# Mahatma Gandhi Institute of Technology

# Analog & Digital communications

## UNIT- 4

## Introduction to Digital Communications

# Introduction

- Digital communication is a mode of communication where the information or the thought is encoded digitally as discreet signals and electronically transferred to the recipients.
- Digital communication information flows in a digital form and the source is generally the keyboard of the computer. A single individual is capable of digital communication and thus it also saves wastage of manpower and is one of the cheapest modes of communication.
- Digital communication is also a really quick way to communicate. The information can reach the recipient within a fraction of a second. An individual no longer has to wait to personally meet the other individual and share his information.

# Advantages

- The digital communication has mostly common structure of encoding a signal so devices used are mostly similar.
- The Digital Communication's main advantage is that it provides us added security to our information signal.
- The digital Communication system has more immunity to noise and external interference.
- Digital information can be saved and retrieved when necessary while it is not possible in analog.
- Digital Communication is cheaper than Analog Communication.
- The configuring process of digital communication system is simple as compared to analog communication system. Although, they are complex.
- In Digital Communication System, the error correction and detection techniques can be implemented easily.

# Disadvantages

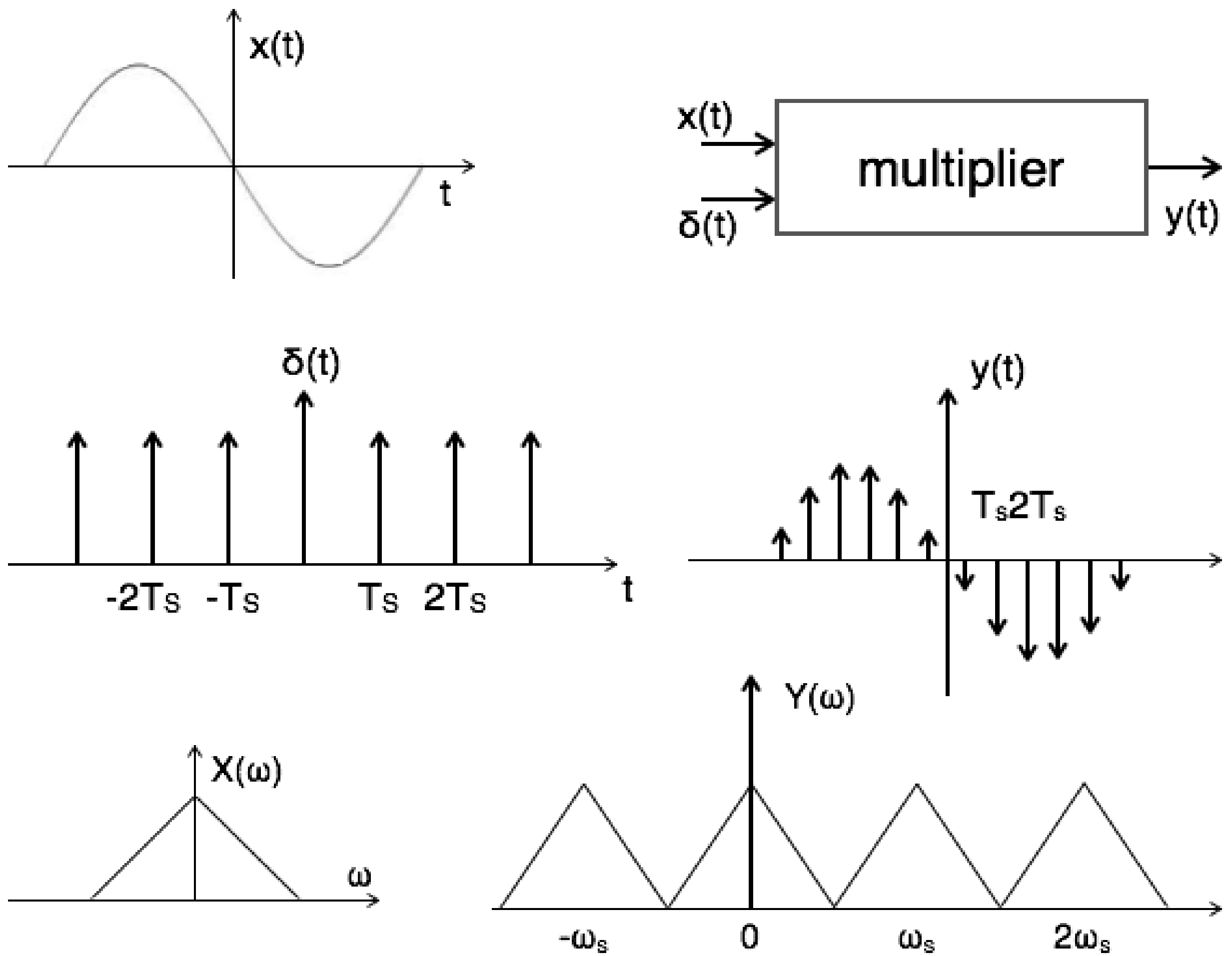
- Disadvantages of digital communication:
  1). Generally, more bandwidth is required than that for analog systems.
  2). Synchronization is required.
  3). High power consumption (Due to various stages of conversion).
  4). Complex circuit, more sophisticated device making is also drawbacks of digital system.
  5). Introduce sampling error

# Sampling theorem

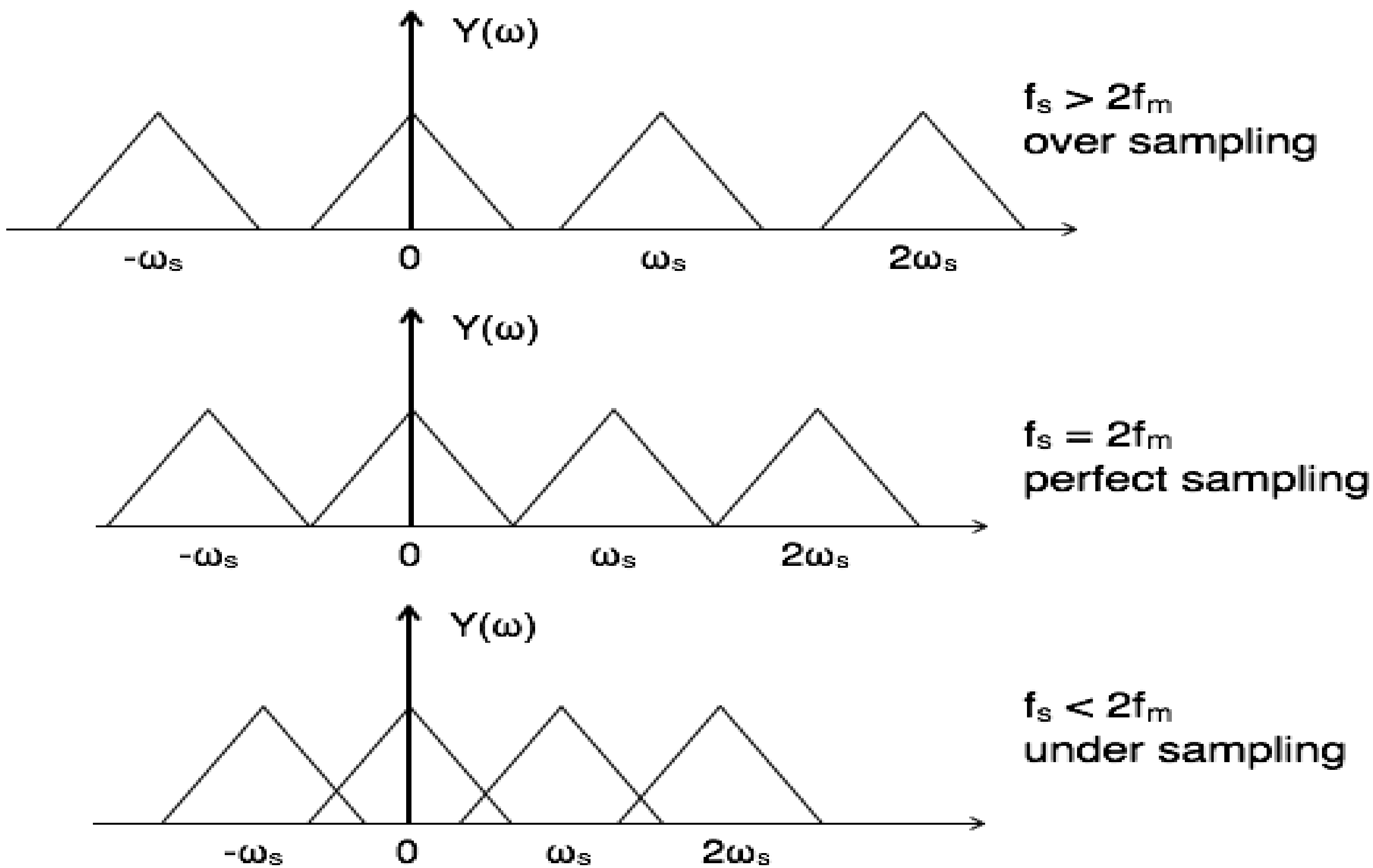
- **Statement:** A continuous time signal can be represented in its samples and can be recovered back when sampling frequency $f_s$ is greater than or equal to the twice the highest frequency component of message signal. i. e.
- fs≥2fm.fs≥2fm.

- **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:


x(t)
t
x(t)
δ(t)
multiplier
y(t)
δ(t)
-2Ts
-Ts
Ts
2Ts
t
y(t)
Ts2Ts
X(ω)
ω
Y(ω)
-ωs
0
ωs
2ωs

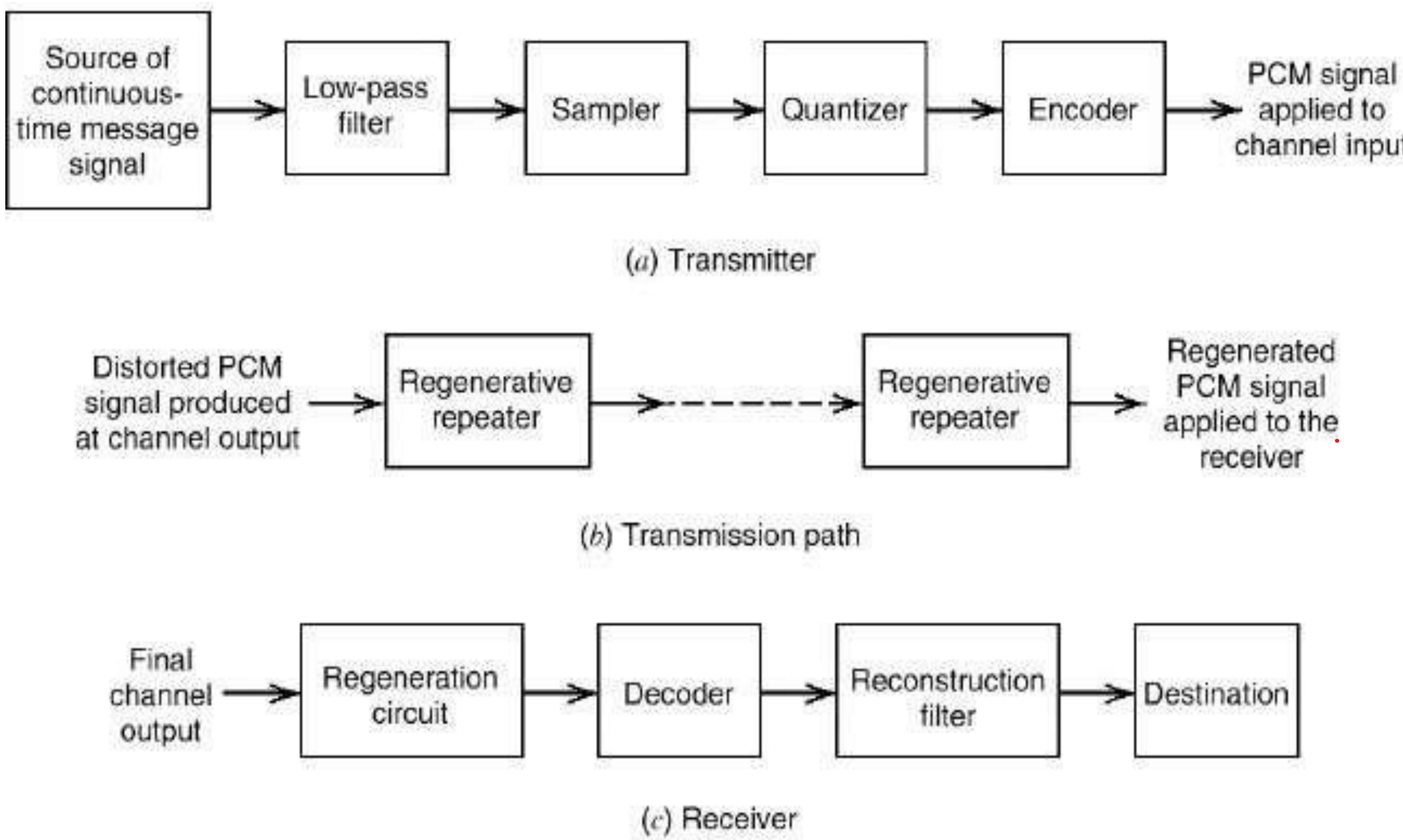
# Sampling effect



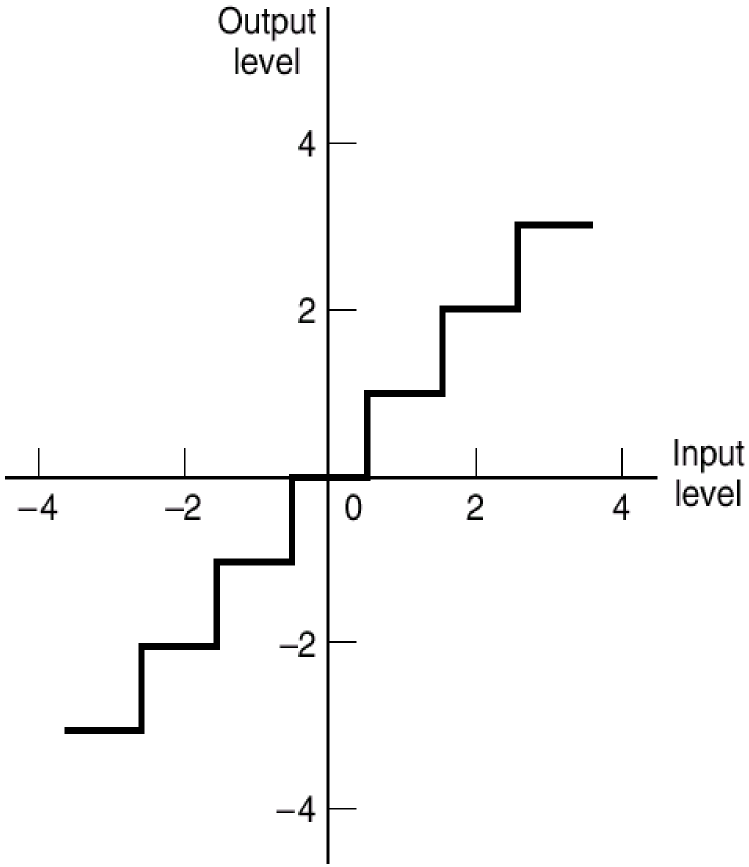
# Pulse-Code Modulation

- In PCM, a message signal is represented by a sequence of coded pulses, obtained from representing the signal in discrete form in both time an amplitude.
- As shown by Figure 3.13 the basic operations performed at the transmitter are: sampling, quantization, and encoding.
- The basic operations in the receiver are regeneration of impaired signals, decoding and reconstruction.
- Regeneration also occurs during the route of transmission.
- In PCM of voice signals, non-uniform quantization is used to allow smaller quantization step sizes form smaller amplitudes and larger step sizes for larger amplitudes so that the $(SNR)_o$ remains quasi-constant for all levels of amplitudes.
- To use non-uniform quantization, the message signal is passed through a *compressor*, then a uniform quantization is applied to the compressed signal. At the receiver an *expander* circuit is used to undo the effect of the compressor.

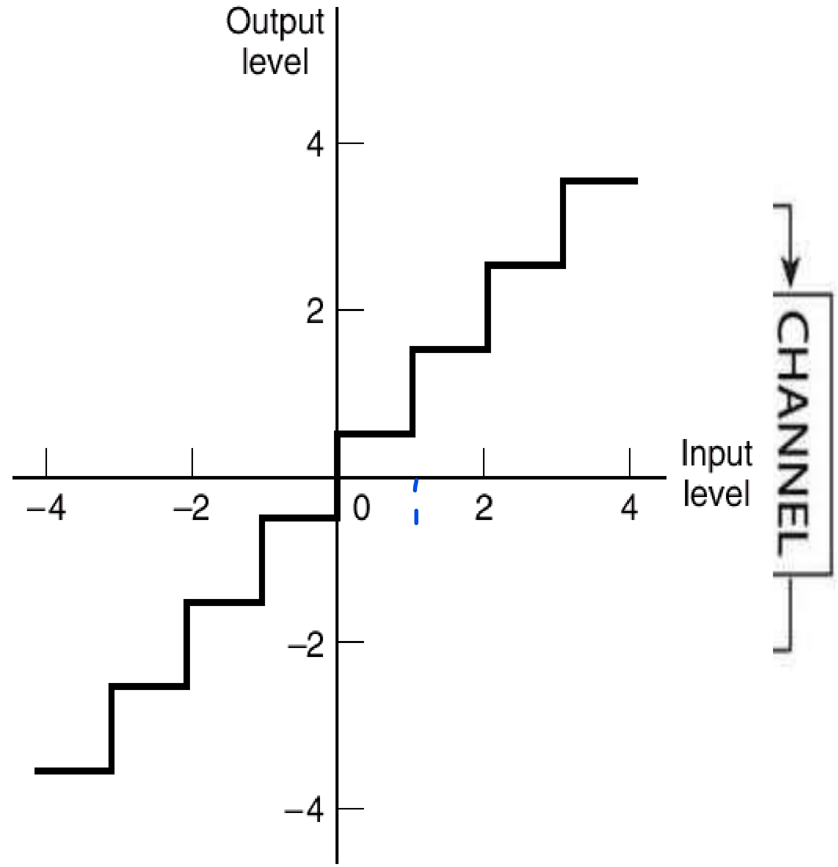
# Pulse Code Modulation(PCM)



# Two types of quantization: (*a*) midtread and (*b*) midrise.

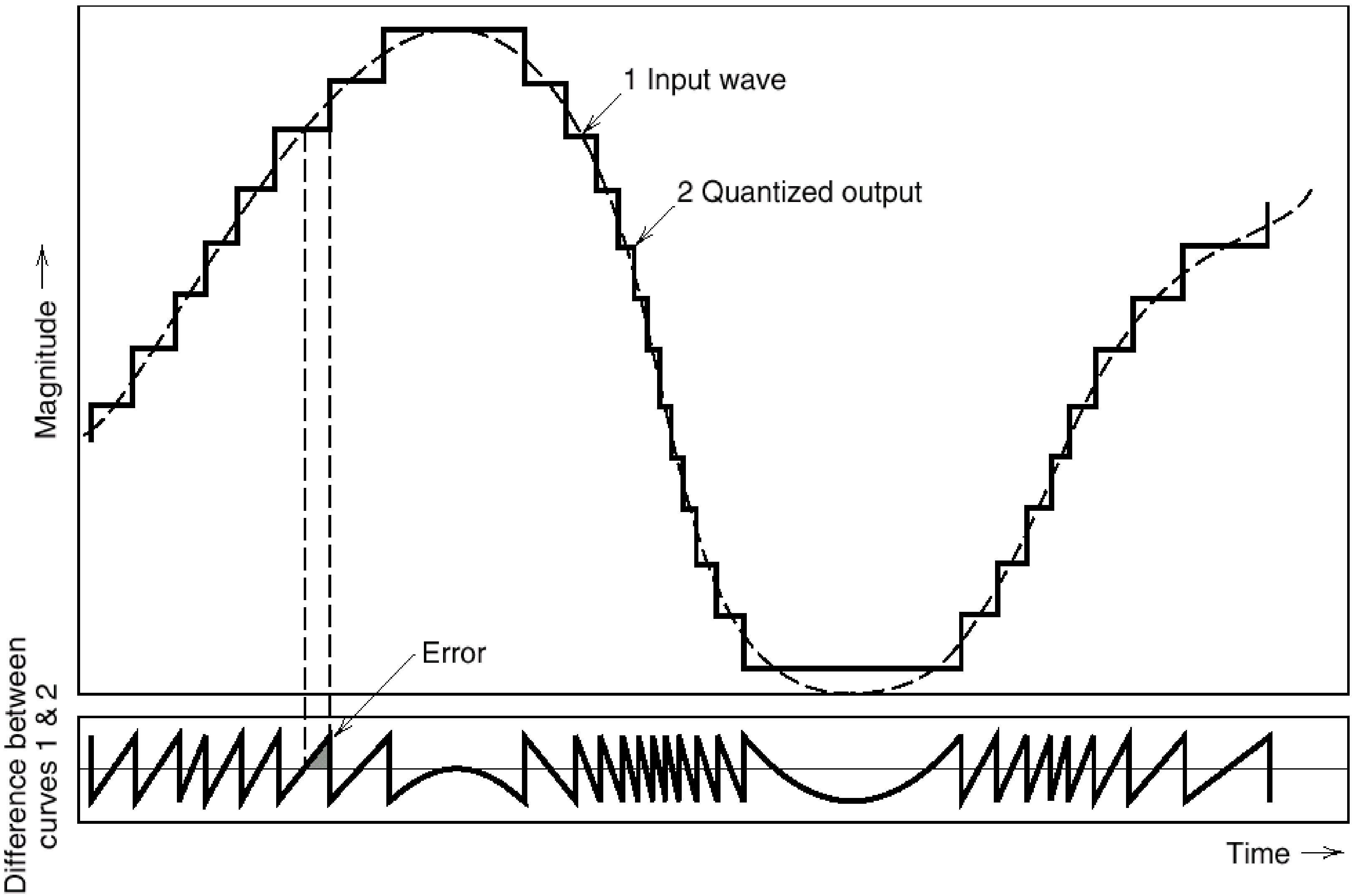



(*a*)



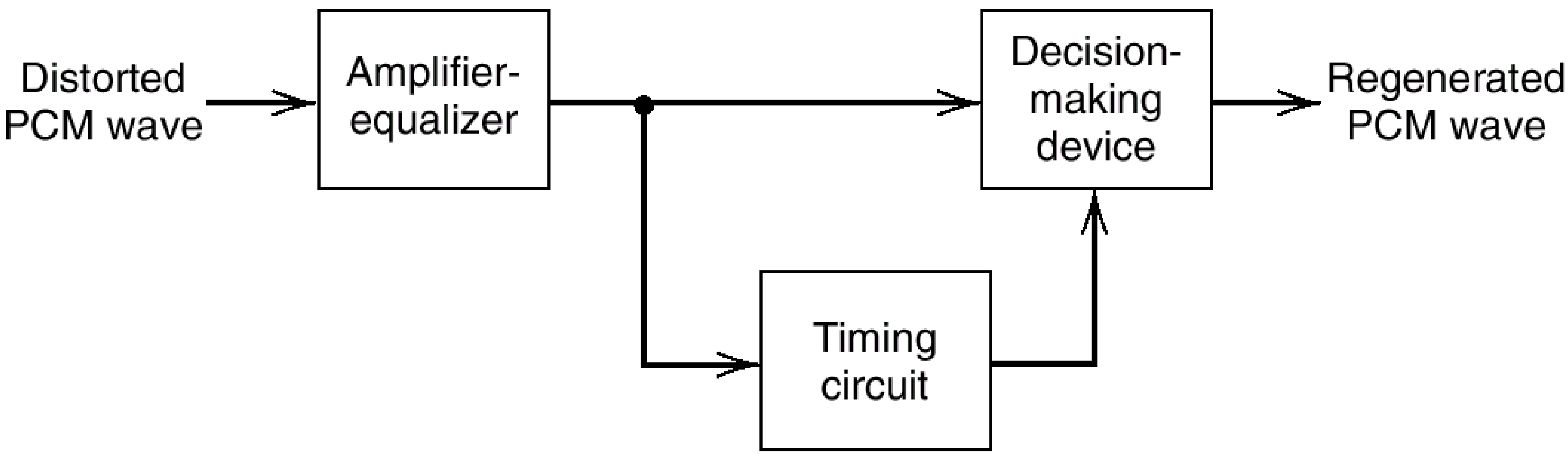

(*b*)

# Illustration of the quantization process. (Adapted from Bennett, 1948,



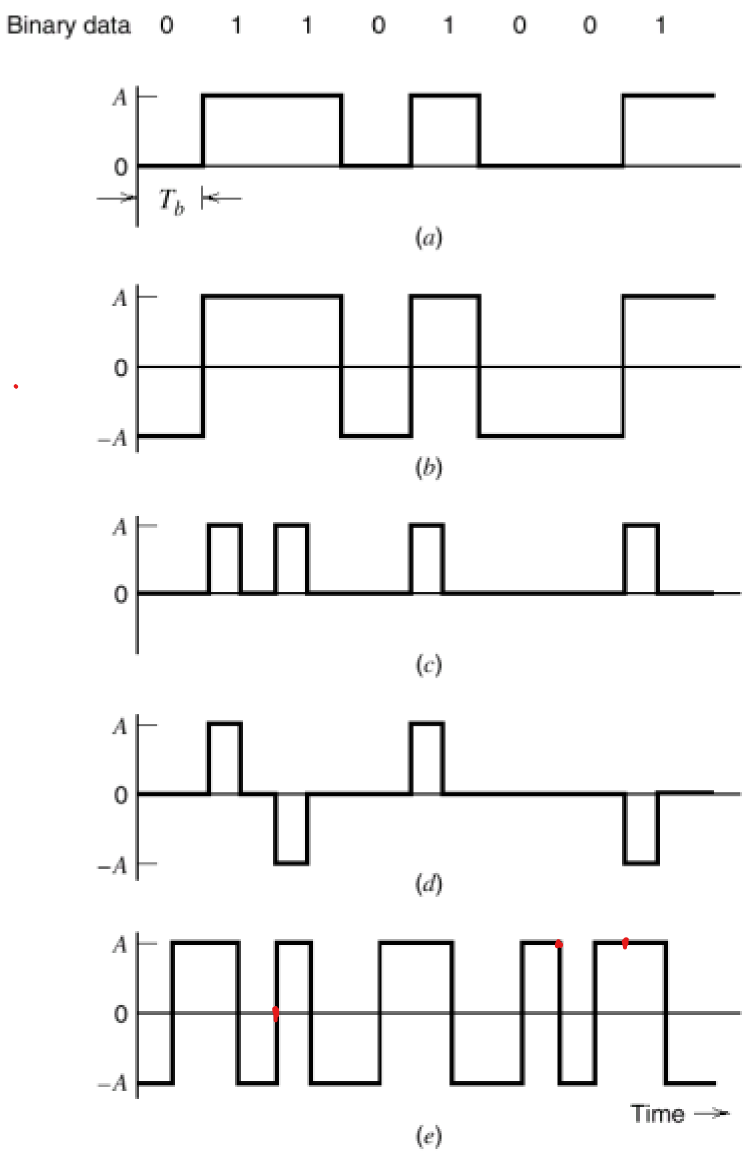
# Figure 3.18

Block diagram of regenerative repeater.

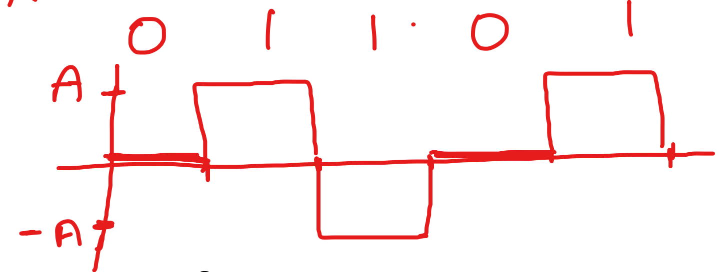


## Regeneration

- A regenerative repeater (see Figure 3.18) consists of (1) an equalizer, (2) a timing circuit, and (3) a decision-making device. The equalizer is used to undo the effect of the transmission channel to get back the pulses in their original shape before transmission. The timing circuit is used to recover the clock of the transmitted symbols (pulses), which is then used in the decision-making process. The function of the decision-making device is to detect the different pulses based on some threshold information.
- The purpose of a regenerative repeater is to clean the PCM signal during its transmission through a channel.




* AMI : Bipolar NRZ *



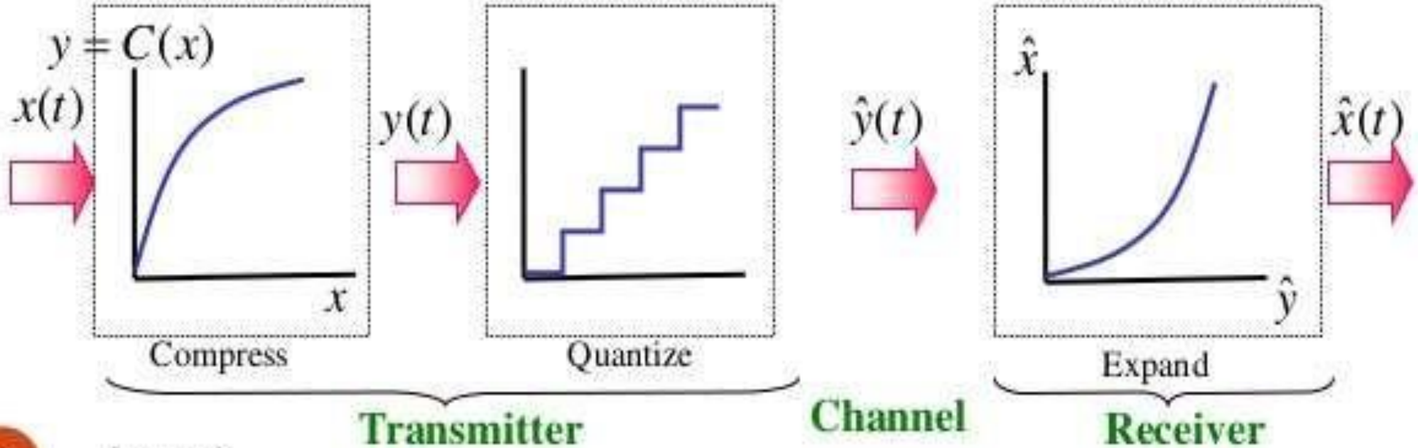

# Figure 3.15

Line codes for the electrical representations of binary data.
(a) Unipolar NRZ signaling. (*b*) Polar NRZ signaling.
(*c*) Unipolar RZ signaling. (*d*) Bipolar RZ signaling. (*e*) Split-phase or Manchester code.

# Non-uniform quantization

- It is achieved by uniformly quantizing the "compressed" signal.
- At the receiver, an inverse compression characteristic, called "expansion" is employed to avoid signal distortion.

compression+expansion ⇨ companding



# Introduction DPCM

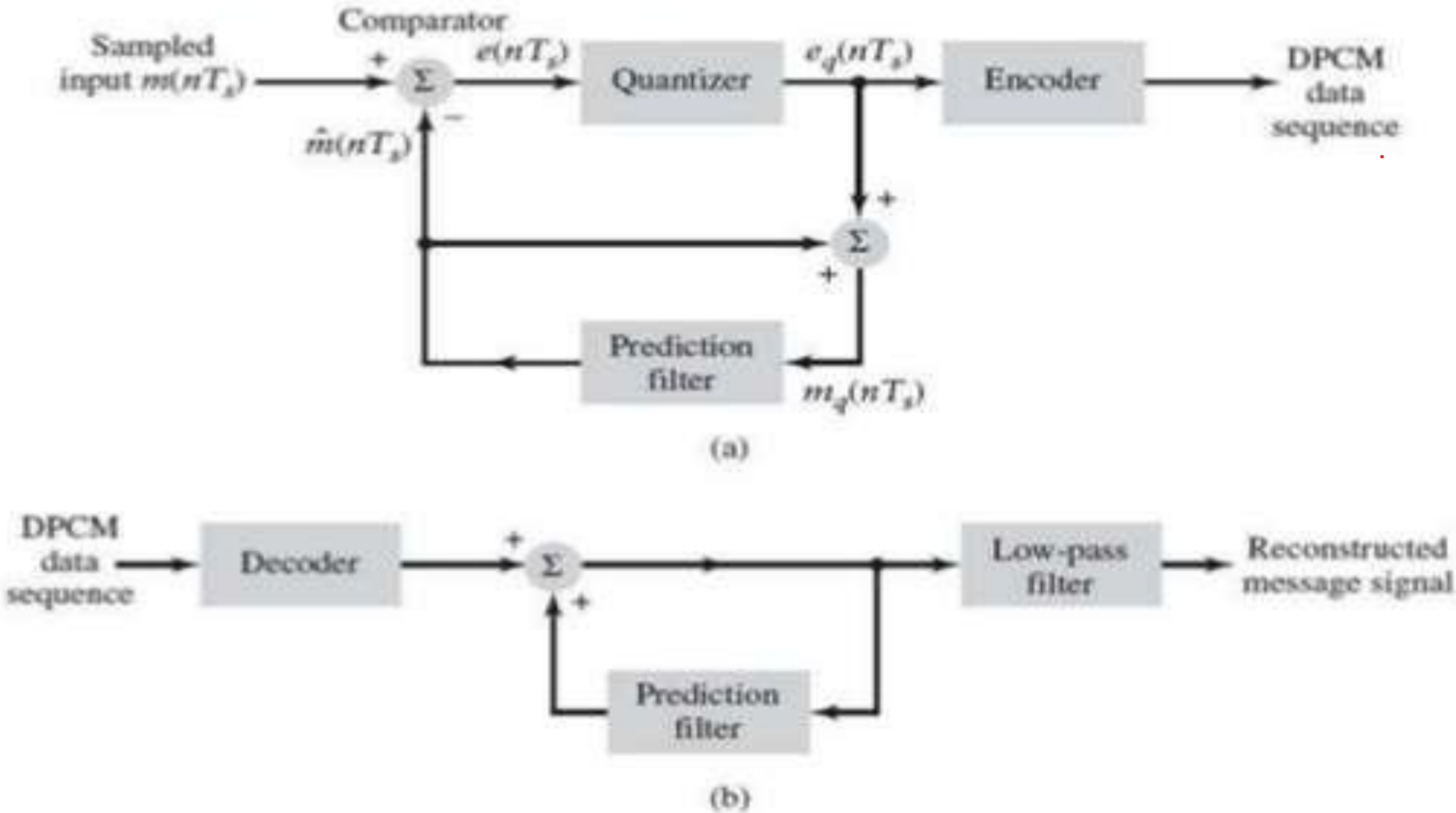



FIGURE 5.18 DPCM system: (*a*) Transmitter and (*b*) receiver.

# $e(nTs)=m(nTs)-\hat{m}(nTs)$

$$e_q(nT_s)=e(nT_s)+q_e(nT_s)$$

quantizing error. According to Fig. 8
led to the predicted value $\hat{m}(nT_s)$ to pro

$$m_q(nT_s)=\hat{m}(nT_s)+e_q(nT_s)$$

3) in (8.64), we get

$$m_q(nT_s)=\hat{m}(nT_s)+e(nT_s)+q_e(nT_s)$$

8.62) we observe that $\hat{m}(nT_s)+e(nT_s)$ is equa
e may rewrite Eq. (8.65) as follows

$$m_q(nT_s)=m(nT_s)+q_e(nT_s)$$

# Prediction Filter

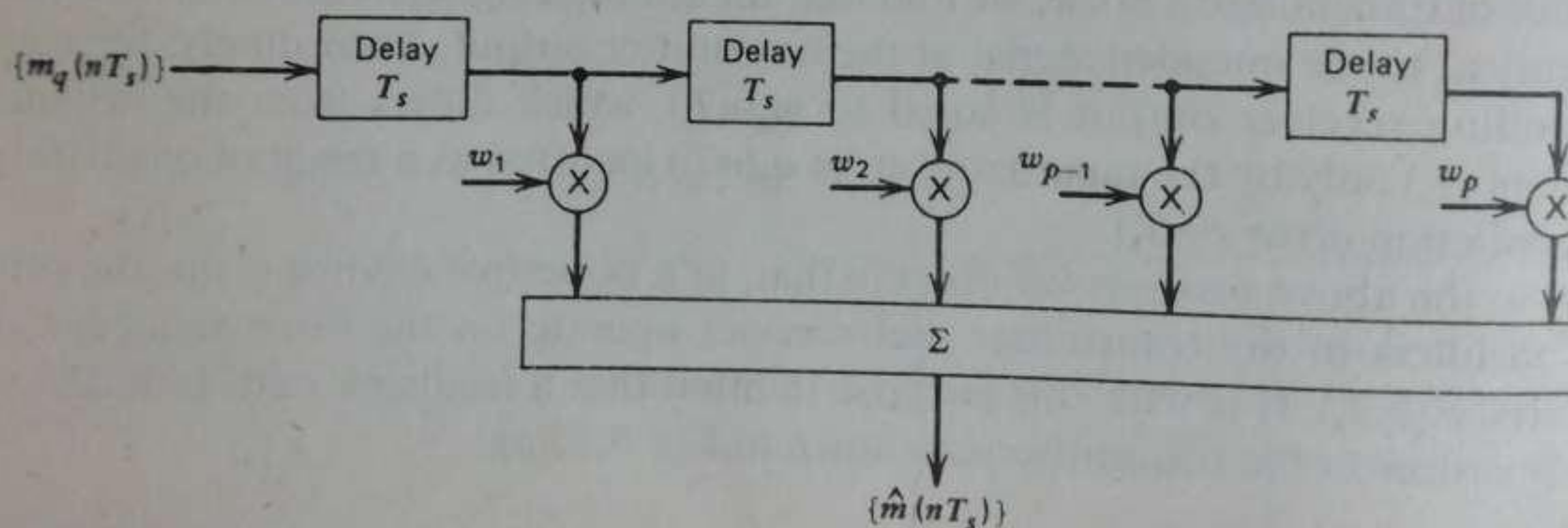



**Figure 8.13** Tapped-delay-line filter used as a prediction filter.

- Voice and video signals represented in PCM exhibit high correlation, which means that PCM signals contain redundant information. The result is an inefficient coding.
- By removing the PCM information redundancy a more efficient coded signal may be obtained. This is done using DPCM.
- In DPCM a linear prediction is performed on samples of a message signal m($kT_s$)=$m$($k$), then the prediction error $e(k) = m(k) - \hat{m}(k)$ is computed and fed to a quantizer to obtain the quantized value $e_q(k)=e(k)+q(k)$, as shown by Figure 3.28a. $q(k)$ is the quantization error.
- The input of the linear predictor of Figure 3.28a is $m_q(k) = \hat{m}(k) + e_q(k) = m(k) + q(k)$,

  which represents a quantized version of the input sample $m(k)$.
- If the prediction is well performed, then the variance of $e(k)$ will be much smaller than the variance of $m(k)$, which results into a smaller number of levels to quantize $e(k)$.
- The receiver as given by Figure 3.28b, consists of a decoder which produces $e_q(k)$, that is added to the output of a prediction filter identical to the one used in the transmitter. The result is the quantized message signal $m_q(k)$.

$$|V_2| = \frac{\log(1+\mu|V_1|)}{\log(1+\mu)} \rightarrow \mu\text{-law}$$

$\mu = 0 \rightarrow$ Uniform Quantisation

$$\left.\begin{aligned} |V_2| &= \frac{A|V_1|}{1+\log A} & 0 \le |V_1| \le \frac{1}{A} \\ &= \frac{1+\log(A|V_1|)}{1+\log A} & \frac{1}{A} \le |V_1| \le 1 \end{aligned}\right\} A\text{-law}$$

$A = 1 \rightarrow$ uniform Quantisation.

$$\hat{m}(nT_s) = \sum_{k=1}^{p} w_k m_q(nT_s - kT_s)$$

$$e(nT_s) = m(nT_s) - \sum_{k=1}^{p} w_k m_q(nT_s - kT_s) \quad \text{—— } (1)$$

# Delta Modulation (DM)

- In DM, the message signal is over-sampled to purposely increase correlation between adjacent samples.
- The DM provides a staircase approximation to the message signal $m(t)$ as shown in Figure 3.22.
- the difference $e[nT_s]=m[nT_s]-m_q[(n-1)T_s]$ is quantized into only two levels $\pm\Delta$.
- The error $e[nT_s]$ is quantized to give

  $e_q = \Delta \text{sgn}(e[nT_s])$.

  The quantity $e_q$ is then used to compute the new quantized level

  $m_q[nT_s]=m_q[(n-1)T_s]+e_q[nT_s]$
- In DM the quantization levels are represented by two symbols: 0 for $-\Delta$ and 1 for $+\Delta$. In fact the coding process is performed on $e_q$.
- The main advantage of DM is its simplicity as shown by Figure 3.23.

# Delta Modulation (Cont'd)

- The transmitter of a DM system (Figure 3.23a) is given by a comparator, a one-bit quantizer, an accumulator, and an encoder.
- The receiver of a DM system (Figure 3.23b) is given by a decoder, an accumulator, and a low-pass filter.
- DM is subject to two types of quantization error: Slope overload distortion and granular noise (see Figure 3.24).
- Slope overload distortion is due to the fact that the staircase approximation $m_q(t)$ can't follow closely the actual curve of the message signal $m(t)$. In order for $m_q(t)$ to follow closely $m(t)$, it is required that

$$\frac{\Delta}{T_s} \geq \max \left| \frac{dm(t)}{dt} \right|$$

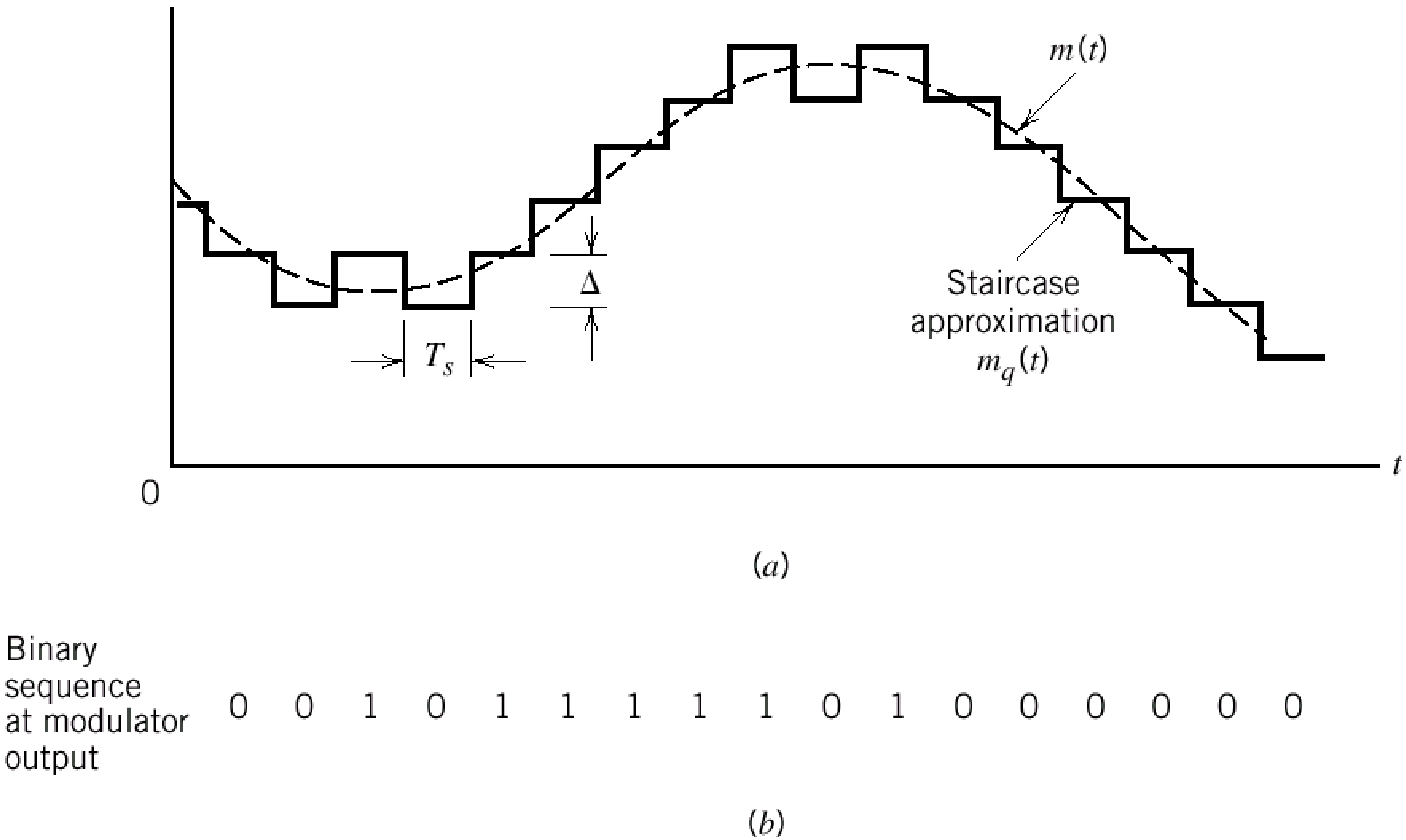
be satisfied. Otherwise, step-size $\Delta$ is too small for the staircase approximation $m_q(t)$ to follow $m(t)$.

# Delta Modulation (Cont'd)

- In contrast to slope-overload distortion, granular noise occurs when $\Delta$ is too large relative to the local slope characteristics of $m(t)$. granular noise is similar to quantization noise in PCM.
- It seems that a large $\Delta$ is needed for rapid variations of $m(t)$ to reduce the slope-overload distortion and a small $\Delta$ is needed for slowly varying $m(t)$ to reduce the granular noise. The optimum $\Delta$ can only be a compromise between the two cases.
- To satisfy both cases, an adaptive DM is needed, where the step size $\Delta$ can be adjusted in accordance with the input signal $m(t)$.

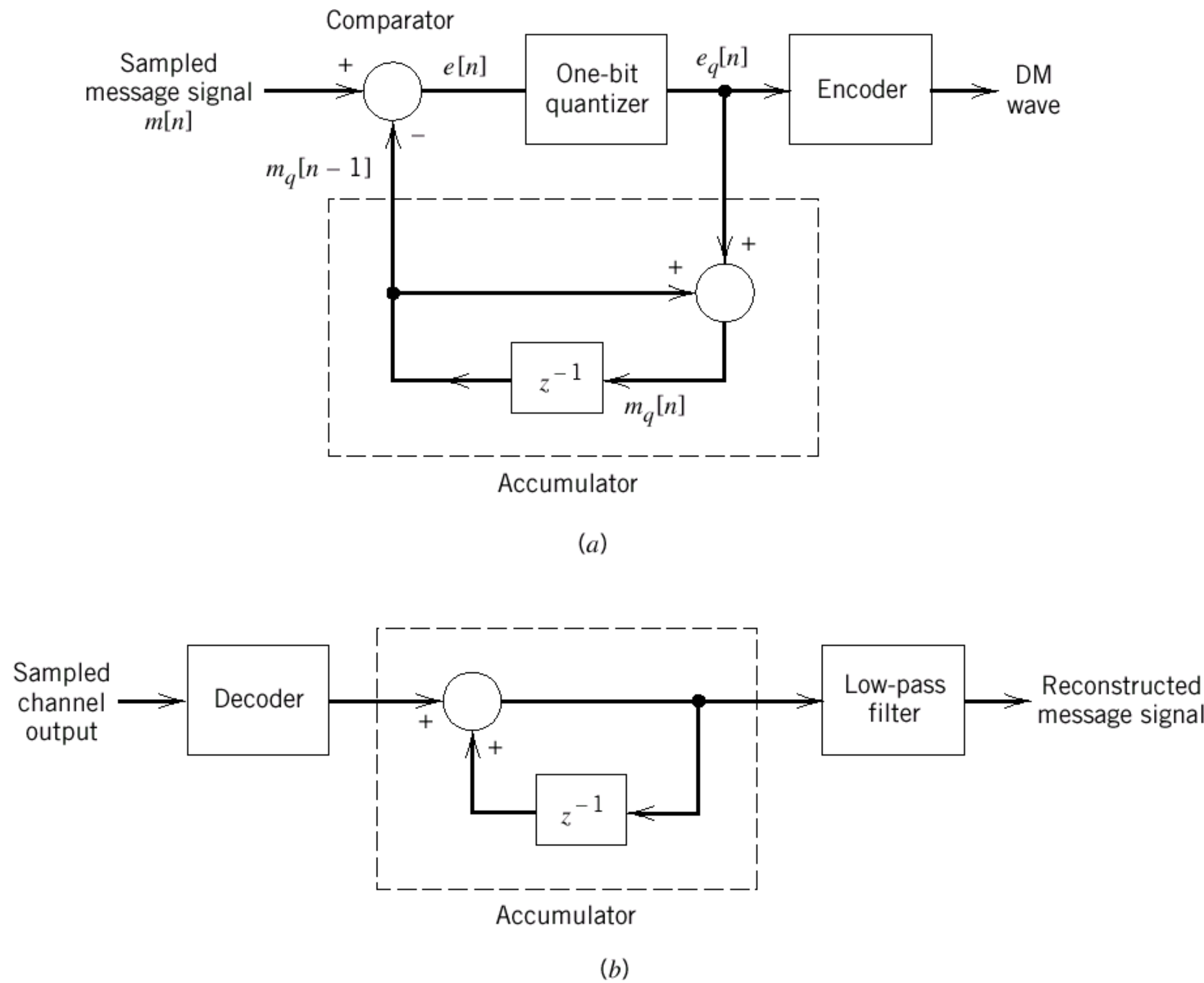
# Figure 3.22

Illustration of delta modulation.



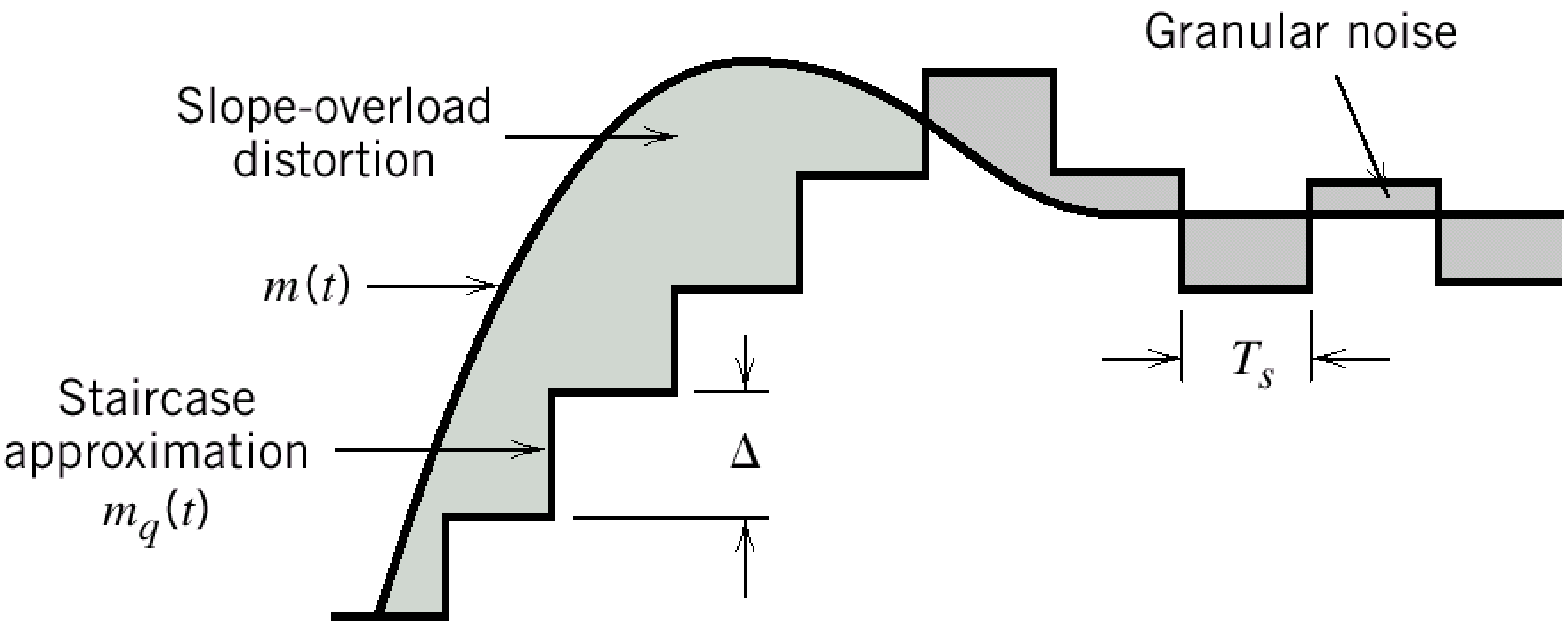
# Figure 3.23

DM system.
(a) Transmitter.
(b) Receiver.

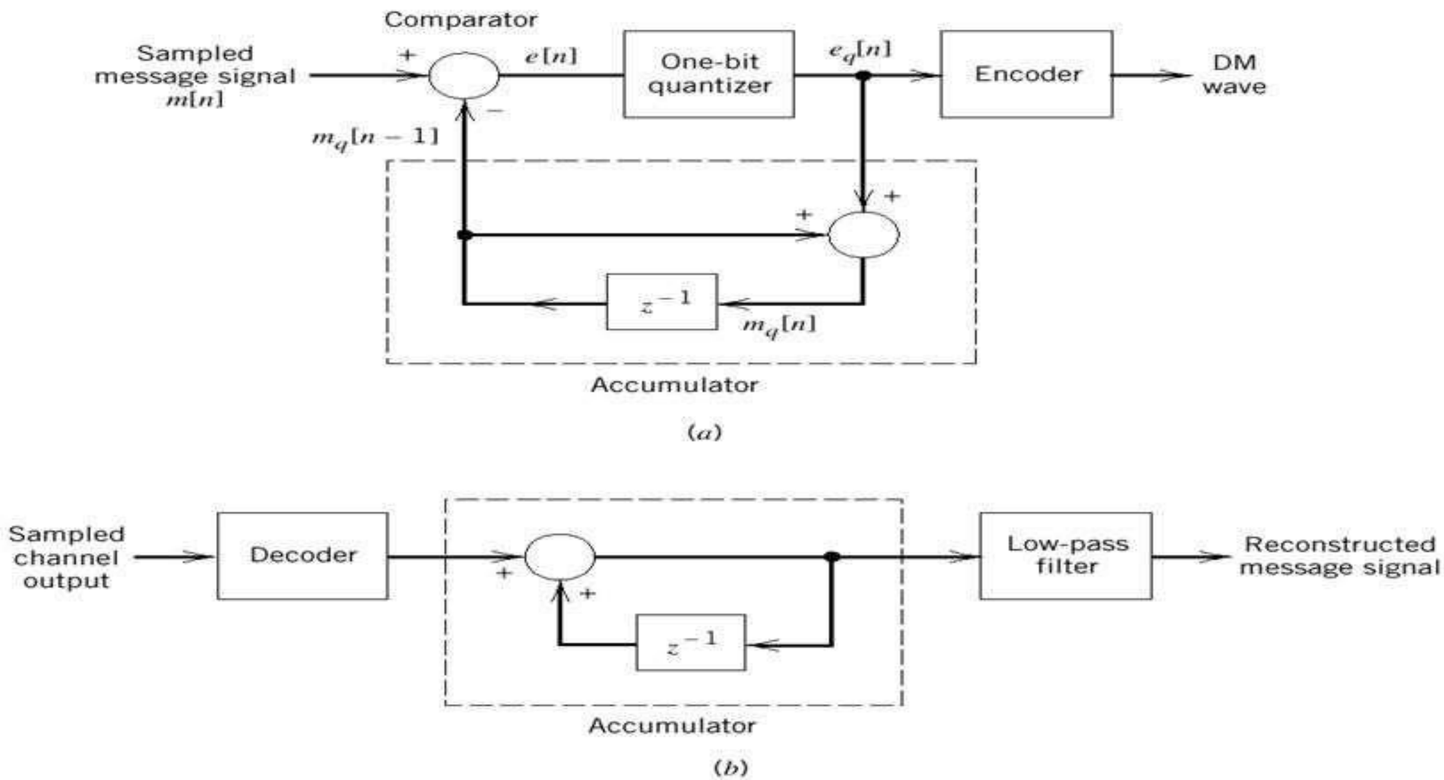


# Figure 3.24

Illustration of the two different forms of quantization error in delta modulation.



# Delta Modulation




DM system. (*a*) Transmitter. (*b*) Receiver.

# DELTA MODULATION (Quantization Errors)

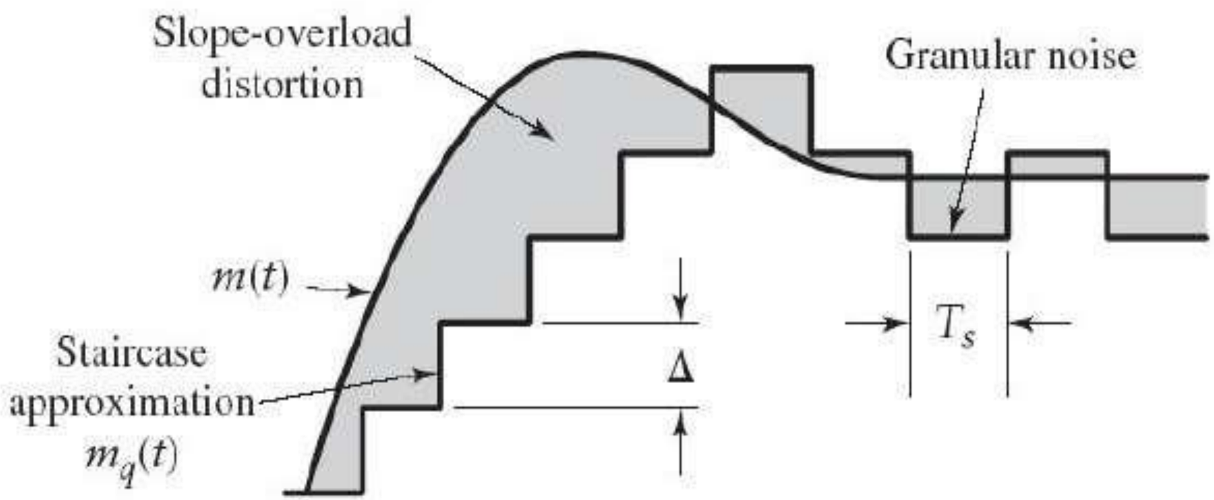
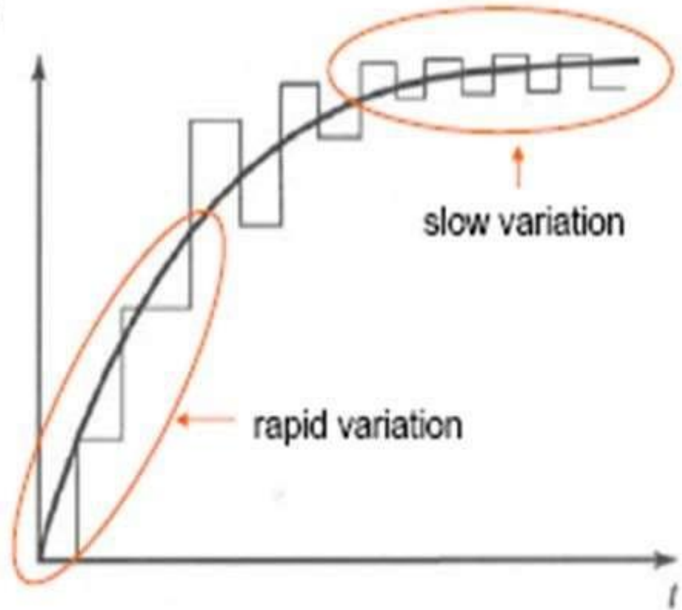



**FIGURE 5.16** Illustration of quantization errors, slope-overload distortion and granular noise, in delta modulation.

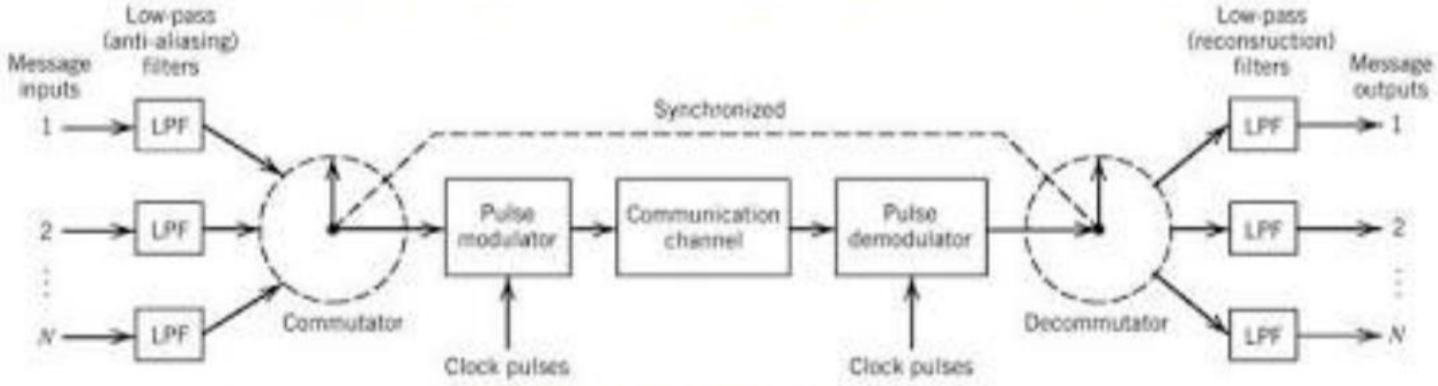
# Adaptive Delta Modulation (ADM)

- Main idea: change step size according to changes in the input signal.
- If the input changes rapidly -> large step size. If the input changes slowly -> small step size.
- How to implement step size change?
- Simple solution: if two successive outputs have the same sign -> increase step size; if they are of opposite sign -> decrease step size.



# Thank You!!!!

# Conceptual diagram of multiplexing-demultiplexing.




**PAM TDM System**