



Introduction to Data Communications (COMP 3721)

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Fall 2021

Learning Outcomes of This Lecture

- **By the end of this lecture you will be able to**
 - Explain the Pulse Code Modulation (PCM) technique for analog-to-digital conversion.

Agenda

- Introduction
- Analog-To-Digital Conversion
- Summary

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Introduction



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- **Analog-To-Digital Conversion**
- Summary

Analog-To-Digital Conversion

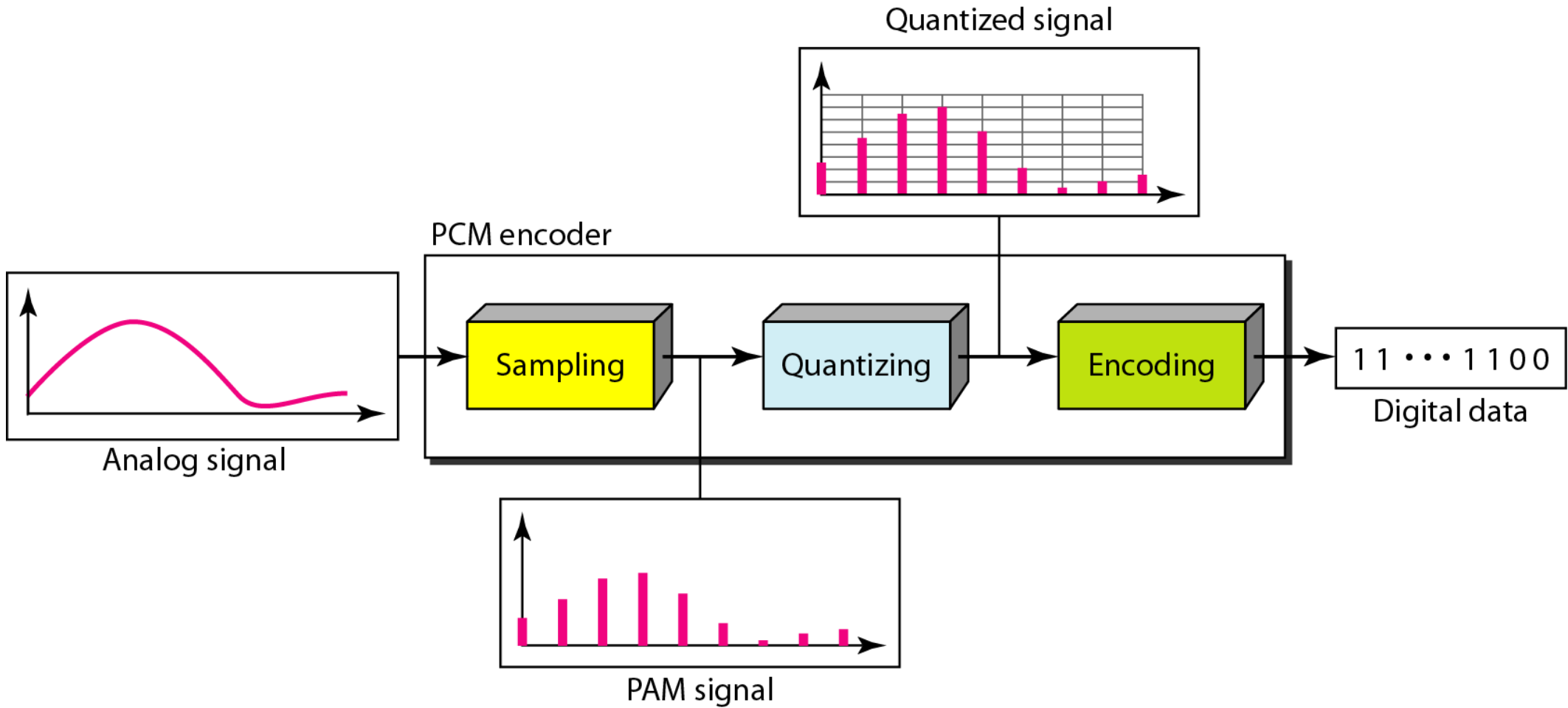
- **Digitization**

- converting an **analog signal** to **digital data**

Analog-To-Digital Conversion

- **Digitization**
 - converting an **analog signal** to **digital data**
- **Pulse Code Modulation (PCM)**
 - the most commonly used technique for digitization.

PCM



Sampling

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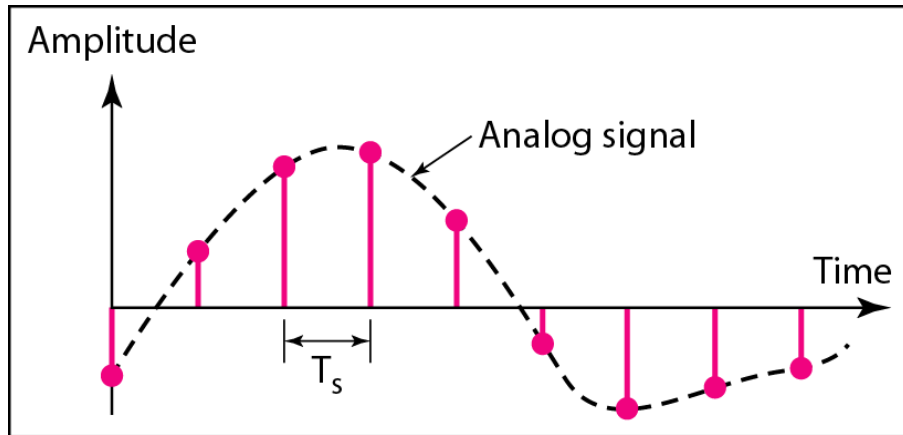
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Sampling

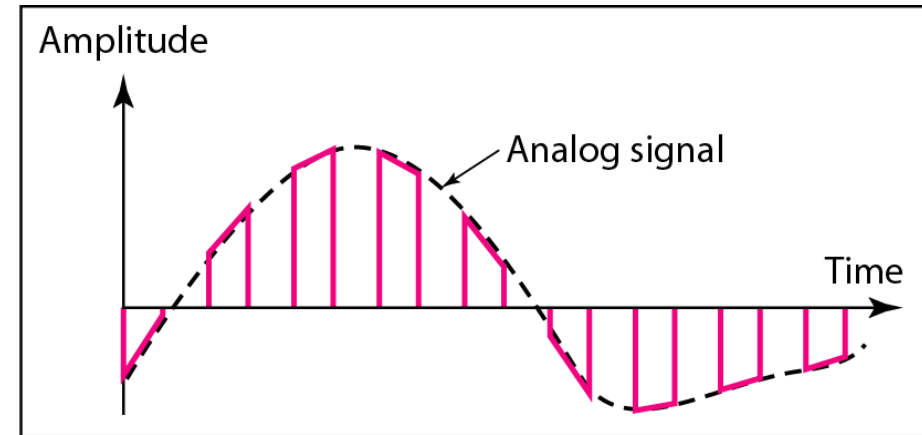
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The result of sampling is still an **analog signal** with **non-integral values** (a series of pulses, with amplitude values between the maximum and minimum amplitudes of the signal)

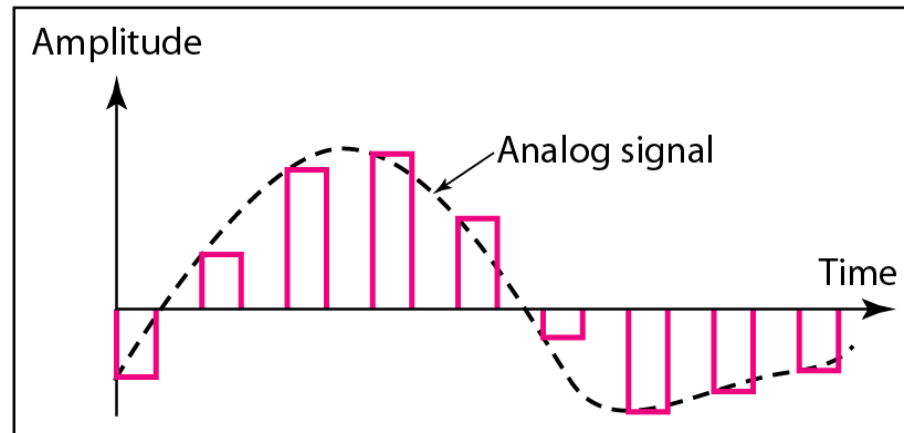
Sampling Methods



a. Ideal sampling

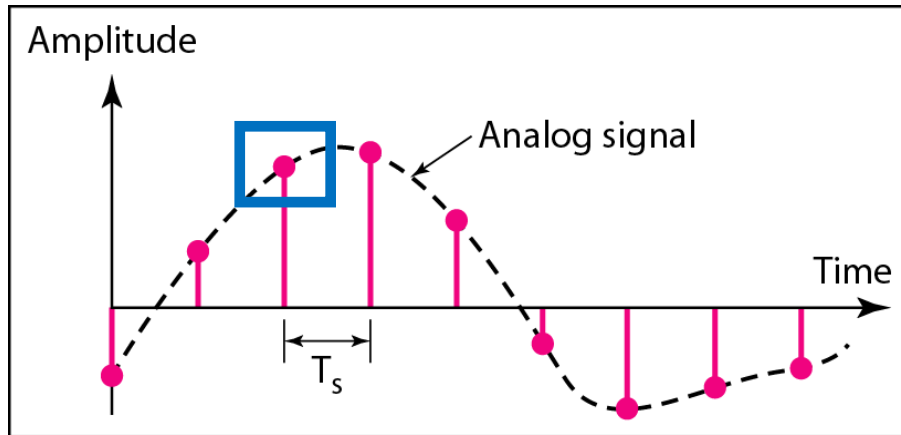


b. Natural sampling

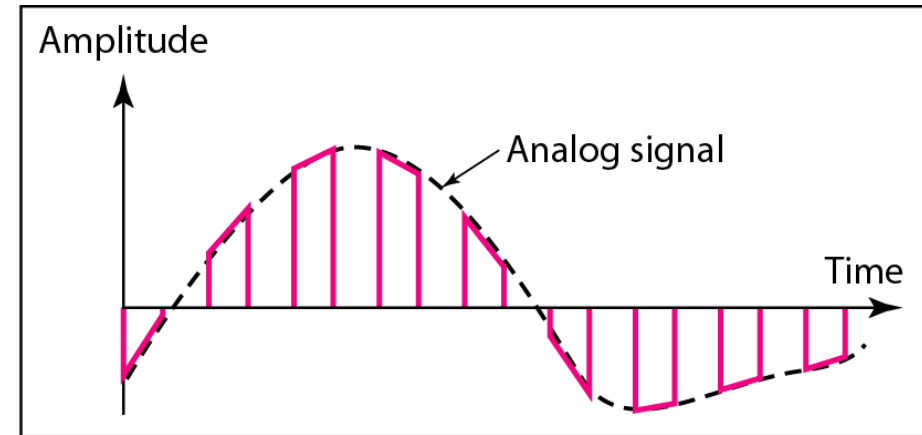


c. Flat-top sampling (sample and hold)

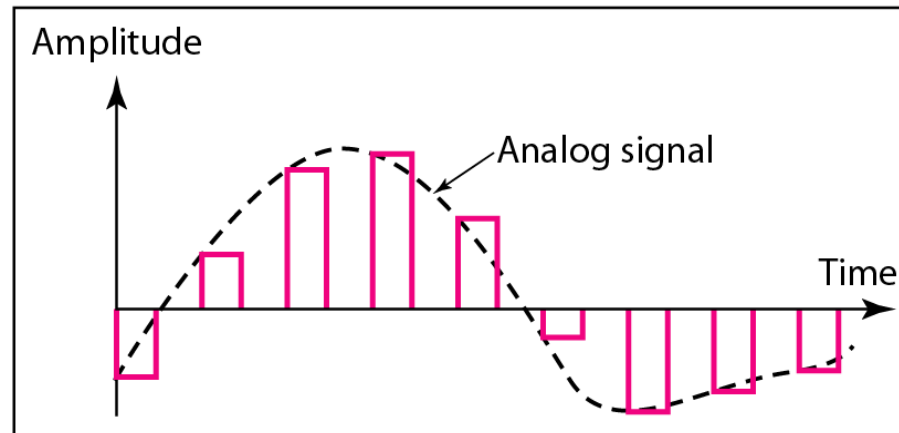
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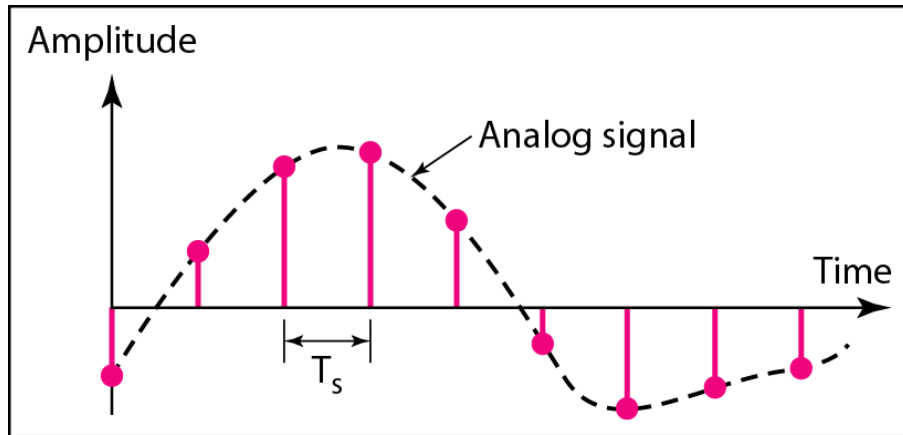


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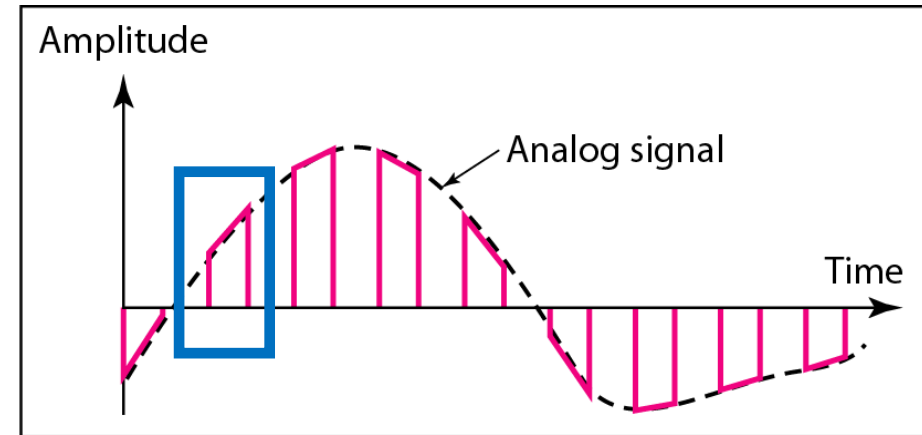


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Sampling Methods

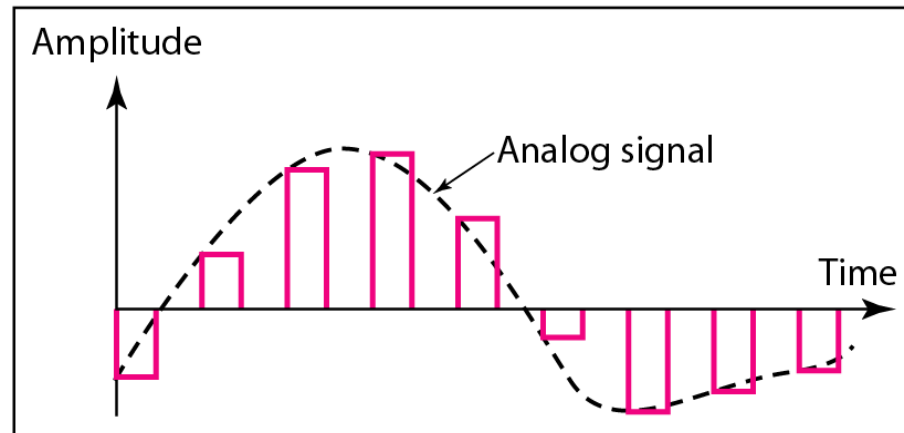


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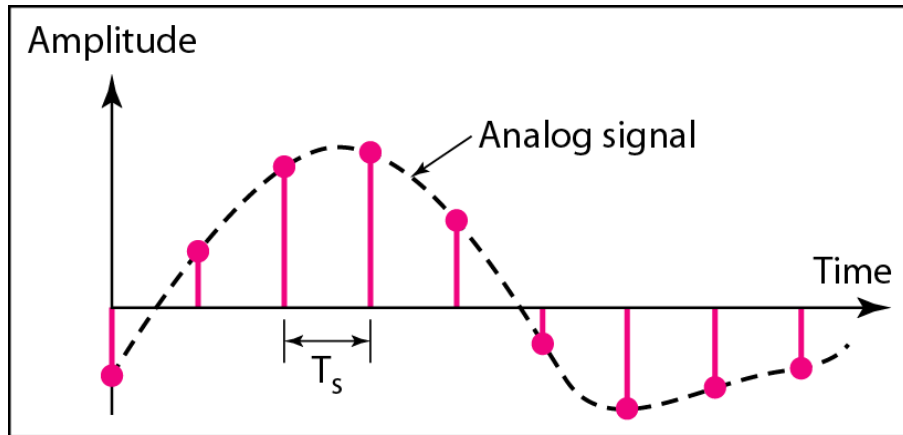
b. Natural sampling

uses a high-speed switch

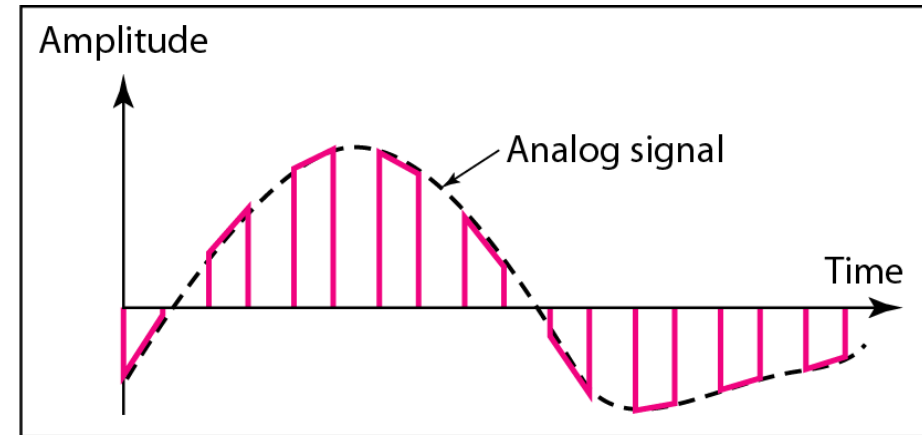


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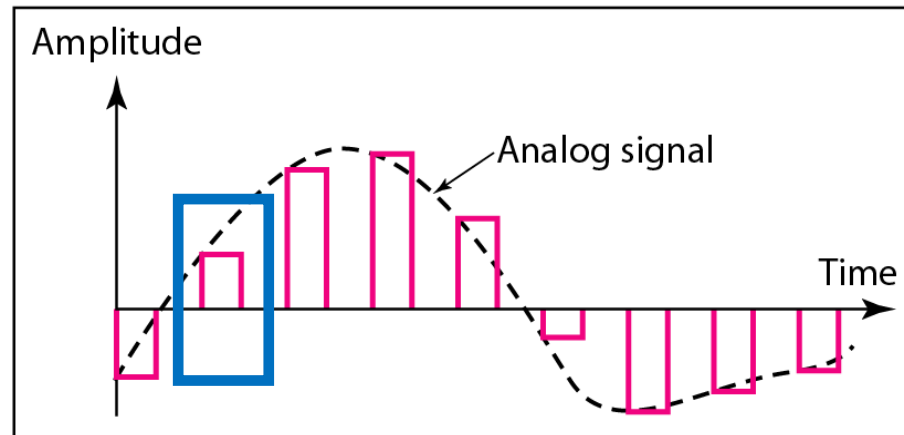
Sampling Methods



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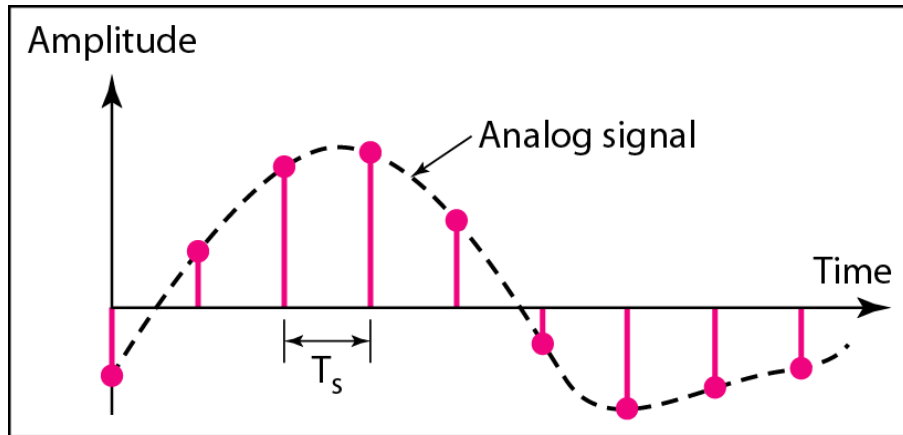


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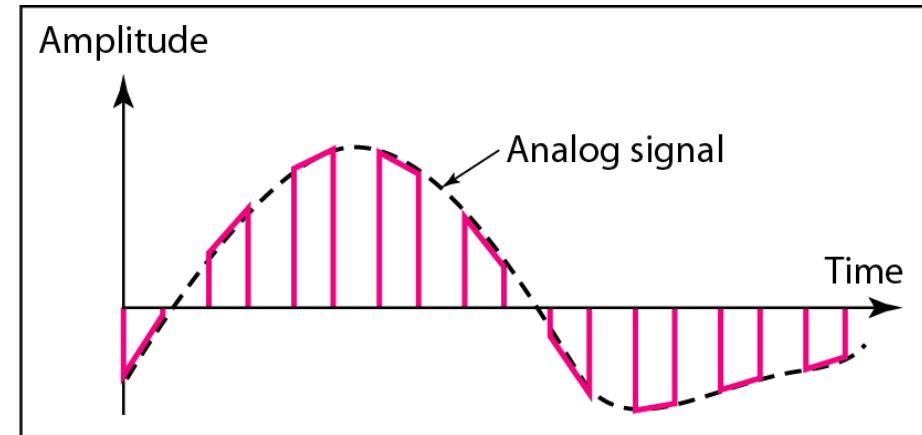
uses a circuit



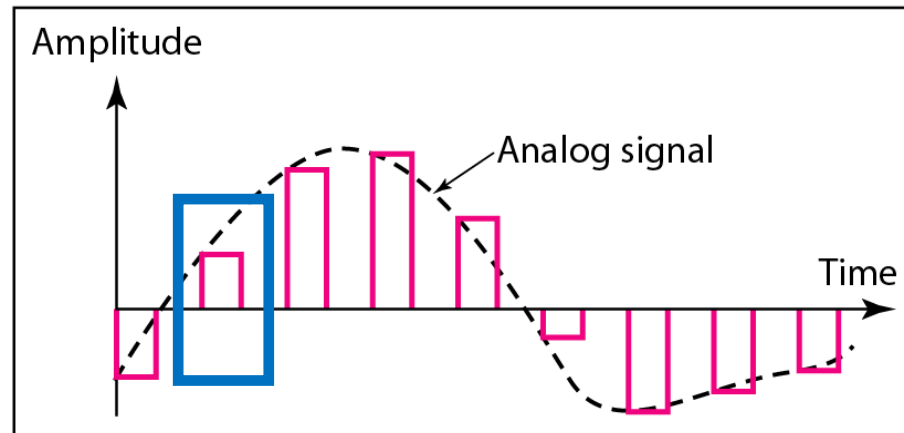
Sampling Methods



a. Ideal sampling



b. Natural sampling



c. Flat-top sampling (sample and hold)

**The most
commonly
used method**

Nyquist Theorem and Sampling Rate

Is there any restrictions on sampling rate (sampling frequency)?

Nyquist Theorem and Sampling Rate

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

Nyquist Theorem and Sampling Rate

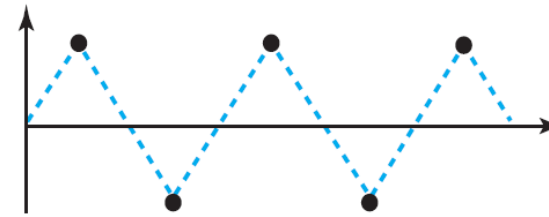
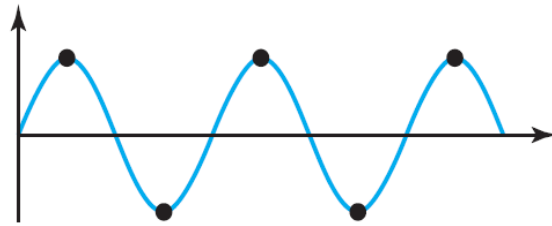
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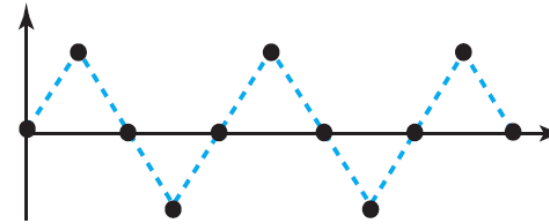
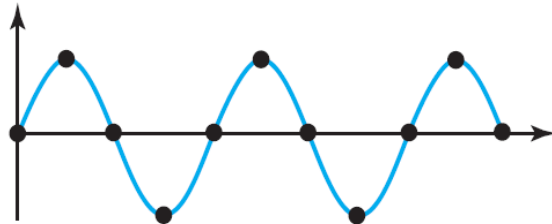
Low-pass analog signal \rightarrow bandwidth = highest frequency

Bandpass analog signal \rightarrow bandwidth < highest frequency

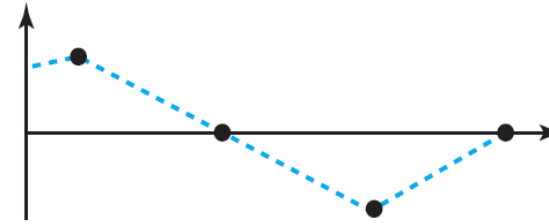
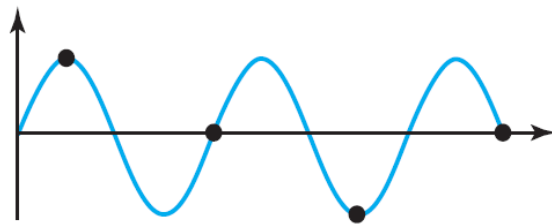
Nyquist Theorem and Sampling Rate – Example



a. Nyquist rate sampling: $f_s = 2f$

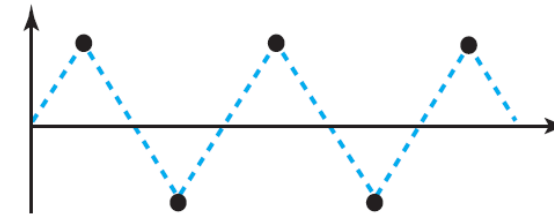
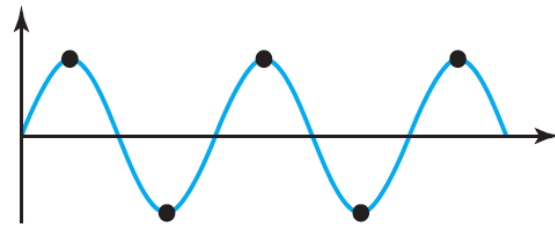


b. Oversampling: $f_s = 4f$

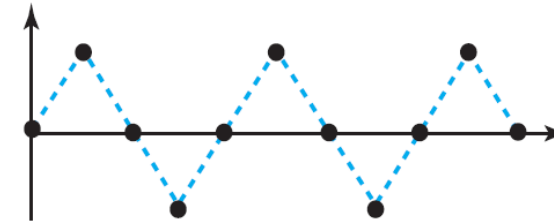
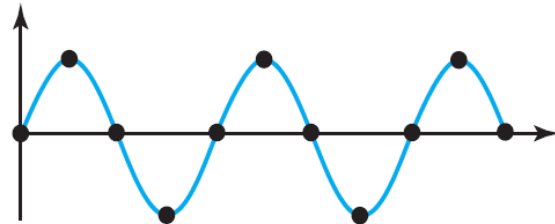


c. Undersampling: $f_s = f$

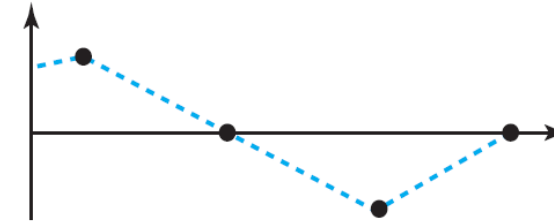
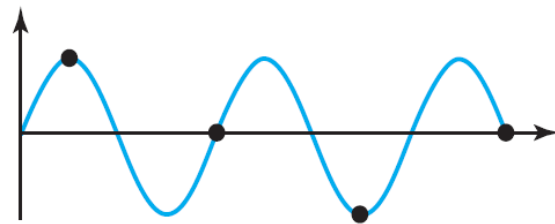
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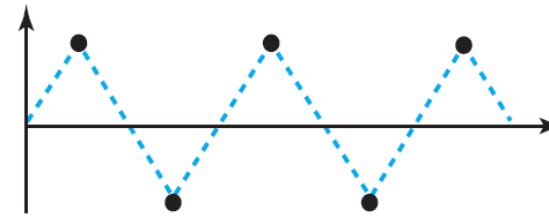
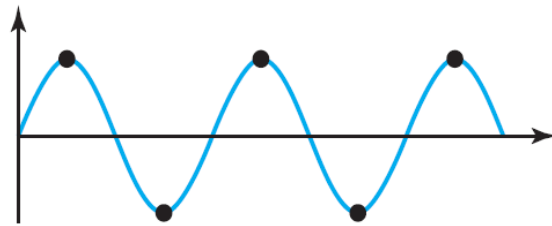


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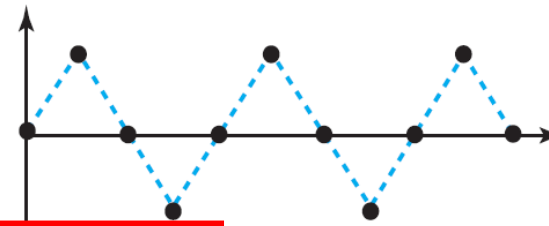
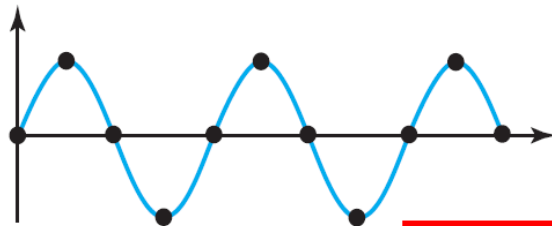
A good approximation of the original sine wave



Nyquist Theorem and Sampling Rate – Example

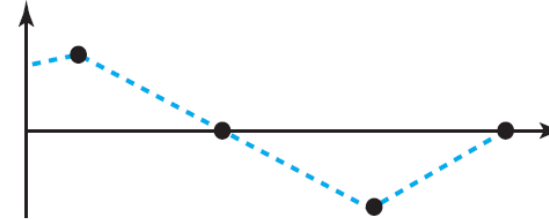
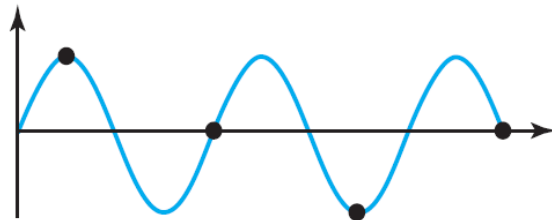


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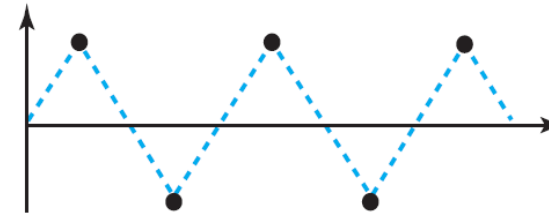
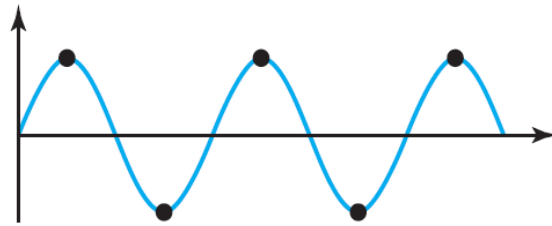
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**Same
approximation
but redundant
and unnecessary**

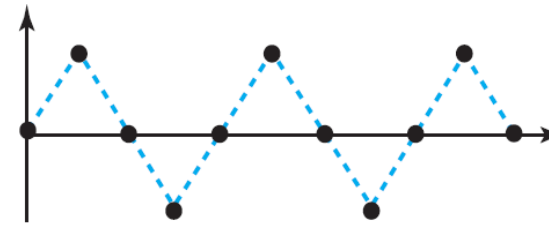
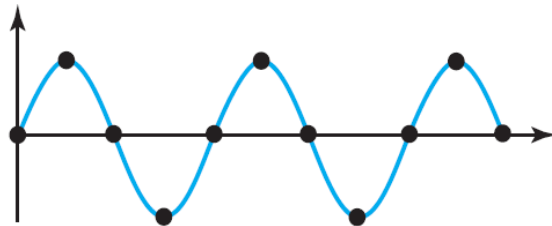


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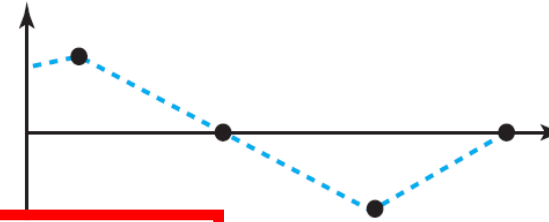
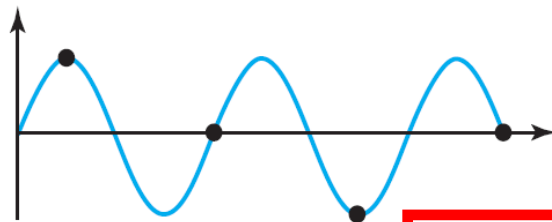
Nyquist Theorem and Sampling Rate – Example



a. Nyquist rate sampling: $f_s = 2f$



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c. Undersampling: $f_s = f$

**Produces a
signal that does
not look like the
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Nyquist Theorem and Sampling Rate – Example

What is the minimum sampling rate for a **low-pass signal** that has a bandwidth of 100 kHz?

Nyquist Theorem and Sampling Rate – Example

What is the minimum sampling rate for a **low-pass signal** that has a bandwidth of 100 kHz?

For a low-pass signal, bandwidth = highest frequency (f_{max})

Therefore,

Minimum sampling rate = $2 \times 100,000 = 200,000$ samples per second

Nyquist Theorem and Sampling Rate – Example

What is the minimum sampling rate for a **bandpass signal** that has a bandwidth of 100 kHz?

Nyquist Theorem and Sampling Rate – Example

What is the minimum sampling rate for a **bandpass signal** that has a bandwidth of 100 kHz?

We cannot find the minimum sampling rate because we do not know the maximum frequency of the signal.

Quantization

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- Quantization is an **approximation** process.

- **Steps involved in the quantization process:**

- 1) Assume that the original analog signal has instantaneous amplitudes between V_{\min} and V_{\max} .

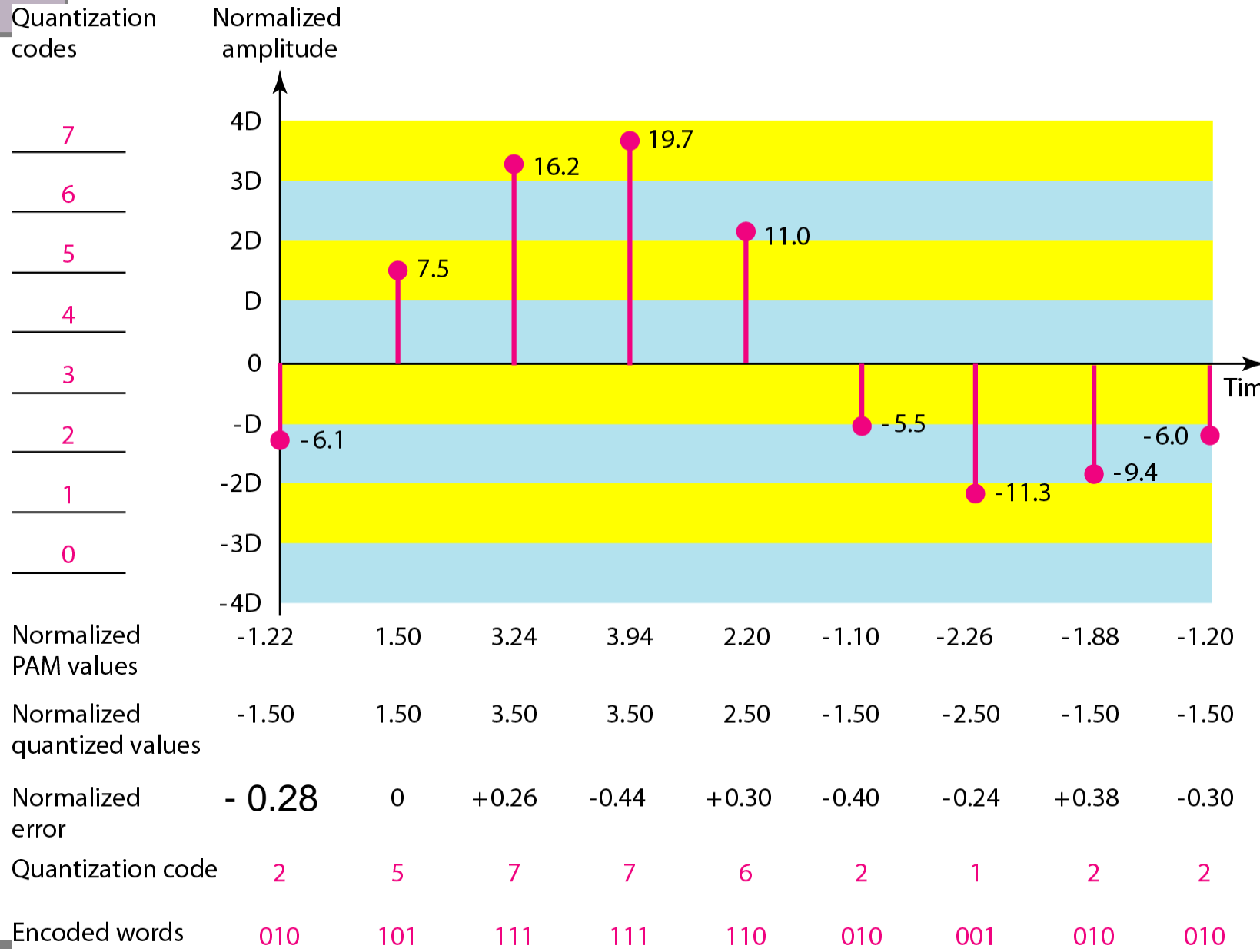
- 2) Divide the range into L zones (L also called the number of quantization levels), each of height Δ (delta)

$$\Delta = \frac{V_{\max} - V_{\min}}{L}$$

- 3) Assign quantized values of 0 to $L - 1$ to the midpoint of each zone.

- 4) Approximate the value of the sample amplitude to the quantized values.

Quantization - Example

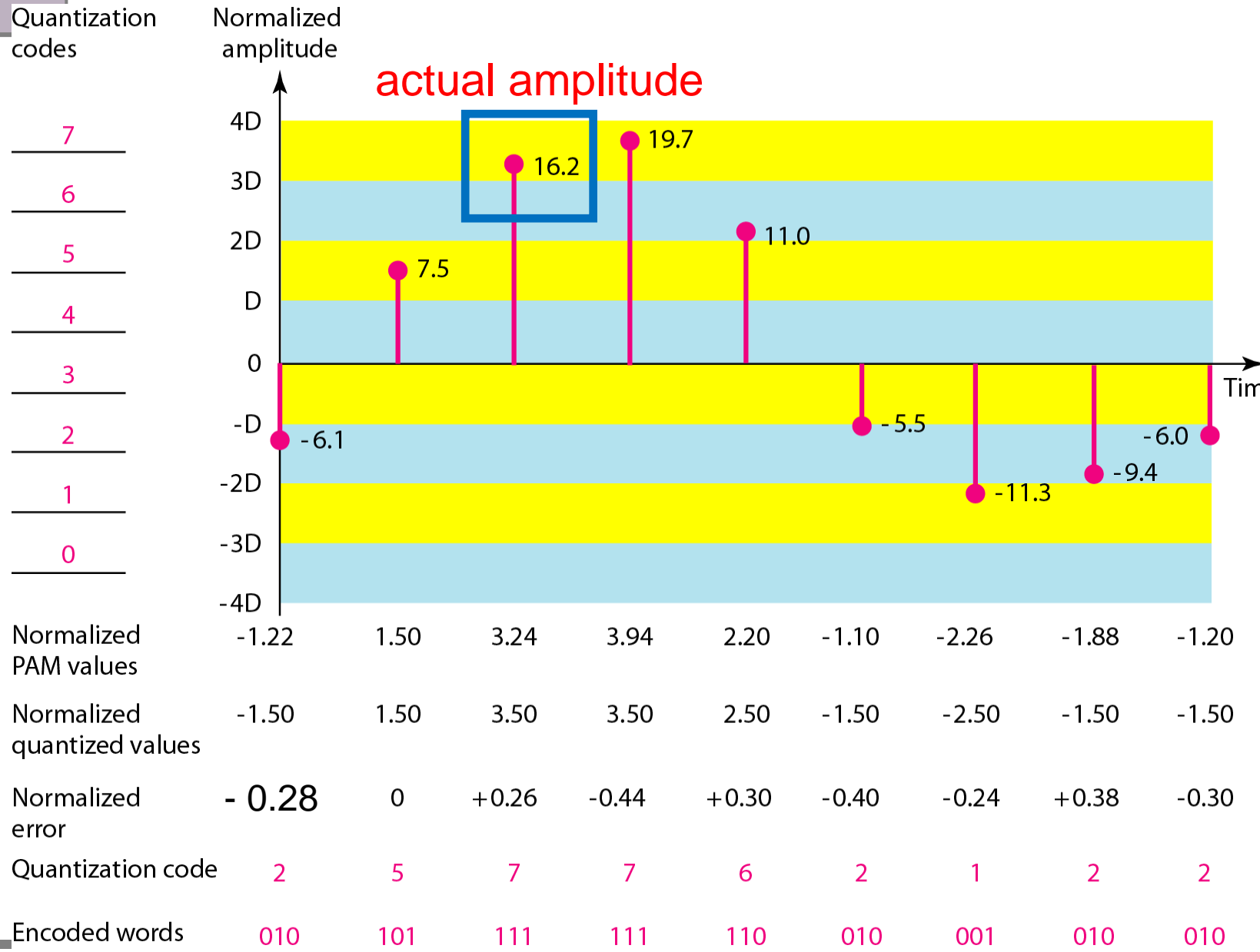


Sample amplitudes are between -20 V and +20 V

$L = 8, \Delta = 5 \text{ V}$

Zones: (-20 , -15), (-15, -10), (-10, -5), (-5, 0), (0, 5), (5, 10), (10, 15), (15, 20)

Quantization - Example

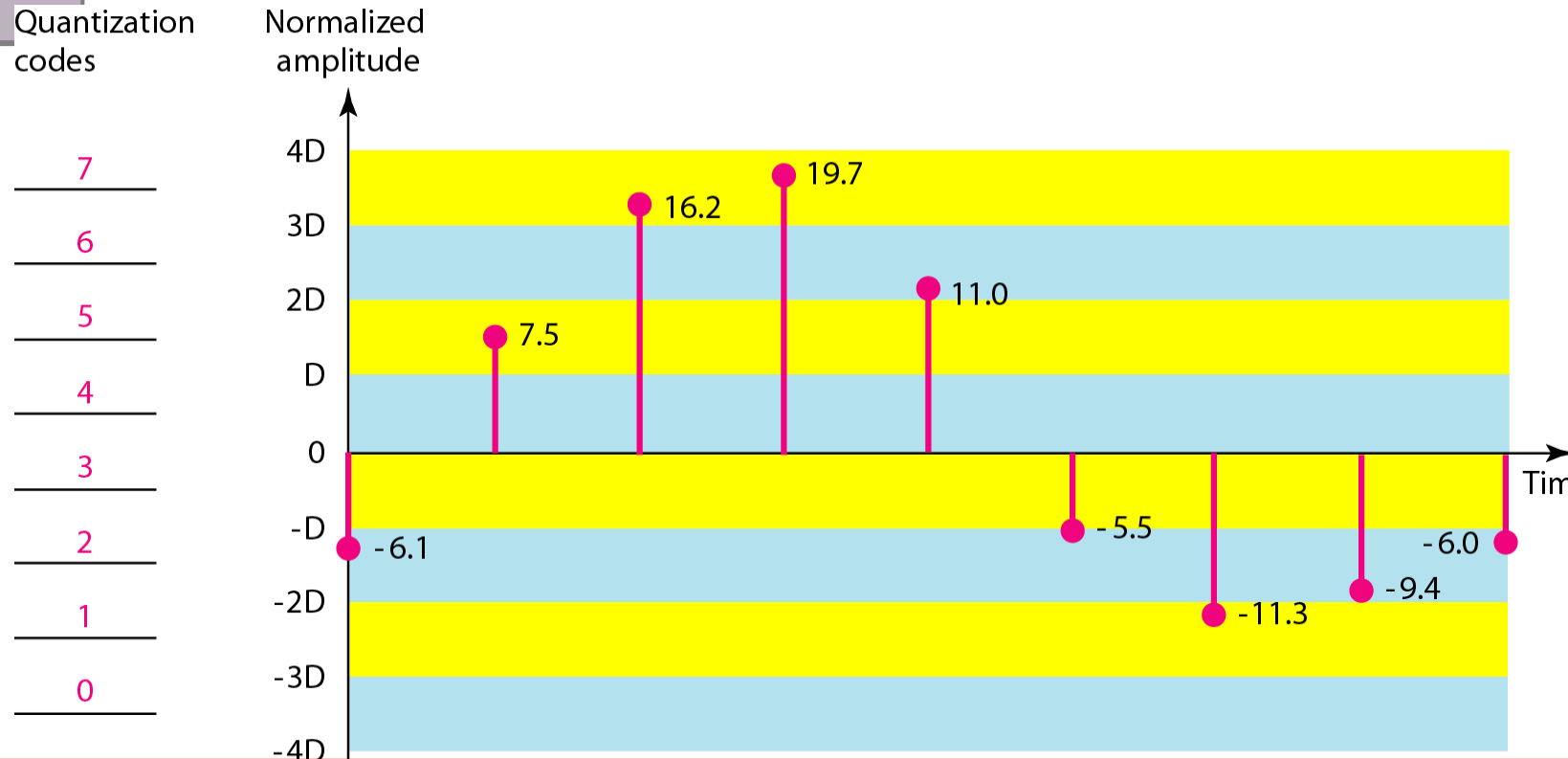


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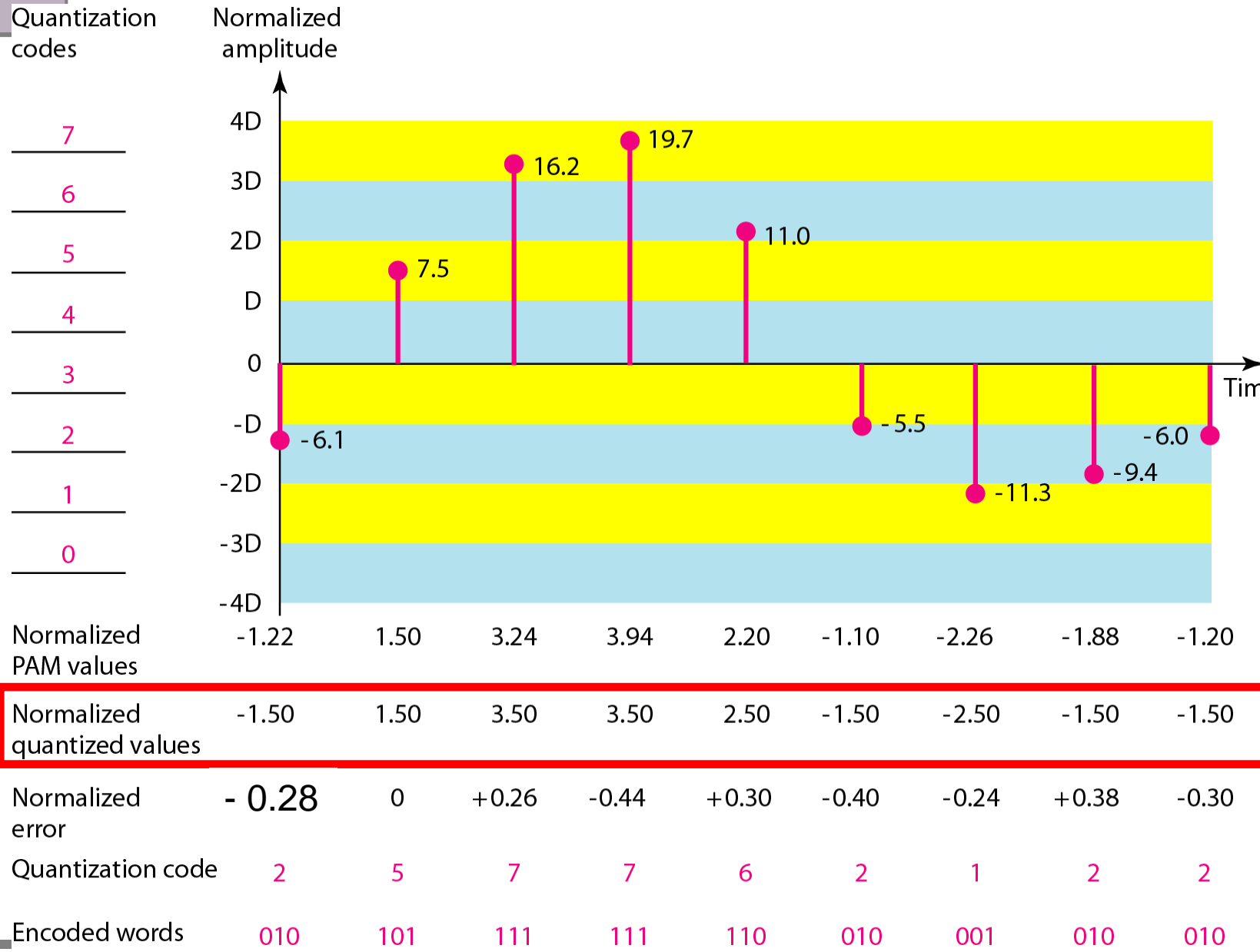
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Normalized PAM values	-1.22	1.50	3.24	3.94	2.20	-1.10	-2.26	-1.88	-1.20
Normalized quantized values	-1.50	1.50	3.50	3.50	2.50	-1.50	-2.50	-1.50	-1.50
Normalized error	-0.28	0	+0.26	-0.44	+0.30	-0.40	-0.24	+0.38	-0.30
Quantization code	2	5	7	7	6	2	1	2	2
Encoded words	010	101	111	111	110	010	001	010	010

actual amplitude/ Δ
e.g., $-6.1 / 5 = -1.22$

Quantization - Example



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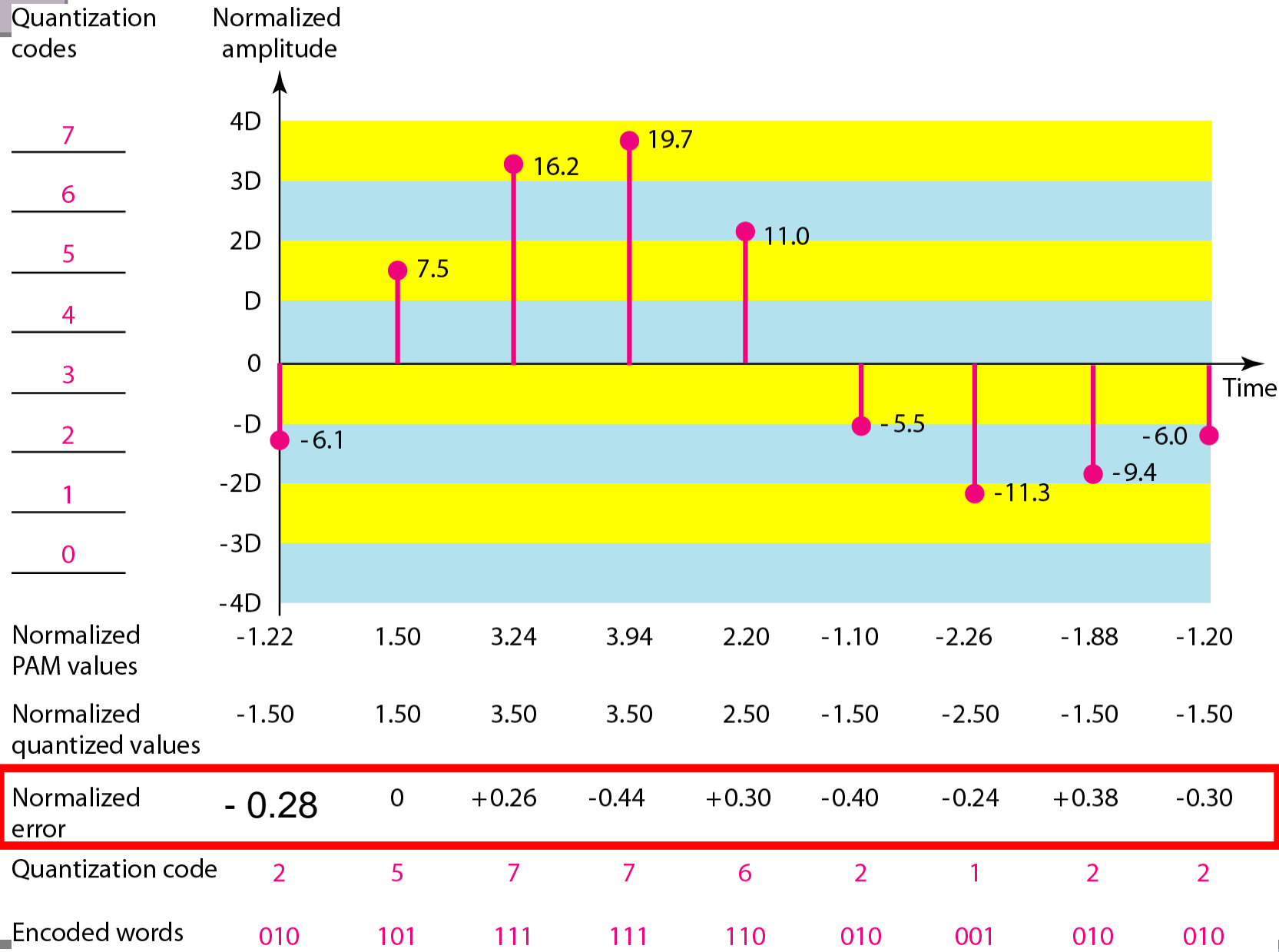
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Normalize zones:
 (-4, -3), (-3, -2),
 (-2, -1), (-1, 0), (0, 1),
 (1, 2), (2, 3), (3, 4)
 -1.22 belongs to zone (-2, -1) → normalized quantized value = midpoint of this zone = -1.5

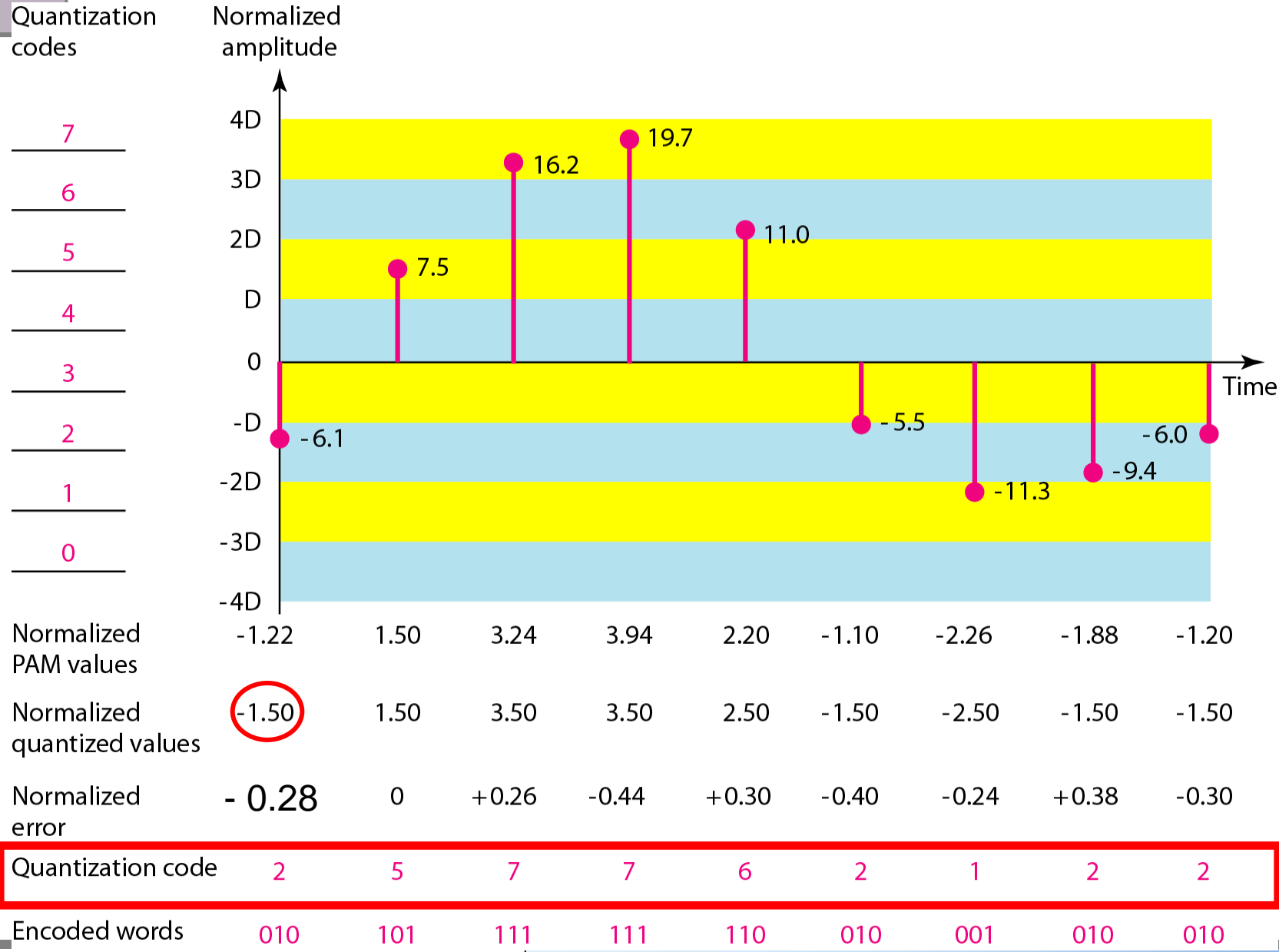
Quantization - Example

Sample amplitudes are between -20 V and +20 V
 $L = 8, \Delta = 5 \text{ V}$



Normalized error is the difference between normalized quantized value and normalized PAM value

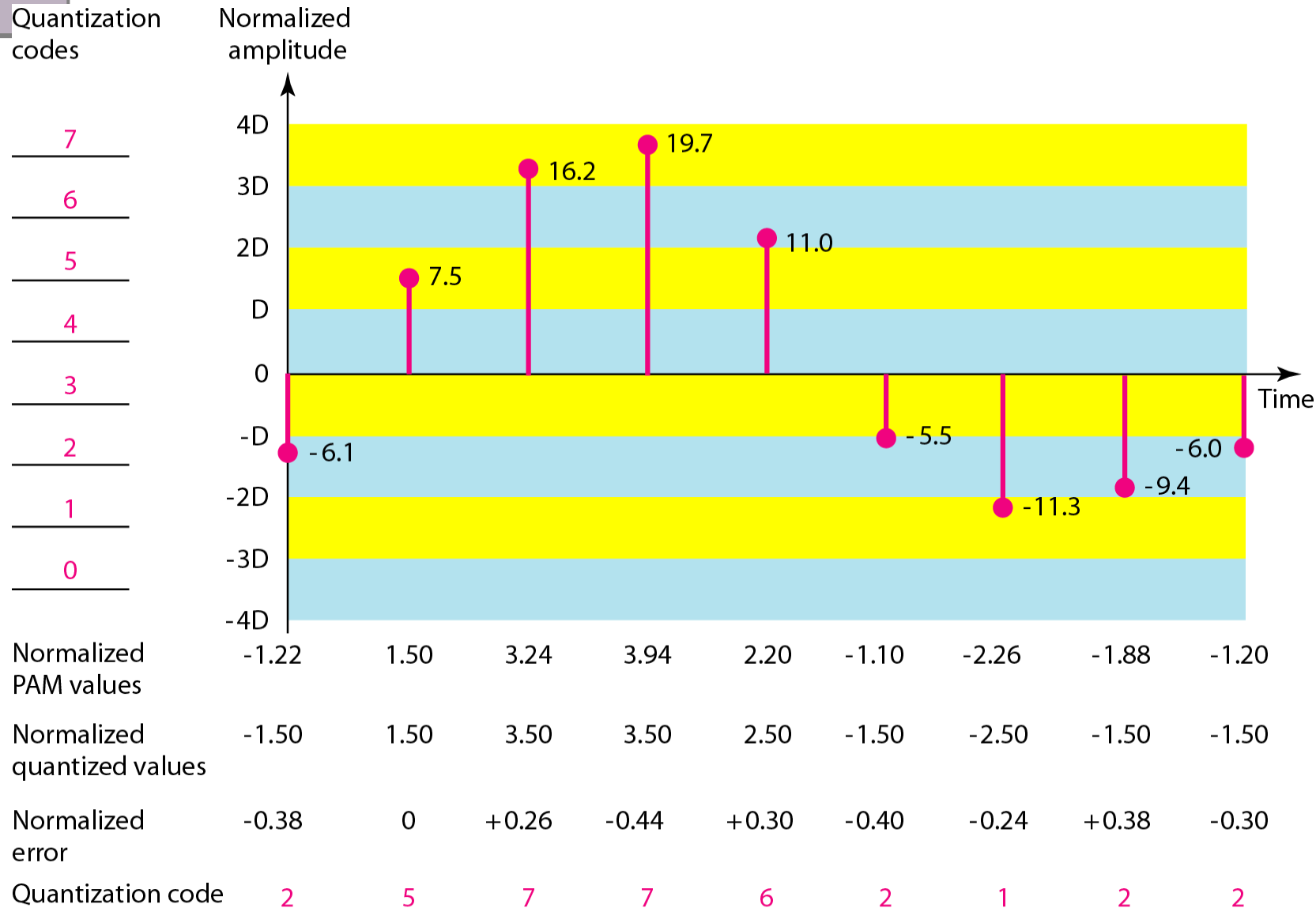
Quantization - Example



Sample amplitudes are between -20 V and +20 V
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Normalized zones are mapped to 0-7:
 (-4, -3) → 0, (-3, -2) → 1, (-2, -1) → 2, (-1, 0) → 3, (0, 1) → 4, (1, 2) → 5, (2, 3) → 6, (3, 4) → 7
 -1.5 → midpoint of zone 2

Quantization - Example



Sample amplitudes are between -20 V and +20 V
 $L = 8, \Delta = 5 \text{ V}$

Encoded words	010	101	111	111	110	010	001	010	010
---------------	-----	-----	-----	-----	-----	-----	-----	-----	-----

Encoding (Last step of PCM)

Quantization Levels

- **Choosing L (the number of quantization levels)** depends on:
 - the range of the amplitudes of the analog signal
 - the required accuracy of recovering the signal
- If the amplitude of a signal fluctuates between two values only, $L = 2$
- For a **signal with many amplitude values**, e.g., voice, more quantization levels are needed.
- In audio digitizing, $L = 256$
- Choosing **lower values of L** increases the **quantization error** if there is **a lot of fluctuation** in the signal.

Quantization Error (Noise)

- $-\Delta / 2 \leq \text{quantization error} \leq \Delta / 2$
- The **quantization error changes the SNR of the signal** \rightarrow the upper limit capacity is **decreased** (according to Shannon Capacity)
- The contribution of the **quantization error** to the **SNR_{dB}** of the signal depends on the number of quantization levels **L** , or the bits per sample **n_b** .

Uniform Quantization

- Issues with **uniform quantization**
 - only optimal for uniformly distributed signal.
 - often the distribution of the instantaneous amplitudes in the analog signal is not uniform.
 - applications such as speech and music (real audio signals) are more concentrated near zeros (lower amplitudes).
 - human ear is more sensitive to quantization errors at small values.

Encoding

- Each quantized sample can be changed to an n_b -bit code word

$$n_b = \log_2 L \quad (\text{L is the number of quantization levels/zones})$$

$$\text{Number of bits per sample} = n_b = \log_2 L$$

$$\begin{aligned} \text{Bit rate} &= \text{sampling rate} \times \text{number of bits per sample} \\ &= f_s \times n_b \end{aligned}$$

PCM Bandwidth

- Given the bandwidth of a **low-pass analog signal**, we want to find the **new minimum bandwidth of the channel that can pass the digitized version of this signal**.

$$B_{min} = c \times N \times 1/r = c \times n_b \times f_s \times 1/r = c \times n_b \times 2 \times f_{max} \times 1/r$$

$$B_{min} = c \times n_b \times 2 \times B_{analog} \times 1/r$$

If $c = 1/2$ (average case) and $r = 1$ (NRZ or bipolar line coding):

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

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If $c = 1/2$ (average case) and $r = 1$ (NRZ or bipolar line coding):

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

The minimum bandwidth of the digital signal is n_b times greater than the bandwidth of the analog signal. This is the price we pay for digitization.

Agenda

- Introduction
- Analog-To-Digital Conversion
- **Summary**

Summary

- PCM is a technique for analog-to-digital conversion.
- PCM includes sampling, quantizing and encoding.
- PCM requires more bandwidth than the bandwidth of the input analog signal.

References

[1] Behrouz A. Forouzan, Data Communications and Networking, 5th Ed, 2013, McGraw-Hill companies.