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A Traffic Measurement Tool for IP-Based Networks

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SUMMARY Today, many kinds of real-time applications are available over IP networks. It is important to measure the network performance for such applications before making use of real applications. Authors developed the general purpose traffic measurement tool for IP networks. This system can generate any kinds of traffic flexibly and calculate the network performance such as throughput, delay and loss ratio according to received packets. In this paper, the concept of this traffic measurement tool is described in detail, and several examples of network measurements are shown.
key words: *measurement, traffic generation/analysis*

1. Introduction

Recently, many people or organizations can use a variety of applications such as videoconference, internet-telephone, www, ftp etc. over IP networks. The other hand, network provider can use a various kinds of link such as radio, xDSL, satellite ATM, frame relay etc. to construct IP networks. In such an environment, network users want to know the behavior of the applications over the network as well as network providers want to know the effect of the user application to the network.

To evaluate any kind of application over IP networks, a flexible traffic generator/analyzer is needed. Many network measurement tools are available today. LAN frame data capturing and generating equipment and software based tools such as netperf [1], ttcp [2], etc. are frequently used to measure the network performances. Unfortunately, these equipment or software tools are able to generate only limited types of traffic pattern with a fixed packet length, a fixed packet interval or fixed data contents etc. With these existing tools, it is difficult to evaluate real time applications such as voice and video.

We developed general purpose traffic generator/ analyzer called KITS(KDD Internet Test System)[3] to emulate the traffic of real time applications and to analyze the effect of the network to such applications. KITS are able to generate any types of IP traffic and measure the network delay time, jitters and data losses. The rest of this paper is arranged as follows. Section 2 describes the requirements of network measurement, Sect. 3 describes the developed traffic generator/analyzer KITS and Sect. 4 describes the examples of experiments for evaluation of IP networks. In the final section we present our conclusions.

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2. Requirements of IP Network Measurement

Before making use of new kinds of network applications over IP networks, usually the following items are measured by users or a network provider. The same items are also measured when introducing new kinds of network infrastructure.

2.1 Throughput (bandwidth)

Some applications require a guaranteed minimum throughput. It is necessary to measure end-to-end throughput. Each network application generates deferent types of UDP or TCP traffic. So, the throughput must be measured by UDP or TCP according to the applications that are planed to be used.

2.2 Delay and Delay Variance

Data propagation delay and delay variance severely affects some applications which require interaction and real-time processing.

2.3 Data Loss Ratio

Data loss may occur according to propagation bit error or buffer overflow in the intermediate network equipments and application terminals. While the propagation bit error ratio is fix value, data loss ratio caused by buffer overflow dynamically changes depending on the network conditions. So, data loss ratio should be measured in appropriate network conditions.

These three values of measurement will depend on the generated traffic pattern i.e. distribution of packet length and intervals. An accurate network measurement requires accurate traffic model of real network applications.

3. Traffic Measurement Tool “KITS”

To evaluate IP networks, it is necessary to generate a pseudo-traffic emulating the target of applications, and to analyze it statistically. But almost all of the existing IP traffic generators or analyzers can not generate traffics flexibly. So authors developed a general purpose traffic generator / analyzer which can generate any type of traffic.

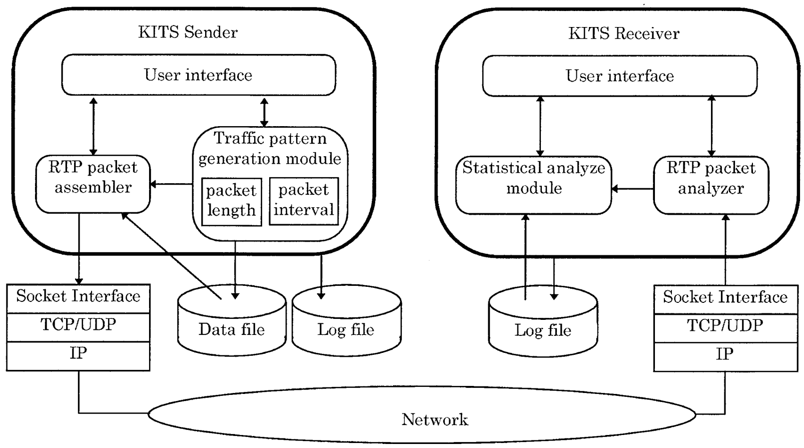


Fig. 1 Functional blocks of traffic generator/analyzer -KITS-.

number of measured packets / total packets: 83182 / 83195
packet size: diffs = 0
min = 24 max = 1472 mean = 402
variance = 217137
standard deviation = 466
confidence interval = -996 -- 1800
delays:min = 100958 max = 543563 mean = 191951
variance = 8122240314
standard deviation = 90123
confidence interval = -78419 -- 462321
packet loss rate: 0.168030
data loss rate: 0.199524
throughput rate (byte/sec) : 132928.753968
packet transfer rate (packet-number/sec): 330.087302

Fig. 2 Output image of KITS analyzer.

3.1 Functional Blocks of KITS

Figure 1 shows the functional blocks of KITS. KITS generates pseudo-traffic as RTP [4] packets. In the sender side, an user sets each parameter via User-interface module. Based on the user defined algorithm or the data file gathered from field experiments, traffic pattern generation module decides a length and an interval for each packet. According to these values, the RTP packet assembler generates RTP packets and hands them to the socket interface.

In the receiver side, received RTP packets are saved to a log file and statistically analyzed. The sample output of the KITS is shown in Fig. 2. Packet size, delays and loss ratio are calculated from the received packets.

At this time, KITS is available on FreeBSD, HP-UX, and SUN OS4.1.3.

3.2 Packet Format

Figure 3 shows the packet format of pseudo-traffic which this tool generates. This packet data are carried as user data of UDP or TCP packets. The meanings of each field is as follows.

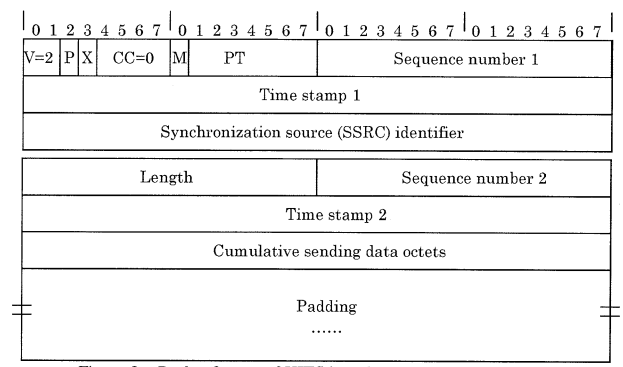


Fig. 3 Packet format of KITS based on RTP packet format.

The first 12 bytes of the packet are same as RTP packet format. The next 12 bytes are KITS original header field.

•Version (V): This value is set to 2 (RTP).

•Padding (P), Extension (X), CSRC count (CC) and synchronization source (SSRC) identifier: In this system, these RTP parameter fields are not used and set to 0.

•Marker (M): When the packet contains "delimiter" information the Marker bit is set to 1. In the next section, role of the delimiter information is mentioned.

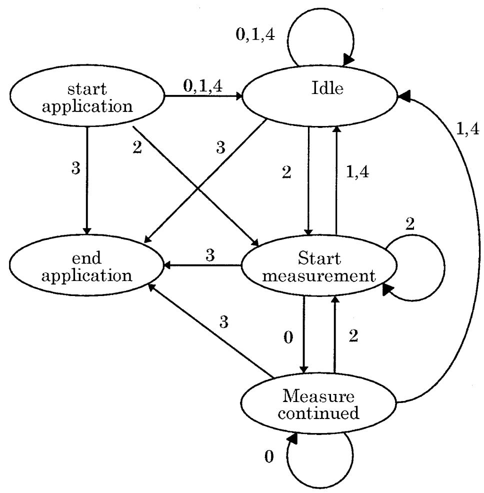
•Payload Type (PT): shows the delimiter information type. The value is 0,1,2,3 or 4.

•Sequence number 1 and 2: In this system, 4 byte sequence number is assigned to each generated packet. Sequence number 1 shows lower 2 byte and Sequence number 2 shows upper 2 byte of the sequence number respectively.

•Time stamp 1 and 2: Time stamp shows the time of the generation of packet. Time stamp 1 contains the day, hour and second information. Time stamp 2 contains micro second information.

•Length: This field represents the length of padding field.

•Cumulative sending data octets: Sum of the data bytes



0:Data packet received (without delimiter information)
1:Delimiter 1 packet received (send start)
2:Delimiter 2 packet received (start measurement)
3:Delimiter 3 packet received (stop application)
4:Delimiter 4 packet received (stop measurement)

Fig. 4 State diagram of KITS (receiver).

of packets which have been sent before.

3.3 Delimiter Information

The delimiter information contained in a KITS packet header are used to control the receiver side KITS application by the sender side. 4 types of delimiter information are defined as follows,

1: Start of data transmission
2: Start measuring
3: Quit KITS application
4: Stop measuring

The delimiter information makes it possible to realize the flexible measurement of network. For example, measurement after the traffic conditions have become stable, measurement with same parameters over and over, and so on. In Fig. 4, the state diagram of KITS receiver side is depicted.

4. Experiment

In the following sections, several cases of network measurement performed with KITS are mentioned.

4.1 Emulate the Traffic Pattern of NV [5]

Now, many kinds of realtime applications are available. NV is one of the most notable network video conference applications. To evaluate the effect of such real time applications on the network, a flexible traffic generator which can generate the pseudo-traffic similar to the target traffic is required.

Authors proposed a traffic model of real-time applications [6]. The following equations (1) and (2) are the traffic models in terms of the packet length of NV.

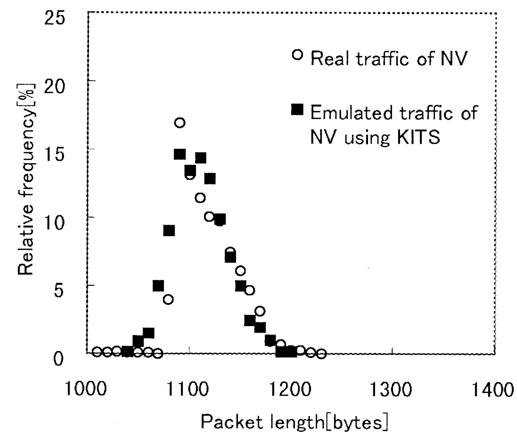


Fig. 5 Emulated traffic of NV.

$$p(s) = 0 \quad (s < s_{\min}, s > s_{\max}) \tag{1}$$

$$p(s) = \frac{(s' - s_{\min})^{\frac{s - s_{\min}}{a \cdot u}}}{\left(\frac{s - s_{\min}}{a \cdot u}\right)!} e^{-(s' - s_{\min})} \quad (s_{\max} \geqq s \geqq s_{\min}) \tag{2}$$

s': average of packet length,
$s_{\min}$: the minimum packet length,
$s_{\max}$: the maximum packet length,
$p(s)$: generation probability of s length packet,
a : coefficient,
u : unit of packet generation

Figure 5 plots the packet length distribution of real traffic of NV and the pseudo-traffic generated by KITS using Eqs. (1) and (2), which shows almost the same packet length distribution is emulated by the model.

4.2 BER (Bit Error Ratio) Measurement

Some network link equipment such as xDSL modem or radio LAN have only LAN interfaces (10BaseT). In that case, it is difficult to measure the link quality with existing BER testers. This clause describes a method to measure the link quality using KITS.

KITS can measure Frame Error Ratio (FER : same as packet loss ratio) . The relationship between FER and BER is shown in Eq. (3).

$$\text{FER} = 1 - (1 - \text{BER})^{\text{Framelength[bits]}} \tag{3}$$

Equation (3) is valid under the following conditions.

Each error frame has only 1 bit error.

Frame or packet layer protocol has only an error detection function. They do not have any error recovery function such as forward error correction or retransmission.

There is no packet loss caused by buffer overflow in the intermediate network equipments except the link to be measured.

To verify the BER measurement method using KITS, the following experiment is performed. Figure 6 shows the experimental configuration of BER measurement using KITS. Two 10BaseT LANs are connected by bridges and a circuit simulator. KITS sender and receiver are connected to each LAN. Circuit simulator connects the bridges in the speed of 1.5Mbps. Random and Burst error are injected artificially.

Procedure of BER measurement is as follows.

(a)Make sure the KITS sender and receiver terminals have enough performance.

(b)Measure the bandwidth of link which will be measured. Netperf, ttcp or KITS can measure the end-to-end bandwidth.

(c)To avoid the packet loss except in the link to be measured, determine the packet length and interval much lower than the bandwidth measured in (b).

(d)KITS sender transmits data packets and KITS receiver analyzes the received data packets and calculate the BER from FER.

Figure 7 shows the results of experiment. When the random error are injected, the injected BER is almost same as the measured BER. When the burst error are injected measured BER is lower than the injected BER. Since burst error dose not meet the condition (a) described before, the measured BER deviate from the injected BER.

We used this BER measurement method in the INA-AINET xDSL field trial [7]. Figures 8 and 9 show the measurement configuration and the result.

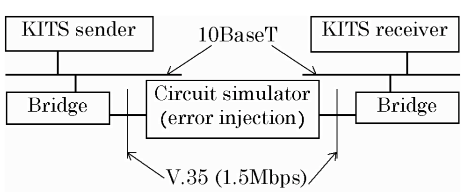


Fig. 6 Configuration of BER measurement.

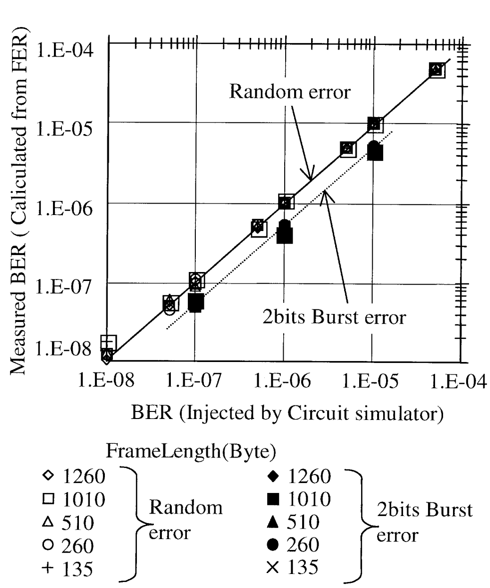


Fig. 7 Injected BER and measured BER.

4.3 Measurement in Multicast Environment

Many multi-cast applications such as video conferences become available now. Usually, each multicast receiver is accommodated in different networks. It is difficult to measure all the parameters of the multi-cast receiving performance simultaneously. In this clause, we describe the method of measurement in multi-cast environments. Figure 10 shows the configuration of the multi-cast environment examined. Sender and Receiver 1 are connected to the same LAN. Receiver 2 is connected to the LAN via a low speed and erroneous link. Receiver 3 is connected to the LAN high-speed long delay link. Sender transmits the multi-cast data and each receiver receives simultaneously and analyzes it. Figure 11 shows the output of KITS analyzer of each receiver. In this network topology, Receiver 2 measured packet losses due to the buffer overflow caused by low speed link, and Receiver 3 measured packet loss cased by link error.

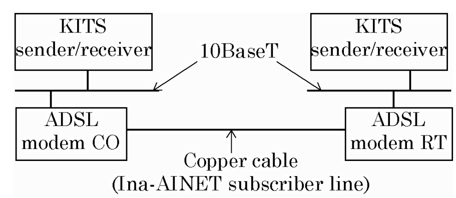


Fig. 8 BER measurement in INA AINET ADSL field trial.

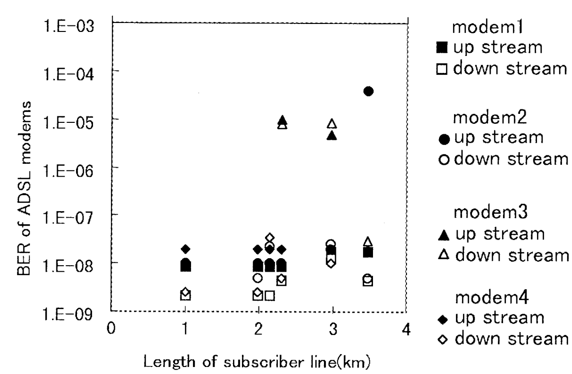


Fig. 9 Measured BER of ADSL modems.

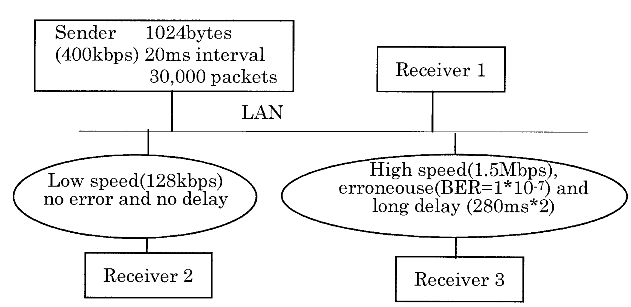


Fig. 10 Measurement of multi-cast environment.

```
number of measured packets / total packets: 2500 / 3122
packet size:    diffs = 0
          min = 1024  max = 1024  mean = 1024
          variance = 0
          standard deviation = 0
          confidence interval = 1024 -- 1024
delays:min = 4902  max = 185641  mean = 5987
          variance = 72875005
          standard deviation = 8537
          confidence interval = -19623 -- 31597
packet loss rate:  0.000000
data loss rate:    0.000000
throughput rate (byte/sec) : 51200.000000
packet transfer rate (packet-number/sec): 50.000000
```

(a) Receiver1

```
number of measured packets / total packets: 739 / 1101
packet size:    diffs = 0
          min = 1024  max = 1024  mean = 1024
          variance = 0
          standard deviation = 0
          confidence interval = 1024 -- 1024
delays:min = 4380248  max = 4477195  mean = 4466197
          variance = 56914257
          standard deviation = 7544
          confidence interval = 4443565 -- 4488829
packet loss rate:  0.704282
data loss rate:    0.704282
throughput rate (byte/sec) : 15134.720000
packet transfer rate (packet-number/sec): 14.780000
```

(b) Receiver2

```
number of measured packets / total packets: 2499 / 3121
packet size:    diffs = 0
          min = 1024  max = 1024  mean = 1024
          variance = 0
          standard deviation = 0
          confidence interval = 1024 -- 1024
delays: min = 292880  max = 378819  mean = 293276
          variance = 6689795
          standard deviation = 2586
          confidence interval = 285517 -- 301035
packet loss rate:  0.000400
data loss rate:    0.000400
throughput rate (byte/sec) : 51179.520000
packet transfer rate (packet-number/sec): 49.980000
```

(c) Receiver3

Fig. 11 Output of KITS analyzer.

5. Conclusion

This paper described the general purpose traffic measurement tool for IP networks called KITS, which can generate any kind of traffic flexibly and analyze the network performance. Using this system, it is easy to measure the network throughput, delay and loss ratio of IP networks. Thus this system will be very useful for network users as well as providers.

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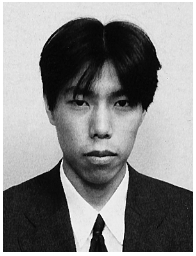

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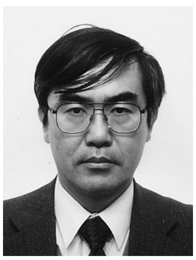

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