# Digital Signal Processing (EC 335)

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# About the course

- Name: Digital Signal Processing
- Code: EC 335
- LMS enrollment code: **769421035**
- The course provides an insight to theory and applications of DSP.
- Upon completion of this course, you should have a solid information in the basics of DSP related to Signals and System Analysis and Design.
- The presented material will describe DSP Techniques, applications, and implementations.
- The course is intended to familiarize the audience with active areas of DSP development, and provide direction of further investigation.

# Learning Outcomes

| S.No | Outcomes | Level of Learning | PLO |
|---|---|---|---|
| 1 | Investigate the different classes of problems in digital signal processing | C4 | 4 |
| 2 | Application of various DSP algorithms such as convolution and frequency-time domain transforms to design filters | C3 | 3 |
| 3 | Investigate and report on a case study that explores the effect of Digital Signal Processing on environment or/and sustainability | C4 | 7 |

# Books

**TEXT BOOKS:**

1. Understanding Digital Signal Processing by Richard G. Lyons (Latest Edition)
2. The Scientist and Engineer's guide to Digital Signal Processing by Steven. W. Smith (Latest Edition)

**REFERENCE BOOKS:**

1. Discrete Time Signal Processing, 2nd Edition (Oppenheim, schafar with John R. Buck)
2. Digital Processing: A computer Based Approach (Sanjit K. Mitra, 2nd Edition)

# Grading System

| S.No | Nature of exam | Weightage |
|---|---|---|
| 1 | Mid Semester Exam | 30% |
| 2 | Assignments | 5% |
| 3 | Quizzes | 10% |
| 4 | Final Semester Exam | 45% |
| 5 | Project | 10% |

# Project

**Course project will be carried out as a group of 3 students each, and the project is divided into three modules**

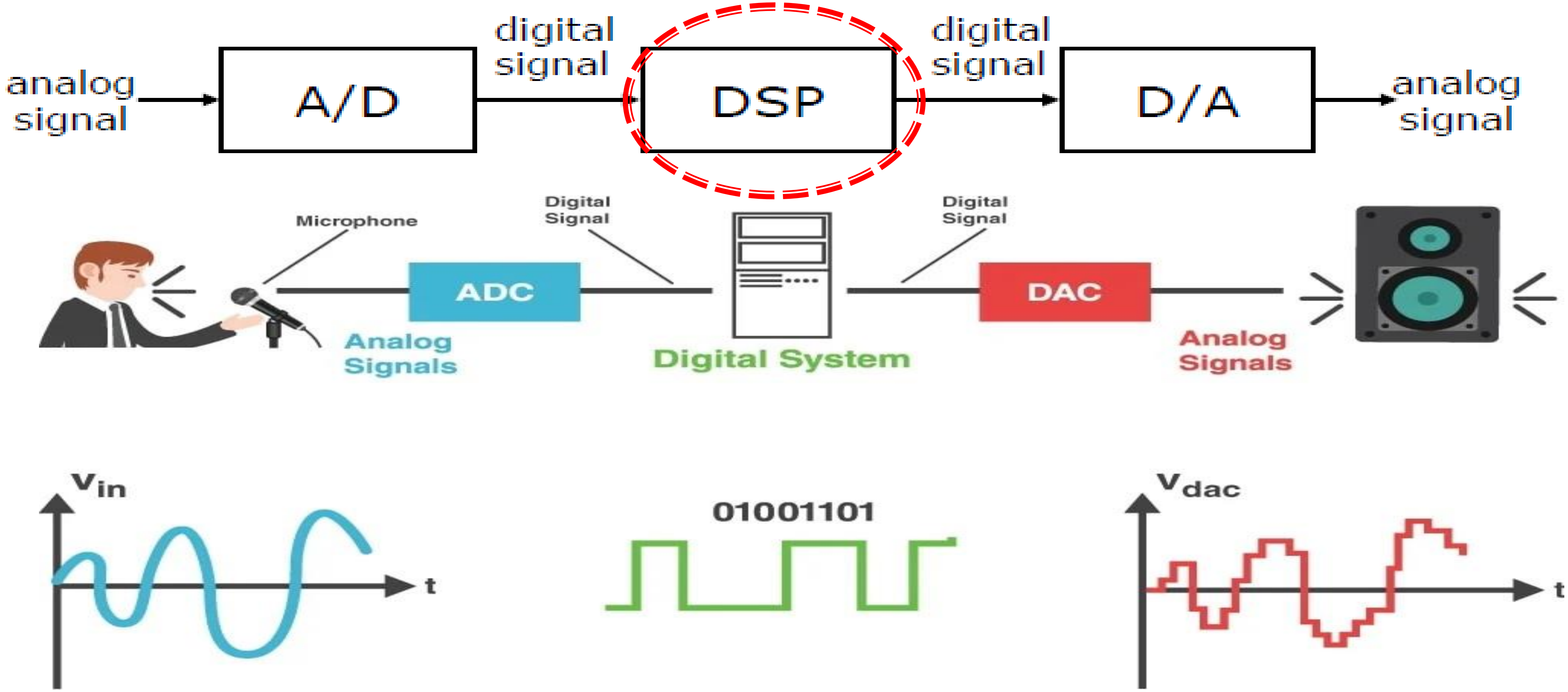
**Module 1: Proposal writing (Deadline 4$^{th}$ week) [Marks: 25%]**

**Module 2: Project presentation, and Hardware/simulation in running condition (Deadline 12th week), [Marks (15% + 35%)]**

**Module 3: Final project report in IEEE template, (Deadline 14th week), [Marks 25%]. The similarity index of the final project report should not exceed 20%.**

# Overview of Digital Signal Processing

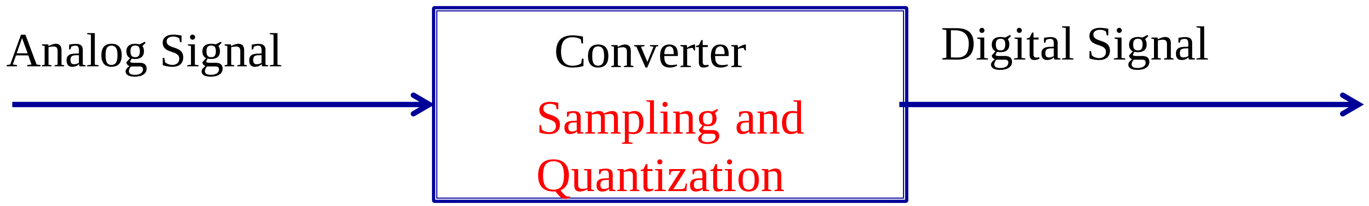
# Digital Signal Processing




https://www.allaboutcircuits.com/technical-articles/an-introduction-to-digital-signal-processing/

# Digital Signal Processing

- We will process Digital Signals
- **What is Signal?**
- **What is the nature of the Signal? Analog or Digital???**
- Most of the practical Signals are Analog.
- **What should be the first step??**
- **Conversion from Analog to Digital…..**




- **Why not process signals in Analog domain???**
- There should be some advantages of processing Digital signal….

# WHY USE DSP ?

- **Versatility:**
  - digital systems can be reprogrammed for other applications
  - digital systems can be ported to different hardware
- **Repeatability:**
  - digital systems can be easily duplicated
  - digital systems do not depend on strict component tolerances
  - digital system responses do not drift with temperature
- **Simplicity:**
  - some things can be done more easily digitally than with analogue systems

# History of DSP

- In 1960s, signal processing was mainly used in defense applications (Radar, Sonar)
- Scientist took Fourier transform of the Radar signal which reflected back after hitting the target, where the computation time was approximately half an hour.
- Cooley and Tuckey in 1965 bring the major advancement in signal processing by demonstrating that same Fourier transform can be computed in few seconds (Cooley and Tuckey FFT), bringing DSP to real life.

# Cooley and Tuckey, 1965

## An Algorithm for the Machine Calculation of Complex Fourier Series

By James W. Cooley and John W. Tukey

An efficient method for the calculation of the interactions of a $2^m$ factorial experiment was introduced by Yates and is widely known by his name. The generalization to $3^m$ was given by Box et al. [1]. Good [2] generalized these methods and gave elegant algorithms for which one class of applications is the calculation of Fourier series. In their full generality, Good's methods are applicable to certain problems in which one must multiply an $N$-vector by an $N \times N$ matrix which can be factored into $m$ sparse matrices, where $m$ is proportional to $\log N$. This results in a procedure requiring a number of operations proportional to $N \log N$ rather than $N^2$. These methods are applied here to the calculation of complex Fourier series. They are useful in situations where the number of data points is, or can be chosen to be, a highly composite number. The algorithm is here derived and presented in a rather different form. Attention is given to the choice of $N$. It is also shown how special advantage can be obtained in the use of a binary computer with $N = 2^m$ and how the entire calculation can be performed within the array of $N$ data storage locations used for the given Fourier coefficients.

# Oppenheim , Schaffer…..

Terminology in digital signal processing
L Rabiner, J Cooley, H Helms, L Jackson, J Kaiser, C Rader, R Schafer, K Steiglitz…
IEEE Transactions on Audio and Electroacoustics, 1972 • ieeexplore.ieee.org

The committee on Digital Signal Processing of the IEEE Group on Audio and Electroacoustics has undertaken the project of recommending terminology for use in papers and texts on digital signal processing. The reasons for this project are twofold. First, the meanings of many terms that are commonly used differ from one author to another. Second, there are many terms that one would like to have defined for which no standard term currently exists. It is the purpose of this paper to propose terminology which we feel is self-consistent, and which is in reasonably good agreement with current practices. An alphabetic index of terms is included at the end of the paper.

A survey of digital speech processing techniques
R Schafer
IEEE Transactions on Audio and Electroacoustics, 1972 • ieeexplore.ieee.org

Digital signal processing techniques are becoming increasingly important in speech analysis and synthesis. These techniques can be implemented using a general purpose computer facility (often not in real time), or special purpose hardware realizations can be constructed. This paper discusses some recent work in speech processing including design of digital filter bank spectrum analyzers, homorphic analyzers of speech, predictive coding, and hardware realization of a digital formant synthesizer. The survey concentrates on those speech processing techniques relevant to the development of sensory aids for the deaf.

**Homomorphic analysis** of **speech**
A Oppenheim, R Schafer - IEEE Transactions on Audio and …, 2003 - ieeexplore.ieee.org
… and termed "**homomorphic** systems," emphasizing their interpretation as **homomorphic** (ie, …
In this paper, an approach to deconvolution of **speech**, based on these ideas, is discussed. …

Chebyshev approximation for nonrecursive digital filters with linear phase
T Parks, J McClellan
IEEE Transactions on circuit theory, 1972 • ieeexplore.ieee.org

An efficient procedure for the design of finite-length impulse response filters with linear phase is presented. The algorithm obtains the optimum Chebyshev approximation on

A **unified approach** to the **design** of **optimum FIR linear-phase digital filters**
J McClellan, T Parks - IEEE Transactions on Circuit Theory, 2003 - ieeexplore.ieee.org
… types and the general procedure (**linear** programming) is comparatively slow. The objective
… the **linearphase FIR** falter **design** problem and to **unify** the **design** of the four types of **filters** in …

# Other Applications

A survey of signal processing problems and tools in holographic three-dimensional television
L Onural, A Gotchev, HM Ozaktas, E Stoykova
IEEE Transactions on Circuits and Systems for Video Technology, 2007 • ieeexplore.ieee.org

Diffraction and holography are fertile areas for application of signal theory and processing. Recent work on 3DTV displays has posed particularly challenging signal processing problems. Various procedures to compute Rayleigh-Sommerfeld, Fresnel and Fraunhofer diffraction exist in the literature. Diffraction between parallel planes and tilted planes can

Power and area minimization for multidimensional signal processing
D Markovic, B Nikolic, RW Brodersen
IEEE Journal of Solid-State Circuits, 2007 • ieeexplore.ieee.org

Sensitivity-based methodology is applied to optimization of performance, power and area across several levels of design abstraction for a complex wireless baseband signal processing algorithm. The design framework is based on a unified, block-based graphical

A subsampling radio architecture for ultrawideband communications
SWM Chen, RW Brodersen
IEEE transactions on signal processing, 2007 • ieeexplore.ieee.org

An impulse radio architecture utilizing a simple analog front end along with digital complex signal processing is proposed to allow a low complexity implementation of a 3.1–10.6 GHz ultrawideband (UWB) radio. The proposed system transmits passband pulses using a pulser and antenna, and the receiver front-end downconverts the signal frequency via

**Signal processing** for **biometric systems** [DSP Forum]
AK Jain, R Chellappa, SC Draper... - ... **Signal Processing** ..., 2007 - ieeexplore.ieee.org
... **systems**. Let us discuss each and outline the associated **signal processing** challenges...
To begin with, why are we concerned with securing **biometric systems** and **biometric** data? ...

The digital filter and speech communication
J Kaiser
IEEE Transactions on Audio and Electroacoustics, 2003 • ieeexplore.ieee.org

The study of speech communication and speech processing systems by simulation on a digital computer, and the design and physical implementation of these systems with discrete components, require a detailed knowledge of the elements known as digital

Frame-to-frame coding of television pictures using two-dimensional Fourier transforms (Corresp.)
B Haskell
IEEE Transactions on Information Theory, 2003 • ieeexplore.ieee.org

Two systems are described for reducing the data rate required to transmit TV signals. Basic to the operation of both systems is the fact that if there is an object in the scene moving more or less in linear translation, then the two-dimensional frame-difference signal

An **experimental 9600-bits/s voice digitizer** employing adaptive prediction
J Dunn
IEEE Transactions on Communication Technology, 1971 • ieeexplore.ieee.org

An experimental model of a coder for transmission of speech over a 9600-bits/s digital channel was built to demonstrate feasibility of an adaptive prediction-coding technique. After analog-to-digital conversion of the speech input, the coder employs digital processing using a computer type organization. Resonances in the short-term speech

# Other Applications

A comprehensive overview on signal processing and artificial intelligence techniques applications in classification of power quality disturbances
S Khokhar, AABM Zin, ASB Mokhtar, M Pesaran
Renewable and Sustainable Energy Reviews, 2015 • Elsevier

**Abstract**
The increasing trend towards renewable energy sources requires higher power quality (PQ) at the generation, transmission and distribution systems. The PQ disturbances are produced due to the nonlinear loads, power electronic converters, system faults and switching events. The utilities and consumers of electric power are expected to acquire ideal voltage and current waveforms at rated power frequency. The development of new

Graph Signal Processing (GSP) is an extension of classical Digital Signal Processing (DSP) to data that lives on irregular structures (graphs) instead of regular domains like time (1D signals) or space (2D images).
In classical DSP: signals are vectors indexed by time or pixels.
In GSP: signals are values defined on the nodes of a graph (e.g., temperature readings at different sensors, activity at brain regions, traffic at city intersections).

**Graph signal processing**: Overview, challenges, and applications
A Ortega, P Frossard, J Kovačević… - Proceedings of the …, 2018 - ieeexplore.ieee.org
… can think of smooth **graph signals** in the vertex domain, that is, **signals** where neighboring nodes tend to have similar values. We can also think of the smoothness of **graph signals** in the …

[HTML] Analysis of **Eeg** Data Using Different Techniques of **Digital Signal Processing**
MM Siddiqui, MS Kidwai… - Biomedical and …, 2024 - biomedpharmajournal.org
… use of **digital signal processing** (**DSP**) techniques to analyses **EEG** data in sleep apnea patients. **DSP** is a set of mathematical algorithms used to process signals, such as **EEG** signals. …

**EEG**-Based Signal Processing Framework for Self-Organized Criticality Detection
M Illeperuma, V De Silva, R Pina - … on **Digital Signal Processing** …, 2025 - ieeexplore.ieee.org
… **Digital signal processing** frameworks have enabled the study of brain activity using more cost-effective and mobile methods such as **electroencephalography** (EEG). EEG signal …

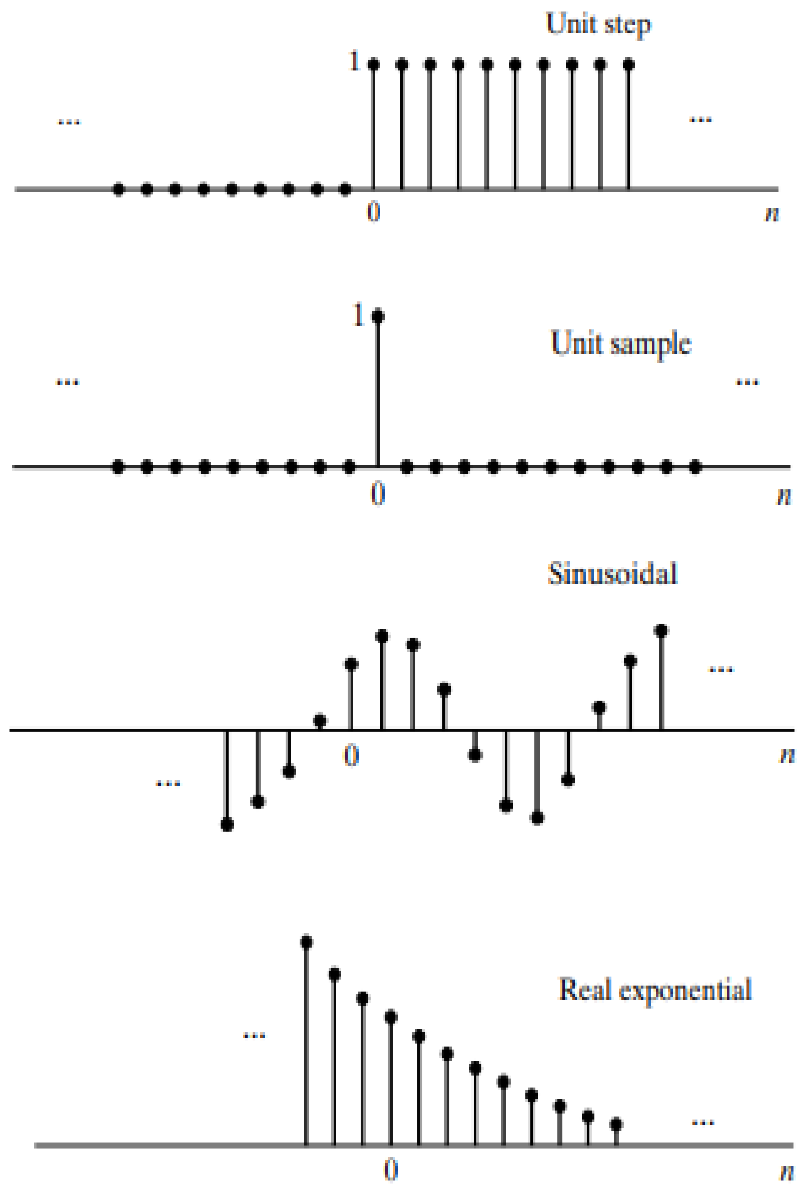
# Basic Types of Digital Signals

Unit Step Signal

Impulse signal

Cosine/Sine Signal

Exponential Signal



# Basic Operation on Signals

Delay

Take any signal $x[n]$, find $x[n-2]$???

-Delay is inherent in all signal processing system. We will acquire signal (samples), then process the signal (samples), and can play the signal. In most signal processing system, we will minimize the delay. If the delay is greater than a certain threshold, then it will cause degradation of the signal.

Addition

Subtraction

Multiplication

# Representation of Discrete Signal

Any Discrete signal can be represented as the summation of the weighted shifted impulses

Example:

$$x[n] = \sum_{k=-\infty}^{\infty} x[k]\delta[n-k]$$

# Homework

Read Chapter 1 of the book "The Scientist and Engineer's guide to Digital Signal Processing by Steven. W. Smith"