Analysis of Continuous-Time and Discrete-Time

Mrs.R. Mahima[1], mahima.ece@krce.ac.in

Faculty, Department of ECE, K. Ramakrishnan college of engineering, Tamil Nadu
Sanjai Kumar S[2], Saravanan R[3], Suresh Krishna S[4], Syed Jameel U[5]
sanjaikumarsk2007@gmail.com[2], rsaravanan.dbs@gmail.com[3], sureshkrizz96@gmail.com[4], syedj7212@gmail.com[5]

Students, Department of ECE, K. Ramakrishnan college of engineering, Tamil Nadu

***Abstract*—** Continuous-time and discrete-time signals are the core concepts that underpin not only signal processing but also automation systems, biomedical instrumentation, and digital communications. Continuous signals, to be specific, are real-time physical quantities, while discrete signals are time-sampled numerical data that can be processed by computers and digital processors. MATLAB, being a high-level scientific computing environment, has almost every tool necessary to analyze, visualize, and experimentally confirm signal theory. The article elaborates on the behavior of continuous and discrete signals, their modeling, sampling, aliasing, reconstruction, spectral representation, etc., with the help of MATLAB. The experimental work covers a wide range of signal types in time and frequency domains, including sinusoidal signals, composite signals, exponential signals, unit-step signals, and impulse responses, as well as their transforms.

***Keywords*—**Continuous-Time signal, Discrete-Time signal, MATLAB, Sampling, Aliasing, Reconstruction, Fourier Transform, DSP, Signal Visualization.

I. INTRODUCTION

Signals are the primary means by which information is conveyed in the systems of engineering and science, and basically, they are the physical quantities that change in time, space, or some other variable. Signals are the core concepts in phenomena and in systems. Signals in nature are for example, speech, heartbeat, earthquake vibrations, and environmental temperature changes. In contrast, signals in engineered systems are electrical communication waves, radar pulses, and outputs of machine sensors. Signals are the main tools to understand, represent, and control information in the real world. Almost every function in the present technological systems of communication engineering, biomedical diagnostics, robotics, industrial automation, avionics, multimedia technology, or intelligent transportation systems depend on the correct performance of signal acquisition, representation, processing, and interpretation.

As a result, Signals are mainly divided into Continuous-Time (CT) and Discrete- Time (DT) signals. Continuous-time signals are those that have a value for each time instant and, in fact, they are the representations of nature of the physical world processes such as analog voice, ECG heart waves, sensor voltage signals, acoustic vibrations, and electromagnetic radiation. These signals have unlimited resolution both in amplitude and time, hence they are perfect models of the real-world phenomena. Nevertheless, continuous- time signals are not directly operable, storable, or transmittable in modern digital systems. Such a drawback is the reason for the conversion of continuous-time signals into discrete-time signals.

Signals are the main ways through which information is transferred in the systems of engineering and science. In essence, signals are the physical quantities that vary with time, space, or some other variable. Signals are core concepts in both phenomena and systems. Signals in the natural world are, for instance, human speech, the sound of the heartbeat, the shaking of the earth during an earthquake, and the changes in the temperature of the environment. On the other hand, signals in engineered systems are, among others, electromagnetic communication waves, radar pulses, and the outputs of machine sensors. Signals are the leading means to comprehend, represent, and control the information that exists in the real world. Almost every function in the current technological systems of communication engineering, biomedical diagnostics, robotics, industrial automation, avionics, multimedia technology, or intelligent transportation systems is dependent on the proper performance of signal acquisition, representation, processing, and interpretation.

Consequently, signals are primarily categorized into Continuous-Time (CT) and Discrete-Time (DT) signals. Continuous-time signals are those that have a value for each time instant and are, in fact, the representations of the nature of physical world processes such as an analog voice, ECG heart waves, sensor voltage signals, acoustic vibrations, and electromagnetic radiation. These signals are of unlimited resolution both in amplitude and time, thus they are ideal models of real-world phenomena. However, continuous-time signals cannot be directly operated, stored, or transmitted in contemporary digital systems. Such an inadequacy is the reason behind the conversion of continuous-time signals into discrete-time signals.

II. LITERATURE REVIEW

Oppenheim, Willsky, and Nawab (1996): Signals and Systems: Oppenheim, Willsky, and Nawab in 1996 deliver an insightful, comprehensive base for the study of signals and systems addressing both continuous and discrete aspects. Their work goes in-depth into LTI system analysis, convolution, Fourier series, Fourier transform, Laplace transform, and sampling theory. Time and frequency domains are presented as dual concepts, and the authors provide a thorough theoretical framework along with a multitude of examples and exercises. This book, because of its rich conceptual content, is often seen as an essential source for beginners as well as researchers who aim to build their foundational knowledge in signal theory and systems.

Haykin and Van Veen (2005): Signals and Systems: The work of Haykin and Van Veen (2005) also covers the field of signals and systems; however, they integrate fewer theoretical aspects and, instead, they put more stress on the practical side, particularly in the areas of communication and control systems. The book is filled with clear descriptions of the classical signal-processing concepts, accompanied by numerous examples taken from real life that demonstrate the interplay of theory and practice. Its methodology permits the students to establish the link between mathematical notions and practical engineering problems, which in turn makes the text a great resource for students who are about to embark on applied research in communications and adaptive systems.

Oppenheim and Schafer (2013): Discrete-Time Signal Processing: Oppenheim and Schafer (2013) provide a deep and detailed study that is limited to only one aspect of signal processing, which is discrete-time. Their work extensively involves representation of discrete-time signals, z-transform theory, discrete time Fourier transform (DTFT), Fourier transform fast Fourier transform (FFT) algorithms, filter design, multirate signal processing, and quantization effects in digital systems. It is well-known for the challenging mathematical basis it establishes, hence is mostly employed at the graduate level. This book is a must- have for any graduate later specializing in DSP theory and algorithm development.

Proakis and Manolakis (2007): Digital Signal Processing – Principles, Algorithms, and Applications: Proakis and Manolakis (2007) outline the DSP techniques and algorithms in their work in detail. Their book covers the fundamentals of discrete-time systems and the transforms (DFT, FFT) as well as IIR and FIR filter design, adaptive filtering, and spectral estimation. The authors' balance between the theoretical concepts of DSP and their computational implementation is one of the major strengths of their book. On the one hand, it is filled with extensive and various problem sets, and on the other hand, it contains deep algorithmic discussions, thus it could serve as a perfect tool for the academic studies as well as a source for the practical digital signal- processing research.

Mitra (2011): Digital Signal Processing – A Computer-Based Approach: Mitra (2011) takes great care in explaining the practical side of the DSP concepts by making use of computer-based tools such as MATLAB. His book not only introduces the fundamental theory of discrete-time processing but also involves computational examples, filter design practice, and signal analysis routines so that readers can perform practical exercises. Hence, the text is a perfect fit for students and engineers who intend to execute DSP algorithms in real-world software environments. Mitra’s focus on simulation-based learning effectively connects the theoretical side of DSP with its practical applications.

Summary

The landmark works discussed here at various levels have essentially charted the intellectual journey of signal and system theory starting from elementary concepts up to sophisticated digital processing and implementation. Oppenheim et al. (1996) and Haykin & Van Veen (2005) provide the core theories of continuous- and discrete-time systems, whereas Oppenheim & Schafer (2013) and Proakis & Manolakis (2007) take the conversation to a more detailed and algorithm-focused DSP level. Mitra (2011) is consistent with this ensemble of publications by highlighting computational and software methods as the most suitable way to a DSP system in practice. So, these works together represent the essential academic groundwork of the present-day research and R&D in signals, systems, and digital signal processing.

III. THEORETICAL AND MATHEMATICAL BACKGROUND

Grasping the concepts of continuous-time and discrete-time signals is only possible with a deep understanding of the math involved. Here, we elaborate on the ideas, categories, features, and changes that are the bases for figuring out how signals act and for putting into practice the digitally based signal processing (DSP) systems.

A. Continuous-Time Signals

A continuous-time (CT) signal is defined for every real-valued time instant:

$$x(t)\quad -\infty<t<\infty$$

Examples include analog audio, temperature variation, and physiological signals such as ECG and EEG.

1. Basic Mathematical Forms

- Sinusoidal Signal
  $x(t)=A\sin(2\pi ft+\phi)$
- Exponential Signal
  $x(t)=e^{at},\quad a\in\mathbb{R}$
- Unit Step Signal
  $u(t) = \begin{cases} 1, & t \ge 0 \\ 0, & t < 0 \end{cases}$
- Unit Impulse Function
  $\delta(t) = \frac{d}{dt}u(t)$

B. Discrete-Time Signals

A discrete-time signal is defined only at discrete time intervals:

$$x[n] = x(nT_s), \quad n \in \mathbb{Z}$$

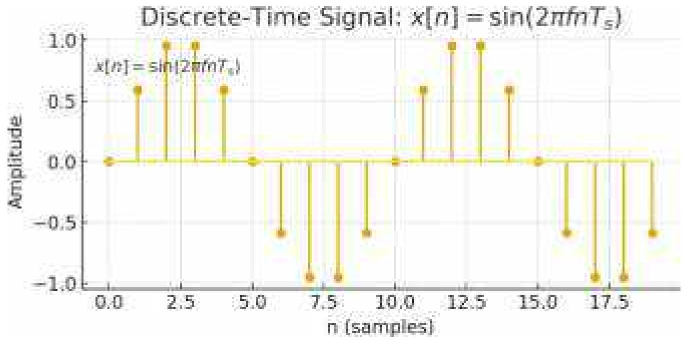
Where

- T_s = Sampling period
- $f_s = \frac{1}{T_s}$ = Sampling frequency

  Examples: Digital speech samples, sampled ECG signals, sensor data streams.

1. Basic DT Functions

- Unit Step
  $u[n] = \begin{cases} 1, & n \ge 0 \\ 0, & n < 0 \end{cases}$
- Unit Impulse
  $\delta[n] = \begin{cases} 1, & n = 0 \\ 0, & n \neq 0 \end{cases}$
- Discrete Sinusoid
  $x[n] = \sin(2\pi f n T_s)$



Discrete-time signals exist in numerical/matrix form, suitable for software processing (MATLAB, Python, DSP processors).

C. Classification of Signals

Category	Types
By Domain	Continuous-Time, Discrete-Time
By Amplitude	Continuous-valued, Discrete-valued
By Causality	Causal, Anti-Causal, Non-Causal
By Time Property	Static, Dynamic; Time-invariant, variant
By Energy	Energy signals, Power signals

IV. METHODOLOGY AND MATLAB SIMULATION ENVIRONMENT

Here the systematic approach that was used to analyze both continuous-time and discrete-time signals is being explained along with the instruments that were used, the simulation setup, and the work process that was followed to confirm the theoretical concepts by conducting experiments in MATLAB.

D. Research Methodology Overview

The methodology followed in this research is a structured simulation-and-analysis approach, which basically includes the following stages:

Identification of Signal Types:

- Sinusoidal, exponential, unit impulse, and step signals were considered as representative test examples for their fundamental role in DSP theory.

Formulation of Mathematical Models

- Both continuous-time and discrete-time equations were written analytically.

MATLAB Environment Setup

- Time vectors and sampling grids were set up for CT and DT signal.

Signal Generation

- CT and DT waveforms were generated with plot() and stem() functions.

Sampling Frequency Variation

- Different sampling frequencies were used to see Nyquist, oversampling, and under sampling conditions.

FFT and Spectrum Analysis

- An examination of the frequency-domain behavior was done through the use of fft() and fftshift().

Reconstruction Analysis

- Signal reconstruction accuracy by interpolation methods was investigated.

Result Interpretation

- A visual and quantitative assessment was done to align with the theoretical predictions.
- Such a systematic approach makes sure that every theoretical concept is substantiated by a simulation and graphical interpretation.

E. MATLAB Environment Configuration

MATLAB was selected as the simulation platform due to its matrix-based computation engine and powerful visualization toolkit.

Component	**Specification**
Software	MATLAB R2024a (Student/Academic version acceptable)
Toolboxes Used	Signal Processing Toolbox
Operating System	Windows 10 / Linux Ubuntu / MacOS compatible
Processor	Intel Core i5 or higher (recommended)
RAM	Minimum 4 GB (recommended 8–16 GB for FFT operations)

The simulation scripts were run in the MATLAB command window and Live Editor environment.

F. Simulation Workflow

Step-1: Define Signal Parameters

- Amplitude A = 1
- Frequency f = 5 Hz
- Phase ϕ = 0

Step-2: Generate CT Sinusoid *x(t)=sin(2πft)x(t) = \sin(2\pi ft)*x(t)=sin(2πft)

MATLAB command:

```
t = 0:0.0001:1;
x_t = sin(2*pi*5*t); plot(t,x_t);
```

Step-3: Sample to DT

*x[n]=sin(2πfnTs)x[n] = \sin(2\pi fnT_s)*x[n]=sin(2πfnTs
)

```
fs = 20;
n = 0:1/fs:1;
x_n = sin(2*pi*5*n); stem(n,x_n);
```

Step-3: FFT Spectrum Analysis Y = fft(x_n);
Y = fftshift(Y); plot(abs(Y));

Step-4: Aliasing Demonstration Sampling at 5 Hz (equal to signal freq): fs = 5;
n = 0:1/fs:1;
x_alias = sin(2*pi*5*n); stem(n,x_alias);

Expected result: distorted/incorrect sinusoid behavior.

Step-5: Signal Reconstruction

Interpolation using **sinc** method or MATLAB built-in functions (interp1, resample, etc.):

```
t_interp = 0:0.0001:1;
x_interp = interp1(n, x_n, t_interp, 'spline');
```

G. Data Acquisition and Validation Approach

To ensure experimental accuracy:

- Each simulation repeated with multiple sampling rates
- FFT compared against analytic frequency response

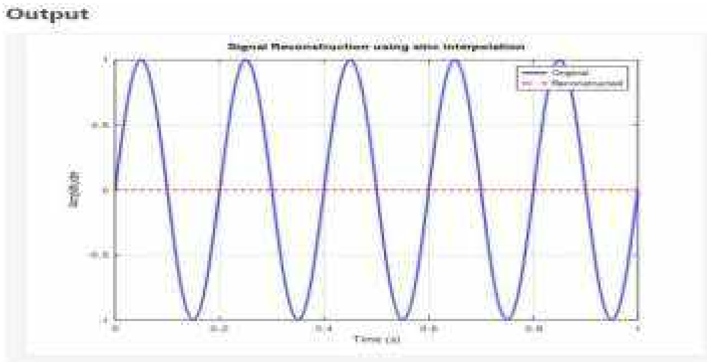


Figure: 1. output

V. CONCLUSION

Using MATLAB to analyze signals in both continuous and discrete time visually illustrates the core concepts of digital signal processing. This study starts with creating and displaying analog signals and then sampling them at various rates to show how the signals behave when they are changed from the continuous-time domain to the discrete-time domain. The findings emphasize the role of the Nyquist sampling theorem, indicating that sampling at or above twice the signal frequency retains the waveform most accurately, while sampling below this limit results in aliasing and distortion. Moreover, the frequency analysis based on FFT illustrates the way in which discrete signals can be represented in the frequency domain, thus, time-domain and frequency- domain representations being interrelated is a concept that is strengthened further. The rebuilding of signals via sine interpolation is a method of demonstrating how the original continuous waveform can be inferred from the discrete samples, hence, the principles of digital-to-analog conversion are confirmed. This project, in essence, portrays the fundamental ideas of signal generation, sampling, aliasing, frequency analysis, and reconstruction. It positions MATLAB as a powerful tool for the visualization and comprehension of digital signal processing concepts, thereby, making these theoretical principles more accessible for interpretation and application in the fields of communication and control systems.

REFERENCES

[1] V. Oppenheim, A. S. Willsky, and S. H. Nawab,*Signals and Systems*, 2nd ed., Prentice-Hall, 1996.

[2] J. G. Proakis and D. G. Manolakis, *Digital Signal Processing: Principles, Algorithms and Applications*, 4th ed., Pearson Education, 2007.

[3] A. V. Oppenheim and R. W. Schafer, *Discrete- Time Signal Processing*, 3rd ed., Pearson Education, 2013.

[4] S. Haykin and B. Van Veen, *Signals and Systems*, Wiley, 2005.

[5] S. K. Mitra, *Digital Signal Processing: A Computer-Based Approach*, 4th ed., McGraw-Hill, 2011.

[6] H. Nyquist, "Certain Topics in Telegraph Transmission Theory," *Transactions of the AIEE*, vol. 47, pp. 617–644, 1928.

[7] C. E. Shannon, "Communication in the Presence of Noise," *IRE Proceedings*, vol. 37, no. 1, pp. 10– 21, 1949.

[8] MathWorks Inc., *MATLAB Documentation and Signal Processing Toolbox User Guide*, 2024. *(Available online: mathworks.com. Accessed: 2025)*