Digital Audio concepts, Sampling Variables, Loss less compression of sound, loss, compression & silence compression, Video representation, Colors, Video, Compression, MPEG standards, MPEG Standard Video Streaming on net, Video Conferencing, Multimedia Broadcast Services, Indexing and retrieval of Video Database, recent development in Multimedia.

Basic Digital Audio Concepts

To distribute recorded speech or music over the Internet, an analog signal must be converted to digital information (described by bits and bytes). This process is called encoding . It is analogous to scanning a photograph to a digital bitmap format, and many of the same concepts regarding quality and file size apply. Some audio file formats (such as MPEG) are compressed in size during encoding using a specialized audio compression algorithm to save disk space. In the encoding process, you may be asked to provide settings for the following aspects of the audio file.

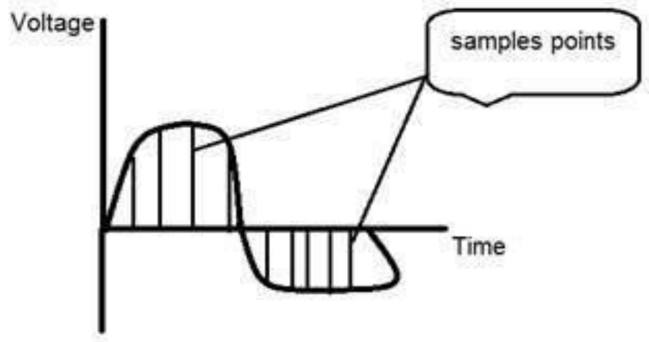
Sampling rate

To convert an analog sound wave into a digital description of that wave, samples of the wave are taken at timed intervals (see Figure 33-1). The number of samples taken per second is called the sampling rate. The more samples taken per second, the more accurately the digital description can recreate the original shape of the sound wave, and therefore the better the quality of the digital audio. In this respect, sampling rate is similar to image resolution for digital images.

Sample rates are typically measured in kilohertz (kHz). On the high end, CD-quality audio has a sampling rate of 44.1 kHz (or 44,100 samples per second). On the low end, 8 kHz produces a grainy sound quality that is equivalent to a transistor radio. Standard sampling rates include 8 kHz, 11.025 kHz, 11.127 kHz, 22.05 kHz, 44.1 kHz, and 48 kHz. The high-end standard is 96K, which may be seen in DVD.

Sampling

The term sampling refers to take samples
We digitize x axis in sampling
It is done on independent variable
In case of equation y = sin(x), it is done on x variable
It is further divided into two parts , up sampling and down sampling



If you will look at the above figure, you will see that there are some random variations in the signal. These variations are due to noise. In sampling we reduce this noise by taking samples. It is obvious that more samples we take, the quality of the image would be more better, the noise would be more removed and same happens vice versa.
However, if you take sampling on the x axis, the signal is not converted to digital format, unless you take sampling of the y-axis too which is known as quantization. The more samples eventually means you are collecting more data, and in case of image, it means more pixels.
Variables sampling is the process used to predict the value of a specific variablewithin a population. For example, a limited sample size can be used to compute the average account receivable balance, as well as a statistical derivation of the plus or minus range of the total receivables value that is under review.

Loss less compression of sound

A lossless compressed format stores data in less space without losing any information. The original, uncompressed data can be recreated from the compressed version.

Uncompressed audio formats encode both sound and silence with the same number of bits per unit of time. Encoding an uncompressed minute of absolute silence produces a file of the same size as encoding an uncompressed minute of music. In a lossless compressed format, however, the music would occupy a smaller file than an uncompressed format and the silence would take up almost no space at all.

Lossless audio compression’s goal is to reduce file size while leaving the original audio untouched. These codecs don’t use any of the permanent compression techniques above, focusing instead on fully-reversible data compression methods. They use lossless compression techniques borrowed from file-compression algorithms like ZIP to remove redundant data while preserving the integrity of the underlying information. Two popular lossless audio codecs – FLAC and Apple Lossless (ALAC) – both use schemes based on ZIP compression.

Silence compression

Silence compression provides a way to squeeze redundancy out of sound files. The silence compression scheme is essential for efficient voice communication systems. It allows significant reduction of transmission bandwidth during a period of silence.

A silence compression scheme includes a voice activity detection (VAD), a silence insertion descriptor (SID) and a comfort noise generator (CNG) module.

Parameters used for silence compression

- Threshold value
- The way to encode silence
- Threshold for recognizing start of silence
- An indication of when silence is over
- A parameter to get a threshold, which means that there is no silence until or unless there is three silence in the rows.