit.
- The transmitted MFSK signal for one signal element time can be defined as follows:

$$\mathbf{s_i(t) = A\cos(2\pi f_i t)\ ,\quad 1 \le i \le M}$$

where, $f_i = f_c + (2i - 1 - M)f_d$
$f_c$ = the carrier frequency
fd = the difference frequency
M = number of different signal elements = $2^L$
L = number of bits per signal element

To match the data rate of the input bit stream, each output signal element is held for a period of $T_s$ = LT seconds, where T is the bit period (data rate = 1/T ). Thus, one signal element, which is a constant-frequency tone, encodes L bits. The total **bandwidth** required is **$2Mf_d$**. It can be shown that the minimum frequency separation required is $2f_d$ = 1/Ts. Therefore, the modulator requires a bandwidth of **$W_d = 2Mf_d = M/T_s$ .**

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We can use four different frequencies f1, f2, f3, f4 to send 2 bits at a time. To send 3 bits at a time, we can use eight frequencies. And so on. However, we need to remember that the frequencies need to be 2 apart. For the proper operation of the modulator and demodulator, it can be shown that the minimum value of 2 needs to be S. We can show that the **bandwidth** with d =0 is

**B = (1 + d) x S + (L – 1) 2 -> B = L X S**

**Example:** We need to send data 3 bits at a time at a bit rate of 3 Mbps. The carrier frequency is 10 MHz. Calculate the number of levels (different frequencies), the baud rate, and the bandwidth.
**Solution;** We can have $L = 2^3 = 8$. The baud rate is S =3 MHz/3 =1000 Mbaud. This means that the carrier frequencies must be 1MHz apart *(2* =1 MHz). The bandwidth is **B = 8 x 1 = 8 MHz.**

## 3.PHASE SHIFT KEYING (PSK)

In PSK, the phase of the carrier signal is shifted to represent data.

**Two-Level PSK:**

The simplest scheme uses two phases to represent the two binary digits (Figure c) and is known as **binary phase shift keying**. The resulting transmitted signal for one bit time is

$$\textbf{BPSK} \qquad s(t) = \begin{cases} A\cos(2\pi f_c t) \\ A\cos(2\pi f_c t + \pi) \end{cases} = \begin{cases} A\cos(2\pi f_c t) & \text{binary 1} \\ -A\cos(2\pi f_c t) & \text{binary 0} \end{cases}$$

Because a phase shift of 180°(π) is equivalent to flipping the sine wave or multiplying it by -1. If we have a bit stream, and we define d(t) as the discrete function that takes on the value of +1 for one bit time if the corresponding bit in the bit stream is 1 and the value of -1 for one bit time if the corresponding bit in the bit stream is 0, then we can define the transmitted signal as

$$\mathbf{s_d(t) = A\, d(t) \cos(2\pi f_c t)}$$

**Differential PSK (DPSK):**

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In this scheme, a binary 0 is represented by sending a signal burst of the same phase as the previous signal burst sent. A binary 1 is represented by sending a signal burst of opposite phase to the preceding one. This term **differential** refers to the fact that the phase shift is with reference to the previous bit transmitted rather than to some constant reference signal. In differential encoding, the information to be transmitted is represented in terms of the changes between successive data symbols rather than the signal elements themselves.

**Four – Level PSK (or) Quadrature PSK (QPSK):**

Uses phase shifts separated by multiples of π/2 ($90^0$). Thus each signal element represents two bits rather than one.

$$\textbf{QPSK} \qquad s(t) = \begin{cases} A\cos\left(2\pi f_c t + \frac{\pi}{4}\right) & 11 \\ A\cos\left(2\pi f_c t + \frac{3\pi}{4}\right) & 01 \\ A\cos\left(2\pi f_c t - \frac{3\pi}{4}\right) & 00 \\ A\cos\left(2\pi f_c t - \frac{\pi}{4}\right) & 10 \end{cases}$$

Figure (below) shows the QPSK modulation scheme in general terms. The input is a stream of binary digits with a data rate of $R = 1/T_b$ , where $T_b$ is the width of each bit. This stream is converted into two separate bit streams of R/2 bps each, by taking alternate bits for the two streams. The two data streams are referred to as the I (in-phase) and Q (quadrature phase) streams. In the diagram, the upper stream is modulated on a carrier of frequency $f_c$ by multiplying the bit stream by the carrier. For convenience of modulator structure we map binary 1 to 1/2 and binary 0 to – 1/2. Thus, a binary 1 is represented by a scaled version of the carrier wave and a binary 0 is represented by a scaled version of the negative of the carrier wave, both at a constant amplitude. This same carrier wave is shifted by 90° and used for modulation of the lower binary stream. The two modulated signals are then added together and transmitted. The transmitted signal can be expressed as follows:

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Figure 12 shows an example of QPSK coding. Each of the two modulated streams is a BPSK signal at half the data rate of the original bit stream. Thus, the combined signals have a symbol rate that is half the input bit rate. Note that from one symbol time to the next, a phase change of as much as 180° (π) is possible.

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Figure 11 also shows a variation of QPSK known as offset **QPSK (OQPSK), or orthogonal QPSK.** The difference is that a delay of one bit time is introduced in the Q stream, resulting in the following signal:

EQ

Because OQPSK differs from QPSK only by the delay in the Q stream, its spectral characteristics and bit error performance are the same as that of QPSK.

**Multilevel PSK:**

The use of multiple levels can be extended beyond taking bits two at a time. It is possible to transmit bits three at a time using eight different phase angles. Further, each angle can have more than one amplitude.

**Bandwidth of BPSK is same as that of binary ASK, but less than that of BFSK.**

**Bandwidth efficiency:** The ratio of data rate R to transmission bandwidth for various schemes

**Example:** Find the bandwidth for a signal transmitting at 12 Mbps for QPSK. The value of d =0.

**Solution:** For QPSK, 2 bits is carried by one signal element. This means that r =2. So the signal rate (baud rate) is S =N x (l/r) =6 Mbaud. With a value of d =0, we have B =S =6 MHz.

## ANALOG DATA, DIGITAL SIGNAL

The process of transforming analog data into digital signals. It might be more correct to refer this as a process of converting analog data into digital data; this process is known as **digitization**. The three most common are as follows:

1. The digital data can be transmitted using NRZ-L. In this case, directly from analog data to a digital signal.

2. The digital data can be encoded as a digital signal using a code other than NRZ-L. An extra step is required.

3. The digital data can be converted into an analog signal, using one of the modulation techniques.

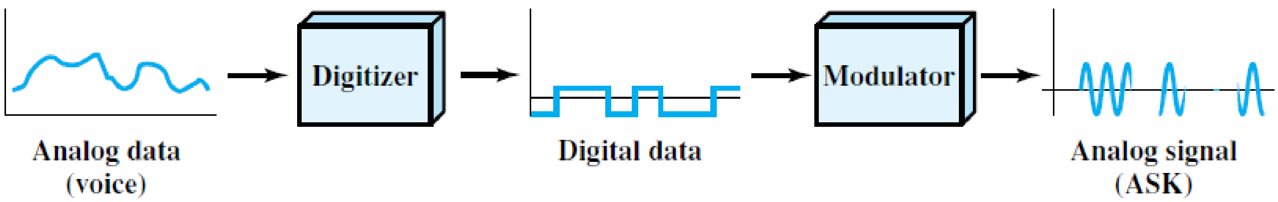


**Fig: Digitizing Analog data**

Voice data are digitized (digital data) and then converted to an analog ASK signal. This **allows digital transmission**. The device used for converting analog data into digital form for transmission, and subsequently recovering the original analog data from the digital, is known as a **codec (coder-decoder).** The two principal techniques used in codecs: **pulse code modulation** and **delta modulation.**

## PULSE CODE MODULATION

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 $f_s$, where $\mathbf{f_s = 1/T_s}$ .There are three sampling methods-**ideal, natural, and flat-top.**

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- 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.

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

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**SAMPLING THEOREM**: If a signal f(t) is sampled at regular intervals of time and at a rate higher than twice the highest signal frequency, then the samples contain all the information of the original signal. The function f(t) may be reconstructed from these samples by the use of a lowpass filter.

Thus, **PCM** starts with a continuous-time, continuous-amplitude (analog) signal, from which a digital signal is produced. The digital signal consists of blocks of n bits, where each n-bit number is the amplitude of a PCM pulse. On reception, the process is reversed to reproduce the analog signal. However, this process violates the terms of the sampling theorem. By **quantizing** the PAM pulse, the original signal is now only approximated and cannot be recovered exactly. This effect is known as **quantizing error or quantizing noise**. The signal-to-noise ratio for quantizing noise can be expressed as [GIBS93]

$$\mathbf{SNR_{dB} = 20 \log 2^n + 1.786\ dB = 6.02n + 1.76\ dB}$$

Thus each additional bit used for quantizing increases SNR by about 6 dB, which is a factor of 4.

**Example:** A telephone subscriber line must have an $SNR_{dB}$ above 40. What is the minimum number of bits per sample?

**Solution:** $\mathbf{SNR_{dB} = 6.02\ n + 1.76 = 40 \ \ -> n = 6.35}$

Typically, the PCM scheme is refined using a technique known as **nonlinear encoding**, which means, in effect, that the quantization levels are not equally spaced. The problem with equal spacing is that the mean absolute error for each sample is the same, regardless of signal level. Consequently, lower amplitude values are relatively more distorted. By using a greater number of quantizing steps for signals of low amplitude, and a smaller number of quantizing steps for signals of large amplitude, a marked reduction in overall signal distortion is achieved (e.g., see Figure).

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The same effect can be achieved by using uniform quantizing but companding **(compressing-expanding)** the input analog signal. **Companding** is a process that compresses the intensity range of a signal by imparting more gain to weak signals than to strong signals on input. At output, the reverse operation is performed. The effect on the input side is to compress the sample so that the higher values are reduced with respect to the lower values. Thus, with a fixed number of quantizing levels, more levels are available for lower-level signals. On the output side, the compander expands the samples so the compressed values are restored to their original values.

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- **Bit rate** = sampling rate x number of bits per sample = $f_s$ x $n_b$

**Example:** We want to digitize the human voice. What is the bit rate, assuming 8 bits per sample?
**Solution:** The human voice normally contains frequencies from 0 to 4000 Hz.

**Sampling rate = 4000 x 2 = 8000 samples/s**

**Bit rate = 8000 x 8 = 64,000 bps == 64 kbps**

**PCM Bandwidth:**
The minimum bandwidth of a line-encoded signal is $B_{min}$ = c x N x (1/r). Substituting value of N,

$$B_{min} = c \times N \times (1/r) = c \times n_b \times f_s \times 1/r = c \times n_b \times 2 \times B_{analog} \times 1/r$$

When 1/r = 1 (for a NRZ or bipolar signal) and c = (1/2) (the average situation), the minimum bandwidth is

$$B_{min} = n_b \times B_{analog}$$

This means the minimum bandwidth of the digital signal is $n_b$ times greater than the bandwidth of the analog signal.

## DELTA MODULATION (DM)

- One of the most popular alternatives to PCM is delta modulation (DM).
- With delta modulation, an analog input is approximated by a staircase function that moves up or down by one quantization level ( ) at each sampling interval (Ts).
- An example is shown in Figure, where the staircase function is overlaid on the original analog waveform.
- The important characteristic of this staircase function is that its behavior is binary: At each sampling time, the function moves up or down a constant amount .
- Thus, the output of the delta modulation process can be represented as a single binary digit for each sample.
- In essence, a bit stream is produced by approximating the derivative of an analog signal rather than its amplitude: A 1 is generated if the staircase function is to go up during the next interval; a 0 is generated otherwise.

Figure 21 illustrates the logic of process - For transmission, the following occurs: At each sampling time, the analog input is compared to the most recent value of the approximating staircase function. If the value of the sampled waveform exceeds that of the staircase function, a 1 is generated; otherwise, a 0 is generated. Thus, the staircase is always changed in the direction of the input signal. The output of the DM process is therefore a binary sequence that can be used at the receiver to reconstruct the staircase function.

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There are two important parameters in a DM scheme:

(a) the size of the step assigned to each binary digit
(b) the sampling rate.

As Figure 20 illustrates,   must be chosen to produce a balance between **two types of errors or noise**.

- When the analog waveform is changing very slowly, there will be **quantizing noise**. This noise increases as is increased.
- When the analog waveform is changing more rapidly than the staircase can follow, there is **slope overload noise**. This noise increases as   is decreased.

The **accuracy** of the scheme can be improved by increasing the **sampling rate**. However, this increases the data rate of the output signal.

**Advantages of DM over PCM :**

1. Simplicity .
2. PCM exhibits better SNR characteristics at the same data rate.

## ANALOG DATA, ANALOG SIGNALS

Analog data are modulated by a carrier frequency to produce an analog signal in a different frequency band, which can be utilized on an analog transmission system. There are two principal reasons for analog modulation of analog signals:

1. A higher frequency may be needed for effective transmission. For unguided transmission, it is virtually impossible to transmit baseband signals; the required antennas would be many kilometers in diameter.
2. Modulation permits frequency division multiplexing, an important technique

The basic techniques for modulation using analog data are:

1. Amplitude modulation (AM)
2. Frequency modulation (FM
3. Phase modulation (PM)

## 1.AMPLITUDE MODULATION (AM)

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Amplitude modulation (AM) is the simplest form of modulation. Mathematically, the process can be expressed as,

$$\mathbf{s(t) = [\ 1 + n_a x(t)\ ] \cos 2\pi f_c t}$$

where, $\cos 2\pi f_c t$ is the carrier and $x(t)$ is the input signal (carrying data), both normalized to unity amplitude. The parameter $n_a$ is known as the **modulation index**, is the ratio of the amplitude of the input signal to the carrier. The “1” in the equation is a dc component that prevents loss of information. This scheme is also known as **double sideband transmitted carrier (DSBTC).**

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The spectrum consists of the original carrier plus the spectrum of the input signal translated to $f_c$. The portion of the spectrum for $|f| > |f_c|$ is the upper sideband, and the portion of the spectrum for $|f| < |f_c|$ is lower sideband. Both the upper and lower sidebands are replicas of the original spectrum M(f ), with the lower sideband being frequency reversed.

An important relationship,

$$P_t = P_c\left(1 + \frac{n_a^2}{2}\right)$$

where, $P_t$ is the total transmitted power in s(t) and $P_c$ is the transmitted power in the carrier. We would like $n_a$ as large as possible so that most of the signal power is used to carry information. However, $n_a$ must remain below 1.

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It should be clear that s(t) contains unnecessary components, because each of the sidebands contains the complete spectrum of m(t). A popular variant of AM, known as single sideband (SSB), takes advantage of this fact by sending only one of the sidebands, eliminating the other sideband and the carrier. The principal advantages of this approach are as follows:

- Only half the bandwidth is required, that is, $B_T = B$, where B is the bandwidth of the original signal. For DSBTC, $B_T = 2B$.
- Less power is required because no power is used to transmit the carrier or the other sideband. Another variant is double sideband suppressed carrier (DSBSC), which filters out the carrier frequency and sends both sidebands. This saves some power but uses as much bandwidth as DSBTC.

The **disadvantage** of suppressing the carrier is that the carrier can be used for synchronization purposes.
A compromise approach **is vestigial sideband (VSB),** which uses one sideband and a reduced-power carrier.

**AM Bandwidth:**
The modulation creates a bandwidth that is twice the bandwidth of the modulating signal and covers a range centered on the carrier frequency.

The total **bandwidth** required for AM can be determined from the bandwidth of the audio signal:

$$\mathbf{B_{AM} = 2B}$$

## 2.ANGLE MODULATION

(a) Frequency Modulation (FM)
(b) Phase Modulation(PM)

Frequency modulation (FM) and phase modulation (PM) are special cases of angle modulation.
The modulated signal is expressed as,

$$\textbf{Angle Modulation,} \quad s(t) = A_c \cos[2\pi f_c t + \Phi_t]$$

For **phase modulation**, the phase is proportional to the modulating signal:

$$\textbf{PM,} \qquad \Phi(t) = n_p m(t)$$

where $n_p$ is the phase modulation index.

For **frequency modulation**, the derivative of the phase is proportional to the modulating signal:

$$\textbf{FM,} \qquad \Phi'(t) = n_f m(t)$$

where $n_f$ is the frequency modulation index and $\Phi'(t)$ is the derivative of $\Phi(t)$.

The **peak deviation** is,

$$= n_f A_m \text{ Hz}$$

where $A_m$ is the maximum value of m(t).Thus an increase in the magnitude of m(t) will increase which, intuitively, should increase the transmitted bandwidth $B_T$.

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**FM & PM Bandwidth:**
The total bandwidth required for FM can be determined from the bandwidth of the audio signal:

$$\mathbf{B_{FM} = 2(1 + \beta)B}$$

Although, the formula shows the same bandwidth for FM and PM, the value of $\beta$ is lower in the case of PM (around 1 for narrowband and 3 for wideband).