Digital Signal Processing

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Lecture - 1

Digital Signal Processing Introduction

Welcome to this course on Digital Signal Processing being taught by me. My name is S.C. Dutta Roy and I am an emeritus professor here. This is the first lecture in which we are going to introduce Digital Signal Processing in general and the course in particular.

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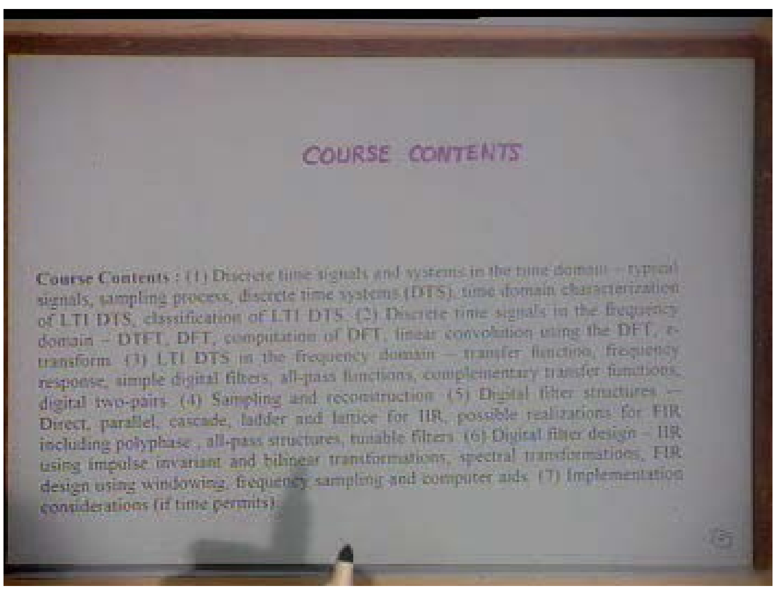

Is the course content readable on the screen? It is okay. I shall read it out. The course contents are given in the sheet that I have circulated. This is the span of the course. Let me tell you I do not expect any prior knowledge of DSP from you. All I assume is that you are acquainted with signals and systems.

However, at least one third of the course shall be devoted to a review of signals and systems because it is extremely important that you understand discrete time signals and discrete time systems and what happens when discrete time signals interact with discrete time systems. The first two topics are: review of signals and systems and discrete time signals and systems in the time domain. We shall discuss typical signals. We shall go into the sampling process and then we shall characterize discrete time systems. We shall then introduce the special class of linear time invariant systems which I abbreviate as LTI and discuss classification of LTI discrete time systems. This will be followed by discrete time signals description or characterization in the frequency domain through various kinds of transforms and we shall introduce the discrete time Fourier transform and Discrete Fourier Transform, abbreviated as DFT.

We shall talk about the computation of DFT; in other words, we shall go into the basics of FFT. We shall talk about linear convolution using DFT and also circular convolution using DFT. Then we shall introduce some new techniques which are not available in the book. We shall introduce Z-transforms in detail and their application in characterizing linear time invariant discrete time systems i.e. system description in the frequency domain. In other words, we shall talk about transfer functions and frequency response. Then we shall introduce simple digital filters, all-pass functions, complementary transfer functions and then digital two-pairs.

We shall go back to sampling and reconstruction and have a small discussion on that. Then will go into digital filter structures that are direct, parallel, cascade, ladder and lattice for Infinite Impulse Response (IIR) filters and possible realizations for FIR or Finite Impulse Response filters, including poly phase. All-pass structures and tunable filters will follow this. We shall spend a considerable amount of time on digital filter design: infinite impulse response using impulse invariant and bilinear transformations, and finite impulse response filters using windowing and frequency response sampling techniques. A short discussion on computer aided design will also be included.

We shall discuss spectral transformations for IIR design. If time permits, we shall also have a brief discussion on implementation considerations. This is what we aim to cover(?). At IITs, we do not cover a course; we uncover at least certain parts of the course. This is the aim and it does not matter how much we can do, but what we do must get imprinted in your mind.

The books to consult: I have adapted S. K. Mitra, Digital Signal Processing - a computational approach, McGraw Hill 2000. This is the second edition; I would advice you to buy the third edition, which is available. A 4th edition is also due to be published. There are major changes which were effected by Professor Mitra in the later editions at the suggestions of a few people, including me.

Those who have difficulty with signals and systems are advised to buy a copy of Oppenheim and Schafer, "Discrete Time Signal Processing." This is a Prentice Hall publication, 2000. This is about what we wish to do in the course.

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Since the topic is digital signal processing we shall explain the meaning of the individual terms and then what the combination means. First we concentrate on what is a signal. These are formal definitions and you must understand them very clearly.

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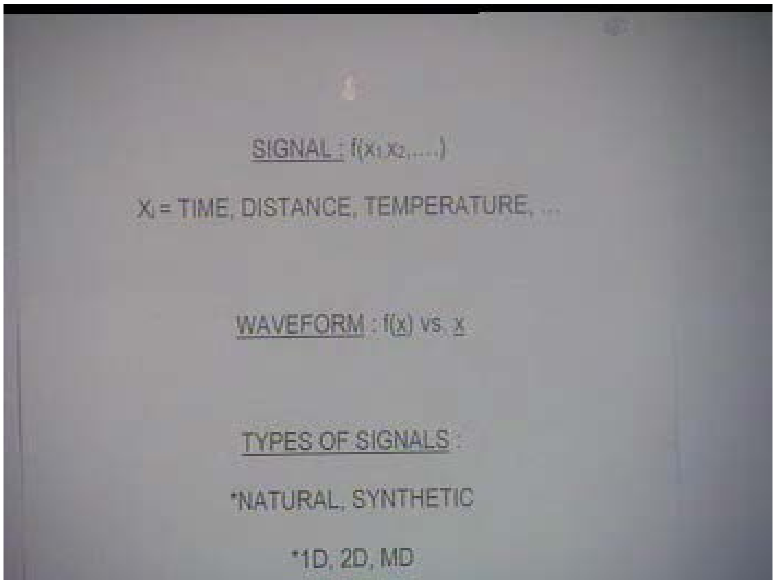

A signal in mathematical terms is a function. A function is a dependent variable of some variables which are independent variables. The number of independent variables can be one or more.

In general, a signal is a function of several variables. These variables are x_1, x_2, x_3 and so on. These variables, for example could be time, distance, temperature or any other physical parameter. In this course we shall mostly be concerned with a function of a single variable and that variable will be time. But because we are talking of digital signal processing, time shall also lose its significance.

Our functions will be functions of numbers and numbers are also restricted to be integers. In other words, in DSP the type of signal that we shall be concerned with are functions of a variable n, where n can take only integer values that are positive or negative. n can be – 15 or it can be 0 or + 13 or + 14, but n = 13.5 is not permitted because time is discretized.

If we plot this function or signal versus the variables, the resulting plot is called a waveform. If the function has only one variable, then a two dimensional picture in a graph paper suffices. The ordinary sinusoidal waveform that you draw on paper is a one dimensional signal and this is called a waveform. In general the waveform can be multidimensional, depending on the number of independent variables.

For example, a picture, which is said to be worth more than 1000 words, has two dimensions. A picture should ideally be three dimensional, because it has three space variables. Dependent variable in an image can be brightness, color, density or it can be any other thing.

So signals can be one dimensional, two dimensional or multi-dimensional. However, we shall be concerned only with one dimensional signal that is f (n). Signals can be natural, for example a thunderstorm or a lightning which are natural phenomena or it can be synthetic. Signals can be generated in the laboratory for communication purposes. Signals can be either analog or discrete. It is the discrete type signals that we are concerned with in this course.

One common confusion is that all continuous signals are also called analog signals. All continuous time signals are analog signals but all analog signals are not continuous time. If the time is discretized but not the amplitude, i.e. If the independent variable is discretized but not the dependent variable, then it is still an analog signal.

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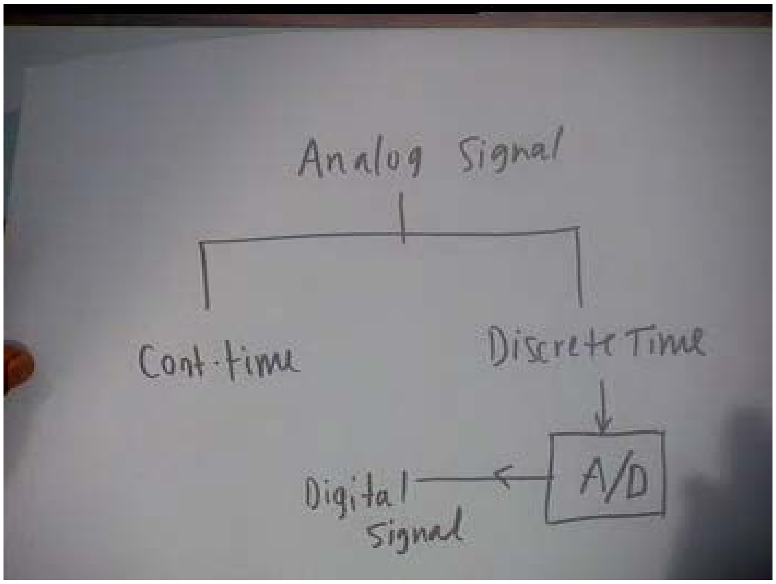

Therefore an analog signal can be either continuous time or discrete time. Discrete time signals are also analog signals. If a discrete time signal is quantized, that is, if in a discrete time signal where the independent variable has been discretized, the amplitude is also discretized, i.e. which is allowed to take on only certain specific values, then it is said to be discretized or quantized. So if a discrete time signal passes through A (analog) to D (digital) converter then depending on the number q of bits, 2 to the power q discrete amplitudes are possible; for example, a three bit A to D converter gives 8 possible amplitudes.

After A to D conversion, the signal is also coded in some form and the most usual form is the binary form. After discretization, the signal becomes a binary number and that is what we call a digital signal. I repeat here that analog and continuous time signals are not one and the same.

As I have said earlier all continuous time signals are analog signals, but all analog signals are not continuous time. Analog signals can be continuous time and also discrete time. If the discretization is limited to the independent variable or the x axis only, the signal still remains analog. The digital signal can be obtained only when both the independent and dependent variables are discretized. A discrete time signal is analog and a digital signal is one in which the amplitude is also discretized. Why should we study signals at all? We study about signals because they carry information. It is information that drives the whole world, the basic steps being generation, transmission, reception, absorption, and action on the basis of information.

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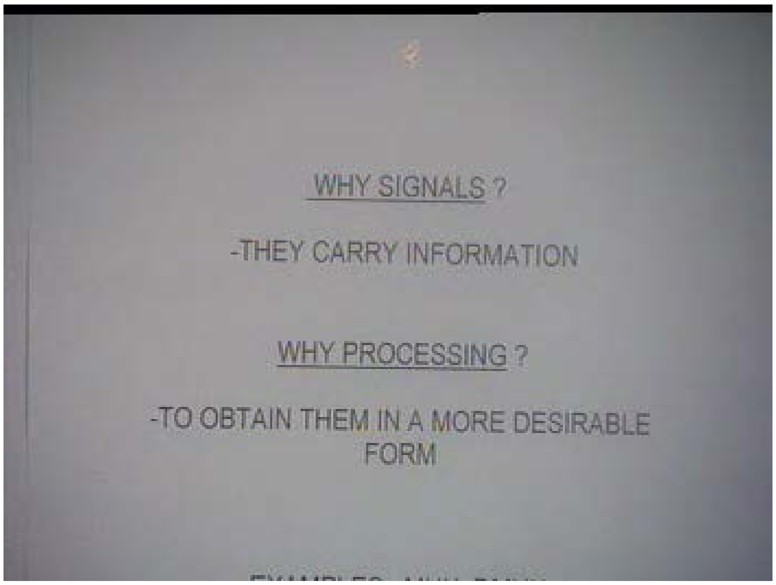

Information is the basic thread of life and therefore we are interested in signals. Why processing? What is the need for processing? Processing is done to obtain the given signal in a more desirable form. For example, if there are a number of signals which are to be transmitted over the same channel, then we do a processing called multiplexing. At the receiver end the signals have to be separated and therefore there is a need for de-multiplexing.

Invariably the signal is corrupted by noise. Therefore our processing may aim at filtering the noise out. I shall illustrate this with the help of a specific case, viz. the electrocardiogram (ECG) waveform.

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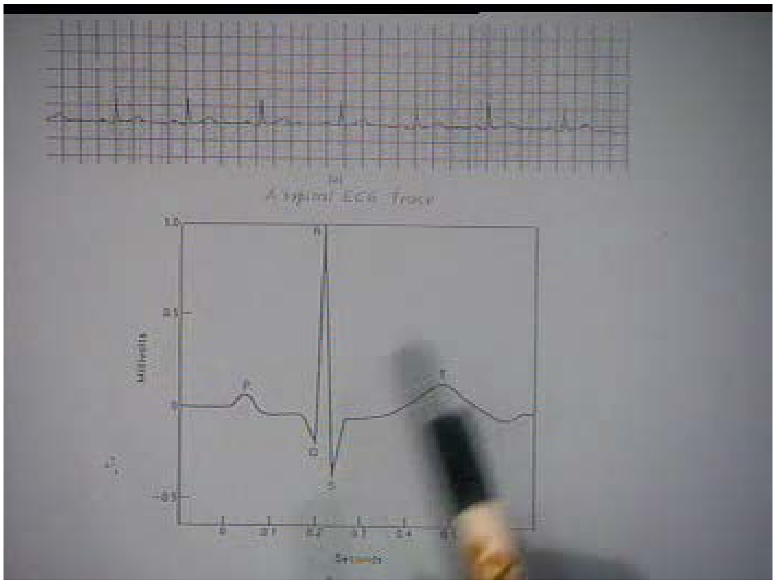

Electrocardiogram waveform usually looks like this. If you take one cycle and blow it up, it will look like this. Doctors are interested where the peaks and dips occur. The standard language used by doctors for them are PQRST. This is a very clean trace that you see here. If this signal is corrupted, the PQRST points may be blurred and difficult to detect.

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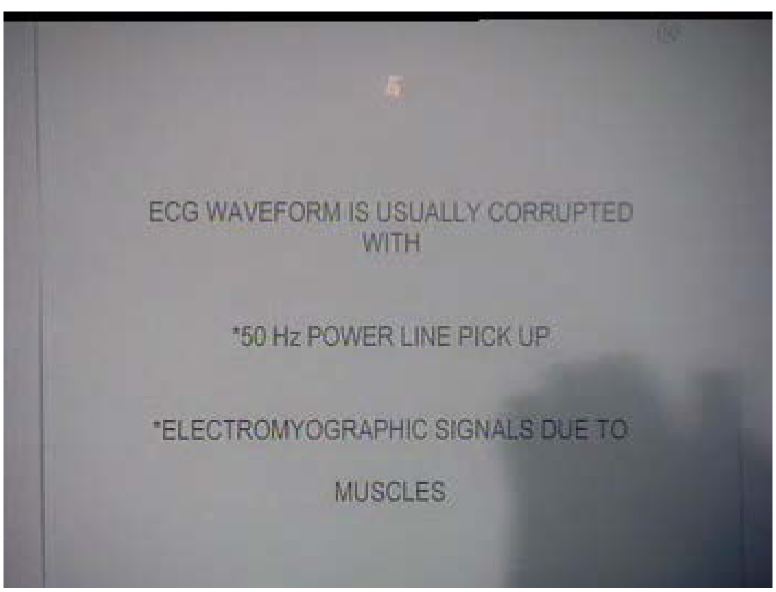

For example, the ECG waveform is usually corrupted with 50 Hertz Power Line Pick Up. The ever present power, 230 volts 50 Hertz can be induced from any nearby equipment. It can be a magnetic pick up or it can be an electrostatic pick up. It can be due to mutual capacitance or mutual inductance. This is ever present and you may like to get rid of the 50 Hertz before you look at the signal. This is done by a notch filter or a band elimination filter which we shall study in this course.

There are also Electromyographic (EMG) signals due to the muscles. The human body is a wonderful electrical machine. If you put two electrodes at any two places of body, it will record an electric potential. The muscles themselves generate electrical signals and they might corrupt the ECG signal. Hence there is a need for processing to get a clean picture.

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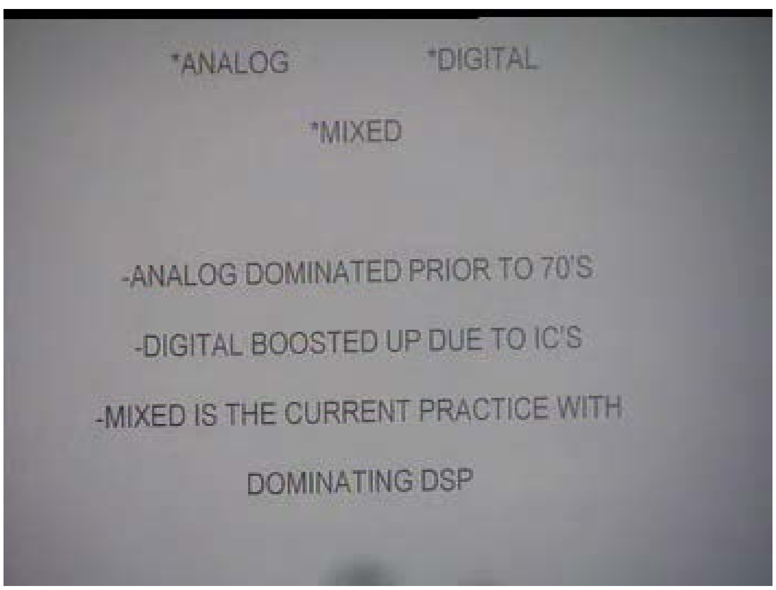

Signal processing basically can be done in three ways: one is analog, that is, do not convert the signal into digital and process it in the analog format itself. The other two are digital processing and mixed signal processing, which is partly analog and partly digital. One might ask here: what is wrong with totally analog processing?

Well, the answer to that is given by the proverb two arrows in a quiver is always better than one, because if one fails, you can use the other arrow to hit the enemy. But then why three? It is three because digital signal processing is so much more convenient and so much more accurate than analog, while most of the natural signals that we generate in the laboratory for communication are analog. Therefore you have to use mixed processing. Analog signal processing dominated the field prior to 1970's but after 1970 because of the easy availability of digital hardware in the form of integrated circuit chips at a very low cost, digital signal processing got boosted up.

The current practice is mixed signal processing, dominated by digital signal processing. In other words at any point in the signal processing chain if you find that DSP can be used, you go ahead without thinking about any other choice. Analog signal processing advanced in quantum jumps with the inventions of new devices and circuits.

We started with Acharya J.C. Bose's coherer and spark gaps. Spark gaps were used to generate signals and coherer, a galena crystal was used to detect signals.

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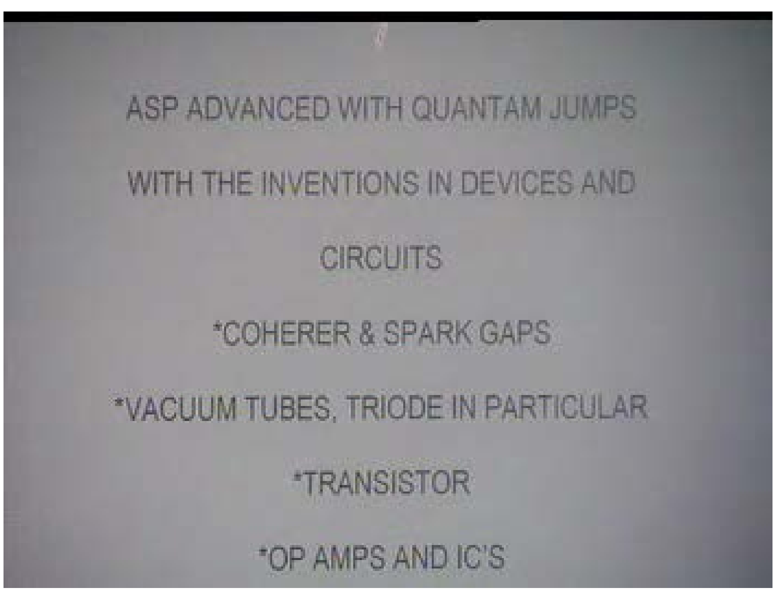

Coherer is a unipolar device and is a crude form of a diode as we know it now. That was what we started with. Marconi's patent on telegraph signals also used a kind of spark gaps with ON/OFF control. He also used an improved form of coherer to detect signals.

Then vacuum tubes came in the picture, triode in particular, which could amplify a weak signal. Then in 1947 transistors came in the picture which made things smaller and made life simpler because circuits were more designable with smaller power supplies. A 5 volt or 12 volt power supply does the job.

Further reduction in size was made possible due to the introduction of operational amplifiers and other integrated circuits, analog as well as digital. Then there occurred this revolution of small scale integrated circuits, medium scale integrated circuits, large scale integrated circuits and finally VLSI, SSI etc. In fact, IC's are only limited by ones imagination as to how much can you go. However, even today, advances in analog signal processing are going in full swing because most signals of interest are analog. The real life signals like the speech, music etc, are analog. The ultimate desired output in most cases is also analog.

Digital telephony has come in. You can process speech digitally and transmit digitally but ultimately you cannot hear 1s and 0s; you have to convert into analog signal. Therefore analog signal processing is also advancing and there is a lot of research that is going on. Mixed signal processing is the trend of the day.