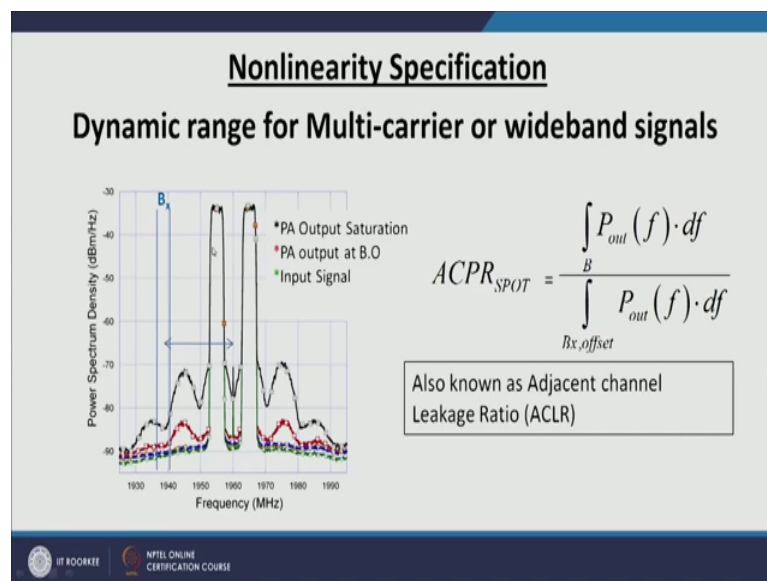


Basics of software-defined radios & practical applications
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Lecture - 11
Power Amplifiers: Nonlinear Distortion in Transmitted Signals

Hello everyone, so in the lecture series of basics of software defined radios and practical applications, today we are covering power amplifiers which are the main source of the non-linear distortion in transmitted signals, till last lecture we have covered the nonlinearity distortion due to the single band and to turn single tone and total signals. So, we have discussed that in the modern scenario mostly our signals are multi carrier signals, so in that case we our criteria of judging the distortion changes a little bit.

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So, in this left-hand side figure, we can see 3 carrier WCDM signal, if you see it has a carrier frequency of 1.96 gigahertz, then it has bands which are 5 megahertz away from the center on the both side, and at the center frequency there is no carrier. So, it is basically 2 carrier signal and both the carriers are broadband broad band and they have the bandwidth of 5 megahertz almost. So, in this case we are showing the input signal, which is shown by the green signal and then when we drive this power amplifier to high power, then we have the power output saturation. So, the black color signal is the output of the power amplifier.

So, distortion is appearing in the in band as well as adjacent band in frequency domain it is easier to visualize the adjacent band distortion, and we can also see if we are driving our power amplifier at a out at a back off it means, we are giving less power which is shown by this red curve, and that blue, and the yellow curve also then the distortion goes down as we keep increasing the output back off of the power. So, in this case when we are using multi carrier wide band signals, our distortion is not measured by signal to noise ratio, but similar kind of metric is used for that which is called adjacent channel power ratio. So, adjacent channel power ratio is the ratio of the powers in the in-band signal and the adjacent channel frequency.

So, it is also called adjacent channel leakage ratio, if you see here that B is the band, where we are having the signal. So, we can choose either this carrier or we can take this carrier, this carrier and take the average of these 2 carriers. So, we are doing the integration of all this band. So, average power of this signal is lying here, and when we do the integration over whole band then we get the power, which is inside this complete signal similarly at the adjacent channel with choose a offset which is given by the this blue arrow and it is called the power offset and normally whenever, we are giving ACPR, which is a particular value at a particular spot, then we always define this offset that add this offset we are measuring this ACPR.

At this particular offset, around that offset we take a particular bandwidth which is represented by B_x , here and then we integrate the power in the channel for that particular band. So, it will be the ratio of the power in this band with respect to the ratio of the power in this particular band. So, as you can see this black one will have a higher noise so it will have lower ACPR the red one will have much higher ACPR, and the blue and the green they will have will be having much more higher ACPR here. So, this was the ACPR of the leakage ratio on at a particular spot.

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Nonlinearity Specification
Adjacent Channel Power Ratio

Dynamic range for Multi-carrier or wideband signals

Total ACPR ($ACPR_{TOT}$) is the ratio between the total output power in the signal bandwidth and the total output power in adjacent channels:

$$ACPR_{TOT} = \frac{P_{in\ band}}{P_{adjacent\ -channels}}$$
$$= \frac{\int_B P_{out}(f) \cdot df}{\int_{LS} P_{out}(f) \cdot df + \int_{US} P_{out}(f) \cdot df}$$

For a particular single sideband:

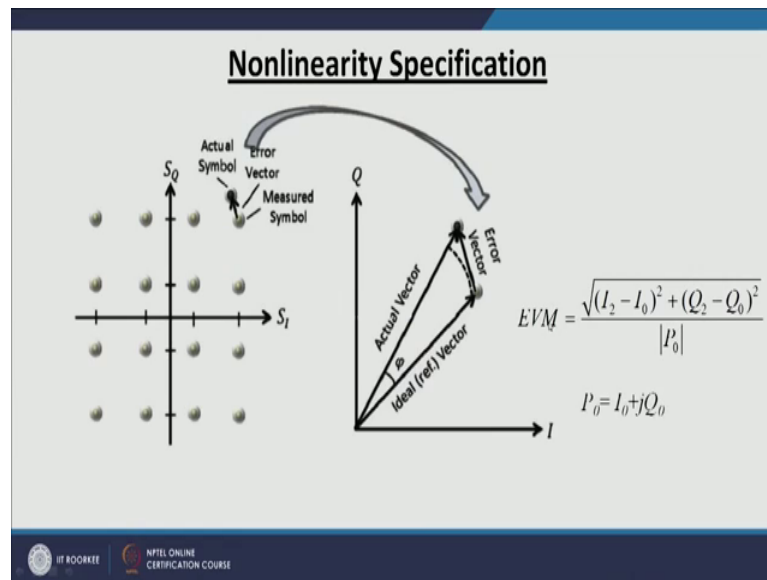
$$ACPR_{LS} = \frac{\int_B P_{out}(f) \cdot df}{\int_{LS} P_{out}(f) \cdot df}$$
$$ACPR_{RS} = \frac{\int_B P_{out}(f) \cdot df}{\int_{RS} P_{out}(f) \cdot df}$$

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Now, when we are dealing with the dynamic range of the multi carrier of wideband signals, you will see in the literature people talking about the total ACPR, when we are talking about total ACPR then it includes both the adjacent channels which means, you will have a low side or upper side. So, in this case lower side will be this below the carrier frequency upper side will be above the carrier frequency this direction. So, we capture our power at this location as well as at the upper side location and then we add them together and we show it with respect our in-band power the output power at this particular band.

Similarly, sometimes we are interested in showing our ACPR power at particular sideband, maybe lower side or upper side because they are interfering with the particular channel. So, we must understand this thing that whenever we are using power amplifier and these are the located frequency to us, wherever power amplify is working the nearby frequencies might be allocated to some other subscriber in that case if this distortion components are appearing there then they will interfere with the signal which that particular subscriber want to send. So, of course, they are undesirable and we have to control them.

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Now, another nonlinearity specification due to power amplifier is EVM it is called error vector magnitude. So, it is represented in terms of the deviation in the constellation diagram. So, if we look at this 16-point signal in the constellation diagram form we have our I phase and the Q phase, and this is the original location of the symbol, but because of the distortion the symbol is not lying at its original location and after demodulation it goes to some other point. So, this is the actual symbol and this is the major symbol or ideal symbol I would say. So, the distance between these 2-idea symbol and the actual symbol this is what we called error vector.

So, this error vector is what we measure for all the symbols and then we take root mean square for all those values and it is called error vector magnitude. So, in this formulation our I_2 is a particular position of the actual symbol and this I_1 is denoting our actual symbol. So, $I_2 - I_1$ and $Q_2 - Q_1$ they are giving their distortion in the I direction and the Q direction and we calculate the distance between these 2 error forms, and then we take root of this value and we normalize with the actual amplitude absolute of that symbol which was originally this one.

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Nonlinearity Specification

EVM calculation for OFDM based signals:

$$EVM_{rms} = \frac{\sum_{j=1}^{N_f} \left\{ \sum_{k=1}^{n_c} \left[|I(i,j,k) - I_o(i,j,k)|^2 + |Q(i,j,k) - Q_o(i,j,k)|^2 \right] \right\}}{N_f n_c I_p P_o}$$

all the frames (N_f) and all packets (L_p) in each frame, and all the symbols (example total data and pilots carriers in each symbol are n_c) in each packet. It is averaged to obtain rms value of the EVM as shown in the EVM_{RMS} equation.

Also known as "Receiver constellation error"

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So, this is the definition and how do we apply it. So, EVM calculation for web-based signals is given by this formula in this formula L_p , is representing the packets number of packets this n_c is number of carriers on which we have mapped all our symbols. So, if we are covering all these carriers then our all of the symbols are being taken care of, now one frame contains this many packets and if we have enough frames then we will have so many $I n q$ position in the constellation diagram for all this data.

So, this is for a particular O d m based signal, we calculate for each carrier then how many carriers were there originally in one packet, we repeat that for all the packets and then how many packets were there in a particular frame, we take the root mean square of all that value and then we normalize it, with all number of frames to have one particular value in the r m s, now in some of the literature if you come across the term receiver constellation error it also denotes to this particular definition it is again the EVM.

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Error Vector Magnitude (EVM)

The error vector magnitude (EVM) is a metric that measures the modulation and demodulation accuracy. It is defined as:

$$EVM = \sqrt{\frac{\sum_{n=1}^N |y_i(n) - y(n)|^2}{\sum_{n=1}^N |y_i(n)|^2}} \quad NMSF = \frac{\sum_{n=1}^N |y_i(n) - y(n)|^2}{\sum_{n=1}^N |y_i(n)|^2}$$

$y(t)$ is the captured signal, and $y_i(t)$ is the ideal signal:

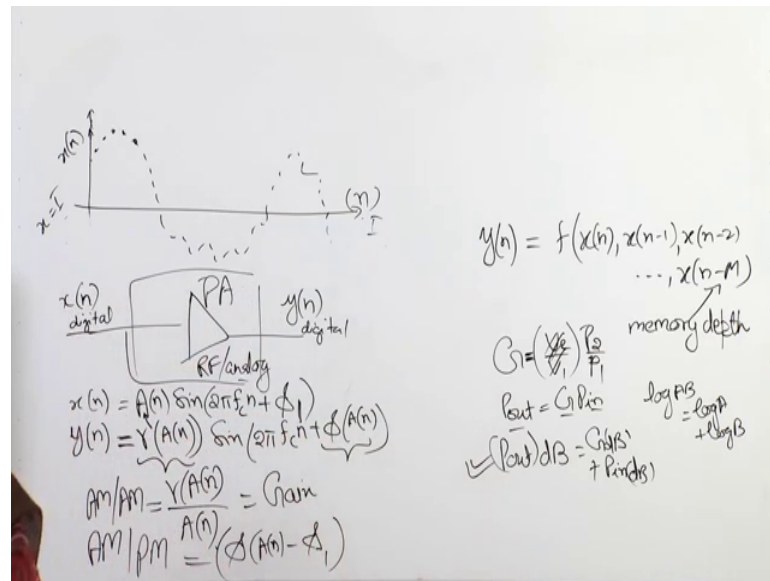
$$SNR_{dB} = 10 \log_{10} \left(\frac{1}{EVM^2} \right) \quad SNR_{dB} = 10 \log_{10} \left(\frac{1}{NMSF} \right)$$



Now, if I multiply this with 100 we can give the percentage EVM, which is achieved by multiplying this EVM r m s by 100 directly, so that was in the constellation domain once we have our signal in digital domain in the baseband domain, but it is not is the in the constellation form, some of the literature they also give the definition of EVM in this term where $y_i(n)$ is the original signal, and I_n is the original symbol which we transmitted and y_n is the symbol which we actually finally, received. So, we take the error between these 2 symbols and take the absolute of that and then we square it some all the data which we have, and then we take the root mean square of it is normalized value.

So, we sum all the data which we actually transmitted and we sum S square of all the errors and this is also denoted as EVM in some of the literature, now there is a term animacy the full form is normalized me is square error, it is simply the squaring of the EVM. So, basically when we are dealing with the baseband data before we demodulated into constellation form in the baseband, we have a data in the discrete time domain. So, after ADC, we have our data which will look something like that for the I as well as q. So, we have both of these data and there in the discrete domain in the n domain and x can be I or q.

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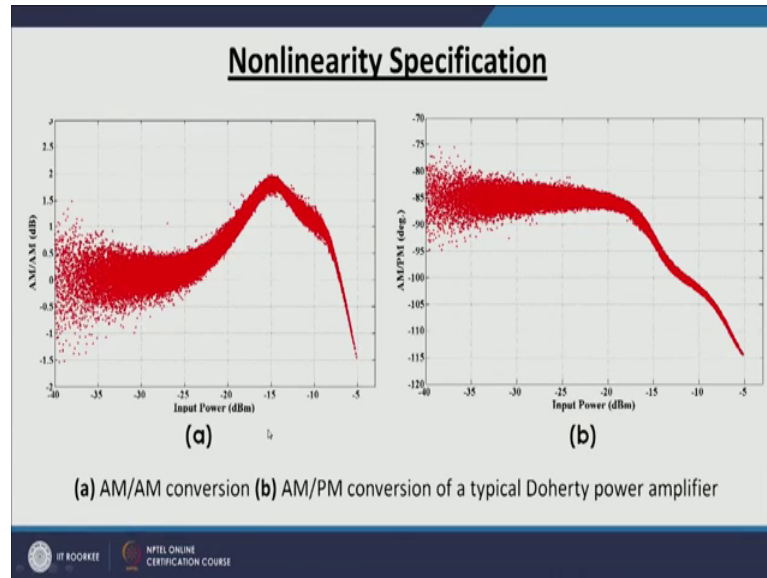


So, then we can actually measure this value, we can actually read these values we have availability to this value just, before we do the route traced cosine filtering again just before that we captured this data it is in a courtesously increasing decreasing form and then, we do the I n plus J q n in the digital domain. So, the data which we transmitted was this one or y I ideal y and data, which we received actual y is I dash n plus J q dash n and we are comparing these 2 values. So, basically, we are just normalizing the error between that and we use this value a lot especially, when we are talking about the power amplifier and transmitter output.

Now, what is the relation between these values with the SNR, this SNR should be actually dynamic range and the relation is given s $10 \log_{10} 1$ upon EVM square. So, of course, if we look at the definition of NMSE, we are dividing the power the summation of the symbol power by the error of this term. So, of course, if we invert this and take $10 \log_{10}$ we will get signal to error ratio. So, if that distortion power is very high in this case then the distortion power was very high, and we are getting this kind of components it is over adding all the noises because, of the phase noise quantization noise temperature noise and then it will be the defining factor here it will give you the ACPR or the dynamic range of the signal at the P output.

So, therefore, our SNR which should be actually ACPR the dynamic range is the inverse of this NMSE, and if you take $10 \log_{10}$ we get the dB of this particular value, now another one of the nonlinearity specification is given in terms of.

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AM/AM conversion and AM/PM conversion of the power amplifier. So, what is AM/AM an AM/PM conversion. So, whenever we are applying any data to PA and there is capturing some data I am showing n discrete domain digital domain basically, but power amplifier works in RF domain right and after that mostly RF analog domain.

But, what I want to represent here is the data which we captured in the digital domain eventually. So, if input data was $x[n]$ which has the amplitude $A[n]$ sign, if this was the input signal then our output signal at the power amplifier is not exactly the replica of the input signal, right? It becomes it has a amplitude which is a function of the amplitude of the input signal and it has a phase which is also a function of the amplitude of the input signal. So, when you want to observe the performance of any power amplifier, we want to have a look at these 2 factors because these are the 2 factors which are differentiating it with respect to the input signal.

So, of course, it will have a gain, but it will also have some kind of distortion. So, ideally when you are sending input signal the output signal gain should be a straight line, why, because for a particular input you should have a particular amplified output. So, when we have a look at this first diagram which is showing the AM/AM or amplitude to amplitude

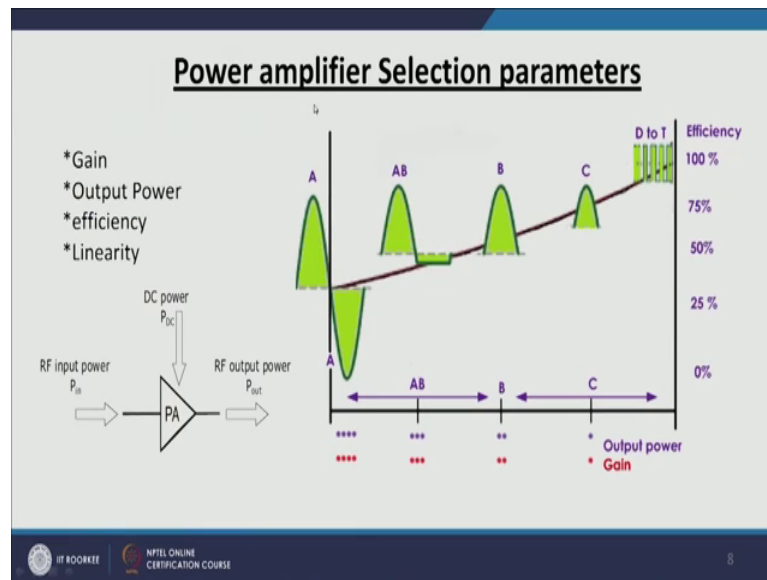
modulation, then we are increasing the input power and we can see that amplitude to amplitude modulation, which is basically simply $r A_n$ upon A_n this is what we call AM/AM modulation.

If you look at this is what is this is basically the amplitude of the second waveform divided by amplitude of the first waveform amplitude of the output divided by amplitude of the input, and by definition it is what simply gain. So, this AM/AM whenever you are talking about AM/AM it is simply the gain of the power amplifier. So, when we look at this diagram, the gain was straight when the input power was very small. So, this was the linear region for a power amplifier, after that when you keep increasing the input power the output power becomes non-linear, it becomes saturated in this particular case it is a power amplifier which has this particular nonlinearity profile. So, instead of giving a straight line the gain it is having this kind of non-linear profile after let us say input power of minus 25 dBm.

Similarly, AM/PM is the phase modulation because of the amplitude. So, here we are basically, simply plotting the output the phase minus input phase with respect to input power. So, AM/AM was the gain and AM/PM is basically, this ϕA_n minus input phase. So, for example, we would have initial phase like, this then we will say this is the AM/PM and we plot it with respect to input power, and that is how it is called with respect to the amplitude of the input signal.

So, if you look at in this diagram also we can see this phase is almost constant till minus 20 dBm, after that there is a great fluctuation and nullity comes into picture and it becomes compressed, now this profile of the gain and phase distortion and modulations they are typical of any particular power amplifier. So, the figure here shown is a for a particular doherty power amplifier and for different devices different classes of power amplifier we will have different kind of profiles. So, what are those classes of the power amplifier basically.

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Whenever, we are selecting any power amplifier we look for 4 things gain the output power V can it can provide efficiency of the power amplifier and linearity, it can provide means after how much input power, it will become saturated if it can become saturated after a larger distance of the larger input power then it is good for us because it will have lower distortion, now typically a power amplifier takes RF input power, which is the signal which you want to amplify apart from that it takes DC power from some other, source some DC supply is always connected there and by using the power of this DC supply, it provides the power to this RF input power and it is having this high output power, which is called RF output power in this range of the frequency which is the RF frequency range.

Now, if you look at this diagram we are showing different classes of power amplifier class amplifier is considered most linear amplifier, if we see it has very high gain and it has good output power, but it has very small percentage of efficiency. So, it has very small efficiency it converts very small portion of this power to output power. So, it requires lots of DC power, now if we do the changeover conduction angle we reach class AB class AB power has much higher efficiency as compared to class A, but because of that we have to choose between A and B, then our output power and gain are less than the class A amplifier class B amplifier, becomes even more non-linear there and it has lower gain and output power is compared to class AB efficiency is higher as compared to the class AB.

Now, class C amplifier is having the higher efficiency with respect to B, but it has lower output power because of its profile and it has lower gain, now this kind of amplifiers are called continuous mode amplifier because you are saying that, they have the conduction angle and they send the original signal into some form or other now apart from this continuous mode power amplifiers, there are some other amplifiers which we call switch mode amplifiers. So, they send the data in the form of switching operation. So, the distortion caused by this kind of amplifiers are separately and they require separate kind of linearization methods. So, these 2 categories are there and we are mostly focusing on these for the time being and class B and class AB are the mostly used power amplifiers in a transmitter.

Now, once we have the idea of the classes based on our requirement we can choose any of these amplifiers and use them, now let us have a look at some of the basic definitions which are relevant to the power amplifier for example, whenever we are calculating the signal power at the input and output we will be provided our V and I.

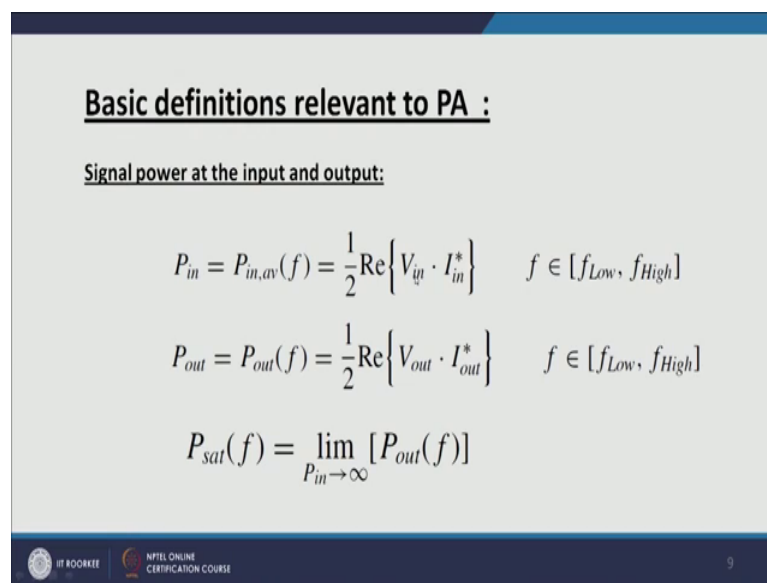
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Basic definitions relevant to PA :

Signal power at the input and output:

$$P_{in} = P_{in,av}(f) = \frac{1}{2} \operatorname{Re} \left\{ V_{in} \cdot I_{in}^* \right\} \quad f \in [f_{Low}, f_{High}]$$

$$P_{out} = P_{out}(f) = \frac{1}{2} \operatorname{Re} \left\{ V_{out} \cdot I_{out}^* \right\} \quad f \in [f_{Low}, f_{High}]$$

$$P_{sat}(f) = \lim_{P_{in} \rightarrow \infty} [P_{out}(f)]$$




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If we know the exact value in the complex form of that V and I, then we can do the multiplication of V input and conjugate of the I, input in the data sheets you will be provided r m s values in that case, we do not need this half value we can simply multiply them, because it will become V upon root 2 into I upon root 2 and V upon root 2 is the r m s value. So, you can directly use that value.

Now, when we are writing that frequency is in the range of f low to f high, then we are just giving the range of the in-band power. So, most of the power is concentrated in the in band the distortion components mostly go out there right. So, in case of the input signal mostly all the power is in the band in the output power, if we do not have any leakage into the adjacent channels in that case the similar kind of calculation can be applied for the output power calculation, and it will devote our band of interest, but in some of the cases some of this power is not part of that band and we have to take that into account.

Now, apart from this input power and output power there is something called saturation power. So, what is the saturation power, if we keep increasing the power of the input signal, then our output power will become saturated this is what we had seen. So, that power where this saturation starts it called the saturation power and of course, it then cannot go to the infinity at some particular input power it will simply burn. So, this is the power after which it becomes start losing it is linearity. So, there is some definition for that which we will review here.

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power levels are expressed in decibels over 1 mW, i.e. in dBm:

$$P_{dBm} = 10 \cdot \log_{10} \left(\frac{P}{1 \text{ mW}} \right) = 10 \cdot \log_{10}(P_{mW}) = 10 \cdot \log_{10}(P_W) + 30$$

$$P_{mW} = 10^{\frac{P_{dBm}}{10}}$$

Power gain $G(f) = \frac{P_{out}(f)}{P_{in}(f)}$

Small Signal Gain $G_L(f) = \lim_{P_{in} \rightarrow 0} [G(f)]$

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Now the calculations of the power, normally when you will see any datasheet commercial components you want to purchase, then they can give you the value in the watts as well as they can give you value in the dBm.

So, if your power in the watts is given by capital P, then if you convert it back into milliwatt first, and then you calculate the it is logarithmic power then it is called dBm power. So, this is the calculation and we have covered it and use it before also that 10 log 10 of the original power plus 30 will give you the dBm power or if you already have the dB power then dB power plus 30 will give you the dBm power, similarly in the milliwatt you can calculate your power by using the this calculation and 10 to the power P dBm divided by 10 will give you P milliwatt power.

So, this definition you have to sometimes many times. In fact, because once you want to convert back from power to voltage you always have to go to this milliwatt of what calculation. So, that you can apply V square upon 2 r formula and get your voltage levels mostly. So, power gain power gain as I am showing in this.

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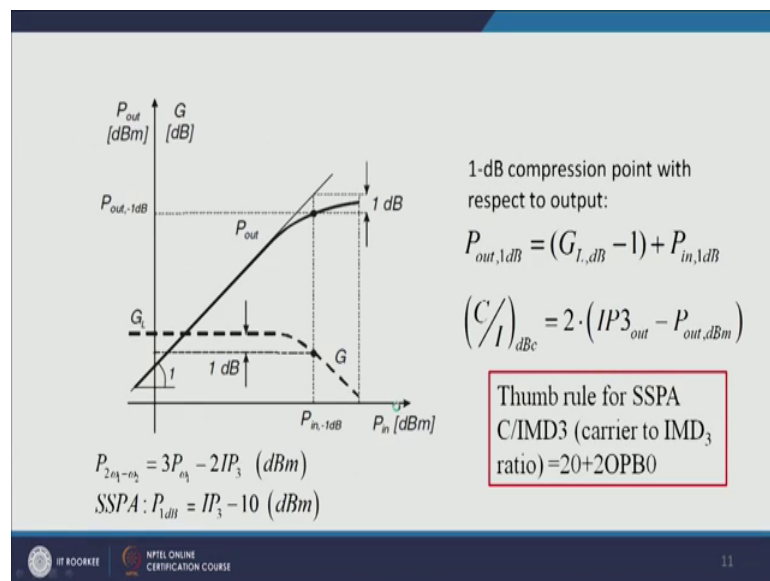


Diagram is similar to your our AM/AM diagram, which I had shown earlier it is the ratio of output power divide by input power in the bed of interest, what is a small signal gain this term is also very common to appear in the literature small signal is the gain, when your P input is 0, again we are showing limit of the input power going to 0 normally we have very small level power, it does not go to 0 because we have noise in the system.

So, as we can see in this diagram, the system was linear for some time and after that it becomes non-linear. So, this range for which it is linear, it is the this gain here is the small signal gain. So, in this case we can see that a very small signal gain is almost 0 for

this diagram, and this scatter point around this they are because of the 2 factors one of them is noise, which appears a lot in the low input power level under the point which we come across in the power amplifier is memory effect. Memory effect comes into picture because of the heating of the devices, moreover we the energy storing components in the power amplifier like inductor and capacitors sometimes to block the DC, we have to use capacitor this kind of elements they store the energy and then they relieve according to their nature according to their time constant.

So, when that happens then the output signal is not only the function of input signal, but the dissipated signal from the previous cycle is also get it mixed. So, y_n which we are giving as a function of x_n is not exactly, x_n it is a function of x_{n-1} , x_{n-2} and so on. This M is called memory depth so by making this relation by choosing the particular memory depth, we are able to find a proper relation between input and output and this diagram shows the data of the practical signal. So, we are able to see this memory affect a lot, and instead of seeing one straight line showing this graph, we are seeing group of data's appearing around that normalize value.

So, if we take the average of all this data and then we plot them, then we will achieve our memory less profile of the power amplifier non amplifier profile, now continuing with our definitions earlier we have covered 1dB compression point, 1dB compression point is the point or input power for which the output power deviates from its original path by 1dB, now because we are talking about the power amplifier we are mostly interested in the output not the input. So, we refer everything at the output of the power amplifier. So, whenever you are reading data sheets most of the time you will see the output of everything P1 dB output power.

So, how do we calculate this output power of the P1 dB, now normally we have this relation between gain and the power. So, gain is basically V_2 upon V_1 , right? And in terms of V_2 upon V_1 it will be actually P_2 upon P_1 , in the sense of power we talk whenever we are talking about the amplifier. So, when this happens we have our gain there, then we can say that we want to make a relation between input and output then P_2 or P_{out} will be G times P_{input} , if it is a linear either simply watts and the gain or if they are in dB then because logarithmic expression it is $\log A$ plus $\log B$.

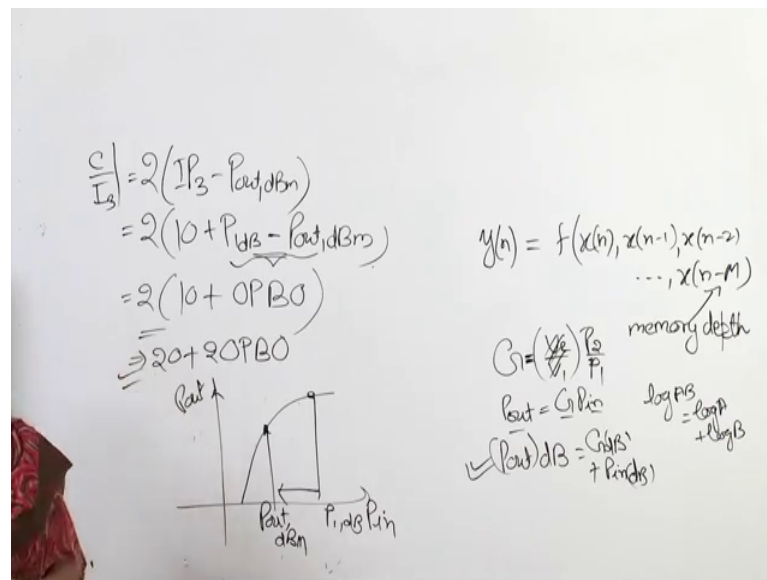
So, we can simply give it in terms of G_{dB} plus P_{in} dB. So, both are logarithmic and they are in the summation. So, when we are dealing the logarithmic scale the output power is simply the gain plus input power, now it might seem to us that we can simply get our output point of the 1 dB compression by adding the gain with the input power of the 1dB, but that will not be true because our data has been compressed by 1dB. So, the relation becomes output power is equal to input power plus the gain at the small signal scale minus 1, Now this small signal scale is the gain which was proposed for the linear range. So, that is why we are deviating from that gain by 1 dB, and we applied that formula to calculate our output power at 1dB compression point.

So, this is the important point because normally, it seems that we can simply add the gain to the input power it will not be the case when we are calculating the 1 dB compression point, now let us have a discussion on the output power back off, we have discussed that when we keep increasing our input power verse amplifier, becomes saturated if we drive our amplifier at some distance from some power previous to the input power, then we have a chance of getting lesser distortion this is what we called output power back off and I had shown you the figure, when we can see the distortion level was less. So, it is interesting to note the C by IMD, this is 3rd order intermodulation for a power at the some particular output power back off.

So, if you remember our original discussion from the last lecture the C by I ratio in dBc is actually twice of $IP3$ out minus P out dBm, and you also remember that our 3rd order intercept point and the signal power has this relation with the I_n d term IMD term and we have gotten this expression for the 2-tone case. So, that is why we are showing it $2\omega_1 - \omega_2$ it is the IMD 3 component again. So, this IMD 3 component is equivalent to C by IMD term here. So, IMD component can be given by 3 times the input signal power at the main frequency one is twice of $IP3$ third order intercept point.

So, from that we can actually calculate our C by I, because C by I is given by this formula then what we can do we can eliminate this $IP3$ from this formula. So, what will become of the formula?

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So, C by I which is IMD 3 basically, it is given by twice off IP third order intercept IP 3 minus your output power in dBm. So, this was the original expression are given there, now if we put our value of SSPA calculation we have given a rule of thumb that P1 dB will be equal to IP 3 minus 10.

So, basically, we can say IP 3 will be 10 plus P1 dB. So, now if you look at this point this P1 dB is the point, where saturation starts and saturation has already occurred because of this the power has been dropped by 1dB. So, we are talking about the current power which is subtracted from the saturation power. So, this is what we what will give us the output power back off. So, the distance is the output power back off from P1 dB point. So, it will become 20 plus 2 output power back off.

So, if you understand this point if it was the input power and it was the saturation curve for the amplifier and first let us say P1dB point here, and we have subtracted OPBO from here and we reached our original power. So, this is what we are representing here output power back off, and because of that what will be our IMD the carrier to IMD 3 ratio is given by this formula which is 20 plus 2 times OPBO.

So, if we know the output power back off, we can simply have some estimate of the character 2 third order inter mode by using this formula. So, in the next lecture we will continue with this specification and we will see how we can actually apply a commercial datasheet to choose any particular power amplifier for our applications.

Thank you.