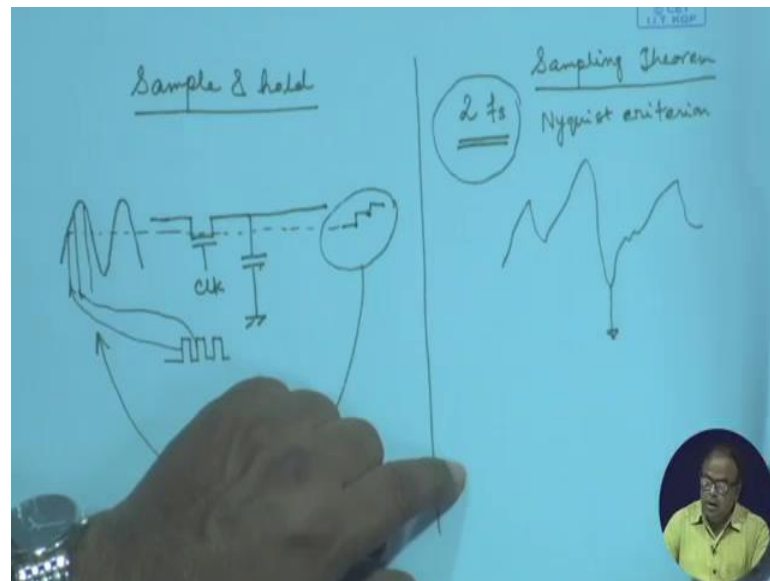


Embedded Systems Design
Prof. Anupam Basu
Department of Computer Science and Engineering
Indian Institute of Technology, Kharagpur

Lecture - 14
Discretization of Signals and A/D Converter

In the last class we have talked about sample and a hold.

(Refer Slide Time: 00:21)



So, we get an analog signal and when we read that we carry out some sample and hold them that we do using a simple circuit, where we clock the circuit as the analog voltage is coming here some sort. And we have got a capacitor, and this capacitor charges and we take the output from here. So, consequently what happens by the frequency of the clock; the frequency at which we trigger the clock this signal is sampled, because these are the parts which are triggered by this clock; which are triggered by this clock are passing through this and we get the output of the capacitor.

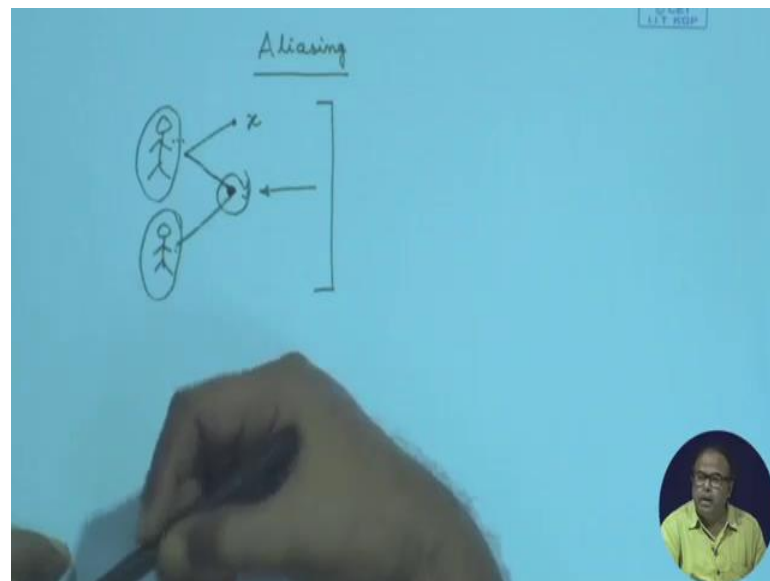
So, at this point I will be getting a signal which is equivalent to these levels. So, I will get this value, then this value, then again there was no sample here so I will get some value intermediate value like that. This sort of signal will come as this being sampled. And we also know that in order that from this sample signal if we want to reconstruct the original signal the sampling frequency needs to be twice the maximum frequency of this signal. Now here I have just shown a sinusoid, but actually you know it can be a signal

like this which is not a sinusoid but, those of you know about Fourier transform you can convert this into a series of sinusoids of different frequency; that apart.

Any such signal, any signal I need not do inside Fourier transform here; any such signal can be sampled at the rate that is twice the largest frequency of this. The context of Fourier transform was coming here because this signal is having multiple frequencies and I have to go above twice of the maximum of those frequencies. In that case I can get a sample signal from which I can reconstruct the original signal alright. So, this is that was the sampling theorem by Shannon, and that is known as the Nyquist criteria. That it should be sampled it twice that thing.

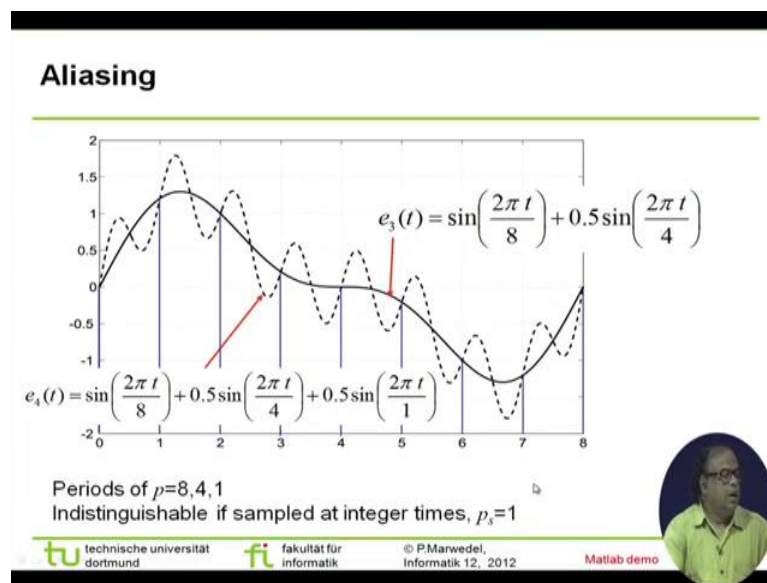
Now, because of this sampling we are getting; so we are sampling now as we sample we are doing some sort of aliasing. What is meant by aliasing?

(Refer Slide Time: 04:01)



Alias means what? The same person has got two names. This person has got x name and is also called as y name. But on the other hand it can be that if there be two persons with the same name then it becomes difficult for me to identify who is. And this person can has taken up this name y because there is another person with this name y, so if he is referred to as y we have to resolve which of these two persons it is. Now how does it reflect to a signaling scenario or inputs or the signals that embedded systems will be dealing with. Let us have a look at this slide.

(Refer Slide Time: 05:06)



So, you can see that here this is again taken from; as you can see here the slides are Professor Peter Marwedel from Dortmund. Say this is a signal; right the bold line which can be expressed in the form of this series. It has got two periods 4 and 8. I was mentioning that a non-sinusoidal complex signal can be broken down into a series of sinusoidal signals. So, here you can see that it has got two periods 4 and 8, so two different frequencies are there.

Now here is another signal e_4 , this one visible here where this dotted signal alright it has got these frequencies to be the same, but in addition there is another frequency. Consequently the waveform is different right. So, we have got two signals e_3 and e_4 . Now why is Nyquist criteria so important? Now if I sample it at these points that are at every integer type one every unit time I am sampling it. So, I am hitting upon these points I am hitting upon these points, this point, this point, and when I hit upon this point I do not know whether the value is coming from this dotted signal or the normal signal.

Therefore, again coming back to this y , this picture when I come to this y I do not know whether this is this person or this person. So that is the phenomenon of alias. And because there is an aliasing here what do I need to do; now you see if I filter it what is the highest frequency of these signals out of which I want to distinguish. The highest frequency is coming from here that is 8. So, if I sample it with a period of 16 then I will come to some points here which are different, will not get the same signal all the time.

Here we are getting even if I double this, I will get say I was doing it here as one if I sample it with a period 4 still it will be indistinguishable, 8 it will be distinguishable because they are matching here. But, if I do it with 16 then; obviously, I will be getting that. So, that is why we have to avoid this aliasing.

(Refer Slide Time: 08:33)


Aliasing (2)


☞ Reconstruction impossible, if not sampling frequently enough
How frequently do we have to sample?

Nyquist criterion (sampling theory):
Aliasing can be avoided if we restrict the frequencies of the incoming signal to less than half of the sampling rate.

$p_s < \frac{1}{2} p_N$ where p_N is the period of the "fastest" sine wave
or $f_s > 2 f_N$ where f_N is the frequency of the "fastest" sine wave
 f_N is called the **Nyquist frequency**, f_s is the **sampling rate**.

See e.g. [Oppenheim/Schafer, 2009]

 technische universität
dortmund

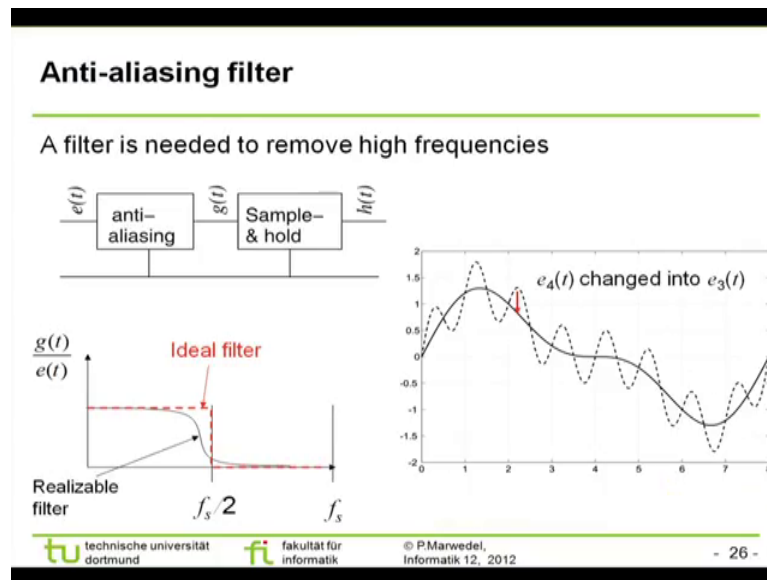
 fakultät für
informatik

© P.Marwedel,
Informatik 12, 2012

- 25 -

So reconstruction is impossible, if we do not sample it frequently enough. And aliasing can be avoided if we restrict the frequencies of the incoming signal to half the sampling rate. So, the incoming signals must be restricted to half the sampling rate. So, higher frequencies I will not take; I repeat I will not take higher frequencies, therefore what do I need? I will need a filter, I do not want higher frequencies to come up, I have frequencies higher than the sampling rate therefore I need a filter. What sort of filter would I need? I will need a filter that is a low pass filter.

(Refer Slide Time: 09:31)

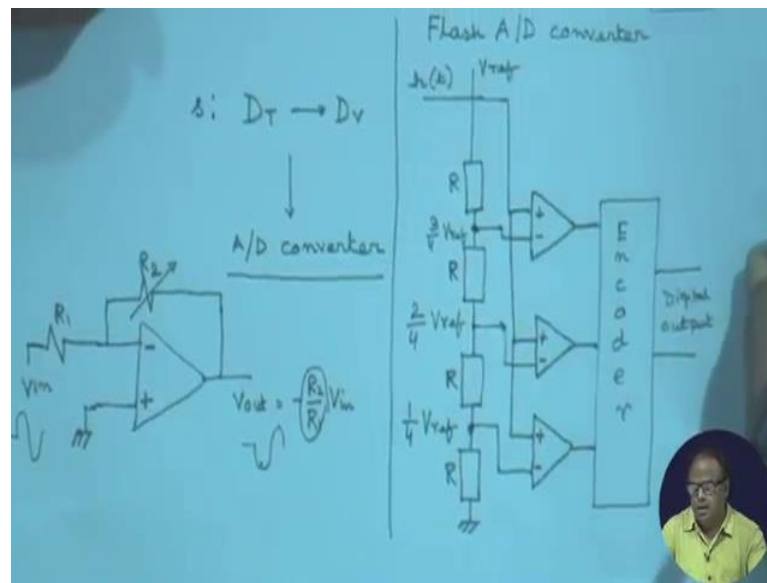


So beyond this I will be; so here. If I want to do a filter is needed to remove high frequencies, so here is a signal coming in and I have got an anti aliasing filter here. And this filter the characteristics ideally is this, that any frequency below this half of the sampling rate will pass anything above the sampling rate will be cut off. Therefore, but the red one is the ideal filter scenario that we actually do not get, so we actually will get this sort of all of you know that realizable filter is this. The better the sharpness of the edge the better it is better cut off.

And if I pass it through this sort of filter anti aliasing filter then I will be able to distinguish between this e_4 . And since the higher frequencies are being cut off, sorry the only the low frequencies are being allowed we will get to this e_3 t. And therefore, next I can feed it to sample and hold.

Now, so summarizing this part what have we done is that- we are dealing with taking the inputs to the embedded system and for that the signals can be analog signals also that are coming up and for them we need to carry out sample and hold in order to digitize them. Still I have not made them digital but I have just sampled, and while I sample I must be careful about the sampling frequency and apply the anti aliasing filter so that whatever samples I get are representative of a particular signal. Now once that is done the next thing is we need to discretize the sampled values.

(Refer Slide Time: 11:57)



Now, discretization is nothing but again I will is a signal I am discretizing that from D T to D V. So, I am taking two from some continuous domain toward discrete domain and that is being done with a A to D converter alright. Now here there are different types of A to D converter. First we are trying to describe one type of A to D converter that is known as flash A to D converter.

What I need here, is I am getting some signal coming here let us call that h t some signal is coming and I have got a reference voltage let me call the reference voltage to a V_{ref} . Now I have got a series of resistances here each of them R , I have got four resistances each of them are of value R . So, there is a voltage drop occurring across each of these resistors. So, here at this point what I am getting is 3 by 4 of V_{ref} , is it visible 3 by 4 of V_{ref} . Here I will get at this point I will get 2 by 4 of V_{ref} that is half of that. At this point I will get 1 by 4 of V_{ref} .

Now there is a magic system called an operational amplifier, which most of you know an operational amplifier is a very versatile device which has got it is an amplifier with infinite input impedance and very low output impedance. And it has got two inputs: one is an inverting input another is a non inverting input. And it is it can be used as a very high gain amplifier if I put a feedback across this, and there is some input resistance coming R_1 and say R_2 . So, depending on the ratio of this R_1 and R_2 we can have say this there is an amplifier configuration. Here if I put in some V_{in} if I put in some V_{in}

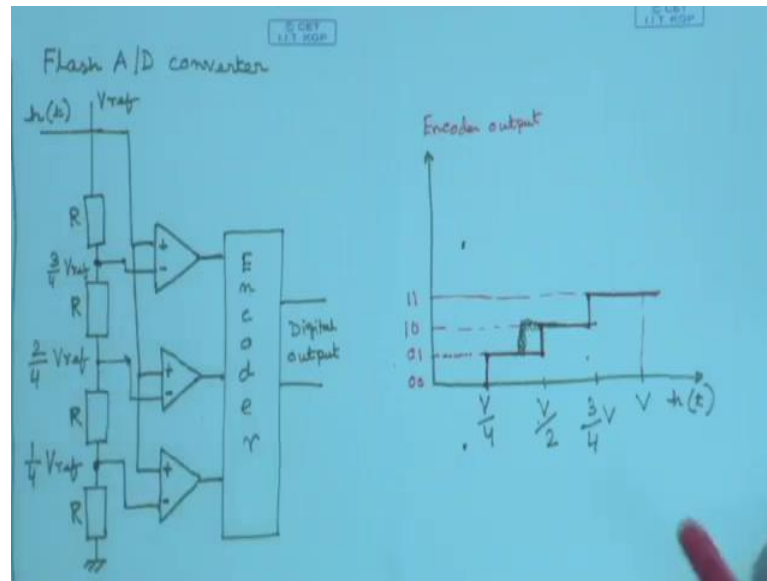
here the V_{out} will be $-\frac{R_2}{R_1} V_{in}$. And why this minus, minus is because if there be a signal here the signal's phase is changed here right phase inversion occurs, but the magnitude will be multiplied. So, by this factor $\frac{R_2}{R_1}$ we can adjust the gain of this amplifier if we may can make it a variable resistance.

However, for our purpose that is the details are not important, but we now feed this can also be used as a comparator such operational amplifiers can also be used as a comparator where there is a little change; the little change is- now I will simply take this if I have an operational amplifier and now feed the reference in the minus here and the signal the actual signal is coming to the non-inverting input in that case if the signal crosses the reference, so I put the reference here and the signal here, if the signal crosses the reference it will be on. So, this is a comparator. But I feed a signal at the non-inverting input and the reference at the inverting input and it will.

Now similarly here now if I have three operational amplifiers and I feed the signal to the where is the signal the signal is coming here to the non-inverting input of all these operational amplifiers right. And the different references I put to the inverting input of this. Now what is the significance? If my signal the signal is higher than this value then this bit will be 1. If it is higher than this value half but smaller than this then it will be 0 this will be 1. If this is higher than this, suppose my reference voltage is 5 volt if it is higher than 5 by 4 volt, but lower than this it will be 0. In between it can be this one can be 1 as well as this can be 1.

So now, if I take the output of all these and put it in an encoder it will take these and it will create the digital outputs, alright. That is how we can do the analog to digital conversion.

(Refer Slide Time: 20:36)



And say for example if I keep this by the side; let us look at this diagram. Here I have got the $x(n)$ that is sorry, this signal $x(n)$ is being shown in this axis. So, here I have got V reference by 4 here I have got V reference by 2 here I have got 3 by 4 V reference and here I have got the V reference right. Now I did the sample and hold right; see this thing is coming after the sample and hold circuit and what is preceding the sample and hold circuit the anti aliasing filter. So, after that I am getting the sample and hold circuit and then feeding it to the A to D converter.

And suppose my voltages are like this then here it goes up, I am sorry it comes here and goes up and goes up to this and here it again goes up. So, once again let me color it up so that this confusion is avoided. So, after sampling the signal has become like this; after sampling whole this is my signal. Now by this means what I will get here this will be 00, this will be 01, why 01? Because when V by 4 only this one this op amp will fire others will not fire.

Why did I need that encoder, because I need it got three but I do not need three; so any one of them. So, 01 then this one would be 10 output of the encoder and this is the 11 there is a reference voltage; reference voltage will be 11. So, this is the output of the encoder output. So, that is how we can get the encoding done; clear. There is a flash type of A to D converter.

Now what is the resolution of this? In the last class we had talked about resolution as a very important part of any sensor.

(Refer Slide Time: 24:26)

Resolution (in bits)

Resolution Q (in volts)

The difference bet. two ~~inputs~~ inputs causing the output to be incremented by 1.

$$Q = \frac{V}{n}$$

2.9 $Q = \frac{V_{ref}}{4}$

$V \equiv$ diff. bet. the max and the min voltage

$n =$ no. of voltage intervals

Speed $\equiv O(1)$ | Hardware complexity $\equiv O(n)$

So now, for this A to D converter we want to find out resolution of this A to D converter in bits. The number of bits produced the encoder output. So, what will be the resolution? The resolution Q in volts is what is that is the difference between the two inputs; input voltage is causing the output to increment by 1 right. So, let me write down resolution is actually the difference in volts between two outputs I am sorry, between two inputs or input voltages, two inputs causing the output to be incremented by 1.

In this case we can see Q my maximum thing was the V reference right. The difference between the largest and the smallest voltage that was there let me call it V there is the largest and smallest voltage and n is the number of voltage intervals, if V is the difference between the max and the mean voltage and n is the number of voltage intervals, that is the dependent on the number of bits; number of voltage intervals, then Q is basically V by n . Now in the case of in this one say in this example that we had done what would be the Q ; in this what would be the Q ? It will be the V reference by 4, the difference between that is V ref. So, for this example where we have taken this in this example it will be for example it will be V ref whatever that was, by 4 because we have taken two bits so Q is V ref by 4 here.

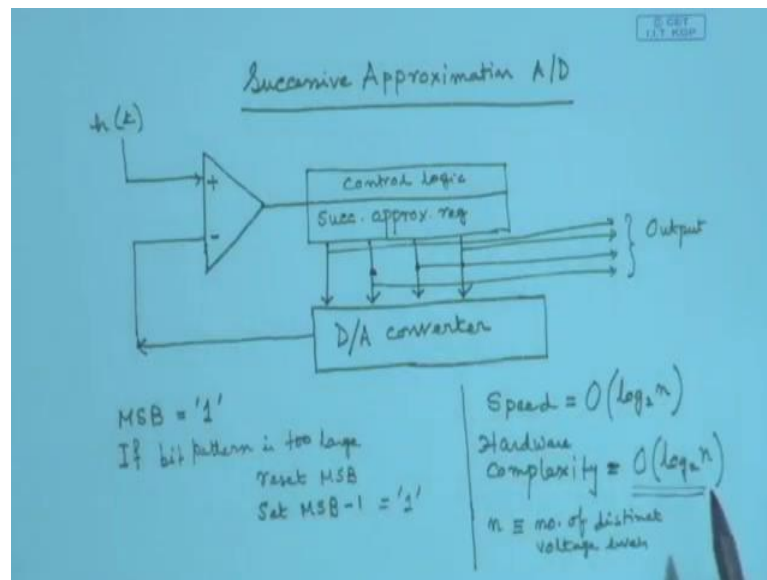
The next thing that we have to see is because we have to select a particular A to D converter. So, what will be the speed of this? The speed of this if I express it just like we express the time complexity of programs, similarly what would be the order for this conversion time complexity. It is a flash A to D converter you get the voltage all these things are being done in parallel.

Therefore, its speed is of the order of 1. The other thing that is important as an engineer; the other thing that you have to think of because for everything that we do we have to pay cost for that. What will be the hardware complexity for this? Hardware complexity if I also express it as the order, the complexity is in terms of the number of bits the resolution that I want is dependent on the number of bits. So, the number of bits is actually n , so if there be n bits that I am trying to deal with then I am having in such ops required here. If I want that, that is what the resolution is that is giving you the resolution.

Therefore, this is of order n . So, as such I can see the hardware complexity is not very small, but the speed is much faster. So, you have to keep in mind these two parameters when you think of selecting whether you want to select a flash A to D converter. One of the applications why it is very much used is video processing. In video processing we need very fast response, because the frames are moving I want to do it very quickly I want to capture them; I need a fast A to D converter. Therefore, for them these flash A-D converters are very much applicable.

But, in video the resolution here is determined by the hardware complexity. Now in video there are applications beyond video where we need higher resolutions.

(Refer Slide Time: 31:52)



So, for higher resolutions we have another type of A to D converter which is known as Successive Approximation A to D converters. The idea is very simple, its resolution is higher. The idea is very simple, we have a comparator where I am giving the signal $h(t)$ as the input and I have got a point for giving the reference. And then I am taking it inside say a module where there is a successive approximation register and there are some control logic; I will explain what this control logic is all about. And we take the output of this successive approximation register and feed it to a digital to analog converter. And the output of this digital to analog converter is fed back here. Now let us see what is happening.

Initially assume that all these are zeroes right and there is some signal, something, say some level of voltage is there. So, all these are zeros are coming over here and so this signal these two are not matching in the comparator, so this is not fired. So, what you do at every stage what we need to do we first convert; so the key idea is just like binary search, I first I do not start with 0 I first set the MSB say whatever be the number of bits here take another bit. The MSB I set to 1 alright.

So the MSB was 1, if that is too large this bit pattern is too large that means the comparator is fired then we reset the MSB. And then we reset the MSB and set the lower bit of the MSB to be 1. Now if it is not too large then it is fine. In that way we give the values here, and accordingly this control logic is actually carrying out this binary search;

you understanding it is actually doing out this binary search for different bit patterns initially with the MSB to be 1, if it is still low I have not matched the h t then I will try to make this one to be 1. Suppose this is too high then I will make this one to be 1. Now it is not too high, therefore I have another scope I make this one to be 1. In that way I go on increasing this and ultimately I want to converge alright. And if this is too large then if after this then I will reset MSB minus 1, in that way I go on. And ultimately when I converge I take out these as my output; this will be my output digital output.

Now, again coming to this therefore, can you tell me just like in comparison to the flash type what would be its speed; the speed will be just like binary search, we are doing binary search actually I am putting up a value and searching whether it is matching or not and I am trying to search for the solution right. So, the speed will be of the order of \log to base n ; $\log n$ to the base to sorry.

So, that will come there is a total, so depending on the number of bits that would be the; and what will be the hardware complexity? The hardware's complexity is if n be the number of distinct voltage levels then I will need $\log n$ bits and that is my hardware cost. So, this will be of for the order of $\log n$. So, what I can find if I compare it with the flash A to D converter that this one is of higher resolution, because I can go on increasing the resolution, I can go on increasing the number of bits because as I increase the number of bits my complexity increases in \log , whereas in the flash one the complexity was or the cost was increasing linearly. However, the speed of that was of order 1, the speed of this is \log ; so it is a little slow.

So, depending on these two we have to select which type of A to D converter will be using. So, we pause here and in the next lecture we will continue with the other aspects.