Signal Processing for mmWave Communication for 5G and Beyond
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Module - 01
Communication system

Lecture - 01

Tx- Rx Structure

[FL]. So, we can now start the formal course on the Signal Processing and Communication for the millimeter wave in the context of 5G and Beyond. So, we will be covering today the communication system and the first lecture is the Tx-Rx Structure. So, in this case we will be covering the digital part mainly and little bit about the analog on Rx part.

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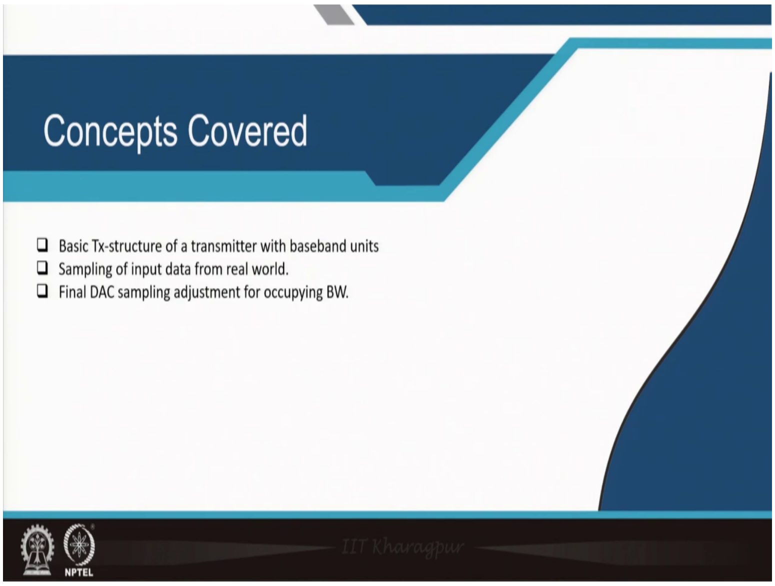

So, this is the first content of the first lecture; so, basic Tx-Rx structure of the transmitter and then sampling of the input data from the real world and then finally, DAC sampling adjustment for occupying the bandwidth.

So, this course will be mainly talking about the signal processing aspect which is mainly the failure because you know in a communications say for example, you talk in a mobile right. So, when you press a button suppose you press a number, so it is not the exact communication that is starting immediately. So, when you press a number, it simply means that you are working on the application layer.

Then you say ok go and put a call. So, what does it happen? So, from the application layer it goes down to various intermediate layers like for example, application layer, then it come back to the transport layer, MAC layer or the network layers and finally, at the end it will come to the physical layer, where the actual DSP or the signal processing will all appear and then it will go through the analog RF and then finally, at the receiver also the reverse generally happen.

In this particular course, we will be mainly talking about the signal processing aspect which is the physical layer part that is the bottom most part of any communication layer ok. Now when you do that the prerequisite of this course I think we have already sent with the content, the prerequisite of this course is that you should have that digital communication and the signal processing aspect. Now, in this case I will diagrammatically brief about how exactly a communication system works in the modern context.

Like for example, 4G and which is another part is a 5G and subsequently probably after say 10 years or 8, 9 years 6G and so, and so, forth. So, what is the basis or what is the fundamental structure of the communication system that appears?

From there we will start and then we will talk about the modem where the actual transmitter and the actual receiver is sitting there how they work and what are the different components sitting there; and then finally, we will enter into the some of the components detail and then subsequently, we will target the channel part, where exactly our interest of this particular course ok.

So, let us brief about what exactly happens when we say speak for example, you have a mobile phone and you are talking over that suppose you make a call and then you are talking over that then what happens? So, that means, you are making a voice signal right. So, voice signal will be something like a say 20 kilo hertz total voice signals spectrum then it will be what happens thereafter?

So, that is exactly what the modem will do and then finally, we will see where the channel part will come into picture because that is the motivation of this complete course content. So, let us say you speak; that means, I start as a voice signal for my complete modem system.

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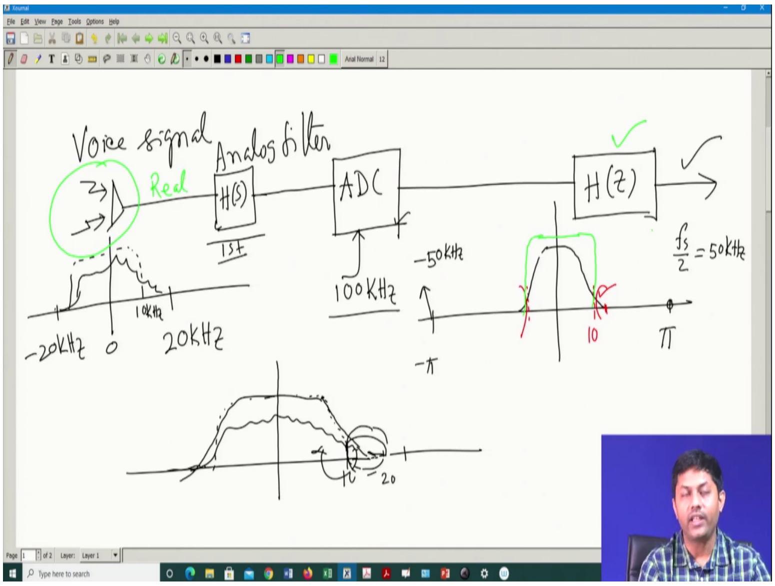

So, let us say I have a speaker here. So, lets say I have a speaker here and I give input as a voice signal say I have a voice right. This is a voice signal. So, what is the spectrum of this signal usually? Usually it will not cross 20 kilohertz right.

So, it will not cross 20 kilohertz in a baseband. So, it will be 0 to 20 kilohertz and there is a minus 20 kilohertz will also be present because of the (Refer Time: 04:35). So, the signal will be anywhere within it depending on your spectrum depending on the male female or if there is a any instrument or a background noise, the spectrum will not cross beyond 20 kilohertz.

So, this is the data which say you want to transmit it and we will see how it can be transmitted and what are the basic components that will be involved here and then we will try to understand and where exactly the channel millimeter web channel and because that is the focus of this particular course. So, let us say I am speaking it ok. So, you are making a voice signal 20 kilohertz voice signal, but that is not the one which will be directly transmitted as it is right because this is an analog signal.

So, but we are in the domain of digital. So, we have to digitize it and we have to do some sort of a signal processing activities, then a lot of channel coding and so and so forth will happen subsequently, now let us see what happens. So, this is the data where I am you know sending the voice signal then what happens?

This signal will be going through some basic analog filters why these analog filters are required? This analog filter is required because the spectrum that you have got probably is not the one which you will be interested to transmit right. For example, in a mobile you may not transmit in the complete 20 kilohertz signal why?

Because you may not have that much bandwidth and also when you usually speak you may not require a very detailed you know very detailed component of the frequency. So, it is your choice to curtain or to I would say to put a cap on the spectrum that you would like to have it. Say lets for example, I put a cap on this spectrum say this much let us say I want to have this much somewhere here ok.

Let us say this can be somewhere around 10 kilohertz, this basic spectrum. So, I may have some sort of a, analog filter which we will just make the first attempt of this 10 kilohertz conversion then what I will do? Then this is still an analog domain then I will make an ADC here. I am going step by step, block by block so, that it will give you a right motivation for the channel part because that is where we lead to, but before that we need to talk about the complete digital system.

Now, this is where the ADC comes into picture. Now this ADC may not sample exactly at the nyquist rate ok. So, for example, the spectrum here we are dealing with is mode of a 10 kilo hertz plus minus spectrum. So, naturally a 20 kilo hertz is minimum required for the ADC, but usually what happens? You take a very large sampling rate that can be it is up to you, but it is not exactly at nyquist rate maybe 2 times, 4 times even 10 times nyquist rate depends on your ADC availability.

Now, in this case, let us say let us for this particular case I am sampling at say 100 kilohertz. Now a natural question come, why should sample it at 100 kilohertz while the signal spectrum is just 10 kilohertz?

So, there are certain reasons why I should sample more than much more than the nyquist rate because the first reason is that this analog spectrum is not perfect right why? Because it is an analog spectrum what should be the order say for example, I give you a very basic thread on this.

Suppose this is my original spectrum. So, let us say I am having this spectrum this is the spectrum I am having it and I would like to have a filter, which is say this is 20 kilohertz, I would like to have it somewhere around say 15 or 10 kilohertz as the example says ok 10 kilohertz I want.

Now you know from 20 kilohertz this particular frequency is not of interest to me right this is what I want to discard it, but the problem is that, if I want to discard it I need a very high ordered analog signal say for example, probably third or fourth order analog signal or a fifth order sixth order analog, because I need a very sharp cut off here, but it is very sharp cut off here ok.

Now it is not possible to have such kind of high order analog signal the main reason is basically the component cost because when you have a very high order say fourth order analog signal how many components are involved in that particular case? Huge; and that increases the cost of it and also analog components are very noisy in nature right.

So, if you have a noisy component too many noisy component that will anyway degrade the signal quality. So, it is not a very wise idea to have a very high order analog filter. So, probably most of the cases it may be just a simple first order analog filter. So, when you have a first order analog filter what will be the cut off in this case? Now, the cut off in this case will be still 10 kilohertz, but there is a slope because it is the first order right the slope will be very wide slope. So, what will happen?

In this case you may not get exactly a very sharp cut off, but rather the filter transition will be very wide. So, it may be like that, but something like that it will come into picture. So, your filter characteristic will be something like. So, because of this wide transition of this filter, what will happen? These are the signals which were not suppose to enter in your spectrum a part of it will enter as a you know aliasing because you are you are inviting that signal you are not really able to cut off using the analog filter.

So, because of that what I do is, I sample the whole signal at a very high ADC rate because now I do not have 10 kilohertz as the cutoff or rather I do not have a 10 kilohertz bandwidth signal because I still have significant component beyond 10 kilohertz. So, if I sample at an nyquist rate with respect to 10 kilohertz all this signal, but whatever I have put as a circle, they will all enter they will all enter as a aliasing.

So, to avoid that what I should do? I should sample at a very high frequency ok. Now it may be 4 times nyquist rate maybe 6 times nyquist rate maybe you know 10 times nyquist rate depends how much you can afford it. So, that is why I have kept a very large frequency 100 kilohertz sampling rate; then what is the consequence of it? The consequence of this kind of signal would be because I am explaining all the components of the modem because that will finally, lead to our channel part ok.

So, to better understand where the channel lead do, I should better start from the modem and that is precise the reason why I start all these components together. Now what I will do? I will sample at a very high rate. So, what will be the spectrum in this case after very high speed sampling? The spectrum would be somewhere like this. So, this will be my pi, this will be my minus pi right because it is now it is the digital spectrum, I cannot see that spectrum in an analog domain because it is now kind of a normalized pi to pi spectrum.

Now, what will be this pi correspond to? How much frequency it will be corresponding to? It is fs by 2 that is my sampling rate by 2 which will be 50 kilohertz. This pi will correspond to particular and this minus pi will corresponds to minus 50 kilohertz ok. Now I have a better or rather I would say it is a larger window to view all this aliasing, but now it is now this signal whichever was not intended for me which was supposed to enter into my system because of the bad characteristic of my filter, that will also be seen here it is still not an aliased filter.

So, what I will see? I will see the spectrum will be something like that ok. Now this part is an unintended spectrum ok probably this is my 10 kilohertz part and this probably up to my 15-20 kilohertz whatever because that precise this part whatever is coming here will enter here clear, but it is not still an aliased signal because my spectrum is viewed in a 50 kilohertz complete window pi to minus pi. Now, I can do a better job in the filtering how? Because this part I want to discard.

This is not what I want right because this is supposed to be discarded. So, what should I do? I basically want to discard this part I have to still do a filter what kind of filter I will now use? Because this is a digital spectrum right; now if it is a digital spectrum, I can use a digital filter, but now I have a liberty what is the liberty? Liberty of using higher order filter. So, this is a digital filter I now use it, but I can now have a very high cutoff.

So, let us say my filter characteristic I can choose a very high cutoff this sharp cutoff. That is sharp cutoff filter I can use know what is the order of this filter? Let it be fourth order, let it be sixth order filter. But now, that filter is a digital filter. So, if it is a digital filter it is not a big deal to use a high order digital filter why? Because the components are just digital components. See if it is a digital component the cost will not increase much compared to what the analog signal would be. So, this is the first idea of my signals characteristic.

So, we will start with the very small bandwidth signal, then it will go through first order analog signal, go through ADC, then it will go through a digital filter where you have a sharp cutoff ok I can create a you know sharp transition of the filter. So, what will happen? I need not incur the cost had I used a high order analog filter. Now I got a digital signal which is sampled at a much higher rate so; that means, what is my final spectrum the final spectrum would be this black colored one.

Discarding the red part because I have used a sharp cutoff filter here then what will I do? Then this signal is a higher order sampled signal. So, what I should do here? I should do I let me go to the next page here what I should do here?

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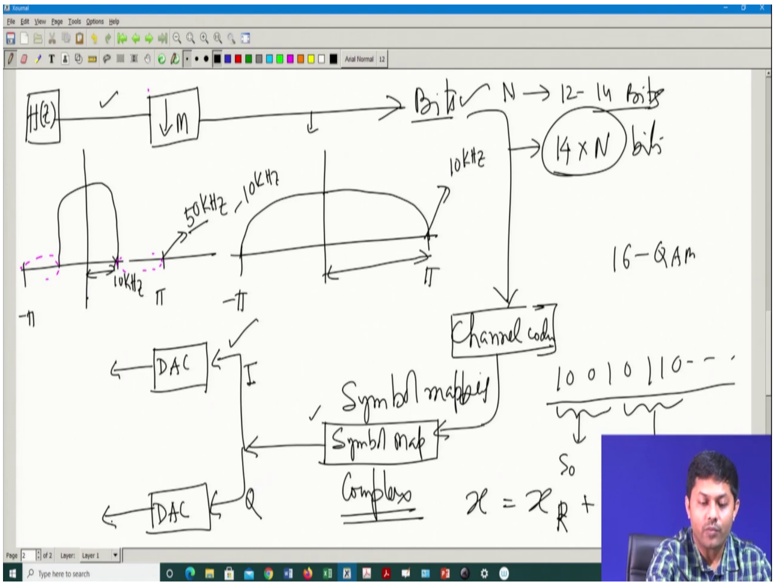

Then this signal which has come from the digital filter, which is a higher order digital filter, but with a very you know sharp transition, but it has a higher rate of signal. So, what is the consequence of it? So, it means that my spectrum will now be like this; this is pi to minus pi, but now the spectrum will not have any extra component because I have used a very sharp digital filter.

So, it will be tentatively 10 kilohertz, it will refer to 10 kilohertz this part ok, but this component will still be 50 kilohertz ok. So, which means that this part you have so much spectrum gap you see this; this gap this is all baseband processing, I am still not gone into RF or analog this is I am in a baseband; what is meant by baseband?

Baseband meaning you are still inside the system wherever bandwidth everything is within your 0 Kilo 0 to something it is not modulated signal. So, you are in a baseband ok; so, this is what happens. Now you have this spectrum gap, so what we will do?

You reduce the spectrum; your spectrum will remains the same, but you reduce the number of data in time domain because you have too many you know too many extra sample that comes into picture because this spectrum its not used for you right. So, why should we use a 50 kilohertz signal while the actual spectrum is just 10 kilohertz.

So, what you do here? You just put a, some sort of a down sampler here say my down sampler order is m what is the down sample order? See it was 50 kilohertz make a one by fifth order down sample. So, what will happen? Now after the down sample what should be my spectrum at this point?

A spectrum will look like the spectrum does not change just the viewing of the spectrum in the window that we are talking. So, this 10 kilohertz will now spread to this point. So, this point is as if like a 10 kilohertz spectrum this is a minus 10 kilohertz spectrum. So, it is as if like I have done a nyquist rate sampling.

But it has not increased or decrease the spectrum, because this is still a 10 kilohertz spectrum baseband spectrum this is 10 kilohertz baseband spectrum, this is also though it is a 50 kilohertz, but effectively it is 10 kilohertz spectrum. So, I have a down sample to reduce the time. Now these are all you are you know digital samples now what do you do? So, you get lots of bits here.

The stream of bits right; each and every symbol will be creating one (Refer Time: 18:19) one bit. So, now, say my ADC is a 12 bit ADC or a 14 bit ADC what does it mean? That means, if you have; say for example, your spectrum has N number of samples what. See you have taken N number of time domain samples after your down sample and each and every sample is say 12 bit or say 14 bits ADC resolution.

So, which means that you may get say 14 multiplied by N number of total bits you get right or 12 whatever your resolution would be. So, that kind of number to be; now at this stage at this bit level it is a stream of bits, then what you do? Next technique will be you put some sort of a channel coding here, after you do a channel coding you do some sort of a symbol mapping here.

Symbol mapping here that this should be your symbol mapper; what is this symbol mapper do? So, you have a stream of bits say 1 0 0 1 0 1 1 0 0 stream of bits right from here you get it, from here you get it, from here you get it. So, what do you do? You map the now these are all bits stream coming here now you map all these bits to the right symbol because that is the symbol you want to transmit; you cannot transmit the bits these are these are logical content.

Logical content you cannot transmit what you can transmit is a actually electrical signals. So, you have to get back the electrical signal model again from this bit stream, how do you get it? You do map it to the symbol. So, what is symbol mapping? Symbol mapping could be you know if you have taken digital communication courses symbol mapping depends on what type of symbol you want; say for example, I may take a BPSK symbol, I may take a UPSK symbol, I may take whatever.

Say 16 QAM or 64 QAM whatever say let us say I take 16 QAM symbol what does it mean? So, each and every four symbol each and every 4 bits I group it to some symbol say S 0 symbol from here, I take it S 1 symbol and so and so forth. So, these are all your symbols that will be coming. Now, from the symbol your job is slightly done because now you are you are getting got back your symbols which are like a single processing component.

Now, what you have to do? You have to now transmit the data. So, now, these are all complex symbol right; have you noticed one part? I started if I if we look back, if you look at the signal what did I start with? I started a voice signal here. Is it a complex signal? No, it is not a complex signal; it is absolutely a real world signal, it is a real signal like for example a video, a voice whatever the real world or some sensor data if it is a sensor data.

They all give you a single electrical value right here also this the microphone will give you a simple electrical value. Now that electrical value is a real signal it is not complex signal, but once it goes through the complete chain of the modem you see how beautifully the signal gets converted to a complex signal.

This must be noted because that will drive to the concept of complex channel we need to talk about the channel ok. So, this is very important to understand because now what you deal with is a complex signal you started with the real signal but once it goes through all the chains ok; I have missed some of the important blocks inside it say like for example, after channel coding you can do rate matching I am not getting into that details, but that is not the focus of this talk; the focus of this talk is more on the channel side.

So, now, I am slowly moving towards that. So, now, you have a symbol mapping which is a complex signal now finally, what I have to do? Because it is the complex signal I have to I will transmit right I cannot transmit a complex signal just like that. So, what I do? I split, I spilt it here ok.

So, one part I call it I part, another part I call it a Q part; say for example, I have a complex number x is equal to say x real plus jx I. So, this is like some complex number ok. So, you can you can think of this I is x R and the Q part is x I that is all. So, what I have do?

This is a digital signal, now this digital signal I will transmit; now what should I transmit? I cannot transmit a digital data right because the real world is it is an analog data real world; real world does not understand what is called a digital system. So, your DSP signal processing everything is inside the machine it is not inside the word ok.

So, I have to convert back to a, analog signal. So, I have to put a DAC here; got it? This is my DAC; now there is a problem here. Problem is that what should be the sampling rate here? Because this drives the bandwidth that you will be seeing in the channel; I am just trying to motivate in the channel what is meant by a channel and what are the components that enter into the channel part. So, those are the things that we need to discuss and understand that.

Now, at this stage when I am in DAC level because DAC needs the data rate because DAC is what? Is basically a digital data comes into its picture and then at a certain rate and then it converts that data to analog waveforms, how it does? Now these are all discrete datas that comes into picture ok; now what should be my DAC output here, DAC output here? That we need to understand it ok.

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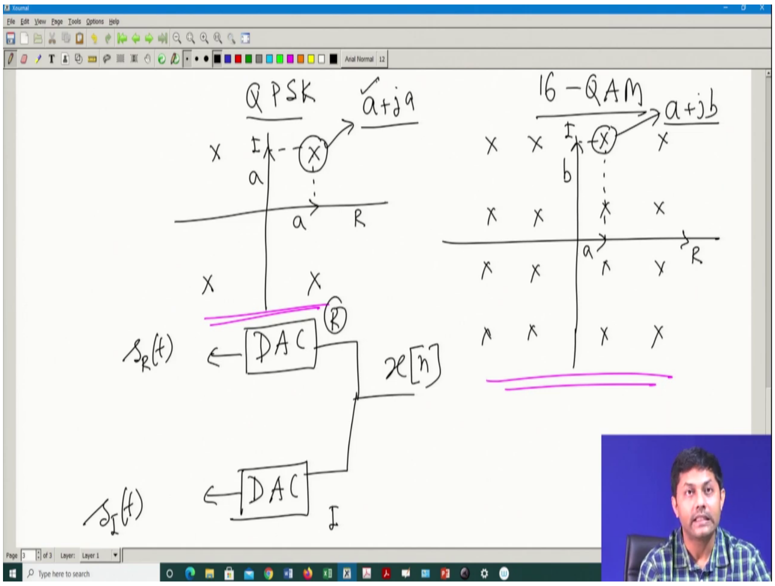

So, for that you have to get to the constellations part ok or some complex number. Say for example, my symbol mapper is a simple case. So, let us say I am taking a QPSK; one case QPSK and I have say another point 16 QAM ok.

So, what are the constellation points? Say these are my constellation points, these are my constellation points, these are my constellation points four points will be there for a 16 QAM;

what should be my constellation point? There will be 16. So, each and every points will have 4, 1 2 there will be 4 more here 16 points here.

So, now these are the points these are like these points let me put a different color here these points and these points are the ones which will be going to the DAC. Now these are all complex number ok; see if it is a complex number, what will happen? So; that means, it will have. So, let us say QPSK scenario. So, it will have a real component here, it will have an imaginary component here right. So, this is my real this is my imaginary component here also this is my real, this is an imaginary component.

Now, let us say I am transmitting this symbol ok let us say this length is a let us say this length is a right. So, what is this constellation? This constellation would be a plus ja ok. So, a plus ja. So, this is just a simple constellation. Let us say in this particular case I am transmitting this constellation 16 QAM constellation ok. So, what does it mean? It means let us say this value is a, and this value is b right.

So that means, this one is your a plus jb; for this constellation a plus b. Now you would like to see in a DAC what exactly my waveforms will be because DAC is creating an analog waveform and how exactly it gets analog waveforms you have to understand ok. So, now, let us say you have the two DACs, one DAC here, which will be mainly my real part another DAC, let us I take the imaginary part.

Let us take the first one QPSK one and I am sending this signal sometimes I am sending this signal I am I can send this signal any of the four and from 16 QAM I can send any of the 16 right. So, what would be my DAC output then? This is what I am observing it right.

Let us call this because it is a, analog signal right you cannot have a any n concept. So, if this is my say x n meaning it is the concept in a DSP signal processing I mean discrete signal, but when I am getting the output of the DAC it should be some sort of a continuous signal right.

So, let us understand. So, let us understand how exactly both the signal will look like right. Now if I send the first one, so how does this s R, s I t will look like? If I draw.

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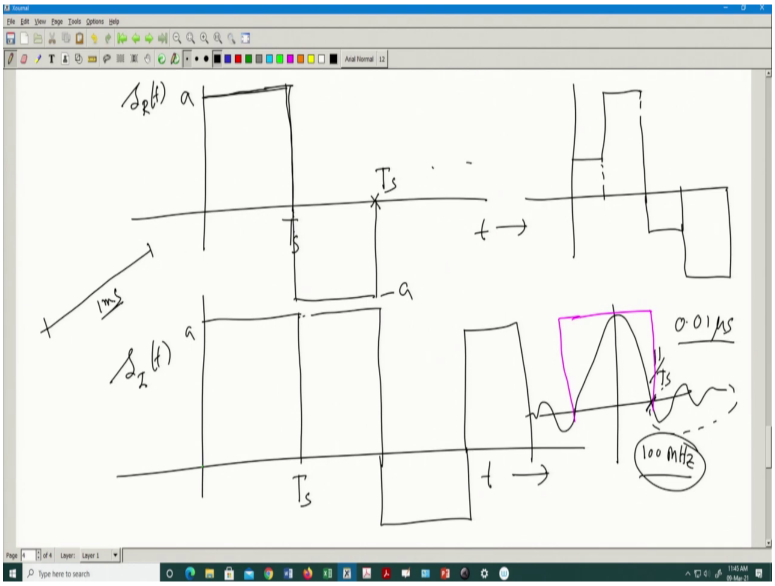

So this is my real part. So, this I am plotting the s R part at different time seeing it and this I am plotting my say s I t ok. So, how does it look like? So, s R t if it is QPSK it will be a, if s I t this will also be a right. Now you may ask the question why like a, why is it like a square pulse? Why not something like this?

Why not like that? Why not like this? Why not like that? Why just a square pulse? We will answer that ok. We will answered it subsequently, but let us assume it is a square pulse. Now this is a, another question that comes what about this time, how long it should hold that pulse? So, it starts. So, I am sending the first point of my QPSK; the first is up to T s it will be holding 1, here also it will be a; let us see in the second case, I am transmitting say for example, this pulse, where for the real case it is a negative, but imaginary it is positive.

So, let us do that real case it is negative. So, which means, real case this signal for the next time stamp this will be somewhere here. So, it will be minus a, but this will be the plus a like that. So, this will be again something like that depends on what exactly you are sending and so on and so forth.

This is how the exact analog waveform will look like. Had it been QPSK? Had it been 16 QAM? How does same signal would look like? The same signal will look like this something like that, then it may go up depends on what I transmit, then it may go down, it may go on further down and so on and so forth.

This is how the envelope will look like right, for the real case and imaginary also a very similar case that will be followed here, ok. Now the next question come how to determine T s? Because if this is a pulse look at this pulse so; that means, at every T s I am sending a pulse. So, what is the waveform of this pulse? What is the bandwidth or how does the pulse look like in a spectrum? The pulse will look like this; it will be something like a sync pulse ok.

How much this point? This is 1 by T s. If I ignore these things the extra part I can say effectively my signal has a bandwidth something like that this is my bandwidth effective bandwidth ok. Now they come; obviously, the actual bandwidth of such pulses are infinite, but effectively I can say it is 1 by T s ok. The biggest question that appears here, has it anything to do with my analog signal that I have send?

What I started? I started with a voice signal what was the spectrum that I took it? It was 20 kilohertz, but I cut into 10 kilohertz ok so; that means, my analog waveform has a 10 kilohertz bandwidth. Does it mean that when I am transmitting it finally, from the DAC and then you know antenna so and so forth, does it mean that this T s has anything to do with that

10 kilohertz? And the answer is no ok; obviously, it should be more than that at least 10 kilohertz, but it has no bearance that it has to be exactly 10 kilohertz.

There is no there is no such requirement. I can have a 100 megahertz also; that means, this T s can be something like a 0.01 microsecond which corresponds to 100 megahertz ok; it all depends on where you want to transmit and how you want to transmit. So, it has no correlation with exactly what will be your received signal because why 100 megahertz? It may happen that you may not get you may not get a bandwidth of just 10 kilohertz continuously.

You are allocated probably 100 megahertz just for a fraction of a second, it may happen right. So, if you are given a 10 kilohertz spectrum availability for the complete time of you know transmission then of course, this T s has something to do with your 10 kilohertz, but that is not the case right. You may be occupying say 20 megahertz spectrum in a particular region and for a fraction of a second say for example, for 1 millisecond you are allocated 20 megahertz that particular user is just given 20 megahertz bandwidth.

Because next times it will be some other users from that precisely what happens in 2G, 3G, 4G. You as an user will never be allocated continuously the same spectrum because there are so, many other users are appearing there right. So, everybody has to be given a chance. So, that is what the troubles start.

So, which means that whenever you are given a chance for transmission that need not will be just 10 kilohertz what your need is ok. If it is 10 kilohertz given for complete say 1 minute, 2 minute or if you speak for 5 minutes then its fine, then your sampling rate can be just as same as 1 by 10 kilohertz that is not the case.

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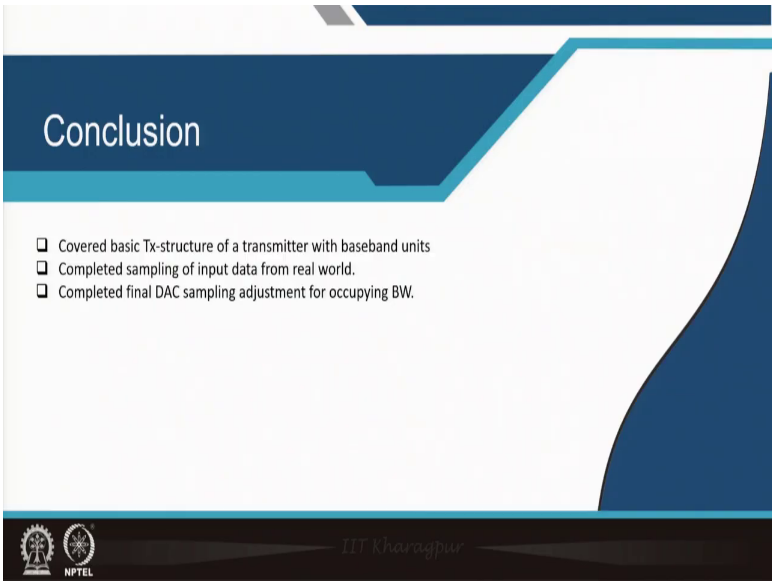

So, we have covered so, far the basic structure the of the Tx and then how to do the input bit input data sampling from the real world like how to get it through the ADC and then adjust the sampling rate and then finally, when it goes to the Tx just before the RF part how it changes the sampling rate again back to the DAC for readjustment of the bandwidth occupation.

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These are the different references. So, I will be mainly following the book number 2 and book number 1 also along with that. Book number 3 will be covered sometimes when the time series and random process come into picture ok. So, now, we understand this part; with this session we will now go to the next session, I will see how exactly our analog goes into the RF and then starts our channel part ok.

Thank you.