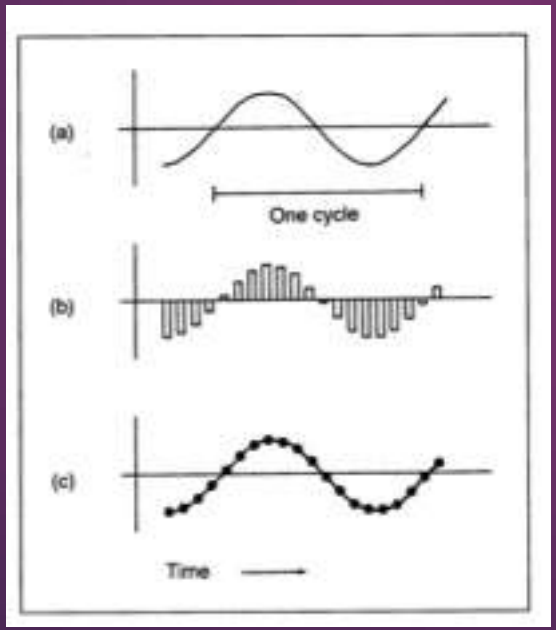
What is Sampling in Digital Audio

It is a digital representation of an analog signal

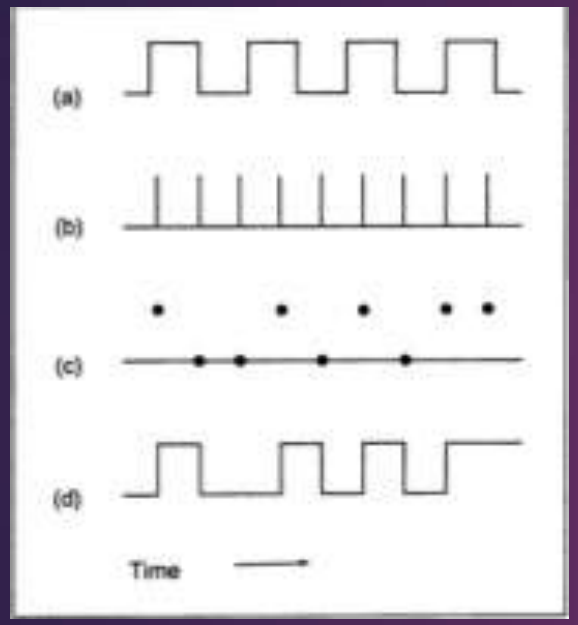
The sampling frequency is expressed in terms of samples per second or the sample rate

Sampling rate is expressed in Hertz Hz.

Check Figure b & c



Reconstruction of DA / Aliasing (Foldover)



If a sample of the waveform is taken at each point in Time in fig b, shown by vertical bars. Each bar creates a sample. Fig c shows the samples are stored as numbers in memory. When reconstructed in Fig d ,the result is different.

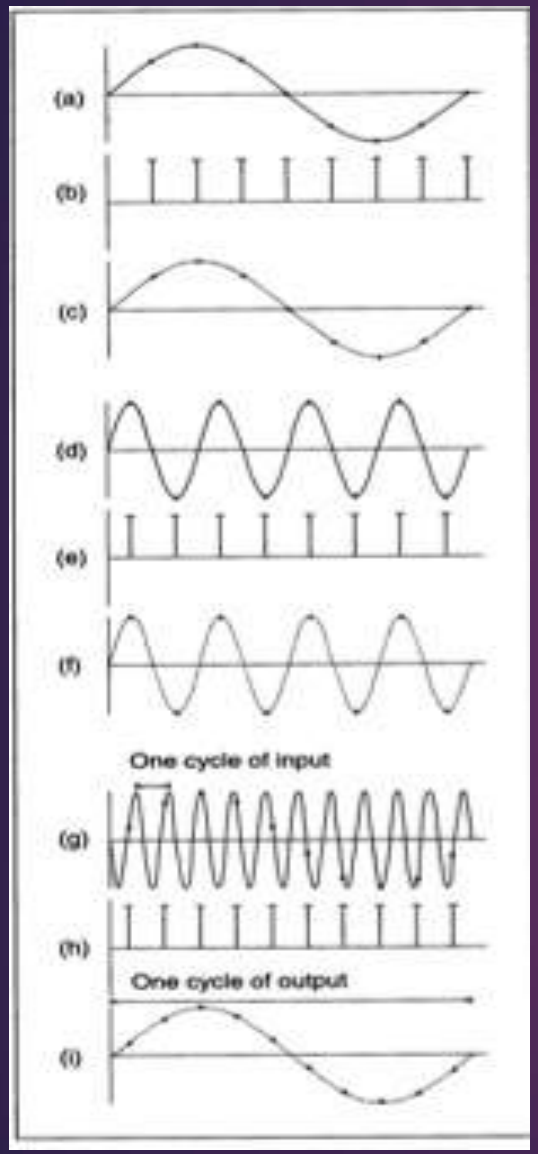


Fig a,d,g are original signal

Fig b,e,h are the samples per cycle of waveform

Fig c & f represent the reconstructed analog signal, but in Fig i, the

Resynthesized waveform is different from the original waveform. (in respect to wavelength)

This kind of distortion is called aliasing or fold over.

Nyquist Theorem or Sampling Theorem

The Nyquist Theorem, also known as the sampling theorem, is a principle that engineers follow in the digitization of analog signals.

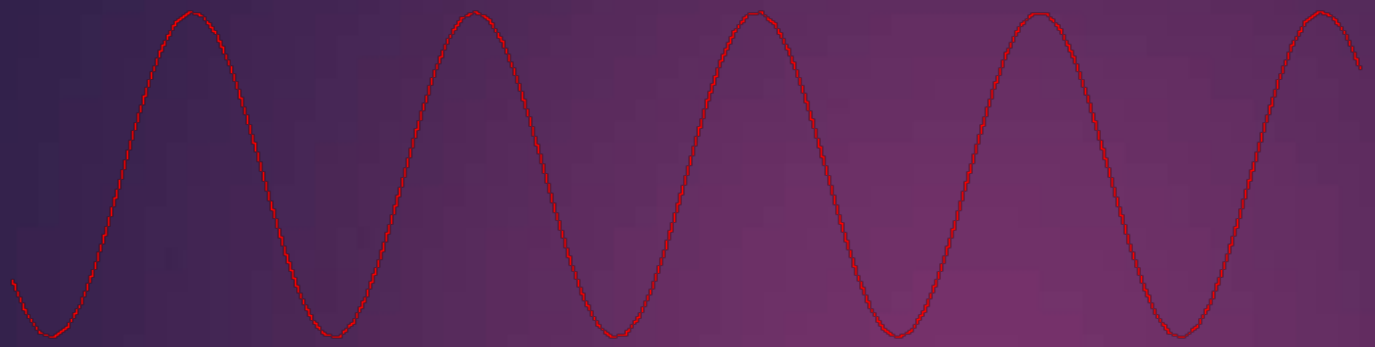
Suppose the highest frequency component, in hertz, for a given analog signal is *f*max.

According to the Nyquist Theorem,
the sampling rate must be at least 2*f*max, or twice the highest analog frequency component.

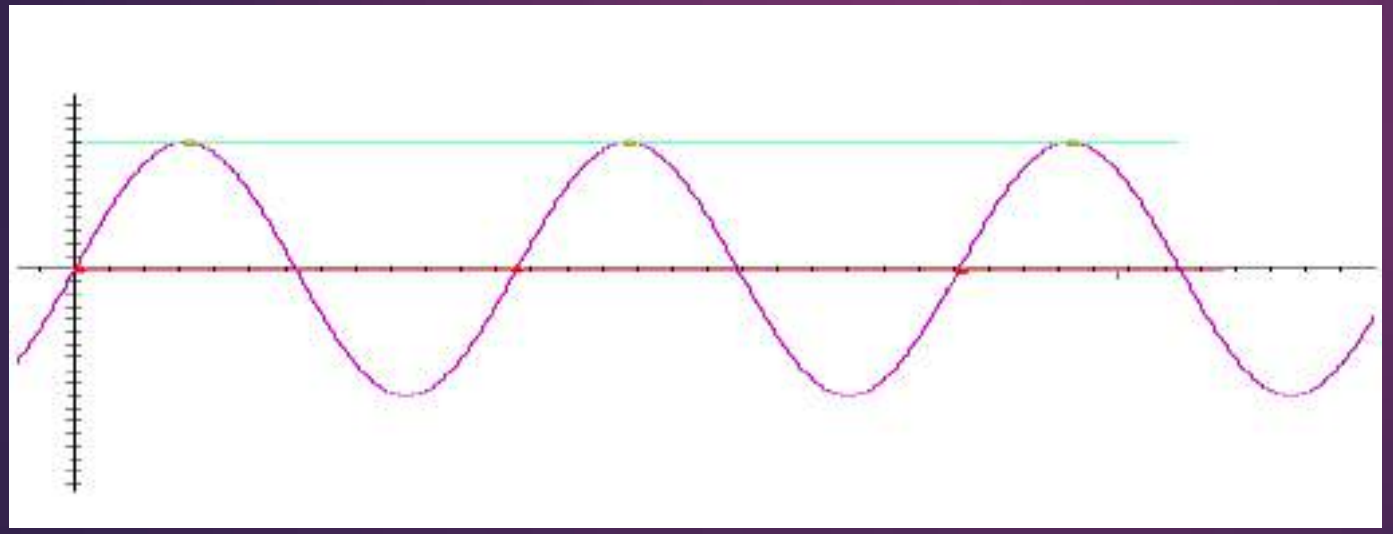
Sampling in an analog-to-digital converter is actuated by a pulse generator (clock). If the sampling rate is less than 2*f*max, some of the highest frequency components in the analog input signal will not be correctly represented in the digitized output.

This undesirable condition is a form of distortion called aliasing.

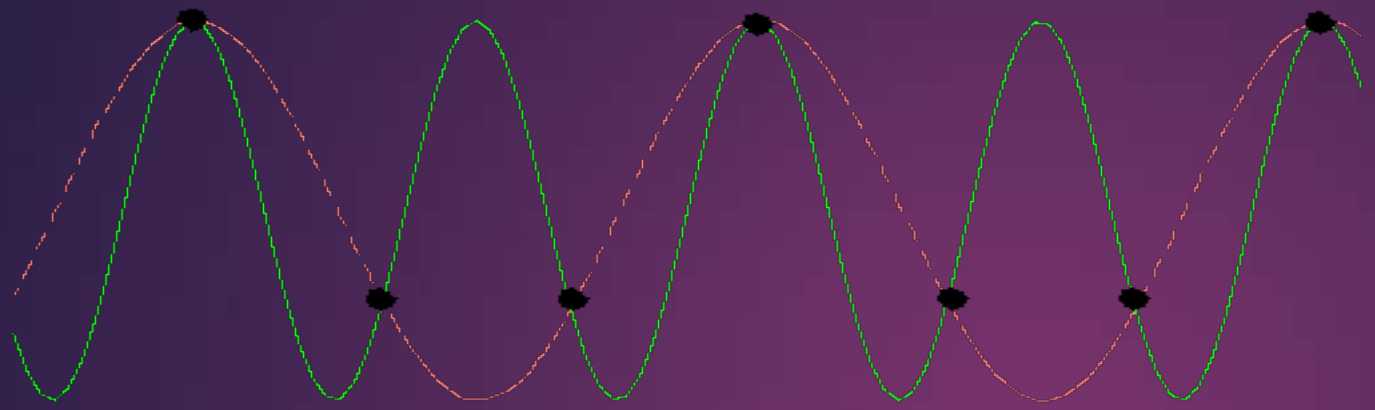
Suppose we are sampling a sine wave



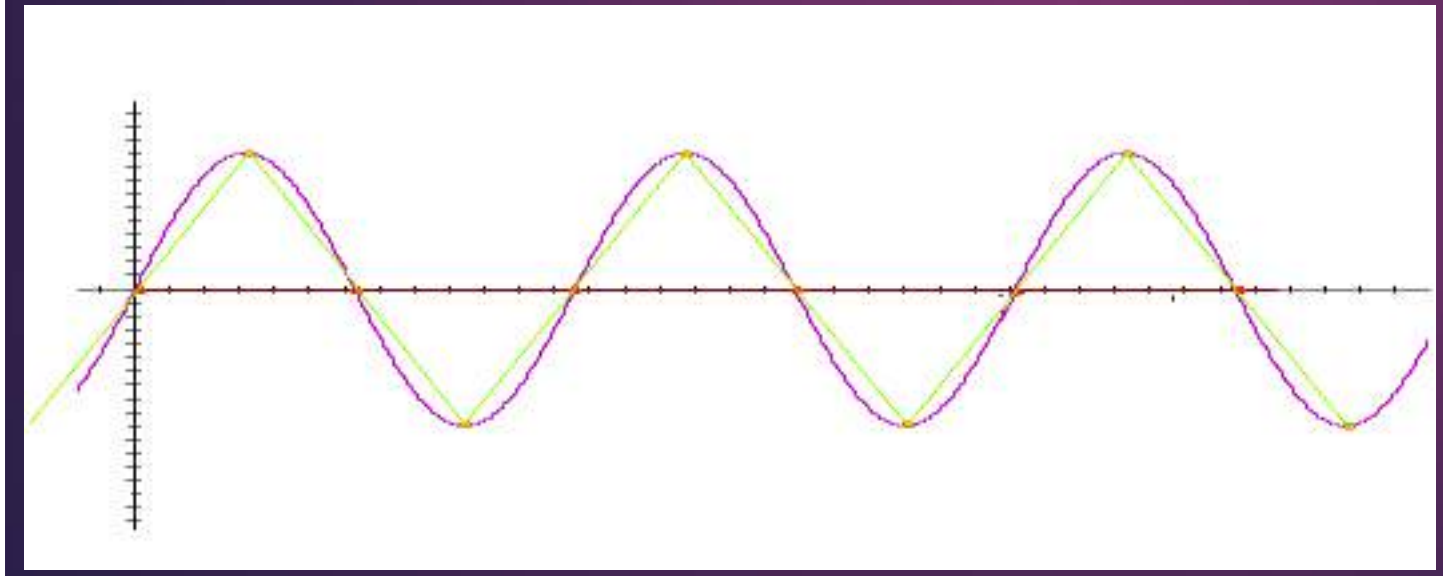
If we sample at 1 time per cycle, we can think it's a constant



If we sample at 1.5 times per cycle, we can think it's a lower frequency sine wave



Now if we sample at twice the sample frequency, i.e Nyquist Rate,



For lossless digitization, the sampling rate should be *at least twice* the maximum frequency responses

Dynamic range of Digital Audio systems

We can define the dynamic range as the difference between the loudest and softest sounds That the system can produce and is measured in units of Decibels.

It is a scale that indicates the ratio of one level to a reference level

db

So we keep the reference level as the threshold level of hearing

The decibel by itself doesn't measure volume, loudness or level.
A decibel is really just a ratio that allows you to compare the value of one unit with another using a logarithmic scale.

Here are some examples of different sound intensities as expressed in dB

180 dB: Rocket at take-off
140 dB: Jet engine at take-off
120 dB: Rock band
110 dB: Loud thunder
90 dB: City traffic
80 dB: Loud radio
60 dB: Ordinary conversation
30 dB: Soft whisper
0 dB: Softest sound a person can hear

Download Frequency Generator on your Mobiles