Signal Processing for mmWave Communication for 5G and Beyond

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Module - 02
Wireless Channel - A ray tracing model-Part-II
Lecture - 09
Wireless channel - A ray tracing model part - II (cont)

Welcome to Signal Processing for Communication for 5G with millimetre Wave. So, today we will be covering the lecture number 9 for the Wireless Channel - A ray tracing model - Part 2. Today we will explain more into the digital channel part. So, last week or last class we have explained more on the IIR to a FIR conversion of my channel right.

So, if you look at the last class note you will see that the channel was actually an IIR filter and we have finally, converted to some sort of an FIR filter by a simple method of windowing ok.

Windowing meaning you just discard the values which are of no important to us, but that should be at the end. Windowing does not mean that in between there are 0 you just discard them there will be 0, but it is just like a delay element kind of things. So, we will talk more on that.

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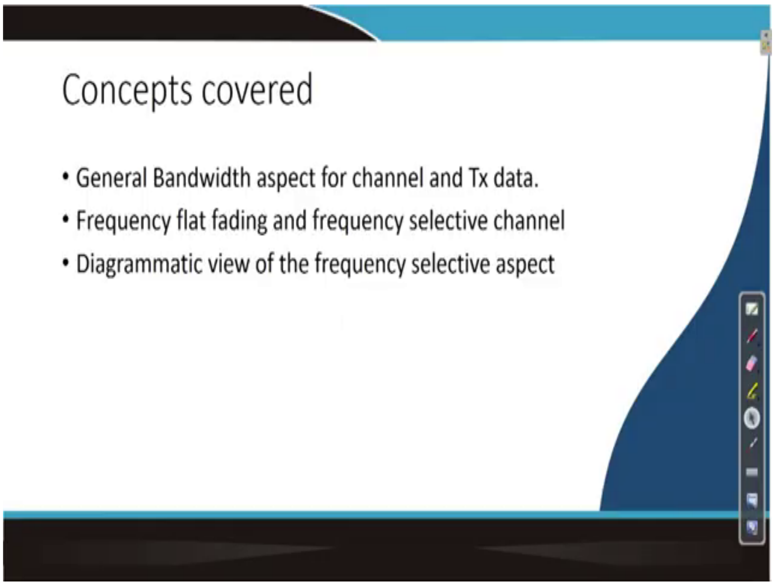

So, things that will be covering are the following general bandwidth aspect of channel and Tx transmitter data. So, we will be showing how exactly they will be related and what happens when one of them is greater another one is smaller so that kind of conceptual aspect. Then I will be talking about frequency flat fading and frequency selective channels and then some diagrammatic view of the frequency selective aspect.

So, what exactly it means for it, for example, there will be a FIR filter IIR filter of the channel models that will be take care of. So, what does it mean? It means we have the channel.

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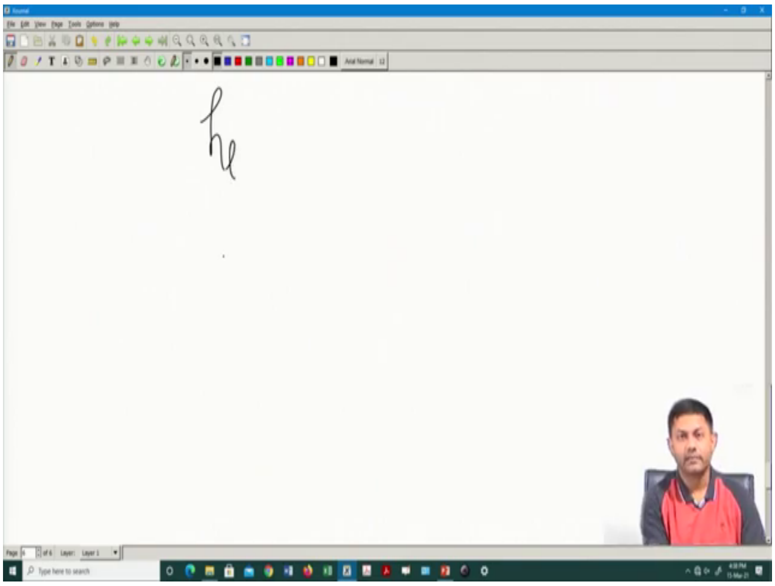

Say let us say, so, I can now straight away think of it as like a system of linear equation just it has the components whatever delay components and the channel components we have this is like a filter. So, when you have a filter you have lot of coefficients into it right. So, each and every tap is as if like a coefficient for that filter that is precisely what is what is there. So, now, let us say this is the lth tap and I let me just generalize it.

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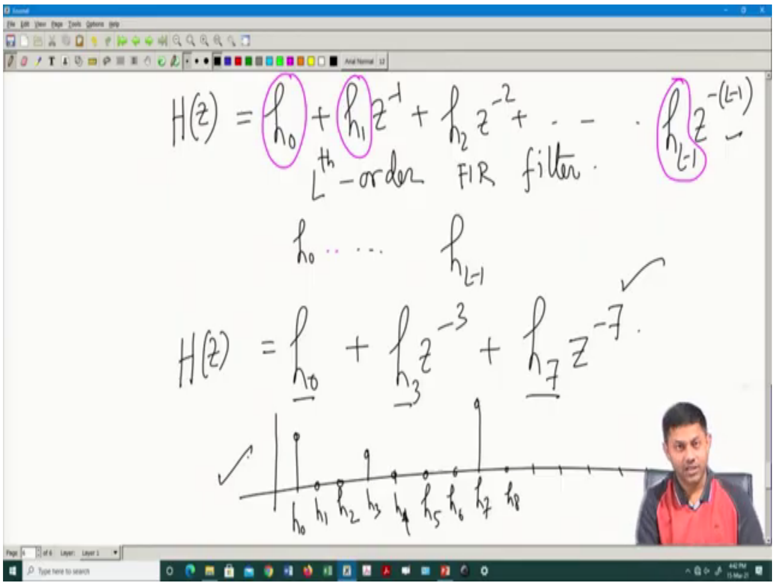

So, instead of h l if it is a filter. So, obviously, I can have some sort of a transfer function of it right. If it is a system of linear equation I can get a transfer function of it. So, this H Z is the Z transformation of that individual h l. So, what does it mean? I can say this is nothing, but h 0 will be there plus h 1 Z inverse plus h 2 Z inverse 2 plus dot dot dot. Say I have considered up to l number of taps because beyond that I do not see any more taps coming up ok.

So, it will be h l minus 1 Z to the power minus l minus 1. So, this is my actual term. So, this is an Lth order FIR filter. This is an Lth order FIR filter ok. But note that though it has its order is L it does not mean that each and every components or the coefficient within this filter of order L will be all non-zero that is not guaranteed. Why?

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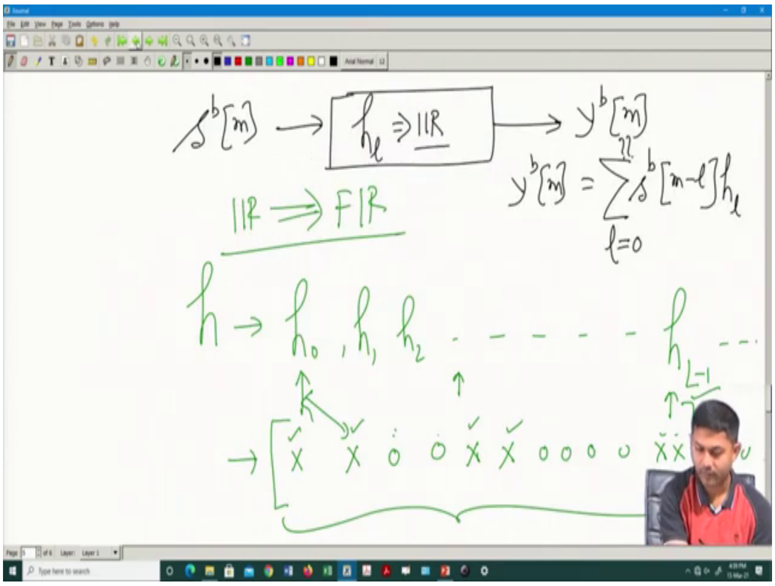

Because look at this, look at this.

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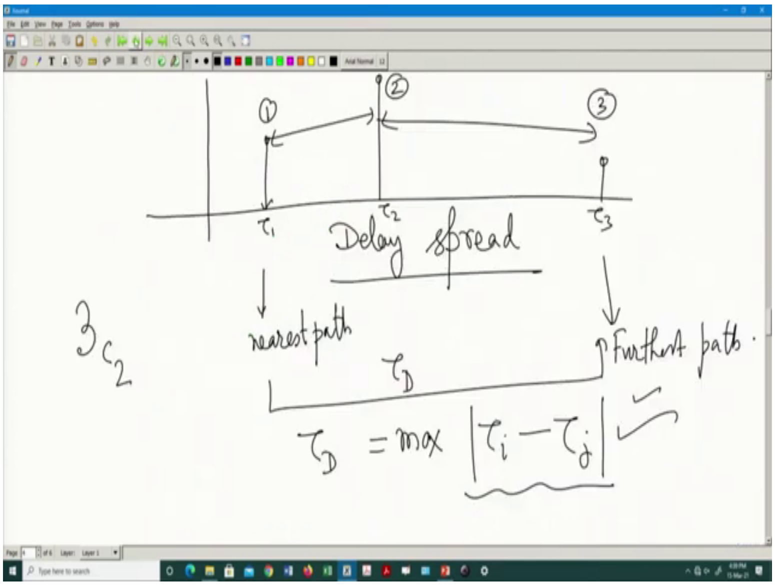

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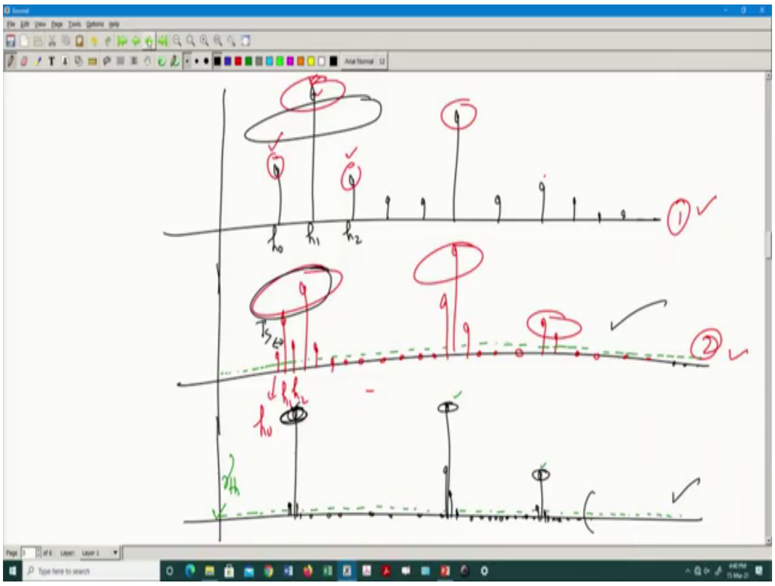

Or look at kind of this diagram, look at this diagram. Let us say I am in the second one second sampling. You will see that initially few taps will be 0 non-zero then again some value will be 0 again some non-zero some will be 0 again non-zero and finally, it will be 0.

So, that mean between the coefficients some of the coefficient can just vanish because of this non existence of the original you know reflectors. If I have sample it even more probably I have to increase my order here order of the filter is more, but you will see lot of 0 in between. Finally, of course, this part is 0 that is fine, but you may see that. So, in a conclusion I can say that though its order is L not every components will these are all like a component.

So, this we call it taps right or channel coefficient if you think from a filter point of view. If you think from a plane filter point of view they are all nothing, but the coefficient of the filter.

So, this is the these are all coefficients of the filter or nothing but the tap what usually I refer to taps.

Now, even if its order is l does not mean that all the h 0 to h L l minus 1, they will be non-zero there is no guarantee in between there will be lot of 0s. So, it may happen that just take an example. Let us say I am taking an example say h 0 H Z is equal to h 0 plus let us say h 3 Z to the power minus 3 plus h 7 Z to the power minus 7, suppose this is the filter. So, what are the coefficients? Does it mean that its coefficients? It has only 3 coefficient? No, it has actually 8 coefficients just that some of them will be 0 because its order is 7.

So, what does it mean? It means that probably h 0 has a value, but h 1 is 0 h 2 is 0. So, this is h 0 this is h 1 this is h 2, but h 3 has some value. Then 4 is a 0, 5 is a 0, 6 is a 0, but h 7 can have some values and h 8 if it exist though we do not consider anything. So, after that every point is 0. So, this is how the tap is, this is how the tap looks like. So, never say that just because order is 7 and some of them are 0 it does not mean that those coefficient do not exist just that it is 0 that is all because it is in between, but of course, beyond the highest point everything else will be 0.

So, this is a typical you know typical channel examples where in between you can have a 0 ok, this one point. Let us do some more characterization of the channel part. Now, as we discussed last time delay spread right. Now, delay spread what happens to the digital cache because in digital the exact time difference between the reflectors as I perceive after the sampling may not be the same right because it is uniform sampling right. So, now, how do I define the delay spread there?

Does the delay spread completely changes here? Well tentatively, but delay spread is an inherent property, it is an inherent property of my channel things right. So, they may not change, but rather I may not be able to find out exact the delay spread just looking at my samples only if I do a higher sample. Because only if it is a higher sample I can exactly somehow point out what is the spread of my you know what is the where the exact reflectors or scatterer they are presented.

But unless I sample it very high speed it is slightly difficult, I think we have just explained it. So, the delay spread becomes some sort of a intrinsic property of my channel.

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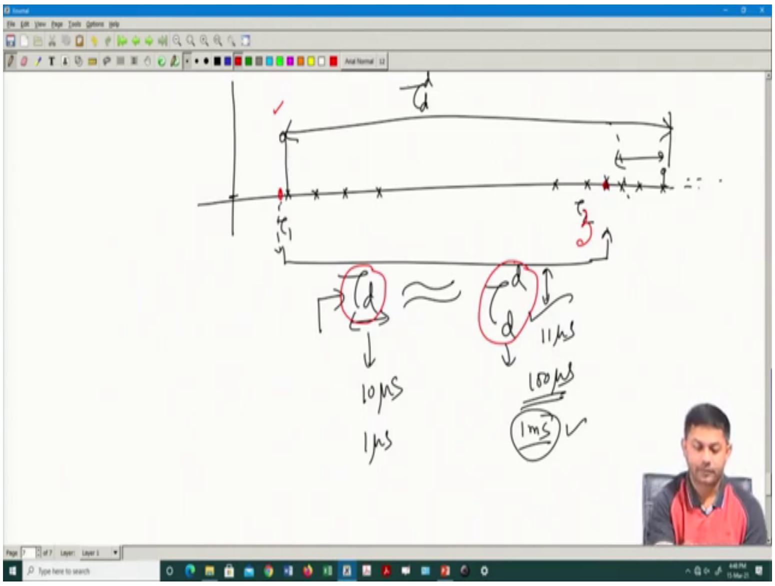

So, even if I sample it even if I sample after ADC they become uniform, but that is ok. So, probably your tau 1 somewhere was here and tau 2 somewhere here. Somewhere your tau 1 may be say somewhere here or maybe somewhere here maybe somewhere here your actual tau 1 and your and your last tau say tau say 3 was somewhere say somewhere here.

So, does not mean that after digitization I will not get anything beyond tau 3. So, I may still get something, but it may die down quickly it may die down quickly it may die down quickly. So, the actual delay spread which we observe for the channel here say tau d and the channel say for example, I observe my first step somewhere here and I observe at least last step here anything beyond that I do not get it.

So, my this is my tau d I would consider because that is the delay between the first step and my second and the last step and that is the definition. So, this is my tau d, but it is actually digital. I say this is this is d, but it is digital tau d of d I call it. Now, there is no guarantee that tau d and tau d d will be equal, but they related ok. So, I cannot say they are equal, but I would say they are of the order naturally there may be extra few sample they may come; obviously, because it is a sinc right. It depends on how you are sampling it ok. So, but there is a correlation.

So, this is my RF analog exact tau d the delay spread give this is after digitizations ok. So, they may not exactly match, but they are related. If this is in the order of say I just give an example 10 micro second, you cannot expect that this becomes 100 micro second that will never happen; obviously, right.

If there is if this is the last step physically and there is obviously, a sinc. So, after one or two extra you may get it one or two extra not more than that right. So, if this is it will never be 10 micro second, but I cannot say how much it would be, but; obviously, it will be very close to that.

So, probably it will be 11 micro second something like that or maybe 9 micro second something like that. So, may not be equal, but it will not be of that huge order or if this is say 1 microsecond you cannot expect that this becomes 1 millisecond ok. It may become 1 millisecond provided you know lower your threshold and probably that can drive what should be my threshold such that it really matches my taps here. So, you cannot expect this kind of things. So, this is my delay spread definition.

So, I will not consider from now it depends where exactly you can think of your delay spread. Now, as you notice in practical sense when you have a modem if it is a digital momentum right, this is difficult to find because how the reason is that everything you observe a system only by sampling right.

You do not do any processing at the analog on RF right. So, there is nothing called analog processing because had I done an analog processing where I can precisely see where my taps are it easy to find out, but I do not do anything in analog or RF.

I do everything in the digital side. So, which means I have to pass through the ADC. Moment I pass through the ADC, it means the concept of tau d does not exist for me. What exists for me is more of this one digital you know digital delay spread that matters to me ok. So, this is one definition that how exactly I will consider, but as we notice they are of the same order tentatively ok. So, we have learnt delay spread. Next some of the parameters that I would like to have it is the following.

So, what about the bandwidth? Bandwidth of what bandwidth of the channel because it is a system of linear equation it is like a filter right. So, if you say filter has a bandwidth either infinite bandwidth or some finite bandwidth everything comes into place. Whatever parameters you can have it in the filter the all of everything will be there.

It may have a ripples, passband ripple, stop band ripple, it may have a delay characteristic, it will have a phase characteristic, it will have a bandwidth concept, roll off factor whatever is there in the filter it will also have the same thing. So, let us characterize that right.

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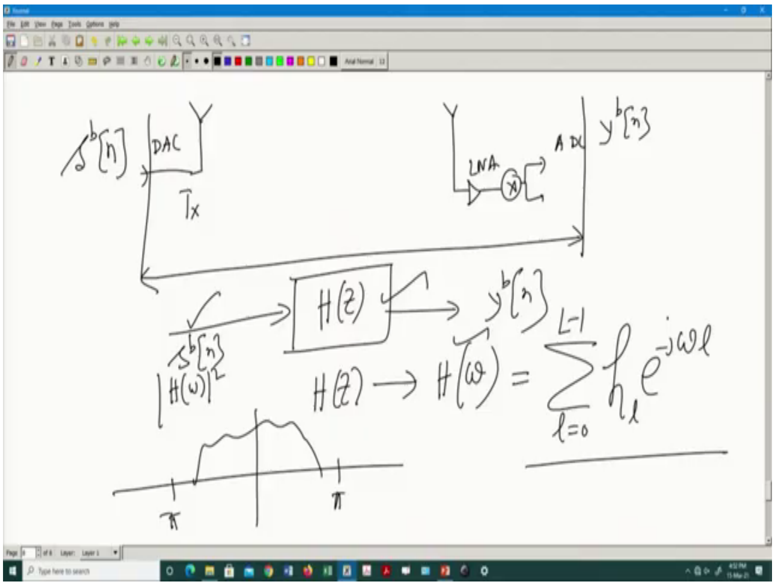

So, now we have a two point. One point is that so, this is my antenna, where I have done the transmission, but as you notice I am viewing the whole system from a digital side. So, I will not draw s t here. What should I draw here? s b n because I am viewing my system s b m or s b n that is ok, you can write n also if somebody is not comfortable with this parameter s b n some digital, pattern because the whole analog RF everything I push it as a channel that is what my channel. So, everything.

So, beyond this point I have DAC, I have different filters, modulator, power amplifier all damp things I push it as a channel and when I have a receiver side I have that you know that LNA, demodulator, bunch of filters and finally, you have the ADC. So, here I am standing right.

So, I have y b n. So, my view of channel is from here to here. This is whole my channel. Here is not just the only part everything else is my channel ok. So, my view is that as if like I am sending s b n and I am viewing or I am getting I am receiving y b n. So, my channel was H z.

So, finite impulse response we have discussed it. So, this is my s b n, this is my y b n, alright. So, what is this how do I how do I estimate my spectrum if this is the finite impulse response? This is the Z transformation. If I know the taps I can easily get it, just replace Z equal to e to the power j omega fine. So, if you have H Z, I can easily convert it to H of omega. How do I convert? I just replace. This was l right, this is if it is a Z transformation is now this will be converted, this will be replaced. This is what it could be.

Now, l equal to 0 to now I have created a finite impulse response just by doing some sort of a thresholding. So, this is my channel spectrum. Now, this is the H omega. Now, you plot it you plot mod of you know H omega we will get some spectrum whatever it is. So, it will be pi to minus pi we will have some spectrum that can happen ok. Let us understand more on that. Now, you have a problem. What is the problem? Problem is that you already have one spectrum from your transmitter side. Why? Because you have a DAC. So, DAC has a sampling rate.

So, which means that you have certain power and certain bandwidth of your transmitted signal that get transmitted through this channel and finally, you get receive you are receiving that. So, the effective bandwidth will be a combination of your channel as well as your input data ok.

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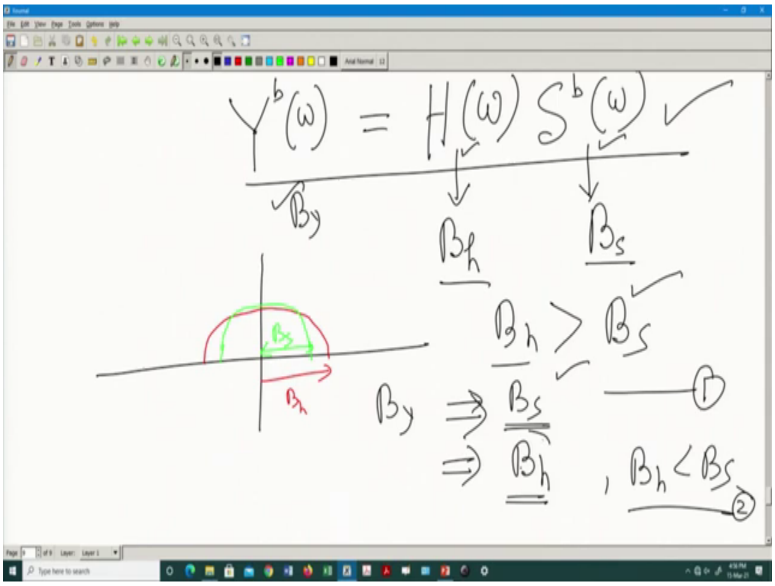

So, can I say if I take if I take Y b omega I can naturally see it will be H omega and S b omega. So, system of linear equation static system. I am not creating any time varying system, otherwise it will not work. If it is a time varying this does not work out ok. If it is a static system this works out pretty well. We will explain how what happens when it is a time varying and that is your Doppler comes into. So, this is what it is.

So, what does it mean? It means that it depends on this bandwidth and this bandwidth based on that this Y b omega comes. So, let us say this gentleman has a bandwidth of B s. So, baseband bandwidth and this gentleman has a bandwidth of say let us say B h that is the baseband bandwidth of my channel because everything is the base band for me. It is a digital baseband for me ok.

What should be the bandwidth here? It depends who is bigger alright. If B if B h is greater than B s, what happens? It means your channel bandwidth is more than your data bandwidth. What happens? It means this is a scenario. Now we need to understand the impact of such kind of scenarios. Let us say you have this B h. So, this is your B h ok and this is more or less this is more or less your data bandwidth ok. So, can I say my B; let us say the bandwidth at the output or receiver side is B y.

So, can I say B y will be nothing but B s it will be equal to B s know? Obviously, right because B h will pass on everything or it will be B h, if B h is less than this is a very standard you know filtering concept nothing great about that, but let us understand what is the impact for case number 1, what is the impact for case number 2?

What exactly it means that your channel bandwidth is greater than your transmission bandwidth and channel bandwidth is less than your transmission bandwidth, what is the impact of it? Naturally if you see that in the case number 1 when the channel bandwidth is more than your B s what does it mean?

It passes most of the component of your transmission system. Obviously, there may be phase distortion, but at least the components go through with some distortion in the amplitude, but look at the second case. Second case channel bandwidth is lesser than your system bandwidth.

So, it discard some of the components of your data which means it degrades the quality a lot right. Now, let us understand what does it translate to in our system when the B h is less than your base? That means, your transmission bandwidth is much greater than your channel bandwidth.

So, channel bandwidth is smaller ok. So, let us understand that part what happens, what is the what is this two cases.

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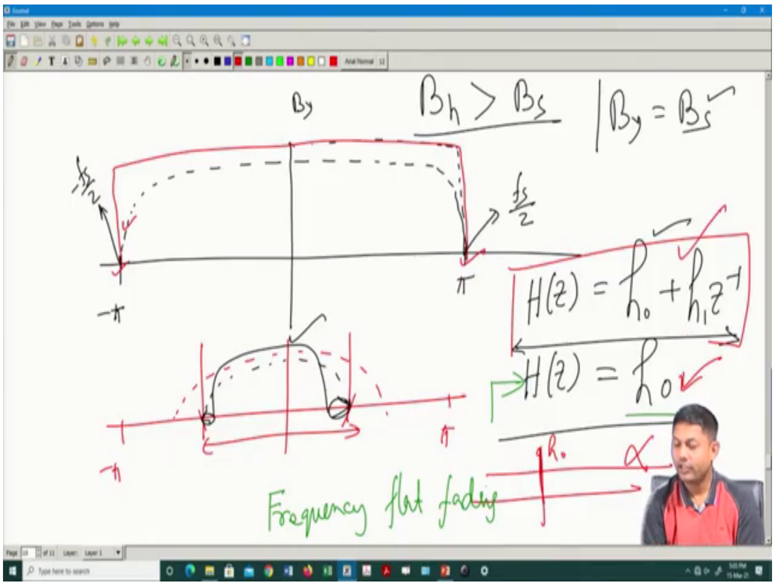

So, first case is your data bandwidth or your rather channel bandwidth is much greater than your data bandwidth. So, your total output bandwidth your B h or your B of y will be equal to your B of s, it passes on each and every signal right. So, what does it mean in every simple sense? That mean I am seeing that whatever I have transmitted varying the phase part phase distortion part each and every frequency component get passed out. Got my point?

What I am saying is when B y is equal to B s what does it mean? It means that each and every frequency component present in my transmission signal they all pass through. When can it happen? What should be the channel or rather than the filter such that it passes on everything? So, can I say if I plot say for example, let us say this is my pi to minus pi, I have adjusted my sampling in such a way that within pi to minus pi I fully can see my B s. So, this is my f s by 2 sampling f s by 2 at the transmitter side.

So, I have adjusted my you know sampling rate such that it completely occupies the input bandwidth part. Whatever input I have sent the same spectrum it just fits into pi to minus pi. Let us assume that I am doing that kind of sampling. So, what does it mean? It means that the whole B s that whole transmission bandwidth is exactly fits in my pi to minus pi without any distortion without any component lost, nothing will happen. When that can happen that when you know whatever I transmit everything I get back. Can I say can I say if a channel is something like that?

So, let us say I have a channel which is just say h 0 plus h 1 Z inverse or I have another channel just one tap. Now, let us understand out of the two case which case we pass on or pass each and every component of my data transmitted data provided within pi to minus pi, I exactly fit my input data. That mean either the filter is as if like this original channel filter then only it can happen. Now, it can happen right. Then only it can have it passes everything because it exactly fits it ok.

This is the worst case scenarios nothing more than that or it can happen that I am sampling at a higher rate so that I can view the data. So, this data see dot dotted data this is my higher sampling rate, my channel bandwidth is larger to it. So, with respect to my transmission window if I just do not look at anything beyond my transmission window what does it mean?

I will see that across my transmission window the channel has presence of frequency something that means channel does not give any distortion or does not cut off any distortion as long as my you know transmission bandwidth is wherever I am transmitting. That mean if I just transmit 10 megahertz and I do not look anything beyond that because that is my transmission bandwidth.

So, what does it mean? Is it is as if like within my transmission bandwidth my channel is just like a it is a it is present everywhere it is omnipresent. What does it mean? It is as if like my channel is an all pass filter correct then only it can happen. So, which means that in my pi to minus pi with the need my channel just fits.

If my data just fits and I am saying that after transmission my data has no distortion. It simply means that within my window of view if it is just exactly tight you know I complete ten megahertz say that is my transmission bandwidth is as if like my channel is an all pass filter. It is all pass filter right. If it is an all pass filter it just passes on each and every component of yours of your transmission bandwidth. So, which means this is one scenario where I can think the channel just passes of everything.

So, what does what is the bandwidth of such signal? Bandwidth is infinite because this is just one tap right one tap. What is the bandwidth? One tap in it is just one impulse, bandwidth is in finite right. If it is just one tap this is your h 0.

So, what is the bandwidth? Bandwidth will be in finite bandwidth. It is not infinite bandwidth. It simply says that within my viewing window can I tap have an infinite bandwidth? Virtually impossible, you cannot have it you cannot have an infinite bandwidth of the channel. It all depends on how you sample it ok.

So, which means that if the channel is just like some sort of a you know multiplier or just a gain increment or decrement obvious gain will decrease the channel is as if like it will view viewed as a infinite bandwidth channel. It simply means that the window I am looking for the channel exist everywhere in that spectrum. The window I am looking for, but channel exist in that window. So, it is like channel just has a like you want it is like a all pass filter.

So, such kind of channels if the channel is just a you know just has just one tap it does not create any distortion in your system in that case. It just passes off everything and it is as if like I am you know I am going through just like an all pass filter. So, such kind of channel I call it frequency flat fading channel frequency flat fading; that means, it does not introduce any frequency separation on your input signal it is just make everything goes through. It is a frequency flat fading channel.

But what happens when my channel is this kind of channel? At least is it has more than one tap. What will happen to it? What will happen to that? It no longer be a frequency flat fading because it is now it may cut down some of the taps, it is some of the frequencies not tap some of the frequencies of your data. So, it simply means that within my viewing window I am seeing some of the component just get.

That mean the bandwidth of this channel will be much lesser than the bandwidth of my transmission. So, what is the impact of it? Then only I can see some of the points or some of the data can be discarded. So, what it means in the time domain?

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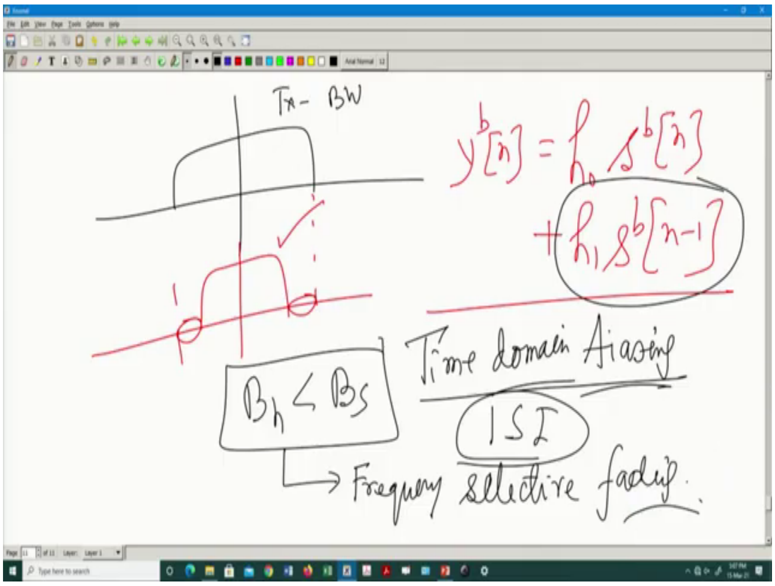

So, if I have a bandwidth of a signal bandwidth of a signal this is my say transmission bandwidth and after I do a reception it is gets shortened the bandwidth is shorter. What is the time domain impact? Time domain impact is that you will naturally see some sort of an aliasing because some of the components are lost right some of the components are lost.

So, what is the impact of it? You will see some sort of a aliasing effect on that because only aliasing can effectively kill some of the you know some of the data's that you are transmitting it. So, when you lose some of the components after transmission it simply means that you will may have some sort of aliasing in your time domain.

So, that is reflected when your channel is having more than one tap. So, it is discarding of some of the frequencies. So, it is as if like as if like your transmission bandwidth what your you know bandwidth of interest your actual bandwidth of your channel is shorter than that.

So, some component is just cut down. So, it simply means that the channel or the received data is having aliasing. Look at this. If you have this kind of filter what does it mean? What is the y b? y b n is h 0 x of n plus h 1 x of n minus not x. What is this? What is this component? That is aliased. It will be creating a time domain aliasing right. So, this is your time domain aliasing ok. So, this is also called ISI inter symbol interference because; that means, the previous data is now get an aliasing in my present data.

So, it kills it simply kills some of the present component and that is precisely what is happening here. So, which means that if the bandwidth of the channel is less than your B s the impact is that you may you will see this come time domain aliasing or ISI; that means, some of the previous data will now come into your present data and that is kind of an interference ok. So, now, this is your such scenarios is called frequency selective frequency selective fading.

So, one is a frequency flat fading it is as if like a all possible as if as if like all possible just one tap. More than one tap meaning the bandwidth is tentatively lower than your system bandwidth. So, it sees some sort of a you know interference time domain interference that will come into picture because you have cut down some of the frequencies right. So, it will create some sort of a time number aliasing or time domain interference and this kind of channel is called a frequency selective channels.

When you say frequency selective channel it simply means you have more than one taps. So, today we will stop it here. We go more details into the next class yeah.

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And the reference will be same as what we have said in the earlier class Fundamental of Wireless Communication by David Tsee P. Viswanathan.

Thank you.