Planet of-tunes

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digital theory

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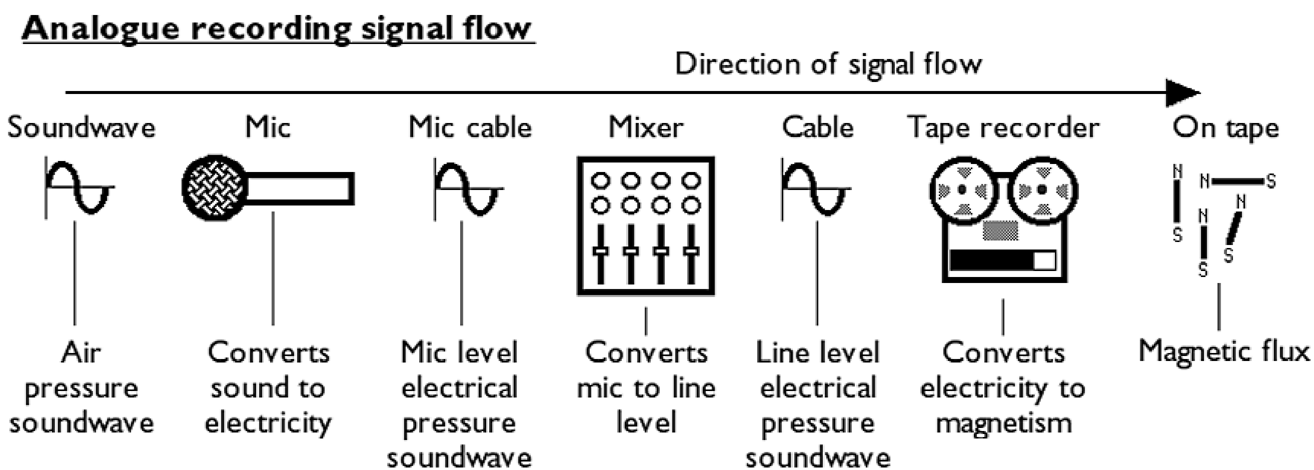
Analogue recording signal flow

Conventional analogue recording involves capturing sound with a microphone, converting it into an electrical soundwave signal, and then recording it onto magnetic tape by converting it into magnetic flux.



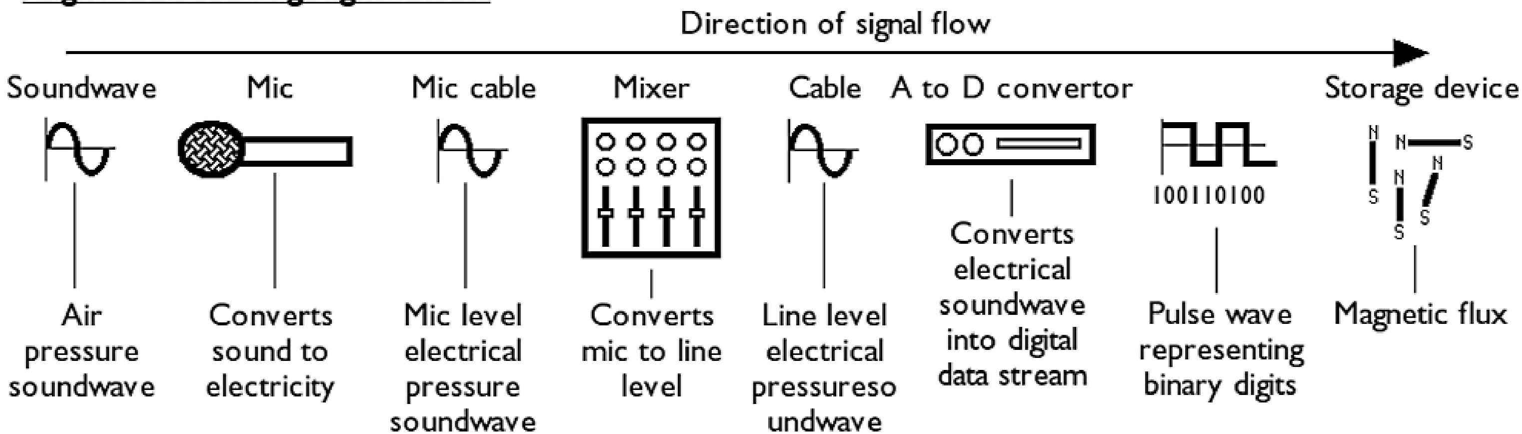
This process has several drawbacks ...

Tape recorders introduce distortion and noise into the signal.
Tape recorders are mechanically complex and expensive to buy and maintain.
Once recorded the soundwave is difficult to edit and manipulate.
Once recorded the tape must be wound back and forth to locate a particular part of the recording.
Recordings cannot be backed up without a loss of quality.

Digital recording signal flow

Converting analogue soundwaves into digital data streams has several advantages for the sound recordist ...

Digital recording signal flow



Once recorded, tape distortion and noise will (almost) never affect the reproduction sound quality.
With non-professional equipment high quality sound is possible.
Digital processors and recorders offers more powerful and flexible creative tools than their analogue counterparts.
Data can be stored, edited and replayed from a hard disk offering non-linear operation.
Data can be cloned and archived.
Data can be recorded to a variety of storage devices including RAM,

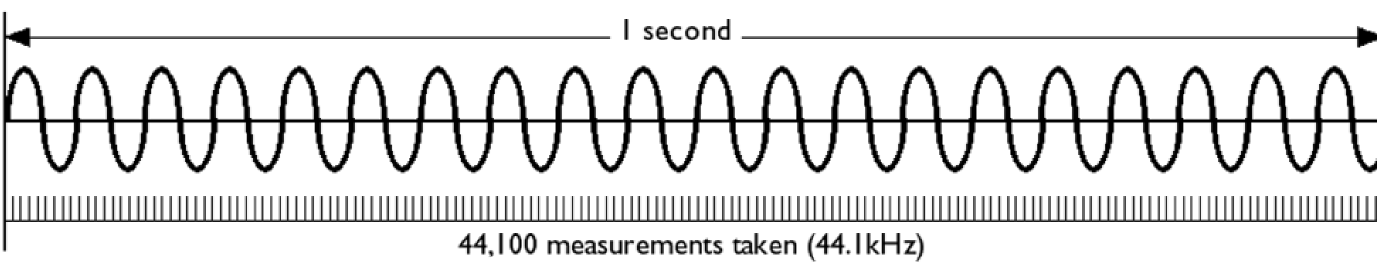
tape, hard disk, and optical media.
Data can be easily transmitted without quality loss.
Sound can be incorporated into multimedia productions.

Analogue to digital conversion

An A to D convertor is a device for digitising an analogue electrical soundwave. This is achieved by measuring the amplitude of the source waveform at regular intervals and recording these measurements (also known as Samples or Snapshots) as a series of 16-bit numbers.

Note: Incidentally, a D to A (Digital to Analogue) convertor is a device for converting the digital data back to an analogue waveform in order to amplify, mix or further process in some way.

Sample Rate (or Frequency)

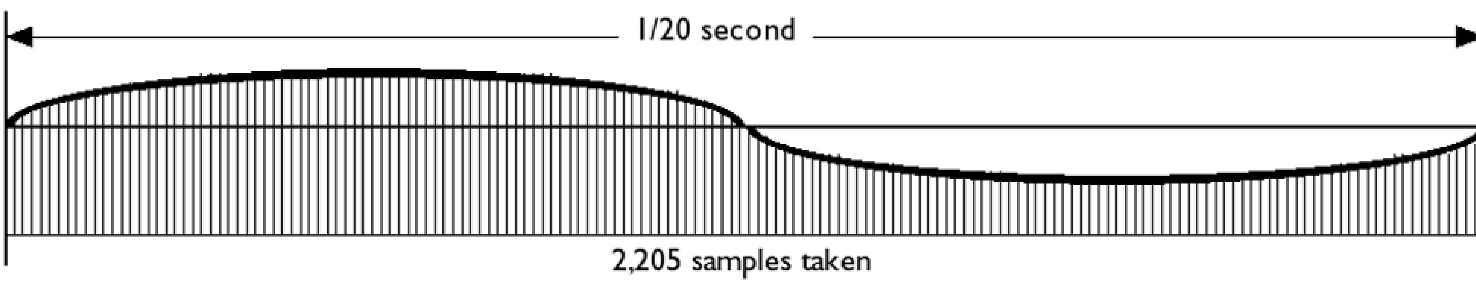


Sample rate (or frequency) The number of times the source waveform is measured every second is known as the Sample Rate. CD quality audio has a sample rate of 44,100 per second (44.1kHz).

The sample rate needs to be as high as practically possible to ensure as little sound distortion as possible. Although the sample rate never changes the quality of the digitised signal partly depends on the frequency of the soundwave being recorded. Because the human audio spectrum is from 20Hz to 20,000Hz (20kHz) the convertor must be able to record all these frequencies well.

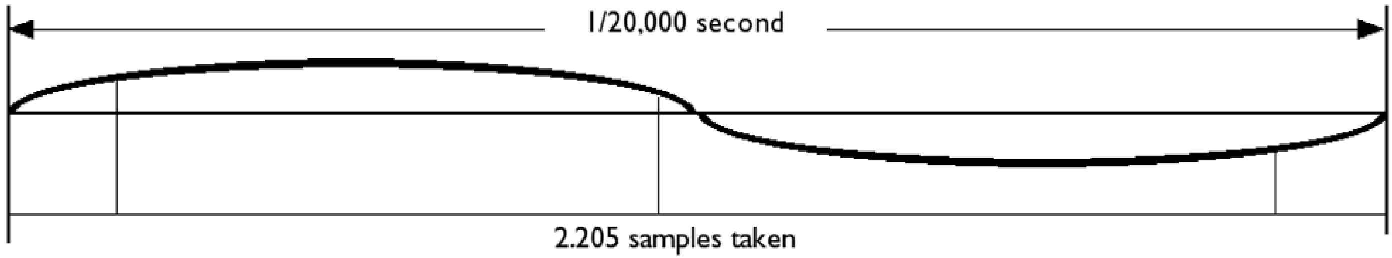
Consider a soundwave at a low frequency of 20Hz. There will be 20 cycles of its waveform every second. This means each cycle will be represented by 2,205 samples. 44,100 divided by 20 = 2,205. So soundwaves at low frequencies are measured thoroughly and the shape of each cycle of the waveform recorded accurately.

One cycle of a soundwave at 20Hz (cps)



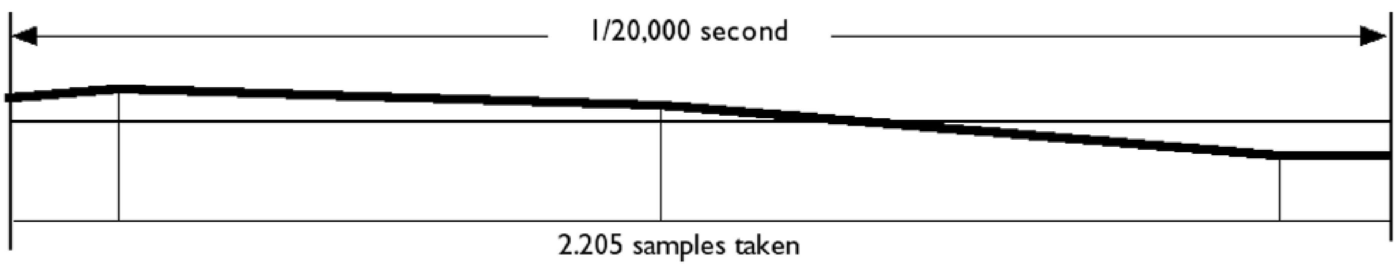
Now consider a soundwave at a high frequency of 20,000Hz (20kHz). There will be 20,000 cycles of its waveform every second. This means each cycle will be represented by 2.205 samples. 44,100 divided by 20,000 = 2.205. So soundwaves at high frequencies are measured badly and the shape of each cycle of the waveform recorded poorly.

One cycle of a soundwave at 20,000Hz (cps) / 20kHz



When the digital data is converted back into an an analogue electrical soundwave, by a D to A convertor, the wave will look different to the source soundwave which originally entered the A to D convertor during recording. This difference between the source and reproduced signal is called distortion.

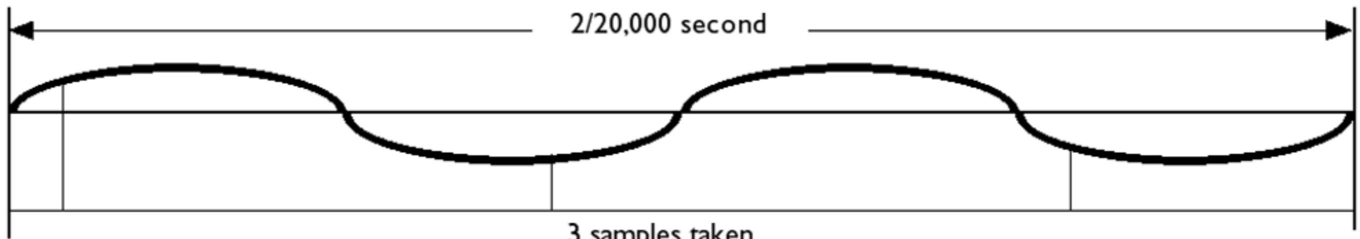
One cycle of a soundwave at 20,000Hz following D to A conversion



Aliasing noise Were the sample rate to be any lower than 44.1kHz for these high frequency soundwaves, then the soundwave produced by the D to A conversion process might look, and thus sound, wildly different to the source soundwave. In fact it would appear as a lower frequency wave and be audible as random noise. This noise is known as Aliasing noise.

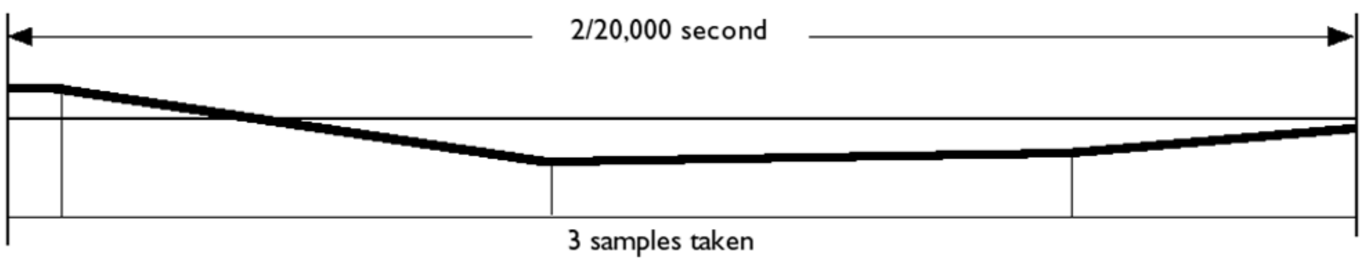
Consider a soundwave at 20,000Hz being recorded at a sample rate of 32kHz. this would mean 1.5 samples per cycle of the waveform. The following diagrams shows 2 cycles and thus 3 samples.

Two cycle of a soundwave at 20,000Hz (cps) / 20kHz



Now look at the soundwave reconstructed by the D to A convertor. The wave shape has changed and the sound has been distorted.

Two cycles of a soundwave at 20,000Hz following D to A conversion



Nyquist theory The sample rate of 44.1kHz was arrived at because it represents a good compromise between the amount of data created (all of which needs to be processed and stored) and sound quality. Should the sample rate be lower distortion would become noticeable and manifest itself as a grainy sound and aliasing noise.

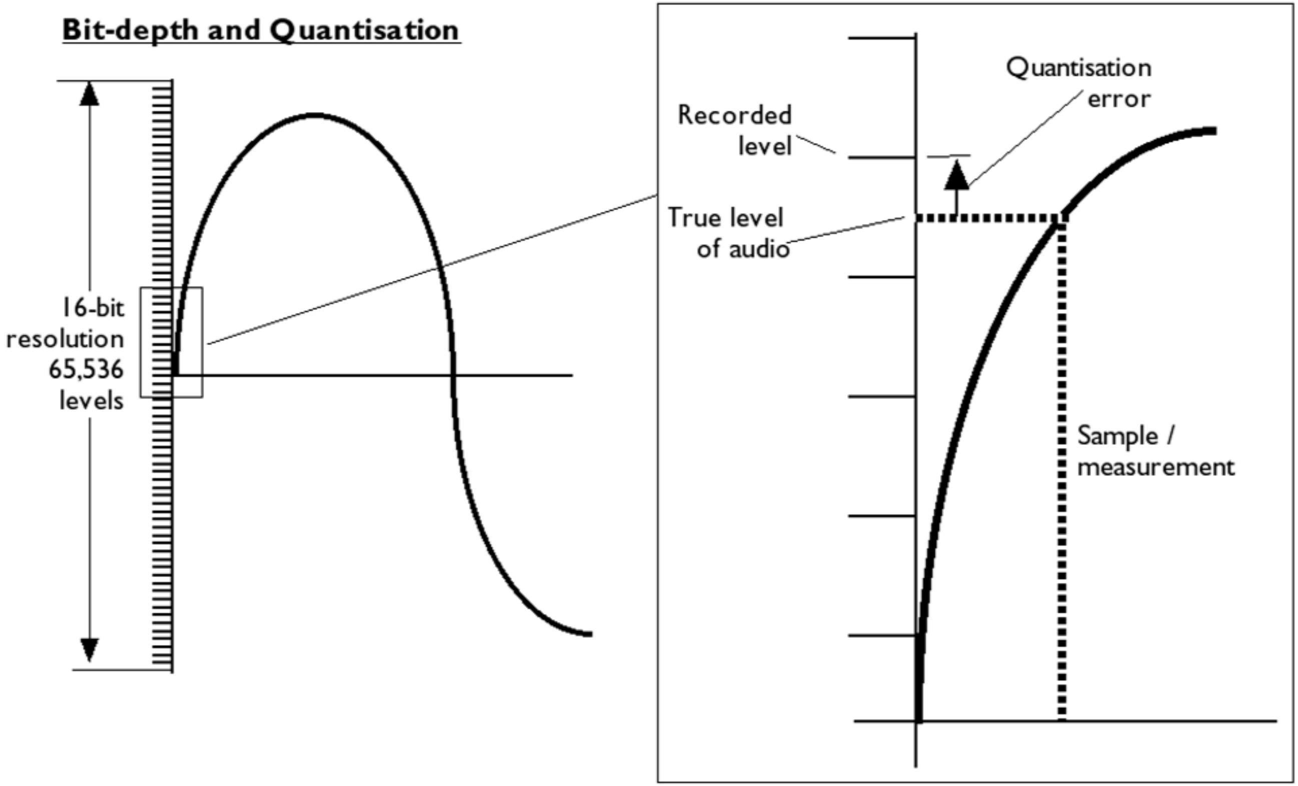
Scientist Harry Nyquist's theory states that: The sample rate should be a little over twice the highest audio frequency to be recorded. Thus a frequency range of 20Hz to 20kHz requires a sample rate of 44,100 per second, or 44.1kHz.

Anti-aliasing filter Should any unheard source soundwaves above 20kHz enter the A to D convertor during recording they would manifest themselves as aliasing noise after D to A conversion. To prevent this happening frequencies above 20kHz are removed by a low pass filter as they enter the A to D convertor.

Bit-depth

The accuracy to which each sample measurement is made is a function of the bit-depth of the system. Taking a sample involves measuring the amplitude of the waveform at a moment in time. 16-bit systems offer 65,536 possible levels (or points of resolution) with which to measure this amplitude.

Quantisation Whenever a measurement is taken and the true amplitude of the waveform falls between two points of resolution a process of rounding up or down occurs, known as Quantisation. Quantisation produces so-called Quantisation Errors which manifest themselves as distortion and noise in the sound quality. This is because the true shape of the waveform is being distorted slightly by the measurement process. Of course sometimes a measurement falls exactly on a point of resolution and no quantisation is required. For these samples there will be no distortion to the soundwave.



Digital sound devices

Digital recording devices have A to D convertors to digitise sound, storage devices to store data, and D to A convertors to replay it. Such devices include ...

Samplers. (Store data temporarily in RAM and permanently on disk drives).
PCI audio soundcard (Store data on the computers hard drive).
Digital mixers (Don't store data).
Digital effects processors (Store data in RAM temporarily).
DAT (Digital Audio Tape) recorders (Store data on tape).
CD-R recorders (Store data on optical discs).
Professional digital multitrack recorders (Store data on tape).
Mini Disc recorders (Store data on diskettes).

Data / file types

Uncompressed CD quality audio requires 10Mb of storage per stereo minute, 5Mb per mono minute. The leading file types are ...

AIFF (Audio Interchange File Format).
Wave, Microsoft's format for PCs.
Sound Designer 2 (SDII), a stereo mastering format instigated by Digidesign, designers of the leading professional non-linear audio software (Sound Designer and Pro Tools).

Some of these file types come in several formats ...

Mono.
Split stereo. Two mono files are linked together but editable individually.
Stereo Interleaved. Two mono files are linked together and cannot be edited individually.