# Simulation Tools for Introduction of DSP in a UG Program : A Teaching Experience

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***Abstract---*** **Digital Signal Processing is a subject often perceived to be difficult to understand. The main reason for this is attributed to the two different domains for the analysis and processing of signals, viz. the time domain and frequency domain. Time domain signals are simpler to analyse as most of the real time signals are in time domain. This makes it easier to visualize the signals. Moreover time domain signals can be viewed through a CRO. But the frequency domain signals look abstract. It is difficult to conceptualise the frequency components in a signal. As a result, students most often miss the concepts of Digital Signal Processing, especially at the undergraduate level, and end up treating the subject as a Math course, without realizing that mathematics is just a tool used! This paper is intended to facilitate the teacher to effectively introduce the fundamental DSP concepts through the interface of real life examples and simulation tools like MATLAB, rather than approaching it as a pure mathematical course.**

***Index Terms--- Digital Signal Processing, Discrete Fourier Transform, MATLAB, Sampling theorem.***

## I. Introduction

Digital Signal Processing has found applications in a vast spread of areas spanning telecommunication, image processing, instrumentation and control, speech and audio processing, biomedical instrumentation, entertainment etc. Recognising the wide range of applications, a course in fundamentals of DSP is introduced in undergraduate courses for all students of electrical sciences. Due to various mathematical operations involved in signal processing, the general trend amongst students is to treat DSP purely from a mathematical point of view, missing out on the essence of fundamental concepts. like sampling, convolution, correlation, frequency transformation, frequency sampling, FFT, etc., which is essential in implementing the knowledge in real life applications. The reason is that most of the faculty also approach the subject through mathematics. A noticeable change can be observed in the perception of the students, if the subject is delivered to them in a different manner. Simple applications are sufficient to clear haziness and introduction using a simulation platform vastly enhances the learning of the student, from the experience of the authors.

A peek into the graphic equalizer of an audio system unveils a world of signal processing where frequencies are segregated, filtered and selectively amplified (in the domain of frequency) from the stream of time domain audio signals, to produce the desired audio effect. DSP, as seen in this scenario would be a magical tool to explore into the process of signal processing, as driven by imagination.

In this paper, an attempt is made to explain the basic concepts with the help of interesting examples. The introduction of a simulation tool like MATLAB will greatly help the students to make the process of learning more efficient. Moreover software simulators act as motivational link between theory and practice [4]. Students will find the subject interesting if the basic concepts are demonstrated with simple examples. In this paper, MATLAB programs have been demonstrated to explain each concept in a simpler way.

## II. Basic Elements of DSP System

A signal is a physical quantity that varies with an independent variable, which in most cases is either time or space. Signal processing is a method of extracting information from the signal. Signal processing can be analog or digital. Digital signal processing is superior due to its advantages [1], [2].

A majority of real time signals are analog in nature, which cannot be processed using a digital signal processor. An Analog to Digital converter (ADC) is used to convert the analog signal into a digital signal. The steps involved in an ADC are sampling, quantizing and encoding as shown in Fig 1.

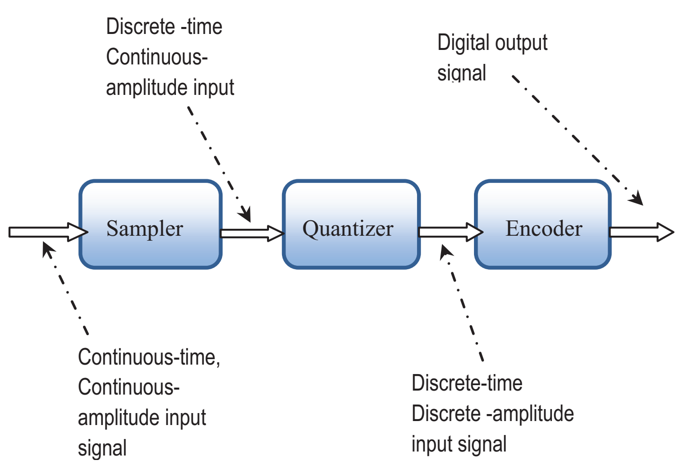


Fig. 1.Block diagram of ADC

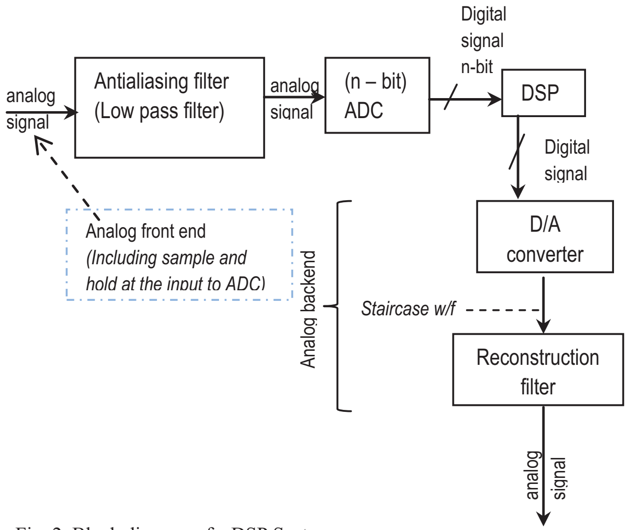


Fig. 2. Block diagram of a DSP System

A sampler takes periodic samples of an analog signal at definite intervals of time which is known as the *sampling time period*. For the effective reconstruction of the original signal, the sampling frequency should satisfy the sampling theorem [2]. The discrete signal obtained by sampling is discrete in time, but can have any amplitude. The quantizer converts a discrete time- continuous amplitude signal into a digital signal by expressing each sample value as a finite number of digits.

In applications where digital output from the digital signal processor is to be given to the user in analog form, say a speech processing application, a digital to analog converter (DAC) is used. Apart from ADC and DAC, the DSP system also involves the use of an antialiasing filter and a reconstruction filter. Antialiasing filter is a low pass filter which prevents the generation of the aliasing noise. The reconstruction filter is provided at the output of the DAC which converts the staircase waveform from the DAC into the analog signal. This is done by filtering out the high frequency components from the staircase waveform to get a smoother analog waveform. The block diagram of a complete DSP system is shown in fig. 2

## III. Introducing DSP In a Class

It is a general trend among faculty to start the subject by explaining the block diagram of digital signal processor. This includes introducing the concept of conversion of analog to digital signal, etc. This is immediately followed by the introduction of frequency transformation and the different algorithms to calculate Discrete Fourier Transform. This situation is perplexing for a student. The signals so far dealt in time domain is suddenly converted into frequency domain. The necessity of analyzing the signal in frequency domain remains unknown to the student [3]. The conversion from one domain to another, itself is confusing, let alone the concepts of sampling in time, sampling in frequency, aliasing etc…..

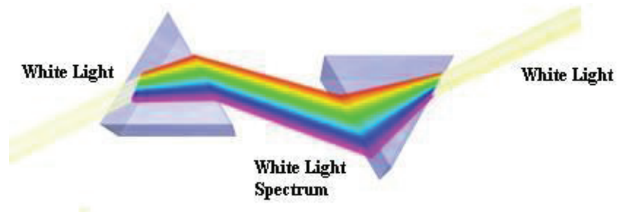


Fig. 3. White Light Spectrum

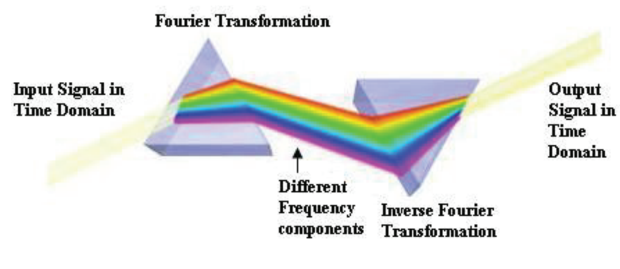


Fig. 4. Analogy

A pragmatic approach to this problem is to introduce the subject with some real life example. A classic example is a music system with an audio equalizer. This can be taken up to introduce the subject to the class on the very first day. The features of the equalizer such as bass and treble adjustment, which exactly is the processing of the frequency components in a sound signal can be explained. The major operations done on the signals in signal processing are selective filtering and amplification of the various frequency components in the input signal. This is achievable only if the frequency components in the signal are extracted. .

The concept of frequency analysis can be introduced by citing the example of a prism creating a spectrum of different colors from white light as shown in fig. 3. An analogy can be drawn in signal processing as shown in fig.4. A signal is decomposed into sinusoidal frequency components (Fourier representation). The time domain signal can be regenerated by performing the inverse frequency transformation, which is analogous to the inverted prism.

## IV. Fundamental Concepts Through MATLAB

This section introduces a few basic MATLAB programs to make the fundamental concepts simple [5].

### A. Sampling

Sampling and the effect of aliasing are the first concepts that should be introduced in DSP. A MATLAB script file can easily demonstrate the sampling theorem, the effect of under sampling, critical sampling and over sampling. Figure 5 shows the results. A sinusoidal signal of 100Hz is taken as the input signal.

Sampling is done at different frequencies and the signal reconstructed. Critical sampling is performed by selecting the sampling frequency as 200Hz (fs=2fm).

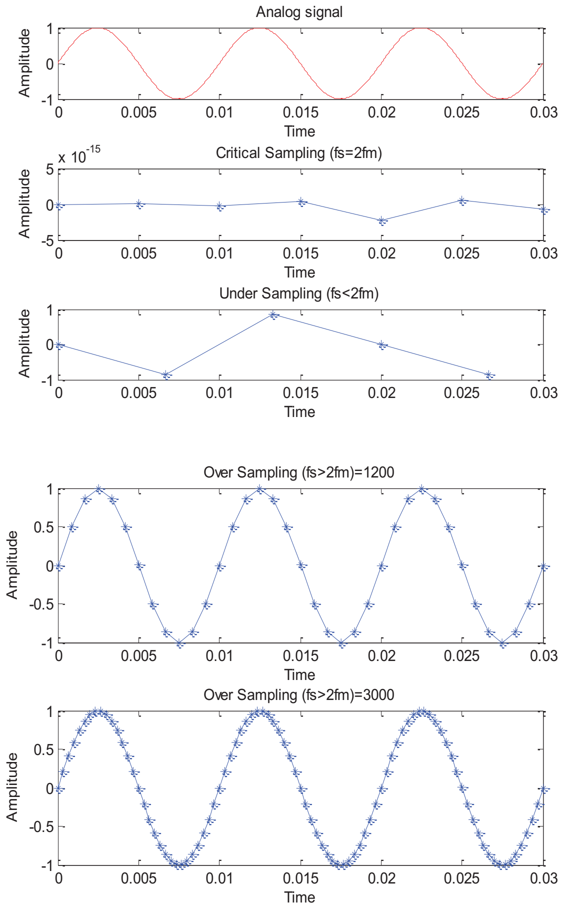


Fig. 5. Demonstration of sampling theorem

For Under-sampling, the sampling frequency chosen is 150Hz (fs<2fm). The reconstructed signals show that the sampling is not effective. Two different frequencies (1200Hz and 3000Hz) are selected for over sampling. It can be seen that as the sampling frequency increases, the quality of the reconstructed signal is improved. The improvement in the reconstructed signal is not significantly different from 1200Hz to 3000Hz. This is the point to make the student aware that over sampling with too large a sampling frequency, leading to computational complexity is unnecessary.

### B. Aliasing

Aliasing is one of the most confusing abstract concepts!! This simple demo, will clarify the concept. Aliasing arises when a signal is sampled at a rate which is less than Nyquist rate (fs≥2fm). A script file is written to show the effect of aliasing. Figure 6 shows the effect of aliasing. A sine signal of frequency 1kHz is sampled at two different frequencies. The reconstructed signal is an aliased low frequency signal of the original. It can be seen from the figure that the frequency of the aliased signal is the difference between the sampling frequency and the signal frequency.

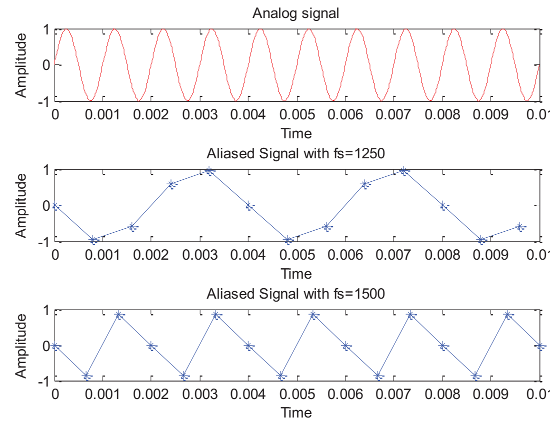


Fig. 6. Effect of Aliasing

### C. Frequency content

To understand the concept of frequency analysis, a student should clearly know the concept of frequency components in a signal. This can be started by taking a pure sine signal and computing the Fourier transform with a MATLAB code. The frequency spectrum gives a single peak at the signal frequency, which shows the presence of a single frequency content. Then a signal with two frequencies can be introduced and the corresponding frequency spectrum can be shown. To explain the concept of phase shift, another signal can be shown by introducing a phase shift to the second signal. Fig.7 shows the graph of a sine wave of frequency 1kHz and the corresponding frequency spectrum. In Fig.8 and Fig.9, the input signal is the sum of two sine signals of frequencies 1kHz and 3kHz. A phase shift is added in fig.7. It can be observed that while the magnitude spectrum remains same, the phase spectrum has changed.

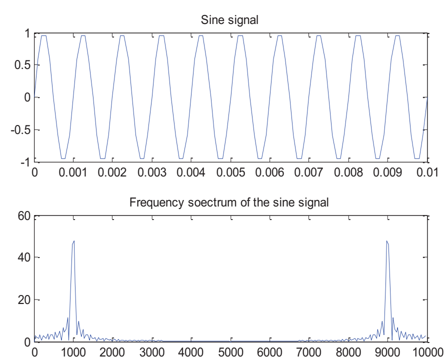


Fig. 7. Frequency spectrum for a single sine wave.

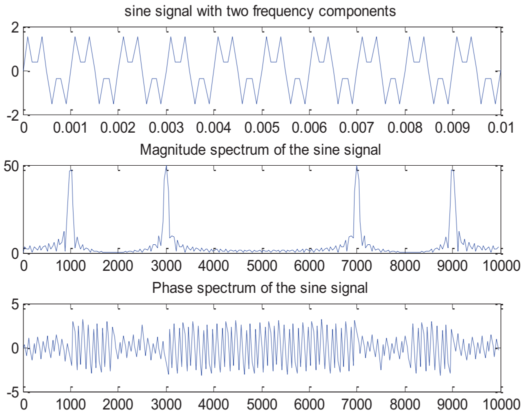


Fig. 8. Frequency spectrum for a sine wave with frequencies 1kHz, and 3kHz.

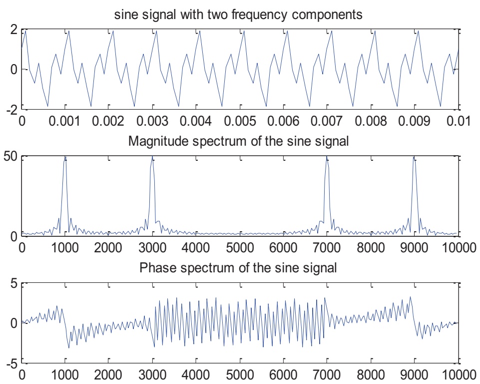


Fig. 9. Frequency spectrum for a sine wave with frequencies 1kHz, and 3kHz with a phase shift.

## D. Need for Frequency Sampling

The need for frequency sampling can be explained by taking the example of a white light spectrum as discussed in section III. It can be noticed from Fig.4 that when a time domain signal is transformed to frequency domain signal, there are infinite number of frequency components. In other words, DTFT of a discrete time aperiodic signal gives a continuous periodic spectrum with a periodicity in ω of 2π or – π to + π. So DTFT cannot be used to the computation advantage. A possible solution is to sample the frequency spectrum at discrete frequency points to obtain Discrete Fourier Transform. The concept of frequency sampling is illustrated in Fig 10 and 11, with different sampling lengths. The reconstructed time domain signal does not change if the samples are increased. This point is to be driven here. However, the frequency spectrum has more resolution as illustrated by the Power Spectrum Density curve.

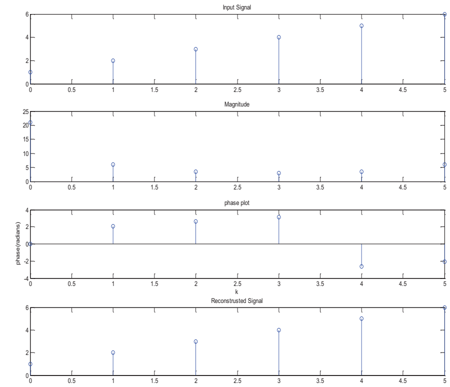


Fig. 10. DFT with N=L

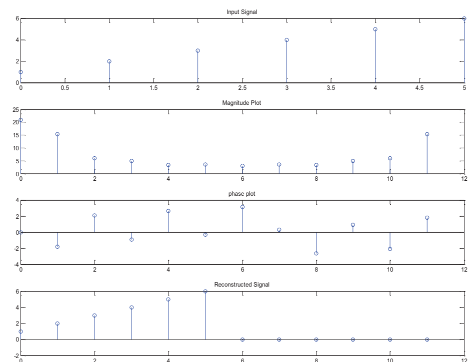


Fig. 11. DFT with N>L

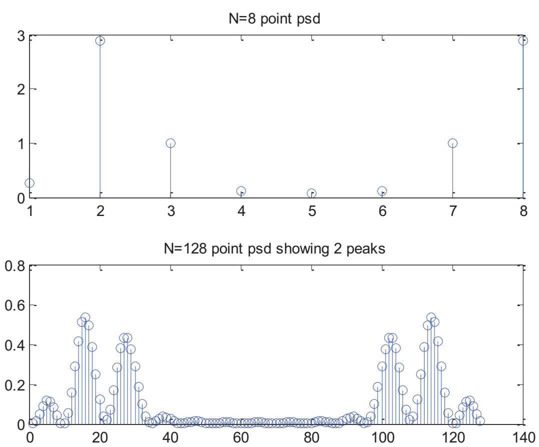


Fig 12. Power Spectrum Density curve using N point DFT

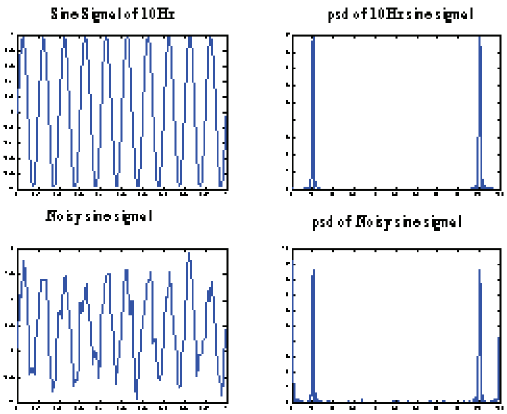


Fig. 13. Power Density Spectrum

Figure 12 shows the power spectrum for a signal using N-point DFT for two different values of N. It can be noticed from the figure that there is only one peak at k=1, indicating poor resolution with N=8. With N=128, the frequency resolution is increased.

### E. *Power Spectrum*

Finite energy signals possesses a Fourier Transform and are characterized by their energy-density spectrum. The other important class of signals characterized as stationary random processes do not have finite energy and hence do not possess a Fourier Transform. Such signals have finite average power and hence are characterized by a power density spectrum. DFT is used extensively in power spectrum estimation.

A MATLAB program is written to generate a 10Hz signal and a composite signal by adding noise. The power density spectrum is plotted for both the signals (Fig 13). It can be seen from the graph that though the sine wave is noisy in time domain with no apparent periodic structure, its PDS is same as that of the PSD of the pure wave. Hence the frequency information can be easily obtained using the PSD.

## V. Conclusions

This paper is an attempt to make the introduction of the subject of DSP simpler to the students. An approach to the subject is proposed where the concepts are explained with real life examples and considering interesting analogies. Employing simple MALAB programs gives the visualization of the fundamental concepts, like frequency analysis, sampling theorem, aliasing and Discrete Fourier Transform. In demonstrating the sampling theorem, different cases are analyzed by taking different sampling frequencies for critical sampling, under sampling and over sampling. By observing the reconstructed signal waveform, the student can easily visualize the effect of sampling.

A faculty can spend the initial few classes to make the concepts clear by adopting different methods discussed in this paper. This makes the students realize the need for performing frequency analysis, convolution, calculating DFT, etc. In the next stage, the various algorithms to perform convolution, Fourier Transform, Fast Fourier Transform, etc. can be introduced to the students. The method proposed in this paper was introduced in the class, and response was very good. The student feedback indicates that the learning experience has vastly improved and the haziness and abstractness about the concepts have diminished.

## VI. References

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