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STUDY OF SAMPLING IN DSP

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Keywords: Signal ,Signal Processing ,Continues Signal, Digital Signal, Sample, Sampler, Digital Signal Processing(DSP), Nyquist Rate, Sample Rate, Digitization ,High Sampling Rate(Oversampled), Low Sampling Rate(Under sampled), Aliasing, Nyquist–Shannon Sampling Theorem (Nyquist–Shannon–Kotelnikov,Whittaker–Shannon–Kotelnikov, Whittaker–Nyquist–Kotelnikov–Shannon, as well as the cardinal theorem of interpolation theory), Thresholds, Nyquist Frequency, Nyquist Criterion(Raabe Condition), Saw tooth wave(extreme case of asymmetric Triangle Wave), Pulse Code Modulation, Analog-to-Digital conversion (ADC), Digital-to-Analog conversion (DAC).

ABSTRACT
In Digital Signal Processing, Analog Signal must be converted to digital format for which Sampling is used. Sampling is process of obtaining amplitudes of a signal at regular intervals. In this paper we will study about the sampling (definition of sampling and all associate concepts such as Signal Processing, Signals, Digital Signal Processing, Samples, and Types of Signal etc), Sample Rate (High Sample Rate, Low Sample Rate and Nyquist Sample Rate), Sampling Theorem (with its related terms as Nyquist frequency, Nyquist criteria and Nyquist rate) and at last we will discuss about the important applications of sampling (such as Audio/Music sampling, Speech Sampling and Video Sampling).Hence we can say that this paper will focus on the overall study of sampling.

INTRODUCTION
We can define the term sampling as
“Sampling is the reduction of a continuous signal to a discrete signal.” A common example is the conversion of a sound wave (a continuous signal) to a sequence of samples (a discrete-time signal).A sample refers to a value or set of values at a point in time and/or space. A sampler is a subsystem or operation that extracts samples from a signal. A theoretical ideal sampler produces samples equivalent to the instantaneous value of the continuous signal at the desired points.[1]

For the detail discussion of sampling we should know the about the terms such as signal, signal processing, continuous signal, discrete signal, Digital Signal Processing (DSP), In short ,we can define these terms as A signal can be referred as function that conveys information about the behavior or attributes of some phenomenon. [2]

The IEEE Transactions on Signal Processing elaborates upon the term "signal" as:
The term "signal" includes, among others, audio, video, speech, image, communication, geophysical, sonar, radar, medical and musical signals. [3]Signal processing is an area of Engineering, Electrical and applied mathematics that deals with operations on or analysis of analog as well as digitized signals, representing time-varying or spatially varying physical quantities. Signals of interest can include sound, electromagnetic, images, and sensor readings.[4]A digital signal/discrete signal is a physical signal that is a representation of a sequence of discrete values (a quantified discrete-time signal), for example of an arbitrary bit stream, or of a digitized (sampled and analog-to-digital converted) analog signal. The term digital signal can refer to either of the following:

“Any continuous-time waveform signal used in digital communication, representing a bit stream or other sequence of discrete values.” [5]

An analog or analogue signal is any continuous signal for which the time varying feature (variable) of the signal is a representation of some other time varying quantity, i.e., analogous to another time varying signal [6]

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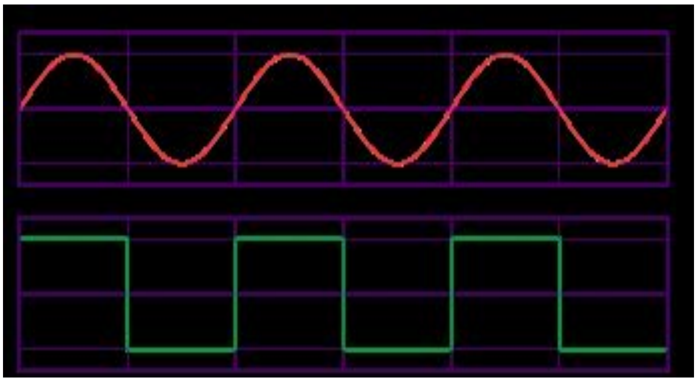

Fig.1: Shows the Analog and Digital Signals.

DSP can be defined as "**d***igital* **s***ignal* **p***rocessing,* which refers to manipulating analog information, such as sound or photographs that has been converted into a digital form. DSP also implies the use of a data compression technique. " [7]

Hence, we know that Sampling is the process of converting a signal (for example, a function of continuous time or space) into a numeric sequence (a function of discrete time or space).In data communication when we send the signal (analog Signal), that signals are converted in the Digital Signals and for no aliasing (for the accurate signal without disturbances), Sampling theorem is used as a standard. In the field of digital signal processing, the sampling theorem is a fundamental bridge between continuous signals and discrete signals (digital signal). [8]

Here, Figure 2 is showing signal sampling. Continuous signal is represented with green color line and digital signals are indicated by blue color vertical lines. [1]

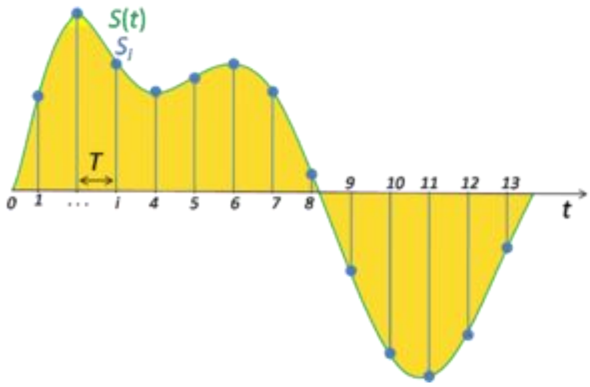

Fig.2: Signal Sampling representation.

To process signals digitally, they obviously have to be presented in the appropriate digital format. Therefore the original analog signal, before processing, has to be converted into a digital one, it has to be digitized. [9]

SAMPLING RATE

Sampling rate is another important aspect of sampling; we can understand it by the below definition.
The sample rate is the number of samples played in each second. Sample rates are measured in "Hertz" (abbreviated "Hz"), which means "per second," or in "kilohertz" (abbreviated "kHz"), which means "per second, times one thousand." [10] Or in other words we can define sample rate as "the sample rate is the number of samples of a sound that are taken per second to represent the event digitally. The more samples taken per second, the more accurate the digital representation of the sound can be." [11]

Our communications are in analog form most of the time when we communicate such as voice signals. As a first step in digitization, the analog signal is converted to a discrete time signal by the process of sampling. While sampling, sufficient number of samples of the signal must be taken so that the signal is completely represented in its samples. Also it should be possible to reconstruct the signal from its samples. [12]

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We can define the term digitization as “Digitizing or digitization is the representation of an object, image, sound, document or a signal (usually an analog signal) by a discrete set of its points or samples. The result is called digital representation or, more specifically, a digital image, for the object, and digital form, for the signal. ” [13]

Below discussing figure 3, 4 & 5 are showing the High sampling rate, Nyquist Sampling Rate and low sampling rate.

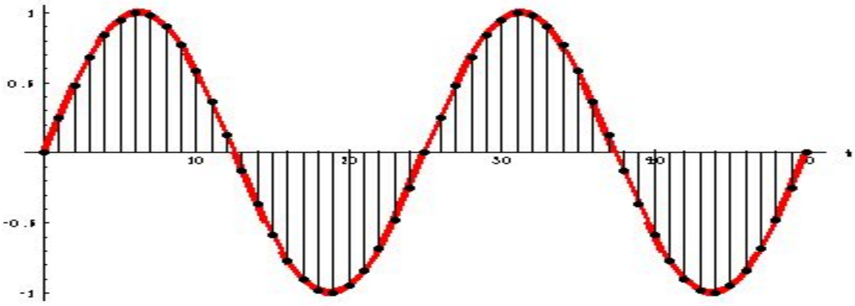

Fig.3:High Sampling Rate (Over Sampling)

- A High Sampling Rate = this is 'Oversampling' that, will take time and will create a large digita file. [14]

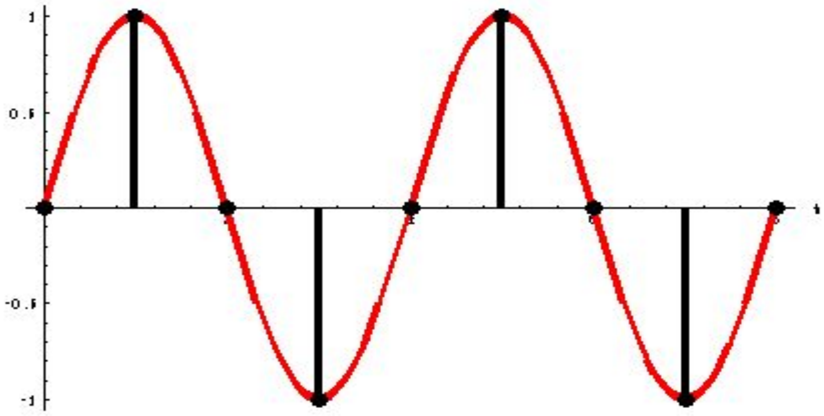

Fig.4: Nyquist Sampling Rate

- Nyquist Sampling Rate = the minimum sample rate that captures the "essence" of the analog information. Note that while Nyquist is appropriate for sampling, it may not capture nuances ir information. But, of course, those nuances are higher frequency, and thus would require a highe Nyquist sample rate. [14]

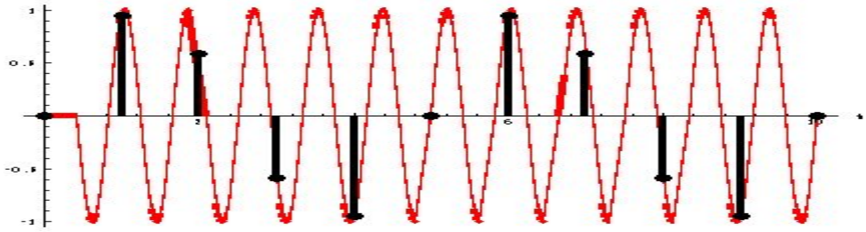

Fig.5: Low Sampling Rate (Under sampled)

- Under sampled: low sampling rate produces results that report false information about the analog data; which does not represent the original. This phenomenon is called aliasing. [14]

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Sampling Theorem

The name Nyquist–Shannon sampling theorem honors Harry Nyquist and Claude Shannon. The theorem was also discovered independently by E. T. Whittaker, by Vladimir Kotelnikov, and by others. So it is also known as the Nyquist–Shannon–Kotelnikov, Whittaker–Shannon–Kotelnikov, Whittaker–Nyquist–Kotelnikov–Shannon, as well as the cardinal theorem of interpolation theory. [8]

Nyquist Theorem: Sample rate > 2 * highest frequency component (of interest) of the measured signal. [16] The Nyquist–Shannon sampling theorem states that perfect reconstruction of a signal is possible when the sampling frequency is greater than twice the maximum frequency of the signal being sampled, or equivalently, when the Nyquist frequency (half the sample rate) exceeds the highest frequency of the signal being sampled. If lower sampling rates are used, the original signal's information may not be completely recoverable from the sampled signal. S14The two thresholds, sample rate and highest frequency are respectively called the Nyquist rate and Nyquist frequency. And the condition described by these inequalities is called the Nyquist criterion, or sometimes the Raabe condition.[1]

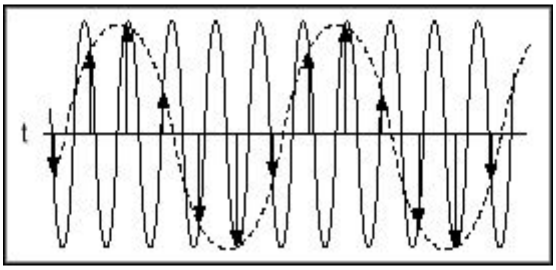

Fig.6: Sine Wave Demonstrating the Nyquist Frequency

Suppose, we are sampling a sine wave

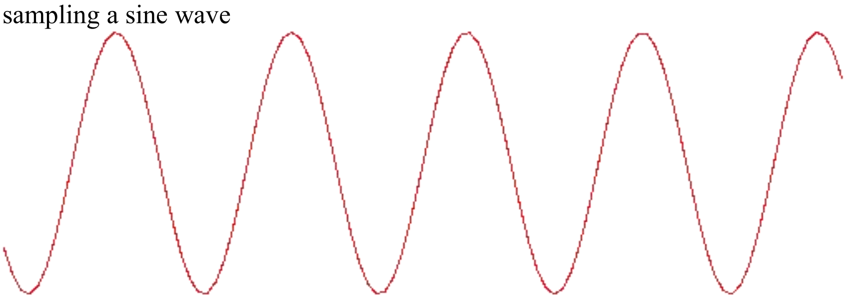

Fig.7: A Sine Wave.

- If we sample at 1 time per cycle,[17]

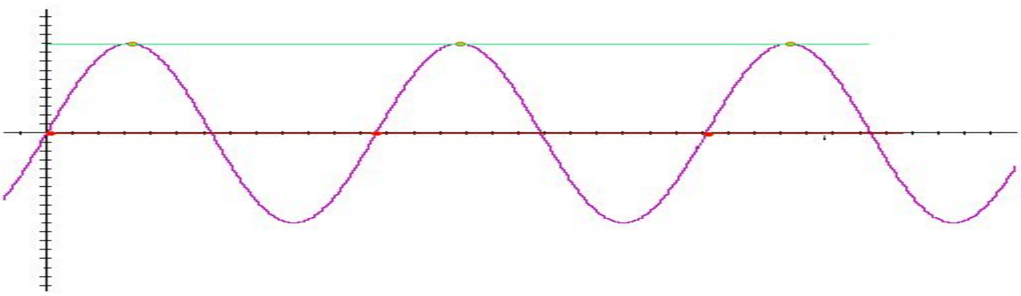

Fig.8: Sampling at 1 time per cycle.

- If we sample at 1.5 times per cycle, we can think it's a lower frequency sine wave.[17]

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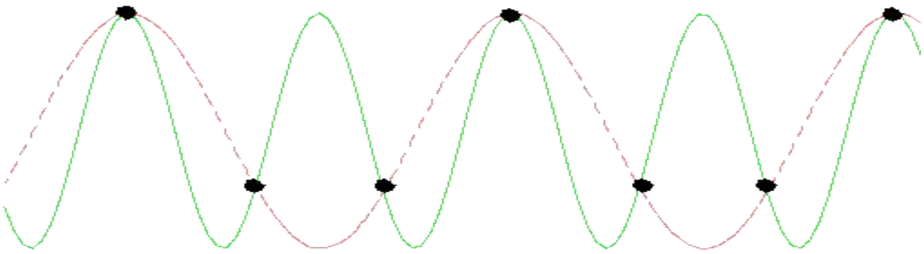


Fig.9: Sampling at 1.5 times per cycle.

- Now if we sample at twice the sample frequency, i.e. Nyquist Rate, we start to make some progress. An alternative way of viewing the waveform (re)generation is to think of straight lines joining up the peaks of the samples. In this case (at these sample points) we see we get a saw tooth wave that begins to start crudely approximating a sine wave.[17]

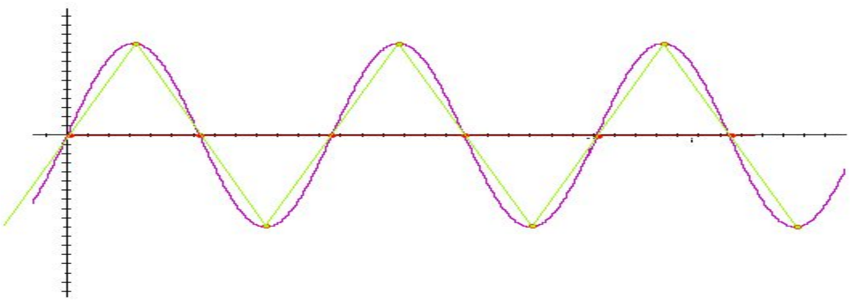


Fig.10: Sampling at 2 times per cycle.

Aliasing does not occur if the sampling rate is greater than twice the frequency of a desired information. [17]

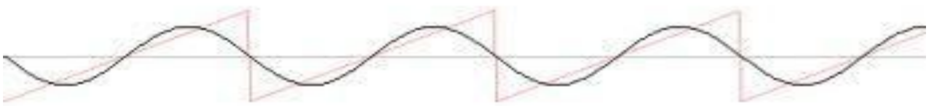


Fig.11: Triangle Wave that begins to start crudely approximating a sine wave.

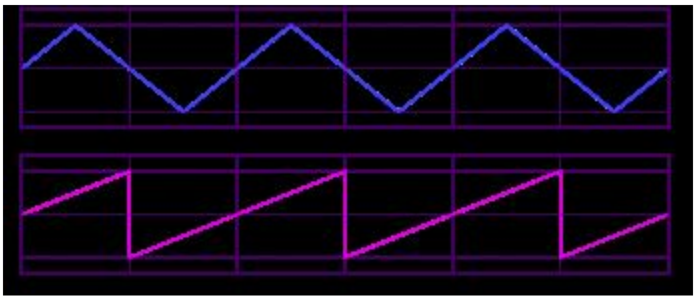


Fig.12: Shows the triangle wave and Saw tooth Wave.

We can define the saw tooth wave as "The saw tooth wave (or saw wave) is a kind of non-sinusoidal waveform. It is so named based on its resemblance to the teeth of a saw". The convention is that a saw tooth wave ramps upward and then sharply drops. However, in a "reverse (or inverse) saw tooth wave", the wave ramps downward and then sharply rises. It can also be considered the extreme case of an asymmetric triangle wave. [18]

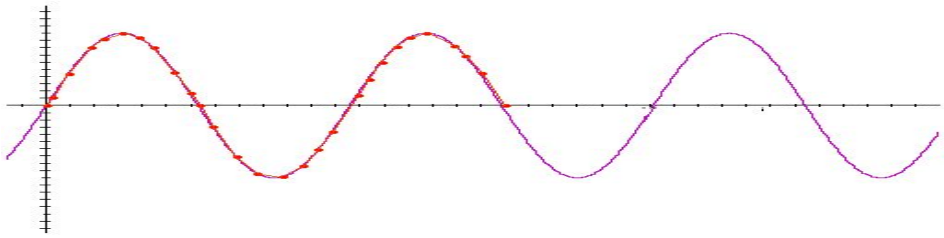
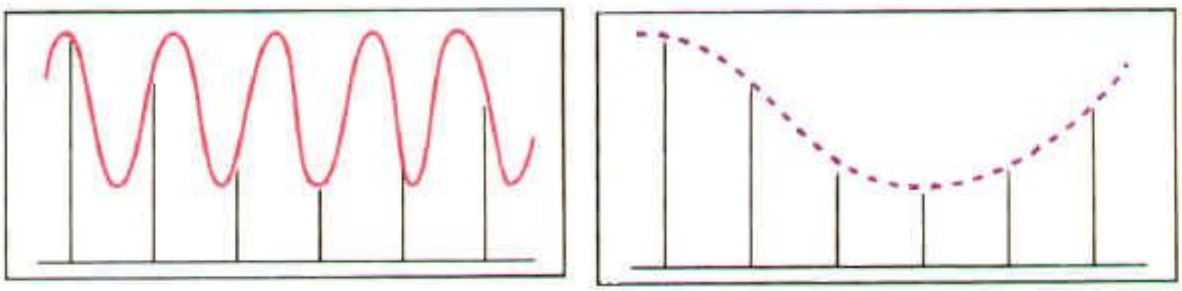


Fig.13: Sampling at many times per cycle.

If the Nyquist rule isn't followed, and the sample rate isn't high enough for the signal were trying to digitize, strange things happen when we try and reproduce the signal back to its analog form. In the accompanying pictures, notice that we have sampled a wave in the first figure (the vertical bars show us where we have picked our sample points.) But when we try to reproduce the signal based on our samples, we get a different wave altogether. This is because we haven't sampled often enough for the high frequency of the original signal. [15]



When good sampling... **...goes bad!**

Fig.14: Shows how Nyquist rule is helpful to reproduce the signal.

Applications of sampling

- ***Audio/Music Sampling**:* Digital uses pulse-code modulation and digital signals for sound reproduction. This includes analog-to-digital conversion (ADC), digital-to-analog conversion (DAC), storage, and transmission. In effect, the system commonly referred to as digital is in fact a discrete-time, discrete-level analog of a previous electrical analog. While modern systems can be quite subtle in their methods, the primary usefulness of a digital system is the ability to store, retrieve and transmit signals without any loss of quality. [1]

BeatCleaver is a sample editor for music sampling from songs, recordings, and other music. Easy slicing buttons make it great for beat slicing and sampling for hip hop. An intuitive design and fluid interface create a fast, frustration-free workflow. [19]

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Fig.15: BeatCleaver, A sample editor for Music Sampling.

- ***Speech Sampling:*** Speech signals, i.e., signals intended to carry only human speech can usually be sampled at a much lower rate. [1]

WASP: Waveforms Annotations Spectrograms & Pitch, A new free program recommended for beginners. WASP is a program for the recording, display and analysis of speech. With WASP we can record and replay speech signals, save them and reload them from disk, edit annotations, and display spectrograms and a fundamental frequency track. [20]

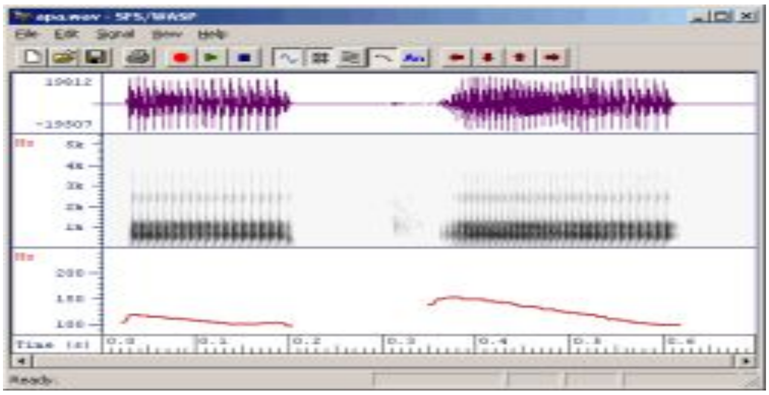

Fig.16: WASP, A sample editor for Speech Sampling.

- ***Video Sampling:*** In order to get images into a form computers can manipulate they must be modified from their analog form (continuous gradients) into a digital form (step gradients) using the technique of sampling. A technique used to record analog information by recording periodic snapshots. If the sampling rate is fast enough, the eye cannot discern the gaps between each snapshot when they are played back. Sampling is the key technique used to digitize analog information such as sound, photographs, and images.[14]

Chroma sampling is the practice of encoding images by implementing less resolution for chroma information than for luma information. It is used in many video encoding schemes, both analog and digital and also in JPEG encoding. [21]

The terminology for the components of digital video is Y' Cb Cr. The Y' represents luma, the ' is important, reminding us that it is non-linear (gamma corrected). The Cb and Cr are chroma components. There are a number of methods of chroma sampling, and they each have terminology that refers to the chroma resolution (the second and third numbers) as compared to the luma resolution (first number).[22]

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- Full resolution luma is represented by the number 4, and as the chroma components Cb and Cr are also 4, there is no reduction in resolution. 4:4:4 sampling is mostly used for RGB images, although it can be used for Y'CbCr, although no camera records 4:4:4 Y'CbCr.[22]

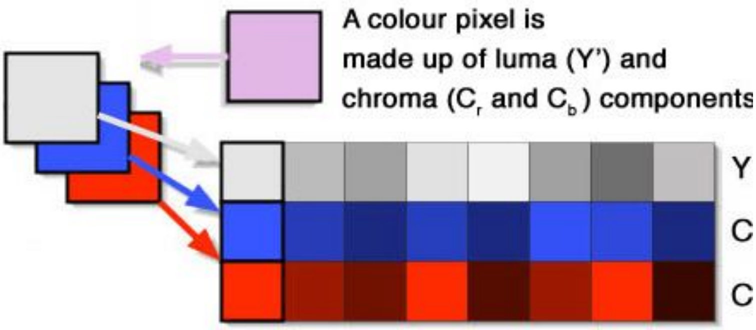

Fig.17:4:4:4 Croma Sampling

- Full resolution luma, and half (2/4 = 0.5) resolution horizontally on the chroma components. This is the traditional broadcast standard for chroma sampling and is used by DigiBeta, DVCpro50 etc.[22]

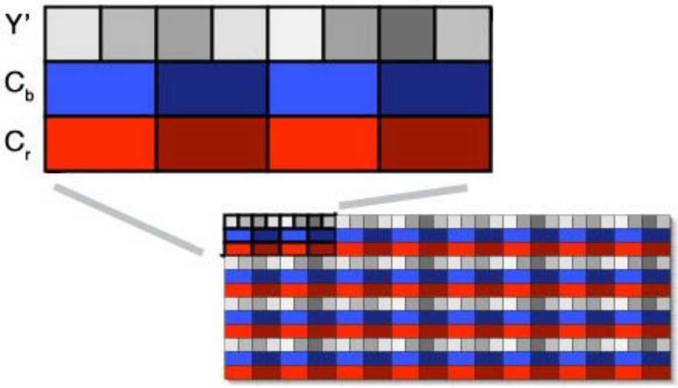

Fig.18:4:2:2 Croma Sampling

- Full resolution luma and quarter (1/4 = 0.25) resolution chroma components. This is the system used by NTSC DV and PAL DVCPro.[22]

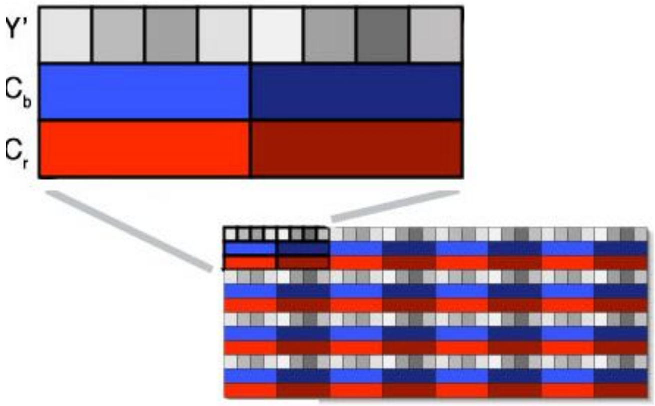

Fig.19:4:1:1 Croma Sampling

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- Full resolution luma, and half resolution in the horizontal direction and vertical direction for the chroma components. 4:2:0 is a very complex chroma sampling with many variants depending on whether the video is progressive or interlaced.[22]

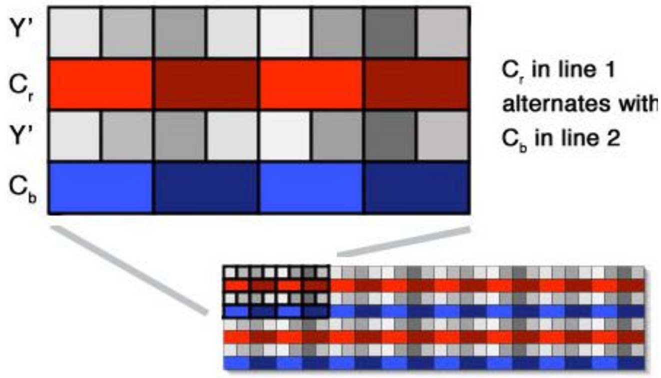

Fig.20:4:2:0 Croma Sampling A1

CONCLUSION

In this paper we have discussed about the Sampling, process of Sampling, related factors such as sampling rate, oversampling, under sampling, Nyquist criteria and the relation between the sample rate of a sample and Nyquist frequency (Sampling Theorem). And at last some applications of Sampling (Audio, Video and Speech Sampling) had discussed. Finally, on the basis of this paper we can conclude that for better Digital Data Processing and for better reconstructive signal (with no aliasing) we should follow the standard of Sampling Theorem.

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