A Scheme to Estimate One-way Delay Variations for Diagnosing Network Traffic Conditions

Makoto Aoki and Eiji Oki

***Abstract*— Real-time applications over the Internet, such as Voice-over-IP and video streaming services, are increasingly being applied to practical communication services so how to evaluate service qualities and how to diagnosis the cause of service degradation has become one of the key issue for QoS management. Various kinds of measurement metrics over IP networks including one-way delay variation (OWDV), inter-packet delay variation (IPDV), and packet losses, have been standardized. In particular, OWDV is useful for analyzing the network traffic conditions since this metric directly reflects the queuing buffer delays generated in the network nodes.**

Unfortunately OWDV is difficult to measure because it requires the use of time information at source and destination hosts. This paper proposes an OWDV estimation scheme that solves this problem; it estimates OWDV values using IPDV values measured at the destination host. The clock difference between the source and the destination, mainly caused by clock skew, can be estimated by using IPDV values. Based on this scheme we develop a tool for diagnosis of network traffic conditions and apply the tool to analyze network traffic conditions in actual networks. The results show that the problems with network traffic conditions can be accurately analyzed using the estimated OWDV values and the cause of the conditions can be clarified.

***Index Terms*— One-way delay, One-way delay variation, Inter-packet delay, Packet delay variation, OWDV, PDV, Packet loss, Quality of Service, Diagnosis of network traffic conditions.**

I. INTRODUCTION

Real-time applications over the Internet, such as Voice-over-IP (VoIP) and video streaming services, are increasingly being applied to practical communication services so that how to evaluate service performance and how to diagnose the cause of service degradation has become a key issue for QoS management [1].

Measurement metrics are categorized by how many information types must be measured into one-point and two-point types. ``One-point" means that measurement need to be conducted only at one point, for instance the destination. ``Two-point" means that measurements must be performed at two different points, i.e. source and destination. The latter raises the issue of a clock reference to measure the delays between the two points.

M. Aoki is with Cyber Creative Institute, Tokyo, 157-0031 Japan e-mail: aoki@cybersoken.com. He is also with The University of Electro-Communications, Tokyo, Japan.

E. Oki is with The University of Electro-Communications, Tokyo, Japan, Email: oki@uec.ac.jp.

Measurements are also categorized by the measurement method used into active and passive measurements. Active measurements inject probing packets into the observed path and measure network traffic conditions that these packets experience. Passive measurements involve only the actual packets being used for communication services. Since our goal is to analyze the behavior of actual packet flows, we focus on passive measurements.

The various kinds of measurement metrics known for streaming services on IP networks include one-way delay variation (OWDV), Inter-Packet Delay Variation (IPDV), and packet loss, all of which have been internationally standardized by IETF (Internet Engineering Task Force) and ITU-T (International Telecommunication Union Telecommunication Standardization Sector) [2]-[4].

OWDV, a two-point type, is defined as the variation in packet delay between the source and the destination; it reflects the change in queuing delays in the IP networks. Since the queuing delays are directly affected by the network traffic conditions, OWDV is useful for analyzing the network traffic conditions. However, OWDV is difficult to measure due to its requirement for clock synchronization between the source and the destination. Most of OWDV schemes assume the use of an external clock synchronization mechanism based on GPS (Global Positioning System). Therefore, OWDV has not been widely used to measure delay variations in actual network paths.

IPDV, a two-point type, is defined as the difference between the delays of the current and the previous packets. Therefore, IPDV has no need for clock synchronization because the clock difference is negligible due to the short inter-packet spacing. This metric, however, can not express the network traffic conditions in detail because it means the difference between the delays of adjacent packets.

Packet loss, a one-point type, is defined as the loss of one or more packets due to transmission errors in the network links or buffer overflow in the network nodes. This metric is also defined as the packet loss rate (PLR), which means the ratio of the numbers of lost packets to the numbers of transmitted packets. The packet loss metric can be easily measured. However, it offers poor sensitivity in measuring the network traffic conditions since packet losses occur only during buffer overflow.

OWDV measurement schemes without external clock synchronization have been proposed by [5]-[7]. [5] and [6] describe a way to estimate OWDV using IPDV values, measured by packet capturing tools [8][9]. [5] and [6] give an estimation method that includes transformation mechanism from IPDV values to OWDV ones under the requirement of the synchronized clocks. The OWDV values, which include the effect of the clock difference between the source and the destination, are compensated to estimate the actual OWDV values to offset clock skew. However, since a truly practical scheme must operate in general conditions, including those in which the source and destination clocks are not synchronized, it is required to give the mechanism in more detail under the general conditions.

The scheme in [7] measures the time-interval values between a pair of packets at the sender and the receiver using dedicated user datagram protocol (UDP) packets and calculates the difference between the values. These values are used to deduce the Fourier-transformed magnitude of queuing delay PDF (probability distribution function); clock synchronization is not needed. This scheme, however, is classified as an active measurement approach and so is not applicable to our OWDV estimation scheme.

Requirements placed on a practical OWDV measurement scheme are summarized below:

- The measurement must be sensitive to network traffic conditions so that we are able to investigate fine changes in delay precisely.
- The measurement must be conducted on a packet-by-packet basis so that we are able to trace the delays of packets which will change widely with each packet reception.
- The measurement must be simple and not require any external clock synchronization mechanism. This will yield wide deployment in actual networks.

One question arises: Is there any measurement scheme that satisfies the above requirements?

This paper proposes a measurement scheme based on OWDV to diagnose network traffic conditions in cooperation with IPDV and packet loss. To estimate OWDV without recourse to a clock synchronization mechanism, the proposed scheme detects and adjusts the clock difference, which is mainly caused by clock skew. In addition, the scheme calculates and removes the offset time, which is determined by the time of the first packet reception.

The reminder of this paper is organized as follows. Section Ⅱ describes the basic definitions for delay measurements and the proposed scheme. Section Ⅲ depicts a developed diagnosis tool for analyzing the degradation in service quality. Section Ⅳ presents our results on the diagnosis of traffic conditions based on OWDV, as experienced in actual VoIP networks. Section Ⅳ describes the related works to our work. Section Ⅴ presents finally our conclusion.

II. Definitions for Measuring Delay Variations

For comprehensive understanding, the packet transmission scheme at the source is defined as shown in Fig. 1. Measurements are conducted during communication on a packet-by-packet basis. The n packets, each of which is denoted as a sequence number i, where $0 \leq i \leq n$-1, are transmitted at a fixed interval and with a fixed packet length during communication. When a packet is transmitted, the transmission time (measured at the source) is set in the packet and carried to the destination. The carried transmission time, called "time-stamp," is used to measure the transmission delay from the source to the destination as described below.

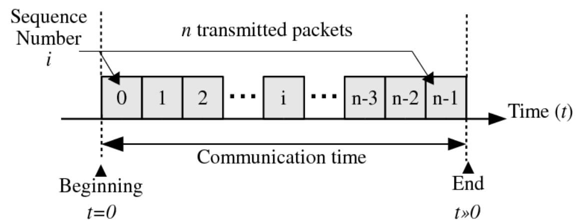

Fig. 1. Measurement is active during communication and is terminated at the end of communication. The n packets are transmitted at fixed intervals.

Fig. 2 shows an example of packet transmission sequences from the source and the destination to aid the consideration of the delay measurement by using different clocks. C_s and C_d are denoted as clocks at the source and the destination, respectively, and the times of the clocks are not equal in general if these clocks are not synchronized using an external synchronization scheme. $T_x(i)$, $R_x(i)$, $TD_x(i)$, and $RD_x(i)$ represent the time values at which the measurements of packet i are conducted

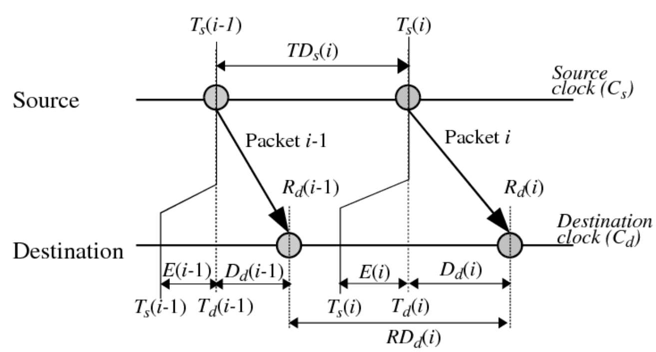

Fig. 2. Delay model for packet transmission from the source to the destination. Polygonal lines in the figure show the transmission times measured by the destination clock (C_d), which have the differences of $E(i$-1$)$ and $E(i)$ at packet sequence numbers i-1 and i.Measurement is active during communication and is terminated at the end of communication. The n packets are transmitted at fixed intervals.

where suffix x shows that time measurements are performed using the source clock (C_s) or the destination clock (C_d), respectively. $TD_s(i)$ is defined as Eq. (1). This means that $TD_s(i)$ is calculated using the interval between transmission times $T_s(i$-1$)$ and $T_s(i)$. For simplicity but without loosing generality, the reference clock is taken to be C_d. If time is measured by C_s, the error, denoted by E, due to the difference between C_s and C_d should be considered. Accordingly, $T_s(i)$ is depicted as Eq. (2).

$$TD_S(i) = T_S(i) - T_S(i-1). \quad (1)$$
$$T_d(i) = T_S(i) - T_S(i-1). \quad (2)$$

Eq. (2) means that $T_d(i)$ measured by C_d can be obtained from $T_s(i)$ measured by C_s and $E(i)$ generated by the difference between C_s and C_d. $D_d(i)$, the transmission delay of packet i, is depicted by Eq. (3), in which Eq. (2) is substituted for $T_d(i)$.

$$\begin{aligned} D_d(i) &= R_d(i) - T_d(i), \\ &= R_d(i) - T_S(i) - E(i). \end{aligned} \quad (3)$$

A. Definition of IPDV based on the delay model

$IPDV_d(i)$ is defined as Eq. (4) in accordance with [10].

$$\begin{aligned} IPDV_d(i) &= D_d(i) - D_d(i-1), \\ &= (R_d(i) - T_d(i)) - (R_d(i-1) - T_d(i-1)), \\ &= RD_d(i) - TD_d(i) - (E(i) - E(i-1)). \end{aligned} \quad (4)$$

Eq. (4) means that $IPDV_d(i)$ can be obtained by measuring two delay parameters:

- The time difference between packet i and packet i-1 of reception interval time based on C_d.
- The time difference between packet i and packet i-1 of transmission interval time based on C_s (This information is transmitted to the destination as time-stamps).

$E(i)$ - $E(i$-1) will be close to zero because E(i) is almost the same as $E(i$-1) given the short interval time period (e.g. 10 or 20 ms). Therefore, Eq. (5) remains valid in practical measurements:

$$IPDV_d(i) \approx RD_d(i) - TD_s(i). \quad (5)$$

B. Definition of OWDV based on the delay model

$OWDV_d(i)$, in the same way as $IPDV_d(i)$, is defined as Eq. (6) in accordance with [11]. $OWDV_d(i)$ is based on the probability distribution of one-way packet transmission delay as shown in Fig. 3.

$$OWDV_d(i) = D_d(i) - \min_{0 \le j \le n-1}\{D_d(j)\}, \quad (6)$$

where $\min_{0 \le j \le n-1}\{D_d(j)\}$ is the packet transmission delay with the lowest value over the observation time (consecutive n packets) and is taken to be the delay without queuing delay in the networks. It should be noted that $\min_{0 \le j \le n-1}\{D_d(j)$ is determined to be a constant value when the communication is terminated. Referring to Fig. 2, $D_d(i)$ is defined as Eq. (7) in the recurrence formula considering the clock difference between the source and the destination:

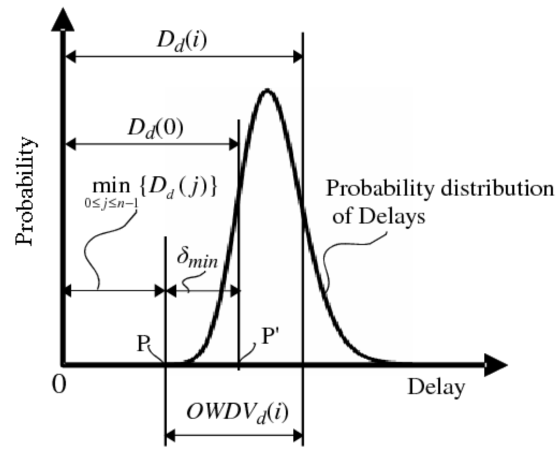

Fig. 3. Probability distribution of OWDV and the definitions of variables for OWDV measurement: $D_d(0)$ and δ_{min} mean the delay of packet 0 and the difference between $D_d(0)$ and $\min\{D_d(i)\}$. P' is defined as the temporary reference point for OWDV measurement, which is determined at the time of receiving the first packet, 0. P stands for the "true" reference point to $OWDV_d(i)$, or $OWDV_d(i)$=0. The change of the reference point from P' to P is processed at the end of measurement.

$$\begin{aligned} D_d(i) &= (RD_d(i) - TD_d(i)) + D_d(j), \\ &= (RD_d(i) - TD_S(i)) - (E(j) - E(i-1)) \\ &\quad + D_d(i-1). \end{aligned} \quad (7)$$

This equation is transformed into Eq. (8) within the range of zero to n-1:

$$D_d(i) = \delta(i) + D_d(0), \quad (8)$$

where let $\delta(i)$ denote the equation as shown in

$$\delta(i) = \sum_{j=1}^{j=i}(RD_d(j) - TD_s(j)) - (E(i) - E(0)). \quad (9)$$

In Eq. (9), $E(i)$-$E(0)$ is the time difference from packet 0 to packet i due to the difference between the source and the destination clocks. $D_d(0)$ is defined as the one-way delay of packet 0. However, its value is not able to be measured because reference point P is not determined at the reception of this packet. Therefore, P', which corresponds to the reception time of packet 0, is defined as a pseudo reference point. $\delta(i)$ corresponds to $OWDV_d(i)$ being measured from the reference point of P' so that its value will be negative when $D_d(i)$ is greater than $D_d(0)$. Using Eq. (8), $\min_{0 \le j \le n-1}\{D_d(j)\}$ is transformed into Eq. (10):

$$\begin{aligned} \min_{0 \le j \le n-1}\{D_d(j)\} &= \min_{0 \le j \le n-1}\{\delta(j) + D_d(j)\}, \\ &= \delta_{min} + D_d(0), \end{aligned} \quad (10)$$

where $\delta_{\min}$ is defined as $\min_{1 \le j \le n-1}\{\delta(j)\}$; its value corresponds to the minimum value of $OWDV_d(i)$ being measured from the reference point of $D_d(0)$. Since $D_d(0)$ is greater than or equal to

$\min_{0\le j\le n-1}\{D_d(j)\}$ holds as shown in Fig. 3, $\delta_{\min}$ remains less than or equal to zero. Eq. (6) is transformed into Eq. (11) by substituting Eqs. (8) and (10) for Eq. (6). Note that the term of $D_d(0)$ is removed by the substitution.

$$OWDV_d(i) = \delta(i) - \delta_{min}. \tag{11}$$

Eq. (11) also means that $OWDV_d(i)$ can be evaluated if $E(i)$-$E(0)$, or the difference between C_s and C_d, can be estimated in some way. We will discuss about how to estimate $E(i)$-$E(0)$ in the next subsection.

C. Removal of the clock difference

The clock differences include clock offset, clock skew, and clock drift. Clock offset is the difference between C_s and C_d, referred to as θ, as measured against the same reference clock, defined as $\theta = t_d - t_s$. Note that θ corresponds to $E(0)$ when i=0. Clock skew, referred to as φ, is the difference between the clock rates (r) of the source and the destination hosts, also defined as $\varphi = r_s/r_d$. Clock drift is the rate of change (ν) in the clock rates. Since clock drift will be assumed to be nearly zero, $E(i)$ is defined as Eq. (12) by introducing a discrete expression in time:

$$E(i) = \varphi i r + E(0), \tag{12}$$

where the time-unit is defined as τ which corresponds to packet transmission interval time measured by C_d. Therefore, the estimation of $E(i)$-$E(0)$ results in determining the slope, or the value of φ, among the values of $\sum_{j=1}^{j=i}(RD_d(j) - TD_s(j))$ in Eq. (9). These values consist of propagation plus transmission delays, and queuing delays: The former holds a constant value when packets with the same length are transmitted, the latter depends on the traffic conditions in the networks. $OWDV_d(i)$ will be close to zero (i.e. not including queuing delays) at higher probability when packets are transmitted under less than extremely high traffic loads (for instance, link utilization $\le$ 0.8). Assuming that we can select $OWDV_d(i)$ and $OWDV_d(i+k)$ (k is natural number), which are nearly close to zero, from among all $OWDV_d(j)$ ($0 \le j \le n$-1), Eq. (11) can be transformed into Eq. (13):

$$E(i+k) - E(i) = \sum_{j=i+1}^{j=i+k}(RD_d(j) - TD_s(j)). \tag{13}$$

Considering $E(i+k)$-$E(i)$= $\varphi k \tau$, Eq. (14) is given by:

$$\varphi = \frac{\sum_{j=i+1}^{j=i+k}(RD_d(j) - TD_s(j))}{k\tau}. \tag{14}$$

Eq. (14) means that the clock skew (ϕ) can be determined by measurable data between packet i and packet i+k. In practice, ϕ will be estimated using linear regression or linear programing to reduce the estimation errors due to misleading outlier (smallest) data. Using linear regression, the error of clock skew estimation is given by less than 3.7% in simulation studies [6].

Fig. 4 summarizes the estimation process of Eq. (11): the data, shown as circles on the long dotted lines represent $\sum_{j=1}^{j=i}(RD_d(j) - TD_s(j))$, which are measured using different C_s and C_d in the same way as in IPDV measurement. The data, shown as triangles on the short dotted lines represent $\delta(i)$-$D_d(0)$. These data correspond to the values that would be measured with synchronized clocks. However, the time errors caused by the time of the first received packet remain. The data, shown as the rectangles on solid lines represent $OWDV_d(i)$, in which the time error, or δ_{min} is canceled.

III. Implemented Measurement Tool for Diagnosis of IP Networks

As mentioned in the previous section, the IPDV-OWDV transformation scheme yields OWDV information at the same time as conducting IPDV measurements. We can use this scheme to create a method that observes the traffic streams on a packet-by-packet basis in addition to packet loss.

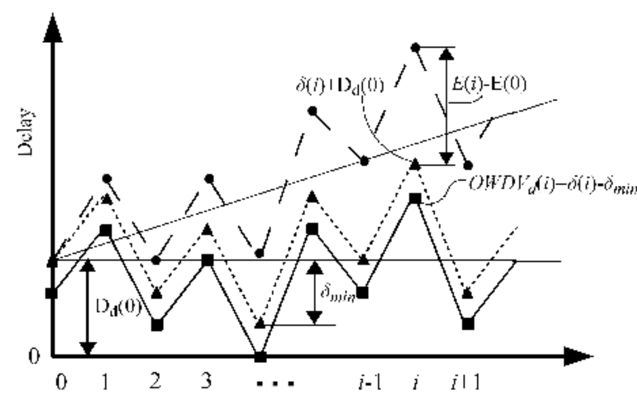

Fig. 4 The slope of $E(i)$-$E(0)$ and δ_{min} are removed from $\delta(i)$ to obtain $OWDV_d(i)$.

We have developed a measurement tool based on this scheme. Fig. 5 shows its block diagram. With this tool, the information of packet streams, including also packet reception time measured by C_d, are recorded on a per communication session basis. The packets to be analyzed are selected from among these collected packets. The selected packets are checked to confirm if they were correctly received. If correctly received, the information of the time-stamps and packet sequence numbers included in the packets is extracted and analyzed. Whereas if incorrectly received due to violation of the communications protocol such as FCS (Frame Check Sequence) errors, the packets are discarded and their sequence numbers remain lost.

These information are transmitted to the functional blocks depicted as an OWDV and packet loss measurement blocks for analyzing the delays and packet loss characteristics. The two blocks output their analyzed information to the display function block, in which displays the two metrics on the same clock and time scale.

IV. Experiments and Results

This section describes the experimental configurations and methods used to assess the proposed scheme and the results obtained from practical voice communications sessions in actual IP networks. Specifically, the relationships between voice quality and the three measurement metrics were analyzed.

A. System configuration for measurements

Fig. 6 depicts the communications paths over several subnetworks. We focused on the network traffic conditions on two communication paths: one is a path, the short-dotted line (a), between two IP-phones over a wireless LAN (IEEE 802.11g) [12], ADSL (ITU-T G.992.3) [13], Internet and VoIP (QoS-managed) networks. The other is a path, the long-dotted line (b), between analog telephones over PSTN and IP phones connected to private VoIP networks. We measured transmission qualities of the voice packets at a network operation center (NOC). The packets being measured are copied and transmitted from probing points (routers) to the NOC. These packets including packet headers are recorded onto storage media at the NOC. We focused on the voice packets which transmitted from the left side to the right side because they experience many of the factors yielding voice quality degradation such as delay, jitter, and packet loss. The recorded data of VoIP calls are selectively analyzed if there are any complaints from users about voice qualities on their conversations. The data are extracted among all of the call data using the information of phone numbers, communication time including the beginning and the end times.

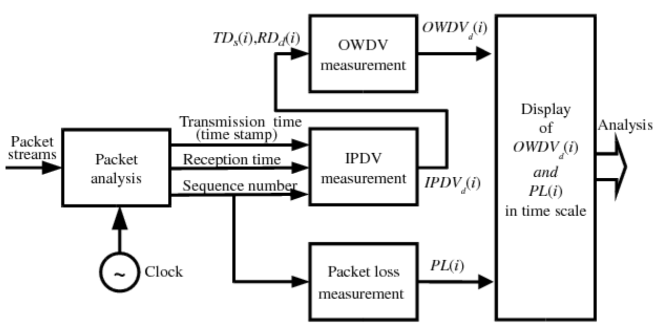

Fig. 5 A block diagram of measurement tool based on OWDV together with Packet loss. In this measurement scheme, OWDV and packet loss metrics can be shown simultaneously on a packet-by-packet basis. This scheme allows us to analyze network traffic conditions in detail.

The networks were assumed to support enterprise-level communications and include around 20,000 IP telephones. Voice signals are modulated based on ITU-T G.711 [16] (modulation rate of 64 Kb/s) and their packets are transmitted at fixed interval (20 ms) from the source to the destination hosts using RTP [10]. The destination host transforms the packets into voice signals and the voice signal's delay is made constant by using a jitter buffer and a partial packet loss can be helped hide by packet loss concealment (PLC) function [17].

B. Measurement analysis for the case of fairly good voice quality

Fig. 7 shows an example of the measurement results of the IPDV, OWDV and lost packets for VoIP communication on path (a). The voice quality of that call is assessed overall by the users as ``fairly good quality," which means ``no appreciable effort required for telephone conversation" [15]. The voice quality is evaluated by user-A and/or user-B, using subjective assessment mainly from the view point of the presence of delay, echo, noise, etc. The primary reason why the voice quality is ``fairly good" is assumed to be that the VoIP packets experienced no packet losses and long delay variations due to the network congestion, transmission errors and the like as shown in the figure. It, however, shows that OWDV exceeds 100 ms with bursty traffic when the communication times (called *CT* in the following) are equal to around 30, 50, 80, and 100 sec. In these cases, delay variations beyond 50 ms will lead to packet discard at the jitter buffer of IP-telephone receivers. The total packet loss rate due to the delay variations beyond 50 ms is estimated to be 3.9% so that the voice quality of this call is assumed to be slightly degraded.

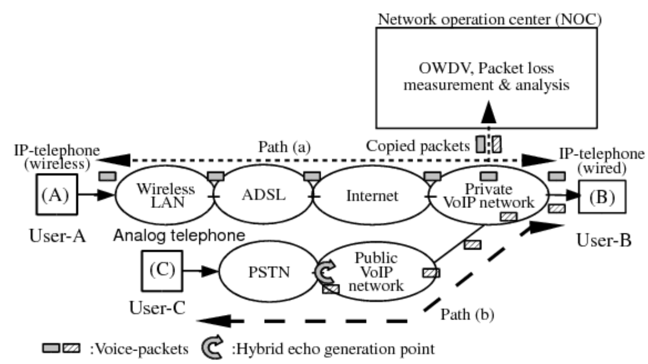

Fig. 6 System configuration for measurement of QoS of voice communications. The packets on observed paths are copied and transferred to the network operation center (NOC) to be measured. The observed paths include two routes; path(a) consists of IP-telephone (wireless), Wireless LAN (IEEE. 802.11g), ADSL (maximum upward rate: 1.2 Mb/s, Internet ,and private VoIP networks. Path (b) consists of PSTN, public VoIP networks and the private VoIP networks.

The IPDV in Fig. 7 appears to match the delay variations of OWDV. However, as shown in Fig. 8 in which the time period between *CTs* values of 25 and 35 sec is graphically expanded, the number of the packets discarded at the jitter buffer is estimated lower value by IPDV metrics to be lower than that estimated by OWDV metrics. The packets that follow a packet with long delay tend to have long delays, therefore these packets will be discarded with higher probability.

C. *Measurement analysis for the case of poor performance*

Here the voice packets experienced longer-delays and higher packet loss rates: the OWDV values were almost flat at around 300 ms and packet loss rate were high at around 4% between *CTs* values of 53 and 99 sec as shown in Fig. 9. The loss included instances of consecutive packet loss. However, between *CTs* values of 99 and 136 sec there was no packet loss and longer delays. In this measurement, *CTs* values of 53 and 136 sec correspond to the beginning and the end points in the VoIP connection. These voice packet data are observed in an actual communication session on the same path (path (a)) as in the previous example. User-A and -B evaluated the voice quality of this call: the user-B felt discomfort in communicating with user-A because of click noises and/or distorted voice. However the user-A did not feel any great degradation in voice quality.

From the analytical point of view, the primary degradation in OWDV metric is presumed to be overloading of the ADSL upward link, in other words, insufficient ADSL upward link capacity. The reasoning is as follows. First. the long and constant delays mean that any queue in the observed path became full when the VoIP connection started using the insufficient resources on path (a) so that packets input into the queue were discarded with higher loss rate. Second, the minimum link capacity over the path is ADSL upward link, of which link rate is estimated around 250 kb/s by the method described in [20]. Third, since user-A did not feel any degradation in voice quality, the path with bad voice quality includes the ADSL upward link mentioned above.

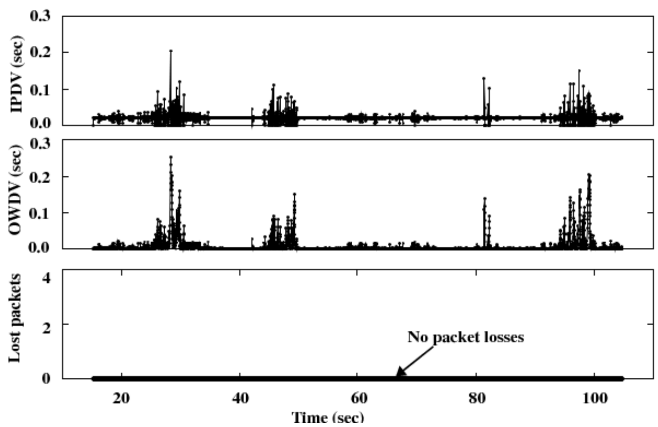

Fig. 7 An example of measurement results of IPDV, OWDV and Packet loss when the voice quality is evaluated as "fairly good quality" on the observed path (a).

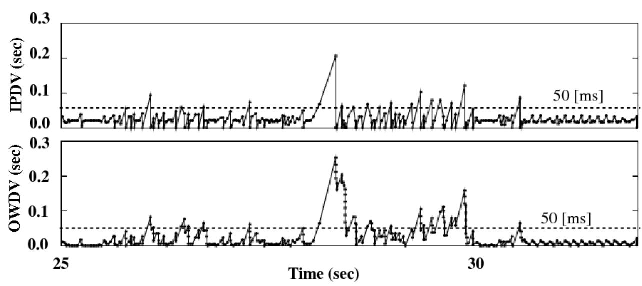

Fig. 8 IPDV and OWDV values between CT_s values of 25 and 35 sec are enlarged from Fig. 7. Focusing on the number of the packets with delay variations greater than 50 ms, it is shown that the OWDV measurements indicate more packets are discarded at the jitter buffer greater than the IPDV measurements.

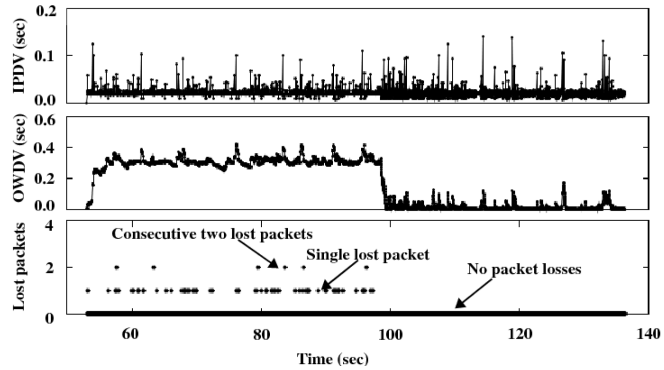

Fig. 9 Another example of measurement results of OWDV and Packet loss. These figures show that quality is estimated to be low over the observed path. The high OWDV values and packet loss rates are caused mainly by the assignment of voice channels to the ADSL link.

We estimate that the cause is the insufficient of ADSL upward link capacity. The bursty OWDV values between *CTs* values of 99 and 136 sec are assumed to reflect the bursty traffic on the ADSL links including that on the wireless LAN because the available link bandwidth is estimated to be around 250 Kb/s. These delays trigger discard of the voice packets at the jitter buffer of the IP-telephone receiver as mentioned in Section Ⅳ-B. Since the packet loss caused by the jitter correspond to packet loss rate of around 3%, the total packet loss rate is estimated to be 7% (= 4 + 3) between *CTs* of 53 and 99 sec and 3% between *CTs* values of 99 and 136 sec.

We should note that the change of OWDV values shows the events generated in the queuing buffer, extremely different from that of IPDV values as shown in Fig. 9. So, OWDV metric can play an important role as a measurement metric in cooperation with packet loss metrics.

D. *Measurement in another case of poor performance*

This case examines dynamic delay experienced by voice packets due to processor overload. The VoIP communication session was established call on path (b), which consists of public and private VoIP networks. Path (b) requires the conjunction of two managed QoS networks. After the call was completed, the users complained about two problems; both of the users sometimes noticed a response delay during the conversation which disrupted communication; User-B often heard an echoed voice. The voice quality of that call was evaluated as ``bad quality" because of recognizing the long response delay and the echoed voice. On path (b), the echoed voice signals could be echoed at the two and four wire transmission converter, which is located at the interface between PSTN (Public Switched Telephone Networks) and the Public VoIP network[1].

[1] Generally, echoed voice signals are canceled by using an echo canceller. However, when transmission delays on the path are long, there is a higher probability of experiencing echoing

As shown in Fig. 10, the peak OWDV value change is 150 ms up to *CT* value of 100 sec, however beyond *CT* value of 100 sec it gradually increases to 600 ms at *CT* value of 135 sec. After that, it decreases to around 300 ms until call completion. There was no packet loss during the call. These phenomena were observed a few times per day, especially during the periods of heavy VoIP traffic, for instance in the morning. The primary issue to solving the problem is to find the cause of the delays starting at *CT* value of 100 sec in this case.

The big difference from previous cases shows that OWDV between *CT* values of 100 and 155 sec increase and then decrease at a small rate (several seconds order), which do not change with bursty traffic. This shows that the queue length at some nodes on the path (b) gradually increase between *CT* values of 100 and 135 sec. The change of OWDV values mainly reflects the events with the change of processing loads in the networks. It is assumed that the loads do not consist of the events generate stochastically on a packet-by-packet basis such as the packet transmission, but the events which involve call processing, per packet protocol processing and the like. Therefore, we paid attention to the processing loads of node equipment in the VoIP networks, which are connected on the path (b). As a result, we found out the node with the interface of public VoIP network, which is assumed to generate longer delays, by investigating a correlation between OWDV values and the numbers of connecting VoIP calls. The node was erroneously assigned to process VoIP protocol conversion

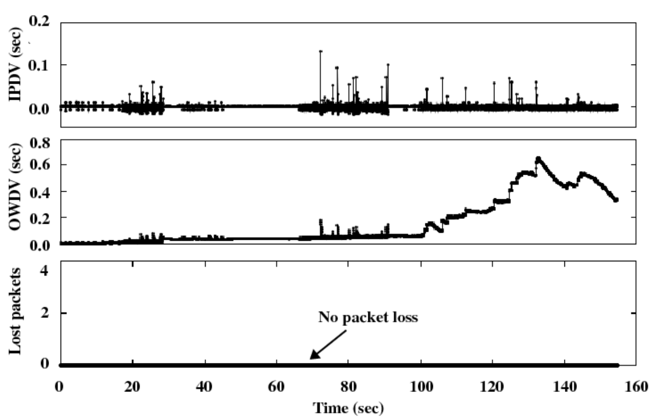

Fig. 10 An example of measurement results of IPDV, OWDV and Packet loss. The measurement was conducted to diagnose malfunction of communication equipment. The result shows that OWDV values changed significantly and increased continuously in the later half of the communication session, whereas IPDV showed no such change.

beyond its performance limit.

As in Section Ⅳ-C, OWDV is shown to play an important role as a measurement metric in cooperation with packet loss metrics.

V. Related Works

Table Ⅰ compares existing one-way delay measurement techniques to our work. The categories are measurement approaches, clock synchronization approach, measurement principles, and applicable areas. OWD including OWDV

measurement has two major challenges to solve: clock synchronization and OWD measurement scheme. There are two approaches to using external mechanisms for clock synchronization: active and passive methods. Several existing active measurement schemes use the global-positioning system (GPS), network time protocol (NTP) [29], or precision time protocol (PTP), also known as IEEE 1588 standard [30]. The introduction of GPS allows us to realize high clock accuracy [1] for delay measurement, but increases equipment costs. Furthermore, GPS is difficult to deploy on legacy equipment.

NTP and PTP require measurement of round-trip time (RTT) to perform clock synchronization, and are even more restrictive since, they require that both paths (from source to destination and from destination to source) have the same OWD to function. In general, each path has a different OWD unless the network was engineered with this feature, and it is difficult to expect a network such as the Internet would be able to meet this restriction.

Active methods that can measure OWD without requiring clock synchronization have been proposed [25][26][7]. In [25], cyclic-paths are selected properly to calculate each one-way delay, while a new relationship between OWD and RTT is introduced in [26]. However, both are based on the measurement of round-trip delays. The scheme in [7], as described in Section Ⅰ, is used to deduce the Fourier-transformed magnitude of the queuing delay probability distribution function and so avoid the clock synchronization problem.

Passive methods use the traffic flowing through the network, most often user traffic, to implement the measurement process.

[19]

TABLE I
ONE-WAY DELAY MEASUREMENT METHODS

Measurement approach	Clock synchronization	External mechanism	Measurement principle	Applicable area	Reference
Active method	GPS	Required	$OWD=R_d - T_s$	Wide area	[21][22]
	NTP				[21]
	PTP			Small subnetworks	[21]
	Dedicated	Not required	OWD estimation using pairs of packets	Wide area	[24]
			OWD estimation based on cyclic-path delay measurement		[25]
Passive method	GPS	Required	$OWD=R_d - T_s$	Wide area	[21][22][23][27]
	NTP				[21]
	PTP			Small subnetworks	[21][31]
	Dedicated	Not required	OWD estimation using pairs of packets	Wide area	[28]
			OWD estimation based on cyclic-path delay measurement		[5][6], this work

R_d and T_s represent the reception time at the destination and the transmission time at the source, respectively

A passive measurement method was recently proposed in [27]. This method uses GPS and NTP for clock synchronization. Another passive OWD measurement scheme is based on the assumption that PTP achieves clock synchronization [31]. A scheme to measure clock skew was recently proposed in [28]. This scheme uses the real-time protocol (RTP) [10] and the real-time control protocol (RTCP) [32] to create fixed packet length and fixed inter-departure gap. This approach is based on a simple model for estimating clock skew, and demonstrates a different perspective from the existing complex approaches to the estimation of clock skew, which, for example, use linear programming. The accuracy of clock skew removal depends on queuing delays being short during the measurement. The contra case, long queuing delays, is not considered in the scheme.

From the perspective of measurement accuracy, only GPS-based schemes are able to lower the error rate of clock frequency determination to under 10^{-8} [27] . Almost none of the methods based on clock skew estimation addressed the issue of measurement accuracy [25][31][33].

From the user point of view, it must be possible to measure the OWD of the user data anywhere with no limitation. For example, this means that the tools offer passive measurement and interoperability with general purpose data capture tools such as Wireshark [9], and that the need for clock synchronization is avoided. OWDV is an important factor in jitter buffer design and the analysis of network conditions. This is because OWDV involves only queuing delays, which strongly depend on network traffic conditions.

Our research targets a passive OWDV estimation scheme that does not need an external mechanism for clock synchronization between the source and destination hosts. Our approach measures IPDV and the combined and accumulated OWDV, which is the sum of IPDV. However, as clock skew is included in the IPDV measurements, we introduce an OWDV-based mechanism to estimate and remove clock skew.

In order to clarify its mechanism more precisely than is done in [5][6], this work analyzes the mechanism including the error due to the clock difference between the source and destination hosts. It also shows that OWDV measurements are useful in analyzing the network traffic conditions in actual VoIP networks.

VI. CONCLUSION

Real-time applications over the Internet, such as Voice-over-IP and video streaming services, are increasingly being applied to practical communication services so how to evaluate service qualities has become one of the key issue for QoS management.

This paper proposed a scheme that can estimate the one-way delay variations (OWDV), which is one of important evaluation metrics. This scheme allows us to estimate OWDV values from inter-packet delays (IPDV) values, which are easily measured.

The paper shows study results from two points of view: theoretical analysis of estimating process in the scheme and experimental results for diagnoses of traffic events in actual IP networks.

First, we analyzed this scheme in detail under the model of different clocks in the source and destination hosts, which corresponds to actual IP networks. The result shows that the scheme is able to be applied to estimate OWDV values and the accuracy of the OWDV estimation is determined by estimating the clock skew between the source and the destination clocks.

Next, we applied this scheme to diagnose traffic conditions in actual IP networks such as Voice-over-IP (VoIP) networks. Our results shows that the scheme is able to estimate OWDV values in actual networks and be applied to diagnose traffic conditions with distinctive characteristics in comparison with IPDV values.

The further work is necessary to study to apply the scheme to high-speed networks, more then ten Gb/s, with high-accurate clock skew estimation.

ACKNOWLEDGMENT

The authors would like to thank Roberto Rojas-Cessa for his

instructive suggestions on performance measurement schemes in IP networks.

REFERENCE

[1] J. Wang, M. Zhou, and Y. Li ``Survey on the End-to-End Internet Delay Measurements," High Speed Networks Multimedia Communications, Springer, 2004.

[2] C. Demichelis and P. Chimento, ``IP Packet Delay Variation Metric for IP Performance Metrics (IPPM)," RFC 3393, Nov. 2002.

[3] ``Internet protocol data communication service ? IP packet transfer and availability performance parameters," ITU-T Recommendation Y.1540, Nov. 2007.

[4] ``Network performance objectives for IP-based services," ITU-T Recommendation Y.1541, Feb. 2006.

[5] M. Aoki, E. Oki, and R. Rojas-Cessa, ~``Scheme to Measure One-way Delay Variation with Detection and Removal of Clock Skew," IEEE HPSR 2010, Dallas, TX, Jun. 2010.

[6] M. Aoki, E. Oki, and R. Rojas-Cessa, ~``Measurement Scheme for One-way Delay Variation with Detection and Removal of Clock Skew," ETRI-Journal, Vol. 32, No. 6, Dec. 2010, pp. 854-862.

[7] W. Lu, W. Gu, and S. Yu, ``One-way queuing delay measurement and its application on detecting DDoS attack," Journal of Network and Computer Applications, Vol. 32, Issue 2, Mar. 2009, pp. 367-376.

[8] http://www.empirix.com/products/hammer.asp.

[9] http://www.wireshark.org/.

[10] H. Schulzrinne, S. Casner, R. Frederick, and V. Jacobson, ``RTP: A Transport Protocol for Real-Time Applications," RFC 3550, Jul. 2003.

[11] A. Morton and B. Claise, ``Packet Delay Variation Applicability Statement," RFC 5481, Mar. 2009.

[12] http://standards.ieee.org/getieee802/download/802.11g-2003.pdf.

[13] Asymmetric digital subscriber line (ADSL) transceivers, ITU-T Recommendation G.992.3, Jan. 2005.

[14] J. Postel, ~``Internet Control Message Protocol," RFC 792, Sep. 1981.

[15] ``Methods for subjective determination of transmission quality," ITU-T Recommendation P.800, Aug. 1996

[16] ``Pulse code modulation (PCM) of voice frequencies," ITU-T Recommendation G.711, 1988.

[17] ``A high quality low-complexity algorithm for packet loss concealment with G.711," ITU-T Recommendation G.711 Appendix I, Sep. 1999.

[18] K. Lai and M. Baker, ~``Nettimer: A Tool for Measuring Bottleneck Link Bandwidth," Proceedings of the USENIX Symposium on Internet Technologies and Systems, Mar. 2001, pp. 123-134.

[19] ``Talker echo and its control," ITU-T Recommendation G.131, Nov. 2003.

[20] M. Aoki, and E. Oki, ``Estimating ADSL Link Capacity by Measuring RTT of Different Length Packets," Globecom 2010, Dec. 2010, 5 pages.

[21] L. D. Vito, S. Rapuano, and L. Tomaciello, ~``One-Way Delay Measurement: State of the Art," Instrumentation and Measurement, IEEE Transactions on Vol. 57, Issue 12 , 2008 , pp. 2742-2750.

[22] A. Hernandez and E. Magana, ~``One-way Delay Measurement and Characterization," ICNS 2007, Jun. 2007, 6 pages.

[23] S. Niccolini, M. Molina, F. Raspall, and S. Tartarelli, ~``Design and implementation of a one way delay passive measurement system," NOMS 2004, IEEE/IFIP, Apr. 2007, pp. 469-482.

[24] J. Wang, j. Yang, G. Xie, Z. Li, and M. Zhou, ~``On-line Estimating Skew in One-way Delay Measurement," PDCAT 2003, Design and implementation of a one way delay passive measurement system," NOMS 2004, IEEE/IFIP, Aug. 2003, pp. 430 - 436.

[25] O. Gurewitz and M. Sidi, ~``Estimating one-way delays from cyclic-path delay measurements," INFOCOM 2001, Apr. 2001, pp. 1038-1044.

[26] D. Kim and J. Lee, ~``End-to-End One-way Delay Estimation Using One-Way Delay Variation and Round-Trip Time," Qshine 2007, ACM Aug. 2007, 8 pages.

[27] T. Zseby, L. Mark, C. Schmoll, and G. Pohl, ~``Passive One-Way-Delay Measurement and Data Export," Proc. Int. Workshop Inter-domain Performance Simulation, Feb. 2003, 6 pages.

[28] B. Ngamwongwattana and R. Thompson, ~``Measuring One-Way Delay of VoIP Packets Without Clock Synchronization," I2MTC 2009, May 2009, 4 pages.

[29] D.L. Mills, ~``Network Time Protocol (Version 3) Specification, Implementation and Analysis," RFC 1305, 1992.

[30] C. Gordon,~``Introduction to IEEE 1588 and Transparent Clocks," White Paper, Tekron, 2009.

[31] M. Cola, G. De Lucia, D. Mazza, M. Patrignani, and M. Rimondini, ~``Covert Channel for One-Way Delay Measurements," ICCCN 2009, Aug. 2009, 6 pages.

[32] C. Huitema, ~ ``Real Time Control Protocol (RTCP) attribute in Session Description Protocol (SDP)," RFC 3605, Oct. 2003.

[33] S.B. Moon, P. Skelly, and D. Towsley, ~``Estimation and Removal of Clock Skew from Network Delay Measurements," INFOCOM 1999, Mar. 1999, 8 pages.

Makoto Aoki received the BS in Electrical Engineering from Tokyo University of Science, Tokyo, Japan, in 1971 and joined NTT laboratory. Since 1971, he worked in the Switching System Department to conduct research and development of remote control switching systems, data communication systems, communications protocol control LSI (X.25 and high-speed packet communications protocol) and so on. From 1998 to 2007, he worked with Infonex and NTT-data Customer Service, where he was responsible for multimedia packet network/system design for the company’s networks. He joined Cyber Creative Institute in 2007 and mainly conducts research/consultation about VoIP networks, next generation network, IPTV and so on. His research interests are quality-of-service networks, network measurements, and high-speed communications protocol control for future Internet. He is an IEEE Member and IEICE Member.

Eiji Oki is an associate professor of The University of Electro-Communications, Tokyo, Japan. He received his BE and ME in instrumentation engineering and a PhD in electrical engineering from Keio University, Yokohama, Japan, in 1991, 1993, and 1999, respectively. In 1993, he joined Nippon Telegraph and Telephone Corporation's Communication Switching Laboratories, Tokyo, Japan. He researched IP and optical network architectures, traffic-control methods, high-speed switching systems, and communications protocols. From 2000 to 2001, he was a visiting scholar at Polytechnic University, Brooklyn, New York, where he was involved in designing tera-bit switch/router systems. He joined The University of Electro-Communications, Tokyo Japan, in July 2008. Dr. Oki was the recipient of the 1998 Switching System Research Award, the 1999 Excellent Paper Award presented by IEICE, and the 2001 Asia-Pacific Outstanding Young Researcher Award presented by IEEE Communications Society for his contribution to broadband network, ATM, and optical IP technologies. He has co-authored two books, *Broadband Packet Switching Technologies*, published by John Wiley, New York, in 2001 and *GMPLS Technologies*, published by RC Press, Boca Raton, in 2005. He is an IEEE Senior Member and an IEICE Senior Member.