Digital Communication
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Module 01
Lecture 26
Demodulators for Binary Waveforms (Contd)

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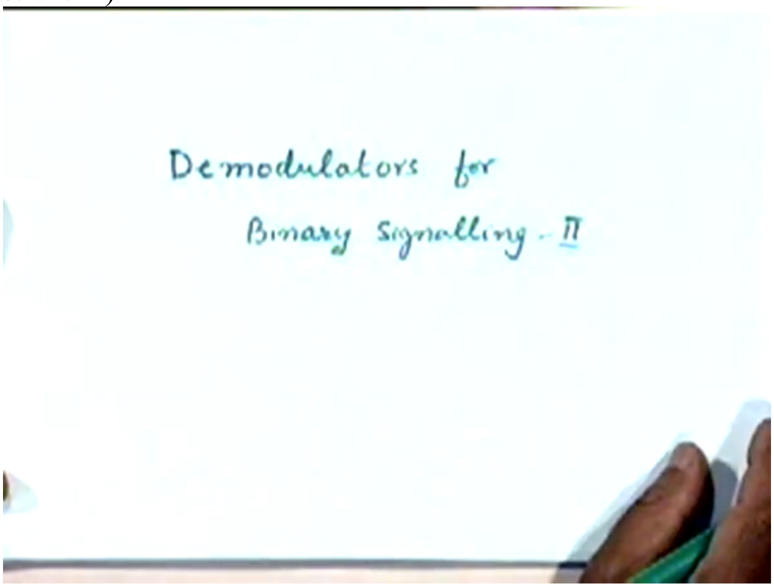

So will continue with our discussion on de-modulators for binary modulation schemes (yeah) maybe one possibility is that we can I think I can try to have it ready by Monday morning yeah no I will try to give you by Monday morning and I think the other thing I will talk about after the at the end of the class ok. So as I said we will continue our discussion on de-modulators for binary signaling just to recapitulate for what you are doing last time we had discussed our coherent receivers for antipodal signals first we discussed a number of possible realizations for it right, starting with baseband realization then passband realization using matched filters using passband and baseband match filters and also a corelator implementation right.

Then went on to looking at the coherent de-modulation of orthogonal signals binary orthogonal signals right and we looked at it very general simple case well the general mathematical result that we looked at was that in this case we need two matched filters a bank of two matched filters, one matched to each of the two waveforms that you are going to use in the orthogonal signaling scheme alright and the overall receiver structure will then have a system by which the incoming signal is match filter using each of this two filters and then mutually compared.

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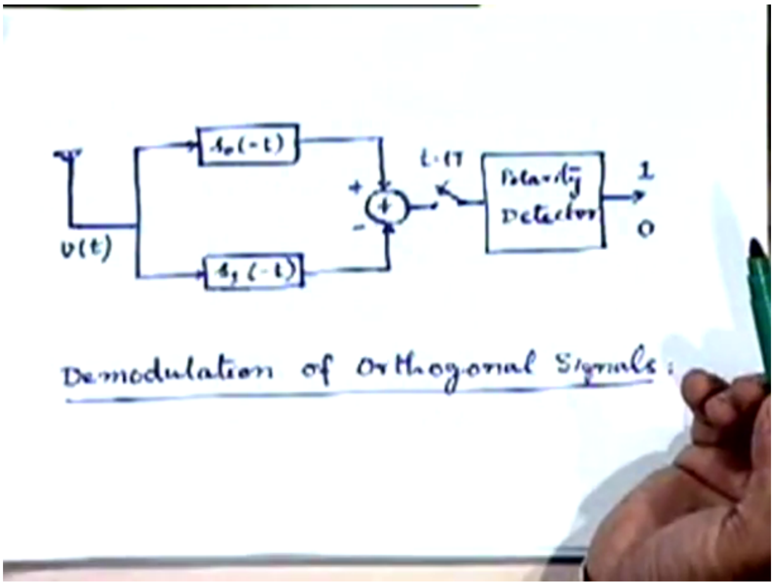

The output the sample output at each signaling instant t equal to L T is compared from both the outputs or a comparison you can even subtract them out and then see which one is larger right. So that was the basic idea and this is a very simple depiction for the case when you are taking your two signals to be real two orthogonal signals to be real for that case when the signals are complex offcourse one can just generalize this picture very easily by just putting a conjugate there right. But one need not do that and I will talk about that a little bit more but let me fully revise for you what we did last time.

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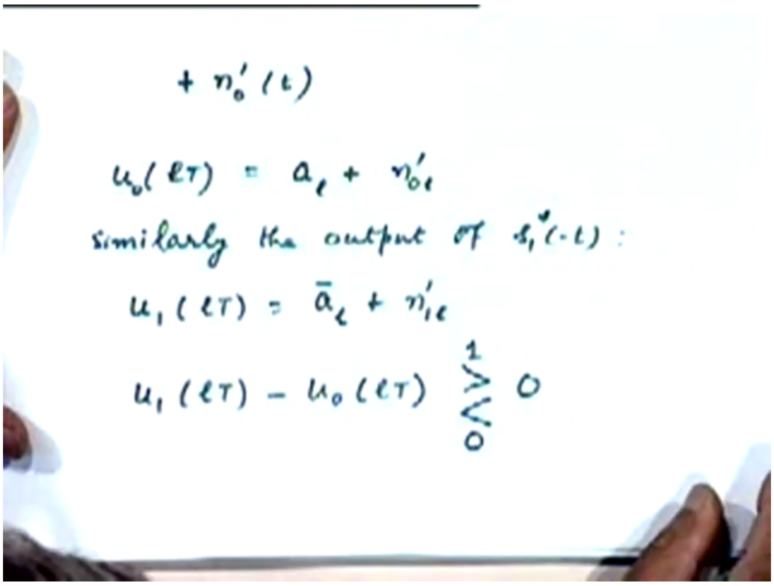

The mathematical equation for this is you compute the output of the first match filter as well as of the second match filter take the difference and depending on which is larger it declare that either a point or a zero was transmitted right.

This was the mathematical equation that we looked at corresponding to that structure. Offcourse you must remember that each of these output quantities U sub 1 and U sub 0 is associated with some noise and therefore there is always a likelihood that occasionally this noise value will disturb or mean they disturb the polarity and make us arrive at wrong decision offcourse we hope that if our signal to noise ratio is reasonably good the probability of occurring such events will be small but we look at the detail performance analysis in terms of error probability a bit later.

So this was what we discussed last time. To continue from here as I said lets first try to look at the general picture when we replace this with complex orthogonal signals right, where the same (())(5:19) is valid except that you put conjugate over here as we just discussed but that would kind of imply that offcourse will also have a complex input signal to deal with at the moment I am simplify the whole thing by just taking a real input signal but then if I take this as complex this as complex and this as complex everything is valid right. But if you recall one of our results that we discussed when we discuss the match filter properties was that at the sampling instant the response is always going to be real right.

So although if you implement the most general form you will have to do a complex valued processing throughout our final interest is not in a complex valued output our final interest is only in a real valued output which occurs at the sampling instance, is it clear? So one even when one has complex valued orthogonal signals here and complex valued input one need not really compute the complete complex valued output so the two filters, it is sufficient if you compute only the real valued outputs because finally we are going to look at this outputs only at the sampling instance at which time instant?

The values anyway expected to be real, the imaginary component is expected to be zero right, so will you not compute that or will you not process in a manner that requires full processing unnecessarily and that is what is shown here.

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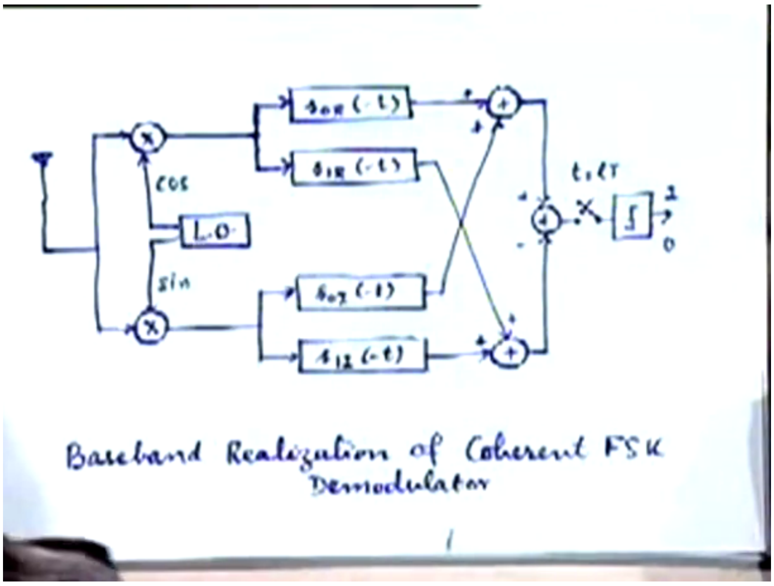

The baseband realization of coherent uhh I am sorry yeah ok that is fine of coherent FSK that is offcourse I am calling any orthogonal scheme as FSK here right. So you see what is happening you have incoming real signal which is first represented in its complex form by quadrature de-modulation right by de-modulation with cosine and sine.

So you get the complex envelope of the input signal which now crosses through a pair of complex filters in such a way that you only produce a real output right that is ideally I would have added a complex input here and a complex output here complex input here and (final) two valued output here but I am only computing a real valued at each of these two filters at the output of each of these two filters. So how do you take the real value, how did you compute the real value? Take the real part of this convolve with it the real part of the impulse response similarly the imaginary part convolved with the imaginary part these added together will give you the real part coming out of the first complex filter corresponding to S minus t, S sub O minus t.

S sub O minus t has two parts the real part and the imaginary part alright. So you basically have the real part computed here for the S O minus t and similarly the real part computed here for S1 minus t ok. So you take the same signal multiply it by S1 or not multiply, convolve with S1 or convolve imaginary part with S1 I and add the two together to give you the

Student: (())(8:53)

Professor: Naturally because one the input is here sure, what how did you get the complex representation in the first instance.

Student: they are the components of the complex waveforms but we have to introduce a J component, I mean in this likely (())(9:19)

Professor: I think I have discussed at the end of the last class also and in general let me discuss the passband representation the significance of tis cosine sine de-modulation the quadrature de-modulation, how do we obtain a complex envelope representation of a real passband signal. So finally what we have here is, this plus J times this. Received signal which is passband in nature is now represented in baseband form by this complex waveform. Complex means this is a real part this is a imaginary part right and what I am now saying is so therefore if you take the analogy basically we are the input now with input is complex.

In this picture the input is complex, this V now corresponds to this pair of signals alright, this pair of signals could have been processed precisely as shown over here by the use of complex valued filters here which you are doing here but we just modifying the processing slightly so as to compute only the real parts of each of these two filter outputs rather than a full complex parts full real as well as imaginary parts. You need to compute two outputs, one corresponding to this filter, one corresponding to this filter.

In general each of these outputs is going to be complex right, but I am only (compu) I am computing two outputs this and this, this constitute the first filter complex valued filter, the real part is here the imaginary part is here is valid on the complex valued input, this processing the complex valued input. But I am only computing the real part right. For the imaginary part I sure I have also done some other combinations right. I should have pass this to this, this to this and then combine those two which are I am not doing right, is it clear?

Similarly for the other filter I am again processing the real part and the imaginary part so as to produce only the real output ok.

Student: output of S0 1

Professor: Yes

Student: S0 t minus T subtract it from the S0 minus T.

Professor: It is a conjugation to be done when you are convolving right, the input signal when your, if you remember the basic equation this V t conjugate the convolution I mean the representation is such that we conjugating one of the signals before carrying out the convolution and therefore that sign (takes care of it) any other problem? So this is a baseband realization of a coherent FSK (de-modulation). I hope you have understood it if not please let me know and we can go over the portion which is giving you trouble

Student: (())(12:26)

Professor: No, that is automatically going to come out of the match filter output I mean, we already seen ok why should we square right then we need for squaring here. Basically I am computing the match filter output and comparing it with thresholds that is all I am doing as I am suppose to do in coherent receivers right and in coherent receivers what is the basic idea that is phase is exactly assume to be known or exactly compensated for, so that we don't have to worry about the unknown phase we have discussed that earlier and will return to this when we go to from coherent to non-coherent.

Deepakar anybody else? Mathematics is same the mathematics is precisely the same as I have earlier there is no difference.

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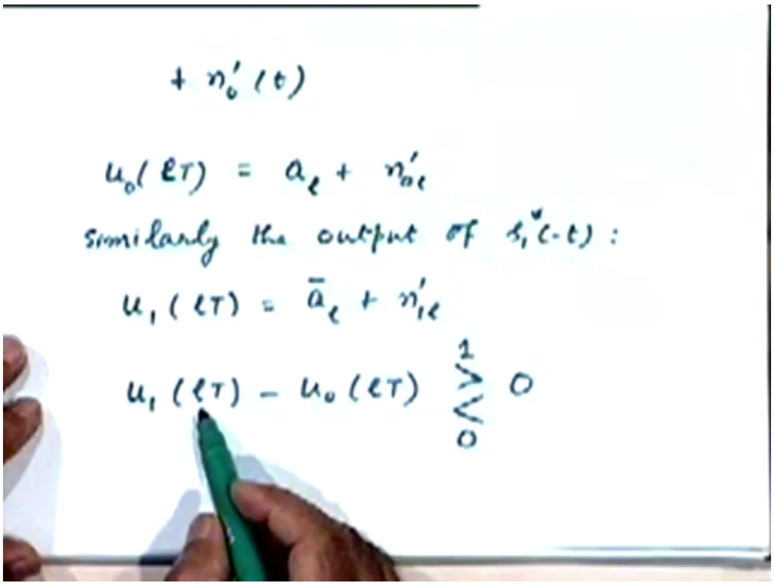

The required equation is precisely this that is I need to compute U1 and need to compute U0 and I need to compute the values at the sampling instance. Now in general this U1 and U0 would be U sub O would be complex valued but at the sampling instance they are real valued therefore I am only computing the real parts that is a mathematics nothing more than that ok. I mean you know how to take compute only the real part of a complex product in multiply the real number with a real part imaginary with the imaginary and either add or subtract the two depending on the nature of the two complex numbers.

Offcourse here we are not multiplying we are convolving that is the only difference right. So that precisely what is happening here ok. So if that is fine that gives me an idea how to do de-modulation of coherent de-modulation of FSK frequency shift keying or any set of orthogonal binary waveform at baseband. First bring the signal to baseband by quadrature de-modulation and then process like this ok.

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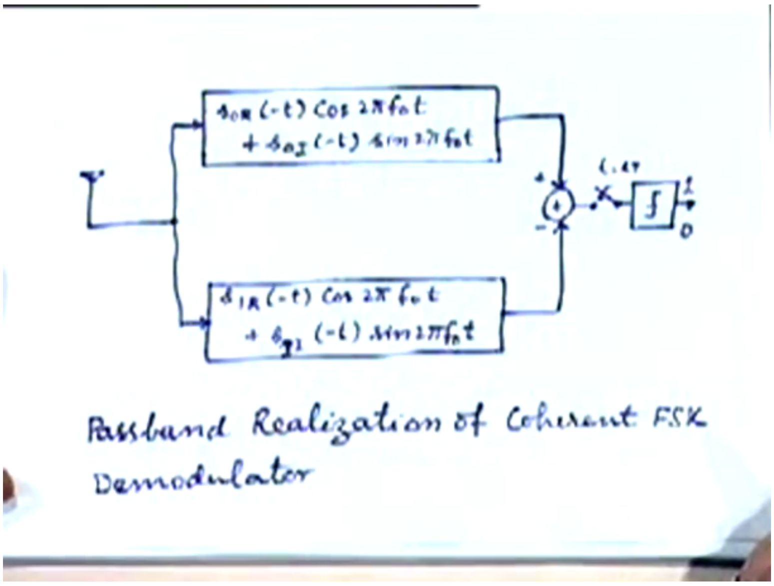

Offcourse alternatively if you wish you could do the whole thing at passband itself and in that case you will have to implement match filters at passband right.

So this is an alternative to this, this is original baseband realization alternative is do everything at passband don't take the real signal which is coming in at passband and process it by two matched filters at passband itself and a general form of these match filters is going to be as shown here isn't it? S sub O R minus t cosine 2 pie F knot t plus S sub O I minus t sine 2 Pie F knot t same thing except that you know how they induces S sub 1 over here. Compare the two outputs at the sampling instant and then decide which one was transmitted 1 or 0, fine, any questions on passband realization?

Basically you have two filters suppose your orthogonal signals were consisting of two frequencies right, which are mutually orthogonal two signals at different frequencies which are mutually orthogonal to each other over the signal duration then essentially it will (()) (16:26) down to having one tune filter at one frequency and the other tune filter at the other frequency this is only to (())(16:35) right. Offcourse I got here a general representation I have not really looked at what is the nature of this orthogonal signals S sub O a source I mean this particular signal but if it is truly FSK then this will balled down to the impulse response corresponding to frequency F1 and this will correspond to impulse response corresponding to frequency F2 right that will make you choose this S sub O r and S sub O I in such a way that this corresponds to an impulse response or a passband filter at F1 and another passband filter at F2.

Student: sir what are the input to the two filters in this case?

Professor: Input is the received FSK signal.

Student: and the filter is real or filter response is real

Professor: The passband yes everything is real it is a whole idea when you are doing passband you are working with real signals throughout right that is why you need to work with quadrature components directly. The whole idea of going from passband to baseband is to go first to a complex representation through quadrature de-modulation and then do complex valued processing. This point you must absorb a very clearly. I thought by now you would have got used to the idea of complex representation because you have been doing it since your previous semester. The complex valued representation is typically used to obtain a low pass representation of a passband signal right.

So when you want to do baseband processing you first convert you passband signal to baseband representation, how do you do that? You do quadrature de-modulation right by cosine and sine of the carrier. However if you want to do a processing at the frequency at this a signal is coming in or at an intermediate frequency some passband frequency you can continue to work with real valued signals as they come in right. There is no need to do any quadrature de-modulation in that case because you are going to work at the tactical frequency and which one would be preferable will depend on the application you are working at.

For example if you are working in a microwave band with a large bandwidth and it maybe very difficult to work at baseband because even if you bring the signal to down to baseband the bandwidth of those signals will be very high and your baseband components may not be able to handle that kind of bandwidth whereas if may be relatively simple to make tune filters matched to those frequencies at microwave itself special components can be perhaps made like saw components or other components of acoustic is many of the possible technologies are available depending on the frequency at which you are working.

You could directly realize this filter with the possible technology at passband itself however in many other (())(19:30) situations it maybe more convenient to first bring the signal down to baseband and do complex valued for same and then you can use digital signal processing.

Student: sir you said that actually these two will be two tune filters

Professor: They could be provided you will have a 2-FSK signal in the sense that this corresponds to transmission of a particular frequency F1 and this corresponds to transmission of a particular frequency F2 right. In that case this complex envelope be so chosen that this essentially becomes impulse response of a tune filter. Well if you I think you have forgotten what we discussed long time ago basic binary orthogonal modulations schemes. Many a times you just refer to nay orthogonal signaling as FSK in a loose way right.

So this is a treatment for that situation right and the specific situation when you are actually transmitting two different frequencies to generate your orthogonal set of signals then essential will balled down to two filters tune to two different frequencies.

Student: this very general (())(20:39)

Professor: No not BPSK, BPSK is not orthogonal, it is antipodal even QPSK is not orthogonal right. You look at other orthogonal signals we discussed there we discussed many possible examples of orthogonal signals right. I give number of possible waveforms which could serve as orthogonal signals right. For example this could be a pulse with impulse response like that and other pulse like that, these two signals will be orthogonal over a particular time interval lets say from here to here right and we discussed many other schemes. But the most important one remains FSK out of all this practically speaking right.

So that brings us to end of our discussion on coherent receivers, coherent receivers for binary signals, what is left is some kind of an understanding of the performance but I like to look at them together both coherent as well as the non-coherent, so I am postponing it slightly right. Will like to look at I mean this so called receiver structures that we have looked at how do they really perform in the presence of noise, what kind of signal to noise ratios are required to obtain reasonable error rates right which maybe acceptable.

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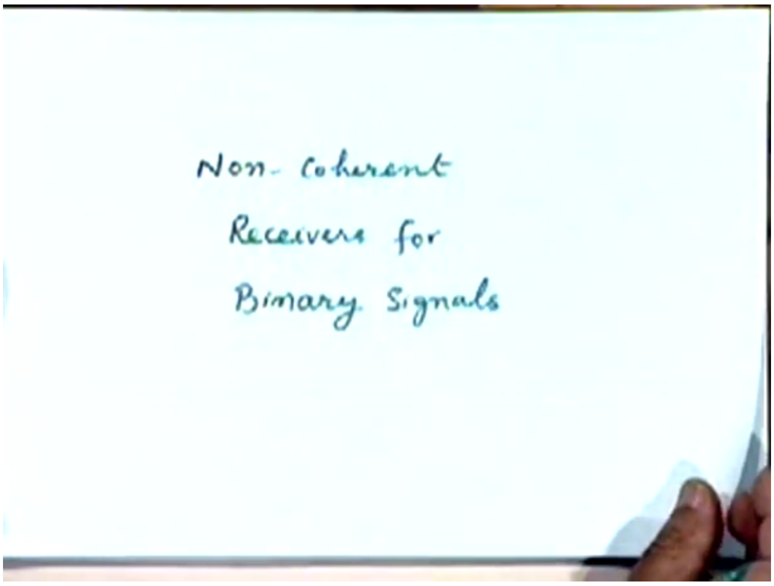

So that kind of analysis we still have to do and will do that after sometime but first let me now go over to the class of receivers which one has to worry about when phase is not known to you properly and that is non-coherent receivers, again at the moment only for binary signals ok. Now if you remember our discussion on coherent versus non-coherent we said that they are some situations where we may not be able either estimate the phase or its estimate they turn out to be too bad and therefore it will be now not possible to compensate for the unknown phase shift that may exists between what is transmitted and what is received right and therefore you don't have perfect knowledge of the incoming received signal atleast to the extent of a phase shift right.

There may be other problems but at the moment we have only discussing of the discussing this problem due to the presence of an unknown phase shift between what is received and what your cart might have been transmitted, because afterall your matched filter impulse response assumes a knowledge of phase, it assumes complete knowledge of the signal only then you can design a match filter too for it right. If you signal is S t the match filter is S minus t S conjugate minus T so you have no S t precisely in every little detail except possibly for amplitude because scaling doesn't matter right.

But in every other sense you must know the signal perfectly and therefore one can use this match filter concept that we have discussed so far only in the coherent situation when phase is either known or can be estimated reliably.

Student: sir phase shift not being on the that means that the signals which actually you are receiving they maybe some non-linear change of shift would be in from signal to sub symbol to symbol

Professor: Some kind of it, it is not exactly you see if it has a pure carrier if it would have been a delay but there is a modulation also in it. So it's it represents some kind of a distortion at the input which carries on or goes on to the output in the form of bad performance which we have try to appreciate it. So example you may remember at the passband, passband coherent receiver if you have a slight delay if your sampling instant is not correctly chosen it is offset by small amount right. Your output will fall considerably so much so that at spacing of 1 by 4 F knot it becomes 0 right. In a unknown phase shift will cause that kind of an effect.

Student: sir is the random phase shift random?

Professor: Typically it is I mean there can be two kinds of models deterministic as well as random both models are used right and wherever convenient you modulate it as random wherever convenient you model is deterministic but unknown. Deterministic means it is a constant value which is not known to you that is a deterministic model the other random model which is typically used is it is a random value which may lie anywhere with equal property lets say between 0 to 2 Pie right, both kinds of models are possible and are used.

Depends on how this phase shift is occurring, it really depends on the mechanism of that phase shift right. For example it maybe due to a fluctuating medium the path (())(26:24) time between the transmitter and the receiver, or they maybe unknown phase shifts sources within your antenna within your receiver which maybe contributing to this phase shift right and so depending on the overall situation one may use an appropriate model. At the moment I will not say more than that but at a later time when we possibly do carrier recovery or something we can take that up in more detail.

But to returning to what should we do when this is a situation, the one thing is clear that in this kind of a situation the rules of signaling schemes (rule) the use of those digital modulations schemes which put information in phase is kind of ruled out isn't it? If to start with in your modulation scheme you are putting information in phase of the carrier and at the receiver you say that your phase is not exactly known then it kind of implies that you should not use those modulations schemes in such situations right.

So use of antipodal signaling schemes like BPSK is ruled out offcourse one can do something even in these situations but I will talk about that separately right (but) therefore really speaking it makes more sense to talk of non-coherent receivers for binary orthogonal signals where typically information is not emburied in phase, information is emburied in signal shape itself right in the possibly in the form of choice of different frequencies or some other attributes but not phase right.

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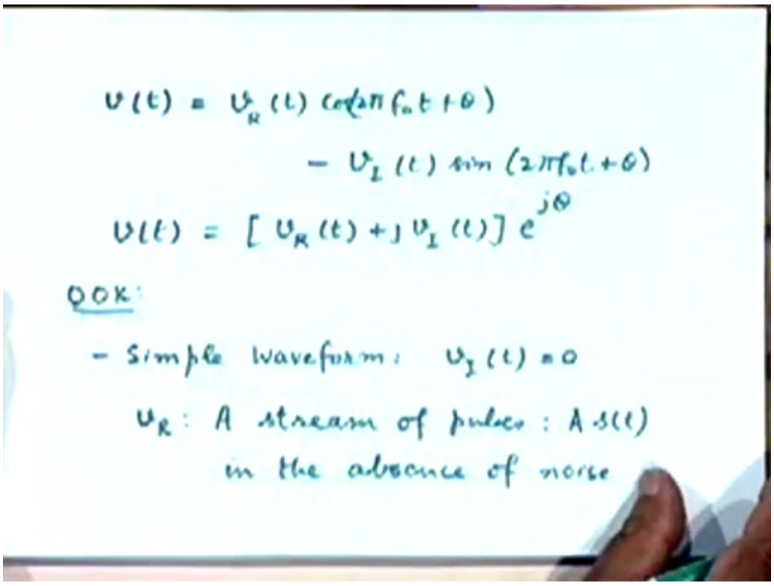

So this is something that you must keep in mind, anyway start with a general passband signal which has some phase shift will have a real part modulating the cosine carrier which is associated with some unknown phase shift theta and an imaginary part with sign 2 pie F knot t plus theta right or if you go to write the same thing in the complex envelope form just for convenience I am not going to use a different notation I think it is clear homo-context what it means. In the complex baseband form I could as well write this as V r t plus J sub I t into E to the power J theta right.

Strictly speaking I should be using different notations for these two right but since it does not cause any confusion I will continue to use identical notations right. so we have this situation in the coherent case we got around this by saying theta is exactly known so I can eliminate this by multiplying this received signal with E to the power minus j theta right or equivalently a passband by doing coherent passband demodulation right fine.

Now let us to understand this case better I will take a very simple syncing scheme the simplest possible and that is On Off Keying, On Off Keying is a special case of remember which kind which class of modulations schemes On Off Keying.

Student: Orthogonal

Professor: Orthogonal right, it is a special case and it is a simplest special case because othogonality is by virtue of using a zero signal as one of the two signals right. So it is one of the simplest possible orthogonal signaling schemes that you can have simple way from very rarely use now but very convenient for the purpose of understanding first simple things. So it is a simple waveform and I will simplify it further because I want to just develop the basic concept, lets assume that it is a very simple scheme in which even V I t is zero so our transmitted signal is V r t cosine 2 Pie F knot t plus theta that is all.

I mean the receive signal right and your V r is obviously going to be stream of pulses lets call them I will call the pulse shape A S t when noise is absent right if noise is present V R t will also contain some noise. Let me consider noise absence situation just to develop the basic concept. So is a model with each your working (())(31:29) all of you we are taking on off keying as the starting example in that also we are simplifying things for other by removing V I t so basically we have V r t which will contain both signal as well as noise.

The signal is lets represent it by A S t which is essentially a stream of this V r t is a stream of pulses in the absence of noise which will be either coming with zero amplitude or with some amplitude A right.

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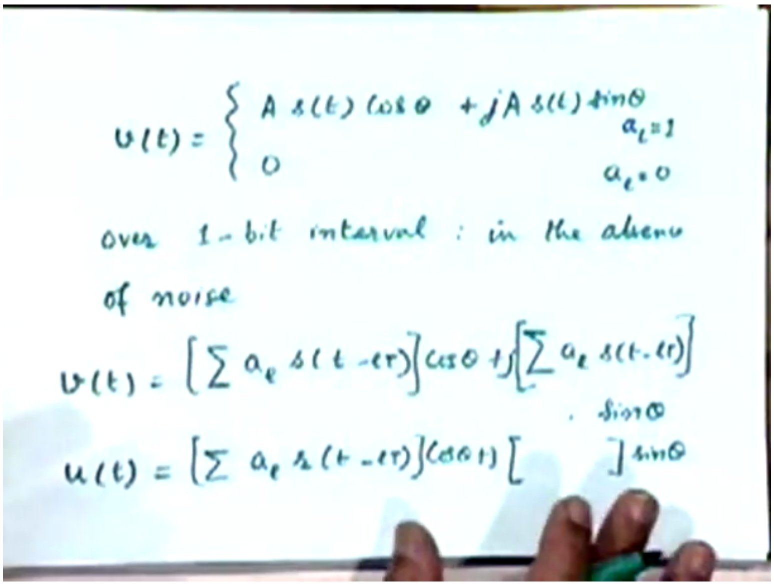

So more precisely I can write the received signal V t in the absence of noise over one bit interval right as either this plus this or zero fine does it call any confusion I am writing the complex envelop of this signal at baseband right, V I t is zero but we have a phase shift which will be in the complex form represented by E to the power J theta right.

So basically what we received is A s t into E to the power J theta which is what I have written here A s t cosine theta plus J A s t sine theta this is when your corresponding bit A L is one and this is when it is zero. If we transmit a one this is how you will be receiving the signal, if we transmit a zero this is how you will be receiving the signal whole one bit whichever bit interval you have selected. Offcourse this is the absence of noise alright. If you want to extend this to multiple bit intervals or general situation well we can then write simply a summation A sub L S how do I modify it, t minus L T cosine theta I am assuming phase shift to be constant throughout this whole thing multiplied by cosine theta plus J times same thing A L S t minus L T into sine theta when offcourse you are transmitting your one.

If you are transmitting your zero this U, after you are done matched filtering on this suppose I go and do the simple plane match filtering that we have been talking about so far ok. What will be the output of that matched filter? Output of the matched filter let me denote that by U t well it will be simply this waveform convolve with the match filter corresponding to the waveform S t, because what you know is S t, not the actual received waveform right and let that response be R t what we have been using so far right.

So this will simply become A sub L R t minus L T cosine theta plus same thing into sine theta alright.

Student: sir in case we use a matched filtering.

Professor: In case we use a matched filter, this is output of the matched filter and finally what are you going to do with the matched filter if you continue to proceed along the strategy we have been we have developed so far, will sample the match filter output at the Kth sampling instant T equal to Kth K t because I have used L as a dummy index here so I am not calling Lth sampling instant right, L is the dummy index in this so I will use some other index K I am going to sample it at sometimes in K t and then decide whether in the Kth interval this or that was transmitted alright.

Let us look at the (sampling) sample value at T equal to K t of this alright what will be the value? What will be the contribution from this summation? If we are using Nyquist pulses then the contribution will only come from A sub K right similarly from this. Assuming that we are using Nyquist pulses that is your S t is such that the response in the match filter is a Nyquist pulse right that then if I sample this output at T equal to K t the contribution from this summation will come only from the term A sub K because all other terms will contribute to contribute as a zero value at T equal to K t, because or K t minus L T would be zero for K not equal to L alright.

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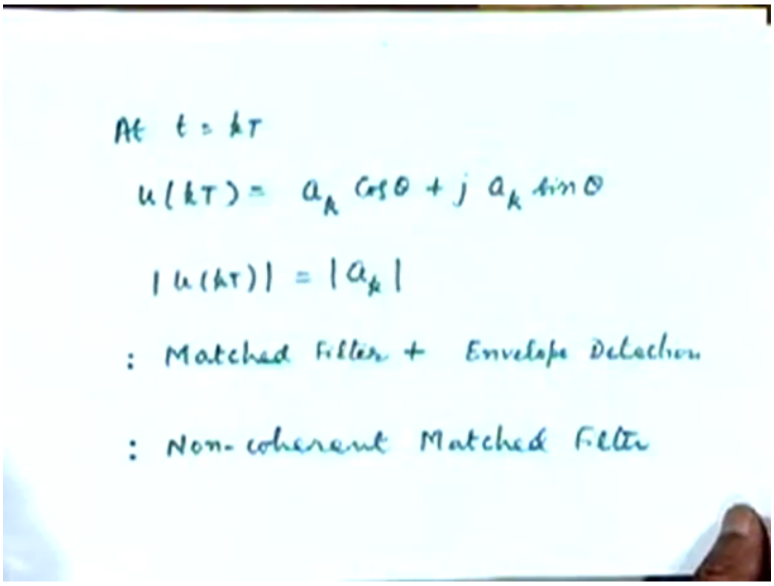

So lets assume that in that case at T equal to K t we can write U of K t equal to tell me the answer, A sub K cosine theta right plus J times A sub K sine theta right, now you know what to do, I don't know theta I can still get what is A sub K by just taking the magnitude right ok. By taking the simple example I wanted to tell you how to obtain your A sub K even when you don't know theta right. So take Mod of U sub K T and that gives your A sub K the value of A sub K. Ofccourse if it is zero that will be zero if not it will be that particular amplitude right.

Offcourse strictly speaking I should write this because when you take the Mod you will have to take the square sum of square and then square root, the sign will really go but then that doesn't bother us we are using orthogonal signaling scheme right either it will be 1 or it will be 0 fine. So what does that tell us? What should be our structure? Our structure is we continue to use our matched filter as before but follow it of with, what is this operation? Taking a modulus of a complex representation envelope detection right, so follow it of with envelope detection that is your optimum receiver. This is sometimes called non-coherent matched filter ok, in which the phase information is ignored by taking the envelop information by doing this kind of processing.

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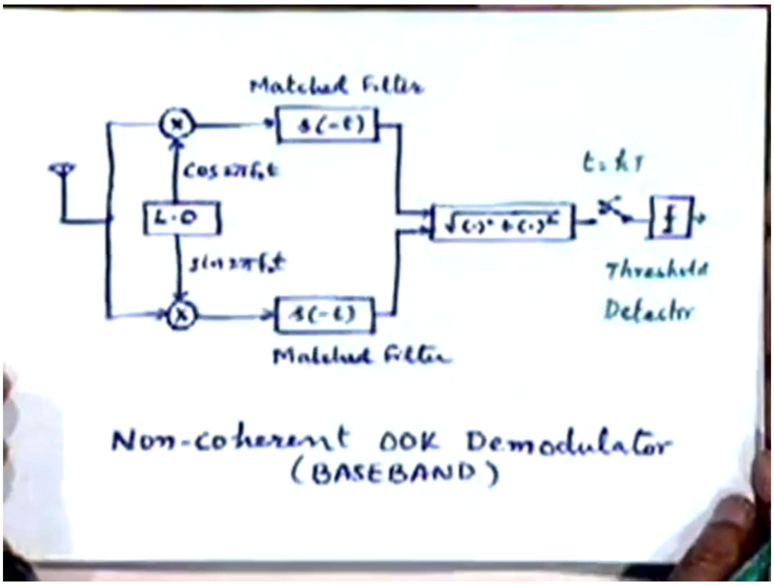

So this is what you do for the case of On Off Keying, this is a baseband OOK receiver on off keying modulation scheme, de-modulation for that non-coherent de-demodulator for that, having incoming passband signal obtain the complex baseband representation through quadrature de-modulation have this matched filter S minus t to obtain the real and imaginary parts and then square the real imaginary parts sum them up take the square root sample at T equal to K T and make your reviver.

Student: (())(40:18)

Professor: Well one can do either, it hardly matters, depends on how you want to implement it whether you want to have samplers here and follow it out with it hardly makes any difference because basically you are interested in the sample value of this angle of that instant whether you continue to have a continuous envelope detector and then sample the output or other way around makes no difference right whichever is easier for you to look.

Student: (())(40:55)

Professor: No actually this just a notation this is a threshold detector in which the threshold will now lie between zero and some amplitude right it will be not at zero the threshold will not be at zero it will not be a polarity detector I have just indicated it as a standard threshold detector , it is a notation for threshold detector, it should be (())(41:19 whether I mean don't let this confuse you right and if you want to do a passband implementation tell me what it will be like so that I don't start from here but I start from here, what do I need? Suppose I want to do this same directly at same passband without bringing the signal down to baseband first, I have a passband matched filter followed by an envelope detector which is very easy to build which you have done in amplitude modulation de-modulation in your analogue modulations schemes.

A simple diode and R C arrangement can be used for that envelope detection right in that situation. Offcourse you can't do that at baseband implementation because for that you have to have a carrier for that to work. In this case you have to implement all this things

Student: (())(42:19)

Professor: A lateral portion this is some basically some non-linearity which you have to implement right one can devise a circuit for it or if the whole thing is been done at baseband the present trend is to do everything digitally right and then this becomes a simple digital computation.

Student: (())(42:45)

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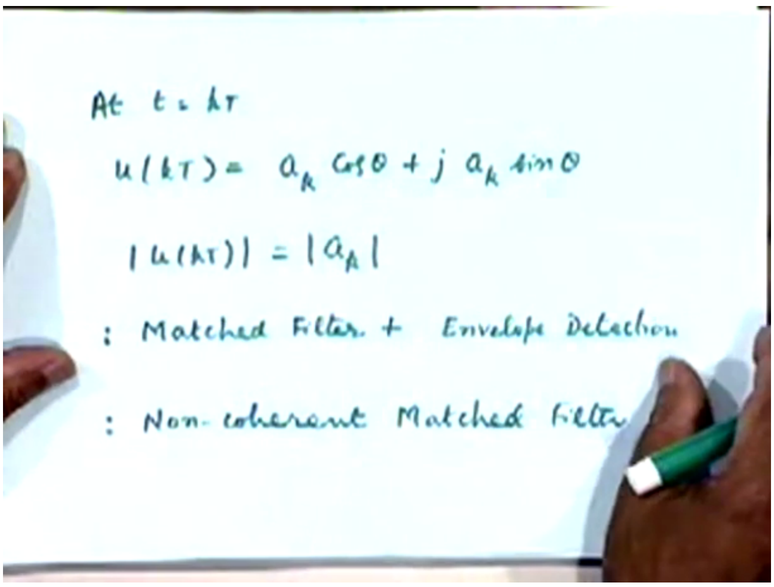

Professor: No but now we are interpreting envelope detection as a more general operation of this kind right, this is our definition of envelope detection in this case that we are taking the (modulation) modulus of this complex quantity right, it is matter of nomenclature you may call it whatever you like but you must understand what you mean.

Student: (())(43:16)

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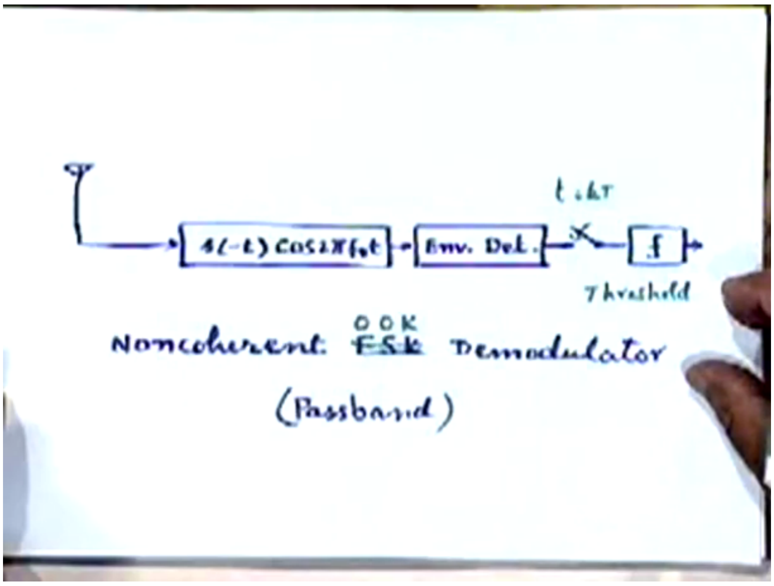

Professor: That is right infact at the moment we are considering a very special situation of orthogonal signaling where one of the two signals is zero right will soon generalize it with a general case where we can have arbitrary to orthogonal signals right, but before that, sorry this is wrong I don't have it here infact this is this only yes this should be OOK this is wrongly written ok, if you want to the de-modulation at passband you have a matched filter (())(44:05) passband follow it of with envelope detector sample it at T equal to K T and have a threshold detector ok that is an alternative realization for the same (prob) ok.

Now let us try to generalize it for general signals yes,

Student: (())(44:32)

Professor: Really speaking it doesn't matter very much because you see I square rooted it because of this equation, I wanted to implement this equation but as far as your (detection) I mean performance is concern it will hardly make any difference whether you square or you don't square it. I mean rather square root or don't square root it but the corresponding threshold will have to be modified right appropriately depending on whether what you do here will have to modify the threshold here.

Student: (())(45:10)

Professor: I don't want to go in that discussion there is some difference it is very minor it is less than a (())(45:21) it is very small difference.

Student: (())(45:25)

Professor: This (())(45:38) brief discussion since you have raised this point, I didn't want to discuss that what we have done here is basically derived the receiver from our intuitive point of view I mean we just looked try to appreciate what the signals are like the output of this matched filter and then what we should do with it so as to obtain what we wanted to. But there is a formal way of obtaining this receiver structure as an optimum receiver structure by following the methods of detection theory or hypothesis testing I briefly talked about (something).

If you follow those techniques formally you actually get this receiver structure as the optimum receiver structure ok. Now whether you do this whether you do the square rooting or whether you don't will hardly matter if you are only considering signal but this may have effect on how the noise will behave right and therefore your overall performance may depends slightly because there is a non-linearity here. If your everything was linear (when you) I mean hardly wouldn't make any difference but there is, this is non-linearity here and non-linearity will behave now a differently with respect to signal and differently with respect to noise right.

One has to care of that because if your square root you may have a Rayleigh distribution here if you don't square root you may have an exponential distribution there for noise right. So that will make any difference to the performance when you look at the performance comparison you will try to appreciate this thing better ok, will come to that little but I think you can see that in the presence of noise the presence of this non-linearity is going to behave differently when you have the square root and when you don't have a square root but fortunately if you adjust its threshold appropriately the difference in performance turns out to be unimportant it is very small ok.

But we wanted to do now was to generalize this to the case of a general orthogonal signaling scheme say FSK that is we want to develop non-coherent orthogonal receivers non-coherent FSK receivers ok. Now lets I think it is simpler if you look at this first and try to generalize this first the passband structure, can you tell me how would I how would you like to generalize it for a situation where I have two possible orthogonal signals for one and zero, 1 signal representing a 1 and the other representing is here. It is obvious now I have two matched filters one matched to each other two's waveforms each followed by a corresponding N1 detector and now I compare the two outputs as before right very simple.

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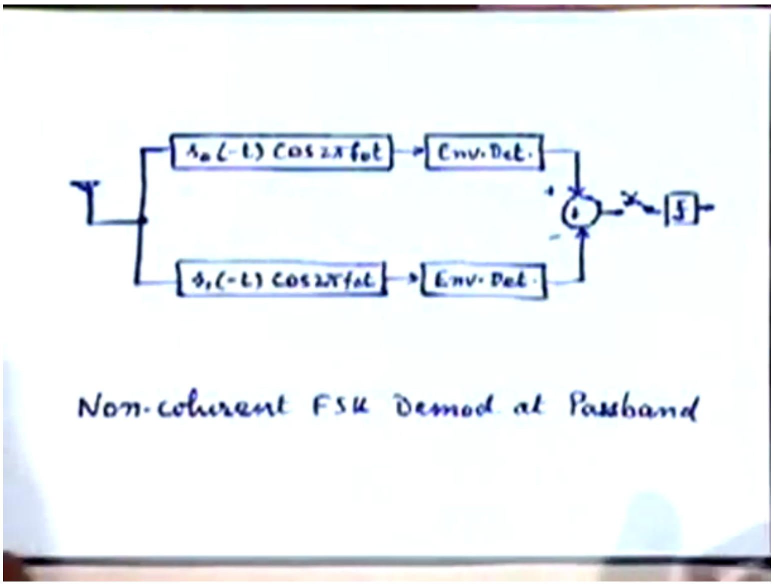

So that's picture here, non-coherent FSK de-modulator at passbands ok.

Student: (())(48:45)

Professor: Yes basically what it means is these two waveforms are these two impulse responses of the corresponding input waveforms are mutually orthogonal alright, each of them followed by envelop detector to get rid of the phase unknown phase problem and then compare the two

Student: also you have taken the (())(49:06)

Professor: In this case when you are taking an envelope, there is no question of imaginary part, no at this point lets see this is a passband implementation so there is no question of an imaginary part anyway.

Student: (())(49:22)

Professor: I am coming to that there is no atleast a passband one doesn't work with a complex representation yes but not at passband.

Student: (())(49:38) the representation in terms of S r and S I

Professor: S r or S I are components of a particular S that is all, that is right, this is what I had earlier

Student: (())(49:55)

Professor: Oh you could do that, you could add that to generalize this further, yes but when you say that ok because there is no imaginary component here it is a real signal, yes you could generalize it further by putting an S0 minus or S I minus this could be S O R you could also add an S O I sine 2 Pie F knot T right you could generalize it further sure ok this is slightly less general but I think conveys the idea equally though. Well basically what is her to do is take this old coherent FSK de-modulator put analogue detectors here that is all you are saying right hat is what I got represented here.

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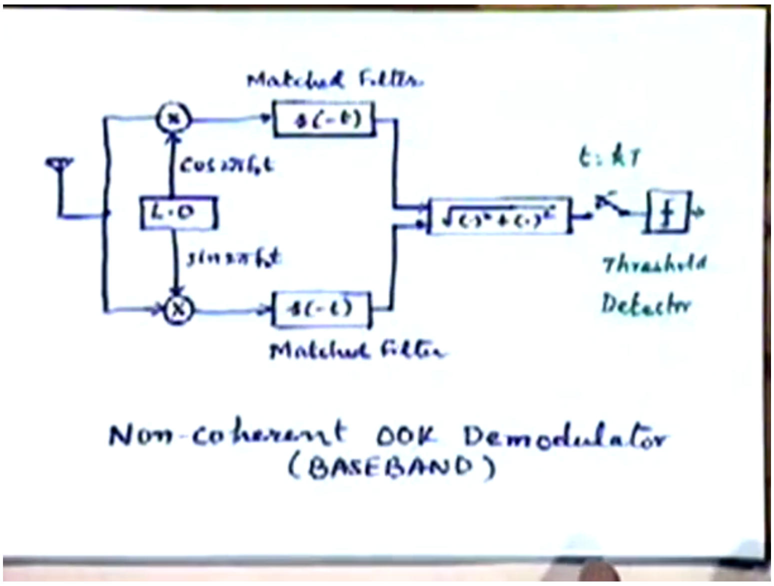

Now will you modify the baseband receiver? Let me show you the baseband that we had just discussed yes this is the one isn't it? This is your complex input signal followed by a complex valued matched filter in general offcourse in this case you call it turn to be a real value matched filter because S minus T is both because I have taken a very special situation and then take the mod square of this, where.

Student: something there is problem, we are taking a magnitude here (())(51:25)

Professor: Then we are comparing the magnitudes from the S0 and S1 terms, then we are comparing the magnitudes (())(51:34)

Ok because one of them is going to be there and the other is not going to be there, you are transmitting only one of the two signals in a particular with into a only one of them will be larger than the other infact one will produce a zero output the property of matched filters right and the other will be producing an amplitude A right which is what you are trying to look at. Same gooder way as you are doing coherent de-modulation except that you wanted to get rid of the unknown phase problem through this magnitude operation.

Just a minute, let see generalization of this. Can you visualize it?

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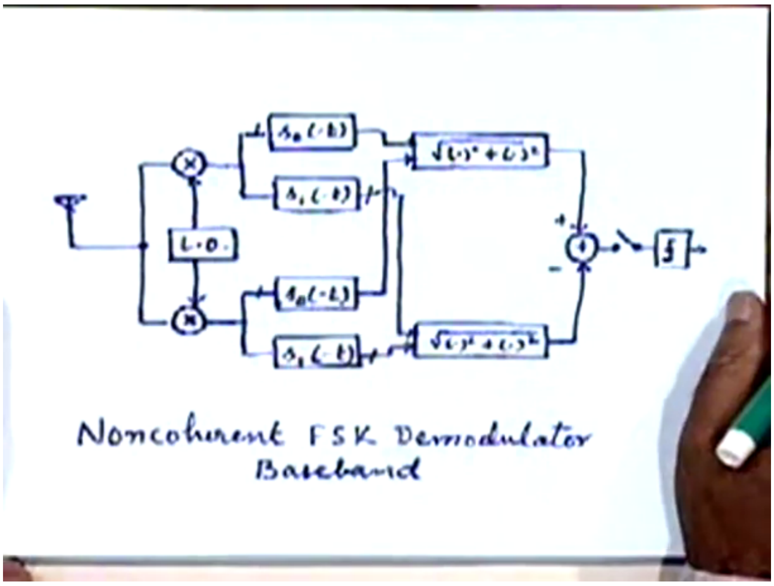

Complex signal coming all I have to do is have, duplicate this one for S sub O and the other for, sorry one for S O1 for S sub 1 right and that is a picture here for you. To start with a complex signal if you look at this you know this part and this part this is doing the operation for S0 and this part and this part is doing the operation for S1 right that is how. Then envelope detection comparison right, this and this corresponds to S0 this and this corresponds to S of 1.

Now this is a complex signal that is why I have got four channels here, ok in this case I have taken real signal and because I have taken S0 to be only consisting of real part, one can make it more complicated by also considering real and imaginary parts right and then take only the real part that will slightly simplify the situation by considering the signal which serve the baseband signal itself to be essentially real and this coming up this representation coming up because of unknown phase right.

I just simplify the situations slightly for you but one can make it more complicated by taking the most (())(53:55) situation, there is no need, ok. I have skipped the maths for this little bit of that but that is very easy, then in any case will take it up when we do the analysis. So next time or I will start with is, analysis of this binary (wave) this coherent and non-coherent receivers we done so far and after we are done that you will go over to a brief discussion of M-ary receivers right. Yes on Monday, one more announcement I have, your minus are ready for delivery.