Basics of software-defined radios & practical applications
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Lecture – 07
Distortion Parameters-Part I

So, in the series of, basics of software defined radios and practical applications. Today we are starting the topic distortion parameters. So, we have already discussed some of the parameter such as, LO leakage, DC offset, IQ imbalance and, in this distortion to reducing kept parameters they have different effect on, different topologies for the transmitter and receivers.

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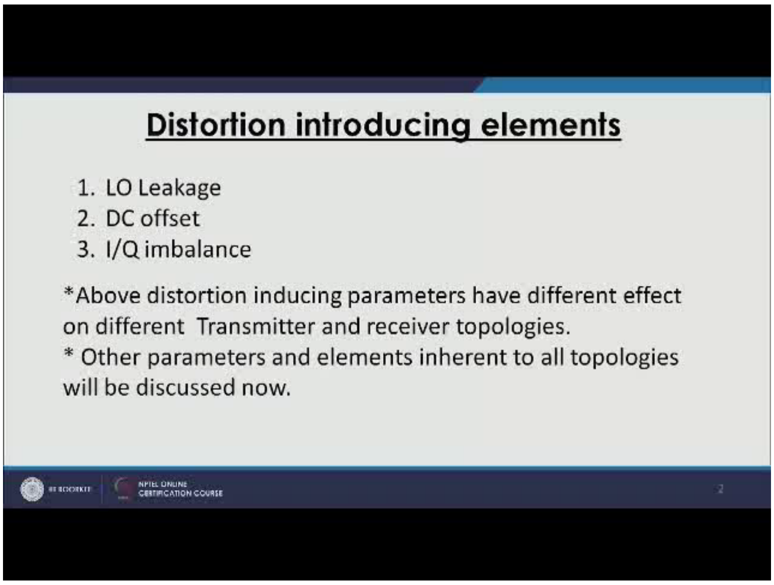

Now there are other than that, some other elements and some other parameters which are inherent to all type of topologies and, their part of all the structures.

So, these are the elements we are going to discuss now. First of all, inherent noise in the semiconductor devices. So, the devices which are used in the power amplifier design or in any of the circuits, of fpz etc. they are some inherent noise there and, this is also known as 1 upon f noise or flicker noise.

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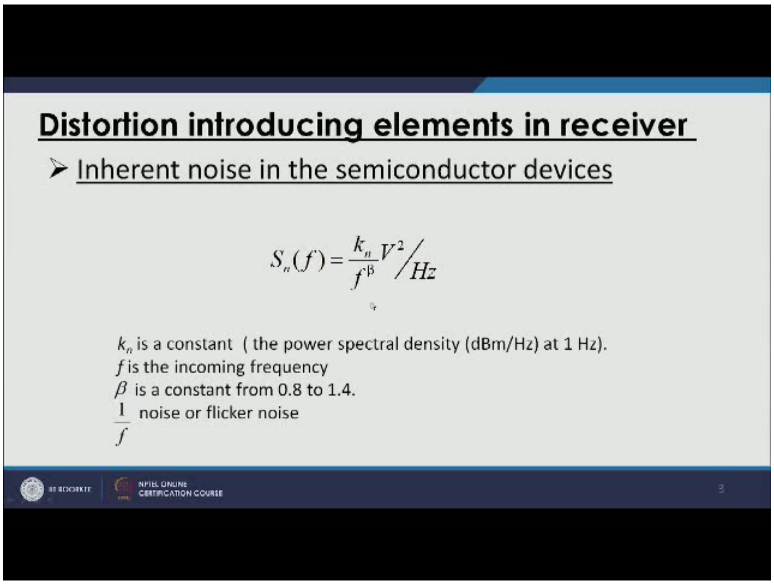

So, this is the formula for this noise. So, this noise if it is shown, as a function of frequency then, it is proportional to a constant k n, which is mostly power spectral density taken at hertz of the frequency. It is proportional to the square of the input incoming voltage applied and, it is inversely proportional to the frequency it to the power beta. Now this beta is a constant, which is taken from 0.8 to 1.4 and 1 upon f is a specifically called flicker noise or 1 upon f noise.

So, this kind of noise it impacts the, early stage of the frequency, if you want to calculate this frequency, in a particular bandwidth from f1 to f2. Let us take beta is equal to n of equal to 1 and then, we can do the calculation by integrating this over that frequency band.

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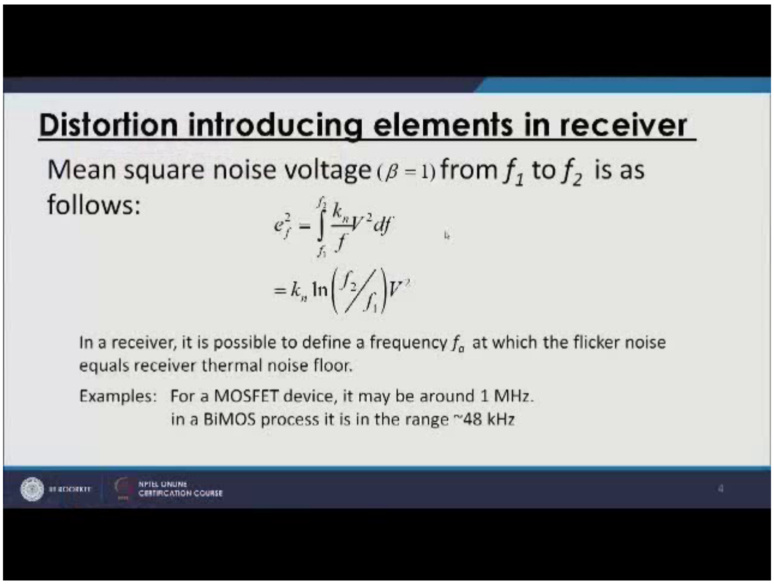

So, we are doing the integration here, k n upon f, beta is taken equal to 1, into V square dfand over the duration of f 1 to f 2 it has been integrated. But, doing the integration of 1 upon f, we are getting natural logarithm of f 2 upon f 1 and k n is common there and, this is the expression, which we get for the mini square noise voltage. So, in the receiver normally, whenever you are talking about a flicker noise, we define it in terms of particular frequency fa, at this particular frequency for that particular device. The flicker noise is equal to the receiver thermal noise 4 floor.

So, by defining this particular fa, we are able to define that noise floor. So, for example, for MOSFET devices, it may be around 1 megahertz and, in the literature, it is found that, in a bimos process, it is in the range of 48 kilohertz. Now, if you are look at it, this fa it is of the order of 1 megahertz and 48kilohertz. So, it is at very lower frequency. So, which kind of architects will be more impacted by this flicker noise, normally the signals, which work near the DC range or, which has the category the profile, which is very straight, very square like then, those kind of signals, they will be mostly affected by this flicker noise. So, when you are you have a signal which is the shape, which is tapering in the nature then, it is not much affected by the flicker noise.

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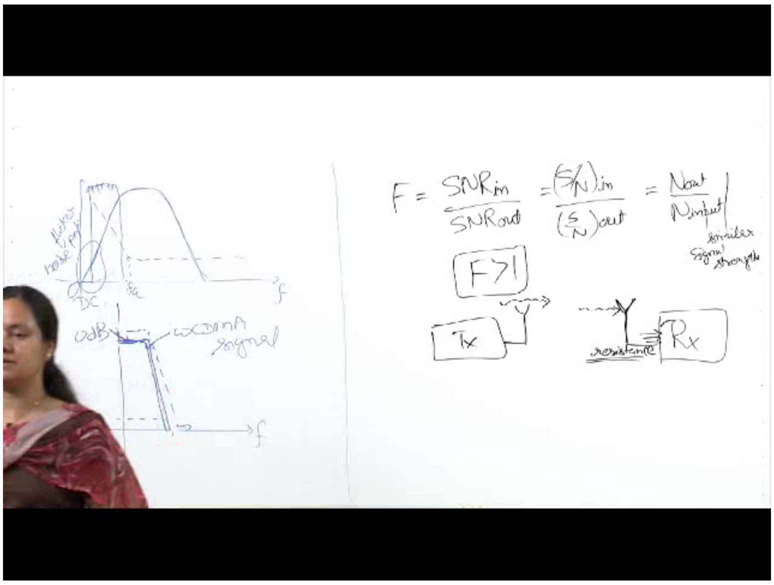

Why is that? Because as we have said let us say, it is fa and for this fa, the noise floor is basically defined. So, we are taking about this flicker, noise, amplitude we are showing it here. So, this is the point where it is actually finding it is threshold.

So, suppose it is more prominent near this region near 0 frequency or DC frequency region. So, whenever you have signal, which has the profile, which is the straight profile, something like that, multi carriers kind of signal then, this flicker noise which is appearing in the lower range it is affected by this. If you have a signal which has profile like this, then this noise affecting a very small portion of that new signal right. So, that is why some of the signals are more susceptible to this kind of noise.

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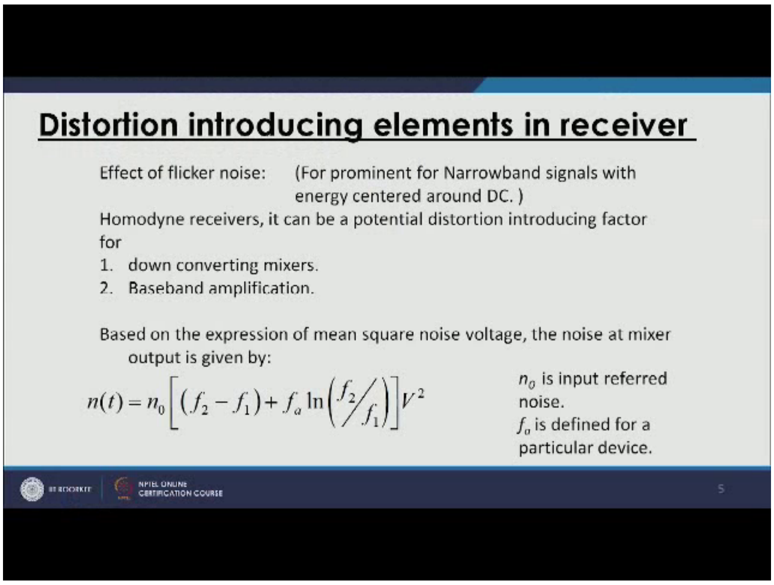

Now, based on the expression of the mean square noise voltage, the noise at the mixer output is given by this expression, where n 0 is the noise at the input. So, at the input whatever noise is being applied referred to input and, fa is the defined parameter which was shown for this 2 per devices, to be 1 megahertz and 48 kilohertz. So, in which kind of architecture impact more, for the homodyne kind of receivers because, in this kind of receivers the signal is appearing near, very near to the baseband or exactly at the baseband.

So, because of that, the down converting mixers, they will be getting impact by this or, if we are doing any amplification in the baseband at the DC level frequency then, it will be impacted by that. So, by looking at this expression, which is the mixer, for the down conversion at the IF, and near IF and the baseband level; we can see that it is directly prefer, proportional to fa and V squared of the voltage.

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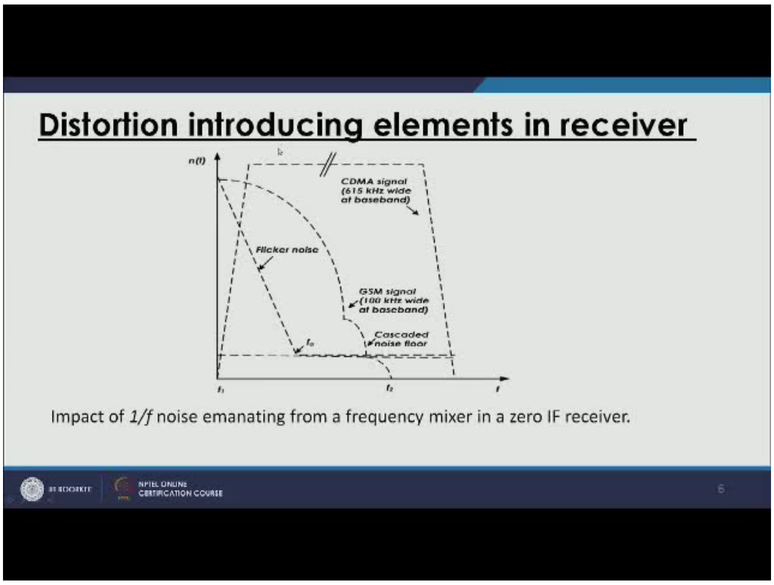

If you look at two particular kind of profiles, this is the flicker noise, which is more prominently at the frequency, as I told you and this is the fa which is defining that boundary of the frequency. Now if our signal profile for example, CDMA, WCDMA kind of signal, they have this kind of profile, trapping profile then, it they are effected, only in the small portion. And if you have a signal, which is actually covering the baseband and it has this kind of profile for example, GSM signals because, they are bandwidth is also lower. So, that is more impacted by this kind of flicker noise.

Now, is another thing is that we can automatically see that, it is more prominent in the homodyne architecture, heterodyne because, we will be working at the IE frequency, which will be at the high frequency result. So, it will not be impacting that much into that higher frequency region. Now, apart from that flicker noise, which is actually because of, the device property we have something called, converter noise. So, what is the converter noise? We have heterodyne structure, we have IF stage, where we do the up conversion or the down conversion of the frequency. And after each stage, if you remember of architecture we put of filter, either it can be band pass. If it is in the analogue and RF domain and can be low pass in the digital domain, mostly we assume that it will remove all the elements interfering signals, from entering to the digital side in the receiver and to the analogue side in the transmitter. But actually, what happens, it stops most of those interferences, but the noise is inherent.

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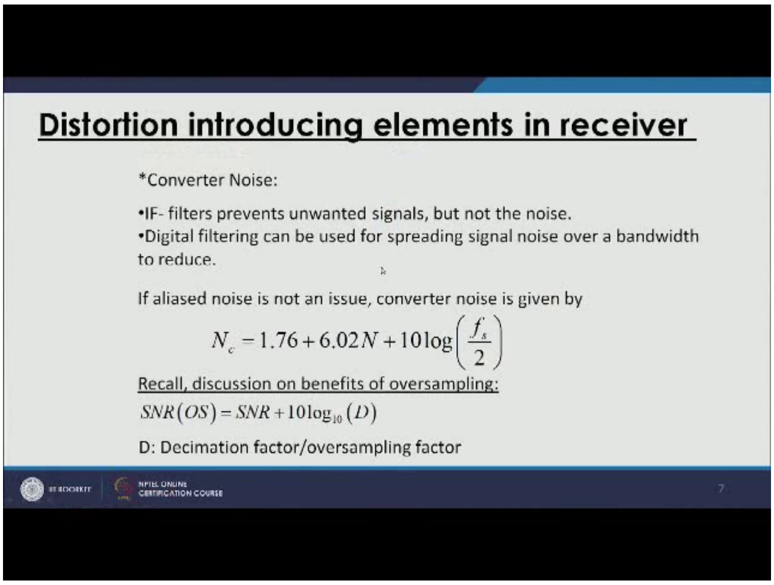

It is distributed in the whole range. So, it is not stop by those filters. In fact, each filter because, it is kind of passive element, it introduces some of it is own noise, thermal noise etc.

So, this noise keep propagating and at the end, we get lots of noise because of this, filters also now analogue filter will add the noise, what we can do? We can actually do the digital filtering so that we can a spread this noise over a particular bandwidth or we can do the boot force filtering in the digital domain. What do I mean by the boot force digital domain filtering? What I mean to say that, once you have achieved, your here signal here and you have done the quantization in the digital domain. It mean the digital domain you are receiving signal like this, let say it is your WCDMA signal. Your analogue filter will be something like that and, it has because it is a passes structure, it is a analogue structure, it will add the noise. But in digital domain, you can simply put a sharp filter there which can go quite below the noise level and it will cut off all the data beyond this, point. And it is possible because, we are doing the post processing in the additional domain.

So, it is called possible to do, it will reduce the noise lot and, you will have your most of the signal having the core information. So now, aliasing noise which is coming onto your baseband, because of the down convergent process from the mixer. From IA filters if those kind of noises are not an issue and we just, think about the noise which is inherent in only converter, then it is given by this formula. This noise 1.76 plus 6.02 times n, n is number of bits, plus 10log10 sampling frequency divided by 2. So, if you look at this formula, this is showing the converter noise, with respect to your voltage, in band. If it is assume to be constant near 0 dB. So, it is the highest point here, if you recall when we were discussing the benefits over sampling, we discuss this pointer also. That by increasing the sampling rate, by increasing the over sampling factor, which is given by D here, it is also called the decimation factor, we can improve our signal to noise ratio.

So, how does that happen, we can see the same thing here, that about noise which is here. We can see that this, distance from the signal can be increased by, increasing the sampling factor. So, we can increase our signal to noise ratio significantly, by using the over sampling there, but it is there if only converter noise is taken into account. Now because of that, the overall SNR gain due to IA filtering process, it can be given by this formulation.

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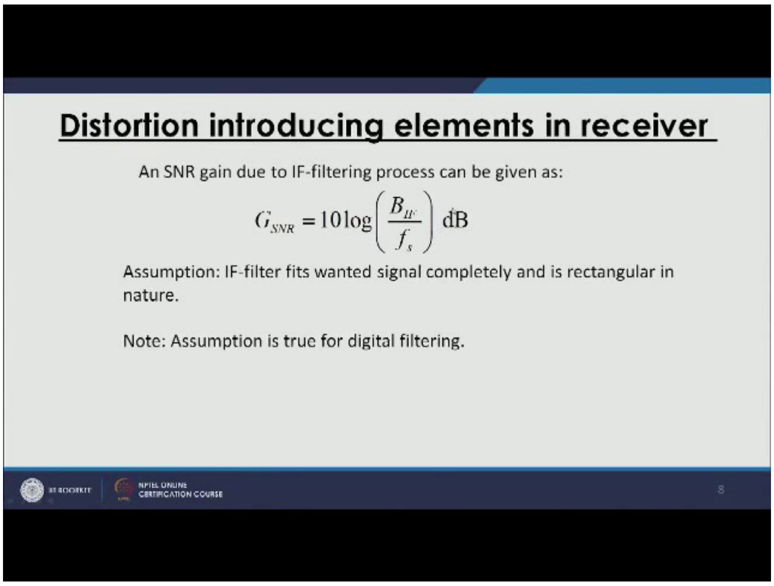

This BIF is actually bandwidth of this, IF filter and fs is the sampling frequency and this, is the gain in the SNR because, of these two factors. Now in this expression, actually we are assuming that our IA filter it fits the signal completely and, it is rectangle in nature; means it is just covering that portion, only and no scope for any other signal. For example, in the same example here, if it was WCDMA signal and we wanted to the filtering then, we are saying that of a new filter is, exactly over our original signal, again if you are doing this in the digital domain, then it is not a problem we can do this because, it is a kind of post processing.

Now, that was a flickering, we have discussed flickering noise and we have seen the noise which will come from the converter size because, side because the IF filter it will not remove the noises.

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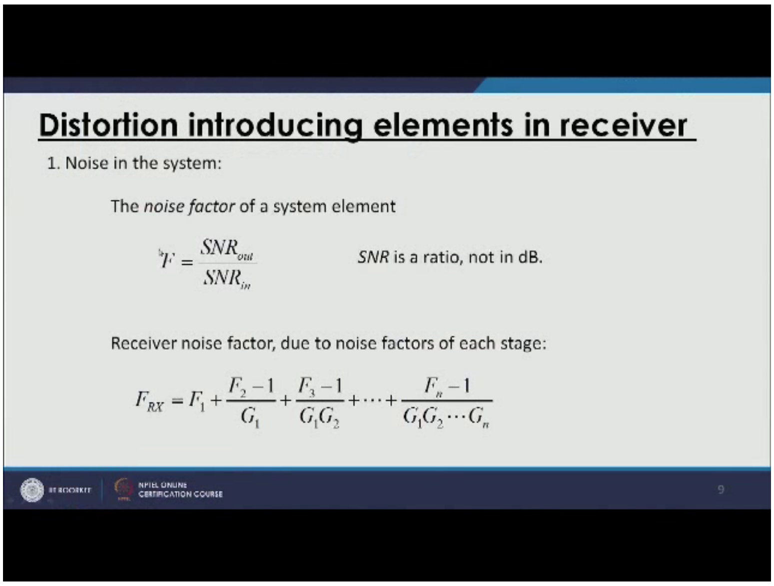

Now, let us have a look at the noise in the overall system, complete system any receiver. So, there is a term called noise factor. The noise factor of a system elements, it is given by F. So, basically these factors is given by SNR in divided by SNR out or signal to noise ratio in, divided by signal to noise ratio out. Inter, if the signal levels are same for the both, then it becomes noise level at the output divided by noise level at the input, for similar signal strengths. So, basically because, if we have equal noise at the input and the output there is no added noise in the system, then we will have factor is equal to 1, but it never happens normally, noise of output is always more than the noise of input.

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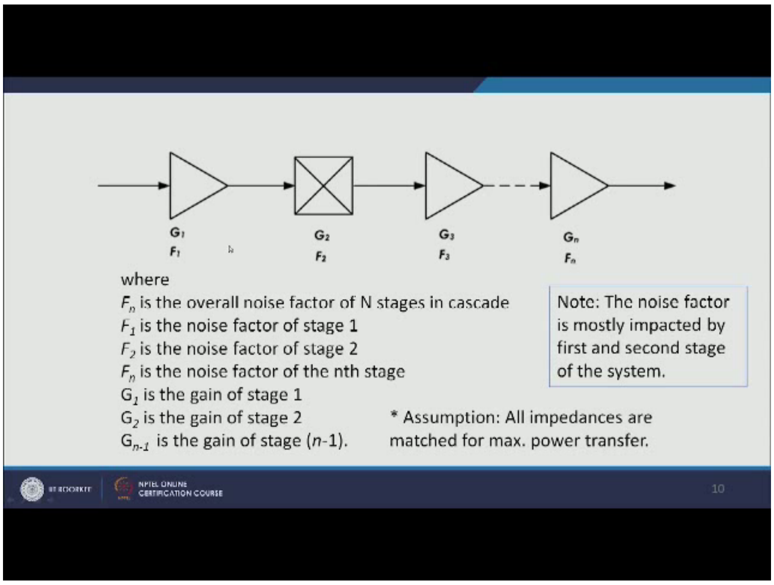

So, F will come out to be always greater than 1. So, noise factor, so, the receiver noise factor because of the noise factor of each stage, is given by this formula F1 F2 F3, they are basically the noise factor of each particular stage. Fn is the overall noise factor of the whole chain which is at the end. G1 G2 and Gn minus 1, they are the gain of the each stage. So, if you look at this expression, what are you saying here that the F1, which is the noise factor of the first stage, it is most prominent here and after that as we keep increasing number of stages, the latter stages the impact of the stages is quite low. So, for the second stage it is divided by G1, for third stage it is divided by G1 G2. So, of course, they are becoming, is smaller in their magnitude. So, what we can say here that, noise factor is mostly impacted by first and second stage of the system, other are negligible in compare to this factors.

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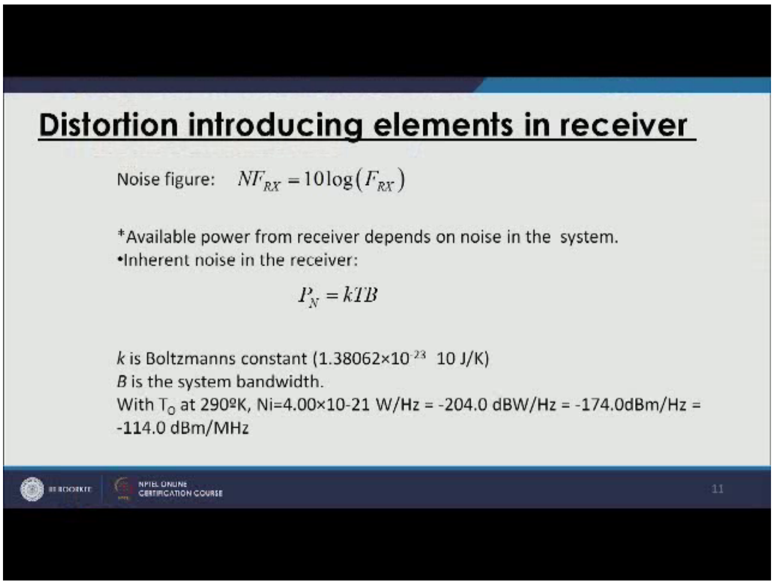

So, whenever we are doing to this calculation, we are assuming that there is no other problem in all the impedances etc. In this system they are perfectly well mesh for the maximum power transfer. So, we are just dealing, we are just concentrating on the noise factor here. Now once we have calculated noise factor, if you take the 10-base logarithmic and multiplied with 10, then it is called noise figure. So, this term noise figure is a heard a lot in the receiver side, we have to keep in to mind that noise in the transmitter is on the other issue and, the noise in the receiver is another issue at all completely; means if you have, transmitter antenna and you have receiver antenna, you might have the transceiver, where you have transmitter receiver in the same system. But mostly these two are independent components. So, once we have transmitted the signal form here, there is signal property the information in that signal that is there, but the noise will be decided independently from this end.

So, when it is reach here, the noise which the receiver has to deal with is actually coming from this antenna resistance and, after that all the elements inside the receiver itself. So, the transmitter noise does not have much impact on the receiver noise, because of the channel etcetera. We will put most of the noise, high level noise and distortions there. So, both of the noises they are treated separately in this case, now what will be the available poll to this receiver. So, as I said that antenna is this that is particular resistance. So, initiation or the input noise is coming from this element there and this is given by PN kTB, where this k is the Boltzmann constant, given by 1.38 into 10 to the power minus

23 joules per kelvin and b is the system bandwidth in this case and T naught, this T is normally taken at the room temperature which is 290-degree kelvin.

If you do the calculation for this one, it is coming out to be minus 204 dB w per hertz. If we converted into dBm by doing watt to dBm convergent and it becomes minus 174 dBm per hertz. Again, if we convert is hertz into megahertz range then it becomes minus 114 tbm per megahertz. So, this is the input noise and after that other stages will add other noises. So, this is one example here.

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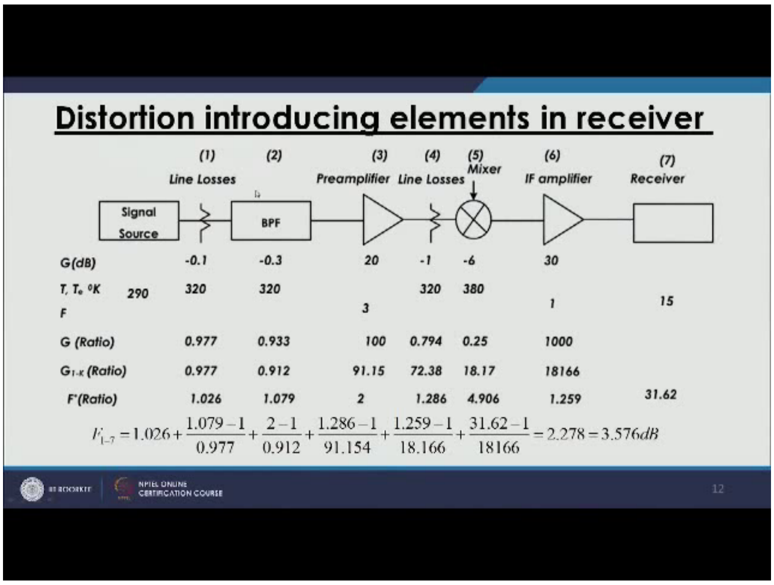

As we can see here, that we have at this signal source and our first input noise is coming from here, then the noise losses, they are making the first stage, there is a band pass filter it is second stage, free amplifier third stage again, there is a line at has losses then it is fourth stage, mixer has the fifth stage, IF amplifier at the sixth stage and then, eventually the receiver in the additional domain it is seventh one, seventh portion.

So, for this system how do we calculate the noise factor? And, eventually we will convert it back to the noise figure. So, you are given the gain of these stage, as you can see the line losses cable, they will not be giving any gain, but they will be giving attenuation. So, that is why it is negative in the sign. So, minus 0.1 band pass filter is also giving the loss. So, minus 0.3 amplifiers are actually providing the gain. So, 20 and 30 dB gain a project by this amplifier and the mixer is again, introducing the loss of minus 6 dB. Now, the temperatures if you see, signal source as the room temperature 290 kelvin. Line losses because of the loss it has the heat in the system. So, it is 320 and the mixer, which is the highest. Because it is doing the multiplication, they are always working, if it is active device then, it will have higher heat. So, different temperatures also defined here. Now frequency is defined for the active elements, where the frequency means much BPF, which is band pass filter the lines they are passive elements for the preamplifier it is working at 3 and 1 get there.

Now, see want to calculate the ratio for this one. So, we are given the particular gains at each point, it is 0.977, 0.933, 100, 0.794 for each stages and, we are given the noise factor for the each stage also. So, how do we calculate the noise ratio for this one. So, we apply the our formula F1, which is our first stage factor, we will apply directly after that second stage minus 1, divided by gain of the first stage after that F3 minus 1 and the gain of the previous one, multiplied with the previous one. So, 1 into G2, it is being shown here, after that this into this, into this and 1.286 minus 1. So, we will keep repeating this process for all this stages here. So, after looking at this stages, we are getting 2.278 as our noise factor. And once we convert it into back into dB then we are getting 3.576 dB. So, noise figure of this whole complete system will come out to be 3.576 dB here.

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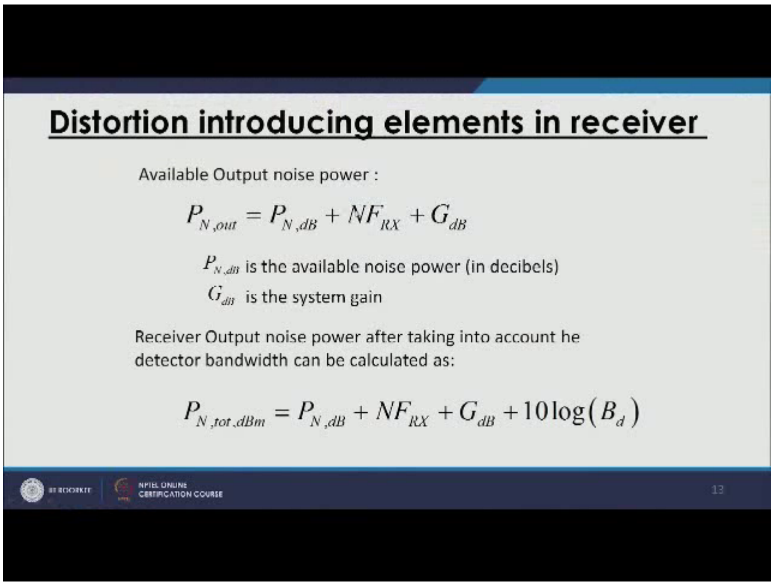

Now, once we have calculated the noise figure, for the complete receiver chain, which is NF RX being shown here and PNdB is the available noise power, which we have calculated from the, previous first stage at the input of the receiver, at the antenna G is the system gain of the complete system, then P out is actually input noise plus gain.

So, this gives us the output noise, eventually at the receiver. Now, these was when we are just calculating at the receiver, but if you look into and take into account the detective bandwidth.

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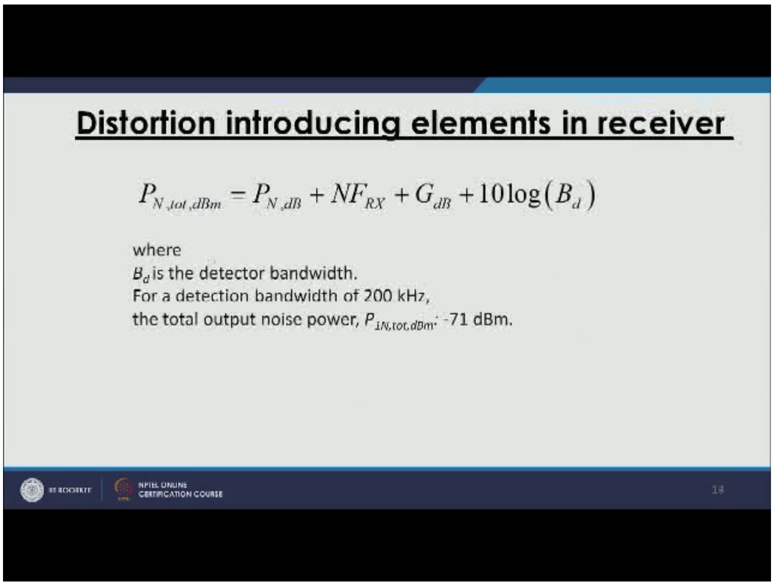

Then the new calculation becomes, when we have the factor of this bandwidth also. So, becomes 10 times logarithmic 10 Bd and, Bd is the detector bandwidth. So, by this calculation, we can calculate our different values. So, if you do calculation for this whole signal, for the given example and we take the 2100 kilohertz of the bandwidth, then we get the total output noise power as minus 71 dBm. So, it keeps dropping in the each of the stages. Now let us have a look at the noise from the oscillators. So, in the oscillator which is basically used in the transmitter, as well as the receiver they are used in the mixer, to up converter down convert the signal.

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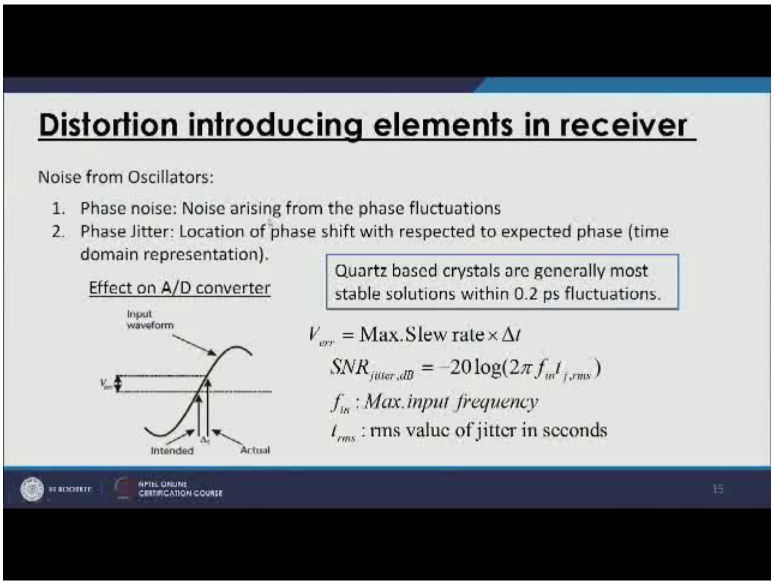

So, phase noise is something, which is a kind of noise, it is the rising from the fluctuations and, if these situations are seen in the time domain and we can compare it with the original point of the phase, then it is called the phase editor.

So, we want to reduce this phase editor, because it is a it will give us very wrong calculation, especially when we are working at high frequencies. So, for example, this error if you see for a sinusoidal signal, it is incoming wave and it was the intended instance. And, because of our error it has been shifted to this one. So, this Verr has come into the picture, now this V error is what? It is basically del t this shift in the phase in the time domain, into maximum slew rate. What is slew rate? Slew rate is basically this slope of is this is incoming signal. So, basically, but just by differentiating it with respect to the time, we can get this rate.

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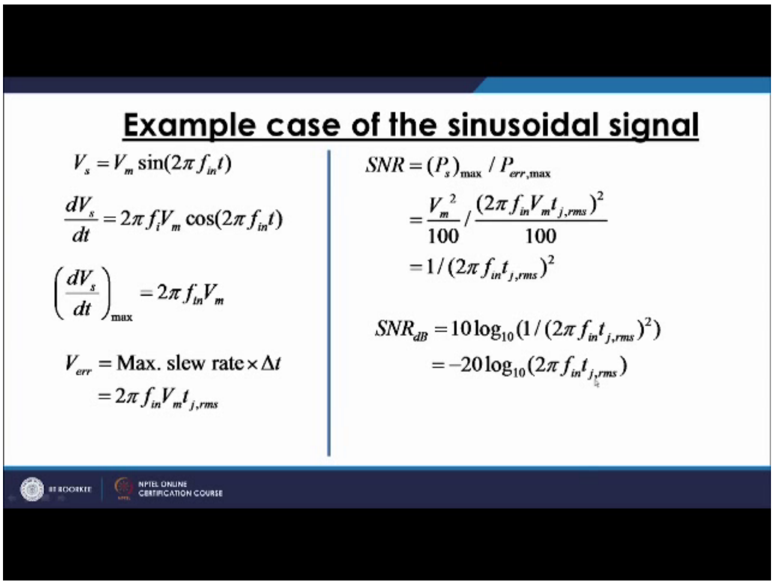

So, let us do an example, here we are seeing the example case of the sinusoidal signal. So, this signal has aptitude of Vm, it has incoming frequency as fin and t is the time, because you want to calculate the slew rate of this input signal, we differentiate with respect to dt. So, we get 2 pi fi Vm cos 2 pi f in t, we want to calculate the maximum slew rate. So, it is d by dt of Vs max, now here the maximum value of the cosine function is actually 1.

So, we have taken it equal to 1 and the maximum of this d by dt of Vs is coming out to be, 2 pi fin Vm. Now, our error voltage because of this jitter is actually maximum slew rate into jitter in time. So, it becomes 2 pi fin Vm. What is the maximum slew rate? And jitter time rms, which is given by tj rms.

Now SNR of any signal to noise ratio, is actually maximum of the signal divided by error, power which is maximum for that distortion signal. So, if calculate like this, our signal is Vs and it has maximum value Vm. So, it is power will Vm square divided by 100 and, it is calculated for R naught equal to 50 ohm, which is the resistance chosen for the Rs systems, now error which we have calculated here is 2 pi fn Vm t jrms. So, we have taken square of this term divided by 100.

Now, this 100 gets canceled out and our SNR, basically becomes 1 upon 2 pi fin t jrms whole squared. So, when we calculate the dB of this SNR term, it is 10 log 10 of SNR absolute value. So, this is the expression here, if we simplify this logarithmic term, it that becomes minus 20 log10; 2 pi fin t j RMS which is the actually expression, which we have shown earlier.

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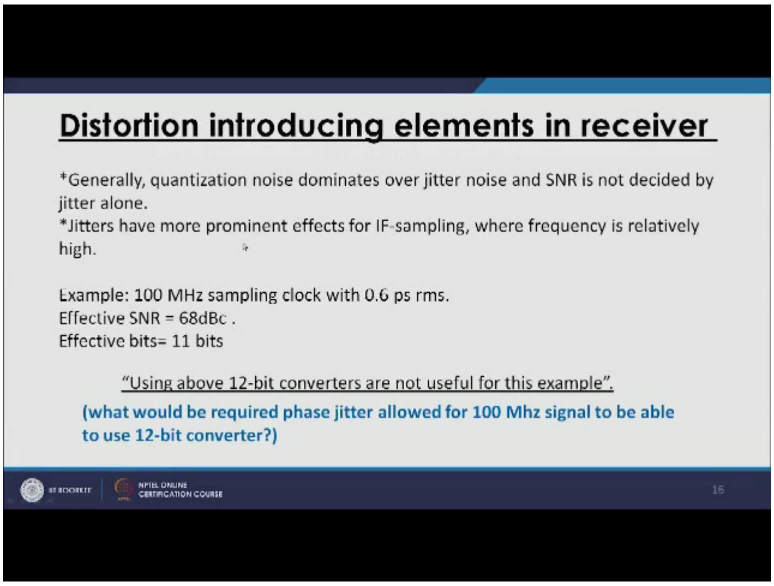

The distortion is more prominent effect at the higher frequencies because, our f input is a factor here. So, if f input is high, then it will become prominent there so, far not DC case, but for the IF sampling case, it is relatively high and if you are talking about the Rs sampling cases, then we have a special take care of this effect. So, I am showing this example of 100 megahertz sampling clock, with 0.6pico second RMS value for delta t.

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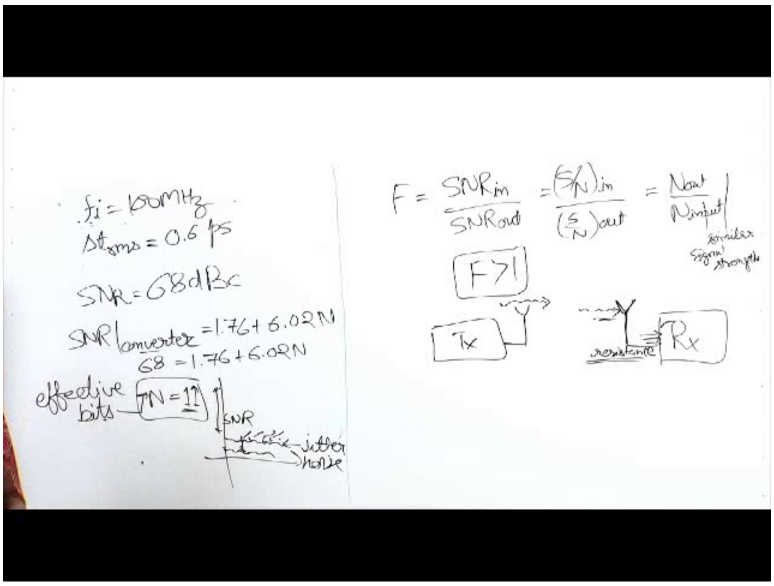

So, what is given to us we have fi 100 megahertz and we are given del t RMS to be 0.6pico seconds. Now, if we calculate you are effective SNR for this one, from the formula it will come out to be 68 dBc. Now this 68 dBc, if we calculate by our formula of SNR, for the converter based on the quantization noise.

So, it will be equal to 1.76 plus 6.02 into N here. So, 68 equal to this one, if you do this calculation this N will come out to be 11. So, effective bits are coming out to be N equal to 11. So, what does it mean? It means, if we have any converter which is of value more than 12, then the it will not have any benefit of that because, the effective bits because, of our jitter noise is coming at to be 11. So, the noise will be fixed at this particular value, because of the jitter noise.

So, even if your quantizer is able to give you the noise level of this one, it will be below that noise. So, it will be the dominate noise right. So, let us given example what will be the required phase jitter, which is allowed for a 100-megahertz signal. So, that it is able to use 12-bit converter, instead of this 11-bit converter. How do you do that? We apply the same formula and instead of N, we are going to use 12 from that we will find this 60 8 dBc SNR and, we will plug this back into the original equation, here and we will find what should be our t jrms.

So, I am keeping this as a one of the assignment and in the next lecture, we will continue from this point and I will reveal the answer to this problem so.

Thank you.