Introduction to Computer Networks - Final Exam Prof. Wanjiun Liao June 17 2003 Virtual Creut 1. (15%) Consider the following design problem concerning implementation of virtual-circuit service. If virtual circuits are used internal to the subnet, each data packet must have a 3-byte header and each router must tie up 8 bytes of storage for circuit-identification. If datagrams are used internally, 12-byte headers are needed but no router table space is required. Transmission capacity costs 1 cent per-100 bytes, per hop. Very fast router memory can be purchases for 1 cent per byte and is depreciated over two years, assuming a 40-hour business week. The statistically average session runs for 1000 sec., in which time 200 packets are transmitted. There is mean packet requires four hops. Which implementation is shown a statistically average session runs for 1000 sec., in which time 200 packets are transmitted. mean packet requires four hops. Which implementation is cheaper, and by how much? (10%) A router is blasting out IP packers whose total length (data plus header) is 1024 bytes Assuming that packets live for 10 sed, which is the maximum line speed the router can operate at without danger of cyclying through the IP datagram ID number space? (10%) Consider the network shown below, and assume each node initially knows the costs to each of its neighbors. Consider the distance vector algorithm and show the distance table entries 1==44-7163. 15 at node E. (20%) Suppose an IP implementation adheres literally to the following packet P destined to IP address(D: if (Ethernet address for D is in ARP cache) send P: MA else (

(a) If the IP layer receives a burst of packets destined to D, how might this algorithm waster IF Disn't in AR Carne, & All pocket will send A resources unnecessarily?

(b) Suppose we simply drop P after sending out a query, when cache lookup fails. How would givery this behave?

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[ B= [ ( (b) 7 (b2) - ] = AB

send out an ARP query for D;

put P into a queue until the response comes back:-

5/(30%) TCP

(a) (5%) Fast retransmit states that the TCP sender starts a retransmission as soon as three duplicate ACKs are received. Why do you think the TCP designers chose not to perform a fast retransmit after the first duplicate ACK for a segment is received? They may just received?

(b) (15%) TCP estimates RTT based on exponential weighted moving average (EWMA).

Suppose that  $\alpha = 0.1$ . Let SampleRTT<sub>1</sub> be the most recent sample RTT, SampleRTT<sub>2</sub> be the next most recent sample RTT, and so on. For a given TCP connection, suppose n ACKs have been returned with the corresponding sample RTTs. Express EstimatedRTT in terms of these n sample RTTs. Then let n approaches infinity. Comment on why this averaging n = 0.1 and n = 0.1 are n = 0.1 are n = 0.1 and n = 0.1 are n = 0.1 are n = 0.1 and n = 0.1 are n = 0.1 are n = 0.1 are n = 0.1 and n = 0.1 are n = 0.1 are n = 0.1 are n = 0.1 and n = 0.1 are n = 0.1 are n = 0.1 and n = 0.1 are n = 0.1 are

procedure is called an exponential moving average.

(c) (10%) We have discussed how TCP estimates RTT and timeous The way of EWMA is used only when an ACK is received for an original segment. Whenever the timeout event occurs and no ACK is received. TCP retransmits the not yet acked segment with the (smallest sequence number) But each time TCP retransmits, it sets the next timeout value to

twice the previous value, instead of using the EWMA method. Why do you think TCP avoids measuring the SampleRTT for retransmitted segments?

6. (15%) Consider the Key distribution Center (KDC) and the CA servers. Suppose a KDC goes down. What is the impact on the ability of parties to communicate securely; that is, who can, and cannot, communicate? Justify your answer. Suppose now that a CA goes down. What is the impact of this failure

# Solution to Final Exam

1. 15 points

2 years = 2\*52 40\*3600 / 1000 = 14976 sessions

VC: Storage: 8\*1cent = 40 cents in 2 years;

transmission: 14976 \* 3 \* 200 \* 4 / 10^6 = 35.9424 cents in 2 years;

Datagram: transmission: 14976 \* 12 \* 200 \* 4 / 10^6 = 143.7696

VC is cheaper in two years by 67.8272 cents.

(2) If the bit rate of the line is b, the number of packets/sec that the router can emit is b/8192, so the number of seconds it takes to emit a packet is 8192/b. To put out 65,536 packets takes (2^29) /b sec. Equating this to the maximum packet lifetime, we get (2^29) /b = 10. Then, b is about 53,687,091 bps.

8192 bit x 2 10

10 points

		VIA	
	В	С	D
Α	6	4	13
В	5	5	14
С	8	2	11

D	9	3	10	

### 4. 10 points

- (a) If multiple packets after the first arrive at the IP layer for outbound delivery, but before the first ARP response comes back, then we send out multiple unnecessary "ARP packets." Not only do these consume bandwidth, but, because they are broadcast, they interrupt every host and propagate across bridges.
- (b) This might, among other things, lead to frequent and excessive packet loss at the beginning of new connections.

### 5. TCP (Chapter 3)

(a) 5 points

Suppose packets n, n+1, and n+2 are sent, and that packet n is received and ACKed. If packets n+1 and n+2 are reordered along the end-to-end-path (i.e., are received in the order n+2, n+1) then the receipt of packet n+2 will generate a duplicate ack for n and would trigger a retransmission under a policy of waiting only for second duplicateACK for retransmission. By waiting for a triple duplicate ACK, it must be the case that two packets after packet n are correctly received, while n+1 was not received. The designers of the triple duplicate ACK scheme probably felt that waiting for two ackets (rather than 1) was the right tradeoff between triggering a quick retransmission when needed, but not retransmitting prematurely in the face of packet reordering.

## (b) 15 points

Please note that the definition of EstimateRTT, is the estimated round trip time after the most recent sample is received if sample 1,...,n is considered.

 $EstimateRTT_1 = SampleRTT_1$ 

 $EstimateRTT_2 = \alpha \times SampleRTT_1 + (1 - \alpha) \times SampleRTT_2$ 

 $EstimateRTT_3 = \alpha \times SampleRTT_1 + (1 - \alpha) \times [\alpha \times SampleRTT_2 + (1 - \alpha) \times SampleRTT_3]$ 

=  $\alpha \times SampleRTT_1 + (1 - \alpha) \times \alpha \times SampleRTT_2 + (1 - \alpha)^2 \times SampleRTT_3$ 

 $EstimateRTT_n = \alpha \sum_{i=1}^{n} (1 - \alpha)^{i-1} \times SampleRTT_i + (1 - \alpha)^n \times SampleRTT_n$ 

 $EstimateRTT_{m} = \alpha \sum_{i=1}^{\infty} (1 - \alpha)^{i-1} \times SampleRTT_{i}$ 

(c) 10 points



If EWMA is used, it is difficult to measure the estimated round trip time since it can't make sure the ACK belongs to the original segment or the retransmitted segment.

#### Chapter 7, #11, p 668

If the KDC goes down, no one can communicate securely, as a first step in communication (see Figure 7.19) is to get the one-time session key from the KDC. If the CA goes down, then as along as the CA's public key is known, one can still communicate securely using previously- issued certificates (recall that once a certificate is issued, the CA is not explicitly involved in any later communication among parties using the

CA's certificate. Of course, if the CA goes down, no new certificates can be issued.

KDC: has secret key of all clients; generate one-time session key (symmetric) for each session CA: create certificate of the sender's public key; the receiver can use the public key of CA to make sure that is really the public key of the sender.