Digital Communication
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Lecture – 14
Precoding for Modified Duobinary Systems (contd.) & General Partial Response Signalling

Quickly recapitulate what you are trying to do yesterday and see if you have any problems about it, see what we said was that by means of partial response signalling what you are trying to do is designing we want, we are trying to design pulse shapes which introduce a certain amount of inter-symbol interference in a controlled manner right, in the sense that at the sampling instance the contribution that one is getting at a particular sample value is not only coming from a bit that is being transmitted in that particular interval.

But also we get a finite interference or a contribution from some other transmissions in the previous bits or succeeding bits, in most of the examples we are discussed through introduction of delays is mostly previous bits right, so for example in the case of duo binary signalling the pulse shape was such that at the designated sampling instance the sampled values were contributed by the present bit and the previous bit, that is xk the sampled value of the waveform at the time instance k T0 plus T0 by 2 is equal to ak plus ak minus 1 right.

This happens because of the specific pulse shape that is bit used that is at T sampling instance these two pulse values, these two pulses contribute non-zero values, all other pulses corresponding to all other transmissions on either side of the k th bit interval contribute a zero amplitude at the sampling instant right.

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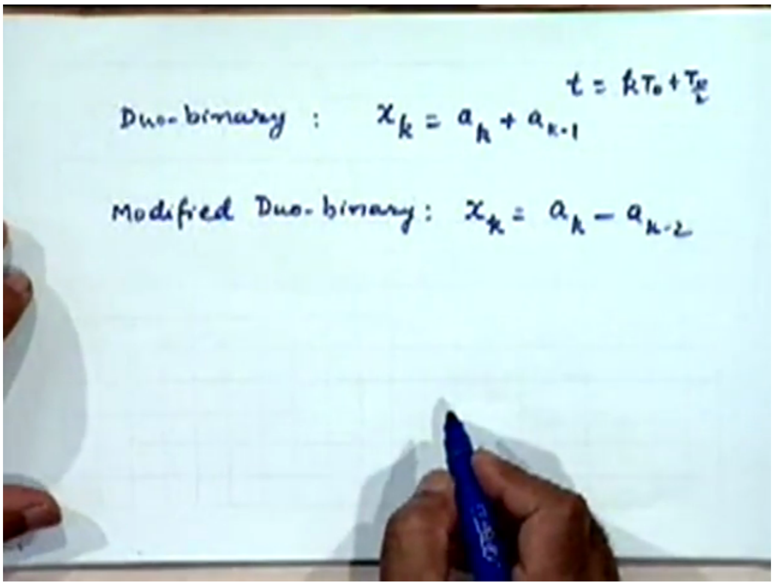

Therefore the net effect is that in the case of duo binary signalling when you sample at the k th instant when I say k th instant it means we are talking about t equal to k T not plus T not by 2 we get a contribution equal to ak and also same amount of contribution from the previous pulse I do not have to show you that figure again which I showed you a number of times before alright which brings all this fact. Similarly in the case of modified duo binary we get, now using a pulse shape which gives us xk equal to ak minus ak minus 2 yes.

Student: In this we are assuming the linearity of the channel (())(04:10)

Professor: Yes yes, if our model with which we are working is everything is linear in fact if you remember the channel is being modelled as a linear filter and what we are talking about as x k here is the output of three filters in cascade right whose input is an impulse sequence and the output is xt right, the three filters that we are referring to are number one at the transmitter which is some kind of a pulse shaping filter, two the channel itself works as a filter with its band limited transfer function whatever it may be.

And three you will have a filter at the receiver to make an optimum decision regarding each bit interval typically a match filter kind of receiver right so it is a sum total of all these three filters that we are looking at over here right cascade of these three filters, yes it is a linear channel we are assuming that assumption is correct okay, does this sound alright to all of you now? No? Where is the doubt?

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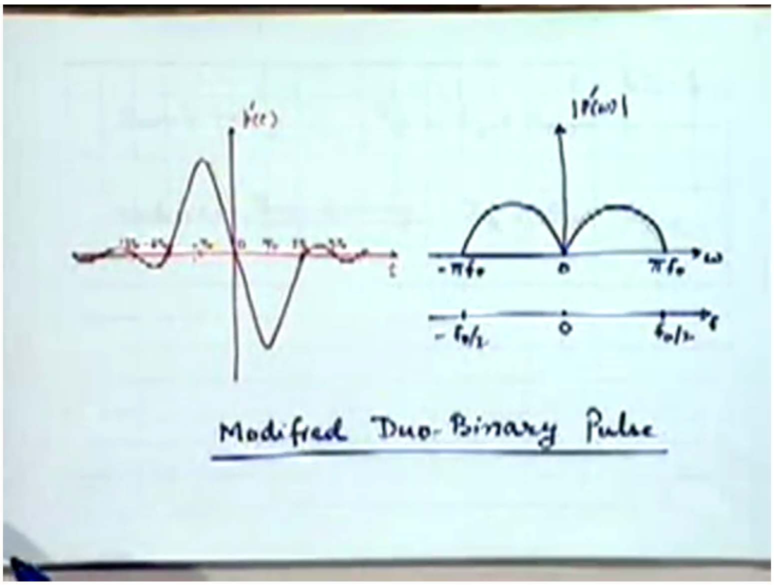

Okay I will have to fill them out of my hat here again, which waveform you would like to look at? The modified duo binary, this is the modified duo binary waveform right, now of course if you remember realization with which you are working as the delay of 2 T0 seconds right because what we are saying is, we are sampling at every T0 seconds, so what you have to really see is if you were to super impose this waveforms one after another every T0 seconds right, what will happen?

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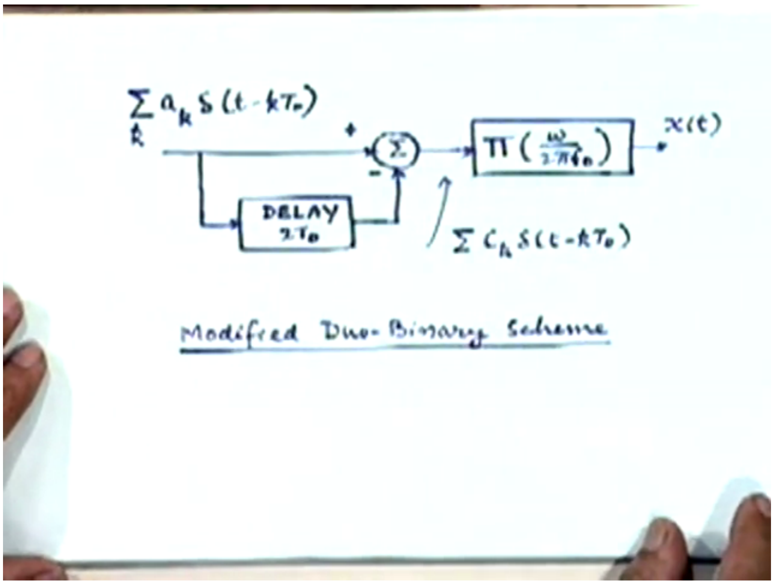

That is a question that we are looking at okay and let me just again pull up something okay remember this picture this is a practical possible method of realizing this wave shape except for the delay of 2 T0 seconds 2 T0 or T0 what was it? 2 T0 okay that means we are, what of course you are going to get here is I should have written p t here, I am distinguishing between p t and x t, x t is the total super imposed waveform after a number of, I mean when you are getting a sequence of impulses whereas p t is a single pulse okay.

So this is a single pulse next to is a super imposition of such pulses corresponding to the specific data sequence that is being transmitted alright so do you remember this equivalence, I mean generation of this spectrum through this kind of a mechanism right what I mean what I try to then tell you is that one can generate a modified duo binary signal scheme this kind of a pulse shape.

No no it is an arbitrary random sequence it is nothing to do with (())(08:08) this has been derived from a mathematical expression for p t if you look at that right, this block diagram has been derived from our understanding of what p t or p omega analysed right and therefore if we interpret this picture properly, (())(08:29) this is modified duo binary what we are talking about.

Student: So I am talking like in duo binary you had shown that ck is equal to ak plus ak minus 1 diagrammatically, going double that is what I want.

Professor: Yes yes okay you want it to be related to this picture right, I think for that you have to just look at some delay, I have to introduce some delay okay suppose I delayed by 2 T0 seconds I think the delay should be T0.

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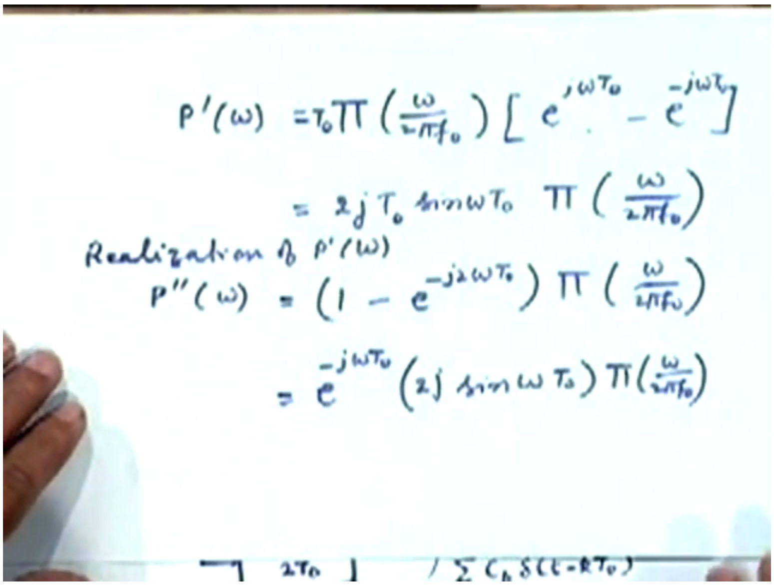

No that is okay in this block diagram it is 2 T0 but with respect to the original expression when I factor it out how much delay was being introduced, was it T0 or, no I am not sure no see okay I think you are, it is okay what I am saying is like this, this is original p prime omega for the duo binary right, this is okay and this is what I get after factoring things out and how do I factor, I basically I am taking e to the power j omega T not out right, e to the power j 2 T not.

Okay let us put it this way this p double prime omega that I am really going to get from this block diagram is this p dash omega into e to the power minus j omega T0 so with respect to the original p prime t what you are getting here is a delayed version with a delay of T not seconds right this 2 T not is coming from this realization.

Okay this p prime omega and p double prime omega are identical spectra, identical pulse shapes except that this pulse shape is delayed with a (())(10:42) by an amount of T not seconds.

Student: Sir what we are plotting modelling in that block diagram is p double dash of

Professor: Basically what you are modelling is p prime t minus T not right so therefore at the k th instant the pulse that is coming to the picture is a delayed version of this pulse just like in the duo binary case I had delay of T not by 2 here I have a delay of T not, now you interpret the waveform so basically this minus T0 becomes your sampling instant right, is not it, if you delay it by T0 seconds, 0 will be shifted I mean this peak will be shift to 0 right.

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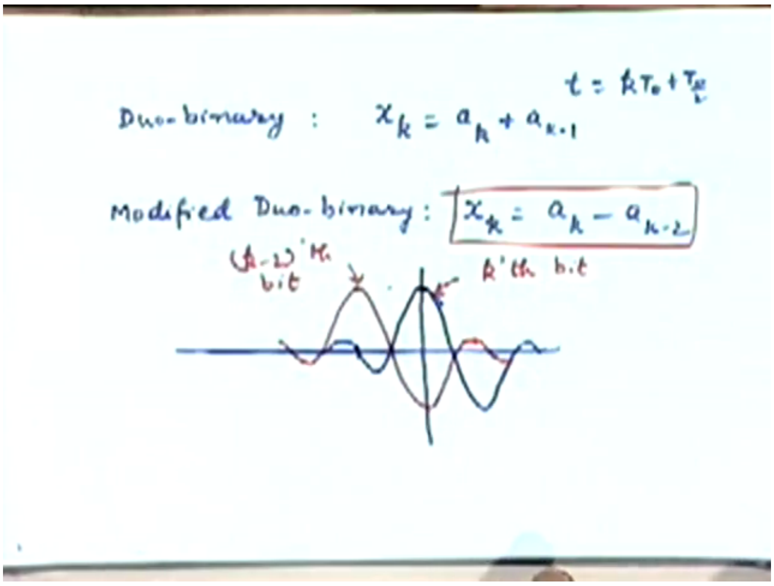

Therefore when I am sampling here I am also getting a contribution from one previously transmitted pulse which was transmitted at 2 intervals behind and that its contribution will be this negative peak right if both are positive right oh okay I will have to draw it for you, no problem, so I have shifted this here this is one okay and the other one is okay this one, this is the pulse that is being looked at in the k th bit right, this was a pulse that comes to the picture for the k minus 2 th bit alright.

Same thing displaced two time intervals, this is one time interval, this is second time interval right. So if you look at this sampled value here basically what you are going to get is this minus this which is, no whether it is, what I have drawn here is a situation where I have assumed ak and ak minus 2 to be both 1 so whether this will be 0 or something else we depend on what is the sequence of bits that comes into play okay.

For example if ak minus 2 was negative this whole pulse shape will be inverted and you will get superimposition (())(13:33) but the important point is this relationship which is what you had doubt about okay, is that doubt resolved now? You wanted that physical picture I hope it is there now okay fine should we return to these two relationships now.

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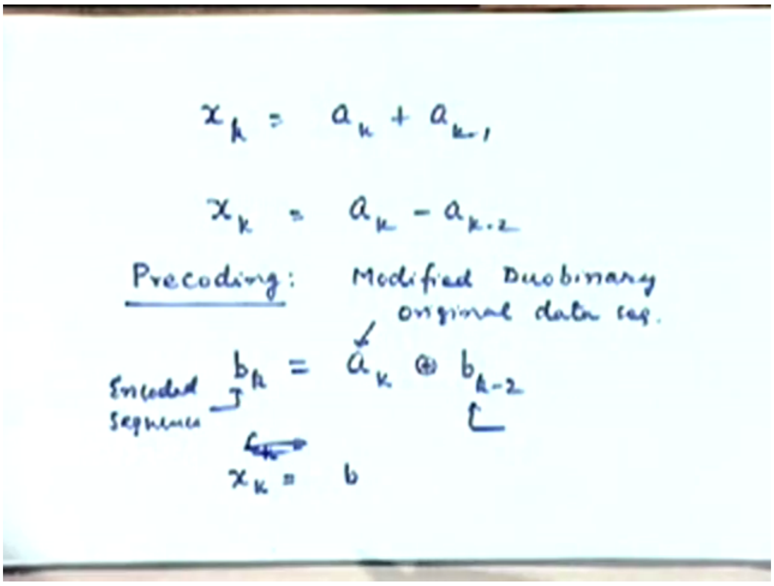

You are all clear that for the duo binary case the sample values are given by this and for the modified duo binary case the sampled values are given by this and we have already seen that this relations enable us to correctly interpret the data from xk's regarding ak's right, you can make correct interpretation of ak's if you just observe xk's right but there is a possibility of error propagation which we will like to avoid through pre coding right.

So and I think we had certain amount of problem yesterday with pre coding so we will go over it again you want me to do pre coding for both the cases or if you do for one I think you can work it out for the other, we will do for the modified duo binary and you can if you still have some doubts about duo binary I am sure you will be able to resolve them yourself, if not we can always talk about it again.

Let us take the pre coding for modified duo binary, I think that is the one which was given more trouble to us yesterday, what we suggested was that we should transform our bit sequence ak to a new bit sequence bk such that bk is ak exclusive or with bk minus 2 right other modified duo binary case okay let us start from here.

Okay then what happens to your ck, actually is this xk or this is well we can call it xk or something, (())(16:19) this original data sequence and this is the encoded sequence or pre coded sequence right fine okay I think I will avoid ck altogether and we will just write xk uniformly fine so xk now becomes what we are now doing is taking this bk's as data coming into your duo binary modified duo binary signalling scheme right so the impulse sequence that is coming into this system is no longer ak delta t minus k T0 but sigma b sub k delta t minus k T0 that is what is being transmitted.

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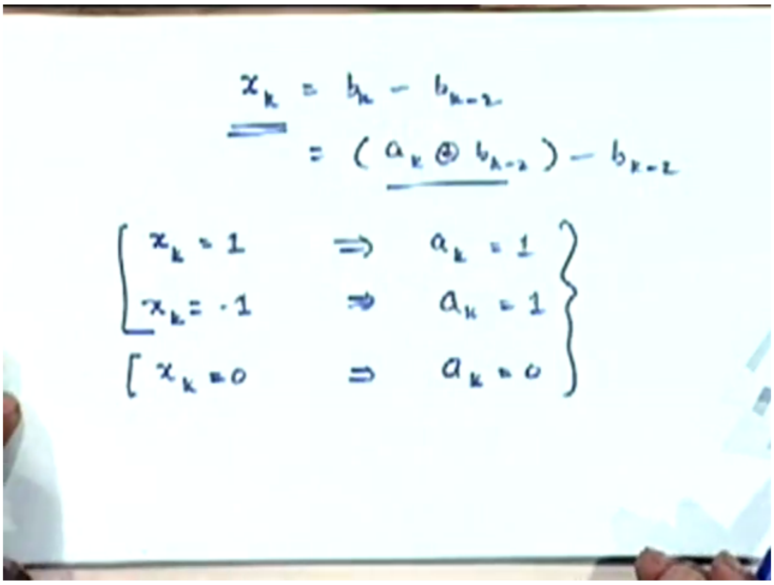

So this xk samples here would be related to this bk's through bk minus b sub k minus b sub k minus 2, so xk is b sub k minus b sub k minus 2 okay, let us start from this neatly now, x k is b sub k minus b sub k minus 2 which we can write as, bk if you remember is ak plus bk minus 2 minus bk minus 2, I think this is what we discussed yesterday also, I am just repeating it, now what we like to appreciate is whether or not we can decode the value of ak from an observation of xk which is what we will be able to observe without inter-symbol interference right.

xk's are the ones which will be obtaining by sampling the received waveform at (())(18:30) sampling instance okay alright, let us talk about the situation when xk is 1 let us talk about one thing at a time, when xk is 1, in what we can get xk equal to 1, if and only if bk is 1 and bk minus 2 is 0 right, if bk is 1 and bk is ak plus bk minus 2 and bk minus 2 is 0 what will be ak? bk is 1 that means this is 1 out of which bk minus 2 is 0, so ak has to be 1.

Is it clear with everybody? Vivek, understand this? Good when xk is minus 1 the only way we can have it is bk is 0 and b k minus 2 is 1 right and the only way we can have that is bk minus 2 is 1 the only way this bk can be 0 is when ak is 1 again because 1 exclusive or with 1 is right, when xk is 0 of course I did not argue out again for you that xk can take three possible values only right, 0 and plus minus 1.

Because of the fact that xk is bk minus bk minus 2 right, if bk and bk minus 2 each take 0 and 1 values then xk can take only three values namely 0, 1 and minus 1 and we are considering all of three values here alright, in this case they have to be equal right that means bk has to be the same thing as bk minus 2 which means this expression must be equal to bk minus 2 which can only occur when ak is 0 right.

Then whatever is the value of bk minus 2 will be the outcome of this exclusive or, this simple thing is what we got confused with yesterday very simple logic there was no need to and that is our decision logic when xk is plus minus 1 conclude that ak is 1 and xk is 0 conclude that ak is 0 and now we can do this data recovery without reference to what was transmitted in the previous bit interval right.

By just looking at a current sample xk we are able to make decision about the current value of the data ak, this xk equal to 0 implies bk equals bk minus 2 only then it will be 0 right which will be, which means that this sum, this exclusive or must equal to bk minus 2 and then will this be equal to bk minus 2, one of the terms is bk minus 2, it has to be that ak is equal to 0 because then 0 exclusive or is either 0 or 1 will give you correspondingly 0 or 1 alright.

But if ak is 1 we cannot get that we cannot get 1 exclusive or with bk minus 2 is bk minus 2 right only 0 exclusive or with bk minus 2 is bk minus 2 right that simple logic for you, fine so I hope that resolves the issue of decoding as far as modified duo binary is concerned and the duo binary is in fact a bit simpler if anything, so you should be able to work it out, if you still have some trouble okay.

Now if all this is clear up there are few comments I would like to make what we have discussed is two examples of partial response signalling and if you look at these two examples a bit carefully you will see a connection, you will see a relationship between both the examples in the sense that they have a very similar philosophy of working, that is we generate a composite pulse shape p t which is obtained by linear combinations of sinc pulses which are all band limited to the nyquist band.

Our primary purpose is to communicate with the minimum possible bandwidth with perturbation tolerance waveforms, so these two examples of generating the so called perturbation tolerance waveforms by linearly combining sinc pulses in the modified duo binary case you took sinc pulse at T0 and minus T0 in the duo binary case you took at T0 by 2 and minus T0 by 2 and things like that.

One can obviously generate other pulse shapes which will have similar desirable features and in fact we have seen that not only we can get perturbation tolerance kind of characteristics like we have gone from duo binary to modified duo binary because something else too we could introduce spectral nulls so in other words this gives us a mechanism for doing spectral shaping just the way we were doing things in line coding.

In fact if you look at think about these thing a bit carefully notice the close connection between what you are doing in line coding and what we are doing here, in line coding we were suggesting methods for representing 1's and 0's when number of 1's come together, number of 0's come together and things like that right excuse me.

Student: Sir modified scheme being (())(24:26) spectral null at 0 and what else I mean what is the other advantage that is the disadvantage over duo binary.

Professor: Of duo binary that is the only advantage, that is a very important advantage for us right but what I am trying to say is that all these things are being brought about by introducing what you can call correlation between successively transmitted waves, if original data ak is uncorrelated we are assuming that this the bits are occurring independently but after being operated upon by this pulse shapes by the introduction of this so called controlled amount of inter-symbol interference, the successive sample value are no longer uncorrelated, it become correlated.

Because xk for example now depends not only on ak but also ak minus 2 right, similarly in the case of modified duo binary depends on okay duo binary depends on ak and ak minus 1 so introducing a certain amount of correlation between what the samples that are finally getting transmitted and this mechanism of introducing correlation again carries out a spectral shaping of the pulse right.

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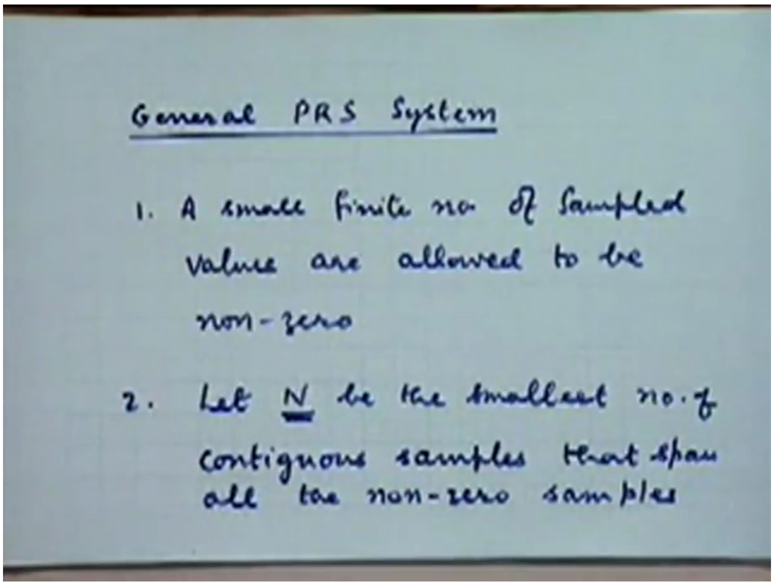

And by linearly combining sinc pulses in other waves you can generate other kinds of partial response signals right, I will just very briefly talk about a for generalization of this concepts and I am generalizing it in the following manner, in the duo binary case we said that two successive pulses will interfere with each other, two successive bits will interfere with each other in the modified duo binary case these two interfering bits were separated by let us say 3 bit intervals right.

No if you look at the span it is k, k minus 1, k minus 2, three spans are coming to the picture it so turned out that k minus 1 will not interfere but if you look at the total span over which inter-symbol interference is taking place, it is three bit interval right and one can generalize it to any amount that you may want right, so let us say we are talking about a generalized system of PRS in which a small of course we would not like it to be too large, finite number of samples or sampled values are allowed to be non-zero.

So in a way this is generalization of this nyquist second criteria instead of only committing two successive bits to be interfering or only two non-zero samples coming up in the sampling instance of the pulse shape you are allowing a finite number of them to be non-zero and the sinc pulse, ideal nyquist first criterion pulse, any nyquist first criterion pulse only one sample is allowed to be non-zero right.

And now we are allowing successively more and more pulses to be, more and more samples to be non-zero and let us assume that this value is N, let N be the smallest number of let us say what we can call contiguous samples which spend all the non-zero samples okay so you understand what is N, N is a symbol denoting the span over which all this non-zero samples exist at the regular spacing of T0 seconds.

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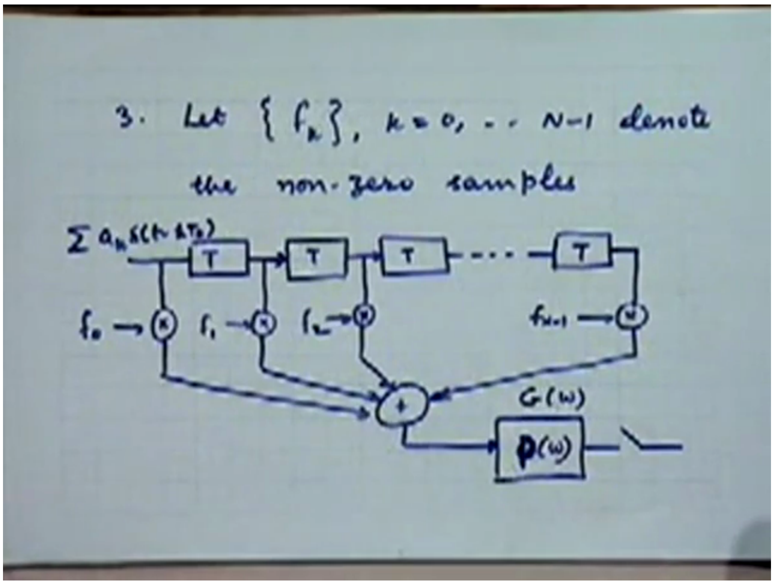

For example in duo binary N is 2, in modified duo binary N is 3 right and what we need to specify further is the sample values in the two examples that we have just done, the samples values were okay what was the sampled values at in the case of duo binary, 1 and 1 or f0 and f0 right, in the case of duo binary, modified duo binary it was f0, 0 and minus f0 right that is over the three bit interval, three values of the non-zero span of three intervals, non-zero sampled values were f0 or 1, 0 minus 1 whatever you like to call it right, so in general let us say let f sub k, k going from 0 to n minus 1 denote the corresponding sample values denote the non-zero sampled values.

Student: (())(29:57) that it will be a complex of the previous one, is not it, compliment sorry.

Professor: This fi's are I am referring to the sampled values of the pulses which will come to the picture at the sampling instance that is nothing to do with data sequence okay maybe I will have to clarify this a bit this was a pulse shaping use in duo binary and our sampled values there are non-zero which are come into the picture is this value and this value right, I think you forgot what duo binary.

At the sampling instance the values of the pulse which will come into the picture, xk's or tk's or whatever you would like to call them and in the case of duo binary the sampled values are these and all these right, the non-zero values are only these two right these are the two non-zero values that come to the picture similarly I will again show you the modified duo binary, if I can find it quickly, no I do not seem to have put it somewhere, maybe I think I can quickly draw it for you.

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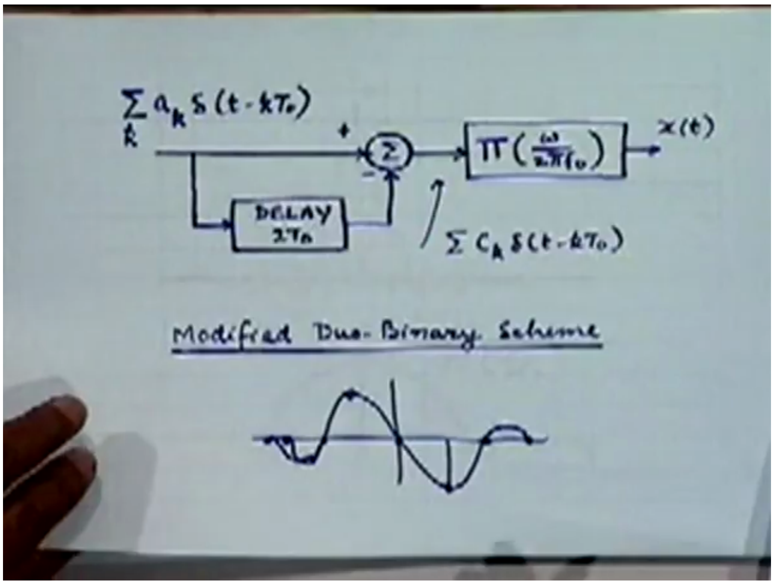

Modified duo binary scheme is in which you have right, sampled values that come to the picture is this, this and this, non-zero sample values okay all others after that will go through zeroes right here, here, here later on it will be all zeroes, ya so I am talking about these three although it is 0, since it lies within the span of non-zero values I am counting it as, I am denoting it by f, some value right some notation yes.

Student: Yeah on this only, the red graph, duo binary pulse shaping, the peak values of f0 by 2, I mean the two separate sinc functions only of that minus T0 by 2 (())(32:40)

Professor: No, I have shown it by minus T0 by 2, this is T0, this is in the middle right maybe the picture is getting a bit small there right, this is minus T0 and this is at minus T0 by 2 okay, no problem, so this f sub k is a notation for the non-zero sampled values that are coming through the picture in the span over which non-zero sample values lie, fine, now suppose I want a thing like this based on your appreciation of what we have discussed so far can you suggest a scheme for generating such a signal?

A signal which has these properties that is inter-symbol interference from n contiguous samples right, generate a pulse shape which will have and the corresponding amount of inter-symbol interference is decreased by these amplitudes, we waited some of sinc pulses with these as co-efficients right, basically that is what you need to do and we can realise this by a system of this kind.

Based on what we discussed with our modified duo binary case I am doing this a bit quickly because there is really no new concept here so I am just going away rather quickly and I am sure you would be able to appreciate it without any difficulty, I think I should call it p omega, precisely, (())(35:20) which question was this, that is a good one I think it will be a good exercise to find that out alright it was not exactly this question and this form.

Student: Just to find out a summation of sinc and error can represent

Professor: It was an example of general (())(35:45) so if you generate it like this where p omega is your, what will be p omega? What will be p omega? Ideal low pass filter, actually speaking rectangle filter right of bandwidth f0 by 2, this f0 and f1 might confuse with that f0, this is a, this your pulse shaping filter, you see there you are getting your impulse train ak delta t minus k T0, see the mechanism now becomes clear.

What you are doing is, this is very good way of visualizing how this controlled amount of inter-symbol interference usually being introduced you are doing it almost deliberately here you can see that, this incoming impulse sequence carrying the data is allowed to interfere with each other through this delay line, more complicated but possible because you know precisely what these fi's are, it is possible to do it right you can set up a set of equations which will allow you to do that equally right.

But before we come to that, what I will like you to appreciate is that one can generalise partial response signalling in this way which clearly shows that your pulse shape comes out as a result of two operations one is introduction of correlation or introduction of inter-symbol interference and finally by passing it through a low pass filter, pulse shaping filter right, you could replace p omega with any G omega which satisfies a nyquist criterion.

It may not be your rectangular filter incidentally right, you could replace it with any other filter which satisfies our nyquist first criterion you continue to have the same advantage or same kind of property but the only problem with that system would be that you will use up bandwidth more than the minimum nyquist bandwidth if you use some other G omega which is not the rectangular locus.

Student: Sir what is the (())(38:05) correlation we have already seen spectral null width modified duo binary

Professor: Okay Parth I am presuming that as far as, I will just come to your question but as far as this concept is concerned, this concept of generalizing it is okay right there is no really any question over directly, there are two questions that have been asked one is regarding the complexity of decoding and the second thing is why should we do a thing like this, what possible gains you might get.

Okay let me talk about both of these very briefly as far as decoding is concerned a certainly becomes more complex, more involved, the corresponding pre-coding if you have to do it in fact most of the time you will do pre-coding because there will be error propagation involved other than this right even pre-coding becomes more involved but it is not that involved that one should not think about it okay, it is manageable.

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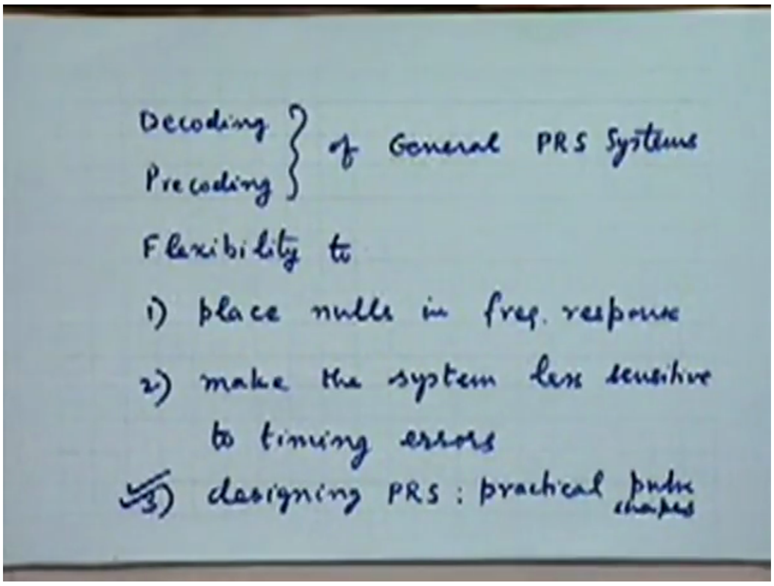

And I would like you to work out the equations for decoding as well as pre-coding of general PRS systems alright excuse me, so I am leaving that as an exercise I am not precisely answering that question because that will take the quite a bit of time and I want to now leave this topic and go on to something else quickly but because there is otherwise so much on each topic that one can do a whole course on each topic which we do not want to do alright.

Now the other thing is why do it? Now basically what we have learn from the experience with duo binary and duo binary is that we get a certain amount of flexibility to do our pulse shaping right by doing things this way so depending on specific application or specific requirement suppose in a particular application somebody says I want to send a pilot tone along with a signal corresponding to the nyquist frequency so that I can recover my clock easily right.

Suppose somebody say that my communication system will work well in a particular situation provided I send a pilot tone of a frequency corresponding to the nyquist frequency which will correspond to the clock rate, bit rate right, now if you want to do that, you want to introduce a sinusoidal component at f0 by 2, I must make sure that at f0 by 2, I have a null right fortunately both duo binary and modified duo binary have that (())(40:56) right so I can easily introduce a pilot tone there.

But similarly I may come up with some other requirement of putting a spectral null at some other place so I could think of investigating whether I could create that kind of a property in the spectrum by using a certain set of waves in the partial response set up right, so basically what we are getting out of this is flexibility and design right which can enable us, this flexibility will enable us to do a number of things.

Flexibility to place nulls in the spectrum, make the system less sensitive to timing errors but in any case the most and overriding, most important and overriding requirement is to design PRS systems which are, which use practical or practically realizable of perturbation tolerant pulse shapes, practical pulse shapes with minimum bandwidth right, that is our most important requirement, all other requirements are perhaps you can think of them as side benefits, the main thing is this that you are getting practical perturbation tolerant waveforms within the nyquist band that is the overriding requirement okay.

So I think with that brief mention of general PR systems I will close the discussion here but I will for those of you who are interested to know more about it I can give you one reference of course there are many books now which leave you this but there is a very interesting paper by Pasupathy, I am figuring the initials it is in IEEE transactions on communications and I am working with my memory and I think I am not able to remember immediately the year but you will find the reference given in your book by Lathi.

If you look at the list of references at the end of this chapter you will find the reference there it is a beautiful reference to waveform for more about partial response signalling if you want to know more okay, now where do we stand now, from our discussion of line coding and nyquist criteria you might feel that we have taken care of all problems of baseband communication of course so far we have been talking about baseband communication right.

Baseband in the sense that we are not introduce to the concept of a carrier okay we are basically working with a pulse shapes to the represent 1's and 0's and they were transmitting them after shaping them appropriately right either shaping them or doing before shaping even some line coding and things like that and specifically they spent quite some time on how to take care of inter-symbol interference that is otherwise likely to be there on a band limited channel right.

And you might get the impression that that is very good right, inter-symbol interference is taken care of but life is not that simple in communications, can you see what problems might practically arise in spite of all these designs that we discussed, nyquist designs, very important point this what we have discussed so far is an overall framework within, if within which we work will get nice properties.

The question is can we create that frameworks so easily right that is we can we really design the overall system without worrying about what the channel is like we have to worry about, of course that we appreciate that we have to worry about the, but we can worry about it or we can take that that into account provided what we know the channel impulse response, we know the channel transfer function if I know the channel transfer function then I can ask a general question okay here is the overall pulse shape requirement for inter-symbol interference elimination, here is a channel transfer function.

Now how should I realise the overall transfer function partly at the transmitter partly, at the receiver far this given channel so as to maximize performance right, one can ask that question and fortunately that kind of a question can be theoretically answered but one has to formulate that question and formulate the answer to that question before I will take up that question separately.

But even that question makes sense or can be useful to worry do you know your channel transfer function right there are many situations in real life communications where there is impossible to know, even in the telephone line system when you pick your telephone either to a transmit voice or data you do not know precisely what is the line that is coming to the picture right how long it is or what its transfer function is going to be, it is going to be different even if same two persons talk to each other at different times because you may go through a different route okay to a same person.

So the assumption that channel transfer function will be known so as to take that into account is also not a very valid assumption in all situations, in some situations it maybe possible but in all situations it may not be possible and therefore it seems almost impossible that we would be able to take care of inter-symbol interference just by doing paper design once for all right we will necessarily have inter-symbol interference and that sect of life will have to face at the receiver.

When you are trying to receive a data you will have to live with that inter-symbol interference and if you are clever you will try to eliminate it at the receiver right and that is done process called equalization, that is whatever you do not know about the channel we try to learn about it from what you receive and then try to compensate for it and that process is called adaptive equalization.

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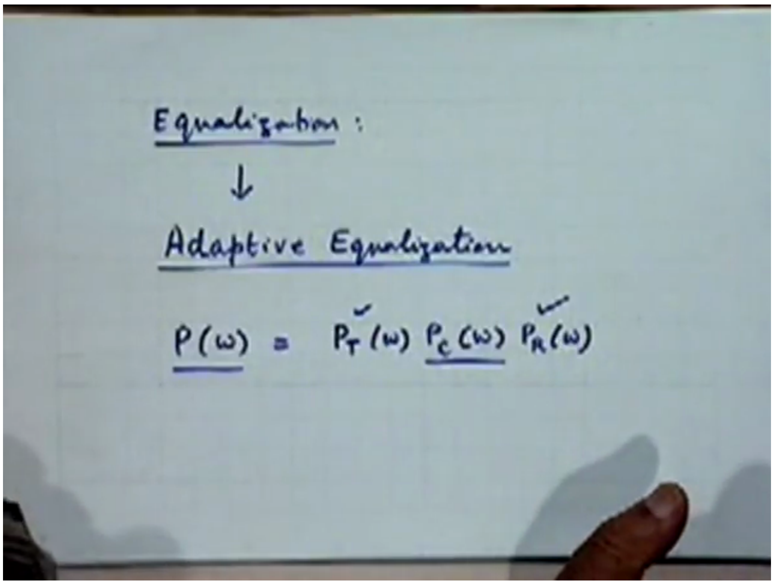

So we have introduced two new terms here, one is equalization which takes care of the fact that it can compensate for departures of the channel transfer function from what you might have assumed provided you know what does departures may be, if you do not know what they are then you have to take a course to first learning about departures and then taking care of them and that process is adaptive equalization.

So estimate a channel and then use a channel estimate to compensate for its correct restricts right, these issues it will make more sense if we take them up slightly later in our course so I will return to inter-symbol interference again later in the course if we have time okay and discuss these issues a little bit further. Yes please?

Student: P omega which you just considered (())(49:25) modified all this schemes that is essentially p, the original pulse shape pass through this channel, because we have said p t equivalent is equal to

Professor: P omega is let us call that, let us call it P t omega P c omega and P r omega right if I know P c omega I can design in appropriate pair of P t and P r such that we get overall P r omega as follows.

Student: Sir overall P omega is what we have considered.

Professor: We know what overall P omega should be so if P c omega is known life is okay, we can work with it somehow the problem is what if it is not known right that is where equalization, adaptive equalization makes sense

Student: Sir earlier you said that (())(50:23) here also those problems arise because we are again sampling at a particular instant of (())(50:34) so when there is a small incurrence, the time so would not they be any problem

Professor: No I think I can understand your confusion it is a very important question that he has asked which shows that some of you may still have some lingering confusion regarding partial response signalling, the confusion that come from the following fact what we have done is we have generated a partial response pulse shape through linear combination of sinc pulses and that is where your confusion is coming from right.

But what we are really doing by this linear combination is generating a new pulse shape that new pulse shape has the features that it, the corresponding spectrum is no longer a sharp transition spectrum a (())(51:31) kind of spectrum, it is a tapered kind of spectrum, the fact it has a tapered kind of spectrum implies that the corresponding pulse shape has small side lobes which also decay very rapidly in time right, so although the composite pulse shape that we are used for construction of our new pulse shape happen to be sinc pulses the resulting pulse shape has nice properties.

Student: (())(51:54) F not will be slightly plus minus delta f the value of

Professor: The effect will be only marginal that is a whole idea of perturbation tolerance that we want to be able to tolerant to perturbations and timing which you will be able to do provided our side lobes are small and the decay very fast with time right, so that has a very important and valid question if this fact was not understood very clearly then although sinc pulses are being used as a basis functions, the final pulse shape is no longer a sinc function it is something else alright okay I think this is a good point to stop, we will talk about digital modulation techniques from next time onwards thank you very much.