Multimedia Systems

II. Introduction to Sound

2.3. Digital Audio

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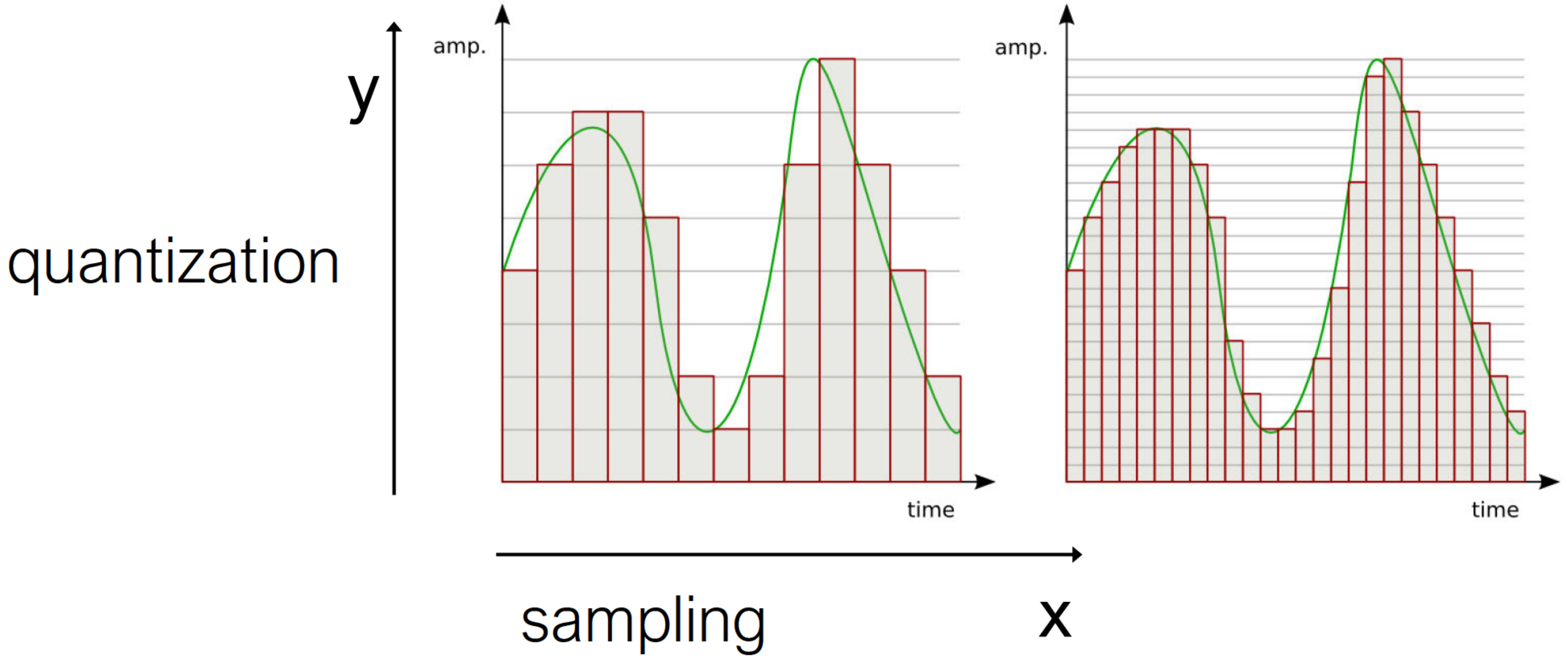
Agenda

- Analog to Digital Conversion
- Sampling and Quantisation
- Pulse Code Modulation
- Audio Coding and Formats
- Assessment of Audio Quality

A/D Conversion

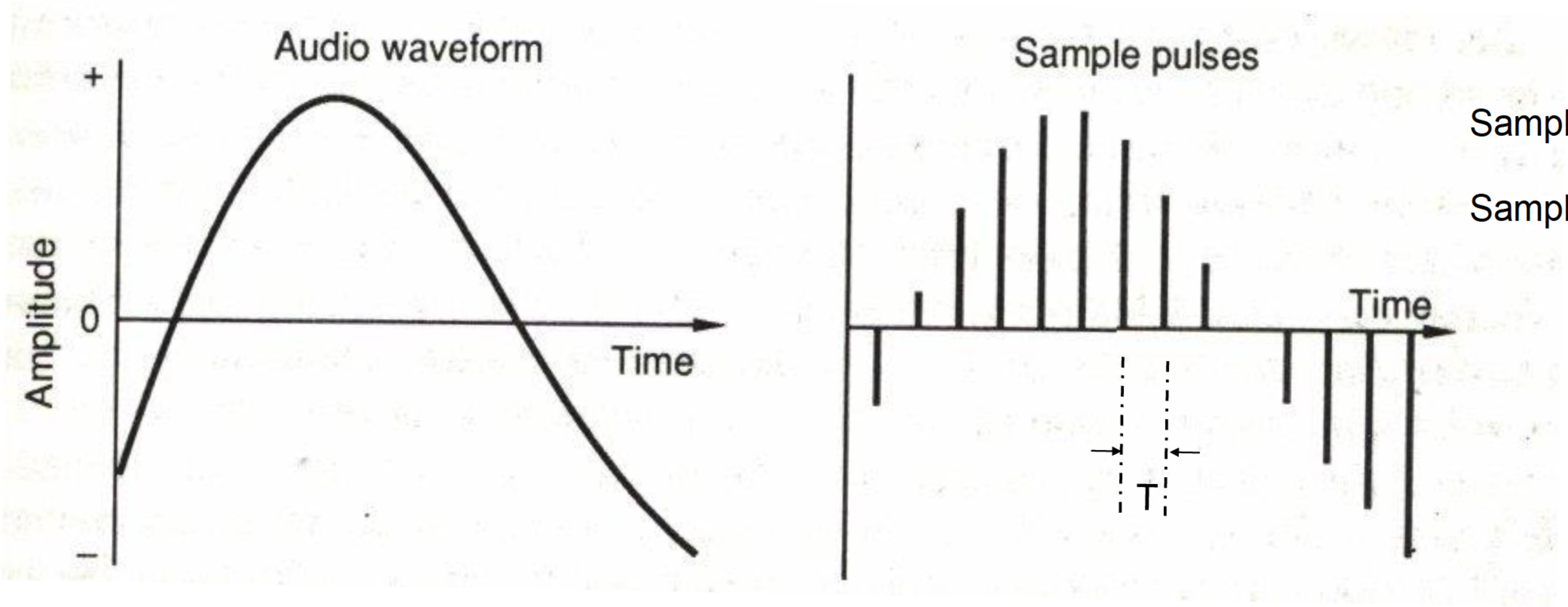
Analog-to-Digital conversion is used to convert the analogue audio signal (a time varying electrical voltage, say, the output of a microphone), into a series of ‘samples’ which are ‘snapshots’ of the analogue signal taken at periodic intervals (known as the sampling period).

The two main steps in digitization of sound are sampling and quantization.



Sampling (I)

Sampling is a matter of measuring amplitude at equally-spaced moments in time (sampling period). The number of samples taken per second (samples/s) is the sampling frequency (or rate). Units of samples/s are also referred to as Hertz (Hz).



Sampling period: T (in second)

Sampling frequency : f = 1/T (in Hz)

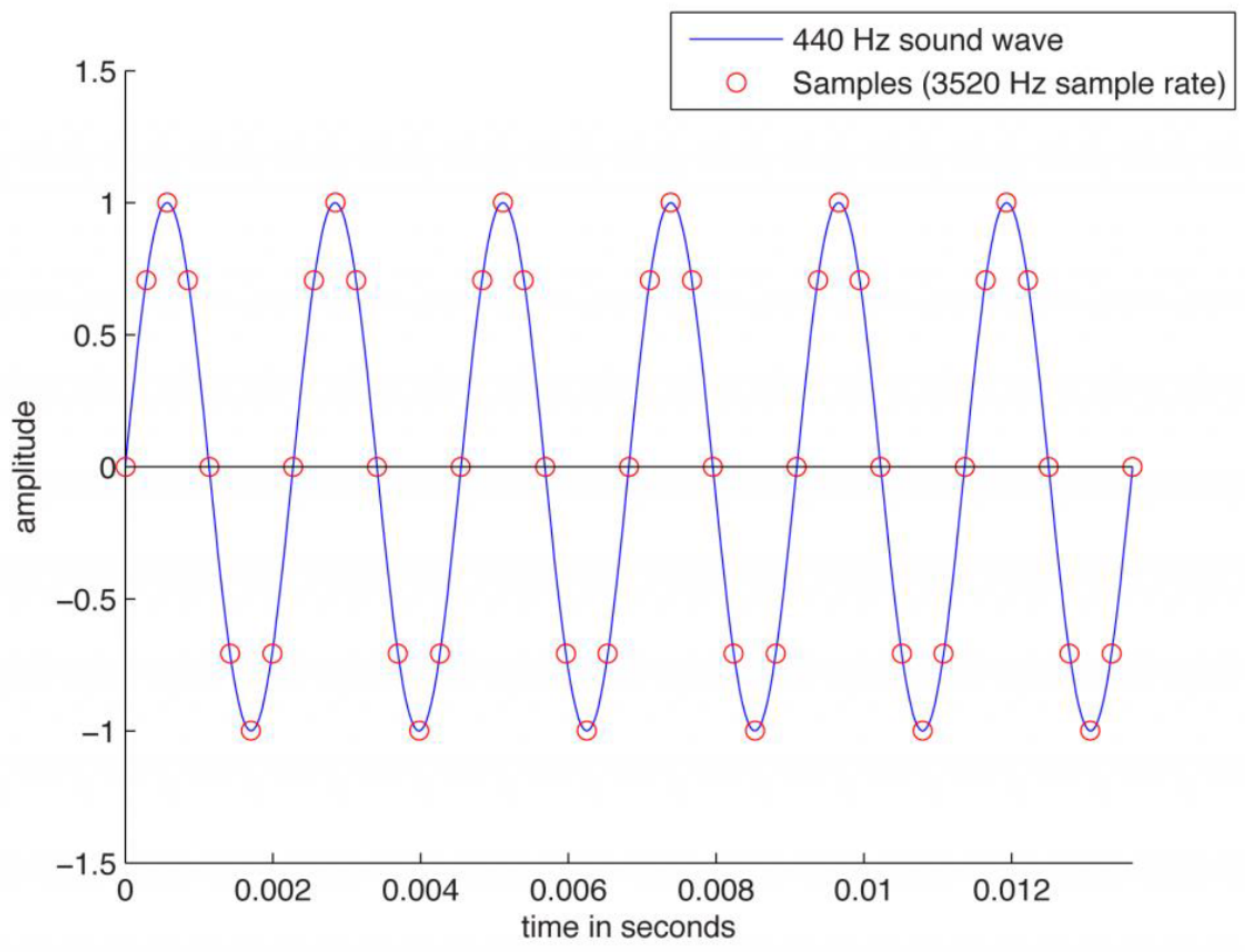
Sampling (I)

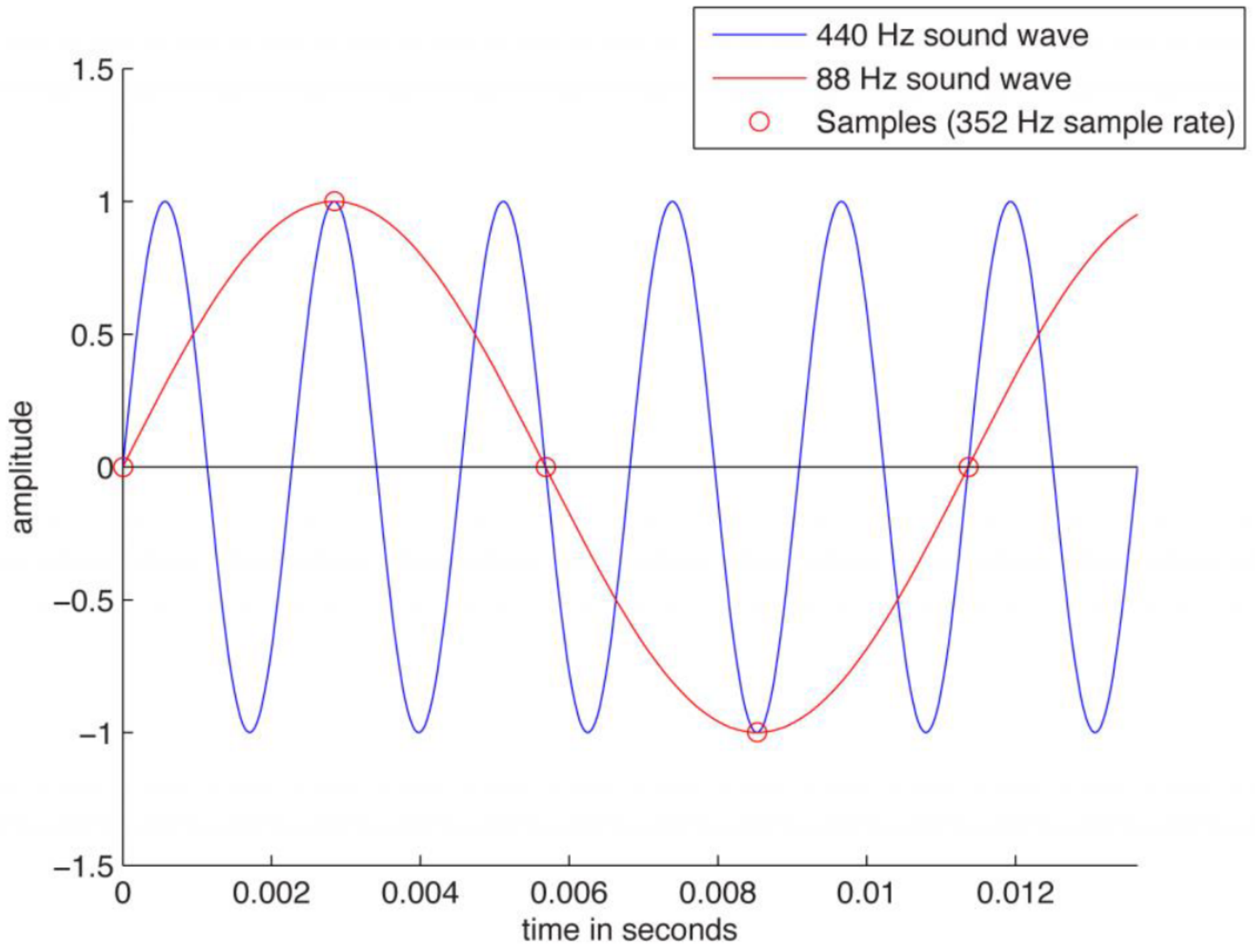
Nyquist-Shannon theorem: To reconstruct the analogue signal perfectly from the samples, at least two samples must be taken per period, ie, the sampling frequency should be at least two times of the frequency of the highest frequency component within the signal.

$$F_{sampling} \geq 2F_{signal}$$

Sampling (II)

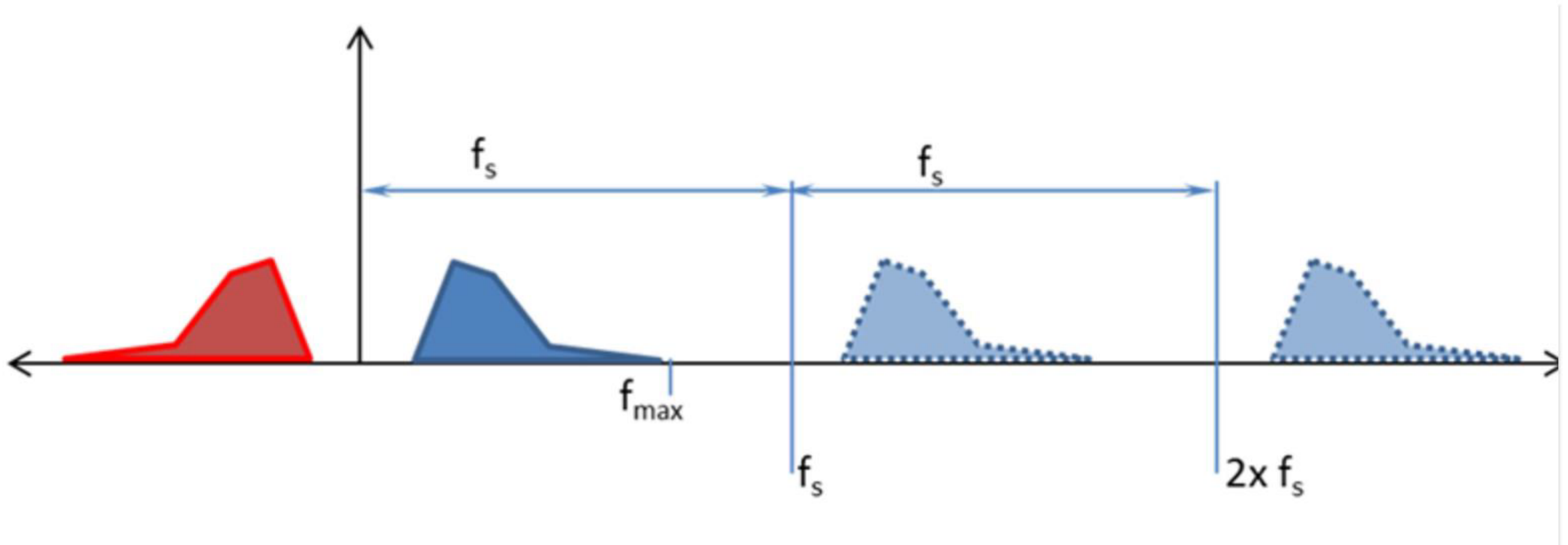
Aliasing Effect (Time Domain)

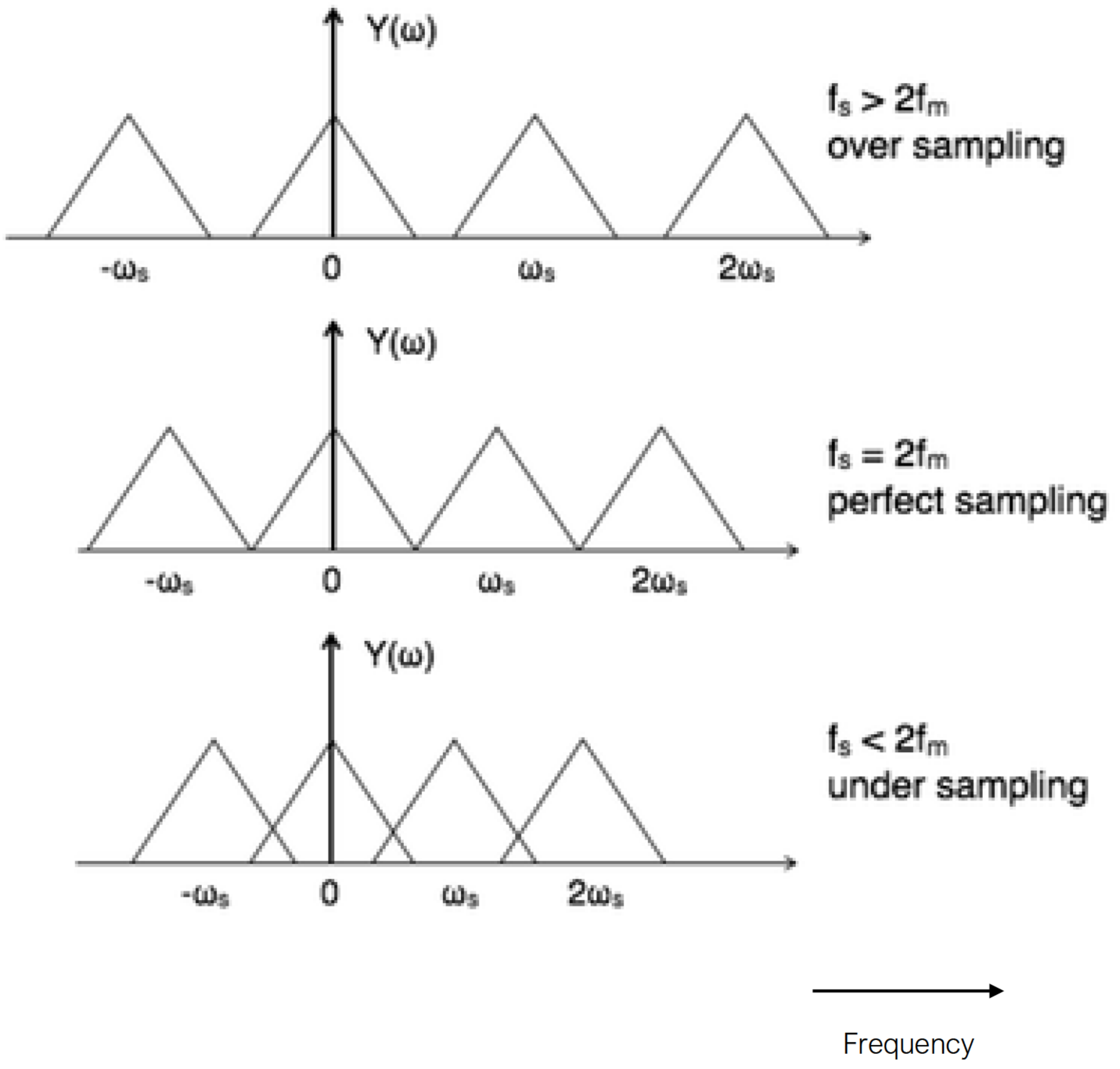




Sampling (III)

Aliasing Effect (Frequency Domain)





https://www.youtube.com/watch?v=0slziGiwZOg

Sampling (IV)

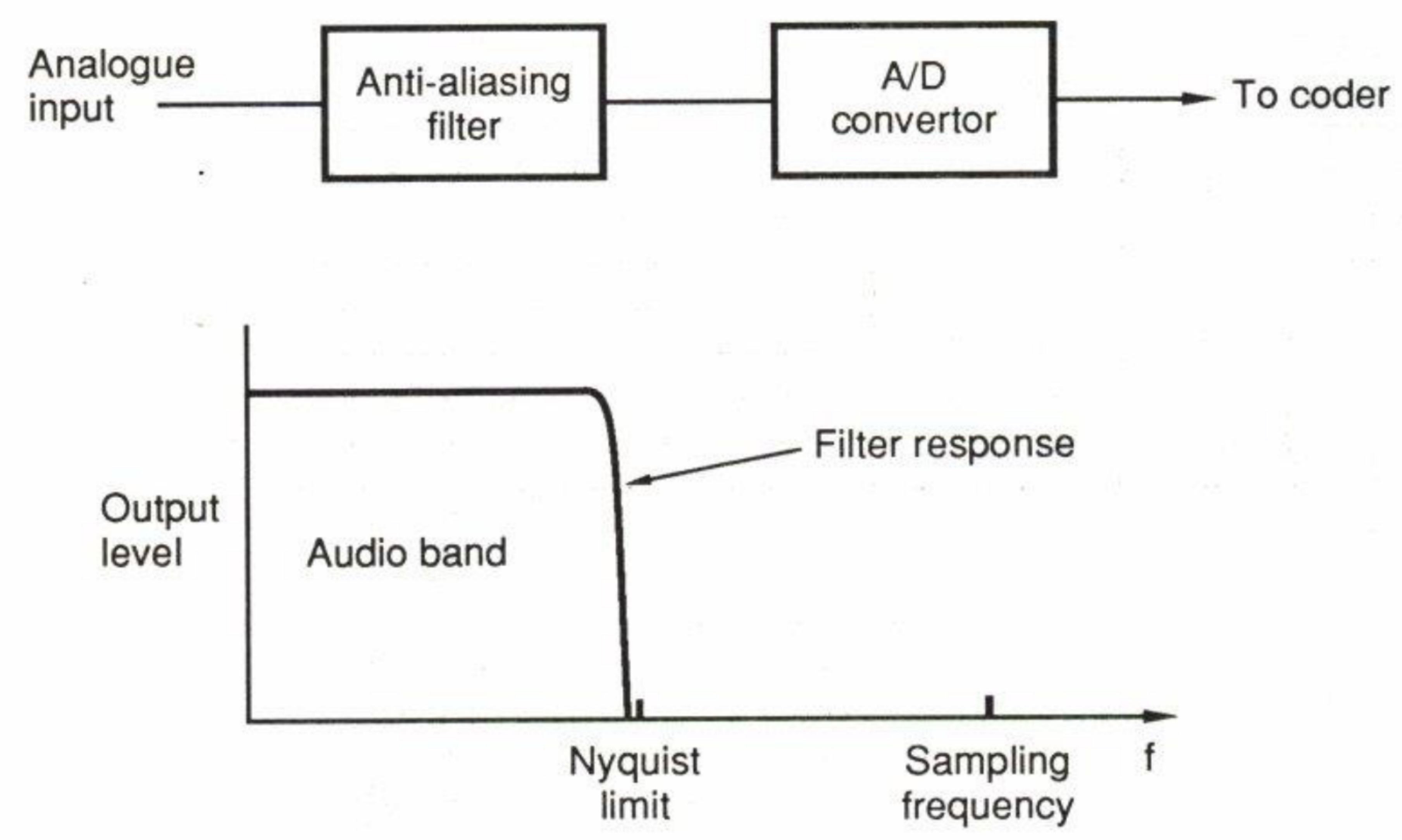
External Content

https://www.youtube.com/watch?v=0slziGiwZOg

Sampling (IV)

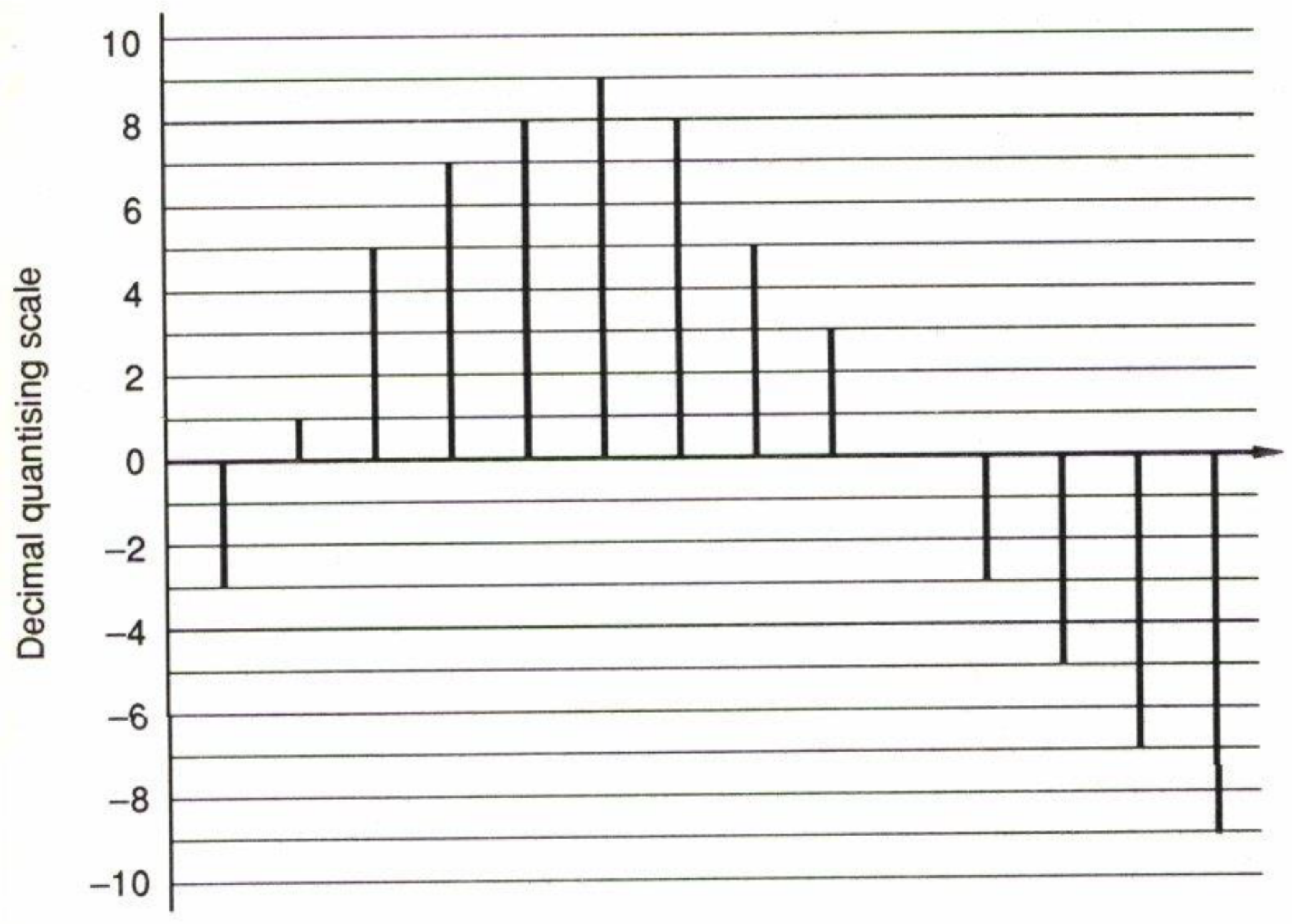
Anti-aliasing

One way to remove the aliasing effect is to make sure the sampling frequency to be **at least twice the highest frequency** in the signal. An “alternative” way is to use an **anti-aliasing filter** to remove the frequency components of the signal whose frequencies are higher than half of the sampling frequency (also usually called **Nyquist frequency**).



Quantisation (I)

Quantisation is a matter of representing the amplitude of individual samples in binary, by quantizing the measured samples into a finite number of discrete levels. The range of the integers possible is determined by the bit depth, the number of bits used per sample. A sample's amplitude must be rounded to the nearest of the allowable discrete levels, which introduces error in the digitization process.



The quantised sequence:

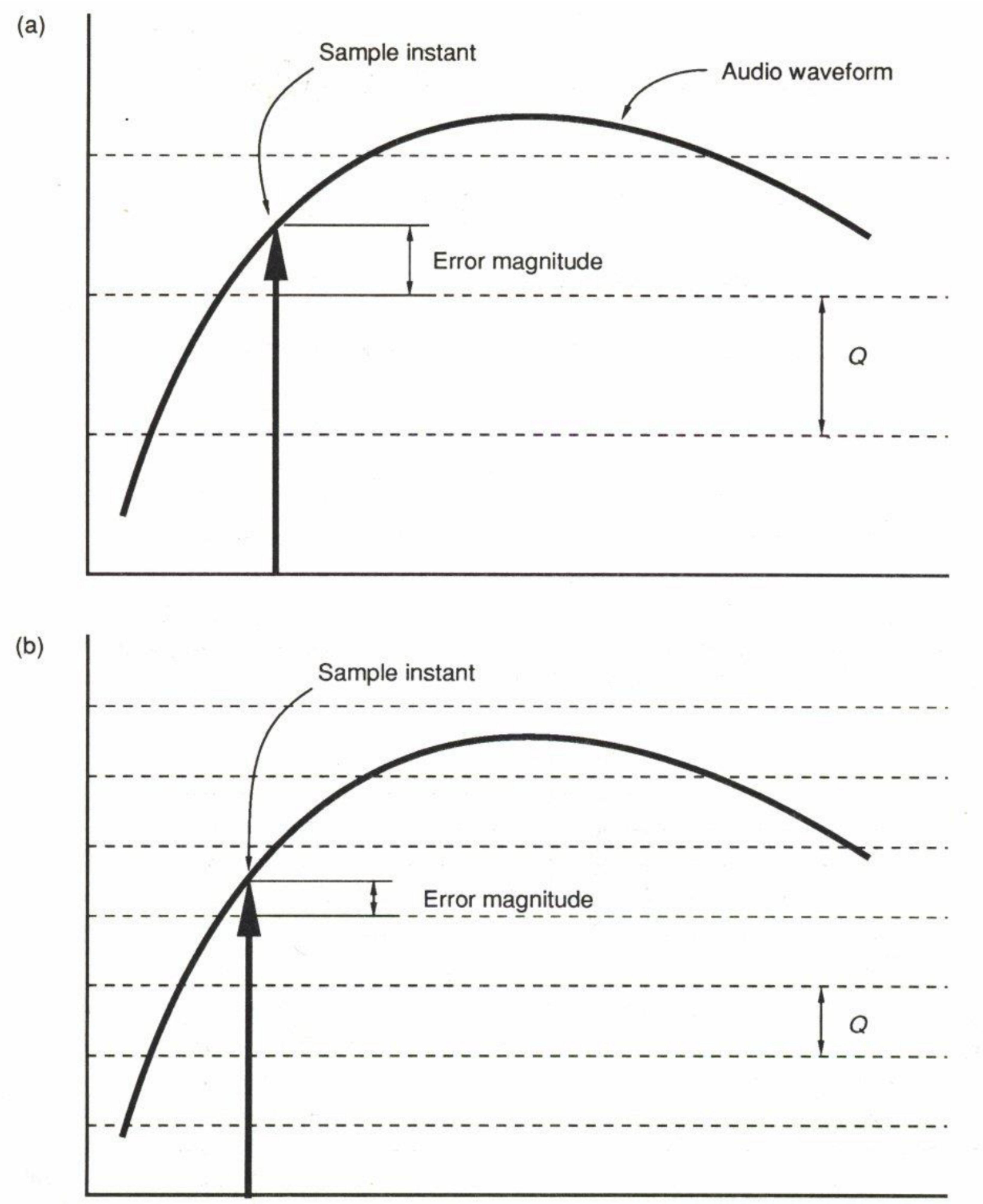
-3, 1, 5, 7, …, -5, -7, -9

Quantisation (II)

The difference between the sample amplitude represented by the numbers and the original amplitude of the sample is called quantisation error.

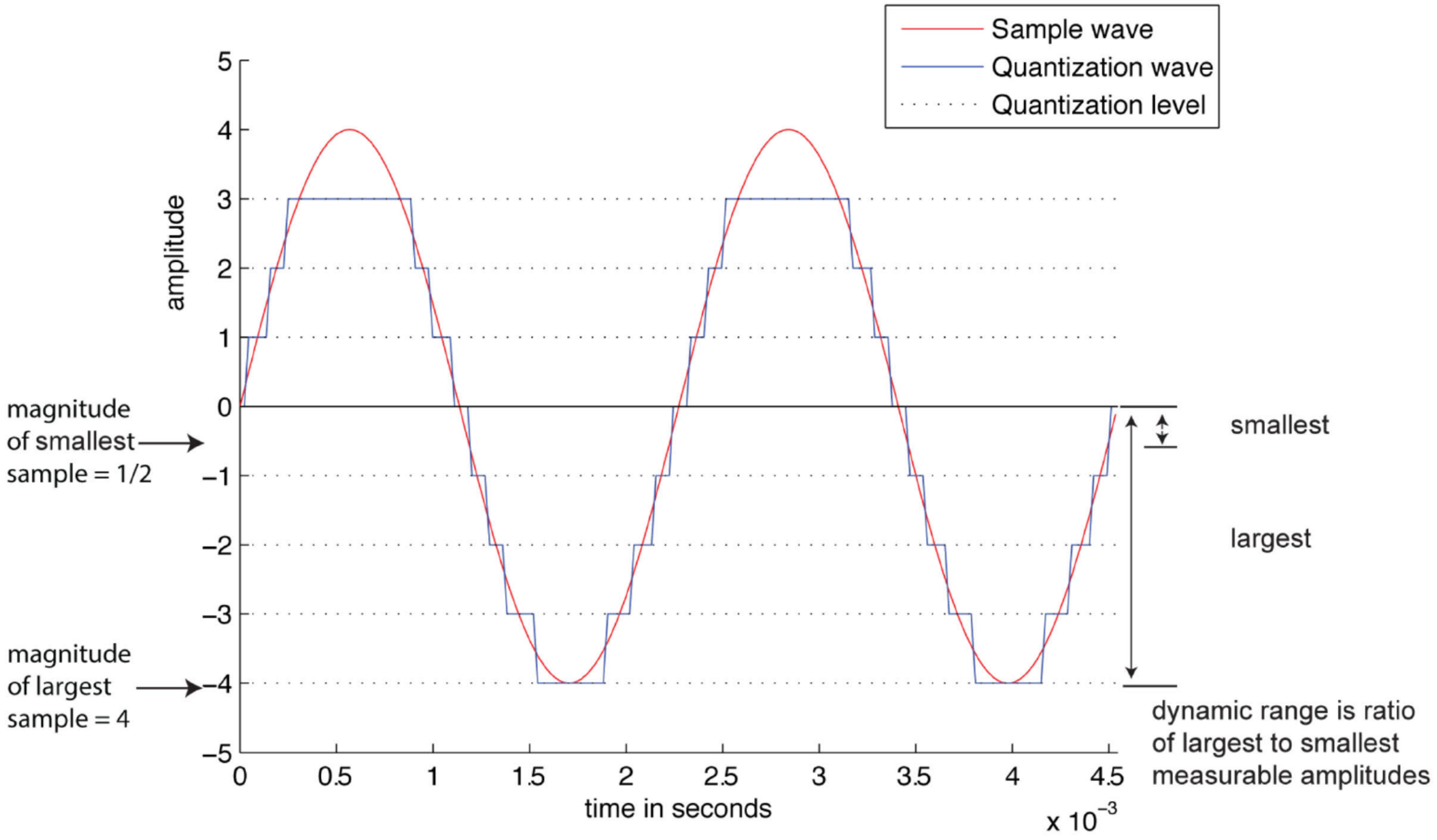
The maximum quantisation error will be half of a quantisation step size, Q.

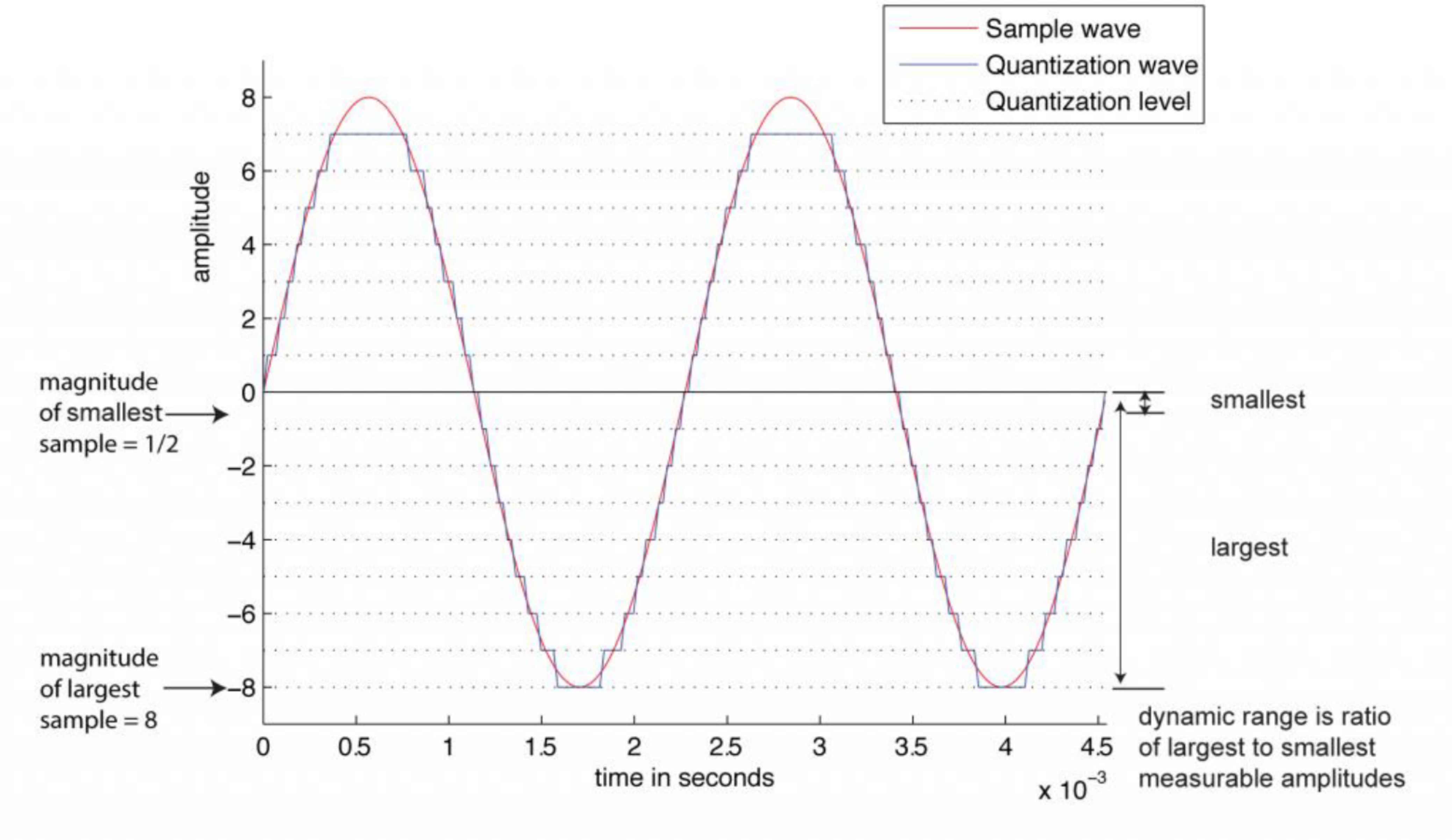
In the subplot (a), there are a fewer number of quantisation steps, therefore, the quantisation error is bigger, as compared to the subplot (b).



Quantisation (III)

The **dynamic range** refers to the possible range of high and low amplitude samples as a function of the bit depth.





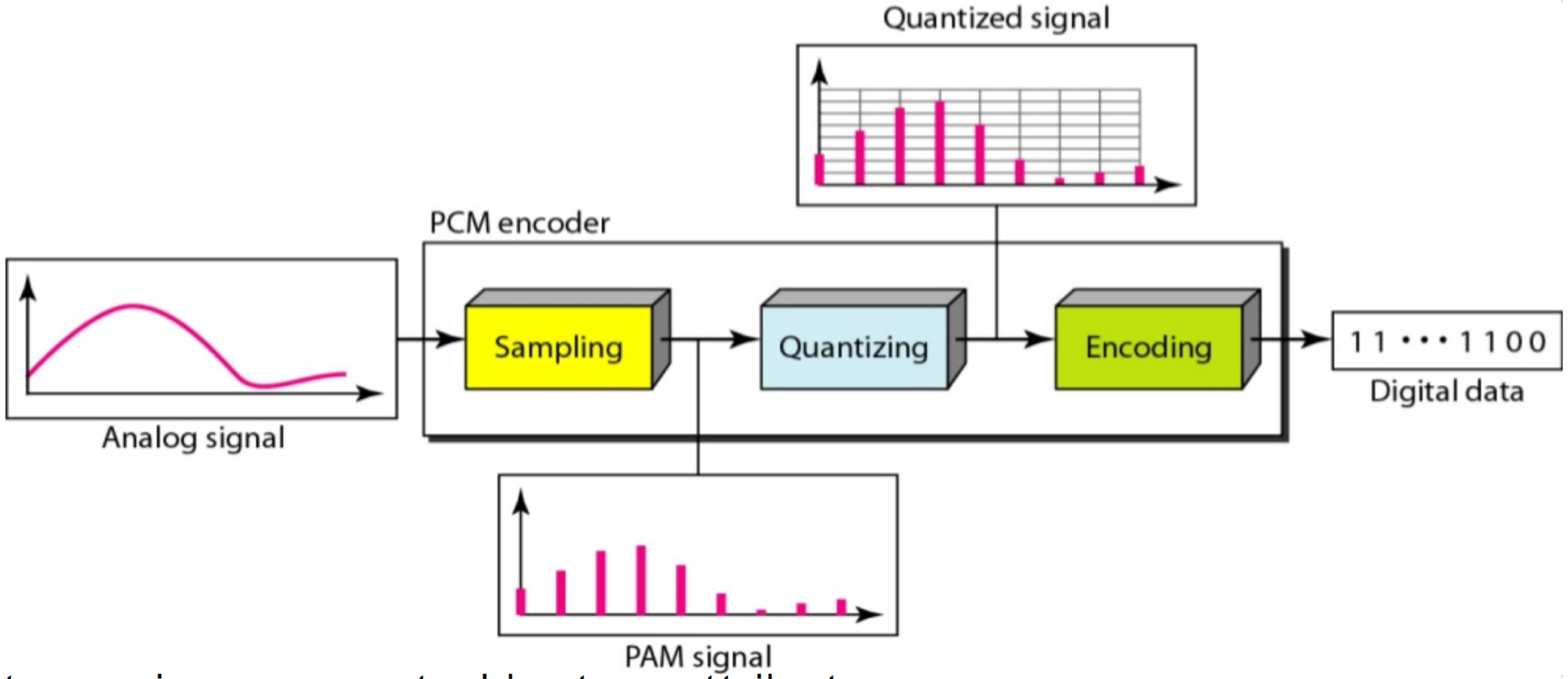
Quantisation (IV)

External Content

Max Patch on Quantisation

Pulse Code Modulation

Pulse Code Modulation (PCM) is a method used to digitally represent sampled analog signals. It is the industry standard for storing analog waves in a digital format. PCM is non-proprietary so anyone can use it for free!

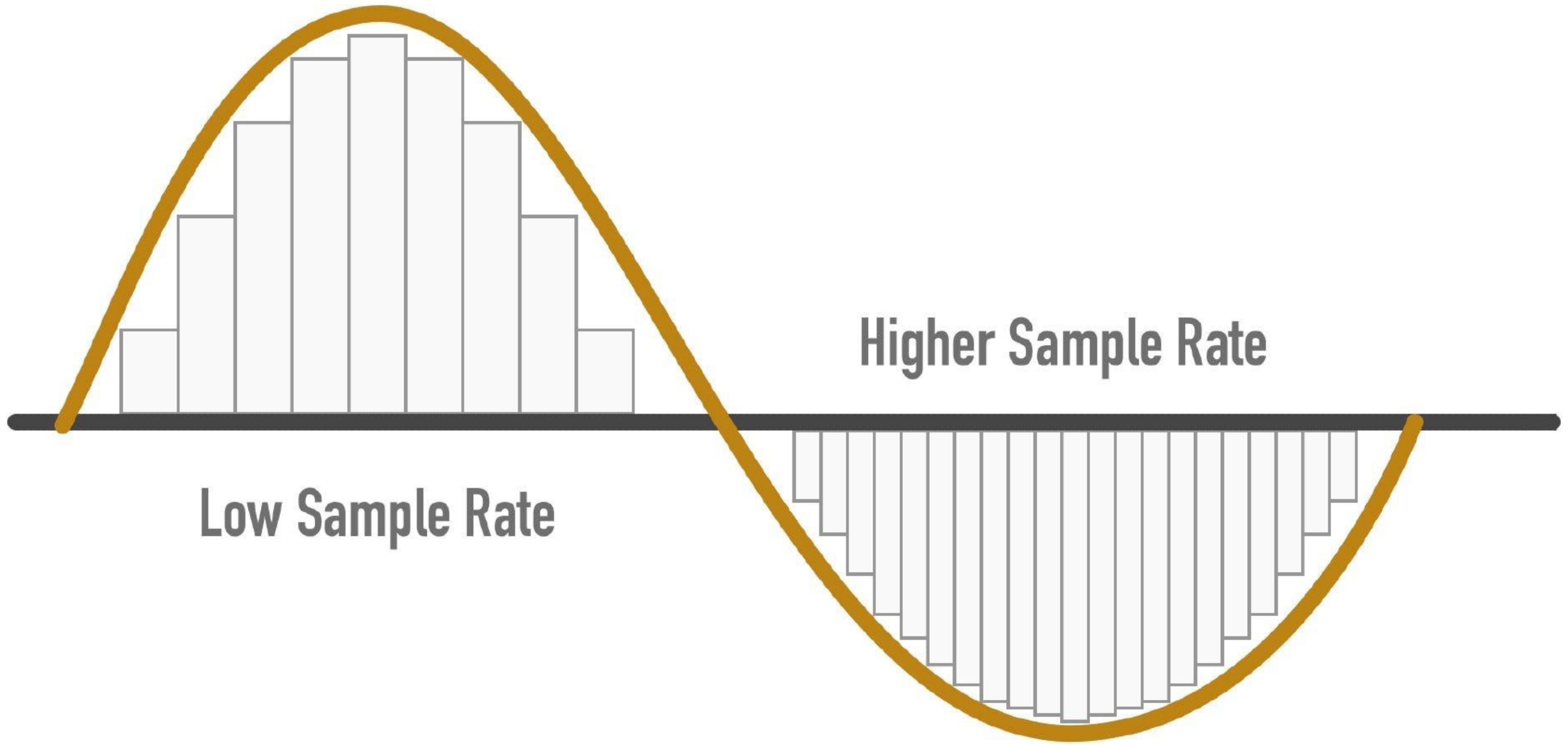


The fidelity/quality of a PCM stream is represented by two attributes:

- Sample Rate
- Bit Depth

Sample Rate

In a typical digital audio CD recording, the sampling rate is 44,100 or 44.1kHz.
Why this sampling rate? (Nyquist-Shannon sampling theorem)
A sampling rate of 44,100 samples per second or 44.1kHz allows for accurate reproduction of frequencies around about 22kHz.



Other examples of common sampling rates are 8KHz in telephones and between 96KHz to 192KHz for Blu-ray audio tracks.

Bit depth

The bit depth determines how much information can be stored, and with what resolution.

A sampling with 24-bit depth can store more nuances and hence, more precise than a sampling with 16-bit depth:

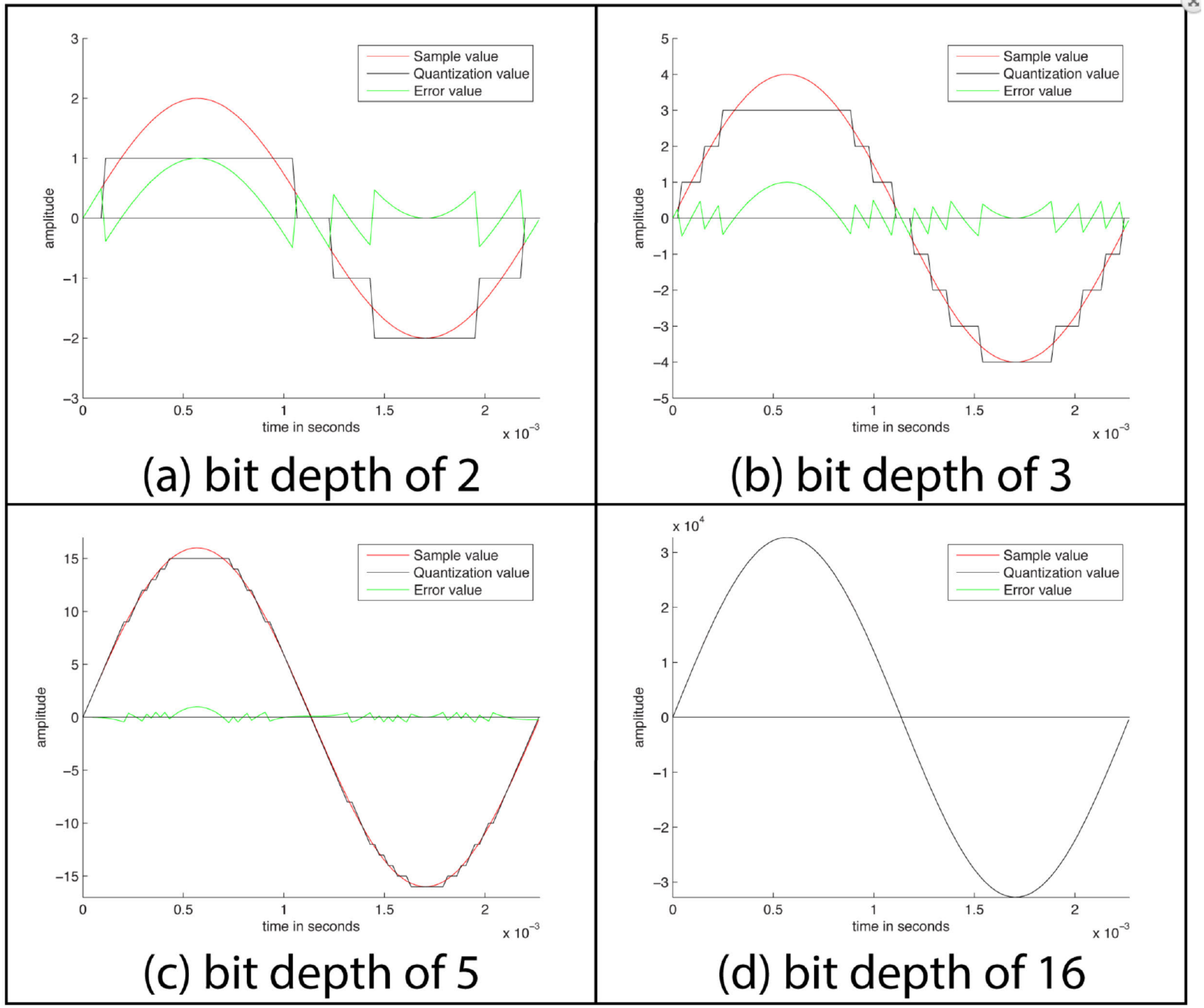
- 16-bit: We are able to store up to 2^{16}=65,536 levels of information, a dynamic range of 96dB
- 24-bit: We are able to store up to 2^{24}=16,777,216 levels of information, a dynamic range of 144dB.

Given a bit depth of n, the dynamic range of a digital audio recording is equal to

$$20\log_{10}\left(\frac{2^{n-1}}{1/2}\right)dB$$

Bit depth

Quantisation



Audio Coding

Key-Ideas

In perceptual audio coding, the main concerns in audio signals representation are:

- removal of **Redundancy**;
- removal of **Irrelevancy**;

Adapted from: Marina Bosi Perceptual Audio Coding, IEEE Signal Processing Society, 2015

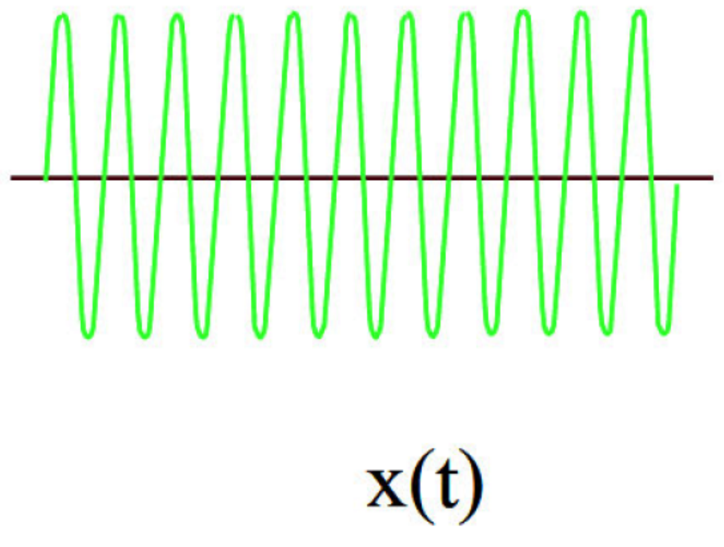
Audio Coding

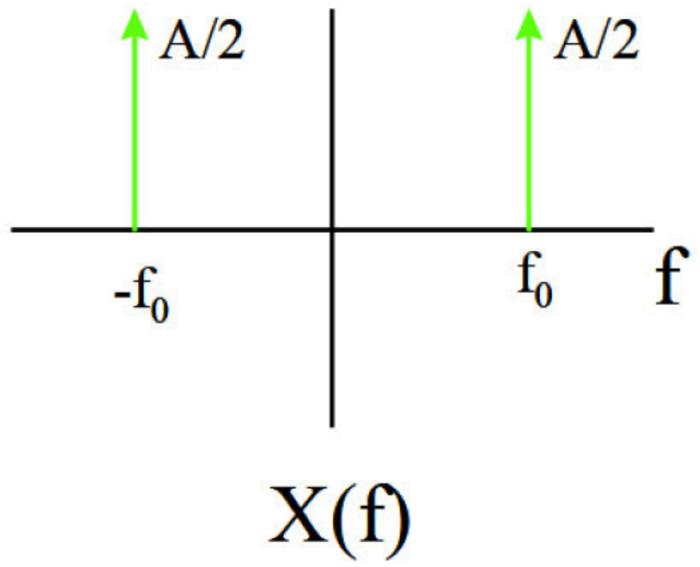
Redundancy

In audio coding, redundant means that the same information can be represented with fewer bits.

Example: represent a sine wave signal in 2 different representations:

- **Redundant**: sample the waveform 44.100 times per second and describe each sample with 16 bits;
- **Concise**: Describe the amplitude, frequency, phase and duration;





The concise representation of the sine wave is basically equivalent to the information in the Fourier transform;

Since music and many other audio signals are very tonal, most coders work in the frequency domain to reduce redundancy;

Adapted from: Marina Bosi Perceptual Audio Coding, IEEE Signal Processing Society, 2015

Audio Coding

Irrelevancy

In audio coding, irrelevant data means that you can't hear any difference in the audio signal if those data are omitted.

The main causes of irrelevancy:
- Hearing threshold;
- Masking;

Exploiting Irrelevancy:
- Don't code signal components you can't hear;
- Only quantise audible signal components with enough bits to keep quantization noise below the level it can be heard;

Adapted from: Marina Bosi Perceptual Audio Coding, IEEE Signal Processing Society, 2015

Audio Coding

Lossless coding

- Based on statistical relation between symbols within the data
- Entropy coding such as Huffman coding, arithmetic coding etc.
- The original signals can be perfectly reconstructed.

Lossy coding

- Based on the perceptual modelling of audio signals (such as psychoacoustic models of hearing), some redundant information within audio signals can be removed without affecting their perceptual quality.
- The original signals cannot be perfectly reconstructed.

Audio File Formats

Format Types	Pros	Cons	Popular formats
Uncompressed	• No data loss • Compatible with older software	• Occupies excessive space	• WAV & AIFF: Store lossy/lossless formats • LCPM • BWF: British Wave Format (used by Tascam)
Compressed Lossless	• Reduces processing time • Retains data and good compression ratio	• Files are sizable compared to lossy format	• FLAC: Reduces files to 50-60% of original size and decompresses to an identical copy • ALAC: Apple-only .mp4
Compressed Lossy	• Small file size • Most popular consumer audio format	• Uses psychoacoustics to lop off "imperceptible" audio info	• MP3 • AAC • WMA • ATRAC

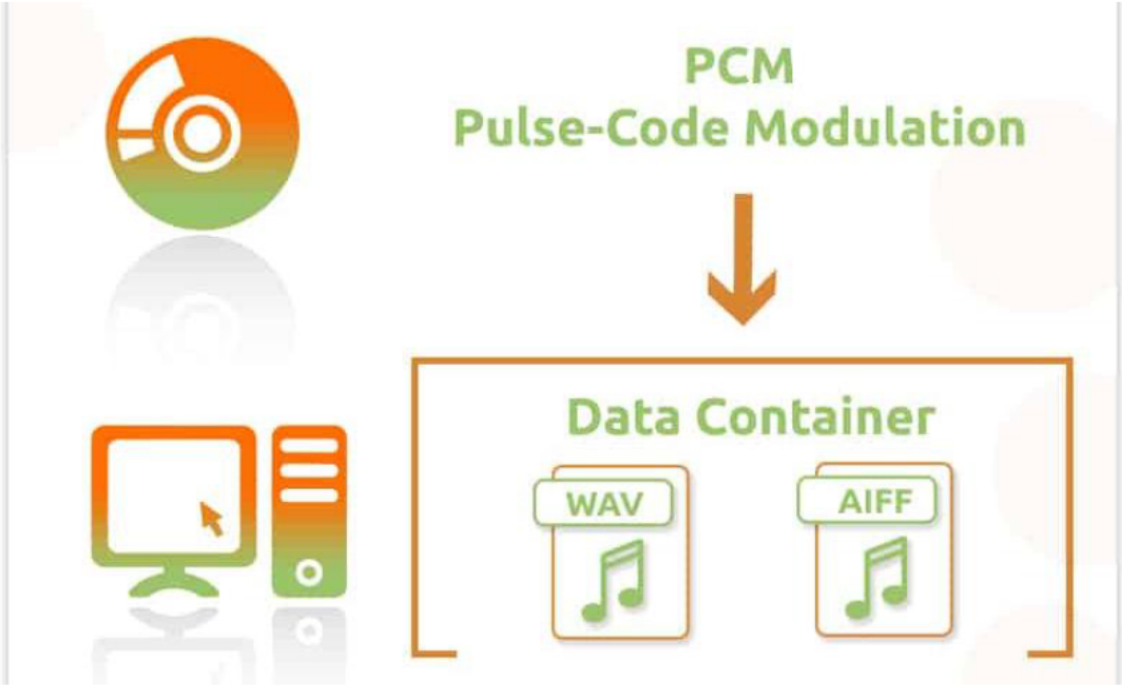
from: soundguys.com

Audio Coding (Codecs and File Formats)

- A Codec is an algorithm that performs the encoding and decoding of the raw audio data.
- Audio data itself is usually stored in a file with a specific audio file format.

- Most audio file formats support only one type of audio data (created with an audio coder), however there are multimedia digital container formats (as **AVI**) that may contain multiple types of audio and video data.

- A **digital container** format is a meta-file format where different types of data elements and metadata exist together in a computer file.
    - Formats exclusive to audio include, e.g., **wav**, **xmf**.
    - Formats that contain multiple types of data include, e.g. **Ogg**, **MP4**.



Audio Coding

History of Standards

1979 - the „Critical Band Coder“
1982 - „classic ATC“ for Music
1985 - MSC
1987 - OCF
1987 - MASCAM
1987 - PXFM
1990 - ASPEC, MUSICAM
1992 - MPEG 1
1996 - ePAC
1997 - MPEG 2 AAC
1999 - MPEG 4 AAC
2002 - HE AAC
Current standardization: MPEG surround, ALS, SLS

Timeline for near-CD-quality

1990 256 kbit/s ASPEC, MUSICAM
would fail today’s listening tests

1992 192 kbit/s MPEG-1 Layer-3

1994 128 kbit/s MPEG-1 Layer-3 (".mp3")
including combined joint stereo coding
bad quality for some signals

1997 96 kbit/s MPEG-2 Advanced Audio Coding
better than MP3 at 128, not fully transparent

2000 64 kbit/s AAC-based MPEG-4

2003 48 kbit/s MPEG-4 HeAAC (AAC+ in 2000)
e.g. used for XM Radio

From: Brandenburg Audio Coding Lecture @ Ilmenau University of Technology

Assessment of quality of coded audio

Objective Quality

- **Traditional objective measure**: The quality of audio is measured, ignoring psychoacoustic effects, through the use of objective performance indexes, such as *Signal to noise ratio* (SNR) and/or *Total block distortion* (TBD).
- **Perceptual objective measure**: The quality of audio is predicted based on a specific model of hearing.

Subjective Quality

- **Human listening tests**: When a highly accurate assessment is needed, formal listening tests will be required to judge the perceptual quality of audio.

Quality measurement of coded audio

Experiment of "13dB miracle" (Johnston and Brandenburg, 1991)

The original signal was injected with noise (with SNR signal-to-noise ratio of 13 dB) that was either:

a) White;
b) Shaped according to psychoacoustic masking models;

Demo

Original

a) White noise

b) Shaped noise

Sounds from: https://homes.esat.kuleuven.be/~compi/demos/

2.3. Digital Audio

MIDI and 3D Audio

HOMEWORK

Make a video presentation* with 3 (maximum) slides explaining what is MIDI and another 3(max) slides for 3D Audio, in a way that is consistent with the rest of classes.

This is a GROUP work and must be delivered in Moodle until next Sunday 19-Apr.

* Like our theoretical classes have been, just slides and audio (your voice narrating). One of the presentations must be narrated by one student and the other presentation narrated by the other.