Real – Time Systems

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Lecture No. # 31

Real – Time Communications

Good morning. So, let start from where we had left last time. So far, we had discussed some very basic issues in real time systems and then we are looking at the operating system issues; we spend time on task scheduling and then we looked at features of a real time operating systems; we looked at commercial real time operating systems and how to benchmark these systems; the aspects that are important for real time applications, we had identified, and now, let us start our discussion on real time communications.

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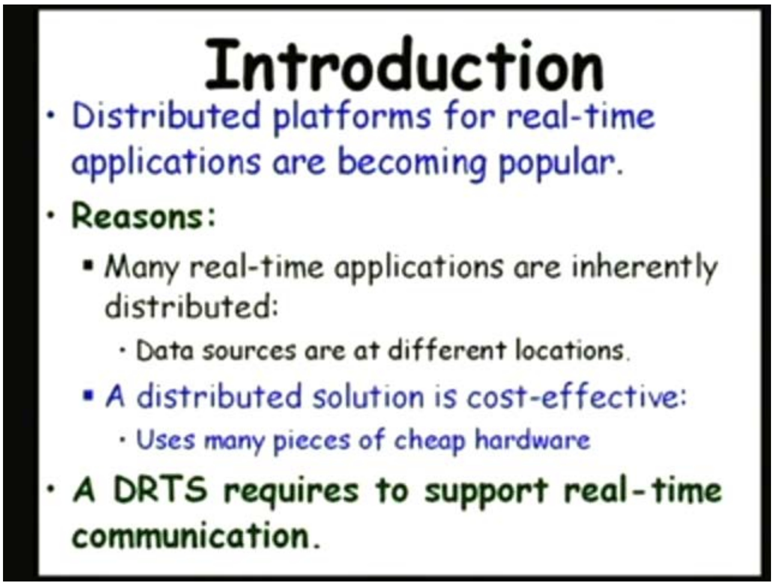

Nowadays, many real time applications, the distributed platforms are becoming popular; there are several reasons behind that, one is that, many real time applications are inherently distributed the events in the system; they arise at different locations and also the activities need to be carried out at different locations, that is one aspect.

The data sources are at different locations, the actions take a different location, and in this situation, a distributed solution is cost effective; we were discussing this last time, that we can process the data as they arise and send only the filtered data or the summary data. And then actions are taken based on that, and therefore, we have processing of the data at different locations, and the central controller also, this solution can be implemented using many pieces of cheap hardware compared to a single powerful computer a distributed solution can be cost effective.

And implicitly, a distributed real time system like this one, which implemented on a distributed platform, would require support for real time communication, to occur between the different processors and the actuators and so on. Besides this, I mean, where we have a really distributed implementation, in geographically distributed locations, but many times we will have situations, where the solution works in an enclosure, may be in a car. There are many pieces of hardware there and they need to communicate and there also, we will need real time communication.

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Let us first get some very basic aspects. One is how does the real time communication differs from traditional communication, which we have all done a first level course, is not it. We know that the traditional communication networks provide best effort service, they do not distinguish between the different applications; all application requests are treated similarly and the best that can be done for all these requests is taken up, there is no consideration, that see one request has a stricter requirements and it should be treated differently, no, that does not occur. Every request is handled similarly, but the network tries to provide the best service without really trying to meet any requirement of any application.

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But in contrast, we will see in next couple of hours that a real time communication network would be required, to support specific quality of service demands from applications.

So, unlike a traditional network, where the application makes a requests to the network to transmit certain data to certain destination in the, a, real time communication; it will not only initiate a request with certain data, but also we will demand certain quality of service, that for this connection, this quality of service is required.

So, the concept of quality of service, in case of real time applications is important. Now, just try to understand is a quality of service, what kind of demands does the applications make on the network, how the network tries to meet them and so on.

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So, the examples of such requests from the application may be the maximum permissible delay, the transmission should occur within certain time the maximum loss rate that can be tolerated by the application and so on; these are some examples of quality of service.

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And in a real time communication, once the network accepts the connection request with specific demands regarding the parameters of the connection, it should guarantee the requested quality, requested service quality should be guaranteed by the network that is the way the real time communication will work; the application requests certain service quality and Yes.

Based on the traditional networks, we will have certain delay after which the packing has to be retransmitted; means, even now you are saying that, it is required here, so, what is extra thing that. Ok. Means demand, ok. To that of additional, see the question, there is that in a traditional network, even when there is a concept of a time out retransmission and so on, so how a real time network is differs from that? So, the answer is that, in a traditional network, the application opens a connection and pumps data in, tries to transmit data and it expects that the data will reach certain time there, but it does not request that, see this data should be delivered by this time or this should be the loss rate, that is permissible or the jitter that is permissible for this.

So, what the application assumes is that, I have made the request to the network; the network will try to do the best that is possible with this data.

Internally, it might be using time out and so on, but externally as far as the application is concerned, it just enters some data to be transmitted and just believes that the network will do the best, that it can and provide the best service that is available, without making a specific demand on the network, that this is the requirement for the connection; we will, we will see more and more of that.

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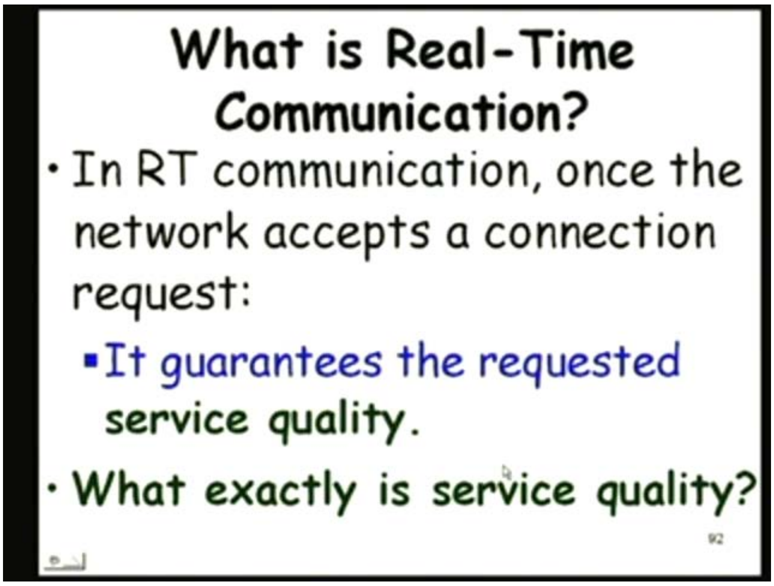

So, the term service quality is appearing again and again and let us elaborate on this, what exactly do you mean by a service quality? That is, requested by the application and also guaranteed by the network.

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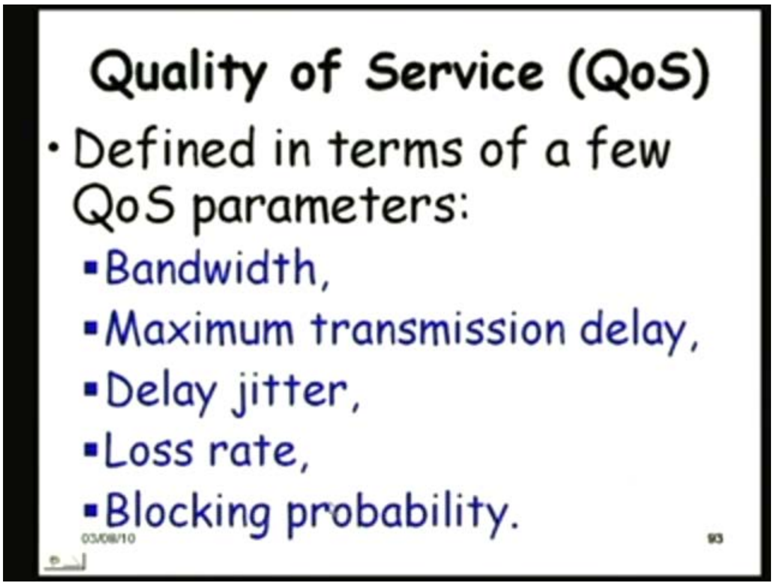

The service quality or the QOS, as it is commonly referred, is defined in terms of a few parameters; the parameters are the bandwidth requirement that is the speed of transmission. Maximum delay that can be permitted for the data to be delivered; the jitter that may be tolerable, the worst case jitter that can occur; the loss rate that may be permissible and the blocking probability that may be acceptable to the application. We will elaborate these terms, we just mention the terms here; we will elaborate them as we proceed in couple of minutes, we will elaborate them.

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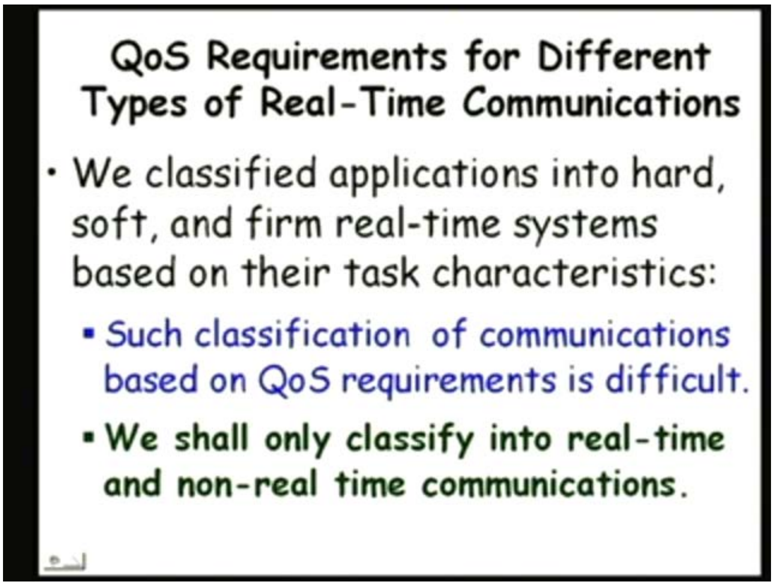

Now, let us see just examples of the quality of service requirement for different types of real time communications. Actually, if we remember that, when we discussed about the task characteristics of a real time system, we could classify them into hard real time task, a soft real time task or firm real time task, based on certain task characteristics.

But such classification of real time communication based on quality of service requirements is difficult, because there are too many parameters here and there may be some application requires some parameter stringent requirement on some parameter and very lose requirement on another parameter. So, a simple classification based on the quality of service requirements is difficult, but what you can do, you can classify into real time and non-real time communications or soft real time, I say real time and soft real time communications, hard real time, soft real time, that is all simple classification; we will use a classification into hard, soft, firm etcetera would be complex, let us not attempt that.

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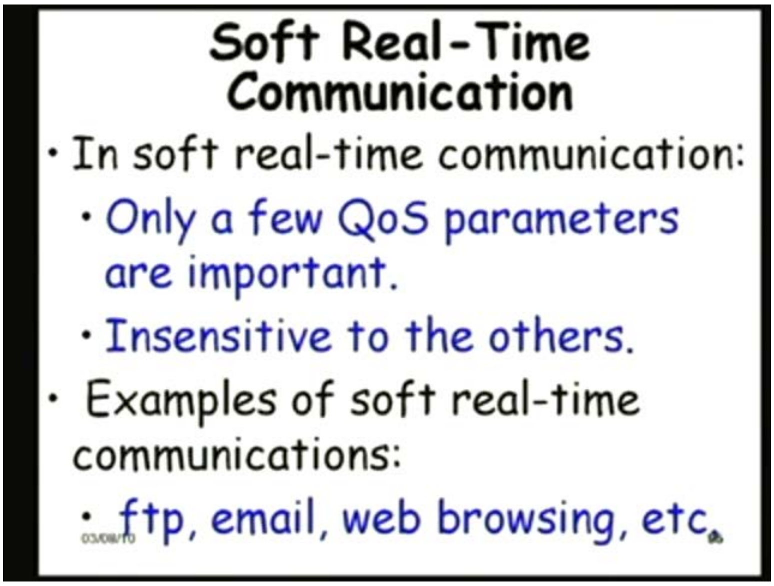

So, a soft or a non-real time communication is one, where certain quality of service parameters are important; it is not that, a non-real time or soft real time communication has no requirement on the quality of service, they have certain requirements.

But they are insensitive to many others, a few quality of service parameters are important for soft or non-real time communications and insensitive to the others; examples of such communications can be an FTP transfer, file transfer, email, web browsing; these are some applications, where the only few quality of service parameters, we will identify they are important here, rest we do not bother.

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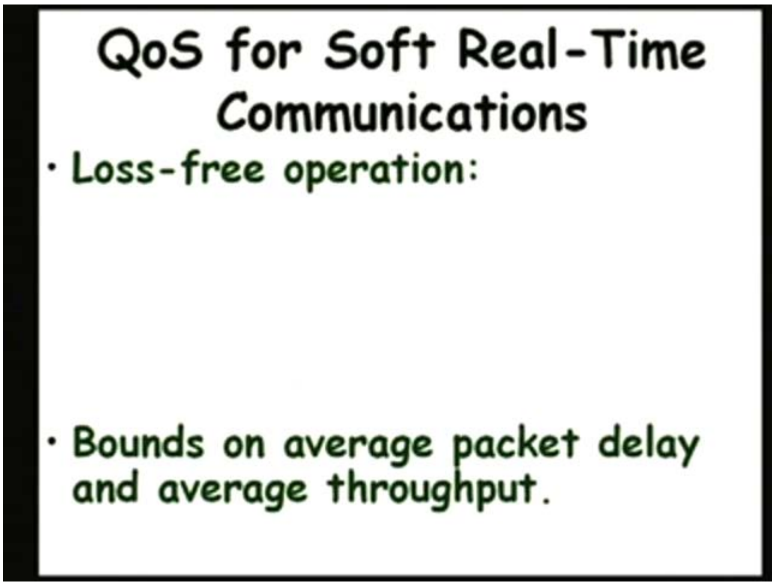

For example, the a file transfer, let us say FTP - file transfer protocol, for operation of a FTP, we, of course, need loss free operation, because if few sentences are (( )) or few sentences do not get transmitted, they are lost; few sentences are lost in a document of, let us say 1000 pages, it is still not acceptable to us; we will rather try to retransmit it, rather than accepting a document of 1000 pages were several sentences, some 20 sentences are missing, it is not acceptable.

Similarly, there is a bound on the average packet delay and the average throughput, just see here, the average word here; there is a bound on the average packet delay and the average throughput.

So, when we initiate a FTP, we expect that, it will complete in half a minute or maximum one minute, but we, what we do not really worry is that, what was the worst case packet delay that occurred or was there a jitter in the packet transmission, those a user would not bother.

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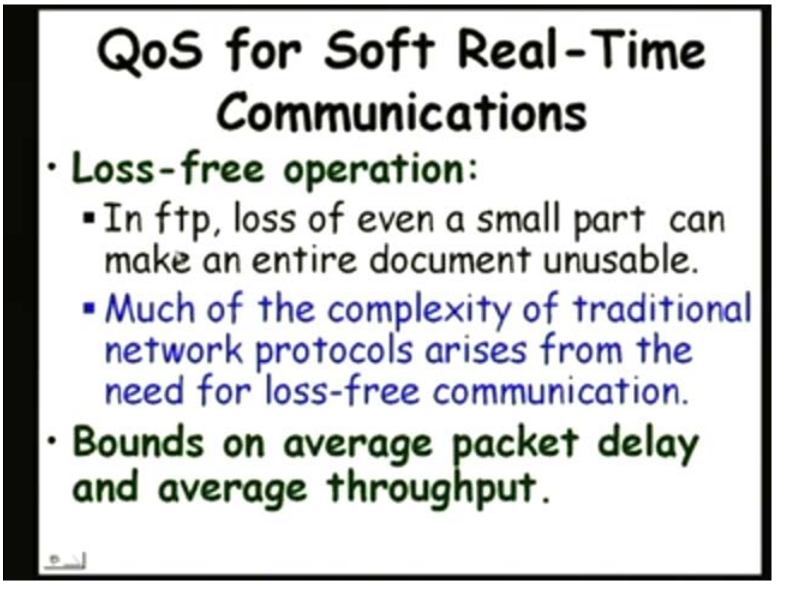

So, in FTP, a loss of even a small part of a document, can make the entire document unusable and if you analyze the traditional network protocols you will see that, much of the complexity of the protocol arises from the need for loss-free communication; there are several checks at different layers of the protocols, checking whether a packet is received correctly at the other end, I mean, it is across the node and then end to end and so on and of course, packet it may be known in different terminology, in different layers, for example, it may be called as frame, in let us say layer two.

So, the concept is that, the check is done at different layers and that is one of the major complexity in the traditional network protocol, how to ensure loss-free, loss-free transmission, you look at TCP or any other protocol, you will see in how many places this aspect is checked and taken care.

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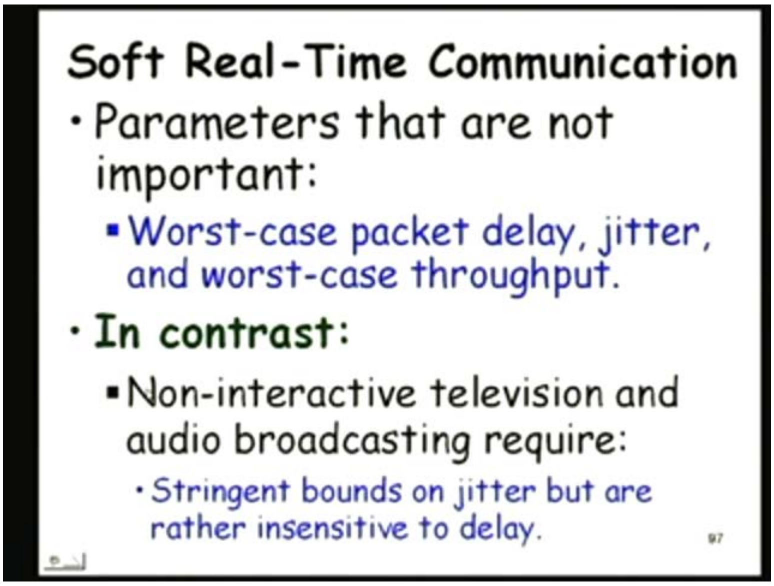

But there are some parameters which are not important, for example, the worst case packet delay, we do not really, it does not really matter if certain packets got delayed, as long as the average throughput is met, we do not bother, if some packets got delayed or they came very fast, we do not bother, jitter we do not bother.

But in contrast, just look at a simple application of a, television broadcasting. So, here, this even a non interactive television, just one side, you know it is a multicast or a broadcast, where there is a stringent bound on jitter, here jitter is very important, because the rate at which frames are shown; if some frames arrive late, it will show up as a glitch or you will see a spike on the screen. So, jitter needs very stringent bounds for a transmission, like let us a, even a television broadcast occurring on a computer network.

But just consider, that this transmission may be insensitive to delay, even though jitter is important for a television broadcast, but its insensitive to delay, because it does not really matter, whether it shows up once the frame is transmitted from the antenna, whether it shows up on the screen in milliseconds or they appear in after 10 seconds or 30 seconds, it does not matter, as long as all the packets, sorry all the frames they are all incur 30 second delay.

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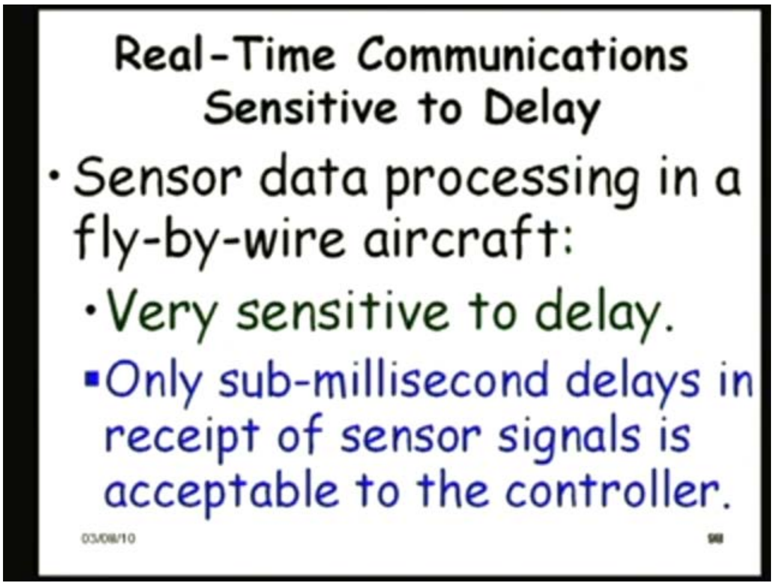

But let us look at another application, which is a sensor data processing in a fly by wire aircraft. So, an aircraft which is automated and there are no pilots in this, say fly by wire, so only ground commands can be given.

So, here there are sensors, which sense the velocity altitude acceleration and so on and then based on that data processing takes place, but if there is delay in the time, by which the data is sensed and it reaches the controller then there will be problem.

For example, an obstacle is sensed and there is a delay in notifying the controller, it will cross the aircraft is not it. So, depending on the speed, I mean for a typical speed of a fly by wire aircraft, sub millisecond delays in the receipt of sensor signal is acceptable to controller, but as I was saying that, even this bound depends on parameter, such as what is the speed of the aircraft; if it is a supersonic aircraft, even this may be too late.

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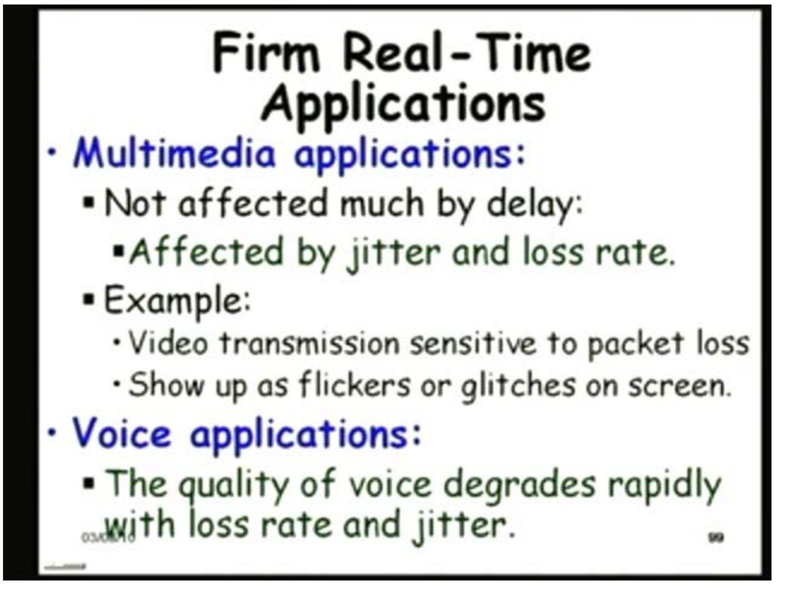

Now, let us look at the requirement of some firm real time applications, we are not saying communication just check here. So, we are saying that our notion of a firm real time application, what kind of requirement they might make on the communication network.

We had from our first discussions we know that multimedia applications are a type of unreal time applications. So, as far as their network requirement is concerned, they are not much of affected by delay, that is, what we are saying about television broadcast.

But they are affected by jitter and of course, loss rate, that few losses occurring they can be tolerated, just 1 frame missed out of 1000 frames can be tolerated, but jitter is one where strict requirement is made by the application. Video transmissions are sensitive to packet loss and if there are too many packet losses occurring, they can show up as flickers or glitches on the screen.

Even in the voice applications, voice transmissions, the quality of the voice would degrade rapidly with the loss rate and jitter, you will start noticing that there is something wrong, when packets are lost and also these are jitter, the packets.

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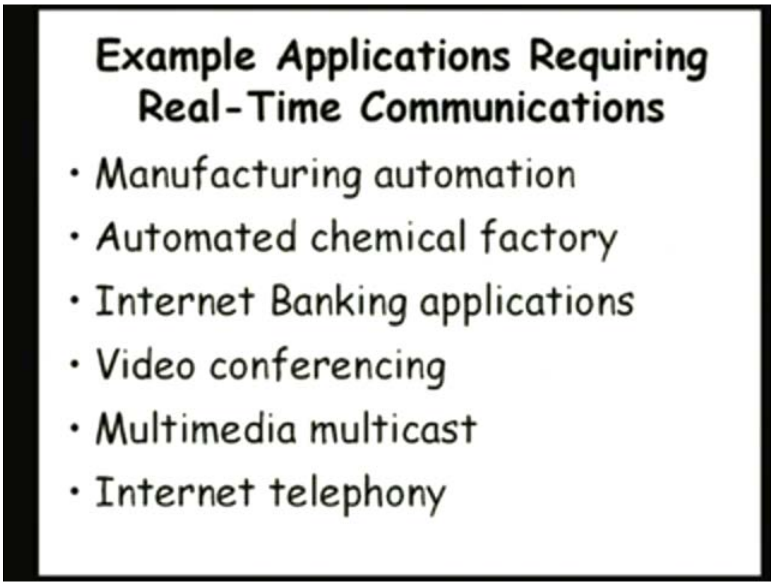

We will, we would have really liked to discuss several applications how they work and what is the role of real time communication there, for example, a manufacturing automation, where a factory, where computer controls everything; there are robots who operate the factory. An automated chemical factory, whereas the different chemical chambers are sensed and the reaction is control by a computer; an internet banking application, it is also a real time application, because unless, you let us say two transactions, the time is considered there can be inconsistency, just look at a railway reservation, similar is with the banking transactions; so, time need to be considered and you know that the banking transactions, they time out very fast.

So, videoconferencing; multimedia, multicast multimedia information, multicast to several receivers, internet telephony, voice over I P, so would really like to have discussed these applications and how the real time communication occurs in them, but considering that our time is (( )), we will not spend time on this just a minute or, so, we will discuss on the manufacturing automation.

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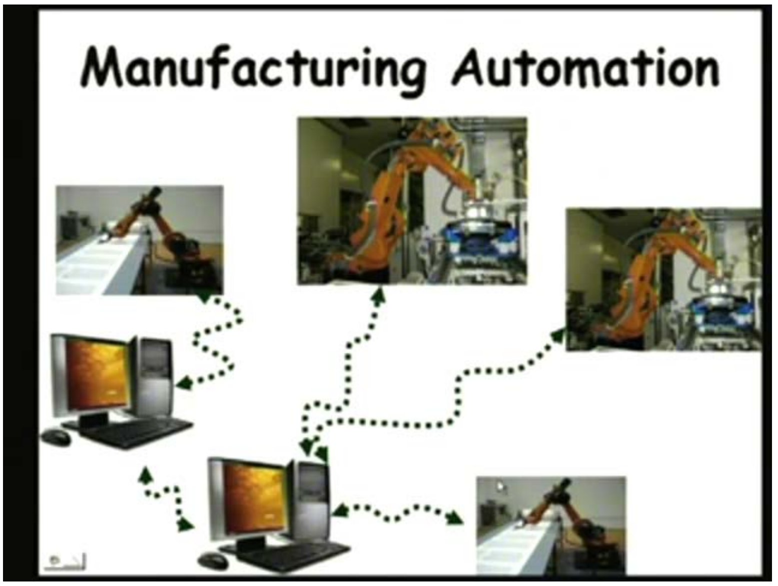

So, just see here, there are various types of robots working in a company and different computers may be controlling the manufacturing at different locations, may be this is occurring in a different building. And this, in another room let us say and the material is operated transferred and it is worked up on by one station passed to another station and finally, finished here and the material is packed and sent out, but here what kind of communication would occur; see here, this is, we have shown here is a wireless communication, but need not be a wireless communication, they can be because the stations, if the stations are fixed station; it can be even wired communication, but the important thing to note here, is that, the robots would sense when certain items arrive.

They would notify the computer, the computer would identify what work needs to be done on that piece of material, may be several types of pieces coming and then it initiates the work, it logs the activities and once it completes the work, notifies the computer and it alerts the other work station, where the work is going to come about incoming work and it starts sensing that and so on and even the computers they communicate among themselves.

And also there may be logging going on; logging is a not a, so critical activity, because even if it is not logged properly or the there is a delay in logging, it does not really matter. So, some of the communication is very important, for example, if a machine malfunctions, so that, the communication regarding that has to be processed at very high criticality compared to a logging or a fire detected in the factory has to be considered with the highest priority. So, we have these notions of a requirement of the different communications from the same station, some are high priority. They have different quality of service requirements and some others are not so high priority and there quality of service requirements is very different.

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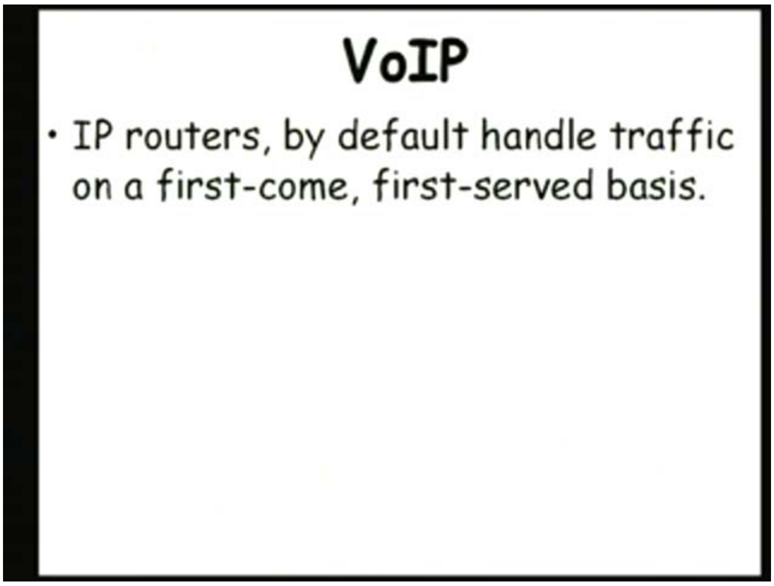

Similarly, in a VoIP application voice over internet protocol, so, which some of you would have already used is that, a IP router handles the traffic and in a traditional I P router, the handling of packets is by first come first serve, it does not distinguish, it is a traditional, we said a traditional IP router; it does not distinguish between which application has initiated the request is it a VOIP, is it a FTP, is it a email, it treats all of them similarly, but we know that, the VoIP have special requirements on the packet delivery compared to let us say FTP or an email. So, in a traditional network, the VOIP would not work very well, which many of you use VOIP must have noticed.

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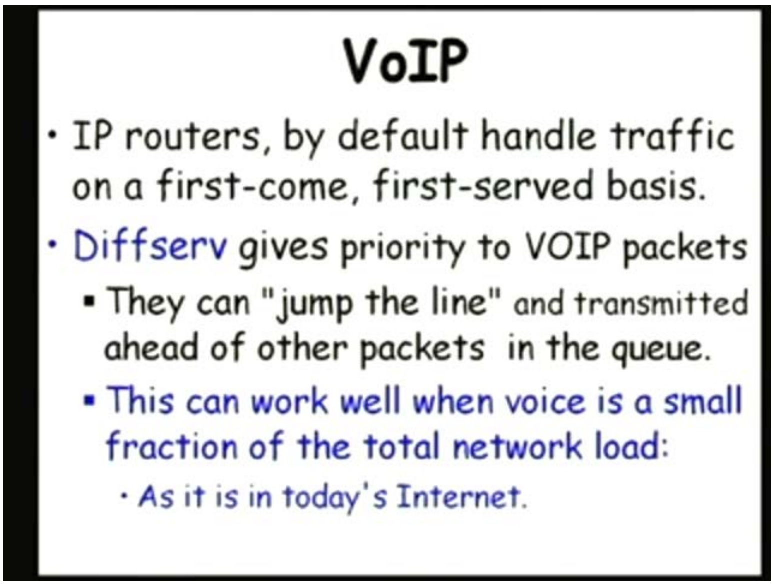

So, as we proceed, we will discuss about Diffserv, I mean, how does some packets can be given a priority in an internet situation. The Diffserv can give priority to VOIP packets, where the VOIP packets can jump the line and transmitted ahead of other packets in the queue.

And this solution, the Diffserv solution which we will be disusing after an hour or 2 hours of other discussions; this will work well, when the voice traffic is small compared to the total network load. So, what you are saying is that, unless there are other kinds of traffic, like email, FTP etcetera, the concept of jumping the queue does not arise.

So, there is significant, other traffic we can jump the queue and the Diffserv would work well and which is the case with the internet. VoIP load is a very small fraction of today's internet load; so, this we will discuss later, but this is just a motivation for what you are going to discuss.

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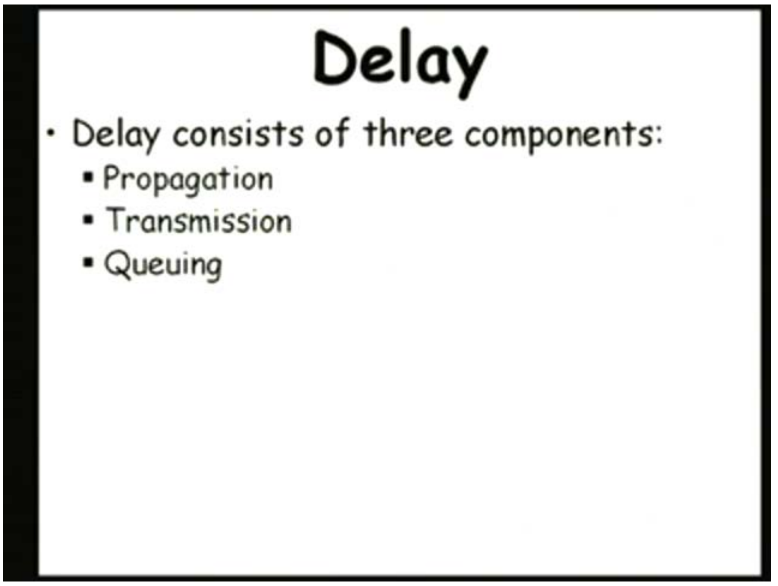

Now, let us look at the different quality of service parameters, that we were discussing; one of the important QOS parameter, we are mentioning was delay. Now, let us just analyze little bit about this delay, when a delay occurs, let us say a communication request, if just requested some data to be delivered at a destination. So, what are the different components that cause the delay, if we understand that, in our later discussion, we will be clear what we are talking about; so, what do you thing, why does the delay arise in a network. (( )).

No, what are the components of delay, see you are transmitting from 1 node, let us say you are just doing a web browsing; now, in the web browsing, you clicked on a URL and then after certain time the page came, one is the processing that occurred on the web server, which are recognized your request got the requested page and then send it; there is a delay there, let us omit that delay; without that delay what are the other components of the delay. (( )).

So, here different persons are saying different components; let us see, there are three components actually which constitute the delay; one is the propagation, a signal electromagnetic signal take certain time to travel and then the transmission, depending on how much bandwidth or what is the speed of the network, there is a certain time require to transmit; if your network is, let us say 1 m p b s or let us say, it is a 64 k b p s link and you are transmitting a 1 mega byte packet, it take certain time for the packet to be read out into the network that is the transmission is not it, because at that speed it will be read out to the network that is the transmission delay.

So, the propagation delay, the transmission delay, the propagation delay depends on the distance the signal needs to travel; the transmission delay depends on the speed of the network or the bandwidth. The speed of the network is typically given in terms of either megabits per second or kilobits per second. So, depending on how much data you transmit, it takes certain time for the data to be read out into a network at that speed.

And the queuing delay, the queuing delay is especially important, if the network passes through several nodes and get queues queued, but even otherwise, it might get queued at the receiver or the sender before it is transmitted, even when it is across to machines there may be queuing delay, because it might get queued either at the receiver or at the sender, but if there are several nodes involved queuing is a major component of the delay.

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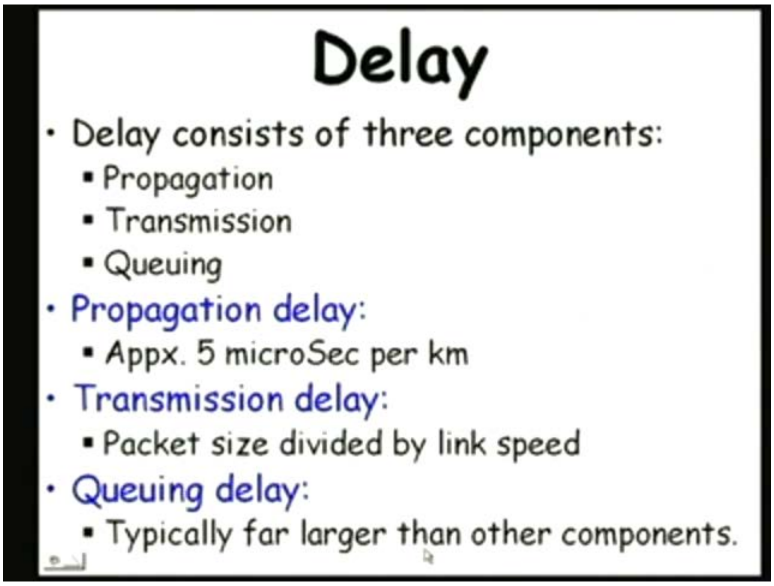

So, have you thought about this, that if you make a request for a web page clicked on the URL, which one, which of these components would be the one that is, dominating, I mean, where the maximum delay would occur? Queuing, it is the queuing delay, which is typically the largest delay in an internet situation, because the propagation delay is approximately 5 micro second per kilometer. So, even if it is 1000 kilometer you are talking of, it is only few milliseconds; a 1000 kilometer will be 5 milliseconds is not it, how many milliseconds micros? 5 milliseconds, so, the transmission delay is the packet size divided by the link speed and the link speed, we said that, it can be 64 k b p s or m b p s and if we are talking of the link between two countries, it can be gigabits per seconds.

The links between a one desktop to let us say router or something, may be that can be kilobits, 64 kilobits or may be megabits, 10 m b p s or something, but after some time as it goes from node to node and then from country to another country, it may be in terms of the speed may be gigabits per second.

So, the transmission speed is also not very large, but the queuing delay is typically much larger than all other components in a typical application. So, let us have these points in mind, it will become important as you discuss more details in real time communication, that a delay consists of three components, the propagation delay is the time the signal takes to travel, travel from the source to the destination, because the electromagnetic wave in free space travels how much, 3 into 10 to the power 10 centimeters per second, but in a, other medium like let us say copper or let us say aluminum its speed will be less slightly less. Similarly, the transmission delay… (( )).

Light speed, in a fiber optic, it might travel at the speed of the, not at the speed of the free space, but it will be slightly lower than that. There is refraction between the… No, not only that, see the free space speed is 3 into 10 to the power 10 centimeter per second, but even fiber optic, it is a medium and it is not really a free space is not it. So, it will travel at a speed less than 3, 3 into 10 to the power 10 is the highest speed, maximum speed at which the signal can travel and other Medias speed will be less, but the speed is so large, is not bother us, but it is the smallest component out of all this. So, the transmission delay arises due to the finite bandwidth or the speed of the link, that is in terms of kilobits or megabits or even gigabits, the queuing delay would also occur and typically the largest of all these delays.

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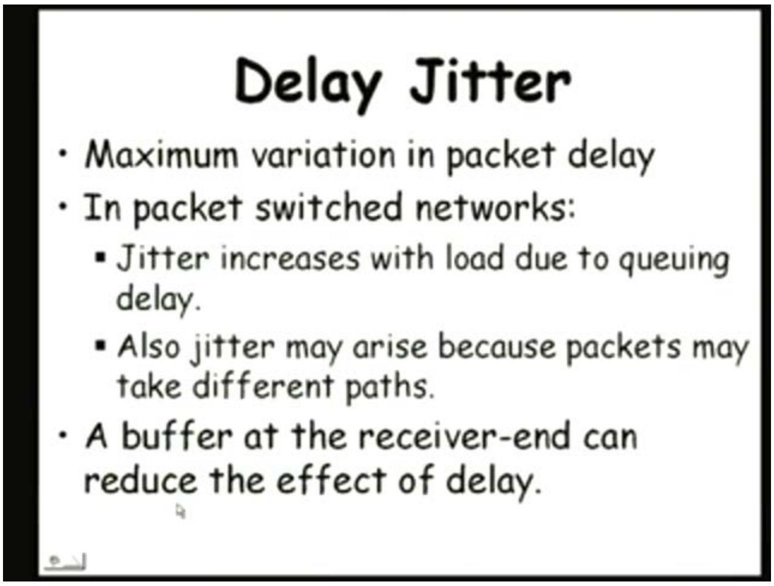

The delay jitter is defined as the maximum variation in the packet delay. So, each packet take certain time to reach, some packet arrive in let us say 10 millisecond, some packet arrive in 11 millisecond and some packet arrive in, let us say 9 millisecond and some packet arrive in 20 millisecond, not 20, let me just say 15 millisecond. So, what will be the jitter, they distributed around 10, let us say some are arriving at 9.

Five (( )), some are arriving at 10 and some are 11, some are 14, some are 15. So, what will be jitter? 5, 5, 15 minus 10, 15, 15 minus 10; I mean why 10? 10 is a, is the actual thing that is… I mean ,what do you mean by actual, see you observed that some took 10, majority took 10, but some have taken even 9 and some have taken 11, some have 14 or 15; so, the certain time that the…We will have some reference about which we should take the jitter.

So, what, what is the reference in this case, I mean, what are you saying, I mean, what will be the reference is it the above 10, we should take then above, so the time that the average packets took and the longest time that the some other packet took, not really jitter is not defined that way is the maximum time some packet took and the minimum time some packet took; so, 15 minus 9 is the 6 millisecond is the jitter, that is, the way the jitter is defined. So, it is the maximum variation in the packet delay, that is, between 15 and 9, there is a variation. So, we will say 6 is the jitter.

Now, let us see why the jitter arises; in a packet, switched network the jitter arises due, due to the load on the network, that is, it undergoes varying queuing delay as the load on the network increases or the run over load; some packets which travelled when there was load was less, they reached faster and as the load on the network increased, the other packets would possibly get delayed. So, that is one reason, the other reason may be that the packets travel different paths, because you know the paths are selected depending on whether some nodes are faulty links or faulty or may be the load on the different; so, this may be another reason why the jitters arise, but one thing good about jitter, is that, in many applications, the effect due to jitter can be smoothened out, just by having a buffer at the receiver; a buffer at the receiver end can smoothen out or reduce the effect of delay, let us just see an example.

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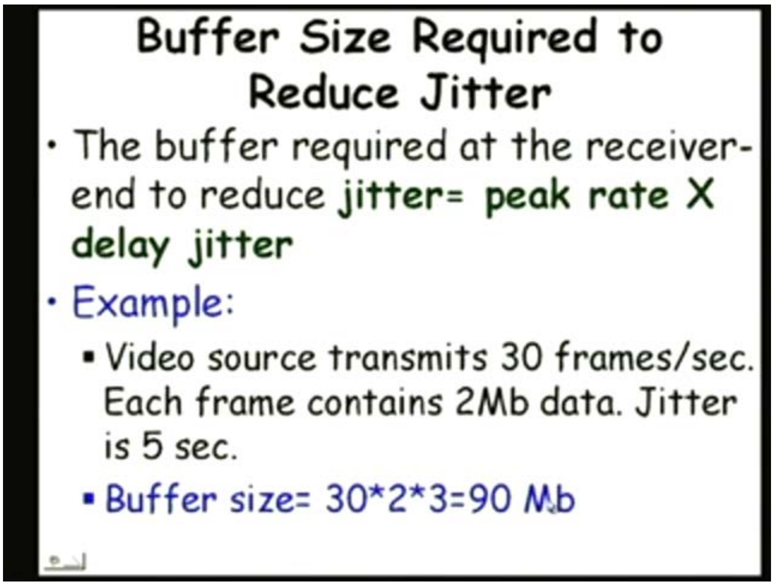

So, as the, we are not saying that the buffer size would eliminate jitter, we are not saying that, we are saying that it will smoothen out the traffic or the application that is using the data, will not feel as bad as when they are not buffered. So, it will reduce the impact of the jitter. Now, the idea is that, as the packet keeps on arriving, it keeps them in the buffer and if some packets are delayed, it can consume the packets that are there in the buffer.

So, if we use that commonsense, we will see that the size of the buffer, that is, required reducing jitter that is a size of buffer is equal to quick rate into the delay jitter. So, if the jitter is let us say 6 millisecond and the rate at which the packets are being transmitted is 1 m b p s, then it will be 1 m b p s into 6, that much data has to be buffered. So that, even if some packets are delayed, then the application can continue running without noticing the jitter, by consuming the packets, that are there in the buffer; simple thing, nothing too much to worry here or discuss here is not it.

For example, let us say a video source transmits 30 frames per second and each frame contains 2 megabits of data, that is the conventional; we will use that small b for bits capital B for bytes; typically used in all books and all literature and the jitter is 5 second, so what will be the buffer requirement at the receiver to smoothen out, the jitter what do you think, do not have to really work out, at least you tell me what you… (( )).

So, it will be 30 frames into, 30 frames arrive per second, so per second 30 frames and each frames contains 2 m b; so, 2 m b into 30 is 60, 60 and we need to store that for 5 seconds to reduce the jitter. So, it should be 30 into 2 into 5, that is, 300, is not it, sorry I think I have done a mistake here, it should be 30 into 2 into 5, it should be 300, sorry a mistake that I have done here, it should be 300 megabits of buffer size required, but what about let me just give you a small slightly different problem, what about let us say the packets, the, the transmit between let us say 100 millisecond to let us say 110 millisecond; the minimum time that the packets take is 100 millisecond and the maximum time that take is 110 millisecond.

So, and let us say the packet size is, let us say 10 k b p s, sorry 10 m b, 10 m b let us say the packet size is 10 m b or let us say simplify 1 m b; so, the packets take 100 millisecond to 110 millisecond and then the packet size is 1 m b and let us say the jitter requirement, the jitter that occurs is let us say 0.5 second. (( )).

Yes, see that the packets that arrive, they have a j, it they, they the delay is the delay, they encounter in the network is between 100 to 110, some come at 100 and some in the worst case come at 110 and the size of the packet or the frame, let us say packet would 1 m b packet looks or we will say frame is 1 megabits. And the, the rate is let us say what, what did you say, one is the packet size we said and then the, we said about the…Time.

That the the delay and let us say the speed of the network is 1 m b p s or something, simple thing, so what will be the buffer requirement? But in between, there are some of the jitters coming. No, jitter you have to compute from the delay, no, that was a mistake, so, jitter you have to compute. So, what is the jitter? 10 millisecond, 10 millisecond and then, (( )) 1 m b p s, 1 m b p s and then, so, what will be the, what will be the buffer size that is what I need? How many seconds is the maximum delay? (( )) 0.01 m b.

0.01 m b would you, I agree with him, you need a buffer size of 0.01 m b, (( )) 10 milliseconds. Now, just think of the answer that you are giving.

What is the frame rate sir? We need the packets are arrive in the frames, are arriving packets the frame the frames are being transmitted and each frame size is 1 m b and then the jitter is we have, we have given the delay that they encounter and then what is the… Frame rate. Frame rate does not matter is not it. Sir, it is bounded by the taken network. So, we can consider only the peak rate as a rate of network.

So, let us a one packet arises per second or something does not matter, 0.01 m b is, it is that, what you are saying, but my only question is that, if you store 0.01 m b, let us say a packet has arrived, so which 0.01 m b do you store? 1 packet, 1 frame has arrived; so, frame is 1 m b and you are saying that you will store only 10 per. (( ))

For this case, I am say, I am not saying that whether realistic or not, but which part of the frame would you store? Dependent on which arrives first, you will store the header of the frame is it. Which arrives first, then is on the receiver and the sender end we can store it at the (( )).

No, you do not have all those options; you have the option at the receiver end. So, basically, what I was saying is that, here the common sense is that, at least 1 frame you need to store is not it, you cannot store a fraction of a frame and get away at least 1 frame you need to store, even if it is coming 0.01 or something, still need to store one frame, otherwise the question arises which part of the frame.

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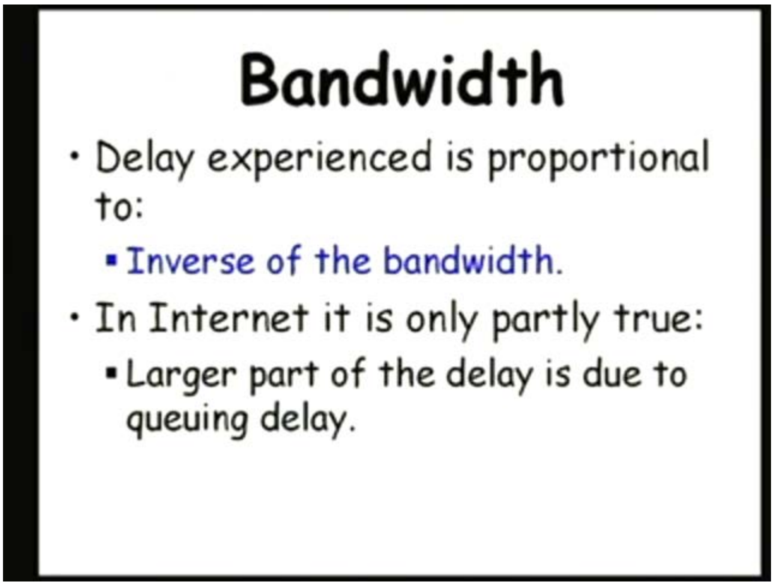

Anyway proceed, bandwidth is also another quality important quality of service parameter, the dual of bandwidth is the speed of the network; the, the speed and bandwidth are dual concepts, those who have studied about transformation between different domains might make more sense to them about how the speed and bandwidth are look at it or related, but still we can think that bandwidth and the speed are dual concepts and the delay experienced is proportional, inversely proportional to the speed or the bandwidth; if bandwidth is more, delay will be less or the speed of transmission is more transmission delay will be less, so, their inverse relation.

But we already saw that, in the case of internet, it is only partly true about the inverse proportional, because the major component of the delay is due to the queuing delay.

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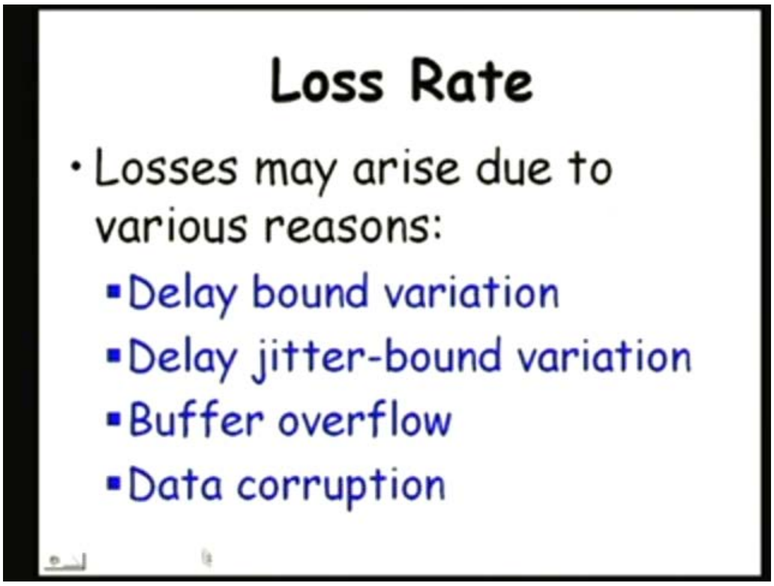

The transmission delay is only a very small part and loss rate is also another quality of service parameter; now, let us try to analyze, why losses occur in the network, what do you think, why losses occur in a network? The losses occur in a network, in a real time situation, due to many reasons, may be due to delay bound variation; the packet arrive too late to be used by the application, as far as the application is concern, the packet has been lost is of, no, use delay jitter-bound variation.

So, the jitter requirement for the application was 0.1 milliseconds, it arrived slightly too late, it was, it was rejected, because the delay jitter-bound variation, even the delay bound was alright. Buffer over flow can occur at the receiver.

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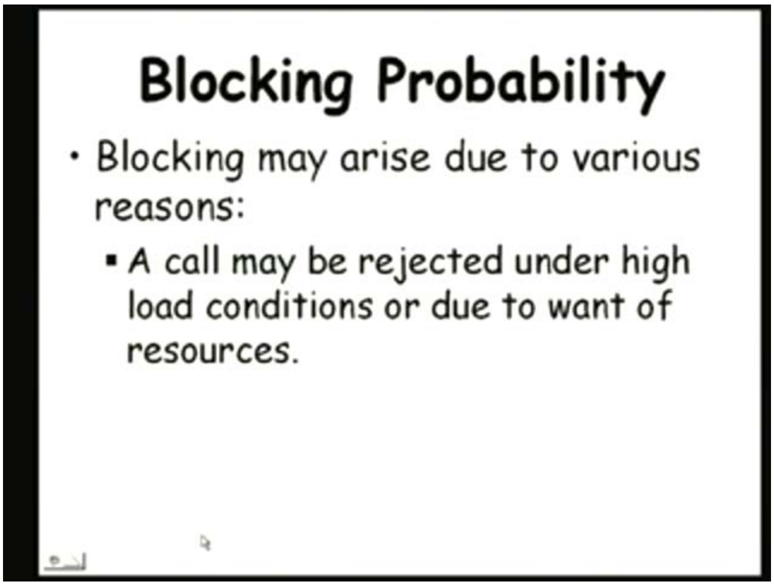

There can be also a data corruption, of course, some protocols they guarantee corruptions do not occur, but certain protocols do not; now, let us see the blocking probability; this is also another quality of service parameter. Blocking arise due to various reasons, a call or a connection request may be rejected by the network, because sufficient resources are not available, for example, if there is a request for certain bandwidth or certain delay and the network cannot meet those under any situation.

So, the network would reject that and that is one reason why the network rejects it, because the demand from the application cannot be met by the network with the given resources the other, is that, temporarily there is too much of load on the network, it cannot meet the specified request. So, the two main reasons, why the call drops or the blocking probability might occur.

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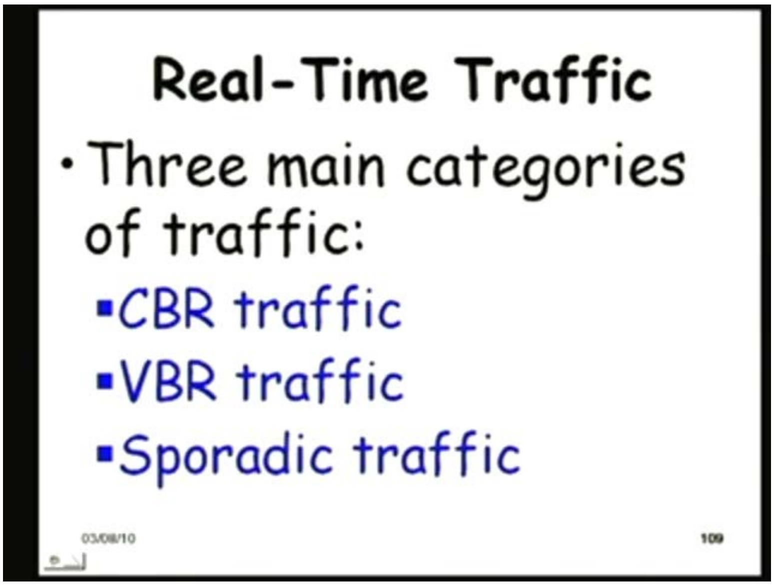

Now, let us look at another very basic concept is about the kind of traffic that typically arise in a real time network. There are three main categories of traffic, that we will talk about in our subsequent discussion; one is the constant bit rate traffic, we will just look at some example sources, we generate this and we will see that, the constant bit rate traffic is a majority of the traffic in a real time network in CBR traffic, then the variable bit rate traffic.

As the name says, here the data generated by the source varies greatly with time; we will see which sources generate that and then the sporadic traffic, where here, if there is almost no data for long time and once in a while then suddenly data arrives.

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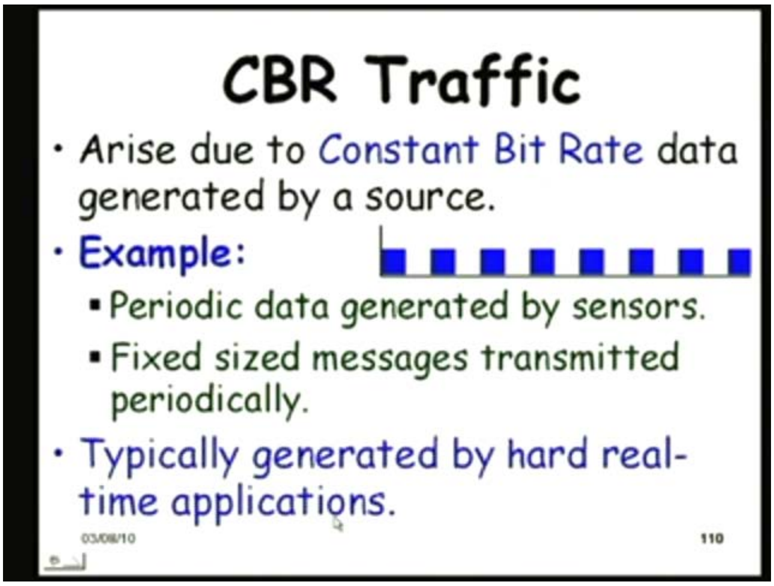

The CBR traffic or the constant bit rate traffic, as you are saying this is the majority of the data in a real time network. So, here the data keeps on arriving certain packet sizes or let us say some size of data, let us say 100 bytes of data or 1 kilo byte of data, may be arriving every 1 millisecond on the network. So, typically, these are the periodic data generated by sensors, for example, the readings of the speed or the altitude transmitted by a sensor mounted on a vehicle, they keep on transmitting the data periodically. So, the fixed sized messages are transmitted periodically and these are the kind of data typically the hard real time applications generally delayed.

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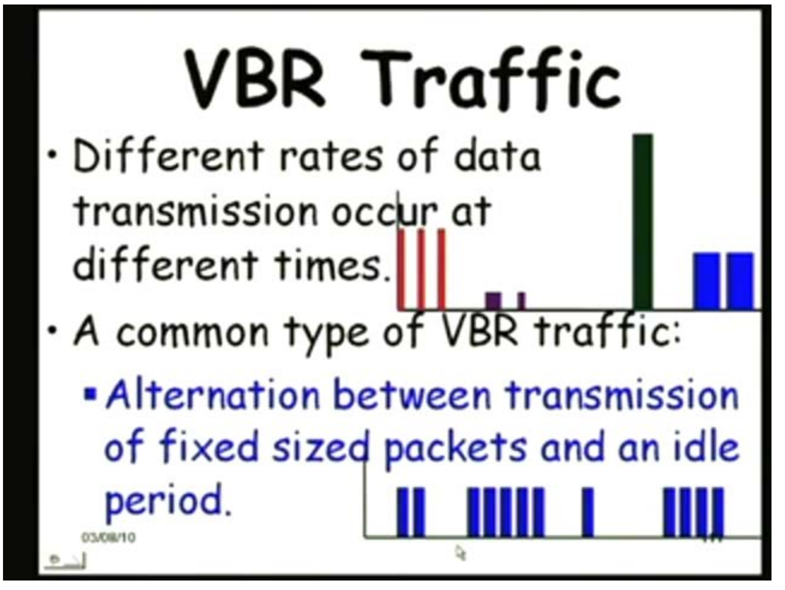

The variable bit rate traffic; here different rates of data transmission occur at different times. Let me not say transmission, we can say that data generation at the source transmission as different meaning. So, here the data generated by the source at certain time is let us say some 10 kilobyte of data generated per millisecond here, but at certain other time, it is 100 megabyte of data or something generated per millisecond.

There are various categories of the VBR traffic, but one type of VBR traffic, that is, that occurs frequently is alternation between transmissions of fixed sized packets and idle period. So, let us say 10 kilobytes of data get transmitted every now and then, but each time, each time the data is generated its only 10 kilobytes that needs to be transmitted, but we do not know, it might occur after 100 millisecond or may be the next data would come after 500 millisecond, but each time that packet size is the same. Sir, under what condition this kind of data generation? We will just look at an example, where such data generation occurs. This is just an illustration of the data, the data size is the same just see here, they keep 2 packets got 2, let us say data samples got generated here, there are some silence here, again some data samples got generated again silence and so on.

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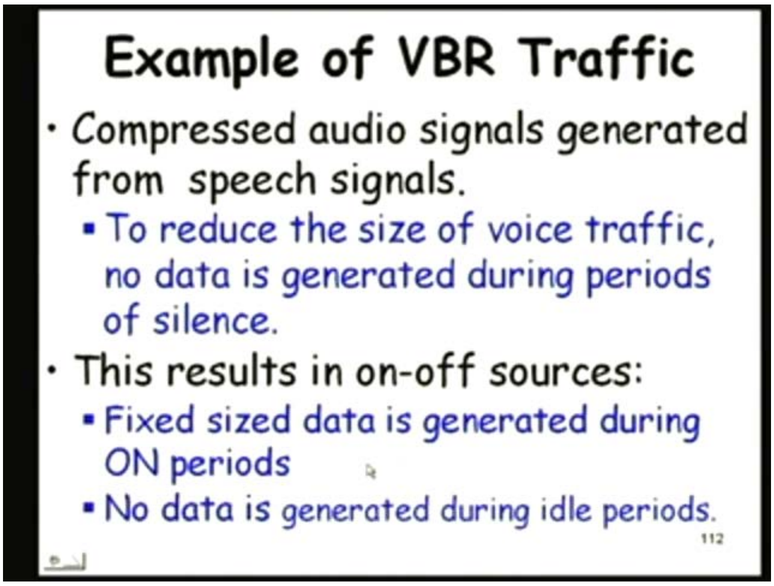

So, an example of a data of this type is the compressed audio signals generated from speech signals. Normally, when the voice traffic is sent on a network, you do not send any data during the periods of silence and possibly that is one of the reasons, why a transmission on a packet switch network is more efficient, then when you do transmit data on a connection based network.

So, establish let us say a telephone line, the traditional telephone line, where you just establish a connection and whether you are talking or not the connection is held up for you So, here data is not generated in this packet switch network, when there is a silence.

So, this would result in a on off source, where fixed sized data is generated during the on periods; when you are actually talking and when you are pausing, no data is generated and it does not, it is not known how long you will pause; sometimes, you pause for you milliseconds sometime up to few seconds.

So, since, the today we are getting the time is getting over for today. We will stop here and we will continue from this point in the next class. Thank you.