Basics of software-defined radios and practical applications
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Lecture – 16
Linearization Techniques for non-linear distortion

So, in the series of basics of software defined radio techniques and practical applications. We were discussing the linearization techniques of power amplifier. Under this technique we have earlier covered different type of models, these kind of mod models such as, sally model, wrath model, Gurbani model. You can simply put into a MATLAB or any other tool because, they are equation based, and then you can use them instead of power amplifier. By keeping those tools in the any transmitter system, you can actually visualize the effect of power amplifier non-linearity.

Now, after looking at the impact on the signal quality now, let us, go into the details of generation techniques, to remove the unwanted effect of power amplifier non-linearity. So, if you look at this slide.

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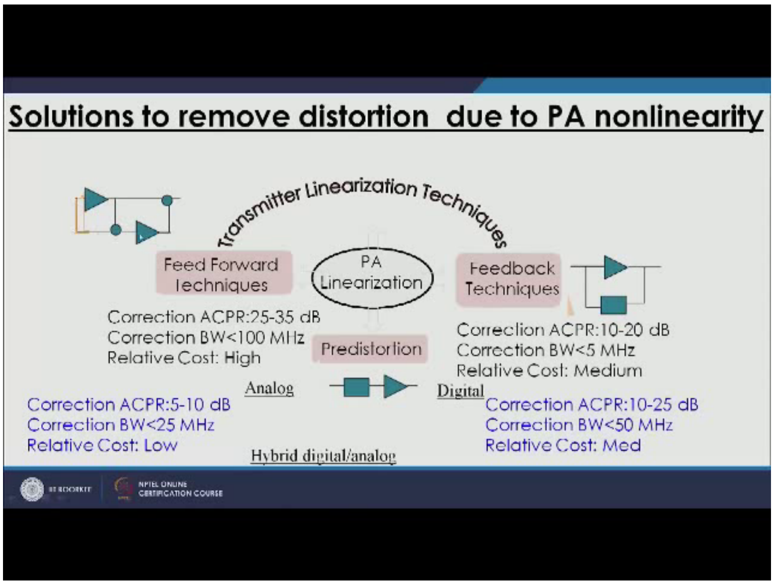

We are showing different transmitted linearization techniques the basically, there 3 types of techniques which are found in literature. One is called feed forward technique. Another is called feedback technique. And a third one is called pre-distortion technique.

So, we will go through these techniques. So, that we have idea of the state of the art, which is currently available in the market. The first of all feed forward technique, which we can see here. The feed forward technique contains 2 power amplifier. One of the power amplifier is the amplifier, which we want to linearize.

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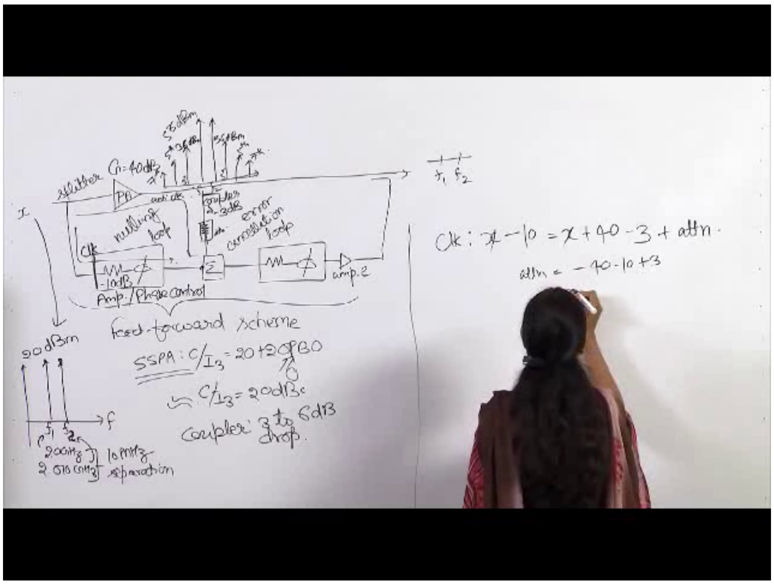

So, let us say, this is our power amplifier, which you want to linearize. So, the schematic of feed forward goes something like this. So, basic signal which is the input signal x it is applied to power amplifier as well as it is sent to a different path to a parallel branch.

In this second branch, you have a attenuator, and a phase shifter which controls that amplitude at the face of this input coming incoming signal. And then, these are some together. And second loop starts. In the second loop, again you have a attenuator, and the and a phase shifter. In second path also, you have a amplifier. And this is the output of this whole system. So, this whole a scheme is called, feed forward a scheme. This is called feed forward because, you take the error signal from one path, and then you feed it to the second path. And the second part does the second type of correction.

So, how does it actually work? You have your input signal here. The first loop is called nulling loop, and the second loop is called error cancellation loop. This power amplifier is the amplifier, which has highest power. This amplifier which we have to use to linearize this power amplifier is actually having a smaller gain, and it does not work into that much higher power level. So, how does it work? We will have a look at this. For example, let us say our input signal x, it is having input power of 20 dBm of each tone. And frequency domain the signal is between f 1 f 2.

So, for example, if you are having 2 gigahertz signal at f1, if they have 10 megahertz separation. The second signal will be at 2.010 gigahertz. So, it will be f 2 and it will be f1. So, let us say it is 20 dBm is the power of the sync incoming signal. This actual power amplifier has the gain of 40 dB and of course, it it is able to provide you very high power. And when you apply the signal here given have 20 plus 40.

So, almost 60 dB m output you should be able to see here. But, actually you you would not see 60 dB been exactly, but, somewhere around that let us say 56 dBm. Because, some of the power is actually going into imd products. So, you are f1, f 2 values. And then, you will have your imd terms, 3rd order imd, 5th order imd, 7th order imd and so on. And then, you will have harmonics. So, it is 3rd order, it is fifth order and it is 7th order.

Now, if you remember, we have for solid state power amplifier. The formula to calculate the carrier to interference ratio of the 3rd order. It is the formula, which we have been using earlier a lot. 20 plus 2 times to power back off. This is the output power back off from the P1 dB point or the saturation point.

So, suppose if it is 0, we are not taking any output power back off then, what basically, we are saying that for any SSPA normally the (Refer Time: 06:36) 3rd order in difference level will be 20 dBc below the original power. So, if it is 56, it will be almost 36 dBm power the. These are continuing almost 36 dBm because, of this formula. It is approximate because, it is the approximate formula thumb rule for the SSPA here. So, this much we can see.

Now, what we want? We want actually our signal to have only our amplified signal, and we do not want these signals. So, here, you want only f1 f 2 component, and not other component. So, what does this feed forward scheme does? In the first nulling loop it removes the original tone, and only imds are remaining here. So, how do we do that? So, this x which was input was this one 20 dBm power and after that at the output we had this path. So, what do we do? First of all we put alternator in this path.

So, that the gain of this path, and the gain of this path they are equal. So, basically if this path let us say have because, we do not have any gain component here, let us say 10 dB drop is happening because,of this resistor in the path and because, of this phase shifter amplitude and phase control unit.

So, let us say, 10 dB is dropping here. So, gain will be minus 10 dB because, it is drop not the gains the negative side here. Then, this coupler which is allowing this signal to come here will have almost coupler. Generally, will have 3 to 6 dB drop again because, it is a analog component you will have some drop, it can vary from 3 to 6 dB anywhere in between.

So, let us assume it is 3 dB loss so, minus 3 dB. So, then what we will do? We will add the gains of this path, and we will decide what will in what should be the attenuation here. So, that the gain of these 2 paths will be equal, and we can balance this game. So, way will check this way and this way, let us say, I call it anti clockwise and clock it let us call it clockwise. So, clk clockwise, if I say see the power here, it will be power of x plus gain of this element which is actually minus, which is actually loss minus 10, and this is the gain from this path the output power here, from this path it should be equal to whatever has been added from this path.

So, x plus 40 then 3 dB drop because, of this coupler, minus 3 then, this attenuation, this attenuation which we have to decide and then, whatever power we are getting here, it should be equal, let us compare these 2 parts and it will be canceled out and it will come here, ah. So, attenuation will be basically, minus 40, minus 10 plus 3, minus 50 plus 3, minus 47 dB.

So, minus sign suggests that, it should be attenuation and not gain. So, this value will be minus 47 dB. When you put it like this, then, the gain of this path, in this path is equal. What does it mean? It means, that we have removed all the gain, and attenuation of this path and now, the signal we are receiving at this point, at this point is basically, the signal at f1 and f 2, and this is the signal at 20 dBm.

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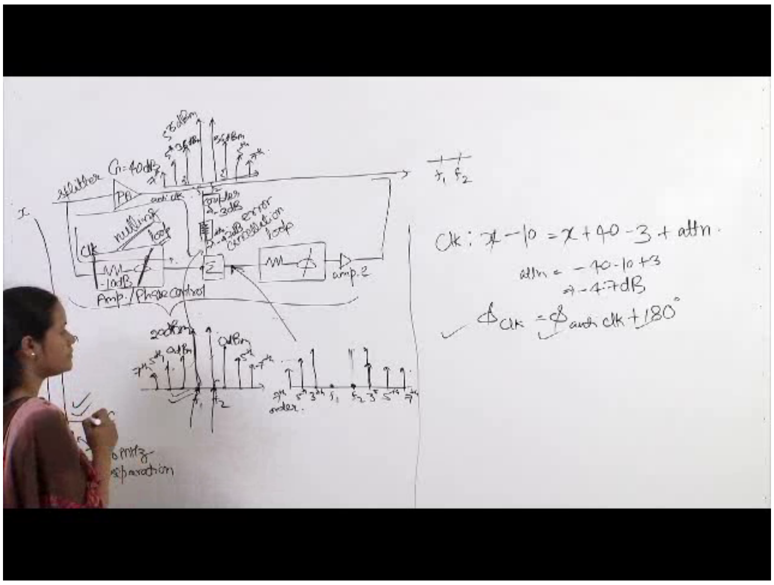

The same power here because, we have removed all the gain which has it has received from here, and other elements. Similarly it will become 0 dBm, for 3rd order we are concentrating and other 5th and 7th terms are even more below here, 5th and 7th.

So, now, at this point this say signal which is coming from here is this one in frequency domain, and the signal which is coming from this path is this signal, right? Now, we start our phase. So, that phase of clockwise direction and phase of anti clockwise direction but, is 180 degree out of phase. So, we select this phase. So, that we can make sure that it happens if this will happen then, these components are it at 180 degree out of phase with this component. It means, this signal at the output of this point will cancel out these terms, and we will be having only these components with 180 degree out of phase condition.

So, at this point in the at the output of first loop we will be having. It is f1, f 2, and if this cancellation is perfect then, we will be having only our imd terms. If our path is only it exactly 180 degree out of phase then, this components will be completely canceled out, and we will not have anything at this point at this point, and only 3rd order, 5th order, and 7th order terms will be remaining.

Now, that is why, we call this first loop nulling loop because, it is nulling the main term. So, what is remaining now, only the error only the imd terms in the second path. Now, comes the second amplifier into picture and these components, this path. The coupler it is giving one portion of the this signal into this path, and the other portion main portion into that path. So, in that path we are a still having all these components with a little bit lower power. Let us say, 3 dB has been lost here. Now, we are getting in the components here.

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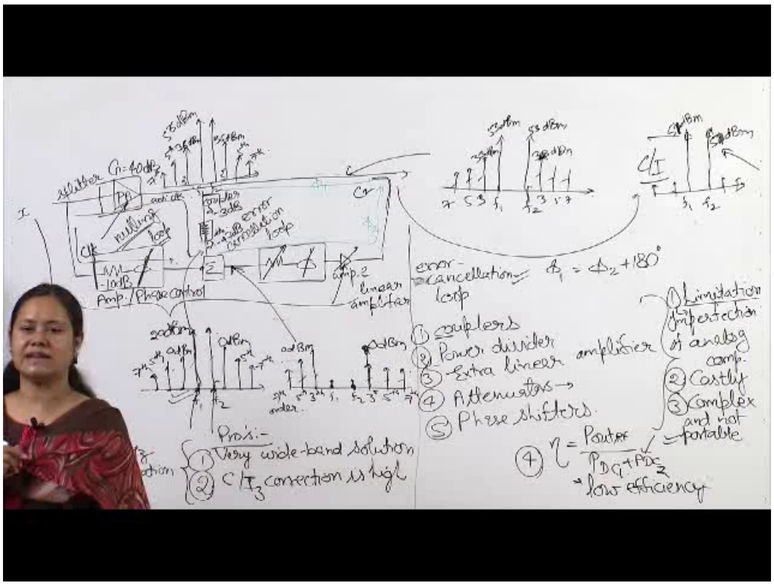

At this point which are f 1 f 2, and this value will be 53 dB. Why because, 3 dB have been lost ? 3 dB has been lost and you have other components, they are also lowered by 3 dB.

So, 3rd 5th, 7th 3rd 5th and 7th are the components here, they are also which was 36 now, it is 33 dBm 33 dBm. This values what should be the role of the second path, the aim of the second path is that we add this imd components to this path. It is the coupler to again it will have 36 dB drop because, of the property of this passive element analog component then, if we are able to add this to this signal and wanted area out of phase then, we will actually cancel all the md terms write all the in terms will be cancelled, and at the output we will be able to get our signal only at f1 f 2 and this is what we require. We want to send a signal, which is a total signal and we want to receive a signal which is again a total signal with some gain.

So, let us see 53 dBm. Now, how will we achieve this? Again, we have to keep in mind that this clockwise direction from here to here phase 1, and the other phase from here to here phase 2. In the ire cancellation loop should be 180 degree out of phase. So, fiber should be 5 to plus 180 degree or minus 180 degree. So, that they are out of phase here, right?.

So, we will measure the phase of this path, and we will measure the phase of this path and we will tune this phase. So, that we can maintain this relation in error cancelation group, error cancellation group. It should allow the cancellation of the imd terms because, they are going to take it out of phase, and there is nothing in the in bed in bend single. So, only this in bend signal will be remaining all the phase will be different, but phase does not matter that can be that can be the compensated later on. So, main requirement is the cancelation (Refer Time: 17:10) imd which we are able to get.

What is another requirement? Of course, 180 degree out of phase will be help us in cancelling these components. But, only when these amplitudes are magic together, right? So, then come the this attenuation and this amplifier into picture. That is why because, this this is to amplify this amply terms to this level, right? So, basically we play with this attenuator, and amplifier to achieve the same level between this imd terms here,.

Now, this amplifier is actually a linear amplifier, it should not introduce it. So, on distortion terms otherwise whole purpose will be defeated. Otherwise it will introduce introduce it is all imd, we do not want that. So, it has to be a linear amplifier [noise, what is the positive point? This amplifier has to amplify imd which are much lower, they are almost 20 dB below the main power level, right? So, it was around 0 dBm here. So, we can use low power amplifier here, and we can use easily it this to be in the linear region of the power amplifier. So, by doing so, we will match the gain of this path, and again the power of these components should be equal to the power of these imd components. So, that the perfect translation will happen ok and we will get our actual signal output.

So, this is the feed forward system. This feed forward system as you can see is the using 2 amplifiers totally in rac one amplifier. So, this was the main amplifier which wanted to linearize and for that which components we had to use, we had to use couplers, we had to use [noi.se] power dividers ok. I think this we had to use extra linear amplifier attenuators, phase shifters. So, all these components analog components are the part of one feed power network.

So, as you can see these are analog components, all of them are a couplers power divider this amplifier, their analog components and they have their own errors, they are not perfect. So, each of them introduce their one errors for example, this coupler will have some losses there. Those losses have to be calibrated properly, this attenuators has to work very fine tuned way why is that? Because, in both of the loops you had seen that, there is at most importance that the power level should make.

For example, in the first level first nulling loop fcr powers are not matching for your P output, and the single output then, this tone signal will not be completely canceled. What you can see, you can see some a small signal here, right? So, because, of this imperfection you will see the in this is small signals here, and when this path it will go and meet with this path then, in turn you will again see some imd terms here, it will not be completely canceled, although C by I ratio might be much higher than the previous one, which for when it was almost 20dbc. Now, maybe it will be 15 or 60 dBc depending on how perfectly systems are. But, because they are the real system they might not be perfect, and you can see some imd terms here.

Moreover in the first path, if you had some terms remaining here, and this term will go here and add with 180 degree with the signal. Which is coming from the other path then, because, they are matching at 180 degree this power will drop because, it will be subtracted from there. So, maybe you will get only 51 dBm.

So, you are losing 2 dBm gain in the main power. So, imperfection of these components is one of the hindrance then, first limitation second limitation, we are using such a bulky escape here. So, it is costly to purchase all this and then, to then to make this system it is complexed, complex and not portable. Once you make this arrangement you do not want to shift anything, you do not want to change anything ah. Because, even this cables, they should be calibrated everything should be calibrated. It is not a portable solution, it is a complex solution. And the 4th one their efficiency of this system will be very low, why is that?

If you remember, efficiency was given by P output in the Arab domain, divided by PDC. So, output power is decided by the signal we are receiving here, right? So, summation of these 2 tones that will be our output power and PDC is the dc power consumed by the power amplifier. But, in this case our PDC is not only conserved by this power amplifier, it is also being consumed by this another amplifier, and this amplifier is not giving us an again it is just helping in canceling these extra imd terms.

So, the P output out of power is same. But, the PDC becomes PDC1 plus PDC2. So, of course, this generator power is increasing an efficiency is going down for no efficiency. Now, these are the limitations what are froze here, very wide band solution. Because, this powerful amplifier it is limited only by the bandwidth of such components phase shifters, your terminator, your couplers and this amplifier. The passive elements are mostly very, very wide, when they do not give us our trouble, but this amplifier has to be broadband, and if it is broadband then it can give you a very wide band solution. So, it was a very wide band solution. So, it has been use for even more than 2 bands. It has been very popular for the implementation the base station when you can, where you can use the bulkier size solution also, but maybe not for the mobile. Because, it will take a space it is very wide band and C by I correction is high.

Because, if you can have these components very C by I 3 because, 3rd order is mostly higher. So, basically, if you are able to get this tuning properly done, even in manually you can achieve very high level of C by I ratio. Because, the imd correction path is independent of the actual signal processing path. So, these are 2 positive points and these are the limitation. These limitation, they call for other techniques and this is how we go to this next technique, which is feedback technique. So, as you can see here, this have a power amplifier, and we are using one feedback. This feedback will contain time delay element ah. So, let us, have a look there.

So, feed forward had a big problem that, it had it was not portable, and it was complex and costly because, it was using many components. So, feedback technique is a solution to that.

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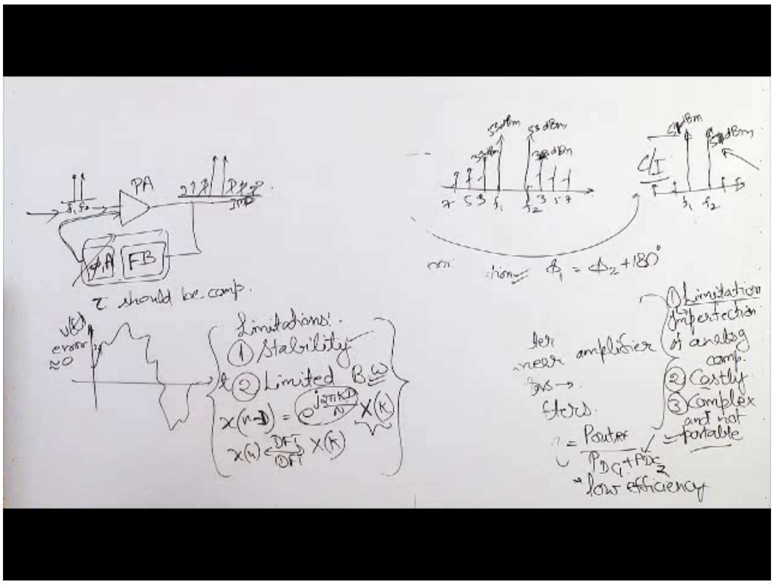

So, in this technique basically, this is your power amplifier and we say, that distortion is occurring at the output power amplifier output. So, suppose these are the input signals. So, at f1 f 2 and these are the imd components at the output

So, instead of using feed forward, we simply take some portion of this sample at the feedback, and with some phase and amplitude correction we apply the signal back here. So, this output signal it contains imd terms and as well as main tones. So, when you get it back, we design our feedback circuit in a way that we will remove the effect of main signal here, delay should be compensated. So, that whenever we are applying the signal here, we are able to get the signal at the same instant when it is being applied, or nearby because, if signal is if our system is not changing very fast, then, this PA will have almost same property for nearby signals.

So, for example, with t, if this is your system is signal at the PA, and we are working very near 2 signals when it is almost the error is very is small. So, it almost excites PA in a same way. So, as long as signals continues, there is no sudden jump, this assumption it holds and because, of that this is Vt amplitude I am I was talking about. So, because of that as long as you are able to composite for tau in a reasonable manner, and you play with the phase and amplitude then, we can control this imd performance a little bit. And because, you can see this same elements, which you are using in a feed forward they are being used only once in one feedback then, it is a very portable solution, and it is less costly solution as compared to feed forward solution.

Again, limitations here, advantage is very clear. But, limitations main limitation is stability. Whenever, we are talking about any feedback solution in from the control theory, you might know that. We have to take into account the stability, if the system can go unstable for particular jitters for particular excitation it might live, it might lose, it is a stable a position. So, stability is one of the concern because, input fluctuation jitters cannot always be controlled.

So, one main concern stability the second one is limited bandwidth. So, what is this? Limited bandwidth because, we are dealing with a system which is time delayed version here, right? So, if from your DFT formula, if you because, you are dealing in the discrete domain, if you remember if you do something a shift of d delay in discrete time domain then, it appears in the sequence domain and this way. So, it is your frequency domain representation, representation of xn, DFT IDFT and it is it is frequency domain representation at the present instance. But, one signal is coming from a delay then, it it shows it appears as a function in frequency domain, a component in the frequency domain.

So, we cannot go to higher bandwidth because, that those bandwidth will be representing actually a delayed signal here,. So, they will interfere with each other. So, if it is a single carrier signal it is. Because, these kind of components will be falling nearby the actual tone, but if it is a bandwidth signal then, you will have distortion components, which will be interfering with this.

So, because, now, we are going for the very hap by higher bandwidth signals when we are moving to moving from 3g, 4g towards 5g so, this feedback scheme, is not that good scheme especially, when we want to have a system, which will keep working continuously, they might be jitters there might be other effects, which we cannot control effect of at atmosphere, temperature etcetera. So, it is not a very reliable solution although it is very portable solution. So, what can be the compromise between these 2 and here, comes the pre-distortion technique.

Pre-distortion technique is kind of combination of feed forward in the feedback technique. In the feed for technique, we remember we were removing the distortion term and then, we were deducting them at the outer loop. Now, in the pre-distortion technique, we put that element which is introducing the imd terms, before our power amplifier. So, this is the scheme, where instead of using in the feedback we can put something.

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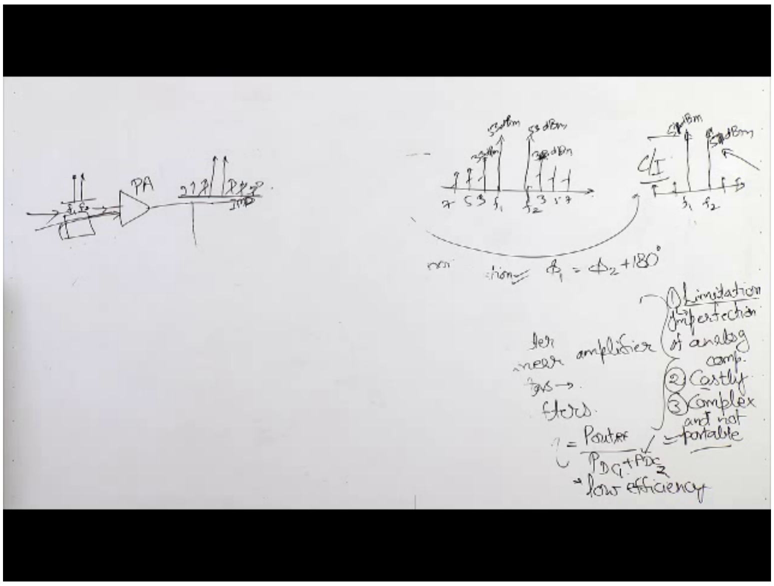

So, this previous distortion is something, we are going to discuss in detail. Because, it is inherent part of digital signal processing seen techniques, which can be used in software defined radios. So, we will be discussing in the next lecture.

Thank you.