Signal Processing Techniques and Its Applications
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Lecture - 01
Introduction

Welcome to the NPTEL Online Certification Course Signal Processing Techniques and its Applications. So, this course is about a 12-week course. So, there will be a 12-week, 30-hour lecture content, and I will cover the topic. The topic will be given to you every day, every week, the title will be given to you, and that assignment will also given to you. However, the main purpose of this course is not just to give you the theory.

So, what I want from your side is that you go through the theory and try to find out its application; it means not only that I only read these lectures and I try to solve that assignment purpose is not; only in remembering information theory, which I will discuss in this course. You cannot solve that assignment problem. So, you have to understand it. Understanding it means you have to develop cognitive skills, not just remembering.

So, in this course, signal processing, most of the students think that signal processing is nothing but a mathematical subject, so there are a lot of mathematics courses. So, here I will be told that not only the mathematics, but you have to know the inside of mathematics, why I write this mathematics, and what the physical significance of that mathematics is. Most of you have already covered the Z-transform. If I ask you why the Z-transformer is required, what is the purpose of the Z-transform?

There is a lot of transformation there. So, why is a particular Z-transform required for signal processing, and what do we do with the Z-transform? So, it is not that given a transfer function, ok, I also said give me a Laplace transform equation I will convert to the Z-transform; give me a time domain function I will convert to the Z-transform, that is not the skill in the signal processing domain. What do I want? I want you to know that once I transfer it, what physically happens in that part is very important.

So, the theory and many concepts I will cover in this application are the digital signal processing application techniques and their application, like the discrete system and LTI system, but everything you have to learn is not that definition of a linear time-invariant system. It is not the linear time invariants; it is the inside of the LTI system. What is LTI? What kind of visualisation do you have? Once, I heard that the LTI system was the kind of cognitive skill I wanted from you at the end of the course.

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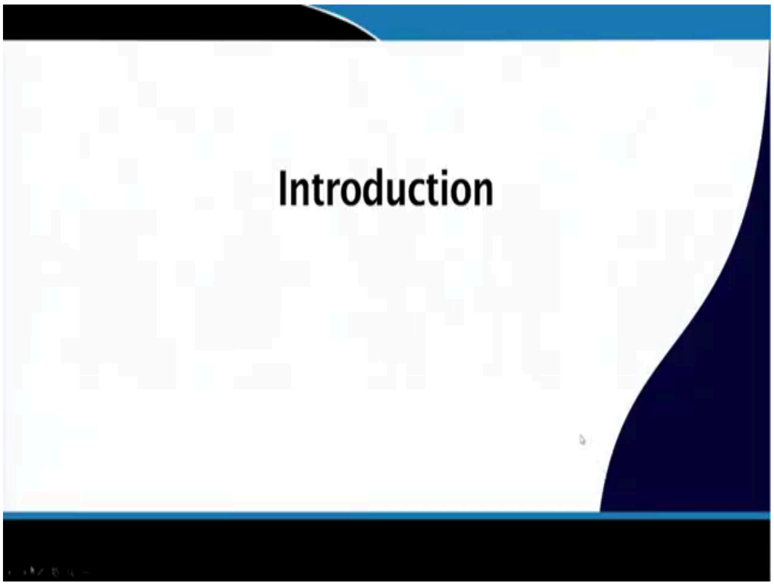

So, the course coverage will be more or less like this.

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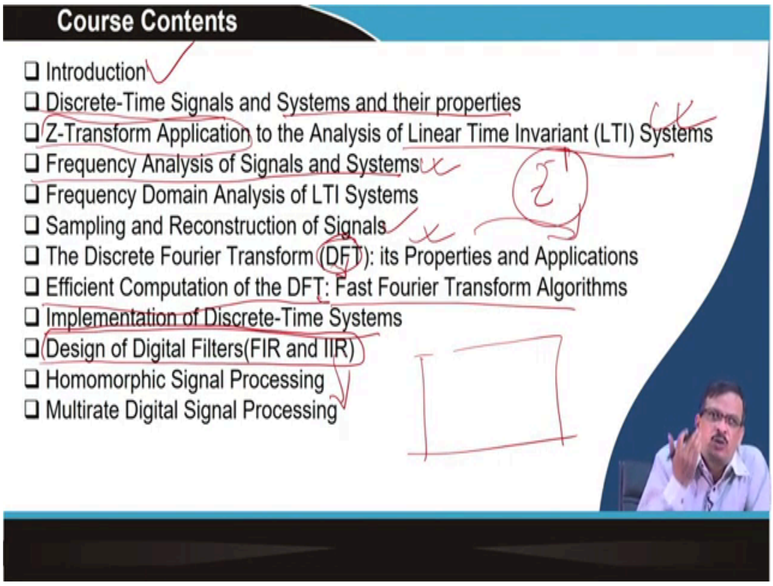

That introduction is the first lecture which I am giving that introduction lecture. So, I will give you the introduction part. Discrete-time signals and systems and their properties. So, I will discuss what discrete-time signals are, what discrete-time systems are, and what their properties are.

You said the signal is a time-invariant signal, signal is a time-variant signal – what do mean by that? What kind of visualisation do you have once you look at the signal you can say ok, this signal is a time-invariant signal; this signal is time in time-variant signal. So, you have to say that. So, that kind of skill you have to develop.

Then, I just use the Z-transform application; I will not cover the general Z-transform. I will only cover the application of Z-transform in LTI systems or the application of Z-transform what is the implementation of LTI systems. If I ask you what is the meaning of a Z^{-1}. You said ok Z^{-1} signal processing meaning is that one sample delay, why? Why do you mean by one sample delay?

So, I will cover that part in Z-transform. I have not covered the normal Z-transform that you learned in mathematics. Then, the frequency analysis of signals and systems. So, what do you mean by the frequency of a signal? I have a signal; what is the frequency of a signal? So, the concept of frequency you have to know and then the signal and systems I have a system what are the frequency domain representation of that system because of the one property of the system. So, that, is what I will cover here.

Then the frequency domain analysis of LTI system Linear Time Invariant system frequency domain analysis of LTI system we will cover. Then, sampling and reconstruction of the signal. Once you say that I have a digital signal, but in the real world, nothing is a digital signal, everything is an analog signal.

Think about you have a music system in your home and when you buy it, they say there is a digital processor is there inside that system. So, it is a very hi-fi music system. It has an active noise cancellation, it has an equalizer, digital equalizer – what do you mean by that? But, if you realise that whatever the music system inside is doing, ultimately, it is producing an analog signal.

Output is an analog signal that goes to the loudspeaker converted to the acoustic signal and input is also an analog signal or it may be a digital signal, but the digital signal also comes from that analog signal. So, somebody is singing that is analog signal singing is an acoustic signal converted to the analog signal by a microphone, and then we digitize it. So, ultimately, analog is the world; in between I have a digital.

So, how do you convert analog signal to digital signal; what are the pros and cons in there and how do you convert the digital signal to analog signal, not circuit level, at least signal processing domain up to what do you mean by a digital signal, what are the restriction is there in a digital signal. So, that part I will cover.

Then, I cover discrete Fourier transform. Discrete Fourier transform all of you know that Fourier transform is the frequency domain analysis. So, I discrete Fourier transform, then discrete domain, discrete-time, and discrete frequency those concepts I will cover and I have not only covered the DFT Discrete Fourier Transform but also the properties of DFT and where I can apply those properties.

Suppose I told you to write a program for the spectrogram I have a time domain signal, you have to write a program that can display the spectrogram of the signal. So, what kind of discrete Fourier transform I will use and how I represent it in that application ultimately it requires a spectrogram. I may know much mathematics of DFT, but unless I am able to apply it what is the meaning of that knowing?

So, you have to know that part. Then I covered the efficient computation of the DFT that you know the Fast Fourier Transform algorithm FFT algorithm. So, I will show you how it is to make it first computation of DFT make first and how it will be implemented in a computer or in hardware. Then the implementation of a discrete-time system supposes, I give you a discrete system I told you to implement it using a computer. So, how do you do that? So, that part is also covered by this course.

And, then, the design of digital filters design of digital filter both FIR and IIR. I will not cover all kinds of IIR, but some kinds of IIR filters I will cover, and then basically, I want to cover the lattice design part. That part is very applicable even if you see that V-coder. You know that big voice coder where the IIR filter implementation using lattice filter is, so I will cover that part.

Then, I cover the homomorphic signal crossing and multirate signal processing. So, I am not covering that adaptive filter design part purposefully, but if I found that 30 hours time is not completed or you and I have time, then I try to cover the adaptive filter design also. So, this is more or less your course content, which we will cover in this course.

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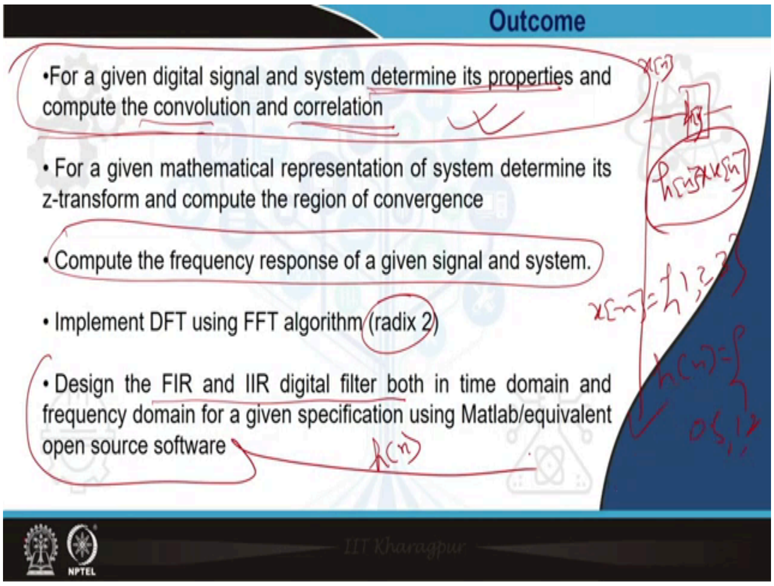

But, what is expected from you as an outcome as a student of this course you have to achieve the outcome. I have written some outcomes, but another two or three outcomes may be added for multirate signal processing and homomorphic signal processing, and I will add one or two from the application side.

So, what do you mean by this outcome? I said that you may say, ok I have gone through the course I do the exam, I get a 90 percent, and then after two days, I forgot everything. So, that means your learning is only memorisation; memorisation is also a cognitive skill, but it is a lower cognitive skill; you have to convert it to the upper cognitive domain skill.

So, what do I want? That I want once you learn, you should not forget it for your lifetime. If I told you to remember the value of pi, you may say after two days, I may forgot it, but if I told you something else not that remembering, apply or do, something or analyse something or design something once you learn the design not only for exam you can given any design at any time you can do it. So, those are the tangible outcomes I want from you.

So, at the end of the course, for a given digital signal and system determine its properties and compute the convolution and correlation. Any digital signal if I give you suppose I told you that when you join in a company or join in an application side you are required to correlate the two digital signals; you may say ok I have heard it, but how do I implement I do not know.

So, here I want, at the end of the course, you should able to write a program to calculate the convolution and correlation for a digital signal and determine the properties of a digital signal, whether it is a time-invariant signal or time variant signal, what is the periodicity of the signal, all kinds of what is symmetricity of the signal, all kinds of properties of the signal system both you should able to do it.

So, not only you have to listen to the lecture, ok I have given some lectures, and you can read it from the book also, but what you require that skill. Yes, give me a digital signal, and I will be able to write a program to calculate the convolution and correlation of that signal or system.

So, when you say a signal is passing through a system, a this is a system, and this is a signal, x[n] let us say digital signal, h[n] is the system. Output is the convolution you know h[n] convolved with x[n], but how do I calculate this? Suppose I give you a signal that x[n] is equal to [1, 2, 3, 4] those are the discrete signal samples and h[n] I have given you.

Let us say [0.5, 1, 2], and then you have to calculate the convolution of x[n] and h[n]. So, I require that skill you should develop for a given mathematical representation of a system that determines its Z-transform and computes the region of conversion. Compute the frequency response of a given signal and system.

Implement – you have to implement not only remember what is DFT what is FFT. Implement DFT using the FFT algorithm in radix-2 or radix-4, which I will cover in this course. I have written radix-2 is the minimum radix-4 is that you have to learn, but the minimum radix-2 you have to learn. Implement not using MATLAB FFT x[n] and t. No, write a C program mathematical computation and implement that part you have to do.

Design the FIR and IIR digital filter both in the time domain and frequency domain for a given specification using MATLAB or equivalent open-source software, for both. So, I can say that I have to design FIR and IIR for both the filter in the time domain and the frequency domain.

Then, I may say the design of a lattice filter, IIR filter means lattice filter; suppose, you said that I know how to design a lattice filter. So, given and transfer function h[n], I can implement it. So, the filter is nothing but the system. So I can implement either IIR or FIR in that system.

So, what are the pros and cons, how do we design it, what kind of modifications have to be made, what kind of specifications will be given to the user, and how will that user specification be implemented? So, all those things will be covered here and at the end of the course, I expect that if I give you a specification of an FIR filter, you should able to design it.

Then, there will be two or three outcomes I will gradually I will be added, which are called multi-rate signal processing and homomorphic signal processing. So, those things and if I get a time, I will cover that adaptive signal processing also, ok. So, those are the expectations from you as a student. Not only is that okay, but I said I have read the signals and systems I have read in that chapter. What skill have you developed that is very important?

And, for all the assignments and evaluations, whatever I take that question, will not prove that $f(x) = g(x)$. No, the questions will all be practical-related implementations. You have to code hard, implement code, and compute that part in your pen and paper, and you have to submit it.

So, on the application side, how does a signal look, what is frequency, how is the discrete frequency, and what is the meaning of discrete frequency? So, you know that frequency per radian, frequency per sec radian per sample radian per second is this concept has to be understood. Not only understand you have to apply that concept in that application side, ok.

So, those are the expected outcomes from you. As a student's you should read and go through this course you should develop that skill. Unless you develop that skill, your course will not be successful. Even if you can ask, you can raise those questions once you watch the video and go through the assignment. If you have any doubt, yes, I cannot understand this part; this part clarity is not there, then and there, you can write to in a forum you know that NPTEL has a forum. So that I get a chance to clear your concept, whatever I know ok.

Thank you. Thank you very much.

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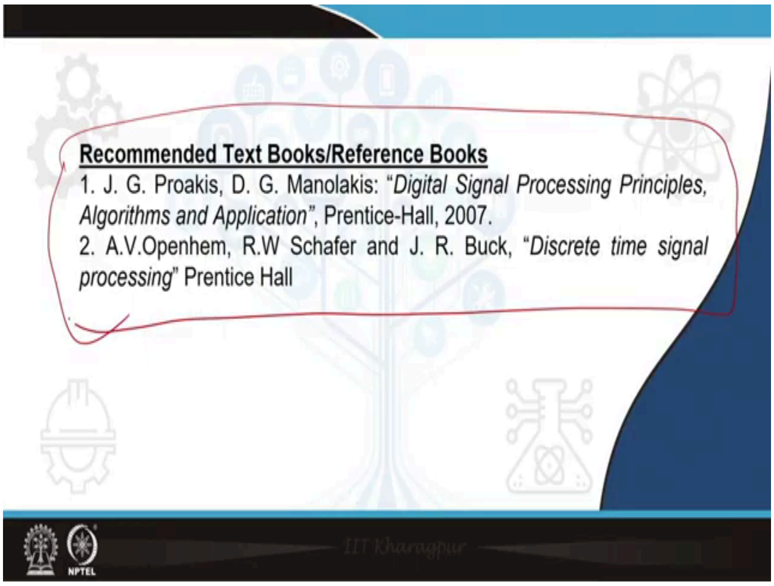

And for this you can go through this books. I will mainly follow these two books, but most of the time may be books are exactly I not follow the book, I may give some example from my experience side or I take a some example from papers or take an example from some my some research whatever I am doing from there I can give you a some example.

Since I am working in the speech and this speech processing also, so, more example will come from speech processing side also, ok.

Thank you. Thank you very much.