

Digital Communication
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Lecture – 17
Passband Digital Modulations – I: PSK & QPSK

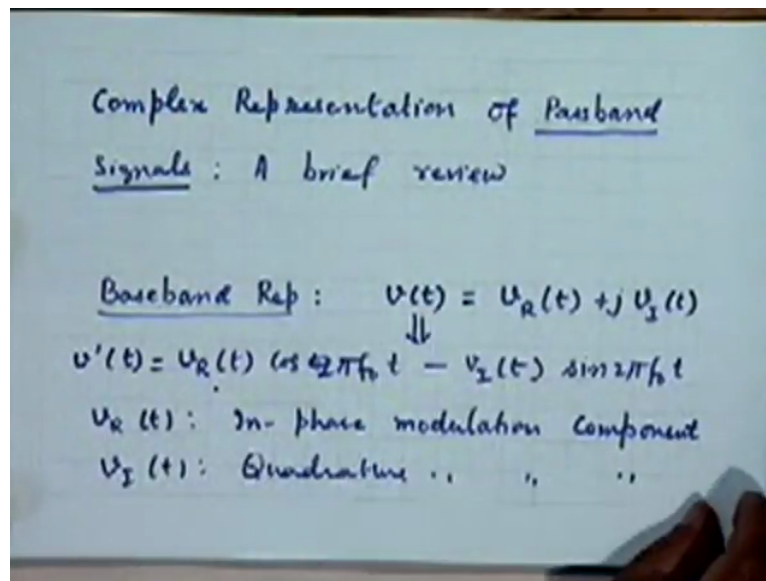
What we have seen yesterday is the case of binary and M-ary signalling in the baseband or binary and M-ary signalling waveforms, baseband signalling waveforms alright we have seen that in the M-ary case we can have the orthogonal, we starting with the orthogonal set of signalling waveforms and give a number of examples of orthogonal signals and since orthogonal signal sets can be taken as basis functions for any other signal set it seems reasonable to derive other signal sets from an orthogonal basis right.

And we in fact derive the other classes from the orthogonal class itself right for example the simplex class and the bi-orthogonal class, the anti-podal signal can be thought of as special case of the simplex class right, okay today we will take up the case of passband waveforms for binary signalling and M-ary signalling of course we will start binary signalling today and perhaps this and the next lecture will be both devoted to binary signalling and then we will go on to the M-ary signalling case.

Now passband signals basically or we cannot call them bandpass signals essentially those signals which have most of the energy or practically all of the energy consensually around a non-zero carrier frequency right that is the basic idea, non-zero centre frequency, that centre frequency definition is a bit vague, there is that contain no components at zero frequency essentially that is what you mean by passband signals.

And you must be remembering from your previous course in analogue communication on the first course you have done that such signals are very conveniently described in terms of a complex representation right and for the convenience that it gives will continue to use that complex representation just to recapitulate for you what the basic features of a complex representation are I will just summarize for you some of the things that we will be, some of the notation and features that we will be using from complex representation okay.

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So I will start with a very brief review of complex representation of pass band signals right, complex representation okay that is what you are doing to just now basically a passband signal can be represented in terms of an equivalent baseband signal using complex representation, this is a underlying point that you have to remember from your previous knowledge that is you can have actually for a passband signal, a baseband representation right which will be generally a complex signal okay.

That is we can represent this as a baseband signal which has a real part and an imaginary part okay the corresponding pass band signal would be $V_R t \cos \omega_c t$ or $2 \pi f_c t$ minus $V_I t \sin 2 \pi f_c t$ right, so corresponding to this baseband representation the passband representation which of course real signal right can be represented as in terms of what is called an in phase component and a quadrature phase component I am sure you done all this and I am just trying to remind you of this basic things.

So $V_R t$ is therefore sometimes also called the in-phase modulation component and similarly $V_I t$ is a quadrature phase modulation component right now briefly they are generally called in-phase and quadrature phase components of the passband signal, strictly speaking if I were to call this let us say $v'(t)$ then you can express this $v'(t)$ in terms of $v(t)$ by using a complex representation as follows.

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$$v'(t) = \text{Re} \left\{ (v_R(t) + j v_I(t)) e^{j2\pi f_c t} \right\}$$

: exhibits both amp & phase modulation.
- Convenient
$$\begin{aligned} & [v_R(t) + j v_I(t)] e^{j\theta} \\ & [\quad \quad \quad] e^{j2\pi f_c t} \end{aligned}$$

$v'(t)$, the passband signal is real part of your $v(t)$ which I will just write in the expanded form here $v_R(t) + j v_I(t)$ multiplied with $e^{j2\pi f_c t}$, real part of this all thing it is quite clear the real part of this will be precisely the passband signal $v'(t)$ namely $v_R \cos(2\pi f_c t) - v_I \sin(2\pi f_c t)$ right and in general this passband signal will if you were to look at it on an oscilloscope will exhibit both amplitude as well as phase modulation right.

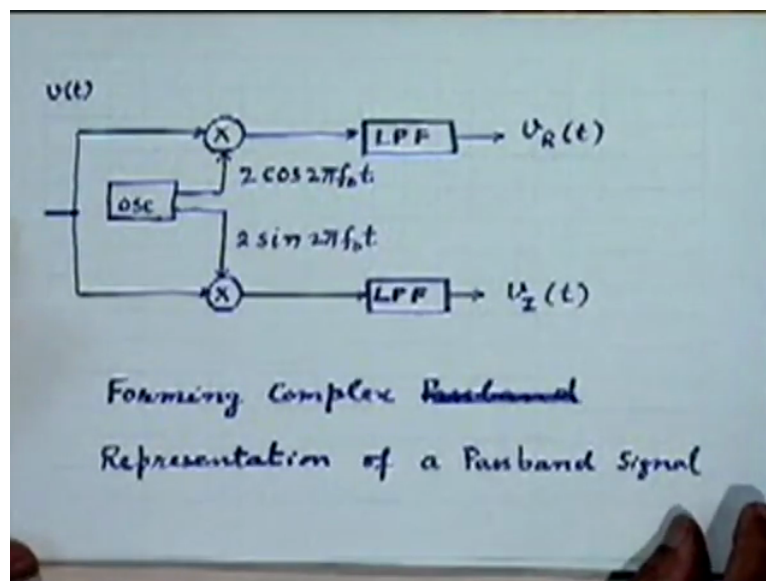
Because that is quite obvious from the nature of this signal right, from the nature of this signal if you were to right this purely in terms of a cosine function then you can write it in terms of an amplitude modulation into term into $\cos(2\pi f_c t + \theta)$ plus a phase modulation term right you can rewrite this in that form by using the standard trigonometric arranged, so in general this hardly needs any elaboration or it will exhibit both amplitude as well as phase modulation, please stop me or if you have any questions please ask the question.

Now the reason why we use this representation this kind of representation is basically for convenience it is very convenient for example suppose you want to study the effect of a phase offset maybe you carrier at the receiver is not perfectly synchronized with the transmitted carrier and you want to study effect of a phase offset in the carrier or the frequency offset in the carrier right all you have to do is multiply this with $e^{j\theta}$ for $e^{j\mu t}$ where μ may be the frequency offset right.

So complex representation is mathematical much more conveniently (9:09) and therefore it is preferred so for example a phase offset could be simply denoted at baseband itself without going to passband right you can take the baseband complex representation let us say this one and multiply it with $e^{j\theta}$ where θ may represent the phase offset right similarly you can have a frequency offset of both for that matter where ω may be a frequency offset.

So without going to messy trigonometry identities which you will have to make use of if you were to continue to work with the real representation it is very conveniently handled in the complex domain okay that is one reason and of course as we will proceed we will see the need for this more and more, any questions so far about the review of complex representation?

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No because it is all really simple and standard and I am sure you do not need much, this is a representation, a block schematic representation of how to obtain a complex representation of a passband signal right all you have to do to obtain the complex representation is to obtain the in-phase and quadrature phase components right because that is for these components basically define the complex representation right so you take your passband signal which I am now calling $v(t)$ right.

You multiply this signal with cosine $2\pi f_c t$ as well as its quadrature carrier sine $2\pi f_c t$ low pass filter these and you will obtain your $v_R(t)$ and $v_I(t)$ right, it is very simple exercise that is if you start with $v(t) = v_R(t) \cos 2\pi f_c t - v_I(t) \sin 2\pi f_c t$ and

if you go through this operations that are shown in this picture at the output here at the output of this low pass filter you will get V_{Rt} , at the output of this low pass filter you will get V_{It} right.

So given a passband signal we can obtain the complex representation practically using this method okay is it a practical method of obtaining a complex representation and then it to continue to work with this complex representation you need a physical real system okay so a complex representation is not only for just convenience in mathematics, it also introduces a lot of convenience in actual processing of this signals as you can see here.

Because what you have done is without loss of any information you have represented the information content of the signal which was earlier centre around f_c not to baseband right and that is a very nice thing to do from a signal processing point of view we appreciate that why?

Student: Sir there is one thing you said transformation from the passband to the baseband is it linear transformation?

Professor: Yes because all the component, okay to the extent that we are doing a mixing operation but that mixing operation you are essentially doing what, it is carrying out a frequency translation nothing more than that so even though strictly speaking it is not a linear operation right but the kind of operation that, in fact even for all practical purposes such an operation is regarded as a linear transformation, it is a linear modulation.

Student: Sir we carry out some operations on V_R and V_I separately and then reconstruct the signal, it will have the desired thing.

Professor: Because as you can see from V_R and V_I , I can go back to V is not it, so therefore I have not lost anything right if I can go back to where I started from that means I will not lost anything right and that is precisely the point here but the question that I had put to you was why should this be of some use to us from the signal processing point of view from actual implementation point of view?

Student: Because we are merging the baseband.

Professor: That is right, that may be very important if for example you decide to do digital signal processing on the signals right because you have to sample the waveform and then process it right and in sampling and all that is taken care of much more conveniently at

baseband than at passband although one can simply in passband also but in sampling rates involved will be higher but in this case one can work with low sampling rates therefore efficiently make use of digital signal processors which may not work at very high frequencies right.

So without any loss of information you can represent it while signal of this kind, any questions on this? Okay. Now in digital communication we are going to have to work with signals which are essentially time limited form in some sense right either strictly or at least for all practical purposes that is we are going to work with pulses so the specific class of signals which are of interest to us here are pulses.

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Passband Pulses

$$s(t) = s_R(t) \cos 2\pi f_0 t - s_I(t) \sin 2\pi f_0 t$$

$$s_R(t) \leftrightarrow S_R(\omega)$$

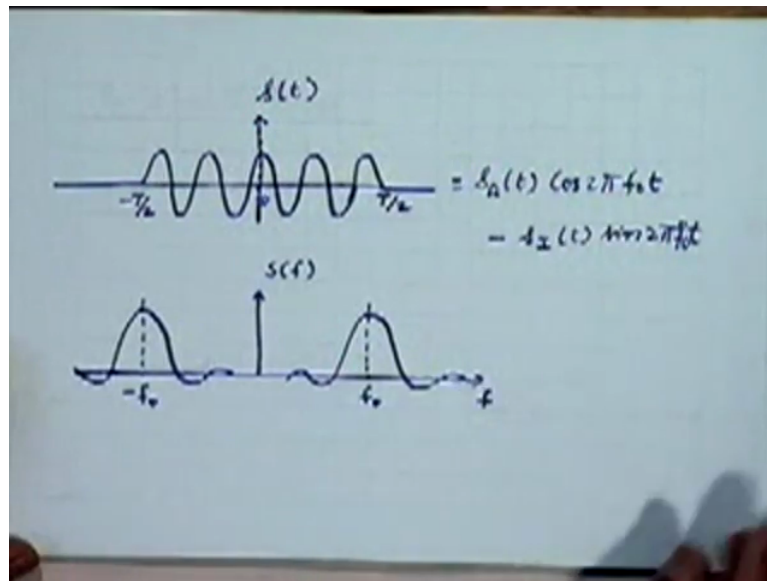
$$s_I(t) \leftrightarrow S_I(\omega)$$

$$S(f) = \frac{1}{2} [S_R(f-f_0) + S_R(f+f_0)] - \frac{j}{2} [S_I(f-f_0) - S_I(f+f_0)]$$

So let us talk about passband pulses and their spectrum, typically I can take a signal so let me I think talk about passband pulses, first of all what will passband pulses will look like you know what do baseband pulses look like just take example of a rectangular pulse you know what a rectangular pulse looks like what will a corresponding passband pulse look like any idea? Anybody would like to volunteer.

Student: Passband pulses will be sinusoidal will make it (())(15:43) modulated by square wave, amplitude modulated, it will be amplitude modulated square wave (())(15:53)

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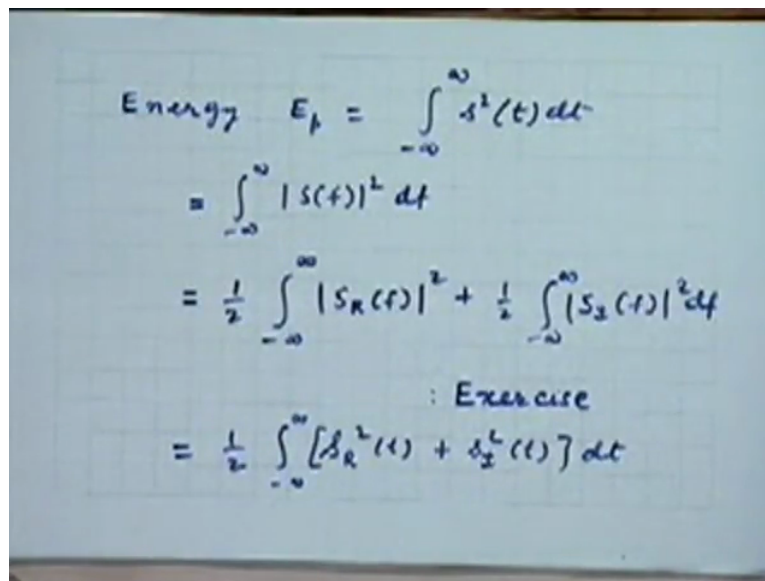


Professor: Yeah but let us talk about a high enough carrier so what will a pulse look like, basically yes you correct right it will be a gated form or a windowed form of a simple sine wave basically that is what it is going to be okay I got a picture here so that you can picture it better right it is basically a finite duration sinusoidal that is all right between minus, let us say between minus $T/2$ to plus $T/2$ right, so gated sinusoidal.

It is 0 outside this interval right so if you were to define a mathematically this sinusoidal function $\cos(2\pi f_c t)$ or $\sin(2\pi f_c t)$ or whatever in the time duration between minus $T/2$ to plus $T/2$ and zero outside okay that is a passband pulse and since in general its phase may be arbitrary we can have a complex representation of this kind, passband representation of this kind alright.

So the general nature of a passband pulse mathematically is a real part or in-phase part pulse, this is also pulse this basically now $s(t)$ is what some kind of a rectangular pulse right, $\cos(2\pi f_c t)$ or $\sin(2\pi f_c t)$ okay what will be the (corres) suppose I denote the Fourier transform of $s(t)$ by $S(f)$ or $S(\omega)$ and that of $\cos(2\pi f_c t)$ by $C(f)$ or $C(\omega)$ what can we say about the spectrum of such a pulse using your properties of Fourier transform.

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$$\begin{aligned} \text{Energy } E_p &= \int_{-\infty}^{\infty} s^2(t) dt \\ &= \int_{-\infty}^{\infty} |s(f)|^2 df \\ &= \frac{1}{2} \int_{-\infty}^{\infty} |s_R(f)|^2 + \frac{1}{2} \int_{-\infty}^{\infty} |s_I(f)|^2 df \\ &\quad \text{: Exercise} \\ &= \frac{1}{2} \int_{-\infty}^{\infty} [s_R^2(t) + s_I^2(t)] dt \end{aligned}$$

This will have a displacement around f not and one around s_R here right, plus f not minus half similarly for s_I , that is right, this should be J by 2, I have put that that is why there should be negative sign here s_I f plus f not okay? Can I proceed further? Fine, let us just complete this discussion by writing expression for the energy of a passband pulse energy E_p , $E_{sub p}$ which is of course defined as s squared t dt between minus infinity to plus infinity.

That Parseval's theorem you know that this will also be equal to integral of mod s f squared df right why did I take the energy in time domain or take the area of the spectrum magnitude square in the frequency domain, now two very simple straight forward exercise in which you express s f in terms of s_R f and you know this expression that we have written here, use this expression for s f and make use of the fact that s f is essentially a passband signal.

That is s_R f minus f not and let us say s_R f plus f not will have no overlapping component right so when you come across the product of these two what will be the product equal to 0 right so making use of these kind of identities if you were to go ahead and do some simplification of these expression it is very easy to check and this will reduce to in terms of the real and imaginary parts.

s_R f magnitude square plus, this I am leaving as an exercise for you to prove yourself simple but interesting exercise to do and therefore in time domain one can again write this as equal to s_R square t plus s_I square t dt again using Parseval's theorem once again okay, so I think

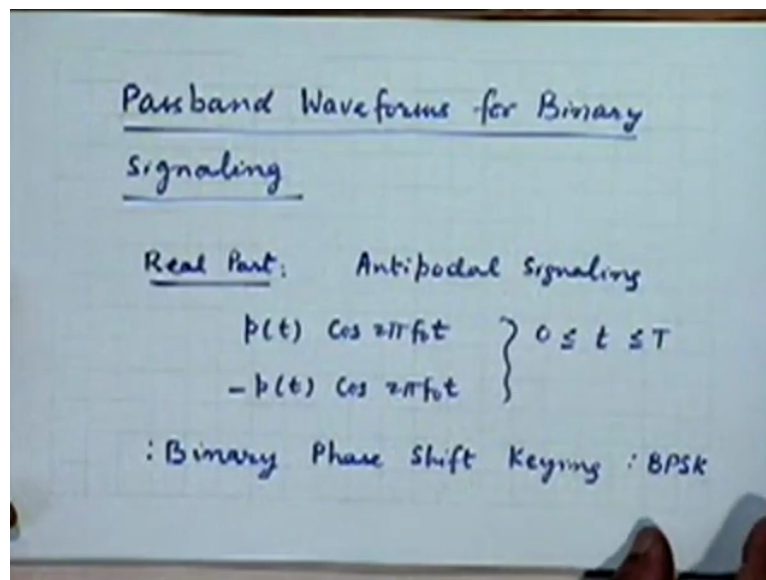
that should be enough for a review of complex representation of passband signals right, you have any questions? Please speak out if you have any questions.

Student: Sir $S(f)$ being band limited sir it (ω) (21:52) mean that $S_R(f)$ and $S_I(f)$ are band limited.

Professor: This is other way around usually, yes of course both way around it is correct because of course the example that I got here of a pulse like this since it has a rectangular envelop right in the sense that it is gated by a rectangular function and it has a strictly finite duration the corresponding spectrum is strictly what kind of spectrum you are going to get, sinc function displaced to f not ω minus f not so it will really ω tend to infinity right.

And therefore the corresponding $S_R(f)$ and $S_I(f)$ will also be sinc functions right they will be similar, the shape of the $S_R(f)$ and $S_I(f)$ will be very closely resemble the shape of $S(f)$ right, one can do that exercise obtain expression for $S_R(f)$ in terms of $S(f)$ it will be very similar to or it will be dual to the kind of expression you have here where we expressed $s(f)$ in terms of S_R and S_I one can go ahead and write S_R and S_I in terms of $S(f)$ right it will be interesting exercise to do. Any other question? Alright.

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Let us return to the main subject of interest to us here namely passband waveforms for binary signalling so we will take up the case of binary signalling first, binary modulations first, okay. Now to do that what is different here as compared to the baseband case in fact if you were to look at the baseband what are the schemes we had that we discussed essentially two in nature.

Broadly there were two classes namely the anti-podal class and the orthogonal class right and I think as far as binary is concerned it is quite clear that the orthogonal class is only of academic interest, of not really of that great in interest and so we will do most of our discussion in relation to the anti-podal class so what did we have there, a baseband pulse shape $p(t)$ and both its that pulse $p(t)$ along with its negative were serving as a two modulation waveforms to map either a 1 or a 0 onto one of this modulation waveforms okay.

Now essential difference that we have is we are going to have to have a pulse which is passband in nature right and therefore the general representation of such a pulse is we already seen it has an in-phase component and a quadrature phase component now we can do one of the following two things we can either use both component of the carrier namely the in-phase and the quadrature or we may use only one right for the construction of the waveform, let us start using only one, let us say we are using only cosine $2\pi f t$ right it is quite clear that I can construct an anti-podal signal from such a waveform by just multiplying this cosine $2\pi f t$ with some pulse shape and its negative right it is trivial.

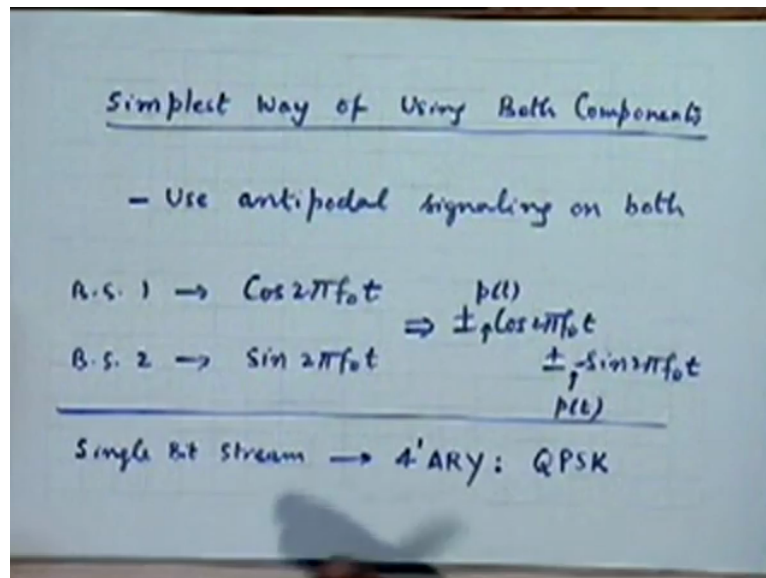
So a trivial kind of extension of the anti-podal signalling that we have done for the baseband case to the passband cases take the same pulses that we are using in the baseband case and multiplied with cosine $2\pi f t$ right so either $p(t) \cos 2\pi f t$ or minus $p(t) \cos 2\pi f t$ right, so let us take the real part of the quadrature representation, so based on the real part representation I could have an anti-podal signalling scheme as the simplest possible scheme for the passband case right.

For example $p(t) \cos 2\pi f t$ has one signal and minus $p(t) \cos 2\pi f t$ has the other signal just for the sake of simplicity you could assume or you could try to visualize $p(t)$ to be rectangular pulse just to visualize what is happening so suppose we assume it is rectangular pulse essentially it is constant over the time duration whatever it is minus $T/2$ to $T/2$ or let us say between 0 and T just for the sake of simplicity.

So what kind of a modulation scheme we get you are familiar with this I am quite sure it is essentially what is generally known as binary phase shift keying, so the anti-podal signalling in this case essentially reduces to, also called simply BPSK okay so because you are taking a carrier cosine $2\pi f t$ and you are either sending cosine $2\pi f t$ with 0 phase or with phase of π or 180 degrees π radians or 180 degrees alright so you could as well have taken of course sine $2\pi f t$ with a similar result, that will also be BPSK.

So that is a simplest kind of passband binary signal and you can think of which is essentially an idea borrowed from baseband signalling only thing is you have to have embedded onto a carrier that is all nothing new about it, really straight forward, in a more general situation I could take both in-phase component and a quadrature phase component together right.

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So the simplest way in which I could use both the components is as follows, when I say both components I am referring to the in-phase and quadrature phase carrier components use antipodal signalling on both the components right that is the simplest way just like I was doing it only on one of the components, now how can we use this, we can use it in many ways we can have two independent bit streams coming in right one modulating doing a BPSK kind of operation with cosine $2\pi f_0 t$ right and a other doing a similar operation on sine $2\pi f_0 t$ right.

So one bit stream there are two independent bit streams let us say bit stream one and bit stream two, bit stream 1 the successive 1s and 0s that are coming along in this stream will decide the successive phases of this carrier right whether in a particular interval, cosine $2\pi f_0 t$ is going to go a positive or a negative sign, similarly the consecutive bit values of bit stream 2 will similarly define the consecutive signs of the sine component.

And whatever the two maybe, two individual signs maybe in a particular interval of t seconds in which both the bits have been looked at, you will have a composite signal which will be plus minus cosine $2\pi f_0 t$ plus minus sine $2\pi f_0 t$ right leading to plus minus cosine $2\pi f_0 t$

$\cos(2\pi f t)$ plus or minus $\sin(2\pi f t)$, assuming of course that we are using rectangular pulses we may be using some other pulse shape so in that case we have to put an appropriate $p(t)$ or $s(t)$ or whatever here.

You can use the same pulse shape no need to use different pulse shapes but the signs of these pulses are independent from bit interval to bit interval so there is one way of doing this the alternate way of doing this would be we have a single bit stream, okay when you have a single bit stream what you can do is you can convert this into a not a binary system of individual bit streams but a 4-ary system right.

By looking at the incoming bit stream two bits at a time right just like the we were doing in M-ary baseband case earlier and now deciding depending on the combination rather or not this has to go with plus or minus and this has to go with plus or minus because there are four combinations and there are four possible set of combinations of these signs here right.

So you can get what is called quadrature phase shift key because essentially depending on what sign you have here essentially you are finally going to have a carrier of frequency f with one of 4 possible phase values is it clear, because $\cos(2\pi f t)$ plus $\sin(2\pi f t)$ will give you 1 phase you have to change the sign combinations you will get 4 different possible phases of the carrier that you might pick up to represent the 2 bit block that you have that you want to transmit, very simple and I do not think you have any questions.

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Representation of QPSK:

$$c(t) = \sum_{l=-N}^N a_l s(t - lT)$$
$$a_l = \begin{cases} (1+j)A & : 11 \\ (1-j)A & : 10 \\ (-1+j)A & : 01 \\ (-1-j)A & : 00 \end{cases}$$

So you can construct either two independent binary modulations out of this or you can construct a single quadrature modulation of this right and because the second way, second method is more commonly the choice we will basically talk about QPSK so let us go through a bit of exercise for a formal representation of QPSK that we have just discussed, we can basically represent this as a single complex waveform baseband waveform even though it is a passband modulation, that is a whole idea of complex representation.

You express in terms of baseband you do not have to keep on talking about the carrier all the time okay carrier can be just thought of as being in the background you do not have to keep explicitly writing it so the transmitted waveform, composite transmitted waveform $c(t)$ is a succession of pulses with complex amplitudes a_{lT} and pulse shape $s(t - lT)$ right mind you this a_{lT} are complex amplitudes what has gone in the background is $\cos(2\pi f_c t)$ and $\sin(2\pi f_c t)$ that is embedded in this representation here.

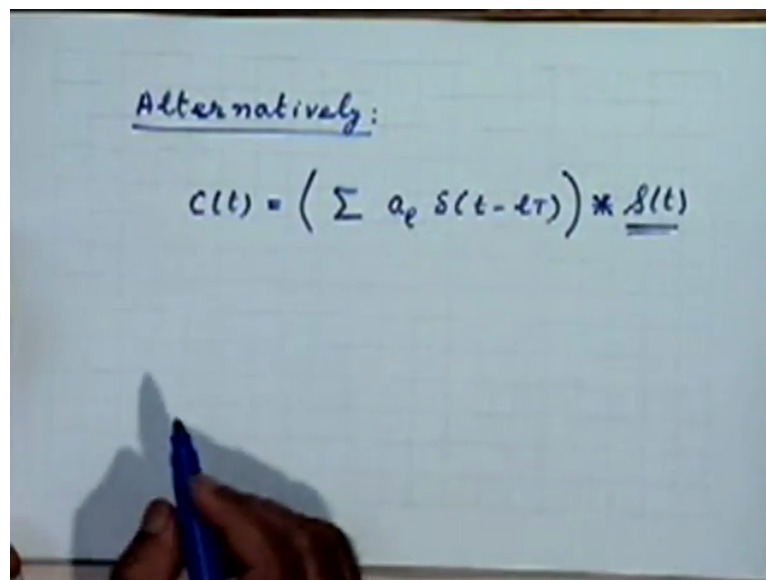
Because I will say a_{lT} would take on one of this possible values it could be $1 + j$ times some amplitude A , it could be $1 - j$ times amplitude A , $-1 + j$ times amplitude A and $-1 - j$ times amplitude A for example this could be the case when your l th bit pair you are considering now l th bit pair which we are assuming is coming over a T second duration right is let us say 11.

I have taken any arbitrary mapping here although there is a practical reason for doing this mapping also in a particular manner call the gray code we will talk about that separately so if

you input bit combination is 1 1 you may choose a_l to be like this 1 0 you can choose a_l to be like this and so on, what is in the (back), so this is only the pulse shape $s(t)$ describes a pulse shape that you may use.

If you take a rectangular pulse $s(t)$ is a rectangular pulse, both in-phase component as well as for the quadrature phase component right and what is gone in the background is the carrier well if you take if you want to bring the carrier in the picture all you have to do is multiply $c(t)$ of this whole thing with $e^{j2\pi f_c t}$ and take the real part of that right even get back your $c(t)$ expression fine, so do you all understand this complex baseband representation of a passband QPSK signal, is there any problem about it anyone, fine.

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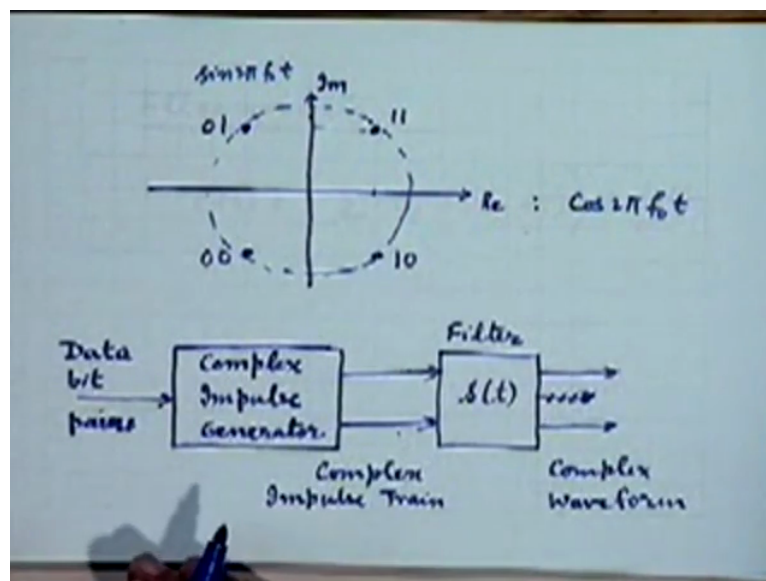
Alternatively:

$$c(t) = \left(\sum a_k s(t-t_k) \right) * \underline{s(t)}$$

Alternatively it is also a bit educative to look at the same thing as follows, instead of writing a sub l into $s(t - t_l)$, let me first introduce impulse functions as if the data is coming in the form of a series of impulses right and then these impulses are shaped by a filter within pulse response $s(t)$ right that is an alternative way of looking at it where this $s(t)$ is what we had earlier put over there.

So this very clearly brings out the role separately of a sub l which represents the data complex, the data is being represented as in a complex form, $1 + j$, $1 - j$, etc corresponds to what particular bit stream, which pair of values you are getting in the l th block right and this brings out the role of pulse shaping at the transmitter now I got a picture which summarizes all this discussion in the form of this diagram.

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I have got two pictures here, this is what is called a complex constellation diagram or a signal space diagram for QPSK right essentially represents the data points the location of the data points of QPSK right more specifically you may remember that 1 plus j will be which point this one right let us say this is 1 and this will be 1 here also so this point is representing 1 plus j so this is a representation for bit pair, value 1 1 do you remember this which I wrote here a few minutes ago okay.

So this is I am just taken any real axis and imaginary axis and implying to the present this complex stage of onto this signal space diagram like this so this point indicates 1 plus j and essentially therefore represents the bit pair 1 1 you therefore what it does is, it gives you a mapping of bit pair values to these, one of these four points on this complex constellation diagram, similarly 1 minus j will be this point right, 1 minus j is this point, this is being represented by 1 0 in that.

This is minus 1 plus j that is 0 1 and this is 0 0, okay so this is one way of representing the QPSK signal you can think of the real and imaginary axis also as essentially an axis which corresponds to the carrier in-phase carrier cosine $2\pi f_c t$ and the quadrature phase carrier sine $2\pi f_c t$ right and therefore now it will come out very clearly that when I take this point what I am really doing is transmitting a carrier phase of how much, what is the carrier phase corresponding to this point?

$\pi/4$, is not it, cosine $2\pi f_c t$ plus sine $2\pi f_c t$ right that will correspond to cosine $2\pi f_c t$ plus $\pi/4$ okay similarly for this will be minus $\pi/4$, this will be $3\pi/4$ and

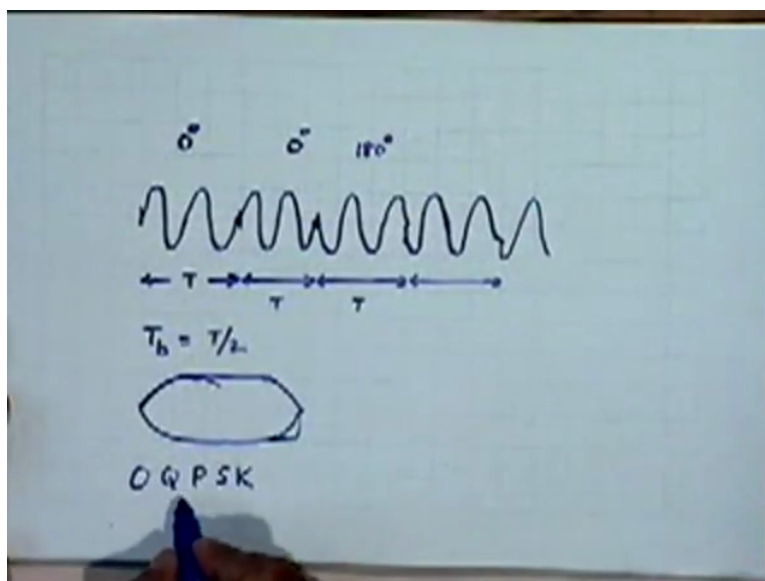
so on okay so you can think of these 4 points as representing 4 different phases actually you can imagine a circle on which they lie right you can imagine a circle on which these four points lie and the circle representing different points on the circle essentially representing the carrier with the different phase right.

This is cosine $2\pi f$ not t with 0 phase, with 45 degrees, 90 degree and so on and this is a picture for the representation I derive later, the data bit there is coming along from which you obtain a complex impulse train $(())(41:28)$ the amplitudes either $1 + j$ or $1 - j$ or $-1 + j$ and that impulse train, this complex thing is being represented by two signals here right the real part and the imaginary part that goes to a pulse shaping filter to obtain the complex waveform.

Of course before final transmission you will convert this complex waveform into a real waveform and how will you do that multiply the real part with cosine $2\pi f$ not t , the imaginary part with sine $2\pi f$ not t and add them up, add them up or subtract them whatever is required right, so you do the processing in the complex domain then come back to the real domain and transmit of course only one signal rather than two signals.

Now if you were to look at typical plot of, of course just to complete the discussion these pulse shapes may not be in general rectangular in practice they may be band limited pulses for example they may be nyquist pulses right so in that case it is more difficult to visualize the picture in terms of a waveform because the phase is not exactly constant throughout right if you have a non-rectangular pulse and the phase is not exactly constant because it is no longer a pure cosine waveform.

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It is a cosine waveform which is modulated by or gated by some kind of a time varying pulse shape right I mean some kind of a time variation depending on the pulse shape, okay now if you were to take the case of, I have got a picture here, try to imagine a QPSK waveform with rectangular pulses just for the sake of simplicity let us see what we will see, I find lot of people feeling very sleepy today, is it my lecture or is it something else? I am putting you off to sleep, then speak highly of what I am doing here, but anyway let us continue.

This is a typical waveform that corresponds to QPSK, I want to bring out an important, very important feature of this waveform which I like you to appreciate nicely it is not a very good plot that I have got here I wish I had a better plot, I have taken a carrier with 0 phase here corresponding to actually apparently I have taken the phases to be 0, 90, 180, 270 and 0 basically these four phases although according to that picture I have should taken 45, 135, then 225 and then whatever it is, I am sure you can all calculate it right.

So I have only done it perhaps wrongly but does not really qualitatively affect the argument that I have want to give here I got 0 phase or T second mind you this T second corresponds to how much time for 2 bits or time for what is this, something called as die bit, right it is not bit period, bit period will be $T \text{ sub } b$ which will be equal to $T \text{ by } 2$ you are looking at 2 bits together and then transmitting one phase of the carrier right then again I have got a 0 degree here and here I got a change of phase, I have got 180 degree phase change.

At this point I have got another change of phase by 90 degrees and so on, so what you will find is every T seconds there is a possible phase change right, there may be a phase change there may not be a phase change depending on the successive bit combinations that you get right and these phase changes can take on values of how much? I think that is very easy to appreciate what are the values of phase changes that you will see in this waveform, come on values of the phase changes that you might see are these points, that is right.

Well 90 and 270 are same thing either plus 90 or minus 90 right it is a magnitude that is really of interest to us to worry about, maximum phase change that you can think of is 180 degree right either plus 180 or minus 180 right alright so what do you see that you can have as large phase changes as 180 degrees at the points of going from 1 die bit to another die bit, now that essentially means that your, in reality you will never get rectangular pulses no matter what you do, is not it?

Because rectangular means infinite bandwidth in reality you are always be doing some band limitation, the moment you do band limitation these are not going to be sharp transitions so what is going to happen to these points when you do a band pass filtering of such a signal before you finally transmit it onto a channel, you will go through a 0 amplitude because there is a you know the envelop really goes become 0 here right.

So you think the envelop will perhaps it will be stay constant here and at this point it will suddenly go 0 because of a 180 degree phase shift, there is something that I think I would like you to appreciate, now which is very undesirable.

Student: Sir this is not clear sir.

Professor: This is not clear?

Student: Sir because after every time period, after every die bit period (0)(48:01) 0 phase gap is on that T_0 .

Professor: Let me put it this way (0)(48:05) there is a discontinuity right that discontinuity will tend to be smoothen out by the filter, the filter is going to do band limitation, in time domain the equivalent effect is smoothening out the sharp transition and how will that be done, by making the waveform round edge, the decay will not be abrupt one but a rounded

one, right. Let us say very crude picture I am giving exact effect you can calculate mathematically right.

Student: Sir the only thing is that the decay will still has to start from the point where the continuity, discontinuity occurs

Professor: Okay there may be a time period, you are right there may be a time period

Student: (0)(48:47) time delay thing has always been problem like we make a thing and take all that (0)(48:52) every time and say there is a time delay.

Professor: Is anything wrong with that?

Student: Sir we will dig it up in some Thursday class.

Professor: Alright I can take it up even here there is no problem, if necessary I can dig it up here but yes it can always be done any anti-casual system can be made casual by introducing sufficient amount of delay in the system right although I am not talking about anti-casual all I am saying is okay what you are saying is of the filter will take some finite time to respond to an abrupt transition sure and therefore physically you will not see this rounding immediately after this occurs but it will continue over some interval I agree with you entirely, nothing wrong with that.

Therefore the precise point of which 0 will come will not be at this point would be slightly later because the filter has a certain response time has a certain time constant which defines as bandwidth.

Student: And by that time the next die bit has started.

Professor: Yes the response to that also will be slow right after all the response time is dependent on the time constant of system everything is dealt by a same amount roughly speaking very crudely speaking right although this is best handled analytically right although this is best handled analytically right, take the filtering pulse response look at the input and calculate the output waveform.

That is the best way to handle it but I am just giving you a very rough crude picture of what you might expect to see waveform wise I am not going to calculation here but what you can expect to see is a zero crossing I mean your waveform is going to pass actually have zero

amplitudes at times this is the important points, I mean normally when you think of a QPSK signal, you think of this as a constant amplitude signal right.

And that is a desirable thing to have in a communication system, a constant amplitude signal, do you know why? Any reason why? See if you look at just this QPSK waveform, what are however badly it will be drawn here, it might first give us an impression rather deceptive impression that it is a constant amplitude signal right and in general it is, next statement of making is it is desirable to have a constant amplitude signal in a communication system, why?

No noise is not the main reason, non linearity is one thing but even more important than the non linearity is the fact that your power amplifiers to which you are going to send these signals are going to work most efficiently if you work them at a specific level of amplitude of power that have been having a large dynamic unit of variation. So power amplifiers will not be very efficient if you allow a large amplitude variation in the amplitude.

Secondly the non linear effect will be more pronounced when the envelop is not constant now that is something that we will take time to appreciate but yes that point is I think we just understood, it is more or less should be clear I mean if there is a non linearity in the system then the system will respond with different gains at different input amplitude levels right rather than a constant gain.

So it is best to have a constant amplitude of operation so that you are working at a specific point on your non-linear system and around which you may expect a linear kind of behaviour without any problem right, so from both in non linearity presence point of view as well as from efficient power amplifier point of view it is preferable to have waveforms in a communication system which are essentially constant amplitude waveforms alright.

QPSK signal to start with looks like such a constant amplitude waveform right but in reality after filtering it no longer remains one right, so which means that you are likely to have non linear problems non linear effect problems and also power efficiency problem, so there is something to keep in mind.

What would you like to do to take care of this now there are number of things people have tried to do but as a simple straight forward improvement one can do something here immediately by slightly changing the scheme of things, you have QPSK you continue to have

QPSK but what you do is instead of having a bit transition every T seconds, phase transition every T seconds the I and Q components are phase offset by half of this T or bit interval.

So that now phase transitions will occur every $T/2$ seconds or every $T/2$ seconds but the magnitude of phase transition will be limited to 90 degrees rather than 180 degrees so that sharp discontinuities of very high very abrupt kind will be avoided.

We will take it up next time because we are running short of time now, when we discuss what is called offset QPSK so it is a child of QPSK which has slightly more desirable properties than QPSK in this manner in the suspect we will start from here next time, thank you.