7- Sampling & Quantization

ITNT323

Sampling & Quantization

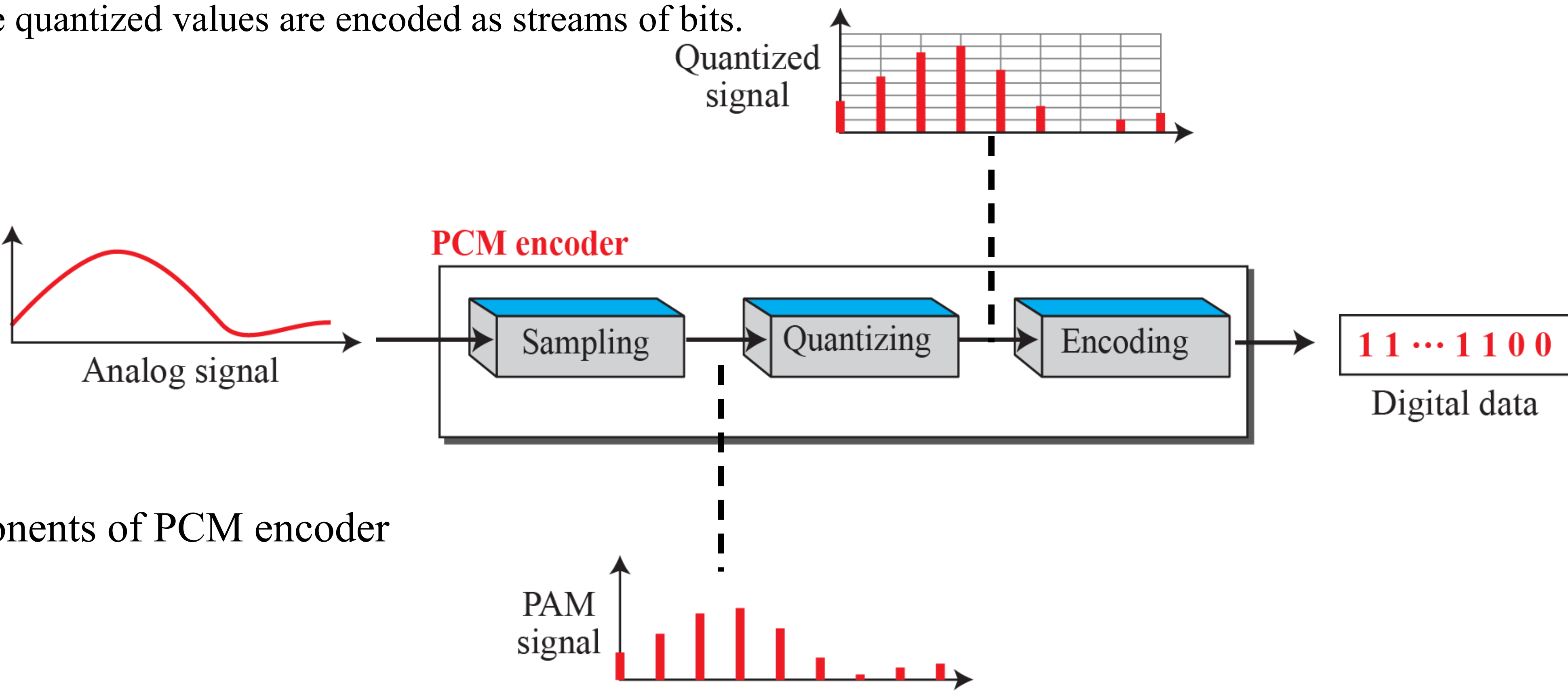
Sampling :–
Obtain signal values at equal intervals (T), (at discrete-time).

Quantization :–
Approximate samples to certain values (L), (with discrete-amplitude).

1 - Pulse Code Modulation (PCM)

The most common technique to change an analog signal to digital data (digitization) is called **p**ulse **c**ode **m**odulation (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.



Components of PCM encoder

(PCM)

Sampling

- **Nyquist Theorem**:
  - “If a signal is sampled at regular intervals at a rate higher than twice the highest signal frequency, the samples contain all information in original signal”
  - eg. 4000Hz voice data, requires 8000 sample / *sec*

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

Sampling

First, we can sample a signal only if the signal is band-limited. In other words, a signal with an infinite bandwidth cannot be sampled.
Second, the sampling rate must be at least 2 times the highest frequency, not the bandwidth. If the analog signal is low-pass, the bandwidth and the highest frequency are the same value. If the analog signal is bandpass, the bandwidth value is lower than the value of the maximum frequency.
Next figure shows the value of the sampling rate for two types of signals.

Sampling

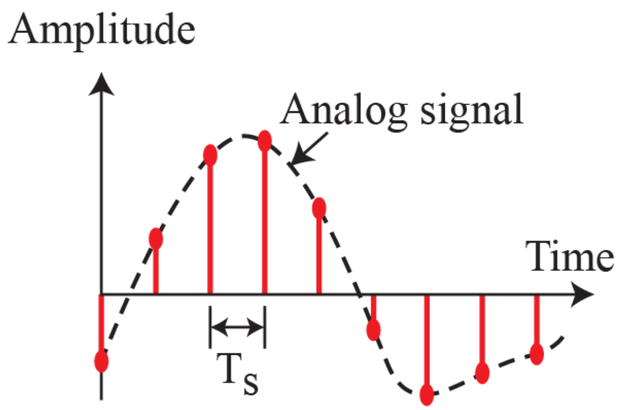
The first step in PCM is sampling. The analog signal is sampled every T_s *sec, where* T_s *is* the sample interval or period. The inverse of the sampling interval is called the *sampling rate or sampling frequency and denoted by* f_s *, where* $f_s = 1/T_s$ *.*

There are three sampling methods:

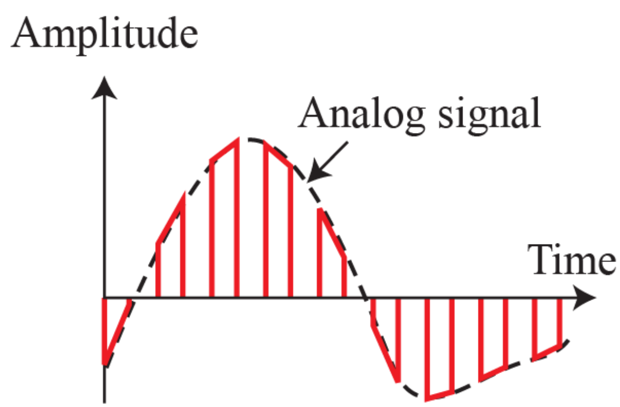
- **ideal**, pulses from the analog signal are sampled. This is an ideal sampling method and cannot be easily implemented.
- **natural**, a high-speed switch is turned on for only the small period of time when the sampling occurs. The result is a sequence of samples that retains the shape of the analog signal.
- **flat-top**. The most common sampling method, called *sample and hold,* however, creates flat-top samples by using a circuit.

(PCM)

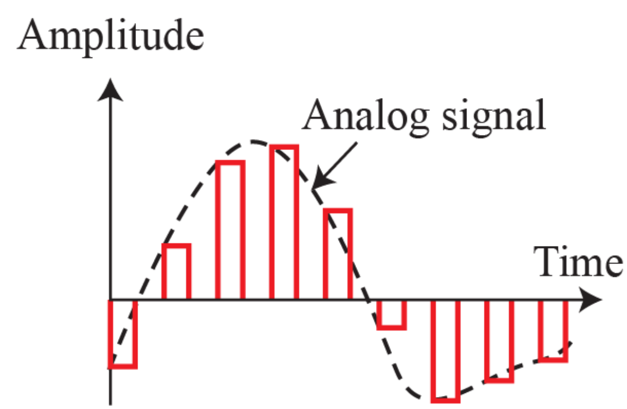
Three different sampling methods for PCM



a. Ideal sampling



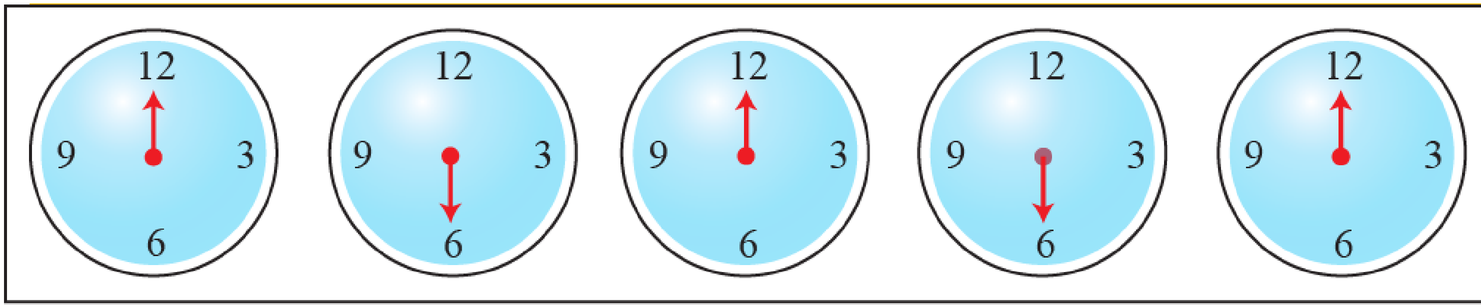
b. Natural sampling



c. Flat-top sampling

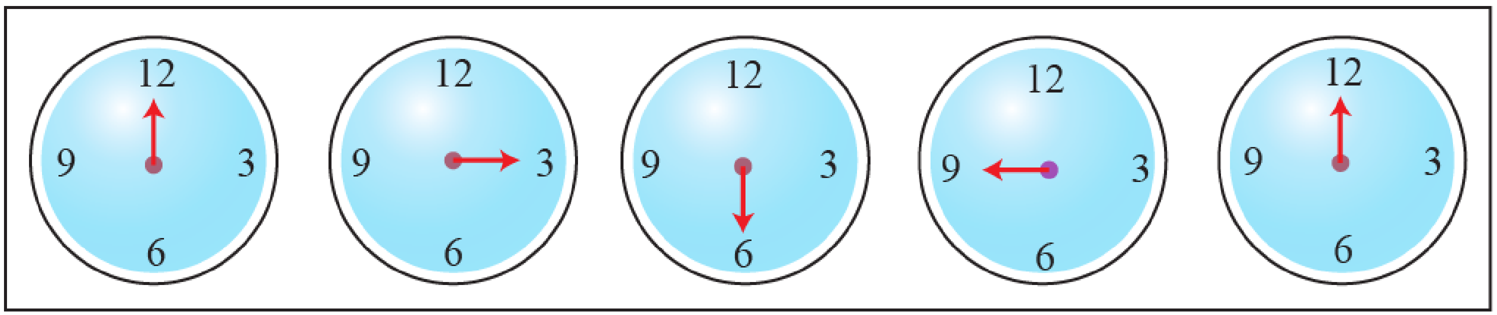
(PCM)

Sampling of clock with only one hand.



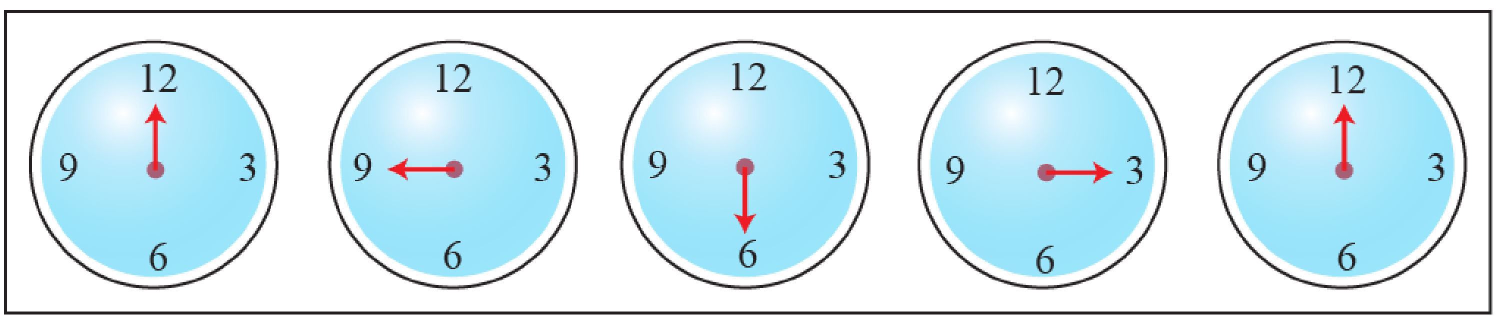
Samples can mean that the clock is moving either forward or backward. (12-6-12-6-12)

a. Sampling at Nyquist rate: $T_s = T\frac{1}{2}$



Samples show clock is moving forward. (12-3-6-9-12)

b. Oversampling (above Nyquist rate): $T_s = T\frac{1}{4}$



Samples show clock is moving backward. (12-9-6-3-12)

c. Undersampling (below Nyquist rate): $T_s = T\frac{3}{4}$

Example

An example related to previous example is the seemingly backward rotation of the wheels of a forward-moving car in a movie. This can be explained by undersampling. A movie is filmed at 24 frames per second. If a wheel is rotating more than 12 times per second, the undersampling creates the impression of a backward rotation.

Example

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.

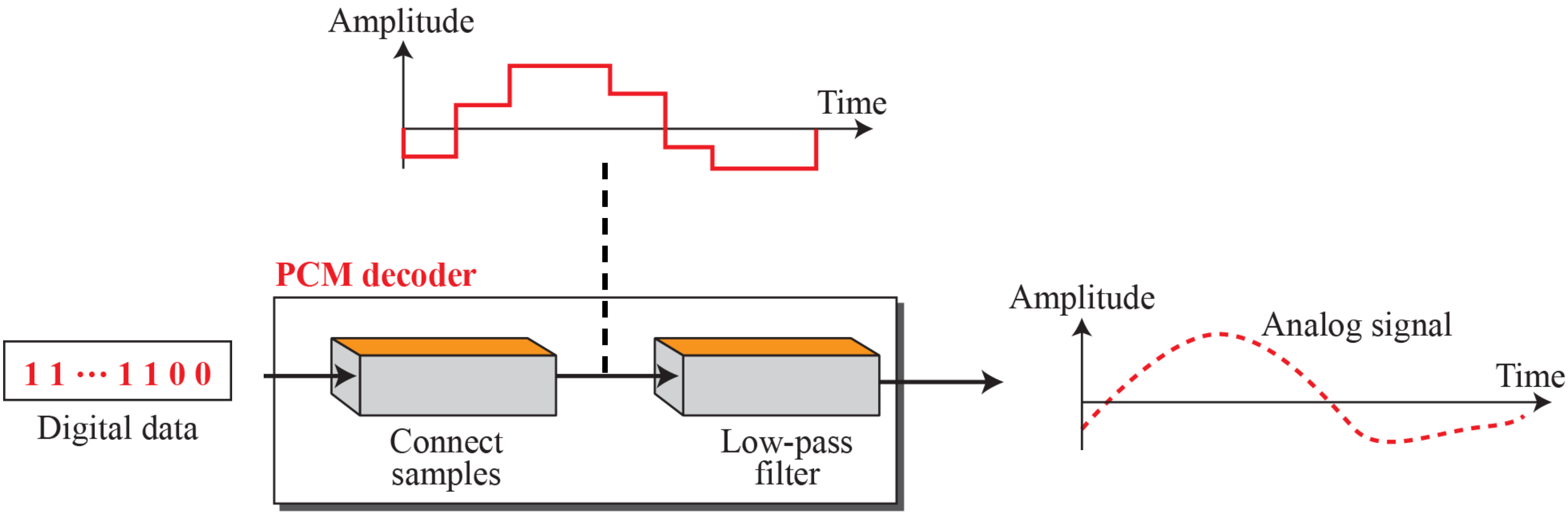
Example

A complex bandpass signal has a bandwidth of 200 kHz. What is the minimum sampling rate for this signal?

Solution

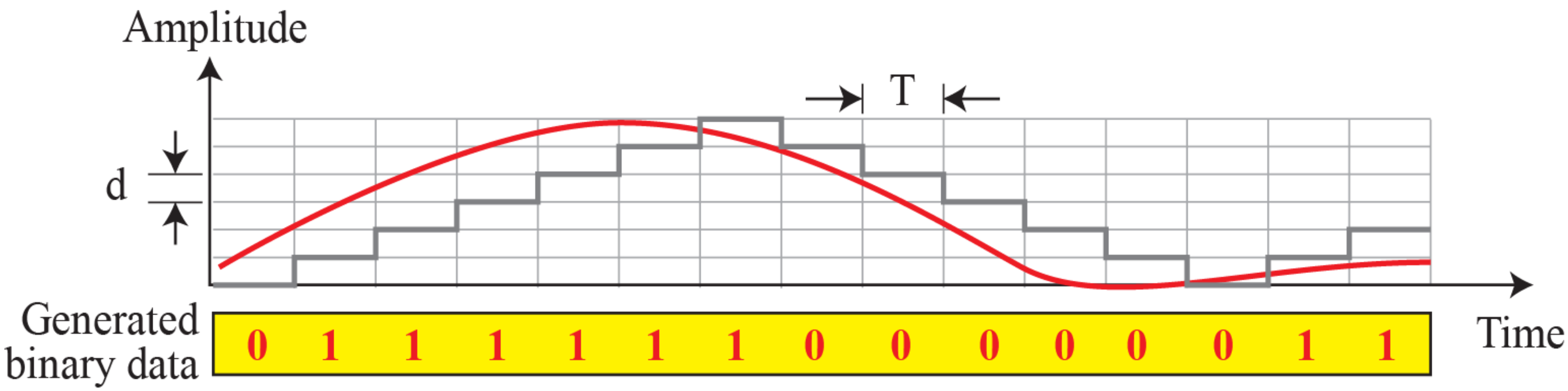
We cannot find the minimum sampling rate in this case because we do not know where the bandwidth starts or ends. We do not know the maximum frequency in the signal.

Components of a PCM decoder

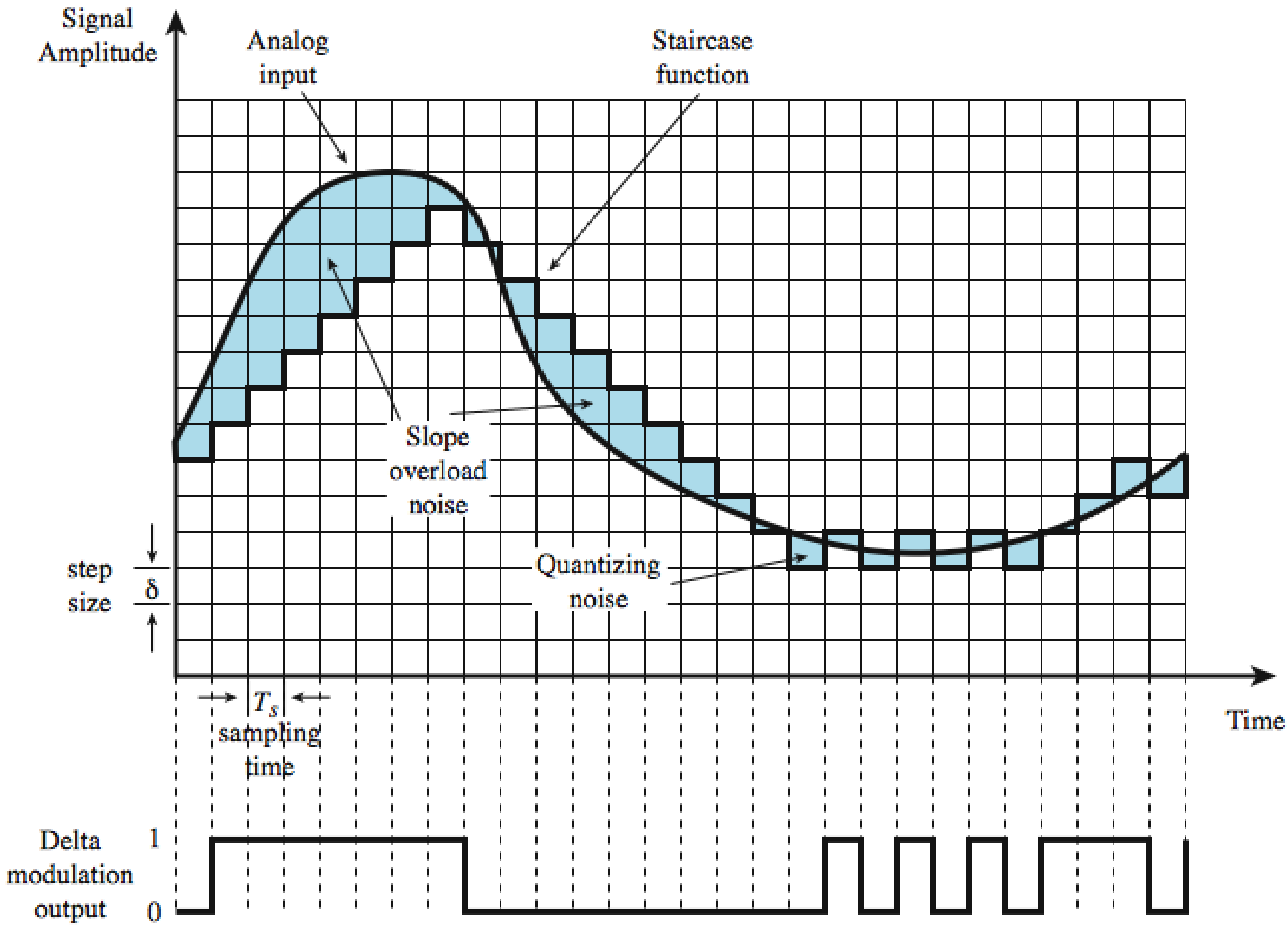


(DM)

The process of delta modulation

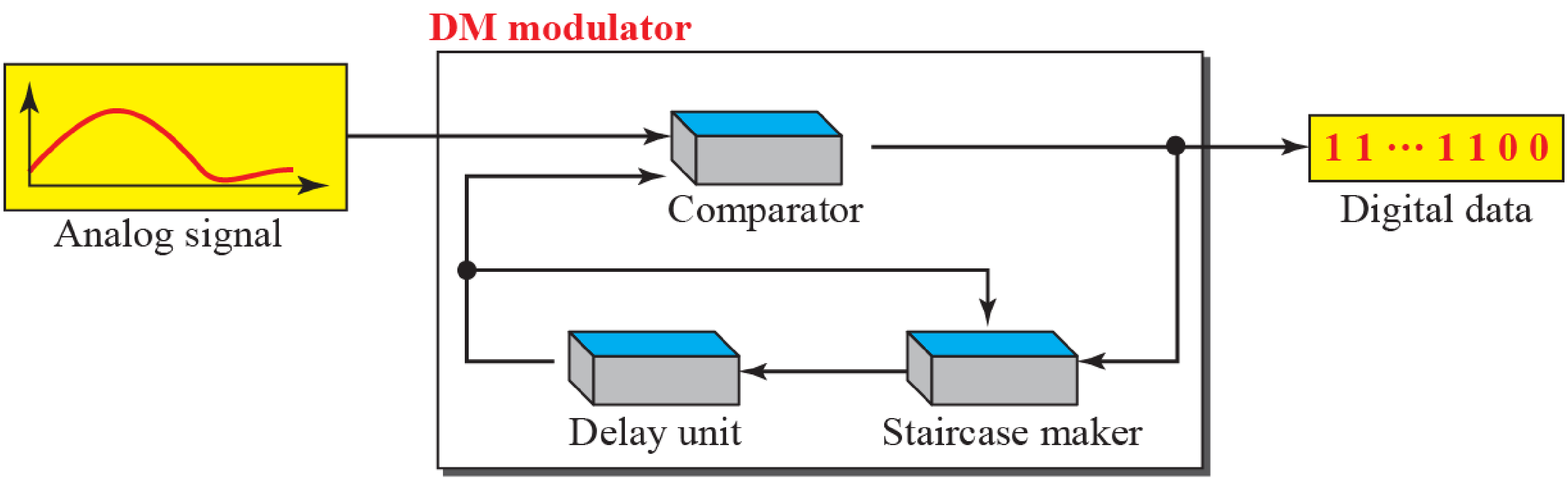


Delta Modulation Example



(DM)

Delta modulation components



The modulator, at each sampling interval, compares the value of the analog signal with the last value of the staircase signal. If the amplitude of the analog signal is larger, the next bit in the digital data is 1; otherwise, it is 0. The output of the comparator, however, also makes the staircase itself. If the next bit is 1, the staircase maker moves the last point of the staircase signal δ up; if the next bit is 0, it moves it δ down.
Note that we need a delay unit to hold the staircase function for a period between two comparisons.

(DM)

Demodulator

The demodulator takes the digital data and, using the staircase maker and the delay unit, creates the analog signal. The created analog signal, however, needs to pass through a low-pass filter for smoothing. Next figure shows the schematic diagram.