Network Traceability Technologies for Identifying Performance Degradation and Fault Locations for Dependable Networks

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Abstract

This paper discusses the per-session quality measurement technology using packet sampling. IP networks are diversified by incorporating various applications and thus systems capable of performing detailed quality measurements at multiple points is one of the most important factors in enabling network dependability. The technology proposed in this paper has solved the problems both of the cost that hinders the installation of a large number of probes and that of the MIB monitoring not being of a satisfactory quality. This innovative approach is expected to contribute greatly to the construction of dependable networks.

Keywords

large-scale broad-area quality measurements, performance degradation location tracing

1. Introduction

One of the essential conditions of network dependability is its high healthiness. Network operations are required to detect faults such as degradation in communications quality either in the overall network or in a specific application. It is also required to be able to identify the cause of a fault and to attempt a rapid restart of services, as well as being capable of diagnosing a fault from the quality degradation trend and to thus be able to prevent faults before they occur. Although these procedures have been implemented in the telephone networks, a radical review is necessary for the Internet which has completely different characteristics from telephone networks in terms of the scale, capacity, switching method and media transfer requirements as well as of the QoS control.

Previously, the main focus in Internet operations has been placed on the type of data traffic. However, as a result of the recent deployment of new applications for VoIP and video distribution and of the increase in unwanted traffic such as virus and DoS (Denial of Service) attacks, the amount of traffic flowing through the network, its characteristics and the requirements for the IP network have been changing significantly. As this trend has resulted in diversification of the factors causing network faults and also that the range and degree of faults are widely variable depending on applications, quick countermeasures against such faults have become more difficult to apply than ever. In order to identify performance degradation and fault locations in large-scale networks by dealing with the problem in the IP network, it is now required to use a technology that is capable of high-speed and detailed measurement and monitoring[1].

In this paper, we will review the main traffic monitoring techniques including the passive probe and MIB monitoring, and will describe the packet sampling measurement technology that is attracting attention recently due to its capability of providing detailed monitoring at low cost. Since packet sampling measurements collect statistical information such as the number of lost packet per router port, or the number of active sessions per router port, it is suitable for the macroscopic measurement of the overall network. It is, however, difficult to use for detailed measurements, such as quality measurements per application or per session. As a result, it is pointed out that, even when the end users notice quality degradation, the networks that use the packet sampling measurement often pass over degradations without noticing them.

To deal with this problem, we propose a method that allows the sampling measurements to estimate the quality per session with high accuracy. This method estimates the quality of each session based on TCP and UDP/RTP characteristics, and is capable of detailed quality monitoring without adversely affecting the high-speed and low-cost advantages of sampling measurements. In this way, the proposed method enables low-cost quality measurements even for large-scale networks and can be regarded as one of the key technological elements for building dependable networks.

2. Quality Monitoring Techniques

In general, passive probe and MIB monitoring are the two major means for traffic monitoring, but these methods present associated problems as described below.

The passive probe technique can monitor detailed information on a per-application or per-flow basis using the traffic measurement devices (probes) in the monitoring target points. However, since installation of a large number of probes in the network is not so practical, monitoring can be performed only at a limited number of network locations and it is not easy to identify the actual fault occurrence location accurately.

On the other hand, the MIB monitoring technique monitors the statistical information collected by the routers. It can monitor multiple points for a lower cost than the probe technique because it makes use of a function that is basically built into the routers. However, since the statistical information collected by the routers is already-processed data such as the 5-minute average packet flow of each port, this technique cannot obtain detailed information on a per-session or per-application basis.

3. Sample Measurement Technology

Packet sampling measurement technology is recently attracting attention because of its applicability as a solution to the above problems. Sampling measurement technology does not measure all of the packets passing through a point but measures only some of them as samples. As a result, the probe technique that adopts sample measurements can reduce the cost and increase the speed of the measurements compared to the ordinary probe technique.

Packet collection systems based on sampling, such as the NetFlow / sFlow[2] have recently been standardized and are being incorporated in routers. The router interfaces collect some of the flowing packets using the sFlow function and send them to another point as shown in **Fig. 1**. Upon reception of the sampled packets, the quality is analyzed in detail. Such a measurement system can monitor the quality by using the sFlow function in the routers, without installing a large number of probes inside the network.

While the sFlow function has to be incorporated in many routers in order to monitor multiple points, the impact in terms of packet sampling cost is not so big because the sFlow function does not have the status of individual flow but simply samples packets according to a simple rule. The load of the server that analyzes the quantity and quality of the measured packets sent from the sFlow routers can be reduced as a result of sampling.

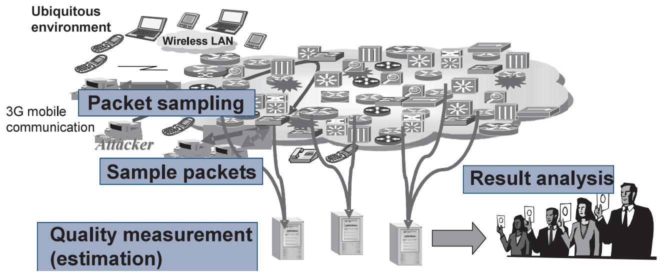

Fig. 1 Monitoring by sampling measurement.

The methods that have been proposed in relation to the packet sampling measurement technology include the technique of estimating the quantity of packets passing during a sampling measurement[3], the technique of identifying elephant flows with higher sending rate and longer duration[3, 4], and the technique of detecting performance degradation at the TCP flow level[6].

However, little consideration has been given to the packet loss rate, which is one of the key indicators in the network management process that is capable of representing the degree of network quality degradation most accurately. The MIB information from the routers can quantify the number of packets discarded at each interface, but it does not identify user quality indicators, such as the number of discarded packets per session or per application.

In this paper, therefore, we propose a method for estimating the packet loss rate of each session based on the packet sampling measurements. In sample measurements, it is generally hard to observe the packet loss rate per session because when a packet of interest is not observed, it is not easy to distinguish whether the packet is not observed because the network discard it or that the packet was simply not sampled. The method proposed by us makes it possible to estimate the packet loss rate of each session statistically and accurately based on the protocol behavior of the session layer such as its TCP or RTP/UDP characteristics.

4. TCP Session Quality Measurement Technique

4.1 Summary of Proposed Method

In this section, we will discuss the packet loss rate estimation method for a TCP session. Large portion of data traffic such as web accessing and FTP use TCP.

When a data packet is discarded at some point between the send and receive terminals in a TCP session, the receiving terminal sends duplicate ACK (phenomenon in which the identical ACK sequence number is sent successively) to the sending terminal. This allows the packet discard to be detected indirectly by detecting the duplicate ACK packets. In consequence, the proposed method captures the ACK packets by random sampling at the probes and transfers the captured packets to the quality analyzer to estimate its quality.

However, the sampling-based capturing process cannot observe all of the duplicate ACK events because it does not measure all of the ACK packets. If the packet loss rate was calculated only from the detected duplicate ACK events, the obtained value would be much smaller than the actual packet loss rate **(Fig. 2)**.

The proposed method therefore statistically estimates the packet loss rate that can be detected when sampling is applied (detection probability). It then estimates the actual packet loss rate by dividing the number of packet losses observed by sampling using a detection probability method.

When the sampling rate is 10% and the packet loss rate of the sample is 1%, the packet loss rate is calculated as being 0.01%, if a correction based on the proposed method is not applied. On the other hand, the proposed method can estimate the packet loss rate more accurately for an error rate of less than 10%. This means that the number of detected packet losses is improved a 100-fold compared to the previous method. For detailed performance evaluation using simulations and laboratory experiments, refer to Reference 7).

4.2 Calculation of Detection Probability

Here, we will call "the number of times the same ACK number occurs successively in a packet loss" the duplicate ACK count. As reported in Reference 7), the duplicate ACK count coincides with the congestion window size in a packet loss and follows the Poisson distribution. The average congestion window size immediately before packet discard, W, can be obtained with the following formula as described in the traditional study in Reference 8).

$$W = \sqrt{\frac{8}{3bp}} \text{ Where: } b = \textit{delayed ack} \text{ (usually 2), } p = \text{packet loss rate.}$$

Based on the above, the duplicate ACK count can be regarded as following the Poisson distribution of the mean value $\lambda = W$.

On the other hand, when packets are sampled with a sampling rate of s, the detected duplicate ACK count should originally be equal to the duplicate ACK count multiplied by s. As a result, the average duplicate ACK count with sampling measurement, λ, becomes equal to $W \times s$. Here, assuming that $P(x)$ is the probability that the Poisson distribution of $\lambda = W \times s$ takes x, and a packet loss is identified when the duplicate ACK packets are observed for n times or more, the detection probability Qn can be defined with the following formula[7)].

The detection probability Qn

= (The number equal to or larger than n in the Poisson distribution of the Mean value $\lambda = W \times s$) / (Total number of Poisson distribution of the Mean value $\lambda = W$)

$$= 1 - \sum_{i=0}^{n-1} P(i)$$

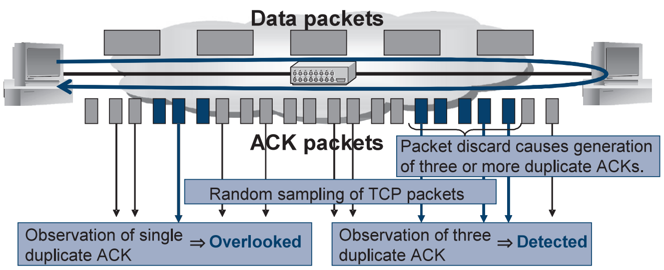

Fig. 2 TCP packet loss rate estimation.

5. RTP/UDP Session Quality Measurement Technique

5.1 Summary of Proposed Method

This section deals with the method for packet loss rate estimation with RTP/UDP sessions. Most real-time traffic including VoIP and IP broadcasting uses RTP/UDP.

As UDP/RTP sessions do not confirm reception like TCP, it is necessary to actually measure the losses of data packets. However, with sampling measurements, the packet loss rate is difficult to measure because it is not possible to distinguish whether the reason for not observing a packet is due to sampling or to packet loss.

Therefore, we proposed a method that refers to the sequence number field in the RTP header. The RTP sequence number increases uniformly and the probability that each number occurs in a field is almost consistent. This means that, when random numbers are generated by reference to this field, the result will have uniform variance and will be suitable for use as the sampling rule.

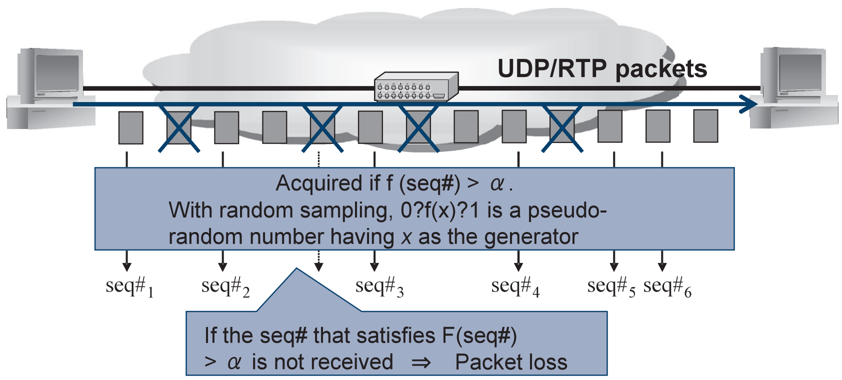

Fig. 3 UDP/RTP packet loss rate estimation.

With the proposed method, the sequence numbers *(seq#)* of RTP/UDP packets are inspected at the probes and the packets are captured only when *f(seq#)* is no more than sampling probability α as shown in **Fig. 3**. Here, function *f(x)* satisfies $0 \leqq f(x) \leqq 1$ and generates pseudo-random numbers using *x* as the generator.

Then, the packets sampled thus are sent to the quality analyzer to estimate the quality. The quality analyzer anticipates that the packets with which $f(seq\#) > \alpha$ are captured. If such a packet cannot be captured at a proper timing in the sequence number train, it judges that the packet is lost. Specifically, assuming that the sequence number captured in a previous session is *seq*#1 and that the sequence number captured last is *seq*#2, the number of packet losses in the period between them, *l*, can be estimated by applying the following formula.

$$l = \sum_{X = seq\#1+1}^{seq\#2-1} | f(x) < \alpha | / \alpha$$

With the proposed method, the packet losses can be estimated accurately when the packet loss rate is no less than 0.3%, assuming that the RTP/UDP flow transports 50PPS, the observation period is 1 minute and the sampling rate is 6.25%. For example, with VoIP, quality degradation is identified when the packet loss rate is more than a few percent, so its accuracy can be regarded to be sufficient. For the results of detailed performance evaluation using simulations and laboratory experiments, refer to Reference 9).

5.2 Measurement Error

With the proposed method, the obtained packet loss rate is an expected value for the actual value, and includes a measurement error. From a statistical viewpoint, the measurement error value is dependent on the number of captured samples and is expressed with the following formula. Here, *c* is the sample count (= Number of packets generated per unit time × Sampling rate × Measuring time) and *d* is the dependable section coefficient (*d* = 196 when the dependable section is 95%).

Error rate (%) =

$$|(\text{Actual value - Estimated value})/(\text{Actual value})| \leq \frac{d}{\sqrt{c}}$$

When studying the results of quality measurements, it is necessary to consider the error rate obtained by applying the above formula as well as the expected value, and also to consider the dependability of the quality results. Also note that the above formula means that the measurement accuracy improves when the bandwidth of the measurement target session is larger or when the measurement time is longer.

6. Conclusion

In the above, we described a per-session quality measurement method that uses packet sampling technology. In the IP networks that are diversifying more and more by accommodating various applications, the provision of a multi-point, detailed quality measurement system is one of the key factors in the building of dependable networks. The technology proposed above is expected to solve the problems with the traditional probe monitoring technique. To be precise, the impossibility of installing a large number of probes due to the costs, and the problem with MIB monitoring, which is the impossibility of performing detailed quality monitoring. The proposed system will thus offer a significant contribution to the construction of dependable networks for the future.

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