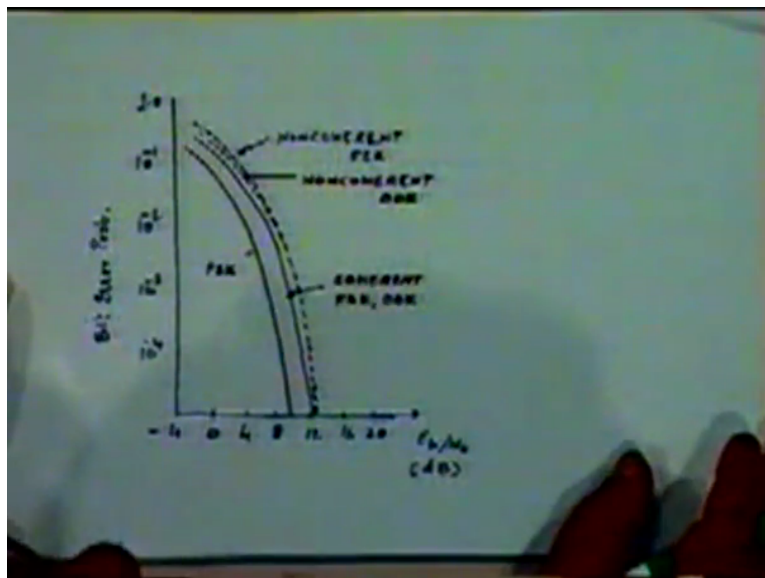


**Digital Communication**  
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**Module 01**  
**Lecture 29**  
**Performance of Non-coherent FSK and DPSK**

So will start with the performance of non-coherent FSK if you remember so far we have looked at the performance of antipodal signaling implying coherent (FSK) coherent PSK and then that of coherent as well as non-coherent On Off Keying and also coherent FSK. The coherent, what is the relative standing of the various modulation schemes that we analyze so far, PSK gives you the best performance because its curve is the lower most. If I can show you those curves that we have discussed right, that is the performance curve showing the bit error rate with error probability versus  $E_b/N_0$  and this is the curve for PSK, it is a lowest and gives you the best performance right.

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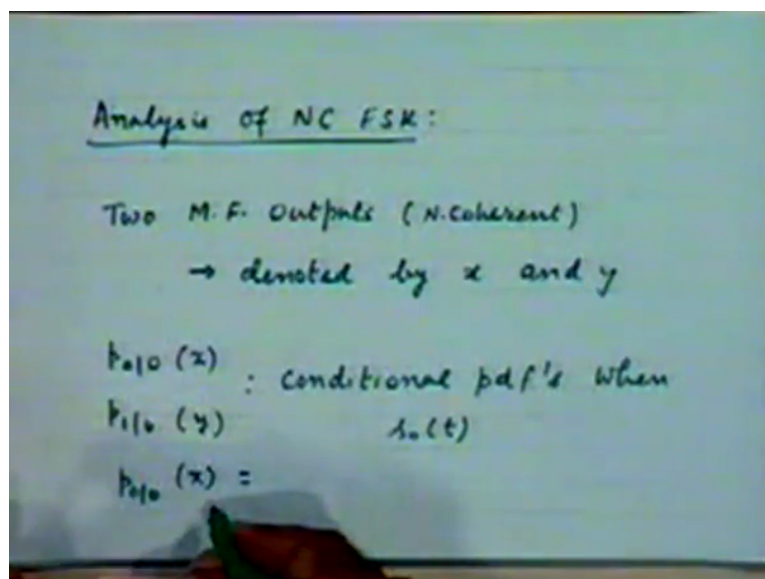
For a given  $E_b/N_0$  it gives you the lowest bit error rate, the coherent FSK and On Off Keying waveforms are effectively 3 dB inferior to that of a PSK waveform, any orthogonal modulation scheme coherent modulation binary coherent modulation scheme (orthogonal modu) is 3 dB inferior right 2 PSK coherent PSK and then it is in the non-coherent On Off Keying it's a followed dB of go down as compare with its coherent performance right. So that is a relative standing of various modulations schemes we discussed today so far and today we

will take up we discussed coherent On Off Keying and non-coherent On Off Keying, coherent FSK and we will discuss non-coherent FSK alright.

Now coherent On Off non-coherent On Off Keying we kind of did not complete the analysis in the sense that I did not give you final expression for the error probability because of the fact that we have we had to leave with slightly messy situations into use involving integrals of the tails of the Rayleigh and the Rician distribution functions but nevertheless it can be done and the results have been plotted over here. Surprisingly however for the non-coherent FSK case we can get a very simple closed form result in spite of the fact that we are still dealing with (non) and that is Rician and Rayleigh Distributions right.

So there is a slight difference in analysis here which will like look at.

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So lets take up analysis of non-coherent FSK, now I dint I have to remind you what non-coherent FSK de-modulator looks like ultimately you have two non-coherent matched filters one matched to each of the two waveforms which you are transmitting for zero and one right, this two waveforms correspond to two different frequencies or could be any set of orthogonal waveforms because for present purpose we are regarding all orthogonal modulation schemes as equivalent to an orthogonal FSK scheme right.

So essentially we have two matched filter outputs to deal with right and these matched filters are going to be non-coherent matched filters so let me write non-coherent matched filters and let me denote the outputs to be  $X$  and  $Y$ . So lets denote by, so matched filter 1 produces

output  $X$  matched filter 2 produces output  $Y$ . Now how many PDF's we have to deal with now, let's see. Suppose we are transmitting a zero right and let's say if you transmit the zero the matched filter which produces a output  $X$  contains the signal whereas the matched filter which produces  $Y$  is orthogonal to that center right.

So  $X$  output is therefore due to the signal and  $Y$  output signal and noise and  $Y$  output is due to signal only noise right, you have two channels the top channels produces  $X$  and it is a result of a signal as well as noise present at its input. Well the same signal is also presented to the lower filter ok. So like I was saying we output  $X$  due to the top matched filter is due to the signal plus noise when the let's say the signal was zero signal transmitted was zero and corresponding to the top matched filter and therefore contains a contribution both from the signal as well as from the noise.

Whereas the corresponding bottom filter matched filter although it has a same signal at its input we will only produce a contribution due to noise right, this is what we are saying. So if I denote the density function of  $X$  I will write it as a conditional density function, I will write it to be very explicit something like this  $P_0$  given 0 and  $P_1$  given 0 by this I mean the density function at the output of the top matched filter this first index refers to the fact that we are looking at the output of the top matched filter and here also this first index refers to the fact that we are looking at the bottom matched filter which is let's say matched to the waveform corresponding to one right and obviously the variable involved here is  $X$  and the variable involved here is  $Y$ .

So these are the two conditional density functions which come into the picture when a zero is transmitted alright when you transmit the waveform  $S_{0t}$  corresponding to transmission of a zero let us say alright and what will be the nature of these two density functions? What can you say about  $P_0$   $P_{0|0}$   $P_{1|0}$   $X$ , what is the density function? We are having a non-coherent matched filter right, when I say non-coherent matched filter it implies all those things right. I am looking at the final combined output that is if it is a passband implementation, it is a matched filter followed by envelope detector, if it is a baseband implementation you first resolve it into I Q form and take the  $I^2 + Q^2$ .

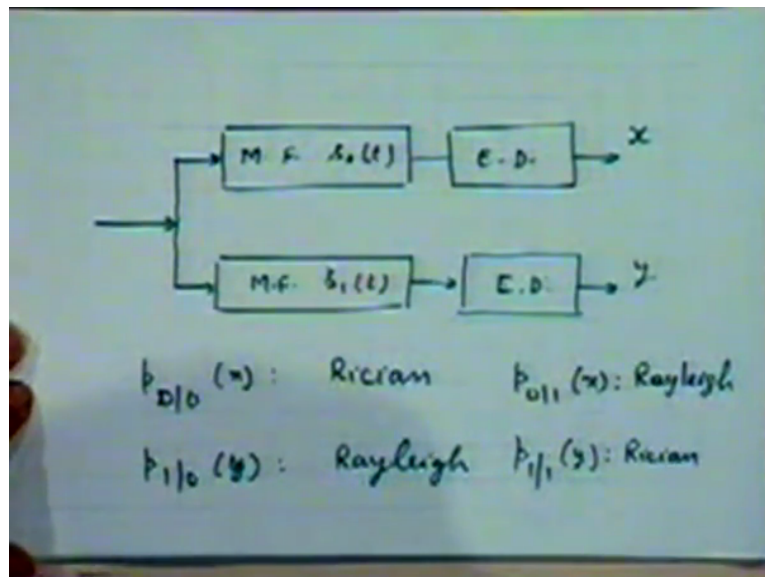
Student: (0)(8:48)

Professor: We have two filters two ok, I think you have not revised what you have been studying so far (0)(9:01) what is an orthogonal, what is a non-coherent matched filter

receiver for orthogonal waveforms, binary orthogonal waveforms? You require two different matched filters, one matched to each of the two waveforms, this is a different thing from the I and Q outputs static reduced from any given waveforms right. So if you are doing a baseband implementation you will be producing two sets of I and Q outputs, one corresponding to well you will produce only one set of I and Q outputs I am sorry, but you will be doing two different sets of matched filtering at baseband on this two I Q outputs, this complex outputs.

So we are looking at the two matched filter outputs after finding it they have been combined by square root of I square plus Q square of which corresponding to the first output as well as you do the second output.

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I don't have a figure here but let me recapitulate for you because obviously you have forgotten what we are talking about, we talking about matched filter for  $S_0$  t right followed by envelope detector for this, this output denoting by X same thing goes to match filter for  $S_1$  t corresponding output of the envelope detector is being denoted by Y. This is the situation we are talking about, now this matched filter envelope detector combination can be realized either at passband or at baseband right.

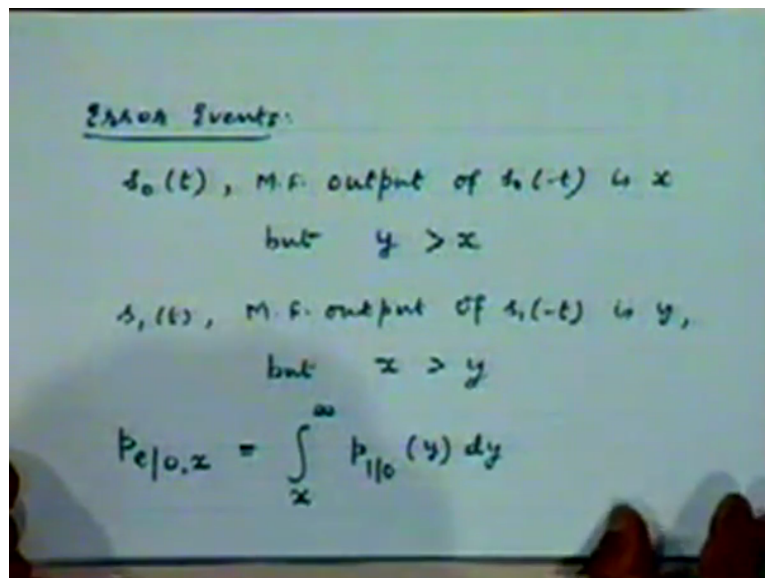
I am not going to the details of that but finally you are producing an output X and an output Y one corresponding to this, one corresponding to this both working on the same input right. So when the input contains  $S_0$  t this density function will be, what will be this density function? No because it is due to signal plus noise, it will be Rician, is that ok? Whereas this one will

produce an output which is dependent only on noise, because  $S_0$  and  $S_1$  are taken to be mutually orthogonal right.

Therefore  $Y$  will be a (Rician) Rayleigh Distributed random variable ok. So therefore let me write down here your (output) I am denoting this is 0 by 0  $X$  is therefore Rician Vikram is your doubt clear? At  $P_1$  given 0 of  $Y$  is Rayleigh does everybody understand this notation and the logic on the basis of which you have written this results. Similarly we can now think of the other situation where you transmit the waveform  $S_1(t)$  right. In that case this would be Rayleigh and this would be Rician. So basically we are looking at two outputs and this case you will have  $P_0$  given 1 that is your transmitting  $S_1(t)$ , this will be Rayleigh and  $P_1$  given 1  $Y$  will be Rician.

The roles will just get an exchange with each other. Is that ok everybody? Any questions about the model underline model based from which will have to do the analysis, this is underline model ok.

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Now lets look at the error events that is the situation under which errors will take place. One type of error corresponds to the situation that  $S_0(t)$  ok before to appreciate that however we processing  $X$  and  $Y$  to make our decisions 0 and 1, we are comparing them right or alternatively we could say we are just directly comparing  $X$  and  $Y$ . If  $X$  is greater than  $Y$  we declare that zero is transmitted if  $Y$  is greater than  $X$  we say that one is transmitted alright.

So error events will correspond to the following situation, you are transmitting  $S(t)$  and the corresponding matched filter output of the matched filter the top matched filter  $S(t) - t$  is  $X$  but  $Y$  happens to be greater than  $X$ , that will be another event right, you transmitting  $S(t)$   $X$  should be greater than  $Y$  right, if you are making correct decision but if  $Y$  happens to be greater than  $X$  that will cause an error right. Similarly if you are transmitting  $S_1(t)$  and the matched filter output of  $S_1(t)$  of the lower matched filter is  $Y$  but  $X$  is greater than  $Y$  these are two possible (15:06)ok.

Lets calculate the probability of each of these two errors, so I am really conditioning the error events here this is only one of the error events is being conditioned by the fact that you are transmitting a zero right and lets say you are also conditioning the fact that the output of the first matched filter is the top matched filter is  $X$  and now given that this value is  $X$  what is the probability that the other filter produce a value  $Y$  which is greater than this value right. So I am assuming that the top filter has given you some value  $X$  which self is also a random variable we must remember that, the value  $X$  itself is a random variable some waveform having to leave it is mysterious.

Lets say it produces an output  $X$  it should have been  $Y$  should have been (greater than) less than  $X$  but it turns out to be less than  $X$  oh sorry, greater than  $X$  and therefore error probability corresponding to this event is which density function comes to the picture? We are looking at the probability that  $Y$  is greater than  $X$  so we are looking at the density function of the lower output given that zero was transmitted  $\int_y dy$  from  $x$  to infinity right and suppose I want to remove this condition on  $X$ , you can think of this as a function of  $X$  it's a random variable right then you have to take the expected value of this with respect to the standard waves right.

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$$\begin{aligned}
 P_{e|0} &= E[P_{e|0,x}] \\
 &= \int_0^{\infty} P_{0|0}(x) \left[ \int_0^{\infty} P_{1|0}(y) dy \right] dx \\
 P_{e|1} &= \int_0^{\infty} P_{1|1}(y) \left[ \int_0^{\infty} P_{0|1}(x) dx \right] dy \\
 P_e &= P_{e|0} + P_{e|1}
 \end{aligned}$$

So your  $P_{e|0}$  given 0 after removing the (conditional) condition on  $X$  will be we can say it is expected value of  $P_{e|0}$  given  $X$  and that will turn out to be integral from zero to infinity, why? We are dealing in only Rayleigh and Rician variables which have only positive values right. Is that ok? Take this multiply it by this density function  $X$  with error of  $x$  (17:44) because we taking the expected value of this value with respect to  $x$  ok.

So that is your expression for  $P_{e|0}$  and it is quite obvious by symmetry that we can write the same kind of expression for the other kind of error event that is likely to happen right, what will be the corresponding expression? Only the density functions will get exchanged this will become  $P_{1|1} X y$  to infinity  $P_{0|1} X dx dy$  this will be y ok also it is quite obvious that your final error probability  $P_e$  if you are assuming that 0's and 1's are equally likely right will be essentially equal to either of this two things right, because actually it will be half of this plus half of this and if they are both going to be equal because of the symmetry of the problem it is obvious that they will be equal, either of them will be equal to the final error probability on the average error probability.

So we need to basically compute anyone of this expressions either this or this. Now we already know the nature of this density functions, it turns out that this is really very simple to evaluate in a closed form and lets that up. I will take up this top expression  $P_{e|0}$ , this density to make things convenient for us will take the density functions in a normalized form right. You know this is going to be Rayleigh and this is going to be Rician.

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$$p_{1|0}(y) = y e^{-y^2/2} \quad : \text{Rayleigh}$$

$$p_{0|0}(x) = x e^{-(x^2 + \lambda^2)/2} I_0(\lambda x)$$

$$\lambda = \frac{A}{\sigma}, \quad \lambda^2 = \frac{2E_b}{N_0}$$

$$p_{c|0} = \int_0^\infty p_{0|0}(x) \left[ \int_x^\infty y e^{-y^2/2} dy \right] dx$$

$$= \int_0^\infty p_{0|0}(x) e^{-x^2/2} dx$$

So will first take the normalized density function of  $y$  which is normalized Rayleigh density function because  $P_1$  given  $0$   $y$  is a Rayleigh density function lets say it is this.

Remember the normalized form of the Rayleigh density function that is what it is. Similarly  $P$  the Rician density function  $P_0$   $Y_0$  normalized form is  $X e$  to the power minus  $x$  square plus  $\lambda$  square by  $2$  into the modified Bessel function of order  $0$  with an argument  $\lambda x$ , where  $\lambda$  was dependent on  $A$  and  $\sigma$   $\lambda$  square being equal to your  $2 E_b$  by  $N_0$  right. So start with this expressions and consider  $P e$  by  $0$  ok, what can you say about this integral? This is  $Y e$  to the power  $y$  square by  $2$   $dy$  from  $X$  to infinity right.  $E$  to the power minus  $X$  square by  $2$  this very easy to evaluate because of the fact that there is a  $Y$  present here, so it is really very simple ok.

And the next part is also very simple because  $P_0$  by  $0$   $X$  is this expression here, now imagine this multiplied by this, what is going to happen? Essentially this  $X$  square by  $2$  is going to become minus  $X$  square, so a kind of change in something here. So happens that by just small manipulation of the expression we can still get Rician density function inside the integral ok and that is what we will see. I will just keep this on top you can see it here ok alright I will keep it like that.



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$$\begin{aligned}
 P_{c10} &= \int_0^\infty P_{010}(x) \left[ \int_x^\infty y e^{-y^2/2} dy \right] dx \\
 &= \int_0^\infty P_{010}(x) e^{-x^2/2} dx \\
 \\ 
 P_{c10} &= \frac{1}{2} e^{-\lambda^2/4} \int_0^\infty \sqrt{2} x e^{-(2x^2 + \lambda^2)/2} \cdot I_0\left(\frac{\lambda}{\sqrt{2}} \cdot \sqrt{2} x\right) d(\sqrt{2} x) \\
 &= \frac{1}{2} e^{-\lambda^2/4}
 \end{aligned}$$

So if I substitute this here and do a little bit manipulation I think it is very easy to check that is why I am keeping this in front of you that we can write this like this,  $2 X^2$  plus lambda square by 4 upon 2 into  $I_0$  as if you have changed  $X$  to square root of 2 times  $x$  and lambda to lambda upon root 2.

If we make this change right and then this product can be written in this form with this factor coming out of the integral because this is independent of  $x$ . Just rewritten the same thing by identifying now this lambda  $x$  as lambda by square root of 2 into root 2  $X$  right. So correspondingly everything else will change, so this will become  $2 X^2$  as lambda by square by 4 and if you take  $E$  to the power minus lambda square by 4 outside you can see that this will reduce to the same thing just a little bit of manipulation or this should be yes you are right, it should be  $D \sqrt{2} X$  there is no need for an extra  $dx$  that was a mistake, fine, very easy to check and what is the value this integral?

It should be trivial, it is a Rician density function, what is the value of any integral involving a density function from 0 to infinity? One, why were you thinking so much about it, no need to think, that is the whole idea, the whole idea of this exercise was to write this as a Rician density function with a different set of arguments but nevertheless it is a density function and the integral under it is 1, so the answer is simply half of  $E$  to the power minus lambda

Student: (0)(25:03)

Professor: Have you taken into account this over here? Let's check yeah I think this should be 2, you are right, (because) that is how you get lambda by root 2 here, thank you.

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$$P_e = \frac{1}{2} e^{-\frac{1}{2} \left( \frac{E_b}{N_0} \right)} \quad \text{NCFSK}$$

$$P_{e|b} = \frac{1}{2} e^{-\lambda^2/4} \int_0^{\infty} \sqrt{x} e^{-\frac{(2x^2 + \lambda^2)/2}{2}} \cdot I_0\left(\frac{\lambda}{\sqrt{2}} \cdot \sqrt{x}\right) d(\sqrt{x})$$

$$= \frac{1}{2} e^{-\lambda^2/4}$$

So we have a very simple close form expression for in terms of your  $E_b$  by  $N_0$  what will this turn out to be?  $P_e$  sub e half of you just invert is  $\lambda^2$  square  $\lambda^2$  square is  $2 E_b$  by  $N_0$  right, so half of minus half  $E_b$  by, that is the expression for the probability of error for non-coherent FSK. So we have a close form result here, not even requiring Q functions, right we don't even require to use Q functions here, it is an exact close form result and that is how it looks like, this blue dotted curve over here and as you can see that as you go from coherence to non-coherence you have a performance loss of about a dB also right.

Bringing between 0.8 dB to what, 1 dB, it is not significant in the case of On Off Keying FSK ok, any questions? So kind of completes the performance analysis of most of the basic kinds of binary modulation schemes that we have discussed so far, both coherent as well as non-coherent. Now at this stage I will like to remind you that for PSK we have not considered a non-coherent situation. Now you may well ask, do we have an equivalent to non-coherent demodulation of PSK, as we said as we discussed earlier it doesn't make sense to talk like that right. However we may have a situation where will transmitting PSK and where we are not able to estimate the phase of the carrier reliably at the receiver.

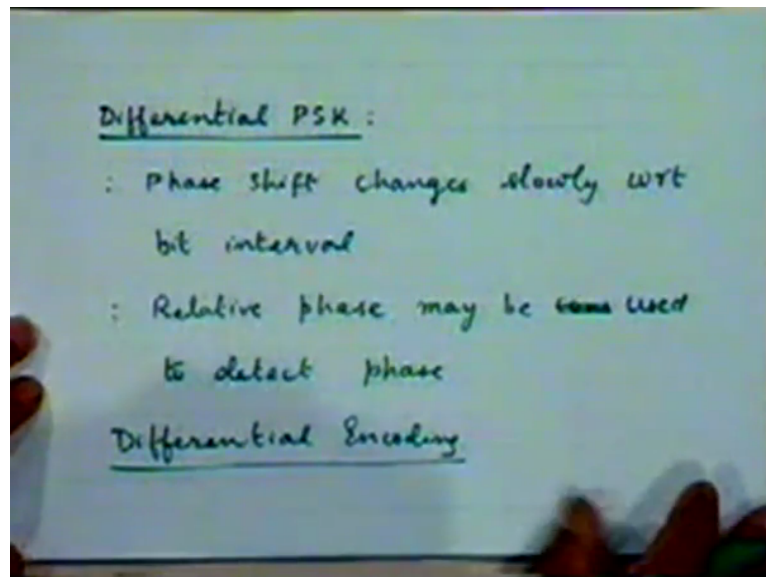
So as to completely compensate for the unknown phase and thereby be able to do coherent demodulation right. Now if the situation is very bad, if the phase is totally unknown and its varying very rapidly and if you know that this is going to happen, it is advisable not to use

PSK right. However the motivation for using PSK is very large as you can see if you are using a binary modulation scheme there is a 3 dB power advantage to have to get, so if possible will try to use PSK wherever possible. So therefore we are now looking for situations where PSK maybe employed even in the situation where phase is not exactly known and but the situation is not so bad that we cannot do anything about it.

And one such situation is, where the phase is varying only with time, right it is a phase variation and this unknown phase is there is constantly changing but the change the rate of change is very slow that is with respect to a bit interval or several bit intervals you may expect the unknown to be same, it is constant.

In that case even though the phase maybe unknown we can exploit the fact that the relative phase bit over successive bits will not be different right and we can make use of this fact that de-modulation of certain kinds of PSK waveform ok, making use of fact that the relative phase in successive bit intervals will not be changing even though the phase is unknown and therefore we can look at the relative phase shifts and thereby carry out the de-modulation.

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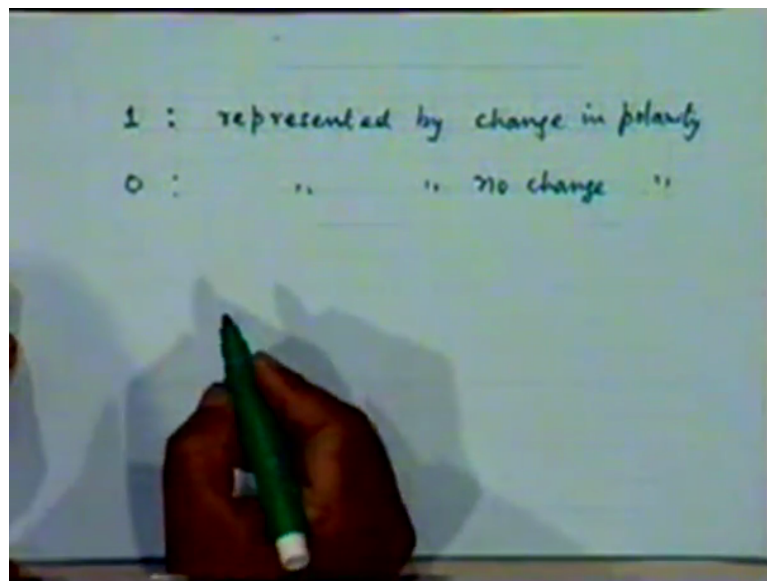
Now the differential phase shift keying is designed to take care of this kind of a situation. So it is something like a non-coherent version of PSK right, it is not strictly non-coherent but because information is in phase but we are not exploiting (0)(30:07) information is in the relative phase rather than in the phase itself, it is a differential phase so this is used where phase shift changes rather slowly with respect to bit interval that is over a bit interval or over

a few bit intervals we can expect the phase to be nearly constant. So under this situation we note that the relative phase is maybe taken to be constant, not constant maybe used to yes.

Because the phase is constant you can (choose) use a relative phase this is for the purpose (()) (31:22) ok. Now we don't exactly do things like this, it is not usually done in this way what we do is, if the data itself is encoded in phase change is rather in absolute values of the phases, rather than detecting phase changes at the receiver the data itself is encoded in terms of phase changes right and that is called differential encoding. You are familiar with differential encoding in the context of baseband signaling waveforms that we discussed earlier (())(32:02) binary waveforms and things like that.

But anyway you can see that there will be similarity with what we are saying here and those things.

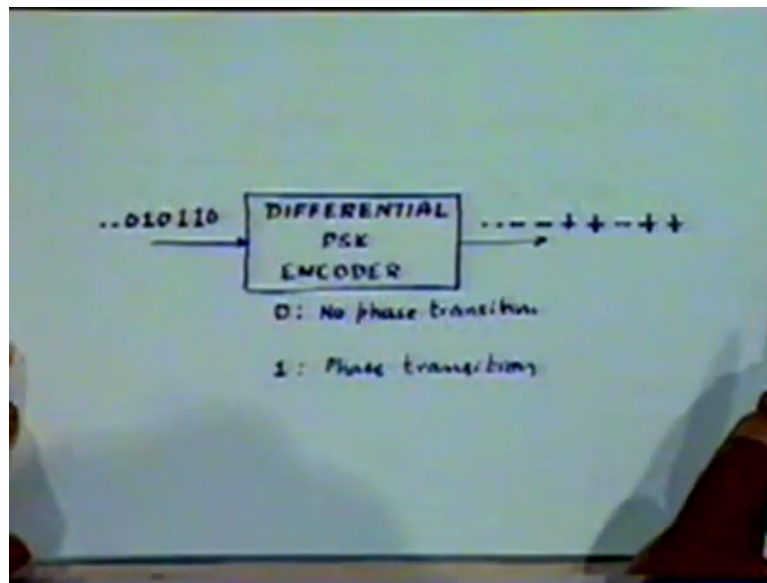
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So lets say differential encoding in operation like this, suppose we want to transmit a 1 this maybe represented now not by an absolute value of some phase value say zero degree of as we have say that if I reduce as earlier, but it maybe represented by a change in polarity, whatever phase you are transmitting earlier you add to 180 degrees to that at  $\pi$  reduce to that, that is it change its polarity or there is a phase transition you can say, it is represented by a phase transition taking place at the transmitter.

Similarly a zero maybe represented by no change in polarity, no phase transition ok. So if you were to look at the differential encoder you may have a situation like this.

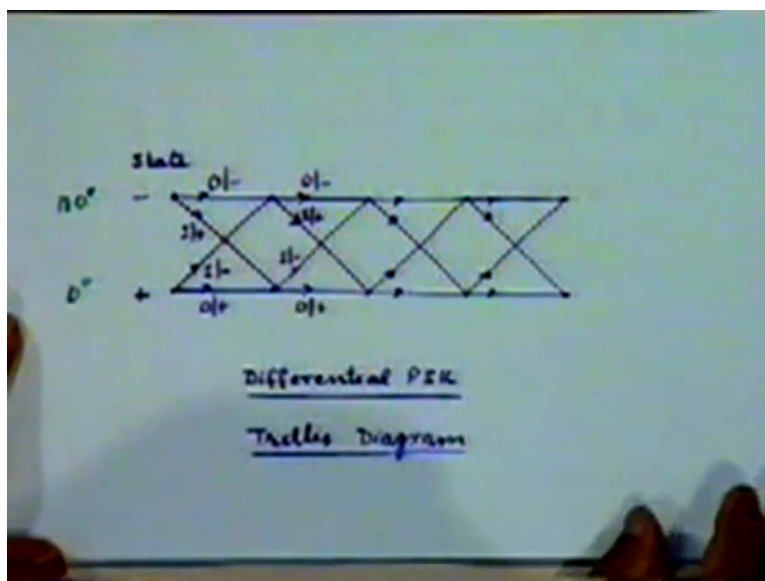
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This is your differential PSK encoder you have lets say bit stream like this coming in and when you are transmitting the zero lets say the previous polarity was negative the new polarity also will be corresponding to this zero will go as the same polarity, when you have a one the polarity will change to plus and you have a zero again it will remain plus because no phase transition is taking place. Next one is a change in polarity, next one again there is a change in polarity and so on right, is that right or?

This corresponds to the zero, this corresponds to right, first minus in a some reference and this zero implies no phase change this one will imply a phase change polarity change, a zero implies no change, one implies a change, this next one implies again a change, this zero implies no change alright.

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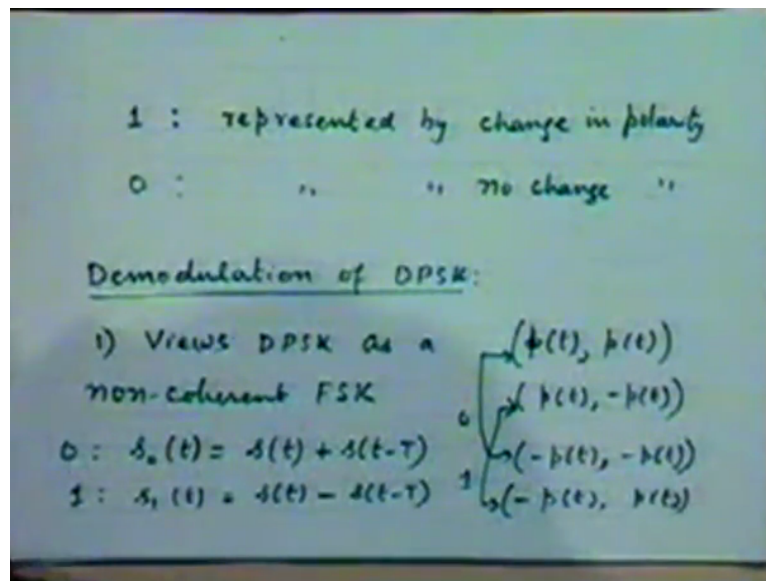


So that is how you do, zero implies no phase transition, one implies (a phase) offcourse could be the other way around without any, you could express this also in terms of a trellis diagram, were the straights are the polarity whether it is negative polarity or plus or it could be zero degree phase or this could be lets say if this is negative we could say 180 degrees this is zero degree right.

So when you are here and a zero comes along you stay there in the same state and when you are here and a one comes along you change the polarity and so on right. How do you de-modulate such a waveform? We look at two different equivalent schemes which gives a very good insight also over the performance of DPSK because if this is going to replace PSK we are interested in looking at whether a not it gives us performance equivalent to that of PSK or not very much worse from it hopefully. So again performance analysis important to appreciate and that will be able to do by first looking at how we can de-modulate a differentially (( ))(36:21) PSK ok.

Because the final waveform is a PSK waveform you can see that right it is described by carrier with 180 degree phase shift or zero degree phase shift right. But now the interpretation of the phase shift is to be taken this, so we have to see them (( ))(36:40) to each other and then decide whether 0 or 1 was transmitted. Basically that is what you have to do, so is clear?

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But we can also look at in a slightly different way and that is what you are going to look at right now for the purpose of seeing how we can de-modulate it nicely. Offcourse there will be one problem also with DPSK, we have seen that kind of problem before, what will the problem? Error propagation, right, and if there is an error in one particular bit then that will cause a further error in a successive bit ok. But will talk about that separately. Now I look at two different kinds of baseband de-modulators, we can also look at the corresponding passband de-modulators I will leave that as an exercise for you to do yourself.

I will look at two types of de-modulators both at baseband ok. Now corresponding passband versions please do it yourself. The first method then this is very interesting, views DPSK as PSK is an antipodal waveform right PSK not DPSK , PSK is antipodal waveform DPSK we cannot directly classify is antipodal or any other class, because we are looking at two bits at a time in some sense or we are looking at two bit intervals at a time we are comparing the relative phase over two successive bit intervals.

However if we look at the waveform over two bit intervals at a time right we can see certain pattern, can you see a pattern in the waveform set we are finally transmitting, if you look at those waveforms as waveforms of duration  $2T$  rather than waveforms of duration  $T$ , can you think of can you see some pattern happening here taking place here? Ok let me try, suppose you are transmitting a waveform  $P t$  in a particular interval and you are transmitting a zero how will this reflect on the waveform that you are going to transmit again in the next interval? Same waveform right at the same polarity this will be  $P t, P t$  right.

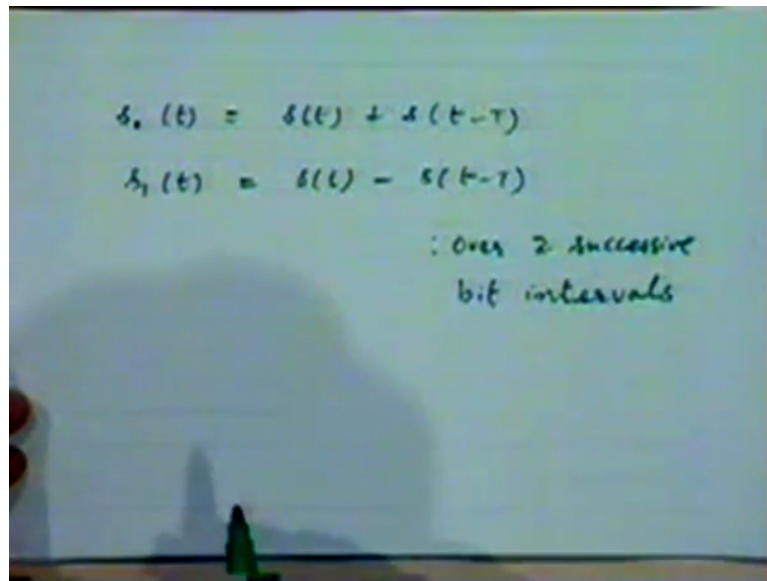
If you want to transmit a 1 what will happen? Minus  $P \cos(\omega_c t)$ , right if this is to start with minus  $P \cos(\omega_c t)$ , for a 0 will be minus  $P \sin(\omega_c t)$ , and a 1 will be plus  $P \sin(\omega_c t)$ . Now do you see some pattern? These are the two zero waveforms right, these are the two one waveforms, do you see any patterns? You have seen this kind of thing before that is why I am asking you. No that is not QPSK, we are looking at this as one try to look at this as one waveform, as if this is one complete waveform over duration of  $2T$  right this is one complete waveform over a duration of  $2T$ , they are mutually orthogonal to each other, a 0 waveform and a 1 waveform are mutually orthogonal to each other right we appreciate that.

That is if I multiply them and integrate over  $2T$  the result will be zero if I take a zero waveform and a corresponding in fact any one waveform right. Let's say take this and this or take this and this, this will become  $P^2 \int_0^{2T} \cos^2(\omega_c t) dt$  minus  $P^2 \int_0^{2T} \sin^2(\omega_c t) dt$  which is 0 and you can check similarly for any other pair. So basically this method of de-modulation of DPSK that we are going to discuss now, views DPSK in this particular manner, DPSK as a non-coherent this I have to explain of course but essentially as an orthogonal set of waveforms and the generic name for the orthogonal class of waveforms is FSK non-coherent FSK waveform with 0 denoted by let me denote the 0 waveform actually I should have written  $S_0(t)$  because I am using  $S(t)$  throughout now.

What is your 0 waveform corresponding to 0 now? It is some waveform  $S(t)$  plus the same waveform delayed by  $t$  seconds over the next  $t$  seconds right. Similarly the one waveform is  $S_1(t)$  equal to  $S(t) - S(t - t)$  right. But these waveforms are to be regarded as waveforms now over two successive bit intervals any two successive bit intervals right, they are defined over two successive bit intervals.



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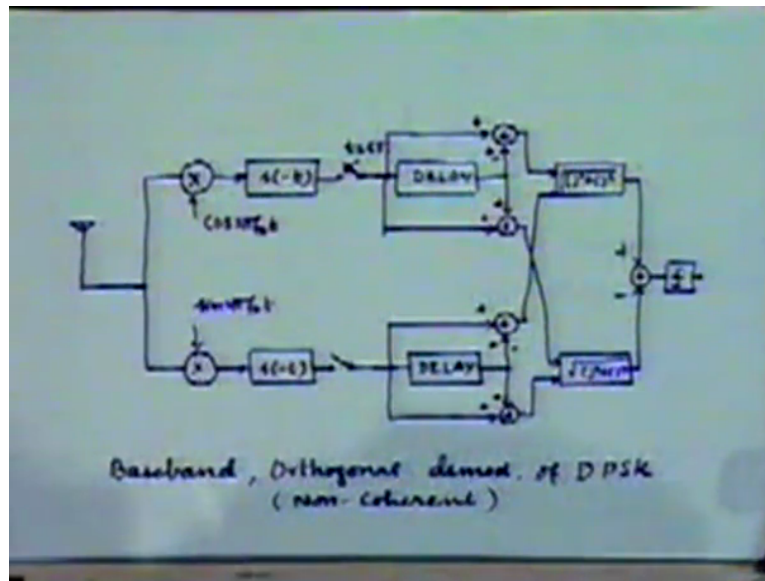


The image shows a whiteboard with handwritten mathematical expressions. The first line is  $s_0(t) = s(t) + s(t-T)$ . The second line is  $s_1(t) = s(t) - s(t-T)$ . To the right of these equations, there is a note: ": Over 2 successive bit intervals".

If you rewrite it for clarity  $S_0(t)$  is  $S(t) + S(t-T)$   $S_1(t)$  is  $S(t) - S(t-T)$  over two successive intervals. So to detect this what should we do? How do we do to check orthogonal waveforms? Two matched filters and how do we detect when phase is not known? Non-coherent detection right.

So the obvious answer would be if you view DPSK in this particular way you can design a non-coherent orthogonal receiver and use that for the de-modulation of DPSK however we can simplify implementation somewhat because we can construct the matched filter corresponding to  $S_0(t)$  by just first constructing the matched filter corresponding to  $S(t)$  because this is also the same waveform by sampling the matched filter outputs at every bit interval I am combining them appropriately we can construct the corresponding output right. So that is the basis for this realization here.

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Look at this carefully you don't have to draw it because anyway I am going to distribute your if you want you can draw it but it will take some time better to try to understand what is happening, I think you can draw it yourself afterwards . This diagram I will I think I have already given you the notes corresponding to this if not then I will be giving you this picture very soon. So this is a baseband implementation as said so you first produce the I Q outputs right, so you got a complex waveform representation, this is a matched filter corresponding to  $S_t$  right.

Now I want to produce from this two matched filter outputs, one corresponding to  $S_{sub 0}$  other corresponding to  $S_{sub 1}$  and what is the relationship, in one case I have to add this after delay of  $t$  seconds in the other case I have to subtract them after delay of  $t$  seconds because if the input waveforms are added the corresponding matched filter outputs will also be simply added. So what I do is sample at  $t$  equal to  $L T$  is going here as well as here the same thing is also available with a delay so if I subtract the two I get the matched filter output corresponding to lets say this waveform if I add the two I get the matched filter output corresponding to (this waveform).

Ok, this should be plus this should be minus, right because  $S_t$  minus  $S_{t-t}$  right. Is it fine? So this produces the real component of the matched filter output for  $S_0$  real component for  $S_1$  similarly imaginary component again it should be plus minus, yeah it really doesn't really matter I agree with that but nevertheless if you want to go by the equation you can change the sign and then the real and imaginary components are combined in the usual way

right and then you finally look at, compare the two and decide whether a 0 or 1 was transmitted. So this looks like the right thing to do, do you understand what is happening here? Which stage? Ok, that is a good question but that is a more difficult thing to do than this, right.

He wants to combine do this combining before the matched filter, surely it can be done, no problem conceptually right, but it is more easy to mechanize this implementation than that because that will require you to do to realize an analogue delay line right. You like to delay an analogue signal by one bit interval not a very easy thing to do, whereas if I have discrete sample which I want to look at  $t$  seconds later much easier to do that right. I can store it in the memory and look at it after  $t$  seconds ok. So the mechanization of this after matched filtering is after sampling  $s$  much easier than working with a continuous waveform itself, is you question answered? Any other question?

Now so what we see therefore is that each bit is non-coherently de-modulated as if you are doing a non-coherent FSK de-modulation right. So you would expect the performance to be the same as that of non-coherent FSK or something different, same with a difference, what is the difference? Each pulse is contributing now to 2 bit intervals right, so effectively you getting twice the energy right, yeah, each transmitted pulse the received waveform the matched filter waveform that you are constructing is corresponding to a interval of  $2T$  seconds right, so for a given energy here of the pulse  $S t$  the final energy that contributes to the matched filter output is twice as much right.

Because it is coming from here as well as from here. So if this is producing an early output lets say  $A$  this is also producing an  $A$  so total output becomes energy becomes twice actually amplitude is only root 2 times.

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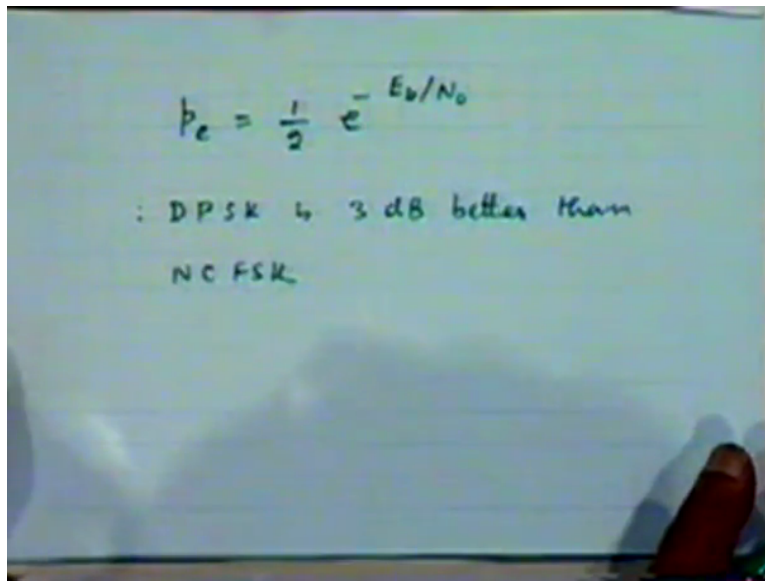
$s_0(t) = s(t) + s(t-T)$   
 $s_1(t) = s(t) - s(t-T)$

: Over 2 successive bit intervals  
: Each pulse  $s(t)$  contributes to two successive bits  
 $E_b$  : half of what that in NCFSK

So the energy is double which is what I want to say here that each pulse each individual pulse  $s(t)$  contributes to two successive bits, therefore if you want a given performance the energy in individual bit interval could be half as much right.

So the required  $E_b$  is half of what it would be in non-coherent FSK, half of that in non-coherent FSK. In other words there is a 3 dB difference in performance between non-coherent FSK and DPSK ok, just like there was a 3 dB difference between coherent FSK and coherent PSK, there is a corresponding 3 dB difference between non-coherent FSK and non-coherent or differential PSK.

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$$P_e = \frac{1}{2} e^{-E_b/N_0}$$
  
: DPSK is 3 dB better than  
NC FSK

So we already have the expression for the performance of non-coherent FSK we just derived it to a morning, that was half of minus  $E_b$  by  $2 N_0$ , so now it will become  $E_b$  by  $N_0$  right. So that is the conclusion DPSK is about 3 dB better than non-coherent FSK, is argument clear?

Student: (( ))(50:41) coding scheme or whatever use a scheme in which you use I mean instead of two three bit intervals and you can keep on

Professor: That is a good thought and it can be done but exactly because you can't easily do differential coding over 3 bits right, but some kind of coding can be done, that is quite true and there are digital modulation schemes in which information about a single bit exists over successive bits, that is what basically what is happening here, in DPSK information about one bit is also contained in the phase of the successive bit right. So if there is a correlation existing between the information about one bit over several bits then the optimal thing to do would be to look at information of all together and then do the de-modulation for each individual bit separately right and there are lot of modulation schemes which fall in the generic class of what are called continuous phase frequency shift keying CPFSK right, which have those kinds of properties and can be made to improve performance over conventional modulation schemes like PSK right.

Infact there is a whole class of new general modulation schemes with much superior performance than the classical one that are known that are usually used. Will see some of them as if time permits but a lot has been a lot has happened in the last decade in the field of

digital modulation schemes and new (( ))(52:21) have come which have brought us closer to the theoretical limits which can be achieved in terms of performance of digital modulation schemes, but we will discuss some of this issues as we (( ))(52:34) thank you very much.