Digital Communication.
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Lecture-5.
Quantisation Noise in Delta Modulation (continued) & Time Division Multiplexing.

Professor: So we were talking about quantisation noise in Delta modulation yesterday. And we have to make a few assumptions regarding the nature of the noise that is produced and one was that the amplitude is uniformly distributed between - Delta and + Delta which is the step size we have assumed for the delta modulator. And secondly we said that the power spectral density function of this noise is also uniform in the frequency domain between 0 and fs or - fs to + fs. Under these 2 assumptions that we made, we could calculate the noise power as seen at the output of the lowpass filter, right.

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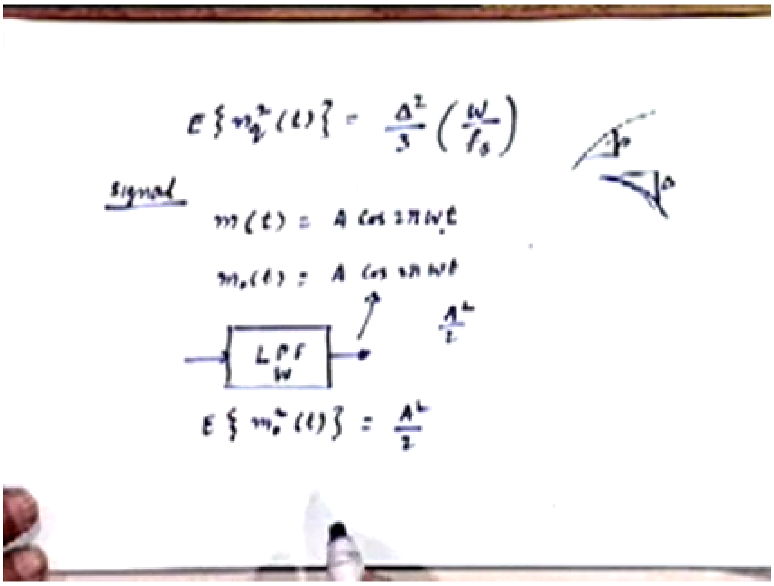

And the expression was… Right. Delta is the step size of the delta modulator, W is the signal bandwidth, that is the highest frequency components present in the signal which is being delta modulator. Fs is the clock rate or sampling rate being used in the modulator. So these are the parameters of the delta modulator and this is an attribute of the signal which is being delta modulaedr.

Student: (())(2:45) - Delta by 2 to delta by 2…

Professor: Now, you could have a situation where you are just, after, let us say, I will just write here. You just reach here and that is Delta, right and you could have another situation where you just reach here while tracking this portion of the waveform. So this is + Delta and this is - Delta, right. So you could have both kinds of situations. This is the waveform being tracked and you could be tracking like that, well this is the waveform being tracked and this will be the tracking delta. So that is why this from - Delta to + Delta rather from - Delta by 2 + Delta by 2. This is not like PCM, you are doing quantisation of the kind you are doing in PCM, you are just trying to track the waveform and that is the reason why the limits are different. Okay.

Student: This is why we are able to track it (())(3:52).

Professor: Yes, the other assumption we made is that we are working within the slope overload limits of the delta moderator. Under these limits that we can make those assumptions because if we have crossed the slope overload point, then our error can be much larger than Delta at any point, right. So it is within, we are assuming here that we are working within the limits of the slope overload of the delta modulator. That is right, I forgot to mention that. Now let us come to the signal, this is as per the noise is concerned, because we would like to calculate the signal-to-noise ratio.

What we will do it, if you remember your design procedure for the delta modulator , we like to fix a step size Delta from slope overload considerations based on the highest frequency components that you would likely to have in your signal. Because the slope overload condition depends on the maximum slope you can have in the single and which will depend on the frequency of the signal as well as on the amplitude of the signal, right. Amplitude of the component of the, frequency component of the signal. Let us assume that the highest frequently component in the signal is W and just for simplicity let us consider a sine wave with that frequency. Right.

So I am assuming the signal to be for simplicity a sine wave whose frequency desires possible frequency that you can expect in that single, right, to get a closed form expression. And therefore obviously lowpass filter is also going to have bandwidth W and the output of the lowpass filter will show as far as signal is concerned the same thing. So what we are talking about is a situation like this, you are delta modulated signal is coming in, which are doing, it has been integrated already. And that has been lowpass filtered, right. So if your original delta modulated signal was this, at the output of the lowpass filter the signal component will continue to be this.

Assuming that this lowpass filter also has a bandwidth W, that is it allows frequencies up to W to go through. So the signal power that we will feel your is going to be A square by 2, that is more precisely expected value of, of course there is no need to talk about the expected value but assuming that there are certain random parameters, in any case the average power is A square by 2, right. And therefore the signal-to-noise ratio is calculated as expected value of M whole square t upon expected value of n q square T and what will be that equal to, I substitute, that is A square by 2 and that was 3 by Delta square, I am taking the reciprocal of the earlier expression into fs by W.

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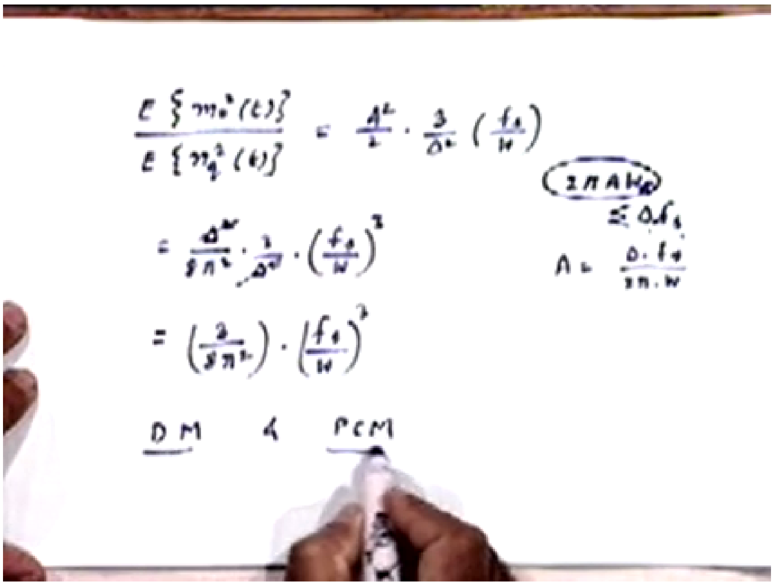

We now would like to use the assumption or the, not the assumption, the restriction under which we are doing this analysis and that is under conditions there is no slope overload. That means therefore there is a restriction on the value of A, if you fixed the value of W, right. What is the maximum value of A can have to avoid slope overload? We know that the maximum value is governed by an equation that, because the maximum slope of the sine wave will be 2 pie A WS, AW, sorry.

Student: (())(8:03).

Professor: That is right, 2 pie AW is that coming from the sine wave and from the delta modulator it is Delta by Ts Delta into fs, right. Maximum value for therefore that you can have for A major one from which this equation is satisfied.

Student: (())(8:23). I mean Delta square W…

Professor: That is right but we did before the computation of the output noise power, at the output of the lowpass filter whose bandwidth this from - W to + W, right. So it gets multiplied by 2W, that is why you get this expression. If you were to look at the notes, you will find that okay. All right. So Delta times fs, so the maximum value of A is Delta times fs upon 2 pie W, right. Let us substitute that here because we are working within that constraint, let us look at the maximum possible value that you can have for A to maximise the signal-to-noise ratio here.

The larger the value of A, the larger the signal-to-noise ratio I will get. But there is a limit on the maximum value of A I can put and that limit is this. So let us substitute that value and that gives you Delta square by 8 pie square into 3 by Delta square into fs by WQ. Cancelling this will give you the final expression which is 3 by 8 pie square into fs by WQ. So we have this expression for signal-to-noise ratio in a Delta modulator, simple delta modulator. It depends essentially on maximum signal-to-noise ratio that you can get, it depends essentially on the sampling rate that you use and the bandwidth that you have for the signal. Right.

Clearly the larger the sampling that you use, the larger the signal-to-noise ratio you will get because we will get a better approximation, staircase approximation to your original signal, right. So there is a clear trade-off between the sampling rates and performance, that is one thing that is quite clear. Now therefore we will like to compare PCM and Delta modulator. Before that do you have any questions about this result?

Student: (())(10:55) if we have to get the maximum amplitude…

Professor: This is clear no, because the slope of the signal is this, right which should be less than or equal to Delta times fs.

Student: (())(11:11).

Professor: To avoid slope overloading because we are doing the analysis under condition of no slope overload. If there is slope overload, this expression is not valid, right. Any other questions? Okay, so now we are ready to carry out some kind of comparison between Delta modulation and PCM. Now essentially when you compare communication systems, they can compare them on the basis of signal-to-noise ratio but it is meaningful to do so provided we are using the same communication resource. And in this case the communication resource is bandwidth, right.

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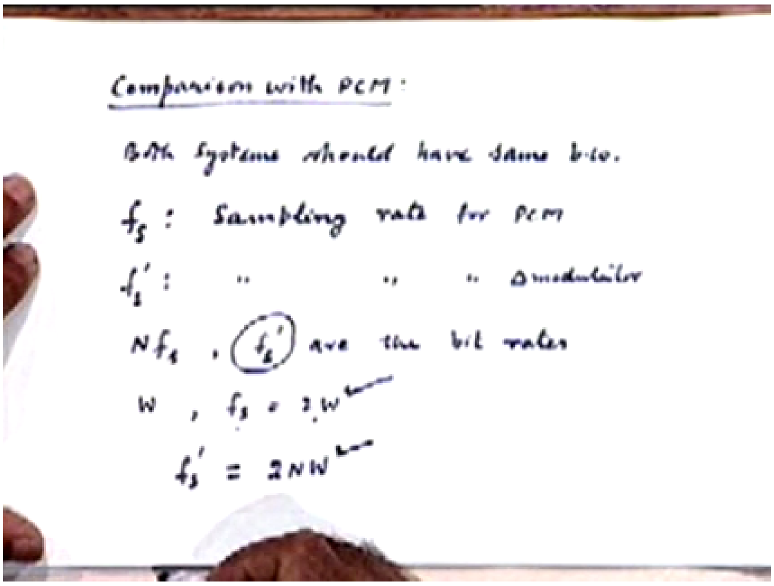

So the 2 will become comparable if we make sure that the bandwidths are, the bandwidths use their identical, only then it is meaningful to talk about their performance comparison, right. So we should do bandwidth, performance comparison by 1st ensuring that we are doing it under similar bandwidth conditions, right. So let us do that comparison. So we like both the systems to have same bandwidth for comparison. Let us say we are using a sampling rate of F sub S for PCM. So let F sub S be the sampling rate being used for PCM. And we know quite well how to select that sampling rate.

It depends entirely on the bandwidth of the signal and it is governed by the nyquist theorem, right. W is the bandwidth of the signal, then fs must be at least 2 W and let us select that, right. Similarly let fs prime be the sampling rate associated with the delta modulator. Now as far as Delta modulator is concerned, the moment you determine your sampling rate, its bit rate is also fixed, because bit rate and sampling rate are same Intel the modulator, right. Whereas in the case of PCM bit rate is n times fs where n is the number of bits being used to represent each word, all right.

So the bit rates in the 2 cases are n sub S, n into fs in one case and fs prime in the other case, right, they are the bit rates. And if your signal has the bandwidth W, then we know that fs is 2 W. So these are the things that we are, therefore it is clear that if you want to systems to have the same bandwidth, then your fs prime should be equal to how much? Fs is 2W, you want the same bandwidth means you want to have the same bit rates, right because identical bit rates you can expect identical or at least similar bandwidth. Right.

Therefore fs prime of the delta modulator should be equal to 2 nw so that PCM sampled at this rate and Delta moderators sampled at this rate have the same bandwidth, okay. Is it okay?

Student: How do we (())(15:36) bandwidth related to the bit rate?

Professor: I thought we have discussed that. It is essentially related to the maximum pulse width you can have. The maximum pulse width that you can have depends on the bit rate that you use. Bandwidth is proportional to the reciprocal of the pulse width and the maximum pulse width you can have is dependent on the bit rate, right. So that is the reason bandwidth depends almost entirely on, at least the large extent on the bit rate. To some extent it may depend on other factors like the pulse shape and things like that but largely it depends on the rate of transmission of information, right. Fine.

So we have determined the relative sampling rates of a PCM system and a Delta modulated system which will have similar bandwidth. For the PCM system if fs is 2 W, then correspondingly for the Delta modulated system it should be 2nw for it to have the same bandwidth. So we can go up to a sampling rate of 2nw without exceeding the bandwidth that you will be using up in the PCM in any case, right, in a Delta modulator.

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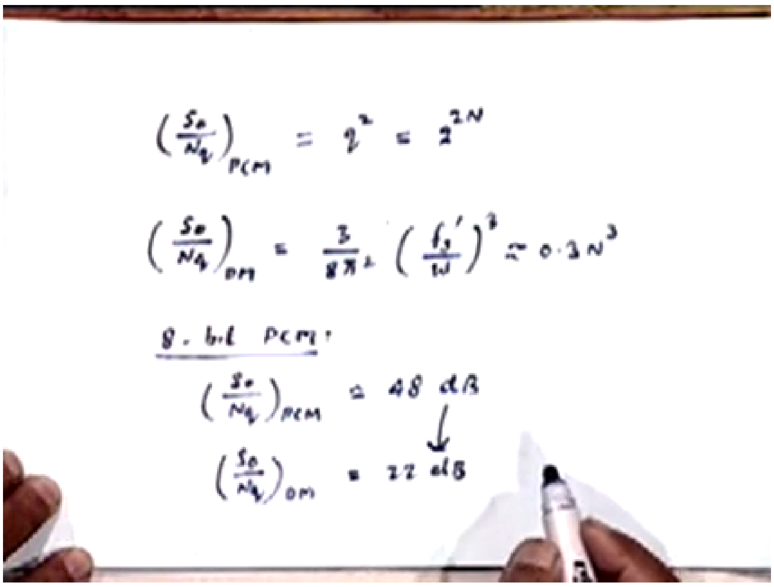

Therefore let us now do the computation for signal-to-noise ratio in both the cases, signal to quantisation noise ratio in both the cases, 1st for PCM, for which we already know the result, the precise result was Q square -1, you remember that, where Q is the number of quantisation levels. But Q is typically large, so we can more or less assume it is equal to Q square and Q is a power of 2, it is actually 2 to the power n, so this becomes 2 to the power 2n, we have discussed that earlier. Right. For the Delta modulator now, we will have 3 by 8 pie square into fs prime by W whole cube.

And fs prime is 2nw, right, for it to have the same bandwidth as the PCM system. So 2 nw by W which is 2n, raised to the power 3, so that is 8 n cube, so this 8 and that 8 will cancel, we will be left with 3 by pie square into n cube. Pie square is approximately 10, slightly less but approximately 10, so that is approximately 0.3 n cube, all right. Of course, so therefore for a fixed bandwidth, what can you say about the relative performance of the PCM and the simple Delta modulator, which is superior?

Student: PCM.

Professor: PCM is obviously superior for almost any value of n, starting from lower values to higher values. Specifically let us take the case of 8 bits PCM which is the normal standard PCM used. Then we know the rule of thumb also 6 bits, 6 db per db 6 db per bit. So for 8-bit PCM the answer is approximately 48 dbs, whereas for Delta modulator if you compute it in db, it will turn out to be something like 22 db. So not only it is poorer, it is considerably poorer, right, there is a 16 db SNR difference between what you can get from 26 db, yes, sorry, 26 db difference between the PCM SNR and the delta modulation SNR.

But mind you this is a situation for a simple Delta modulator, we never use a simple Delta modulator. We already understand quite well that a simple Delta modulator is not a very good one. Right, so it is quite clear now quantitatively how poor it is, how useless it is for say in comparison with PCM. On the other hand an adaptive Delta modulator, a suitably designed adaptable the modulator can be made to perform at the same bandwidth fairly similar to PCM. This result I am just quoting or giving to you, we are not really doing an analysis of adaptive delta modulator but there have been theoretical as well as experimental investigations on adaptive delta modulation.

And people have been able to realise performance as good as PCM using adaptive Delta modulation schemes. Therefore the net conclusion is that we cannot get the same kind of performance from Delta modulator as from PCM but we can do so by using a suitable adaptive Delta modulator, right. Any questions?

Student: So is adaptive delta modulator better than PCM?

Professor: No, you cannot get it better than PCM, you can just about managed to get equal to PCM. Well, you can get very close to it may be this way but that way it is really not much better, right, it does not score over PCM. So what is the implication of this? Historically the implication was that Delta modulator, particularly in the adaptive form was a preferred kind of conversion technique because of, still the simplicity of adaptive delta modulator is still considerably more than that of the PCM system. But over the years and more recently, in fact PCM is becoming the de facto standard because technology has made implementation fairly easy and there are other advantages of representing a sample by let us say binary representation in a normal way which are not available to you at least obviously in Delta modulator.

Because in Delta modulator your representation is being obtained in a rather indirect way, it is not, there is no direct correlation between the bits that were coming and the sample values, right. Whereas in a PCM there is a direct correlation in the sample values in a word and binary pattern that is being transmitted. And if therefore you want to do some arithmetic operations or digital signal processing along the way, you can do it much more easily using the PCM format than a Delta modulator format. So because of other advantages and because technology has made the simplicity of a Delta modulator rather unimportant the PCM is becoming the de facto preferred mode of digitisation rather than Delta modulator.

But it does not mean that Delta modulator is no longer of interest, there are many applications where it is still of interest. I think that brings us to the end of our discussion, planned discussion for analog to digital conversion methods, that is methods for representing analog signals digitally. But we will like to, we are still not really quite finished with PCM, one of the primary purposes by which we do PCM besides the other advantages of digital communication between discussed. If you remember one of the advantages was Time division multiplexing that could be carried out.

That is if you had a cable or a communication system which could accommodate more than 1 voice signal, right. Now you could do so by simply multiplexing a number of voice signals in an appropriate manner. One possible technique of doing this is frequency division multiplexing which we have done in our previous course, which we have done in our previous course. Here we will be talking about Time division multiplexing. Basically the idea of time division multiplexing is that you have a lot of typically you will not be using the entire bit period for transmitting one bit or one word.

We may use the intermediate period for stuffing additional information from other voice channels, right. So if we have a cable with large bandwidth which can accommodate not just 64 kilobits per second but maybe much larger than 64 kilobits per second because it has the capacity to transmit at a higher bandwidth, then you may like to put an additional bits, right. You may like to transmit a higher bit rates and these bits can be interlinked with each other, that is the basic idea of multiplexing in time domain. So let us talk about Time division multiplexing in the context of PCM.

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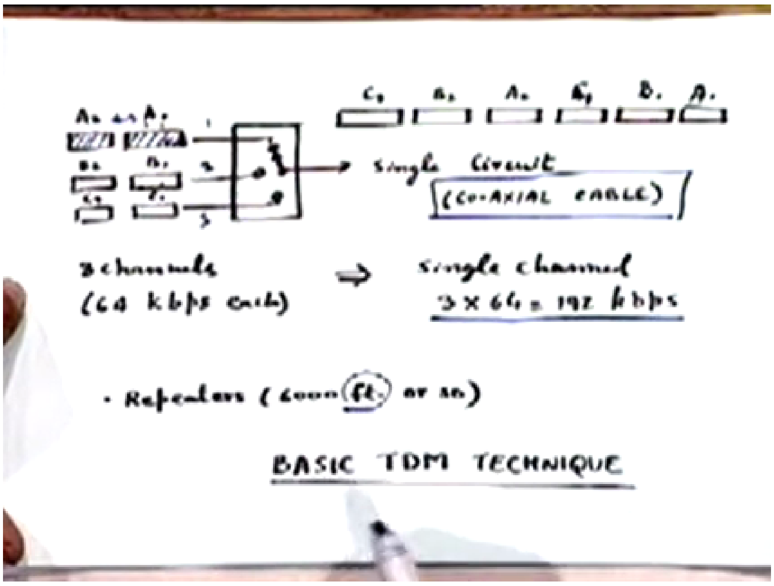

The basic idea, the basic points is that there are many circuits like cables, high bandwidth cables or like satellite channels which can, which give you a lot of bandwidth, right. And if you, even if you are doing PCM transmission for one voice channel, let us say you are only interested in transmitting voice signals, because that is one of the most important kinds of communication traffic, voice signals, we are only using up 64 kilobits per second of the bandwidth, much more maybe available. Then we would like to do, make use of it by using some kind of a multiplexing technique.

And all these will be multiples of the basic PCM rate, 64 kilobits per second, right. So we use time division multiplexing which is essentially a form of bit interleaving of the pulses from a number of voice channels. This interleaving maybe at the bit level or at the byte level, right, let me illustrate by a picture. There is a simple scaling. Let us say I have 3 voice circuits, in each voice circuits I am going to use, let us say they have identical bandwidth which is let us say 3.3 kilos hertz therefore the sampling is being done 8 kilo hertz to provide for the guard band and all those things, usually you leave a sufficient margin.

And therefore the sampling is done at twice, not 3.3 but twice of 4 kilos, the gap between 3.3 and 4 is the guard band. Therefore each of these signals has to be sampled at a rate of 8 kilo hertz, right. There are 3 signals coming in…

Student: (())(28:07).

Professor: There are 3 voice signals, that is what I have said, this is 1, 2, 3, right. Each has to be sampled at the nyquist rate because all of them have voice signals, they will have identical nyquist rate requirements and let us say each therefore has to be sampled that 8 kilo hertz. And immediately after sampling it is encoded into a digital word of 8 bits, right. So this is the 1st sample encoded into 8 bits, this is the next sample encoded into 8 bits so individual bytes corresponding to the individual samples from… And the gap between A1 and A2, the 1st and the 2nd bytes from A signal is obviously 8 kilo hertz, the reciprocal of 8 kilo hertz, which will be I think 125 microsecond, right.

So the time interval between this is 125 microsecond. In between the samples of A1, that is from A1 and A2, I could put samples of other signals as shown over here, I could take the sample from here, 1st sample from there, next sample from here, next sample from here, provided that I make sure that this feature which is back here within 125 microsecond, right. Which is the time at which the next sample is required to be taken. Similarly it reaches here within 125 microsecond of the previous sample of this, within 125 microsecond of the previous sample of this.

So that is what you get at the output of such a multiplexer. This is bit, a multiplexer is therefore nothing but a switch, a commutator which rotates at the sampling rate, at multiple times of sampling rate, depending on how many channels you want to multiplex. In this case it will rotate at 3 times the sampling rate. And therefore the multiplexed byte sequence that you will get is A1 followed by B1 followed by C1, A2 followed by B2 followed by C2 and so on. And this multiplexed sequence which will now be obviously at this bit rate, individual channel was producing 64 kilobits per second, right, overall bit rate will be three-time this, 192 kilobits per second.

And this can be now put on a single circuit which can accommodate such a bit rate, for example a coaxial cable, right. And on the cable as the propagation is taking place, you place a repeaters on the way, typically the gap is around 6 kilometres between repeaters.

Student: (())(31:09).

Professor: I am sorry, 6000 feet, I thought it was metres but it is feet, around 6000 feet or so. So this is the basic TDM technique, just in between successive bytes we put other bytes from other voice channels, all right. This is a very important, it is a very simple kind of thing to do, I mean now things in principle, sizes not much to explain about it, it is a very simple concept. But since this is one of the most important features, it is a very important feature of the communication system when you are doing PCM because we do not deal with in practice 1 or 2 or 3 voice circuits.

In a, when you are carrying traffic out let us say from one city to another city, the number of voice signals that you may have to carry can be very large, right. And therefore the equipment that you use for this purpose must be designed to handle number-one, it should be able to handle large number of voice circuits, right. The cable that you use therefore should be of sufficiently large bandwidth to be able to handle that kind of bandwidth. Equipment that you use should be standardised so that you can handle different number of circuits. Small number of circuits if you are servicing through a small exchange and larger number of circuits if you are servicing through a larger exchange.

And this has to be more or less standardised across a country and across continents, right. So that people can understand or work with each other’s signals, different exchanges. I hope it is obvious that you will not typically do PCM transmission of a voice signal within a city, within small distances, right. Where you do not really require the advantages of complex digital systems. If I am talking to somebody else in the same city on a telephone, the distance involved is not so much and analog communication, a simple 2 wire line is sufficient for carrying out the communication.

Usually we need to take advantages of such PCM kind of techniques only when you are doing long-distance communication. And that is typically the case let say between the cities or between the countries, right or when you are going through satellites. So it is a trunk route we are talking about and a trunk route means we are handling large number of circuits and it must be handled with standard equipment. In order to make sure the things are done in a standardised manner, the organisation both have at the national level as well as international level which carry out such standardisation.

And make sure that the equipment manufacturers manufacture that equipment according to that standard. So we look at the multiplexing standards that exist, there are actually 2 kinds of standards that you will find in multiplexing. Just like everything else, Americans have their own standards for everything and so do they for multiplexing as compared to the rest of the world.

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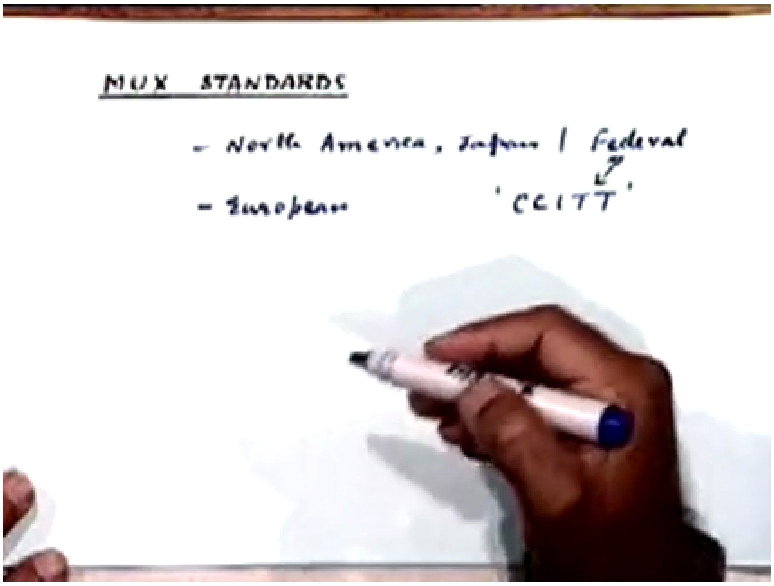

So you will have one set of standards which is applicable in North America including Canada and countries like Japan in the Far East, right. And some other countries which have close technological relations with northern America. And much of rest of the world uses another standard, let us call it the European standard. Actually there are 2 standard making bodies that every important, one is the CCITT, I do not think I remember the complete explanation for this, which is a European body. Actually it is an internationally recognised body and the other is the federal standards of America, so they are governed largely by the federal standards.

There is a lot of interaction between these 2 at least these days so that many of the standards that are coming up now are now common. But many of the standards that have evolved in the past are not so common. So I will look at only the European standards in the multiplexing context. There are quite a few differences between the European and the American standards but we have to basically learn how these things thing are done, it is unimportant what the specific details of various standards might be.

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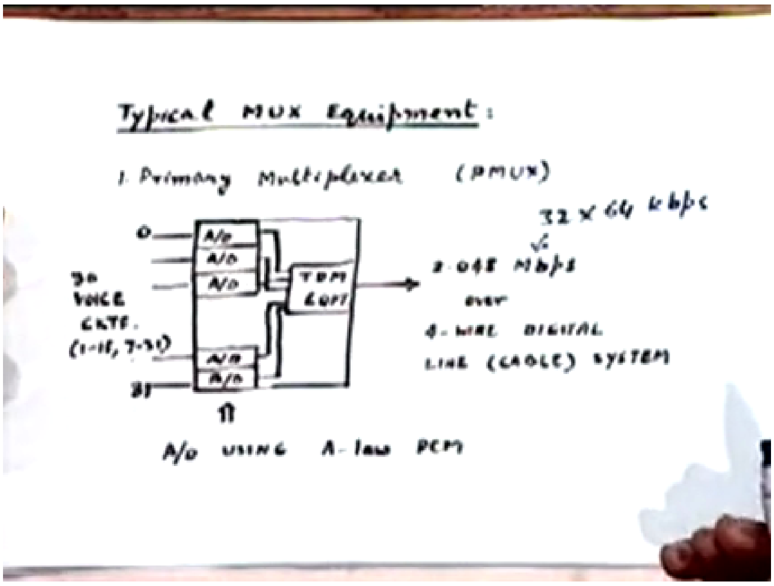

Now the basic multiplexing equipment in this standard is what is called the primary multiplexer or primux, now which I have shown schematically like this. It can handle 32 circuits, it has the potential of handling 32 circuits. So you find started to A to D converters shown over here, 0 to 31. Actually out of these 32, only 30 are used for voice circuits, that is circuits numbered 1 to 15 and 7 to 32. So multiplexing 30 channels, 30 voice channels into a single circuit, right. So followed by these each of these A to D converters, we have the TDM equipment, multiplexing equipment which is essentially a high-speed switch, all right.

So finally you get a data rate at the output which is 32 times 64 kilobits per second, right which is 2.048 megabits per second which could be transmitted over a 4 wire digital line which could be, that is the cable system or it is transmitted over, even a satellite link. The A to D converters as rightly pointed out by our friend yesterday, in the European standard uses A law PCM, in the American standards it uses mu law PCM. So this is, this is the basic multiplexing equipment.

Student: What do you mean by 4 wire digital line?

Professor: Okay, the 4 wire system, we will talk about it separately sometime. A 4 wire system essentially is the one in which you transmit on one pair of lines and receive on another pair of lines, which is unlike the 2 wire system that you are normally familiar with in the telephone system. I am sure you know something about telephones, a telephone receiver handset that you have has a microphone as well as an earphone. So ideally to listen and to speak you require separate wires, right but they are combined together into 2 wires for actually propagating onto telephone line which is a 2 wire line.

We both transmit and receive on these 2 lines but within the handset, within the instrument we do something so that your local transmitted signal does not bother you at the ear as well as it does not bother the other person when he is receiving your signal, right. So there has to be a 2 wire to 4 wire conversion in such devices. In these instruments we have 2 wire to 4 wire convergence which is done through some kind of hybrid transformer. But in trunk lines it is a bit dangerous to use hybrids, good quality trunk lines would prefer not to use such hybrid transformers.

They would rather do separate transmit, transmission in a separate pair of lines and reception on a separate pair of lines flies. So when I transmit it is done on a separate pair of lines, when the other guy transmits it is coming on a separate pair of lines, not in the same pair of lines, so that we do not interfere with each other, right. On trunk lines it is important to do so is feasible. Many trunk lines also could be of the 2 wire kind but it is preferred to have 4 wired trunk lines. We will talk about that in our extra class sometime.

Now the multiplexing equipment I have shown you so far can take you to multiplex 32 channels. Actually only 30 of these are used, we will see why in a few minutes. But many a times you may have situations where you may like to put more number of channels. For example if you are using a satellite link, you may not do it only for a city or one major centre from a city, you may like to combine information coming from a number of nodes in the city. Right. So you may like to have a hierarchy of multiplexing to go from multiplexing of 32 voice channels too much higher number of voice channels.

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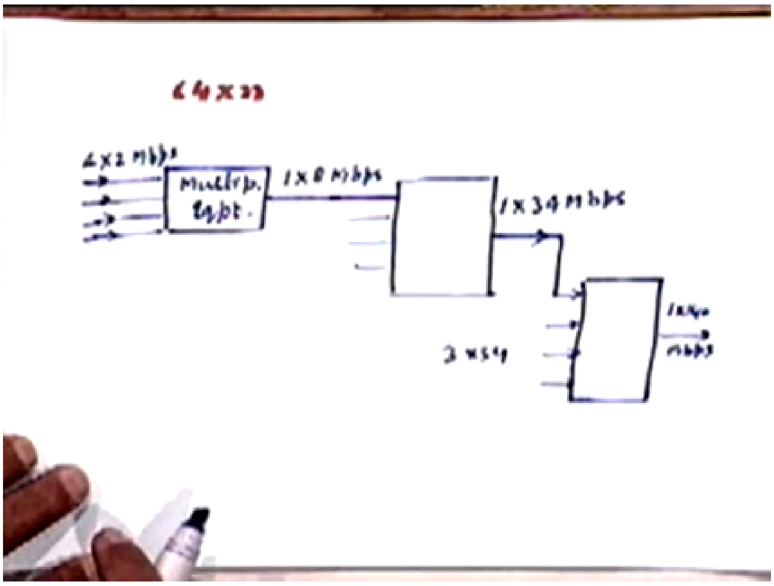

There has to be flexible it in terms of number of channels that I can put in this mode, right. So the standard also specifies the hierarchy of multiplexing which can allow you bit rates which are multiples of 64 times n but much higher than 2 megabits. And the hierarchy is something like this. We have the basic multiplex equipment which is in the form of primux which gives just discussed. Let us say we have a number of primuxes. The next hierarchy from primux onwards is, you can put 4 of these, each of this is the output of one primux, okay.

I can put 4 of these into the next hierarchy of multiplexing to produce a single line of. So this is 4 into 2 megabits per second, I finally get 1 into 8 megabits per second, right. I can multiplex outputs of 4 primary multiplexers into the next level of hierarchy to get say total of how many channels, 128 channels.

Student: 120.

Professor: Well, 128, data rate corresponding to 128 channels, though out of which only 120 will be utilised for voice. We will soon see what happens to the others in a few minutes. At the next level of hierarchy I again put 4 of these, to get, now logically it should have been 32 megabits per second but actually it is 1 into 34 megabits per second, okay and so on. Similarly at the next hierarchy I have 4 of these, so 3 into 34 and again it is not precisely 4 into 34, some actually 1 and 3, so it is total of 4, it is 1 into 140 megabits per second.

Student: What is the reason for this?

Professor: I will just mention, we will discuss that reason in a few minutes. So this is the hierarchy that we use in the CCITT standard that we talked about. The American standards again are different at this level. Just to mention the American standard to you, the basic primary multiplexing equivalent of the American standard multiplexes 24 voice channels instead of 32, right. Also the manner in which synchronisation and all is handled is quite different in that. Similarly at each hierarchical level the American standard is different from the corresponding CCITT standard.

Student: All intercontinental communication…

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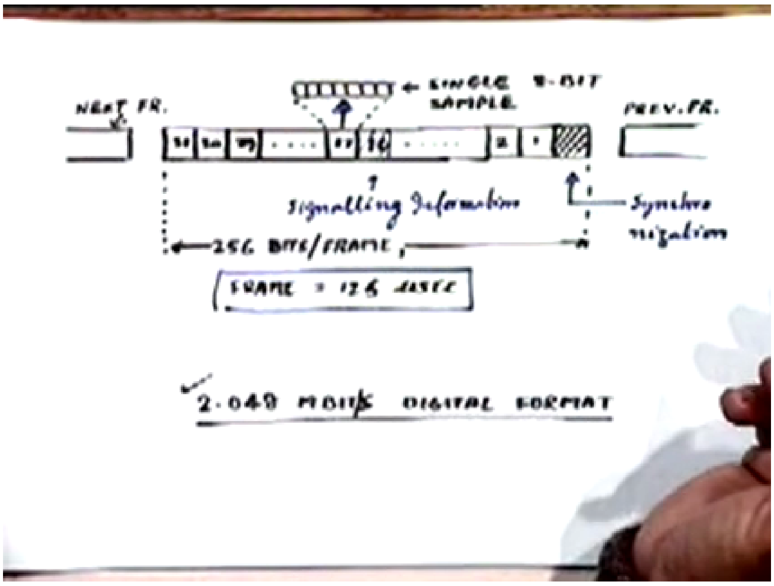

Professor: They have a lot of problem and they have to have what is called internetworking equipment which can convert signalling, signals from one format to another format, so they require all that between Europe and America, they have all those problems, they require that. Now let us come to how each, let us look at this multitasking at the primary level in slightly more detail and then I will also come back to the question which you have asked as to why those are not exact multiples, right, which we talked about in the hierarchical structure.

Let us come back to our basic time division multiplexer which is a 2.048 megabits digital data stream. As we said, we have 32 time slots there, each of these small little squares indicate one such timeslot number from 0 to 31, okay. Out of which slot numbers 1 to 17 and slot numbers 19 to…, sorry, 1 to 15 and 17 to 31 are used for voice circuits. Slots number 0 and 16 are used for other purposes, but the total duration of this slot is has to be worth 125 microsecond because after you have sampled the 1st channel, 1st sample of the 1st channel, you have to get to the next sample within 125 microsecond. So the total duration of the slot has to be 125 microsecond and within that we have 256 bits coming in corresponding to 32 slots over here, 32 into 8 is 256.

That is the previous frame, this, this unit of 32 time slots constitutes a frame, we call this a frame, this is one frame of information, this is actually a snapshot of one frame, right. Each slot consists of 8 bits, right. Each slot has one sample of a voice circuit contained in it represented in the form of 8-bit word, all right. That is how the formatting is done, now slot number 16 is, 1st let me talk about slot numbers 0, slot number 0 is reserved for synchronisation purposes. So there is a special bit sequence that you use your, which will even vary from one frame to another frame but it repeats itself after several frames, which will help you to know the frame boundaries, right.

And that is very important for you to know, we must know where one frame ends and another begins. And therefore you require special arrangement to do that, this is done by transmitting specific signal sequences in the 0th timeslot, so that is reserved for that purpose. Similarly frame, slot number 16 contains what is called signalling information, okay. Typically the signalling information will enable a particular set of, let us say a particular bitstream that is coming along to reach its proper destination. So it contains information regarding the destination.

Destination is taken care of in quite a different way in analog communication being the destination. In fact it is done on an additional wire, additional circuits. Whereas in analog communication, in digital communication, this is done by putting the destination information as a bit of information along with the rest of the information. So it will tell as to which exchange it has to be, this information has to go through. Particular when you are talking about the hierarchy of equipment like this, different bitstreams at 2 megabits level maybe all destined to different places, right.

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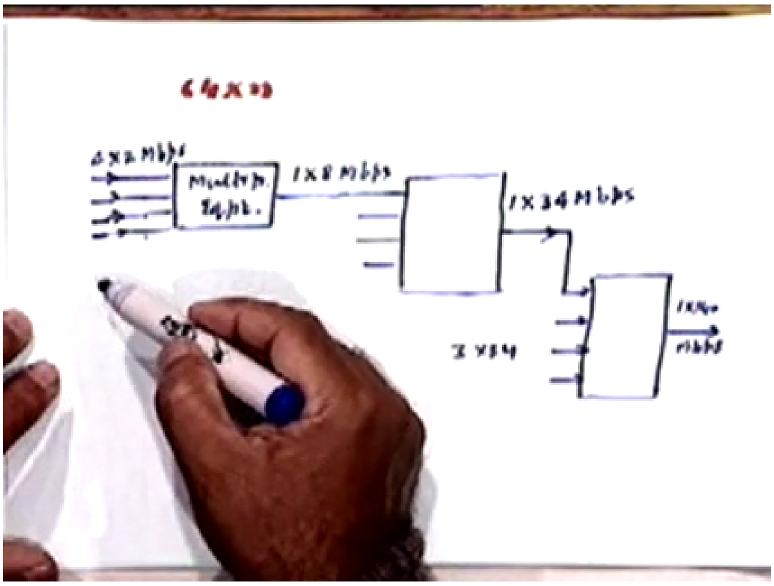

So each of them contains its own destination where it has to go, in the 16th timeslot of each frame, okay. And obviously this destination information will change from time to time for a particular…

Student: So this destination information is for the whole frame or it is for the individual elements of the frame itself?

Professor: No, it is for the whole frame because we are not doing it on the individual level, we are doing this typically at the trunk level, multiple levels.

Student: (())(50:27).

Professor: Obviously the grouping has to be done properly. When you do a courier service and or a Postal Service, you do not put information arbitrarily.

Student: But even if (())(50:40) exchange to exchange but from exchange (())(50:46).

Professor: Yes then that is the problem of the exchange itself. There will be, see there will be levels of destination information, one at the level of exchange and then other levels of information which will take you right down to the actual user where the information as to go.

Student: What is the information specific to each user?

Professor: Actually I am not discussing here the signalling standard, I will really have to get into the signalling standards then. We will skip that for a moment, let us assume that there is a mechanism by which we can embed the signalling information in the 16th timeslot that is available to us. Right, I am not discussing that mechanism because that is a slightly different kind of thing to discuss and again it gets us deep into standards. It is just a matter of knowing how it is to be done and to be taken care of.

So we will, our emphasis here is on principles rather than getting into those specific details of each standard. Okay, this I think, this is where I can stop today. Just to see that the synchronisation information and signalling information are done in a very simple manner in this standard, in the American standard this done in a slightly more complex. One last point that somebody asked as to why for example this is not 32 and this is not… What should it be?

Student: Additional information (())(52:18) 136.

Professor: 136, right. So instead of 136 it is 140, the reason is obvious, I think from what we have discussed now, you require additional information regarding synchronisation and regarding signalling, right to be embedded into each of these, which has to be taken. So that is the reason why it is not exact multiples of this. Okay, we can stop here, thank you.