Foundations of Wavelets, Filter Banks and Time Frequency Analysis.
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Week-1.
Lecture -1.1.
Introduction.

A very good morning, let me introduce myself 1st, my name is Vikram M Gadre and I am a professor in the Department of Electrical Engineering at IIT Bombay. I am going to deliver a series of lectures on the subject of wavelets and multirate digital signal processing. In this 1st lecture or more appropriately exposition, I would like to bring before you the reason why we should be studying this subject or rather the reasons, the inspirations that have driven researchers and scientists to understand the concept of wavelets and multirate digital signal processing, the broad theme that we are going to address in this course and finally a broad outlines or the structure of this course.

So before I begin a formal statement of structure, it would be appropriate to put before you the thrill or the fascination of wavelets that I have or the inspiration that leads me to look deeper at this subject which I would also like to convey to you in a few words and inspire you to study the subject in depth. Typically this subject follows a basic exposition to signals, systems theory, perhaps digital signal processing or discrete time signal processing and other basic courses in communication and single processing.

So I would pick this course at an advanced level in signal processing. I would not intend to convey from this that the concepts are difficult to understand. As you will see in some of the subsequent lectures, the concepts are actually very easy, very simple, perhaps even simpler than some of the concepts that we learn in a basic course. In a basic course, the whole subject is new to us, when we study signals and systems, the whole idea of abstraction is new, abstraction of a signal, abstraction of a system, abstraction of a transform, abstraction of a domain.

So we assume that we have crossed that step, we have overcome that obstacle. We have agreed that abstraction is a good thing, in that sense this course is simple. We are moving one level higher where we are moving a little closer to reality. In a basic course on signals and Systems, or in a basic course on discrete time signal processing, we tend to be idealistic, we tend to want to oversimplify. In this course we will refrain from doing that, at least in Toto. What do I mean by this?

In a basic course on signals and Systems, we assume that signals last forever, for example, when we talk about the Fourier transform, we think of basic functions or functions on which the transform is based, we start from time T equal to - infinity and go upto T equal to + infinity, T denotes time. In this course we shall recognise that this is a silly thing to do, no signal in the world, at least no single that we can deal with realistically is going to last forever, it is going to have its lifetime, it is going to begin somewhere, end somewhere. After all the analyser, the person who analyses is also going to begin somewhere and end somewhere.

So the 1st thing we understand is we must deal with finite domains, we understand the finite domain as far as the natural domain goes very well. What I mean by the natural domain is, suppose for example we are talking about an audio signal, the natural domain is time. So an audio signal begins somewhere in Time, ends somewhere in Time, we do not listen to the piece of music forever. But for that limited period of time in which we wish to listen to a piece of music, we would like to understand better the content of the music to which we listen, we also wish to be able to develop the ability in a signal processing setup to enhance what we want out of the music and to suppress what we do not.

We also wish to be able to characterise a system to does so and all the while recognising that we are not going to deal with the signal forever, we are going to deal with over a limited range of the natural domain. So here as I said, in audio signal, the natural domain is time. Let us take another example of a natural domain. Suppose I wish to deal with the picture, I wish to take a picture in which I have a face and naturally many facial features the eyebrows, the forehead, the nose, the lips and other features that are associated with a typical face.

When I wish to isolate a certain feature, I would imply being able to localise in the spatial domain. Now this is an example of a 2 variable domain, unlike the other example of audio where it was a one variable domain, only time, here I have 2 spatial variables. Localisation is the common thing, let me spend a couple of minutes in explaining that little better. Suppose I take a piece of audio, let us assume that in that piece of audio, a number of notes are sung, if I take recourse to the Indian system of description, you could have a Raaga in which there are several different notes, the components of the Raaga.

Suppose I wish to build a signal processing system that takes the rendition of this Raaga and identifies the notes that compose it, the different frequencies so to speak that come together and play to form that piece of audio. What do I need to do in such a circumstance from a signal processing perspective or from the point of view of analysing signals? One thing is obvious, I need to segment in time, so I need to be able to say that in this part of the time axis, I have this note prominently played, perhaps only that note was played.

In a different part of the time axis, a different note was played. Now when I say different parts of the time axis, they are not only separated, but also might be different in length, so one note could be longer, another note could be shorter. It is not just a question of which note have been played but also for how long. Now what exactly do we mean when we talk about notes, if I take a signal processing perspective I mean? After introduction to the fundamentals of signals and Systems or even for that matter basic course on discrete time signal processing, all of us are comfortable with the idea of a frequency domain.

So we accept that we can think of continuous signals or discrete sequences as having embedded inside them a collection of sine waves. If it is a continuous signal that we are talking about, we have continuous sine waves. If it is a discrete sequence that we are talking about, we have sampled sine waves and the whole philosophy of the Fourier transform is most reasonable signal that we deal with can be thought of has comprised of or composed of a collection of sine waves.

In fact in principle if the signal is not periodic, then we are talking about an infinity of sine waves with frequencies ranging all the way from 0 to infinity. If it is a periodic signal that we are talking about, then we have a discrete set of sine waves, possibly infinite, possibly finite, which we call the Fourier series representation. Anyways, what I am trying to emphasise here is that there is a different domain to which we go when we wish to analyse the signal better.

So in the language of signal processing, if I were to take that Raaga, that rendition of the audio piece, the elementary audio segment as understood in Indian music and if I were to query what notes are being played and equivalently asking what is the frequency domain content of this audio piece, if I visualize a frequency axis, what points on that frequency axis are occupied, what are the locations where the transform is prominent?

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