Chapter 4
Digital Transmission

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P47 A2D CONVERSION

ANALOG-TO-DIGITAL CONVERSION

- PULSE CODE MODULATION (PCM)
- DELTA MODULATION (DM)

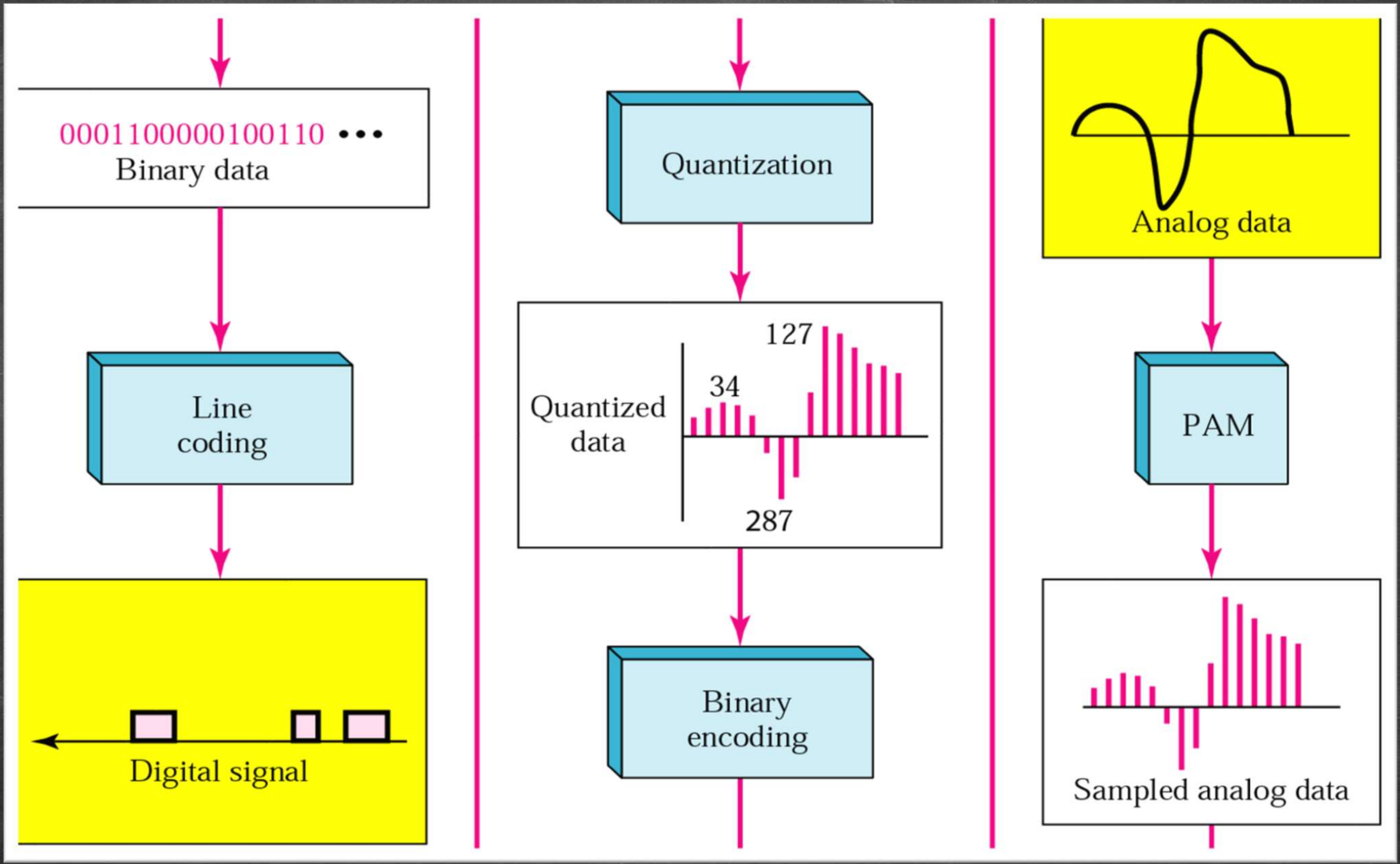
Pulse Code Modulation (PCM)

- The analog signal is sampled.
- The sampled signal is quantized.
- The quantized values are encoded as streams of bits.

sampling หาคือ ตัวอย่าง

Pulse Code Modulation (PCM)

Sender



Pulse Code Modulation (PCM)

- Pulse Amplitude Modulation has some applications, but it is not used by itself in data communication.
- However, it is the first step in another very popular conversion method called Pulse Code Modulation.

Pulse Code Modulation (PCM)

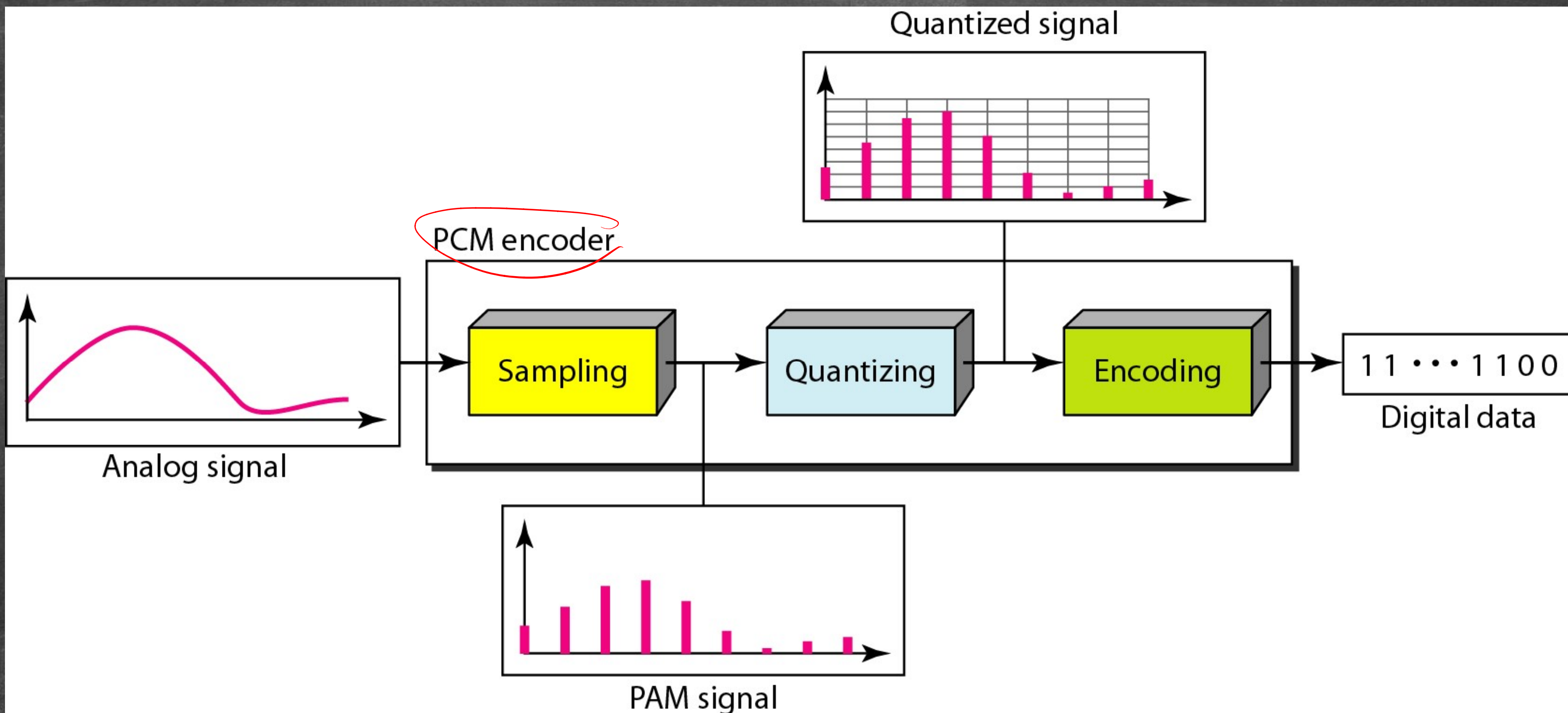
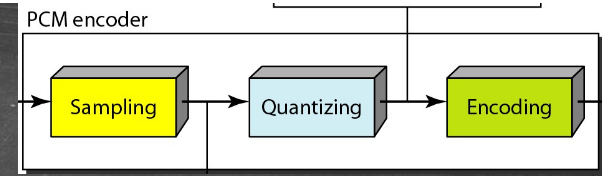


Figure 4.21 Components of PCM encoder

Sampling



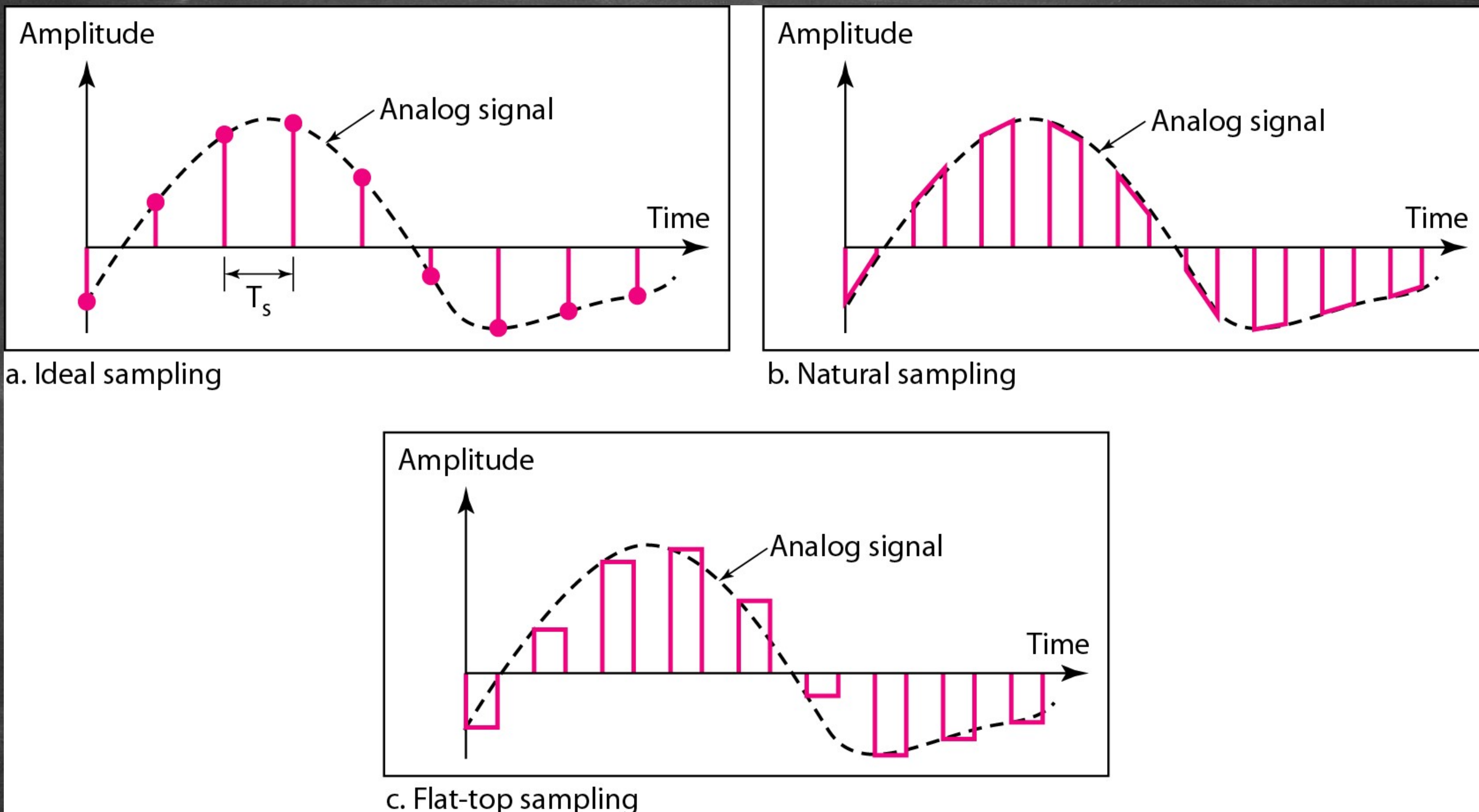
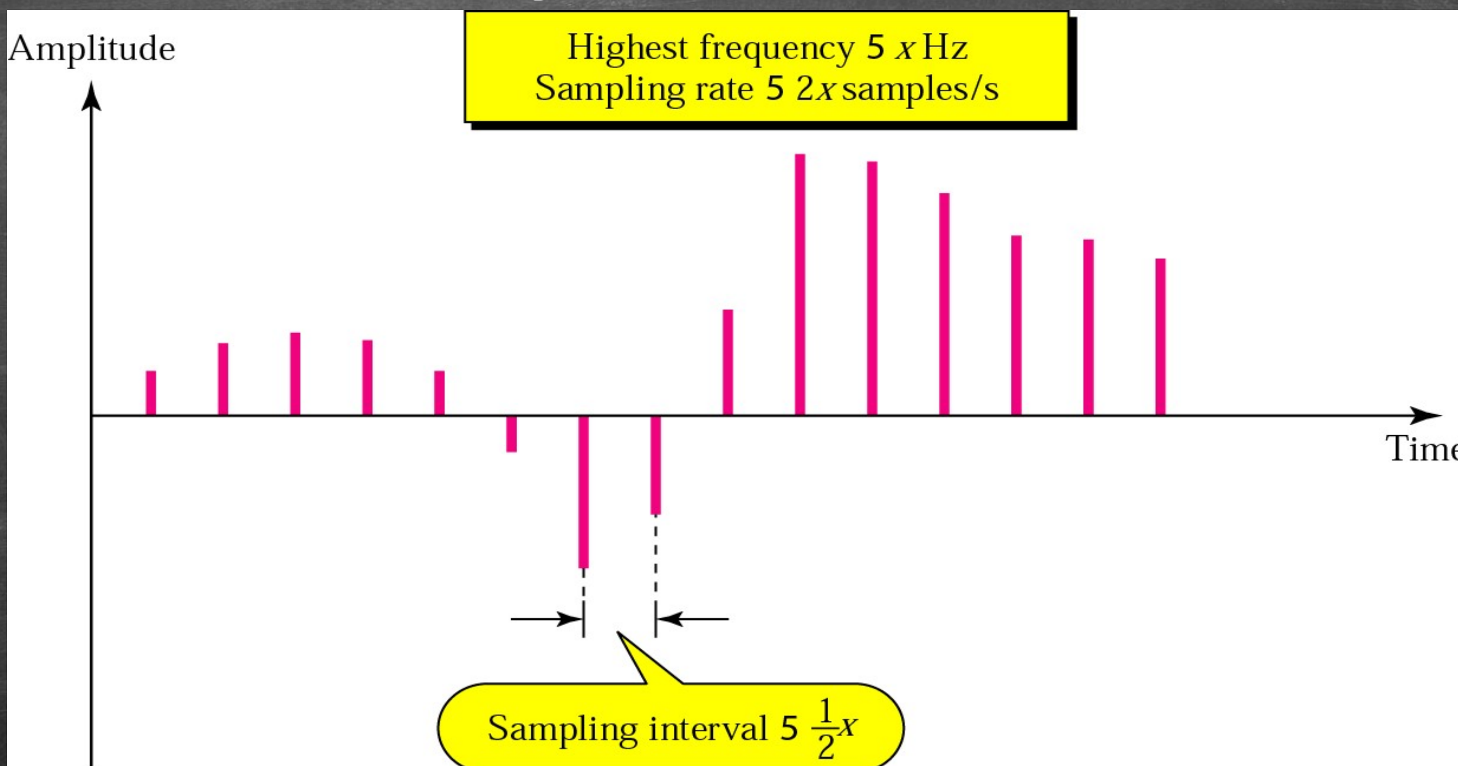


Figure 4.22 Three different sampling methods for PCM

Sampling

- According to the Nyquist theorem, the sampling rate must be at least 2 times the highest frequency contained in the signal.



Sampling

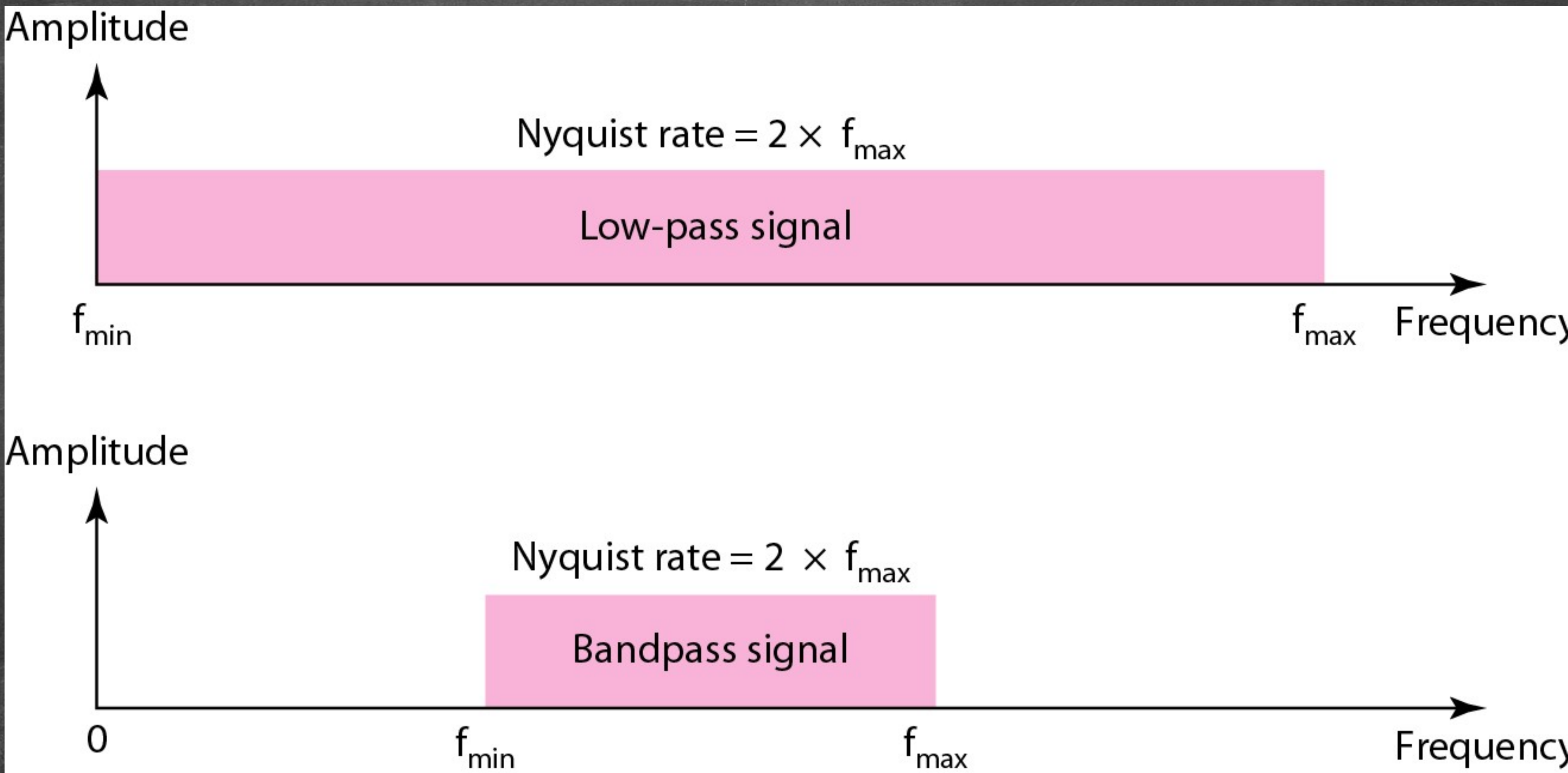


Figure 4.23 Nyquist sampling rate for low–pass and bandpass signals

Example 4.6

- *For an intuitive example of the Nyquist theorem, let us sample a simple sine wave at three sampling rates: f_s = 4f (2 times the Nyquist rate), f_s = 2f (Nyquist rate), and f_s = f (one-half the Nyquist rate). Figure 4.24 shows the sampling and the subsequent recovery of the signal.*
- *It can be seen that sampling at the Nyquist rate can create a good approximation of the original sine wave (part a). Oversampling in part b can also create the same approximation, but it is redundant and unnecessary. Sampling below the Nyquist rate (part c) does not produce a signal that looks like the original sine wave.*

Example 4.6

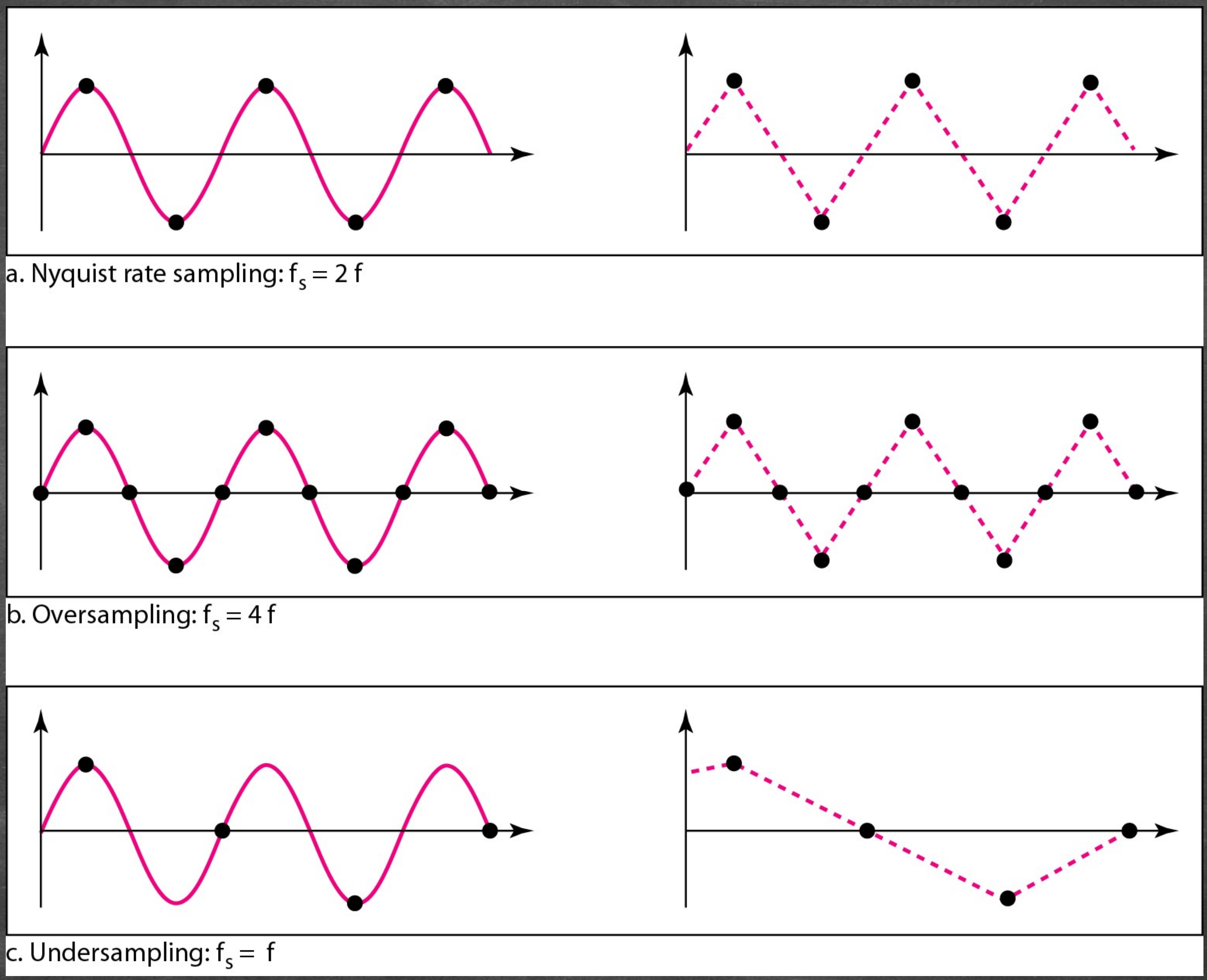


Figure 4.24 Recovery of a sampled sine wave for different sampling rates

Example 4.7

- *Consider the revolution of a hand of a clock. The second hand of a clock has a period of 60 s.*
  - *According to the Nyquist theorem, we need to sample the hand every 30 s (T_s = T or f_s = $2f$). In Figure 4.25a, the sample points, in order, are 12, 6, 12, 6, 12, and 6.*
  - *In part b, we sample at double the Nyquist rate (every 15 s). The sample points are 12, 3, 6, 9, and 12. The clock is moving forward.*
  - *In part c, we sample below the Nyquist rate (T_s = T or f_s = f). The sample points are 12, 9, 6, 3, and 12. Although the clock is moving forward, the receiver thinks that the clock is moving backward.*

Example 4.7

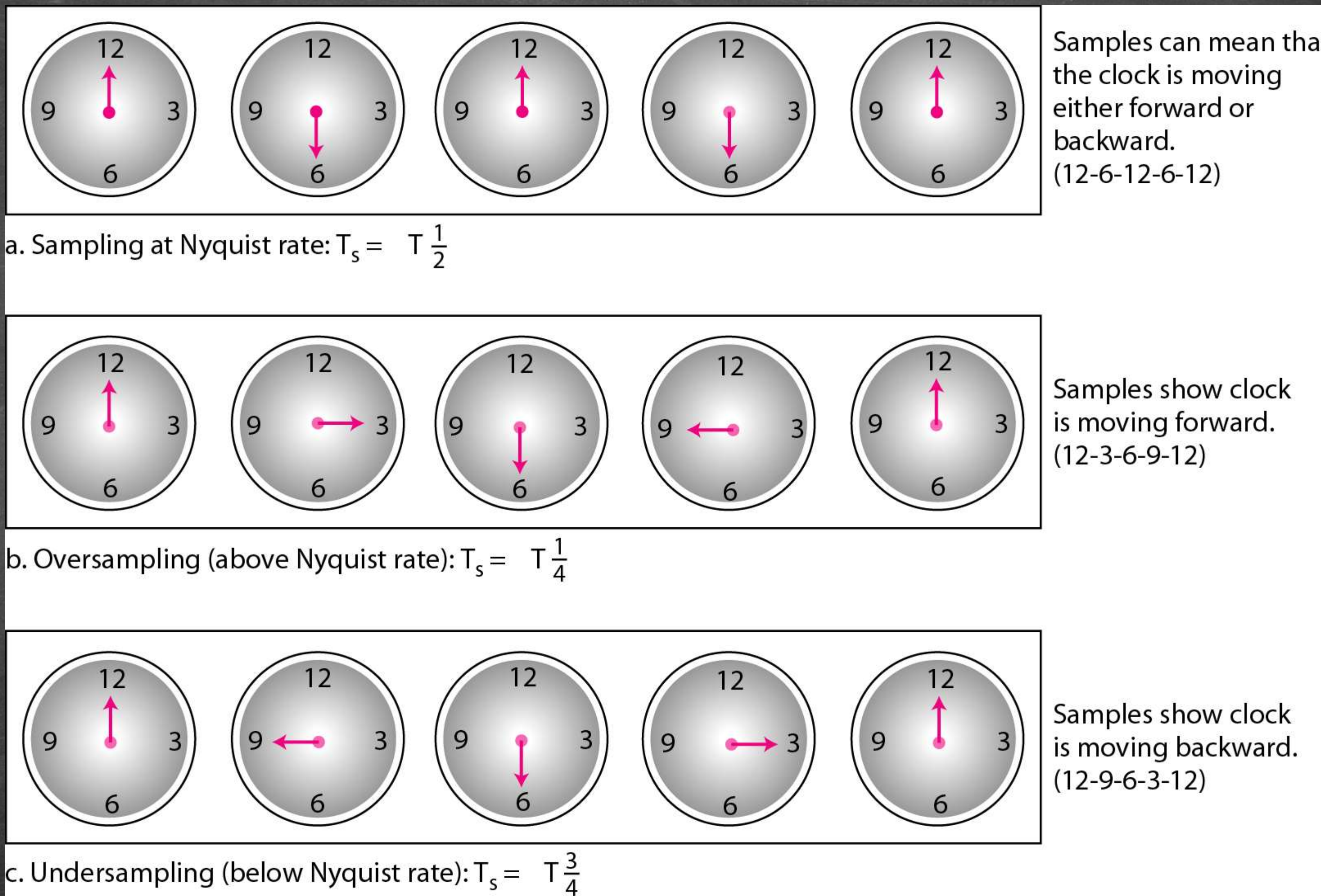


Figure 4.25 Sampling of a clock with only one hand

Example 4.9

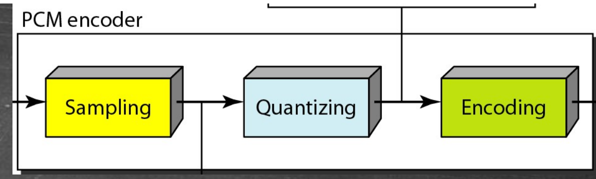
- ***Telephone companies digitize voice by assuming a maximum frequency of 4000 Hz. The sampling rate therefore is 8000 samples per second.***

Example 4.10

- *A complex low-pass signal has a bandwidth of 200 kHz. What is the minimum sampling rate for this signal?*
- Solution
  - *The bandwidth of a low-pass signal is between 0 and f, where f is the maximum frequency in the signal. Therefore, we can sample this signal at 2 times the highest frequency (200 kHz). The sampling rate is therefore 400,000 samples per second.*

Example 4.11

- ***A complex bandpass signal has a bandwidth of 200 kHz. What is the minimum sampling rate for this signal?***
- *Solution*
  - ***We cannot find the minimum sampling rate in this case because we do not know where the bandwidth starts or ends. We do not know the maximum frequency in the 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.

Quantization

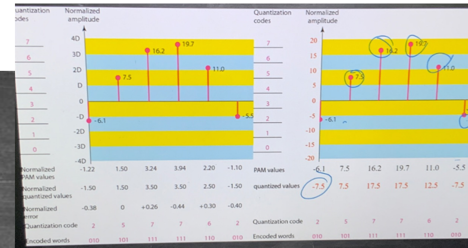
- amplitudes of the signal $[V_{max}, V_{min}]$
- divide the range into L zones

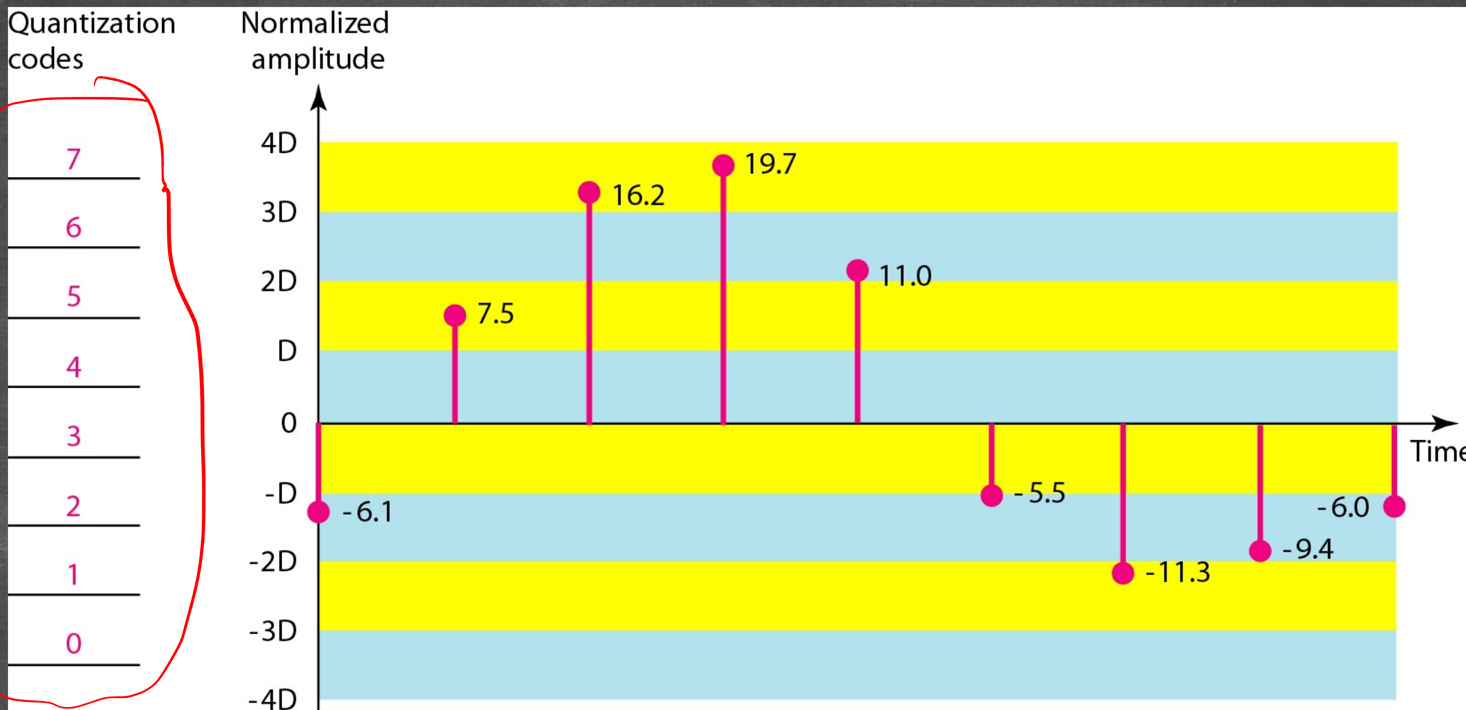
$$\Delta = (V_{max} - V_{min}) / L$$

- assign quantized values of 0 to L-1 to the midpoint of each zone
- approximate the value of the sample amplitude to the quantized values

$\Delta = (20 - (-20)) / 8 \Rightarrow 5$

Quantization





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ขนาด 5 จากครึ่งแบบ

แบ่งช่วง ตามจำนวน

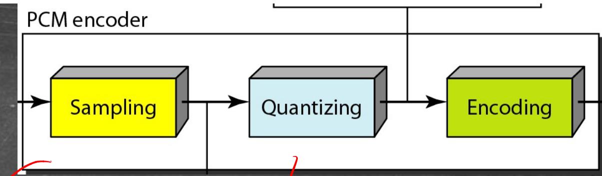
ความต้องการ

ที่จะแบ่ง

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

Figure 4.26 Quantization and encoding of a sampled signal

Encoding



$$n_b = \log_2 L$$

$$\text{Bit rate} = f_s \times n_b$$

Example 4.14

- ***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. So the sampling rate and bit rate are calculated as follows:***

$$\text{Sampling rate} = 4000 \times 2 = 8000 \text{ samples/s}$$
$$\text{Bit rate} = 8000 \times 8 = 64{,}000 \text{ bps} = 64 \text{ kbps}$$

Original Signal Recovery

- Note that we can always change a band-pass signal to a low-pass signal before sampling.
- In this case, the sampling rate is twice the bandwidth.

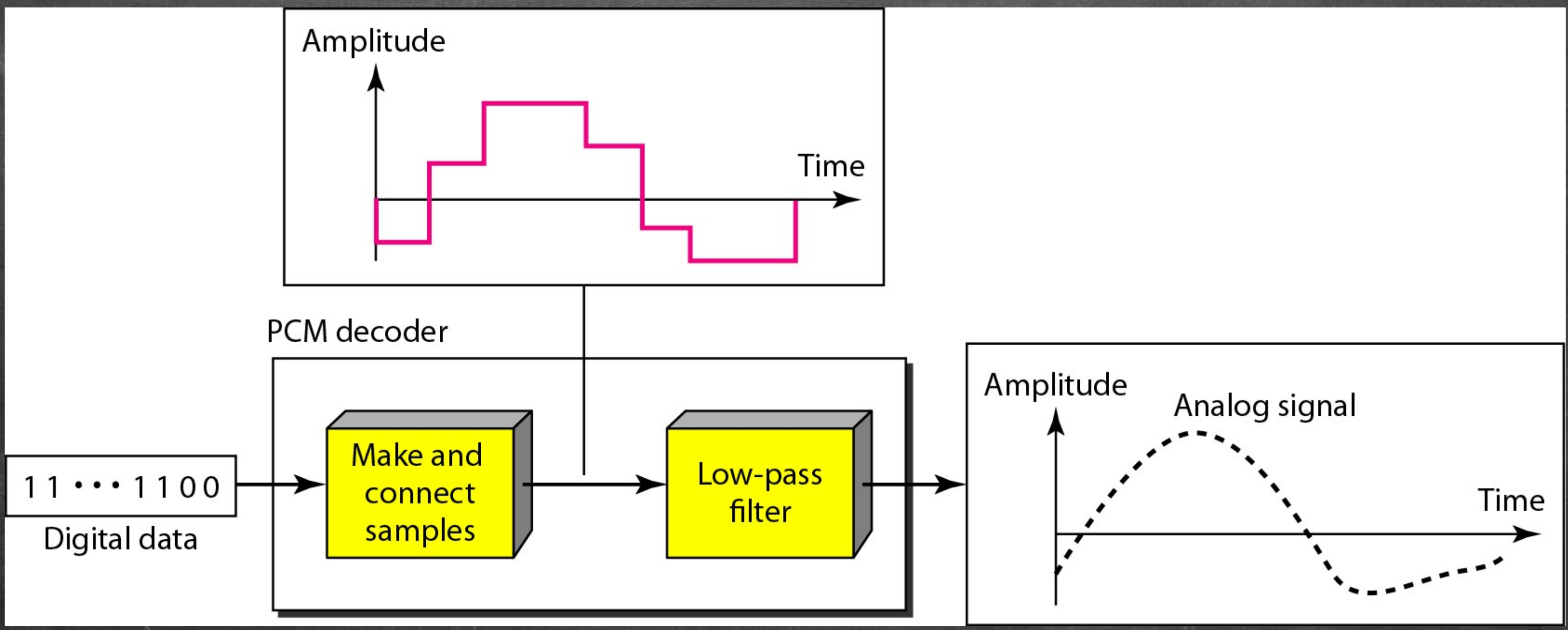
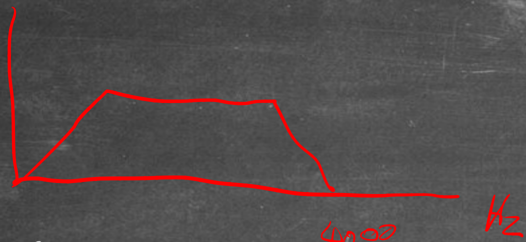


Figure 4.27 Components of a PCM decoder



Example 4.15

- ***We have a low-pass analog signal of 4 kHz. If we send the analog signal, we need a channel with a minimum bandwidth of 4 kHz. If we digitize the signal and send 8 bits per sample, we need a channel with a minimum bandwidth of 8 × 4 kHz = 32 kHz.***

$f_s = 2 \times 4000 = 8000$

$N = 8 \times 8000 = 64$ kbps

$S = c N \frac{1}{r}$

$= \frac{1}{2} \times 64000 \times \frac{1}{1}$

$= 32$ kbs

Delta modulation (DM)

- developed to reduce the complexity of PCM
  - PCM finds the value of the signal amplitude for each sample
  - DM finds the change from the previous sample.

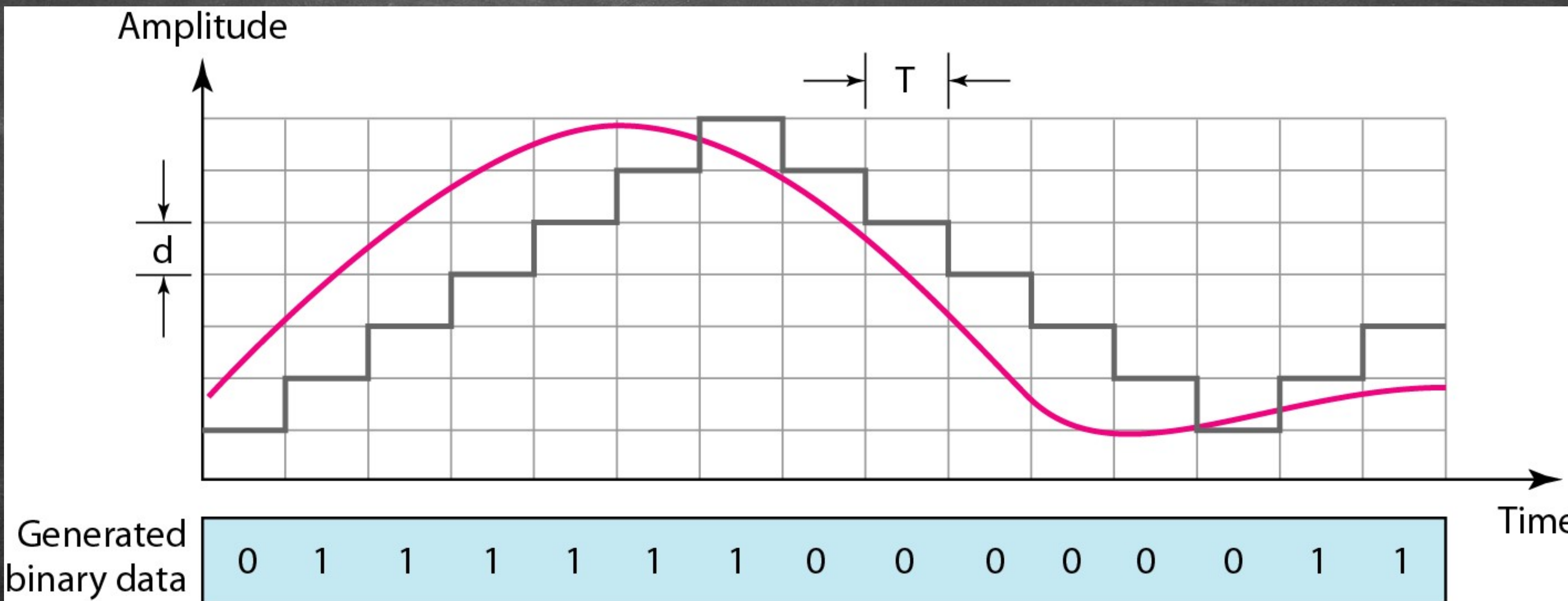


Figure 4.28 The process of delta modulation

Delta modulation (DM)

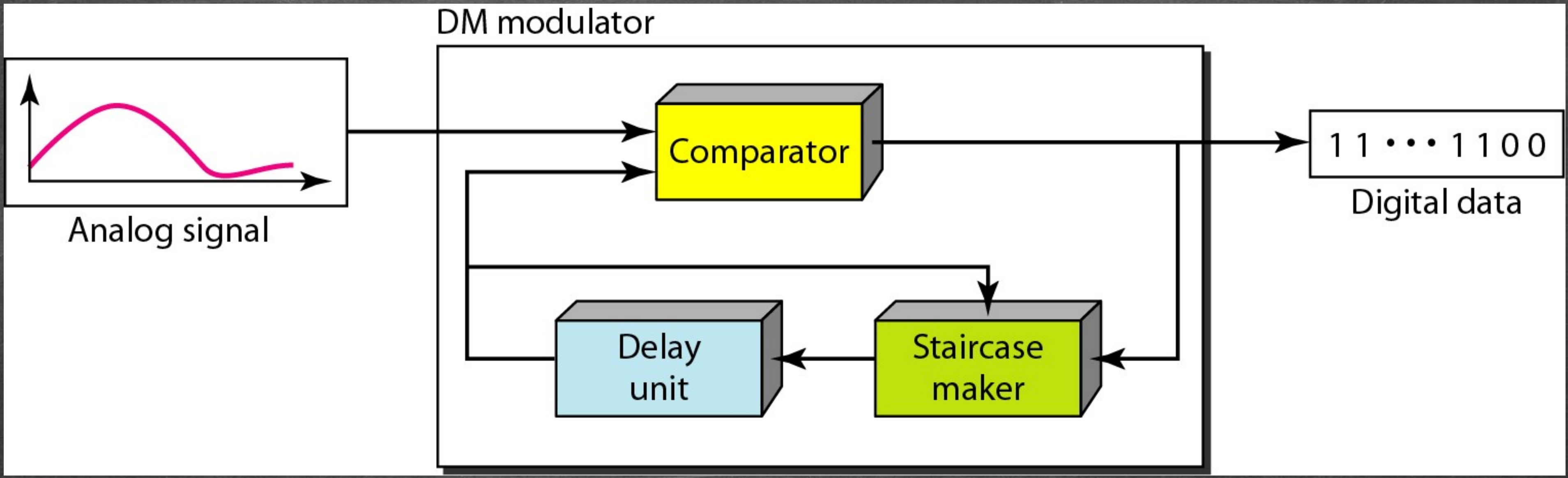


Figure 4.29 Delta modulation components

Delta modulation (DM)

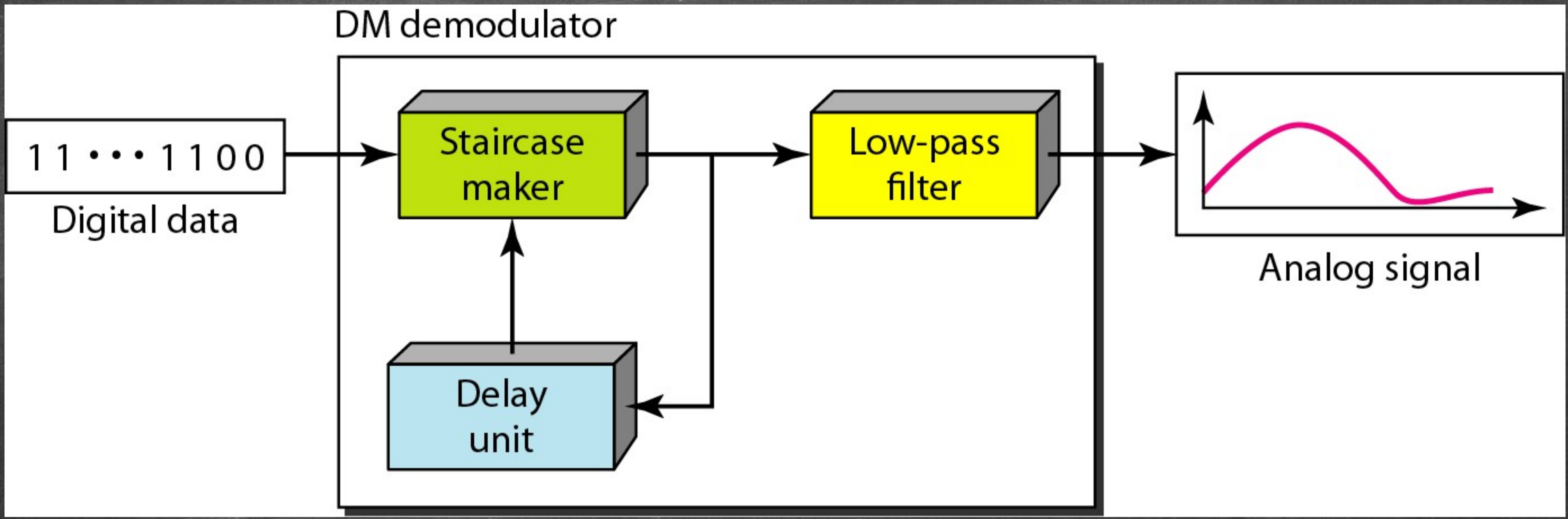


Figure 4.30 Delta demodulation components