

Broadband Networks

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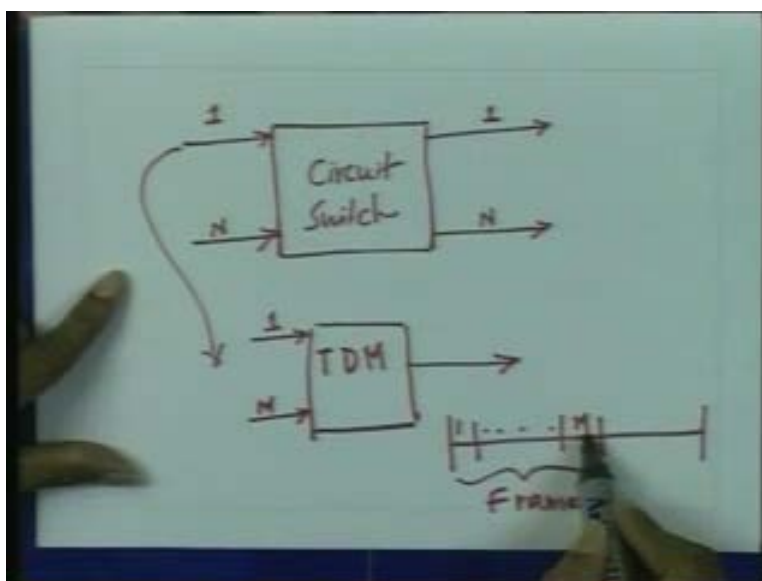
Indian Institute of Technology, Bombay

Lecture - 1

Various information transfer modes: traditionally, the oldest information transfer mode which was used in the telecommunication switching is circuit switching. So, let me first explain what is meant by circuit switching which also essentially uses the principal of time division multiplexing.

So, let us understand what is meant by circuit switching on a time division multiplexing. Typically in a circuit switching, a certain portion of the bandwidth is reserved for a particular connection or what is called as a circuit.

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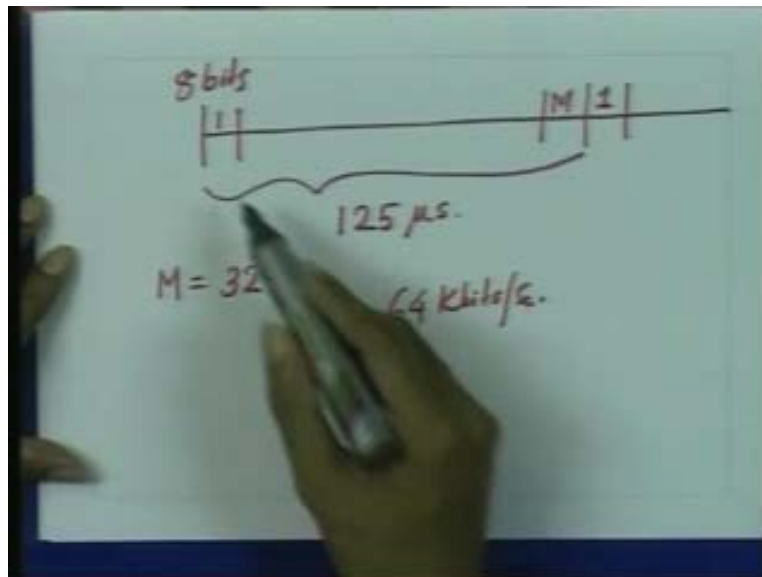


So, let me just explain what is meant by the circuit switching and the principal of time division multiplexing. Typically, in a time division multiplexing we will have multiplexers followed by a switch. So, let us say that this is a circuit switch which **which** could be an N cross N switch. So, where there are 1 to N input ports and 1 to N output ports.

So now, on each of these input ports, there may be multiple connections which could be multiplexed. So, let us see, **you know**, how this, each of this input ports may look like. This itself could be TDM multiplexers; a time division multiplexers which could be multiplexing, let us say 1 to M input ports.

So, typically on this input port, therefore the time axis will be divided into a frame. Now, this frame itself is divided into 1 to M slots. The user number 1 is given the slot number 1, the user number 2 is given the slot number 2 and the user number M is given the slot number M.

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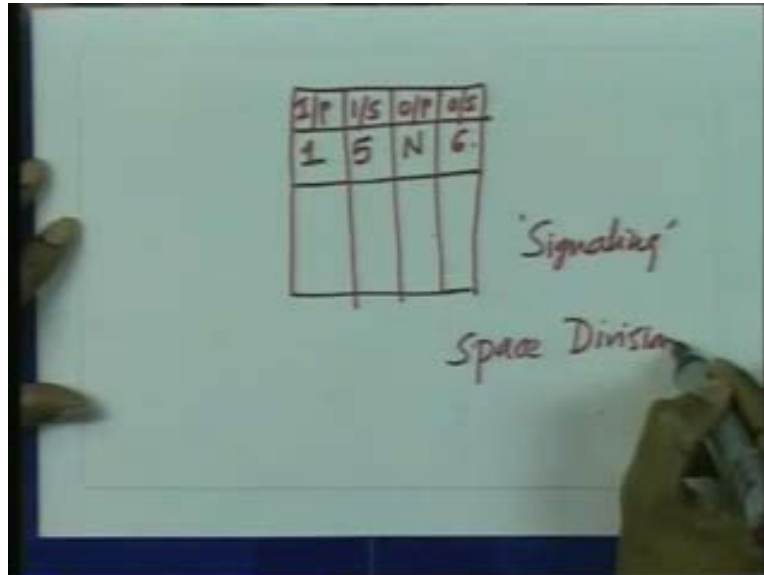
Now, typically this frame duration is; let us say, 125 micro seconds. So, **let us** let us understand that there is this time frame which is having 1 to M slots. The frame durations could be of 125 micro seconds, a typical example and the value of M, let us say is 32. So, each user may be allowed **each user may be allowed** to transmit 8 bits in a particular slot.

So, user number 1 transmits 8 bits in slot number 1 and the next 8 bits he will transmits in the slot number 1 of the next frame. So typically, a user ends up transmitting 8 bits every 125 micro seconds and that results in a data rate of 64 kilo bits per second. So, the value of M is 32. Then we have the total data rate to be about 2 mega bits per second.

So, what we are saying now is that there are 1 to M users which have been multiplexed on this TDM frame, M is equal to 32, each user is allowed to transmit 8 bits in slot and **he transmits 120** he transmits 8 bits every 125 micro seconds and that gives him a data rate of 64 kilo bits per second.

Now, in this particular circuit switch that we have, now what happens is then on each of these input ports, these 32 users have been multiplexed. Now, the bits which are there in a particular slot on this input port may have to be switched to a particular slot on the output port.

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Now, the time signaling **you know**, this circuit switch will have some kind of a translation table which will look something like this that there is an input port, input slot, output port and output slot. So, you will say; what is the input port, what is the input slot, what is the output port, what is the output slot. So, that means typically in an input port of one input slot of let us say, 5 may have to be switched to output port of let us say, N and some output slot may be equal to 6.

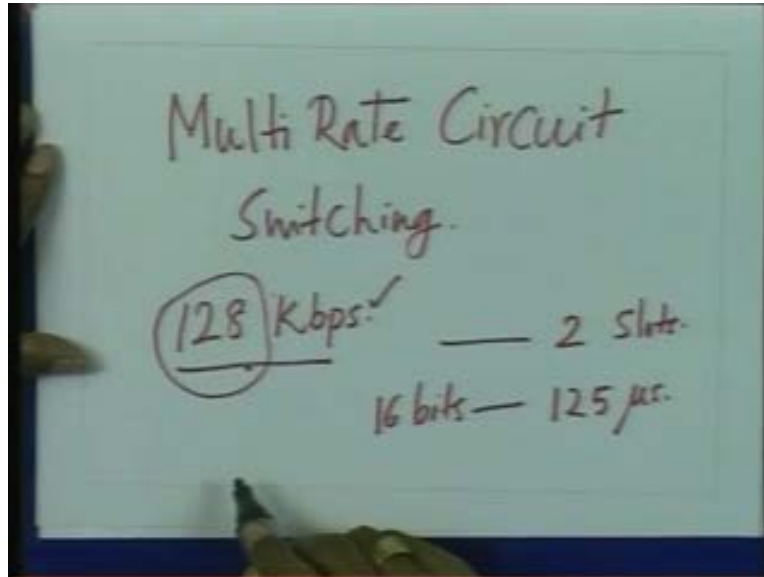
So, **this** typically this table is built at the time of signaling and then the users will be switched by following the principle of some space division, space division switching which will exist here. Now, you can see from from this that since each user has been allotted 8 bits for transmitting in 125 second, the data rate of this user is fixed and that is 64 kilo bits per second.

Now, clearly the disadvantage of this circuit switching is that this is inflexible, the user cannot transmit more than 64 kilo bits per second, each user has to transmit only 64 kilo bits per second because the slot duration is fixed and users are allowed to transmit only 8 bits in a slot and the frame duration is also fixed which in our example has been shown to be 125 micro seconds.

Now, the question then arises is that what happens if the users, different users have different data rates? Now, one of the advantages of the circuit switching is that except for certain delays which may occur in the switching, of the space division switching, there are almost no latencies involved. So, except for the switching latencies, there are no delays and therefore the service is typically suited for the real time services, likewise video.

The example the particular example which I have just drawn is suited for the PCM - pulse code modulation multiplexing of the digital voice. However, as we have just mentioned that if different users have different data rates to transmit, then circuit switching is clearly unsuitable.

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So, what do we do if different users have different data rates to transmit? Then we can say that we will have what is called a multi rate circuit switching. Now, in multi rate circuit switching, what we can say that in circuit switching; for example, we had allotted 1 slot to every users and we had asked every user to transmit 8 bits.

In multi rate circuit switching what we can do is we can allocate multiple slots to a particular user. Now for example, if a user wants to transmit let us say, 128 kilo bits per second and if we have the basic channel rate built as a multiple of the or particular multiple connection if we have as a built up of the basic channel rate; then to sustain 128 kilo bits per second, we can allocate him 2 slots.

So, as a result, a particular user who wants to transmit 128 kilo bits per second will end up transmitting 16 bits in a duration of 125 micro second. So, if we transmit 16 bits in a duration of 125 micro seconds, he gets this the data rate of 128 kilo bits per second and we can as a result accommodate, **you know** different users with different data rates. Now, what is the difficulty of the multi rate circuit switching?

The difficulty of the multi rate circuit switching is that if we have a spectrum of users starting from the low data rates like 1 kilo bits per second to a high data rate like 1 mega bit per second, then the basic difficulties that how to choose, **you know**, the basic channel rate; for example, **you know**, in this example we have chosen a basic channel rate to be 64 kilo bits per second. Now, if you choose the basic channel rate to be 64 kilo bits per second and if we have to transmit a user with 128 kilo bits per second, we can accommodate it by giving him 2 slots in that frame duration of 125 micro seconds where each slot duration is 8 bits.

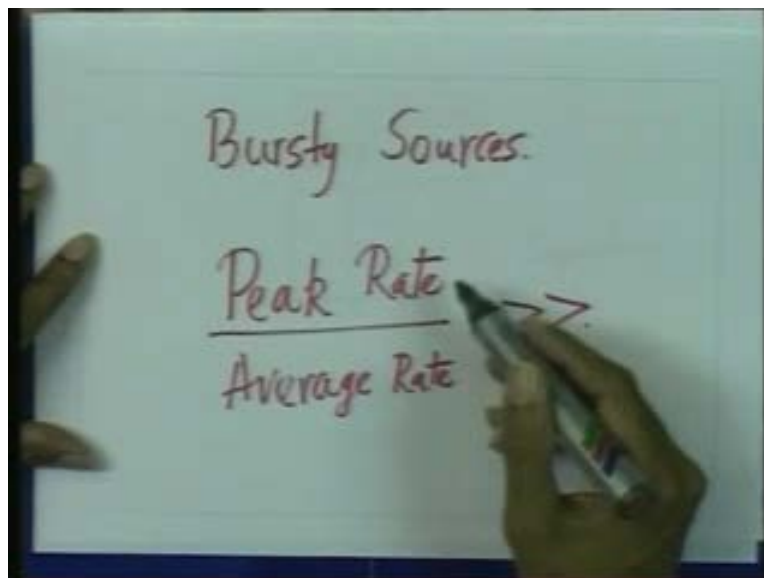
However, if we have to transmit a data rate of 16 kilo bits per second, then **you know** every slot in 4 frames; every slot in 4 frames typically will be occupied and the reaming **you know** 3 frames will remain unutilized because the basic channel rate is 64 kilo bits per second. Now however, if

you have chosen the basic channel rate to be the lowest rate, let us say the lowest rate is 16 kilo bits per second; then to transmit 128 kilo bit per second, we have to have large number of slots which leads to complexity of hardware and complexity of slot synchronization.

So therefore, while multi rate circuit switching this is able to accommodate users with different data rates. It is efficient only if we have a spectrum of users with data rates which varies only from let us say, 64 kilo bits per second to 256 kilo bits per second or 512 bits per second.

In other words, the range of services to be multiplexed is narrow and it is not **you know** it is not broad. Now, one more thing is that that the circuit switching is also unsuitable if we are considering the bursty sources.

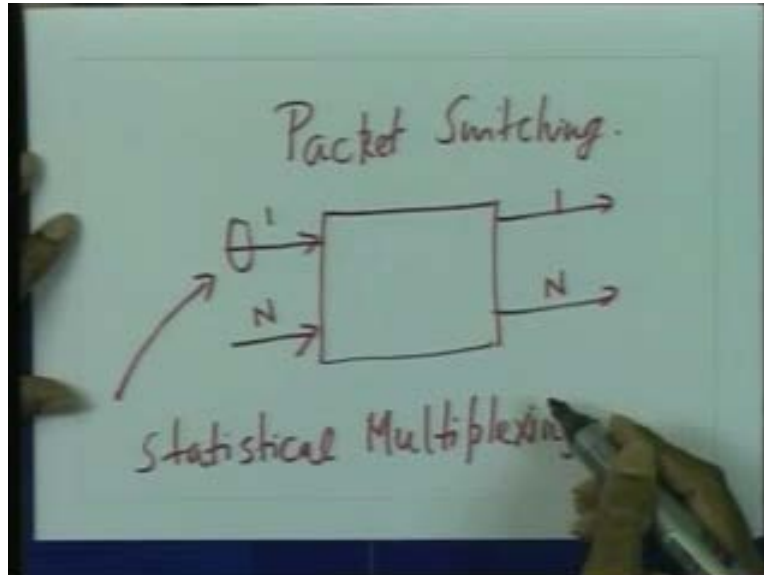
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Now, what is meant by a bursty source? By bursty source we mean that if the ratio of the peak rate, if the ratio of the peak transmission rate to the average transmission rate of the user is very high. Now, such sources we call it to be the bursty sources.

Now, if we allocate a particular slot to a bursty source, then a user may have something to transmit in that slot for few frames but it may not have anything to transmit for the rest of the frames resulting therefore in under utilization of the slots or non-utilization of a slot. Now, this leads to inefficient channel utilization. So therefore, **you know** the circuit switching or the multi rate circuits switching **you know** even if we allocate multiple slots to a particular user; it is clearly inefficient for the bursty sources. So, for the bursty sources we then have the packet switching. Now, what is done in a packet switching?

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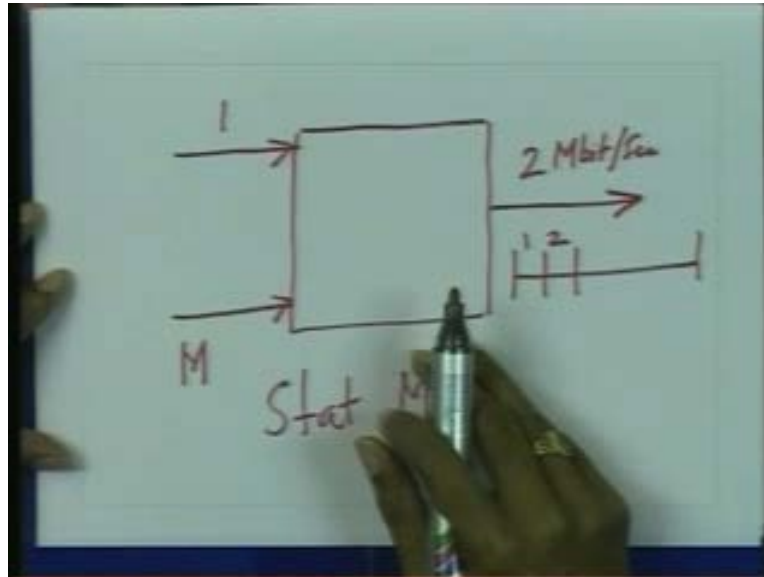


Packet switching; they also have similar to what we have in a circuit switch. It may also have 1 to N input ports and 1 to N output ports with however a very important distinction and that important distinction is that in this case, unlike in a circuit switching where we are having time division multiplexing, in a packet switching we will have here the statistical multiplexing.

Now, what is meant by statistical multiplexing? In time division multiplexing, we had allotted particular slot to a particular user. Now, unlike that in a statistical multiplexing, there could still be a framing structure; in the sense that the time axis may still divided into frames and the frames may also be divided in term into slots. But unlike in time division multiplexing where a particular slot has been given to a specific user, the slots are not allotted to particular user.

So, let us understand what happens in a statistical multiplexing. So, let us say that this is like a statistical multiplexer.

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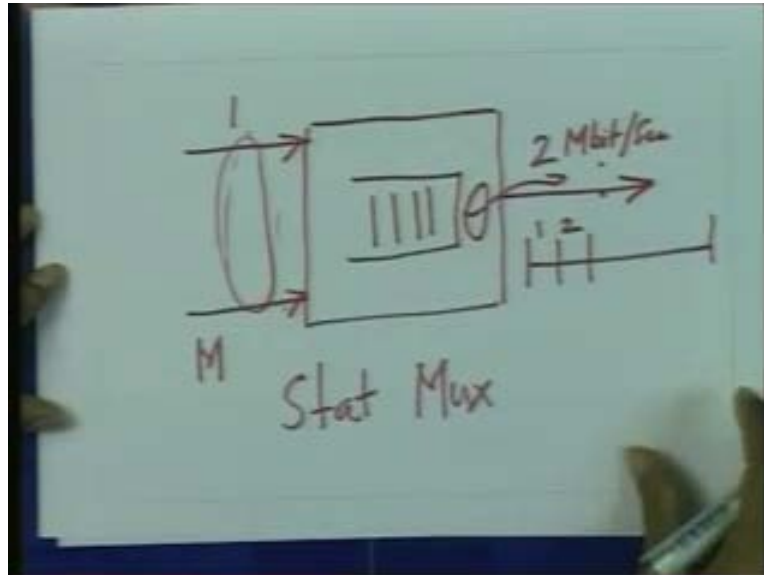
Now, like in a time division multiplexer; in a statistical multiplexer may have several input ports. Let us call them 1 to M but this M number may not be the same as we had considered in the previous example and there is an output port. Now, this output port in the time axis may be divided into slots - 1 to M. But there may not be any framing structures or there may be a framing structure.

Now, let us say for example that the output data rate of this may be 2 mega bits per second. Now, in a statistical multiplexer, unlike in a time division multiplexer where a particular slot has been allotted in statistical multiplexer; we are not allocating these slots to any of the users, we are not allocating to any of these users.

Now, what we are doing however is that there is no particular admission control, any number of users can transmit. In time division multiplexing, in a particular example that we had considered, our frame was 125 micro second and each user could transmit 8 bits and we were admitting 32 users.

Now, what happens in a time division multiplexing is that if all the slots are occupied and if another user dials that number then he will get a busy tone because all the slots are occupied; unlike that in a statistical multiplexer, we allow all the users to transmit. Obviously, if the combined data rate of all the user exceeds the output capacity, then either some of the bits have to be dropped or they have to be queued or buffered.

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Now, what happens we will see this statistical multiplexer is that let us say that there is a buffer, now, what we are doing is that each user transmits his bits in a particular format which we call it to be packets and these packets come and get queued in this buffer. Now, there will be a scheduler who schedule these packets on to the output link; **picks one packet one by one**, picks a packet one by one and then puts it onto the output link.

Of course, if the combined data rates of transmission of these users are less than the output link capacity, then obviously there will not be any queuing. However, if the combined data rate increases the output capacity, then these packets will be queued.

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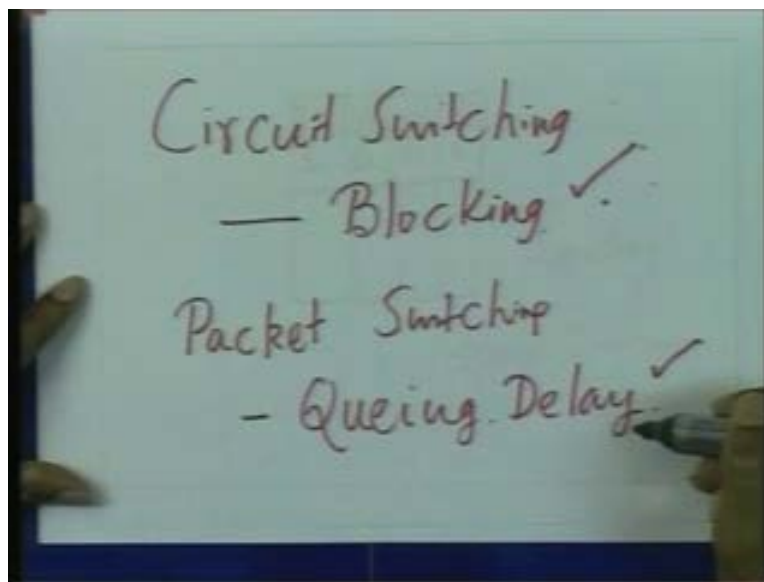
Circuit Switching
— Blocking .

So, as we have seen, while circuit switching was characterized by blocking; by blocking I mean that if all the slots are occupied, then the users are blocked. If all the 32 slots in our time frame are occupied or **or or** all the M slots are occupied, then if N plus 1th user attempts to transmit, then he gets a busy tone or what we call he gets blocked.

Therefore, the performance characteristic of a circuit switch network is probability of call blocking. By that we mean that what is the probability that if a new user makes a call and he finds that none of the slots is free or all the slots are occupied. As opposed to that in a statistical multiplexing, we do not have any concept of admission control. We are allowing all the users to transmit and these users will come and sit in a buffer or in a queue.

Now, the fact, the thing is that since we are not having any concept of admission control, there is no blocking. But there is queuing and therefore the packet switching, packet switching are characterized by what we call as the queuing delay.

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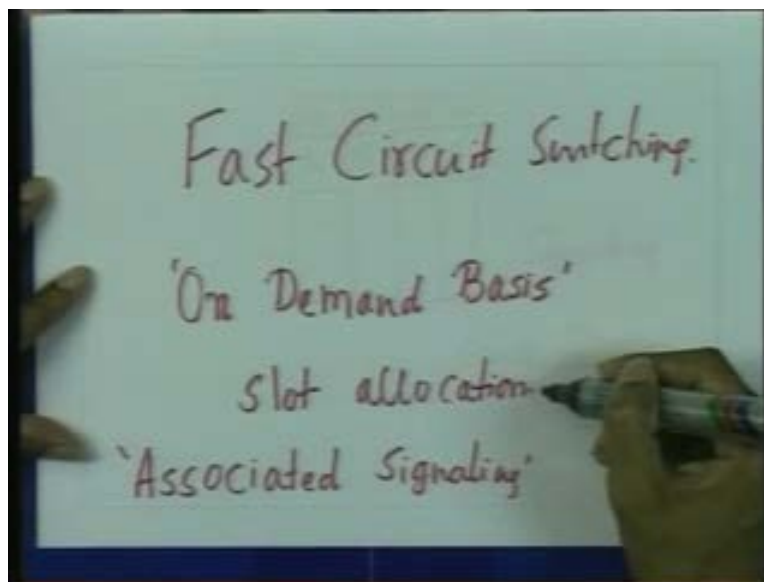
While circuit switching is characterized by the probability of call blocking, the packet switching is characterized by the queuing delay. Now, since there is no admission control, since there is no admission control on the number of users that can be admitted, there being no admission control, this queuing delay could be inordinately large, there being no control on that.

So therefore, we say that the packet switch network is a best effort networks. By best effort networks we mean that the network will make every attempt to transmit a packet but will not offer any quality of service guaranties in terms of delays. Remember that in circuit switching there are no queuing delays. The only delays **that are** that are possible in circuit switching are the switching latencies, switching latencies that occurs in the space division switching. Except for that there are no delays.

So, while circuit switching may be suitable for the real time services which require no delays. On the other hand, the packet switching or the statistical multiplexing is suitable for the bursty sources where the circuit switching is inefficient; because, for bursty sources, the ratio of the peak rate to the average rate being very large, the permanent allocation of a slot can lead to under utilization of the channel. But at the same time, these bursty sources must be capable of tolerating some delays because a packet switching **will** may lead to queuing or buffering of these packets.

Now, before however, we go to what **is the disadvantage** are the other disadvantages of the packet switching etcetera; historically, when circuit switching was found to be inefficient, both circuit switching and multi rate circuit switching were found to be inefficient for the bursty kind of sources, an intermediate transport technique was also suggested and which was called fast circuit switching.

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So, let me just briefly explain it and then we will go to the details of the packet switching more. Now, in fast circuit switching, remember, what was the disadvantage of the circuit switching? The disadvantage of the circuit switching was that a slot is permanently allocated to a user and therefore, if a user is bursty **that means if a ratio** that means if he has nothing to transmit for some of the slots and if he has data to transmit in some other slots, then those slots where he does not have anything to transmit, they will go waste.

So therefore, it was suggested in the fast circuit switching that we may do the allocation of the slots on demand basis. So, in fast circuit switching, we are doing that on demand basis slot allocation. The idea being that by using some kind of associated signaling, you can allocate the slots on demand basis.

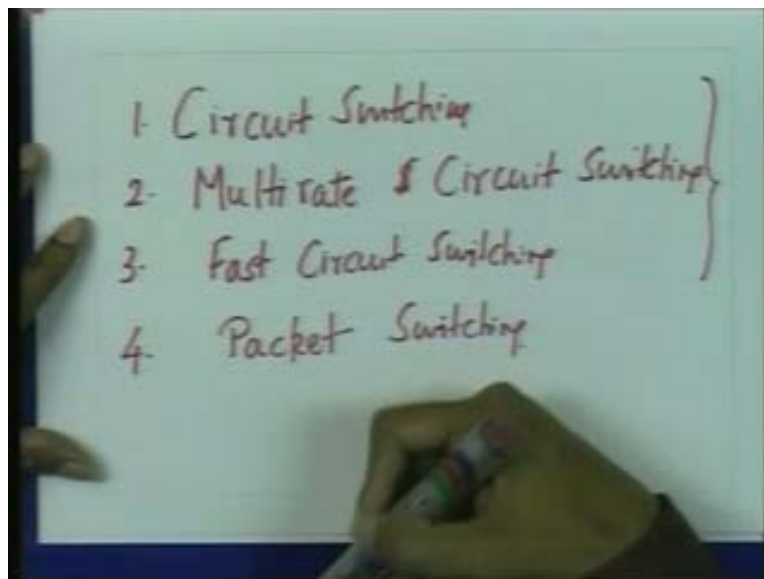
Now, the associated signaling works something like this that if a use transmits in a particular slot and in that particular slot, he can indicate by some kind of a signaling bit whether he requires

some slots in the next consecutive frames or not. If he does not require, then those slots will be released and may be given to new connection. However if he requires, then the slots may be allocated to him. That means the network is doing the allocation of the slots on demand basis; the idea being that that we utilize the slot in an efficient basis and do not have either wastage of the slot or **you know** non allocation of the slot to a particular user when he needs it.

Now clearly, the efficient allocation of the slots on demand basis will lead to lots of complexity, both in terms of the network management as well as the signaling that needs to accompany the particular user's data for indicating to the network its demand.

Now therefore, this fast circuit switching was discontinued or it did not become popular for the transport of the bursty sources and historically then fast packet switching that is another transport mechanism was evolved. Now, let us see, let us review what kind of information transport modes that we have discussed till now.

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One mode that we have discussed is circuit switching. We said that circuit switching is inefficient for the bursty sources and also for the transport of those sources where the data rate requirements are different. Then we have studied multi rate circuit switching, third we have studied fast circuit switching and fourth we have studied packet switching.

Now, if you see, these various forms of circuit switching are suitable for the real time services. They are more suitable for the real time services. On the other hand, the packet switching is suitable for the bursty sources which can tolerate some fixed amount of delays. Now, another disadvantage of the packet switching is that in any practical statistical multiplexer, the buffer size that will be there will be finite or limited.

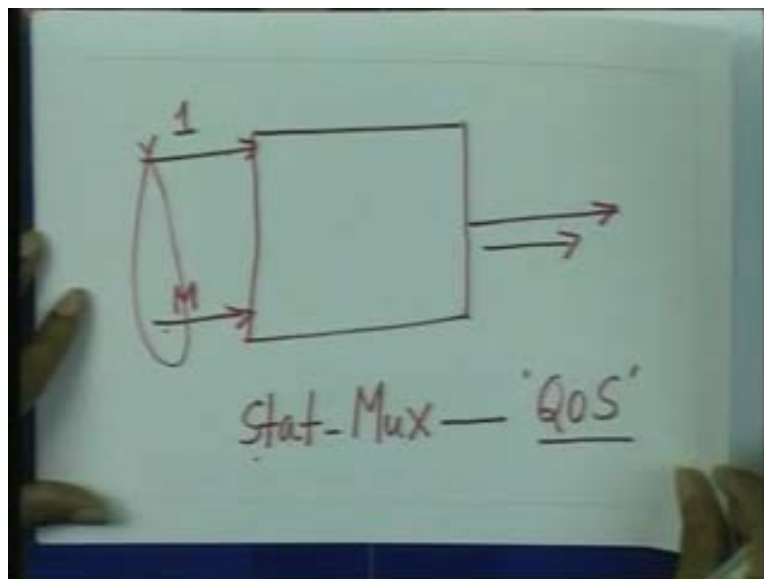
Now since, there is no admission control in packet switching or in statistical multiplexing; it may so happen that the buffer may get full and when the buffer gets full, the packet may also get

dropped. As a result, in packet switching not only there is a queuing delay but also there is a packet loss.

Now, this kind of transport mechanism therefore have not been found to be suitable for the real time services which requires some guarantees on delays or packet loss. Obviously, if you want to have a transport mechanism which is efficient for carrying the bursty sources as well as at the same time is capable of transport of real time services which requires certain quality of service guarantees, then you require a new transport mechanism which has the advantages of both the circuit switching as well as the packet switching.

A new transport mechanism to achieve these objectives was proposed and that was called an asynchronous transfer mode or atm which was also called at that time as fast packet switching. Now, let me just explain you what is the principle of the fast packet switching or the asynchronous transfer mode, shortly.

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So, now here we are talking of a statistical multiplexer, Stat-Mux which can offer certain quality of service guarantees. So, we are saying a statistical multiplexer which can offer certain quality of service guarantees or what is called as QOS.

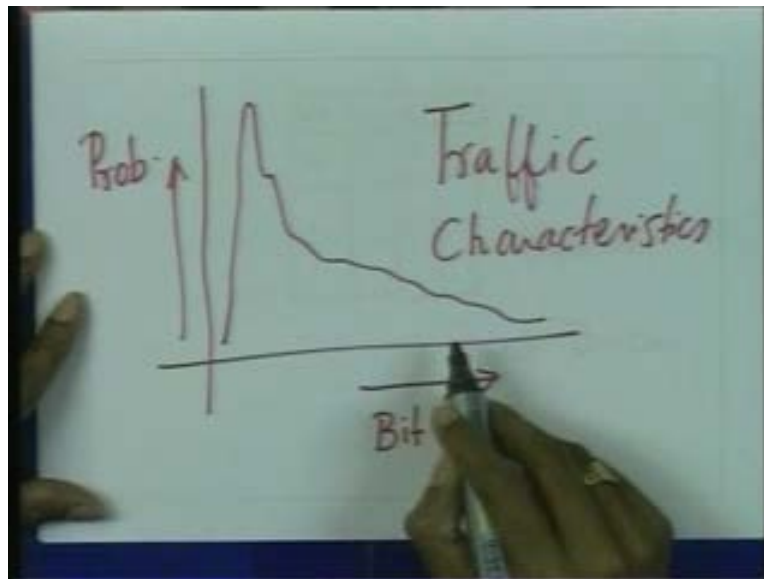
Now, we assume that these M sources, M traffic sources have the data or the traffic which are statistical in nature. So, these traffic sources have the data which are statistical in nature. So, they do not transmit a constant periodic bit rate but their bit rates are variable in nature and that is where we will show that there is an advantage of statistical multiplexing.

If indeed, all the users are transmitting at constant bit rates, then we know that multi rate circuit switching or circuit switching may be considered as more efficient. So, we are saying that these M sources or these M traffic sources are statistical in nature; their bit rates are varying and let us

assume that their bit rate can be characterized by some kind of a probability density function or a pdf.

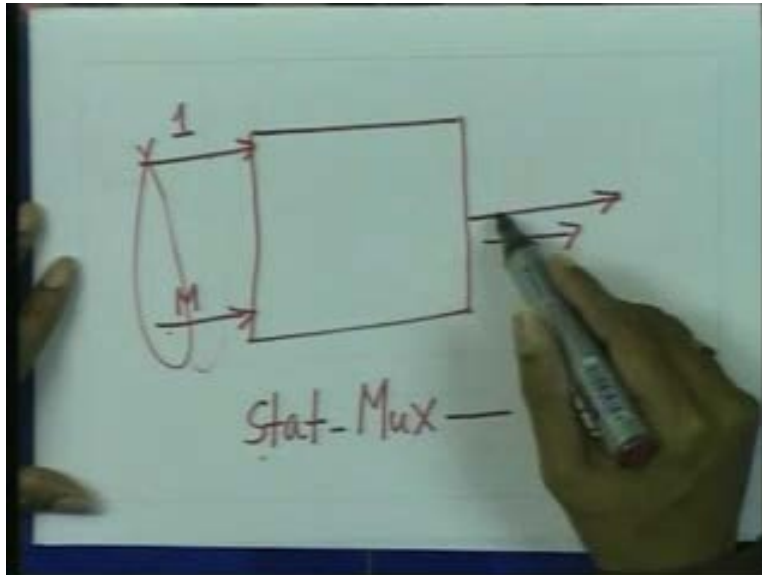
So, **let me just** let me just show you that how that can be done and so what we are saying is that here is this, here is this statistical multiplexer and here these m sources are there. Now, these sources are statistical in nature and they will transmit it on this output link. So, we are saying that each of these traffic sources **have the** are statistical in nature and therefore their bit rates can be characterized by some kind of pdf. So, let us say how that looks like.

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So, here on the x axis could be the bit rate and here it could be **you know** the probability. So, here is a source whose the traffic characteristics may be **may be** represented by this density function. We are of course assuming that traffic source is statistical or stationary in nature and this is the density function of this bit rate.

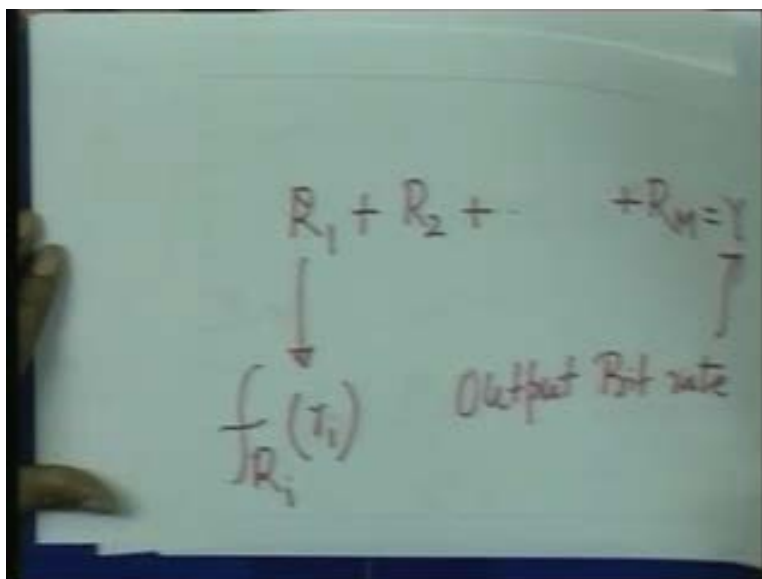
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So, now what we are saying; here is the traffic source, each of this traffic sources and this each of this traffic source are characterized by this kind of probability density function. So, now we are saying that when these sources are multiplexed whether we can have some kind of admission control which can offer you quality of service guaranties.

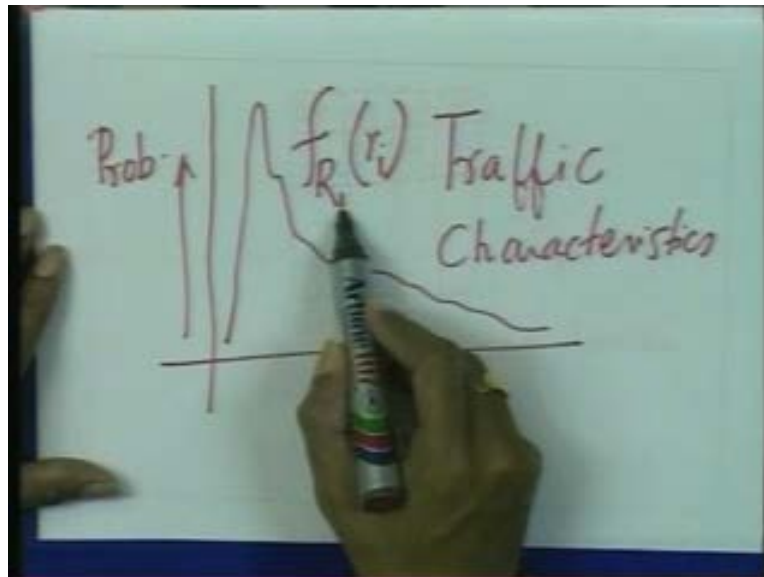
So, let us represent these bit rates of these individual sources by say, $R_1 R_2 R_m$. So, what we are saying is that these individual bit rates can be represented, let us say, by $R_1 R_2 R_3 R_m$ and so on. So, let me just explain you.

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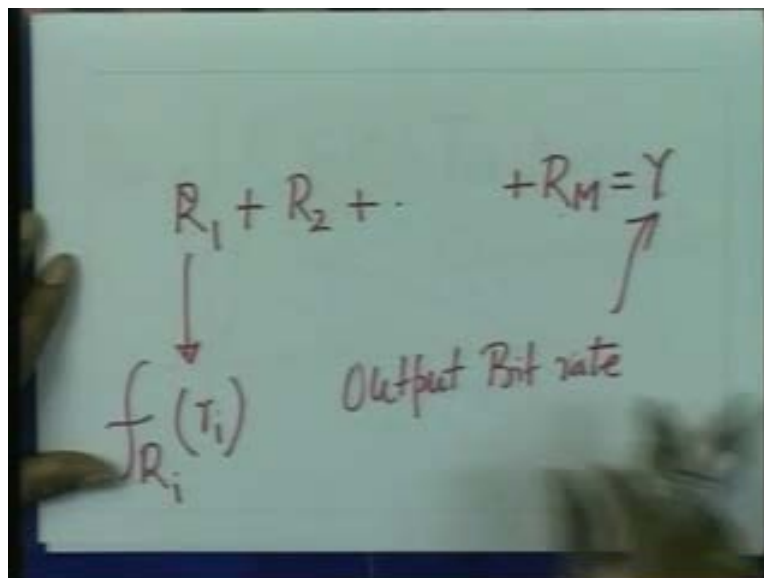
So, what we are saying is that this R_1 plus R_2 plus so on R_m that is equal to Y , where Y is the output bit rate. Now, each of these R_1 R_2 so on till R_m ; each of these bit rates are varying, they are statistical in nature and the density function of let us say each of these R_i which we denote by this has been **you know** shown to be something look like this.

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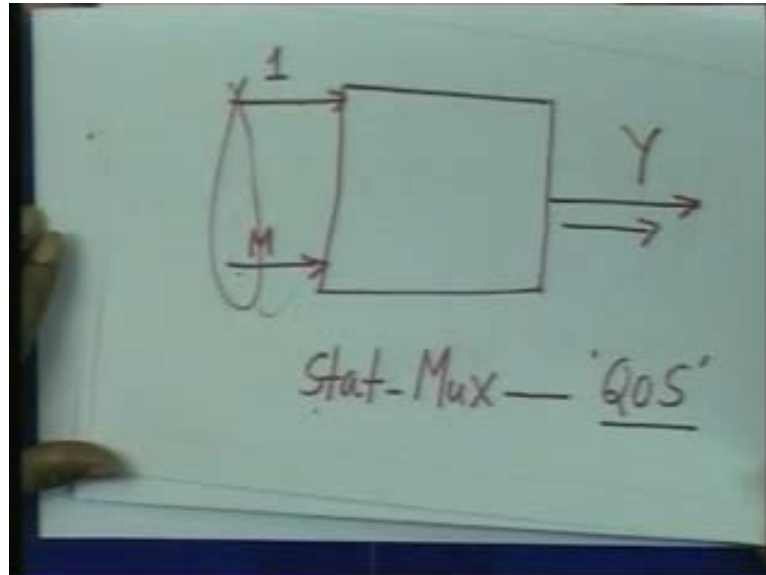


So, we can say that this is a density function, probability density function of a source whose bit rate in this case is R_i .

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Now, our problem is that these sources, these different sources are statistically multiplexed over here and here is the output bit rate Y and we may ask this question that if these sources have to be multiplexed and if the multiplexed bit rate becomes Y , then what is the probability that there may be a certain packet loss or a delay.

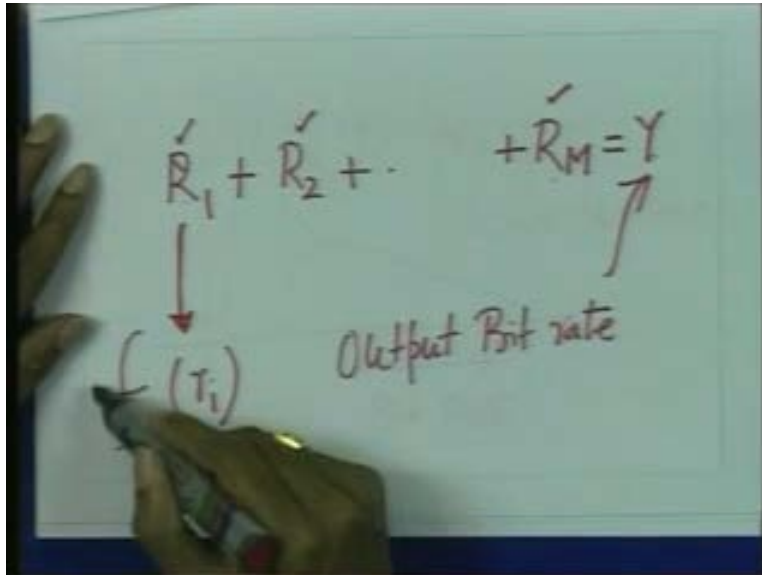
So, we will just review that how to offer quality of service guarantees in such a scenario. As we have seen that in a packet switched networks; typically when the users are statistically multiplexed, there is no concept of an admission control. Any users can submit his data to the packet switch. If the combined output rate of all the users exceeds the channel capacity, then the packets get buffered in the queue and if the queue length increases the buffer capacity, then the packets may get dropped also.

So, as a result there is neither a control on the queuing delay nor a control on the packet loss. However, the network will make every attempt to transmit the packet in the best possible manner and therefore a packet switched network is called a best effort networks.

On the other hand, for the real time services we want certain guarantees or certain quality of service guarantees either on the packet loss or on the delays or on both and then we were trying to just to see how those quality of service guarantees can be given in an otherwise statistical multiplexer, otherwise in a statistical multiplexer based multiplexing.

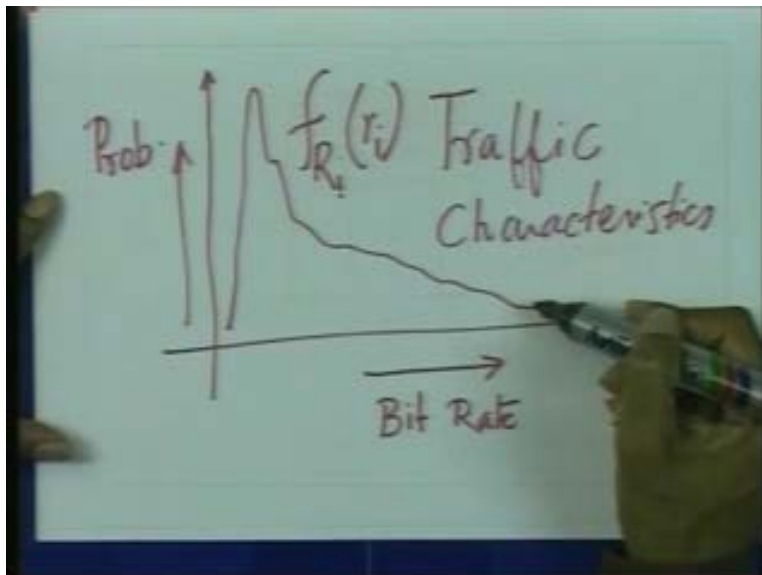
So, I was just explaining **explaining** you that here is a statistical multiplexers that you have seen that there are 1 to M users which have been multiplexed here and the the combined output rate is Y .

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So, we had considered that each of these users have the data rates $R_1 R_2 R_m$. These data rates, remember, are fluctuating. They are not constant bit rates but they are a variable bit rates and each of these data rates density functions is denoted by $f_{R_i}(r_i)$ which typically may look something like, something like this that we have just plotted here the bit rate versus the probability.

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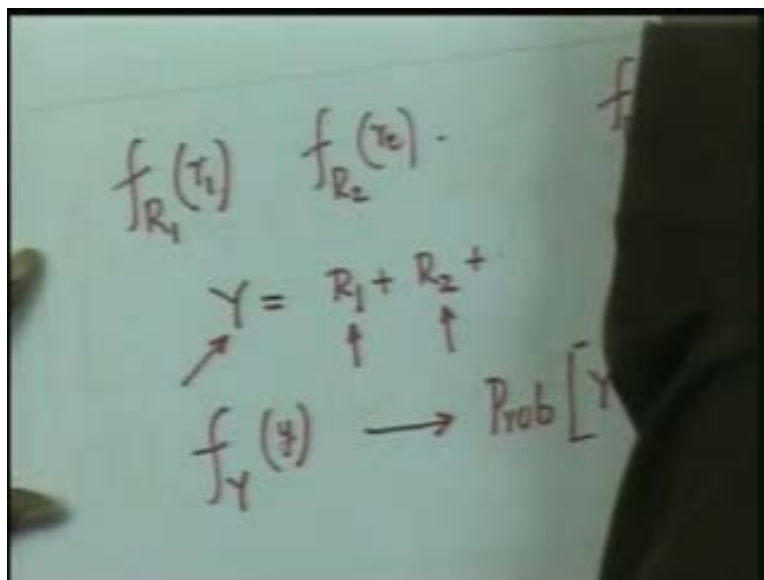
This is just an arbitrary realization, the density functions may vary from one user to another user.

Now, whenever we have to give a quality of service guarantees, we have to consider some kind of an admission control. Now, let us consider a hypothetical situation; **where**, of course, this is a hypothetical situation, this will not be there in practice. But for the time being, let us consider a hypothetical situation where each user knows his probability density functions. So, that means what we are saying is that each user knows how this curve **you know** how his curve **you know** looks like. So, what we are saying is that each user can submit its density function to the network.

Now, let us say that these are M users; each of the users has given its traffic characteristics to the network. Now, the network's problem is that whether to admit the user or not. That means the network wants to determine that how many such number of users can be admitted such that the packet loss, **such that the packet loss** remains below a certain tolerable limit.

So now, we are considering a situation where there are no delays. Let us say that there are zero delays, zero queuing delays. But each of the users each of these M users can tolerate certain packet loss.

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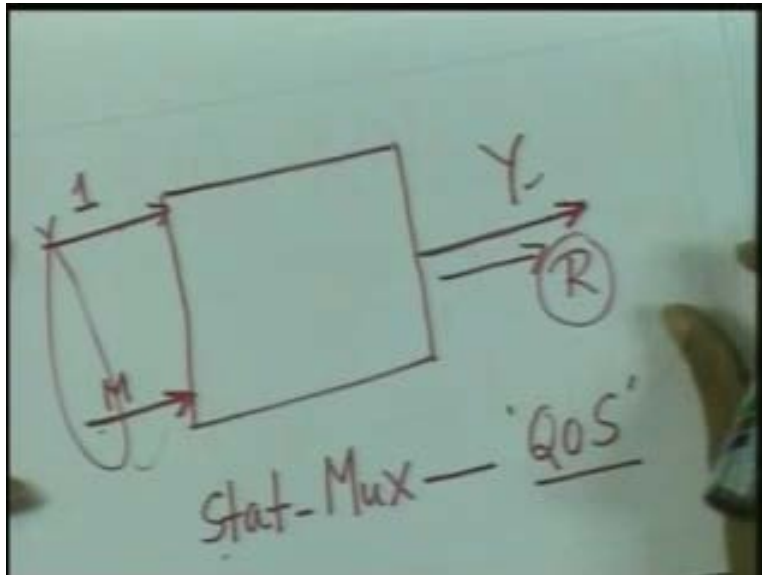


So now, what we are saying is that each of these users has submitted their density function that is f_{R_1} , f_{R_2} and so on f_{R_m} . Now, **the user** the network determines what is the density function of the output. That is Y which is Y is actually equal to R_1 plus R_2 plus R_m .

Now, given the density function of each of these individual users, the network can determine the density function of Y by simple convolution of the density functions of these individual bit rates. So, the network can determine what is $f_Y(y)$.

Now, from this density function we can easily determine, what is the probability that Y will exceed R, where R is the output link capacity.

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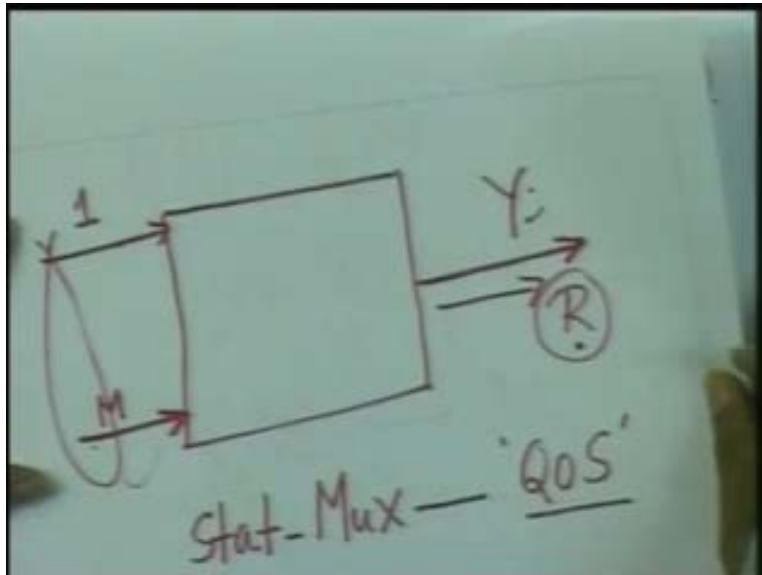
That means the R is the output link capacity of this.

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Handwritten mathematical expressions on a whiteboard. At the top, three terms are written: $f_{R_1}(r_1)$, $f_{R_2}(r_2)$, and $f_{R_M}(r_M)$. Below these, the equation $Y = R_1 + R_2 + \dots + R_M$ is written, with arrows pointing from each R_i term to the corresponding f_{R_i} term above. Below the equation, the expression $f_Y(y) \rightarrow \text{Prob}[Y \geq R]$ is written, with an arrow pointing from $f_Y(y)$ to the probability expression. The R in the probability expression has an arrow pointing up to the circled R in the diagram above.

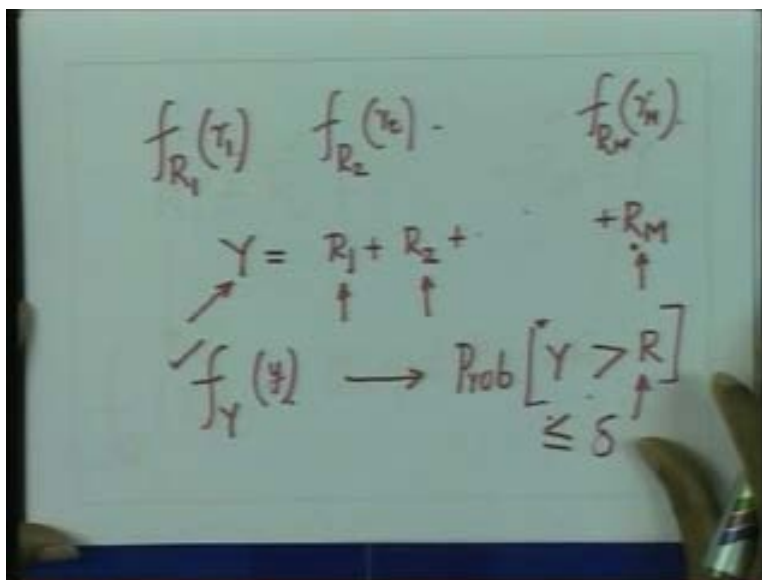
So, this can be determined by determining $f_Y(y)$.

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The moment it happens that Y exceeds R , there will be a packet loss.

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Now, if this packet loss is less than or equal to some tolerable limit, let us say δ , then we can admit all the M users. So, a typical quality of service - QoS mechanism would work something like this that each user submits its probability density function to the network.

Now, in this example we have shown that the user 1 submits its density function f_{R_1} , user 2 submits its density function is f_{R_2} and so on. By convolving these different density functions, the network can determine what will be the probability density function of the multiplexed output

which is Y , which in this case we have said $f_Y(y)$. From that f_Y , the network can determine what is the probability that the bit rate of the multiplexed output exceeds the channel capacity which is R .

Now, whenever the multiplexed output bit rate increases the channel capacity, there will be a packet loss because we have assumed that there are no buffers. Now, **if did packet loss** if this packet loss is tolerable, then it is clear that all M user can coexist. If this packet loss is not tolerable, then the M users cannot be admitted. So, in practice how this admission controls will work?

First, say let us say that there are no users. Then the user number 1 comes. The user number 1 gives his probability density function. From that density function clearly, the network can determine whether **you know** the probability that the output will exceed the channel capacity within tolerable limits. If it is within tolerable limits, yes, the user number 1 can be admitted.

Now, the user number 2 comes. The network asks him that what is its density function. So the user 2 says its density functions looks like this. Then the network determines whether by admitting this multiplexed output density function is such that the multiplexed bit rate is less than the channel capacity with a probability that is within tolerable limit, then yes, the user number 2 can also be admitted and so on.

And, after sometime when it so happens that by admitting another user, the quality of service guarantees given to the other users in terms of packet losses is getting affected or the quality of service guarantees given to this particular user is not met, then the user will be blocked or he will not be allowed to transmit his data.

Now, in such a manner we can have a large number of users multiplexed in a statistical fashion. Now, remember that if each of these users says that they cannot tolerate any packet loss. That is the packet loss they can tolerate is a zero packet loss and at the same time they cannot tolerate any delays that is zero delays, then the admission control scheme will degenerate to a **nive** admission control scheme where the users can be admitted based only on the peak rates.

So, in that case what will happen is that the network will ask each user to specify what its peak rate and then admit as many number of users such that the combined **the sum the** sum of the peak rates for each of the users is less than or equal to the channel capacity.

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$$p_1 + p_2 + \dots + p_M \leq R$$

↑
Peak Rate of user 1

That is **you know** a ... case where if p_1 p_2 p_m are the peak rates of the individual users, then they should be less than R equal to R, where this is like **you know** the peak rate of user 1. Now, this is the worst case scenario in a **nive** admission control scheme. Obviously, the number of users that can be admitted based on this scheme would be much less than what can be admitted **you know** based on a hypothetical admission control scheme like this. But here, again as we have shown that there will be a probability of packet loss that will occur with this.

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$$p_1 + p_2 + \dots + p_M \leq R$$

↑
Peak Rate of user 1

No loss / No Delay
↓
Circuit Switching

However, in this case if you have guaranteed that the number of users that have been admitted is such that p_1 plus p_2 plus p_m is less than or equal to R , then there will be **you know** there will be no loss and of course that there will be sort of no delay.

Indeed, if you are admitting a user based just on the peak rates, then the scheme essentially degenerate to **you know** some kind of circuit switching. On other hand, **you know** this scheme allows you to have some kind of statistical multiplexing. So, this is the case of statistical multiplexing where we can give you certain quality of service guarantees provided, the users can express their traffic characteristics by some kind of a **some kind of a** density functions which is like this - f_{R1} f_{R2} and f_{Ry} , if the users can express their density function.

Now, in practice however, we know that traffic source cannot specify its probability density function, indeed, it is impossible for an online traffic source to specify its probability density function. It may be possible for an off line source like **stored video or a** stored video or a video server. It may be possible to characterize its traffic characteristics ... and therefore we can know its probability density functions. But it may not be possible for an online traffic source.

Now, even if it were possible for a source to specify its traffic characteristics and submit it to the network; in any network which offers quality of service guarantees, it is necessary that a network must be able to verify that the source is confirming to its advertise traffic descriptors or its advertise traffic characteristics. If this does not happen, then the source may under specify its requirements or its traffic characteristics and may get admitted into the networks and afterwards it may start transmitting data at a higher rate than what it had, what it had advertised. So therefore, in any network which offers certain quality of service guarantees, it is necessary that the source specifies its traffic characteristics in such a manner that the network must be able to verify when the transmission is going on that the source is confirming to the advertise traffic descriptors.

Now, as a result, this means that the source really **you know** cannot specify its characteristics in terms of a probability density function because it will be very difficult for the network to verify that the source is indeed confirming to the statistical characteristics which have been specified by its density functions. Now, we will then see that what kind of traffic descriptors were standardized in an atm network **which was the** which was one of the earliest packet switch networks with quality of service guarantees and we will see later on, how its variants have now come into the internet also which offers quality of service guarantees.

But before we go into that, one thing is clear that unlike in the packet switching which was not offering any quality of service guarantees, there was no admission control. However, the disadvantage was that the traffic sources were suffering queuing delays or packet losses which were not guaranteed.

On the other extreme we have the circuit switching where there were an admission control, as a result the users were subjected to probability of call blocking. But at the same time there was no delays, no losses. There was a perfect quality of service guaranty. In between now, we have a statistical multiplexing with quality of service guarantees which combines the advantages of both

that is statistical multiplexing without QOS on the one hand and circuit switching with QOS but with less multiplexing gain on the other hand.

In a **circuit switching**, in a statistical multiplexing which offers us quality of service guarantees, it exploits the fact that its traffic characteristics are variable in nature, they are bursty in nature and therefore we can multiplex them by exploiting their statistical characterizations. At the same time these sources are amenable to some packet losses and some delays. They do not exactly want zero delays or zero packet loss.

As a result, we can not only substantially increase the network revenue by admitting large number of users than what would have been that otherwise possible in a pure circuit switch networks; but at the same time by using some admission control procedures, we can do a little better than pure packet switchings or pure statistical multiplexing which is not offering any quality of service guarantees.

Now, the problem however is that how do we achieve this statistical multiplexing game such that we have an increase in the channel utilization, an increase in the multiplexing gain but not at the expense of a complex admission control or a complex traffic characterizations. We would like to have a simple admission control and a simple traffic characterization and we will see in the next lecture how these can be achieved.