

EITF25 Internet--Techniques and Applications

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L2 Physical layer (part 1)

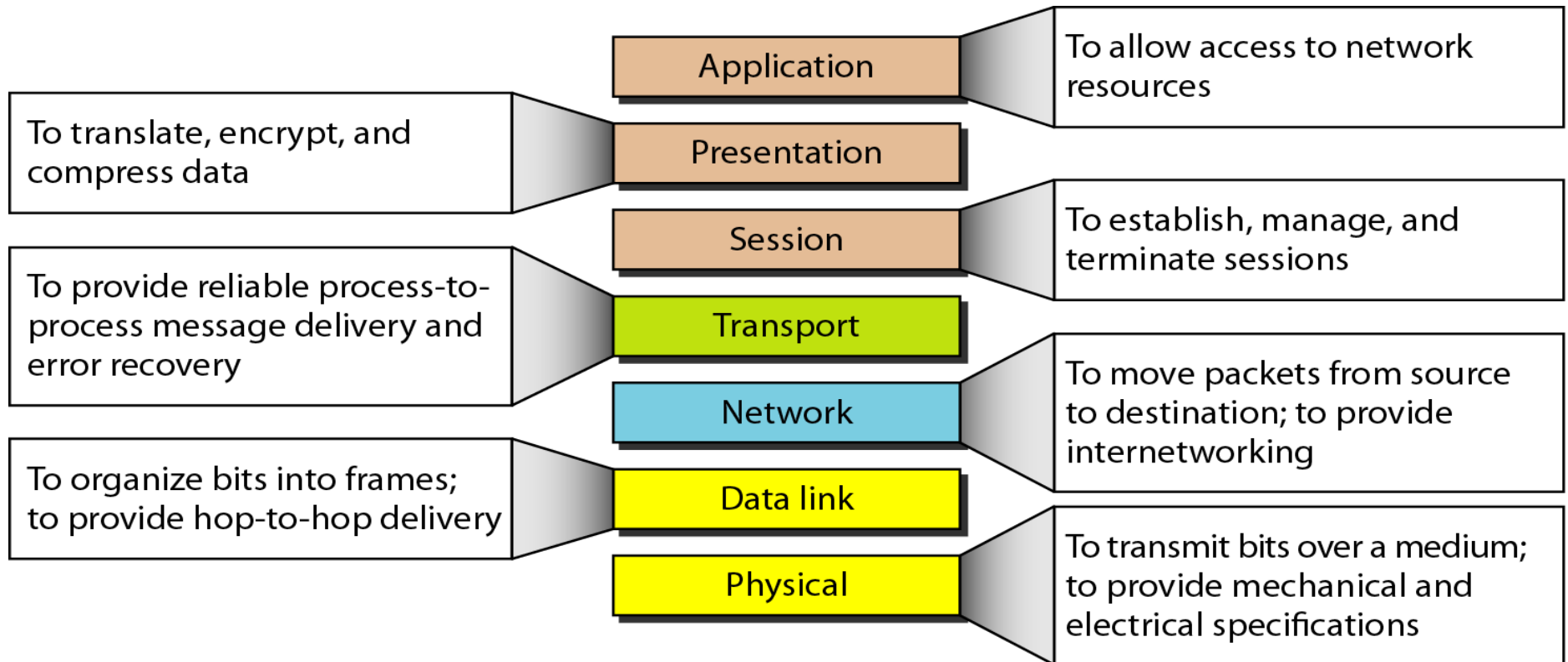


LUND
UNIVERSITY

OSI model

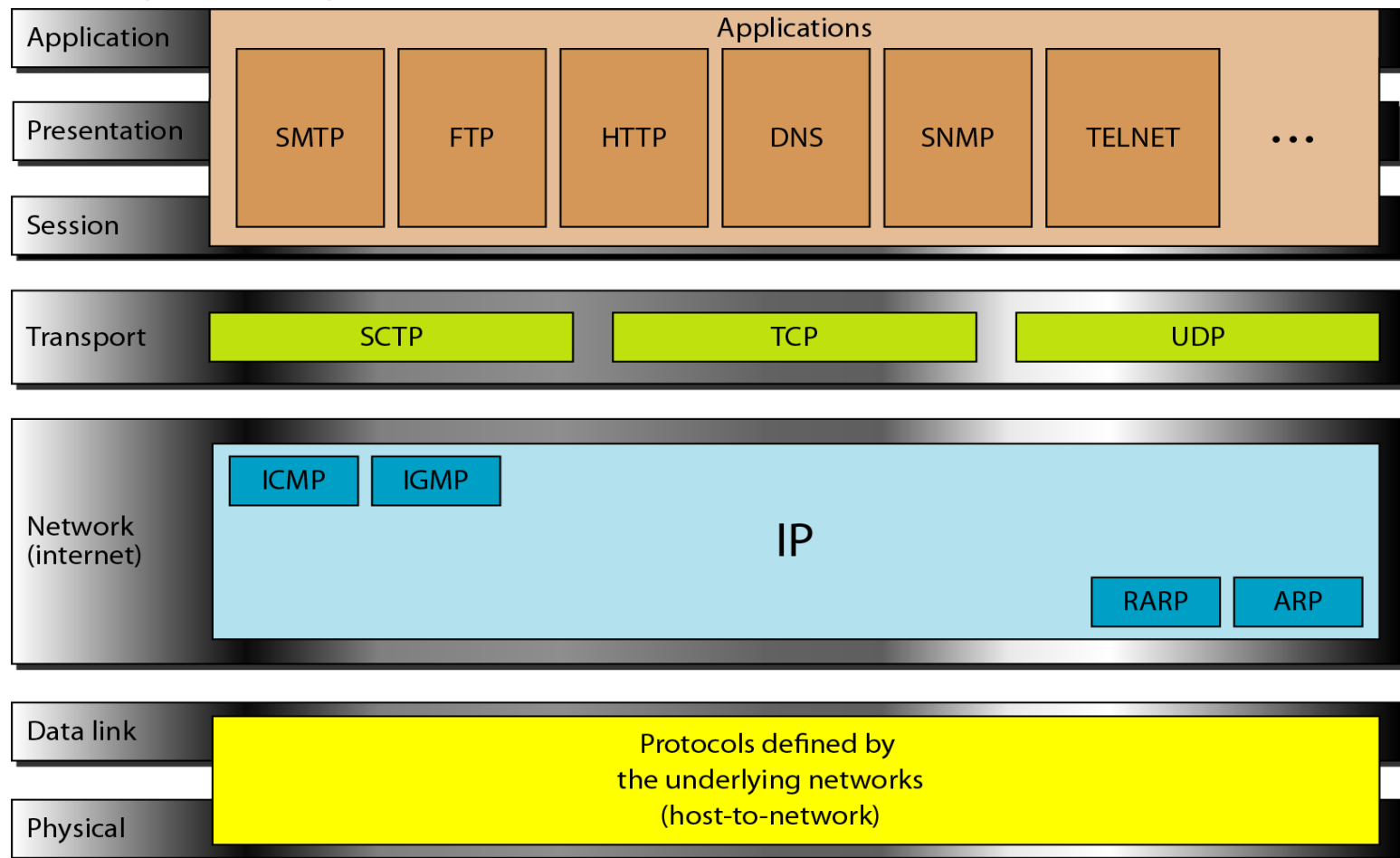
Open Systems Interconnection

- Developed by ISO, 1970~



TCP/IP model

- Developed by DARPA, 1970~

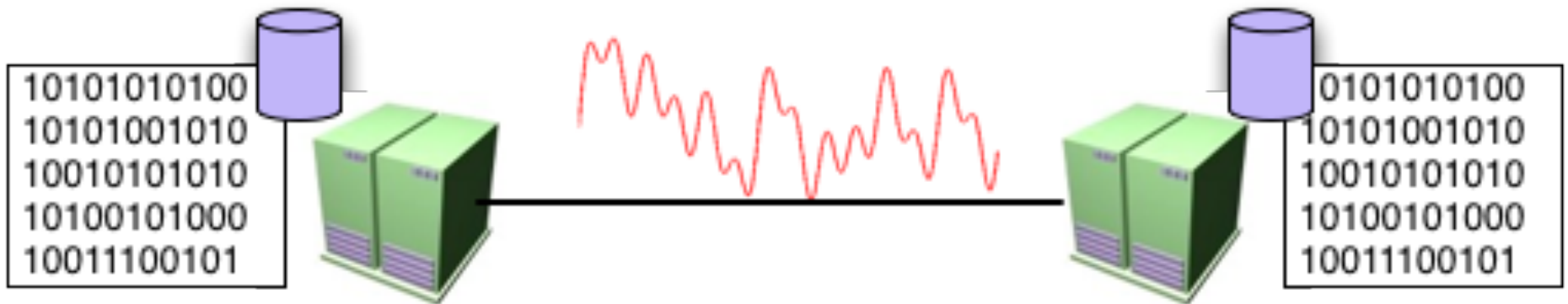


Physical layer

- Analog vs digital signals
 - Sampling, quantisation
- Modulation
 - Represent digital data in a continuous world
- Transmission media
 - Cables and such
- Disturbances
 - Noise and distortion

Data vs Signal

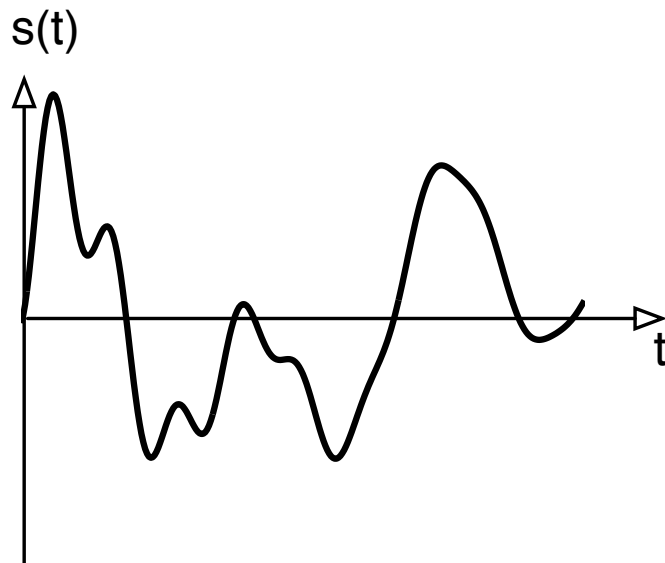
- Data: Static representation of information
 - For storage (often digital)
- Signal: Dynamic representation of information
 - For transmission (often analog)



Analog vs digital

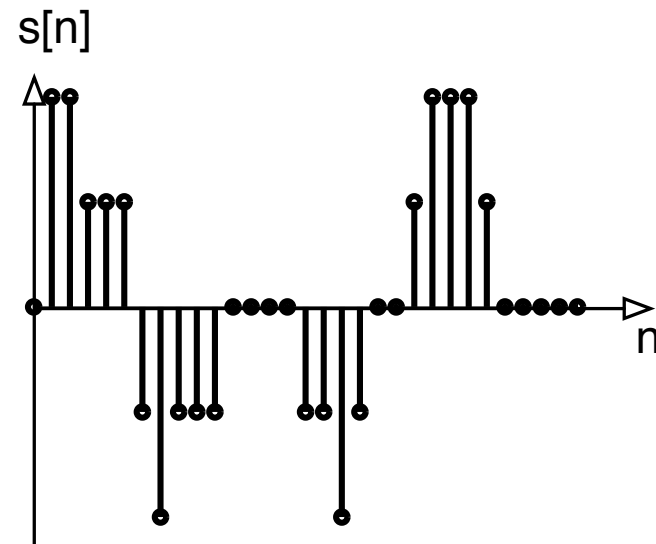
Analog

- Continuous time and amplitude signal
- Electrical/optical domain



Digital

- Discrete time and amplitude
- Binary representation



Digitalization of analog signals

Performed in three steps:

1. Sampling

Discretization in time

2. Quantization

Discretization in amplitude

3. Encoding

Binary representation of amplitude levels

Sampling

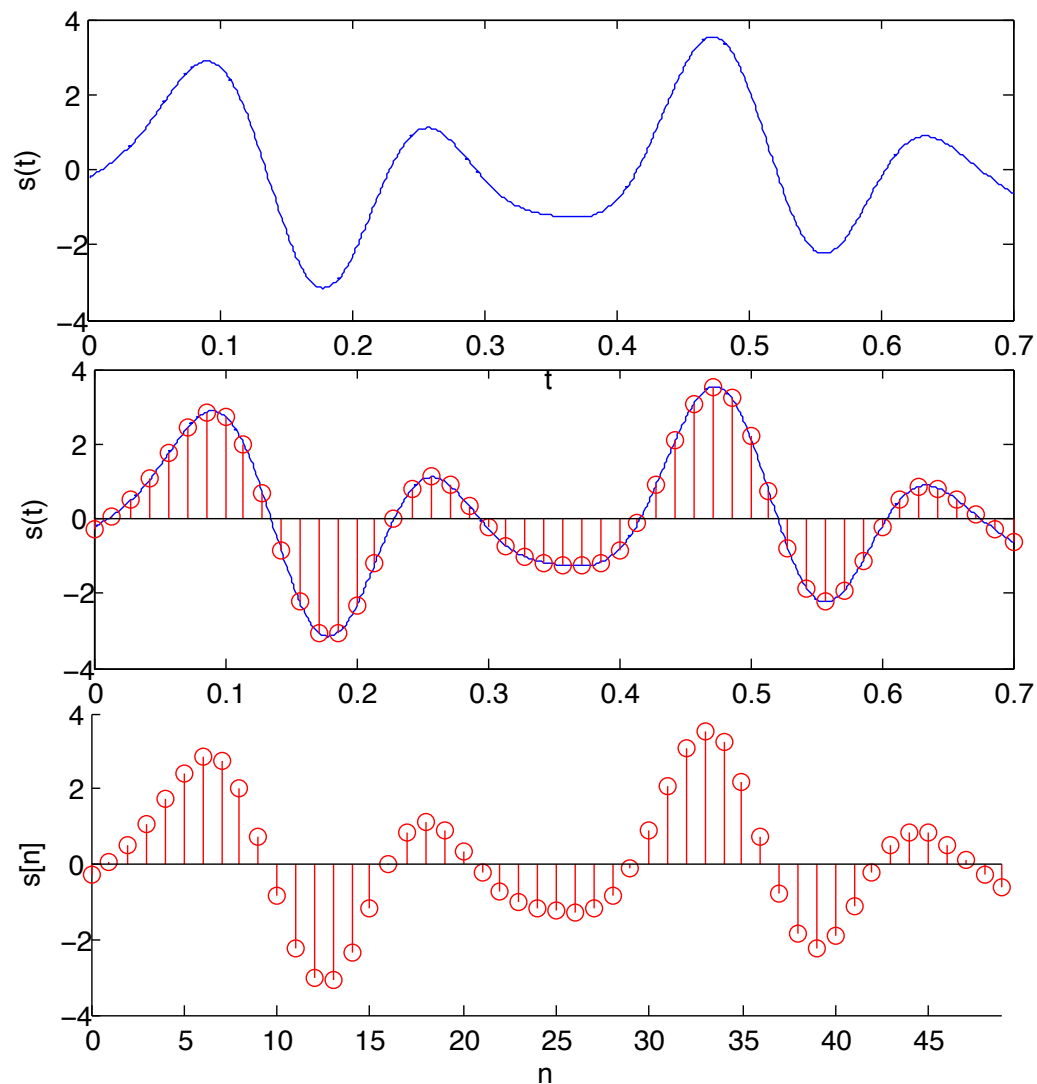
- The process of discretizing time of a continuous signal.

$$s[n] = s(nT_s)$$

- Sampling time: T_s
- Sampling frequency:

$$F_s = 1 / T_s$$

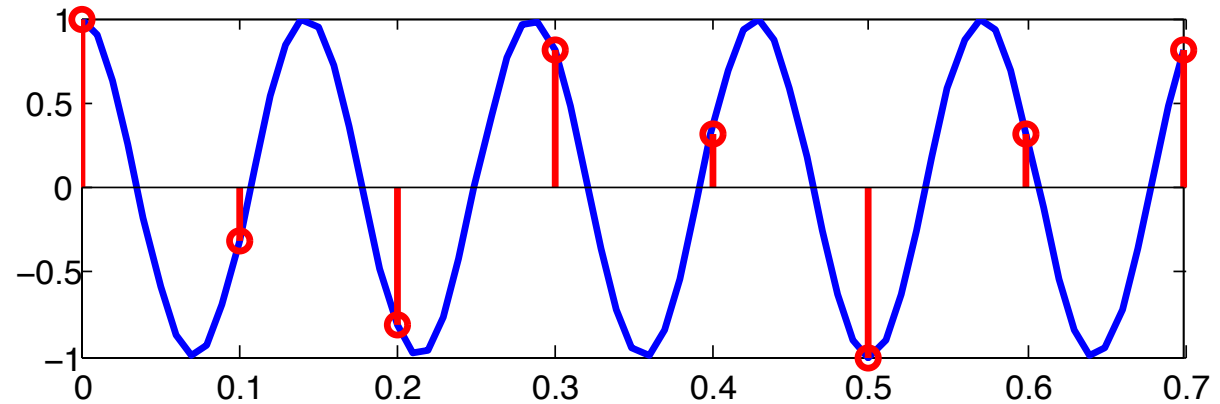
- Loose information about time



Aliasing

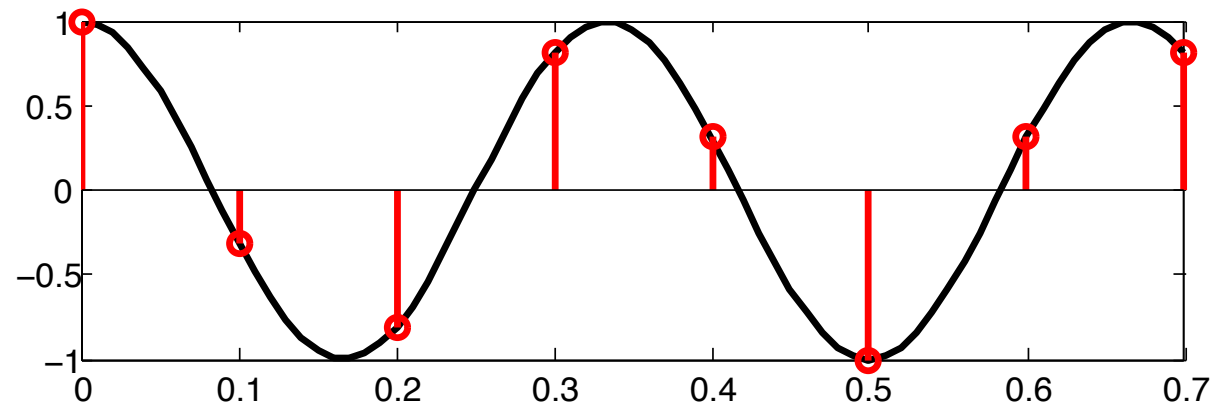
$$y(t) = \cos(14\pi t)$$

$$F_s = 10 \text{ Hz}$$



Reconstruction to lowest possible frequency

$$y(t) = \cos(6\pi t)$$



Shannon-Nyquist Sampling Theorem

If $s(t)$ is a band limited signal with highest frequency component F_{max} , then $s(t)$ is uniquely determined by the samples $s[n] = s(nT)$ if and only if

$$F_s = \frac{1}{T} \geq 2F_{max}$$

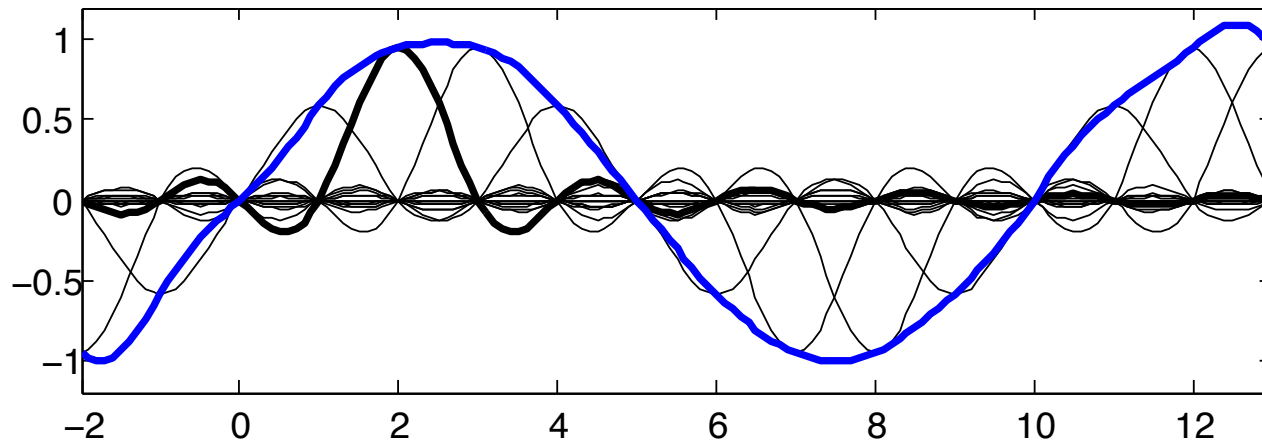
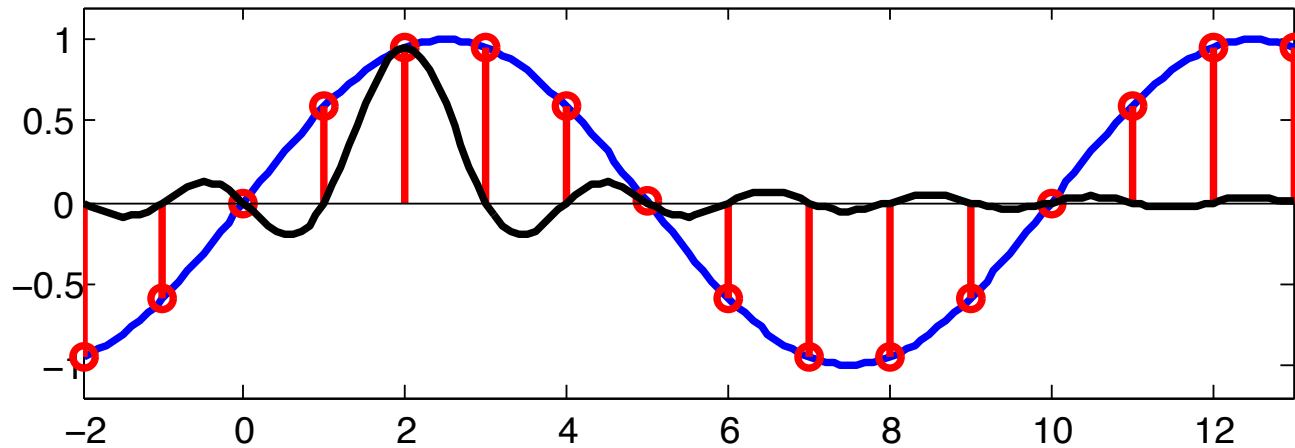
The signal can be reconstructed with

$$s(t) = \sum_{n=-\infty}^{\infty} s[n] \operatorname{sinc}\left(\frac{t - nT_s}{T_s}\right)$$

$F_s/2$ is the Nyquist frequency and $2F_{max}$ the Nyquist rate

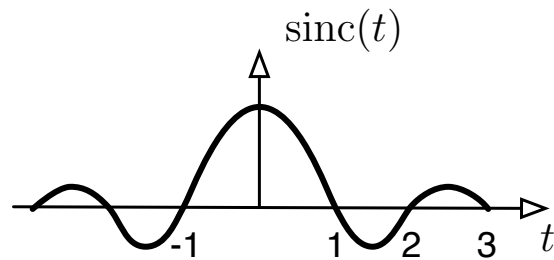
Reconstruction Example

$$y(t) = \sin(2/7\pi t), \quad F_s = 1\text{Hz}$$

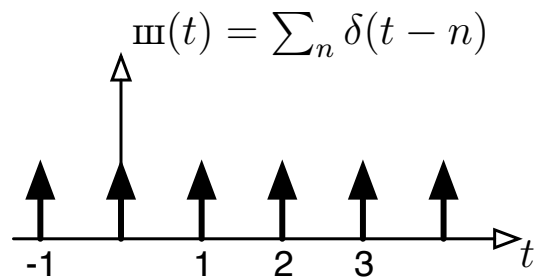
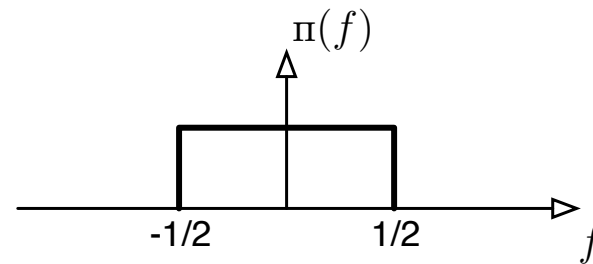


Sampling theorem proof

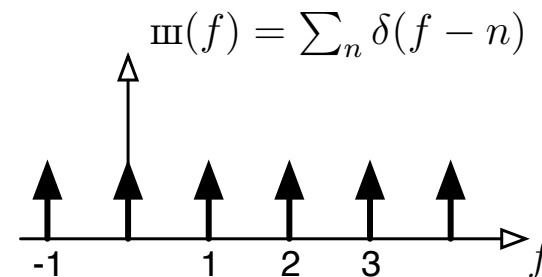
Two important transforms



\mathcal{F}

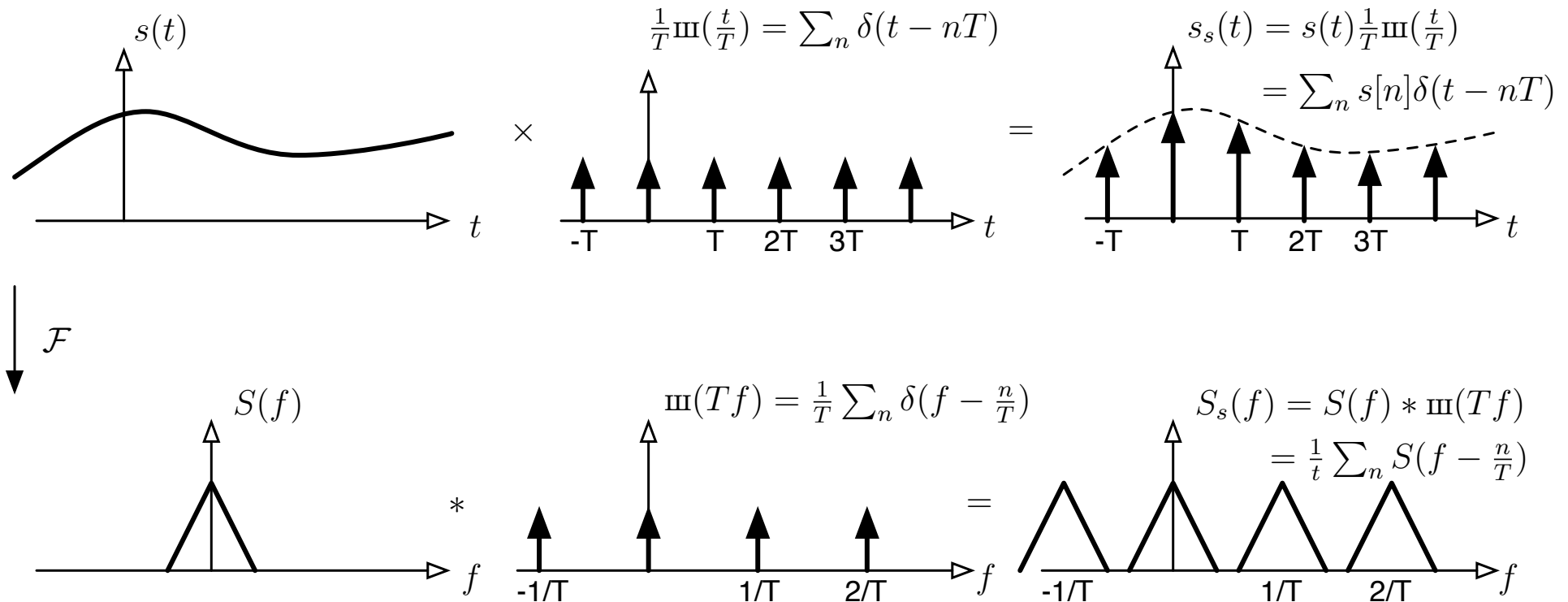


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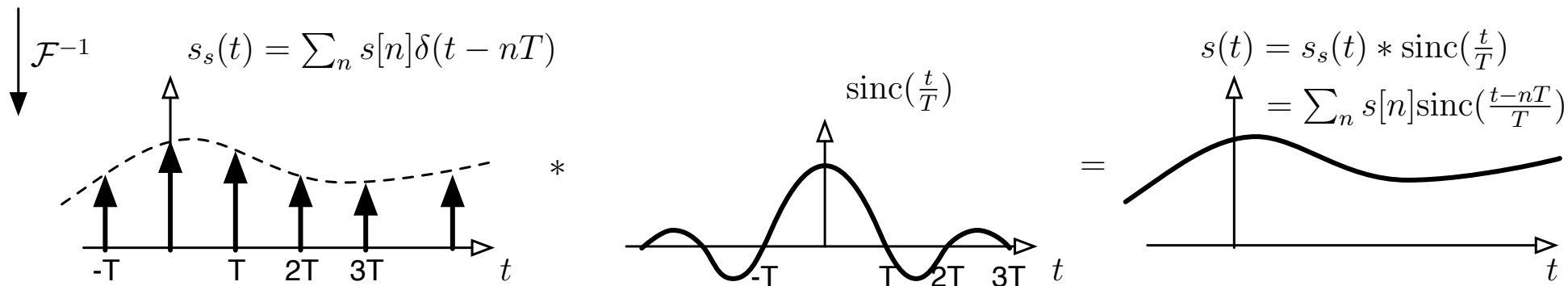
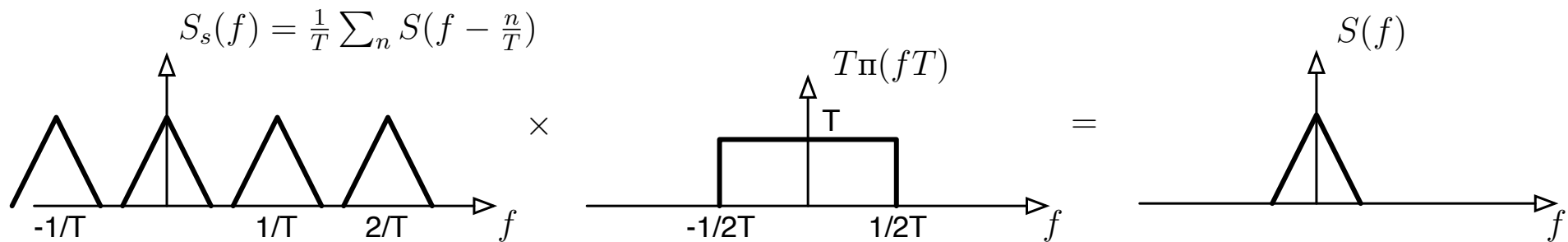
Sampling theorem proof

Mathematical description of sampling



Sampling theorem proof

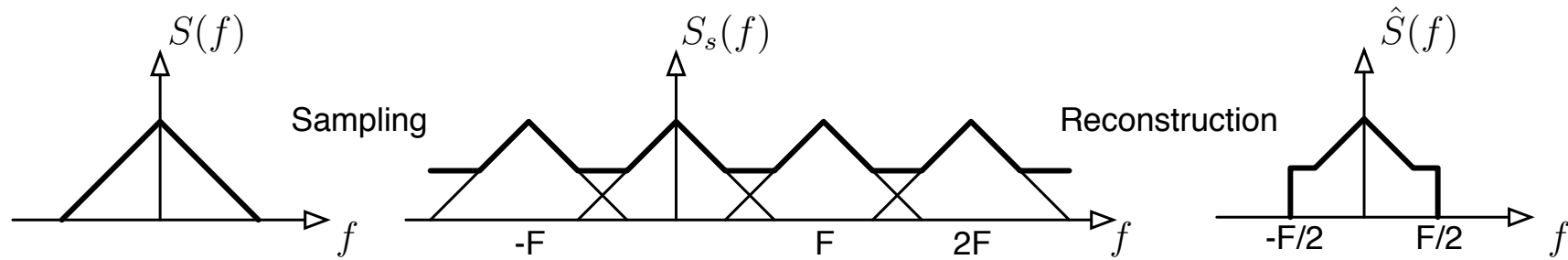
Reconstruction



Sampling theorem

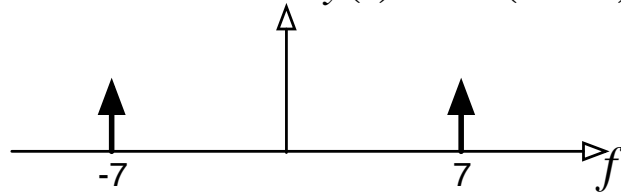
Aliasing

Let $F_s < 2F_{\max}$

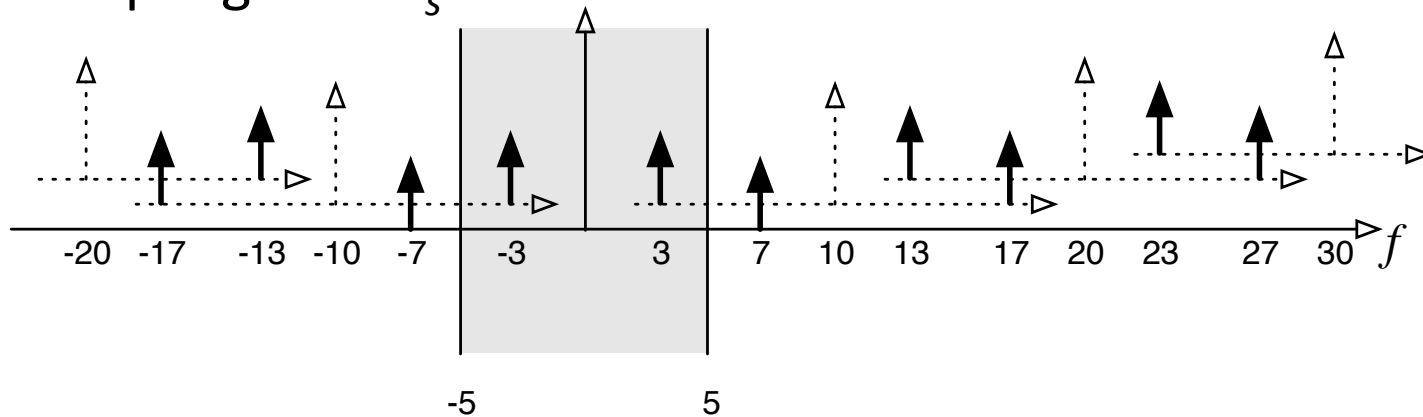


Example

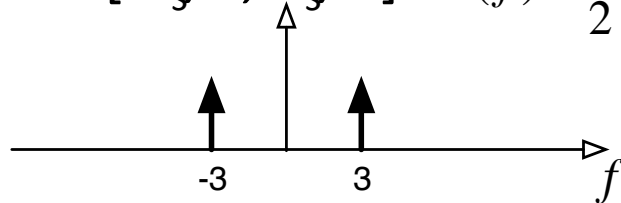
$$y(t) = \cos(2\pi 7t) \rightarrow Y(f) = \frac{1}{2}(\delta(f + 7) + \delta(f - 7))$$



Sampling with $F_s = 10$ Hz



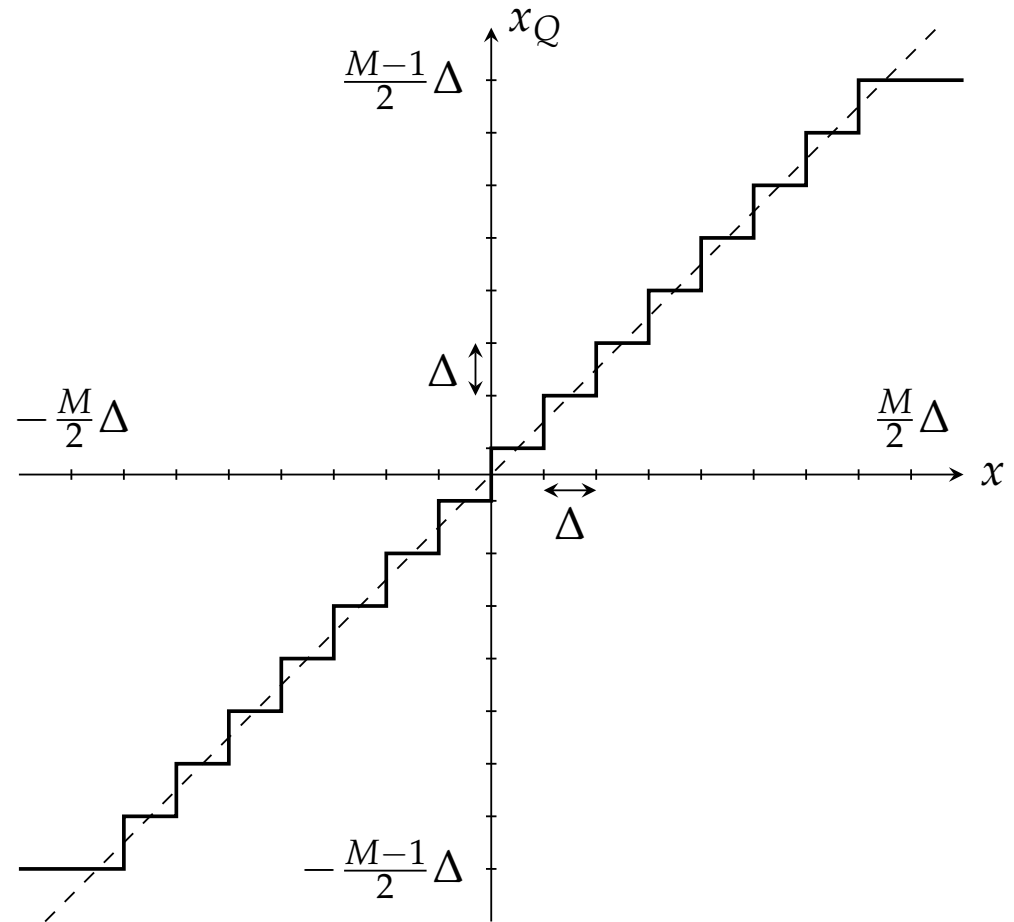
Reconstruct in $[-F_s/2, F_s/2]$: $\hat{Y}(f) = \frac{1}{2}(\delta(f + 3) + \delta(f - 3)) \rightarrow \hat{y}(t) = \cos(2\pi 3t)$



Quantization

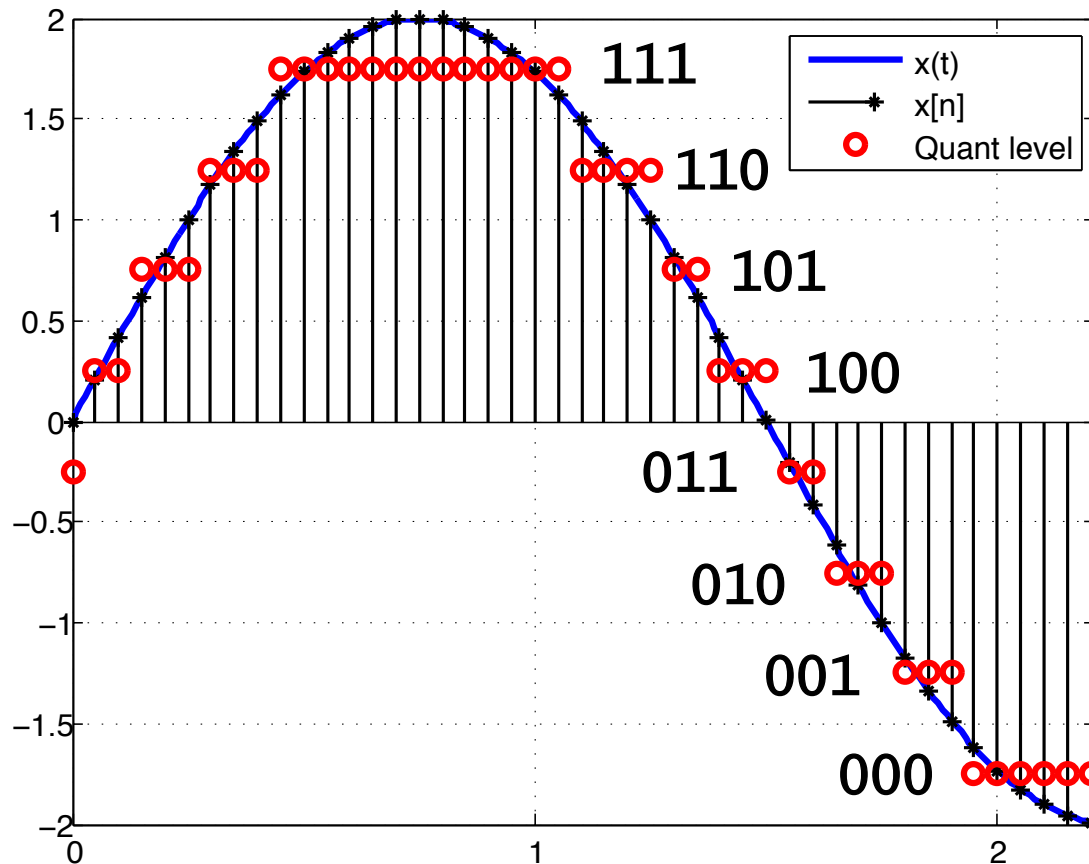
Linear Quantization for k bits

- $M=2^k$ equidistant levels
- Represent sample with k bits



Encoding

Representation of quantized samples in bits



$x=0111001001011011011111111111...$

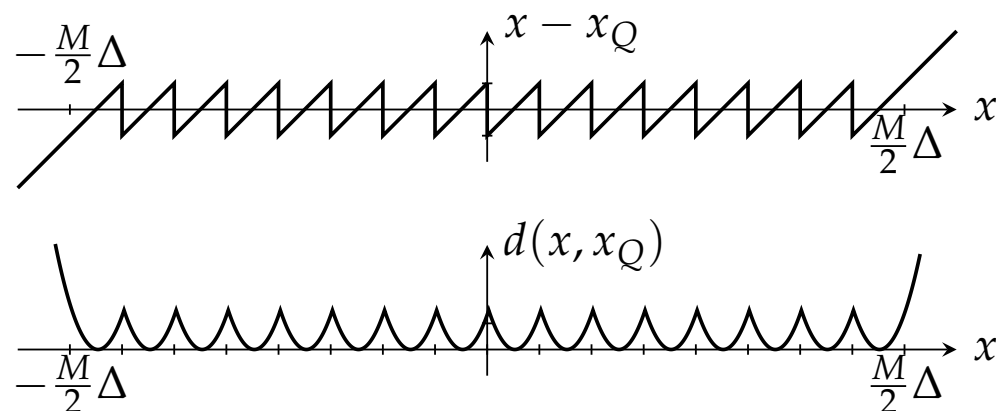
Quantisation distortion

Distortion:

$$d(x, x_Q) = (x - x_Q)^2$$

Average distortion for uniform input:

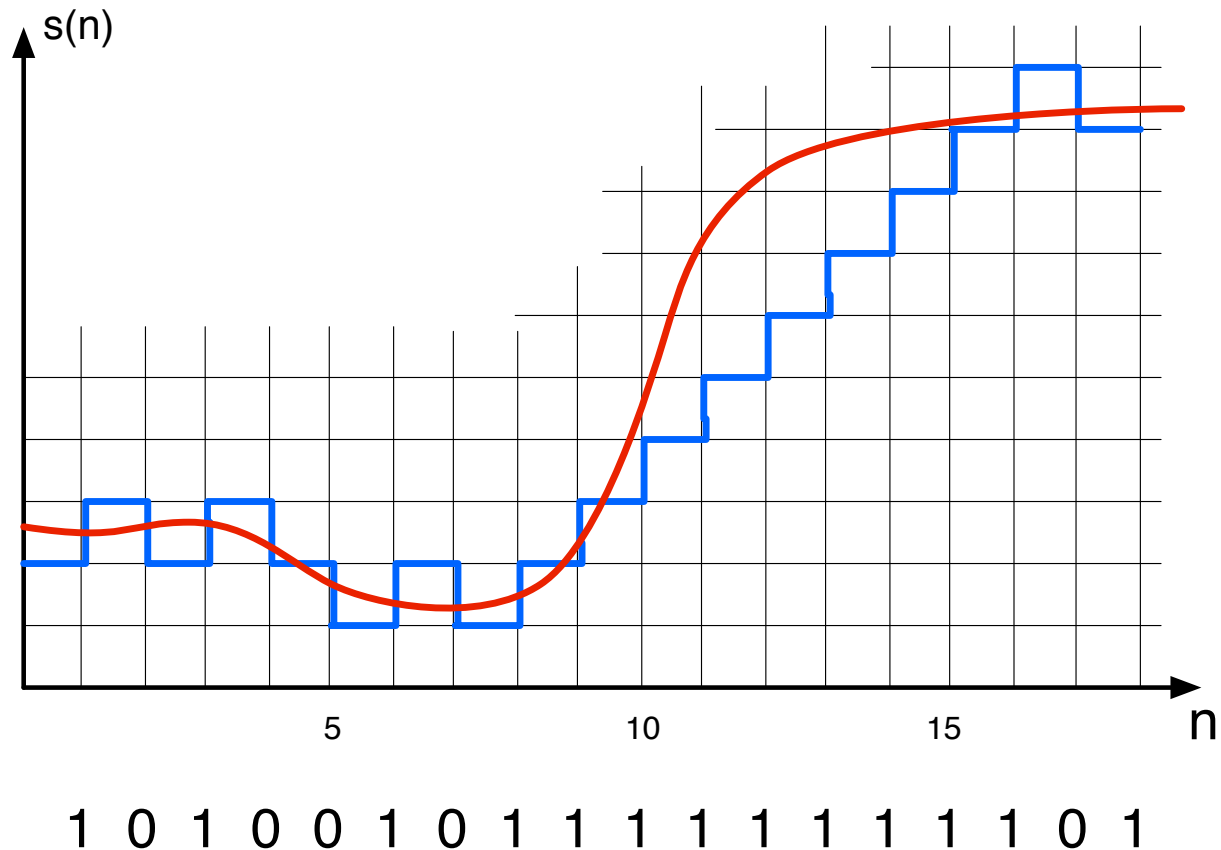
$$E\left[(X - X_Q)^2\right] = \int_{-\Delta/2}^{\Delta/2} x^2 \frac{1}{\Delta} dx = \frac{\Delta^2}{12}$$



Quantization

Delta modulation

- Represent change in amplitude with 1 bit
 - 1: +1
 - 0: -1
- Must be faster sampling



Examples

Telephony

$$F_{\max} = 4 \text{ kHz}$$

$$F_s = 8 \text{ kHz (samples per sec)}$$

$$8 \text{ bit/sample} \Rightarrow 64 \text{ kb/s}$$

CD

$$F_{\max} = 20 \text{ kHz}$$

$$F_s = 44.1 \text{ kHz (samples per sec)}$$

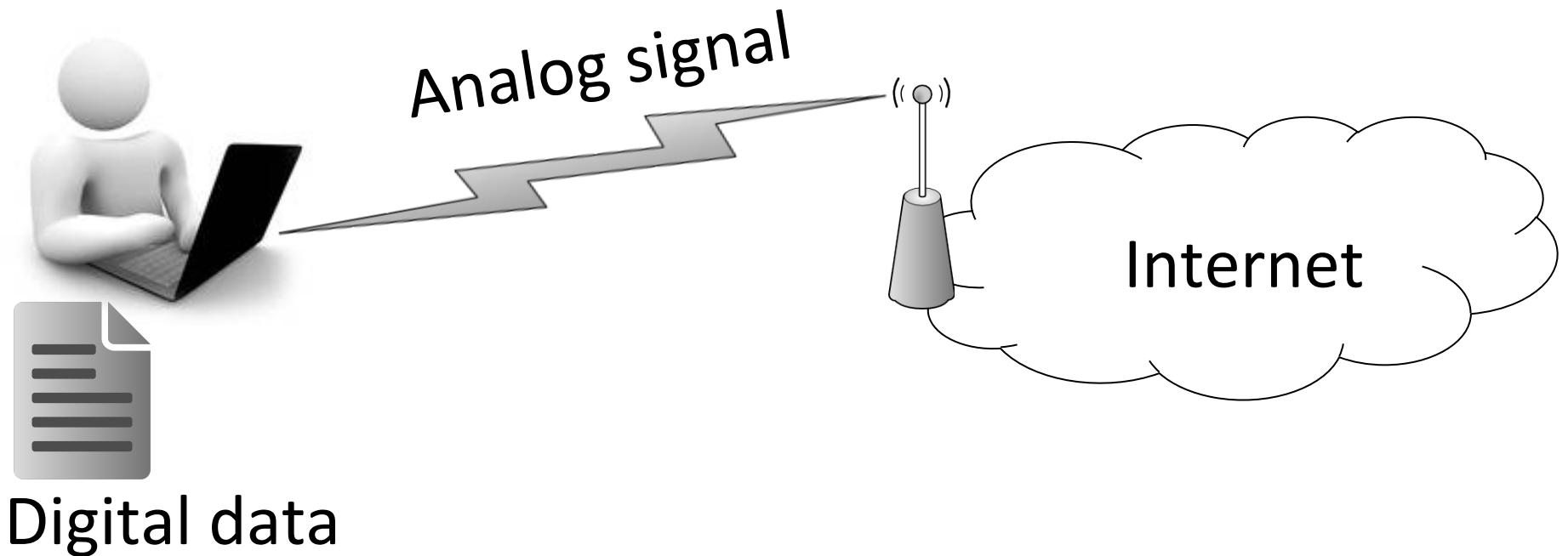
$$16 \text{ bit/sample} \Rightarrow 705.6 \text{ kb/s}$$

2 channels (stereo)

$$\Rightarrow 1.4 \text{ Mb/s}$$

From bits to signals

Principles of digital communications

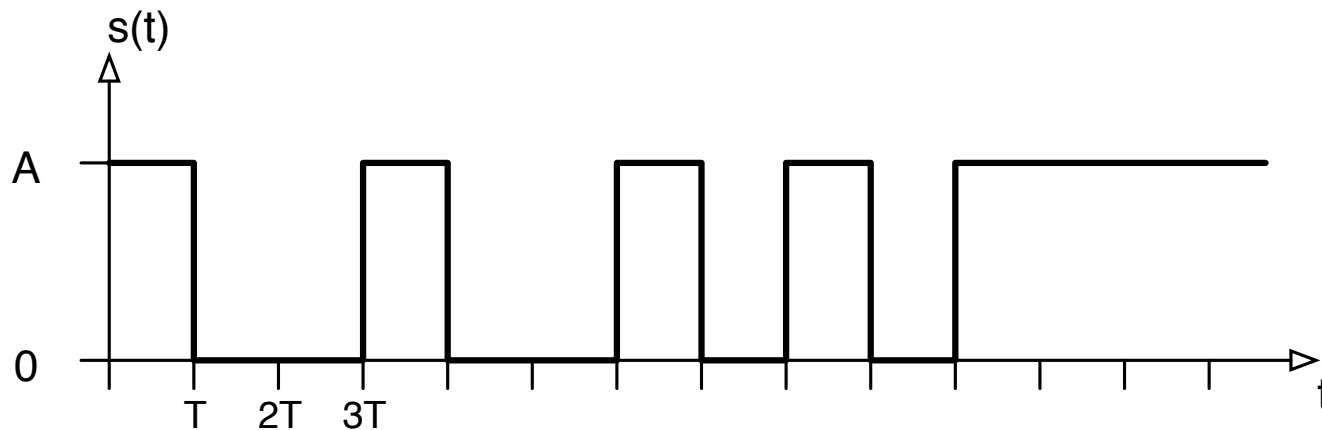


On-off keying

- Send one bit during T_b seconds and use two signal levels, “on” and “off”, for 1 and 0.

Ex.

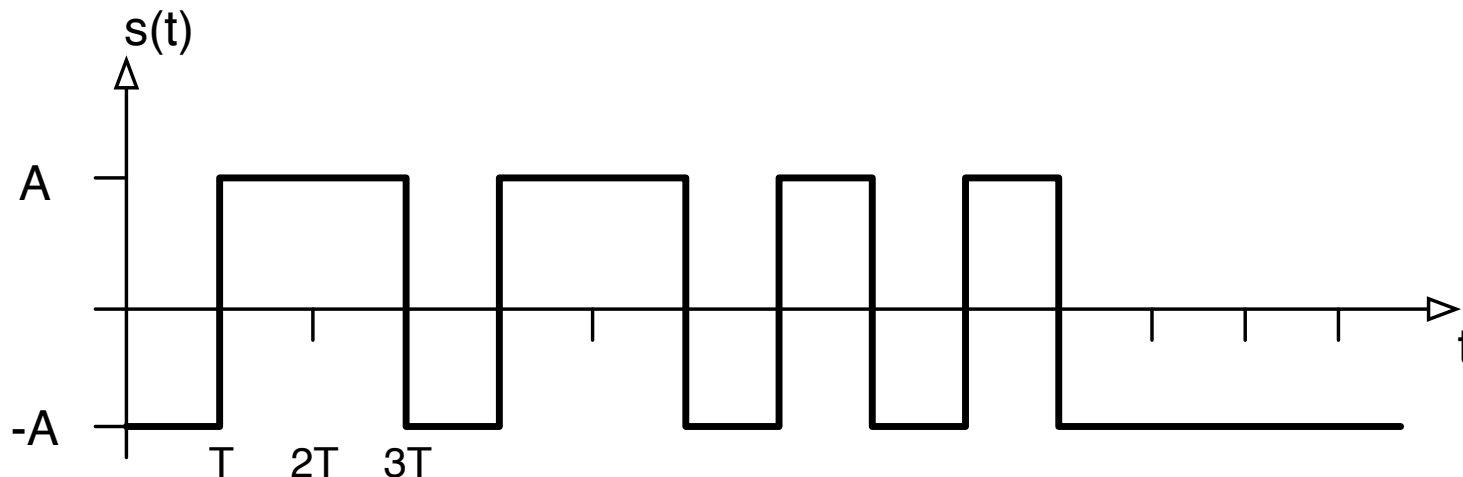
$x=10010010101111100$



Non-return to zero (NRZ)

- Send one bit during T_b seconds and use two signal levels, $+A$ and $-A$, for 0 and 1.

Ex. $x=10010010101111100$



Mathematical description

With $g(t)=A$, $0 < t < T$, the signals can be described as

$$s(t) = \sum_n a_n g(t - nT)$$

- On-off

$$a_n = x_n$$

- NRZ

$$a_n = (-1)^{x_n}$$

Two signal alternatives

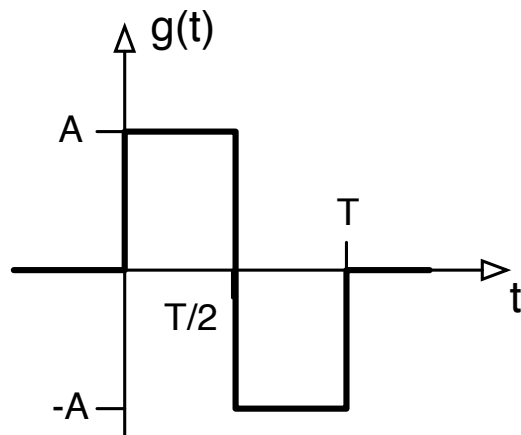
- $s_0(t)=0$ and $s_1(t)=g(t)$

- $s_0(t)=g(t)$ and $s_1(t)=-g(t)$

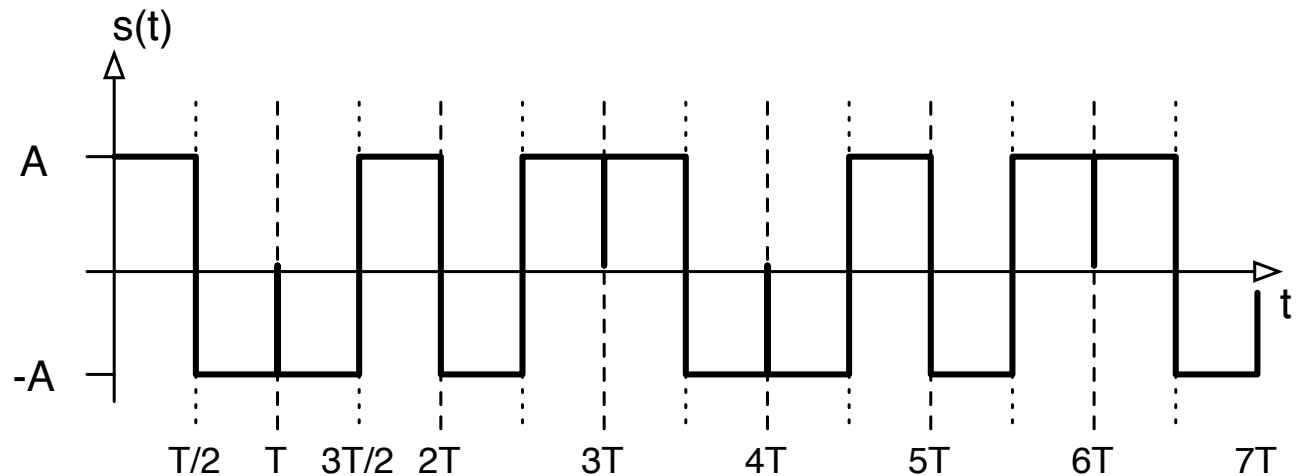
Manchester coding

- To get a zero passing in each signal time, split the pulse shape $g(t)$ in two parts and use +/- as amplitude.

Ex.



$x=10010010101111100$



Multi level modulation

- To transmit k bits in one signal alternative of duration T_s , use $M=2^k$ levels.

Ex. Two bits per signal

$x=10\ 01\ 00\ 10\ 10\ 11\ 11\ 10\ 00$

