Basics of software-defined radios & practical applications
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Lecture – 17
Predistortion Techniques for non-linear distortion in SDR

Hello everyone. So, in the series of software defined radios and it is practical applications, we were discussing power amplifier nonlinearity and the generation techniques to overcome it is negative effects. So, today we will be discussing pre-distortion techniques. We already discussed feed forward technique and feedback techniques.

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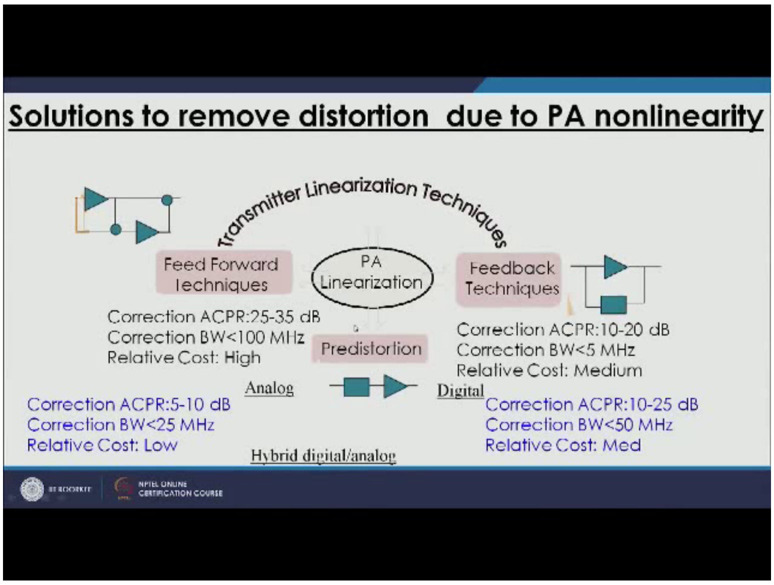

These are basically unlock techniques and we were discussing pre-distortion where it contains the elements of feed forward technique as well as feed feedback technique.

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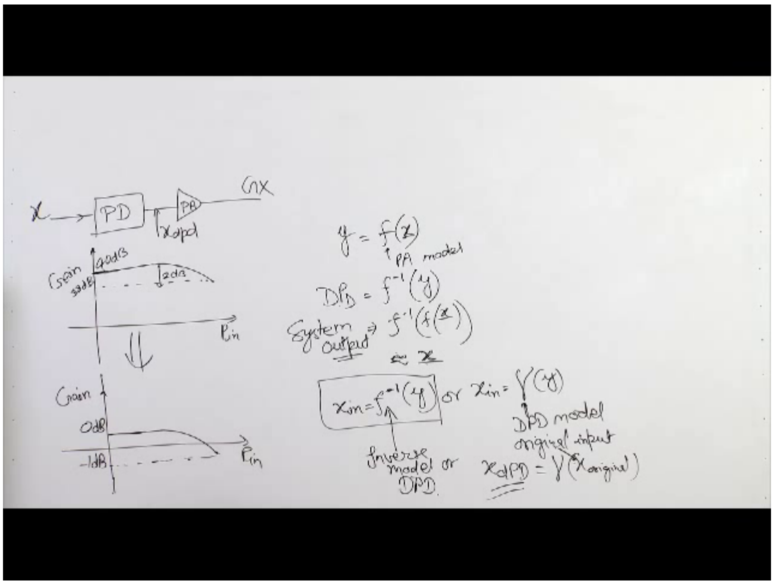

So, in this kind of techniques, if this is your power amplifier this pre-distortion element is before power amplifier. So, the imb terms which we were generating in feed forward by using different loop. We get this imb terms by using a feedback in an offline manner. So, it becomes a combination of feedback technique as well as feed forward technique.

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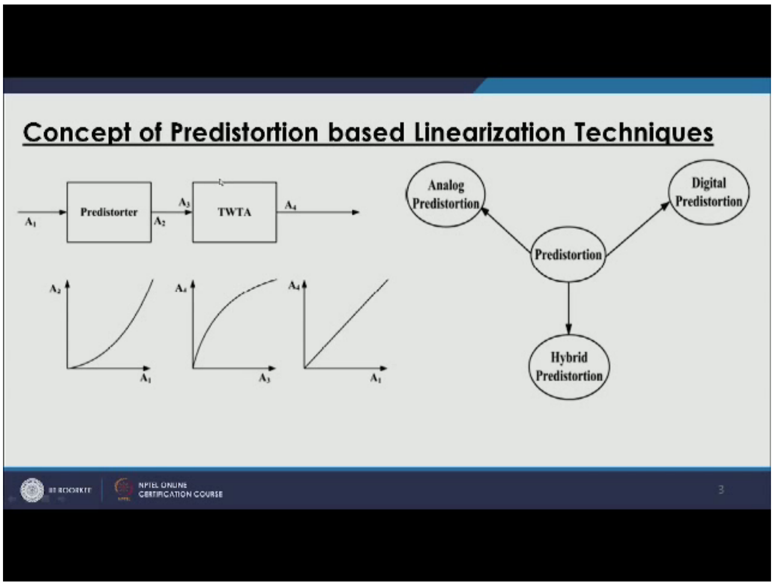

So, once we have our data of the device. For example, in here you can see tm TWTA as our power amplifier traveling wave tube amplifier. And we can see the A3 versus A4 which is the input power versus and the output power curve we can say it is saturating after some time. So, we get this information that what is the output power with respect to input power? And then we make a characteristic which is a opposite of that one. If, we can put the element we change the characteristic which is opposite to that one before this element then the combination of these 2 give us the an linear region.

So, again this predistortion techniques, where we are putting element before power amplifier it has a correction bandwidth of from 10 to 25 dB. So, it is better than feedback techniques and little bit lower than the feed forward technique, but it has relative co cost because it is not using that many component which feed forward techniques for using. And it is corrector correction bandwidth is better than the feedback techniques.

Now, because our requirement for the bandwidth which is increasing. So now, we have to have a different look again, that which of the pre-distortion techniques whether it is analog digital the hybrid will be better. So, let us have a look at this techniques 1 by one. So, let us start with digital pre-distortion technique and this is the backbone of the software defined radio. In software defined radio, we can define our Parameter Parameters in the software in the DSP of the system.

So, that is why the digital pre-distortion is something we wish to discuss in detail.

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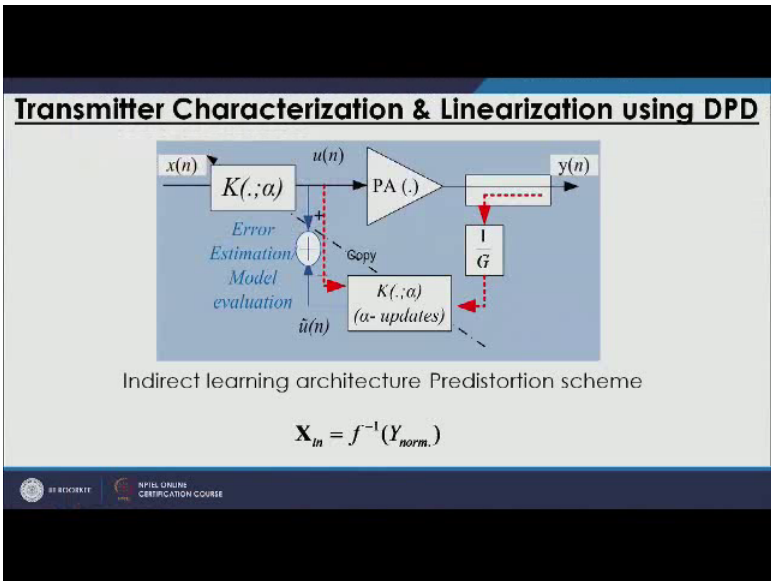

So, here you can see the power amplifier is here , we apply our signal which is xn and. We have one element which is in the digital domain and initially it has gained 1. So, the output of the digital domain is u n. So, when there is nothing, when there is no coefficient upd updation here. In that case, xn is equal to u n. So, basically this xn is being applied to power amplifier. the power amplifier output is captured here and then because PA output will have some gain we remove that gain by dividing it by a small signal gain.

So, how if it look so, if it is P in versus gain which we also called amm characteristic we have discuss it before and this gain is small signal gain is let us say 40 dB and after sometime it is starts getting saturated. So, with respect to 40 dB it is getting saturated and it has 2 dB compression. So, what should be the power of this point? it will be 38 dB. So, this g is this gain the small signal gain, when it is flat, but we divide our signal by this a small signal gain then what will what happens? Our output signal will look like it is 40 dB m. So, let us say it will be 0 dB because we have divided with respect to 40 in voltage domains. So, in dB domain we are basically subtracting this. So, around 0 dB and we can see a still 2 dB compression which will be which will be giving as this point. So, the output after this PA after 1 upon g will be having 0 dB gain it is small signal at saturation it will be showing minus 1 dB gain.

Now, using our yn and un we update our coefficient and we make a non-linear model. So, that that non-linear model will make these imd terms which are opposite to that of the imd terms of the power amplifier and once we get this information, we update our K alpha which is our pre-distorter element. So, for first iteration when there is nothing only once are there u n is equal to xn, but when we update these equations of our pre-distorter then, this un will be different from xn and un will be some kind of non-linear function. As we have only shown that it will be non-linear, but this if this will be non-linear in the opposite sense of that of power amplifier.

So, basically what we are trying to do here to get the inverse modeling here, our function of PA. So, if f is the modeling function which is representing our PA nonlinearity after remove removing the gain, let us say y was given by f x upon g because we want to model only the nonlinearity set we can remove the gain and then we can try to model the nonlinearity.

Now, this is the effect this is the function of the PA. So, PA model we can call it the inverse function DPD output is actually f inverse function which is applied on y. So, that the whole system because y is actually f x upon G. So, digital pre-distortion output is f inverse y and system output will be actually f inverse and y is actually fx upon y x upon G. So, it should be able to gives a give us the actual signal. and of course, because we have removed this G we can again multiply it back you will get, you are have to signal. So, this effect of g we can remove. So, that we it is conversion feed this we are assuming that they are both at the same power level after removing the G and if we assume that then, our system output is actually x and this is what we wanted we wanted our system to have an x and we wanted our output to have Gx without any distortion.

So, keeping this in mind our x in and y if you want to do the inverse modeling, we do the modeling in opposite way it makes it means we try to model this relation and because of that whatever function we get it is our inverse model or DPD model.

So, again you can see we require some kind of model here instead of writing f inverse I can give it a new name. I can say that, I want to find a function let us say gamma function of y and this will be the DPD model.

So, by capture data we will set all the output at one side and keep all the input we have captured at one side and we try to find this model. Once we find this model we will actually apply original x here. So, x DPD will be actually gamma of x original. So, for finding the model, we put x n is equal to gamma y and for apply then we put original x instead of y and the x DPD which we will get here will be actually pre-distortion version. Which will when passed a power amplifier will give you linear system. How can we implement such a system in the real life? One example I am showing here.

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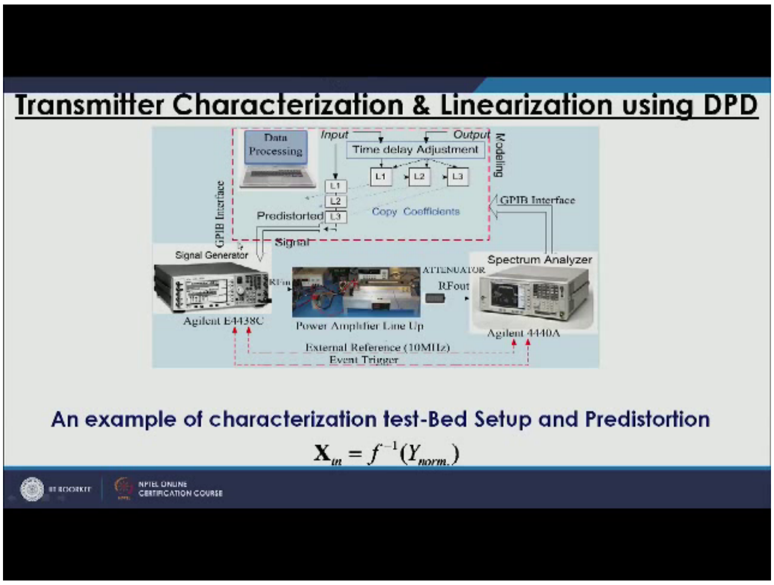

So, this is one example how proof of concept in the laboratories are done you can see here we are using a signal generator which is from Agilent it is called Keysight now. since, this signal generator signal generator what does signal generator do ?

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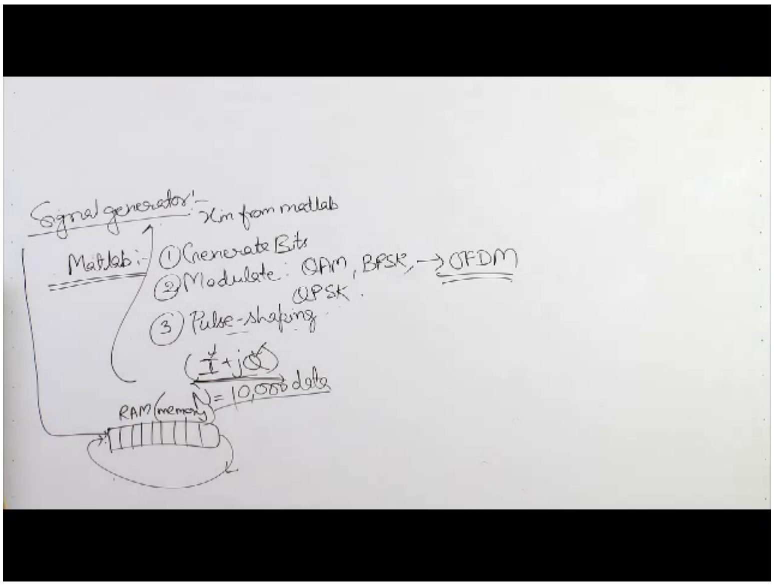

First of all, we choose a set of data; we load the data in our personal computer using MATLAB. Now, when you have decided that x in from MATLAB within MATLAB, generate bits module in Particular format. It can be QAM BPSK QPSK and so on. So, modulation is done there once you have modulated over your signal, then you do the pulse shaping, if you want to apply OFDM. You can apply after this symbol once you have decided your symbols. Then you can apply OFDM if you want to and then you do the pulse shaping then you have your signal with the basement, that signal which is in the form of I plus JQ. Pulse shape diagram which is continuous in nature not in pulse shaped signal. This is of a particular length let us say n equal to 10 thousand data is loaded into this signal generator.

now this data is generated by us. So, we have the information of this data and once you have loaded into the signal generator. What this signal generator do? It takes a format where you can give it the TXT file of I and j columns or it can accept the data which is of the this complex format I plus Jq. And that data can be read by this instrument why this instrument read this data it starts this data it saves this data in it is ram or memory within this signal generator. And after that it keeps repeating this data again and again.

So, for one it should once it well read from 1 to 10 thousand data after that again from 1 to 10 thousand and so on it will keep repeating that data again and again. that data is being sent to the dac which is inside this signal generator. That dac will convert it into a log domain you will have knobs in this kind of instrument which in which you can choose the carrier frequency. So, up conversion to the carrier frequency using mixer is is already happening inside it. And the RF you can get from the output from this signal generator

Now, this RF is applied to the power amplifier which is working in that range. So, this is the power amplifier lineup which is a very high-power amplifier root 3 power amplifier. Now it is RF output is taken into the spectrum analyzer, a spectrum analyzer will have some range for example, it can take only 20 dBm power that is why we are showing attenuator here this attenuator they apply to keep the power range low enough to keep our spectrum analyzer safe. So, spectrum analyzer shows us the spectrum, if it is vector spectrum analyzer then we can also get inq data from within this analyzer.

So, what is the analyzer does this analyzer does the down conversion to baseband frequency and then it does analog to digital conversion and after that we are able to get our data.

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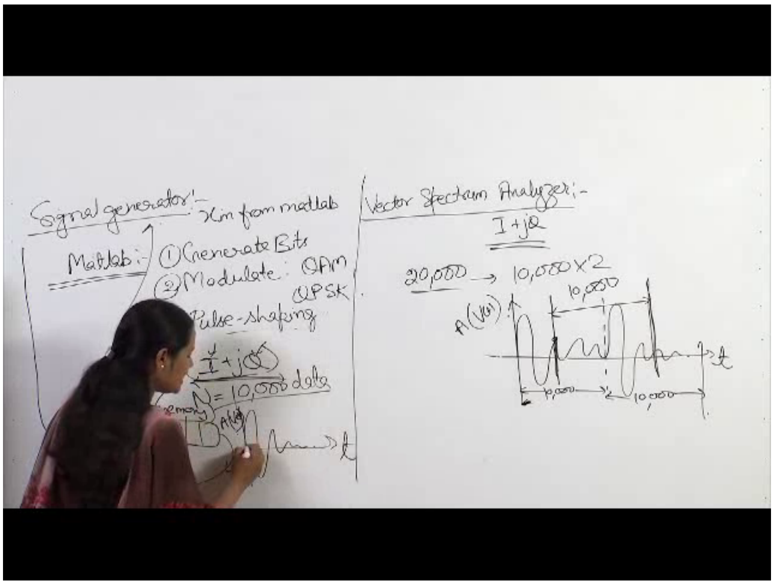

So, vector spectrum analyzer which will again provide you I plus jq rata data at the PA output.

Now, you have originally providing the 10 thousand data now it keeps repeating. So, suppose you capture 20 thousand data from this vector spectrum analyzer. it means, you have received this 10 thousand data 2 times because, this is what you have loaded into generator. So, it is basically kind of training sequence you are sending and you are capturing outside. What do we do after that? For example, with the respective time, this is the amplitude of signal, amplitude of v t. and this is how it looks like?.

So, basically the same data which you have sent if you have captured for the twice of the previous time, you will have 20 thousand data, but it has the same information twice. So, you only need to capture 10 thousand data at a time.

So, originally let us say this was the single sent I am again drawing the amplitude only. it was the send signal and it is the received signal you know need only 10 thousand delta. Suppose, you have captured 20 thousand data you cut this data anywhere. So, that it contains only 10 thousand data. right if you cut it like this what does it mean? You are capturing your data you are starting to read your data here, but you will notice it after some time it is again showing the data which was there before. Which we are not actually capturing, but it is repeating. So, you are not losing any data

What is the main thing here? Our data it is starting from here. if I want to compare this data with this data we will have a problem of the time delay because, starting is starting from here. So, this is the delay to remove the confusion let us say this is our capture data.

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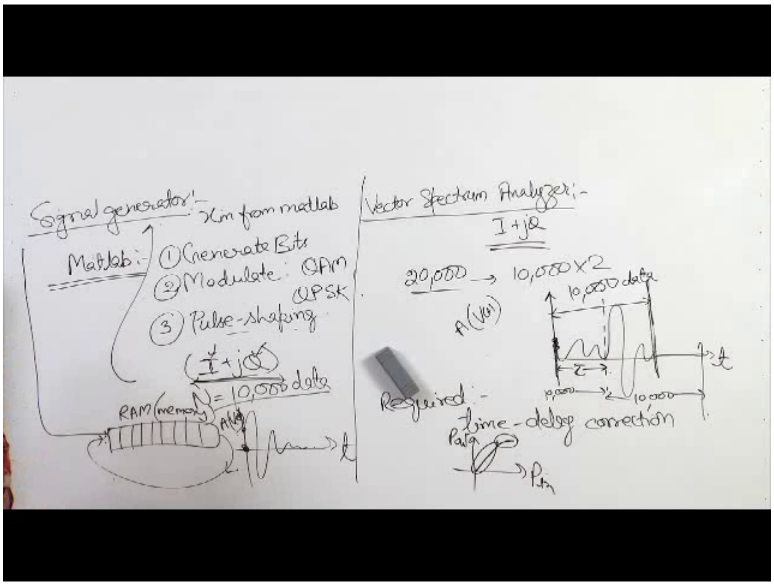

Of the 10 thousand data window at the VSA and this is the time lag, but in real life because the system is continuously running and you will suddenly start capturing your data. So, you do not know stand it will start capturing. So, this delay will be there and you will not be knowing it.

So, we require time delay correction. Because, whenever you want to characterize any data then you should know one to one mapping of that data, you should know the input and output relation. And that cannot happen if this delay is there otherwise for this data you will be reading this data for this next data you will be reading this data which is a wrong information.

So, how can we do that? We know that power amplifier characteristic if we plot it with respect to P in. How does it look? It look linear for a long time and after that is saturates. So, most of the time the data is in the linear region only for a smaller time it will have the n effect of distortion there.

So, relying on this fact we do the cross-correlation bit of the input signal and the output signal. And whenever we get the peak of our signal that point we will know that it is our delayed point and then we had just over output data and we will put all these data after this and this data and this data will be actually time aligned.

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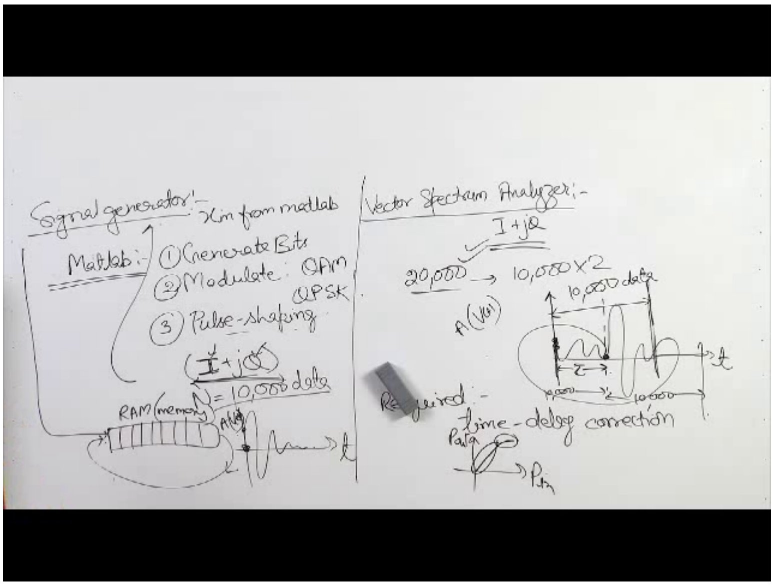

So, we do the cross correlation to know this time delay and compensate for that

So, this is what we are showing here that this both of the vector signal generator and vector spectrum analyzer they are connected to our digital signal processing unit which is laptop in this case. As we are showing here, they are connected using GPIB here once we have our data in the input in the form of I plus jQ which was this data which we sent and we created in the MATLAB. So, we already have it and this I plus jQ which we captured from the VSA from the output of the power amplifier they contain the information of the nonlinearity.

So, after time delay removal we have done the compensation. Now, we can actually start to see the relation between these 2. So, again, after compensation this point is shifted.

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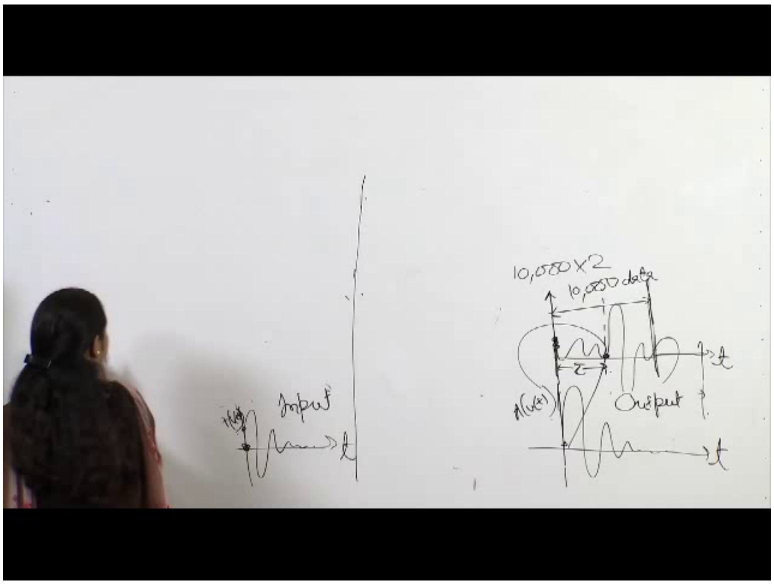

To this point right. So, this is how we will read it. And now this is the input signal, to the power amplifier and this is the output signal from amplifier. And digital domain we have in digital domain we have information of both of these values.

So, let us remove this and now concentrate only on input and output values.

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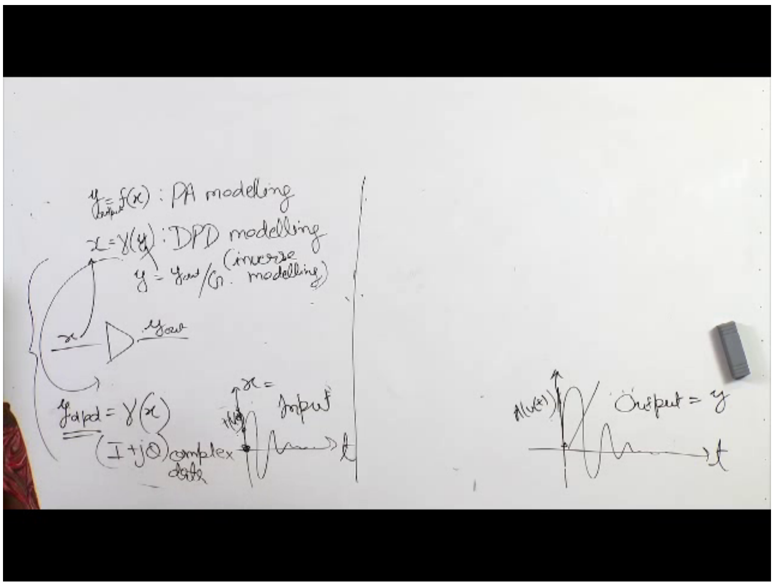

So, from eyes they have some similarity in their shape, but because system also have nonlinearity, you will see it is not exactly same it will have some similarity in the eyes, but somewhere some appear will not match. So, when it is input and output and we do the modeling and let us say it is x and let us say it is y, y is equal to fx is called PA modeling. Because it will give you the relation between PA input and output. So, if you are able to model this function then this models can be used in MATLAB or any other processor. We had seen the parametric model earlier in the previous lectures and they are doing the same thing they are giving us the PA model.

But we were discussing digital pre-distortion and digital pre-distortion we do the digital pre-distortion modeling. And it actually is let us call gamma y this is this function only this y is actually y output divided by gain gain is removed because, we want to get the nonlinearity effect, but not the gain after removing the gain.

So, this modeling is called DPD modeling it is also called inverse modeling because it is kind of inverse of the original relation. So, similar to the modeling same models can be applied here, by exchanging the input and output. So now, output so, input of the input of PA. it becomes output and the output divided by G it becomes it becomes your input the thread is called inverse modeling once you get your gamma then you know your model.

So, how do you apply your DPD once you know your gamma then you say DPD model is known and you again apply your actual input which is x original x is applied through this gamma and this y DPD comes into picture which is then again loaded to the signal generator using MATLAB.

So, please remember here, that your x and y they all are complex quantities here. So, this y DPD will again have the stream of I plus jQ data. this data which is generated inside MATLAB is loaded into this vector signal generator like we have loaded our original x signal here. And then again sent to power amplifier and then we can observe the spectrum at the receiver here.

So, this is the process of doing the pre-distortion, as you can see the application of this distortion is happening inside MATLAB. In this case, or in our that is why it is called digital pre distortion and once we do this then we have to take care of our a to d and d to a requirement, which we did not have to look into when we were dealing with the feed for in a feedback system because it is DSP system here.

So, let us have look at some of the models which we can use for this modeling. Again, I will mention, all of these models can be used for modeling as well as inverse or DPD modeling.

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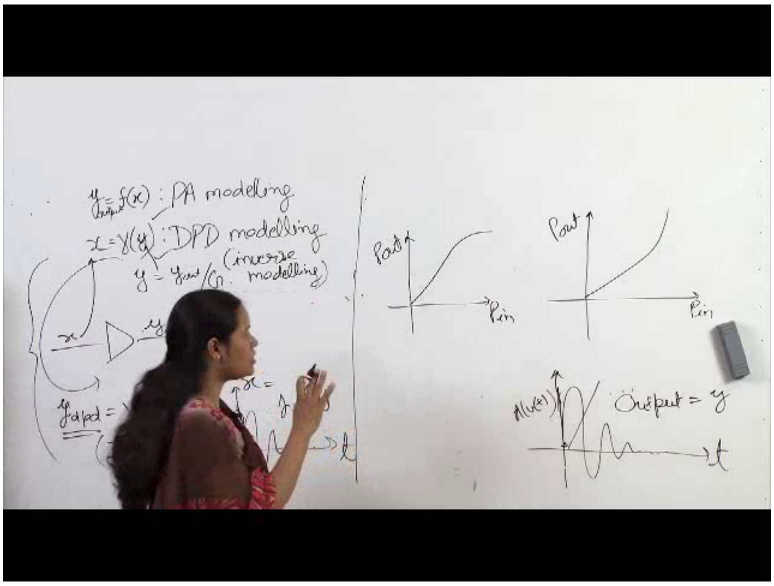

Only difference is that in one of the cases which is modeling case we are modeling this kind of characteristics and were for inverse modeling, we are modeling this kind of characteristics both of them are non-linear nature, but they are different kind of nonlinearity.

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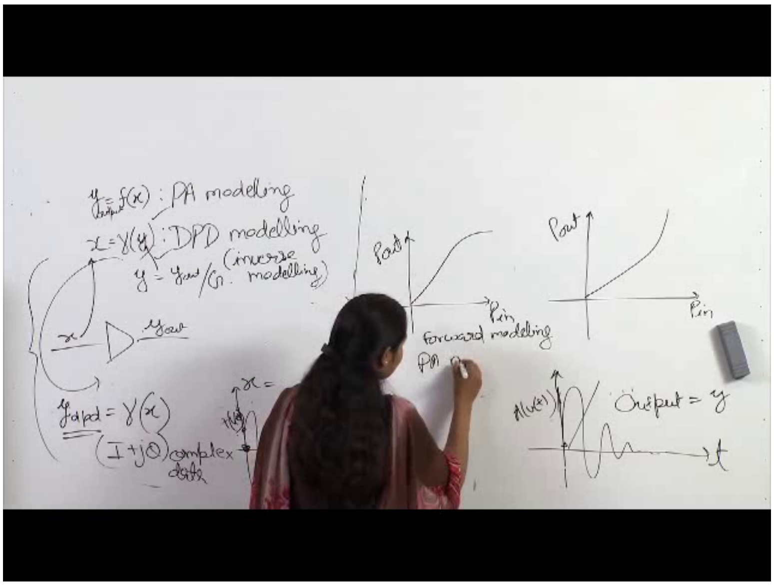

So, it is for forward modeling or PA modeling or PA modeling.

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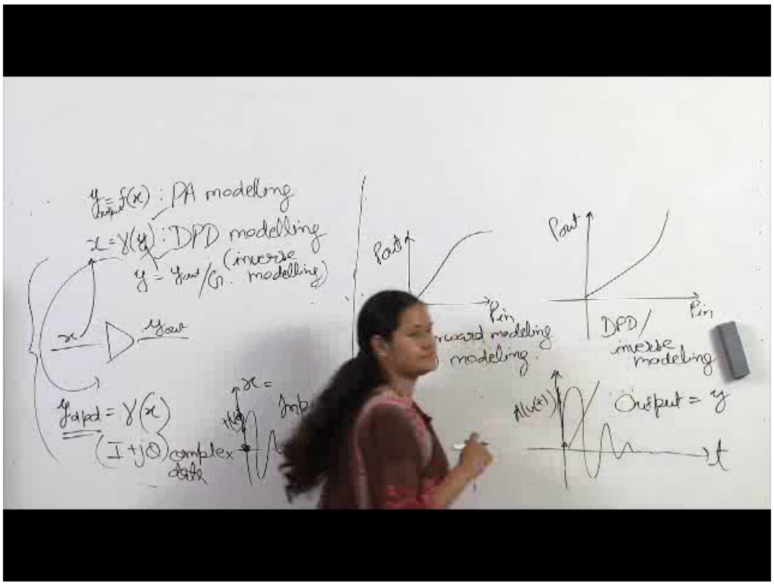

And this is power this is for DPD or inverse modeling.

So now let us have a look at these models.

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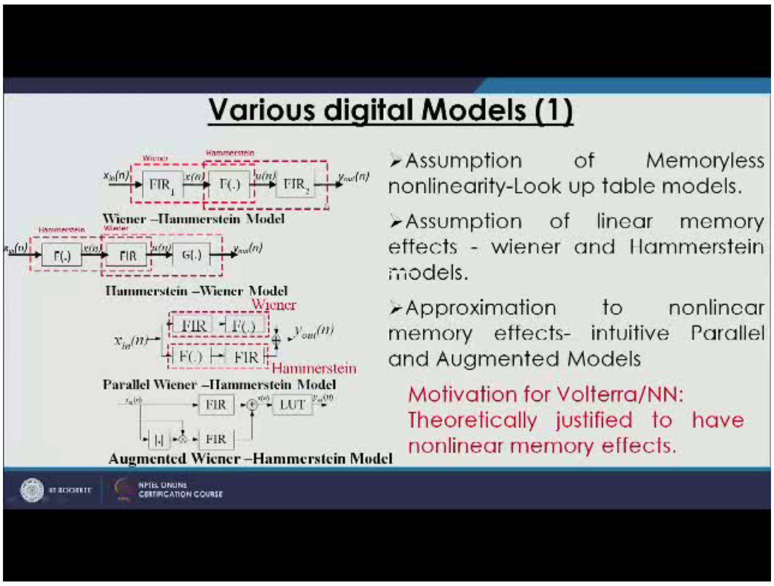

And first of all let us look at very established models in the literature. For example, wiener model or the Hammerstein model. This model assume that nonlinearity can be used by using lookup table we have used lookup table earlier also. So, what is lookup table?

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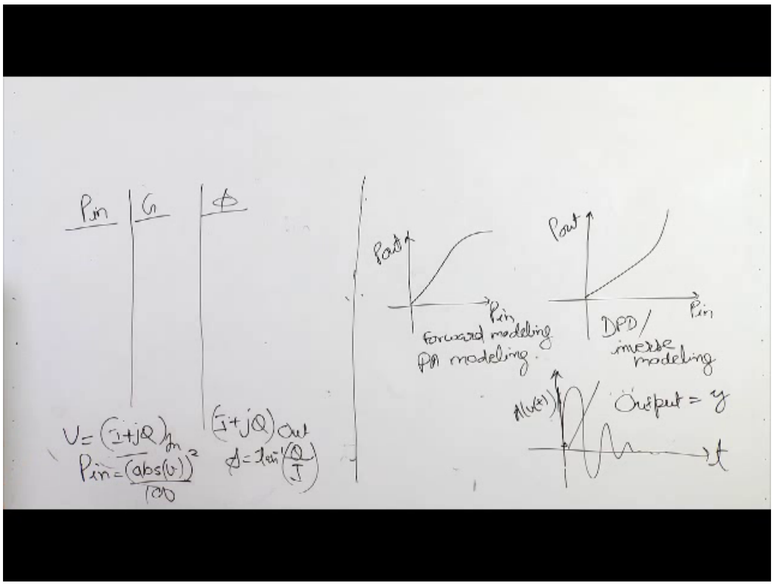

Whenever you have your data P in versus gain and phase if you have your input I plus jq and output I plus jq in and out of course, we can easily get this information P in can be the calculated from here. So, P in will be absolute of v whole square divided by 100. if you want to convert into dB then do the 10 log 10 of this plus 30 and you can calculate those values and put here.

Respective gain, you can calculate by comparing this with respect to this v 2 upon v one and then you can convert into dB and gain will be here. Similarly, phase 10 inverse q upon I gives your phase. So, phase of output minus phase of input you can put in here. So, these are LUT lookup table methods you can just see and a Particular value and see the gain and phase.

Similarly, if I want to do the mapping this respect to this then my phase will be not output by minus input it will be input phase minus output phase. And again, gain will be also opposite it will be input divided by output and that will be LUT for the inverse model. So, LUT form is something which gives us the modeling performance. So, there are different models and in the next lecture we will continue this discussion with different kind of models. So, keeping this in mind these all models can be used here, we will come back and discuss. This setup again with low cost setup ah. So, that it can be used easily in the laboratory environment.

Thank you.