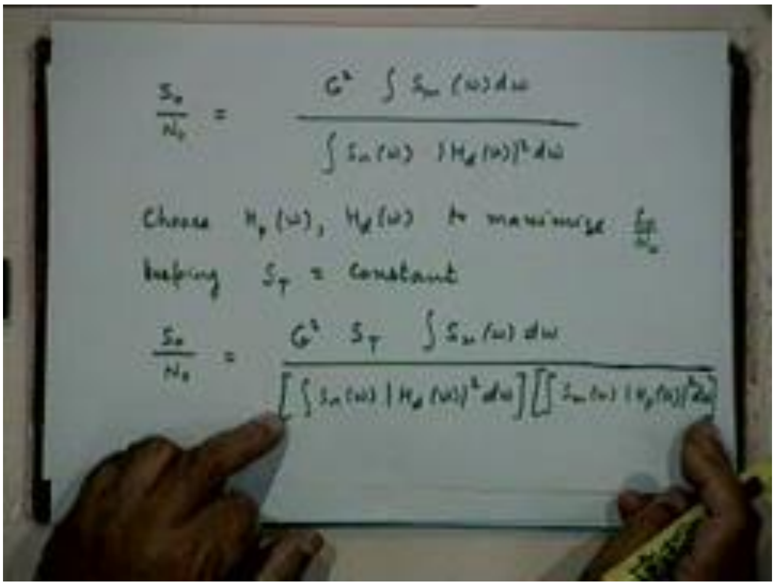
Communication Engineering
Prof. Surendra Prasad
Department of Electrical Engineering
Indian Institute of Technology, Delhi

Lecture – 38
Preemphasis – Deemphasis (contd.)
Pulse Modulation System

Let us recall that we were discussing the importance of optimum, optimization of Preemphasis, Deemphasis filtering. In the general context, we are not specifically referring to the FM systems or angle modulation systems although, they are most predominately used in angle modulation systems. However, the basic ((Refer Time: 01:36)) between baseband systems can be more or less easily carried to the case of FM systems, with some simple modification, which we will probably have not time more time to discuss.

So, let us complete the discussion that we were carrying out yesterday and you may recollect that, we have come to a stage we had come to a stage where we had written the expression for the output S n r. You know we have the block diagram, we are looking at it consists of three blocks, the transmitter, preemphasis filter followed by the channel, which also adds some noise channel has some transfer function h c omega and at the receiver we have the deemphasis filter.

(Refer Slide Time: 02:25)



And we were looking at the signal noise ratio at the output of deemphasis filter and the expression we have obtained for signal to noise ratio was this, I think this is the point we have left yesterday is that. And our problem we want to solve is, we want to choose a set of a pair of filters preemphasis filters and deemphasis filters, we have to choose a pair of these a combination of these two of course we know that this combination satisfies certain constrain, what is the constrain that the product of all these transfer function should be a constant, H p omega H c omega and H d omega.

But, within that constrain we want to choose these two filters, so as to maximize the output signal to noise ratio, keeping there is a weather constraint keeping the transmitted power a fixed value to be a fixed value, so it is a constant. So, as I said in general solution of these problems is carried out through using the calculus of variations, but for this case it is easy to solve the problem by making use of very simple well known inequality in mathematics called Schwarz's inequality, anyone know of Schwarz's inequality.

Student: ((Refer Time: 04:07))

Does not matter, I will just give it to you now, so to do that what we do is we slightly rewrite this expression, I multiply and divide this expression with transmitted power S sub T. So, this will become G square S sub T ((Refer Time 04:29)) of course, into S m omega d omega divided by, so I have this denominator which i have earlier and instead of S of T as a second term let me substitute for S of T what was S of T,

Student: Integral of S.

Integral of S m omega into...

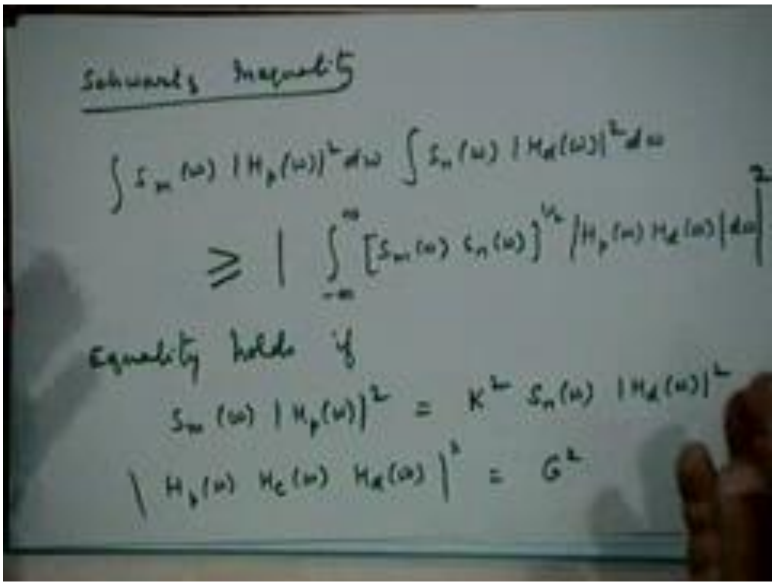
Student: H p omega.

H p omega mod square d omega, so I get this into this this is nothing but, S T the transmitted power is the signal is the power of the signal which is coming out of the preemphasis filter, input of preemphasis filter has a power spectrum density function S m omega. So, the power spectrum density at the output will be S m omega power H p omega mod square and area under the power spectrum density of the output is the total power transmitted. The power spectrum is given by this product and the total power being transmitted is the area under the power spectrum density is it okay all of you all of you with me on this, so we want to maximize this keeping S t constant.

Now to do that, if we can suppose you keep the ((Refer Time: 06:06)) if see really G square is constant S of T ((Refer Time: 06:12)) fixed, S m omega is a cons a property of the signal there is not much to, it does not depend on H p omega H c omega at all literally because, keeping S T omega constant in that sense. However, therefore what we can do is we can maximize the quantity by minimizing the denominator in some sense, but let me do not jump to any conclusions we will just see what that means.

To do that once again, there is a very famous inequality called the Schwarz's inequality, which we used to uh what is the minimum value of this product, we can say this product will always be greater than certain number. And it will achieve value this produce will achieve a minimum value under certain conditions, that inequality ((Refer Time: 07:06)) under what conditions that will happen is called Schwarz's mechanism, let me take a fresh page.

(Refer Slide Time: 07:13)



So, this is what it states this two integral S m omega mod H p omega square d omega into S n omega mod H d omega actually, ideally speaking I should have be using different w variables to avoid the confusion, we could do that if you wish, if I am using omega here I could use omega prime here. But, does not matter as long as you know the context there is no problem, this product will always be greater than or equal to mod square of this integral, what in this new integral I take the square root of both these terms the square root of the term here the square root of the term here and multiply the 2.

This becomes S m omega S n omega square root of that into H p omega H d omega modulus of that d omega, and take the square of the modulus of this integral. This is what the Schwarz's inequality state that is, if I have two integrals like this which are being multiplied their product will always be greater than the value of the single integral, whose integral is basically the square root uh the multiplication of square roots of these two integrals modulus square of these two integrals.

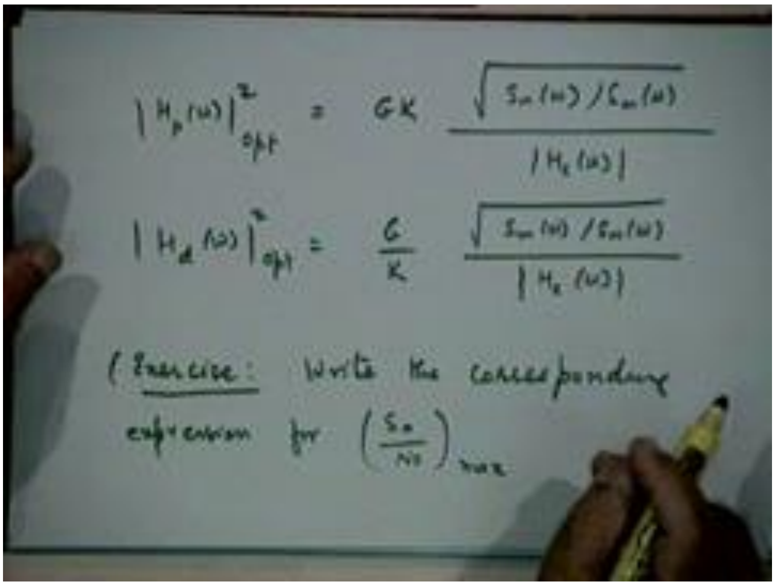
So, the modulus square of this integral will always be less than or equal to if you separate them out into these two integrals, that is Schwarz's inequality. And this also this is not the complete statement of inequality the complete statement is that, these inequality sign will become an equality sign these two will become equal under certain conditions and that condition is, that these two these qualities that we are using in the two integrals they are more or less the same except for a scalene factor.

If there are a scale version of each other if there scalene version of each other than this becomes equality, so equality holds if this is some constant term I am calling the constant denoting the constant by K square. The other term other integral S n omega into H d omega mod square that is, why you had an equation to use Schwarz's inequality somewhere other in the maths courses, because this is often keeps many applications, so it is not anything new for you.

So, that is the condition under which this product is equal to this, so what does it mean that means, when this condition is satisfied this product will have the least possible value which will be equal to this value, is not it if this condition is not satisfied this product will be always greater than the number on the right hand side. And the smallest value of this product is on the right hand side and that smallest value will be under this condition, so that really solves our optimization problem.

We are solving this optimization problem, we are trying to maximize this and in the process we know that the maximizing condition is basically given by this. This ratio will be maximum, when this denominator is minimum this denominator will be minimum, when this condition is satisfied and that specifies our requirement is that. So now, you can solve for H p omega uh to do that you will also need this condition H p omega into H c omega into H d omega modulus square is equal to G square, this was constrain that we have on filters, if you look at these two equations, together and solve for H p omega you can express you want to eliminate H d omega solve only for H p omega.

(Refer Slide Time: 12:35)



So, if I do that using these two equations and very easy it is check that your optimum value of H p omega, optimum function the preemphasis filter should have transfer function, which is equal to GK times square root of S n omega by S m omega upon mod of H c omega. Basically, using these two equations I am eliminating H d omega and solving for H p omega, that is very easily, because you substitute for H d omega square from here, you will get H p omega this entire expression will be in terms of H p omega mod square, which you can then solve for H d omega.

Actually, H p omega will come in the denominator it will become the power 4 and then you take the square root of that and that is why you finally get this expression. I have skipped one or two steps very simple and general steps, yes please...

Student: ((Refer Time: 13:36))

Why, ((Refer Time: 13:44))

Student: ((Refer Time: 13:46))

G has a dimension of H q, this is the ratio you will be left with H mod of H square, so it is fine I think there is nothing wrong. Similarly, if you solve for H d omega eliminate H p omega from these two condition these two equations, you can solve for H d omega and that turns out to be G by K into S m omega upon S n omega upon mod of H c omega all right. ((Refer Time: 14:58)) what will be the corresponding maximum value of the S n r by using that condition I will not write that expression, but you can write it down as an exercise, write the corresponding expression for the maximum value of S N, that is when y s condition is satisfied right that is very simple, so I will just skip that part.

I think, I will end this discussion by looking at the physical significance of these filters, so this is what the optimum preemphasis filter should be like, this is what the optimum deemphasis filter be like. If you look at these expressions carefully they will make a lot of ((Refer Time: 15:59)) our discussion, that H c omega the channel transfer function is an ideal transfer function, that is ((Refer Time: 16:08)) magnitude is equal to some constant, let us say equal to 1. So, this goes outside out of our consideration from from this discussion, so what we are we saying that the preemphasis filter will have a transfer function magnitude square root is proportional to the ratio of noise pass spectrum to the signal pass spectrum.

So, what does it mean, at least I must emphasize basically this this magnitude this transfer function will have large values at those frequencies, where the signal is relatively small is not it signal power is small compared to the noise power, remember noise will be added later in the channel, deemphasis ((Refer Time: 16:58)) transmitter. So, noise has certain distribution at certain frequencies, which is significantly high I must prepare the signal ready the signal to take care of that, I must enhance the signal to ((Refer Time: 17:14)), where I can expect the noise to be large.

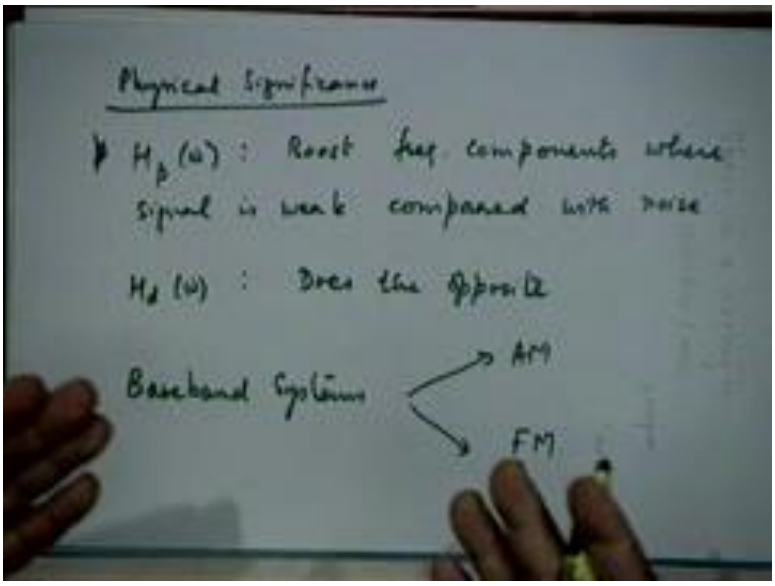
So, that I do not signal to noise ratio even under I must boost up those frequencies at the transmitter, where I expect a large number of noise, that part of the spectrum should be emphasized, ((Refer Time: 17:33)) we try to do. Optimum preemphasis fliter has to be this, similarly at the receiver I must do the other way round, the deemphasis filter is proportional to the ratio of the lesser spectrum power spectrum to the noise pass spectrum, we want to suppress the noise.

So, where ever the signal to noise ratio is good, which ((Refer Time: 18:01)) you put more voltage to that portion of the spectrum, where the signal ((Refer Time: 18:06)) less voltage to that portion of the spectrum. So, you emphasize where the deemphasis the portion of spectrum, where the signal to noise ratio is poor. And, because the constraint is satisfied you will ensure that there is no distortion, because of the product of the three transfer function is equal to one there will be no distortion, so this is what physical feature...

Student: Before, the preemphasis filter have no idea about the noise, it is going to ((Refer Time: 18:40)).

So, you have to have some idea, if you want to do optimal design you got to have some idea for example, you may you may even the noise the white Gaussian noise is also helpful, if it is white Gaussian noise you what it is, if want you know what it is you can optimize further. You have to have some idea in order to use this theory and ((Refer Time: 19:03)), so the physical significance I will just summarize.

(Refer Slide Time: 19:11)



So, p lets say H p omega will boost frequency components, where the signal is weak with compared to noise and ((Refer Time: 19:52)) suppress, where the signal is strong. H d omega will be the other way round does the opposite, but ((Refer Time: 20:08)) signal ((Refer Time: 20:09)), because your both H p omega and H d omega are together designed in such a manner, the product that of the three transfer function is constant, any questions.

Now, the other day I have carried out the discussion for the case of baseband systems only, extending the discussion to both A M as well as F M is possible, for AM it is almost similar, because all you are doing to AM is taking the frequency back. So, your filters if you want you can make these filters band pass signal, or you can still keep the filters at the baseband at the both transmitter and the receiver, because you you have done the demodulation in the process.

And, you have brought the signal back to the baseband, because the A M only carries out frequency translation the theory does not change at all, theory remains the same except some very minor detail about, where the which filters are your channel will be a band pass channel here, rather than that nothing else really changes. For F M things are little more complicated, but it ((Refer Time: 21:26)) giving a very simple argument you all can rest reduce yourself into same solution.

I will not go into those details, I like to leave that as self reading exercise, how to extend these ideas to the case of A M and F M primarily they remain the same. But, for FM they are particular this concept of preemphasis and deemphasis filtering is particularly important in view of the fact, that we have already discussed when we discussed in the context of tonal interference for example.

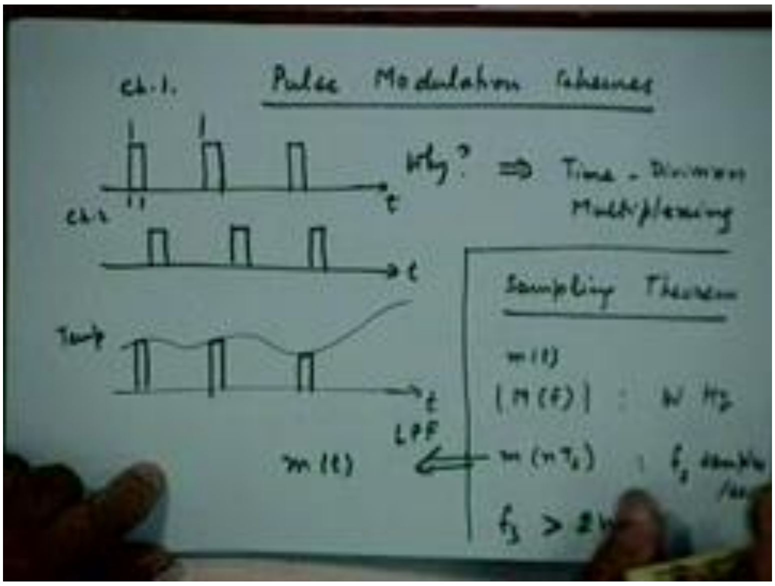
We discuss it in more detail and anyway the fact, that the threshold is likely to be, the likely make make use of every ((Refer Time: 22:02)) possible to make sure that your ((Refer Time: 22:05)) threshold value, so you optimize the performance ((Refer Time: 22:10)). So, now also if you remember the differentiation in the in the F M demodulation process, there is a differentiation that you effectively do at the demodulation. And that differentiation enhance the noise in fact you power spectrum, if you remember is a parabolic spectrum.

So the noise, that noise spectrum gets enhanced towards the edges and we can make use of this theory to use a deemphasis and preemphasis, sort of filters which will ((Refer Time: 22:49)), so I leave those details for self reading because of time factor. So, we move on to now a different sort of modulation scheme so far, all the modulation scheme that we discussed basically, three types of mod really two two types modulation schemes namely, various kinds of amplitude modulation and various kinds of angle modulation or explanation modulation.

In all these modulation schemes that we discussed, the carrier was a sinusoidal signal, now we can have situations where we use non sinusoidal signals and one of the important kinds of non sinu ((Refer Time: 23:38)), which are used as carriers or pulses. So the pulses are used to carry the messages of interest, of course we are still in the ((Refer Time: 23:46)) analogue modulation.

And pulses are absolutely basic to the transmission of different information, I will not talking about digital transmission here, transmission of digital information here and still ((Refer Time: 23:58)) transmission of analogue information, continuous signal continuous in amplitude and continuous in time. Even for transmission of such signals, sometimes we use pulse modulation, pulse is a carriers.

(Refer Slide Time: 24:16)



And when you do that ((Refer Time: 24:15)) resulting modulation scheme as pulse modulation scheme. Now, in pulse modulation schemes the first thing that we need to understand before we go into pulse modulation scheme is the fact, that you know ((Refer Time: 24:42)) carrier ((Refer Time: 24:45)) why...

Student: ((Refer Time: 25:05))

No, it is really application dependent,

Student: Reduce noise,

Reduce noise not really directly ((Refer Time: 25:19)), actually let us look at this one suppose we are suppose, we are measuring lot of data is somewhere, we monitor the conditions on the ((Refer Time: 25:33)) monitoring the conditions somewhere else, or in some ((Refer Time:

25:38)) remotely. So, you have a number of measurement devices and these devices all have to pass on the information to some central station.

So, there are two ways of doing it, you keep collect each of the cells that you have this monitoring devices that you have, it generates electrical information and modulates a suitable carrier and carries out a frequency division multiplexing of all the information and transmits it to central station.

Student: ((Refer Time: 26:09))

It is more convenient in such situations, because we do not need really a large number of carriers becomes the equipment becomes unnecessary complex, there is much simpler and it so happens, that most of the time your measurement rate is not typically that high in many many applications it is quite. If you measure the temperature at one time instant then measure something else, maybe pressure maybe, something else so on, and so forth and then come back and measure the temperature again in the mean time the temperature would not have varied too much.

So, what we are really doing is you are measuring each quantity passing on the information and coming back to ((Refer Time: 26:54)), you are multiplexing a lot if information by transmitting each information in a small interval of time. So, during this pulse interval let us say you measure the value of the parameter, maybe you modify amplitude and transmit ((Refer Time: 27:14)) amplitude.

And after some time, you come back to the same value same parameter you re measure it maybe the value has changed, so you ((Refer Time: 27:23)) different amplitude, ((Refer Time: 27:26)) and then just transmit the amplitude again. And in between these two, you have working ((Refer Time: 27:33)), a second set of pulses measure the second parameter and that parameter is transmitted in these slots and so on, and so forth.

You have depending on how width is how ((Refer Time: 27:57)), that you can make and how much time interval you have between pulses, that is how frequently you have transmit this information, you can multiplex ((Refer Time: 28:09)) information one after another. So, during this time interval, you are let us say you are ((Refer Time: 28:19)) channel 1 during this ((Refer Time: 28:21)) interval ((Refer Time: 28:22)) channel 2 and so on, and so forth channels are now different time slots.

So, basic ((Refer Time: 28:30)) why is ((Refer Time: 28:34)) simple way of multiplexing multiple messages for the same transmission medium. So if these two time ((Refer Time: 28:46)) alternative method of multiplexing information, one can as use frequency division multiplexing even in this context, but many situation in many situations in this ((Refer Time: 29:02)) much easier and much more convenient to do than frequency division multiplexing.

So, that is the basic implicit in this whole thing is the fact, that even if you are let us say temperature, so the temperature has a certain profile as a function of time exists temperature ((Refer Time: 29:32)) profile. So, it was implicit that you will not be able transfer this continuous information, now at best what you will doing is you are looking at this temperature information, at this time interval then looking at this time interval and so on and so forth.

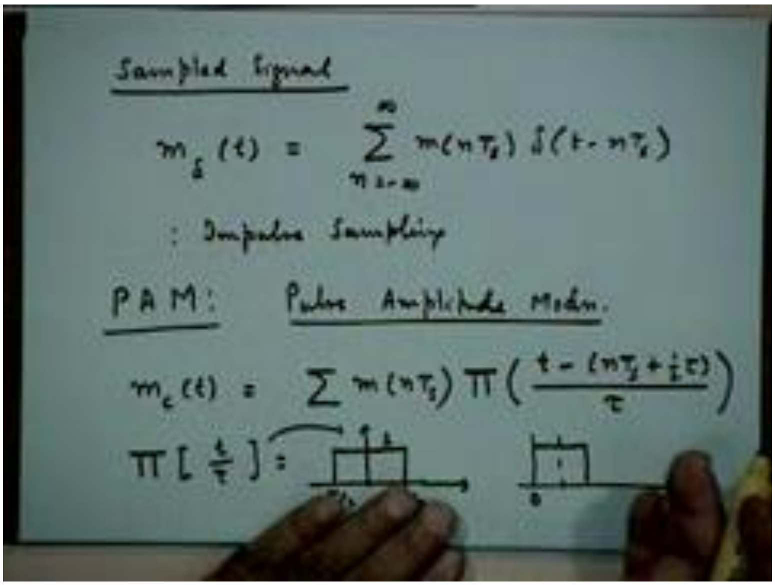
So now, you will say that you are losing information, you are not really getting all the information that you want, the answer is correct in general yes, but I am sure you know the sampling theorem. If the sampling theorem is, satisfied by this signal right and if you sample this parameter at a sufficiently fast rate you would ((Refer Time: 30:21)), you can reconstruct this entire roof form by just ((Refer Time: 30:25)) of the samples of specific time instance.

So, central to the theme of pulse modulation schemes, is the sampling theorem which all of you know, what does the sampling theorem state. The sampling theorem states that, if a signal if a message m t has a spectrum M of f, which is limited in bandwidth let us say to W hertz. So, sampling theorem is really for ((Refer Time: 31:08)) signals the spectrum is from minus W to plus W it has no specific components above W.

Then it is possible to reconstruct the continuous m t from, it is samples taken at intervals of t s seconds or taken at the rate of f s samples per second. If you just have the samples of the signal at the rate f s samples per second here, f s f sub s is greater than two W, if you choose a sample greater than 2 W, then it is possible to reconstruct m t from the samples and what is the reconstruction process, it will pass the samples through an ideal low pass filter.

So, the operation is low pass filter of bandwidth, this is summary of the sample theorem if you have forgotten it quickly review it, because the entire pulse modulation scheme is based on this result sample theorem.

(Refer Slide Time: 32:38)



So, the methodical uh description of sample theorem would be as follows, if I you impulse functions per second, you simplify the simplest form of sampling is if you were to if this each of this pulses becomes an impulse. Mathematically, the entire ((Refer Time: 33:00)) is very simple in that case, so the sample signal when used in pulses for sampling will be given by this, these are the sampled values m of n times T s delta t minus n T s.

So, the ((Refer Time: 33:21)) associated with the an impulse function located at, ((Refer Time: 33:25)) n T s n goes from minus infinity to plus infinity, so this is called impulse sampling of course this is not very practical. First ((Refer Time: 33:44)) modulation we discussed is by pulse amplitude modulation, so as you see now everything else is the basic concepts are the same we have pulse frame, which is your carrier. Instead of, a sinusoidal signal of some frequency you have a pulse ((Refer Time: 34:00)) carrier.

So, you want to carry out a modulation basically, what does it mean? You want to embed the information that you want to convey on this carrier. Again how can you do that, you can modify some parameter of the carrier in proportion to the information, that you want to embed. So, you want to make either the amplitude proportional to the message signal at the various time instances, what are the other parameters of a pulse.

Student: ((Refer Time: 34:29))

Width pulse, width is another parameter you can make the width of the pulse proportional to the signal amplitude at that point. You can make the position of the pulse, proportional to the signal amplitude at that point, these are the things you can vary there are the three parameters, which is very similar to the things, that we do in sinusoidal carriers. Modify, the amplitude give you amplitude modulation modify the phase, that is like shifting that is like, phase modulation is somewhat similar, modify the width that is like changing the frequency of the carrier ((Refer Time: 35:10)) in the I am just giving analogy these aren't exactly the same thing.

So, just like you have three parameters, where you you can play around you can also play around with these parameters and come up with different kinds of pulse modulation pulse, amplitude modulation being the simplest. Pulse amplitude modulation is essentially sampling, so the sample signal in this case is ((Refer Time: 35:45)) m sub c t c denotes the carrier, the carrier is the pulse here and that same thing as this into, instead of a impulse function you have a rectangular pulse function of the ((Refer Time: 36:05)) doing here ((Refer Time: 36:08)).

So, this is a notation for rectangular function, instead of writing r e c t, sometime just use this notation which denote a rectangular function, with some argument. And the argument is t minus n T s plus half tau upon tau, could you explain the notation pi t by tau is essentially ((Refer Time: 36:50)) function, this is ((Refer Time: 37:03)) and this function when shifted by this much, so basically you have to ((Refer Time: 37:11)) samples. You know, you are shifting this pulse to a general time instant n t s, also these pulses because of this ((Refer Time: 37:21)) implies, that we are shifting probably a special extra time 0.

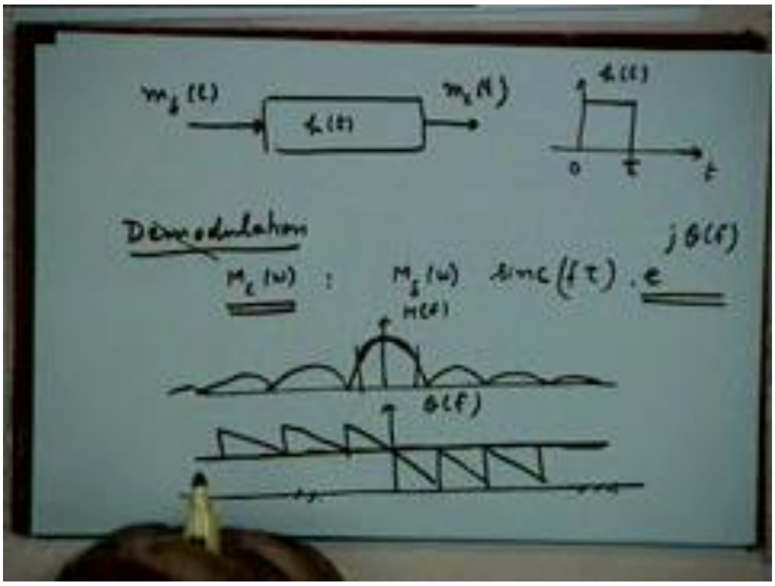
So, centre the pulse the first pulse is not zero, but it is ((Refer Time: 37:30)), can I can you tell me how can how I can get this kind of a signal from this kind of a signal. This is your impulse sample signal, how can you get a uh E M signal here this is a PM signal as you can you see if the n the nth pulse ((Refer Time: 37:54)), which is your rectangular pulse located at time n T s has an amplitude proportional to the message value at that time instance.

So, this is what we call a PM signal, now how can I obtain this signal from this signal.

Student: ((Refer Time: 38:10))

Basically, you have to go through a filter, which converts the sample impulse in to this rectangular pulse is that it that is what we have to do.

(Refer Slide Time: 38:29)



So, if I pass this signal pass this impulse sample signal to a filter, whose is impulse is cos h t looks like this, then if I have m delta t here, I get m c t here. So, I can generate of course direct generation is much easier, but mathematically this ((Refer Time: 39:00)) model because this helps you to understand the nature of PM signal. So, I am when you did your sampling theorem, we talked about impulse sampling natural sampling flat top sampling what kind of sampling is PM.

Student: Flat top sampling.

Flat top sampling PM, is essentially a kind of flat top, flat top sampling as we can see that is, what the definition is ((Refer Time: 39:25)) you are having the flat top pulse this is the pulse, which we are transmitting this amplitude is governed by the message value of that ((Refer Time: 39:35)). What happens in natural sampling, this pulse is not flat top, if amplitude varies in proportional to the variation of the message in that time interval, but that is what we call pulse amplitude modulation, pulse amplitude modulation is basically flat top sometimes, now if you this kind of a sampling what happens to recovery process.

How to demodulate it, so the modulation process is really very simple, modulation process is generating these pulses, these amplitudes are proportional to the message signal. So, to demodulate suppose you ((Refer Time: 40:29)), suppose I really had impulse sampling, I know the process of recovery the process of recovery is one of ideal low pass filtering, that the process of recovery that is all.

What do we what how ((Refer Time: 40:46)) change, when I have this signal coming in rather than this signal coming, have I done some modification here. Yes, I have multiplied the spectrum of this signal by the Fourier transform of this rectangular pulse.

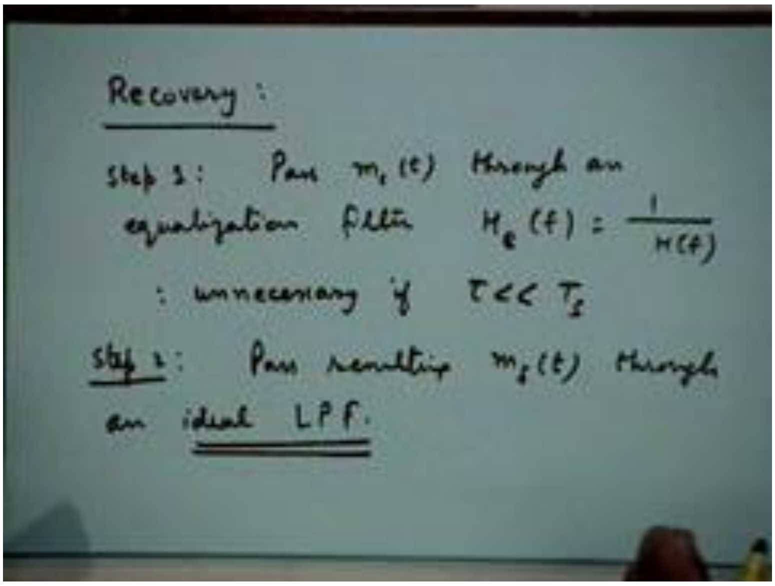
Student: ((Refer Time: 41:05))

So, you have to first modify have a receiver filter, which will have an inverse transfer function which will cancel the effect of this pulse modelling as you carry out ((Refer Time: 41:21)). Of course, you are not holding till the next time, but in the next impulse to a finite pulse and that modifies the spectrum how do we modify the spectrum, before I discuss demodulation, how before I ask you what is, how does M c omega differ from M delta omega.

Answer is, there is a product of this with what is ((Refer Time: 41:58)) transfer of this same function, of course there will also be phase function associated with this, because it is center at 0. So, if you pass some j theta f, which ((Refer Time: 42:19)) ignoring, there will be ((Refer Time: 42:24)), because of the phase shift, because of the time shift in the time domain uh more specifically H of f will actually look like this.

Modulate of H of f etcetera, value phase function also keep this as a reference will look something like this, you can verify that that is it ((Refer Time: 43:20)). So, the important point is this, that the spectrum of the received signal is not just the message spectrum that we are looking for in your it is very modified by this shape here, some kind of filtering is being done by this filter h of h of t.

(Refer Slide Time: 43:53)



So, to equalize for that so, that it is equivalent to coming back to your impulse sampling, so ((Refer Time: 43:56)) equal to your demodulation will be as follows. Actually, a two step process in step one, you pass the modulated signal through an equalization filter, which transfer function let us say H e f equal to 1 by H of f, so mind you this really would become unnecessary ((Refer Time: 44:38)), if you are pulse width is very small is that it.

Suppose, your pulse width is really very small then this ((Refer Time: 44:47)) will have very large width and it would be more or less flat in the dimension of the message signal, so in that case it will be not necessary. Because, let us see ((Refer Time: 44:59)) because, if pulse’s width is very small you are back to more or less impulse sampling. So, pulse width is large then this becomes important equalization becomes important, so unnecessarily, if tau is much less than T s.

And step would be pass the resultant impulse sample signal through a low pass filter through an ideal low pass filter, any questions here. So, that is all there is to pulse amplitude modulation, pulse amplitude modulation in summary is essentially a flat top sampling, so just like you recover the signal for a flat top sampling case, you are doing the same thing here nothing more in that you in the flat top sampling case, you do require an equalizer followed by ideal low pass filter.

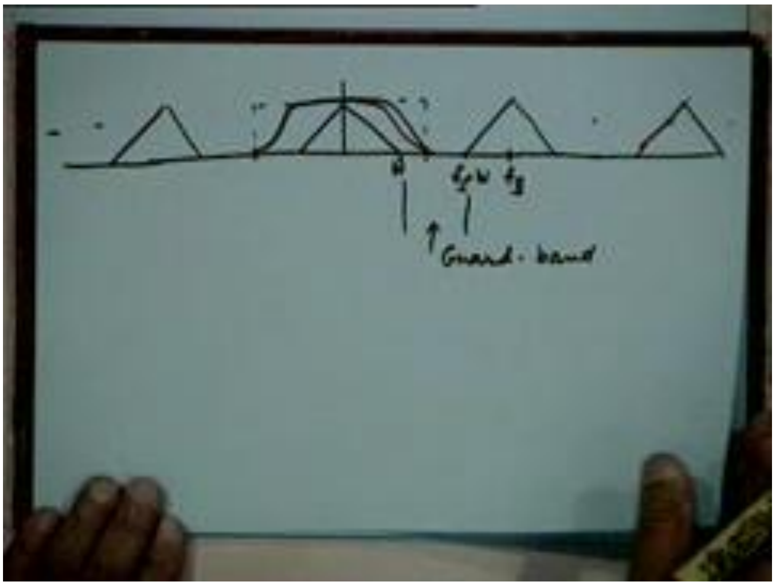
Of course, this requirement of ideal low pass filter can be a little stringent, because we do not cannot physical realize an ideal low pass filter typically, this is handled by how do you handle this situation. The fact is that low pass filter will be not be really ideal, so in practise how will you handle this situation.

Student: ((Refer Time: 46:37))

Your sampling rate, should be somewhat larger than 2 W you remember what, because I have not gone into the details of sampling here. And there I assuming that all of you know the reason why we require an ideal low pass filter. The ideal low pass filter comes, because if sample signal has a replicated spectrum, the spectrum of the sample signal replicates at intervals if 1 by t s it is a it is a periodic spectrum.

And you when you want to reconstruct continuous time signal back you have to keep only the central portion of the spectrum, removing all the other ((Refer Time: 47:13)) portion of the spectrum, which can only be done by ideal low pass filter if ((Refer Time: 47:19)) you make sure that the replicated components are well separated from the desired components, then you have a slightly better situation in terms of design of your low pass filter, let me just recap for if you are not recollecting.

(Refer Slide Time: 47:35)



Suppose, your message signal has this kind of a spectrum, then the sample signal has this kind of a spectrum ((Refer Time: 47:48)) infinity like that, extends up to infinity both sides. So if you want to reconstruct this signal back you have to pass it through a low pass filter, if you have sufficient gap between these then you have a possibility that is a ((Refer Time: 48:06)) ideal low pass filter, you can ((Refer Time: 48:09)) slightly non ideal filter.

Basically, that what the situation, where you difficulty in designing ideal low pass filter is critically handled by making sure that the sample rate is somewhat larger than 2 W. Because, this is sampled at f s, so this will be f s minus W, we do not want this point and this these two points to be close to each other there should be guard band. If you have a sufficient guard band, that guard band provides for designing more practical filters rather an ideal low pass filter that is one thing.

The second thing is if you do not provide for guard band, certain kind of distortion occurs which we call the ((Refer Time: 49:01)) distortion, this ((Refer Time: 49:03)) merge with this ((Refer Time: 49:05)) and we will have to do the separate the 2 out. In a low pass filtering, some portion of spectrum from the next term will also interfere with the ((Refer Time: 49:16)) and you are resulting reconstructed filter would be a distorted version of the ((Refer Time: 49:20)) signal.

And how ((Refer Time: 49:22)), but these are things you have already discussed in the sample theorem, so I have not, I am not going in to details of this, so that is as far as pulse amplitude modulation goes any questions in that. So, we quickly come to, we will stop here we will discuss the other two ((Refer Time: 49:42)) of modulation pulse width and pulse position modulation next time.

Thank you very much.