Digital Communications I: Modulation and Coding Course

Term 3 – 2008

Catharina Logothetis

Lecture 2

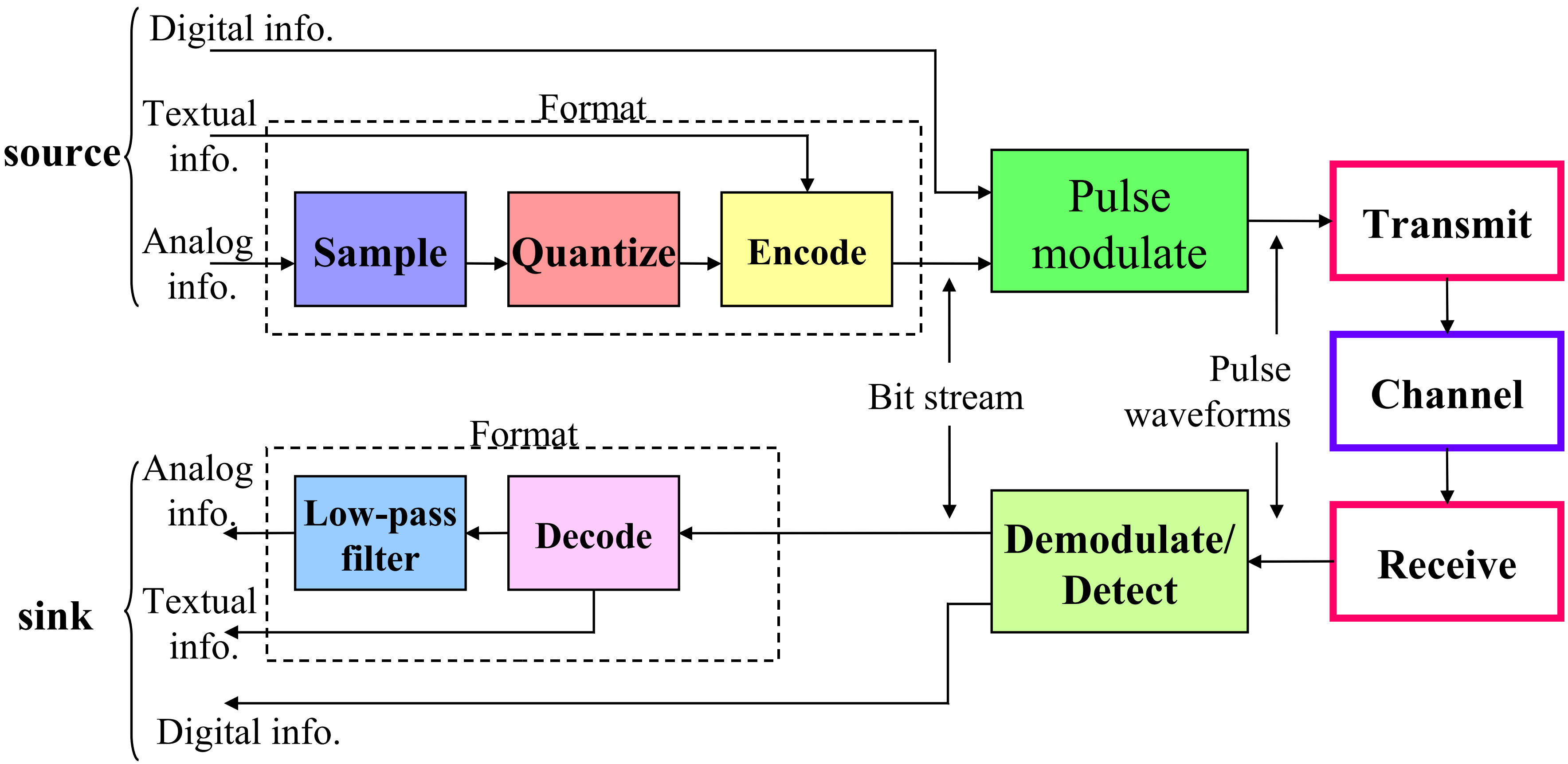
Last time, we talked about:

- Important features of digital communication systems
- Some basic concepts and definitions such as as signal classification, spectral density, random process, linear systems and signal bandwidth.

Today, we are going to talk about:

- The first important step in any DCS:
  - Transforming the information source to a form compatible with a digital system

Formatting and transmission of baseband signal

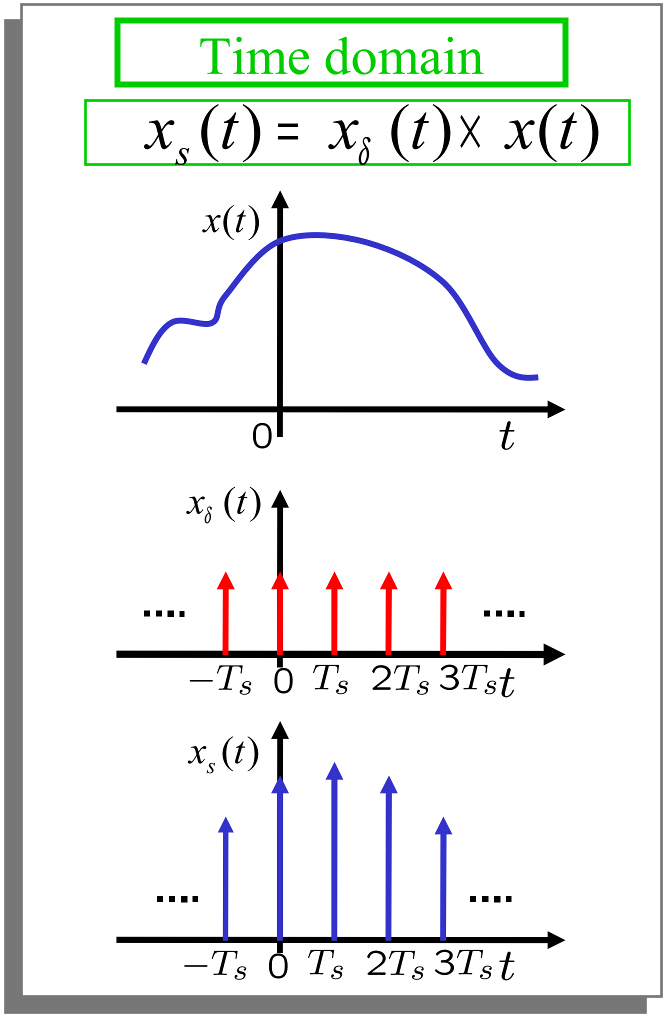


Format analog signals

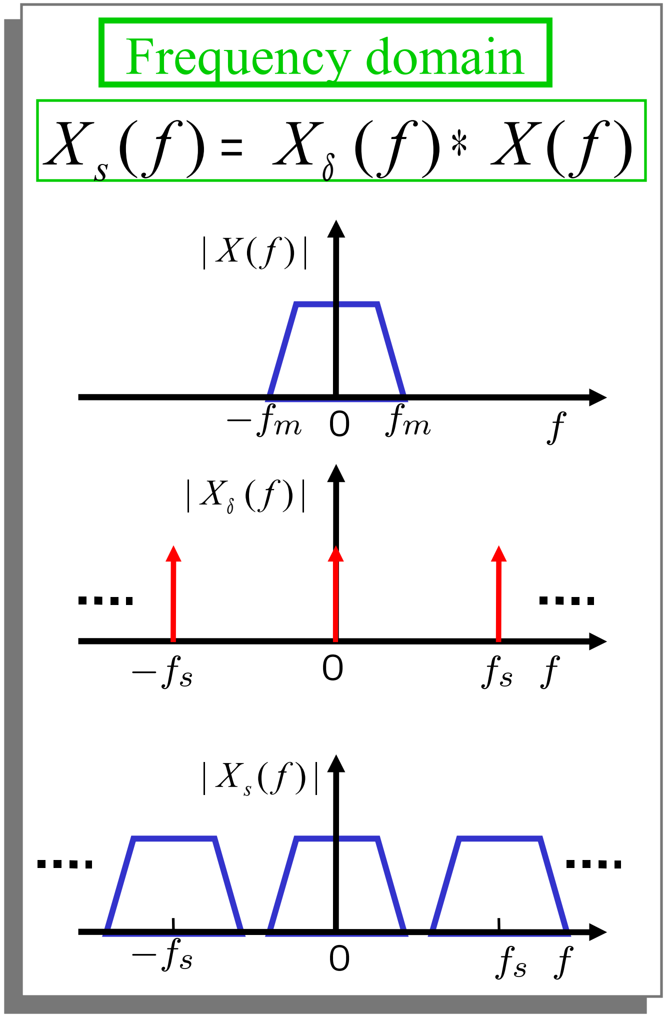
- To transform an analog waveform into a form that is compatible with a digital communication system, the following steps are taken:

1. Sampling
2. Quantization and encoding
3. Baseband transmission

Sampling

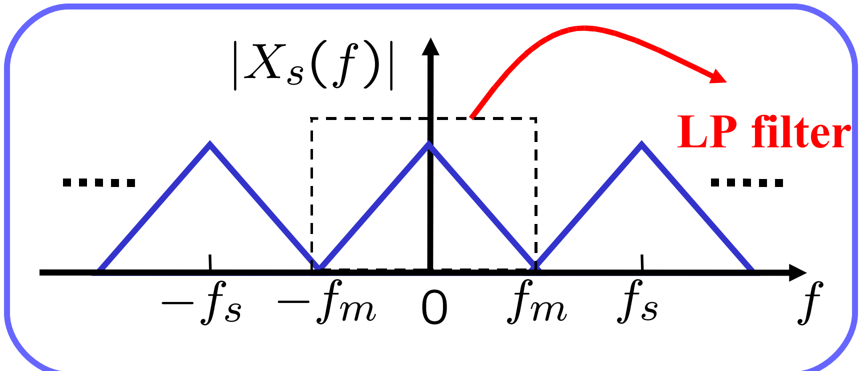


Time domain
$x_s(t) = x_\delta(t) \times x(t)$
$x(t)$
0
t
$x_\delta(t)$
....
....
$-T_s$ 0 T_s $2T_s$ $3T_s$ t
$x_s(t)$
....
....
$-T_s$ 0 T_s $2T_s$ $3T_s$ t



Frequency domain
$X_s(f) = X_\delta(f) * X(f)$
$|X(f)|$
$-f_m$ 0 f_m f
$|X_\delta(f)|$
....
....
$-f_s$ 0 f_s f
$|X_s(f)|$
....
....
$-f_s$ 0 f_s f

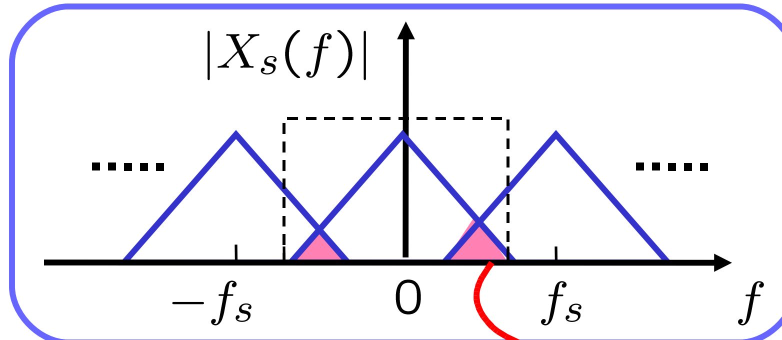
Aliasing effect



LP filter

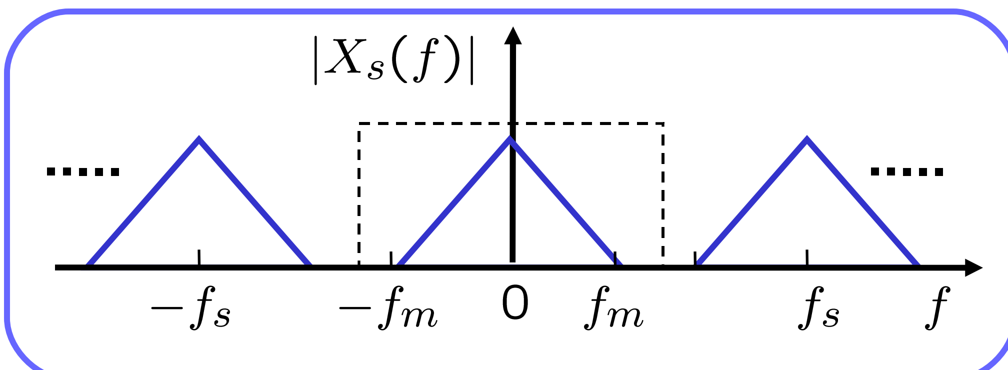
$$f_s = 2f_m$$

Nyquist rate



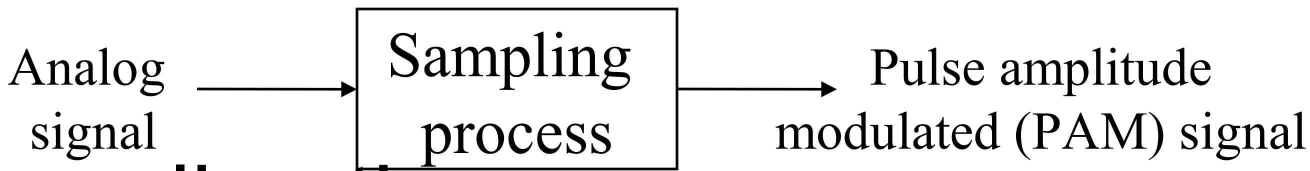
$$f_s < 2f_m$$

aliasing



$$f_s > 2f_m$$

Sampling theorem



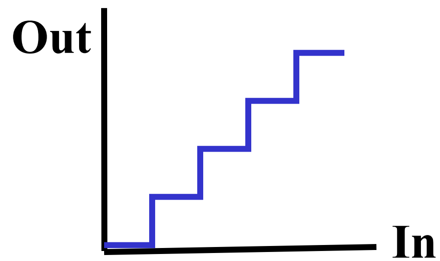
- Sampling theorem: A bandlimited signal with no spectral components beyond f_m, can be uniquely determined by values sampled at uniform intervals of

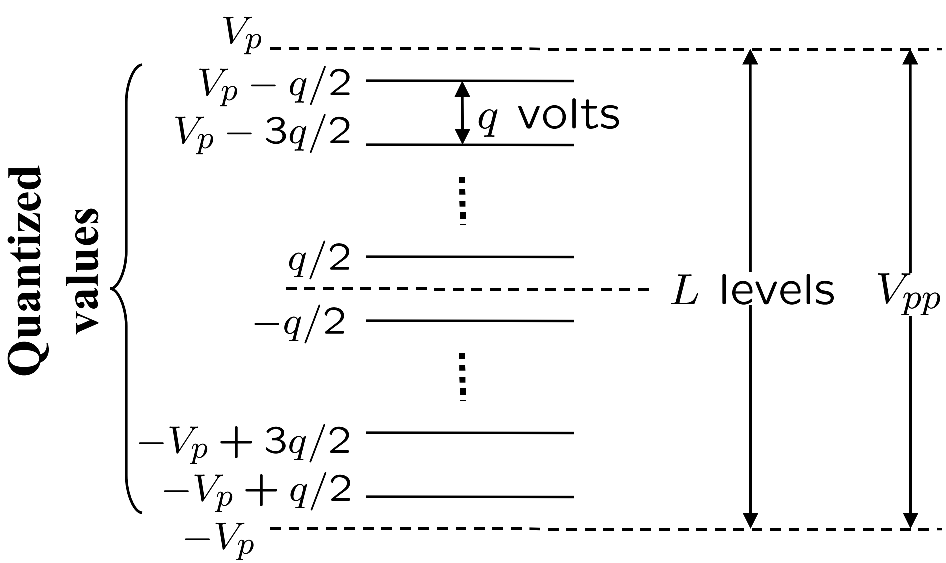
$$T_s \leq \frac{1}{2f_m}$$

  - The sampling rate, $f_s = \frac{1}{T_s} = 2f_m$ is called Nyquist rate.

Quantization

- **Amplitude quantizing:** Mapping samples of a continuous amplitude waveform to a finite set of amplitudes.





- Average quantization noise power

$$\sigma^2 = \frac{q^2}{12}$$

- Signal peak power

$$V_p^2 = \frac{L^2 q^2}{4}$$

- Signal power to average quantization noise power

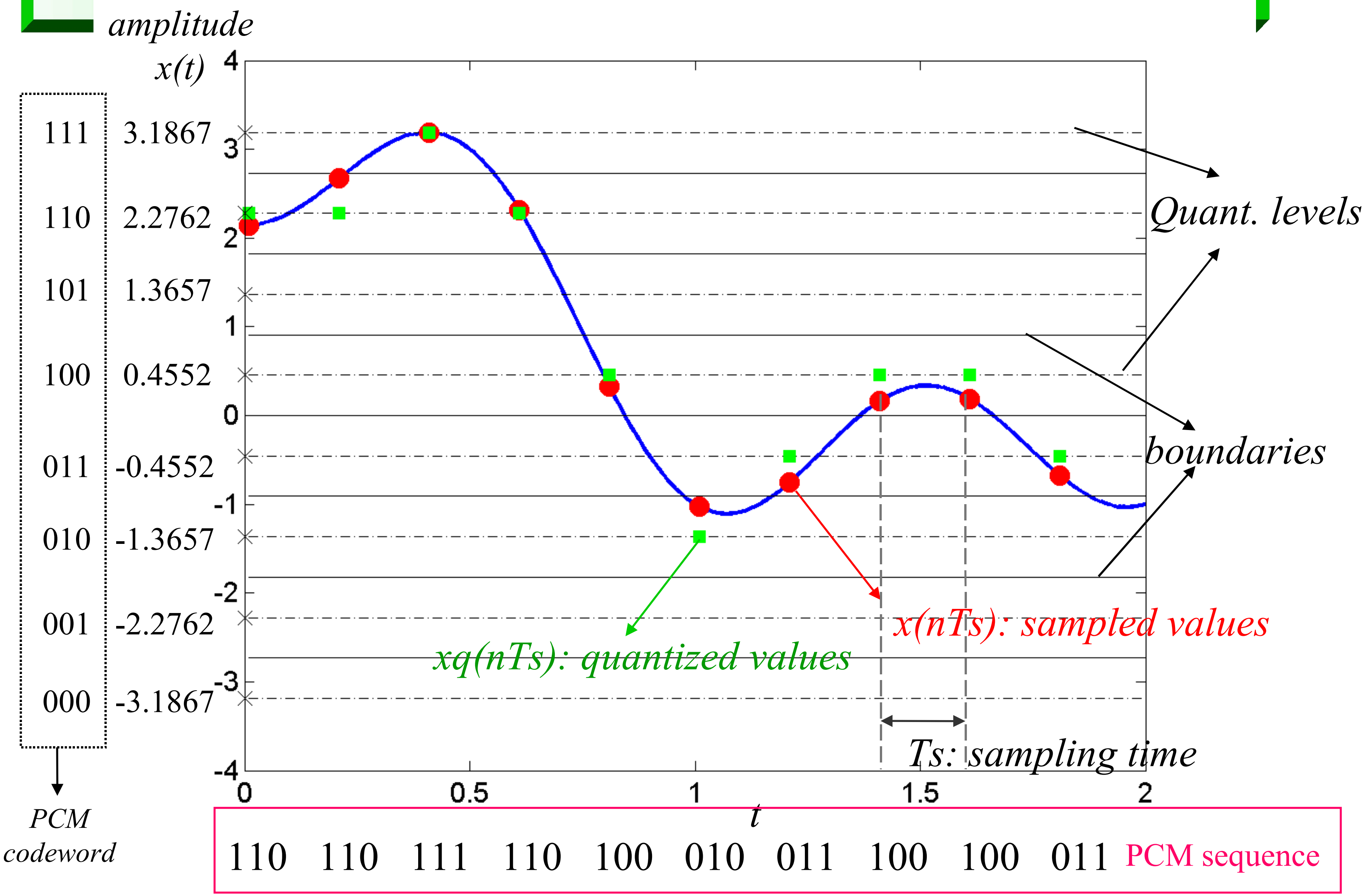
$$\left(\frac{S}{N}\right)_q = \frac{V_p^2}{\sigma^2} = 3L^2$$

Encoding (PCM)

- A uniform linear quantizer is called Pulse Code Modulation (PCM).
- Pulse code modulation (PCM): Encoding the quantized signals into a digital word (PCM word or codeword).
  - Each quantized sample is digitally encoded into an l bits codeword where L in the number of quantization levels and

$$l = \log_2 L$$

Quantization example

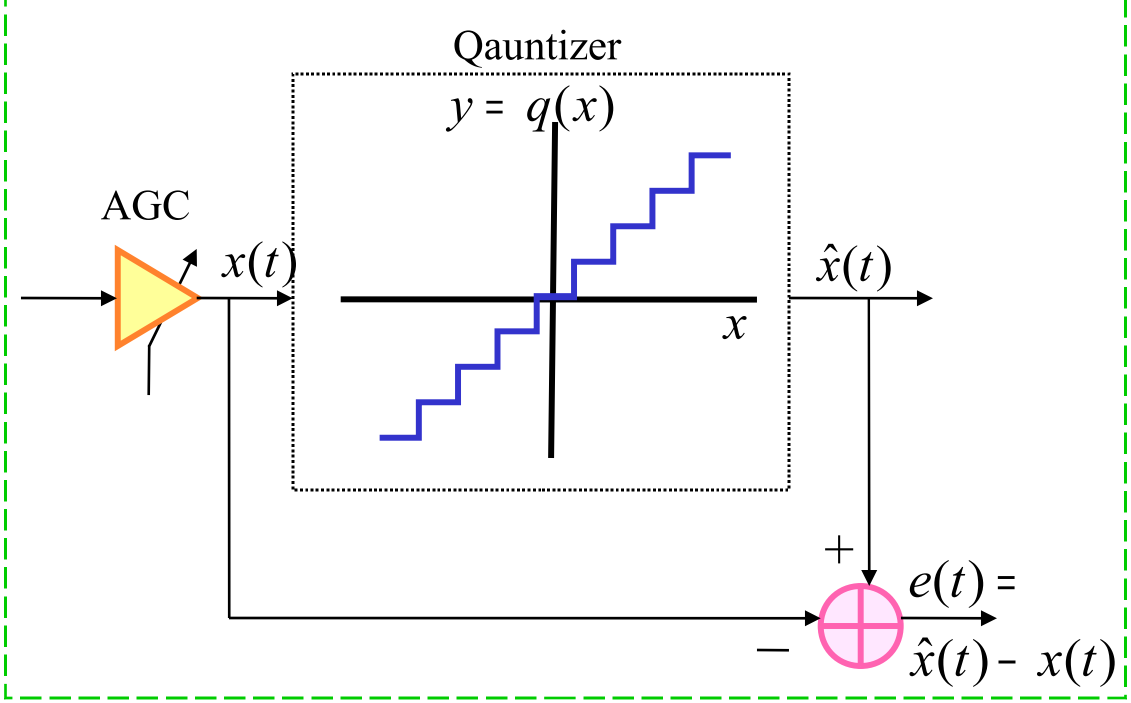


Quantization error

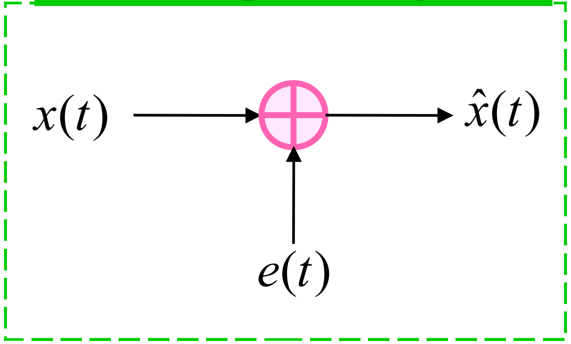
- Quantizing error: The difference between the input and output of a quantizer

$$\Rightarrow e(t) = \hat{x}(t) - x(t)$$

Process of quantizing noise



Model of quantizing noise



Quantization error …

- Quantizing error:
  - **Granular or linear errors** happen for inputs within the dynamic range of quantizer
  - **Saturation errors** happen for inputs outside the dynamic range of quantizer
    - Saturation errors are larger than linear errors
    - Saturation errors can be avoided by proper tuning of AGC
- Quantization noise variance:

$$\sigma_q^2 = \mathbf{E}\{[x - q(x)]^2\} = \int_{-\infty}^{\infty} e^2(x)p(x)dx = \sigma_{\text{Lin}}^2 + \sigma_{\text{Sat}}^2$$

$$\sigma_{\text{Lin}}^2 = 2\sum_{l=0}^{L/2-1} \frac{q_l^2}{12} p(x_l) q_l \quad \xrightarrow{\text{Uniform q.}} \quad \sigma_{\text{Lin}}^2 = \frac{q^2}{12}$$

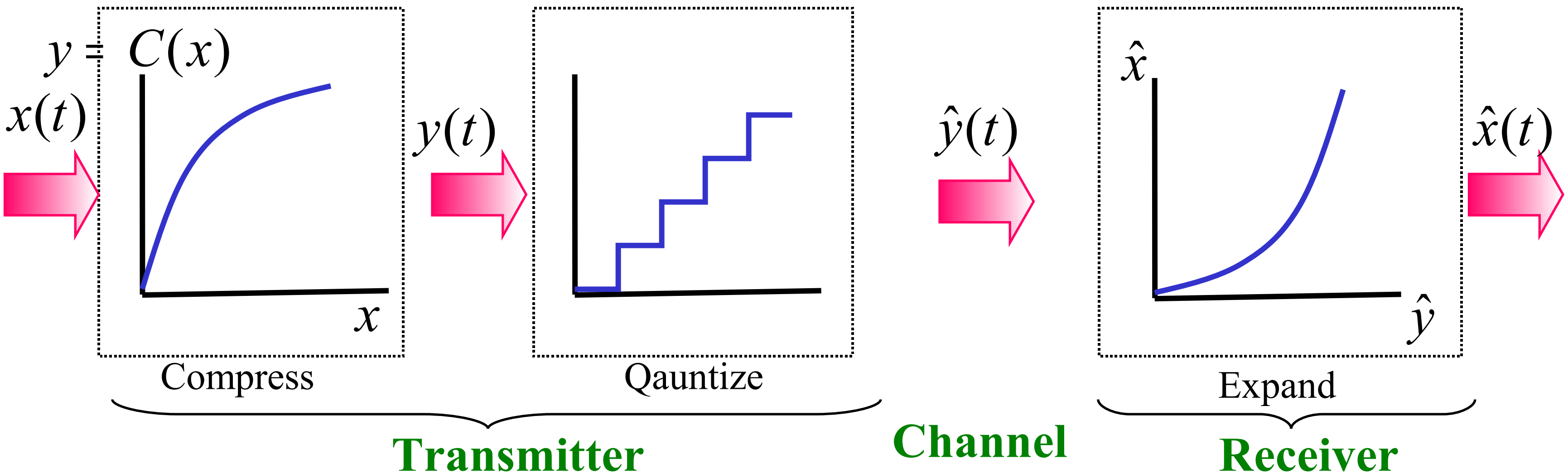
Uniform and non-uniform quant.

- Uniform (linear) quantizing:
  - No assumption about amplitude statistics and correlation properties of the input.
  - Not using the user-related specifications
  - Robust to small changes in input statistic by not finely tuned to a specific set of input parameters
  - Simple implementation
- Application of linear quantizer:
  - Signal processing, graphic and display applications, process control applications
- Non-uniform quantizing:
  - Using the input statistics to tune quantizer parameters
  - Larger SNR than uniform quantizing with same number of levels
  - Non-uniform intervals in the dynamic range with same quantization noise variance
- Application of non-uniform quantizer:
  - Commonly used for speech

Non-uniform quantization

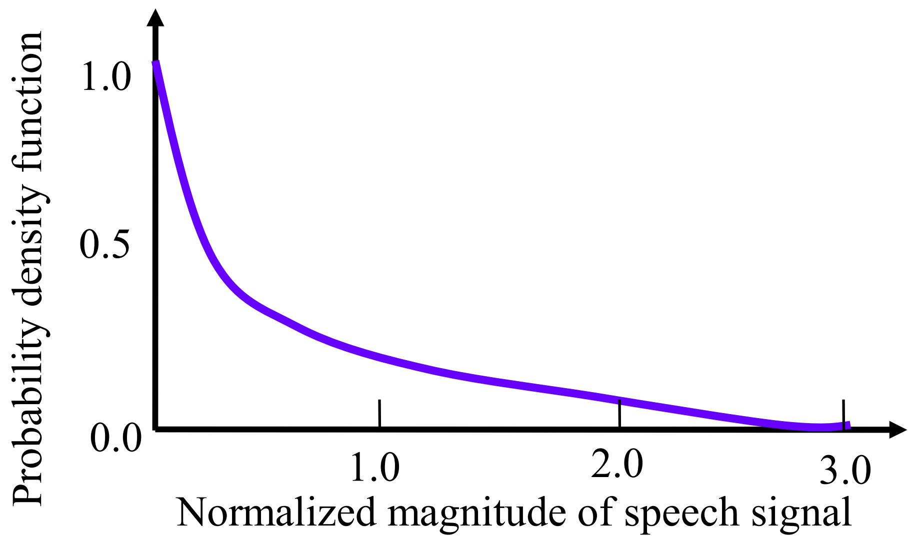
- It is achieved by uniformly quantizing the "compressed" signal.
- At the receiver, an inverse compression characteristic, called "expansion" is employed to avoid signal distortion.

compression+expansion ⇨ companding



Statistics of speech amplitudes

- In speech, weak signals are more frequent than strong ones.



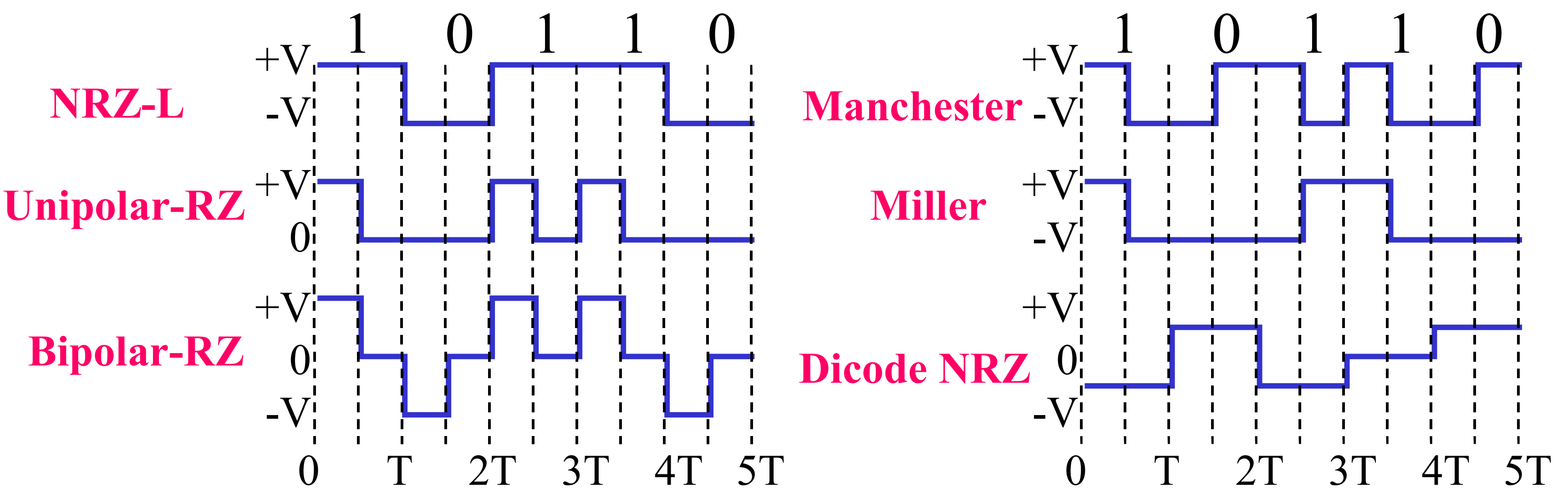
- Using equal step sizes (uniform quantizer) gives low $\left(\frac{S}{N}\right)_q$ for weak signals and high $\left(\frac{S}{N}\right)_q$ for strong signals.
  - Adjusting the step size of the quantizer by taking into account the speech statistics improves the SNR for the input range.

Baseband transmission

- To transmit information through physical channels, PCM sequences (codewords) are transformed to pulses (waveforms).
  - Each waveform carries a symbol from a set of size M.
  - Each transmit symbol represents $k = \log_2 M$ bits of the PCM words.
  - PCM waveforms (line codes) are used for binary symbols (M=2).
  - M-ary pulse modulation are used for non-binary symbols (M>2).

PCM waveforms

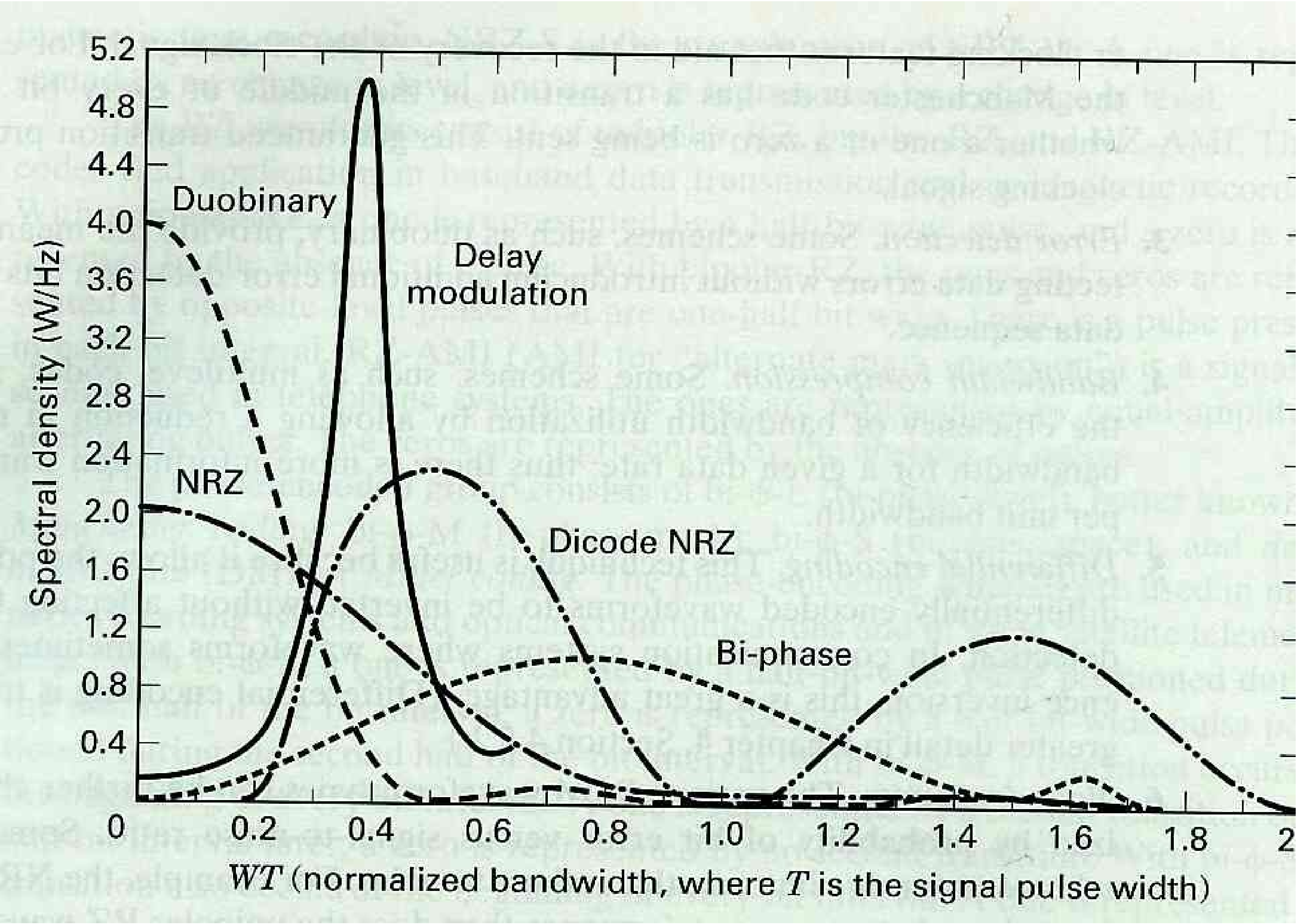
- PCM waveforms category:
  - Nonreturn-to-zero (NRZ)
  - Return-to-zero (RZ)
  - Phase encoded
  - Multilevel binary



PCM waveforms …

- Criteria for comparing and selecting PCM waveforms:
  - Spectral characteristics (power spectral density and bandwidth efficiency)
  - Bit synchronization capability
  - Error detection capability
  - Interference and noise immunity
  - Implementation cost and complexity

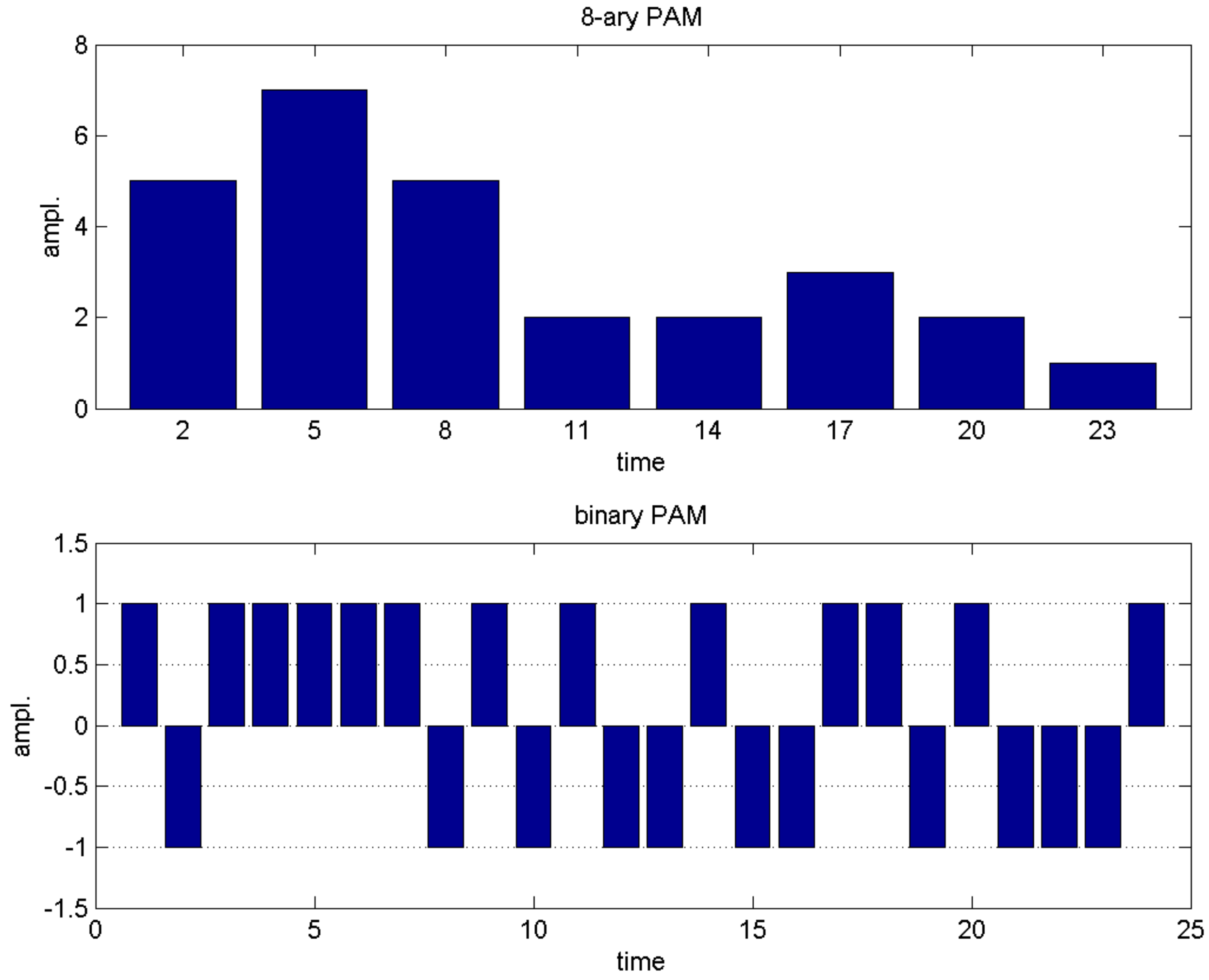
Spectra of PCM waveforms



M-ary pulse modulation

- M-ary pulse modulations category:
  - M-ary pulse-amplitude modulation (PAM)
  - M-ary pulse-position modulation (PPM)
  - M-ary pulse-duration modulation (PDM)
- M-ary PAM is a multi-level signaling where each symbol takes one of the M allowable amplitude levels, each representing $k = \log_2 M$ bits of PCM words.
- For a given data rate, M-ary PAM ($M>2$) requires less bandwidth than binary PCM.
- For a given average pulse power, binary PCM is easier to detect than M-ary PAM ($M>2$).

PAM example


8-ary PAM
ampl.
8
6
4
2
0
2
5
8
11
14
17
20
23
time
binary PAM
ampl.
1.5
1
0.5
0
-0.5
-1
-1.5
0
5
10
15
20
25
time