ISSN: 0974-2115
www.jchps.com Journal of Chemical and Pharmaceutical Sciences

A Tutorial Review on Digital Signal Processing Using Matlab

V. Elamaran*, K. Narasimhan, G. Rajkumar, M. Chandrasekar
Department of ECE, School of EEE, SASTRA University, Thanjavur, India.
***Corresponding author: elamaran@ece.sastra.edu**

ABSTRACT

The primary focus of this study is to enhance the understanding skills and to improve the research attitude among the student community in the domain of digital signal processing (DSP). This paper exemplifies some of the DSP concepts through fundamental exercise problems in two sections. Nyquist theorem concept, typical block diagram of a DSP system, advantages of DSP over analog signal processing, analog and digital frequency with sampling period concepts are explained in one section. The second section includes finite impulse response (FIR) DSP systems, infinite impulse response (IIR) DSP systems, moving average low pass and high filters and the significance of pole/zero location in filters. This study portrays all the simulation results for relevant theory using Matlab and Simulink software tools.

KEY WORDS: Digital signal processing, FIR filters, IIR filters, Matlab, Pole/zero locations, Simulink.

1. INTRODUCTION

Though the most real world signals are continuous by nature, the processing of signals in analog domain is difficult. Analog circuits/systems are not repeatable due to the tolerance values, ageing of components and temperature variations. Analog circuits/systems do not produce accurate results due to the above said reasons. But DSP systems are repeatable i.e., they do perform identically at all times. The DSP systems are physically stable since they are not affected by ageing or temperature. DSP systems do have the advantages of programming and hence many applications like replay the audio/music, fast forward/backward audio/music, etc. are possible in digital domain. This kind of versatility is not possible or difficult in analog domain.

DSP systems play a vital role in all kind of consumer electronics applications like mobile phones, MP3/DVD players, digital televisions, digital cameras, etc. The processing of data in digital domain is easy like addition, subtraction instead of integration, differentiation happens in analog domain. The analog systems do need to replace the components for different specifications or needs. For example, in a low pass filter R and C components should be replaced if the required cut-off frequency is different. This may not be a problem for one or two cases in a design, but it is tedious if thousands should be changed in a system. This problem doesn't happen in DSP systems by means of modifying few parameters in a computer program.

DSP techniques are often used in audio, still images, moving images, communications etc. Audio sample rates and conversion, audio in the frequency domain, audio signal compression, pitch extraction and delta-sigma conversion are performed using DSP principles and algorithms. DSP principles are used in still image applications like filtering an image, image compression using discrete cosine transform (DCT) and discrete wavelet transform, image enhancement, image scaling, edge detection and pattern recognition. Video processing applications like deinterlacing, video standards conversion, MPEG video compression and motion estimation demand DSP techniques to solve. Digital modulation techniques and spread spectrum schemes exploit the principles behind DSP for effective communication systems design.

2. MATERIALS AND METHODS

DSP basics and digital filter theory and design are explained in the section with relevant examples.

DSP Basics:

Nyquist theorem concept: Nyquist theorem is visualized with moving a second hand of clock which performs one rotation at every 60 seconds around the clock. If the sampling begins at 12 o'clock with a sample at every 10 seconds, the timings are 0, 10, 20, 30, 40, 50, 0, 10, 20, 30, 40, 50, 0, etc. If the sampling begins at 12 o'clock with a sample at every 20 seconds, the timings are 0, 20, 40, 0, 20, 40, 0, etc. Both the results convey that the movement of a second hand in forward direction. Suppose the samples are taken at every 30 seconds, the results are 0, 30, 0, 30, 0, 30, 0, etc. This is difficult to say the samples are towards in forward or backward direction. If the sampling is done at every 45 seconds, the timings are 0, 45, 30, 15, 0, 45, 30, 15, 0, etc. It looks like they are moving backward, but actually they are in forward direction. This is termed as aliasing. So aliasing is happened here due to insufficient sample in 60 seconds i.e., samples at 45 seconds. There is no aliasing in the first and second cases due to adequate samples i.e., samples at every 10 seconds and 20 seconds respectively. Similarly, if the sinusoid signals are sampled at more than twice the frequency, there is no aliasing; also the signal can be reconstructed back to its original form without loss. Importance of Nyquist theorem is demonstrated in Figure 1 with Matlab script and the resulting signals are plotted in Figure.2.

Typical block diagram of a DSP system: Digital signal processor is the heart of the DSP system. Anti-alias filter is a low-pass filter with cut-off frequency Fs/2. Then the band-limited analog signal is converted into digital data using ADC which becomes the input to the DSP processor. The filtered or processed signal comes out from the DSP processor and are given to the DAC section. The analog signal is further processed by reconstruction filter which is a low-pass filter with cut-off frequency Fs/2 to produce the output signal as in Figure.3.

Advantages of DSP over analog signal processing: DSP systems are programmable i.e., the changes can be made easily to perform different tasks. They are repeatable i.e., they do process signals with identical performance. DSP systems avoid the effects of temperature, component ageing and tolerance problems. So, they are physically stable. Since all kind of processing is done using computer programs, DSP systems are versatile. That is adaptive filters, performing functions which are not possible with analog domain and software updates, etc. can be made easily with DSP systems.

Real and digital frequencies: The analog frequency is termed as real here. The real frequency to digital frequency conversion is made through the sampling interval T (seconds/sample). The relationships among these terms are made in Equation 1, 2, and 3.

$$t = nT$$
$$\text{seconds} = \text{sample} \times \frac{\text{seconds}}{\text{sample}} \tag{1}$$

$$\omega = \Omega T$$
$$\frac{\text{radians}}{\text{sample}} = \frac{\text{radians}}{\text{second}} \times \frac{\text{seconds}}{\text{sample}} \tag{2}$$

$$\omega = 2\pi f$$
$$\frac{\text{radians}}{\text{second}} = \frac{\text{radians}}{\text{cycle}} \times \frac{\text{cycles}}{\text{second}} \tag{3}$$

DSP Systems:

FIR DSP system: Digital filters are mainly classified as finite impulse response (FIR) and infinite impulse response (IIR) filters. The concepts behind these filters are explained in this section with examples. If the system difference equation is (Equation 4), then the impulse response can be determined as in Equation 5.

$$y(n) = 0.5x(n) + 0.4x(n-1) \tag{4}$$
$$h(n) = \{0.5, 0.4\} \tag{5}$$

This is FIR DSP system since the impulse response is finite i.e., only two samples. This is FIR DSP system.

IIR DSP system: If the system difference equation is (Equation 6), then the impulse response can be determined as in Equation 7.

$$y(n) = 0.5x(n) + 0.4x(n-1) - 0.4y(n-1) \tag{6}$$
$$h(n) = \{0.5,\ 0.2, -0.08, 0.032, -0.00128, 0.00152, -0.002048, \ldots\} \tag{7}$$

This is IIR DSP system since the impulse response has infinite samples.

Simple low-pass FIR filter: If the system difference equation is $H(z) = 1+z^{-1}$, the DC component (zero frequency) is allowed and the Nyquist frequency component (Fs/2) is not allowed since it is a low-pass filter. These are portrayed in Figures.4 and 5. If the filter coefficients are 1/2 & –1/2, then the DC component is allowed without amplification as in Figure.6.

Simple high-pass FIR filter: If the system difference equation is $H(z) = 1-z^{-1}$, the DC component (zero frequency) is not allowed and the Nyquist frequency component (Fs/2) is allowed since it is a high-pass filter. These are portrayed in Figures.7 and 8. If the filter coefficients are 1/2 & –1/2, then the Fs/2 component is allowed without amplification as in Figure 9.

Pole/zero locations and the frequency response: The pole/zero locations and the frequency response of the filter, $H(z) = 1+z^{-1}$ are plotted in Figures.10 and 11. Figures.12 and 13 show the pole/zero locations and the frequency response of the filter, $H(z) = 1-z^{-1}$.

3. RESULTS AND DISCUSSION

This study demonstrated DSP fundamentals through examples and Matlab script for better understanding. Understanding Nyqusit theorem and real/digital frequencies concept would help a lot to design a DSP system perfectly. Also the concept behind simple low-pass filters and high-pass filters described in this study would help to understand even much complicated filters in real time applications.

```
t1 = 0:0.001:0.4;
y1 = sin(2*pi*10*t1);

subplot(211);
plot(t1,y1);
title('sampling frequency is more than twice the signal frequency');

t2 = 0:0.01:0.4;
y2 = sin(2*pi*100*t2);

subplot(212);
plot(t2,y2);
title('sampling frequency is less than twice the signal frequency');
```

Figure.1. Simple Matlab script demo for Nyquist theorem

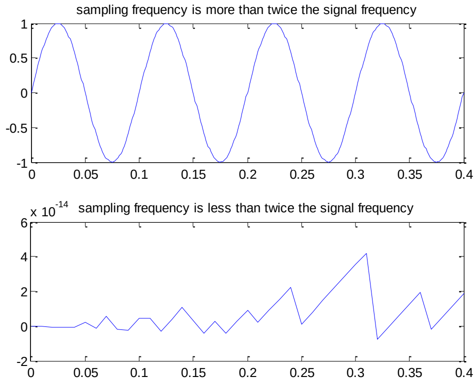

Figure.2. Nyquist theorem and its violation

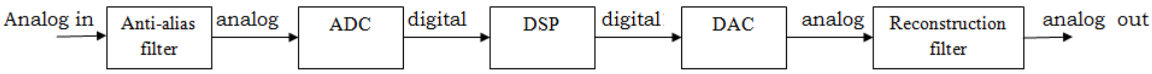

Figure 3. Typical DSP system block diagram

$\{2,2,2,2,2,....\} \rightarrow \boxed{H(z)=1+z^{-1}} \rightarrow \{2\ or\ 4,4,4,4,4,....\}$

```
>> filter([1 1],1,[2 2 2 2 2 2 2 2 2 2])

ans =

     2     4     4     4     4     4     4     4     4     4
```

Figure.4. DC component is allowed (amplified) through low-pass filter

$\{2,-2,2,-2,2,-2,...\} \rightarrow \boxed{H(z)=1+z^{-1}} \rightarrow \{2\ or\ 0,0,0,0,0,....\}$

```
>> filter([1 1],1,[2 -2 2 -2 2 -2 2 -2 2 -2])

ans =

     2     0     0     0     0     0     0     0     0     0
```

Figure.5. Fs/2 component is not allowed through low-pass filter

```
>> filter([1/2 1/2],1,[2 2 2 2 2 2 2 2 2 2])

ans =

     1     2     2     2     2     2     2     2     2     2
```

Figure 6. DC component is allowed through low-pass filter

$\{2,2,2,2,2,2,...\} \rightarrow \boxed{H(z)=1-z^{-1}} \rightarrow \{2\ or\ 0,0,0,0,0,0,0,0,....\}$

```
>> filter([1 -1],1,[2 2 2 2 2 2 2 2 2 2])

ans =

     2     0     0     0     0     0     0     0     0     0
```

Figure.7. DC component is not allowed through high-pass filter

$\{2,-2,2,-2,2,-2,...\} \rightarrow \boxed{H(z)=1-z^{-1}} \rightarrow \{2\ or\ 0,-4,4,-4,4,-4,4,....\}$

```
>> filter([1 -1],1,[2 -2 2 -2 2 -2 2 -2 2 -2])

ans =

     2    -4     4    -4     4    -4     4    -4     4    -4
```

Figure.8. Fs/2 component is allowed (amplified) through high-pass filter

```
>> filter([1/2 -1/2],1,[2 -2 2 -2 2 -2 2 -2 2 -2])

ans =

     1    -2     2    -2     2    -2     2    -2     2    -2
```

Figure.9. Fs/2 component is allowed through high-pass filter

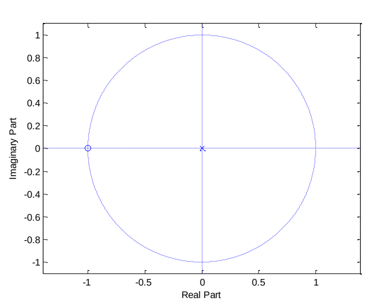

Figure.10. Ploe/zero plot of filter, H(z) = 1+z^{-1}

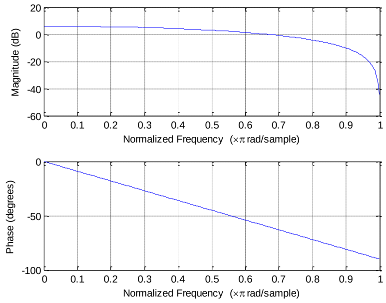

Figure.11. Frequency response of the filter, H(z) = 1+z^{-1}

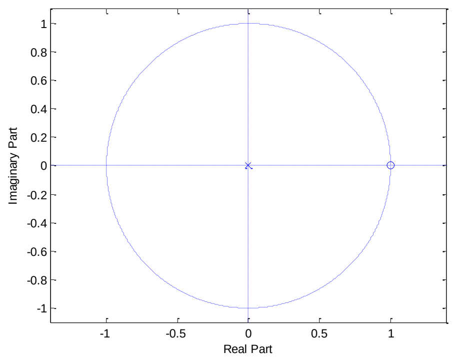

Figure.12. Ploe/zero plot of filter, H(z) = 1–z^{-1}

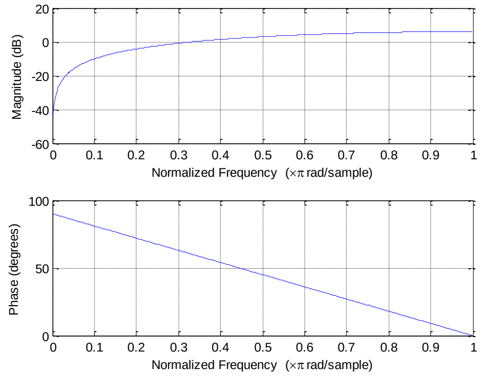

Figure.13. Frequency response of the filter, H(z) = 1–z^{-1}

4. CONCLUSION

This study further can be extended to fundamentals of adaptive signal processing, 2-D signals and systems, multi rate signal processing, radar signal processing, signal processing for random signals, speech signal processing, etc.

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