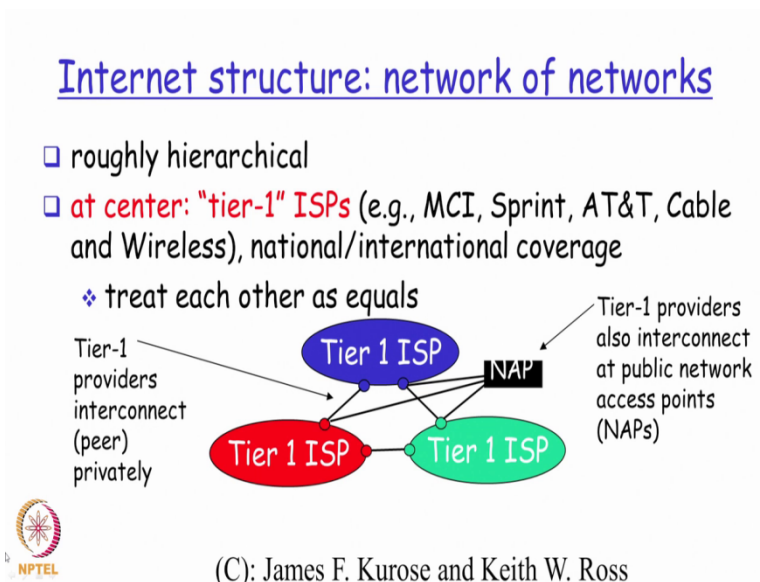


Information Security 3
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Module 49
Structure of ISP & Packet delay

So this module we will basically talk about ISPs, so ISP stands for internet service provider, how they are typically structured in a very hierarchical model and also talk about that the different types of delays that a network packet can possibly incur when it goes from a source mission to a destination mission travelling across the different ISPs and what kind of care we need to take in terms of trouble shooting any kinds of network problems that we might have.

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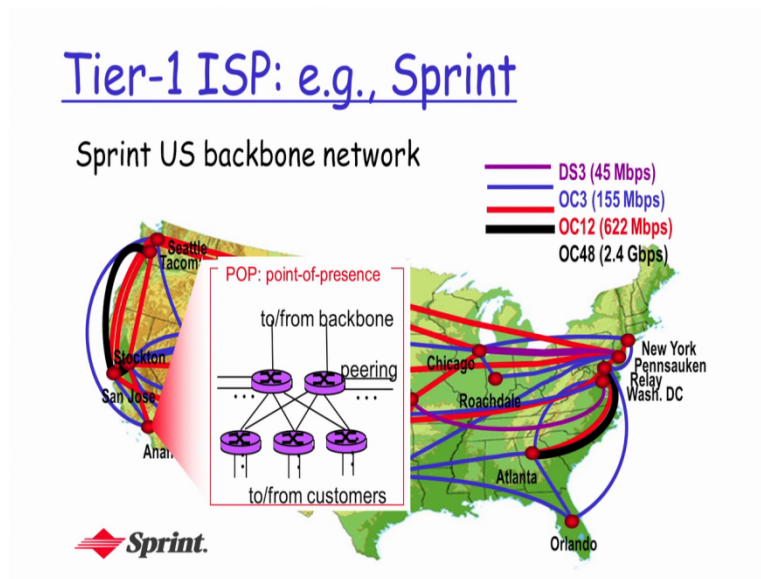
So the internet structure which is basically talking about when you are dealing with an ISP is a sort of a hierarchical in nature where I have a different tiers of ISPs so at the center we would typically have tier 1 ISPs which actually has invested in the physical laying of different cables typically fiber optic cables across the ocean.

So some of the very commonly used tier one ISPs in the world are some some organizations like Mci At&t sprint and so on where I could have the tier 1 ISPs having done the physical infrastructure laying up and the tier 1 ISPs will all be connected to each other over what is called

as a network access point, so these network access points are public and these will serve as a sort of a connecting point between the different tier1 ISPs right.

So the the connectivity between the two tier 1 ISPs will be provided by two network access points one on each of those sides and these network access points will be very critical for these two ISPs at the same tier 1 level to be communicating with each other.

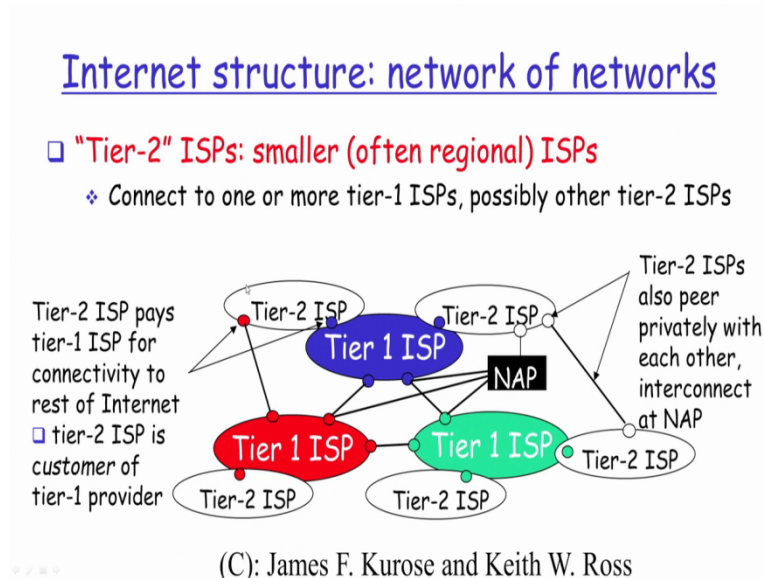
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So if I really look at an example of a sprint as a tier 1 ISP they have a huge backbone across the US for example if you see there are different types of links ds3 oc3 oc12 and so on, each type of a link is actually having different capability different capacity bandwidth capacity that it could actually be used for the transmission part, right?

So this is just an example of 1 tier on ISP of how they have actually physically laid cables either under land or under the oceans to being able to have the entire internet connected with each other.

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Now tier 2 ISPs who could who would typically be the regional ISPs they will be actually connected to one or more tier1 ISPs so they have connectivity to more than one tier1 ISP typically to have a fault all rent setup in just in case the connectivity to 1 tier 1 ISP goes down they they are not actually left stranded without being able to reach the internet right.

So the tier2 ISP will basically be paying the corresponding amount that is been built by the tier1 ISP for the amount of data that they have actually sent and received according to the policy right,

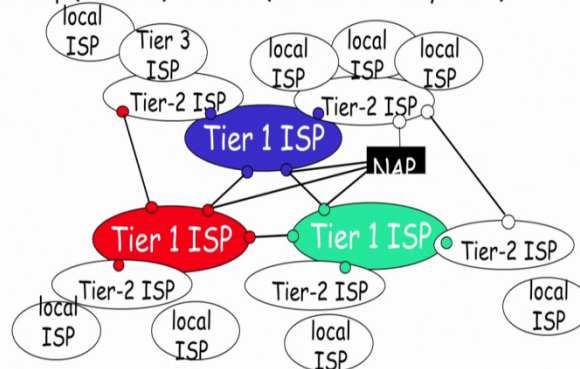
So I have a tier 1 ISP, the tier1 ISPs are all connected over each other using the network access points that we talked about then at the next level in the hierarchy have a tier2 ISP who can be typically classified as a regional ISP and these tier2 ISPs are connected to tier1 typically each tier2 ISP will have connection to more than 1 tier ISP as we discussed for fault all rent purposes as well as for possibly load balancing.

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Internet structure: network of networks

□ "Tier-3" ISPs and local ISPs

❖ last hop ("access") network (closest to end systems)

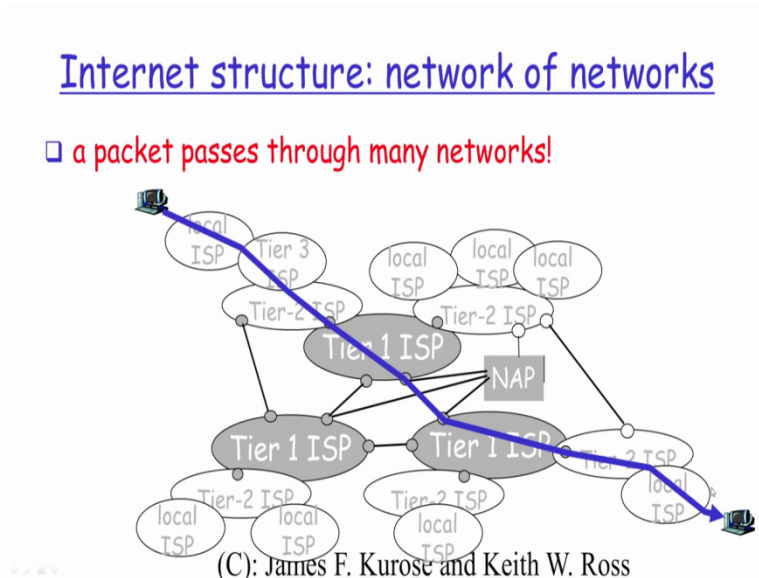


Then the tier 2 ISPs Will then be connected with tier 3 ISPs and the local ISPs who will be basically providing the the the last byte connectivity to the end users so the last byte connectivity here to the end user as far as the local ISPs is concerned could be an organization or it could be a home user likewise they will be the ones who will be dealing with the end user

So an example of a tier3 ISP could be national carrier like a bsnl or an mtnl or it could be some sort of a service providers like the airtel and the reliance and the tata's of these world right. So in strict sense if you find you have a hierarchy where there is always a sub leasing of capacity, so you find here the tier1 ISP as leased the capacity to tier2 ISP, the tier2 ISP has reached the capacity the sub lease the capacity to tier3 ISP and so on

Till they the lasts level of ISP in the range hierarchy range actually gives you a policy wherein they believe saying that for using this much of data plan that you'll have to subscribe to and then pay every month right. So this is basically how the entire internet is typically structured across the globe and this is not something very specific to one part of the globe or whatever it is but this is typically the hierarchy that is actually followed for the internet connectivity across the entire globe.

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So when you have a host mission connected to one local ISP in India and you want to basically talk to a host mission that is actually connected to a remote ISP across somewhere in the internet right, a packet which is actually originating from here from this host has to actually pass through so many different networks, right?

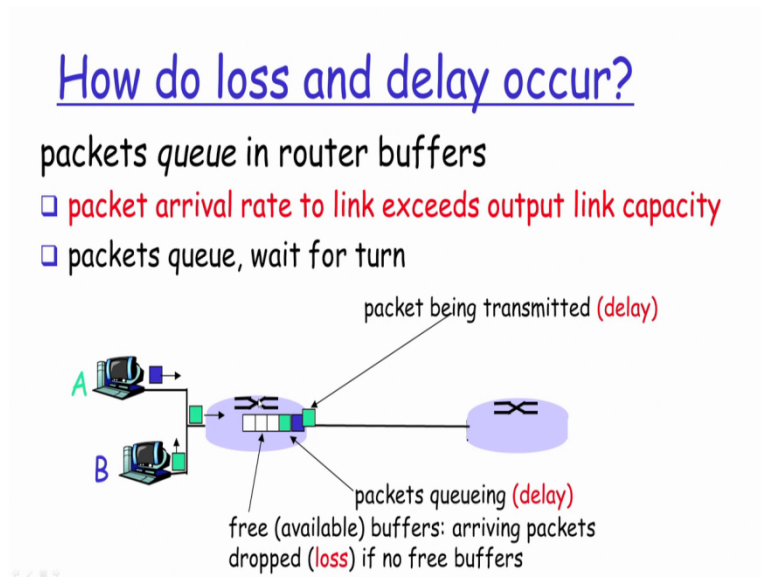
Till it reaches the final destination mission and if you see here as part of this route the packet is basically taking you find that it actually travels us all the different types of ISPs typically speaking unless and until there is a possibility that some of the traffic that you are originating needs to be serviced possibly by host connected to your local ISP itself.

So that is the possibility but predominantly you will find that most of the packets that are originating from a host needs to travel across different levels of ISPs different hierarchical levels of ISP before it reaches a final destination wherever it is intended to be right. So giving you an example sitting on this front of this pc you are actually running a network application like a browser trying to go to

Let say a google.com webpage and typed www.google.com on this pc the packets that are going to be generated by the browser is actually going to go across all these different levels of ISPs because typically if you are if you are accessing this particular page for the first time and it is not

cache and all that your packets will be going across the different levels of ISPs since the google.com web server will be on a remote host which will be reachable only across the the the different levels of ISP only after the packet travels across the the different tiers of ISP.

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So because of the fact that my packet is actually going to go across the different levels of ISPs over so many devices my packets are going to be incurring certain delays and also possibly losses, now why do these delays occur and what leads to a final loss of my packet right. The main reason is that the packets basically have got to get queued in my router buffers, right?

So when a packet is actually coming in the my packet is actually going to get queued in the router device because there are multiple hosts in my network and all these hosts are actually generating the data and the the generated data is actually going to be going into the router for it to be sent on the appropriate path towards the destination,

so because the router will be having other packets which has reached it earlier then the current packet I will always have a queue that my router will need to maintain right so the reason why the packets get queued is router buffers is basically because of the fact my router device is currently busy servicing the packets that has reached before this existing packet because of which I have to queue the incoming packet right.

So the packets basically queue in this router device before they are taken up for processing and why this queue is being maintained is also because of the fact that my arrival rate into my router device could actually be greater than whatever the link capacity that I have here.

so the the the lesser the link capacity that I have the lesser will be these peer which we will be able to carry because of which I will need more time to push one packet outside onto my link because I need more time to push one packet outside my link I am going to have the the packets that are coming in after this particular packet that is getting pushed out all needing to be stored in the cube right,

Waiting for their turn to come to the head of the cube and then started to be getting processed in terms of getting pushed into the cube right, so this is one reason why the packets basically incur certain amount of delay from the time it gets originated at the host till the time it basically goes to the destination right.

So that is basically the transmission delay that we are talking of here, so there is going to be amount of time that I am going to spend for pushing the first bit of the packet to the last bit of the packet and that is going to be taking time to in my overall delay component and that delay we are basically going to call it as a transmission delay right.

Now we just talked about the queuing part of it because there is already a packet that is actually getting processed so the packet which is incoming into my router device will need to be in the cube waiting for its term turn to come because of which there will be certain amount of time the packet will spending in the cube and that part of the delay component is we are going to refer to as the queuing delay.

Now, when I don't have many free space many free buffers available in my router device why will I not have free space available in my router device because I have allocated originally a certain amount of memory as my buffer because of the fact that I have too many packets coming in and also maybe because of the fact that my output link is not that very fast,

I have too many packets that is actually getting queued because of which any packet that is coming in will not be able to get stored in the pre allocated memory buffer because the originally

allocated memory buffer has actually been used for storing all the packets that has reached me before and since my output link is also not very fast,

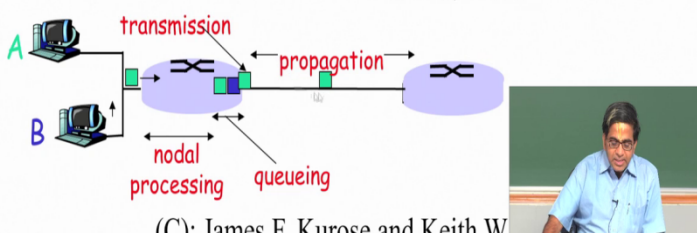
I am not able to push those packets out as quickly as required because of which the packets have got queued in my buffer because the packets have got queued in the buffer any incoming packet that is coming in newly from my connected host devices they will have to be just simply dropped.

So what we mean by drop is that I will just discard the packet and that is basically why you are referring to that particular packet being lost in my network right. So when will the packets get lost to the network whenever I don't have a free buffers in my router device, right?

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Four sources of packet delay

| | |
|--|---|
| <p>❑ 1. nodal processing:</p> <ul style="list-style-type: none">❖ check bit errors❖ determine output link | <p>❑ 2. queueing</p> <ul style="list-style-type: none">❖ time waiting at output link for transmission❖ depends on congestion level of router |
|--|---|



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So other than this transmission delay and the queuing delay you also have a processing delay where the every packet that is coming inside the router device will be processed so some types of processing that will be typically done is it will basically check whether the the packet is having any kind of a bit errors, whether the packet is still a valid packet, right?

So there are so many checks that needs to be done by the processor on the router device, so all these kind of computing time is basically classified as a processing delay so and apart from the

queuing delay and the transmission delay that we talked about there is also a propagation delay, now what is this propagation delay.

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Delay in packet-switched networks

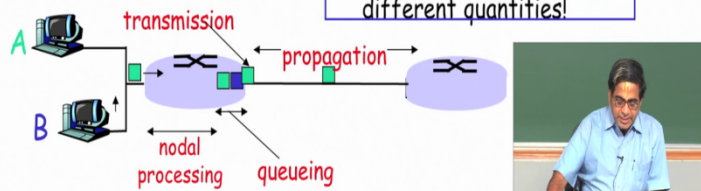
3. Transmission delay:

- R = link bandwidth (bps)
- L = packet length (bits)
- time to send bits into link = L/R

4. Propagation delay:

- d = length of physical link
- s = propagation speed in medium ($\sim 2 \times 10^8$ m/sec)
- propagation delay = d/s

Note: s and R are very different quantities!



The diagram illustrates the delay components in a packet-switched network. It shows two source nodes, A and B, connected to a central switch. The switch is connected to a destination switch. The delay components are labeled: 'transmission' (time to put bits on the link), 'propagation' (time for bits to travel across the link), 'nodal processing' (time spent at the switch), and 'queueing' (time spent waiting for service at the switch). A small inset video shows a man in a blue shirt speaking in front of a green chalkboard.

So the time that I basically take for the packet to be sent from the source end of the link to the destination end of the link right. So the time that it takes for the packet the entire packet size the bit the entire set of bits of the packet to move across the link to reach the destination is what we are referring to as a propagation delay, right?

So please understand the difference between the transmission delay and the propagation delay, the transmission delay is basically the time that is sent, the time that I incur to sent all the bits into my link on the source side right, so from the time I inject the first bit to the last bit of my packet is the transmission delay, right?


Now I have injected all the bits into the into the physical link these bits are going to take time to propagate across to reach the destination right. So the propagation delay is basically the time that it takes for all the bits from the source end of the link to reach to the destination is the propagation delay component, right?

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Nodal delay

$$d_{\text{nodal}} = d_{\text{proc}} + d_{\text{queue}} + d_{\text{trans}} + d_{\text{prop}}$$

- d_{proc} = processing delay
 - ❖ typically a few microsecs or less
- d_{queue} = queuing delay
 - ❖ depends on congestion
- d_{trans} = transmission delay
 - ❖ = L/R , significant for low-speed links
- d_{prop} = propagation delay
 - ❖ a few microsecs to hundreds of msec



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So in a sense if you look at it, we defined something called as a nodal delay which is summation of my processing delay, my queuing delay, my transmission delay and my propagation delay right. So each of these delay components can be either a negligible or it could be substantial right,

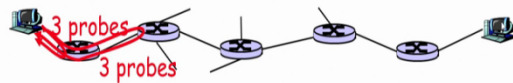
So when will it be negligible when I can sort of discarded so for example if my network is very least loaded right, my queuing delay will possibly be very negligible because my router buffers are all going to be pretty much free if not completely free at least the maximum component of it is free and because of the fact that my router is now going to have only packets as compare to ten packets the amount of time that I spend in the queue inside my router for coming to the end of the queue is going to be much lesser right,

So similarly if my if I am going to have my output link as a very high bandwidth link then my transmission delay will be very very less, my propagation delay might also be very less depending on various factors so accordingly we introduce the component called as a nodal delay which is basically the summation of my processing delay, the queuing delay, the transmission delay and the propagation delay, right?

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"Real" Internet delays and routes

- What do "real" Internet delay & loss look like?
- **Traceroute program**: provides delay measurement from source to router along end-end Internet path towards destination. For all i :
 - ❖ sends three packets that will reach router i on path towards destination
 - ❖ router i will return packets to sender
 - ❖ sender times interval between transmission and reply.



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So that's basically how I really try to understand the different types of delays that my packet can go through from the source mission to the destination mission. So in reality how do I measure my delay time right so I need to know if my if I am basically trying to reach a server like google.com server available on some remote mission in the my entire inter network what time am I going to be getting spend for my package to go from my source mission to that destination mission and for the response to comeback to be, right?

So this is basically what we call as RTT, so what is RTT ? rtt stands for round trIP time so my packet from my source mission as to reach the destination mission and I have to get the response back of that packet back to me and that is when I will consider the transmission of the packet to be successful right,

So the time that is taken for the packet to reach the source from the source to the destination and the response coming back from the destination mission to the source, the total time put together is basically what we refer to as an Rtt duration so the round trIP time duration and for me to effectively measure the the delay that my packets are going to incur for reaching a particular destination there is a program called as a tray suit program,

So it's a command that is actually available pretty much in all operating systems today by default so it basically provides me a delay measurement from the source to the router along the entire internet path that I have to travel for reaching the destination right.

So how does it basically work so it send three different probes from the source mission to the destination and every time when I when I hit the router on the path, the router will basically return the packets back to the sender and I basically calculate the the time that I that that time difference between the time that I had send the packet to the time I had received the response back right,

So this times difference gives me the delay that my packets are encountering from going from my source mission to the first hop in the path towards the destination right. So likewise the next time I do I basically try to reach the second hop in my destination and then I find out the time that I take it the response back from the second hop.

And the time difference between the two from the time I sent it and the time I got the response now will give me the total delay from going from my source mission to my second hop on a network likewise I will keep calculating the time difference till I reach the final destination and I will find out what is the total delay that it is getting encountered on the complete internet path, right?

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"Real" Internet delays and routes

traceroute: gaia.cs.umass.edu to www.eurecom.fr

Three delay measurements from
gaia.cs.umass.edu to cs-gw.cs.umass.edu

```
1  cs-gw (128.119.240.254) 1 ms 1 ms 2 ms
2  border1-rt-fa5-1-0.gw.umass.edu (128.119.3.145) 1 ms 1 ms 2 ms
3  cht-vbns.gw.umass.edu (128.119.3.130) 6 ms 5 ms 5 ms
4  jn1-at1-0-0-19.wor.vbns.net (204.147.132.129) 16 ms 11 ms 13 ms
5  jn1-so7-0-0-0.wae.vbns.net (204.147.136.136) 21 ms 18 ms 18 ms
6  abilene-vbns.abilene.ucaid.edu (198.32.11.9) 22 ms 18 ms 22 ms
7  nycm-wash.abilene.ucaid.edu (198.32.8.46) 22 ms 22 ms 22 ms
8  62.40.103.253 (62.40.103.253) 104 ms 109 ms 106 ms
9  de2-1.de1.de.geant.net (62.40.96.129) 109 ms 102 ms 104 ms
10 de.fr1.fr.geant.net (62.40.96.50) 113 ms 121 ms 114 ms
11 renater-gw.fr1.fr.geant.net (62.40.103.54) 112 ms 114 ms 112 ms
12 nio-n2.cssi.renater.fr (193.51.206.13) 111 ms 114 ms 116 ms
13 nice.cssi.renater.fr (195.220.98.102) 123 ms 125 ms 124 ms
14 r3t2-nice.cssi.renater.fr (195.220.98.110) 126 ms 126 ms 124 ms
15 eurecom-valbonne.r3t2.ft.net (193.48.50.54) 135 ms 128 ms 133 ms
16 194.214.211.25 (194.214.211.25) 126 ms 128 ms 126 ms
17 ***
18 ***
19 fantasia.eurecom.fr (193.55.113.142) 132 ms 128 ms 136 ms
```

trans-oceanic link

* means no response (probe lost, router not replying)

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So if I basically do a traceroute this will be a typical example that will be a that that could come as an output right so I have the three delay measurements here one each for each of the probes that has been sent and if you find that it tells me the first hop is this particular mission and their IP address is this particular IP address.

The second hop is this particular mission and the IP address of the second hop is this much and then it tells me the total time that it is being taken for reaching that particular hop, so likewise I keep getting different hops and then I finally reach the destination right because I am trying to find the route from this particular mission that is my source mission to this particular mission that is my destination mission.

So the destination mission finally is reached here and it tells me here that for the first probe I took 132 milliseconds, for the second one I took 128 and third one I took 136 so I could possibly take get an average of what is the approximate time that will be incurred for my packets to go from the source to the destination.

So if you see the difference between the two the seventh and the eighth hop you find there is a drastic jump because till 7th hop I was only incurring a delay of around 22 milliseconds approximately from my source mission but whereas for the 8th hop I have now incurred a 104 milliseconds for my as my delay from the source mission right for my packets.

Now one possibility of this particular huge jump from 22 milliseconds to around 100 milliseconds could be because that from the 7th hop to the 8th hop I could be possibly traversing a transoceanic link, right? So when I traverse a transoceanic link possibly the propagation delay component of my entire nodal delay that we talked about could be much much higher because I have to propagate across a huge distance, right?

Which is actually increased my total nodal delay that we are talking of here or other possibility is instead of it being a transoceanic link I might have a very very low bandwidth link from the 7th hop to the 8th hop because of which my transmission delay is possibly very high in this scenario because of the fact that my transmission delay is very high possibly my end to end delay from the source mission to the 8th hop as possibly increase, right?

So this basically gives you an idea of the different types of different amounts of delay that is getting encountered in the path of the source to the destination and through this details that application command is providing to you, you could possibly try to identify where are the possible bottle necks in your network topology or in the network path for a particular destination and address it right.

So if for example if it was basically because of the transmission delay here, because of a low bandwidth capacity that you have from the 7th hop to the 8th hop, you could possibly talk to the ISP right and try to see if you can actually have a higher bandwidth allocated between those two hops so that your end to end delay from the source mission to that particular target mission is drastically reduced right or any other kind of a character action that is required you'll be able to do specifically for that particular path of your entire path from source towards the destination right.

So that's basically the beauty of the output that this command provides to you in order to do a trouble shooting of where exactly is the potential performance problem of the of your entire network connectivity path to a particular destination

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Packet loss

- ❑ queue (aka buffer) preceding link in buffer has finite capacity
- ❑ when packet arrives to full queue, packet is dropped (aka lost)
- ❑ lost packet may be retransmitted by previous node, by source end system, or not retransmitted at all

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So as we were talking the packet loss could typically happen because the queue that I have actually allocated internally that is basically my memory buffer inside my router device has only

a finite capacity of memory that is available so obviously it is not possible to have an infinite amount of memory allocated just for my memory buffer alone,

So essentially when the packet is arriving and I already have a queue that has that is full with all the packets that are arrived prior to this new packet inside my queue then I don't have any other option other than to have the packet drop so the moment packet is dropped that is basically what is refer to as a lost packet at my higher levels right, So a lost packet has to be necessarily retransmitted by the previous node or the source node if my application is actually running over something like a reliable service that we talked of in our earlier module right,

Or if it is running over an under liable service the packet will never be retransmitted at all and then it basically becomes a responsibility of application to handle scenarios where where the packet have been got have have got lost in the path towards a destination and then there has to be a correction mechanism handled as part of my application level protocol,

So in this module we have basically seen the different types of ISPs that are potentially a packet needs to be traversing through as part of this traversal across so many different tiers of ISP, what kind of delays could encounter and how do you calculate those delays and measure them and also give you an idea of how you can really troubleshoot and sort of fix the problems that could be there in some very specific sub path in your entire path from your source to the destination.

Thank you.