NETWORK AND TELECOMMUNICATION

ACADEMIC BACKGROUNDS:

- 1987-1993 Georgia University of Technology (Former USSR) **Specialize: Radio Transmitting Device of Satellite Telecommunication Systems** (Master of Science).
- 1997-1998 Advanced course at the Saint–Petersburg State University of Technology in computer simulation of ground stations Modem for Sputnic communication (Russia).

PREVIOUS EMPLOYMENT:

- 2002-2018 The World Bank Cambodia (IT Analyst, Client Services).
- 1999 -2001 Worked as Systems Engineer at VIRTU International Limited.
- 1995 -1997 Worked as assistant manager in operation and technical department at CAMINTEL.
- 1993 – 1995 Worked as engineer in Operations and Technical Department in HUB-station (ex-UNTAC Networks) at Ministry of Post and Telecommunications of Cambodia.

Teaching Experiences:

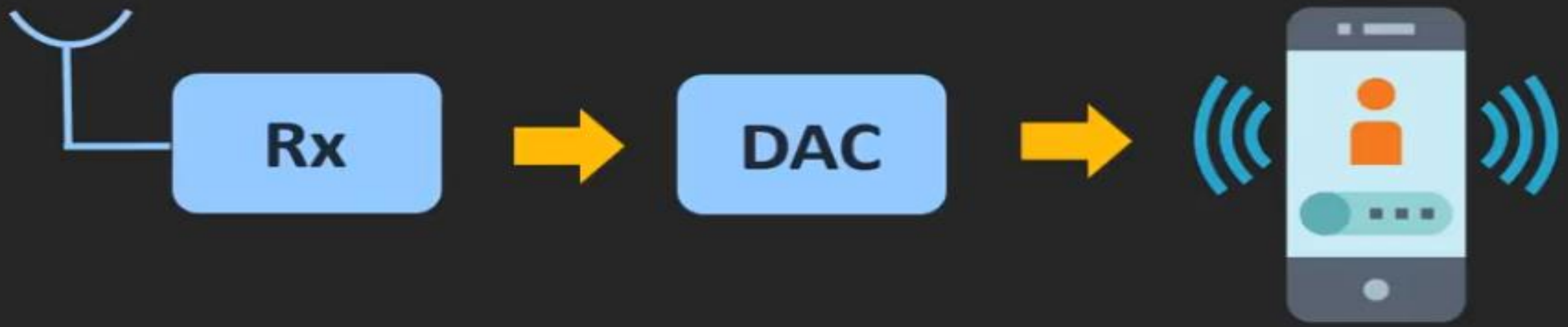
- 2000 Royal Academy of Cambodia (MSc.IT).
- 2002 Build Bright University (MSc.IT).
- 2019 National Polytechnic Institute of Cambodia (BSc.Telecom).
- 2020 Norton University (BSc.IT)
- 2023 Cambodia Academy of Digital Technology (BSc.Telecom).

Introduction

- The digital processor are only capable to detect binary/ digital signals. They do not understand analog signals, hence to interface/ process analog signals using digital processor (microprocessor, microcontroller) we must use an ADC i.e. analog to digital converter. Which converts a continuous physical quantity (generally voltage) to a digital number that represents the quantity's amplitude.
- The conversion involves sampling and quantization of the input to produce digital output.

ADC and DAC

ADC and DAC



Why we use ADC and DAC ?

Analog Signal → Transducer → Electrical Signal

Analog Signal →

- Susceptible to Noise
- Difficult to Process in Analog Domain
- Difficult to Store in Analog Domain

Digital Signal →

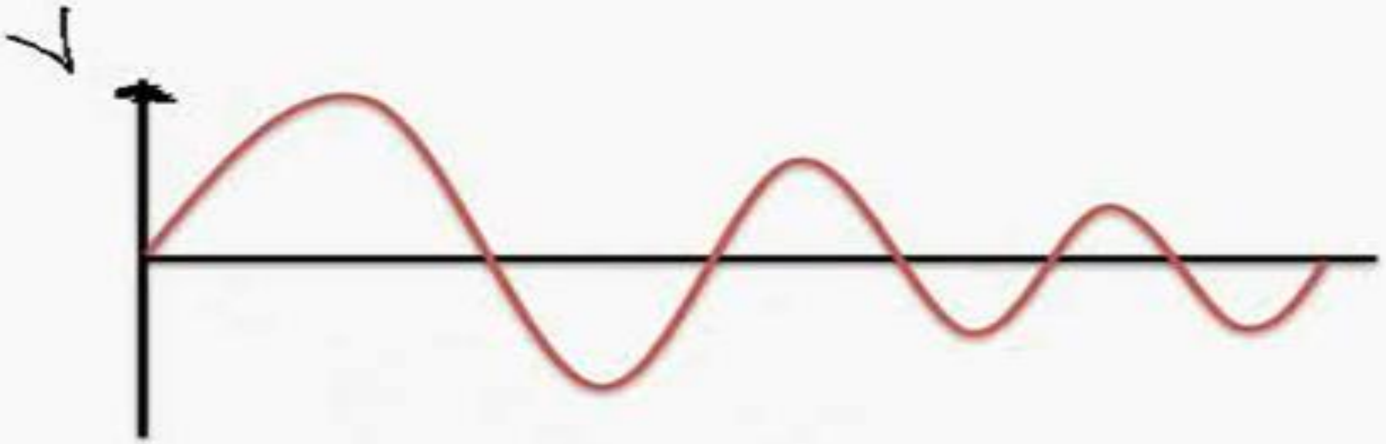
- Less Susceptible to Noise
- Easy to Process in Digital Domain
- Easy to Store in Digtial Domain

ADC and DAC

ADC - Analog to Digital Converter

Digitizing Voice

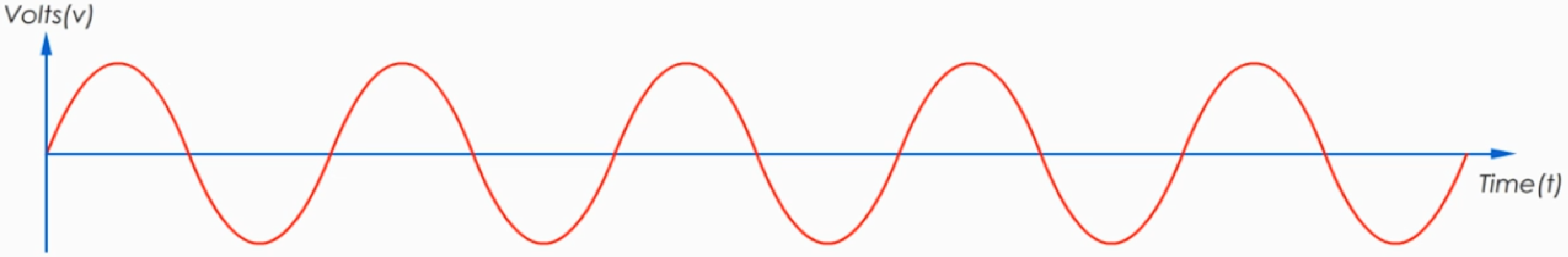
Step 1. Sample the Signal



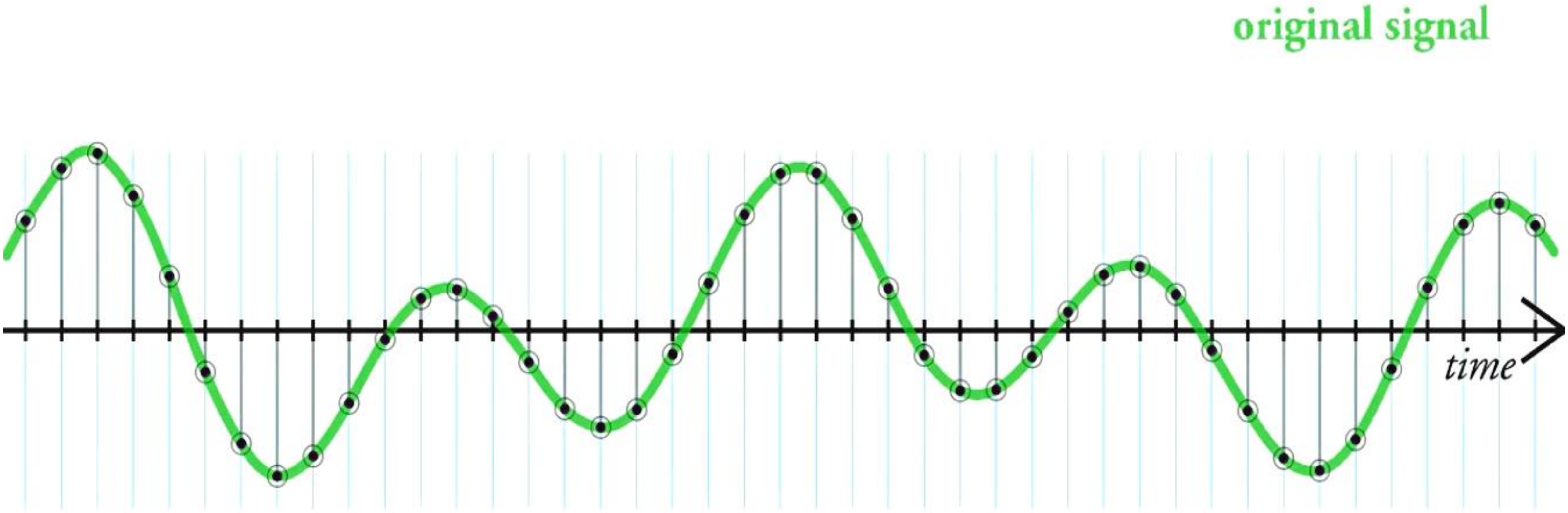
- The famed Dr. Nyquist formula:
- If you sample at twice the highest frequency, you can accurately reconstruct a signal digitally
- Common Frequencies:

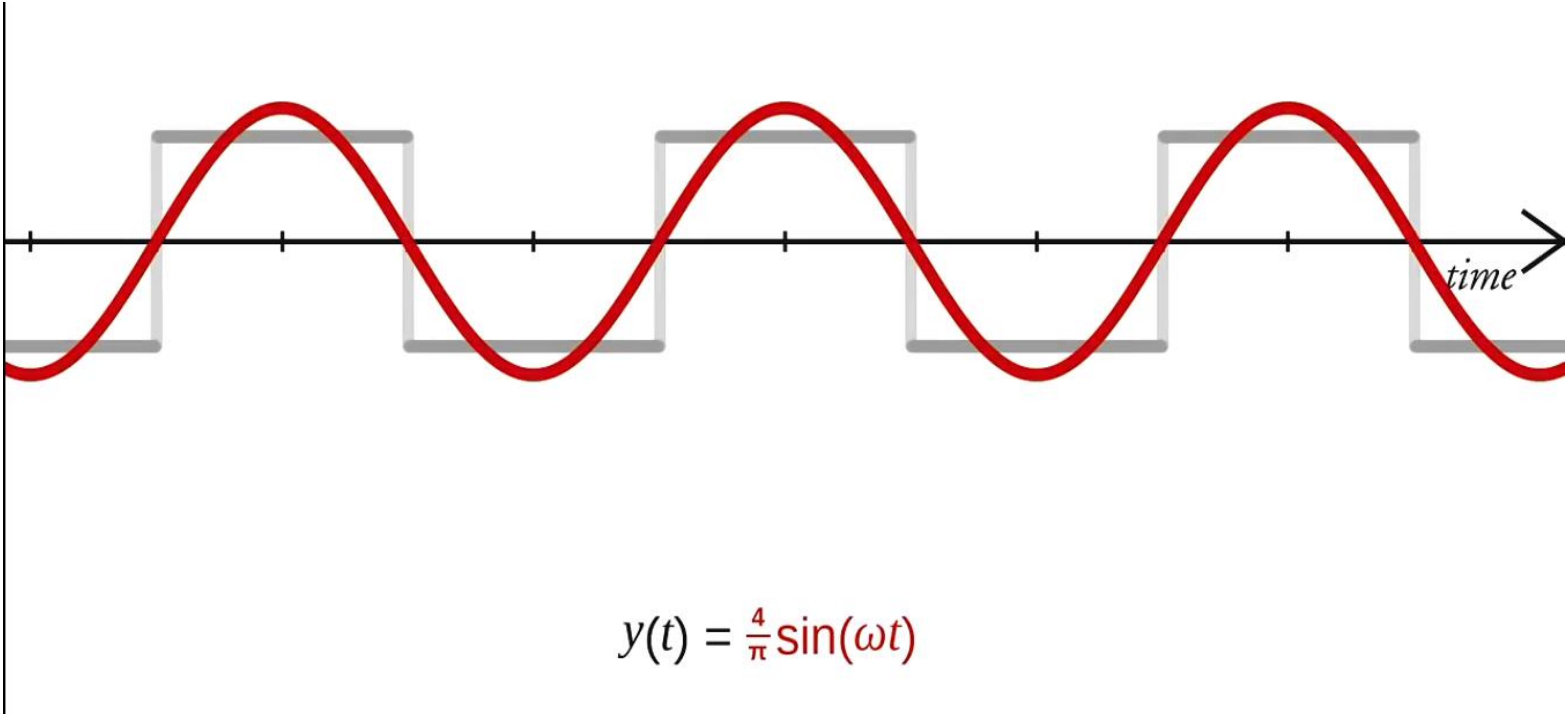
  - Human ear: 20 - 20,000Hz
  - Human speech: 200 - 9,000Hz
  - Nyquist theorum: 300 - 4,000Hz

An Analog Signal Wave

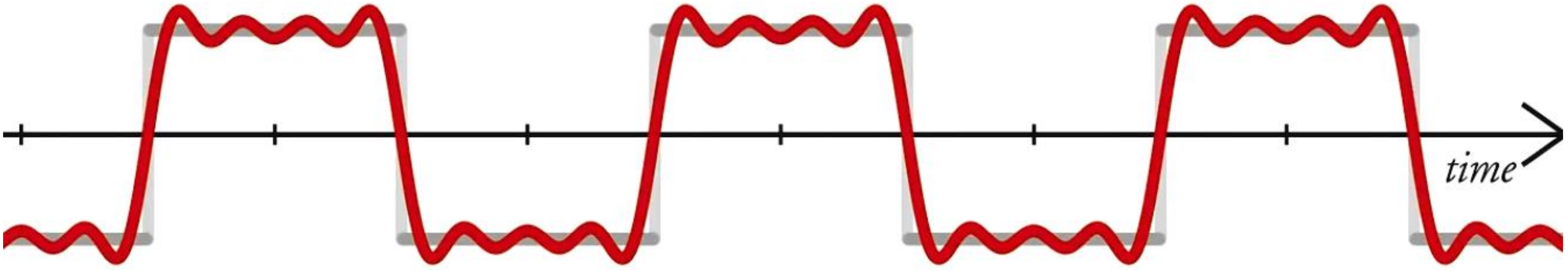


If we use voltage to represent energy, analog signal wave should be smooth and continuous over time.

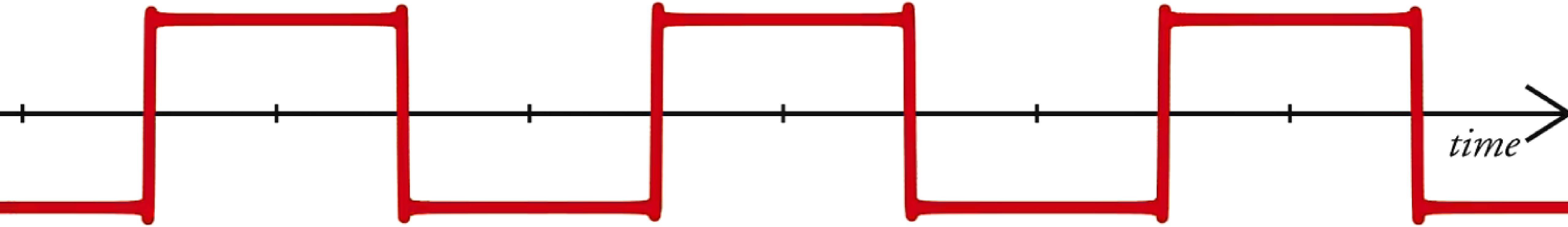

original signal
time



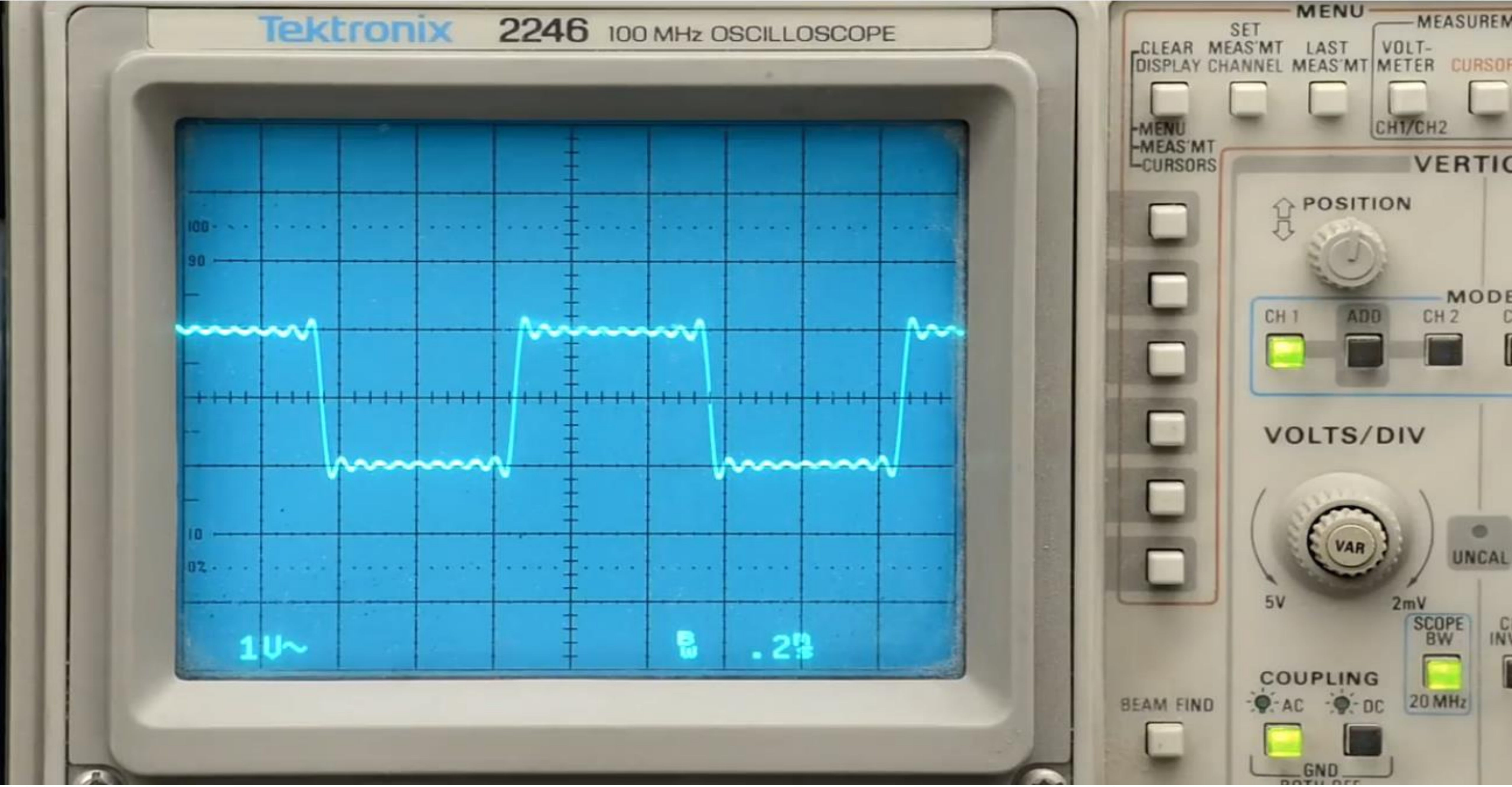
$$y(t) = \frac{4}{\pi}\sin(\omega t)$$



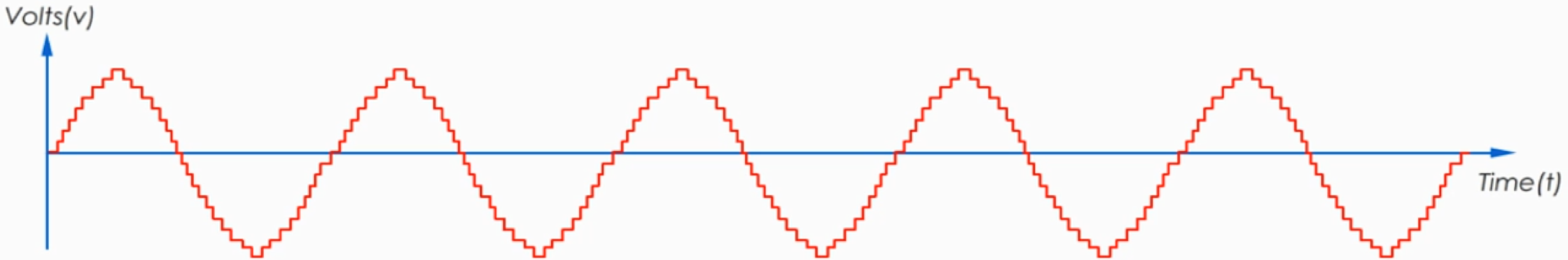
$$y(t) = \frac{4}{\pi}\sin(\omega t) + \frac{4}{3\pi}\sin(3\omega t) + \frac{4}{5\pi}\sin(5\omega t) + \frac{4}{7\pi}\sin(7\omega t)$$



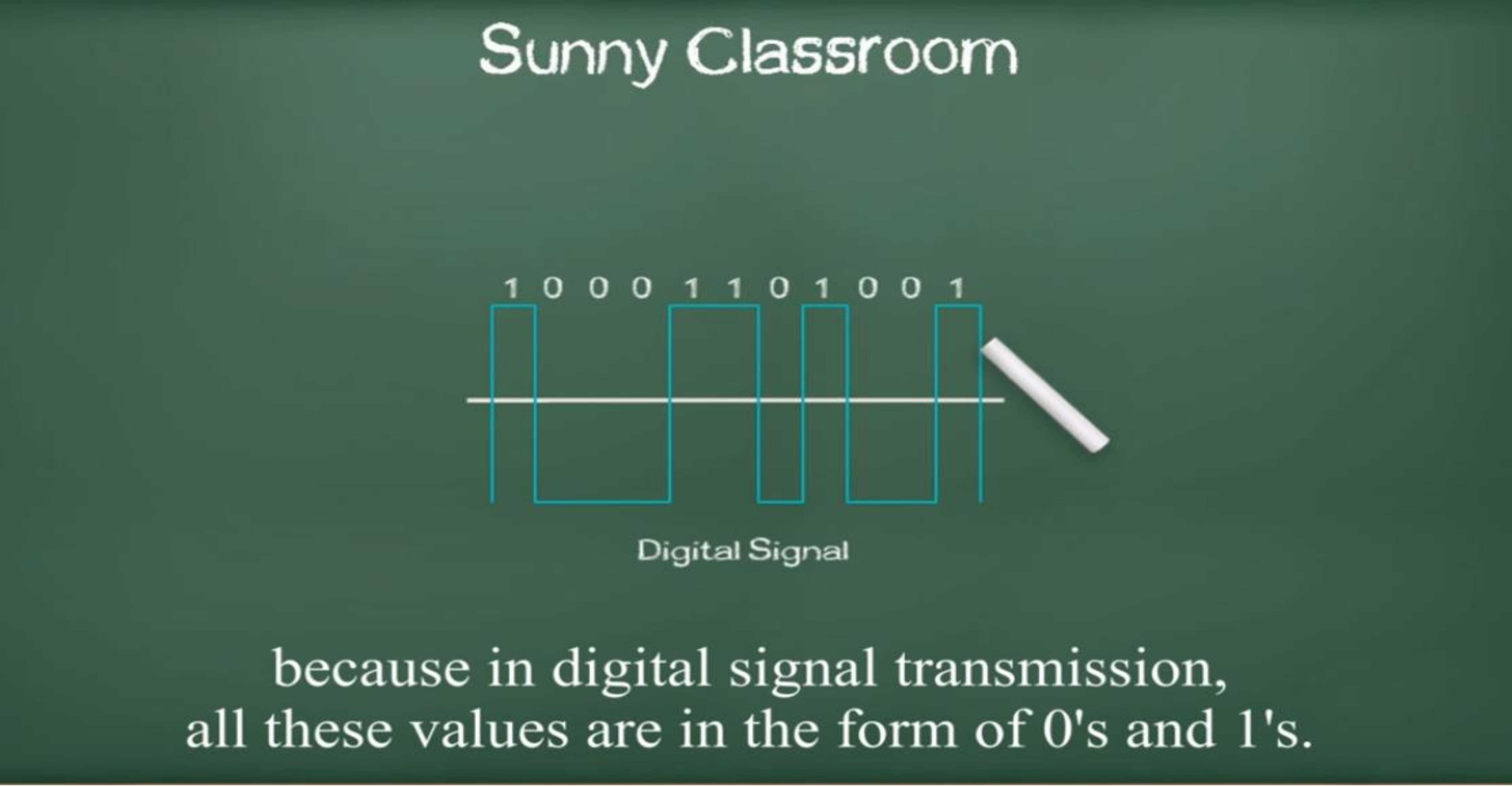
$$
\begin{aligned}
y(t) = {} & \tfrac{4}{\pi}\sin(\omega t) + \tfrac{4}{3\pi}\sin(3\omega t) + \tfrac{4}{5\pi}\sin(5\omega t) + \tfrac{4}{7\pi}\sin(7\omega t) + \tfrac{4}{9\pi}\sin(9\omega t) + \\
& \tfrac{4}{11\pi}\sin(11\omega t) + \tfrac{4}{13\pi}\sin(13\omega t) + \tfrac{4}{15\pi}\sin(15\omega t) + \tfrac{4}{17\pi}\sin(17\omega t) + \tfrac{4}{19\pi}\sin(19\omega t) + \\
& \tfrac{4}{21\pi}\sin(21\omega t) + \tfrac{4}{23\pi}\sin(23\omega t) + \tfrac{4}{25\pi}\sin(25\omega t) + \tfrac{4}{27\pi}\sin(27\omega t) + \tfrac{4}{29\pi}\sin(29\omega t) + \tfrac{4}{31\pi}\sin(31\omega t) + \\
& \tfrac{4}{33\pi}\sin(33\omega t) + \tfrac{4}{35\pi}\sin(35\omega t) + \tfrac{4}{37\pi}\sin(37\omega t) + \tfrac{4}{39\pi}\sin(39\omega t) + \tfrac{4}{41\pi}\sin(41\omega t) + \tfrac{4}{43\pi}\sin(43\omega t) + \tfrac{4}{45\pi}\sin(45\omega t) + \\
& \tfrac{4}{47\pi}\sin(47\omega t) + \tfrac{4}{49\pi}\sin(49\omega t) + \tfrac{4}{51\pi}\sin(51\omega t) + \tfrac{4}{53\pi}\sin(53\omega t) + \tfrac{4}{55\pi}\sin(55\omega t) + \tfrac{4}{57\pi}\sin(57\omega t) + \tfrac{4}{59\pi}\sin(59\omega t) + \tfrac{4}{61\pi}\sin(61\omega t) + \tfrac{4}{63\pi}\sin(63\omega t) + \\
& \tfrac{4}{65\pi}\sin(65\omega t) + \tfrac{4}{67\pi}\sin(67\omega t) + \tfrac{4}{69\pi}\sin(69\omega t) + \tfrac{4}{71\pi}\sin(71\omega t) + \tfrac{4}{73\pi}\sin(73\omega t) + \tfrac{4}{75\pi}\sin(75\omega t) + \tfrac{4}{77\pi}\sin(77\omega t) + \tfrac{4}{79\pi}\sin(79\omega t) + \tfrac{4}{81\pi}\sin(81\omega t) + \tfrac{4}{83\pi}\sin(83\omega t) + \tfrac{4}{85\pi}\sin(85\omega t) + \\
& \tfrac{4}{87\pi}\sin(87\omega t) + \tfrac{4}{89\pi}\sin(89\omega t) + \tfrac{4}{91\pi}\sin(91\omega t) + \tfrac{4}{93\pi}\sin(93\omega t) + \tfrac{4}{95\pi}\sin(95\omega t) + \tfrac{4}{97\pi}\sin(97\omega t) + \tfrac{4}{99\pi}\sin(99\omega t) + \tfrac{4}{101\pi}\sin(101\omega t) + \tfrac{4}{103\pi}\sin(103\omega t) + \tfrac{4}{105\pi}\sin(105\omega t) + \tfrac{4}{107\pi}\sin(107\omega t) + \tfrac{4}{109\pi}\sin(109\omega t) + \tfrac{4}{111\pi}\sin(111\omega t) + \\
& \tfrac{4}{113\pi}\sin(113\omega t) + \tfrac{4}{115\pi}\sin(115\omega t) + \tfrac{4}{117\pi}\sin(117\omega t) + \tfrac{4}{119\pi}\sin(119\omega t) + \tfrac{4}{121\pi}\sin(121\omega t) + \tfrac{4}{123\pi}\sin(123\omega t) + \tfrac{4}{125\pi}\sin(125\omega t) + \tfrac{4}{127\pi}\sin(127\omega t) + \tfrac{4}{129\pi}\sin(129\omega t) + \tfrac{4}{131\pi}\sin(131\omega t) + \tfrac{4}{133\pi}\sin(133\omega t) + \tfrac{4}{135\pi}\sin(135\omega t) + \tfrac{4}{137\pi}\sin(137\omega t) + \tfrac{4}{139\pi}\sin(139\omega t) + \tfrac{4}{141\pi}\sin(141\omega t) + \cdots
\end{aligned}
$$

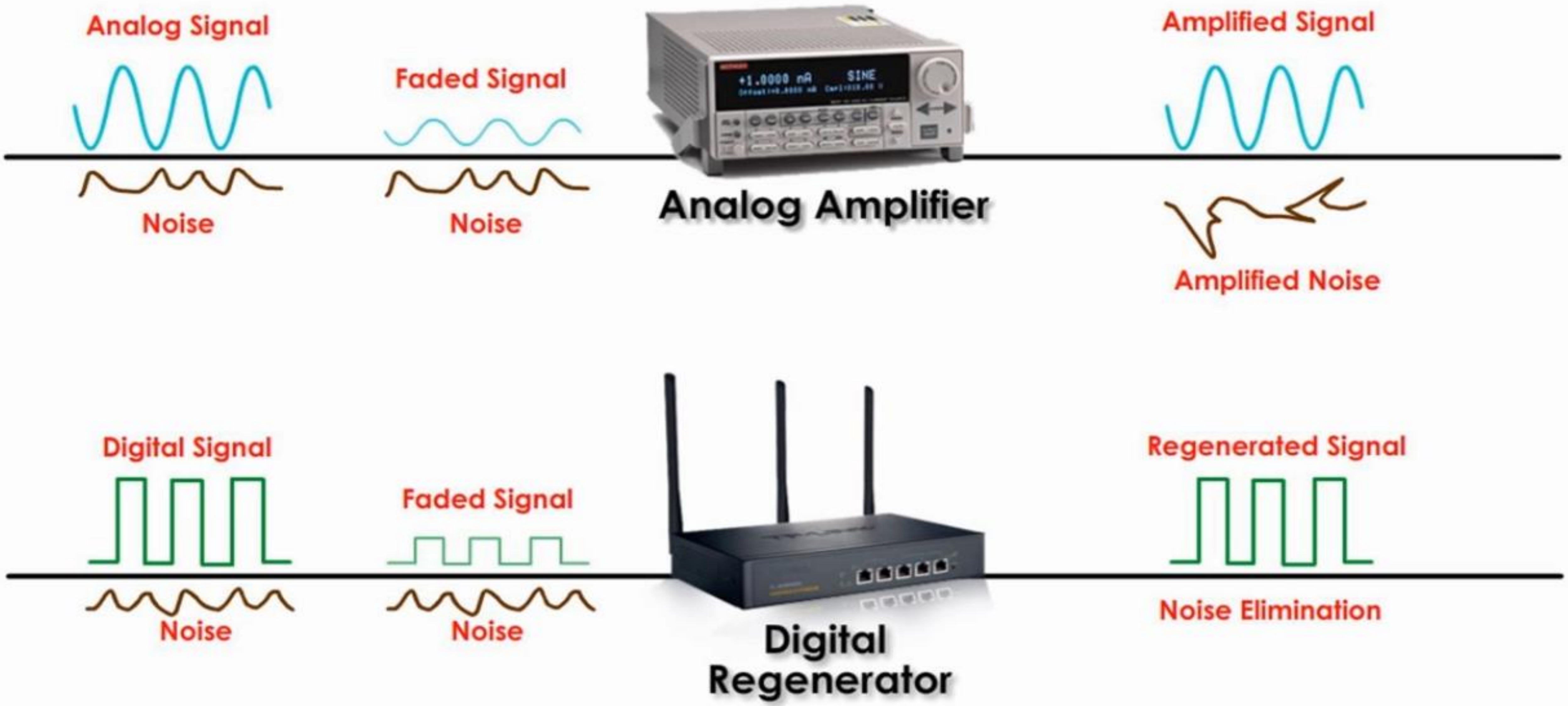

Tektronix
2246 100 MHz OSCILLOSCOPE
MENU
MEASUREM
SET
CLEAR MEAS'MT LAST VOLT-
DISPLAY CHANNEL MEAS'MT METER CURSOR
CH1/CH2
MENU
MEAS'MT
CURSORS
VERTIC
POSITION
MODE
CH 1 ADD CH 2
VOLTS/DIV
VAR
UNCAL
5V 2mV
SCOPE BW
20 MHz
COUPLING
AC DC
GND
BEAM FIND
100
90
10
0%
1V~

A Digital Signal Wave



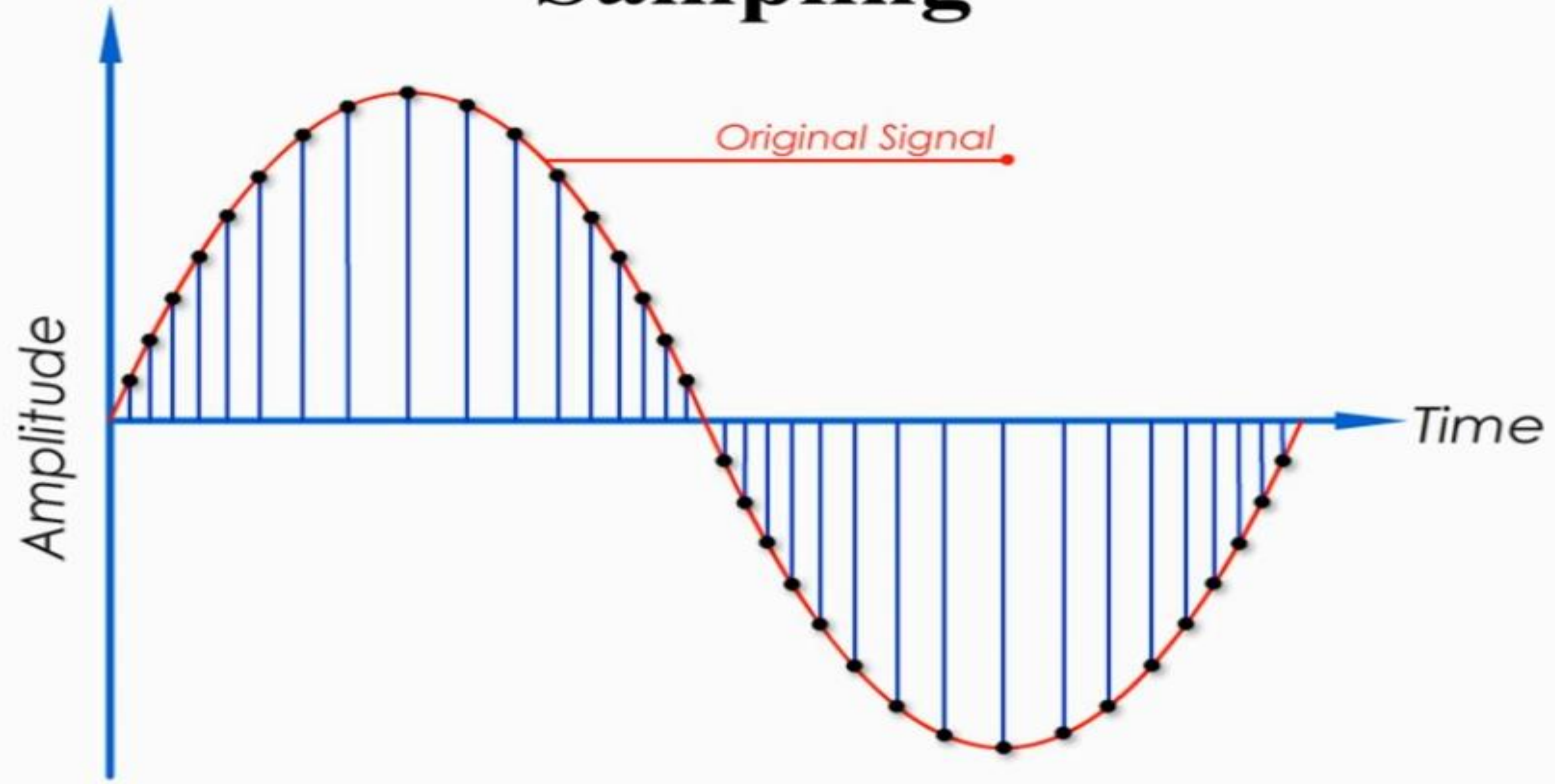
if we draw the graph of voltage over time.


Sunny Classroom
1 0 0 0 1 1 0 1 0 0 1
Digital Signal
because in digital signal transmission,
all these values are in the form of 0's and 1's.


Analog Signal
Faded Signal
Noise
Noise
Analog Amplifier
Amplified Signal
Amplified Noise
Digital Signal
Faded Signal
Noise
Noise
Digital Regenerator
Regenerated Signal
Noise Elimination

so that original signal can be represented by those samples completely

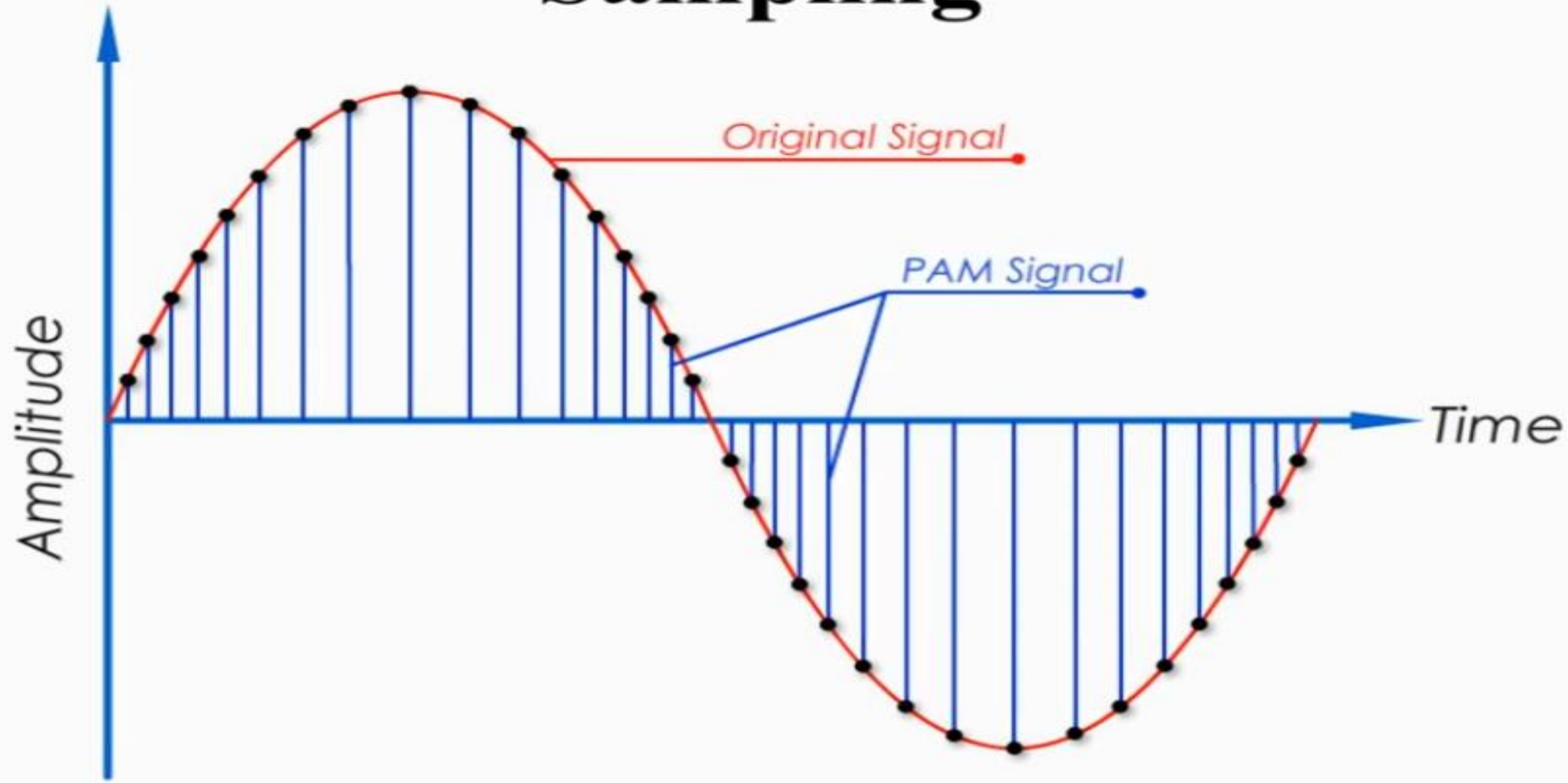
Sampling



amplitudes with a regular interval over time.

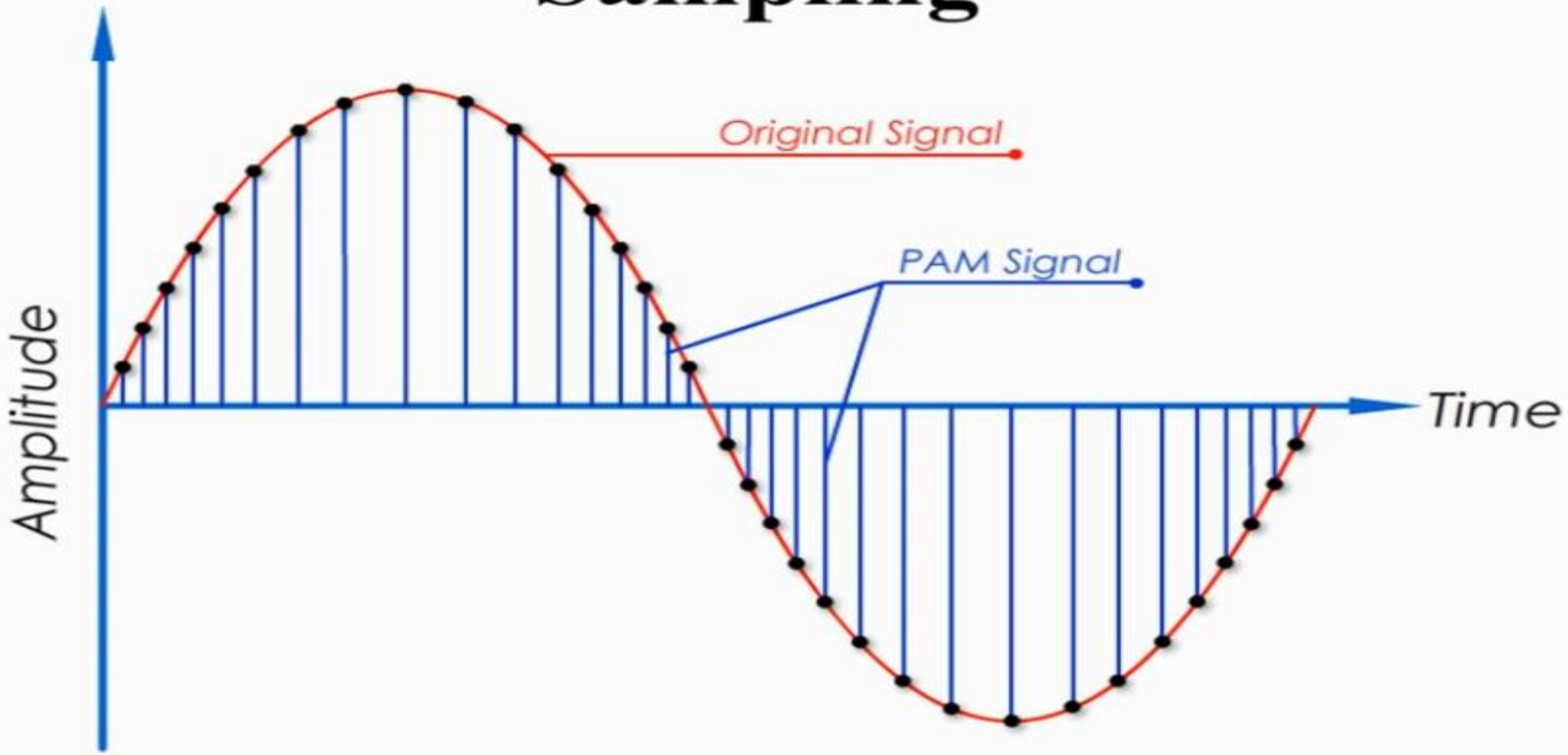
That is why the process of sampling is also called PAM

Sampling



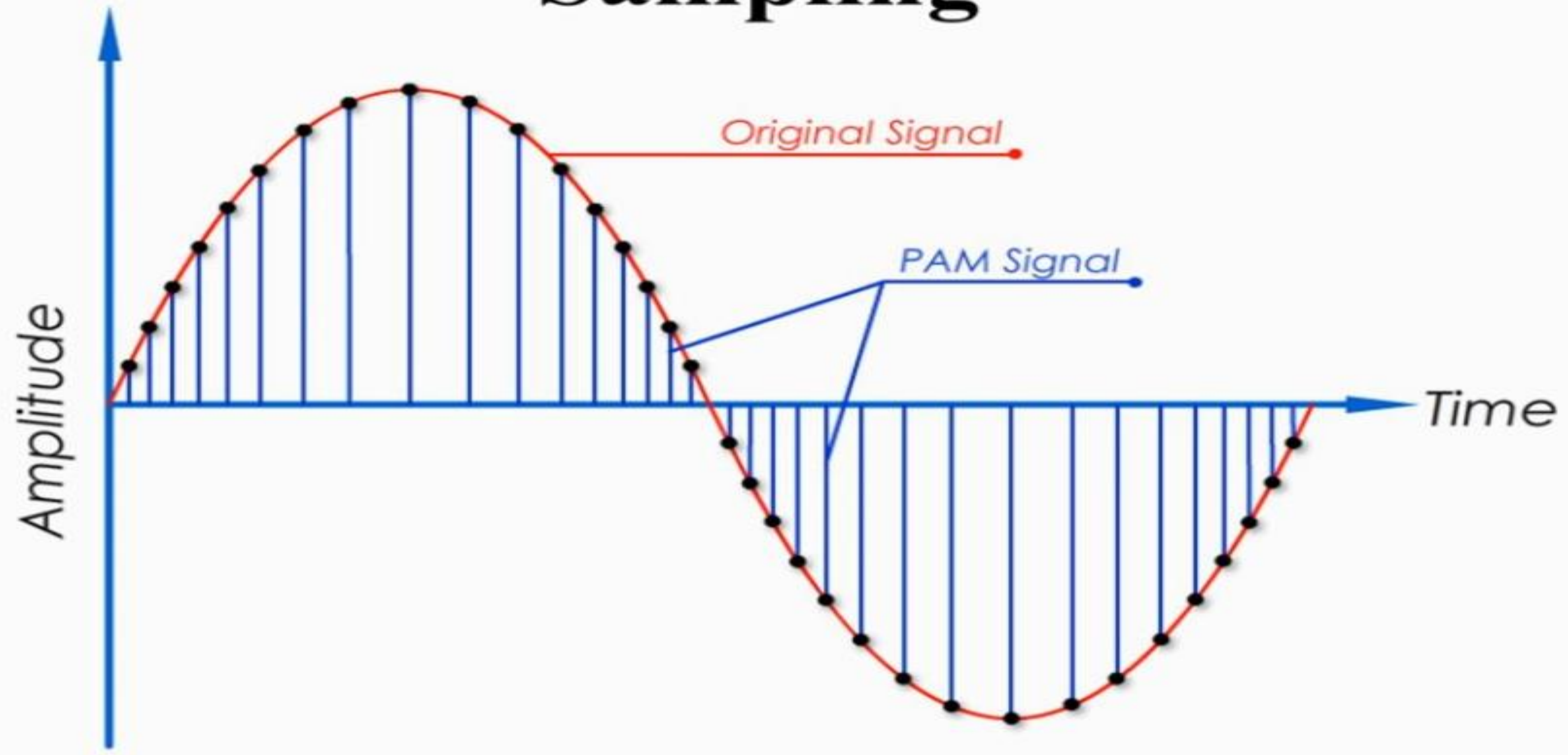
PAM signal is simply the result of a series of these discrete sample values.

Sampling



We use Hertz to measure sample rate.
e.g. one sample per second is 1Hz.

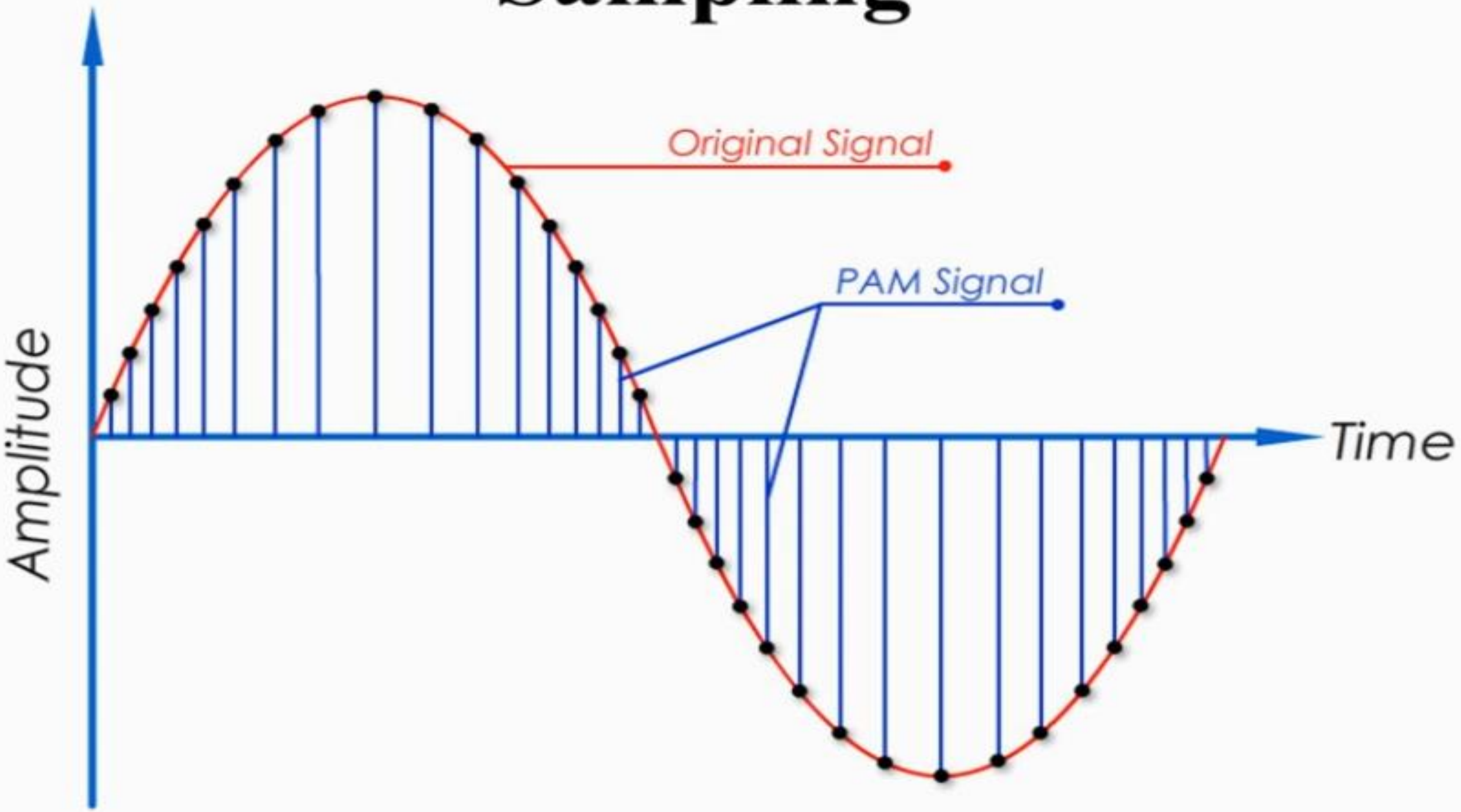
Sampling



1,000 samples per second is 1 KHz

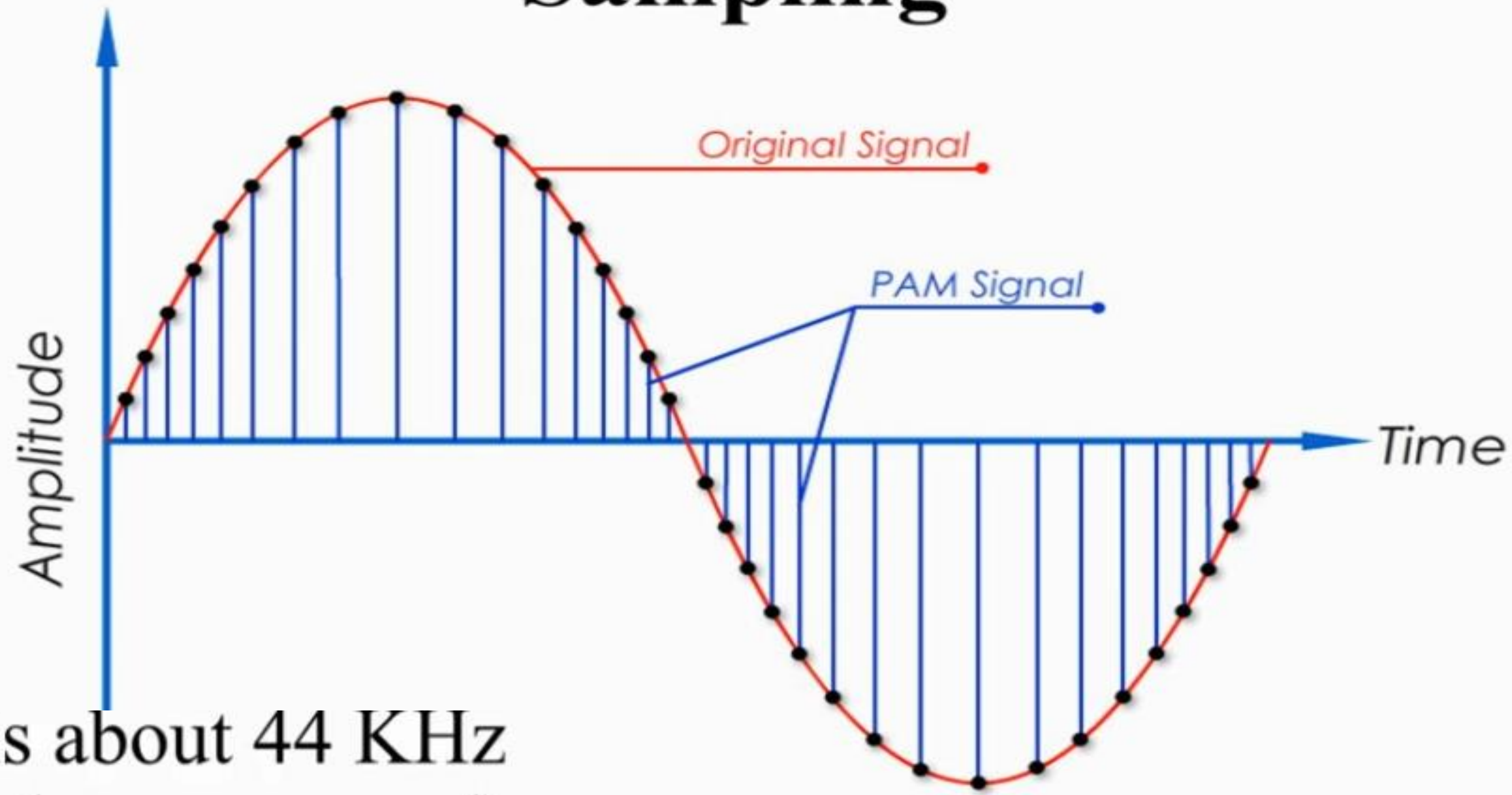
For telephone, sample rate is 8 KHz (8,000 samples per second).

Sampling



For Voice over IP(VoIP),
the sample rate is double: 16 KHz.

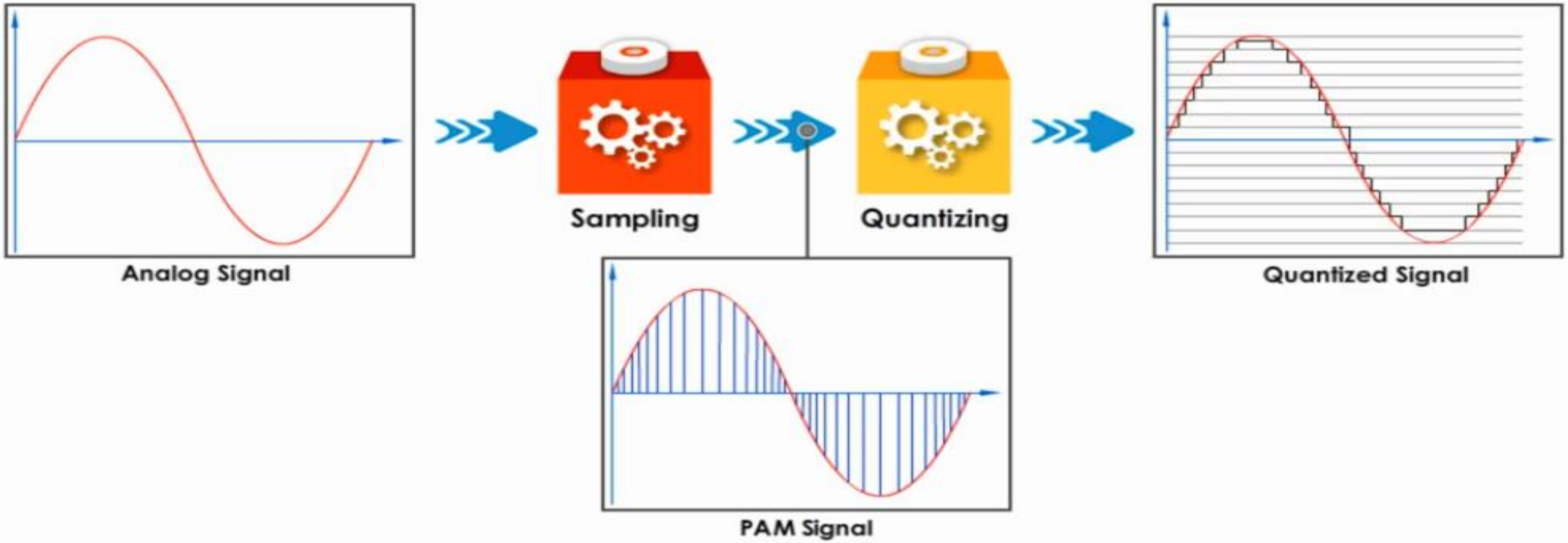
Sampling



sample rate is about 44 KHz
(44,000 samples per second).

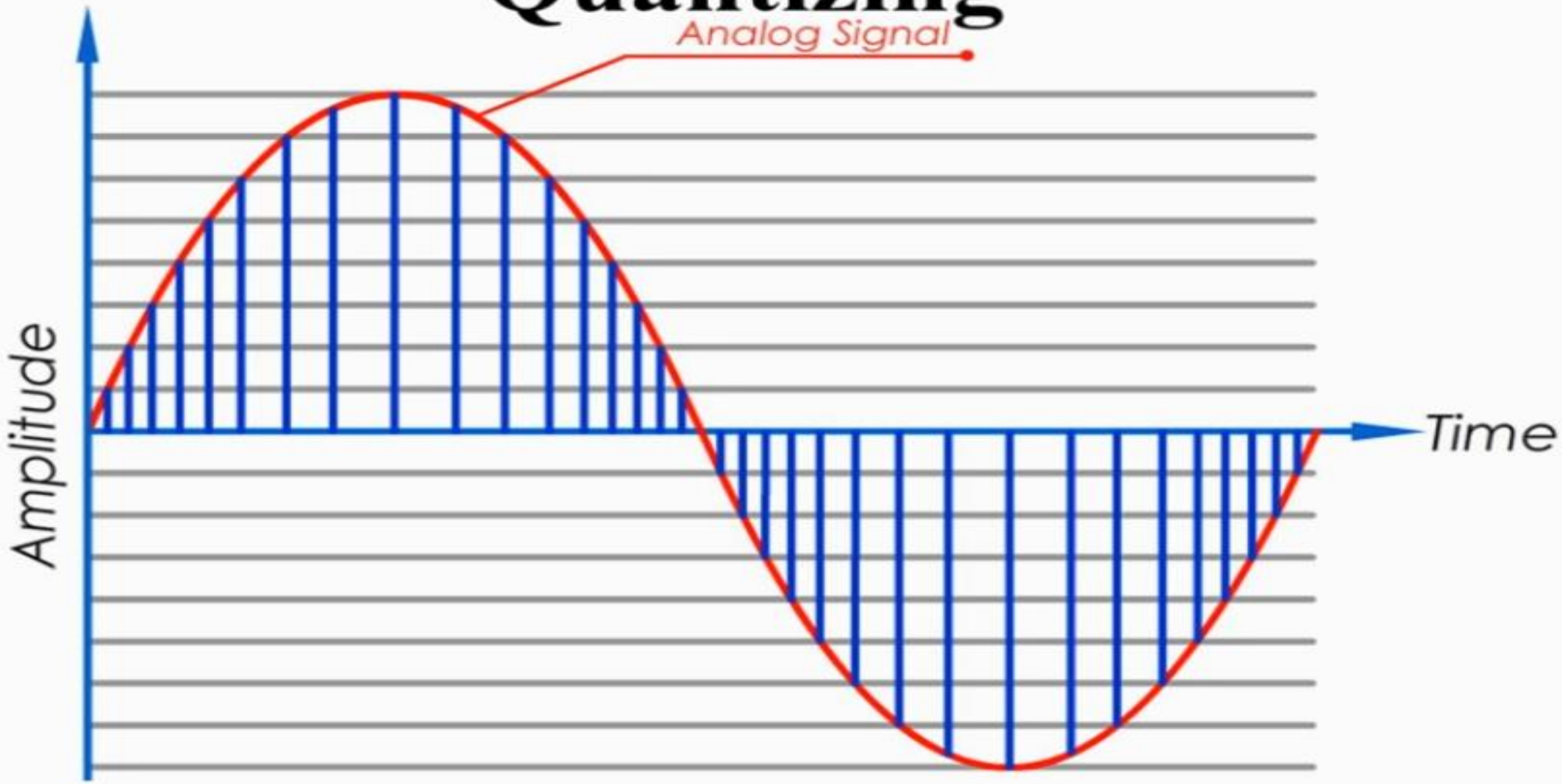
For audio CD or MP3,

Quantizing



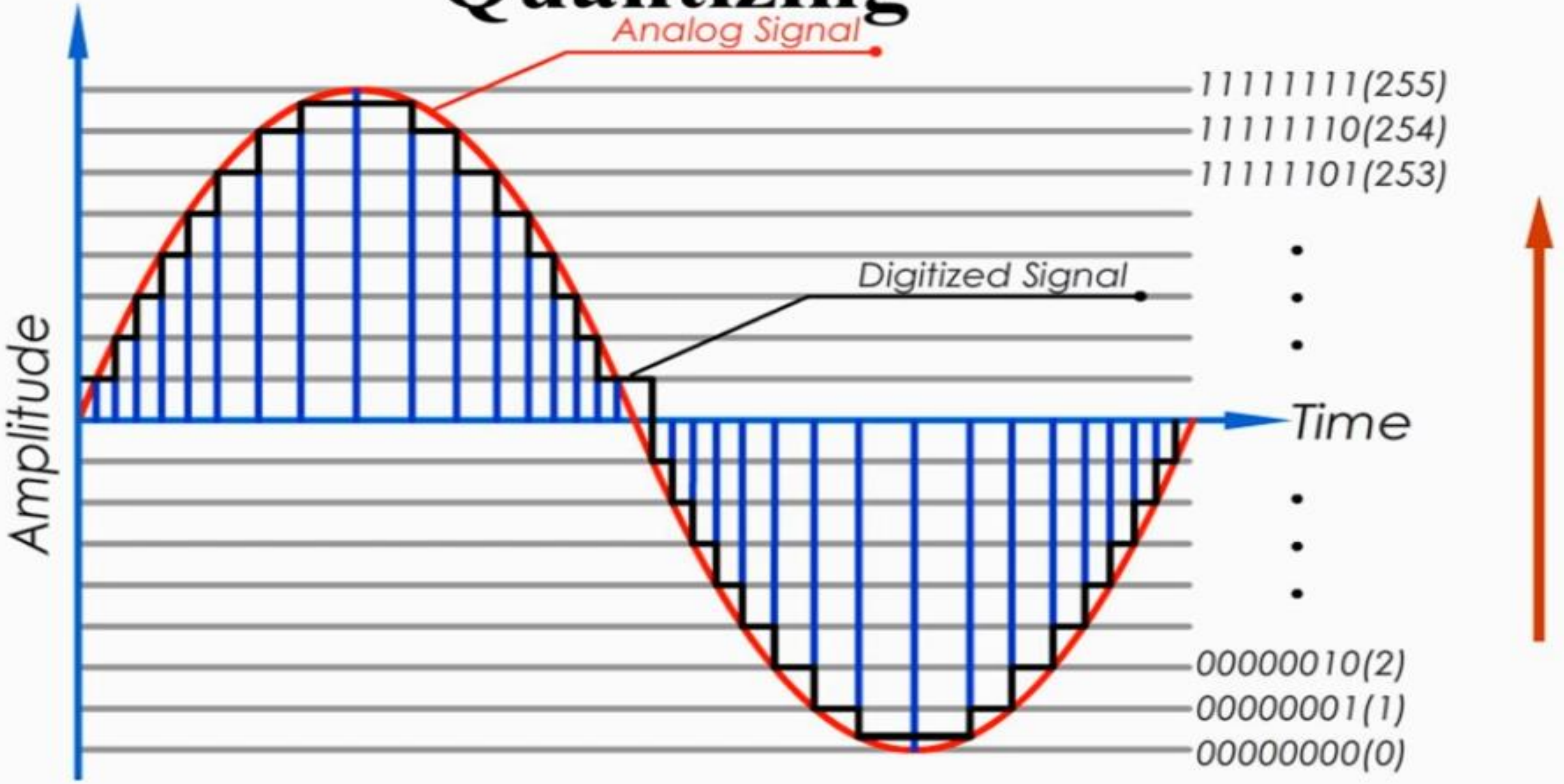
Sampling
Quantizing
Analog Signal
PAM Signal
Quantized Signal

Quantizing



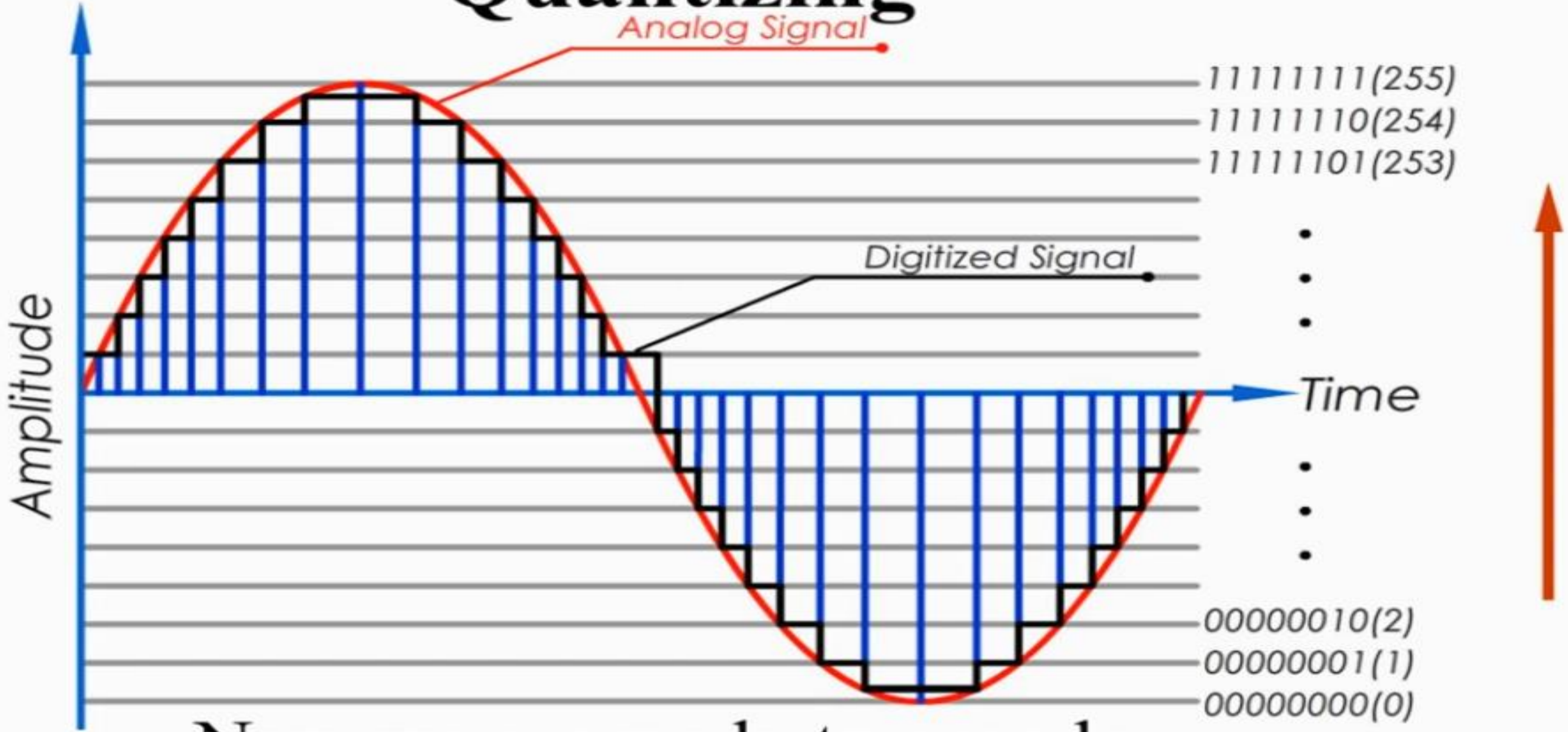
Sampling converts a time-varying signal

Quantizing

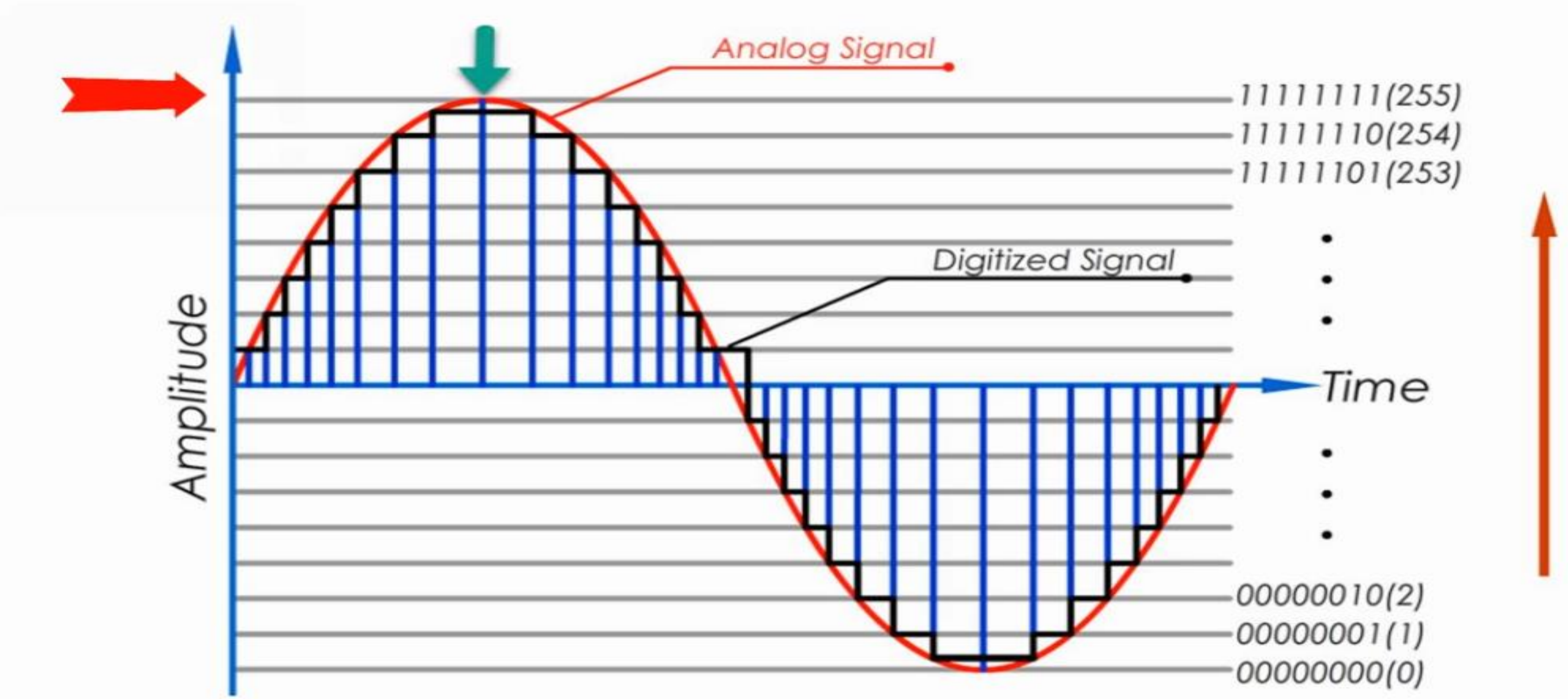


It means there are 256 levels,
Each level is associated with a specific bit value.

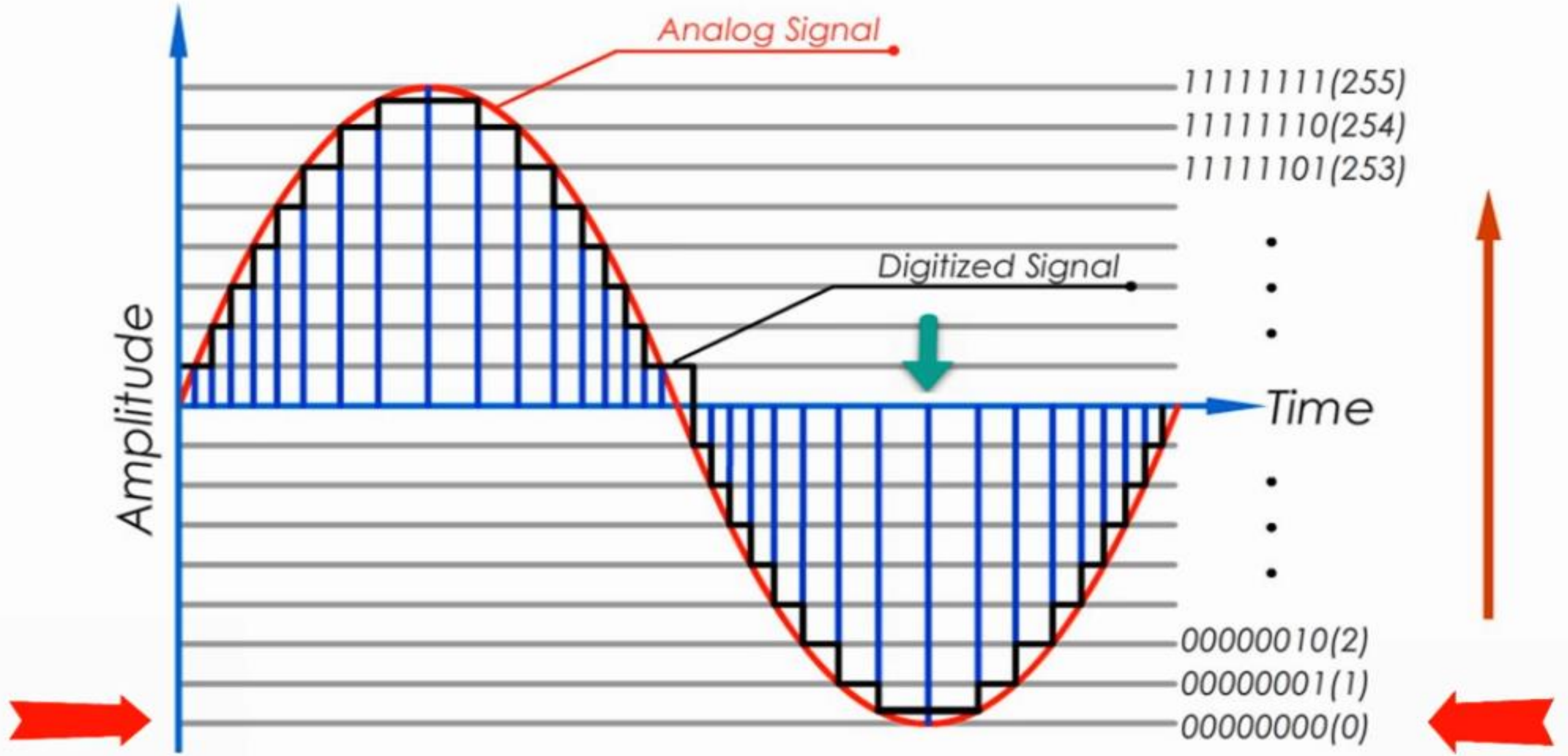
Quantizing



Now we are ready to encode these samples with 8-bit depth.

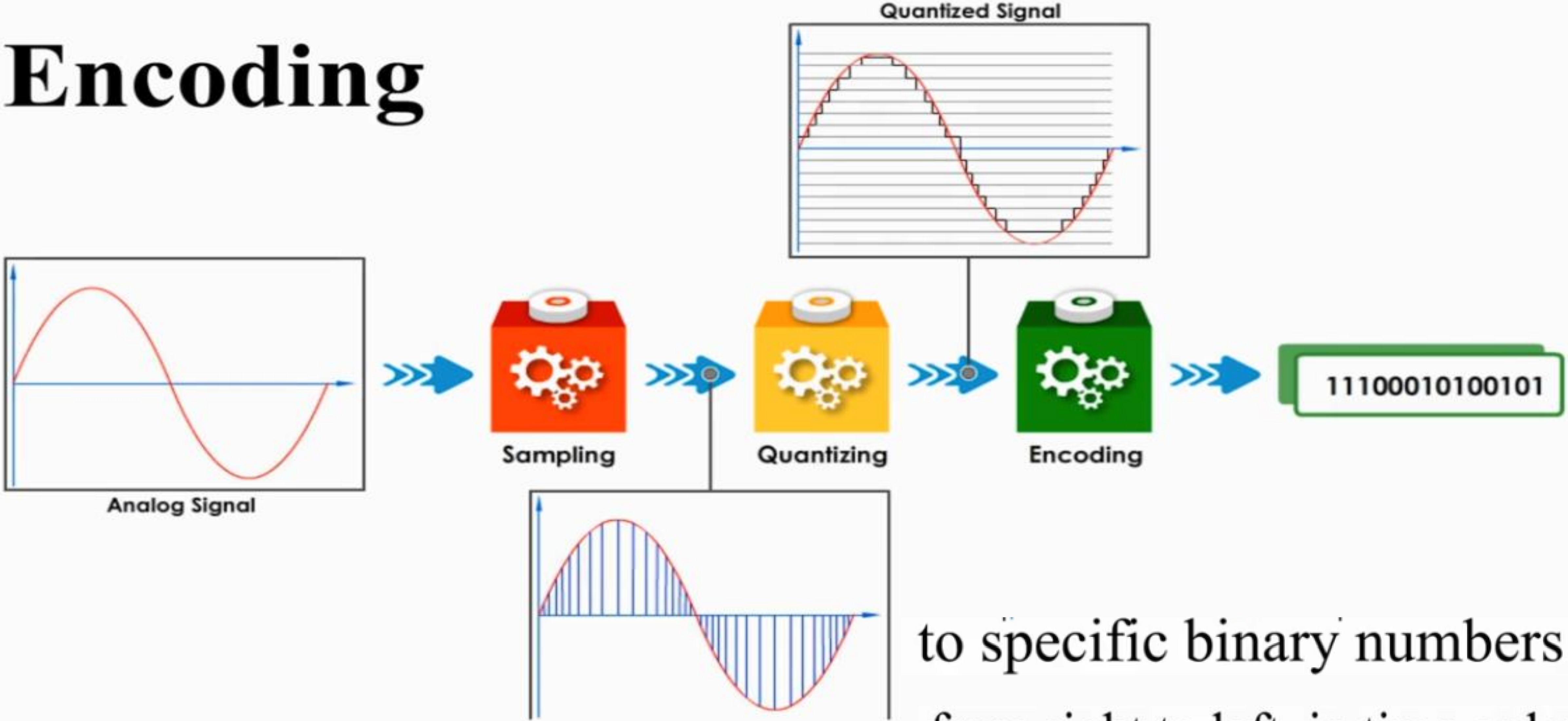


We can see it would match decimal number 255 which is equal to the binary number 11111111.



with decimal value 0 (00000000 in binary).

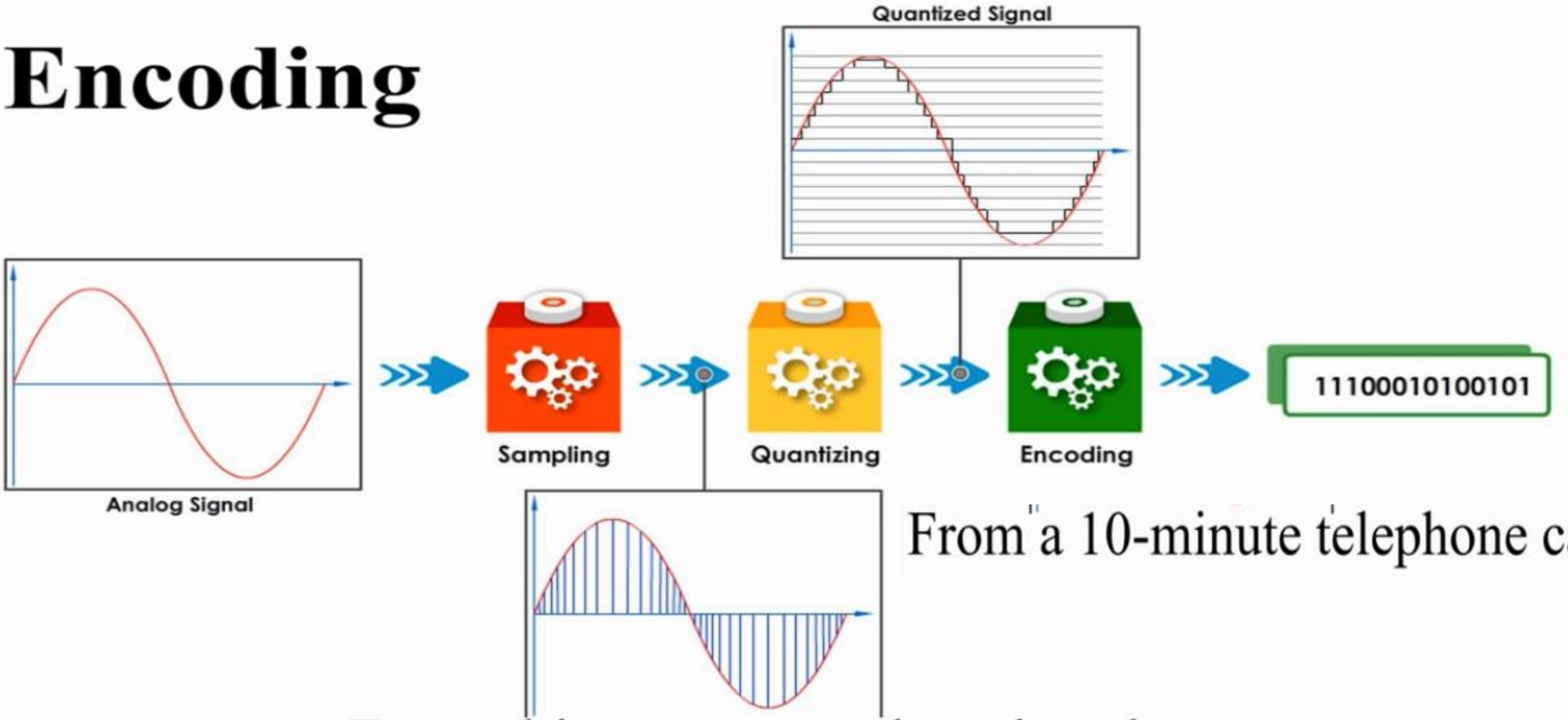
Encoding



to specific binary numbers.

In encoding step, we will cc from right to left, in time order,

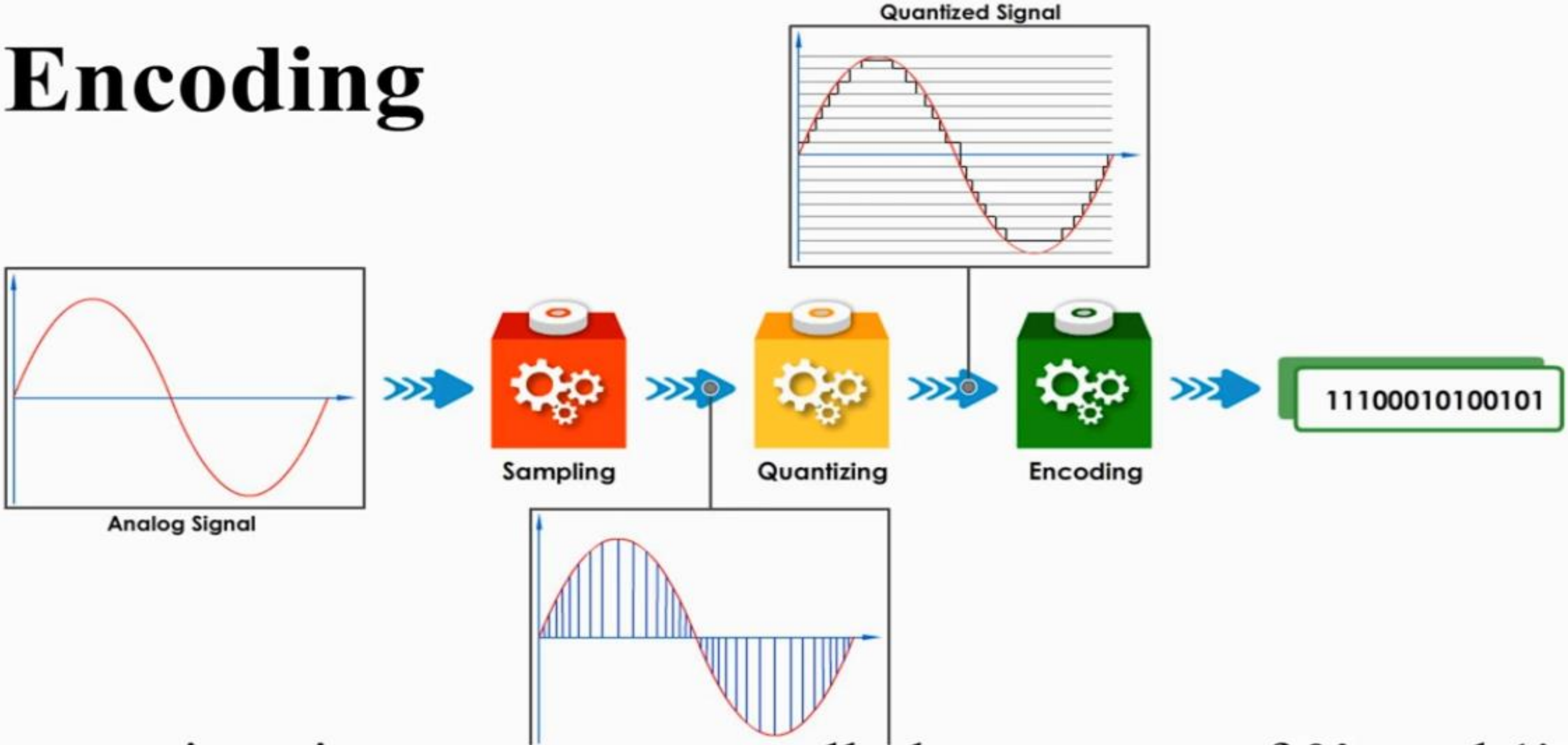
Encoding



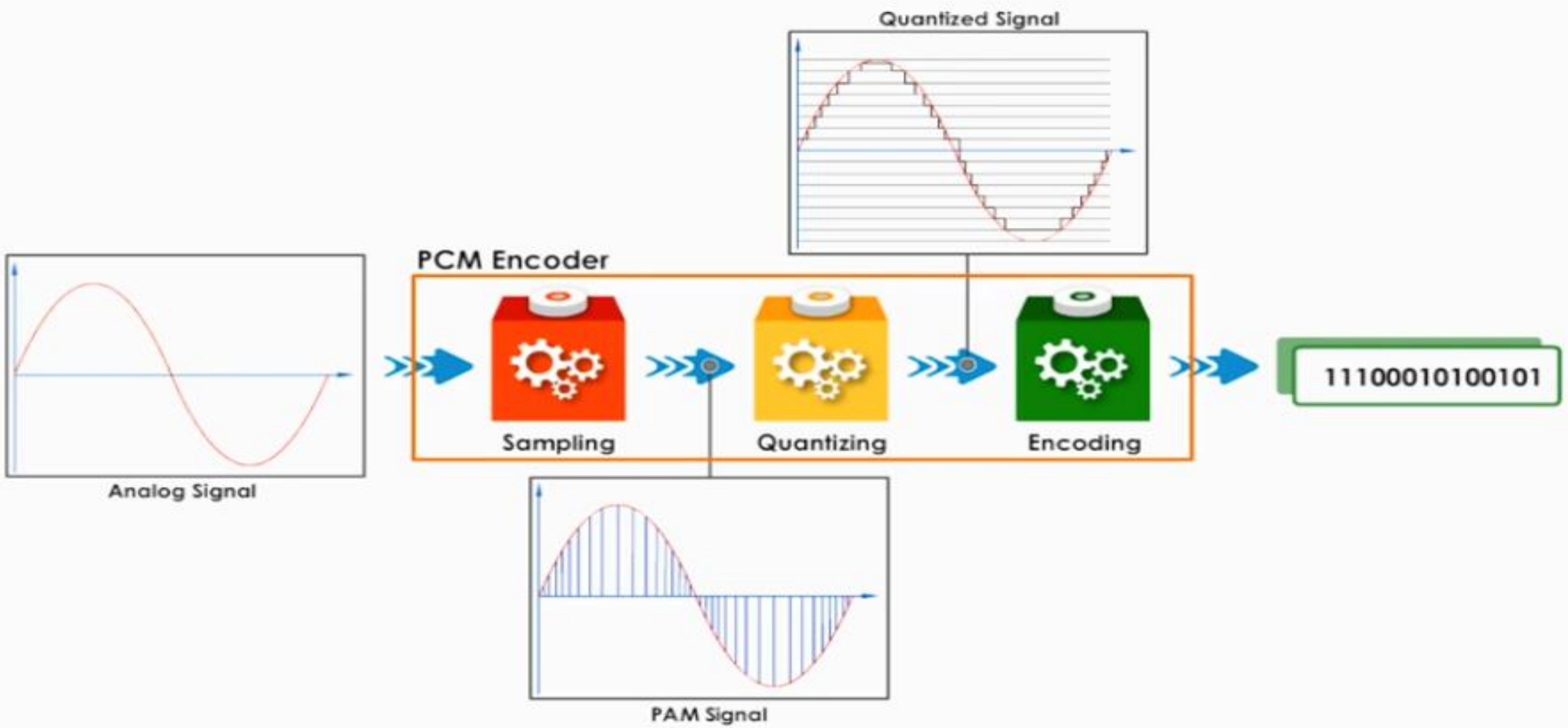
From a 10-minute telephone call,

From this one-second analog data,
we can get a long stream of 0’s and 1's.

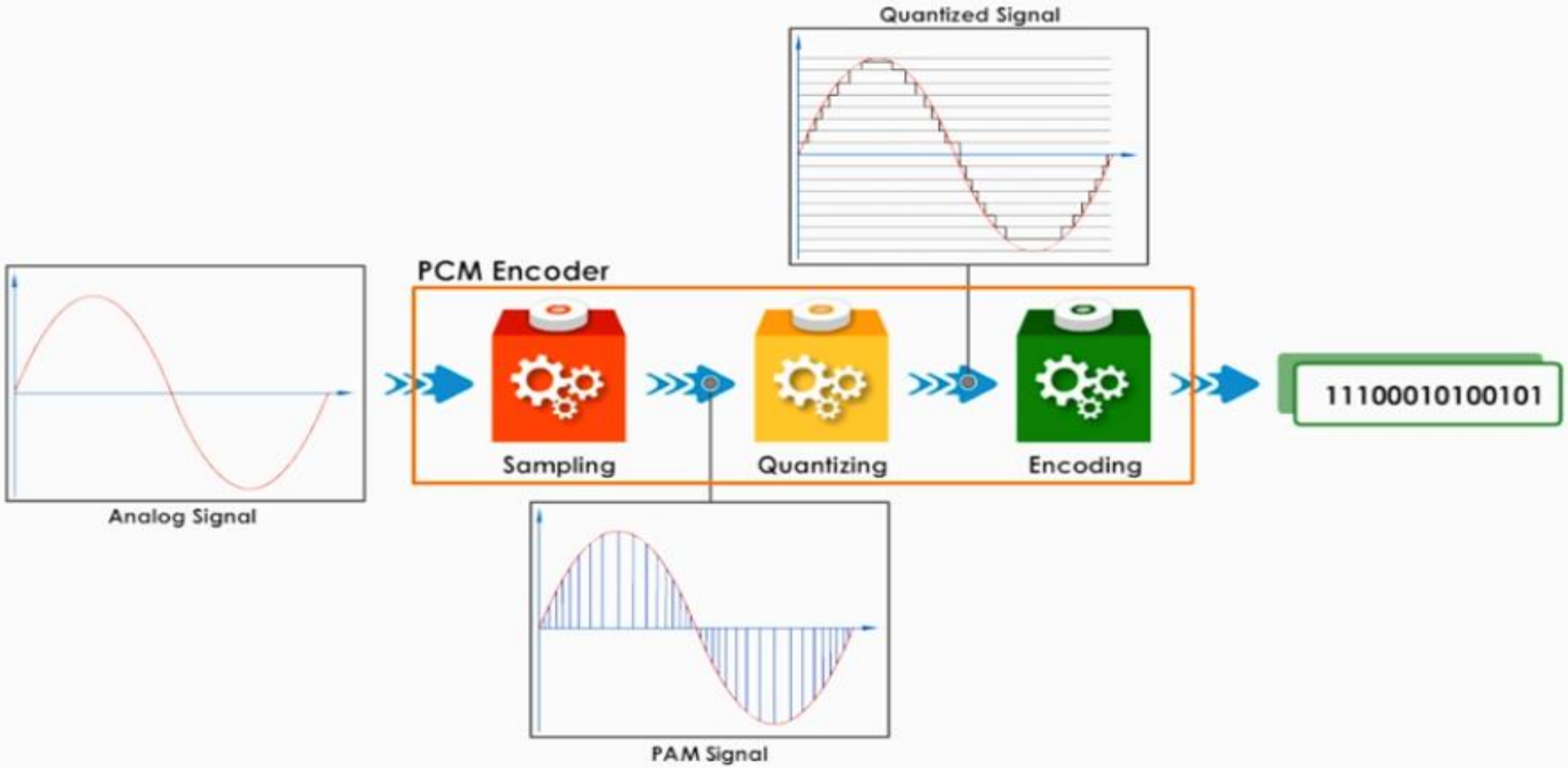
Encoding



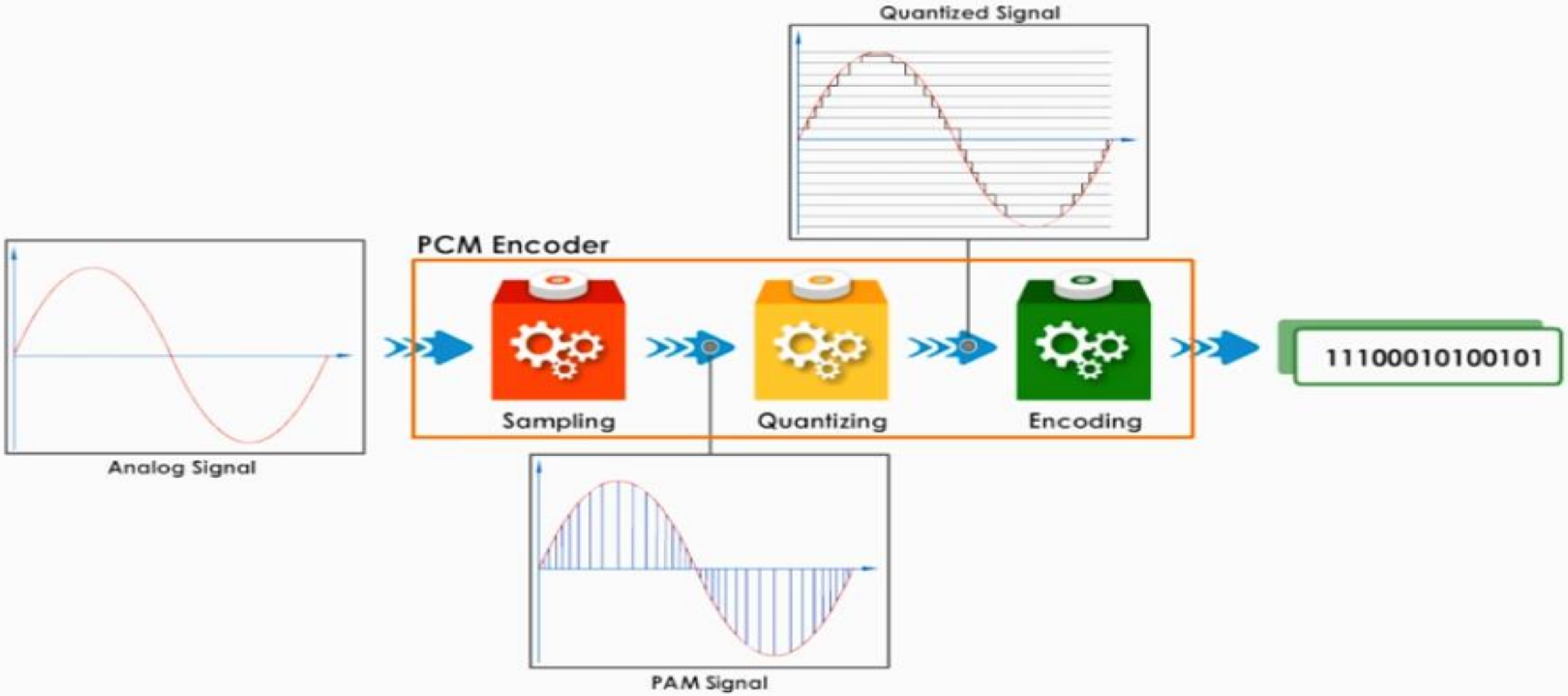
we can imagine we can get a really long stream of 0’s and 1's.



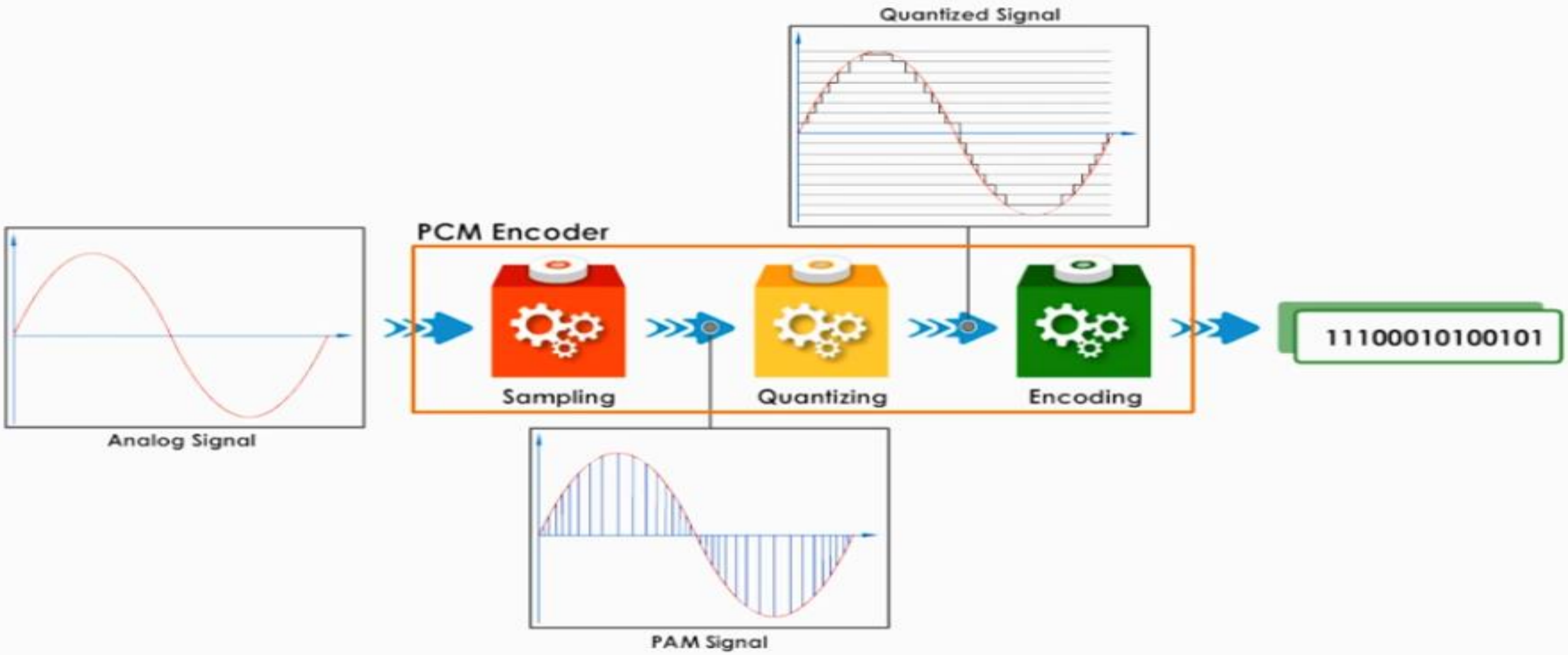
In summary,sampling converts a time-varying signal into a discrete-time signal, a sequence of real numbers.



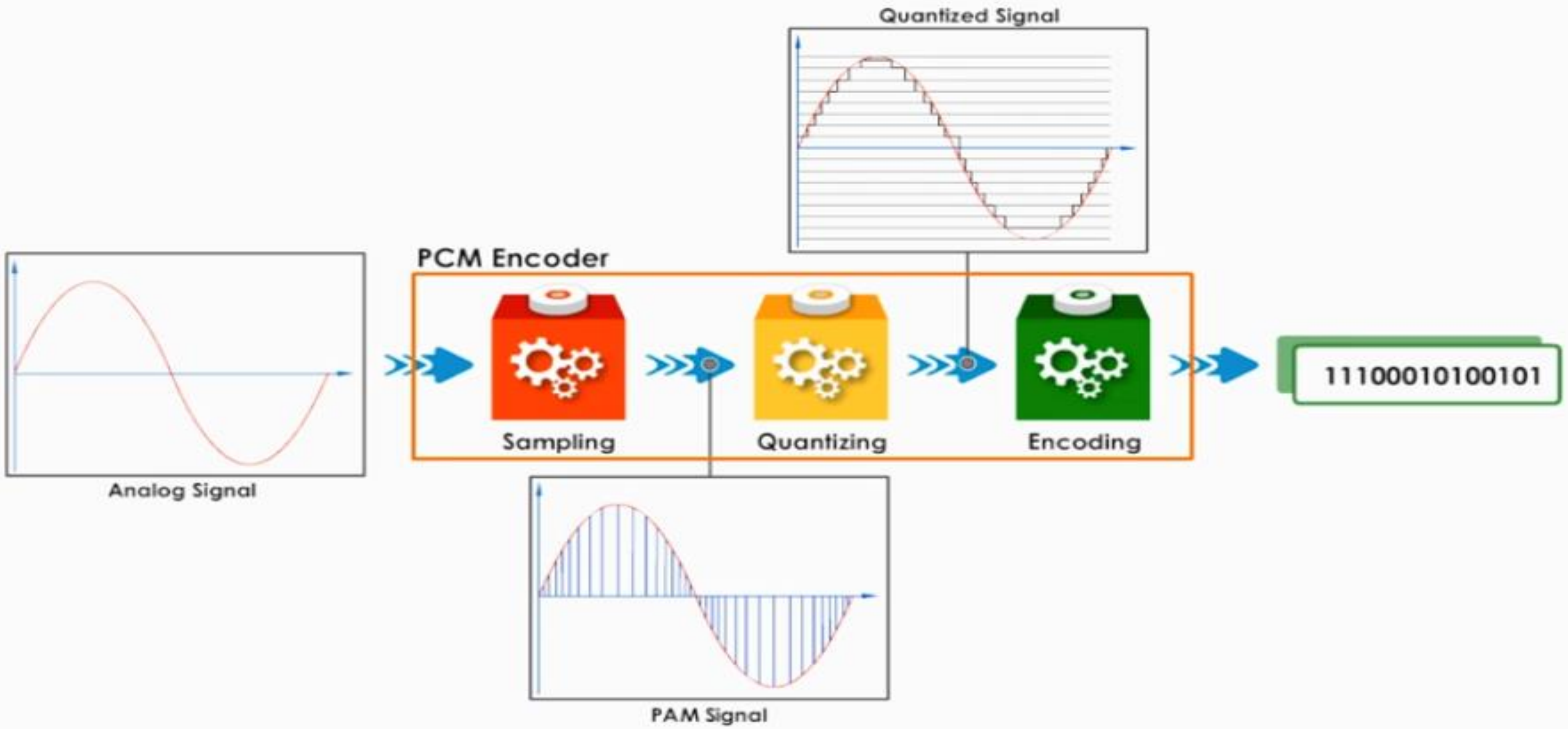
Sampling is like drawing vertical lines with a regular interval.



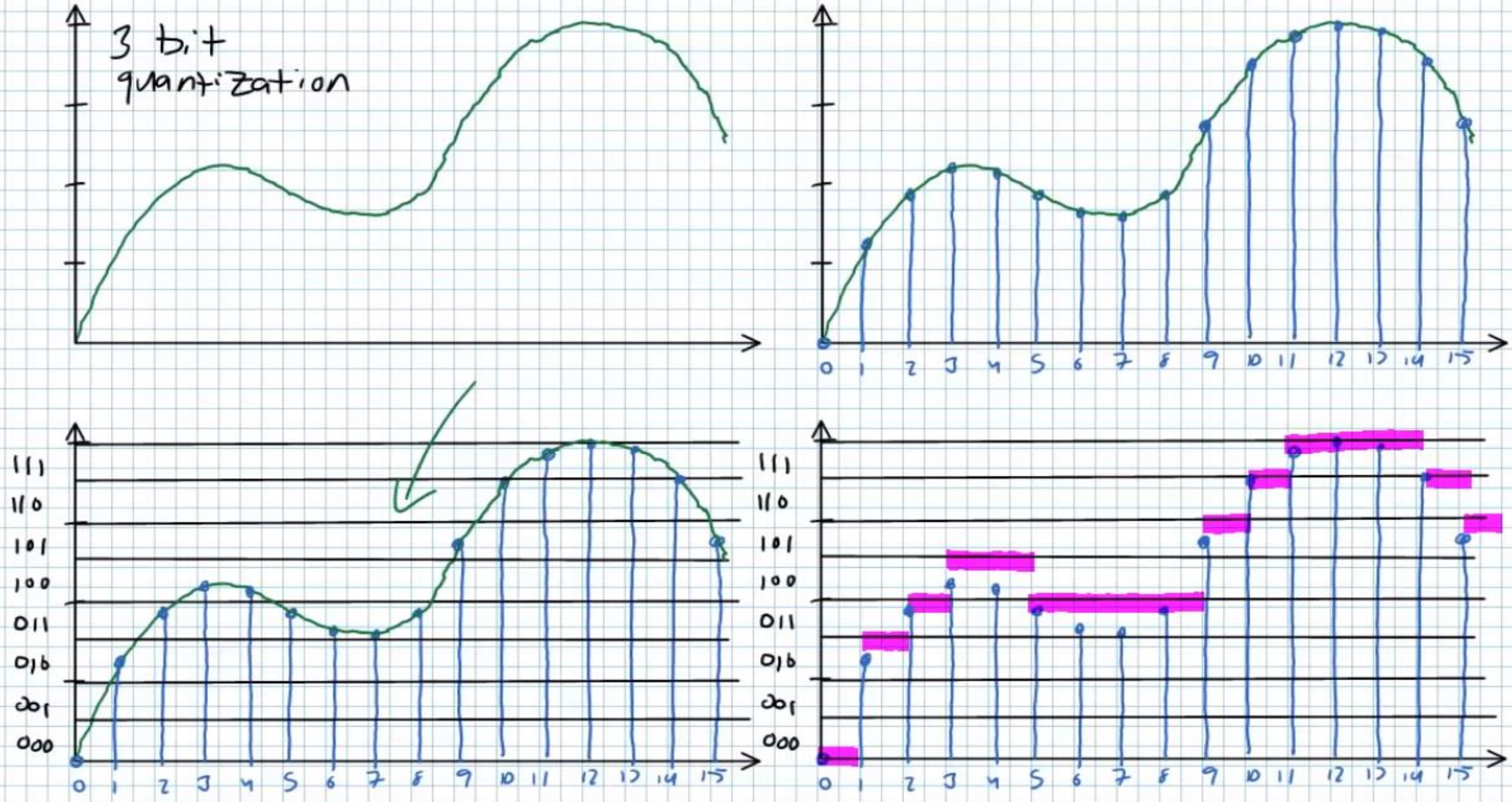
Quantizing process is like drawing horizontal lines and each line has a specific measurable bit value,

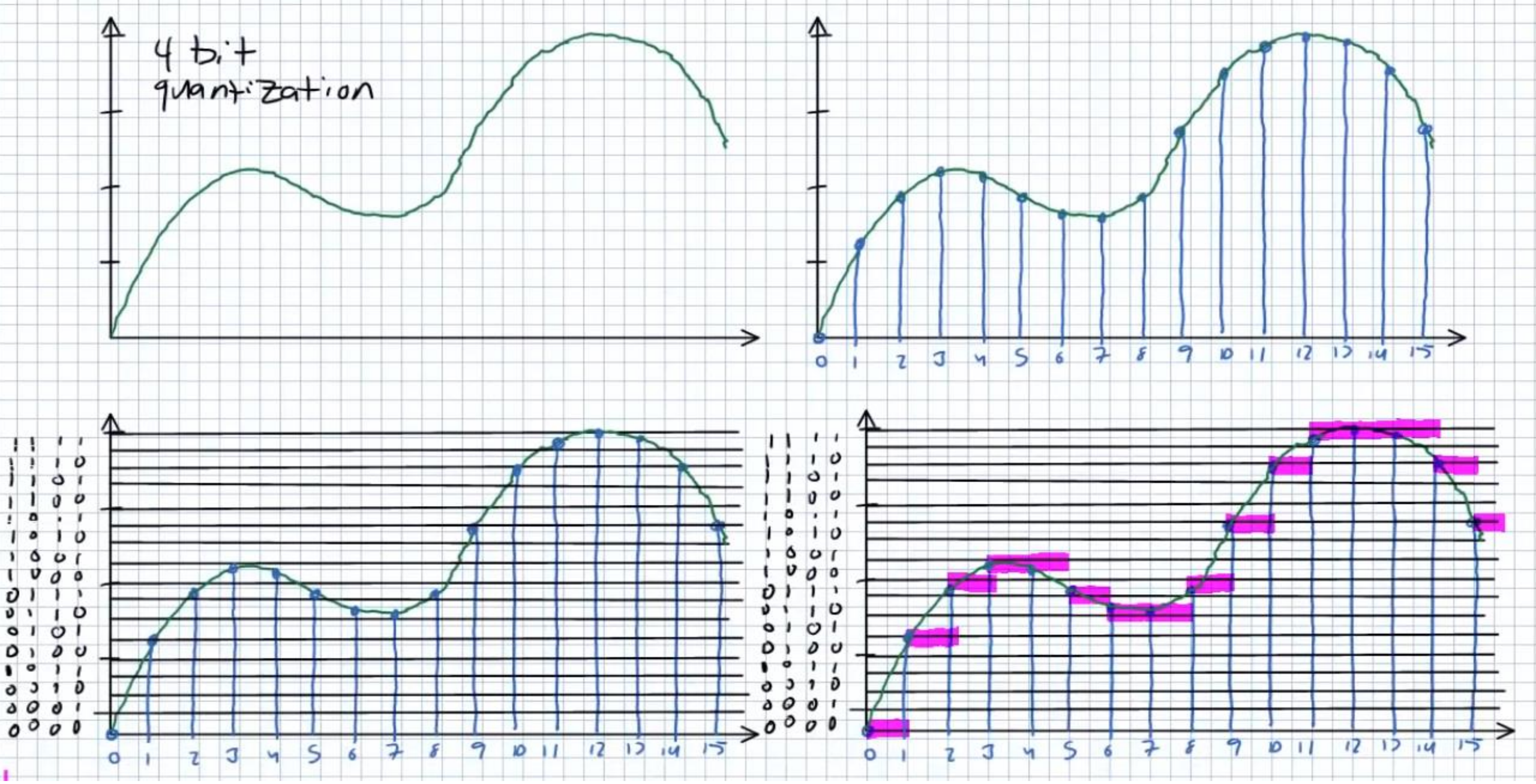


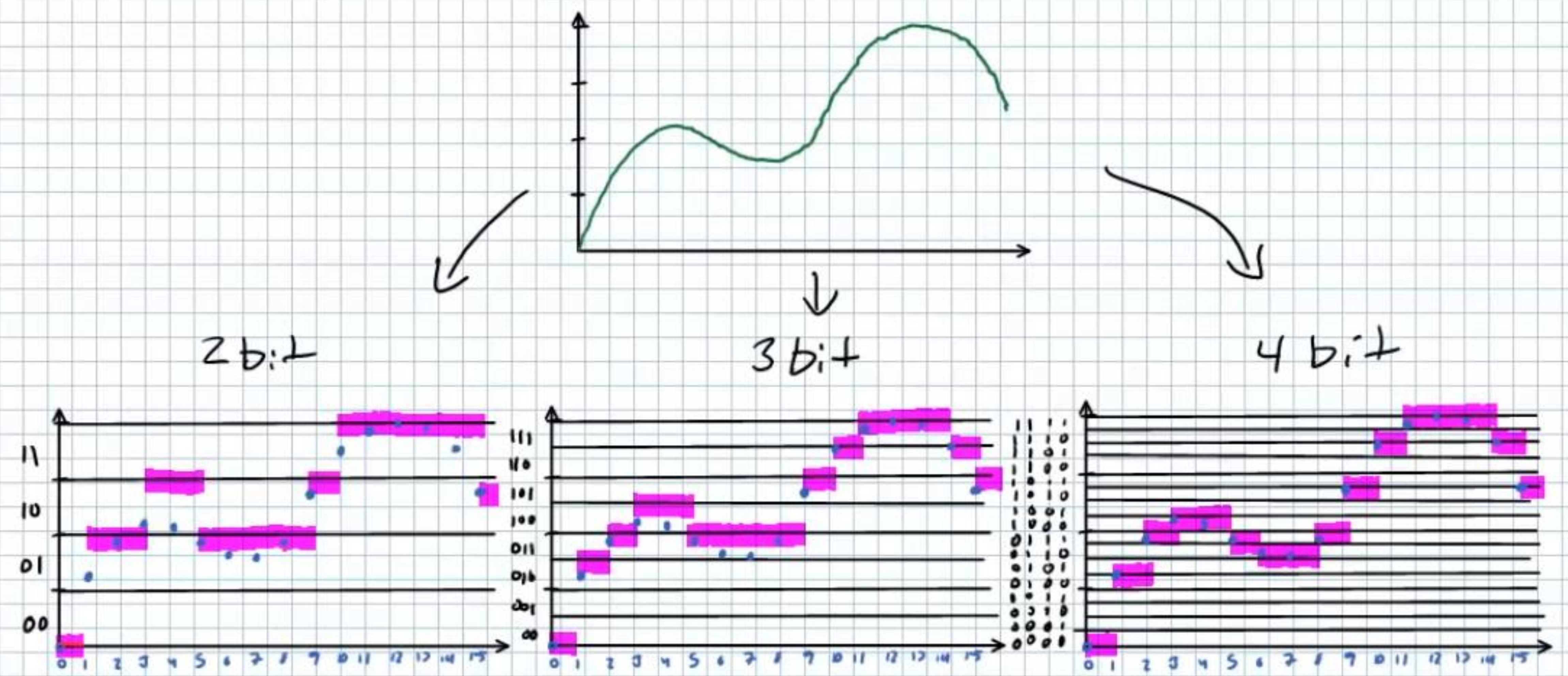
and then make each sample flat top
match a specific horizontal line.



In this way, each sample can be uniformly encoded with specific bit values.


3 bit
quantization
0 1 2 3 4 5 6 7 8 9 10 11 12 13 14 15
111
110
101
100
011
010
001
000
0 1 2 3 4 5 6 7 8 9 10 11 12 13 14 15
111
110
101
100
011
010
001
000
0 1 2 3 4 5 6 7 8 9 10 11 12 13 14 15


4 bit
quantization


2 bit
3 bit
4 bit

PCM

- PCM consists of three steps to digitize an analog signal:
  1. Sampling
  2. Quantization
  3. Binary encoding
- Before we sample, we have to filter the signal to limit the maximum frequency of the signal as it affects the sampling rate.
- Filtering should ensure that we do not distort the signal, ie remove high frequency components that affect the signal shape.

Figure 1.1 *Components of PCM encoder*

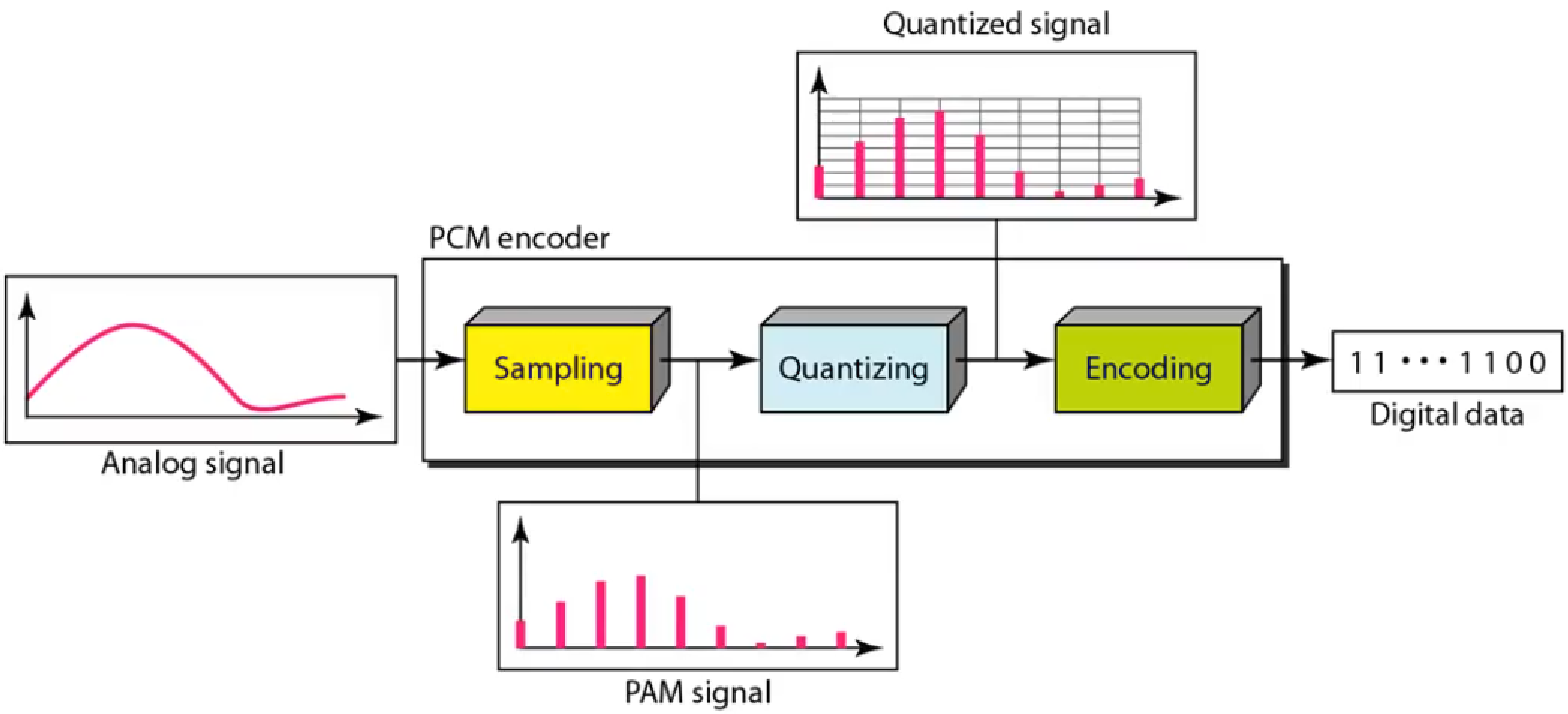


Figure 1.5 *Components of a PCM decoder*