SAMPLING

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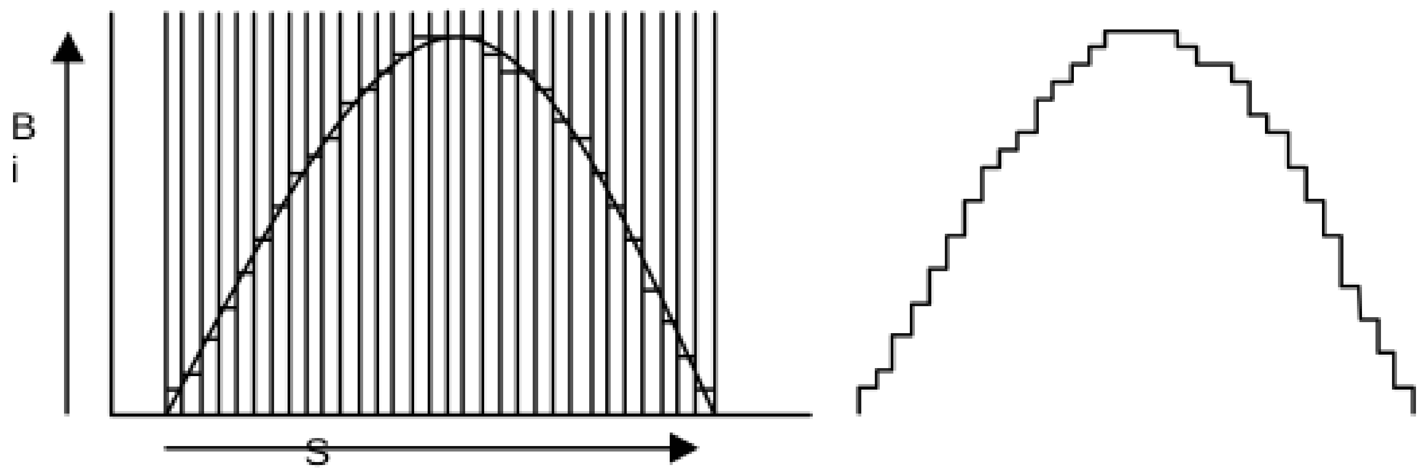
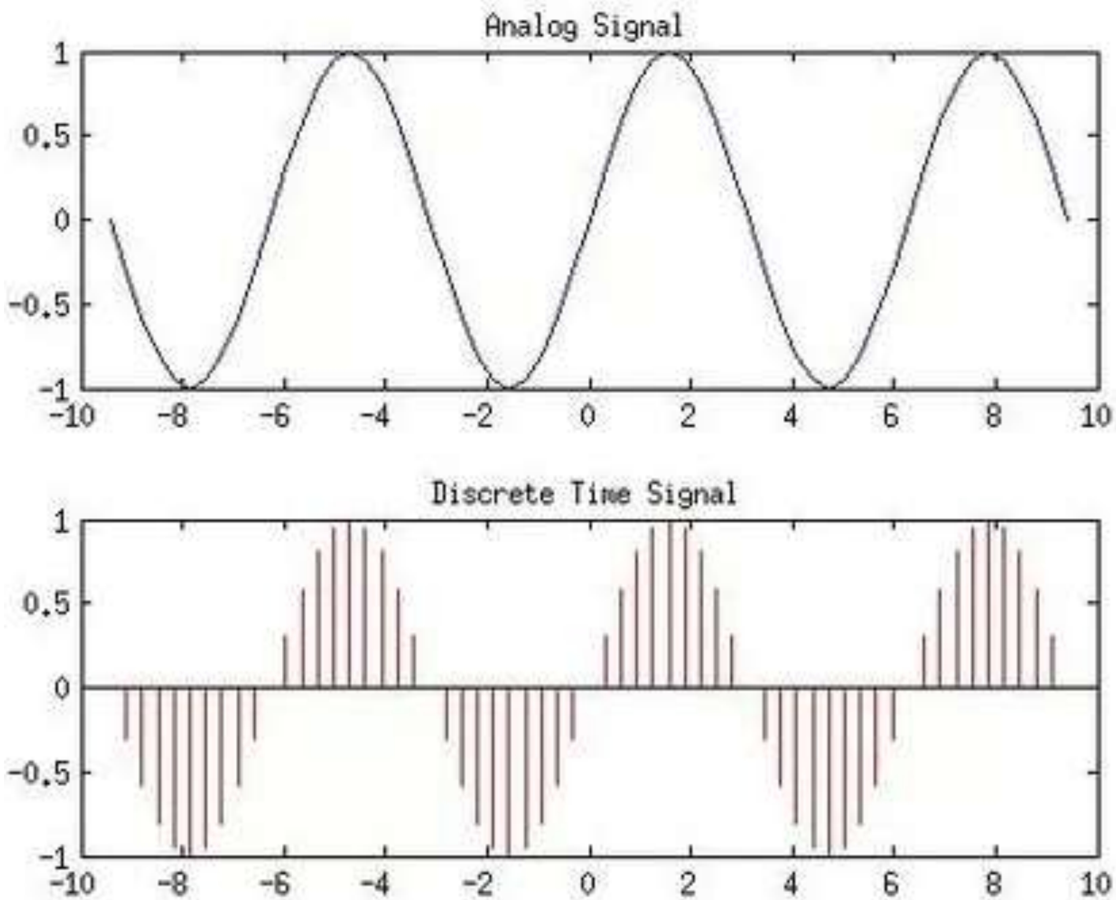
Sampling

- In the digital realm, recordings of analog waves are done through periodic sampling. That means when a sound wave is recorded, snap shots of the wave are taken at different instances.
- These snap shots are later examined and given a specific value (a binary number).
- This process is called discrete time sampling.

- The sampling rate of a digital system is defined as the number of snap shots or samples that are taken in one second.




Analog Signal
Discrete Time Signal
1
0.5
0
-0.5
-1
-10
-8
-6
-4
-2
0
2
4
6
8
10

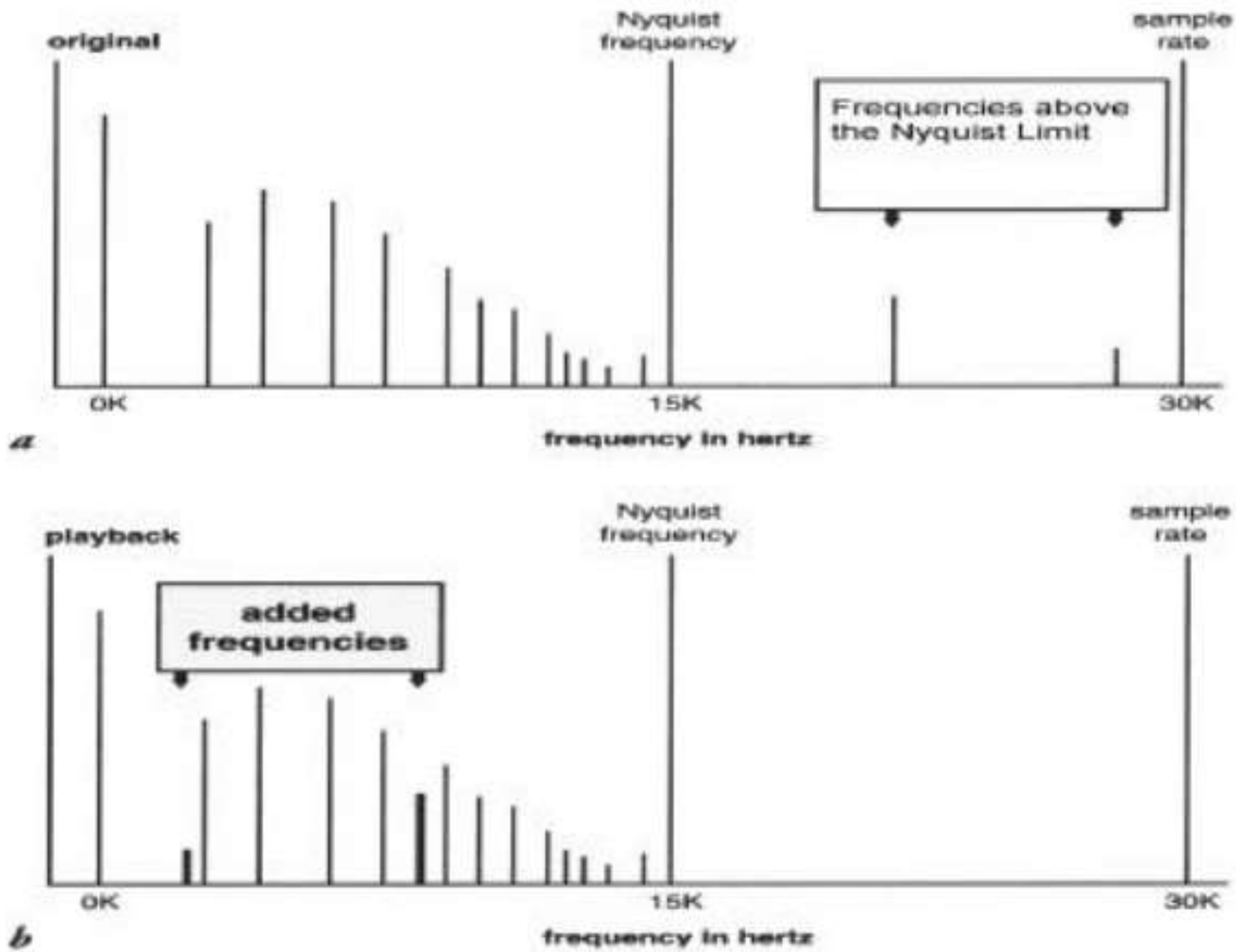
Nyquist Theorem

- According to the Nyquist theorem, S samples per second are needed to completely represent a waveform with a bandwidth of S/2 Hz.
- Therefore for a bandwidth of 20Hz-20kHz, one must use a sampling frequency of at least 40 kHz.

Aliasing

- Alias frequencies are produced when the input signal is not filtered effectively to remove all the frequencies above half of the sampling frequency.
- it is a necessity to send the recording signal through a low pass filter before the sampling circuit to act in accordance with the Nyquist Theorem, and avoid aliasing.

- When the input signal is not filtered, the sample continues to produce samples at a fixed rate, outputting a stream of false information caused by deviant high frequencies.
- his false information takes the form of new descending frequencies, which were not present in the original audio signal.
- These are called alias or fold over frequencies.


original
Nyquist frequency
sample rate
Frequencies above the Nyquist Limit
0K
15K
30K
a
frequency in hertz
playback
Nyquist frequency
sample rate
added frequencies
0K
15K
30K
b
frequency in hertz

Quantization

- Quantization is the measured amplitude of an analog signal at a discrete sample time.
- The accuracy of quantization is limited by the system's resolution, which is represented by the word length used to encode the signal, i.e. the number of bits in a signal such as 8,12, 16, 20, 24 bits.

Quantization Error

- This is the difference between actual analog value at the sample time and the chosen quantization intervals value i.e. the difference between the actual and measured values.
- Quantization error is limited to +- 1/2 interval at the sample time.
- At the system output, this error will be contained in the output signal. This error will sound like white noise that is heard together with the program material.

Dither

- Although quantization error occurs at a very low level, its presence must be considered in hi fidelity music. Particularly at low levels, the error becomes a measurable distortion.
- To fix quantization errors, a low-level noise called dither is added to the audio signal before the sampling process.
- Dither randomises the effect of quantization error. Dither removes the distortion of quantization error and replaces it with low-level white noise.

Pulse Code Modulation

- A method of encoding digital audio information which uses a carrier wave in the form of a stream of pulses which represents the digital data.
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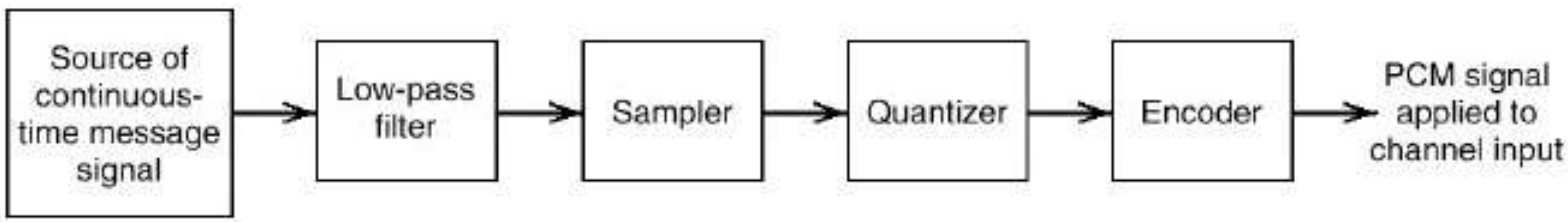
- Binary numbers are sent to the storage medium as a series of pulses representing amplitude. If two channels are to be sampled the PCM data may be multiplexed to form one data stream. The data is processed for error correction and stored. On playback the bit stream is decoded to recover back the original amplitude information at proper sample times and the analog waveform is reconstructed by the digital to analog converter (DAC).



(*a*) Transmitter

Linear PCM Recording

- Dither Generator
  - An analog noise signal is added to the analog signal coming from the line amplifier.
  - The dither causes the audio signal to constantly move between quantization levels.
  - The noise should resemble noise from analog systems, which is very easy on the ear.
  - Gaussian white noise is often used.

- Input Low Pass Filter
  - The analog signal is low-pass filtered by a very sharp cut -off filter to band limit the signal and its entire harmonic content to frequencies below half of the sampling frequency. The ideal LPF would have a "Brick wall" cut off, but this is very hard to achieve.
  - In professional recorders with a sampling frequency of 48kHz, the input filters are usually designed for 20Hz-20kHz.

- Sample and Hold
  - The S/H circuit time samples the analog waveform at a fixed periodic rate and holds the analog value until the A/DC outputs the corresponding digital word. Samples must be taken precisely at the correct time.
  - In audio digitization, time information is stored implicitly as samples taken at a fixed periodic rate, which is accomplished by the S/H circuit. An S/H circuit is essentially capacitor and a switch.
  - Maintaining absolute time throughout a digital system is essential. Variations in absolute timing called jitter can create modulation noise.

- Record processing
  - After conversion several operations must take place prior to storage:
  - Multiplexing - Digital audio channel data is processed in a single stream. However the A/DC outputs parallel data, i.e. entire words. The multiplexer converts this parallel data to serial data.
  - Data coding - Raw channel is properly encoded to facilitate storage and later recovery. Several types of coding are applied to modify or supplement the original data.
  - A synchronisation code is a fixed pattern of bits provided to identify the beginning of each word as it occurs in the bit stream. Address codes are added to identify location of data in the recording. Other specifications such as sampling frequency, table of contents, copyright information, even Time Code can be added.

- Analog to Digital conversion
  - This is the most critical component of the entire system. The circuit must determine which quantization increment is closest to the analog waveform's current value, and output a binary number specifying that level.
  - This is done in less than 20 microseconds.
  - In a 16-bit linear PCM system each of the 65 536 increments must be evenly spaced throughout the amplitude range so that even the LSBs in the resulting
  - word are meaningful.
  - Thus the speed and accuracy are the key requirements for an A/D converter.

Digital to Analog Conversion

- The D/AC is the most critical element in the reproduction system - determining how accurately the digitized signal will be restored to the analog domain.
- A DAC accepts input digital word and converts it into an output analog voltage or current.

- Output sample and hold
  - When the DAC switches from one output voltage to another, false voltage variations such as switching glitches can occur which will produce audible distortion. The output circuit acquires a voltage from the DAC only when the circuit has reached a stable output condition.
  - The S/H circuit holds correct voltage during the intervals when the DAC switches from samples. Hence false glitches are avoided by the S/H circuitry.

- It operates like a gate removing false voltages from the analog stream and like a timing buffer, re-clocking the precise flow of voltages.

- Its output is a precise "staircase" analog signal, which resembles the output of its counterpart in the recording conversion.