Real - Time Digital Signal Processing

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Lecture - 01
Introduction

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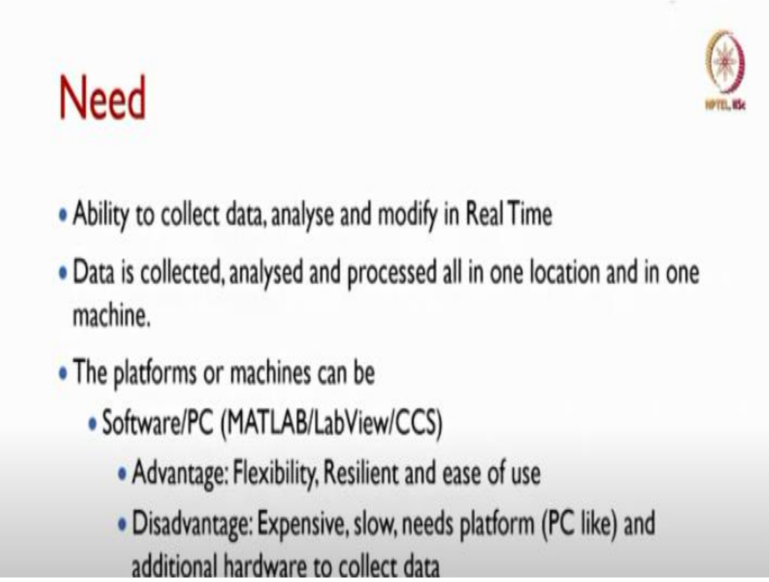

Good morning. So, in this course we will be covering real time digital signal processing. So, we will see first, what is the need for the course. Ability to collect data, and then we are going to analyse it and then see how we can do the modification in real time. So, whatever data collected, we will be analysing it, and then we will be processing, so all in one location and in one machine we call it as hardware.

So, the platforms for this machine can be either through software that is we can use PC, MATLAB or the software or using LabVIEW or Code Composer Studio these are the things what we will be using it. The advantage if we do it in the software is flexibility. It is resilient to all conditions and then all of us know that it is very easy to use that. The disadvantage part of it is because all software's we know that it is expensive, and then it is slow in nature. And then we need platform like PC, and then additional hardware to collect the data from the software.

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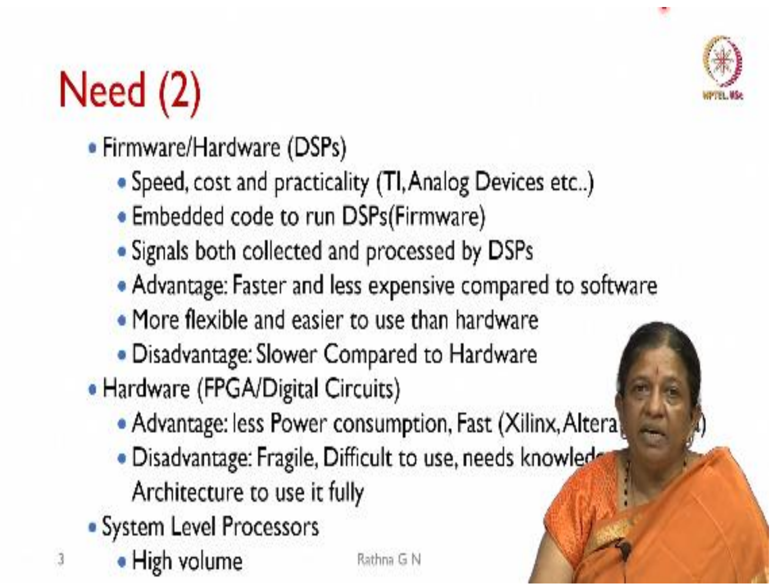

So, the next one what we can use is the firmware or hardware basically. In this case, we will be calling it as DSPs. So, the advantage of using this firmware or hardware is speed, cost and then practicality. So, we know that most of the DSPs were manufactured by TI analog devices and then other companies are there in this. The other advantage is embedded code basically to run your DSPs which we call it as firmware, so the signals both collected and then processed by DSPs.

The advantage of using this hardware or firmware is faster and less expensive compared to software and then it is more flexible and easier to use than the hardware. So, in this case, we will be discussing the hardware in a little while. The disadvantage in this case is also it is going to be slower compared to hardware. So, the completely use the solution to use the hardware is basically either field programmable gate arrays that is FPGA or we can go for digital circuits that is basically VLSI design what we can do the thing.

The advantage in this case is because all of us know hardware uses or FPGAs use less power consumption wherever the application is there or we have more number of units which have to come out then better to go for complete hardware design. And then it is going to be fast in running of your algorithms or any application. So, as an example, FPGAs we know that Xilinx Altera and then now recently Intel is coming with FPGAs.

The disadvantage in hardware is because it is a fragile what we call it and then it is difficult to use because people should know how to design to the core level. And then you should know the knowledge of architecture, basically so whether you are using the Siemens architecture or Enmos architecture, so you should know core level how to design your circuits.

So, the other level of it is what we can have is system level processors what we call it, so when we have high volume requirement, then we go for the system course processor design.

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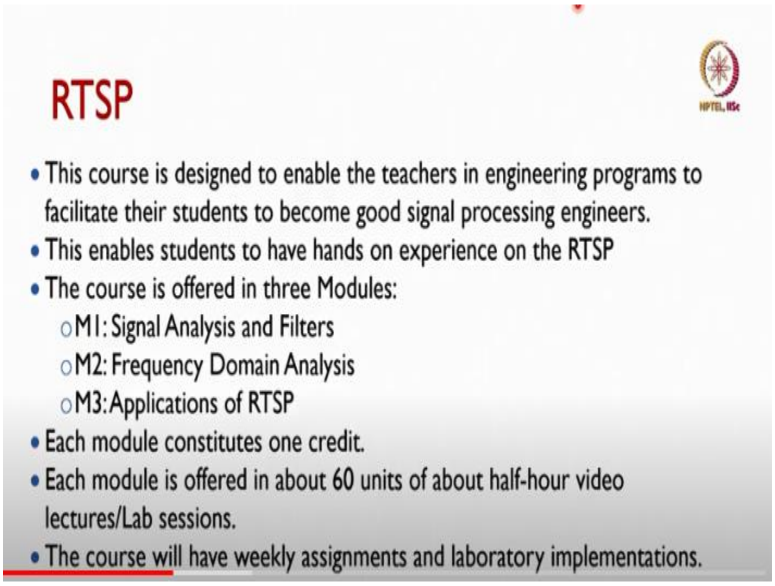

So, in the present lectures, that is in real time signal processing, it is designed to enable the teachers and engineers who are in that program, and then facilitate the students to become good signal processing engineers. And then this enables students to have hands on experience on the real time signal processing. The course contains both theory as well as lab component. The course is going to be offered in 3 modules, what we call it. The first module, M1 we call signal analysis and filters.

And then the second module we will be covering frequency domain analysis and then the third module covers applications of some of the real time nature architectures and other things what we will be considering in the real time signal processing. And then we have each module will be consisting of one credit and then each module is offered in about 60 units of about half an hour video of lectures which have lab sessions also.

So, we are going to have weekly assignments and laboratory implementations because most of the colleges have DSK 6713 board hardware so they can use them in the college or we are going to provide hardware interface so those who want to do it remotely can be used the hardware which is available in our lab.

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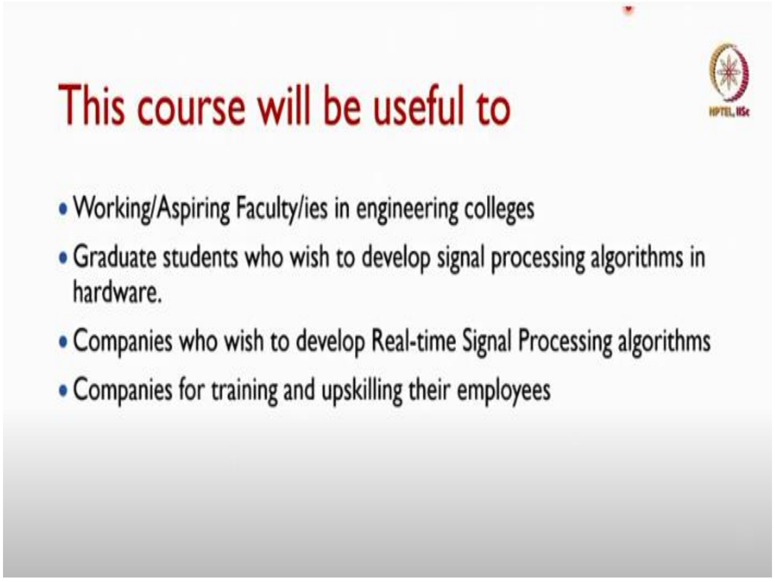

So, how this course is going to be useful? It can be for working or aspiring faculty or faculties in engineering colleges. So, because they will be having little difficulty in interfacing the board and then running some of the experiments so it will help all of these people and then even the graduate students, so they want to develop some signal processing algorithms in hardware, it will be of help to them.

Companies who wish to develop real time signal processing algorithms, because we will be covering not just C programming or any other languages. So, we will be little bit going deeper into the core level design, so that people will get the hands on information how it has to be coded in the high-level languages also. And then companies can use it for training and then upskilling their employees.

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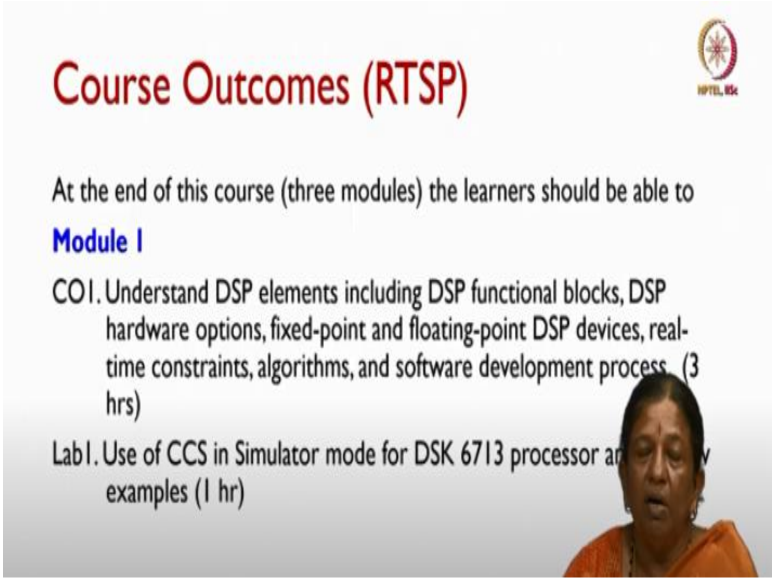

So, what will be the course outcomes that is what will be seeing in the slide, at the end of this course, that is 3 modules what we are covering it, the learner should be able to that is basically module one, so which you will be seeing the sub modules understand DSP elements, including the DSP functional blocks, DSP hardware options we will be discussing fixed point and floating point DSP devices and what is the advantage of using fixed point number system compared to floating point.

Real time constraints, what we will be discussing about it some of the algorithms what we will be covering, will be covered in this course and software development process. So, this will take us somewhere around 3 hours to complete this module. And then followed by what we will have is the lab one, so use of code composite studio in simulator mode initially we will start for DSK 6713. So, the software is simulator software is going to be provided and then you can run few examples those who are unable to access the hardware board, so initially we will do in simulator mode.

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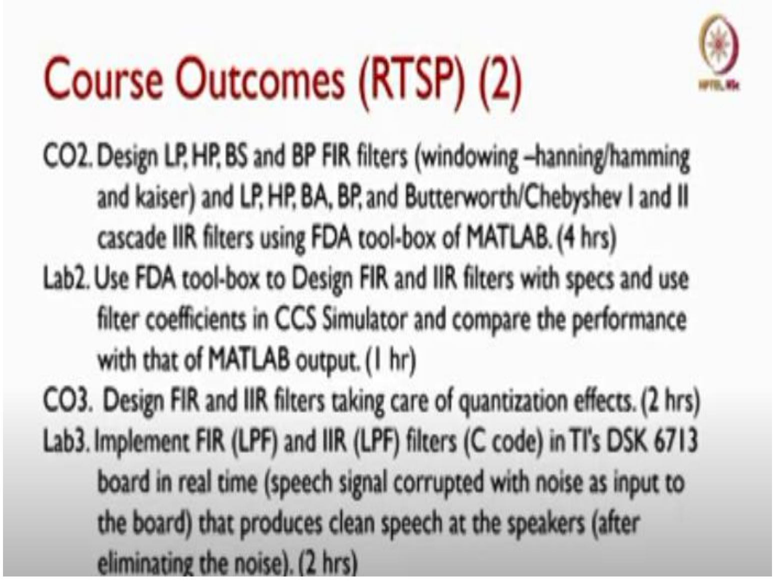

So, the next course outcome, what we will be seeing it, design, low pass, high pass, band stop and band pass, FIR filters, using windowing hanning or hamming are will be using the case of window. And then same thing what we will be designing using Butterworth or Chebyshev 1 and then 2 IR filters in cascade mode using initially MATLAB FDA tool box, I think it will be renamed as filter design tool box in the latest versions. So, whatever version is there, students can use that because it is available in the net.

So, we are going to accompany this lab session with how to use the FDA tool box to design these filters. How we are going to specify the specifications, and then how will be using these filter coefficients because the design is going to happen using MATLAB and then we will use this filter coefficients to your FIR filtering it can be real time FIR filter what we will be using it in either initially we can do it in the simulator, later on we can compare with the MATLAB code how it is going to work whether it is in a similar fashion or not.

Later on, we will be using the hardware to design the same thing. So, input will be using some noise signal with the original signal and we will see how the filter is going to perform in its best way to get as noise stream out from the input signal. So, the third module what will course outcome is going to be design of our FIR and IIR filters. That is basically we will be taking care of quantization effects because in the analog mode, we know that we do not have any quantization problems creeping in.

But when we come to digital domain, we know that quantization is going to play havoc in so many other cases. So, we will be seeing that how these effects are going to modify the filter characteristic so that we will be able to check that whether we are in the permitted flavour or not. So, the same thing we will be implementing in the lab with low pass filter using C code using the DSK 6713 board and real time scenario as I was mentioning in the previous outcome also, here what we are going to use is the speech signal corrupted with some noise.

So, that is going to be fed into the board and then we will see these filters how it is going to remove the all of us know that noise is high frequency component which is present in the low frequency component, which is going to be removed and then we will be seeing whatever the speaker's what he want speech is going to be clean speech what it will be coming out of it. So, we will because it is going to take little interfacing and other things what we have to discuss in hardware. So, we will be taking around 2 hours to discuss about it.

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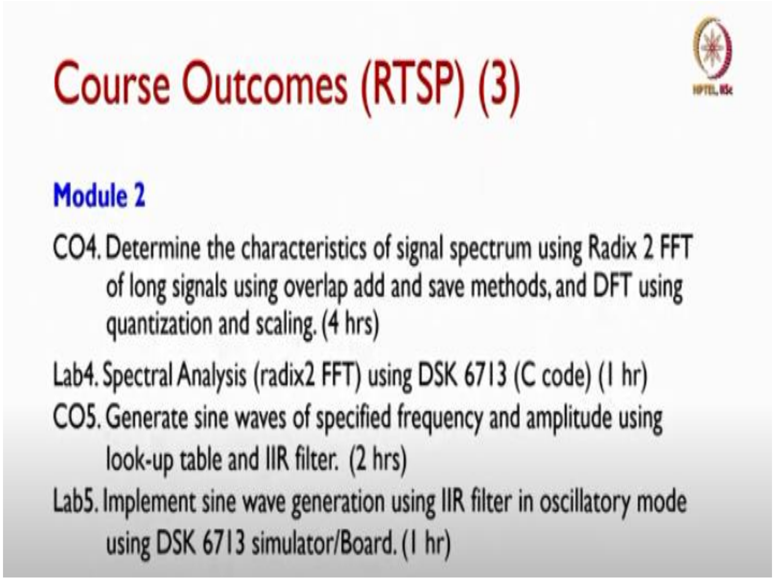

So, coming to this we saw that we had signal analysis and then filters in the module one, so module 2 goes out comes out are going to be as follows. So, first we will be determining the characteristics of signal spectrum using Radix 2 FFT and then we are going to apply it for long signals using overlap add and save methods and we will be using the DFT comparison with respect to FFT how it is going to be faster.

And then we will have the because we know that when we are in the digital domain, we said that how the quantization is going to affect in the filters same way how in doing our frequency domain analysis and then comparing design part of it how quantization and even the scaling of inputs, how it will be affecting the signals and how we are going to take care of it is going to be discussed in this module.

So, the accompanying lab part of it is going to be will be analysing the spectral basically whatever is given using the Radix 2 FFT. In this case we will be using the DSK 6713 board either in simulator mode or in the hardware. So, the accompanying code what we will be writing is in C basically. So, the next one is because how to generate different sine waves signals what will be checking up, because we know that IR filter when we put it in oscillatory mode, once the input is removed, it can oscillate itself.

So, we can use this concept to generate sine waves with specified frequency and amplitude. What we can run it and the other way of implementing is we can use the lookup table by generating sine or cos functions signals can be generated. The component for the lab is will be using the IR filter in the oscillatory mode to generate sine wave using the DSK 673 it can be used both in simulator as well as board.

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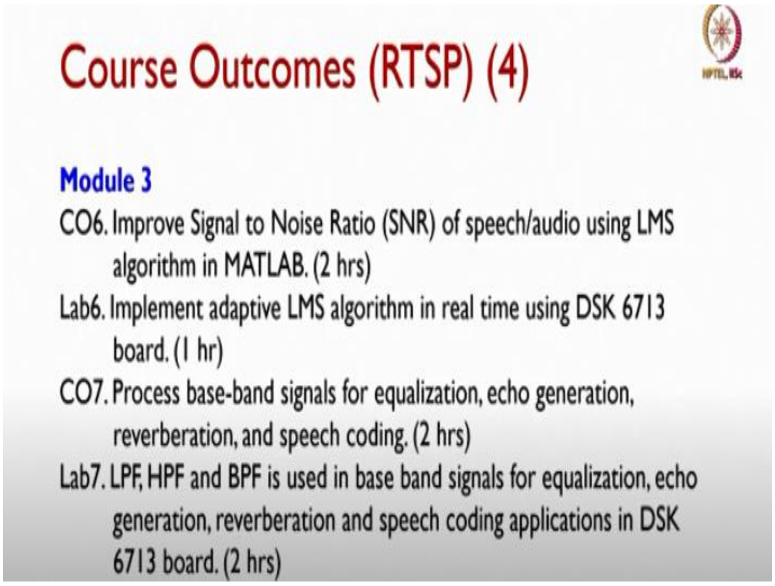

So, coming to the next module that is 3 actually. So, we will be checking little bit of applications what we have specified. The first one is going to be how to improve signal to noise ratio. That is basically will be speech or audio signals using LMS algorithm first in MATLAB what we are going to use it. And then later on the lab component will be implementing in real time using DSK 6713.

So, we know that our FIR and IIR filters have fixed coefficients basically but if there is any noise in the input or in the surroundings, they will be not performing whatever we would like to, so how we can do the adaptive filter will be seeing it using LMS algorithm. So, the next outcome is how will use this some of the design filters to generate synthetic output. That is basically process this is base band signals for only what we will be doing equalization.

So, all of us know that how we can make all of you have heard about the karaoke or whatever it may be the thing so in your mobiles and other things, whatever we are using it. You will be suppressing the voice and then you will be having the instruments basically how you can record your own voice along with the instruments by suppressing some of it by doing equalization what we will be looking at one of the application.

The other one we know that if we go for outstation in a hilly region or wherever we speak about it, we know that how echo is going to be generated. Can we generate that synthetically, that is what we will be looking as an one more application. Next one is reverberation, basically lot of people will be speaking multiple things, how it can be multiple echo's what we can have it what we call it as reverberation how we can generate it and then later on we can see that how this can be eliminated also.

So, the next one is all of us know that speech basically coding. That is whatever the speech we are recording or whatever it may be the thing there are different ways of storing them, how to do the compression and other things. So, one of the example we will take it, how we can implement it or we will be looking at basically in this module. The other one we will going to have lab component after learning about the theory.

So, how will use or whatever filter we have designed earlier that is low pass, high pass and band pass filters in the base band signal for the equalization for the echo how we are going to delay the speech and then generate a echo and even the reverberation how multiple peoples are speaking what we can generate it and in the speech coding also some of the things what we will be looking using DSK board.

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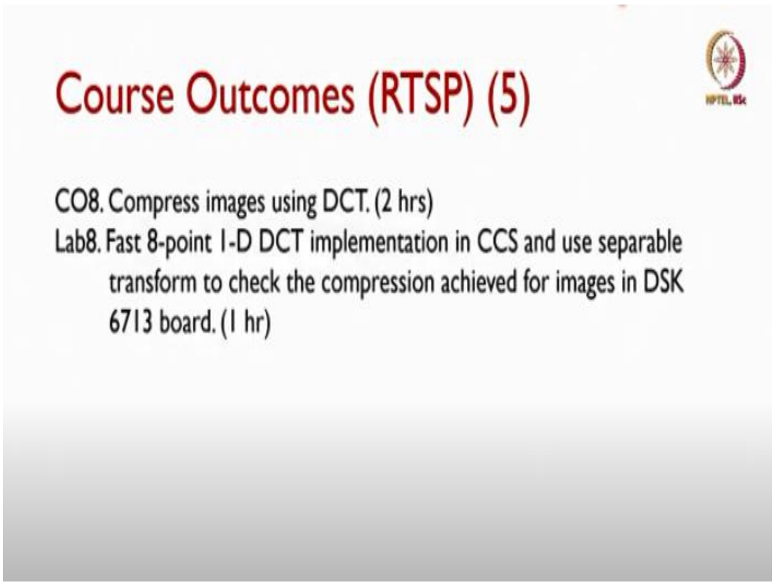

So, this is what this course outcome is. The last one what we will be looking in this module is going to be little flavour on the image processing. So, in this case, only discrete cosine transform what we will be looking at, so we will be seeing how fast 8-point 1-D DCT implementation in using Code Composer Studio we can implement and then check that how this can be used in the 2-D domain that is using separable transform to check the compression achieved for images in DSK board.

So, these are the course modules what we have it, so in the next session, what we will be taking starting with the regular course basically, basics of signal processing as a module one what we will be looking at. So, what I will say is happy learning and then thank you with introduction of this real time signal processing. Thank you.