**Analogue to Digital Conversion**

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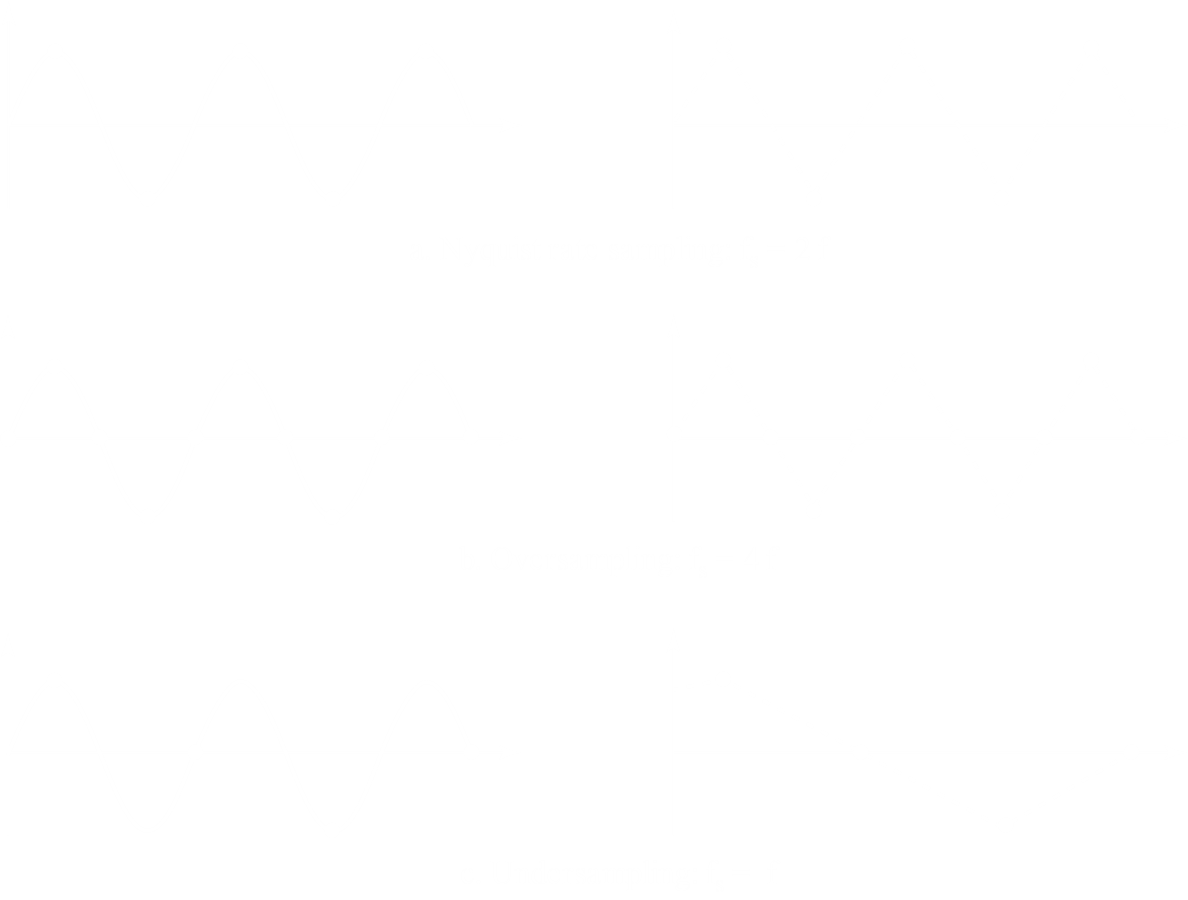
Analogue signals are converted to digital data through a series of steps. We take an analogue signal, process it and covert it to digital data using a converter. This data can now be handled by a processor. On the other end, another converter converts the digital data back to analogue signals that are again processed to remove noises and inaccuracies and finally given as output. The reason we need to convert back to analogue signals is because humans will be unable to make sense of purely digital data.

## Sampling

The first step is called sampling. We take samples of the analogue signal, which consists of an infinite number of frequencies, at regular intervals, and these samples can be used to reconstruct the analogue signal.



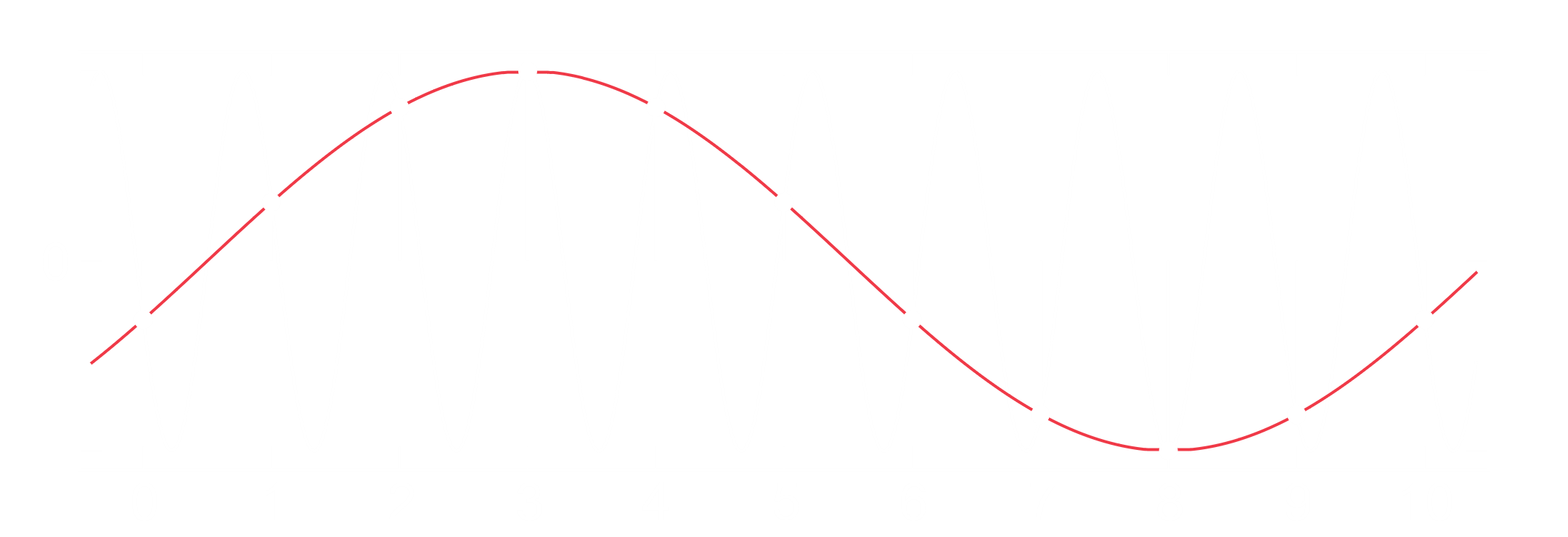
The frequency with which we take samples is determined by the Nyquist sampling theorem, which states that we need to take samples at a frequency that is at least as high as the twice the frequency of the highest frequency component of the analogue signal. Taking samples at a higher frequency, called oversampling, will not improve the results, and would increase costs since we would need more sophisticated hardware. Taking samples at a lower frequency, called under sampling, will make the results less accurate.



Closely related to sampling frequencies is the sampling rate, which deals with the number of samples taken per second. The sampling rate should be at the Nyquist rate.

## Aliasing

Aliasing occurs when a system is measured at an insufficient sampling rate. A false lower frequency component appears in the sampled data of a signal due to under sampling.



One of the ways to avoid aliasing is to use an anti-aliasing filter. This is a low-pass filter that is used to eliminate all frequencies in the signal that are above the Nyquist frequency before sampling is done. This causes some data loss, but ensures a good reconstruction.

## Resolution

Resolution is the degree of accuracy to which changes in the analogue signal are detected.

## Accuracy

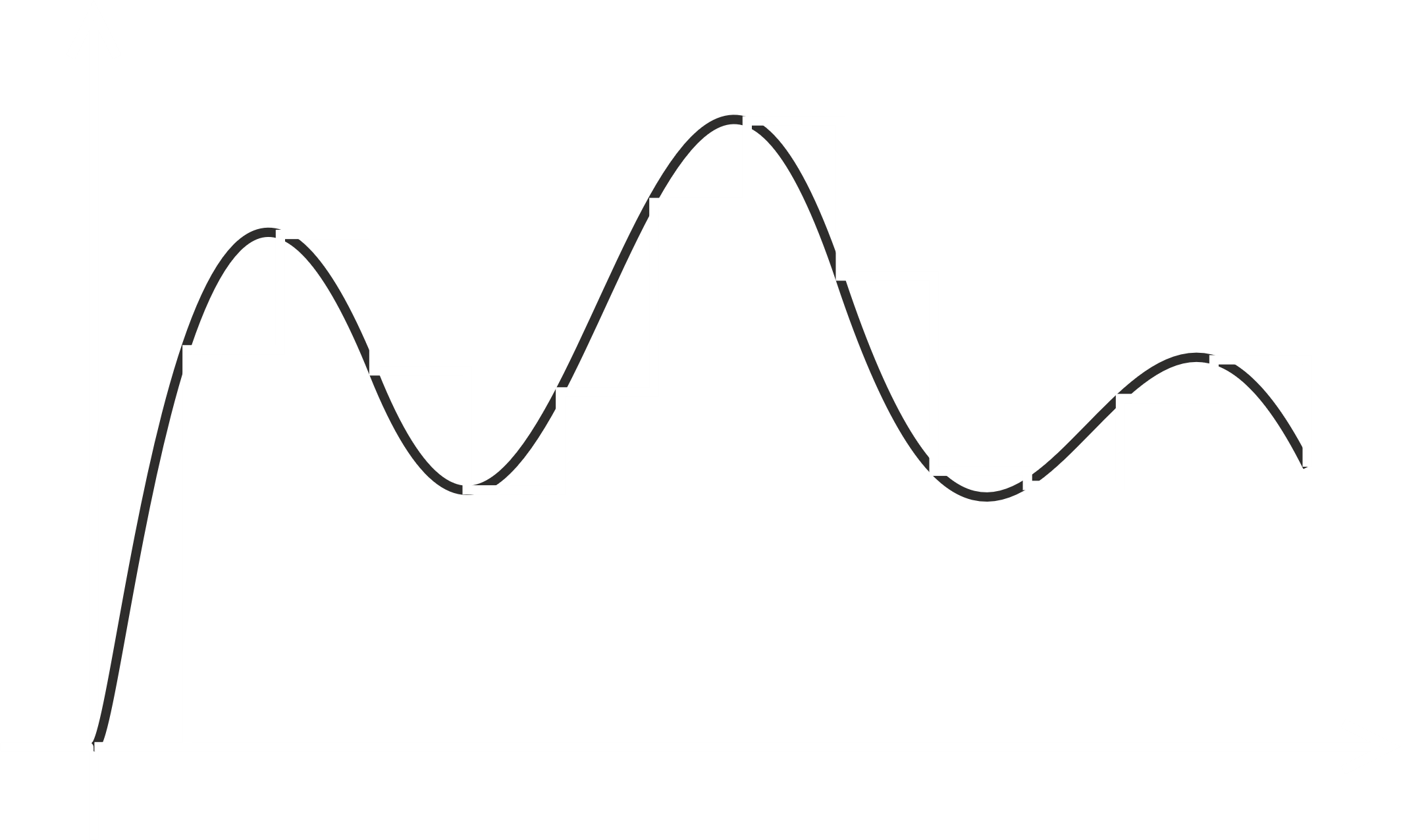
The accuracy of the sampled data depends on the sampling rate and the resolution.

A higher sampling rate means more samples are being taken per unit time. This, in turn, results in us being more likely to catch smaller changes in the signal. For example, if a constant signal has a sudden change to in between, a low sampling rate would miss the point where that change occurred, but a high sampling rate would not.

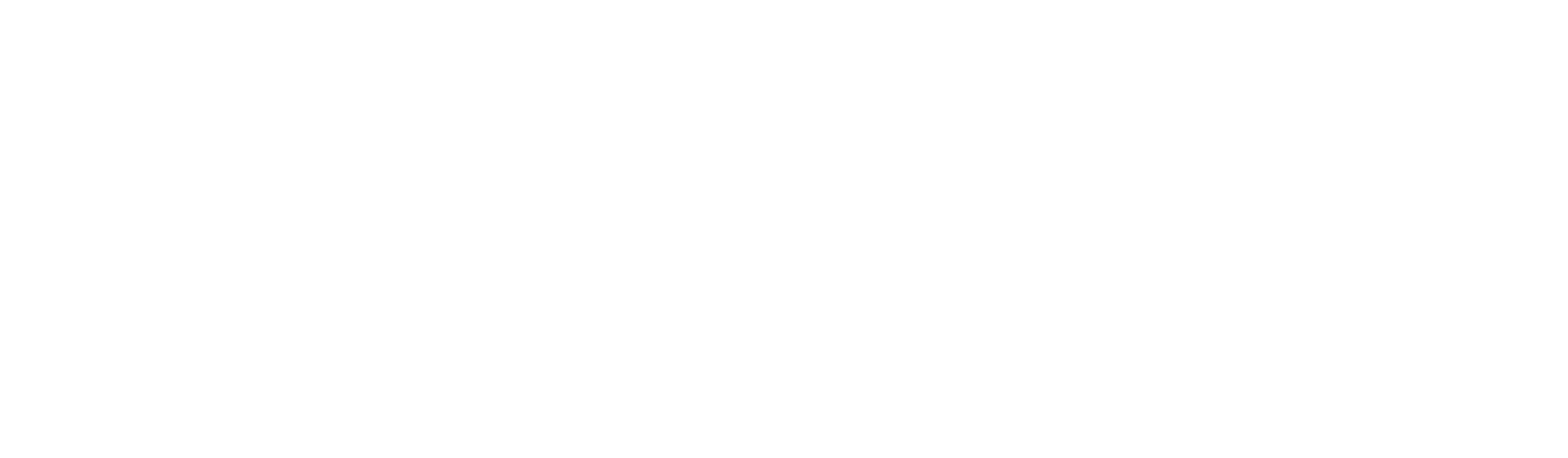
A higher resolution means we can take each of our samples more accurately. Again, this results in us noticing smaller changes. For example, if a signal has a point where the voltage is , but our sampling resolution only allows us to detect signals to an accuracy of , we would detect this point as , which would be ever so slightly inaccurate. If we increased our resolution and could now detect signals to an accuracy of , we would be able to read this point exactly.

## Sample and Hold Circuits

In sample and hold circuits, the input signal is sampled and the sampled value is held until a new sample is taken. This results in a staircase shape instead of just vertical bars. The holding period may be between a few milliseconds to several seconds.



This process makes use of a FET.



## Quantization

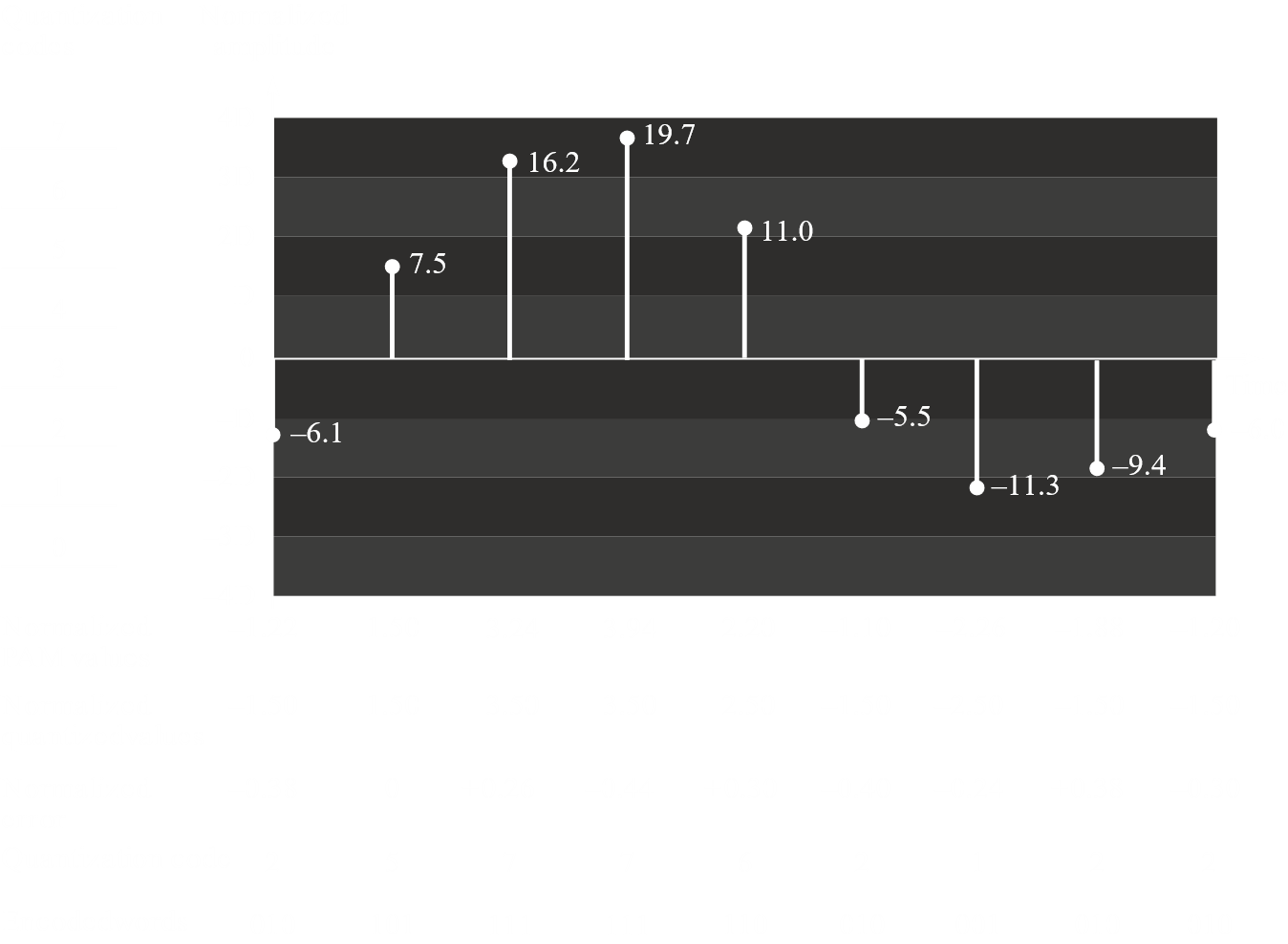
The samples we have taken still fall under the category of an analogue signal. We need to convert it to digital data, and to do that, we need to decide on how accurately we want to represent the samples taken. The number of bits used to represent each sample is called the quantization level. The more bits we use, the more levels we end up with and the more accurate the representation.

In the original signal, we have a maximum and minimum amplitude. We could have an infinite number of readings between these two limits, but we need to take a finite number of readings. Thus, we divide the range into zones, each with a particular height, .

The larger the value of , the smaller the value of and the more accurate the representation.

The value of each zone is given by its midpoint. Each sample falling into a zone is given the value of this midpoint.

After this, each zone is given a binary code. For zones, the number of bits needed is given by



The above diagram has values that are normalized before being quantized. This means, the values are divided by .

Thus, the value becomes , and since this falls into the zone between and , it is given the mid value, .The difference between the normalized value and the quantized value is the quantization error. is the fifth level, and thus its code is .

The quantization error can be reduced by using a smaller , which is the result of using more levels. However, this would mean more bits have to be used, which would increase the bitrate.

The quantization error can also introduce some noise. The relationship between the resolution and the quantization noise can be expressed using the signal to noise ratio (SNR).

where is the resolution in bits.

Alternatively, the SNR can be approximated as

The more bits we have, the higher the SNR. This means the signal is boosted while noise is suppressed.

The SNR is sometimes also represented as .

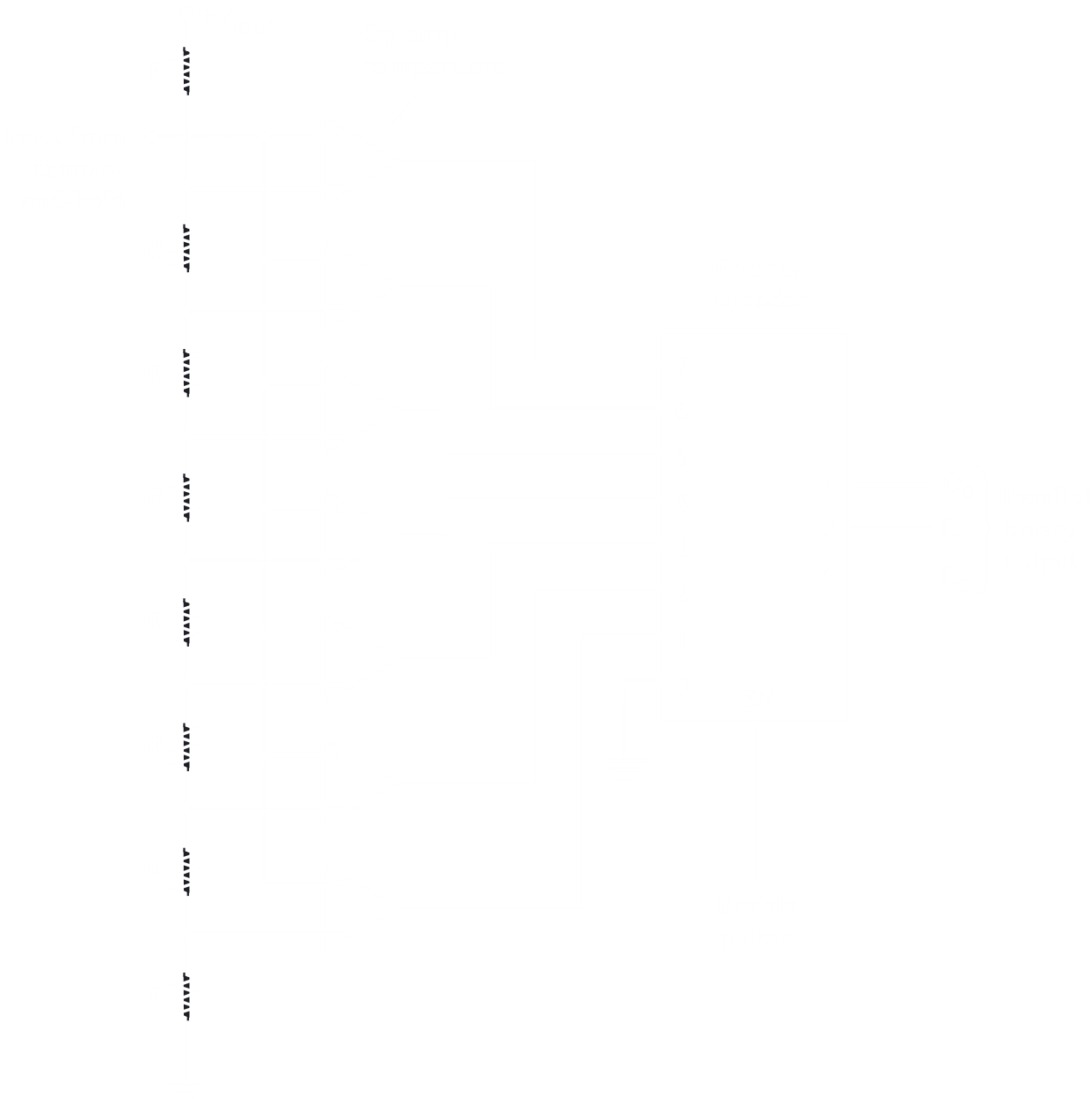
## Analogue to Digital Convertors

Analogue to digital converters (ADCs) can be classified into a few groups:

* Flash ADC
* Sigma-Delta ADC
* Duel Slope Converter
* Successive Approximation Converter

We will only be covering Flash ADCs and Successive Approximation Converters.

### Flash ADCs



Flash ADCs use multiple comparators as shown. An input signal is given to each of the comparators, as well as a reference voltage. This reference voltage decreases from the top to the bottom. Thus, the comparators at the top are for higher voltages, while the ones at the bottom are for lower voltages. Following this thinking, a priority encoder is also used. As a result, the highest voltage received is encoded.

Keep in mind that the input voltage is not a pure analogue signal, but one that has been converted from a digital signal, such as a signal from the sample and hold method.

The advantage of flash ADCs is that they are very fast. However, they have several disadvantages. A huge number of parts are needed for higher numbers of bits, since we need one comparator per bit. This makes it expensive and consumes a lot of power. Additionally, the resolution is lower.