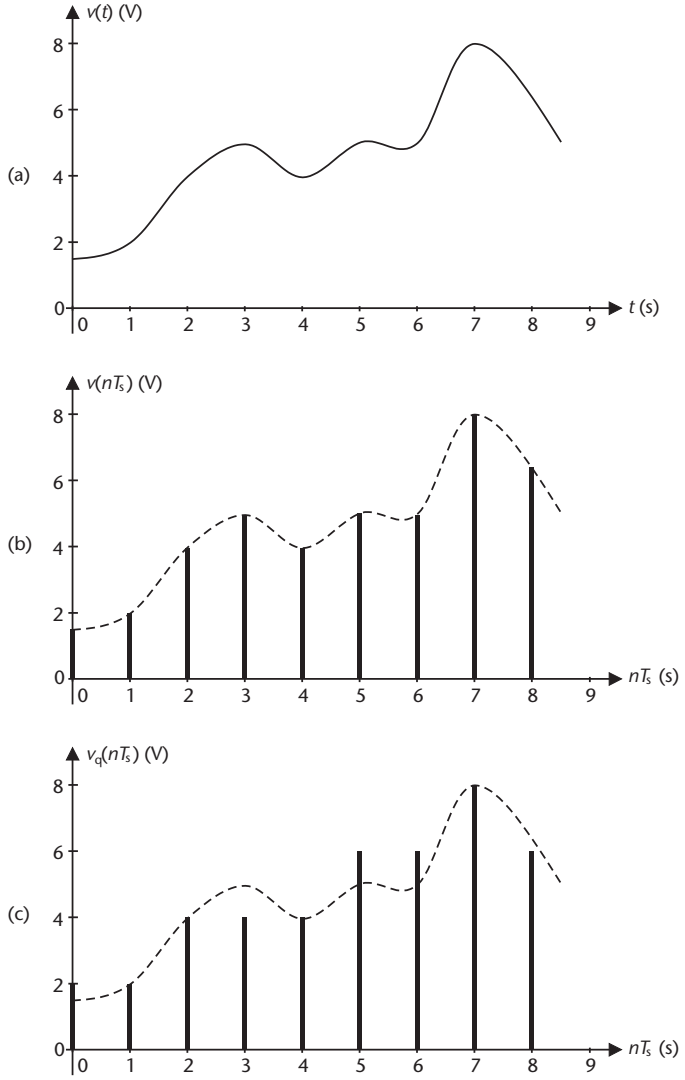
ELEC1207 Communications

8 ADC

Analogue to Digital Conversion (ADC)

Figure 1.18

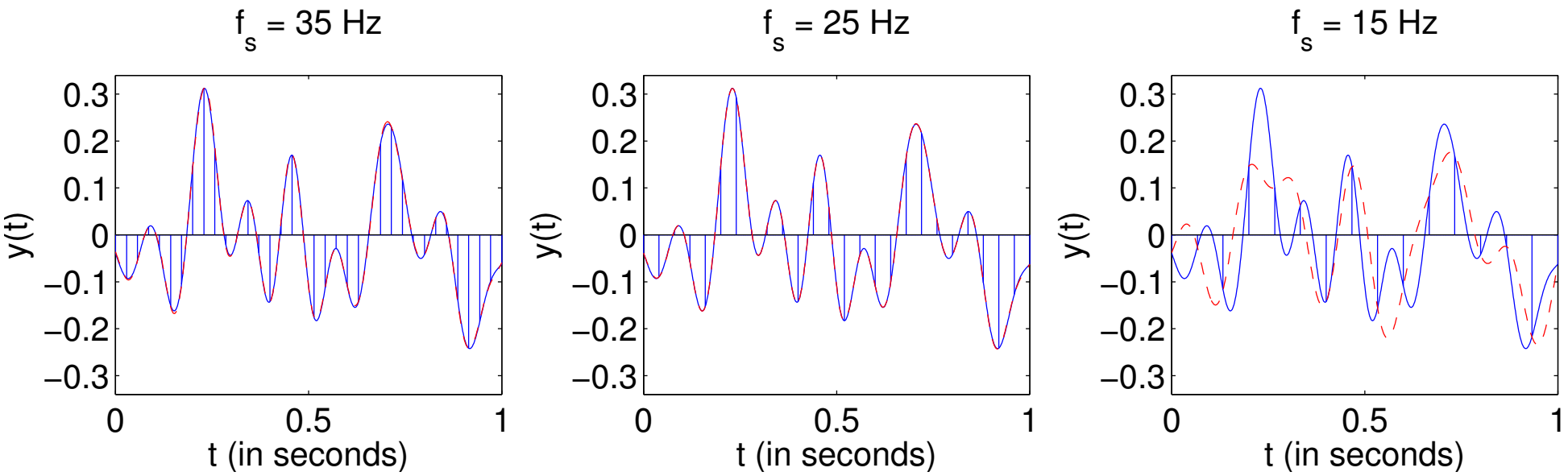


Taken from *Communication Engineering Principles*, © Ifiok Otung, published 2001 by Palgrave

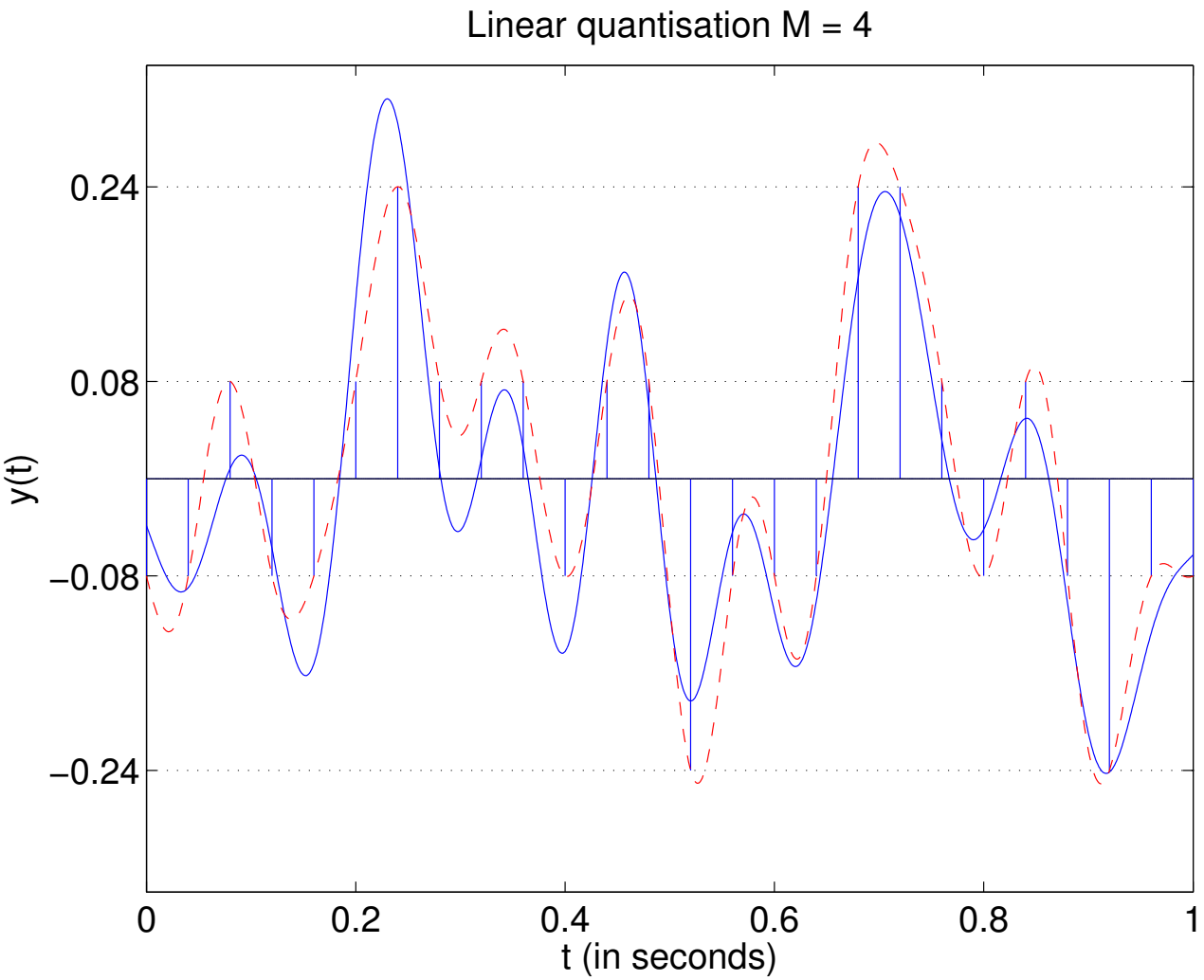
- (a) An analogue signal $v(t)$, which is continuous in time and continuous in value.
- (b) Sampling makes the signal discrete in time by only retaining the value of the signal at time instants t that are integer multiples of the sampling period $T_s = 1$ s. Sampling can be reversed by using a Low Pass Filter (LPF). To avoid aliasing distortion, Nyquist theory states that the sampling frequency $f_s = 1/T_s$ should be at least double the highest frequency in the signal $v(t)$.
- (c) Quantisation makes the signal discrete in value as well. This rounds each sample to the nearest value from the limited set $\{2\text{V}, 4\text{V}, 6\text{V}, 8\text{V}\}$.

The Nyquist theorem and desampling

- If $f_s > 2f_{max}$, the original signal can be perfectly reconstructed by low-pass filtering the samples using a cut off frequency of $f_s/2$.
- Despite discarding the signal value in between samples, sampling is lossless.
- The signal cannot be reconstructed if $f_s < 2f_{max}$, owing to alaising.
- A sampled signal is discrete in time, but continuous in value.



Linear quantisation

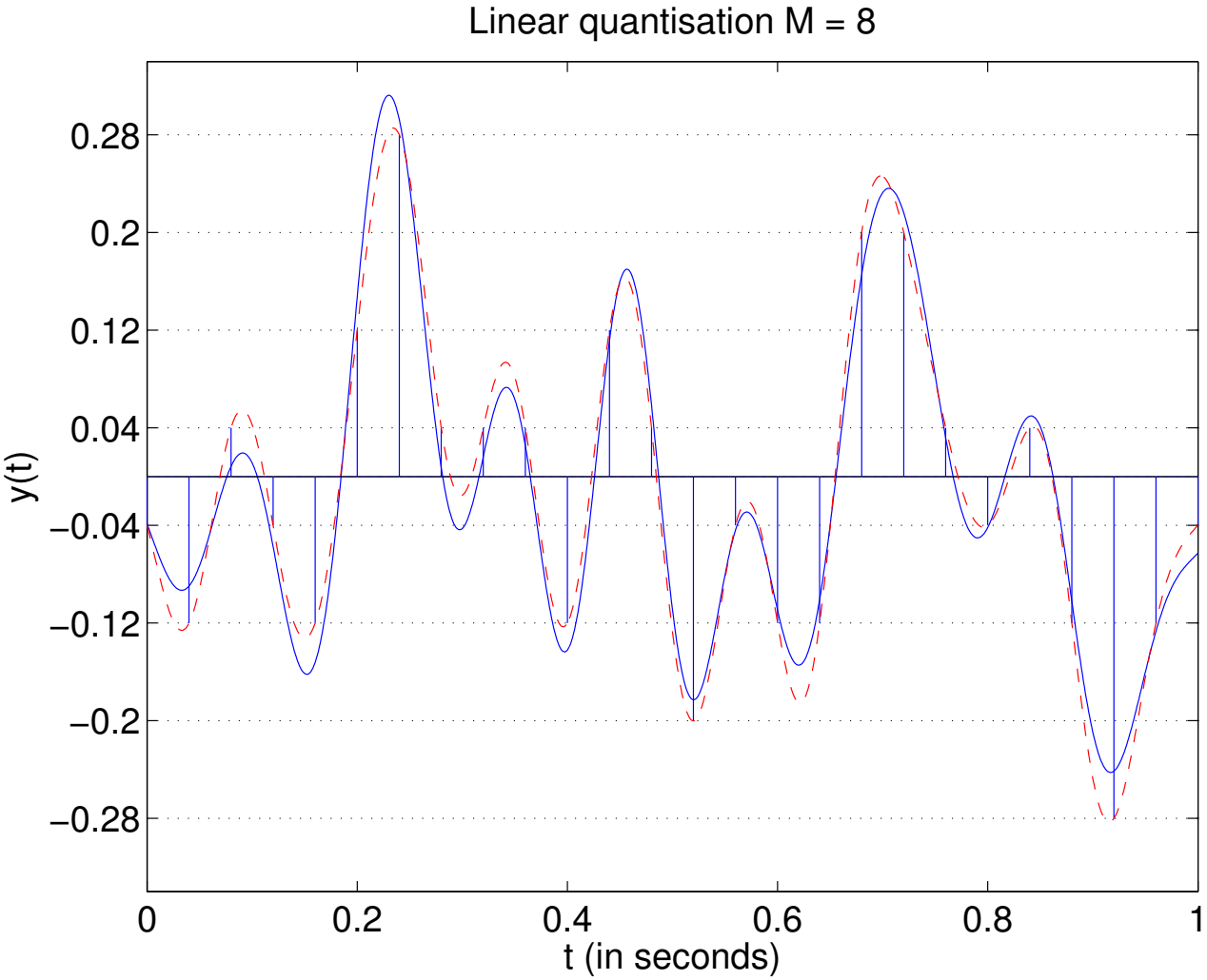


- The samples can be made to be discrete in value by quantising them to one of M quantisation levels.
- In linear quantisation, the quantisation levels are equally spaced.
- e.g. $M = 4$ levels with spacing of 0.16

Symbol	Level
1	+0.24
2	+0.08
3	-0.08
4	-0.24

Resultant sequence of symbols is 3,3,2,3,3,2,1,2,2,2,3,2,2,4,3,3,3,1,1,2,3,2,3,4,3

Number of quantisation levels

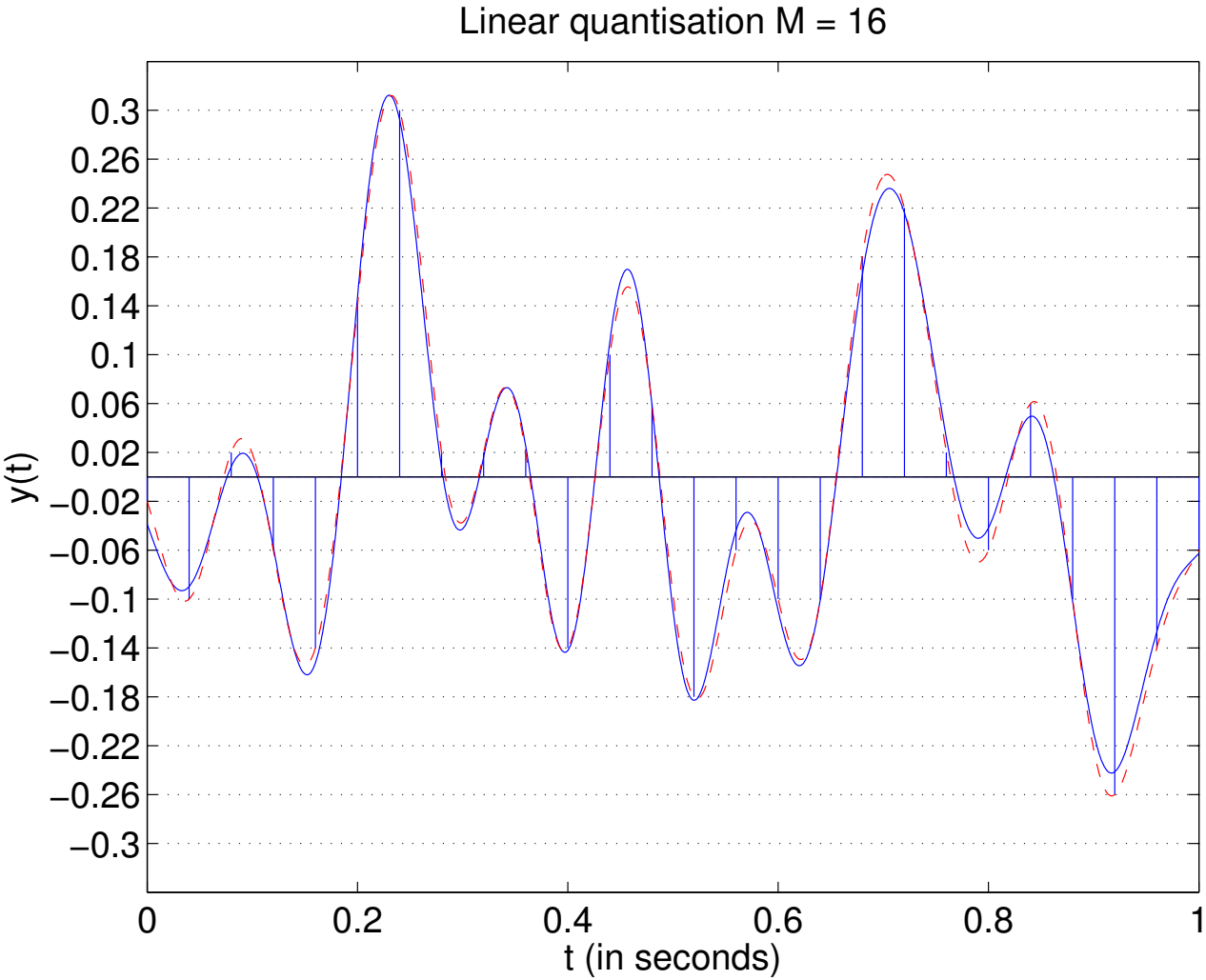


- Less quantisation distortion results by using more quantisation levels.
- e.g. $M = 8$ levels with spacing of 0.08

Symbol	Level
1	+0.28
2	+0.20
3	+0.12
4	+0.04
5	-0.04
6	-0.12
7	-0.20
8	-0.28

Resultant sequence of symbols is 5,6,4,5,6,3,1,4,4,4,6,3,4,7,5,6,6,2,2,4,5,4,6,8,6

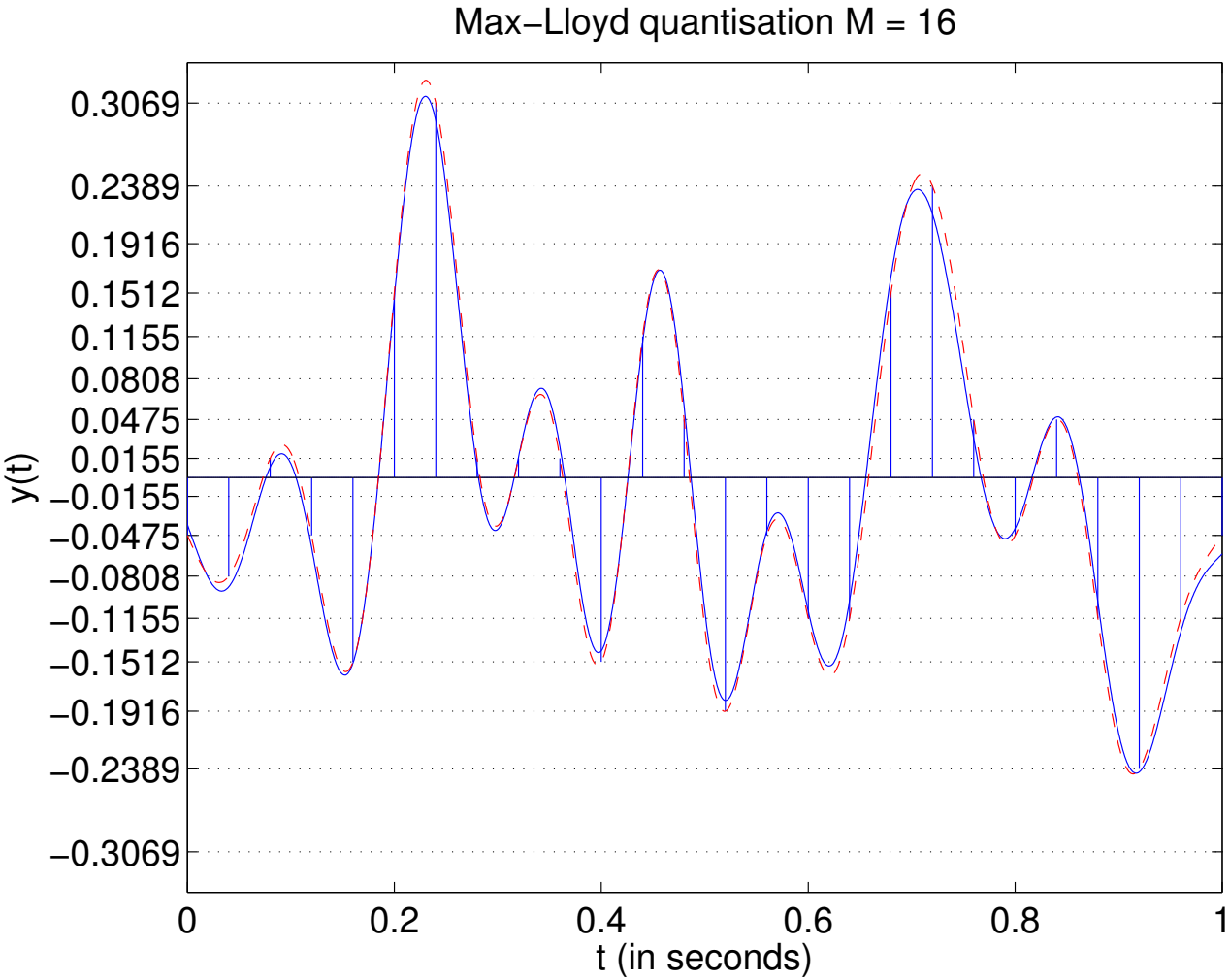
Quantisation is inherently lossy



- Quantisation is inherently lossy, since perfect reconstruction is only possible if an infinite number of levels is used.
- e.g. $M = 16$ levels with spacing of 0.04

Resultant symbols are 9,11,8,10,12,5,1,8,8,8,12,6,7,13,10,11,11,4,3,8,10,7,11,15,12

Non-linear quantisation



- Low amplitudes typically occur more frequently than high amplitudes.
- Quantisation distortion is reduced by spacing the quantisation levels closer together at low amplitudes.
- Max-Lloyd quantisation puts the quantisation levels in the non-linear positions that minimise the quantisation distortion.

Resultant symbols are 10,11,8,10,13,4,1,8,8,8,13,5,7,14,10,12,12,4,2,7,10,7,12,15,12

Pulse Coded Modulation (PCM)

Symbol	Codeword
1	0000
2	0001
3	0010
4	0011
5	0100
6	0101
7	0110
8	0111
9	1000
10	1001
11	1010
12	1011
13	1100
14	1101
15	1110
16	1111

- PCM converts each symbol in a sequence into a binary codeword.
- Each unique symbol value is mapped to a unique codeword.
- The binary codewords can be concatenated to form a bit sequence.
- If the symbols have values in the range 1 to M, then the corresponding codewords will comprise $k = \log_2(M)$ bits.
- Conversely, codewords comprising k bits can be used to represent symbol values in the range 1 to $M = 2^k$.
- e.g. $M = \{2, 4, 8, 16\}$ corresponds to $k = \{1, 2, 3, 4\}$.

PCM example

Symbol	Codeword
1	0000
2	0001
3	0010
4	0011
5	0100
6	0101
7	0110
8	0111
9	1000
10	1001
11	1010
12	1011
13	1100
14	1101
15	1110
16	1111

- Previously, we used a sampling frequency of $f_s = 25$ Hz and $M = 16$ Max-Lloyd quantisation to obtain the sequence of 25 symbols
  10,11,8,10,13,4,1,8,8,8,13,5,7,14,10,12,12,4,2,7,10,7,12,15,12
- We have one symbol per sample so the symbol rate R_s is equal to the sampling frequency f_s, giving $R_s = 25$ symbols per second.
- The corresponding $k = 4$-bit codewords can be concatenated to obtain the sequence of 100 bits
  1001101001111001110000110000011101110111100
  0100011011011100110111011001100010110100101 1
  0101111101011
- We have $k = 4$ bits per symbol so the bit rate is given by $R_b = kR_s = 100$ bits per second.