CS-224

(Computer Networks)

Assignment-4

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Solution 1:

Requirements needed for the newly designed TCP variant:

(a) It is purely an end-to-end congestion control protocol, that is, it does not require any special information from routers or explicit information about other TCP flows in order to perform congestion control.

(b) It behaves like TCP-Vegas if all other competing flows on its network path employ TCP-Vegas. Let us assume that all TCP flows have the same RTT on the network path.

(c) It behaves like TCP-Reno if there are some competing flows on its network path that employ TCP-Reno.

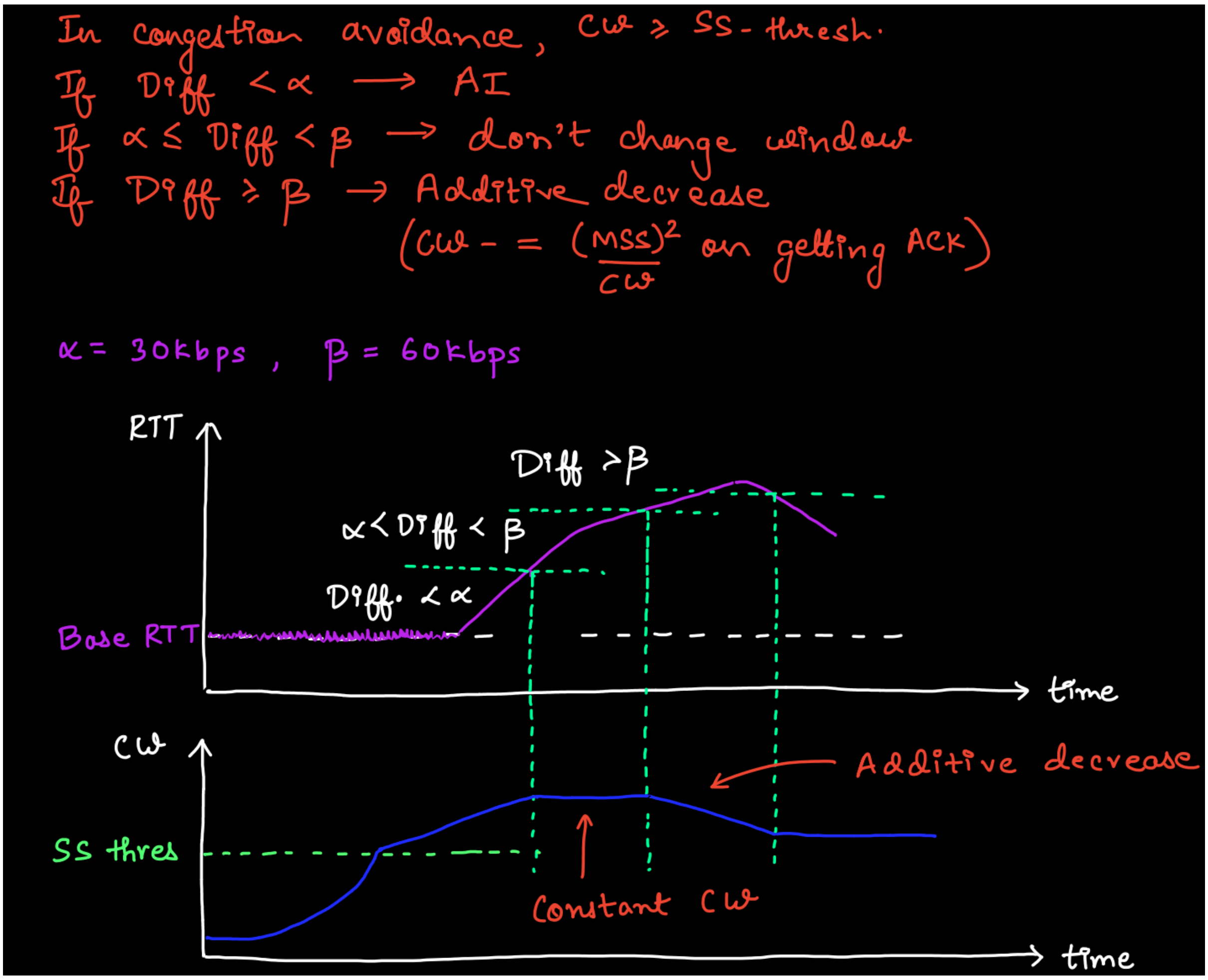
(d) It adjusts its Congestion_Window based on inference of packet losses and/or queuing delays in the network.

My proposal for such a TCP-variant:

TCP_New: It behaves like TCP-Vegas whensoever all of the other flows in its network are employing TCP-Vegas. If not, then we switch it to TCP-Reno.

Now, as we know that packet loss doesn't occur if all the other flows are using TCP-Vegas for congestion control, therefore if a packet loss is detected, it simply implies that someone among the flows has switched to TCP-Reno. This is exactly when we switch our protocol to TCP-Reno. In this manner, we ensure that criterions b) and c) are met. Also, since we are employing a mix of TCP-Reno and TCP-Vegas in a tactful manner with the help of packet-loss detection, it is purely an end-to-end congestion control protocol (working without the aid of any explicit information from the routers).

Now, congestion window is adjusted following the rules just as in TCP-Vegas



but with an additional feature, that is:
During packet loss, `CW = 1 MSS` and `SS_Threshold = CW/2`, thereby adjusting its `Congestion_Window` based on inference of packet losses and/or queuing delays in the network.

Solution 2:

Sol 2] To check for "RTT-fairness" = (do two TCP flows give same bandwidth for different RTTs) we compute and compare the total amt. of data transmitted by the two TCP flows b/w consecutive "loss" events.

For flow 1:

$$\frac{dw_1(t)}{dt} = \frac{1}{T_1} \Rightarrow \int_0^t dw_1(t) = \frac{1}{T_1}\int_0^t dt$$

$\because$ given that $w_1(0) = 0$, $\therefore$ $w_1(t) - w_1(0)^{\to 0} = t/T_1$

$$\therefore \boxed{\text{Bitrate}\Big|_{flow_1} = \frac{w_1(t)}{T_1} = t/T_1^2}.$$

Similarly, for flow 2:

$$\frac{dw_2(t)}{dt} = \frac{1}{T_2} \Rightarrow \int_0^t dw_2(t) = \frac{1}{T_2}\int_0^t dt$$

$\because$ given $w_2(0) = 0$, $\therefore$ $w_2(t) - w_2(0)^{\to 0} = t/T_2$

$$\therefore \boxed{\text{Bitrate}\Big|_{flow_2} = \frac{w_2(t)}{T_2} = t/T_2^2}.$$

Now, as given in Qn,
both flows facing "loss" at time 't_L' $\Leftrightarrow [w_1(t_L)/T_1] + [w_2(t_L)/T_2] = C$ (bits/sec)

$$\therefore \Leftrightarrow C = t_L\left(\frac{1}{T_1^2} + \frac{1}{T_2^2}\right).$$

Also, given that the RTTs T_i of both the TCP flows are constant,

$$\therefore \boxed{t_L = \frac{C \cdot T_1^2 T_2^2}{(T_1^2 + T_2^2)}} \longrightarrow \text{CONSTANT} ✓$$

Now, computing the total data loss up until t_L of both the flows to compare:

(TDL_1): Total data loss$\Big|_{TCP_1} = \int_0^{t_L} \frac{t}{T_1^2}\,dt = t_L^2/2T_1^2$. $T_1 \to (RTT_1)^2$; $t_L \to$ CONSTANT :)

(TDL_2): Total data loss$\Big|_{TCP_2} = \int_0^{t_L} \frac{t}{T_2^2}\,dt = t_L^2/2T_2^2$ $T_2 \to (RTT_2)^2$

From the derivation, clearly, $\boxed{TDL_i \propto \frac{1}{RTT_i^2}}$

(Total data loss is inversely proportional to the square of RTT) and thus,

TCP Reno is NOT 'fair' as the throughputs of both are not the same for different RTTs.

Observations:

$RTT_1 < RTT_2 \Rightarrow TDL_1 > TDL_2$

$RTT_1 > RTT_2 \Rightarrow TDL_1 < TDL_2$

$RTT_1 = RTT_2 \Rightarrow TDL_1 = TDL_2$