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

Lecture - 39
Pulse Modulation Schemes - PWM and PPM

We will continue with our discussion on Pulse Modulation Schemes, that we started last time, you may recollect that we talked about pulse amplitude modulation. And this to quickly recap pulse amplitude modulation is essentially manifestation of sampling process. We are basically flat top sampling the message signal and the amplitudes at those discrete time instances at the sampling instance are transmitted. In terms of amplitudes of either impulses, which is not practical more practically in terms of finite width pulses.

Basically the kind of sampling process that you go through is a flat top sampling process, the value of the signal that you pick up could be any value in the pulse duration. But the example that we took up, the development that we carried out was based on the leading edge modulation scheme. That is the amplitude of the pulse was proportional to the amplitude of the signal at the leading edge of the pulse. That is why when we pass this impulse sample signal to generate the flat top sample signal through a filter.

The filter impulses passed up a troop was a pulse function with a width of tau starting from 0, 0 to tau and we found how to demodulate the signal, if the demodulation had two aspects. One was the equalization process, which equalize for the fact that flat top sampling involves some spectrum distortion, in the base transition to compensate for that spectral distortion we first equalize the spectrum.

And then, pass it through an ideal low pass filter to reconstruct our continuous time signal back, just the sampling theorem revisited for you and that is also pulse amplitude modulation. I went through it rather quickly, because I was sure that all of you already know these stuffs and I just have to recap it for you, in case you have forgotten some of it please do recap. More importantly we will now turn to the two other kinds of pulse modulation schemes that are commonly used. Namely pulse width modulation and pulse position modulation, let us look at pulse width modulation first.

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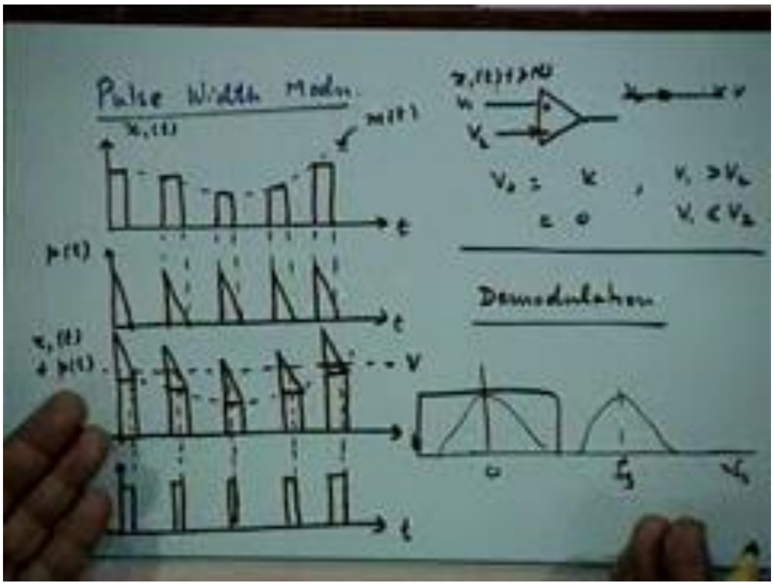

As the name obviously implies the parameter which we modulate here is the width of the pulse, that is you want to make a instantaneous pulse width at each pulse duration proportional to the, amplitude of the message signal, so that much is obvious. One of the obviously in the first question is how can we do such a thing, how can we implement this pulse width modulation. So, to do that let us just go through a series of pictures which would illustrate one way of by which you can generate a pulse width modulated signal.

So, let us say you have a message signal like that, to start with you generate a PAM signal, Pulse Amplitude Modulation signal which is very easy to generate, because all you need to do design is a sampler, I have not given you any circuits for designing a sampler, but that is very easy to do. All you to give the sampler what you need, all you need is a switch which will allow the signal to either pass through or not to pass through.

If you are giving the on period of the pulse, it will allow to pass through and of course, you will have to clip it if you want to have a flat top sampling signal and when the switch is off it is now allowed to pass through, so essentially it is a switch. So, you first generate a pulse amplitude modulated signal in the same as we discussed earlier, that is the first step. So, you generate this is the message signal m t and this is a PAM signal corresponding to this message signal.

Now, to this signal the simultaneous, let me call it as x 1 t, this simultaneously generate another set of pulses of constant amplitude, but the shape is triangular in nature and synchronize with these pulses, so this is what they look like. So, sort of triangular pulses this shape is generated, again they are very easy to generate, there are methods of generating sort of pulses and that is what we do. So, as a function of time, this is what the two way function ((Refer Time: 06:18)), let us call it p of t and then, I add up these two waveforms.

Student: ((Refer Time: 06:29))

Some constant amplitude, it is not important, so that we selected so as to satisfy the requirements, it is not very important. Now, you simply add up these two sequences, so that you will see you let me redraw that PAM signal, so on top of this pulse you will get this triangular pulse, you add up these two. On top of this pulse you get a similar triangular pulse, but this will now go up to slightly lower, because in the pulse that the PAM pulse to which you added has a lower amplitude.

Similarly, here you will get like that this will be the sum of the two, so like that you generate a waveform like that when you add up these two, having done that all you need to do is slice this waveform at some point. Have a threshold and it is only the signal it is passes through a comparator let us say, you have heard of a comparator, a comparator is a device basically a open loop operational amplifier is also a comparator, where ((Refer Time: 08:11)) comparator.

One of the inputs is this signal which is x 1 t plus p of t, that is one of the inputs x 1 t plus p of t and the other input is, the reference voltage is equal to this constant voltage let us say V. So, the output will be 1, the output of the comparator will be positive output for if the input signal is above the threshold value, above the voltage V either 0 may be depend on how you design the circuit, let us say if it is 0, then the input voltage is less than that.

So, what will this produce, this comparator output it will generate pulses of constant the amplitude, because it only give a either a positive pulse or 0 pulse, either the output will be positive or 0. So, during this interval you will get a pulse width like that, at this point you get a pulse width of this kind, now this will be even narrow pulse and so on, and so forth.

Now, if you can see the width of the pulse that you generate will be proportional to the amplitude of the pulse over which it is added, on which the triangular pulse is adding, that is precisely that you want. So, this becomes your PWM waveform, are you with me, in fact it is obvious because of the fact that this is a triangular pulse, the point of intersection of this constant voltage line with this added waveform, which generate different duration. During which the pulse amplitude, the sum amplitude is larger than the pulse width.

Here it is larger than the value V for much larger time than here than here and so on, and so forth. So, you will get a large pulse width here, a smaller one here and a still smaller one here. The pulse width will increase here again will increase further here, because during this time this voltage is greater than the reference voltage. So, you will get a pulse of constant amplitude, but it is duration will depend on the duration above which this remains other this threshold voltage, so that is really as simple as that. So, this is one way of creating the PWM waveform, there are others, this is the one way.

Student: ((Refer Time: 11:10))

In the width of the pulse, so that is what you will transmit, now

Student: ((Refer Time: 11:20))

Amplitude will be constant because comparator output, this produces a fixed amplitude output of some value, simply depending on whether this voltage is more than this or less than this. That is what a comparator I will do, because comparator has to tell you whether this one is larger or this one is larger and this it does by generating a voltage value which is either positive some fixed value or 0.

Basically the output will be a binary waveform here; the comparator output will be a binary waveform, which is what we have drawn here. It is operation is suppose you have call this V 1 and we call this V 2, the output voltage V naught is some fixed value. Let us say k volts for, let me write it like this. V naught is k for V 1 let us say greater than V 2, it is equal to 0 for V 1 less than V 2, that is what it does, that is what we have got here, that is what a comparator will do.

What about demodulation, if you look at how do you get the original message waveform back. So as you can see this is a different kind of sampling, you are still doing sampling, but the sampled information the amplitude in the waveform is not directly undergoing the amplitude of the pulse. It is underground the width of the pulse, now to look at the demodulation what was the demodulation based on for the case of PAM, it was based on the spectrum of the PAM signal, remember that.

The spectrum of the PAM signal, if you look at that or the impulse sample signal is essentially replicated spectrum. This is your basement spectrum, and then it gets replicated at the sampling rate, along the sampling frequency way around 2 F s and so on, and so forth. It is a periodic spectrum and the demodulation was based on the fact that the spectrum of interest for the continuous signal lies in the central node, around the base value, so all you have to do is remove the periodic components.

Remove the components other or the other the value for the signal by passing it through a low pass filter and you have the continuous signal back, continuous type signal back. So, to do the demodulation for PWM signal, therefore either you should look at the spectrum of the PWM signal.

Student: ((Refer Time: 14:01))

No, our objective is to generate embed the amplitude information of these pulses into the width information of constant amplitude pulses. That is our limited objective that we have able to do.

Student: ((Refer Time: 14:26))

Yes, so obviously you have to design the values of this triangular pulses and this amplitude of these pulses have to be designed in such a manner, that this operates properly, that is a matter of detailed design, surely. We do not want to have a situation where the sum of these two is below the threshold value of this. So, the value of threshold voltage will be decided by the range of the sum pulses, you have to choose them carefully.

So, that you always get a pulse of some width corresponding to the amplitude of interest, that is the method of design which I am not getting into I am sure it is easily done. So, coming back to demodulation if you want to do a similar thing ground, we have to understand what is the spectrum of the PWM signal. Unfortunately the spectrum calculation of the PWM signal is very involved and it is beyond the scope as well as the time available to us to do it.

I will just give you a reference for those of you who are interested in going through the calculation and it is like the spectrum calculation of an FM signal or PM signal is much more involved than the spectrum calculation of the AM signal. This is similar kind of computation, in face a little more involved in that and there is a very famous book in the area of communication it is a classical book, it is a communication classic for all time.

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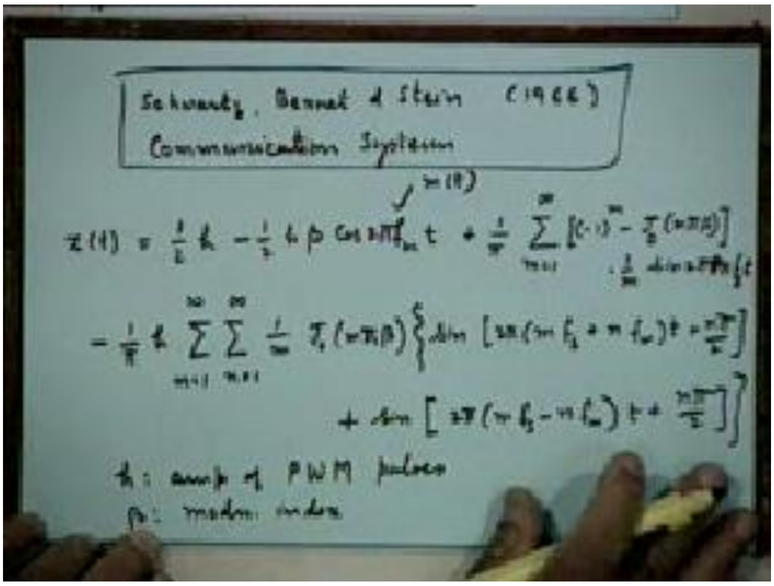

And that is the book by Schwartz, Bennett and Stein book from McGraw Hill and I think it is 1966 book, so very old book, but still regarded as a communications classic book it is called communication systems something like that. So, those of you who are interested in detailed spectrum analysis of the PWM waveform can go through this book and look at the derivation, but let me just recap the result for you.

It is little complicated, but we will give some insight and that insight is important, I am just coping the result from this book, I am not designing this results here, simply we do not have a time or this out of the scope of this discussion. The PWM signal call it x of t, if I call this PWM wave from as x of t can be expressed in terms of a series expansion like this, this is a constant value of half of h.

Basically you are doing some kind of a Fourier series the presentation of this, actually it is not the periodic signal, and one can see that it is not a periodic signal in general, but still it is possible to cover with some kind of a harmonic light representation. It is not exactly a harmonic representation as you are going to see and which looks like this, half h minus half h there is a parameter beta which I will defined shortly cosine 2 pi f m t, we are assuming that your modulated signal m t is ((Refer Time: 17:53)).

So, m t is cosine 2 pi f m t, so that appears in this expansion plus 1 by pi sigma m equal to 1 to infinity there is a large number of terms here, m th coefficient is minus 1 to the power m minus quite surprisingly or not so surprisingly the Bessel's function appear again. This is like when you go from AM to FM Bessel's function appear, J 0 of m pi beta into 1 by m sin 2 pi m sub s t, m f s is the sampling rate of the pulses, sin 2 pi m f sub s t, that is one term.

Here the, whether the one complicated huge term, but let me write it just for the sake of completeness, it may appear to be waste of time, but it is accepted. So, therefore, I will just write it, there is a general solution now, m pi beta once again I hope you can read it. sine of 2 pi m f sub s plus m f sub m into t plus m pi by t plus sine of 2 pi m f sub s minus n f sub m t plus n pi by 2, is it comfortable enough for you.

Let us see some pattern in this magnus if there is anything, h here refers to the amplitude of the PWM pulses and beta is a kind of modulation index which I will not define formally, but basically it is like this. Beta would be equal to 1, if the maximum possible pulse width if you use a maximum possible pulse width that you can, suppose we what is the rate of these pulses, what is the interval between pulses it is 1 by f s.

So, what is the maximum possible pulse width you can have, 1 by 2 f s, this is 1 by f s, so what is the maximum possible pulse width you can have without causing interference with the next pulse, it is half of this pulse width, pulse interval. So, 1 by 2 f s. So, when we choose a maximum pulse width to be equal to 1 by 2 f s beta is said to be 1. So, that is like 100 percent modulation that is what basically, so beta is this parameter.

If you choose the maximum pulse width to be less than 1 by 2 f s beta is suitably scaled down appropriately, so that is what beta is, it is a modulation index, based on modulation index. Now, let us see whether there is any method in this magnus, what you will notice is the twist signal has a DC component, has the desired necessity m t has one of the additive terms. And then, has components corresponding to the sampling frequency of the pulses as well as sum and difference of these frequencies.

M f s plus n f m and m f s minus n f n with all possible values of m and f. And of course, the amplitude of these components will somehow depend on the Bessel function values; if you want to plot this the spectrum waveform will look something like this.

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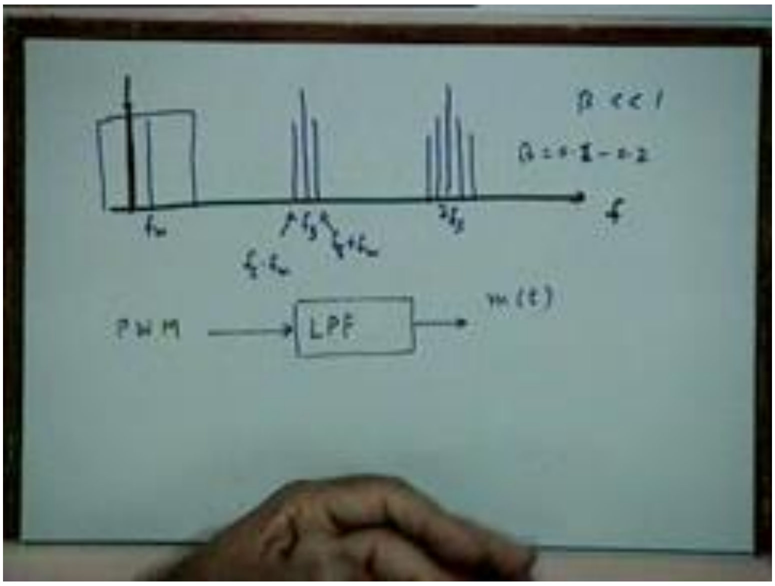

You are plotting it as a function of frequency, so as you can see there is a DC component. Let me use a different color, there is a DC component this term and then, there is a component at f m, then you have a component at f s and then, you have components around f s sum and difference. In terms out that how many components you have around f s, m equal to 1 will depend on various factors like beta and h etcetera, what is called the beta the modulation index.

So, the beta is very small that is you are using only a fraction of the maximum possible width that you can have, when beta is very small then around f s also you have any two components namely f s plus f m, this is like narrowband FM. It becomes like narrowband FM, so f s plus f m and f s minus f m and then, you have components around 2 f s and these amplitudes will be different and so on.

Then there can be additional terms here, this like we have many components in reference signal just rather only two side bands you have many components here around f s, 2 f s, 3 f s and so on, and so forth, not just plus f m and minus f m. However the interesting fact is and the useful fact is that as far as the basement part is concerned it contains only a DC component and the desired signal component. So, what kind of spectrum you have around f s and 2 f s is not very important as long as my basement signal is intact.

So, to recover the basement signal what should I do, again just put a low pass filter, because all these signals will simply disappear once again all these components will still. So, the demodulation for a PWM signal is therefore, once again the same low pass filter that you are using for the case of PAM or sample signal, they can of course, this is true when beta is small. So, when beta is let us say typically much less than 1 this spectrum is for example, typically between 1.1 to 0.2, the way I have plotted it.

As beta increases you can have larger number of components around f s and that imply that the guard band that you have for your problems like aliasing to occur, it will become smaller, you will have to watch out for that. But, for small values of beta the basement spectrum more or less remains intact and all you have to do is recover the signal that you want, the basement signal that you want by passing it through the low pass filter. So, PWM waveform is coming in pass it through a low pass filter and that will recover you message signal plus DC component which you can get it, any questions.

Student: ((Refer Time: 26:15))

Yes, half of h.

Student: ((Refer Time: 26:21))

Yes

Student: ((Refer Time: 26:24))

What I am saying is let it come; eventually I can always remove a DC component by coupling capacitor or something at the output. So, let the DC component come, it will come if you have a low pass filter the DC component will pass through that. So, let it come it does not matter, it does not do were it harm to us that is all I am saying ((Refer Time: 26:54)), yes please.

Student: In transforming modulating waves what about frequency axis scaled with the message band ((Refer Time: 27:02))

So, that is the good question, unfortunately I do not have time to go into the details of the answer to this question, roughly it is like this, typically you use pulse modulation schemes where you are not worry too much about bandwidth concern. Also secondly, of course, one can always come back and say that hardly in a system which you are not worried about bandwidth. So, second part of the answer is that I have drawn all these pictures using rectangular pulses, in fact is we very rarely use rectangular pulses. Rectangular pulses will have this problem of bandwidth, of course your concern is slightly different, your concern is that you have these repeated spectrum, because of the sampling itself.

Student: ((Refer Time: 28:02))

So, the very fact that you have agreed to use pulses implies that you have sufficient bandwidth to transmit this spectrum, because that is implicit in time division multiplexing you cannot do much about that. So, basically we are saying that our data rate of interest is typically much smaller or information rate is much smaller than the bandwidth available to us. It is only that, it is convenient to work with time division multiplexing in this case rather frequency division multiplexing.

So, bandwidth becomes a secondary issue in this particular situation right now, really speaking the bandwidth is not secondary, because until when we you must remember that certain bandwidth when we transmit it is give number of messages that will still happen. It will turn out the bandwidth efficiency of time division multiplexing and frequency division multiplexing will be of the same order will not be very different, but that is a matter of detailed. Unfortunately we do not have too much time to go into it, so may be some self reading will be required for you to take care of that, but that is a good question.

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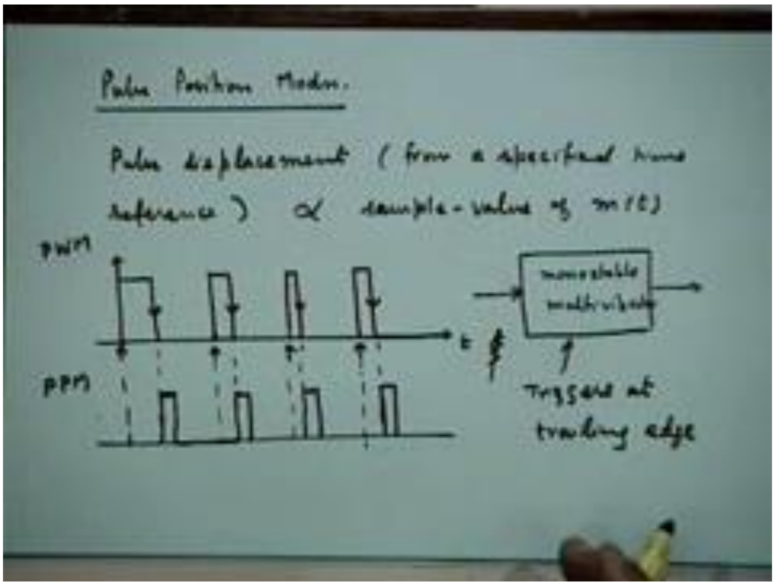

Let us now come to pulse position modulation, so what is pulse position modulation, as the name implies you will not have variable width pulses. You will not have variable amplitude pulses you will have pulses whose positions from some fixed reference position is proportional to the message signal. So, that is how you define pulse position modulation, pulse displacement from some specified reference point, specified time reference is proportional to the sample value of m t.

Of course, if you want this to be true typically, either you will have to have positions from both sides of the reference if you want positive or negative values of m t or best thing is to keep m t always positive by adding some positive bias to it, so that you always go in one side of the reference. Let us see that in detail how do you generate this pulse in such things suggest, can you suggest something, any suggestion for generating PPM waveform, anyone.

Suppose, let me give you a hint, suppose I have a PWM waveform can I convert this into a PPM waveform just like I convert a PAM waveform into a PWM waveform, is there a simple method of converting the PWM waveform to a PPM waveform.

Student: ((Refer Time: 31:20))

Let us look at that, let us suppose this is a PWM waveform, I am just drawing some waveform arbitrarily, how could I convert this into a PW. So, this is PWM this time interval is 1 by f s which is fixed. Now, you want to generate a pulse, remember this width is proportional to the amplitude or the message. So, basically where you want to produce a pulse at this point, here you want to produce it at this point, here reference point is this.

So, the reference points, time reference points are these, what you have to do is, generate pulses this position here is this and the pulse width is constant and the pulse amplitude is constant, this position here is refers to this position is this and so on, and so forth. This we finally get, this is your PPM waveform, so any idea how to generate that, is it clear please say what you want, yes

Student: ((Refer Time: 32:56))

Yeah, this is true this is our information about pulse starting

Student: ((Refer Time: 33:09))

If you differentiate in these two pulses positive edge and negative edge, what you do with that, this is your positive edge and you have a negative edge and that has to be converted into a positive pulse of this kind. A very simple way of doing it by using a device called monostable multivibrator, I discussed that in some other context may be no, I will discuss that stable there.

You do not have your digital circuits yes or no; you have a bistable, about flip flops.

Student: ((Refer Time: 33:48))

We have bistable and multivibrators, there are two stable states, there are two stable states, as you trigger it goes from one stable state to another stable state, in monostable multivibrator it is like a flip flop except that it has only one stable state. So, when you trigger it runs into an unstable state and after some time it comes back to the stable state, that is what a monostable circuit is. I will not go into the circuit there, because we do not have enough time this is not a circuit class, but that is what a monostable multivibrator does.

So, you have a trigger here and the trigger will make it go in to the unstable state, it will stay in there for some time and will come back to the stable state. So, let us say unstable state corresponds to a one output and the stable state corresponds to a zero output. So, temporarily produces a one output after some fixed time it will come back to which will that you want, temporarily to go to this side. One output after some time you want to come back to this steady state which is a zero output, it is a straight line till the next trigger pulse.

Obviously, the trigger pulse it should come at the PWM pulses, so basically you need a monostable multivibrator which triggers at say this triggers like this. You can always design mono stables which trigger at the leading edge or mono stables which trigger at the trailing edge.

Student: ((Refer Time: 35:33))

Impulses are very difficult to handle, they will have very large bandwidth that is why you always prefer finite width pulses, that mean you have finite width pulses, you are actually reducing the bandwidth required, and impulse is of infinite bandwidth. So, always have finite width pulses, the larger the width that you can have the better it is, but the width will be decided by how many signals you want to multiplex in time.

Suppose, within this duration you want o transmit many more signals, so then obviously there is a limit even to the pulse width you can have for individual signal. Suppose see this 1 by 2 f s maximum width I talked about was when you are transmitting only one signal. Suppose you are transmitting 10 signals and the maximum pulse width you can have is 1 by 20 f s, because this second circuit will also take it is range of 5 seconds.

So, it depends on the pulse width that you will end up using will be finally, dictated by the number of signals you want to multiplex in time, so that is a PPM signal, that is how you can generate a PPM signal. So, the demodulation once again one has to go through the spectrum I think you would not like me to write another expression for that kind. But without writing the complete expression I will say the first terms expression, which are useful to understand.

To understand what the nature of the demodulator would be, actually PPM and PWM bear the same relationship to each other that FM and PM bear to each other. If you think about it you can convince yourself about this issue and you will find that there is a similar relationship here.

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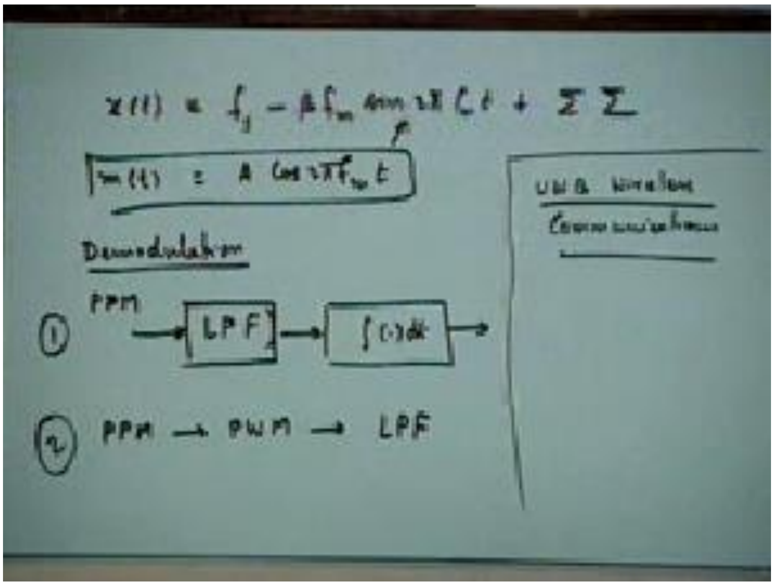

So, if you look at the PPM signal in detail and I am only looking at the first two terms in the expression please do not disturb like that. x of t is equal to f s minus beta f sub m sine 2 pi f sub m t plus all those horrible looking expressions, which I have not write which basically contains terms like MFS plus NFS these were and your message signal was A cosine 2 pi fm t. So, what do you see, this is your message signal, this term is not cosine 2 pi fm t, but it is derivative actually as you can see.

So, in a way when you go when you do PPM, in fact if you remember somebody said that you can differentiate a PWM signal and come close to a PPM signal. So, you can see from this spectrum expression also and therefore, that also gives you a clue to how to demodulate. Now, can you give a clue?

Student: Integrating ((Refer Time: 38:52))

Therefore, demodulating the first pass it through a low pass filter and then, integrate it, you have a PPM demodulator and the output of PPM demodulator is integrated to give you the. Your PPM demodulator output will give you sine 2 pi f m t or low pass filter essentially will give you sin 2 pi fm t, if you low pass filter it you will get this term to get cosine 2 pi f m t you will be getting like that.

You integrate this f m sin 2 pi f m t will become cos 2 pi fm t, so demodulation of a PPM signal is really very straightforward, pass it through a low pass filter that how PPM demodulate. There is another way of doing it, you can convert PPM signal first to PWM signal if you wish and pass it through low pass filter. I will leave that as an exercise for you to work out, how to convert a PPM signal to a PWM signal, so that is the second possibility, PPM to PWM to low pass filter.

This is the second possibility, so this is one method; this is second method think about this method on your own, so that is a very brief introduction to pulse modulation schemes. This analog pulse modulation schemes that I have just mentioned we have covered them very briefly they are largely used in telemetry applications largely or they where till very recently there was picture, of late they have become important also in other kinds of applications.

Example, this pulse modulation schemes have become very important in optical communications. I just like to mention some of these things for your benefit, in optical communications where you try to modulate the intensity of light in proportional to the message signal. Sometimes you like to transmit in the pulse mode, when you are transmitting in the pulse mode it is useful to consider all these options, width of the pulse amplitude of the pulse that is intensity of the pulse and position of the pulse that is one.

Another application where it has become recently very important is some of you have heard of these wireless communications, which are called ultra wideband wireless communications. You know transmitting over very long distances may be you are designing a limited area or actually it is presently more used in defense applications of some kind, basically we do not have a carrier once again. We are transmitting basement pulses in air, very narrow basement pulses, it is very difficult to do because antenna has to be very corresponding antenna is very difficult to design.

Nevertheless they are very secure for various reasons, because the average spectrum flow is low, because it is pulse widths are very small they are ultra wide band. And therefore, the signal level is, so low that somebody who wants to intercept it cannot see it; that is one advantage in that average spectrum floret is. So, these have become very important of late and now research is currently happening in this area.

And when you do this, basically you have a since you are transmitting basement pulses information in that can be embedded, either through amplitude modulation PA or pulse width modulation or pulse position modulation. So, just some of the model about these pulse modulation schemes just briefly mentioned before we close this topic, but remember what we are discussed is analog modulations.

We discussed analog modulation of sinusoidal carrier and we discussed analog modulation of pulse carrier waves why analog. Because the message amplitude is embedded in some parameter of the carrier which is again analog in nature, it is width or height or position.

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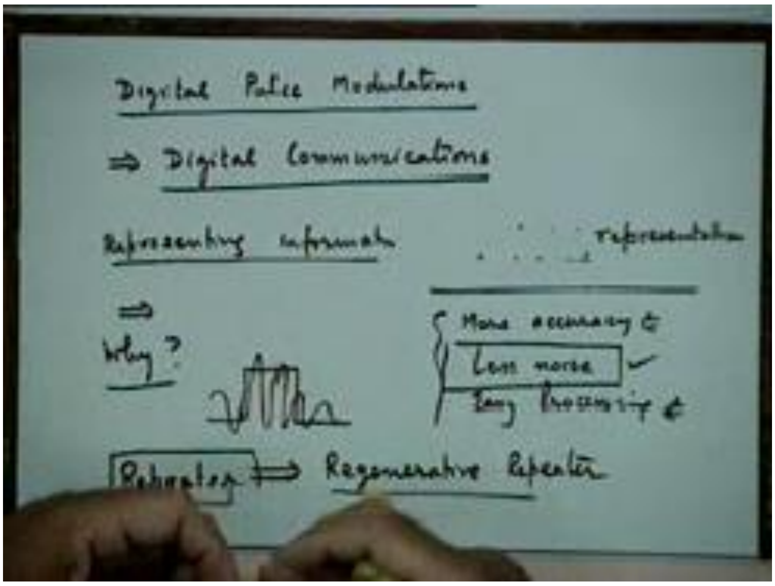

Now, as against, that as against analog modulations we can do this embedding pulse wave digitally, so you can also have digital pulse modulation, what is meant by digital about it. Basically what we mean by this that the final pulse waveform if you transmit on the channel is essentially binary in nature. Now, it is only a one zero information that you transmit and somehow information about the message has to be encoded into this one zero form ((Refer Time: 44:38)).

In fact, that is what eventually leads to digital communications which of course, you will learn in much more detail in a subsequent phase. But I will just try to give you a flavor of what happens in this communication at the end of this course. So, basically you are representing, your information in which is presently message signal which is basically analog in nature via some digital representation.

That is by digital I mean something which has a fixed number of levels, rather analog wise you can have any amplitude in a continuous range from minus infinity to plus infinity. When you talk about digital you are saying that no, we will not have any possible value you will have values which are discrete in nature, basically digital means discrete amplitudes first.

So, when I refer to the word digital it refers to discrete amplitude, the simplest example of discrete amplitudes is to have only two amplitudes 1's and 0's. But more general representation will have several amplitudes may be 4, may be 8, may be 16. You have can have a fixed number of amplitudes, you work only with the fixed number of amplitudes, now how do you represent a signal which has continuously varying amplitude with another signal, which has only a discrete set of amplitudes to work with.

And that when you do that you are able to convert that analog information into digital information, why should we do this, I think that we first discuss that, why. How of it is something that perhaps that some of you already know, perhaps you have learnt about analog to digital converters, so how of it is something that may not be dilute to you. Although there are two methods that I want to discuss here, the one which correspond roughly to your A to D conversion.

The second is much simpler it is called delta modulation. I have not discussed both of these very briefly, but more important today I want to discuss why of it, what benefits are possibly acquirable when you go from representing information of interest which you want to transmit from analog to digital, can you try to answer this question.

Student: ((Refer Time: 47:19))

More accuracy, let me write down the answers that here, more accuracy, less noise.

Student: Easy processing

Easy processing anything else, now if you have to vote for the most important reason for these three, which one would you vote for…

Student: ((Refer Time: 47:51))

As communication engineers you should have guessed the answer, it has to be less noise, because noise is our bane in some sense, we have to worry about noise all the time and it is this feature which is very important. So, basically you remember when you went from AM to FM what did you buy, it traded bandwidth with performance in the presence of noise. So, noise is of constant concern to us and that is the major advantage you worried upon, accuracy yes and no, depends on how you look at it.

Easy processing also yes and no, depends on how you look at it, but yes less noise is the crucial thing, now how can that come about this, noise in the channel will be the same. No matter whether you transmit binary pulses over it or you transmit analog messages over it. How can assembly come down? Can you think about that?

Student: ((Refer Time: 48:53))

Let us see, here is a digital pulse and you have this noise, where is your digital now at the receive what will happen by itself it does not do anything.

Student: ((Refer Time: 49:14))

The key to the answer is this is correct and when it is nearly there, but I am not sure whether you are fully understanding the total point is the following. You do not let noise do the real harm to you and recover the information before that. So, your noise is getting more and more as you are traveling, as your signal is traveling, signal is becoming weaker and the noise that is getting is becoming larger and larger.

That eventually happen when you actually transmit the signal over any communication medium, there are some clean media and not so clean media, but invariably this is the behavior, whether it is cables wires or wireless or any other medium. This is what will eventually happen, signal will become weaker and weaker as it travels, as it propagates and noise will become stronger and stronger. When you come to this situation you are already in a hopeless, you may not be able to do very much.

So, what we can do is make sure that the before it becomes too weak and before the noise overrides the signal too much, you recover the signal and make it absolute clean signal once again, you regenerate the signal, regenerate the clean signal through a simple device which is called as the repeater. A digital repeater, a repeater normally in digital sense only means amplifier, if the signal becomes weak you amplify make it stronger and retransmit it, but an amplifier will not do, because the amplify both signal and noise. So, repeater here would have to be of a special kind it would have to be a, what is called a regenerative repeater. In the sense that you have to clear up the pulse and transmit a fresh absolutely clean pulse once again.

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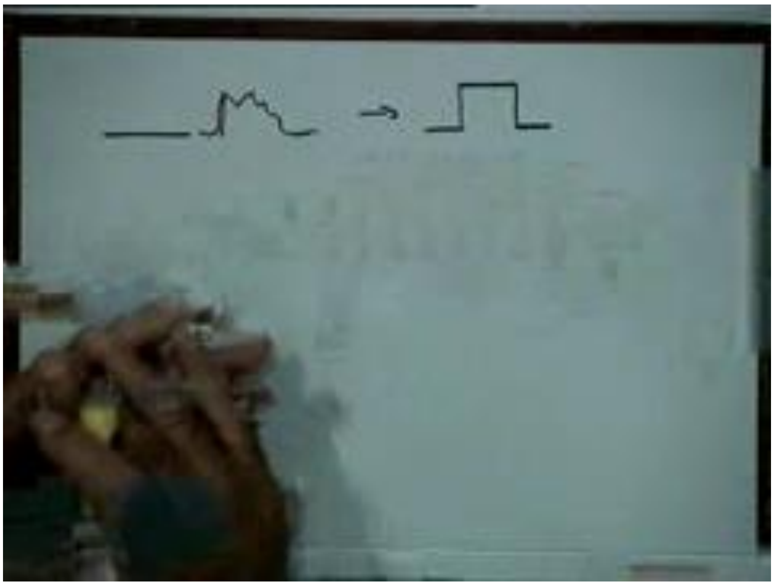

So, as the signal is propagated it becomes a little noisy on the way, so the pulse becomes like that, you clear up this pulse and convert this into a pulse like that, by a very simple clipping and work of clipping kind of operations. You retransmit it and keep doing this, keep doing this regeneration as and when required and if you are able to do that, you eventually get reasoning. And that is why you are able to get this course in digital communication.

And that is the essence of treating of bandwidth with noise once again, because the transmitted pulses rather continuous type signals, however it is clear that cannot do this with analog transmission. You cannot carry out such a cleaning operation ((Refer Time: 52:19)) message; because we have to separate the two, there is no way to convert that into something else.

Student: ((Refer Time: 52:25))

No, converting a pulse back to a pulse the clean pulse that is what we mean, that will be a single pulse that is coming. In which it is somewhat noisy is being converted to clean pulse in the same position, so the information is intact.

Student: ((Refer Time: 52:42))

Sometimes it happen, I mean this is I am giving idea ((Refer Time: 52:47)) as well, obviously you would not perfect answer, but that is much better than what you can get through analog. So, that is the reason why digital communication is so important as compared to analog. So, I thought stop at this point and we will discuss two methods of converting analog information into digital information in the next two classes

Thank you.