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Computer Networks
Homework #8

Chapter 5:

53)

The packet loss rate can tell the server how often packets are being missed, and the jitter data can tell the server how out of sync it is with the client.
I would think these could be used for a variety of things, but the first thing that comes to mind is that the server could use the packet loss rate and jitter rate to determine which packets to send to the client again. The packet loss rate can tell the server how often packets are being missed, and if there's congestion, and the jitter data can tell the server how out of sync it is with the client.

Chapter 6:

10)

a) For fair queuing, you simply look at the time that each packet would arrive based on the fact that each flow has its own queue. The order would be:

1, 7, 8, 5, 2, 3, 6, 4.

b) For weighted queuing when flow 2 has a weight of 4, flow 2 would send 4 times as many bits with each transmission. The order would be:

5, 6, 1, 7, 8, 2, 3, 4.

14)

a) The router would send each packet at the following clock times (just add the time from the last packet sent, with queues going in the order of A,B, and C):

i.) Flow A: 1, 6, 18, 35, 58, 85, 117.
ii.) Flow B: 3, 12, 26, 46, 70, 100.
iii.) Flow C: 4, 14, 29, 51, 76, 107, 125.

b) Here, flow C has twice the capacity of the other ones, so it would send two packets every time it has a turn.

i.) Flow A: 1, 8, 26, 53, 79, 100, 125.
ii.) Flow B: 3, 14, 34, 64, 91, 115.
iii.) Flow C: 4, 6, 17, 22, 40, 47, 72.

25)

In order to establish that no slow start was used, look at the number of packets sent to the router and compare that number to the number of ACKs sent to the client. If slow start is being used, the client will increase the number of packets sent each time. Thus, if the number of packets sent is greater than the number of ACKs sent back, then slow start is being used. After a timeout, if slow start is not being used still, the client will not continue to increase the number of packets sent. To detect this, look at the number of packets sent immediately after timeout and compare that number to the number of packets sent after a few ACKs have been sent back.

29)

a) When we start a connection, we will send 3 segments and then will drop one. We will keep losing packets, so TCP will assume this is due to congestion. The window size will be increased, and we WILL enter the linear increase phase of congestion avoidance.

b) No, it would not.

c) It would be feasible; TCP would simply have to increase the window size by a larger amount if it was discovered that a segment was lost due to congestion rather than a poor link.

34)

a) If MaxP = 0.01, then p is half of that, or 0.005. The drop probability will be (0.005)/(1-0.005) = **0.00502** if count =1. If count is 100, then it will be **0.00005**.

b) Using excel and explicitly calculating the value I was able to obtain the probability as **0.748743719.**

38)

There is a conflict between tolerating burstiness and congestion control because allowing burstiness implies that you are allowing extra traffic on to a particular link, which is fundamentally what congestion control aims to prevent. If you tolerate influxes of traffic, or burstiness, the quality of service will go down and the network will be congested. If you aim to control congestion, you have to limit the amount of burstiness that's tolerated.