**CMOS RF INTEGRATED CIRCUITS**
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**Module - 12**
**RF Power Amplifiers**
**Lecture - 39**
**Class C, D, PWM Amplifiers, Linearization**

Welcome back to CMOS radiofrequency integrated circuits. So, we have been talking about power amplifiers. And, in the first lecture I thought we would cover class A B and C amplifiers, I did not I have just managed to cover class A amplifiers. In the second lecture; I thought I would cover class B C and D amplifiers I did not, I managed to cover B and mostly class C amplifiers.

We actually; just studied class C amplifiers as a generalization of class A class B and class A B amplifiers, right. In the previous lecture what we studied was the class C amplifier; it is conducting during a small period of time out of the entire cycle. I have arranged the bias voltage V bias, such that; only during a small angle of time, small portion of the period, only during very small portion of the period the transistor is actually on, the remaining time the transistor is off.

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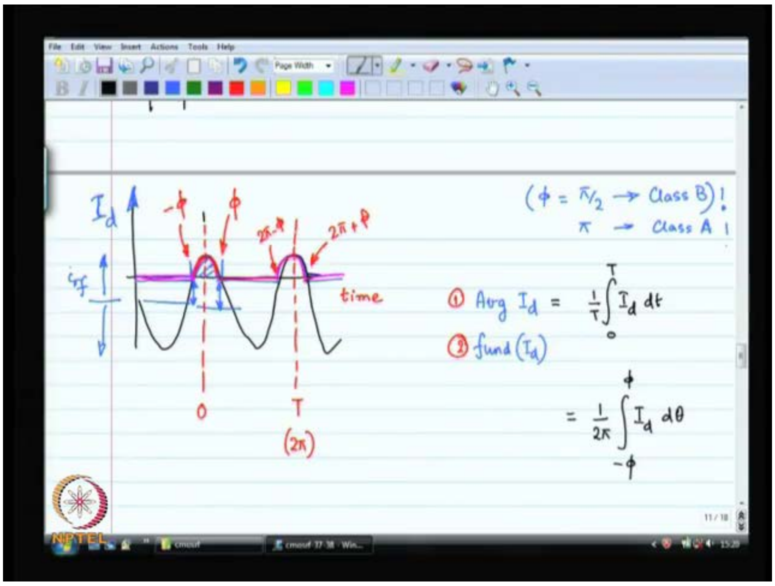

So, this is the general idea, right. Now, first of all this was the profile of the current going through the drain then, we figure out what is the average current going through the drain, what is the fundamental of the current going through the drain. Now, the RF choke allows the average current to go through everything else is rejected, which kind of tells me what is the power consumption of the circuit is. The fundamental component cannot go through the RF choke; it also cannot come from the tank because the tank looks like the infinite impedance at the desired frequency. So, therefore; the fundamental component is completely delivered to the load and therefore, I can figure out what is the power delivered to the load. So, that is basically what we did in the previous lecture.

And, we had 2 expressions; 1 for the power delivered to the load best case power delivered to the load. And, 1 an estimate of what is the power consumed from the power supply. So, with these 2 power estimates we figured out that is the efficiency, alright. Now, we saw that the classy amplifier as you decrease the conduction angle, the efficiency becomes more and more and things look very exciting, but the tragedy is you that it is you are not really delivering power to the load. As you are improving your efficiency, at the same time you are reducing the total power that you are delivering to the load.

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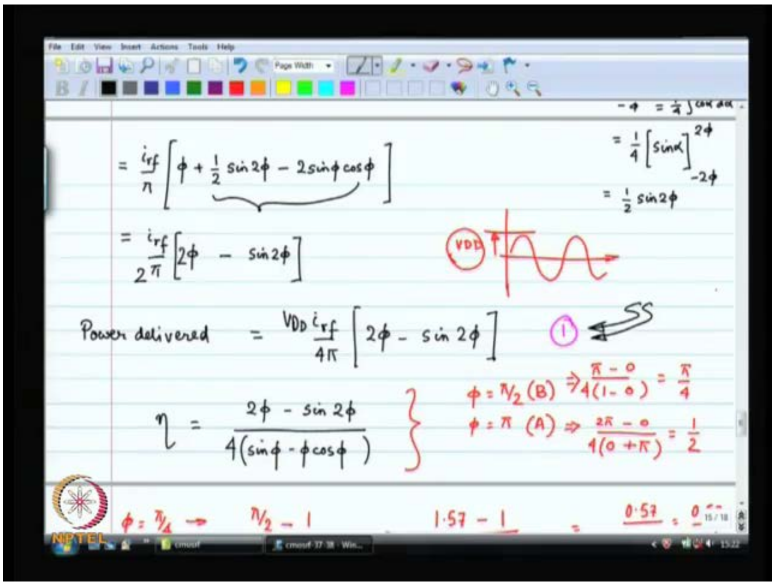


What is the total power that you deliver to the load? It is only so much, and for a certain phi, let us say phi equal pi by 4. If phi is that is why; we got an outstanding efficiency, right. At lesser conduction angles things are going to be even better even closer to 100 percent. So, twice phi is basically pi by 2. So, it is something like 1.5 right between 1.5 and 1.6. So, 1.55 and sine 2 phi is 1 pi by 4. So, 2 phi is pi by 2, sine of pi by 2 is 1.

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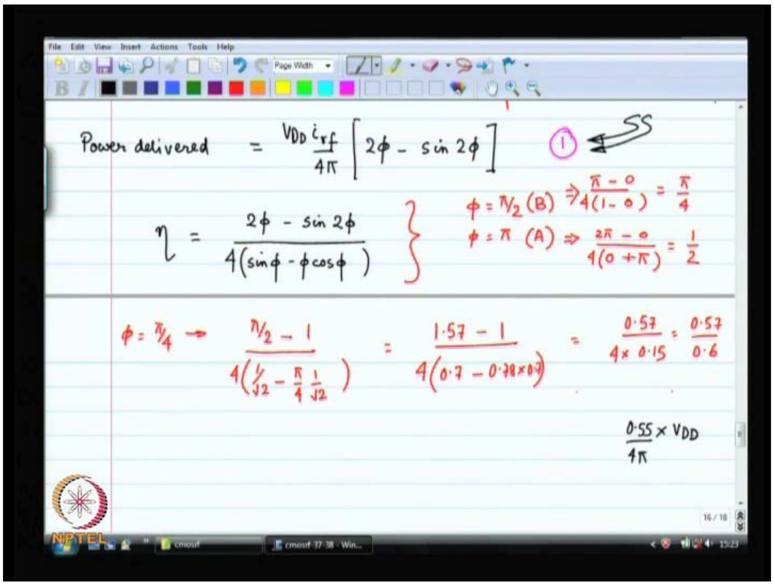


So, this will give me something like 0.55 times 0.55 by 4 pi times V D D times i r f, Right. 0.5 5 by 4 pi factor of 4 pi is something like 12. So, you have got almost 124 of V D D times i r f. So, the amount of power that you are delivering finally, at the end of the day is also going down. So, this is the unfortunate part of the story. So, there is only so much power that you can deliver. Now, in our applications the efficiency of course, is very important, but the amount of power that you are delivering to the load is also extremely important right, because you are designing to deliver certain amount of power to the load.

So, typically what happens is at the end of the day as far as the radios are concerned, radios or cell phones are concerned you almost never make class C amplifiers, you do not even make class B. You typically make something that is a class A B amplifier this is more or less the accepted story. And, class A B amplifier means the conduction angle is more than. So, the time during which the transistor is on is more than half of the time, but less than all the time.

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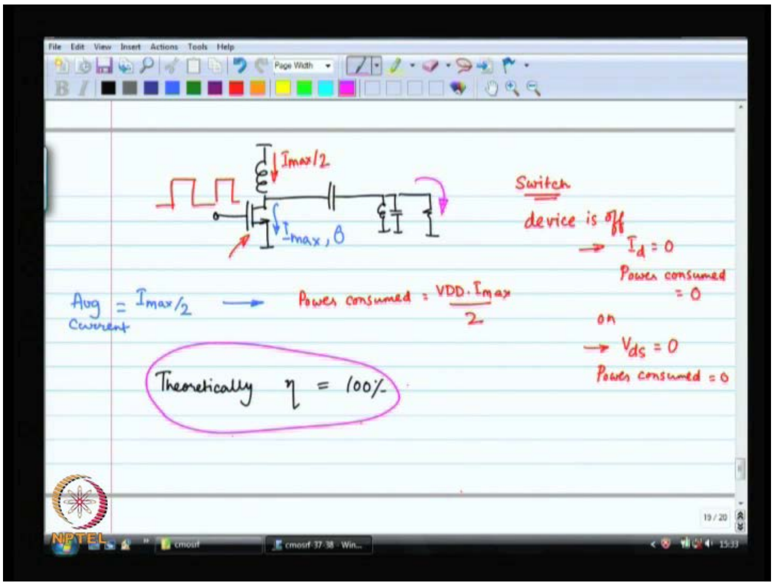


Now, what does this mean? So, between class A B and C typically class A B is the best compromise. So, we will stop at this and we will look at a couple of other architectures. One is the class D amplifier now, a class D amplifier is a push pull amplifier the idea is this. Let us once again draw our schematic now, the thought is this that if I drive this transistor hard, hard enough that it works like a switch, if I can make the transistor work like a switch, when the device is off the current through it is 0. So, the power consumption is 0, when the device is on; there is current, but the voltage across it is equal to 0, which means once again the power consumed is equal to 0, alright. If I can operate this particular MOSFET as if it is a switch, then it is never going to consume any power.

So, I have to drive the voltage at the input of the MOSFET very hard. So, this is like a square wave, aright. So, if I drive the input of the MOSFET like a square wave, when the device is on it is not consuming any power because the voltage across the device is 0 then, the device is off once again it is not consuming any power because the current through it is 0. So, where is all the power going? Suppose on the average forget the average, suppose when the device is on it is taking a current of I max when the device is of it is taking a current of 0. So, the average current that it is taking is assuming 50 percent duty cycle is I max by 2, which means; that the average consumption of power not the average consumption of power consumption from the power supply, think about it. This was not allowing any current other than D C.

So, the current through this is equal to I max I 2, assuming 50 percent duty cycle, it does not really matter what the duty cycle is. So, that basically means that the power consumed from the power supply is V D D times I max by 2. Where is all this power going, the inductor does not take any power, the capacitor does not take any power, the MOSFET we just proved is not taking any power. So, where is all the power going, where is it going? It is going to the load; there is nothing else, right. There is just 1 MOSFET which is not consuming any power; we just proved that it is a perfect switch. And, if it is a perfect switch; it does not consume any power, inductor does not consume any power, and capacitor does not consume power the tank circuit does not consume any power.

But my voltage my power supply voltage is delivering power. So, clearly all that power is being consumed by the antenna, there is no other way right. So, what does that mean? That theoretically; the efficiency is 100 percent. Now, is this correct, think about it? Once again all the power is being delivered to the load. So, therefore; what I am saying is efficiency is 100 percent; there is a little bit of a flaw here, quite a few flaws first of all there is no such thing as a perfect switch. So, in the absence of a perfect switch the MOSFET is going to consume power. And, therefore; the efficiency is never going to be 100 percent, fine.

Let us say that you do have a perfect switch does this argument hold water even then, is it correct, is it correct to say; that the power delivered to the load has an efficiency of 100 percent. Well, you are justified in saying that the power delivered to the load is equal to the power consumed. But is all of this power at the desired frequencies? Yes, no, what is the kind of wave form that you are going to see across the load, the spectrum of that wave form.

You are going to see the fundamental; you are going to see that second harmonic, third harmonic, etcetera. So, all of those harmonics also have power in them, which means; that this 100 percent of power that you are delivering, you are taking some power from the power supply translating to a lot of different frequencies, your fundamental, your second harmonic, third harmonic pumping them into your load, alright.

But the desire portion of this set is the one that said the fundamental frequency not, at the other higher frequencies, right. So, even the statements theoretically eta is equal to

100 percent is kind of flawed. We have not done the Fourier analysis that we did before, we did not find out fundamental portion of the current going into the load. How much is the fundamental portion? You can find out, it is a square wave, right. So, it is not terribly difficult to find out the fundamental of the current going in to the load, this is the filtering action which is why I am hesitating to do it on paper, do not want to do it on paper. You have to include the filtering action over there, otherwise the current going in over here is straight forward, you can find out alright, let us just do it.

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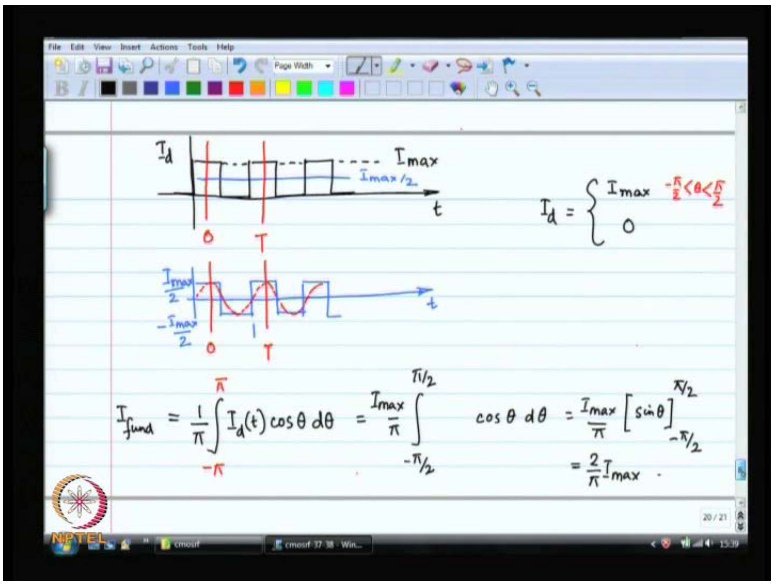


Let us say that this is the wave form of the current going in to the drain of the MOSFET. Now, as we said before the current going through the RF choke is going to be the average of this, which is I max by 2 which means; that the current going in to the load, load tank rather load tank and load is basically going to look like this. It is a square wave period T, alright. And, then what we are trying to figure out is what is the fundamental component of this, how much is the fundamental component of this? I suggest that you do not go by I estimation over here, and actually do computation.

So, your current, you do actually work on the previous 1; what was our so, this is fundamental component and, actually I am going to shift this a little bit. Actually, let us shift our axis a little bit. So, let us not say this is 0, let us say this is 0 and this is T, because; that is really what is going to give me the correct result, alright. So, that is what I have got. And, you take out I max outside the integral and you work on this integral of cos theta D theta is basically sine theta. Sine theta going from minus pi by 2 to pi by 2 sine of minus pi by 2 is minus 1, sine of pi by 2 is plus 1. So, I have got plus 1 minus 1.

So, that is equal to 2. So, the fundamental component is really 2 by pi times I max, right. So, that is why I did not put I max by 2 over here, it is 2 by pi times I max. So, this fundamental component is not going into the tank because the tank looks like infinite resistance at the chosen frequency. So, therefore; it is only going to be delivered to the load which means that the power delivered to the load is the amplitude of the fundamental squared by 2 times r, right.

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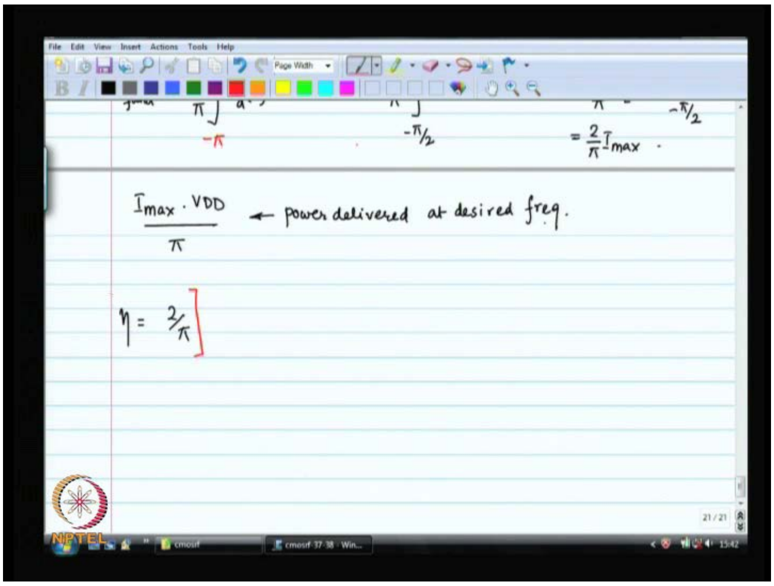


And, basically you are going to get I max times V D D by pi, right. This is the power delivered to the load. And, the next step is to make sure that our consumed is I max times V D D by 2. So, therefore; deficiency is basically going to be power delivered divided by the power consumed. So, you are going to be equal to 2 by pi. So, this is the unfortunate part of the story that here it looked very promising when I started off got a theoretical that seemed like I got an efficiency of 100 percent. But when you recall that the power delivered to the load has a lot of different frequency components, it is not just the fundamental then, you realize that this argument is flawed.

And, really going by our previous discussions the way we did the class C amplifier, find out; what is the fundamental component current going in to the load and what is the amplitude at the load etcetera. And, you do the derivation you will realize that the maximum efficiency that you are going to get is like 2 by pi. The next thing is these are basically classified as class D amplifiers. You have to realize that the input to class D amplifiers is something like a pulse width modulated signal. The input to the class D amplifier is a pulse width modulated signal, what I am suggesting is that by changing the width of this pulse you can achieve whatever signal you want.

And, basically; your filter at the output is going to make sure that you just have a narrow band of frequencies around your center frequency. So, the class D amplifier is a kind of almost synonymous with pulse width modulated amplifier. You can conceive of having sigma delta modulator as the input, driving the input of class D amplifier, right. A sigma delta modulator is a pulse width gives out a pulse width modulated wave form, we saw this before, when we were talking about frequency synthesis. So, it is popping up in all the wrong places sigma delta modulation, but anyway. So, you could conceive of a pulse wave modulated wave form that is the input to the class D amplifier and power is transferred out.

The catch is you go to remember the catch; the catch is all the power is not at the fundamental frequency. So, beware of that; the efficiency is 100 percent as long as the switch is perfect. Of course, the switch is also not going to be perfect ok. So, these class D amplifiers are not commonly seen in cell phones. The primarily the problem is the pulse width modulated wave form you need at the input of the class D wave form. So, that is a big stumbling block especially when you are talking about extremely high frequencies.

However, the class D amplifier is very popular at audio frequencies. It is very popular at audio frequencies, because it is so easy to build and you can conceive of having an audio sigma delta converter which is also easy to build. And, the output of the audio sigma delta A to D converter, you got a 1 bit A to D can directly drive the power amplifier. And, you have got a pulse width modulated wave form the input up the amplifier, you treat it as a class D amplifier, and figure out your analysis, all the power get delivered to the load. Of course, there are harmonics and you cannot avoid them. Now, in all of these amplifiers class A, B, C, D, A B whatever you want, we have achieved efficiency we can get whatever efficiency we want, we can deliver power the problem the basic problem with all of these amplifiers is the linearity.

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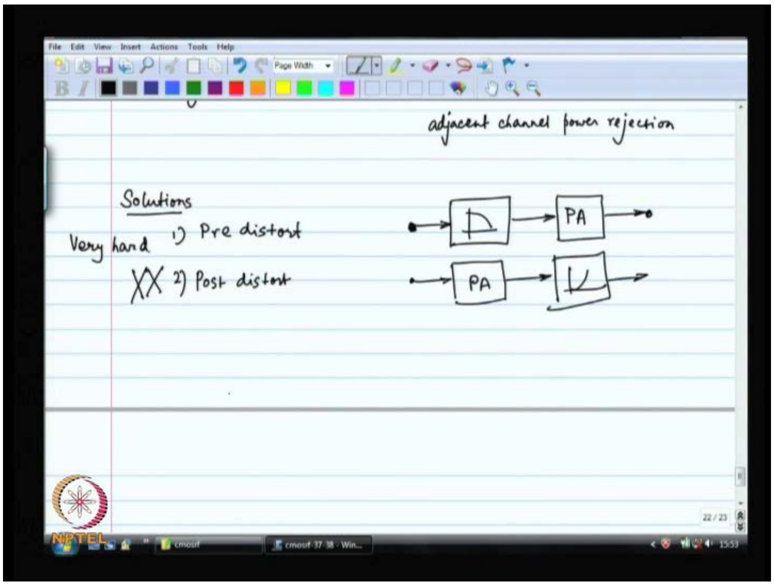


And, linearity is a very important concern because in modern wireless systems. Your system has to be very good very linear as in you are not allowed to transmit at channels that do not belong to you, right. Your base station has instruct you to transmit at a certain channel, you cannot violate. So, there are some terms that are commonly used; 1 of the popular terms are A C P R adjacent channel power rejection. So, this is the acronym that is used in all the standards documents, right. And, we saw that just by a squaring operation you basically smear the signal to twice the bandwidth right. Similarly, of course, when you do a squaring operation on RF signal you actually come down to base band and to high frequencies.

So, squaring is not really the problem when you do a third order linearity. You will actually see the smearing, any kind of odd order non-linearities will smear your signal and will give you power in adjacent channels. So, that is bad for us therefore; we want a perfectly linear amplifier. How are you going to make a perfectly linear amplifier, is it even feasible under the present circumstances. So, what we have got is a power amplifier that is horribly non-linear. You input signal over here, and the output the relationship between the input and output is not at all linear, which means; if the input is a perfect sinusoid output will have a lot of harmonics, alright.

We do not want these harmonics, in fact, the harmonics are not really our problem, our problem is when the input is a sum of 2 sinusoids that a close to each other. Then, the output will have a lot of other unnecessary components that we should not have been having. So, if my input is sine f 1plus cosine f 1 plus cosine f 2, the output because of a third order linearity is going to have 2 f 1 plus minus f 2 it is also going to have 2 f 2 minus f 1 and of course, it will have f 1 and f 2. Now, f 1 and f 2 are desired signals, but 2 f 1, f 2 2 f 1 minus f 2 is very close to f 1, it is not something that I want 2 f 2 minus f 1 is also very close to f 1 it is not something that I want.

So, this is what basically creates the problem. How do I solve this? Well, 1 thought is why do not we pre distort the signal. So, I will put a block over here that is also non-linear then, I am going to send it to my PA, and I am going to say that between these 2 points I have got perfect linearity. Now, such a system is very hard to build. So, is a system called post distortion, so this is also very hard to build, and it is not, they might not be feasible. In fact, because the PA the characteristics of the power amplifier depends on a lot of different things, depends on temperature, it depends on the process, etcetera.

And, characteristics of this distortion circuit, pre distortion or post distortion circuit will also have to change accordingly. So, the post distortion circuit is almost never feasible because you really want, you are really planning to drive the antenna instead. Now, you are going to drive the post distortion circuit the post distortion circuit will have to drive the antenna. So, it is really not something which is advisable the pre distortion is still as a system, but you know it is very hard to build. What else; could we do is there anything else that we could possibly do anything else that we could possibly do any thoughts well you can think of using feedback.

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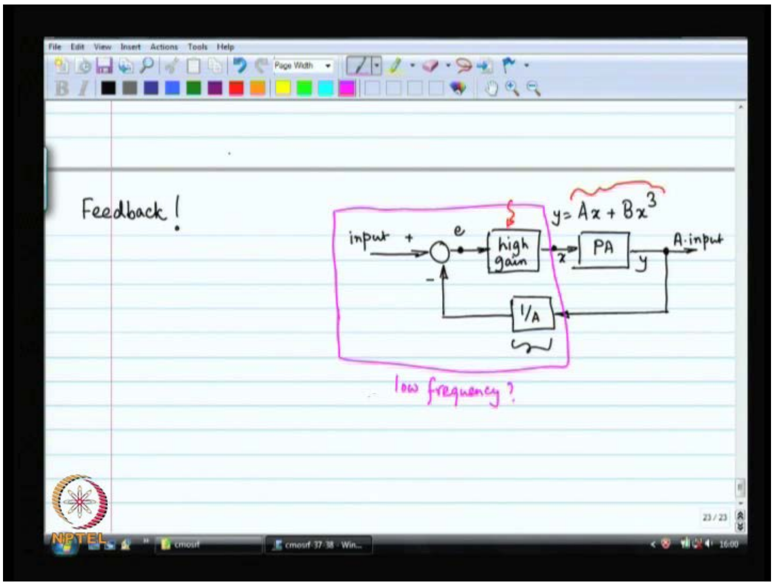


So, we have done this a couple of times, this is a general block diagram for a feedback system. Our plant over here is a power amplifier. So, if somehow if I could sense the power output of the power amplifier or if somehow I could sense the quality of the signal at the output of the power amplifier and compare it to what I want. Then, that is kind of going o give me the linear system. Feedback make sure that the feedback is going to make sure that distortion is also minimized is it or is I not going to make sure of that, how is distortion reduced when I apply feedback. What are your thoughts? No thoughts think of this as a high loop gain system, you always want large loop gain.

So, let us say that this is not linear A x plus B x cube, alright. This is x is the input, y is the output has some characteristics that looks like this. Now, I want a total gain of A. So, I sense the output, I divide the output by factor 1 by A, and I compare it with the input, alright. And, this of course need a plant here, I am sorry, is of course, need a controller there. This comparison gives result to an error signal, alright. And, if the loop gain of my system is very large then, the output is equal to A times the input, if the loop gain is very large the e has got to be 0, because if e is not 0 then, x is going to be infinitely large. Let us say that this has extremely high gain.

If e is large then, x is out of range, x is infinitely large and you cannot really tolerate an infinitely large I mean it does not make sense right. It is never going to be infinitely large because there is negative feedback. And, therefore; that basically tells you that e is equal to 0, and if e is equal to 0 then the output has got to be equal to A times the input by virtue of this, by virtue of the feedback path. So, we are basically relying on the linearity of the feedback socket, right. So, we can kind of it really does not matter what that polynomial over here is, it does not matter what this polynomial over here is. As long as you have got high gain over here, this will always work, what do you think, is this conceivable, can you do this? We could do this; the problem is with the high gain block.

How are you going to get a high gains at an RF frequency, right? It is hard to make a gain of 10 and, you are asking for extremely high gain of infinity it is not really conceivable to make such high gains at RF frequencies unless we put a tune circuit over there. Now, if we do put a tuned circuit over there; that has extremely high gain at chosen frequency then; that means, only at that is the feedback system making any sense, at other frequencies the other can be whatever else it possibly wants to be agreed. So, you could make a high gain amplifier at attuned set of frequencies, right. That is kind of conceivable not, at all frequencies. So, this is 1 thought, 1 direction of thought, the next thought is it possible that we do not deal with high frequencies, we translate this entire portion for a low frequency, is it possible to have translate the entire portion to a low frequency and then, up convert it and then broadcast it through the power amplifier.

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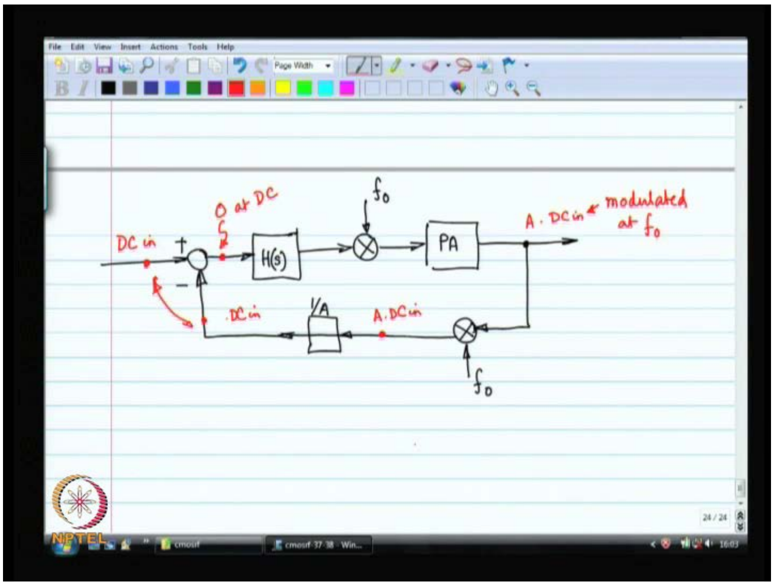


So, what I am suggesting is why do not we build something like this? H(s) is an integrator over here; I forgot to put the 1 by A portion over here. Now, what is going to happen, it is easy to build an integrator an integrator means an infinitely large gain at D C. So, around DC the error signal over here is 0, what does that mean; that means that at DC these 2 signals are equal. That means; I am sorry that means, that the output of the feedback mixer is equal to A times the desired output, right. And, that is forcing the output to be what you want. Now, this is also pretty nice this kind of strategy is very nice of course, you have to make sure that make sure of a few things. First thing that you have got to make sure is the DC offset of the mixers that is a huge problem especially of the feedback mixer.

So, you are almost never going to choose DC as your frequency of choice maybe you are going to choose something offset from DC by so much frequency. In that case; your H(s) has to be designed to be infinite gain at that particular frequency at the higher frequency. So, it is not too difficult to build such an H(s) long as it is not DC it is can be some small frequency. The next thing is of course, a fact you have in band and quadrature phases and you do not want them to get mixed up, right. So, you can modify this architecture a little bit.

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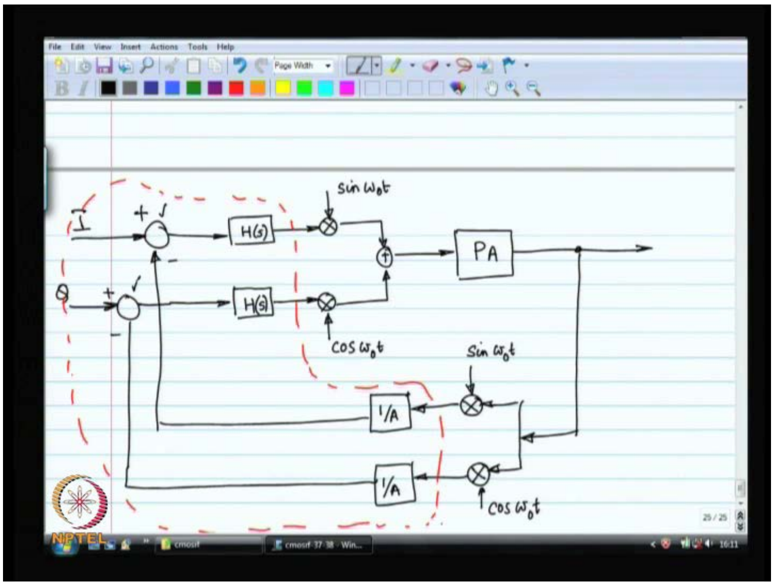


So, this is modification of the previous architecture accommodated for in band and quadrature phase. This is actually a very popular architecture as far as the transmitter is concerned. So, the signals I and Q are directly coming from D to A converter. And, they are a little bit not exactly at DC because it is hard to make the mixer the feedback mixer function at direct down conversion it is hard to make it function. So, it is a little bit offset from DC. However; you have to remind yourself that these signals are further a digital system. So, it is not terribly difficult to have some small frequency over there.

So, it is all very straight forward it is done digitally now, this is he saving word for everything. That you cannot do you do digitally it does not mean at the end of the day it is really going to be easy anyway we are going to say that it is being done digitally and we would not worry about it. So, these I and Q signals are at some low frequency and the error signal has got to be 0, right.

This error signal has got to be 0 that is even better is that this entire thing can work digitally just that you need to build 2 more A to D convertors in that case. So, it is feasible to make this entire block in the digital domain. But you will need 2 extra A to D converters which may or may not be justified. So, over here what we need is some sort of an analog subtraction, it is not terribly difficult to make an analog substractor at low frequencies.

The filter what I have termed H, it is a continuous time filter H has to filter out all garbage all the other higher harmonics, and it has to make sure that within the band of where, you want the output to be faithful to I and Q inputs signal passes right through. So, that is what you have to make sure when you design H(s). And then the quadrature mixer is of course, something that we have studied before the power amplifier can be a non-linear amplifier it does not matter the rest of the system is straightforward. Now, what is of great concern here is the mismatch of I and the Q path. So, if the I path you have made an H(s) for the in phase we have made an H(s) for quadrature. If these 2 are not identical then, you have got a problem similarly; you have made an attenuator 1 by A for the in phase you have made an attenuator 1 by A for quadrature.

If they are not perfectly equal to each other you have got a problem similarly, the subtraction this particular unit over here, right. These 2 also have to be perfectly matched of each other otherwise, mean phase and the quadrature phase are not going to be exactly what you want you are going to get some constant time in phase plus some other constant times the quadrature phase which is not what you want. This particular error kind of broadly called it has a term which you will see in your standards documents.

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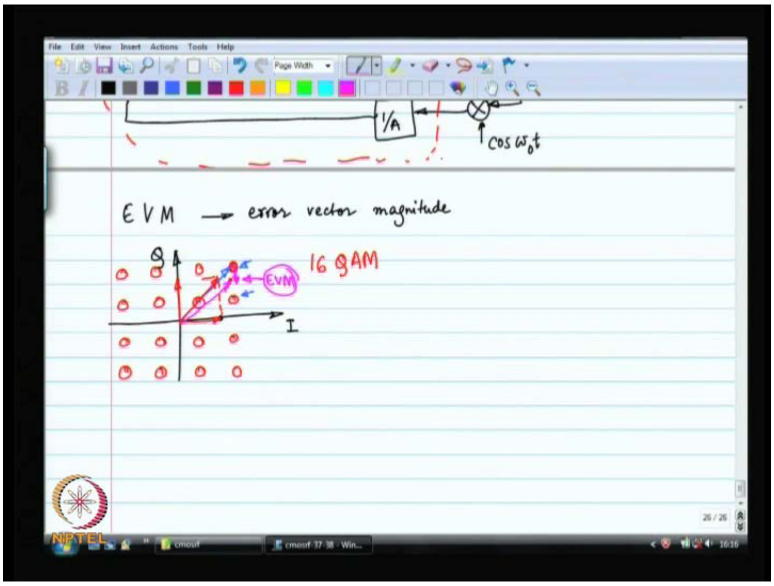


It is quantified as E V M ok, so it is like this. The I, this is the in phase I and this is the quadrature Q, alright. These are the in phase and the quadrature phase phasors that we are talking about think phasors now. So, the phaser of the in phase is a certain magnitude, the phaser of the quadrature is a certain magnitude means; that the net signal is the vector addition of the 2 and the next signal has a certain magnitude, and phase alright. Now, these in your communications books you will see a constellation diagram suppose, you are doing something like binary shift key then, your constellation diagrams will have these 4 points.

If you are doing some other modulation scheme; the more difficult this is easy to realize I mean the in phasors are either there or not there quadrature phase is there when in phaser is not there etcetera. The other more difficult schemes are when the constellation diagrams are all over the place. Actually; I have drawn it incorrectly. So, this is called 16Q A M, there are 16 symbols, 16 possible symbols at every instant of time. You are actually sending 1 of those 16 symbols which means you are sending 4 bits at a time. So, it improves the data rate of communication 16 Q A M is used quite often in modern systems, modern cell phone systems.

And, if you have made a mistake in this phaser if the I path has gain that is different from the Q path. So, you probably wanted to go over here, but let say you r I path has more gain than the Q path. The Q path has less gain, which means; you will come somewhere over here. Now, the decision as to where this particular symbol belongs to this or this. If that decision is clear then you have met your E V M requirement, if that decision is wrong then, you have not met your E V M requirements. So, the modulation scheme is going to tell you what is the maximum error in terms of I path and Q path gains are concerned, you can tolerate.

So, the error vector magnitude is; let us say this was the desired phaser instead I have achieved this phaser. So, the error vector magnitude is this much, alright. Then, we will tell you what this percentage can be, what percent of your total amplitude this can possibly be that is what they are going to tell you. So, this is why; matching between the I and the Q path is very important and this is going to play an extremely crucial role in this scenario. So, this kind of feedback system is called a Cartesian feedback system. This is very popular in modern radio architecture in G S M systems. The amplitude is always the same it is a sine wave.

So, you can also employ other different architectures which use polar feedback. So, instead of comparing x and y you compare the magnitude and the phase and create some sort of feedback loop using polar feedback. So, I am not going to discuss polar feedback in the class, but you can I will point out a reference, which you can look up as far as polar feedback is concerned. So we are going to stop the discussion over here. So, we have discussed power amplifiers A B C D amplifiers pulse width modulation class A B amplifiers of course, we went in to the details of efficiency. How much power is delivered to the load, and then we have studied feedback as linearization technique, right.

So, thank you for your attention.