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NPTEL ONLINE CERTIFICATION COURSE

Course
on
Analog Communication

by
Prof. Goutam Das
G S Sanyal School of Telecommunications
Indian Institute of Technology Kharagpur

Lecture 58: FDM Vs TDM

Okay, so, so far I think we have started discussing about sampling theorem, so why we started discussing this is because we have already done the amplitude modulation and the frequency and phase modulation. So this is something we have already seen that how that works, and what are the relative benefits or what are the pros and cons of these things. And then after that we have promised to start another form of analog modulation which is probably the boundary towards digital modulation.

So this is called the pulse modulation, so there are few types of pulse modulation that we have talked about, one is called pulse amplitude modulation where the information actually is being carried over a pulse, but it is on the amplitude of the pulse. So basically if you see instead of using career in pulse modulation like analog or all other analog version like amplitude or frequency modulation, what we are trying to do is.

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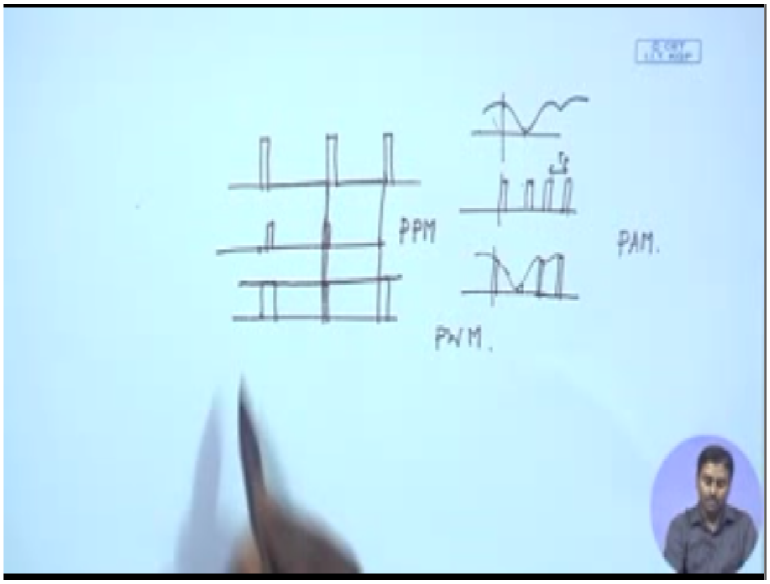

We have a pulse strength which looks like this okay. So we have a pulse strength it might have different shape, so that is something we will explore. So we have this periodic pulse strength, and then we want to modulate our information on this pulse, so pulse is almost working like a career. So what we can do is there are three types of pulse modulation, we have already discussed that.

One is pulse amplitude modulation, so in pulse amplitude modulation what we do, suppose we have a message signal okay. So depending on and suppose this is the pulse strength which goes like this okay, it will have uniform period of either T or Ts let us say. So basically what we do, we modulate the amplitude of the pulses according to the input signal. So it will be let us say this becomes the envelope of the pulses and the pulses are just see the tip like this.

So basically the amplitude of the pulse or the relative amplitude of the successive pulse actually carry out the message information. So this is one kind of pulse modulation, this is called pulse amplitude modulation of PAM, so that is something we have seen. We have also, we have talked about very briefly other two version of pulse modulation, one is called the pulse width modulation.

So like in the career we had amplitude, we had frequency and phase, for pulse also there are few things which actually can tell us about modulation or possibility of modulation. So one is of course the pulse amplitude that can be varied. Also the pulse position can be varied according to the amplitude of the message signal. So what we can do this, because it is a periodic one, so it has a definite pulse position.

Now we can start varying this position of the pulse keeping the pulse width and amplitude same according to the message signal. So let us say we can have this convention that whenever the pulse amplitude is higher then probably we will deviate from the center point little bit, whereas whenever the amplitude is lesser will deviate much lesser and something like this. So with this we can have a pulse position modulation where the pulse width remains the same, pulse amplitude remains the same.

It is the pulse position related to where the pulse should be, original pulse should be, original periodic pulse should be. So if we do that then the position of the pulse is actually carrying the information of the, about the message signal. So that is one way of doing it, another way of doing it is now because we have three parameters one is the pulse amplitude, other one is the pulse position and the third one is the pulse width.

So we can also do pulse width modulation okay, so this one was called as pulse position modulation, and in width the pulse remains in the same location as the input sorry go from here okay as the input. But the width will be now varying according to the amplitude, so this might have a higher width, next might have a lower width and so on, this might have a bit higher and so on.

So basically the pulse amplitude remains the same, and the position remains the same that is starting point of each of the pulse that is fixed at after every Ts duration the next pulse will be starting. But the width of the pulse will be now varying according to the message signal. So these are the three possible modulation that can happen. So we actually call these kind of modulation as discrete time modulation.

So basically earlier whatever we are doing the signal was continuous time, because we had a career and that was existing for all the time instance. Whereas, pulse that exist for some duration of time, rest of the time there is nothing. So we can call this as discrete time modulation, whereas the amp means the information that it carries that can be means that is not discrete that can take any value so either it is or width or position or amplitude that can take any value so that is why it is still a analog modulation because the information that it carries that is in analog form that is not discredited okay.

So to develop the theory of this modulation what we have started is something like this initially we have talked about theorem which is well know and all of you might be already knowing in some version so we have given a theorem which is called the inquest sampling theorem okay and at that point what we have taken.

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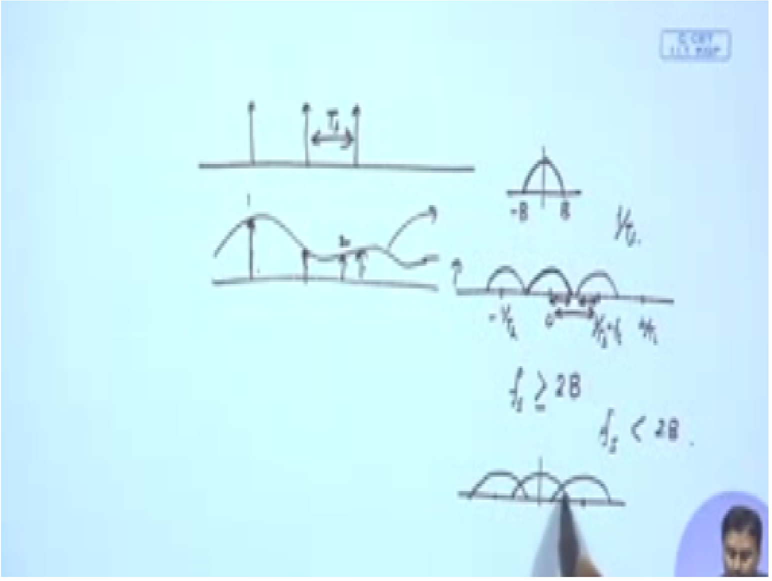

So I am just quickly recapitulating what we have done so inquest sampling theorem means the pulse is where we still have pulse train but the pulse where impulse okay so individual pulse was just a impulse function or we should say conical δ function okay so it is just a δ function continues time δ function so which is defined has the pulse okay with that we started actually means started doing almost like pulse amplitude modulation or we where rather telling that because it is a impulse function so we are actually telling them as if we were talking samples of the signal.

So if w had a signal we are just multiplying by this so what because if you multiply by δ it will just pick the sample at that instance so it was just picking this instances the sample value at that instances and these sample values where now the where represented as discrete time continuous amplitude so it is still analog signal okay.

So what we could see from inquest sampling that basically the spectrum suppose this particular signal just the signal has a spectrum like this it is a band limited signal from –B to + B if we just sample it goes down by so suppose this is 0 goes down by actually 1/Ts the Ts is this or 1/Ts is the frequency of those pluses so basically it goes down by 1/Ts and it gets repeated at the sampling frequency so basically if this is 1/Ts so this is will be 1/Ts next will be 2/Ts and so on and similarly it also gets repeated over – 1/Ts or you can call that as Fs okay.

So this is what happens and then whenever you do sample why we are sampling it we want to reconstruct the signal so that was our target that if we sample it we should have provision to extract the signal it is also most similar like that we modulate we must have a means whenever we do modulation we also have some mechanism to actually demodulate it so that we get the original message signal that here also we are trying to do same thing.

We are trying to see if we sample it or let us say with impulse train we are doing impulse amplitude modulation the we need to have a Mechanism which takes my signal out on disturbing so if that has to happen then has you can see this spectrum of this signal is remaining in tacked has long as we have already proven and this FC see this is B this is B that Fc is greater than 2B or I should say greater than equal to that equal to becomes that inures sampling where okay.

So long as that is happening there is no problem if it is not true that fs is less than 2B then what will happen when ever I do this sampling so that means actually you are under sampling so you take a sample over here because fs is less than 2B so what you do you take sample over here so this is will be the second sample probably this will be 3rd sample so if you under sample it then what will happen this things.

So because fs is now no longer so this is fs no longer greater than 2B so there will be a interpenetration of these 2 spectrum and then what we have a distortion over here okay so this is called aliasing okay so it will the spectrum will look like because these two will super impulse so it will be smoothen out over here so it will look like something like this okay so if you now see this is not equivalent to our message signal, so if I satisfy the Nyquist criteria that means my sampling frequency is greater than to twice of the band width of the message signal.

Then I can always say that I can employ a ideal low pass filter okay which will exactly take out because there is no aliasing there is no super imposition of the neighboring means that spectrum that is being created side spectrum so there is no over lapping so the spectrum remains in take and I can put a ideal low pass filter of band with d and I can take my expect my signal of so of this nyquist criteria is satisfied I can get the signal of the whole thing in a way we can also explain this physically,

So if suppose I have a signal what we mean by highest band width okay so highest band is suppose be we are saying that means that is the highest frequency so over a unit time because this is the highest frequency so they will be means of rate of change of the signal so basically if my sampling rate is somewhere related to this particular band width so basically I will ensure that I sample at instance where I can still have the tracking of the signal, so what does that means that actually mean.

Suppose I have a signal like this and because this variations are there so a particular highest frequency component is already there if I start sampling it like this then what is happening if I try re construct it this sample this sample and this sample the easier method is interpolation and if I interpolation and if I interpolate them we will see probably it will b smoothen out so all this variations are not getting captured, whereas Nyquist criteria is saying that I have to depending on the bandwidth I have take enough number of sample.

So that this particular variation within time is being captured by those samples and it nyquist criteria tells what are the minimum number of samples in depending on the bandwidth of the original message signal are required to represent the signal okay so this is something we have already done Nyquist theorem also and we have done another part of it which is called the interpolation theorem, so basically what we have shown through Interpolation theorem so I am just recapturing the capturing all these things we are not again mathematically proving it because that part we have done.

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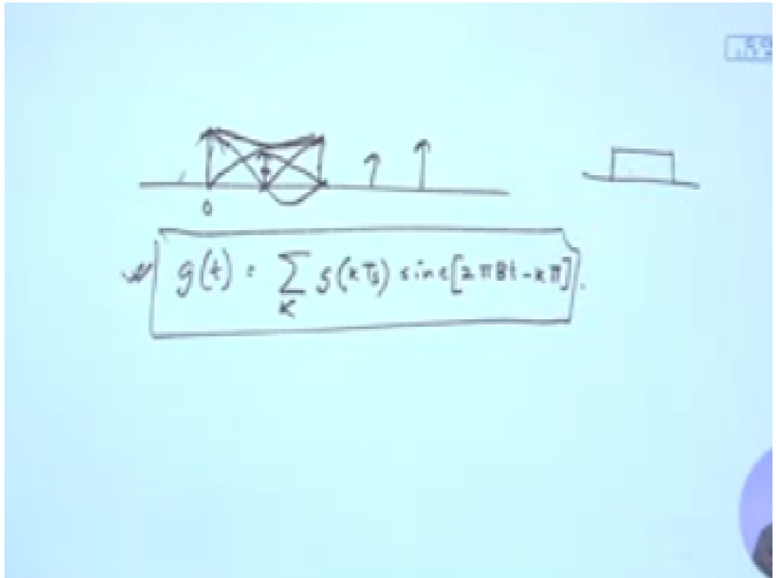

So what we have said suppose I now have just the samples okay whenever we say we pas it through a low pass filter so low pass filter it is see the transfer function that is a basically a box function, so what will be the corresponding Fourier inverse transform that should be a sink function so if this impulse is or getting passing through this particular impulse is consider should be a convolution and everywhere a sink function will sit together and sit one on top of one another and we have also proven that the 0 are exactly over the other samples.

So if you start putting all those sink function and we have proven that if you just a every sample you start putting the sink function corresponding sink function with this amplitude, so you will see the whole signal those sink function getting added will create the original signal original message signal which will just go through those steps okay, so this is the reconstruction theorem so we have already told that our gt a message signal can be represented as this g of KTs.

So these are those sample values multiplied by sink function okay sink function appropriate the delayed so depending on the value of k so that is why you are putting a sin function over here over here depending on the value of k if k is 0 you put a sin function over here k is one you put a sin function over here and so on okay. So accordingly you fill up all the sin function and that should give you the original message signal back okay.

So this is call the interpolation theorem so basically Nyquist sampling theorem and interpolation theorem together they tell us that those samples whenever you follow the criteria of Nyquist sampling those samples are good enough to represent the signal okay, if you just pass it through a those samples if you just pass it through a the pass filter we probably get your signal back so this is a very important result and this also gives us some possibility of again multiply signal so that something let us try to see what happens exactly.

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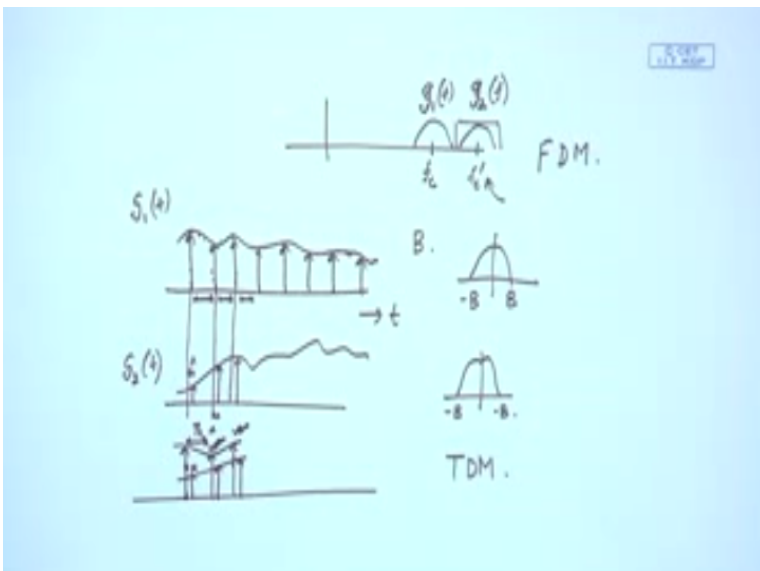

So basically suppose the multiplex was one of the important thing that was required for our transmission so we have said that suppose I have multiple signals okay and I want to simultaneously transmit that over a same media so let us say here is the media or if I have a wired media so everybody wish to actually simultaneously communicate through that same media okay so this is something we have done in amplitude modulation or even frequency modulation, so or general modulation theorem.

So what we are doing we were actually in the same media we were modulating with the different frequency of a carrier so what was happening in frequency domain, they were suppose this is corresponding to f1 this is f1 f okay this is f2 or let us say g 1g g2 f and so on so this is at fc this is at fc- so we put them in different frequency so that they co-exist in different frequency band and whenever we have to demodulate we first employ a band pass filter we have already seen that band pass filter is probably the first part of the modulation.

So we employ band pas filter which is exactly matching with the desire signal so basically the desire signal will be at some frequency if I already have understanding means I know already that frequency so I will be putting my band pass filter centered around that so that I can just filter out the desired signal and I can reject all the other signals okay. this have to purpose we have already we have seen that one is that it actually means take the desire signal and also it is reject some of the noise which are present in the entire band.

So this is something we have done and this particular technique was named as FDM frequency division multiplexing okay, so we were trying to do frequency division multiplexing over there so that we can reuse the channel for in actually putting simultaneous transmission. This is something we have already discussed we have done that through modulation. Now by this sampling theorem we have another opportunity so what is happening in sampling theorem, we suppose I have a signal let us say that is again g1 t okay I put few samples to represent it now if you just see it in time domain these samples we have already proven through interpolation theorem that they are the sole represented if of the entire signal.

So I do not need all the description of this signal for the entire duration of time I can just take those few samples that means discrete time description of the signal is good enough for me to represent a whole signal okay. if the signal is band limited as like visit as already a point it out so if this is something which is happening then I know that these samples are good enough to represent the signal so basically what this is doing? This sampling theorem is doing it is freeing of some of the time which are not be utilize now for transmission of the signal.

So whenever we transmit the signal now after doing this sampling we will be only transmitting those samples so only few instance of the entire time line I will be transmitting those samples rest of the time my transmitter as well as my receiver will be ideal so I can actually employ these thing to multiplex other signals okay so that is my purpose now what I want to do suppose I have another signal g2t which is having a different shape so I also can samples so this was the sampling instance so these two signals simultaneously exists so time has unite definition and what I do because I know that these particular parts are free in time domain.

And my sampling does not say exactly were from I start sampling it can be anywhere but only the regular interval should be maintained so if this two have equivalent band width okay so that means this also is having B may be the shape is little bit different so –b to +b and this is probably another noise signal suppose let say this is noise signal so this 3.4 kilo watt I have another noise signal where the shape is little bit different.

But it is still –b to +b is that happening when the sampling rate will be same equivalent or both the signals but what I can do I can actually start sampling it little bit opposite so let say that is Δ off set so then because the sampling rate is same so it should be second sample also should be Δ offset and so on for every other samples.

Now what I can do I can actually simple impost these things add these two signals and transmit in the channel so how that will look like so there will be one sample from here and next sample will be from here then one sample from here next sample will be from here so if you just keep doing that one sample from here and next sample from here if we start joining the corresponding sample tip you will get the original message signal back but these samples will co exist in the channel.

Now what I have to do is if I do this at the other end I need to know exactly the timing is very important the earlier it was happening when I was doing frequency dividing multiplex in the frequency was important the frequency information should be note that means which central which carrier frequency I have modulated I have to put my band pass filter accordingly.

Here the multiplexing is being done in time domain so that is why it is called TDM time division multiplexing and the time information is very important so I need to precise it know where my first signal sample start if I know that I only take this sample and then skip all the samples which are intermediate which are due to multiplexing other signals actually.

So I skip this side signal and I can pick this one just after that Ts deviation sampling or inter sampling deviation so I can pick this signal as long as time synchronies then I will pick this one so I will be picking my signals and then I reconstruct that means I just pass it through low pass filters I reconstruct the whole signal and I get back my original signal okay.

So this is something which is possible so this has been made possible because of sampling theorem so you have see there was probably in communication or analog communication I should say there are two fundamental theorems one is called the modulation theorem which actually if you multiply with the carrier it translate the signal to a different frequency.

And if you sample it properly knowing the band with and then following criteria then you are actually freeing up sound the time between samples and where you can multiplex other signals so these are potentially to multiplexing schemes that has been employed in communication you will see later on that mostly your aim modulated signals fm modulated signals goes through this.

Because it is already with the pulse so it goes towards the t multiplexing scheme so it can still be done width fdm and sometimes the mix of tdm and tm is being employed so these are the things which can be done but pulse modulated signals they are closer to time deviation multiplexing and that is why tdm is employed for multiplexing in pulse modulated signal okay.

So this is something which we have understood now let us try to see another every fundamental thing in communication or this kind of pulse modulated signal so what is happening I have a signal I sample it right let say these are ts and my sampling frequency is 1/ts okay which satisfies the criteria so that means fs must be greater than equal to 2B okay.

So this is something which is being satisfied so minimum I can take to make it let say because if I take because one my samples will be closer because sampling frequency increases so ts will be decreasing because its samples will be closed less time I will get between the samples so the best I can do is if fs is just to be and I am sure that the highest frequency whenever the band with I defined highest frequency does not have any means corresponding all component okay.

So that is something which is required because otherwise there will be again so if the highest frequency does not have any significant power component or it does not have any impulse or in kind of carrier so if I just sample it at 2b then what will happen this will be 1/2b so that TS becomes because FS becomes 2b and this becomes TS becomes 1/2b okay so this is something I already know now let us try to see how much of information I am transferring okay with this so basically whenever I am transferring how much information I am transferring see how many samples I am transferring each of the sample actually carriers information okay.

So if the frequency is the time is 1/2b frequency is 2B so basically 2B samples I am transferring per second right and that occupies how much bandwidth whenever I am transfusing it off course because of sampling it is go over here over here and all those things but these things are not required for me when I will be at the demodulated side I will be putting I will be putting a filter over here okay so basically my signal is confirmed over this okay so B what does that means so basically this 2B samples I can now transmit with D bandwidth because this 2B samples I am transferring these are the information.

So 2B information for second I can transmit with b bandwidth okay or B arch bandwidth if it is represented in F domain not in ω domain so with B, 2B information I can transmit so for hurts how much information I can transform basically 2 information per second so this is the very fundamental result of later on we will see for digital communication this is probably the most fundamental result that every hours if you do sampling every hours possibly carriers two information whatever that information is per second for every hours is capable of carrying two information for second this directly comes from the sampling curve okay if you do sampling little bit means if you do over sampling probably this will be little lesser.

But this is the best you can do every hour at most can transmit to hertz per second okay so this is the very important and fundamental restriction we should say whenever we employee sampling and then each samples are information for us okay so later on we will see that when the from pulse amplitude modulation to pulse forted modulation the PCM we told this is actually curios the amount of bandwidth that is required for a PCM transmission.

So that is something we will see later on but that is for now for the time being we should be happy about this result so basically in this particular class we have discuss three fundamental result side by side one is the modulation techniques which is the multiplication with the carrier frequency and there is a translation which helps in FDM because the division multiplication the next one is the sampling theorem which helps us in this is actually sampling and freeing up some of the time.

So any signal does not have to represented all the time it just needs to sample you discrete time representation of the signal so that is how we get time division multiplication or we have opportunity to do the division multiplication and then we also understood whenever we do this sampling and each sample becomes information for us so every hertz is capable of transmitting two information per second okay so that is the relationship between bandwidth and the information transferred okay so this relationship will keep on talking about later on and whenever you will be doing the digital communication probe this is the most important and fundamental process that will be required okay.

So with this background what we will do we will start now exploring some of the more generic practical sampling that can be happening because this impulse sampling is something which is in practical because each impulse will have a infinite energy we have already proven that so that is not possible we cannot really generate infinite power so we need to see if we employees some practical which are the restriction of circuit if employee those kind of sampling what happens okay so that will be our next discussion thank you.