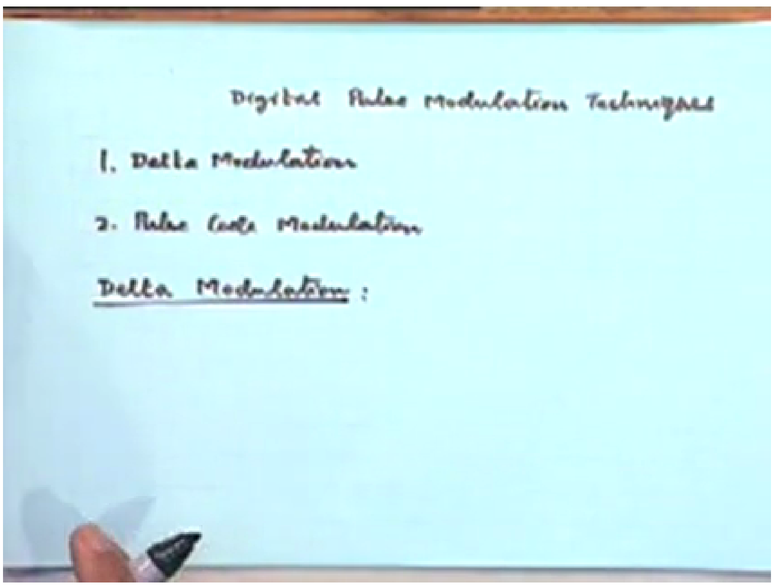
Digital Communication.
Professor Surendra Prasad.
Department of Electrical Engineering.
Indian Institute of Technology, Delhi.
Lecture-2.
Digital Representation of Analog Signals: Delta Modulation.

Professor: We are going to talk about digital communication in this course. And the very name implies that the information is transmitted in a digital form. Now in many applications information is naturally available in digital form. For example information coming from computers, if it has to be conveyed from one place to another place, it is directly available in a digital form in binary 1-0 form or whatever. But in many situations the information of interest is not naturally in the digital forms, right. For example most of the message signals that we are working with normally in our day-to-day life are not naturally digital signals, they are not in the form of ones and zeros, some are but many are not.

the voice signal through which I am conveying messages to you is analog signal, the pictures are there that you may want to convey also is an analog signal. many of the elementary signals that you have to work with when you are making measurements at a distance are also analog signals. so all these applications we are interested in communicating information or the information is not naturally in digital form. In all these applications, before we can make use of digital communication techniques, we need to 1st bring the information into a digital form so that we can use digital communication.

(Refer Slide Time: 3:11)



And therefore one of the requirements, therefore we can use digital communication is to have a technique by which the information of interest if analog has to be brought into the digital form, right. Now there are various kinds of techniques for doing this. they are broadly classified as digital pulse modulation techniques. Now I believe you are at least have heard of Analog pulse modulation techniques, if not, done it in your class, if you have not done it in your class, we will take it up separately in the extra slot. But if you have done or if you have at least heard, you must have heard of things like pulse amplitude modulation, pulse width modulation and pulse position modulation.

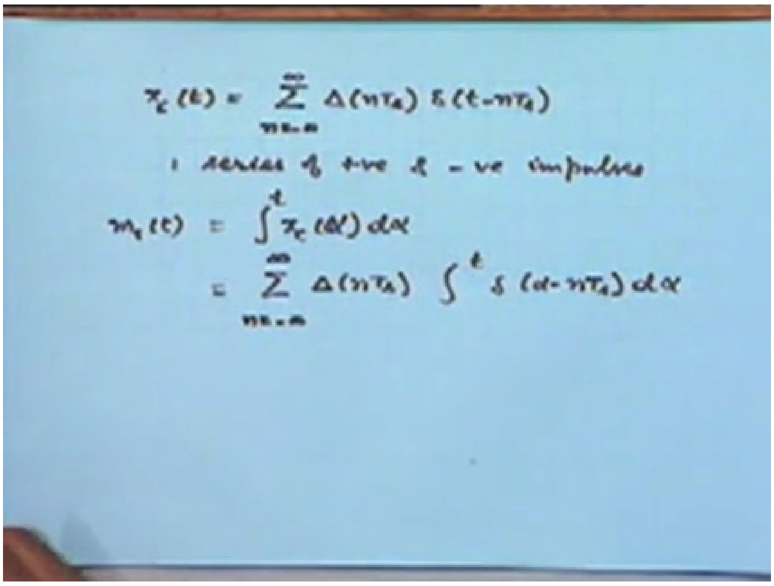
Basically in analog pulse modulation schemes will use a pulse train as a carrier, as against this sinusoidal carrier that you are familiar with for a normal analog modulations seem right. so here the carrier is in the form of a train of pulses. And the parameters, some parameters of the pulse train is picked up for modulation, for conveying information. typically could be the amplitude, it could be the pulse width which is made to vary in proportional to the secularist would be the position of successive pulses with respect to some reference frame of positions that is made variable or proportional to the message signal.

And we can take up the discussion of these if required in a separate class. In digital pulse modulations, instead of making some of these parameters analogously dependent on the message signal, the message signal sample value as sampled by any of the pulses is represented with discrete form. therefore that is essential difference between analog and digital. Analog is essentially a continuous thing, and digital is essentially discrete thing, so you know pulse train again as in analog modulations but the information conveyed by the parameters or let us say the amplitude is not a continuous amplitude, it is a discrete valued amplitude, that is also we shall refresh.

It could have finite number of discrete values, the samples of the pulses can go through a finite number of discrete values rather than a continuous set of values over either a finite or an infinite interval, right. Basically that is the difference between analog and digital pulse modulation schemes. In analog we have continuous variations possible, in digital only discrete variations are permitted, is that okay? Please feel free to interrupt if you have any doubts. Now there are 2 primary kinds of digital pulse modulation schemes that we do, they are to generate kinds of pulse modulation schemes.

One is called Delta modulation and the other is called pulse code modulation, okay. Of these Delta modulation is very simple to implement, extremely simple to mechanise and therefore we will consider Delta modulation 1st. so we will start by discussing Delta modulation. It is a very simple technique to encode a message signal into a binary 2 valued signal, where each sample, each pulse sample take only one of 2 possible polarity, positive or negative. And for the sake of discussion I will assume that the pulses that we are talking about at ideal impulses, just for simplicity of bringing out the concepts.

(Refer Slide Time: 7:51)



this is a block diagram of a simple delta modulator. As I describe it, maybe you can copy it if you want to. It is a very standard picture which is available. Is it readable? I will explain. the input to the this block, this system is the message mt which you want to represent in a digital form, all right. Now this is compared with a reference signal m st, m sub st which have somehow direction within this block and we will see how it is being generated, what is the nature of m st in a few minutes, right. so basically you look at these 2 signals and take the difference of these 2 signals which are denoted by dt.

so dt here denotes the difference between mt and m st, so let me write it here maybe. dt is mt - m sub st, where m sub st is a reference waveform which is internally being generated by the system. this difference signal is hard limited, I hope you understand what I mean by the term hard limited, that is this goes to a block whose output becomes +1 if dt is positive and becomes -1 if dt is negative. that is we, that is the operation of hard limitation, right.

Student: Sir what (())(9:33).

Professor: Well, we never go through a 0, we are assuming that dt is always positive or negative, I always nonzero, always positive or negative. And in real life systems one does not have really to worry about it too much. And this hard limited signal is being denoted by Delta t, right.

Student: Delta…?

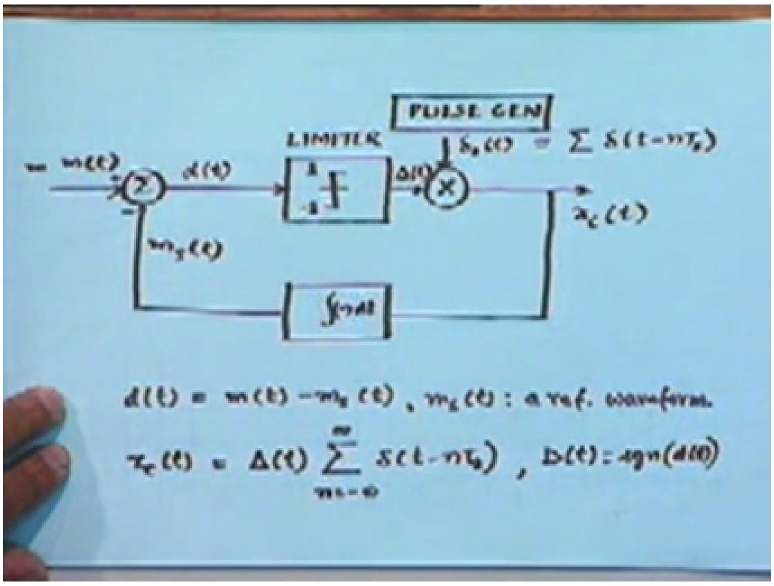
Professor: Delta as a function of t. so d is being transformed to Delta, D is analog signal, analog difference signal and Delta is obviously binary value discrete valued signal. Again it is a representation of the same difference signal but it has gone through a non-linear operation of hard limitation. this Delta t is being multiplied with an impulse train as you can see, coming from a pulse generator or impulse generator. similarly that is represented as sigma Delta t - nt sub s where t sub s represents a period of the impulse train, right. so these 2 are being multiplied and that is the output your delta modulator.

so the difference single Delta t, the hard limited difference signal Dell t multiply this impulse train and produces the delta modulus output. so what is the nature of delta modulator output? It is a sequence of positive and negative impulses, right. Each impulse that is going out is either of a positive amplitude of a fixed value or a corresponding amplitude but with a difference, negative polarity, all right. Now the same delta modulator signal is taken also to generate the reference waveform m st by simple integration of this through this process. Whereas the, how we can interpret this m st? Let us look at the delta modulated signal x ct, you can write this as the product of Delta t and the impulse train Delta t - nts, right, where Delta t is signum of Delta dt. Is that okay? I can remove the sheet?

Student: (())(12:23) What is signum of dt.

Professor: Signum is, you are familiar with signum function? that is if dt is positive Delta t is +1, if dt is negative, Delta t is -1, that is the mathematical representation of hard limitation, right. Fine? Okay.

(Refer Slide Time: 12:53)

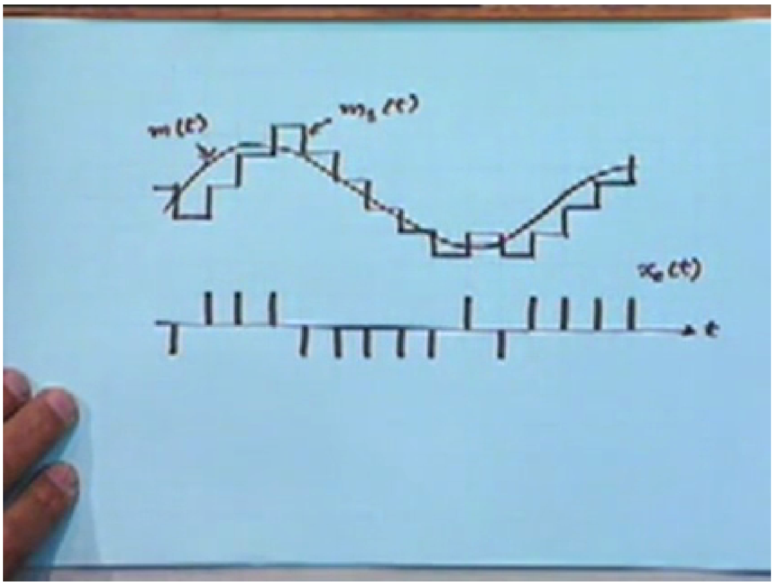


Or I can also write x ct as equal to, let us take the thing inside, because when I multiply any function of, any function like Delta t, even if t is continuous but I am multiplying it with your delta function, I need to really only worry about the values of the function at t equal to nts, the sampling instance, right. so which is essentially a series of impulses, series of positive and negative impulses. And whether it is positive or negative will depend on whether the difference signal is positive or negative, right, at the sampling instance. Difference signal as sampled at the sampling instance, that is what is actually…

Let us look at the reference signal m st, which is essentially integral of x ct, that means integral Xe Alpha d Alpha integral up to t, which you can write as Delta nts integral of Delta Alpha - nts d Alpha, okay. straightforward substitution and taking the integral inside because Delta nts does not depend on t. What is this, how do you interpret this? You can interpret this, what will be just the simple functions after integration? It will become a unit step function, right. so basically you have sigma Delta nts multiplied by unit step located at nts, multiples of ts, right.

so what will this create? this will essentially creates a staircase kind of waveform, in fact a staircase kind of approximation to your original signal nt, right. Let me illustrate by an example. I will come back to this line if you are out of this, I am removing it and that is what you have.

(Refer Slide Time: 15:35)



this is your original message signal mt and this staircase kind of waveform that you are seeing is the integrated signal or the reference signal m st. And the waveform at the bottom is the output of your delta modulator, this is x ct. so this is our delta modulated signal, a sequence of positive and negative impulses. Let us let us go through this waveform a bit carefully and see that indeed this kind of behaviour conforms to the diagram that we have just discussed.

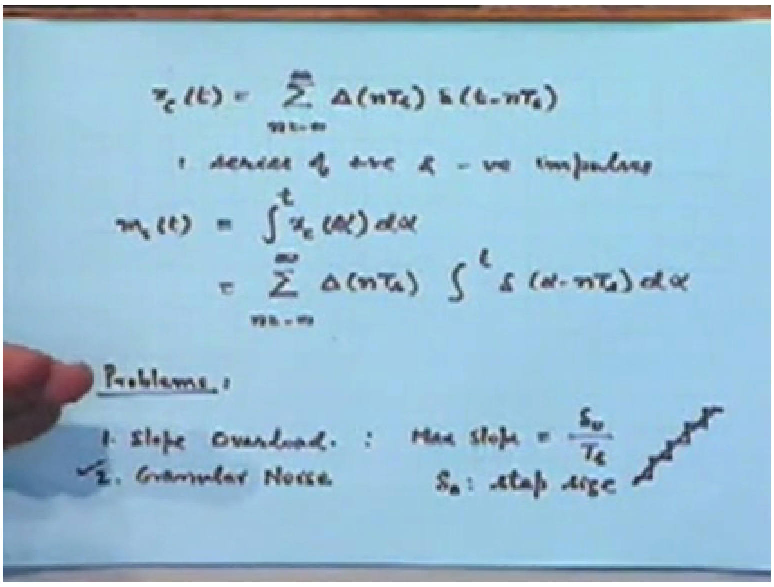
Let us say at this instance, this is your reference waveform mt, this is your value and this is the value of the, this is the input signal, this is a reference value signal and you are taking the difference mt - m st and therefore the polarity is negative, right. And therefore you produce a negative impulse at the output, all right. that negative impulse after integration is becomes a negative step, right. that negative step continues and at this next sampling instance, , this is compared with… Yes, so you look at mt - m st again which is a positive and therefore if positive impulse is created which is integrated again to produce a positive step. Right, you all follow this and this process continues.

And this process then generates a reference waveform which is essentially a staircase approximation to your original continuous signal mt, right. Also it simultaneously produces the delta modulated output, which is the sequence of positive and negative impulses. so that is a simple process of digitisation, you have to start with a continuous signal varying in any arbitrary manner but this x ct is the representation of the same continuous signal. Why is it, you can consider to be digital representation of mt. Why, because from this digital representation I can reconstruct the staircase approximation of the original continuous signal, agreed, by a simple operation of integration.

so given this x ct I obtain m st which is some kind of approximation of mt. Of course I have to make sure through appropriate design that m st is a good approximation of mt and we should also understand under what situation that it is a good approximation and when we will have problems, right. Given those problems which we will have to understand within those constraints, this seems like a reasonable way of converting a continuous time analog signal into a discrete time binary valued digital signal. A simple way of doing it, in fact as you can see the only components that are needed are a comparator, integrater and some kind of modulator, pulse modulator, right.

Which are all very simple things to implement, extremely easy to mechanise. And as far as the modulation is concerned, it is trivial. Given this, as the other hand all you need is an integrator because that will produce this. Of course you may like to do some kind of smoothening of this m st so as to produce a nice approximation to mt and a simple lowpass filtering operation is usually adequate for that purpose. so the demodulator will consist of an integrator followed by an appropriate lowpass filter so as to smoothen the output. Okay.

(Refer Slide Time: 20:25)



Now we said that we must produce a good approximation and to produce a good approximation we must understand what kind of problems might arise in the situation. Problems with the simple delta modulator, there are a few problems, in fact there are some serious problems, problems and difficulties. Primarily I would say there are 2 kinds of problems and they are both slightly contradictory to each other. One is called the slope overload problems, other is called the granular noise problem. Okay, let us look at what these problems are and how they are created.

Okay, I think we can look at here itself. Consider 1st the slope overload problem. You are approximating a continuous waveform mt with the staircase approximation, right. As a staircase approximation has a basic limitation, it can build up at a maximum possible rate which is governed by 2 factors, what are these?

Student: sampling rate (())(21:45).

Professor: The step size and the sampling interval because there is a limit on the maximum slope you can generate from a staircase, from staircase of these 2 fixed parameters, right. maximum value of the slope that you can have in fact is equal to Delta 0 upon ts where Delta 0 is the step size, right. And the step size will depend on the amplitude of the impulses that are going out, the strength of impulses that are going out, right. Because impulses are being integrated (())(22:28) steps. the stronger the impulses the larger the step, the weaker the impulses the smaller the step. Right.

But once you fix the impulse train the step size is fixed in that particular Delta modulator and it is going to be constant. And it is constant, let that constant value be Delta 0 and the maximum slope that you can build up is obviously Delta 0 by ts.

Student: But sir a larger step will mean more error?

Professor: Largest…?

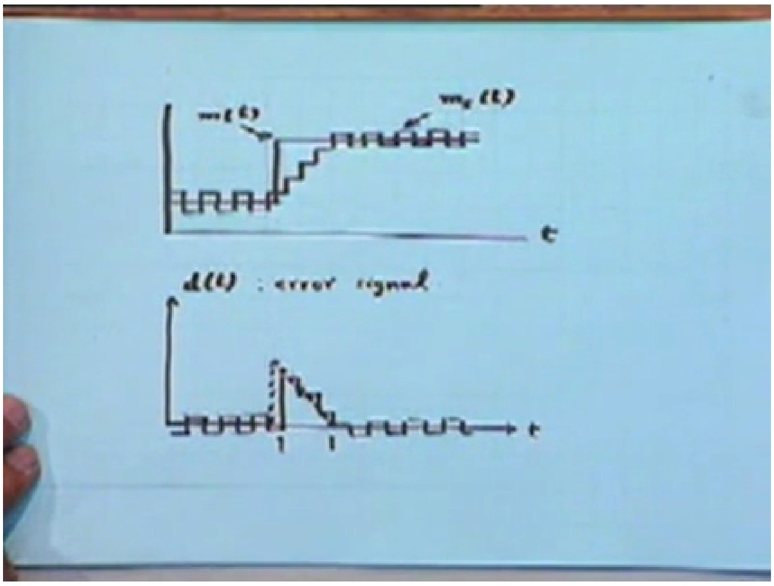
Student: Larger step.

Professor: The question is will a larger step mean more error.

Student: I mean when you are trying to approximate, suppose you go from a lower value to a higher value.

Professor: Let us talk of one problem at a time, that will, you are quite right, the obvious solution is that if you want to avoid the slope overload problem, you increase the step size and what you are saying is that if you do that, that will create more error, indeed that is correct and that is in fact the 2nd kind of problem that you are going to talk about, the granular noise problem. Right. so these 2 problems are, if you try to improve one, the other one gets worse, this is precisely what I am good value in a few minutes, but let me come back to it. that is 1st at the moment let me answer to the slope overload problem and will come back to the 2nd problem in a few minutes.

(Refer Slide Time: 24:15)

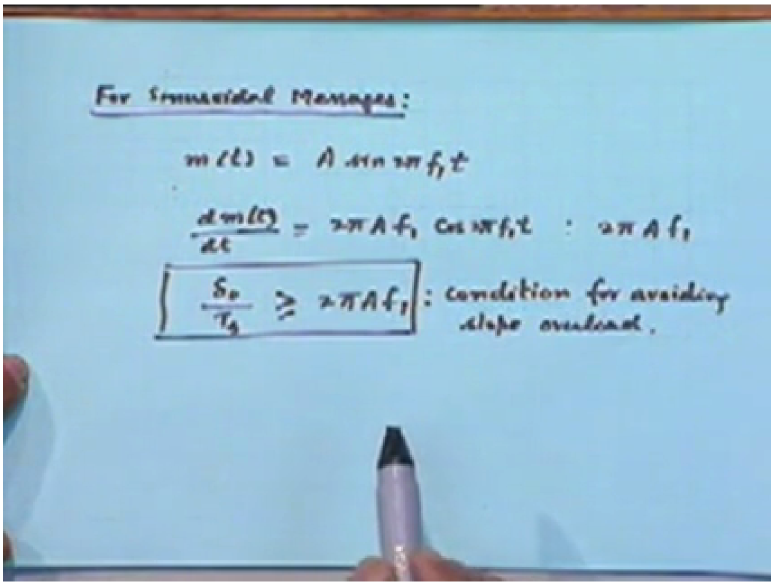


You all at least appreciate the problem of slope overload now, why it occurs. I have got this captured on this diagram here nicely, which shows a situation where your message signal mt has this kind of a nature, a unit step kind of a nature. Right. And obviously at the point of change, you have a very large slope, in fact infinite slope, right. Your Delta modulator which was approximating this like before the steps came along, suddenly finds the slope to be too much and cannot track that slope, it can only go as a staircase or the maximum possible slope governed by the step size and its sampling input.

Eventually it is able to track onto the new level but there is a, there is a timespan over which it is not able to track the waveform because of the slope overload situation. this is the error signal, the error signal looks like this, in this timespan you have a larger. Eventually it again becomes small but there is a finite time span over which the error is very large and that is due to slope overload. so you can see that the duration of the error due to the slope overload obviously will depend on these 2 parameters, the step size and the sampling interval.

there are 2 ways to reduce this, either increase your step size or increase your sampling rate, right, decrease the sampling interval, okay. so both the things, both the options are available to us. You need to look at this problem and write properly or it is okay?

(Refer Slide Time: 26:43)



Let us look at another kind of message signal, have looked at unit step kind of message signal here, let us look at another kind of message signal, the more standard kind that you normally deal with. Let us take a sinusoidal message, where a message mt is of the kind that say that A sin 2 pie F1 t, where F1 is some appropriate frequency. Dm by dt, the slope at time t is given by 2 pie times A times f1 cosine 2 pie F1 t, all right. And the largest value of the slope that you can encounter for such a message signal is obviously the maximum value is 2 pie A times F1.

so it depends on 2 things, it depends on the amplitude of the sinusoid and also depends on the frequency of the sinusoid, right. For larger frequencies you will get a larger value of the maximum possible slope for a fixed amplitude, right. therefore if you want avoid slope overload for a sinusoidal signal, what you have to do is make sure that the Delta modulator that you design has a combination of step size and sampling interval ts says that Delta 0 by ts is greater than equal to 2 pie times AF1, right, this is the condition that you must satisfy to avoid slope overload problem. Okay. Any questions?

this is the necessary condition so that m st follows mt faithfully when a large step comes along or for it to be able to track the maximum possible slope that it is likely to encounter in the sinusoid, right. Now let us just discuss this in a different way. Let me raise a question of sampling here. ts is some kind of sampling pulse, the sampling period in your delta modulator, right. You are familiar with sampling theorem, when you normally sample a signal and create a signal like PPM signal or something by sampling, you know that you have to be governed by the so-called Nyquist criterion, right.

The sampling interval should be greater than twice the maximum frequency containing the signal. Let me ask, maximum frequency containing the message…

Student: sampling frequency…

Professor: The sampling frequency should be greater than, should be greater than twice the highest frequency in the message, right, absolutely right. What do you think should be the value of ts here, should be of the same order or the different order, can you make an intelligent guess?

Student: It should be different.

Professor: Should it be lower or should it be higher?

Student: 1 by ts should be larger than this thing F1. It should be lower.

Professor: Why?

Student: Higher frequency. (())(30:24).

Professor: It should be sampling rate be higher in Delta modulator than the Nyquist rate or it should be lower, that is the question.

Student: Higher.

Professor: It should be higher. Why, so as to create , so as to further eliminate the possibility of slope overload. see, what are we trying to track here by sampling? We are tracking the difference signal, right. the difference signal, what is the difference signal, it is basically some kind of a derivative, it is the representation of the derivative of the signal because you are taking differential of the signal over small intervals of time, right. this is going to be proportional to the derivative of the signal. the bandwidth of the derivative of the signal in general will be much larger than the signal, bandwidth of the signal itself.

So for an appropriate, adequate representation of that signal you typically require much larger sampling rates. That is the intuitive picture, why you require large sampling rates. Of course the more exact picture is here given by the slope overload condition, right. The sampling rate that you should use is such that that for the maximum frequency that you have, F1 let us say, for the maximum amplitude for that frequency you may have, that frequency may have, right, the step size and the sampling interval should be related in this way. so this is some kind of design equation for the delta modulator, very important design equation.

Now you may say that if you want to reduce the slope overload problem, you can simply increase Delta 0, the step size, you can do that. Because the moment you do that you will have a capacity track larger slopes and go off to higher frequencies or larger amplitudes of the sinusoids, right. However this makes it difficult to track slow variations or small variations. For example, suppose we have a signal like that, let me let me just complete this, then I will come back to your question. When we have a large step size, you will now be able to track any slow variations, okay. so all these variations of the signal are killed because you have, as far as the lowpass filter output of this will be concerned, it will be more or less a constant value, okay, in this interval.

Student: Basically all step will have very high sampling rate.

Professor: Or we have to increase the sampling rate which is, in which also you have some limit. Let us say you have some limitation on the maximum sampling rate that you can use, then at that sampling rate the only thing you can do further beyond that is increase the step size, okay. Yes, now can I come back to your question please? Is it answered? Okay. so suppose I want to take care of both the things, that is, incidentally this problem that is the inability of the Delta modulator to track small variations or slow variations due to a large step size is precisely the 2^{nd} problem that I mentioned in the beginning, this is called the granular noise problem, okay.

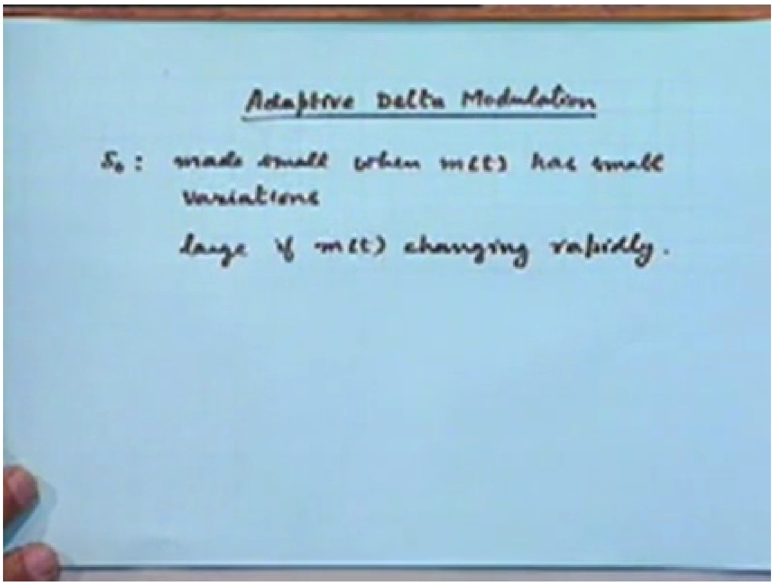
That is you have a kind of noise which is there and it does not represent any kind of variations in the signal, right. Small variations in the signal are just killed due to this kind of noise, okay. And this becomes more serious if your step size is large. So ideally it gives us some kind of an idea, intuitive idea as to what we should do to build a different kind of Delta modulator which will have less serious versions of both these problems, both the slope overload problem and granular noise problem. That is one which is able to track fast rapid changes as well as one that is able to track slower small changes, right.

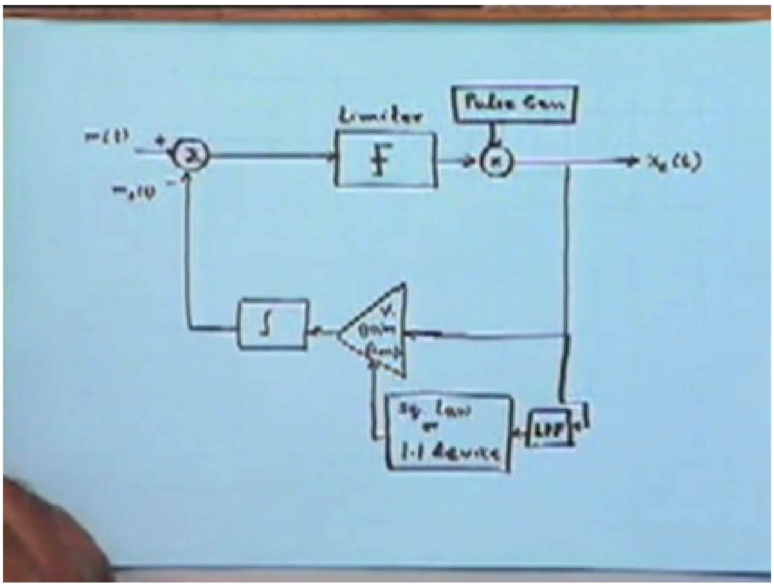
That is what I would like, I mean a signal, we do not know at what time you will have small variations or slow variations and what time you will have rapid variations. And we like to be able to take care of all kinds of situations, how do we do it? An ideal answer would be… Yes?

Student: (())(35:36) will it really matter if it is slow…?

Professor: Well, they are related, if it is small, it will also be slow, if you are talking of a fixed time span, right. so in the loose sense that I am speaking, slow or small, but they are related. Is a small variation takes place in a fixed interval of time, it is also slow, while is a large variation takes place in a fixed interval of time, it is also rapid, right. so in a very loose sense are talking about the similarities between slow and small. Is it okay? Let me come back to the point that I was trying to make. therefore I really wish to have a system in which which should have a small step size is where having slow variations in the signal and we should have a large step size if we are encountering rapid variations in the signal.

(Refer Slide Time: 36:55)





So if we could build a system which could do that, we could avoid both slope overload and granular noise, right. And that is precisely what we are able to do by a variation of a Delta modulator which is called Adaptive Delta modulator. the idea is extremely simple. the step size is no longer kept fixed, it is made to vary or it is made to adopt with the nature of the signal, right. so Delta 0 is made small when mt is nearly constant or has small variations, small or slow or whatever. It is made larger if mt is changing rapidly, okay. One possible way by which you can do this is the modification of the Delta modulator shown here.

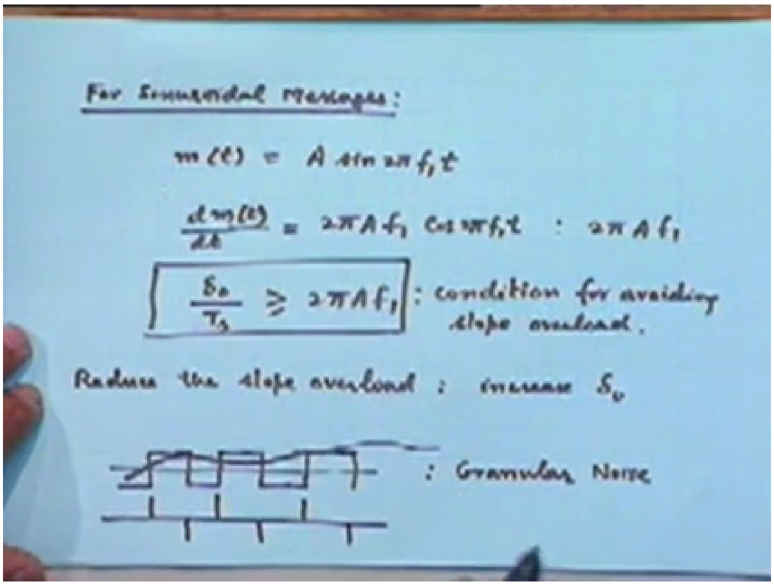
Everything is just about the same as it was earlier with one small difference. And the difference is I have introduced here is variable gain amplifier in the feedback path, okay, everything else is the same, this variable gain amplifier is an addition in the system. What does this variable gain amplifier do? It changes the impulse and that is going into the integrator, right. By multiplying the impulse by a gain you are changing the impulse train and therefore it is able to change the step size after integration. Is that okay?

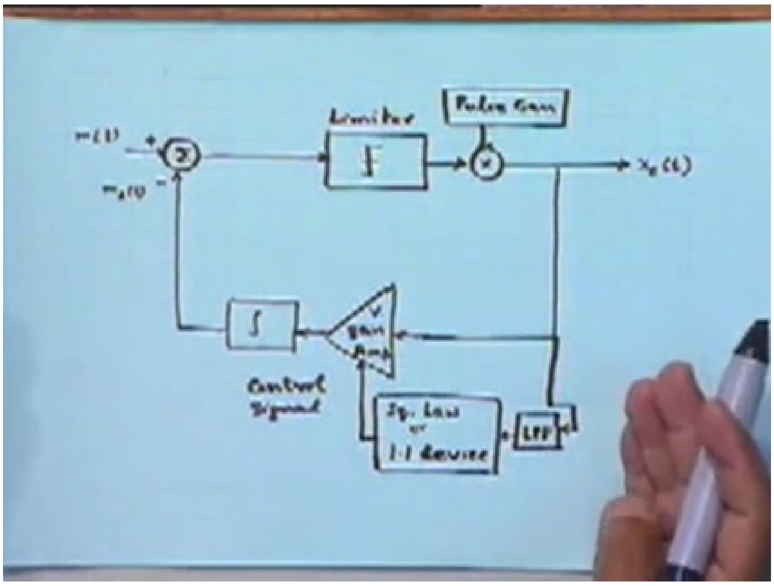
The gain itself is controlled because it is variable gain, it has to be controlled somehow by an electrical voltage which is also being generated from the Delta modulator output. This is the control signal going into the variable gain amplifier. This control signal will decide whether the gain is large or small, okay. When the gain is large, the step size will be large, when the gain is small the step size will be small. And what we would like to do is that the gain should be large when rapid changes are taking place and gain should be small when slow variations or small changes are taking place.

to understand therefore what we must do, let us look at the picture that we have seen earlier, I will come back to this picture in a minute.

Student: What is this (())(40:07).

(Refer Slide Time: 40:35)





Professor: Okay, this block here says square law modulus device that is indicated by modulus symbol here and this one is a lowpass filter, right. Remember this was the situation of small variations. What is the Delta modulated output is going to be? A sequence of positive and negative impulses, right? For this kind of situation we are going to have a sequence of positive and negative impulses one after another. Alternate impulses are going to be of different polarity. When we are undergoing a small variation, then we have a large step size this is what is likely to happen.

And what we would like to do is in this situation, reduce the step size, right. so when the Delta modulator finds this situation happening, this situation presents, that alternate impulses are positive and negative. You can interpret it to mean that the existing step size is too large, you must reduce it, right.

Student: But sir the limiter output is at a certain level, where are we reducing the step size?

Professor: the step size is controlled by the variable gain amplifier.

Student: (())(41:49).

Professor: Okay.

Student: (())(41:55).

Professor: One person at a time please.

Student: (())(41:59).

Professor: Alright, I think you have to still understand operation of the adaptive Delta modulator, please try to understand this. Nothing is changing yes, these are of fixed amplitude, they have to be of fixed amplitude because we only allow 2 levels. the output of x ct must be only binary valued, we cannot have variable value that the output of the Delta modulator.

Student: Demodulator will do?

Professor: But within the demodulator I have provision for a changing the impulse train that is coming from the Delta modulator output.

Student: that will be the same block will be used in the demodulator?

Professor: Of course. the demodulator… I will come to the demodulator later, at the moment I am only trying to understand how to track rapid as well as slow variations, right. Obviously the feedback portion will also become your demodulator later.

Student: A better way would have been to directly take the derivative of mt and then have that control, this is an indirect way of physically getting a derivative. What you are doing is just a crude way of (())(43:07) lowpass filter smoothening things and all. 1st we have the difference which you, which is approximately the derivative, then you smoothen it using the LPF and then physically you are applying that to control.

Professor: Let me ask you, when you say I should take the derivative, how do you go about doing it? this is precisely what we are doing here. the whole idea of adaptive Delta modulator is or Delta modulator for that matter is that indirectly it is taking a derivative as well as creating a modulated signal. And within that framework we want to have a very simple implementation.

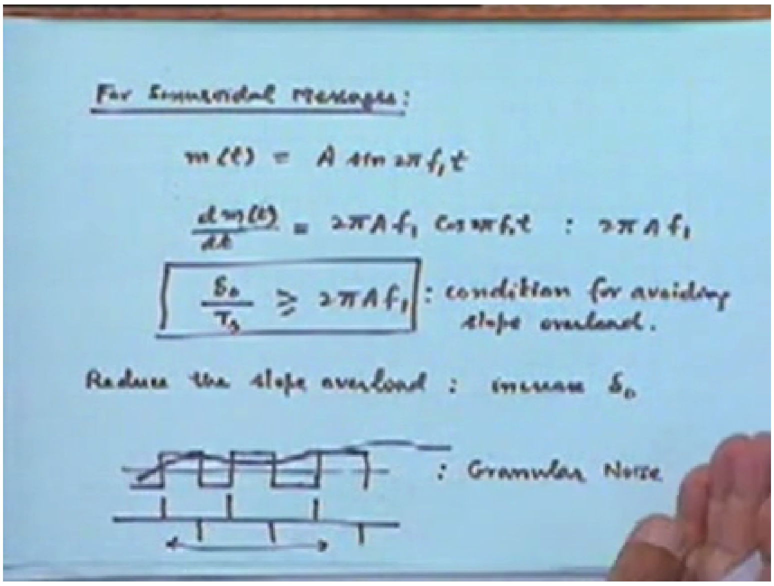
Student: so why do not we have the direct analogue…

Professor: the whole idea is that to be able to have a very simple mechanisms which is sensitive to the nature of the signal, right and this is able to do it. Incidentally this is obviously not the unique implementation, many possible variations of this implementation are there and are possible, right. And we can consider some of those which you are suggesting separately. But we would like something which is within the feedback loop, right, for various reasons. so because we would like to have a closedloop system, part of which also serves as the demodulator, right. All these conditions are nicely satisfied by this system which will be difficult to satisfy if you are not doing things in this way but not necessarily begin always cover with a different way of doing things which will work as well.

But have you understood the operation? We were trying to understand the operation of this and to understand, to verse that, what I was trying to suggest that when the data modulator produces an alternate signal of positive and negative impulses, this system will interpret it to mean that the step size as it exists is too large and it must be made small, right. And therefore what must be done in this system is to reduce the value of this control signal, right. And that is precisely what will happen because this lowpass filter will produce an output which is in this period where the output is the sequence of positive and negative impulses, it produced a nearly 0 DC kind of value. Right, small value.

Student: (())(45:29). How is the signal impulses are going to leave…

(Refer Slide Time: 45:46)



Professor: Lowpass filter is essentially some kind of an integrated over a certain period of time, right. so suppose it is looking at this time constant, it is time constant is about this much, so it is looking at a sequence of for 5 or 6 impulses and looking at the average value which is going to be, which is going to be very small because alternate impulses are positive and negative. they are going to cancel each other's effect, right.

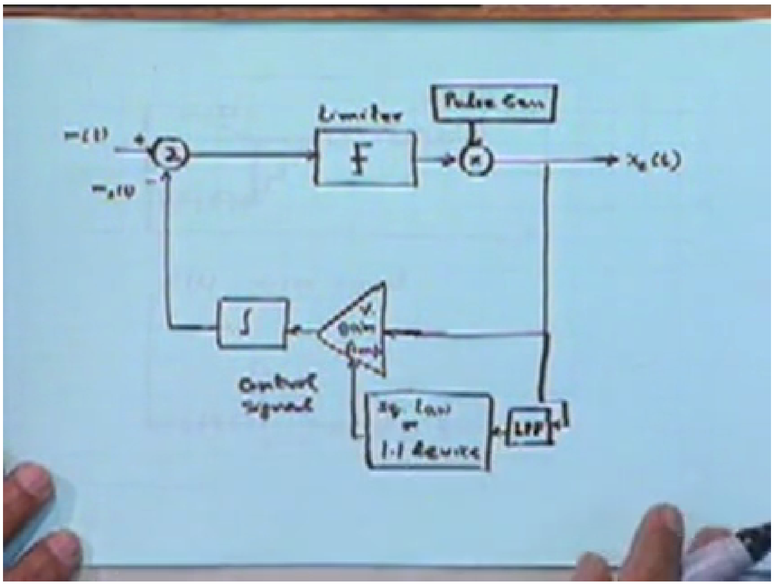
Student: (())(45:58).

Professor: there is no need to that, otherwise I just want to give you a concept, okay. We can go into implementation details later.

Student: How fast is this (())(46:13) like? After how many such things can decide that it is…?

Professor: that is the design parameters and that is what will decide time constant of a lowpass filter. A lowpass filter that you must design should be such that it is able to do this over a, over the required number of sequence of alternate impulses, positive and negative impulses. But there is a matter of details which at the moment I am skipping because I just want to introduce the concept to you, I am not going into that much detail at the moment.

(Refer Slide Time: 46:55)



On the other hand if you have a slope overload situation of this kind, right. this is our slope overload situation, what will you get as the delta modulator output here, a long sequence of positive impulses or a long sequence of negative impulses. For example if the step was in the other direction, you will get a long sequence of negative impulses. so in that case the lowpass filter average value output will be large in magnitude, but of course it could have any sign, positive or negative.

And that is the reason immediately followed by the lowpass filter I have a square of device or modulus device because I am not interested to know whether the difference is, whether the error is large positive or large negative. I know that is large and therefore I must increase the step size, the slope overload exists, I must increase the step size. And therefore the control signal becomes large and the variable gain amplifier gain is increased, step size is increased and that is the basic principle of adaptive delta modulation.

Student: If there is alternating, how will we reduce the step size? I mean the lowpass filter is giving a DC, so how will be reducing the step size?

Professor: A DC signal, the small DC signal results in a small value of the gain because it is controlling the gain. And the manner of controlling the gain is that when the control signal is small the gain is very small, when the control signal in large the gain is very large, that is all, it is controlling the gain of this amplifier, this is a control signal only, right. After all we need some parameter to indicate whether this kind of condition exists or that kind of condition exists, based on that we have to take action in the form of changing the gain of the amplifier appropriately. that indication is coming to you through this control signal. Fine.

As far as receiver is concerned, obviously the receiver is going to be precisely just this. take this out at the other end and use this as a receiver.

Student: Probably smoothening this thing after (())(49:11).

Professor: Of course, after the integrator you can have for the step of smoothening with an appropriate lowpass filter. Okay. so this finishes my discussion on basic principles of delta modulation and adaptive delta modulation. We can take up some of the issues that you have raised in the class tomorrow that we will have.