Computer Networks Assignment #5 Zhang Zhexian (0545080) zhangzhexian@outlook.com

- 1. [Chapter 5 Hands-on 1] Ns-2 is the most popular simulator for TCP research. It includes a package called NAM that can visually replay the whole simulation in all timescales. Many Web sites that introduce ns-2 can be found on the Internet. Use NAM to observe a TCP running from a source to its destination, with and without buffer overflow at one intermediate router.
 - Step 1: Search the ns-2 Web site and download a suitable version for your target platform.
 - Step 2: Follow the installation instructions to install all the packages.
 - Step 3: Build a scenario consisting of three cascaded nodes, one for the Reno TCP source, one for an intermediate gateway, and one for the destination. The links to connect them are full-duplex 1 Mbps.
 - Step 4: Configure the gateway to have a large buffer. Run a TCP source toward the destination.
 - Step 5: Configure the gateway as to have a small buffer. Run a TCP source towards the destination.

For all the Reno TCP states that the Reno TCP source in the preceding two tests enters, screen dump them and indicate which state the TCP source is in. The figures should be correlated. For example, to represent the slow start behavior you may display it with two figures: (1) an ACK is coming back; (2) the ACK triggers out two new data segments. Carefully organize the figures so that the result of this exercise is no more than one A4 page. Only display necessary information in the screen dumps. Pre-process the figures so that no window decorations (window border, NAM buttons) are displayed.

Steps I have performed:

- o Downloading the ns-2 NAM from http://www.isi.edu/nsnam/ns/ns-build.html
- Downloading Cygwin as it is needed for Windows platform
- o Downloading gcc4-g++ packages as it is needed for ns-2 NAM installation
- o Modify the ./install file in ns-2 NAM to allow checkpoint to be passed
- After countless tries, error still persists:

```
Zhexian@DESKTOP-N696679 ~
$ cd ..

zhexian@DESKTOP-N696679 /home
$ cd ..

zhexian@DESKTOP-N696679 /
$ cd cygdrive/d/ns-allinone-2.35/ns-allinone-2.35/
bin/ install otcl-1.14/ tclcl-1.20/
cweb/ INSTALL.WIN32 README tk8.5.10/
dei80211mr-1.1.4/ nam-1.15/ sgb/ xgraph-12.2/
gt-itm/ ns-2.35/ tcl8.5.10/ zlib-1.2.3/
zhexian@DESKTOP-N696679 /
$ cd cygdrive/d/ns-allinone-2.35/ns-allinone-2.35/
zhexian@DESKTOP-N696679 /cygdrive/d/ns-allinone-2.35/
s./pinstall
```

/cygdrive/d/ns-allinone-2.35/ns-allinone-2.35

```
puter Networks Assignment #5

| Assignment | December 2.35 | Chang Zhexian (Cygdrive/d/ns-allinone-2.35 | Chang
       -o tclsh.exe
/cygdrive/d/ns-allinone-2.35/ns-allinone-2.35/tcl8.5.10/unix/libtcl8.5.a(tclEnv.o):tclEnv.c:(.text+0x110): undefined reference to `cygwin_posix_to_win32_path_li
      /cygdrive/d/ns-allinone-2.35/ns-allinone-2.35/tcl8.5.10/unix/libtcl8.5.a(tclEnv.o):tclEnv.c:(.text+0x110): relocation truncated to fit: R_X86_64_PC32 against un defined symbol `cygwin_posix_to_win32_path_list_buf_size' /cygdrive/d/ns-allinone-2.35/ns-allinone-2.35/tcl8.5.10/unix/libtcl8.5.a(tclEnv.o):tclEnv.c:(.text+0x132): undefined reference to `cygwin_posix_to_win32_path_list'
     /cygdrive/d/ns-allinone-2.35/ns-allinone-2.35/tcl8.5.10/unix/libtcl8.5.a(tclEnv.o):tclEnv.c:(.text+0x132): relocation truncated to fit: R_X86_64_PC32 against un defined symbol cygwin_posix_to_win32_path_list' /cygdrive/d/ns-allinone-2.35/ns-allinone-2.35/tcl8.5.10/unix/libtcl8.5.a(tclEnv.o):tclEnv.c:(.rdata$.refptr.__cygwin_environ[.refptr.__cygwin_environ]+0x0): und efined reference to cygwin_environ' collect2: error: ld returned 1 exit status make: *** [Makefile:569: tclsh.exe] Error 1 tcl8.5.10 make failed! Exiting ... For problems with Tcl/Tk see http://www.scriptics.com
               hexian@DESKTOP-N696679 /cygdrive/d/ns-allinone-2.35/ns-allinone-2.35:
```

2. [Chapter 5 Hands-on 4] Linux Packet Socket is a useful tool when you want to generate arbitrary types of packets. Find and modify an example program to generate a packet and sniff the packet with the same program.

```
Raw TCP packets
#include<stdio.h> //for printf
#include<string.h> //memset
#include<sys/socket.h> //for socket ofcourse
#include<stdlib.h> //for exit(0);
#include<errno.h> //For errno - the error number
#include<netinet/tcp.h> //Provides declarations for tcp header
#include<netinet/ip.h> //Provides declarations for ip header
    96 bit (12 bytes) pseudo header needed for tcp header checksum calculation
struct pseudo header
{
   u int32 t source address;
   u int32 t dest address;
   u int8 t placeholder;
   u int8 t protocol;
    u_int16_t tcp_length;
};
    Generic checksum calculation function
unsigned short csum(unsigned short *ptr,int nbytes)
    register long sum;
    unsigned short oddbyte;
   register short answer;
    sum=0;
    while(nbytes>1) {
        sum+=*ptr++;
        nbytes = 2;
    if(nbytes==1) {
        oddbyte=0;
        *((u char*)&oddbyte)=*(u char*)ptr;
        sum+=oddbyte;
    }
    sum = (sum >> 16) + (sum & 0xffff);
    sum = sum + (sum >> 16);
    answer=(short)~sum;
    return (answer);
int main (void)
```

```
//Create a raw socket
int s = socket (PF INET, SOCK RAW, IPPROTO TCP);
if(s == -1)
    //socket creation failed, may be because of non-root privileges
   perror("Failed to create socket");
   exit(1);
}
//Datagram to represent the packet
char datagram[4096] , source ip[32] , *data , *pseudogram;
//zero out the packet buffer
memset (datagram, 0, 4096);
//IP header
struct iphdr *iph = (struct iphdr *) datagram;
//TCP header
struct tcphdr *tcph = (struct tcphdr *) (datagram + sizeof (struct ip));
struct sockaddr in sin;
struct pseudo header psh;
//Data part
data = datagram + sizeof(struct iphdr) + sizeof(struct tcphdr);
strcpy(data , "ABCDEFGHIJKLMNOPQRSTUVWXYZ");
//some address resolution
strcpy(source ip , "192.168.1.2");
sin.sin family = AF INET;
sin.sin port = htons(80);
sin.sin_addr.s_addr = inet_addr ("1.2.3.4");
//Fill in the IP Header
iph->ihl = 5;
iph->version = 4;
iph->tos = 0;
iph->tot len = sizeof (struct iphdr) + sizeof (struct tcphdr) + strlen(data);
iph->id = htonl (54321); //Id of this packet
iph->frag off = 0;
iph->ttl = 255;
iph->protocol = IPPROTO TCP;
iph->saddr = inet addr ( source ip ); //Spoof the source ip address
iph->daddr = sin.sin_addr.s addr;
//Ip checksum
iph->check = csum ((unsigned short *) datagram, iph->tot len);
//TCP Header
tcph->source = htons (1234);
tcph->dest = htons (80);
tcph->seq = 0;
tcph->ack seq = 0;
tcph->doff = 5; //tcp header size
tcph->fin=0;
tcph->syn=1;
tcph->rst=0;
```

```
tcph->psh=0;
   tcph->ack=0;
   tcph->urg=0;
   tcph->window = htons (5840); /* maximum allowed window size */
   tcph->check = 0; //leave checksum 0 now, filled later by pseudo header
   tcph->urg ptr = 0;
   //Now the TCP checksum
   psh.source address = inet addr( source ip );
   psh.dest address = sin.sin addr.s addr;
   psh.placeholder = 0;
   psh.protocol = IPPROTO TCP;
   psh.tcp length = htons(sizeof(struct tcphdr) + strlen(data) );
   int psize = sizeof(struct pseudo header) + sizeof(struct tcphdr) + strlen(data);
   pseudogram = malloc(psize);
   memcpy(pseudogram , (char*) &psh , sizeof (struct pseudo header));
   memcpy(pseudogram + sizeof(struct pseudo header) , tcph , sizeof(struct tcphdr) +
strlen(data));
   tcph->check = csum( (unsigned short*) pseudogram , psize);
   //IP HDRINCL to tell the kernel that headers are included in the packet
   int one = 1;
   const int *val = &one;
   if (setsockopt (s, IPPROTO IP, IP HDRINCL, val, sizeof (one)) < 0)
       perror("Error setting IP HDRINCL");
       exit(0);
    }
   //loop if you want to flood :)
   while (1)
        //Send the packet
       if (sendto (s, datagram, iph->tot_len , 0, (struct sockaddr *) &sin, sizeof
(\sin)) < 0)
           perror("sendto failed");
        //Data send successfully
        else
        {
           printf ("Packet Send. Length : %d \n" , iph->tot len);
    }
   return 0;
//Complete
```

Compile by program by doing a gcc raw_socket.c at the terminal.

Use a packet sniffer like wireshark to check the output and verify that the packets have actually been generated and send over the network.

3. [Chapter 5 Hands-on 8] What transport protocols are used in MS Media Player or RealMedia? Please use wireshark to observe and find out the answer.

The wireshark capture I obtained is from http://www.pcapr.net/, and here is the exact URL: http://www.pcapr.net/view/bwilkerson/2008/10/3/14/Viewpoint_Media_Player_for_IE_3.2_Remote_S http://www.pcapr.net/view/bwilkerson/2008/10/3/14/Viewpoint_Media_Player_for_IE_3.2_Remote_S http://www.pcapr.net/view/bwilkerson/2008/10/3/14/Viewpoint_Media_Player_for_IE_3.2_Remote_S <a href="http://www.pcapr.net/view/bwilkerson/2008/10/3/14/Viewpoint_Media_Player_for_IE_3.2_Remote_S <a href="http://www.pcapr.net/wiewpoint_Media_Player_for_IE_3.2_Remote_S <a href="http://www.pcapr.net/wiewpoint_Media_Player_for_IE_3.2_Remote_S <a href="http://www.pcapr.net/wiewpoint_Media_Player_for_IE_3.2_Remote_S <a href="http://www.pcapr.net/wiewpoint_Media_Player_for_IE_3.2_Remote_S <a href="http://www.pcapr.net/wiewpoint_Media_Player_for_

As shown in the capture above, the transport protocol used in media player is TCP.

4. [Chapter 5 Written 4] A mobile TCP receiver is receiving data from its TCP sender. What will the RTT and the RTO evolve when the receiver gets farther away and then nearer? Assume the moving speed is very fast so that the propagation delay ranges from 100ms to 300ms within 1 second.

When the receiver gets farther away, the RTT will increase (from 200ms minimum to 600ms maximum), as RTT is proportional to propagation delay which is affected by physical distance. The RTO will increase as well to prevent wrongly recovering packets that are not lost, but just take a long time to transmit.

Similarly, as the receiver gets nearer, both RTT and RTO will become smaller.

Since RTT is affected in real time as the physical distance changes, but RTO needs to be adjusted by algorithm, the RTT will change at a faster rate than RTO.

5. [Chapter 5 Written 6] Given that the throughput of a TCP connection is inversely proportional to its RTT, connections with heterogeneous RTTs sharing the same queue will get different bandwidth shares. What will be the eventual proportion of the bandwidth sharing among three connections if their propagation delays are 10ms, 100ms, and 150ms, and the service rate of the shared queue is 200 kbps? Assume that the queue size is infinite without buffer overflow (no packet loss), and the maximum window of the TCP sender is 20 packets, with each packet having 1500 bytes.

Since the throughput of a TCP connection is inversely proportional to its RTT, and the RTT is proportional to their propagation delay, the eventual bandwidth sharing is inversely proportional to their RTT too.

With delay of 150 ms, a sender transmits 1 packet every 300 ms (RTT is twice of delay); the sender with 100ms delay transmits 1.5 packets every 300 ms. A sender with 10ms delay transmits 15 packets every 300ms.

Total packets = 1 + 1.5 + 15 = 17.5;

150ms sender: 1/17.5 = 6%; 100ms sender: 1.5/17.5 = 8%; 10ms sender: 15/17.5 = 86%.

6. [Chapter 5 Written 8] If the smoothed RTT kept by the TCP sender is currently 30ms and the following measured RTTs are 26, 32, and 24ms, respectively, what is the new RTT estimate?

By using dynamic RTT estimator, which adopts the exponential weighted moving average (EWMA), which takes 1/8 of the new RTT measure plus 7/8 of the old smoothed RTT value to form the new estimate of the RTT.

(1/8)*26+(7/8)*30 = 29.5 (1/8)*32+(7/8)* 29.5 = 29.813 (1/8)*24+(7/8)* 29.813 = 29.086

Thus, the new RTT estimate is 29.086ms.

7. [Chapter 5 Written 13] Suppose you are going to design a real-time streaming application over the Internet that employs RTP on top of TCP instead of UDP. What situations will the sender and the receiver encounter in each TCP congestion control state shown in Figure 5.21? Compare your expected situations with those designed on top of UDP in a table format.

State	RTP over TCP	RTP over UDP
Slow start	Slow start aims at quickly probing	UDP does not require connection
	available bandwidth within a few	between sender and receiver, so the
	rounds of RTTs. Since three-way	proving time will be shorter.
	handshake is required in TCP, it will take	
	a longer time to complete the probing.	However, since no ACK is sent from
		receiver to sender, cwnd is always set
	Once bandwidth probing is done,	to 1. Due to the same reason,
	several bandwidths can be utilized	bandwidth sharing is not possible in RTP
	simultaneously for efficient RTP	over UDP so the transmission may not
	transmission.	be as efficient.
Fast retransmit	Fast retransmit targets transmitting the	RTP itself does not address resource
	lost packet immediately without waiting	management and reservation and does
	for the retransmission timer to expire.	not guarantee quality-of-service for
		real-time services. RTP assumes that
	Triple duplicate ACK will trigger	these properties, if available, are
	retransmission, thus resulting in faster	provided by the underlying network.
	resending of loss packet. The packet	
	loss rate will be lower resulting in better	However, such retransmission
	quality streaming. Nevertheless, the	mechanism is missing in UDP, so
	time taken for retransmission affects	quality-of-service may be severely
	streaming speed.	affected in RTP over UDP.

Networks Assignment	. 113	Zhang Zhexian (0343000)
Congestion Co	ongestion avoidance aims at slowly	No formal congestion control
avoidance pr	obing available bandwidth but rapidly	mechanism is implemented. Thus, in
res	sponding to congestion events	times of network congestion, RTP
		packets may be dropped, affecting
Or	nce congestion is detected, the cwnd	streaming quality.
wi	ill be quickly reset to 1 to trigger the	
fas	st-transmit state, and the real time	However, since no cwnd size change,
str	reaming speed will suddenly be much	the streaming speed will not fluctuate
slo	ower.	much during streaming.
Retransmission Re	etransmission timeout provides the	No retransmission timer is maintained
timeout las	st and slowest resort to retransmit the	in UDP sender, because no
los	st packet.	acknowledgement will be sent from the
		receiver.
lt i	is implemented to TCP to ensure a	
hig	gher quality-of-service for RTP. Since	
th	e timer is designed with RTT, it	
mi	inimizes the possibility of resending	
alr	ready-transmitted packets that wastes	
ne	etwork resources.	
Fast recovery Af	ter a packet loss, fast recovery state	RTP itself does not address resource
pr	eserves enough outstanding packets	management and reservation, as such it
in	the network pipe to retain TCP's self-	does not retransmit lost packets.
clo	ocking behavior.	·
lt a	allows more packets to be sent even	
wh	hen packet loss is detected, thus	
ind	creasing recovery speed for RTP.	

- 8. [Chapter 5 Written 16] The text goes into some detail introducing the different versions of TCP. Find three more TCP versions. Itemize them and highlight their contributions within three lines of words for each TCP version.
 - a. TCP Hybla: it aims to eliminate penalization of TCP connections that incorporate a high-latency terrestrial or satellite radio link, due to their longer round-trip times. It implements the necessary modifications to remove the performance dependence on RTT.
 - b. TCP Westwood+: it is a sender-side only modification of the TCP Reno protocol stack that optimizes the performance of TCP congestion control over both wireline and wireless networks. It significantly increases throughput over wireless links and fairness compared to TCP Reno/New Reno in wired networks.
 - c. Compound TCP: it is a Microsoft implementation of TCP which maintains two different congestion windows simultaneously, with the goal of achieving good performance on LFNs while not impairing fairness.