

# Sampling & Quantization in Digital Audio Systems

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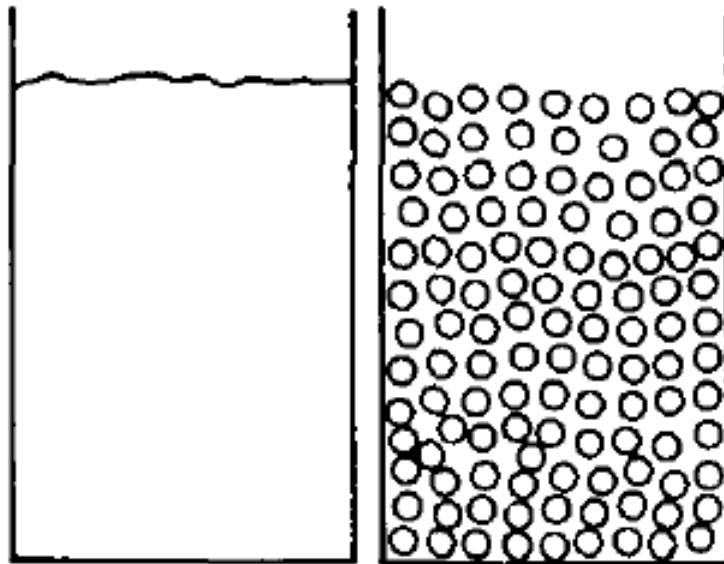
## Digital Audio

- Combines
  - Sampling
  - Quantization

## Quantization

- Turning a level into a number

Imagine you have two water glasses. One is filled nearly to the top with water. The other is filled to the same level with marbles.



Analog

Quantized

The glass with water is analog and can have any amount of water in it. The amount can vary over the range from empty to full in a continuous way. You can add or remove any fraction of a glass full of

water. However, water can spill and/or evaporate out of the glass, so the exact amount of water may not be a constant. Vibrations can disturb the surface of the water and make it difficult to determine the exact amount of water with accuracy.

The glass with marbles is quantized, where each marble represents a certain amount of water. Marbles can only be added and removed in discrete steps of one marble. You can't have a fraction of a marble, it is all or nothing. Therefore, the number of marbles have a quantization error when compared with the amount of water in the water glass. It is hard to lose a marble, the amount of marbles is robust. The number of marbles can be counted, and the value is always an integer. While it is possible to make an error in counting the marbles, this is far less likely than being able to measure precisely the amount of water in the water glass.

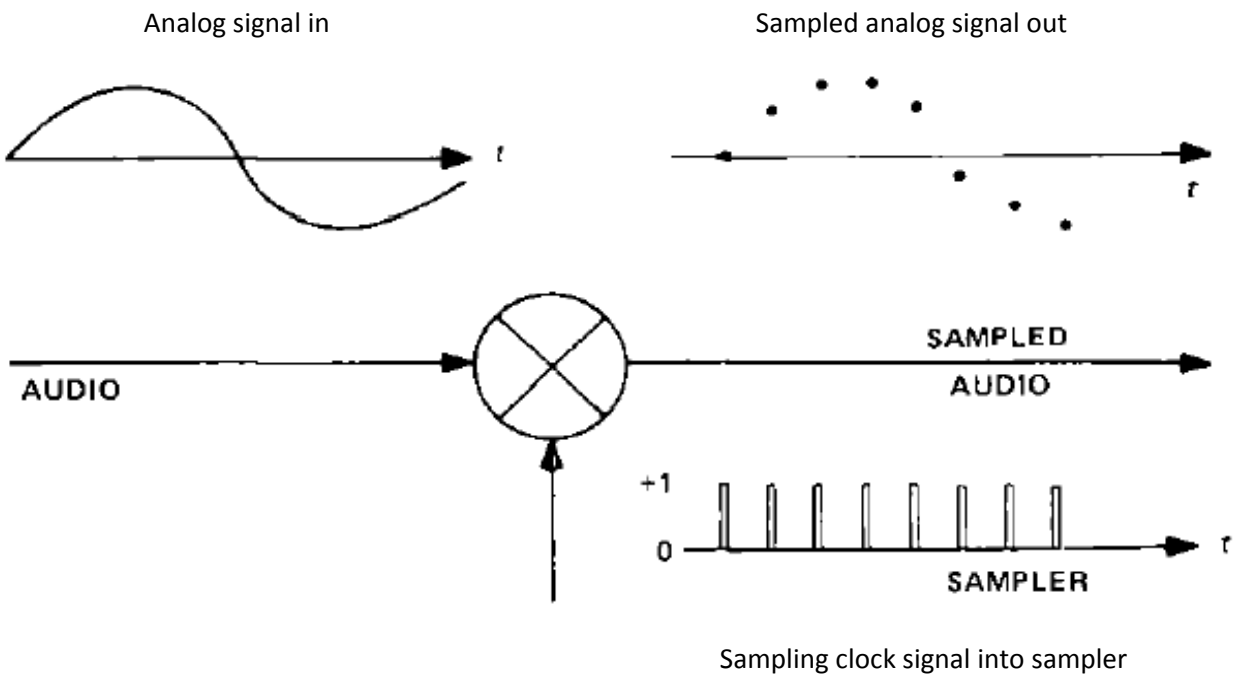
Quantization errors (distortions) can be minimized by using a small quantization step size (smaller marbles). Each time the number of bits used to represent a digital audio level increases by one (say 12 bits to 13 bits) the quantization step size is reduced in half.

Quantization errors (distortions) can be eliminated by the proper use of "dither". Dither is adding a small amount of random noise to the signal **BEFORE** it is quantized. The amount of dither added is important and will depend on the quantization step size. If too little dither is added the distortions will not be totally eliminated. If too much dither is added the distortions will be eliminated but there will be excessive noise. Please note that dither actually eliminates the quantization distortions, and does not merely mask them.

If dither is added to the signal **AFTER** it is quantized, no reduction in quantization distortion happens. Adding dither after quantization only results in added noise.

# Sampling

- Taking representative samples
- At uniform time intervals



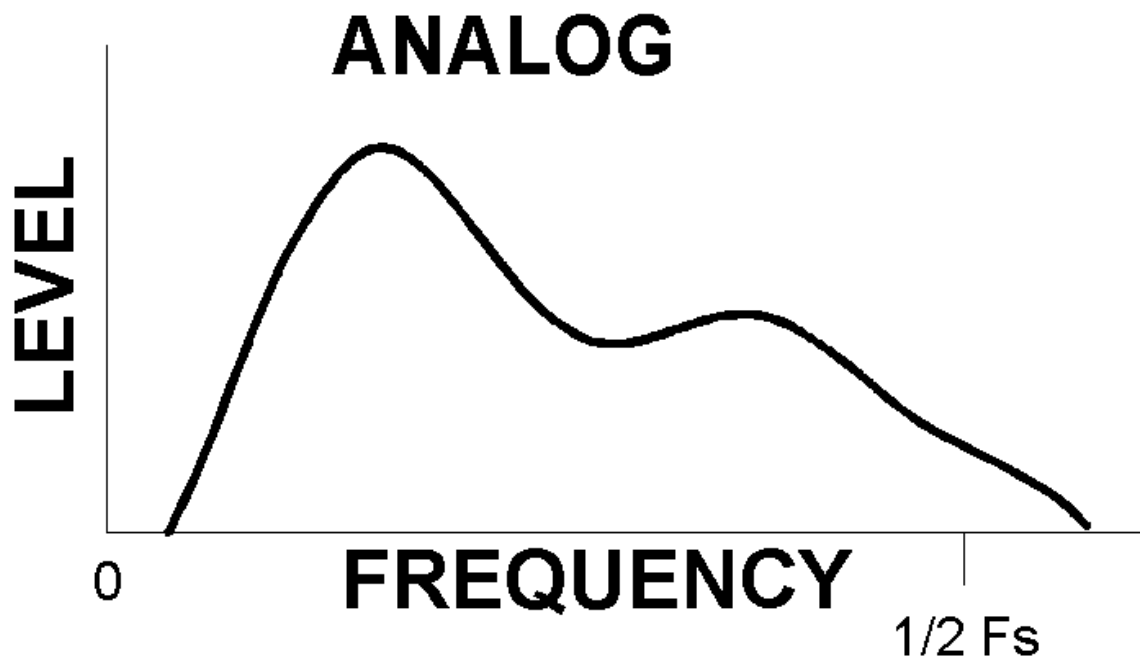
The continuous analog signal comes into the sampler. The output of the sampler is a series of analog values taken at uniform time intervals. The series of pulses at the bottom right is the sample commands to the sampler (circle with the X in the middle).

Sampling can be done independent of quantization. The output of a sampler is still analog. One example of sampled but not quantized audio is a bucket brigade (charge coupled device) signal delay. These were common before digital audio became cost effective. They were used for simple signal delays, pitch sifters, and flangers.

# Nyquist's Theorem

- Uniformly taken samples contain all the information in the original signal if:
  - There is no information in the original signal above  $\frac{1}{2}$  the sampling rate
- $\frac{1}{2}$  the sampling rate is called the Nyquist frequency

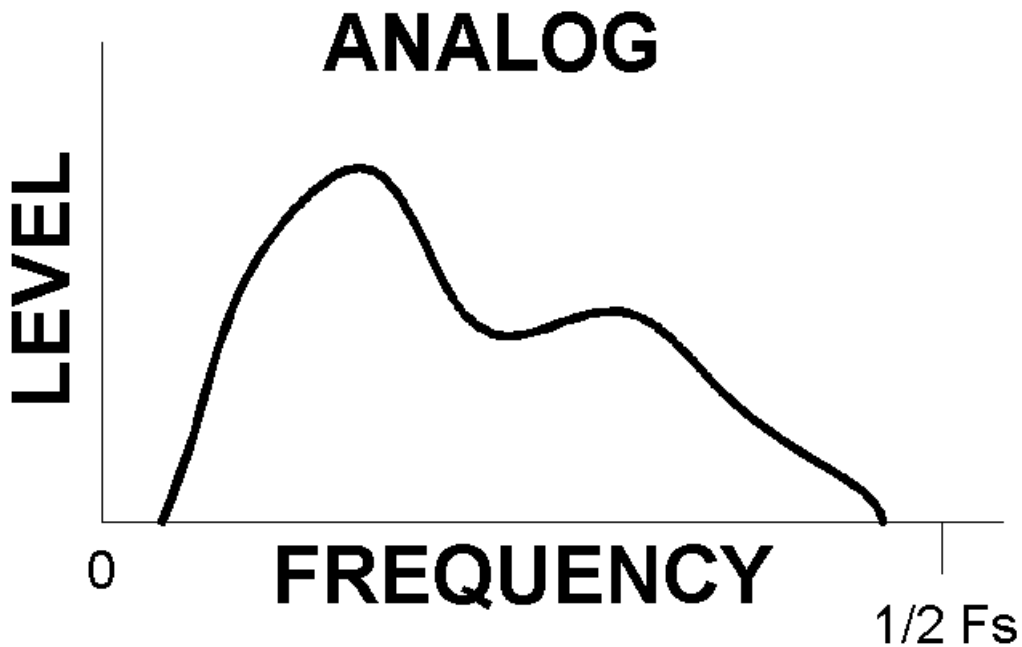
Analog signal **NOT** frequency limited



This is a frequency spectrum. Level on the vertical axis, and frequency on the horizontal axis.

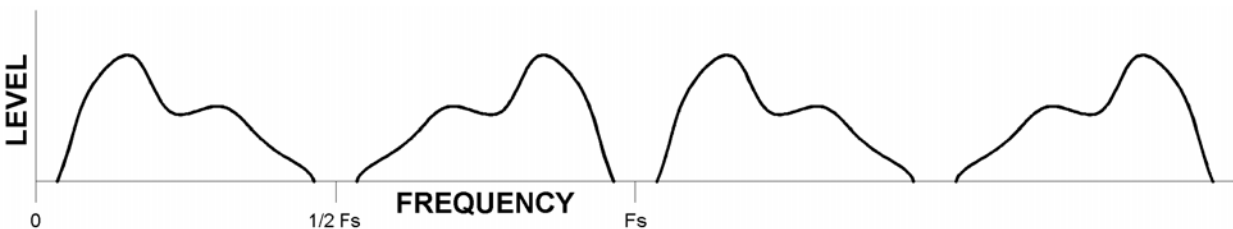
Note the spectrum extends past the Nyquist frequency.

Analog signal frequency limited



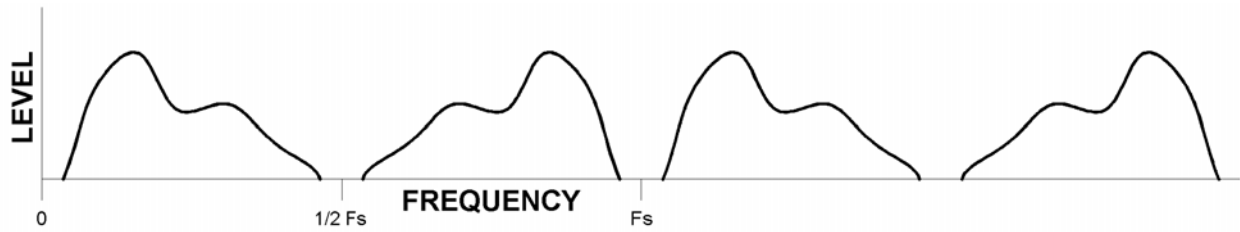
Note that since the extreme highs have been filtered out, there is no longer any energy above the Nyquist frequency.

- Sampled frequency limited signal



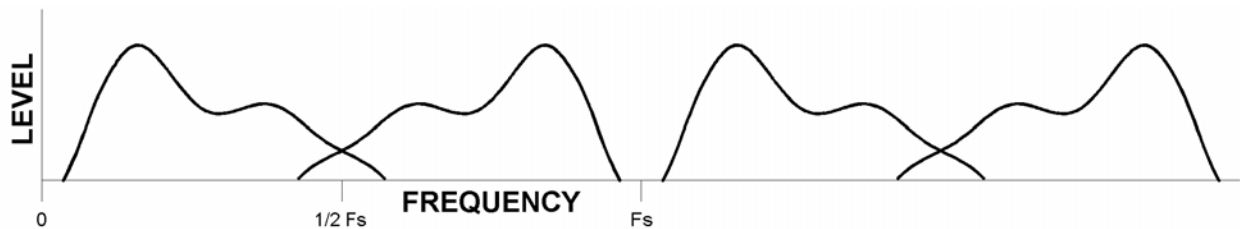
When a signal is sampled, the spectrum of the sampled signal contains the original signal, plus at higher frequencies copies (images) of the original signal. Between the Nyquist frequency and the sampling rate is a frequency shifted (original frequency plus the Nyquist frequency) and frequency inverted copy of the original signal – the highest frequencies of the original are lower in frequency and the lowest frequencies of the original are now highest in frequency. Above the sampling rate is found a frequency shifted (original frequency plus the sampling rate frequency) copy of the original signal, followed by a shifted and inverted copy. The drawing only shows the original plus three copies, but the images continue out to infinitely high frequencies.

- Sampled frequency limited signal
  - Original signal can be recovered by **filtering**



Since there is no overlap of the frequencies, the higher frequency copies of the original can be removed by filtering.

- Sampled **not** frequency limited signal
  - Original signal **can't be recovered**
  - This is called **aliasing**



Note overlap of frequency inverted image with the spectrum of the original signal. Since the 1<sup>st</sup> inverted copy overlaps the original signal, they can't be separated by filtering. The portion of the 1<sup>st</sup> image that is below the Nyquist frequency is known as aliasing distortion.

The audibility of aliasing is highly dependent on the program material. In general material with greater amounts of high frequency energy will have more audible aliasing.

# Digital Audio

- To recover the original signal we must:
  - Frequency limit the incoming signal
    - Anti-Aliasing filter
  - Filter out the unwanted images
    - Reconstruction filter
- In a properly designed digital audio system, all the quality losses take place in the initial Analog to Digital conversion. Modern A/D converters combine:
  - Anti-aliasing filters
  - Sampling
  - Quantization
- In a single chip