Anatomy of Computer Time and its Application to a DRM Scheme

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***Abstract*— This paper presents the computer timing mechanism and its relationship with the packet transmission delay at the protocol stack. The experiment platform uses two general purpose personal computers connected directly via a network interface. This setup allows us to ignore the influence of end-to-end network devices between the hosts which can cause delay variation. In trials we measured the round-trip time (RTT) between the two hosts in various parameters, such as packet size, packet interval, burst length and burst interval. We show the various RTT characteristics and recommend several parameters that can render precise and stable measurement. Furthermore, we present a proposal for a location-based identification system in the Digital Rights Management (DRM) scheme for a secure digital content distribution via the Internet. This scheme allows the user storage system to be identified by its network location, among other personal identification systems, for its strict content management and prevention of illegitimate activities on the Internet.**

I. INTRODUCTION

Because of the heterogeneity of the Internet being a packet-switched network, variation of packet transmission delay poses a great challenge in network measurements. Packet delay variation, or jitter, are attributed to queuing in intermediate nodes [1], path routing behavior [2], distance between the hosts and the underlying transmission medium [3], cross traffic effects [4], and the processing of packet at the end host [5,6].

Measuring round-trip time (RTT) is simplifier than one-way delay. To precisely measure one-way delay, the end hosts must first be synchronized using network time protocols or by external synchronization scheme using GPS. However, the accuracy of network time synchronization protocols in general configuration still cannot be guaranteed to be better than few tens or hundred microseconds. In RTT measurement, time offset and clock frequency difference between the hosts does not affect the result because time stamps are taken in the same clock.

The most common network tool known as Ping, measures RTT using ICMP echo request/echo reply mechanism. There are other ways to measure RTT, such as the UDP and TCP based methods. Several studies have also been made regarding RTT measured across the Internet spanning in large distances, however, their aim is to characterize network traffic and end-to-end behavior [1,7,8].

The aim of this study is to determine the optimal packet transmission parameters that will result to precise and stable RTT measurements. As a first approach, we used a simplified experiment platform - using two computers directly connected by a cross cable via a network interface. This approach allows the analysis of the packet transmission delay at the protocol stack, while ignoring the effect of the network devices. In our experiments, we investigated the RTT characteristics in various parameters, such as packet size, packet interval, burst length and burst interval.

Specifically, we intend to employ the precise and stable RTT measurement to a Digital Rights Management (DRM) scheme [9,10]. DRM is a framework of copyright protection particularly for digital content distributed over the Internet. We propose a new DRM system consisting of the user mobile device, user storage system, and the content servers. To strengthen the DRM scheme, we propose a location-based identification system to authenticate the user storage device and to strictly enforce content management. In this paper, we briefly describe our DRM concept and its implementation.

II. PACKET TRANSMISSION MODEL

In this section, we describe the terminologies for probe packet transmission model. This model is based on the traditional method of measuring RTT, that is, a two-way transmission. In this method, the sender sends a probe packet to the receiver, and the receiver returns the packet to the sender. However, we have modified several packet transmission parameters that gave us various RTT characteristics which we describe in the next section.

A. *Cyclic Transmission*

We refer a two-way transmission between a sender and receiver as cyclic transmission when the transmission is repeated after the sender completely receives the returning packet from the receiver. The period of one cycle depends on

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the actual round-trip packet transmission time. This period may be modified by introducing a time interval between two transmissions. When only one packet is sent per transmission, it is practical to refer it as cyclic single-packet transmission.

Sending multiple packets successively in per transmission results to a burst transmission. Each burst consists of equal number of packets. We refer this model as cyclic burst transmission.

Cyclic burst transmission was conceived based on the observation that the first few RTT data in a time series usually have higher values. This happens when the operating system has been idle for a certain period, thus invoking it may take a considerable time.

B. *Packet Transmission Parameters*

We modified and varied several parameters of the packet transmission model. The objective of varying the parameters is to determine the packet transmission condition that will result to precise and stable RTT measurement. These parameters include:

- Packet size: The number of bytes of one-length packet.
- Packet interval: The interval between the arrival time of returning packet and the departure time of the next probe packet.
- Burst length: The number of packets transmitted successively per burst.
- Burst interval: The time interval between burst packet transmissions.

Fig. 1a shows the two-way packet transmission parameters, while Fig. 1b shows the burst packet parameters. In Fig. 1b, the head packet is represented as P1 and tail packet as Pn, where n is the number of packets in a burst.

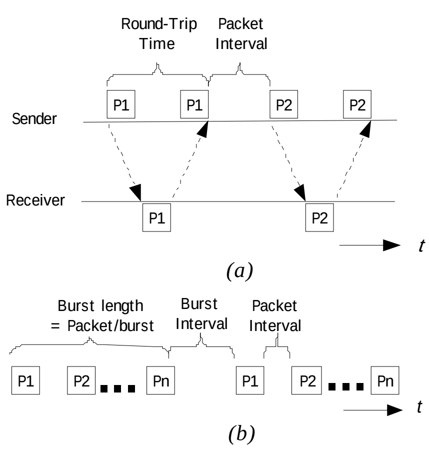

Figure 1. (a) Two-way single packet transmission parameters, (b) Burst transmission parameters.

III. Experimentation and Analyses

We used an experiment platform that is ideal for measuring RTT at the protocol stack. We directly connected two computers having the same specifications using a 2-meter cross cable via a 1Gbps NIC. The two computers are both Dell Optiplex GX520 Pentium 4HT 3GHz and run by Fedora Core 4 – 2.6.17 kernel. The software tool for measuring RTT is based on UDP client-server program. Time stamping is done by an rdtsc function call, where the TSC count values are converted to absolute time.

A. *Effect of Packet Size to RTT*

In digital transmission, the time to transfer data is directly proportional with the length of data transferred at a given data rate. To serve as benchmark, we measured the RTT for different packet sizes sent cyclically and without packet interval. Results show that RTT increases proportionately with packet size, provided that packet size does not exceed the Maximum Transmission Unit (MTU) of the interface. When we performed the same experiment using other platforms, we got the same characteristics but the figures were different, which shows that protocol stack delay is hardware and software dependent.

B. *Effect of Transmission Interval to RTT*

We examined the effect of introducing a time interval between packet transmissions. When the interval is varied, the transmission rate also varies correspondingly. At higher rate, that means with a short time interval, the RTT characteristics shows multi-modal behavior because of the appearance of several peaks. Based on observation in several trials, highest peak does not appear constant at certain peak order; it may appear at first, second or even at third peak. Peak to peak distance varies depending on the packet interval. At lower transmission rate, however, the data distribution shows only one peak and is skewed to the right. But the distribution becomes similar to a Gaussian curve as the packet time interval increases beyond 1sec.

Fig. 3 shows the effect of varying the packet interval to the RTT value and its deviation. It shows that RTT increases logarithmically with respect to the packet time interval. However, for a packet time interval greater than 1sec, RTT have different characteristics. It was observed in 100sec packet interval, that at every 6 RTT measurements or about 600sec, the value is around 3-4msec, when actually the peak RTT is only around 100usec. Also, for a 1000sec packet interval, RTT measures 6-10msec. This erratic effect on RTT measurement is due to the slower response of the receiver when the kernel has moved to an idle state. Moreover, the deviation of RTT measurements is observed to vary for different transmission intervals. At shorter transmission interval, the data deviates from the peak value around 300-350nsec for each peak. But at longer packet time interval, deviation is considerably higher.

C. *Effect of Burst Transmission to RTT*

We further investigated the effect of sending multiple packets per transmission,which we called a burst transmission. We varied some parameters such as the number of packets per

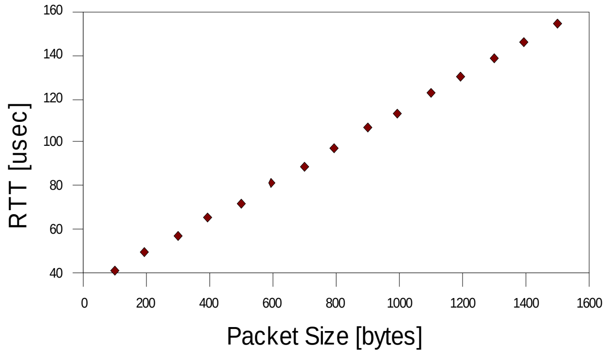

Figure 2. RTT of different packet sizes on 1Gbps interface.

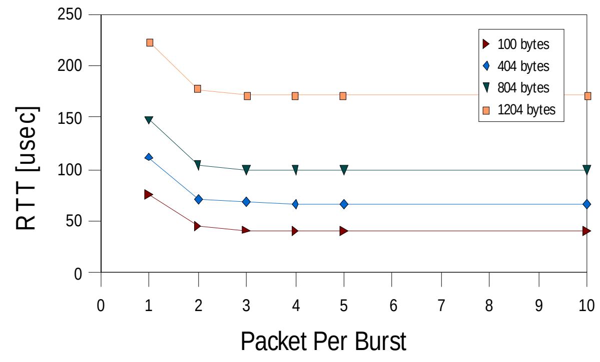

Figure 4. Effect of burst transmission to RTT at different packet sizes. Burst interval is 1sec.

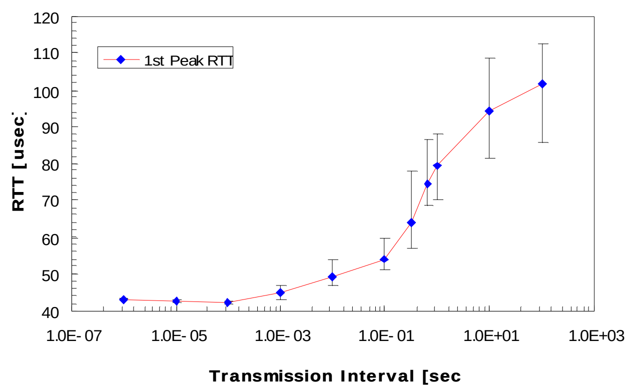

Figure 3. Effect of varying the transmission interval to RTT on 1Gbps interface using 100-byte packet.

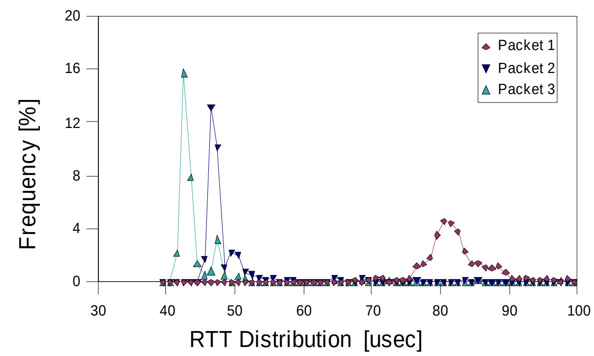

Figure 5. Per packet RTT distribution from a 3-packet/burst transmission at 1sec burst interval.

burst and burst interval. Fig. 4 shows that sending multiple packets can reduce the RTT value compared to a single-packet transmission. This is due to the combined effect of short and long packet intervals because packets in a burst transmission are sent successively, and then there is a long burst interval. Based on the graph, sending 3 packets per burst or more, the peak RTT vary slightly. This value was taken from the first peak found in the frequency distribution, and not from the total distribution. Similar characteristics were observed for longer packet sizes with an increase in RTT which is reasonable.

In a burst transmission, we verified which packet can offer a precise and stable peak value. As an example, Fig. 5 shows the RTT distribution when 3 packets are sent per burst. The RTT distribution for all first (head) packets is located at far right side of the second and third (tail) packets. Several outliers are seen from the peak going to the minimum value. The distribution from the tail packets gives a very narrow distribution that characterizes a stable and precise delay measurement.

D. Ping RTT and UDP RTT

As shown in Fig. 6, we compared our results to the RTT data gathered using Ping network tool. Compared to 1-packet

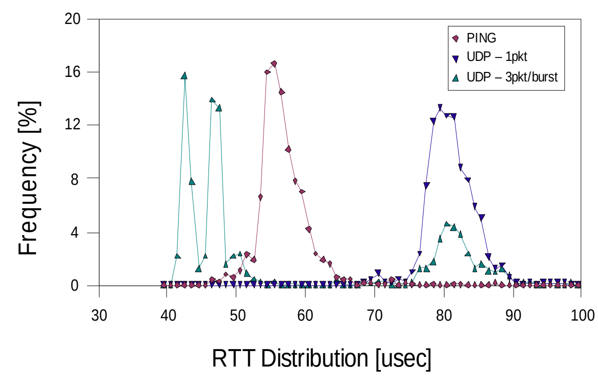

Figure 6. Comparison between Ping, 1-packet UDP and 3-packet/burst UDP using 100-byte packets. All have 1 sec transmission interval and measured on a 1Gbps interface.

UDP, Ping data has lower RTT value. However, this is reasonable because the Ping receiver works at layer 3, while our program is based on UDP and works at the user space. Comparing Ping and the 3-packet/burst UDP, the latter shows high probability of providing precise and stable measurement because of narrower distribution exhibited by the first peak that corresponds to 27% of the total distribution.

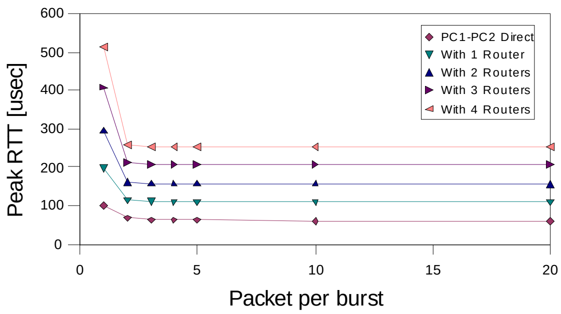

Figure 7. Router influence to RTT on 100Mbps Ethernet interface using 100byte packet at 1sec burst interval.

E. Router Influences to RTT

In order to examine the effect of sending multiple packets in measuring RTT across a real network, we performed a simulation using several routers linked via Ethernet interface at 100Mbps. Fig. 7 shows the peak RTT vs. packet per burst on different network configurations. In this experiment, we used 100bytes probe packets sent every 1sec burst interval. As shown in the figure, each router introduces transmission delay, but burst transmission has the same effect on RTT characteristics even when delays have been increased. It was observed that first peak produces the precise and stable RTT measurement.

IV. APPLICATION

In this section, we briefly describe the target application of precise RTT measurement, particularly in the Internet-based digital content distribution that uses DRM scheme.

A. A New DRM Scheme

Digital Rights Management (DRM) has been the answer to the pressing problem on infringement of copyrights, particularly for unauthorized copying of digital contents such as music, video, and other similar valuable digital data. More recently, the specification of DRM does not only sought to solve the issue of illegal content reproduction, but has restricted what the user can do with the content such as viewing, copying, and many other things that you can possibly do with digital content [9]. These control actions are all handled by a company provided software on an Internet-based permission scheme.

Currently, the general way to identify a user is by using a unique personal identification number (PIN) or by a password. It is, however, difficult to maintain a perfect management of such data. Identification data can be used illegitimately after they have been stolen and tampered.

A new DRM scheme proposed in [10,11] provides secure content distribution via networks to end users in ubiquitous environments. The DRM system configuration consists of three components, namely the user mobile device, the user storage system and the content servers. This DRM scheme encompasses a comprehensive content management method based on the location information of the user mobile device, as well as the user storage system. Location information of the user is utilized to produce a unique content associated to the user. Digital content is vulnerable to unauthorized actions by the end user, thus making it user-identified can help deter illegal copying.

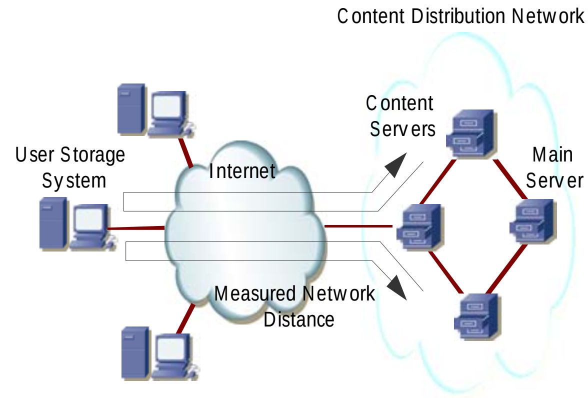

Figure 8. Network location identification system using network distance estimated from round-trip time (RTT).

B. Network Location Identification

In Fig. 8, we present our proposed storage system identification system for the DRM scheme in Internet-based content distribution network. The content distribution network consists of content servers that are strategically located in different locations and a main server that manages the implementation of the DRM scheme. In the storage system identification process, we propose a network location based on a 2-dimensional coordinate system. Conceptually, a location can be accurately derived from a set of distances from that unknown location to a minimum of three (3) reference points by triangulation (lateration) method. In the Internet domain, however, we may not be able to get the host-to-host geographic distance, but rather a network distance, which is a function of the network topology and routing path, and is not dependent on the instantaneous network traffic load. To estimate network distance, the valuable metric is based on packet round-trip time (RTT). In this proposed system, we consider that a precise and stable RTT measurement ensures the accuracy of network location estimation.

In our DRM scheme, storage system identification process is carried out every download and playback processes, where the user invokes the content server by sending a particular request. Upon receipt of the request that also contains other user identification data, a number of content servers sends probe packets to the storage system, where it has to send back a reply to each. This mechanism allows RTT measurement from the storage system to the content servers. These data are then sent to the main server that computes the network location of the storage system. In the download process, this network location information is converted to a key used to encrypt the content. While in the playback process, the newly probed network location is used to issue a key for decryption.

V. SUMMARY

We have presented the effects of varying several parameters of the packet transmission model in measuring RTT between the two hosts. Our purpose is to determine the best parameters that could give us precise and stable RTT measurement across the Internet. The experiment results we have presented were based on the ideal platform, but it was useful in understanding the capability and responses of the host machines in handling several packet transmission parameters.

We have described that multiple packet transmission exhibits the combined effect of short and long transmission intervals. However, we cannot specify how many packets would be considered an optimal value as we have not tested this method on a practical network. On the other hand, we have presumption that, in the presence of cross traffics, more packets per burst is better.

It was presented that the RTT distribution of tail packets in a burst transmission exhibits the first peak. In the implementation, when the tail packet is to be considered, then it could reduce the cost of data manipulation. The advantage of selecting tail packet is that it offers narrow distribution, thus few measurement is required.

In this paper, we also have presented our proposed network location identification system for the DRM scheme employed in Internet-based digital content distribution. An on-going design and implementation schemes are being mapped out to verify the efficiency of the proposed system.

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