Real-time Application Traffic Monitoring System for Streaming Data Quality Evaluation

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Abstract

Our innovative traffic-monitoring system lets operators observe data-communication traffic as an oscilloscope-style trace, which provides an efficient way of directly evaluating the quality of data streaming services. The monitoring system has a high time-resolution (1 ms) sampling function and a real-time data representation/recording mechanism that operate in synchrony.

1. Introduction

The popularity of broadband network access services (FTTH and ADSL) will inevitably lead to the spread of data-streaming-based services. These next-generation network services include TV, movie, and radio streaming. They require real-time data transfer as well as received data integrity, which means that the received data arrives on schedule and delayed data is discarded. Unfortunately, the conventional data retransmission mechanism that ensures data reliability fails to ensure received data integrity because it tolerates propagation delays in data transmission between servers and clients.

If we are to develop a high-quality data streaming environment, we need a new traffic monitoring system that can provide a more detailed view of the traffic environment than is possible with conventional monitors. In particular, time resolution is a key factor for observing the behavior of traffic in the network. We chose a time resolution of 1 ms and examined some actual streaming applications. We found that this time resolution is sufficient for streaming applications.

2. Conventional traffic monitoring system

Conventional network analyzers, which are widely used for monitoring networks, generally have a traffic monitoring function and an in-depth telecom data analyzer. The traffic monitoring function periodically measures the network line utilization ratio and plots it on a chart similar to a stock price graph. The time interval is set to more than one second. The in-depth data analysis function is essentially a protocol-regulation-based data correctness evaluation in which each elementary telecom datagram is evaluated in a packet-by-packet manner.

The conventional traffic monitoring function is used to collect reference data for the design and maintenance of network facilities. The traffic is periodically sampled at intervals greater than one second, which is short enough for proper maintenance. MRTG, which is a well-known traffic measurement system, reports traffic values every five minutes [1].

Likewise, commercially available traffic monitoring systems do not have high enough time resolution to be able to observe streaming data traffic in detail.

3. Our traffic monitoring system

We developed a high time resolution traffic shape monitoring system that can evaluate data streaming quality in real time. Our monitoring system can pick out a specific traffic flow from a network line carrying a tremendous number of data communications and plot the flow's traffic volume versus time on a display in real time, similar to an oscilloscope trace (Fig. 1). It has three main features:

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- High time resolution: The traffic is sampled every 1 ms, which is sufficient for observing the characteristic shape of the traffic.
- Traffic-flow selection: A series of particular telecom packets can be selected from the line traffic according to their attributes.
- Real-time data plotting and storage: The periodically sampled traffic is shown on a display and stored in a storage device in real time.

The high time resolution and real-time traffic plotting function of our system are unique features that have never been implemented in any other traffic monitoring system. Figure 2 compares our monitoring system with other forms of traffic measurement and monitors. The time resolution is determined by the balance between the streaming source media's behavior and networking line bandwidth. For example, NTSC-based video is equivalent to almost 30 still-frames per second. High-quality video streaming services aim to support at least this frame rate. Every still-frame must be transferred within 33 ms. Standard cinematographs use 24 still-frames per second

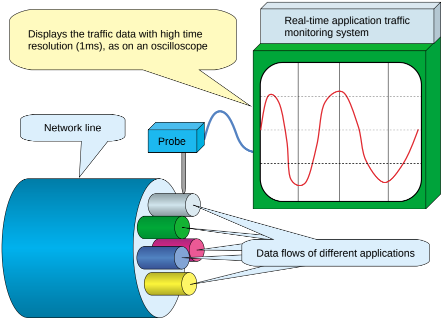

Fig. 1. Functionality of our traffic monitoring system.

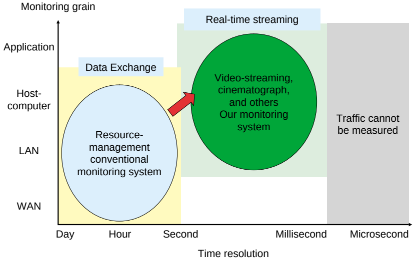

Fig. 2. Relationship between our monitoring system and others.

for which the frame interval is 40 ms. Therefore, we can observe and evaluate their behaviors with a 1-ms traffic sampling interval.

From the perspective of network bandwidth, on the other hand, we need a statistically meaningful number of packets to measure traffic volume. The Gigabit-Ethernet (GbE) can transfer a large datagram within about 12 µs. If we sample a GbE line every 10 µs, the datagram can be observed, but the traffic's bit rate cannot. This sampling period is too short to observe the bit rate. Accordingly, we chose 1 ms for the traffic sampling interval.

This time resolution can also handle higher-quality and forthcoming movie streaming based on HDTV (High definition TV) over the GbE network. An MPEG2-encoded HDTV frame uses around 1.5 Mbit, which occupies 1.5 ms on the GbE network line. This can be observed with our time resolution.

4. Applications of the traffic monitoring system

Case 1: MPEG-2 streaming system evaluation

We evaluated two streaming systems using an MPEG-2 codec, which is widely employed for broadband video streaming over networks (Fig. 3). Each system was evaluated in terms of its multi-channel streaming ability on a shared media network. The systems were connected to a network concentrator (hub) and the shape of the streaming traffic was observed with our monitoring system (Fig. 3(a)). The shape of system A's traffic shows that the streaming data was sent consistently at under 10 Mbit/s. The shape of system B's traffic, on the other hand, was discontinuous, with peaks exceeding 40 Mbit/s being generated at 20-ms intervals.

This result shows that we can use system A for a multi-channel streaming service and can connect five systems of type A to one 100-Mbps shared media network with a safety margin. However, if we do the same with system B, we will lose data because of traffic overflow if three or more traffic peaks overlap on a 100-Mbps shared media network. Thus, we conclude that system A is the better choice for multi-channel streaming on a shared media network (Fig. 3(b)).

We measured the traffic shape with a conventional traffic monitoring system to compare it with our monitoring system. The results are shown in Fig. 4. The conventional monitoring system could not observe any difference between the two systems. We can conclude that our monitoring system is the only one that has sufficient time resolution to evaluate a streaming system.

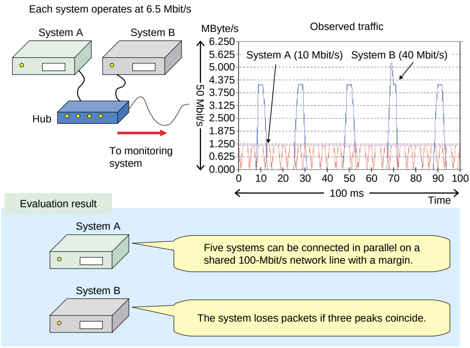

Fig. 3. MPEG-2 streaming system evaluation.

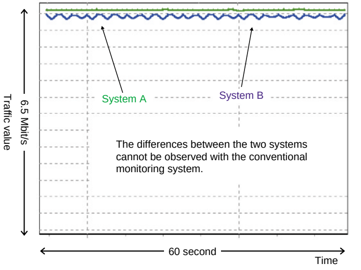

Fig. 4. Measurement results for a conventional monitoring system.

Case 2: Ultra-wideband streaming application development

We used the monitoring system in a ground-breaking experiment that attempted to distribute a Super High Definition (SHD) [2] movie from Chicago to Los Angeles over a wideband experimental network (called the Internet2 [3]) via TCP/IP. An SHD movie requires a bandwidth of at least 300 Mbit/s. The round-trip-time (RTT) between the cities was 60 ms. Here, we describe how the monitoring system works to aid streaming system development.

The blue line in Fig. 5 shows the "un-tuned" traffic shape of the SHD movie in a LAN environment measured with the new monitoring system. The RTT was just a few microseconds in this case. The average traffic volume, measured by a conventional monitoring system, was around 300 Mbit/s. However, the peak traffic value reached almost 1 Gbit/s, which can be seen if the peaks (groups of them) can be observed every 40 ms.

We tried to distribute an SHD movie through Internet2 using this "un-tuned" traffic shape without any tuning up. This trial failed because of the effect of the large RTT, which determined the data transfer performance of TCP. To solve this problem, we used multiple TCP connections between the SHD server and client with a certain bandwidth allocated to each connection. The TCP data transfer behavior was observed with our monitoring system and its shape was smoothed by an engineer using the monitoring system to tune the traffic's shape (red line in Fig. 5). Without our monitoring system, this trial would also have failed. This clearly demonstrates the advantage of our monitoring system for streaming application development.

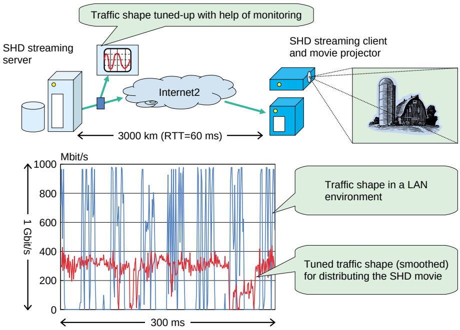

Fig. 5. Ultra-wideband streaming experiment over Internet2.

5. Architecture

The architecture of the monitoring system is schematically illustrated in Fig. 6. The system consists of two GbE line interfaces, a packet classifier, and two packet length accumulators for each GbE interface, a real-time data read module, a measured-data transfer interface, and a data plotting display. The network line is connected to the monitor's line interface using an optical coupler or a network hub that splits the line signal into two line signals. The line interfaces can be used as optical cable when the system is in pass-through mode. In this case, the monitor can be inserted into a network line directly. One of the split optical signals is inserted into the monitor's line interface. The communication packets are extracted on the line interface and classified according to pre-configured packet classification rules in the packet classifier. The classification rules use 128-bit-wide data which can contain existing protocol packets header formats. The packet classifier can have two rules for selecting the packet length accumulator. The packet length is measured and the value is added to the corresponding packet length accumulator. These functions are done in hardware and work in a packet-by-packet manner without any packet loss. The real-time data read module reads the accumulated value in each accumulator every millisecond. The interface module sends bundles of these measured values to the display module. The display can be located at a remote site, as it communicates with the monitoring system through the TCP/IP-based network.

6. New standard in network measurement

Our new monitoring system sets a new standard for network quality measurements. Monitoring network usage based on a service level agreement (SLA) is one of its new applications (Fig. 7). Usable bandwidth is an important element of the SLA contract, and users generally run a best-effort-based network application over network lines that are shared with other users. This means that when an application sends traffic that momentarily violates its SLA, the bandwidth available to other users will be reduced. These momentary SLA violations cannot be detected by conventional monitoring systems. Our monitoring system can detect them and provide information about how to solve the problem. This ability will be very useful for network management and traffic engineering.

References

[1] http://people.ee.ethz.ch/~oetiker/webtools/mrtg/

[2] T. Fujii, M. Nomura, D. Shirai, T. Yamaguchi, T. Fujii, K. Hagimoto, and S. Ono, "IP transmission system for digital cinema using 2048 scanning line resolution," IEEE Global Telecommunications Conference 2002 (GLOBECOM2002), Taipei, Taiwan, Vol. 2, pp. 1643-1647, 2002.

[3] http://www.internet2.edu/

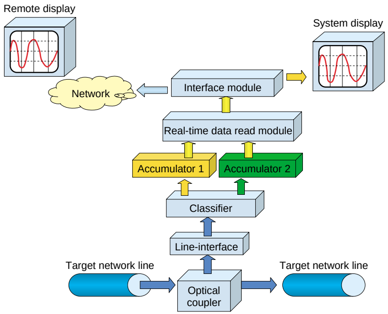

Fig. 6. Architecture of our monitoring system.

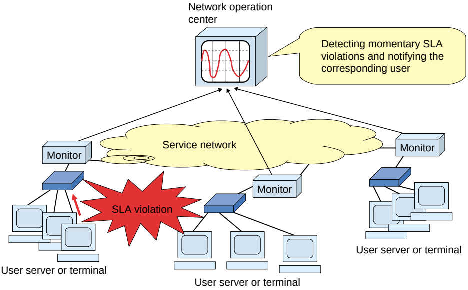

Fig. 7. Application for monitoring SLA violations.

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