Foundations of Wavelets and Multirate Digital Signal Processing
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Module - 1
Lecture - 1
General Introduction

Let me introduce myself first. 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 first 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 concepts of wavelets and multirate digital signal processing.

The broad theme that we are going to address in this course and finally a broad outline and 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 wavelengths 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 this subject in depth.

Typically this subject follows a basic exposition to signals, system theory, perhaps digital signal processing or discrete time signal processing and other basic courses in communication and signal processing. So I would pick this course at an advanced level in signal processing. I do 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 over simplify. In this course we will refrain from doing that, at least in total. 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 the basic functions or functions on which the transform is based, which starts from time (t) equal to minus infinity and go up to t equals to plus infinity. T denotes time.

In this course we shall recognize that this is a silly thing to do. No signal in the world, at least no signal 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 analyzer, the person who analyses is going to begin somewhere and end somewhere.

So the first thing we understand is that we 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 an 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 don't listen to a piece of music forever.

But for that limited period of time, which we wish to listen, to that 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 of the music and to suppress what we don't. We also wish to be able to characterize a system which does so and all the while, recognizing that we are not going to deal with the signal forever.

We are going to deal with it over a limited range of the natural domain. So here as I said, in an audio signal the natural domain is time. Let us take another example of a natural domain. Suppose I wish to deal with a 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 localize in the spatial domain. Now this is an example of a two variable domain. Unlike the other example of audio, where it was a one variable domain, only time. Here I have two spatial variables.
Localization is the common thing. Let me spend a couple of minutes in explaining that a little better.

Suppose I take a piece of audio. Let's 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 Raga, in which there are several different notes. The components of the Raga. Supposing I want to build a signal processing system, that takes the rendition of this Raga 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 from such a circumstance from a signal processing perspective? Or from the point of view of analyzing 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 had 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 the different parts of the time axis, they are not only separated but also might be different in length. So one note could be longer and another note could be shorter. It's not just a question of which notes have been played, but also for how long.

Now, what exactly do we mean when we talk about notes, if we take a signal processing perspective I mean. After introduction to the fundamentals of signals and systems or even for that matter a 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 sample sine waves. And the whole philosophy of the Fourier transform is, most reasonable signals that we deal with can be thought of as 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 infinite T of sine waves, with frequencies range from all the way from zero 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.

Anyway, what I'm trying to emphasize here, is that there is a different domain to which we go to, when we wish to analyze a signal better. So in the language of signal processing, if I were to take that raga, 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, I'm 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 predominant is prominent.

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