Communication Engineering
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Lecture - 24
Frequency Compressive
Feedback Demodulator

We will continue with our discussion on Frequency Compressive Feedback that we started last time. I hope you recollect the basic idea, we just introduce the block diagram at that stage.

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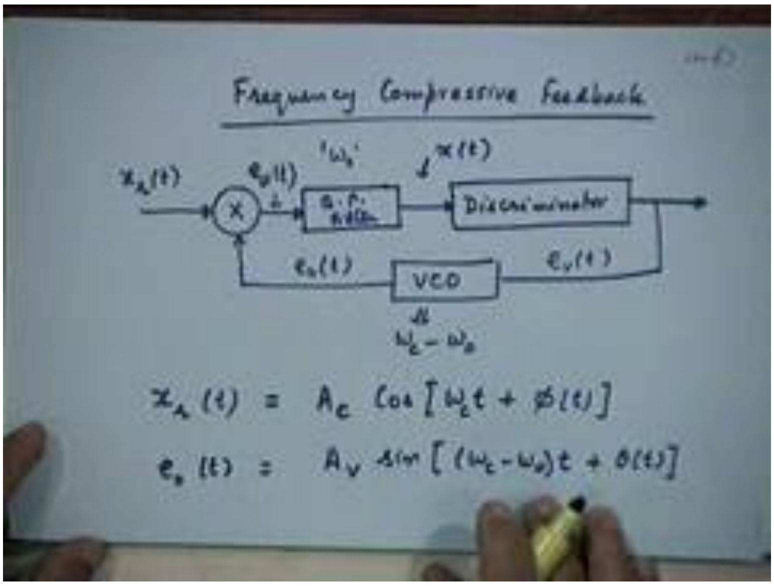

And the block diagram look something like this, you have the incoming FM signal here this is followed by a multiplier and a bank pass filter, this is slightly different from what we have in a phase locked loop, where the multiplier is followed by a low pass filter. So, that this combination acts like a phase detector, even now with more or less will do very similar thing, except that the difference signal will not be a base band signal. But, will be a signal at the difference frequency between the incoming signal frequency and the VCO frequency.

The VCO frequency is tuned not to omega c, but to a slightly is somewhat offset from omega c to some other carrier frequency omega 0. So, there is an offset ((Refer Time: 02:07)) amount of omega 0, so that the difference frequency here is omega 0. So, this passes through this bank pass filter and as we will argue today at the output of this bank pass filter we have the, we again will have some kind of an FM signal present here, what kind of FM signal will try to see that.

So, that the demodulate be require a discriminator and the discriminator output is fed to the VCO to close the loop. So, the essential difference between the phase locked loop and the frequency compressive feedback system here is that the VCO is offset from the incoming signal frequency by an amount omega 0. And as a consequence the filter following the multiplier is not a low pass filter, but a filter tune to omega 0, the amount by which the VCO is offset.

Now, how does this help to start the analysis of this to understand how it works or what it is able to do for us, consider the received signal as a signal of this kind A sub c cosine omega c t plus phi of t, where phi of t denotes your modulation. Then, the VCO output here we say would be let us say A sub v sin of omega c minus omega naught t plus some phase modulation theta of t. Where, once again the phase modulation will be dependent on the input signal here that is for example, in a VCO we know that the model is at the instantaneous frequency is proportional to the input voltage.

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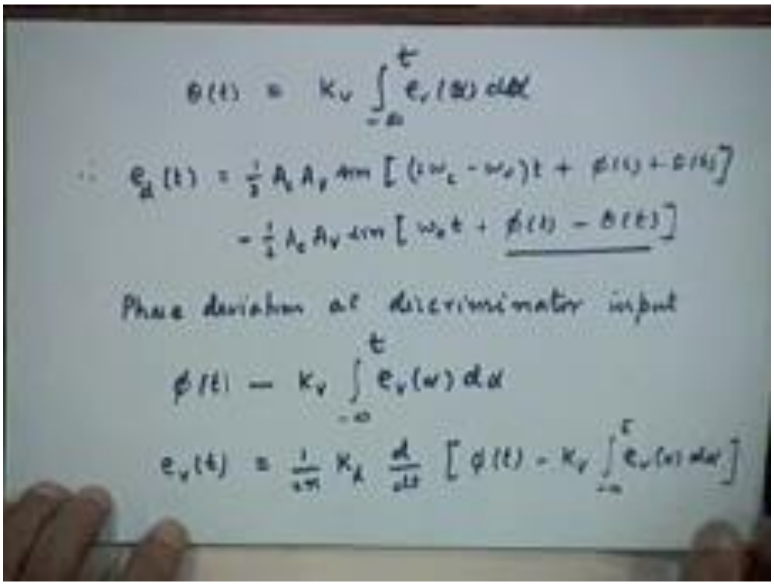

So, the model for theta t is the usual VCO model theta t is let us say some constant K sub v integral of e sub d t or where K v is the VCO constant. So, all the expressions are same except that the VCO output is not at omega c, but omega c minus omega naught t, so what happens to let us call this signal, the signal at the multiplier output as ((Refer Time: 05:02)) e sub d t signal here. So, what will be the nature of the signal e sub d t, you multiply x sub r t and e naught t it will be a cosine into sin term and there will be a some component and there will be difference frequency component.

So, e sub d t becomes half of A sub c, A sub v some component will be sin of 2 omega c minus omega naught t plus phi of t plus theta of t, this is a some argument. And the difference argument will be minus half A sub c, A sub v sin of omega 0 t plus phi of t minus theta of t, this is straightforward application of trigonometric activities. So, what will be the bank pass filter output, bank pass filter output will be proportional to this quantity, I mean well it is basically the second term.

Because, that is that frequency omega 0, the first term 2 omega c minus omega 0 is it a much higher frequency and we will remove with the bank pass filter. Therefore, if I look at the phase deviation at the input of the discriminator, what can we say about that the phase deviation is really you know this part phi of t minus theta t is not either phi of t or theta of t. There is a difference between these two which is it what requires in the case of PLL also except for this term.

Now, let us look at this quantity phi of t minus theta t, theta t is from here K sub v integral e sub v alpha d alpha minus infinity to t. ((Refer Time: 07:36)) This is being followed by discriminator and what could the discriminator do here, this discriminator was not present in the PLL instead of a discriminator at most we had a loop filter. So, we have a discriminator here and what is a job of a discriminator, it produces an output voltage proportional to the in input frequency deviation, instantaneous input frequency deviation.

So, we have the output e sub v t output of the discriminator is let us say 1 by 2 pi the constant K sub d which is a discriminator constant into d by d t of incoming phase deviation, which is phi of t minus K sub v integral e v alpha d alpha, if you about to continue this manipulation.

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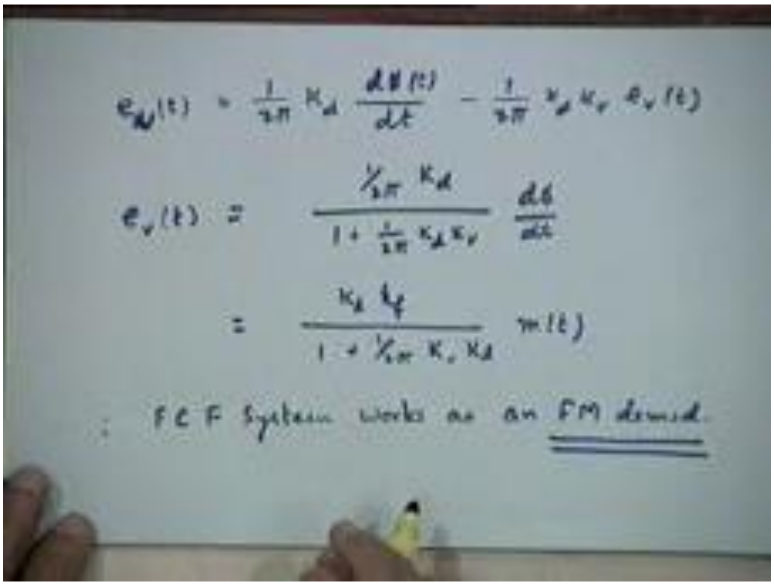

So, this will become e sub d t as 1 by 2 pi K sub d d phi by d t minus 1 by 2 pi K sub d, K sub v e v t this should be e v t my mistake, the left hand side is also e sub v t that is what it was, let us continuing on the next sheet, you can move this e v t, e sub v t to the left and solve for e sub v t. So, if I do that I will get e sub v t as 1 by 2 pi K sub d upon 1 plus 1 by 2 pi K sub d, K sub v into d phi by d t, that is quite obvious take this on to the left hand side, combine these two terms solve for e sub v t this is what you will get.

Let me write this as in terms what is d phi by d t; d phi by d t is the instantaneous frequency deviation of the input FM signal which is going to be proportional to the message signal. So, this becomes K sub d, K sub f upon 1 plus 1 by 2 pi K v, K d m t, so therefore, the demodulation is done, the output is proportional to m t as we would like this is a constant. So, the output is proportional to that, here I have taken the FM modulator constant to be 2 pi K f it is a unit slightly different from what we are been considering, so far.

So, therefore, it is clear that the system is frequency compressive, feedback system works as an FM demodulation. What is a big question here that it works is fine then what is a big question here, there is a certain kind of question that arises here. That if you are going to have a discriminator here anyway ((Refer Time: 11:39)) you might as well have just the discriminator and that will produce your FM output.

So, what are we achieved, no to see that let us go through some more mathematics a little bit of mathematics to understand why it helps or why is it useful to do the way we have just done. It is to try to write an expression for x t, where x t I am defining the signal here, signal at the discriminator input. Why, we just said that the signal here is basically due to this term ((12:25)) the signal here is that due to the difference term, the signal and there the instantaneous phase has this form phi of t minus theta of t, where theta t is this. Let us try to understand this a little more, so let us write down an expression for x t, x t is this expression.

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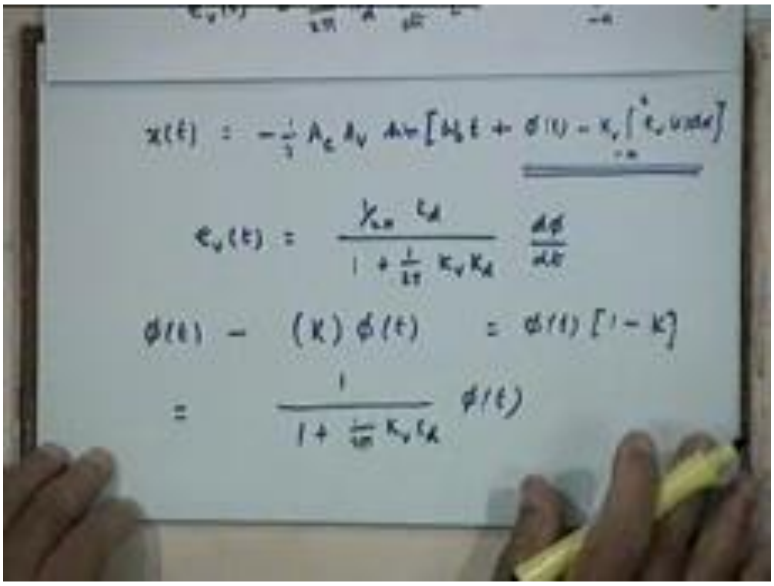

So, x t, x t is the signal at the discriminator input, let us see what kind of signal do we have at the discriminator input, it is minus half A sub c, A sub v sin of omega 0 t plus let me substitute for this quantity. Let me first write it like this plus phi of t minus K v nothing new there, I want to simplify this is there a way to simplify this any suggestions for that. Look at this, we just got this expression for e v t what was it, e v t was 1 by 2 pi K sub d upon 1 plus 1 by 2 pi K sub v, K sub d d phi by d t.

So, if I integrate this what do I get, I essentially get this constant times phi of t you already have phi of t. So, two together can be express in terms of phi of t what it will be phi of t into 1 minus this constant. So, this argument which I have underlined here can be written as phi of t minus the integral of this which is let me just rewrite this constant as some K into phi of t, where K denotes this constant.

If I carry out the subtraction this becomes 1 upon 1 plus 1 by 2 pi K v, K d phi of t, just a simple algebraic manipulation. Any questions on this? 1 minus capital K is basically this constant, as this I am writing as phi of t into 1 minus capital K, where capital K is this constant you carry out the subtraction we get this.

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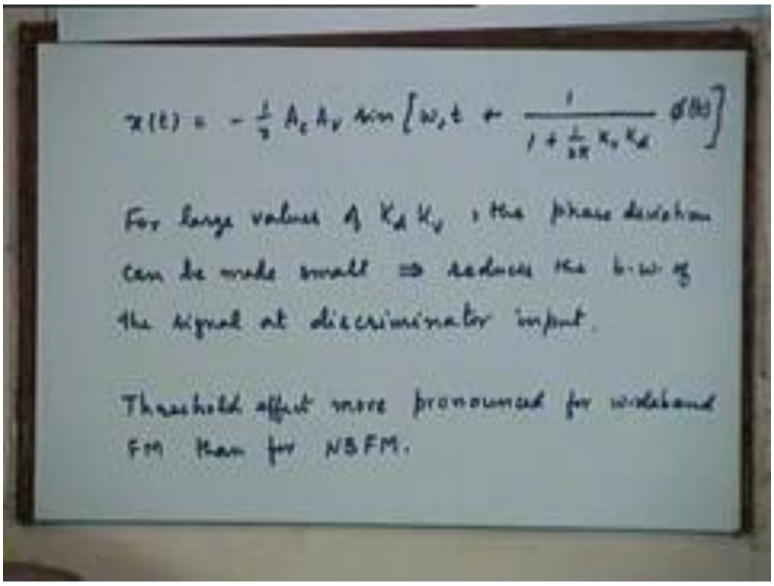

So, therefore, this expression for the signal x t becomes minus half A sub c, A sub v sin of omega 0 t and the term phi t minus theta t becomes 1 upon 1 plus 1 by 2 pi K sub v, K sub d phi of t. Now, that is the point at which we can try to gain some understanding or insight into what we achieving here, what was the input FM signal like. The input FM signal was similar except that you had omega c t here, some other carrier frequency plus phi t this constant was missing, where omega c t plus phi t.

So, the FM signal at the input or the PM signal for the input was this plus phi t of course, you have a different carrier frequency. But, the carrier frequency is of no consequence, but because the discriminator will be designed to operate at the carrier frequency of the input signal, if you are directly used the discriminator the discriminator would have been design around omega c. Since, you are using the discriminator inside the loop; this discriminator would be working around the frequency omega 0 that is your bank pass differentiator will be around omega 0.

But, it is a same discriminator except that the range of frequencies, which it operates on, is different. But, important point is this constant, what is the nature of this constant is it a very small quantity or a very large quantity, it is a small quantity. Because, typically will we can choose very large values of K v and K d and typically to be done. So, what we are doing essentially is multiplying the incoming phase to by a small number, what is that mean. That means, the discriminator that we are going to design for use within the loop has to work with a narrow band FM signal, rather than a broadband FM signal.

So, even if the input signal is a broadband FM signal, because of this division here of phi t by some small constant, it converts that broadband signal into effectively narrow band signal or a signal with a smaller value of the modulation index beta. So, if you choose large values of the two constants here that is the product K sub d into K sub v is large the phase deviation can be made small. So, this reduces the bandwidth of the signal at the discriminator input. Effectively it reduces the modulation index, modulation index of course, when talks about only for a sinusoidal modulating signal in this case.

So, in general basically converts this broadband signal into narrow band signal, that is what it reduces the bandwidth of the input discriminator, why should that be important for demodulation is not obvious at all, how does it help us. Now, unfortunately the details of this I cannot discuss with you right now, but let me just mention something which I mentioned earlier, this has to do with a peculiar characteristic in the demodulation of FM signals. Particularly, when you are dealing with noisy input signals, I mention this earlier very briefly.

The FM system works extremely well in noisy conditions, provided the signal is considerably larger than noise, provided the signal to noise ratio is above a certain threshold value. However, it is interesting to load will discuss how that happens later, because we are not right now discussing the performance against noise, as we do that will appreciate that the FM signal actually becomes very bad, when the input noisy conditions become very bad.

So, instead of a graceful degradation as the noise increases, which is what we normally expect any good system to work with to behave like, the as the input signal becomes worse, the output al also becomes worse. But, usually we would expect systems to have graceful degradation, as the input signal degrades in quality, the output also should degrading quality proportionally FM does not behave like that. It behaves extremely well when the input signal is good, that is signal to noise ratio is good.

But, the moment the signal to noise ratio becomes smaller than some threshold value it starts to behave extremely badly. The system almost breaks down, that is why this phenomenon is called threshold effect and it is advised to use FM always above the threshold value, the signal to noise ratio should be maintained to be larger than some threshold value. So, this threshold effect is something will discuss in more detail when we discuss effect of noise later, but this is an important fact of life when we consider FM.

Now, this effect is more pronounced this threshold effect that I have just discussed is more pronounced for wide band FM, than for narrow band FM. And that immediately gives us the motivation for going for this circuit, because the discriminator here is working with the narrow band FM signal. So, the effect is the implication is that it will be less sensitive; it will be less prone to threshold effect, than the discriminator which would have had to operate on the wide band input FM signal directly.

So, this is a real advantage of going for frequency compressive feedback that is to reduce the incoming wide band FM signal, to narrow band FM signal for which the threshold effect is less pronounced. What do we mean by the threshold effect is less pronounced means, that the input signal can be ((Refer Time: 23:05)) threshold is lowered, the value of the threshold SNR at which it degrades this proportionately become smaller. So, in a way we carry out a kind of threshold extension using the frequency compressive feedback system, the threshold is move to the left.

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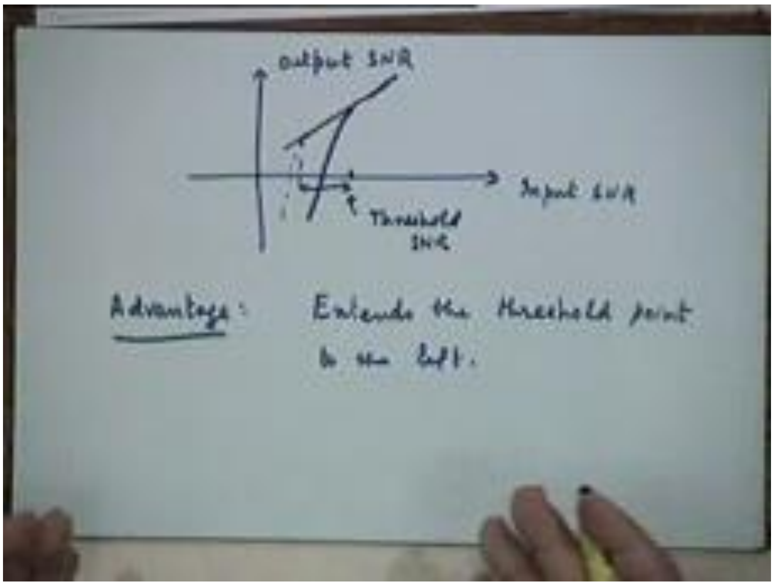

Let me just let us try to argue this out pictorially any communication system. If you have to plot mean we are not yet done this analysis, output SNR versus input SNR. SNR stands for signal to noise ratio, signal power to noise power ratio. Typically, the behavior would be like this as an input SNR is decreased, the output SNR would decrease that is expected there will be some SNR gain depending on what kind of system you have.

But, the output SNR will; obviously, decrease as the input SNR decreases, there is nothing wrong with that, because that is likely to happen this will happen for example, in AM this will happen in FM. But, what happens in FM additionally which is that I am talking about now, that as long as the input SNR is above some value you will find that this output SNR is very good, it will be much more than what you can get in AM system.

But, the moment it falls below some value let us say this value, the input SNR falls below this value. Instead of degrading gracefully like this which is what would have been ever thing to do. It suddenly starts to behave like this, the output SNR becomes much smaller than the linear decrease that I have shown here. So, that is the threshold effect and this SNR at which it happens is called the threshold SNR. That is you have to maintain the input SNR above this value it is therefore, desirable that the value of this threshold SNR should be as low as possible and that is what we are trying to do.

For working with narrow band FM discrimination, rather than wide band FM discrimination, even though you are transmitting a wide band FM signal, we are trying to move this threshold point to the left. So, that instead of this behavior starting here, it can start here and you have this margin by which you have carried out threshold extension that is the purpose of this system.

So, that is a main advantage of frequency compressive feedback, so the in one line the advantage is it extends the threshold point to the left. However, I am merely making a statement here, because I am not true to any of these things that I mentioned, at this point we cannot do that. But, that is the main advantage; just take it from it right now, any questions? Both the phase locked loop and the frequency compressive feedback system, essentially help us to do this carry out the threshold extension, that I have just mentioned.

So, that brings us to the end of discussion of FM demodulation using feedback circuits, we discuss two feedback circuits, the phase locked loop and the frequency compressive feedback. Both have similar working principles, but they differ in detail this is what we are discussed. Now, let us look at the performance of FM signals I am not yet ready to discuss the performance of any kind of modulation signals in the presence of noise because, we need to build our background about describing noise, because noise alike signals that we are dealing with normally are non deterministic signals.

So, I do not have yet the necessary back have not yet developed in the class, the necessary background that is required to discuss the performance of all this systems in the presence of noise. Noise is a very, very important phenomenon that every communication engineer has to deal with communications is broadly made difficult. Because of the presence of noise, various kinds of noise which we are discussed long time ago.

However, it is possible to get a feel for how the system performance will be even without knowing, how to characterize noise. And that is by looking at the situation of deterministic kind of interference, which we did for example, also for the case of AM signal. If you remember, we discuss what happens if there is a nearby carrier to the AM signal and it gets through your RF and IF filters, the image frequency comes in the form of a slightly, a signal which is close to your actual frequency and it comes in.

So, it becomes a kind of interference and we saw what happens in the case of AM, what happen there also there was a kind of threshold effect, if you notice. Particularly, for the envelope detector, for the synchronous detector there was no problem, there was a graceful degradation. But, for the envelope detector as long as the input is signal to interference ratio was large. The output was similar to the input signal except that the interference was proportional to the input interference; output interference was proportional to the amount of input interference.

However, the moment the interference became very large if you remember what happen, the output became totally different it had no resemblance to the input. So, even without considering noise, one can get an intuitive feel for what is going to happen, now will not discuss to that detail here. But, to some level of detail will discuss the effect of sinusoidal interference, which may be present in the receiver. And will try to see how FM and PM differ from each other in the presence of these interferences, which one is better; that is a natural question to ask.

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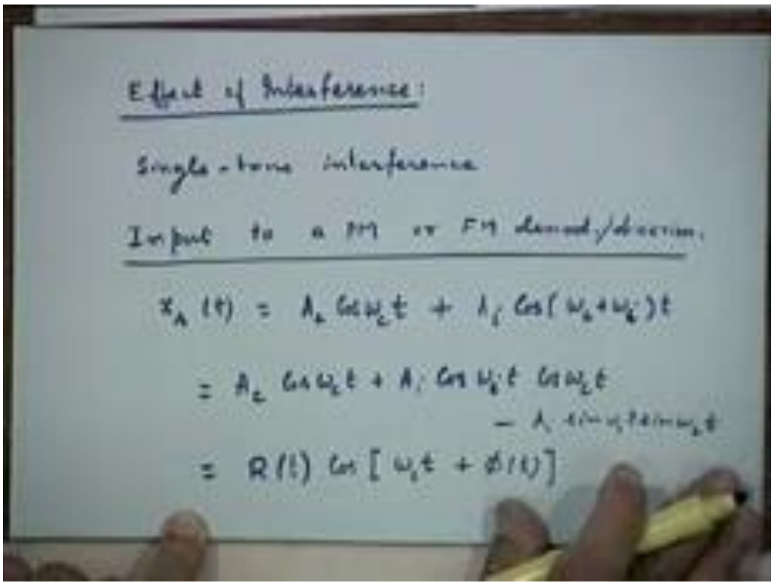

So, we will try to address this question by looking at the effect of interference and will model the interference here as a single tone interference, that is a sinusoidal interference. So, you can proceed at this discussion as some kind of a ((Refer Time: 30:33)) to our discussion to understanding the effect of noise on FM system. So, only thing is about noise here is not a random signal, but a deterministic interference sinusoidal signal for different frequency.

So, let us consider an input to either a phase for PM demodulator or FM demodulator or discriminator, whatever you like to call it, what you are saying is that, this input contains two parts for the sake of discussion. At the moment ignore all the modulations; just otherwise the discussion becomes too complicated. Let us say, you have a desire signal carrier frequency of course, along with the modulation and all that if that carries. Let us just assume that we are ignoring a modulation, there is no modulation present.

So, it is you know slightly tricky thing here, because FM is not a linear system, so it is behavior with and without modulation would be slightly different. But, broadly we can also the fact that if I know the response to one and no other response to the other, it does not immediately tell you what will be the response to the sum. Because, it is a non-linear system, it is to really go through the motions and to keep the discussion simple I am as at the moment ignoring the modulation here.

And let us see, interfering signal has an amplitude A sub i and has a frequency which is slightly different from the carrier frequency. Basically, what we are saying is that your input RF filter and the IF filter bandwidth or whatever are not sufficiently good enough to reject some interfering signal, which may be present close by to the desire signal. So, there is a interfering signal close to your carrier signal, close to your signal of interest and that ((Refer Time: 33:00)) through the i th stage, that comes along with the desired signal through the i th stage, that is what we are trying to understand.

So, how will the FM discriminator respond to this kind of a signal, that is the issue suppose this is the output of the i th stage, how will the FM discriminator or the PM demodulator respond to this simple signal, that is the question will address and try to understand gain some understanding. So, we can write this as how will you do this, we will do it exactly in the same way that we did the discussion for the AM signal, convert this into a signal with some amplitude modulation and some phase modulation and then proceed from there onwards.

So, we can write this in terms of quadrature components A c cosine omega c t plus A i expand this into cosine omega i t cosine omega c t minus A sub i sin omega i t into sin omega sub c t, which we can write in the form of an envelope or amplitude modulation R of t into cosine of omega c t plus phi of t.

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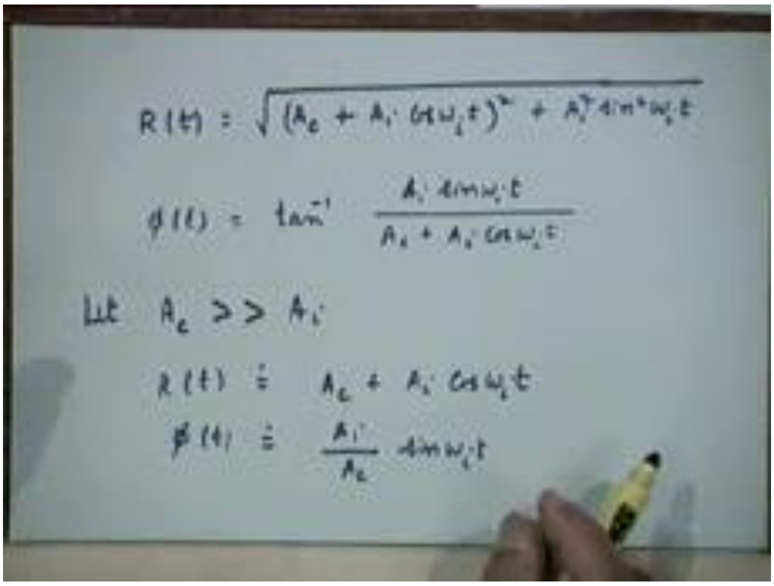

So, see this is, this is the input to the discriminator, in the presence of interference where phi of t what is R t and what is phi t, R t is what is a in phase component A c plus A i plus sin omega i t this whole square plus A i square sin square omega ((Refer Time: 35:10)). Look at it from here, this plus A i cosine omega it is a phase component A i sin omega it is a quadrature phase component.

And what is phi of t, that is tan inverse of A sub i sin omega it upon A sub c plus A sub i cosine omega t that is standard tan inverse of the quadrature component upon the in phase component is the value of the instantaneous phase. To keep the discussion let us consider the discussion primarily again for the large signal to noise ratio, that is assume that your desired sig carrier frequency component K sub c amplitude is much greater than A sub i the interference.

Then, we can simplify this expressions as we can ignore this, essentially it becomes A sub c plus A i cosine omega i t and phi of t becomes A sub i upon A sub c, you can ignore this term into sin omega i t.

Student: ((Refer Time: 36:44))

Any case it is not going to make any difference to our discussion here, so let us keep it here ((Refer Time: 37:00)). Because, your phase demodulator or frequency discriminator is not going to be sensitive to R of t, so it is of no consequence, you are right if you wish you could ignore this, because A sub i is small, here it is easier to ignore than here. Because, there is a squaring involved here, so if you wish you can keep it if you wish you can remove it, because it will not make any difference to our discussion from here onwards.

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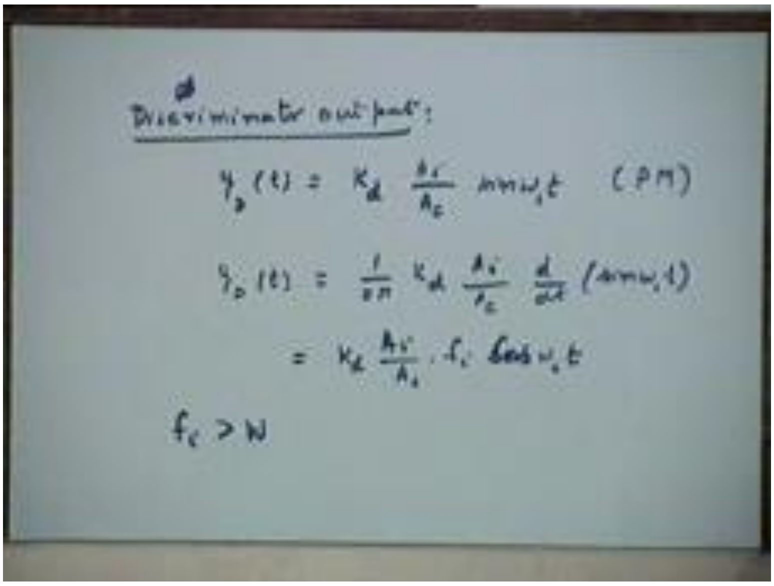

So, what will be a discriminator output, so you consider phi of t as given by this, so a discriminator output by discriminator I mean frequency discriminator. Let us say, let us use the discriminator in a more general size; it could be phase discriminator or frequency discriminator. For the phase de modulation, we will call it a phase discriminator for the frequency demodulation we call it a frequency discriminator, so it is an ideal discriminator. So, that which can for the phase modulated signal y sub d t would be proportional to the instantaneous phase deviation.

So, it will be the detector constant, time instantaneous phase deviation that is an instantaneous phase deviation. So, k sub d into A sub i upon A sub c sin omega, so as you might expect the output contains the interference of course, it will come along with the modulation, along with the modulating signal m t, but I am ignoring the m t here just looking at the effect of interference. How will the interference appear at the output of the demodulator and I am doing this in the absence of the modulation, in the absence of just looking at the effect of the interference alone.

Since, there was no message signal present; obviously, there is no message signal present at the output. But, output should not have been 0; instead of it is 0 there is a signal present, something coming from the interference a signal coming at frequency omega sub i. So, this amplitude it is strength depends on the ratio A sub i upon a sub c and if A sub i is small which is what we assume this will be a small interference, the discriminator output will show a spurious signal which is not present in the desire input in this portion.

What will happen, so this is for the phase demodulator for the PM case, what will you see in the FM case. You will see the derivative of this an ideal discriminator will produce 1 by 2 pi K sub d A i by A c into d by d t of sin omega i t etcetera.

Student: ((Refer Time: 40:21))

That is a good question, ((Refer Time: 40:30)) how do tan inverse disappear.

Student: ((Refer Time: 40:35))

So, basically we are ignoring which term.

Student: ((Refer Time: 40:51))

We are way using the approximation the tan inverse theta is approximately theta for small values of theta tan theta is theta and then for tan inverse theta is also a ((Refer Time: 41:11)) tan inverse sum x is also equal to x, basically we are using that approximation. So, this becomes K sub d A sub i upon A sub c these an additional term constant here, the additional constant is this is 2 pi f i 2 pi will cancel into f sub i sin omega it cos.

So, one interesting observation that we make here is whereas, the output amplitude for the output amplitude interfering signal at the discriminator output is constant for the case of the phase demodulator, it is a frequency dependent function for the case of frequency discriminator. For the frequency demodulator, that is the larger the input frequency difference due to omega c and the interference signal, the larger the amplitude of the interference signal at the discriminator output, the smaller it is the smaller the amplitude.

Of course, the values of f sub i's, which are greater than the bandwidth of the message itself will be of no consequence is not it, f i greater than w are of no consequence.

Because, then it is you can put a low pass filter at the output of the discriminator to remove any higher frequency components, because the message signals contains signals only up to minus w to plus w.

So, if the interference signal exist at the frequency f i greater than w that will be anyway removed. You do not have to worry we only have to worry about f i's value of f sub i's which are between minus w to ((Refer Time: 43:04)) basically f i less than w this are of no consequence.

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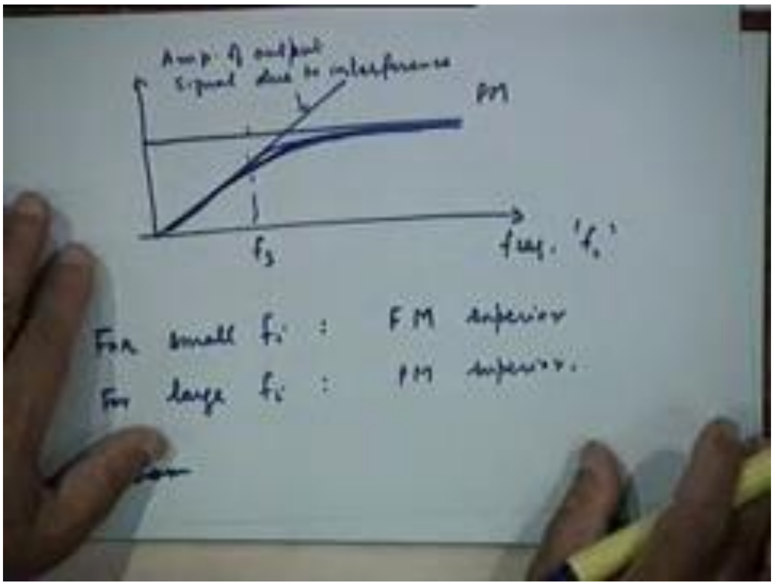

So, what do we see suppose I summarize this behavior pictorially what we are saying is the following. If I plot the amplitude of the discriminator output signal which can be describe to the interference present at the input, that is due to the interference versus input frequency the input interference frequency, that is I am talking about f i the interference frequency f sub i. For the case of PM it has no dependence, that is a constant value, what is a value of this constant this K sub d into A i upon A c that is the behavior for phase demodulator.

For the FM it is a function like this, the amplitude of the output is frequency dependent for smaller frequencies the amplitude is small for larger frequencies the amplitude is large. So, FM seems to have an advantage as long as your f i is a less than certain value. But, if f i is more than certain value then PM as the advantage, for higher the interference frequencies PM has an advantage for lower interference frequencies FM has an advantage.

Student: ((Refer Time: 45:07))

That is why it is a sinusoidal interference, so as long as omega i is close to omega c FM is an advantage. For omega c plus omega is close to omega c FM has an advantage, omega c plus omega e far away from omega c PM at an advantage of course, omega i or f sub i greater than w is of no consequence, we are talking of all this values lying between 0 to w.

So, for small f i FM is superior for larger values of f sub i PM is superior; however, it is something that you know we would like to get the best of both the words if possible. So, how can we do that, that is we would not like to we will like to attain the advantage of FM signal, which you get for the lower f i frequencies, but not never exceeds let us say what your phase modulator can give you.

So, I do not I want to may be except this curve I do not want to take this portion of the curve. So, what can I do something, so that I live with this advantage, but never allow the output amplitude to exceed, what would come in the case of PM signal. Indeed that can be done, but one has to use a filter at the output of the discriminator. So, that you get this kind of a behavior, if you use such a filter which is called a de emphasis filter to emphasize larger frequencies.

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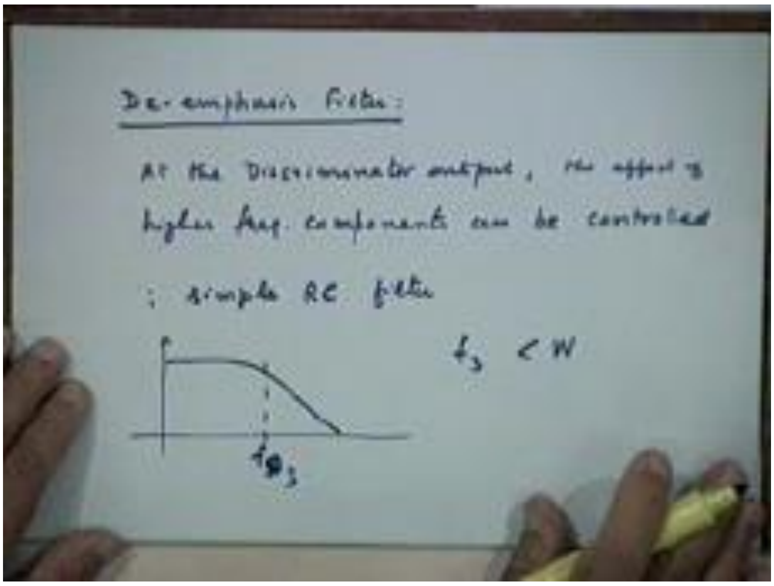

So, if we take a fresh page, so we use a filter ((Refer Time: 47:45)) let us say a low pass filter, basically what we want to do, we want to make sure that as the amplitude increases with frequency, the filter will reduce the filter response will be smaller and basis proportionately. So, the FM discriminator response is increasing like this, so I will allow the filter response to decrease linearly in opposite direction, so that this becomes a constant, the product of these two become a constant.

Such a low pass filter will be which what we call it de-emphasis filter, to de emphasize the higher frequency components present due to the interference, it de emphasize the higher frequency component. So, by using the de-emphasis filter at the discriminator output, the effect of higher frequency components can be controlled, so typically use a simple low pass RC filter.

Student: ((Refer Time: 49:12))

Very good of course, it will, so we will have to do something about that, but let me just come to that in a few minutes. So, you one typically use a simple RC filter of this kind, the frequency response is like that choose the 3 dB cut off frequency let us call this some value f sub 3 ((Refer Time: 49:40)) which is somewhere here. So, that when you combine this linear response with this linear response this becomes a direct response like that, the combination of this about this filter response gives you an overall response like this.

And obviously, the value of f sub 3 would be less than w, because we are anyway considering frequencies less than w. So, choose an appropriate value of f sub t the cut off frequency of this filter, so it will ((Refer Time: 50:24)) use 3 it should simply say f sub c cut off frequency, that is the reason perhaps I think will let it ((Refer Time: 50:35)). So, f sub three it is less than w, choose an appropriate cut off frequency less than the bandwidth of the message signal and ((Refer Time: 50:45)).

But, yes as ((Refer Time: 50:48)) pointed out there is problem here, the problem is that if the input signal has a bandwidth larger than this cut off frequency, these higher frequencies will get attenuated, we do not want that to happen you want the output to be replica of the input, the filter should not distort the message signal itself. So, what can we do about that, basically what are we doing we are attenuating at the discriminator output the higher frequency components it may be present in the input signal, input message signal.

So, to take care of that the simple thing that you can do is to put a compensation at the transmitter itself, you pre distort the message keeping an account taking to account the fact that we are going to do this attenuation at the discriminator output. So, you have a filter before you carry out your FM modulation, frequency modulation you have a free emphasis control which has the reverse or opposite characteristics of that of the de emphasis control.

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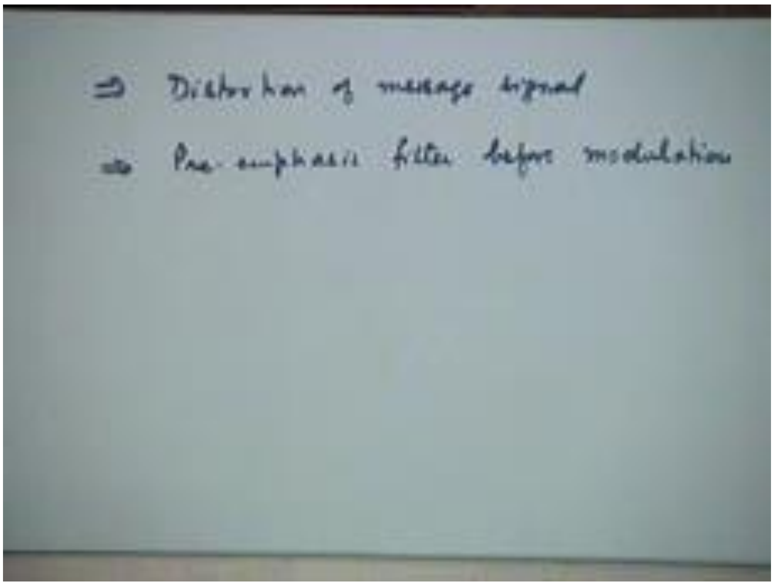

So, to take care of, so this will lead to distortion of the message signal and which can be avoided by using a suitable pre emphasis filter at the transmitter before modulation, so for example, we will start from here next time.

Thank you very much.