Design Principles of RF and Microwave Filters and Amplifiers

Prof. Amitabha Bhattacharya
Department of Electronics and EC Engineering
Indian Institute of Technology – Kharagpur

Module No # 4
Lecture No # 18
Quantitative Characterization of Nonlinearity for Large Signal Amplifier(Contd,)

Welcome to this lecture we have already seen the RF amplifier design for conjugate maximum gain specified gain then low noise amplifier design then broad band amplifier design. Now we will another very important part because you know in any transmitter finally before sending the RF signal to free space. We have a power amplifier where we try to give as much power we can is possible to give power. So that is a power amplifier and that power amplifier generally has a very high signal level.

But over till now whatever amplifier design we have seen in their we assume small signal model but in power amplifier it is not small signal it is quite large amplitude of signal and you know that in power amplifiers many time operate it at class C etc. So generally, non-linearity may come in power amplifier so if non-linearity comes how do you go about in RF power amplifiers. So we will have to see power amplifier design.

Actually, amplifier design we have already seen but in power amplifier there are certain characteristics which are not we do not pay any attention to them in low small signal amplifiers, but in power amplifiers or large signal cases we need to pay attention to them because if we do not put that, if we do not restrict ourselves to within those constants then what happens due to non-linearity. We generate some other spurious frequencies etc also saturate the amplifier.

So its gain etc that falls so it is essential to understand this power amplifier non linearity so that will see. In this first lecture there will be the next two three lecture will be on this. So first one will be power amplifier design first we will discuss about the non-linearity at large signal Values. I already said that at power amplifier which is the final stage of before antenna the final stage of any wireless transmitter they are this due to large signal the non-linearity may come.

(Refer Slide Time: 03:13)

Linearity of Amplifier

At small signal levels linearity maintained

Due to interconnects, joints etc., at large signal values, nonlinearity appears

How to characterise non-linearity of Amplifiers ?

So we let us see those non-linearity so at small signal levels linearity will maintain. But due to interconnect joint etc at large signal values non linearity appears. How to characterize the non-linearity of any amplifier.

(Refer Slide Time: 03:32)

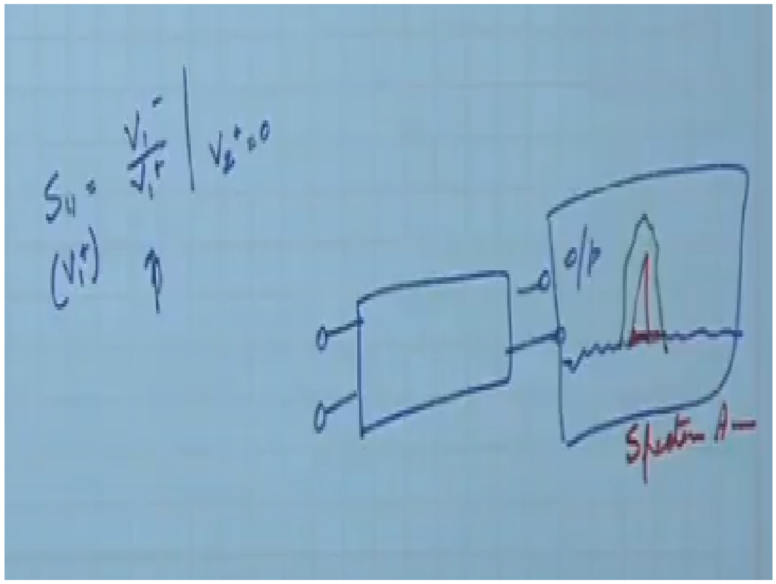
Absurdity of S-Parameters

- S parameters for small signal do not depend on power level
- For large signal, due to nonlinearity, s-parameters become level dependent
- So, S-parameters cannot characterise power amplifiers.

Now you see that S parameters characterization for large signal is an absurdity why because S parameters for large signal do not depend on power label there we have seen that any voltage any current a reflector incident that ratio that we take.

But for larger signal due to non-linearity S parameters also becomes level dependent. We have already seen their frequency dependent but now that time also you said that they are bias dependent at large signal they are highly on signal dependent.

(Refer Slide Time: 04:19)



So if it becomes as signal dependent then suppose am writing that S11 is equal to V1 by V1 + when I have the V2 + is 0. Now if this ratio is again a function of this is the ratio but if it is a function of let us say V1 + then it is very difficult to have this parameter because for every level then i will have to define this that is why for the large signal people do not use S parameters to characterize power amplifiers or characterize any block which is using any large signal levels.

(Refer Slide Time: 05:00)

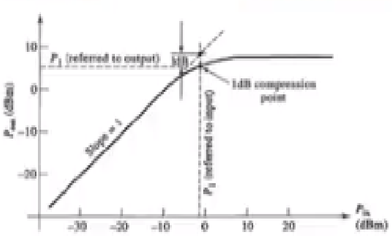
Non-linearity Figures of Merit

- 1 dB gain compression point
- Dynamic Range
- Spurious Free Dynamic Range
- 3rd Order Intercept Point

So what will do will see there are various figures of merit parameters for characterizing non linearity one them is 1 db gain compression point another is called dynamic range, another is called spurious free dynamic range, another is called third order intercept point. So this 4 are very important characteristic that is why we will see that power amplifiers generally they are specified in terms of this not in terms of our usual S parameters or other those gain and other values.

(Refer Slide Time: 05:43)

Gain Compression



Definition of the 1 dB compression point for a nonlinear amplifier.

1 dB gain compression Point

→ The input or output Power point

→ Gain has fallen 1 dB from linear small signal gain

$$G_{1dB}(dB) = G_0(dB) - 1$$

Now let us first see what is gain compression we know that in a linear thing if we plot the transfer characteristics the power transfer characteristic of any two port network particularly amplifier. So let us say that in the X axis we are putting input power in dbm in Y axis we are putting output power in dbm. Now in the linear zone it is slope of 1 we know we have a linear thing. So as we are increasing the power in the input output is following that after certain point to a C that it starts falling.

Now why its starts falling we will later explain but all systems are these that this it would have gone infinitely with the dotted line. But in all practical systems we see that this curve instead going like this linearly it starts this. So this is when it starts deviating from this normal. We can say that it become non-linear so that transfer is no more non-linear now where I will say that it as become non-linear here or here obviously when it is flat I know that it is a non-linear thing it is saturated.

But here where so people made this that we will not say that it is here we will say that here it has become clearly non-linear. This point is called 1 db compression point, so what is this definition now it can be in terms of input power or output power because this point if I say this point as an X coordinate as well as this point as Y coordinate that mean this can be reference to input power level or output power level now that is why am saying that input or output power point is1 db compression point.

If it is input side we call it 1 db gain compression point if it is output side we call it output 1 db compression point. Now what is the definition of this point because this point need to be precisely defined to say that ok I have already entered the non-linear region. Where this is the point where the gain as fallen 1 db from linear small signal gain. You see I was having some gain here because P out by P in was my power gain now that, would have been something in the linear case but here you see I have got it.

Please remember this is a db scale that is why we are writing slow one that does not mean the gain is not here. In the db scale if I have the slope of 1 that means it is linear but what it says that there is some gain. Now the linear gain is something but here it is fallen by 1db from that, you see that is why this is called gain as fallen 1 db from linear small signal gain. So in mathematical terms we can write that G at 1 db gain compression point that means here G.

In db scale is 1 db less than this points power gain so this point power gain is linear one so from that it has fallen to 1 db. So I know always this point that where it is and I check that whether this is 1 db because had I checked it here I would have seen that ok still this difference between the linear thing and actual thing that is not 1 db.

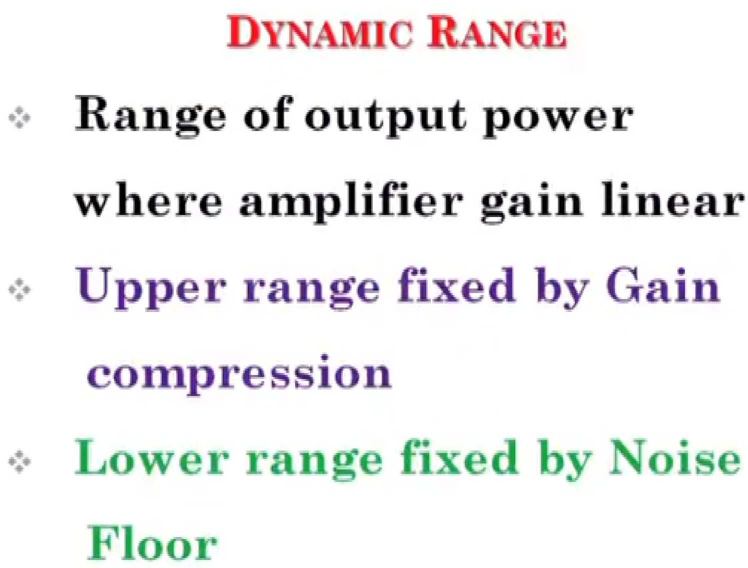
But the moment it becomes 1 db this point we characterize that ok in the actual graph this point represents the 1 db compression point, this points also represents 1 db compression point, this is the input 1 db compression point this is the output 1 db compression point. Now you can ask that why it is compression it could have been that non linearity could have been gain could have been increased yes it may increase also but we see that generally this is compression not the expansion, but theatrically it is possible and in some system it may expand also.

Generally we do not encounters the systems but obviously with newer applications one day sometimes we may see expansion also ok. So now if we see this that means now I can say that from here to here I can use the system. Because I want to have in amplifiers we need to have linear things so if the transfer characteristic is not linear then amplifier will have problem. So we need define a range of input or output power labels where the amplifier will be behaving as linear.

So we say that this is the onsite of non-linearity so from this 1 db gain compression point to maximum minimum usable level I will say range that is called dynamic range of any amplifier or any 2 port network now it is a very useful thing it is the range of output power here am saying in terms of out power dynamic range is also defined in terms of output power. So range of output power where amplifier range is linear.

Now upper is fixed by range compression we are not express to shown what is the lower range? Now lower range is fixed by noise floor. Even if you do not give any input to any 2 port network here suppose I have a 2 port network and I am not giving any input here but suppose output these output I have no input here. But output I have connected to any oscilloscope or spectrum analyzer or something a measuring device.

(Refer Slide Time: 12:15)



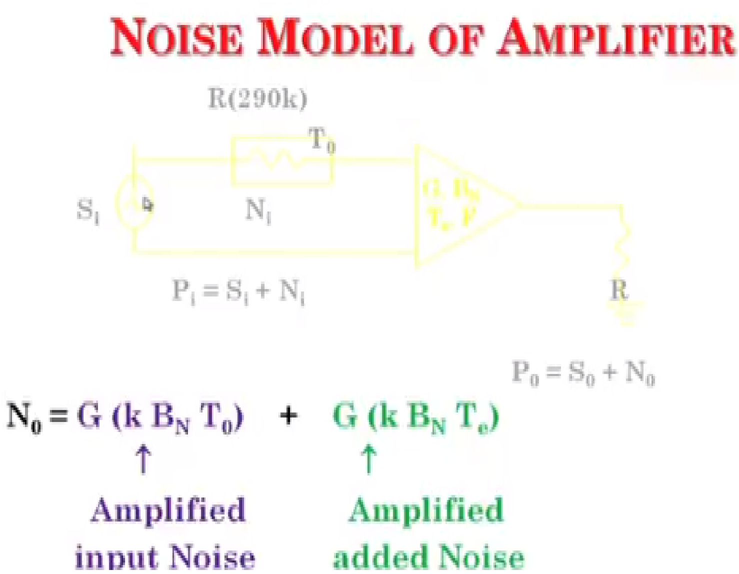
So there will see always without signal a also I have this if I have a signal then I get somewhere here suppose I am looking at a spectrum analyzer then you will see there is a signal but in other places where no signals are there. Still this what is this because there are actually noise is always present particularly block it is adding some noise that is the noise floor of the device in oscilloscope in multimeter spectrum analyzer everywhere you have noise in multi meter you have a visual display of that but in oscilloscope or spectrum analyzer you will see that there are noises.

Now that noise float so my signal should be sufficient care of that noise float. Actually several db up from that noise float so minimum usable range is always that noise float plus some extra db so that I can recognize at signal because if my signal is such that instead of this the signal was also this. I think you understand this red one that instead of this if I have signal like these I would be able to recognize as a signal even though it is as a signal.

So minimum level of input signal should be sufficiently above the noise float typically it is depending on what you prefer that you may call these as signal. I may say no unless and until it becomes like this. This green one I wont say so that green one is saying that his definition of signal is so much a bit higher db then the red ones thing but whatever you decide that ok this is signal.

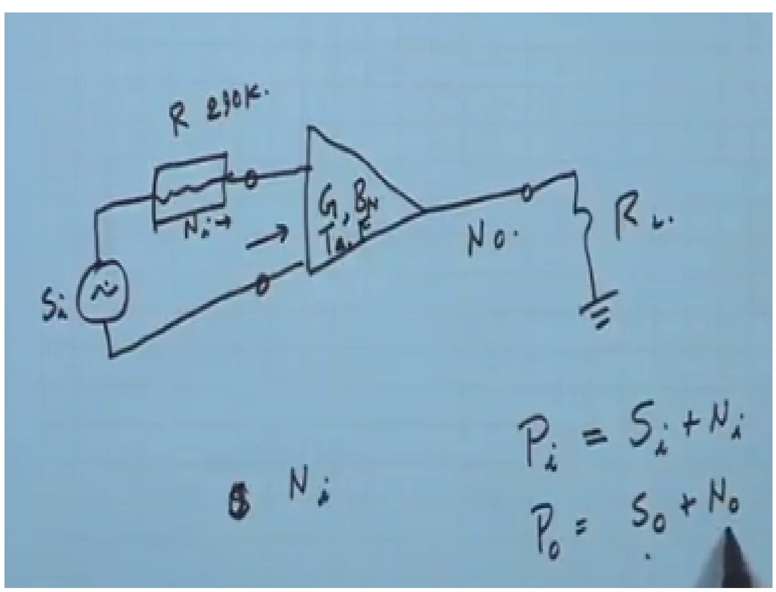
So that means there is above noise float you have something so lower range of the dynamic range is fixed by noise floor.

(Refer Slide Time: 15:00)



Now let us see the noise model of an amplifier you see I have a I think the picture is not visible.

(Refer Slide Time: 15:14)



So I have a amplifier, the amplifier has its gain also it has some noise equivalent bandwidth this terms I think you are familiar noise equivalent bandwidth it is not the actual bandwidth but that means a rectangular gain function that whatever bandwidth is (()) (15:39). Then it has some temperature and noise equivalent temperature and also it has the some noise figure.

Now when you have the input obviously what we do the source that will have some resistance and we assume that source is a pure thing it does not have resistance whatever source resistance we have whatever noise it produces because this will also produce noise that we in abstract we say the temperature here resistance here whose this temperature is at two ninety degree kelvin and the noise this whole signal part a source part that is producing that is giving NI I think.

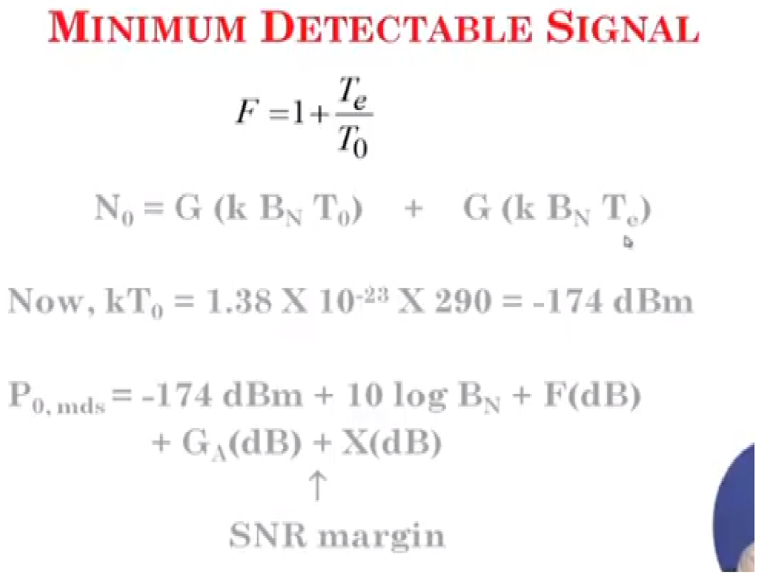
Then the amplifier will be adding noise now output of the amplifier is typically on a load it. So if I write what is the to this amplifier what is the input noise let us call this NI and output noise is No. So I can say that PI which is the input power to this amplifier that will comprise of one is the signal power another is the noise power.

Because I will get in the input side both of this generally they are additive so PI is SI+NI. Similarly at the output I will get PO which is SO which is the signal power though actually always it is added. So we cannot separately recognize it. But in mathematics we say that SO is the signal powers level and N0 is the noise powers level. You know this is a deterministic level, this is a random signal so but when they are mixed we cannot separately say but for the analysis we do like this.

Now obviously you can now look at that what is the value of this output noise power since this amplifier as gain G. So it will multiply this whatever this input noise power is coming that this G into NI. So that we are recognize NI we know that from this definition it is K B and T0 and also it will add its own power it that it is G into K BN T. That whatever amplifier added noise and this is amplifier input noise this should be amplifier added noise.

So that input noise it is multiplier with the gain and its own noise that also is generated here that also getting multiplied by its gain.

(Refer Slide Time: 19:18)



Now minimum deductible signal is what because ultimately we will have to deduct the signal. So what is the minimum deductible signal now this you know that noise figure and equivalent noise temperature they are related by this. So if you put this the N0 becomes this that we have already seen. Now KT0 K is the (()) (19:43) constant T0 is the two ninety degree generally in European or America or advanced countries they fix this standard as seventeen degree centigrade which is two ninety kelvin.

So this is the constant term so that turns out to be minus one seventy four dbm. So we can write that output minimum detectable signal that expression will be minus one seventy four dbm plus you see from that this PO it is SO + NO. So now minimum value is given by this that POmds is - 174 dbm + 10 log Bn + F is the noise figure + GA + X this is the SNR margin I was talking about noise floor (()) (20:47).

So this is the minimum detectable signal or this is the output side. So it is always given by this so knowing the your amplifiers noise equivalent band width knowing its noise figure knowing its gain and knowing that SNR margin you fixed is typically if nothing fixed take it either 3 or 4 db above the noise floor. So you can find the POmds once you have that then you can define the dynamic range.

(Refer Slide Time: 21:17)

DYNAMIC RANGE

$$DR = P_{0,1\,dB} - P_{0,mds}$$

You see dynamic range is now analytically what is it previously we have seen it graphically now we say that it is P output this is the output dynamic range. PO output from 1 dbm compression point minus POmds, we know how to calculate mds, we know how to we have seen in graph this later will find out ho to find this point then you can always find the dynamic range. What is the physical significance of dynamic range ?

That means within this zone if my output power level goes depending on what gain I have so I can have the corresponding input power level also. Input dynamic range so within that level if we keep my system, if I keep my amplifier it will be linear so all whatever we discussed before of designing a small signal based amplifier concept that design concept you can put. So you need know that in the power amplifier I will have this much dynamic range.

So I can barring my input level or output power level same thing so that within this range I am linear so all my amplifier design concept will tally. So this is the definition of the dynamic range ok now you see that there are if I have a single spot frequency then that dynamic range concept is sufficient. But as I said generally we have in bass band suppose when am talking this bass band the voice bass band that is from 20 hertz to 20 kilo hertz.

(Refer Slide Time: 22:54)

INTER MODULATION DISTORTION

- Voltage Transfer characteristic of PA

$$V_0(t) = a_1 v_i(t) + a_2 v_i^2(t) + a_3 v_i^3(t) + \ldots\ldots$$

$a_1 \rightarrow$ linear gain coefficient

- A tone will generate upto n^{th} order harmonic of w_0
- Whether harmonic content significant or not depends on a_2, a_3 etc

So I have various frequency components presents over a band also if I have a video signal I have typically 4 megahertz but if I want to play it or band width for that also I have various frequency components presence. Now let us see the transfer characteristic of any power amplifier device. So output voltage V0T that I can express as a power series because in general I will try to make it as linear but when I have as I said various interconnect, joints etc. so they are all not linear component so they are that linear means electronic means we always say super position principle.

So for them super position principle is not valid that means the additive property and scaling property. Those are not valid so there I can also have this components that the square of the input that Cube of the input etc with the proper coefficient. Now usually this A2, A3 they are the this components this coefficients of the non-linear components they are small so in small signal values you see if I have Vit small then this value is not much this value is further smaller.

So that is why we that time assume that it is always this but large signal cases I will see that we cannot neglect all of this. So A1 is called linear gain coefficient because this VOT is equal to A1 VIT then this is linear thing that is why it is linear coefficient. A tone will generate upto Nth order harmonic you see if I have a tone and if the system is non-linear. So this B is square T that will make the second harmonic come.

Then I have this Bi3 that means If A 3 is significant I will also have third harmonic also get generated because you know that if Vit is COS omega T it will generate a a COS square. COS square means there is a COS 2 omega term present similarly if this is present i will have COS cube. COS cube means I will have COS 3 omega T so we say that if we have non linearity like this. Let us say upto nth order then a tome will generate upto here we are saying nth order.

So if we have non-linearity upto N that I have a sizable thing VI to the power N then upto Nth order harmonic will get generated. Now whether content significant or not that depends obviously this coefficient A2, A3 etc this is called a second non-linearity component. This is called as third non-linearity component. If they are significant then we have significant harmonics otherwise not.

(Refer Slide Time: 26:44)

TWO TONE TEST

$$v_i = A_1\, Cos\, w_1 t + A_2\, Cos\, w_2 t$$

$$\begin{aligned} v_0(t) = {} & a_1 \left(A_1\, Cos\, w_1 t + A_2\, Cos\, w_2 t\right) \\ & + a_2 \left(A_1\, Cos\, w_1 t + A_2\, Cos\, w_2 t\right)^2 \\ & + a_3 \left(A_1\, Cos\, w_1 t + A_2\, Cos\, w_2 t\right)^3 \\ & + \ldots\ldots \end{aligned}$$

So to have that how much non linearity we have let us consider it 2 tone test. What is a 2 tones test? Though we have many frequencies let us assume that we have 2 frequencies 1 is omega 1 angler frequencies correspondingly F1 and F2 so we have this. With this is their coefficient that means in my input signal I have this capital A1is the coefficient for omega 1 component capital A1 is 2 omega component also later we will see in 2 tone test we assume that omega 1 and omega 2 they are different but they are not very much far apart.

So omega 1 and omega 2 they are nearby but to distinct component. Now assuming that nonlinear system what will be my output VO output will be A1into Vit so in plus Vit I can write this + A2 Vit square. Since Vi is this I can write this + A3 Vi cube so if more that is why dot. Now let us see what happens here so if you now make the terms that what is the DC term what is the omega 1 term there will be omega 2 term then there will be 2 omega 1 term there will be 3 omega 1 term etc.

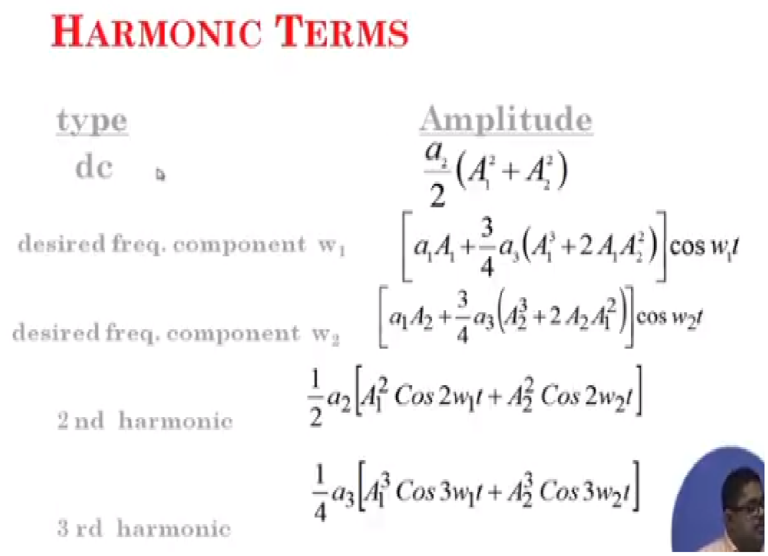
If you list them please do it at your home you have this please have it you will see that DC will have amplitude this. You see in the input there was no DC component our input was typically this you see there are no DC component with the non-linearity etc this DC as come there is a DC. Then desired frequency component omega 1 is this, omega 2 component this second harmonic 2 omega 1 is this, then third harmonic is this etc.

(Refer Slide Time: 29:16)

INTERMODULATION TERMS

type	Amplitude
2nd order IM	$a_2 A_1 A_2 \, [Cos(w_1 + w_2)t + Cos(w_1 - w_2)t]$
3rd order IM	$\frac{3}{4} a_3 A_1^2 A_2 \, Cos(2w_1 - w_2)t$
3rd order IM	$\frac{3}{4} a_3 A_2^2 A_1 \, Cos(2w_2 - w_1)t$
3rd order IM	$\frac{3}{4} a_3 A_1^2 A_2 \, Cos(2w_1 + w_2)t$
3rd order IM	$\frac{3}{4} a_3 A_2^2 A_1 \, Cos(w_1 + 2w_2)t$

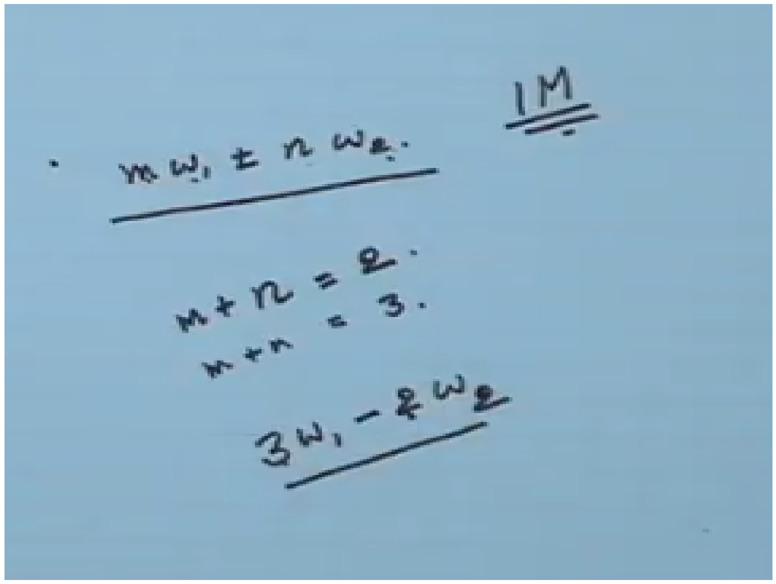
(Refer Slide Time: 29:17)



Also not only harmonics there will be some other term you see if i have this term let me see it. Now when I have this term present you see this A1 A2 etc now there will be if I have this then there will be some terms which will be also omega 1 + omega 2 or omega 1 – omega 2 or 2 omega 1 + omega 2, 1 omega 1 – omega 2, 2omega 1 – 3 omega 2 etc that means harmonics are those that omega 1.

So for omega 1 I have omega 2 sorry 2 omega 1, 3 omega 1, 4 omega that means integral multiple or (()) (30:19) half omega 1 or one third omega 1 or 1 forth omega 1 etc. But when I have 2 tone not only them there will be mixture of omega 1 and omega 2 that means I can write that. Now this is not an harmonic this is harmonic of omega 1 mixed with another harmonic of omega 2. This is called IM or inter modulation product so let us see this intermodulation terms. Now type what is second order IM now if this M whatever may be this plus minus thing but M + N if that M and N are integers M + N if it is 2 it is called second order IM if M + N = 3 it is called third order IM.

(Refer Slide Time: 30:38)



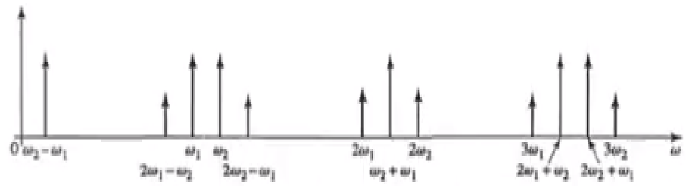
So you see that when i have omega 1 + omega 2 basically I have M is 1 and N is 1. So 1+1 2 this is also even though it is Cos omega 1 – omega 2 but that actually this coefficient is 1 this is also 1. So this second order IM then third order IM you see 2 omega 1 – omega 2 so 2+ 1 that is why it is third order IM. Here you see 2+1 that is why it is third order IM here you see 2, 1 so 3,1,2 so 3.

So all this are third order IM so that means frequency components 2 omega 1 – omega 2, 2 omega 2 – omega 1, 2 omega 1 + omega 2 omega 1+ 2 omega 2 all these are examples of third order IM . Similarly if I could have 3 omega 1 – 2 omega 1 now what is this? This is actually which IM it is it will be fifth order IM sorry this is omega 2 because this is 3 this is 2 so it is a fifth order intermodulation product. So here we have listed what are the amplitudes you can cross check you do it yourself all this comes that.

So amplitude will be actually I have written cost to show you the frequency part. Amplitude will be simply 2A1 A2 here 3,4 this. So you see this is interesting that this let us see the harmonics. In harmonics I have A1 etc A3 etc this is the desired components in harmonics you have A2 into this in third harmonics you have this.

(Refer Slide Time: 33:45)

SPECTRUM OF INTERMODULATION PRODUCTS



Output spectrum of second- and third-order two-tone intermodulation products, assuming $\omega_1 < \omega_2$.

Then intermodulation if you have this and if you see the spectrum of intermodulation product you see that my actual desired frequency is this omega 1 as well as omega 2 because this is my actual band now you see that DC is far away generally this omega 1 omega 2 are RF frequencies so this is far away. Omega 2 – omega 1 that means this is the second order intermodulation that is near this DC because omega1 and omega 2 are nearby.

Similarly, 2 omega 1 it is double of that so it is far from this tone. 2 omega 2 that is also far away omega 1 + omega 2 faraway, 3 omega 3 omega 2 this is third order intermodulation here this third order intermodulation but you see along with my desired zone this third order intermodulation product. Some of the intermodulation product which are bearing this minus sign 2 omega 1 – omega 2, 2 omega 2 – omega 1 these two are nearby.

Now what is the problem with this that typically what we do when we have this finally we put a filter and filter out with this portion but this portion as also will come nearby. So when I try to filter it is difficult to remove this two intermodulation products with this this third order intermodulation product with this you see second order intermodulation product is far away so I can remove it these third order intermodulation product this third harmonic they are further away.

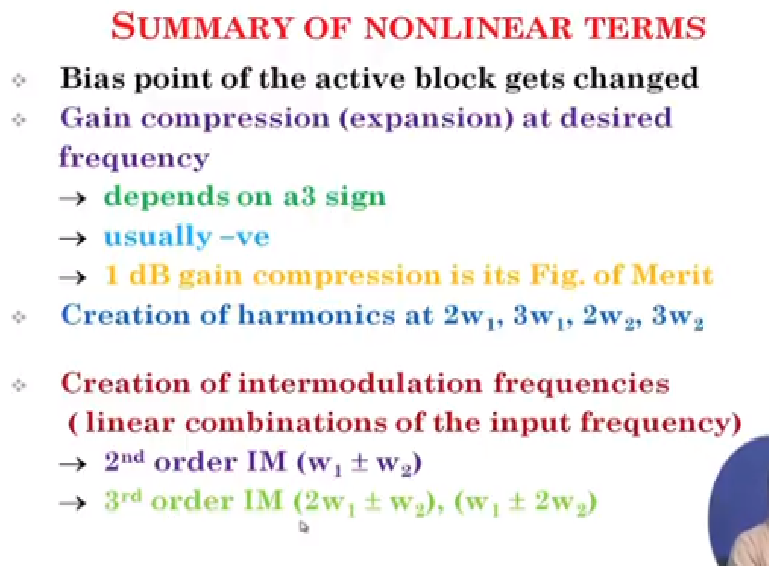
Fourth harmonic fourth order intermodulation product they will be further away, I can always filter that but this third order intermodulation product given by 2 omega 1 – omega 2. And 2 omega 2 – 1 they are in the same frequency band as my desired frequency band. So by filtering I cannot remove them that is the problem that they will come and they will disturb my system because this are not desired frequencies. So third order intermodulation product where from it came because of non-linearity it came.

If the system was linear these two are not generated now non-linearity also generated harmonics but by filtering I always remove them but this I cannot remove so they will disturb me that is why I will have to more clear when we have high large signal level. I have non linearity presence in small signal also non linearity presence but due to small signal level its value is much less compared to my desired signal.

So always even if its present it does not disturb but at high values they are comparable with my desired frequencies and by filtering I cannot remove the third order intermodulation. So third order intermodulation is dangerous I will have to characterize my system that third order intermodulation should not come because if it comes it will disturb me.

So it is at the low value of the input signal it is not significant but will see in some point high value its starts becoming significant and I should not take my amplifier into that region or that level where it comes. So this we have seen so summary of this non-linear terms this lecture will finish so we have seen the danger of third order intermodulation. We have seen that due to non-linear terms bias point of active blocks get changed. We did not have in the any input DC thing but the DC thing as changed.

(Refer Slide Time: 37:43)



So the bias point of active block also get changed so should be aware gain compression also I said expansion thought generally we do not see expansion at desired frequency depends on A2 sign this gain compression depends on A2 sign usually A3 is negative. Remember A3 is the component the by which we can characterize the nonlinear system A3 into Bit cube that A3 typically is negative that is why we get compression and 1 db gain compression is figure of merit of this A3 sign.

Creation of harmonics we have seen that due to non-linearity all the harmonic gets created. Creation of intermodulation frequencies linear combinations of input frequency we have seen second order IM we have seen third order IM and we have seen the danger of third order IM. So in the next lecture we should quantify this third order IM and then we will try find out that how to characterize our system so that this type of third order intermodulation product do not come. Thank You.