Basics of software-defined radios & practical applications
Dr. Meenakshi Rawat
Department of Electronics & Communication Engineering
Indian Institute of Technology, Roorkee

Lecture – 06
Software-defined radio architecture Part IV

So, in the series of basics of software defined radios and practical applications, we are covering the 4th part of the software defined radio architectures. Previously we had seen heterodyne, homodyne architectures, their combination digital life hydrogen structures, which contains the benefits of homodyne structures also. Now we want to have a look at the concept of SDR, when we want to do the sampling directly at the RF frequency. So, what are the requirements? What are the limitations? This is what we will be covering in this lecture.

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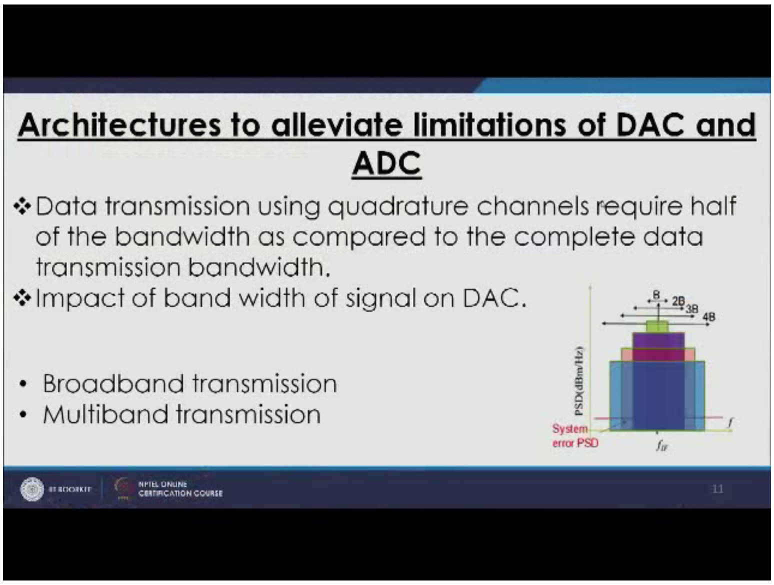

So, architectures to a limitations of DAC and ADC, by using the quadrature channels, we have already reduced the bandwidth requirement by half; means I and q channel both of them are carrying half of the bandwidth, which was required for the full complex signal.

So, we are already reducing some of this limitation, now what is the impact of bandwidth of the signal of the DAC? Till now most of the discussion we have done they were in the terms of signal band a sinusoidal signal is coming, what will be the of conversion? Down conversion? How we will choose the frequency this is what we have covered.

Now, let us have a look at the bandwidth impact on DAC and ADC performance, first of all let us remember our Nyquist-Shannon theorem. So, Nyquist theorem we had seen maximum frequency component; if we have, then our sampling frequency should be more than the twice of our incoming maximum signal frequency.

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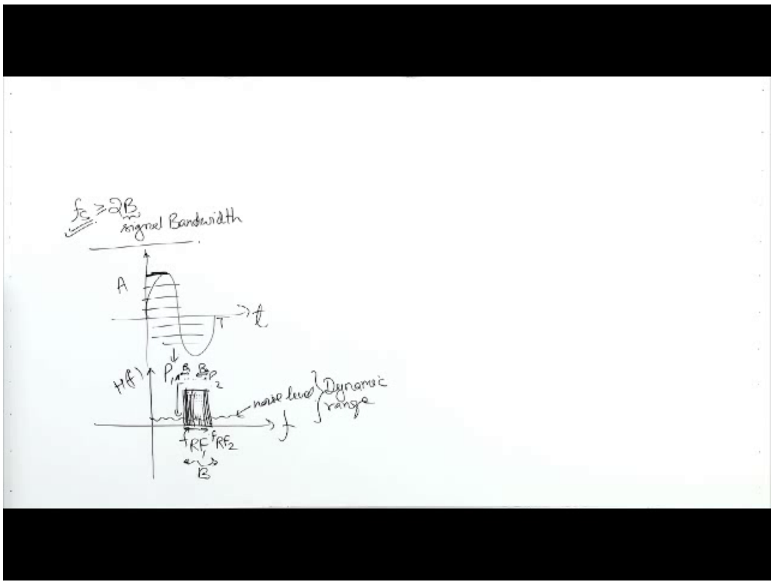

Now Nyquist-Shannon theorem says; for the band limited signal, the sampling frequency should be more than equal to 2 times of signal bandwidth. Now we when we say signal bandwidth it is the band limited signal. So, we are talking about minus B by 2 to B by 2 if it is at the base band. So, this we have to remember.

Now, when we are talking about the band limited signal, another thing whenever we are sending any signal an for throw DAC sampling it and or we they are we are receiving in a receiver and we are getting back our digital data from the analog domain using ADC, then whatever we have discussed before it was in terms of single band signal. So, we were talking about a single bad signal and then we have said this is a signal and let us with the quantization etc.

So, it was the attitude, now this inverse it suppose it has a bandwidth how will it impact in the digital in the frequency domain. So, for this signal which is the sinusoidal signal, in the frequency domain it appears as let us say it is fRF as a single tone all right. So, your DAC which is able to take this highest amplitude this highest amplitude is defining the power which it can take.

So, for a single band that power is concentrated in this frequency, now suppose you have total signal then or we can say that this becomes the bandwidth of the signal, then this both of these signals they contribute to that power and the ADC is reaching their highest level of aptitude, because of these totems. So, suppose this was the noise level and this was the P1 which was because of only single tone this was the dynamic change, now when you have 2 signals the power is distributed between these 2, right? So, it will be reduced for sure because it is distributed between these 2 tones and this is your new P2. So, P1 has higher SNR for the single carrier even case number of carrier the power is distributed. So, the highest point of power with respect to noise is coming closer to the noise it is becoming lesser and lesser.

Now, this distance from the highest power to the noise power it is called dynamic range of any ADC or DAC. So, as we can see here, if we have a single carrier this is being shown here by this arrow, as you keep increasing our bandwidth the amplitude the average power will go down, because the whole power is contained within this signal now. So, the dynamic range goes down so what is the problem with that the sensitivity of our signal will go down, what is the sensitivity we have defined it before that the ADC should be able to distinguish the actual signal with respect to noise or the interfering signals. So, if this amplitude keep coming closer to our noise level, then it is difficult to distinguish which is our main signal and maybe our interference signal is higher, than this suppose it has a lower bandwidth it will have more chance of saturating over ADC, then it will be creating problem in detecting the actual signals.

So now, bandwidth can be of 2 types like, I said if you have 2 tone signals you have 2 tones and you are still covering the bandwidth B, this is multi band kind of operation there are 2 bands there are 2 frequencies on which it is working this was enough example of only 2 tone signals, but it is possible that it is a multi-carrier signal here, let us say photons here and also now they have their own bandwidth B1, B2 where it is containing it is carriers it is own stones and it together by using these 2 they are again making this bandwidth B.

So, this B contains B1, B 2 and the distance between this stone, but now what is the sensitivity requirement for this one, when it was a single tone we have very high dynamic range, when we have this 2 band system the dynamic range will go down and if I want to fill this whole is frequency range with the carrier you want to transmit at all these frequencies, then whole band it is being used and then in this case our dynamic range will further go down, because it is saturating our ADC and DAC with it is whole power.

So, that with ever is going down, now when we are covering all the frequency phase, then we call it broadband transmission and then they are using selectively some of the frequencies that it is called multi band transmission. So, let us keep this in mind we will come back to it later the STR, now let us go to the concept of sampling, how it effects ours architecture is there. So, you can see here the relation between.

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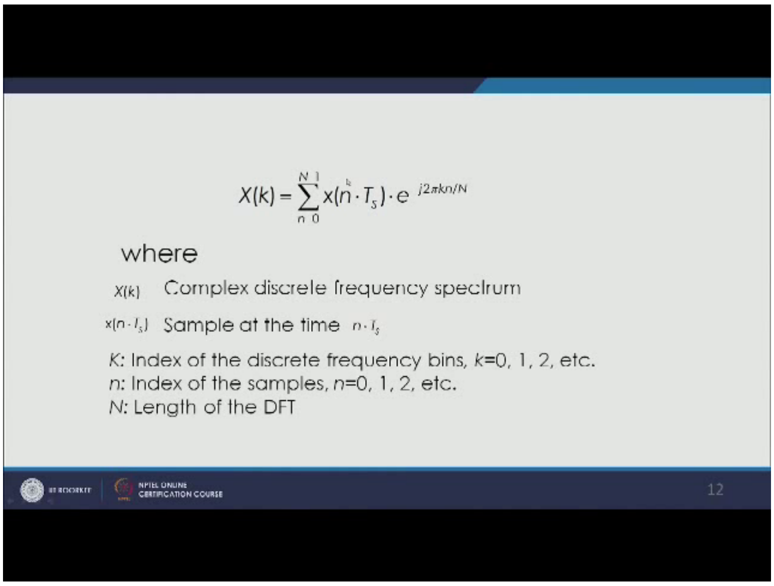

The time domain signal and how it is converted back into the frequency spectrum X k is representing your complex discrete frequency spectrum, where k is the index of discrete frequency bins in right hand side again the time is sample time and time Ts, and Ts in your sampling frequency. So, whenever we are having the multiple of that sampling frequency, we are sampling at a particular points where one time a duration is being finished.

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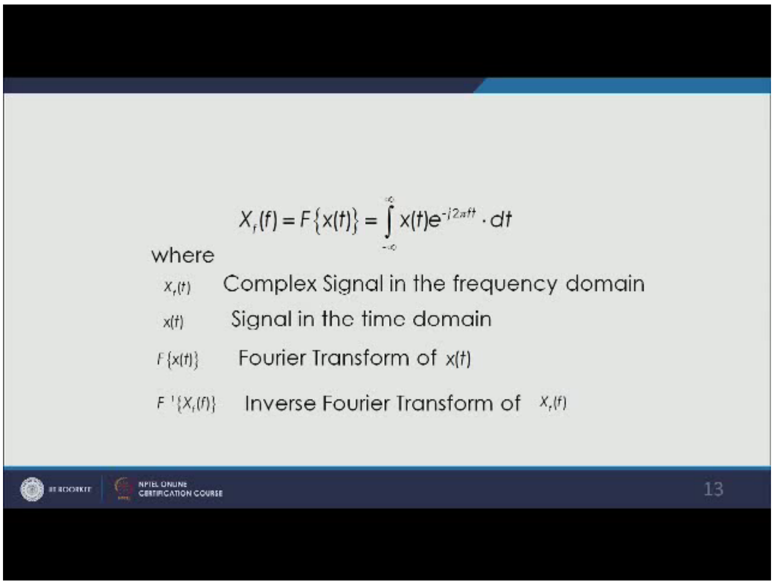

If we see in terms of continuous frequency, in time domain and we want to see what will be the complex signal frequency domain then it is given by the integration as opposed to the summation normally we use there.

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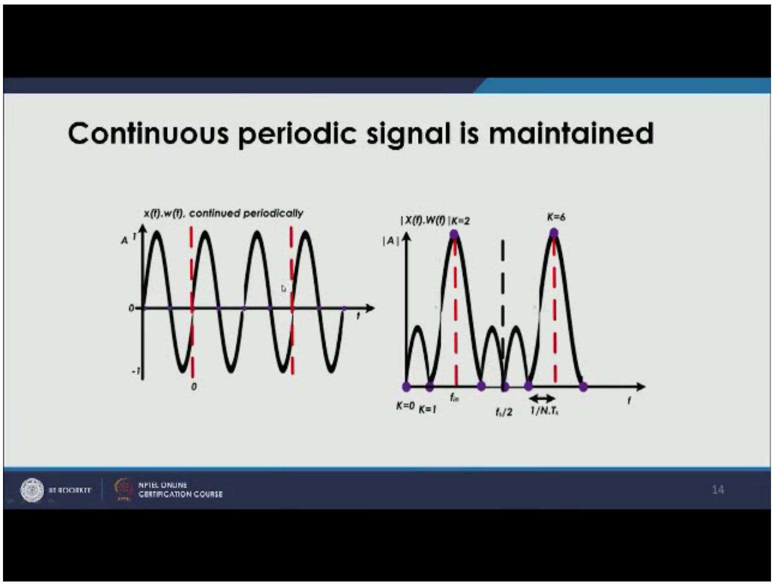

So, whenever signal is being sent and it is a limited signal, let us say this red lines they are representing one signal duration it is a limited signal, then corresponding to that we will have a sinc function because it is sinusoidal signal. So, because of the windowing function, we can see at f in which is at the instance of 1 upon NTs our sampling points.

So, at exact point our data we are able to sample, if it is a multiple of Ts in the time domain and 1 upon NTs in the frequency domain.

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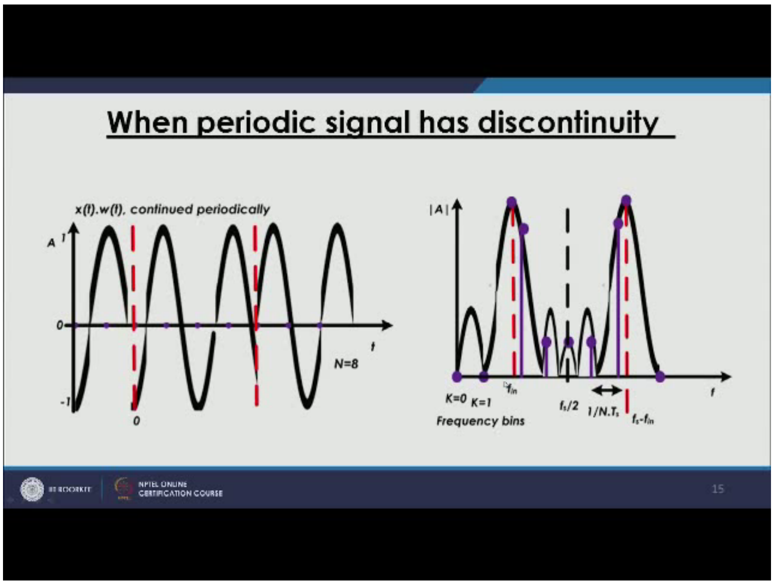

So, these are the particular points we are getting here, if our signals are periodic in nature, but they have disrupted nature, there is a sudden jump as we can see in this case then we can see the shift because we use the interpolation filter in the frequency domain, there is a shift here in the data, right? And we will be losing some of the peak points. So, peak points we have lost here, we are not getting the exact frequency location we are losing some information there.

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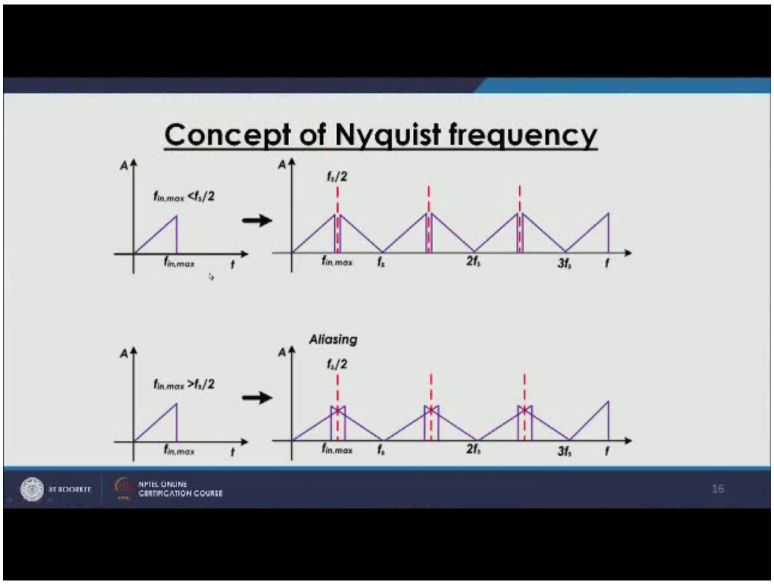

Now, keeping those concept in mind, let us go to the nyquist frequency criteria, we know that of our sampling frequency should be more than the twice of f in max. So, in this figure you can see that this is our signal in frequency domain and the max will be decided by this boundary here, right? So, when it is sampled at a particular for sampling frequency which is more than twice of f in max, then how do we get our signal, our signal is here which falls in the boundary of 02fs by 2.

Now this 02fs by 2 is called first nyquist zone from fs by 22fs, we have second Nyquist zone, where we see the image of this signal. So, conjugate of that everything will be negative here. So, you can see this is here from fs2, 3fs by 2 is actually the 3rd Nyquist zone and from 3fs by 2 to 2fs, we have 4th nyquist zone. So, in each zone you can see the your signal you can receive and you can receive the image of that signal, also now suppose our signal f in max is more than fs by 2 then what happens as you can imagine that the signal will start from 0 to fn max.

So, the signal is here, but because we are not taking care that our fs more than twice of the maximum frequency these of some earlier sing off the image and the original signal. So, keep it in this concept it mind they are. So, benefits of over sampling and under sampling architectures which I want to discuss.

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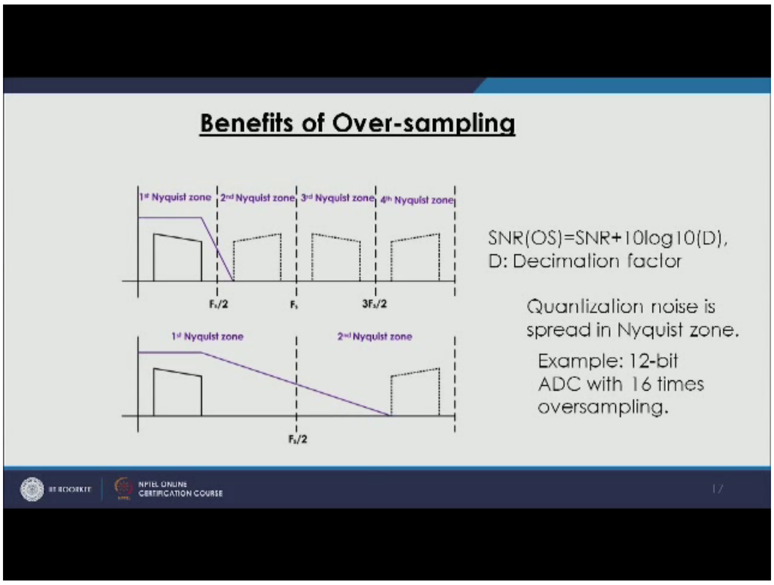

What is the benefit of over sampling for example, this is the first nyquist zone as I shown earlier, it is the second one then third and the 4th one and we want to retrieve our signal, some here this is our signal and we have filtered it.

Now, this filter location is very new to the image frequency, if it becomes fs become even little smaller then this image will have some part inside that filter, because filters are not ideal filters right they do not have sharp cutoff they can gradually go down. So, to keep our signal intact to be able to filter it properly, over sampling is taken into account in the over sampling what happens because we increase the number of samples we increase the sampling rate, then by the Nyquist criteria our signal is here, but fs by 2 is lying.

Now further from the signal and the image single is much further. So, we can have some loose constant on the filter bandwidth, we can use a filter bandwidth which a with a large role of factor, it can goes slowly away and in the paternal conditions most of the filters are like that, they do not have sharp cutoff they will be gradually decreasing in magnitude. So, more than that because we have increased our Nyquist zone.

So, the thermal noise it is distributed in the Nyquist zone. So, initially it was distributed in this area the same power now it distributed into this area. So, noise goes down. In fact, the effective SNR in the case of over sampling is given by original SNR plus 10 log 10 of D.

Where, D is the decimation factor or the oversupply factor. So, you will here the decimation and over sampling these 2 terms are used in the place of each other and both mostly mean the same thing. So, let us take one example, suppose you have a 12-bit ADC and let us do the 16 time over sampling with this. So, what will be the actual SNR of this system. So, 12-bit ADC what will be the dynamic range of that one we have SNR for the single, carrier signal original SNR what was the formula 6.02 into number of bits plus 1.76.

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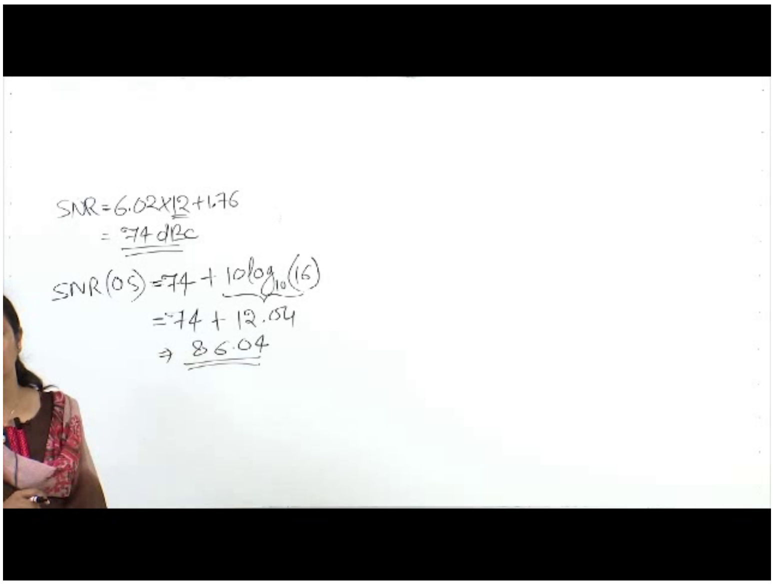
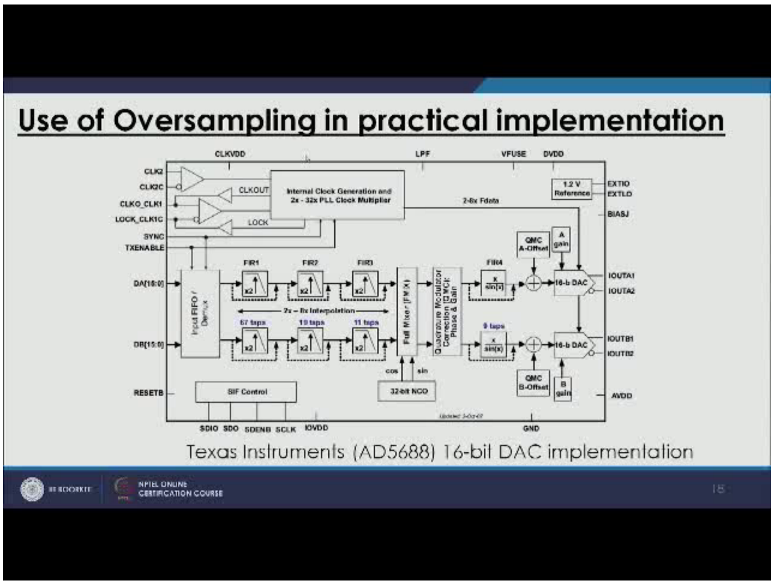


So, it will be around 74dBC. So, signal to noise ratio 74dBc for the 12 bit, now we have done the over sampling by 16. So, what will happen the new formula of the what is happening SNR will come into effect, we says the original SNR right plus 10 log 10 of the decimation factor which is 16.

So, the summation becomes 74 plus 12.04. So, SNR has increased almost by 10 dB or. So, for the single carrier of course, for the multi carrier or the broadband signal it will be lesser, but we can see the impact on the calculation here. So, over samples has the certain benefits and which should be taken advantage of whenever we are designing any system. (Refer Slide Time: 15:34)



Here we are seeing the use of over sampling in practical implementation in the 16 bit a DAC AD5688 from the texas instrument. So, as you can see first of all our data which is coming for the INQ channel and it is 16-bit data being applied here, we are having from 2 to 8x interpolation options are there. So, we can choose this interpolation of option maximum we can have 8, and minimum we can have 2 or we can simply avoid this portion, but increasing interpolation gives you good signal to noise ratio. So, it is recommended that we use it, then 32-bit NCO is being used here.

So, as I told before now in this scenario by using choosing a appropriate sampling frequency coding it there, we are able to up convert in the digital domain this over sample data which is keeping it is image away quite nicely, now phase and gain correction are happening here, because we have 2 different branches.

So, we put this linear correction by our hand and we will be dealing with this later in detail, now because we have our system in the first form and we have the sync kind of architecture at the output whenever you have a rectangular window there, right? So, to remove the fact of that rectangular window we multiply with the inverse of that. So, because rectangular window will actually give you sink kind of function sin x upon x. So, we want to multiply with the x upon sin x to remove the effect of that one, and whatever we are getting at the output, this is what we applied to our digital to analog converter to be converted into analog domain.

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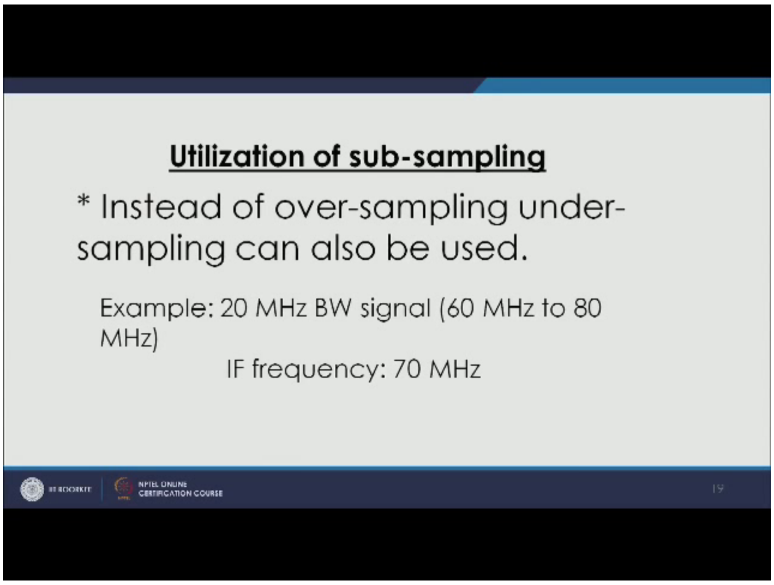

So, this is a practical example and we can see that we are using the NCO as well as we are using the over sampling in the actual setup. Now this was the example for the over sampling, how we can use the sub sampling or under sampling. So, let us take one example of signal which is 20 megahertz bandwidth signal, which is available from 16 megahertz or 18 megahertz and our if' frequency is 70 megahertz so our RF signal.

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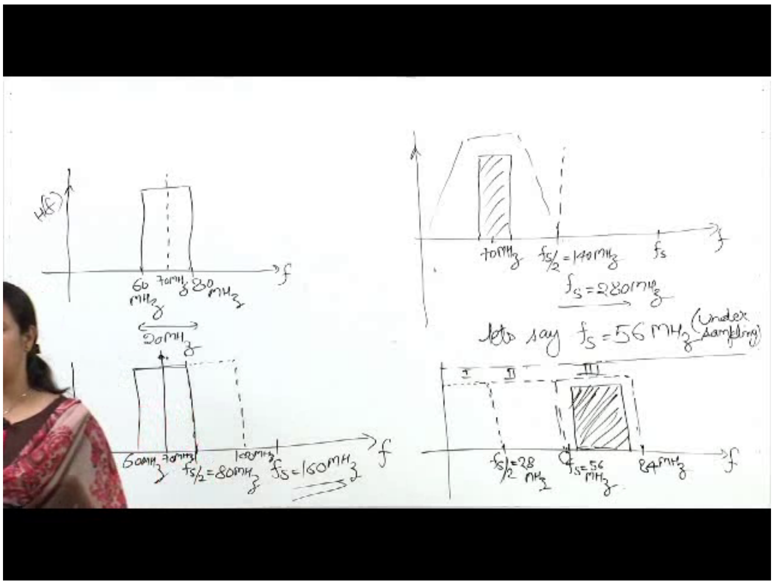

So, what we are saying that in the frequency domain from 60 to 80 megahertz, our RF signal lies. So, bandwidth becomes 20 megahertz we are told that our RF frequency is 70 megahertz. So, 70 megahertz is the IF frequency. So, how to choose our sampling frequency so that we can have the signal properly here. So, the highest frequencies highest f is 80 megahertz in this signal, if suppose I take it to the basement this will be the highest frequency there.

So, we have to have at least twice of that, let us say. So, our sampling frequency should be getting equal to twice of f max. So, it should be more than 160 megahertz all right. So, suppose I take the one say 160 megahertz sampling frequency. So, f s is equal to 160 megahertz. So, f s by 2 will be 80 megahertz.

So, in this case if I do the down conversion my signal will be on the boundary, right? It will be 80 megahertz it will be 60 megahertz. So now, it is at boundary and it is image will be starting right over here, it will be 100 megahertz. So, we want to avoid that we want to have it in between. So, let us take our Frequency sampling frequency s to 80 megahertz. So, fs is equal to 280 megahertz, it means f s by 2, 140 megahertz and this is our first Nyquist zone right and our signal will be in between this Nyquist zone, right?

So, we are able to get the signal properly by using this particular band pass filter and if our if is at 70 megahertz, we will be directly getting it back at the baseband signal. Now this is good we have used the over sampling and what we have is achieved, we have whichever had very clean signal in our first Nyquist sampling rate.

Now let us say I use f s equal to 56 megahertz, which is under sampling right because we had said that at least 160 megahertz was required to get it at the boundary. Now we have gone to even lower. So, it is under sampling it is also called sub sampling. So, again it is 56 f s equal to 56 megahertz. So, f s by 2 will be 28 megahertz and this will be first Nyquist zone, it will be second one with 56 and the third one, when it at 28 to this one.

So, 84 megahertz it will be the third zone, now our signal which is starting from 60, 280 will be here. So, basically if it is folded back on into the first Nyquist reason, if it is at the 0, if it move down convert it and bring it to 0 it is folded back, then basically just by sampling in the third Nyquist zone we are still able to achieve our signal, if you do not actually go have to go further over sampling even with the what is happening we are able to get our signal only thing is that we have to get our signal in the third Nyquist zone.

So, this is the example of our under-sampling example, because for a over sampling we have to have high frequency, we have to put considered on the ADC and DAC instead, we can use the under-sampling sub sampling and by choosing the different Nyquist zone, we can a still recover were signal. Now start finishing with that we have seen the usefulness of the sub sampling and that is under sampling and the concept of RF DAC, when we are directly sampling our frequency, right which is the RF frequency.

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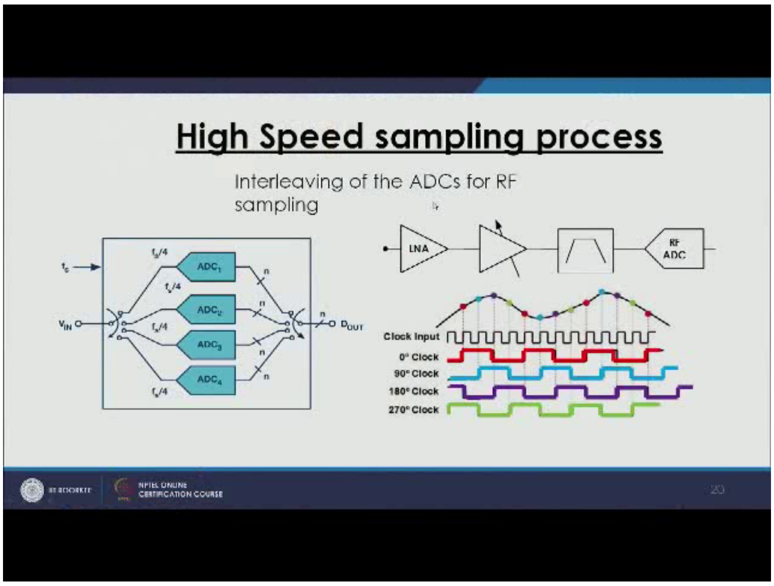

So, if we could do that then it is actually the ideas software defined radio architecture, now this is one example which is called RF sampling process in this case in the receiver side we are actually having directly the RF signal, we are filtering and we are directly sampling that, we have discussed before that it is not possible, because our ADC and DAC has to qualify the Nyquist criteria it has to have more than twice of the RF frequency.

So, it was frequency of the RF signal is one gigahertz then our sampling should be more than 2 gigahertz. So, it was not possible, but by interleaving of ADCs we can actually increase the sampling rate. So, for example, our one a lot digital converter it can support up to 500 megahertz by using 4 ADCs and you interleaving of them we can get 4 times of that sampling frequencies.

So, this is the architecture which we are showing here, our input voltage it is applied input signal is applied here, and they are applied to 4 different ADCs at different sampling instances. So, in the right-hand side diagram you can see that this is the clock input in the black color and we have a clock which is starting at the 0-degree phase, and according to that at every rising edge it takes the sample right. So, the red one it is showing that sample, which is it is taking we put the next clock which is 90-degree phase shifted with respect to previous 1 and again at the each rising edge it is sampling one of that sample. So, if the output of all this clocks are added together basically, we are having 4 types of the sampling with respect to the previous 1. So, basically if this was the sampling instance, with the red by using all the sampling instances of different colors, we are having multiplied by 4 times of sampling rate.

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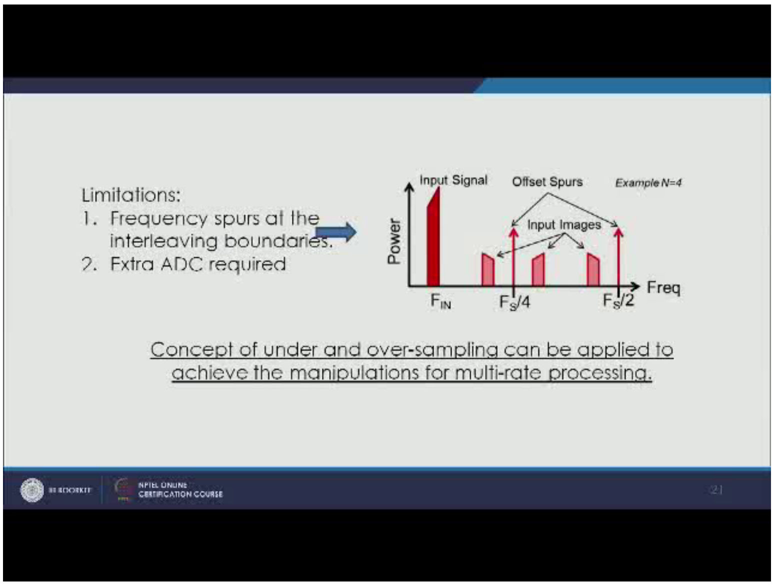

So, it is called interleaving of ADCs for increasing the RF frequency in the advanced architectures we are trying to achieve that many RF companies they are actually working on this principle and giving very high subject frequency of 2 gigahertz and beyond.

Now, what is the limitation first of all obvious limitation is that we require the extra ADC instead of one we require the 4. So, if the cost of the ADC is lower than it is a good architecture and in some cases, we cannot avoid this complicity if we have to have that high sampling rate for example, in the 5 j under the 6 gigahertz range it is proposed that we might have to go for the signal bandwidth from 200 megahertz to 400 megahertz bandwidth signal. So, by Shannon Nyquist theorem we have to have 400 to 800 megahertz sampling frequency. So, this kind of a structure we cannot avoid any more, now what are the another limitation, because they are using interleaving it might not be perfect.

So, there are might be frequencies purse at the interleaving boundaries. So, force for example, we have 4 levels we will see this 4 images there which are at the distance which is the input frequency distance from the basement from the 0 IF. So, we were seeing this kind of images, there if you have 2 level of ADC we will see only single of this image here to conclude the concept of under and over sampling can be applied to achieve the manipulations, for multi rate processing and to achieve the advanced kind of architectures.