EE1 and ISE1 Communications I

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Lecture sixteen

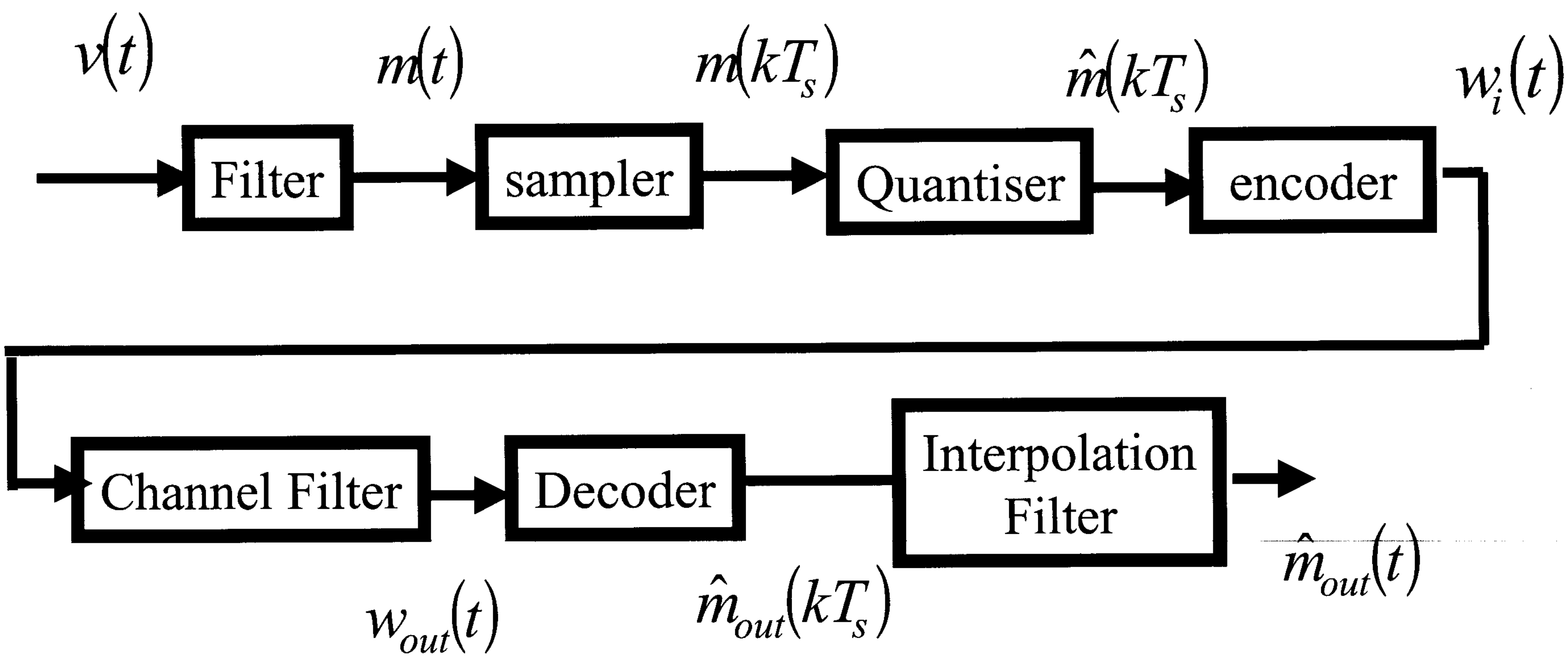
Lecture Aims

- Outline digital communication systems

Why digital modulation

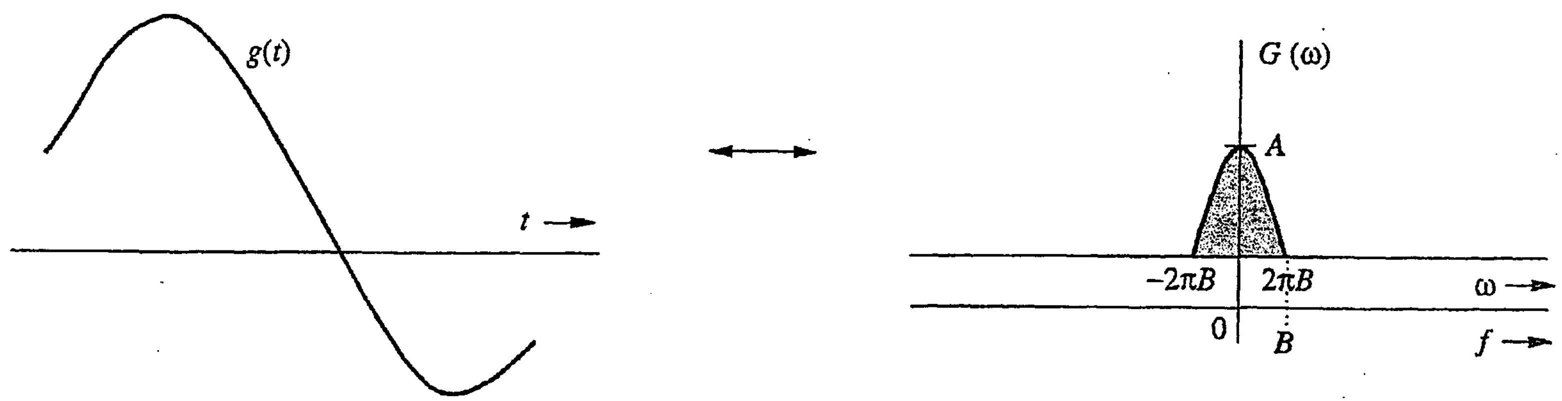
- More resilient to noise
- Viability of regenerative repeaters
- Digital hardware more flexible
- It is easier to multiplex digital signals

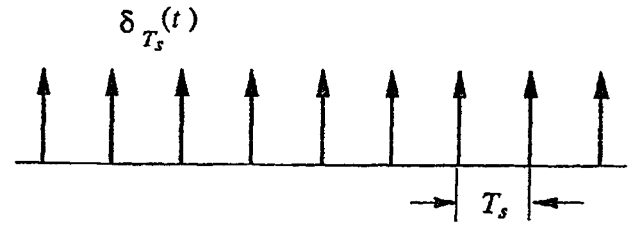
Digital Transmission System



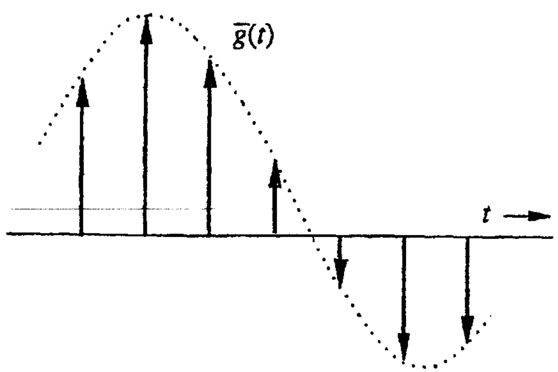
Analogue waveform

- **Analogue waveform and its spectrum**

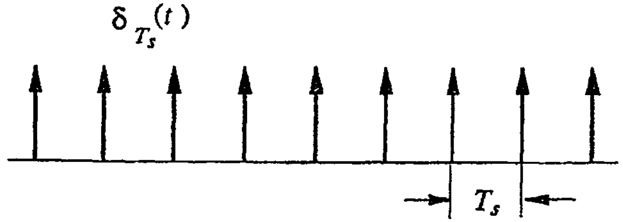




Sampling waveform

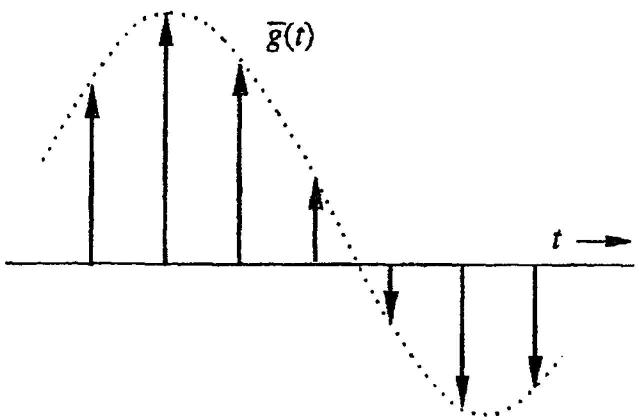


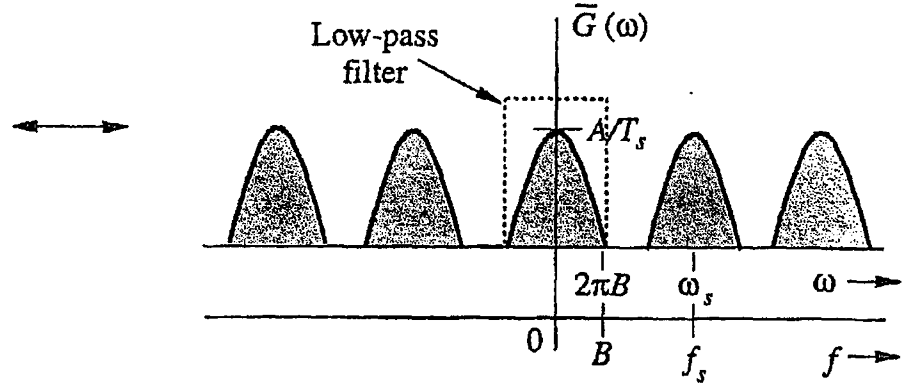
Sampled waveform



$$\omega_s = \frac{2\pi}{T_s} = 2\pi f_s$$

$$\delta_{T_s}(t) = \frac{1}{T_s}[1 + 2\cos\,\omega_s t + 2\cos\,2\omega_s t + 2\cos\,3\omega_s t + \cdots]$$

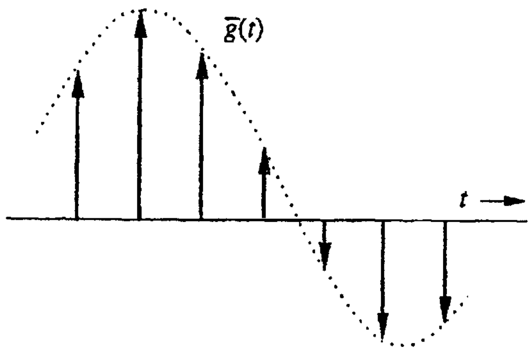


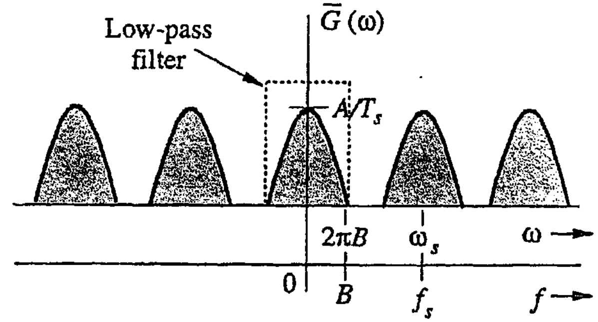


$$\overline{g}(t) = g(t)\delta_{T_s}(t)$$

$$= \frac{1}{T_s}[g(t) + 2g(t)\cos\,\omega_s t + 2g(t)\cos\,2\omega_s t + 2g(t)\cos\,3\omega_s t + \cdots]$$

Sampled signal spectrum





Samplig frequency

$$f_s = 1/T_s \text{ Hz}$$

Samplig time

$$T_s = 1/f_s$$

Also have

Sampled signal spectrum

$$\overline{G}(\omega) = \frac{1}{T_s} \sum_{n=-\infty}^{\infty} G(\omega - n\omega_s)$$

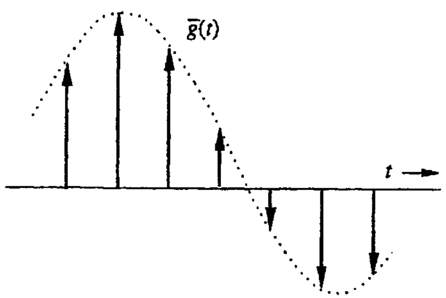
Sampling frequency must satisfy

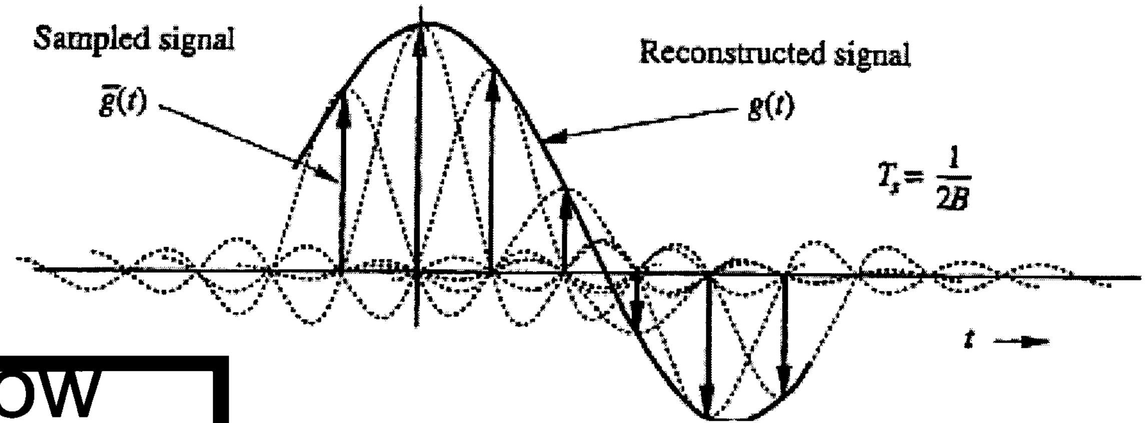
$$f_s > 2B$$

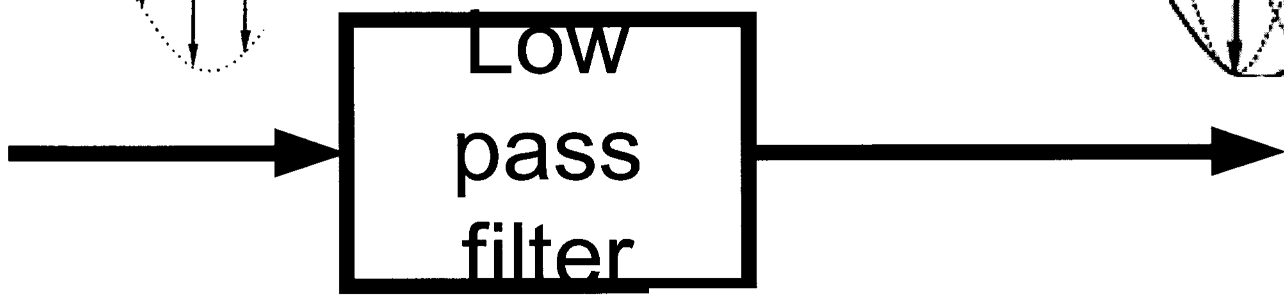
$$T_s < \frac{1}{2B}$$

Signal construction using better filter

$$2BT_s = 1$$





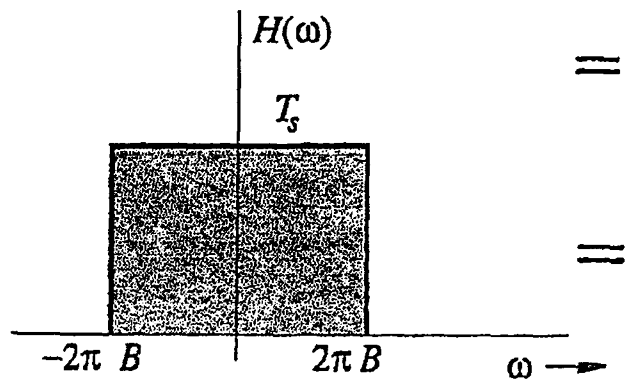


$$h(t) = \text{sinc}(2\pi Bt)$$

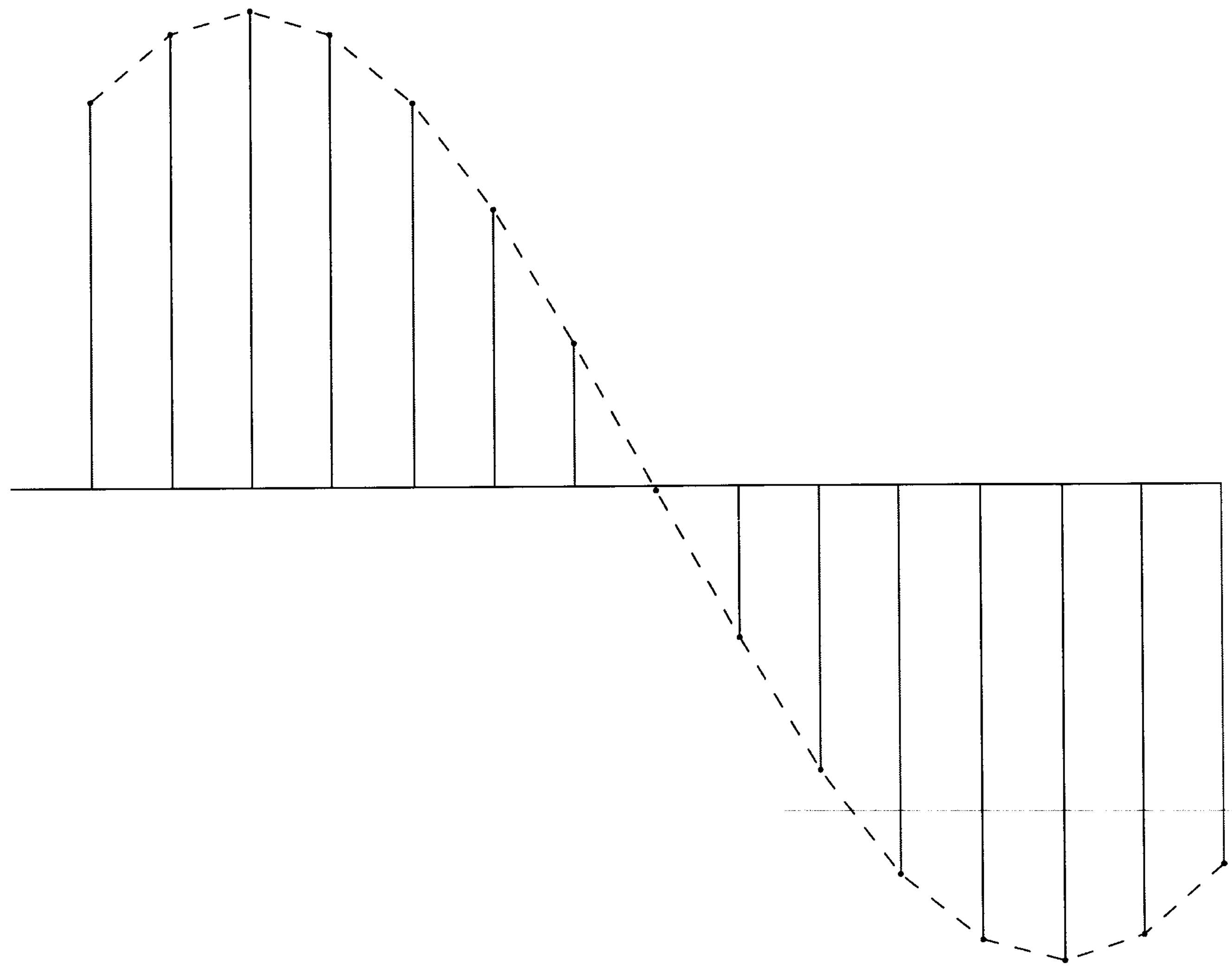
$$g(t) = \sum_k g(kT_s)h(t - kT_s)$$

$$= \sum_k g(kT_s)\,\text{sinc}[2\pi B(t - kT_s)]$$

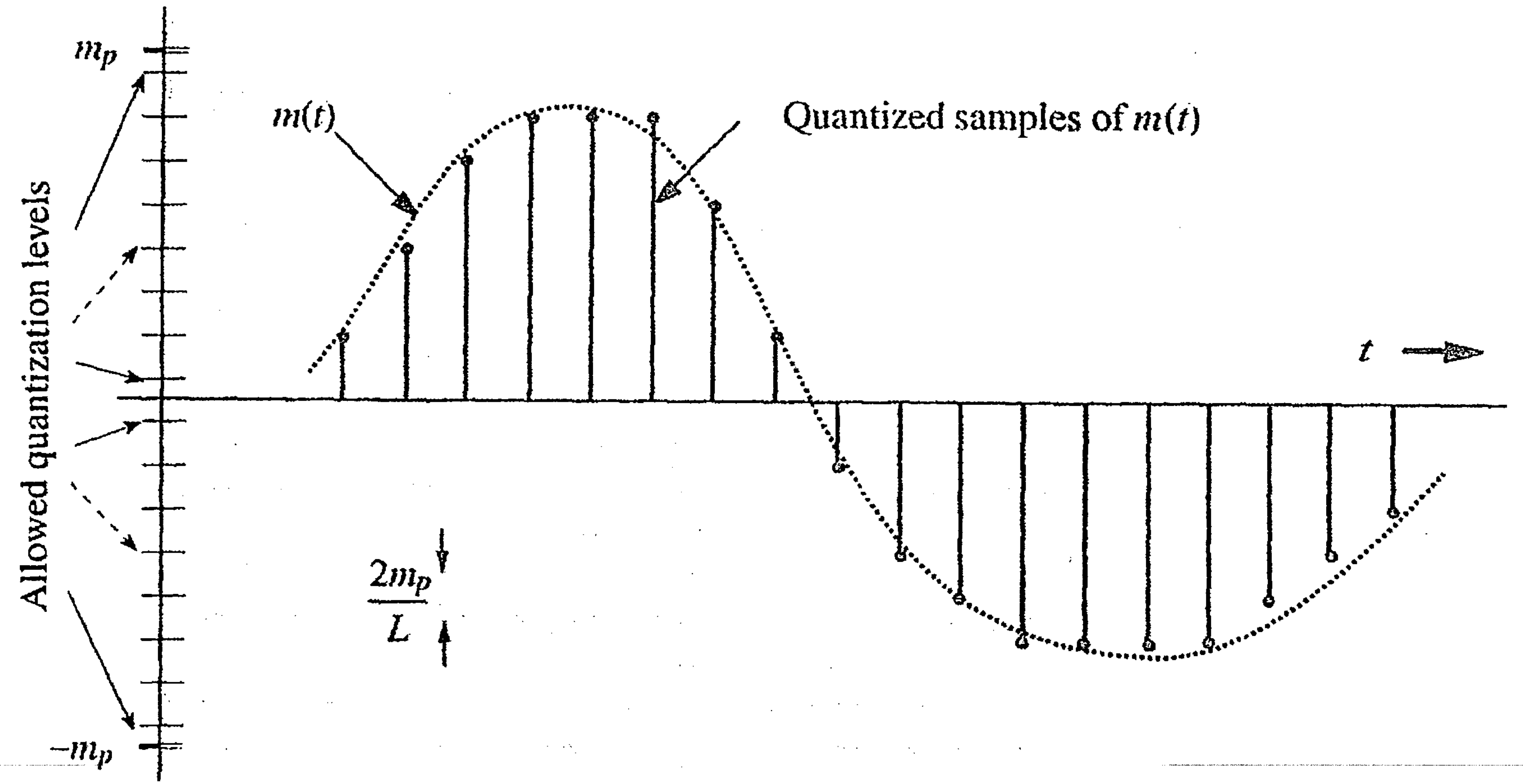
$$= \sum_k g(kT_s)\,\text{sinc}(2\pi Bt - k\pi)$$



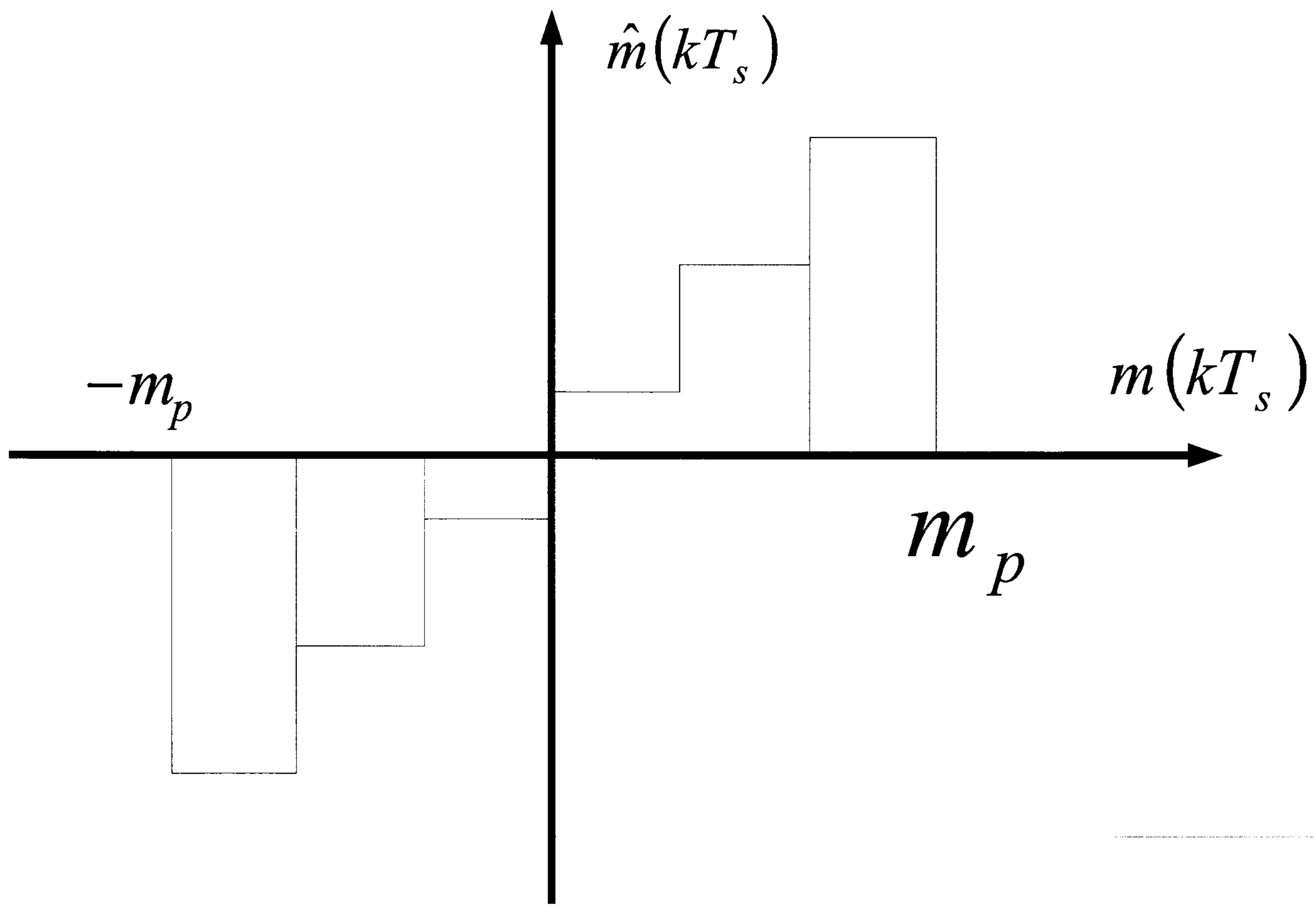
Sampled Waveform



Quantized waveform



Uniform quantizer



- Minimum and maximum voltages

$$\max\left(m(t)\right) = m_p$$

$$\min\left(m(t)\right) = -m_p$$

$$n = \text{number of bits}$$

$$L = 2^{n} = \text{number of levels}$$

$$m_i = \text{voltage boundaries}$$

$$i = 0,1,2 \cdots L$$

$$m_0 = -m_p \qquad\qquad m_L = m_p$$

Voltage range values

$$\Delta = \text{step size} = \frac{\max(m(t)) - \min(m(t))}{\mathrm{L}}$$

$$m_0 = \min(m(t)) = -m_p$$

$$m_i = \min(m(t)) + i\Delta$$

$$m_L = \max(m(t)) = m_p$$

$$\Delta = \frac{m_p - (-m_p)}{L} = \frac{2m_p}{L}$$

$$m_i > m(kT_s) \geq m_{m-1}$$

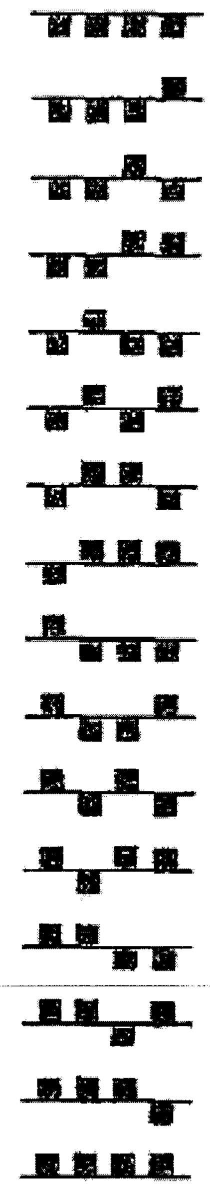
$$\hat{m}(k\ T_s) = \frac{m_i + m_{i-1}}{2}$$

Quantization and binary representation

- Assume the amplitude of the analog signal $m(t)$ lie in the range $(-m_p, m_p)$.
- with quantization, this interval is partitioned into L sub-intervals, each of magnitude $\delta u = 2m_p/L$.
- Each sample amplitude is approximated by the midpoint value of the subinterval in which the sample falls.
- Thus, each sample of the original signal can take on only one of the L different values.
- Such a signal is known as an L-ary digital signals
- In practice, it is better to have binary signals

- **Alternatively we can use A sequence of four binary pulses to get 16 distinct patterns**

Digit	Binary equivalent	Pulse code waveform
0	0000	
1	0001	
2	0010	
3	0011	
4	0100	
5	0101	
6	0110	
7	0111	
8	1000	
9	1001	
10	1010	
11	1011	
12	1100	
13	1101	
14	1110	
15	1111	



Examples

1. Audio Signal (Low Fidelity, used in telephone lines).
   - Frequency from 300Hz to 3400Hz, we assume bandwidth $B = 4$kHz.
   - Sampling frequency $f_s = 2B = 8$kHz that means 8000 samples per second.
   - Each sample is quantized with $L = 256$ levels, that is a group of 8 bits to encode each sample $2^8 = 256$.
   - Thus a telephone line requires $8 \times 8000 = 64000$ bit per second (64kbit/s).
2. Audio Signal (High Fidelity, used in CD)
   - Bandwidth 15kHz, we assume a bandwidth of $B = 22.05$kHz.
   - Sampling frequency $f_s = 2B = 44.1$kHz, this means 44100 samples per second.
   - Each sample is quantized with $L = 65536$ levels, 16 bits per sample.
   - Thus, a Hi-Fi audio signals requires $16 \times 44100 \simeq 706$kbit/s.

Modulation

- The process of modulating a digital signal is called *keying*.
- As for the analogue case, we can choose one of the three parameters of a sine wave to modulate.

  1. Amplitude modulation, called *Amplitude Shift Keying* (ASK).
  2. Phase modulation, *Phase Shift Keying* (PSK).
  3. Frequency modulation *Frequency Shift Keying* (FSK).

- In some cases the data can be sent by simultaneously modulating phase and amplitude, this is called *Quadrature Amplitude Phase Shift Keying* (QASK)

Conclusions

- Highlighted digital communication systems
- Importance of digital communication
- Sampling and quantization
- Modulation of digital signals